Meridian 1

Network Planning Guide

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Terms and abbreviations

#

Also known as octothorpe, hash, pound and number sign. There is a key with this symbol on Digitone-type and digital telephones. When a user presses this key, the system treats the call as if the End-of Dialing timer has expired. In Europe, this key is called the fast-connect key.

*

Also known as asterisk or star. When used in programming, it represents a three second pause.

64 K clear

The term given to the transmission rate possible on B-channels and D-channels used in ISDN networks. Previously, with in-band signaling, there was some bandwidth per channel used for signaling. With the introduction of ISDN out-of-band signaling that no longer happens. The entire channel is clear for use in voice and data transmission at 64 kbps.

Access Restriction

This is sometimes called the Class of Service of a telephone. It is only one component of the Class of Service. The access-restriction type controls the types of calls which can be made from a telephone. For example, if the access-restriction type is programmed as Toll Denied, the telephone cannot be used to make any calls where the digit 1 or 0 is the first or second digit following the access code digits.

There are many different access restriction types that have different levels of control of the types of calls that can be made from the telephone.

ACD

Refer to Automatic Call Distribution

Terms and abbreviations

ANSI

American National Standards Institute

Asterisk

The term used for a star (*) symbol. When used in programming, it represents a three second pause.

Attendant

The main answering position on a system. From the attendant position, incoming calls are transferred to internal telephones. The terminal used at the attendant position is called a console. There can be one or more attendants on a system, although some systems do not have any attendants. Attendants can answer incoming calls for the main listed number, recalls which have not been answered, calls from users who are dialing incorrectly, and calls from users who are attempting to place restricted calls.

Automatic Call Distribution (ACD)

Application software that puts incoming calls in a queue to one or more telephones referred to as agent telephones. The longest-waiting call is sent to the agent telephone that has been idle for the longest time. Incoming calls can be given a priority and answered at a prioritized telephone as an option. There are enhancements which can be added to basic ACD functionality to help with management and reporting tasks and also to customize and enhance the treatment incoming calls are given. ACD software packages A, B, C, D, and Custom Controlled Routing offer enhancements to management tools and reports, in order to add to the functionality of basic ACD.

B-channel

B-channels are the channels carrying (bearing) the voice and data calls in ISDN networks. Each one has a data rate of 64 kbps.

Backup answering

Otherwise known as Call Redirection, backup answering refers to the answering of calls done at a telephone or voice messaging port when the originally dialed caller is busy, not answering, or does not wish to be disturbed, and features like Call Forward are active.

Terms and abbreviations

Backup D-channel

If the primary D-channel fails, there can be a secondary D-channel configured to take over the signaling function.

Basic Rate Interface (BRI)

An international standard for connecting terminals or trunks to a system. One BRI connection is composed of 2 B-channels at 64 kbps each, and 1 D-channel at 16 kbps.

BNE

Business Networking Express (BNE) is a term that refers to a group of different EuroISDN network functionalities. BNE provides a Virtual Private Network (VPN) solution for Meridian 1 systems through the EuroISDN public network.

BRI

Refer to Basic Rate Interface.

Call Detail Recording (CDR)

Printouts made when calls are dialed, giving information about the number dialed, the telephones involved, and the duration of the call. Additional information can be printed when CDR enhancements are installed on a system.

Call Redirection

Also known as backup answering, call redirection refers to the answering of calls done at a telephone or voice messaging port when the originally dialed caller is busy, not answering, or does not wish to be disturbed, and features like Call Forward are active.

Camp-On

This feature allows an attendant to extend a call to a busy telephone. The user of the telephone hears a tone indicating a call is Camped-On. When the user hangs up, the Camped-On call rings the telephone. If the user does not hang up within a programmed amount of time after hearing the tone, the Camped-On call recalls to the attendant.

Terms and abbreviations

CCSA (Common Control Switching Arrangement)

A service offered by AT&T for private networks that allows any telephone in the network to call another using a seven-digit number.

CDR

Refer to Call Detail Recording.

Central Office (CO)

A large telephone switching system that provides service to subscribers located over a large geographic area, usually as part of a public or military switched telephone network.

Central Office trunk (COT)

This is a physical carrier of voice and data traffic to and from a local Central Office. It can be copper wire, cable, or optical fibre. Refer to Appendix 1, *Introduction to trunks*, for more information.

Centrex

A type of telephone system that usually resides in the central office and can serve telephones distributed over a wide area. The Nortel Networks system that provides this functionality is called the DMS. It provides services and features that are similar to those of a PBX and some that are different.

Channel

A transmission path capable of carrying voice or data.

Class of Service

There are many different capabilities and features which can be activated or deactivated for each telephone or trunk. A programming term is used to refer to these capabilities and features collectively. It is Class of Service. For example, the Last Number Redial feature is activated in the Class of Service.

The term Class of Service is often used for a feature called Access Restriction. When you program Trunks, Meridian Mail channels, Authorization Codes, and Direct Inward System Access ports, you assign a class of service which is the access-restriction type.

Terms and abbreviations

CO

Refer to Central Office.

Conventional main

A switch that is connected to an ESN node and equipped with no special features.

COT

Refer to Central Office trunk.

CPU (central processing unit)

The card that controls the functions of the other system components, following instructions it gets from the system memory. Some systems have one CPU; others have two.

CSA trunk

Common Control Switching Arrangement trunk. Refer to CCSA.

Customer group

A group of users with their own trunk groups, attendants, features and Numbering Plan. A system can be used by one or more unique customer groups.

D-channel

D-channels are the channels used for signaling. Each one has a data rate of 64 kbps (if Primary Rate Interface is used) or 16 kbps (if Basic Rate Interface is used). Information related to call set up, call release and feature activation is carried on the D-channel.

DDI (Direct Dialing In)

External callers dialing internal telephones directly, without the intervention of an attendant or interactive voice response system. Refer to Appendix 1, *Introduction to trunks*, for more information.

DID

Refer to Direct Inward Dialing.

Terms and abbreviations

DID trunk

A trunking feature that allows telephone callers connected to the public exchange network to dial directly to a telephone connected to the Meridian 1 system. DID happens without the intervention of an attendant or interactive voice system. Refer to Appendix 1, *Introduction to trunks,* for more information.

Digital subscriber loop

Any one of eight physical Basic Rate Interface (BRI) ports on a BRI line card. Each port has two B-channels and one D-channel.

Digital telephones

A telephone which uses digital signaling. An analog voice is converted into a digital signal within the telephone. A Macintosh, IBM-PC or other data terminal can be connected to the telephone. The data to and from that terminal is multiplexed on the same set of wires used by the telephone for voice calls.

Digital Trunk Interface (DTI)

The interface to a digital trunk carrier. In North America the carrier is a T1 transmission facility with 24 channels, each with a data rate of 64 kbps. In Europe the carrier is an E1 transmission facility with 32 channels, each with a data rate of 64 kbps.

Digitone

A signaling system that uses audio tones to transmit information. The tones are a combination of two frequencies, a high tone and a low tone. Tones are used for signaling the digits 0 through 9, # (octothorpe) and * (asterisk) and on specially configured telephones, A-D.

Digitone receiver (DTR)

A card with Digitone receiver units on it. The DTR units translate analog Digitone signals into a digital format.

DIP

Also known as Dial Pulse. A signaling system that uses electrical pulses to transmit digits.

Terms and abbreviations

Direct Inward Dialing (DID)

External callers dialing internal telephones directly, without the intervention of an attendant or interactive voice response system. Refer to Appendix 1, *Introduction to trunks,* for more information.

Direct Inward System Access (DISA)

An incoming trunk port configured to allow external callers to use the system as if they were internal users. Only trunks that provide disconnect supervision and ground start signaling can be used for DISA, prior to Release 15.

Directory Number (DN)

A dialable number assigned to telephones connected to a Meridian 1 system.Thisisthenumberinternalcallersdialtoringaninternaltelephone.

DISA

Refer to Direct Inward System Access.

DN

Refer to Directory Number.

DTI

Refer to Digital Trunk Interface.

DTMF

Dual Tone Multi Frequency. Refer to Digitone.

DTN (Digitone)

Refer to Digitone.

DTR

Refer to Digitone receiver.

E&M

A short form for Ear and Mouth, used to describe separate wires on TIE trunks used for receiving and transmitting, respectively.

Terms and abbreviations

E1

A digital transmission facility with 32 channels, each with a data rate of 64 kbps.

ECMA

European Computer Manufacturer's Association

Enbloc dialing

All digits in the call are collected before the call is processed. All digits are sent together in one "block".

ESN

The Electronic Switched Network (ESN) is a private communications network intended for use by large business customers with distributed operating locations. However, users at a stand alone Meridian 1 can benefit from the installation of ESN software.

ESN software packages include the following:

- Basic Alternate Route Selection (BARS)
- Network Alternate Route Selection (NARS)
- Coordinated Dialing Plan (CDP)
- Network Speed Call (NSC)
- Network Authorization Code (NAUT)
- Network Signaling (NSIG)
- Network Call Transfer (NXFR)
- Network Traffic Measurements (NTRF)
- Priority Queuing (PQUE)
- Off-Hook Queuing (OHQ)
- Coordinated Call-Back Queuing (CCBQ)
- Call-Back Queuing to/from Conventional Mains (CBQCM)

You can implement the packages that suit your needs.

Terms and abbreviations

ESN main

A switch equipped with Network Signaling (NSIG) software that is connected by means of TIE trunks to a single ESN node. An ESN main can also be equipped with the Basic Alternate Route Selection (BARS) feature to provide alternate route selection capabilities for calls placed to satellite switches that are connected to it.

ESN node

A Meridian 1 equipped with Network Alternate Route Selection (NARS) software.

ETSI

European Telecom Standards Institute

Exchange network

The global network made up of telephone switches operated for the public by telephone utility companies and governments.

Far end

The remote PBX at the distant end where your Tie trunk or Private circuit terminates.

Features

Capabilities assigned to the terminals which allow the users to do more than make and receive basic calls. Features range from basic Call Transfer to something as complex as Network-wide Message Waiting. Features are provided by system software which is sometimes basic to every system or packaged as separate options which are either chargeable or non-chargeable. For regular telephones, there is an element of programming referred to as the “feature prompt” that controls certain capabilities which can be activated for that telephone. Some capabilities are activated in the Class of Service.

Terms and abbreviations

FEX

Refer to Foreign Exchange Trunk.

Foreign Exchange Trunk (FEX)

The physical carrier of voice and data communications to and from a remote or foreign central office (exchange office). Refer to Appendix 1, *Introduction to trunks,* for more information.

Gateway

A means of connecting two different signaling schemes.

Group Call

The name of a feature that allows one user to press a key on a proprietary telephone and automatically call several telephones. When users answer these telephones they are automatically bridged into a conference connection with other users on the same Group Call.

ISA

Refer to Integrated Services Access.

ITA

Refer to Integrated Trunk Access.

Immediate start

A signal used on trunks to control the transfer of dialed digits. After a trunk seizure, the originating switch may start outpulsing digits to the terminating switch after a minimum delay of 70 milliseconds.

Inband signaling

Signaling made up of tones which pass within the voice frequency band and are carried along the same circuit as the talk path that is being established by the signals.

Terms and abbreviations

Integrated Services Access (ISA)

The Meridian 1 can send and understand call set up messages that specify the type of route a channel will be used for, on a call by call basis. ISA is sometimes called Call by Call Service and Dynamic Channel Assignment.

ISA eliminates the need for dedicated B-channels for each service route. The channels are assigned to routes on a call-by-call basis and used as needed. This capability is only provided in North America.

Integrated Trunk Access (ITA)

This capability is provided by PRI, and allows a T1 facility to include trunks that are controlled with out-of-band signaling and in-band signaling trunks (called“A+B bit” signaling trunks).

Intercept treatments

Invalid or denied actions coming from a telephone, TIE trunk, attendant, or CCSA/DID trunk are given a treatment called an Intercept treatment. These treatments are defined customer wide. For example, if a user who is Toll Denied tries to dial a toll call, then the Intercept treatment for that situation determines what will happen to that user. The possible treatments are: the user will hear overflow tone, or be routed to the attendant or will hear a recorded announcement.

Interworking

A term used when capabilities offered by different protocols can work together to provide users seamless functionality even though they are connected to two different parts of the network.

In-WATS

In North America, these trunks are rented by companies from the Telco to allow callers to call in without paying a long distance fee. The purchaser of the trunk pays for the calls, generally at a reduced long distance rate. Refer to Appendix 1, *Introduction to trunks*, for more information.

Terms and abbreviations

ISDN (Integrated Services Digital Network)

A digital telephony network that allows the transmission of voice and data using approved protocols.

ISO

International Standards Organization

LD (Load)

An abbreviation for the term load, otherwise known as overlay program.

Link

1. Another name for a communications channel or circuit.
2. A button on certain types of telephones that users can press when they want to perform a switch-hook flash, instead of pressing the switch-hook under the handset.
3. A connection to another system, as in the Meridian Link application.

Loop

A transmission path within the system. Line cards and trunk cards share the transmission path using 32 timeslots.

MCDN

Meridian Customer Defined Network – A proprietary Nortel Networks out-of-band signaling protocol that provides ISDN features and services between Meridian 1 and DMS-100, DMS-250, SL-100, ESS #4, ESS #5 and Meridian 1 systems.

Main Distribution Frame (MDF)

The panel where wires from telephones and trunks are interconnected with corresponding wires from the system’s line and trunk cards. It is sometimes called the cross-connect panel or jumper panel.

Terms and abbreviations

MDF (main distribution frame)

The panel where wires from telephones and trunks are interconnected with corresponding wires from the system's line and trunks cards. It is sometimes called the cross-connect panel or jumper panel.

Meridian Mail

The voice mail system manufactured by Nortel Networks to be compatible with the Meridian 1. Meridian Mail is provided using an external application processor.

Meridian Passport

Meridian Passport upgrades the Meridian 1 system into a multimedia enterprise networking system. It allows voice, data, image, and video traffic between enterprise locations to be consolidated onto a multimedia wide area network.

Message Center

A configuration where telephones are programmed to redirect calls to either a specific telephone, the attendant, or voice mail when calls are not answered or the telephone is busy.

Mnemonic

A code used as a memory aid.
Mnemonic codes are also used in programming.

Multiple Appearance DN

A DN that is programmed to appear on more than one telephone or more than one key on one telephone.

NANP (North American Numbering Plan)

The North American Public Exchange Network has been divided into geographical areas with three digit codes which precede the seven digit local telephone number of the subscriber. Previous to 1995, the three digit area codes assigned to each geographical area were in the format where the first digit was any digit between 2-9 and the last digit was any digit between 0-9. The middle digit was either 0 or 1. As of January 1, 1995, the middle digit can be any digit between 0-9. This increases the capacity of three digit codes available.

Terms and abbreviations

nB+D

Use the nB+D configuration if you need more than one T-I or E1 facility to handle the connections between two switches. n represents a number between 2 and 16.

NCOS

Refer to Network Class of Service.

NMS

Refer to Network Message Services.

NE

Refer to Network Equipment.

Near end

The local Meridian 1 system where your Tie trunk or Private circuit terminates.

Network Class of Service (NCOS)

A class of service that determines network access.

Network Equipment (NE)

The part of the Meridian 1 that serves to interconnect terminal equipment. Network Equipment also provides services such as conferencing and tones.

Network Message Services (NMS)

There are two types of Network Message Services available, Message Center and Meridian Mail.

With Network Message Services, messages can be taken at a node for users at several remote sites as well as users at the node where the message center or Meridian Mail is installed.

New Flexible Code Restriction

A software package that allows numbers to be restricted when calls are dialed with or without ESN software installed.

Terms and abbreviations

Node

A Meridian 1 equipped with Network Alternate Route Selection (NARS) software.

NPA (Numbering Plan Area Code)

The North American Public Exchange Network has been divided into geographical areas with three digit codes which precede the seven digit local telephone number of the subscriber. It is used when dialing a long distance call. For example, when someone in a city in the 205 area code wants to direct dial someone located in the 613 area code, they dial 1613 followed by the person’s seven digit telephone number.

NTP (Northern Telecom Publication)

The manuals that are published by Northern Telecom that describe how to engineer, install, program and maintain all the features, services and components of Meridian 1 systems. Many of these manuals are shipped with every system. Some of the manuals are optional and can be ordered.

Numbering Plan

The leading digits which are assigned to Directory Numbers, trunk route access codes, and the Special Prefix (SPRE) code for feature activation within one customer group database. The same digits cannot be assigned to two different numbers or codes. This is called the “leftwise unique rule”.

For example, it is permissible to have access code 11 for SPRE and access code 130 for paging trunks but it is not permissible to assign access code 55 to a TIE trunk route and access code 552 to a dictation trunk route.

An example of a typical numbering plan follows.

Terms and abbreviations

A typical Numbering Plan:

- 0 Attendant
- 11 Special Prefix Code (SPRE)
- 2XXX Directory numbers (DNs)
- 3XXX DID Directory numbers
- 4 Unassigned (for future use)
- 5 Unassigned (for future use)
- 6 Unassigned (for future use)
- 7X Access codes for TIE trunks, paging trunks, dictation trunks
- 8 Access code to Automatic Route Selection calls
- 9 Access code to COT trunks or local calls

NXX (Public Network Exchange code)

The first three digits of a seven digit telephone number assigned to each subscriber in North America. These digits identify the Central Office to which the subscriber is connected.

Octothorpe key

The key labelled with a # (hash) symbol. In Europe, this key is called the fast-connect key.

Out-of-band signaling

Signaling that is separated from the channel carrying the information - the voice, data and video for example. The signaling includes dialing and other supervisory signals.

Outpulse

To transmit digits on external trunks to other systems. Also, telephones can outpulse digits to the system to which they are connected. The user causes either one of these types of outpulsing to occur by dialing digits on a telephone.

Terms and abbreviations

Overlap receiving

The destination address digits are received one digit at a time or a few digits at a time, as they are dialed, or after a short delay, rather than waiting for all digits to be collected first. Several signaling messages are needed to complete the incoming digit strings.

Overlap sending

The destination address digits are sent out one digit at a time or a few digits at a time, as they are dialed, or after a short delay, rather than waiting for all digits to be collected first as with Enbloc signaling.

Overlay programs

The programs used for entering the data required to customize a system include data for such things as features, telephones, trunk groups, hardware, data devices, and Automatic Route Selection, to name a few.

PBX (Private Branch Exchange)

A telephone switch that serves trunks and telephones.

PRI

Refer to Primary Rate Interface.

Primary Rate Interface (PRI)

A standard for connecting telephone switches. In North America, a PRI connection is composed of 23 B-channels at 64 kbps each, and one D-channel at 64 kbps. In Europe, a PRI2 connection is composed of 30 B-channels at 64 kbps each, and one D-channel at 64 kbps.

Prime DN

The DN programmed on key 0 of a telephone.

Private line service

Also known as leased-line service or point-to-point service.

Terms and abbreviations

Private network

Trunk connections between PBXs and Centrex systems that carry calls between users who reside on different systems belonging to one organization or company. The private network trunks may be used for calls that end up on the public network, if that is allowed in the area where the systems are installed.

Prompt

A mnemonic presented by the system when you are programming or issuing commands to the system.

PSTN (Public Switched Telephone Network)

Otherwise known as the public network. The global network made up of telephone switches operated for the public by telephone utility companies and governments.

Public network

Refer to Exchange network.

Redirection

Redirection refers to the diversion of calls to a telephone or voice messaging port when the originally dialed DN is busy, not answering, or redirection features such as Call Forward are active.

Release

When used in terms of Generic X11 software, a Release is a version of software that contains new features or capabilities compared to the previous Release of software.

Response

A mnemonic you type, in answer to a prompt, when you are programming.

Restriction

Preventing telephone users from making certain types of calls or accessing certain features.

Terms and abbreviations

Ring Again

A feature that allows a telephone user to queue for a busy telephone or trunk group.

Service change

A term used, when you are programming, in administration overlay programs.

Slot number

A numbered designator that indicates where cards are located within the card cage of the modules.

Software package

A component of software that, if equipped, provides certain features and capabilities. Software packages are listed by a mnemonic or a number or both.

Starting arrangement

The protocol used between Meridian 1 PBX and a Central Office to control digit collection. Refer to immediate start, delay dial and wink start.

Superloop

A transmission path within the system. Intelligent line and trunk cards share the transmission path using 128 timeslots.

T1

A digital transmission facility with 24 channels, each with a data rate of 64 kbps.

TDS (Tone and Digit Switch)

A card that the system uses to provide many different tones to users.

Terminal Number (TN)

A physical or hardware location address, consisting of a network loop number, PE shelf number, PE card number, and unit number.

Terms and abbreviations

TN

Refer to Terminal Number.

TIE trunk

A dedicated circuit that connects two Meridian 1 systems or a Meridian 1 system and another switch. May also be called a private network trunk. Refer to Appendix 1, *Introduction to trunks,* for more information.

Time slot

An interval of time during which you occupy a shared transmission path during an active call.

Traffic

A measurement of the level of activity of a specific resource.

Tributary switch

A switch in an ESN network that is connected to a main with TIE trunks.

Trunk

One or more pairs of wires that connect one system to another. There are many types of trunks, distinguished by the types of calls they are designed to carry and the types of systems they inter-connect. Trunks are grouped together by type into trunk groups. Some examples of trunk types are TIE trunks, central office trunks, and Foreign Exchange trunks.

Trunk group

A defined set of trunks that can be used interchangeably by the system to reach a specific destination. Also called trunk route.

Trunk route

A defined set of trunks that can be used interchangeably by the system to reach a specific destination. Also called trunk group.

Terms and abbreviations

Virtual Private Networks

In Europe, the EuroISDN Trunk -Network side configuration, offers network users reduced telephone bills by using their Virtual Private Network instead of the public network. The Meridian 1 may be used as a “remote point of presence (RPOP) in a VPN, closest to the customer premises. It supports many different trunk connectivities and it has a lower cost than a public network switch.

In another type of VPN configuration, a Central Office can be configured to accept Location Code dialing on public network trunks from a Meridian 1. The users at the Meridian 1 are able to make calls using a Uniform Dialing Plan without the need for TIE trunks. The Central Office routes calls on the public network as if there are private network TIE trunks to other locations in the customer’s network. Where there is a separate VPN CO and a Telco CO, the Meridian 1 can be programmed to use the VPN first and the public network CO as a second choice, if that is economical.

WATS (Wide Area Telephone Service) trunk

In North America, a circuit between a public exchange network switch and a Meridian 1 system. WATS telephone calls are billed at a reduced rate. Refer to Appendix 1, *Introduction to trunks*, for more information.

Wink start

A signal used on trunks to control the transfer of dialed digits. The terminating switch finds and attaches its digit collection equipment, then sends a 140 millisecond off-hook, on-hook pulse to the originating switch that requests the digits to be sent.

Terms and abbreviations

About this guide

Who should use this guide

This guide is intended for telecommunications planners, network administrators, Meridian 1 engineers, and sales representatives. Use it when planning for implementation of networking on Meridian 1 systems.

How to use this guide

This guide contains background information, advice on implementation, and detailed instructions for planning networking on Meridian 1 systems.

The content of this book is focused on the information you need to know in order to plan the implementation of a network solution. It complements the technical and programming information presented in the NTPs. It does not cover the programming and installation of hardware already explained in the NTPs.

How the sections of this guide work

Terms and abbreviations section

This section provides a definition or explanation of the technical terms and abbreviations used in this guide.

About networking

The chapter called *About networking* presents several sections. There is an overview of the tasks involved in network planning, a diagram of the layers of network-related software that build on each other, trunk transmission guidelines, information about Call Detail Recording and Network Traffic studies. This information will improve your understanding of the material in the rest of the book.

About this guide

Network configuration

This chapter is comprised of individual sections devoted to the major networking configurations involved with Electronic Switched Networks (ESN) and Integrated Services Digital Networks (ISDN).

ISDN networks build on an ESN platform. Therefore, information on ESN is presented first. You should become familiar with ESN before you learn about ISDN.

The *Introduction to ISDN* section contains a summary that gives the basic characteristics of ISDN networking in different parts of the world. Each type of ISDN network also has its own section with more detailed information.

Services and functions

Features and applications that build upon ESN and ISDN network platforms are detailed in the sections included in this chapter.

Interworking and gateways

The information in this section will be of interest to anyone whose network combines various systems using different protocols.

Appendixes

The appendixes provide basic or advanced information that expands on material in the book, in a detailed way, or provides answers to questions you may have. There are also tools and job aids provided.

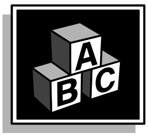
Index

Use the index to look up topics of your choice related to networking. Page number references are listed for each topic.

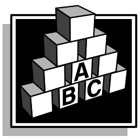
About this guide

How icons and symbols are used

This symbol is used to alert you to information that is of major importance. The text that appears beside the symbol can vary from one situation to another, and it is important that you read it.



This icon illustrates basic building blocks. It appears in the Basic configuration part of each section. It symbolizes that the information is basic to the implementation of the feature or function being discussed.



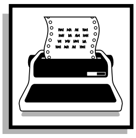
This icon illustrates basic building blocks with additional blocks added. It appears in the Improving feature performance part of each section. It symbolizes that the information concerns the enhancements or optional capabilities that you can apply to the feature or function being discussed.

This icon illustrates a person who is directing traffic. It appears in the Control tips part of each section. It symbolizes that the information helps you improve control of the system operation, costs, and security because of the feature or function being discussed.

This icon illustrates a graph showing improvement trends in system efficiency. It appears in the Administration tips part of each section. It symbolizes that the information is related to the administration of the system with respect to the feature or function being discussed.

This icon illustrates a person doing telephone training. It appears in the Training tips part of each section. It symbolizes that the information is related to the use of training with respect to the feature or function being discussed.

About this guide



This icon illustrates a printout at a maintenance terminal. It appears in the Maintenance tips part of each section. It symbolizes that the information is related to maintaining a network with respect to the feature or function being discussed.

Availability of product

This guide is intended for a global audience. The information in this book identifies the regions of the world where different solutions are offered. If you are an end user telecommunications planner or network administrator, confirm with your system supplier whether products are available in your region before you plan the details of your implementation strategy.

How the sections apply to market regions

Section of the book	Region(s) where applicable
Network planning	global
Implementation diagram	global
Transmission guidelines	global
Call Detail Recording	global
Network Traffic studies	global
NARS and BARS	global
Coordinated Dialing Plan	global
Flexible Numbering Plan	global
R2MFC	global
Automatic Number Identification	global
Introduction to ISDN	global
Primary Rate Interface	global
ISDN Signaling Link	global
Basic Rate Interface trunks	outside North America
— continued —	

About this guide

Section of the book	Region(s) where applicable
DPNSS1	global
DASS2	UK
EuroISDN	Europe
QSIG	global
NI-2	North America
Virtual Network Services	global
Authorization codes	global
Network Signaling	global
Basic Call Service	global
ESN on ISDN	global
CLID and Name display options	global
Attendant-related features	global
Restriction-related features	global
Redirection-related features	global
Network applications	global
Optimizing trunks	global
Queuing	global
Other network features	global
Interworking and gateways	global
Introduction to trunks	global
ESN Expansion feature	global
Software and benefits	global
Common questions and answers	global
Compatibilities chart	global
Testing a network	global
Trouble reports	global
Troubleshooting flowchart	global

About this guide

Language standards and translation

This guide is written to North American English standards.

For versions of this guide in languages other than North American English, please check with your system supplier or with Nortel Networks.

Additional Meridian 1 documents

There are a number of other Meridian 1 documents that offer information you may find useful while planning. They may also help you find solutions that meet complex network requirements.

Where possible, the binder names are listed, for easy identification of the books.

North America

- *X11 Software Features Guide*
- *Networking Features*
- *X11 Input / Output Guide*
- *System Planning and Engineering Guide*
- *ISDN Basic Rate Interface*
- *X11 System Management Overview, Applications, and Security*
- *Controlling Access Privileges* - coil bound book
- *Networking Feature Document* - coil bound book
- *Library Navigator* - coil bound book

About this guide

International

- *X11 Software Feature Guide*
- *Networking*
- *X11 Software Input / Output Guide*
- *Planning and Engineering Guide*
- *Planning and Engineering (UK) Guide*
- *ISDN Basic Rate Interface*
- *DPNSS1*
- *DASS2 and DPNSS1 (UK)*
- *Software System Management*
- *International Networking Feature Document* - coil bound book
- *International Customer Documentation Catalogue* - coil bound book

About this guide

Network Planning

Introduction

Network planners and administrators can be involved in a wide variety of different tasks. Their work can involve complex projects such as organizing separate sites into a more unified, sophisticated network or simple projects such as adding more trunks to a route at one site.

Most network planners perform their duties in a particular sequence, no matter how complex or simple the project is.

The sequence of duties presented here is very general. Use it as a framework for your planning activities. Make modifications to suit your needs.

Sequence of network planning duties

1. Assess your business objectives
2. Assess your existing network
3. Learn about networking solutions
4. Investigate other solution requirements
5. Calculate the cost of the solution
6. Select the network solution that fits
7. Plan your implementation strategy
8. Plan the administration of the solution
9. Plan the maintenance of the solution
10. Implement the solution

Assess your business objectives

Network planners who work closely with senior executives are more likely to manage the network in ways that help to fulfill the goals of the executives. It is wise for network planners to be aware of the short-term and long-term plans the executives have for the company. This way, the network infrastructure can be in place, as required, to support these plans.

Some typical goals are:

- to save money
- to increase employee productivity
- to increase security
- to manage expenditures more closely
- to share resources
- to present a progressive image

Network planners can assist in attaining these goals. Once the executives view the network planner as a team member with a shared vision, proposals for network changes and improvements will be accepted more readily. Optimizing the network will be seen as an important part of company strategy instead of an expense that should be minimized.

The network solutions covered in this book are presented in such a way that planners can understand what objectives each solution supports.

Assess your existing network

Careful network planning requires current, complete information from each site about its:

- geographical location
- working trunks and routes
- network-related tariffs (present and pending)
- local carriers
- network costs
- traffic patterns
- billing systems
- job functions
- existing system hardware and architecture
- existing network-related problems
- suggested changes
- future growth
- future plans

Geographical location

Use a large map to indicate where each site in your network is physically situated, relative to the other sites in your network.

Network Planning

Figure 1- Network location map

= Node

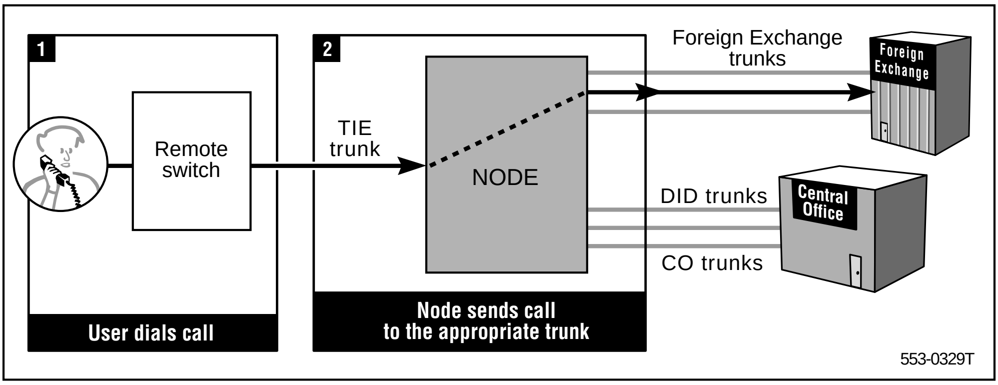
= ESN Main or Conventional Main

From the map, and with the help of your service provider, you will be able to calculate the distances between the sites in your network. You can use this information to choose the switches that would make the most sense to serve as nodes, from a cost point of view. Although cost should not be the only factor that determines what switches you will configure as nodes, it is usually a major factor.

Nodes and TIE trunks

Node switches have nearby remote switches connected to them with TIE trunks. When a user dials a long distance, chargeable call at a remote switch, it is sent to the node on a TIE trunk. The node sends the call out on one of its public network trunks.

Figure 2- Remote sites and nodes



Networks are usually set up in such a way that there are clusters of remote switches connected to a central node.

TIE trunks are usually billed as fixed rate trunks. In other words, you are billed the same amount for them monthly, regardless of the number of calls that are routed on them. The amount you are billed is usually based on the distance between the two termination points (switches). The farther apart the two switches are, the greater the monthly bill.

Network Planning

Network planners look for sites to serve as nodes that will result in the best utilization of TIE trunks between the node and the remote switches. The analysis is done based on the fact that TIE trunks will be used for **on-net** calls (inter-switch) and **off-net** (public network) calls placed through the node.

Since the bill for a TIE trunk is fixed, increasing the number of calls on it decreases the expense of each call.

The trunks and equipment at the node are set up to handle a large volume of long distance calls for users at the node and the remote switches. It is usually more economical to concentrate a high volume of calls from several switches on a pool of trunks at a node than to route calls out to the public network directly from each site.

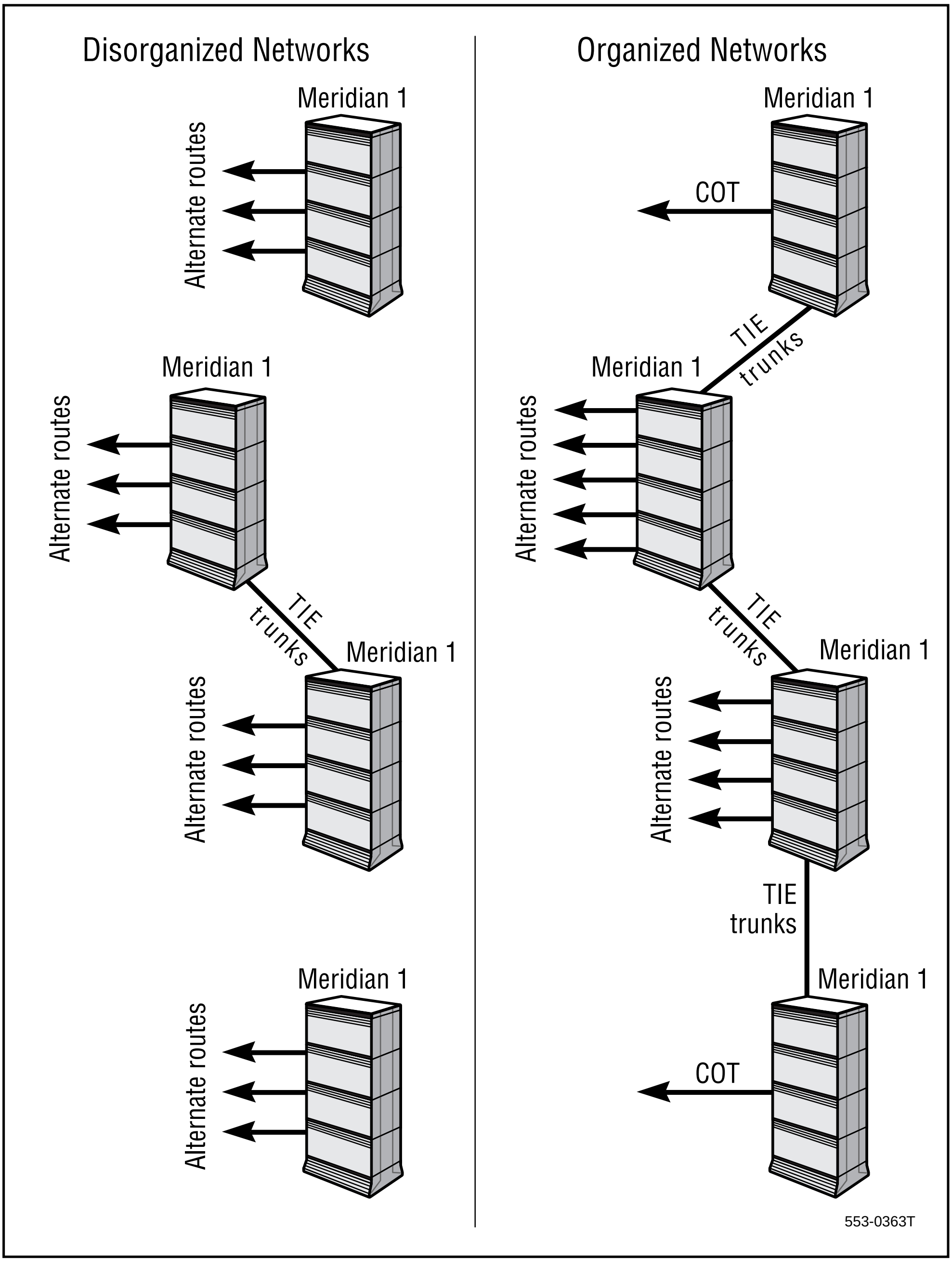
Studies have shown that you will need fewer public network trunks if you concentrate them into a centralized pool of trunks at the node than if you have public network trunks installed at all sites.

Bear in mind that there must be sufficient TIE trunks (private network trunks) provisioned between the node and each remote switch to adequately handle the call volume expected.

When a node and another site are separated by a great distance, the monthly bill you would pay for extra TIE trunks to handle off-net calls through the node would probably exceed any savings you would achieve from centralizing the public network trunks at the node.

It is the planner’s task to find the point at which it makes sense to install different public network trunks at the node and at the remote site, provision sufficient TIE trunks for on-net calls only and install software at the node and the remote site to handle public network calls cost-effectively. (Refer to the sections on BARS, NARS, and CDP).

Figure 3- Organizing a network



Network Planning

Working trunks and routes

- Network planners must have an accurate inventory of the number and types of trunks that are actually working at each site. Unfortunately, billing records may be out of date, inaccurate and unsuitable for this purpose.
- Plan to involve the local technicians in doing the inventory and testing that is required to get accurate information.
- Take advantage of this opportunity to clean-up the billing records.

Tariffs

- Make sure you understand how each type of trunk is designed to be used. For example, if there is a Foreign Exchange trunk at a site, find out what exchanges are in the free calling area of the distant Central Office to which the trunk is connected.
- Collect information about the fixed, monthly charges for the different kinds of trunks that exist at each site.
- Find out about how calls are billed, 24 hours a day, seven days a week for each kind of trunk that exists at the site.
- Find out what the service charges are for adding new trunks, if your plan will include expansion.

- Find ways to keep yourself informed. Rates change and new services are offered constantly. Optimizing your network costs and functionality requires constant attention from network planners and administrators. When rates change or your network call traffic changes, you may want to adjust your programming and trunks so the calls at each site are being routed in the most cost-effective way possible.

Carriers

- Use this opportunity to do some cost comparisons between carriers, if there is a choice of carriers in your area.

Network costs

- Collect long distance billing records for several busy or average months.
- Investigate how the chargeable calls are billed. Telephone companies do not usually bill for the exact duration of the calls, they round off the usage in some way.
- Use Call Detail Records (CDR), if they were collected during the same billing period, to find out what the trunk usage really was. You will need this type of data when you analyse calling patterns and when you provision trunks, later in the project.

Traffic patterns

Some of the major responsibilities of a network planner are: to find the most appropriate switches to serve as nodes, to configure the proper number of TIE trunks from the remote switches for the expected traffic, to provision other trunks at the remote sites if required, and to provision the proper type and number of trunks at the nodes to serve users across the network.

When you first configure a network and on an ongoing basis, you need accurate trunk usage data from each site so you can provision trunks properly. Traffic study data gives you that information. Refer to the section called *Network Traffic studies* for more information on how to use Meridian 1 traffic studies.

- Before provisioning trunks at a site, you must decide if you want to provision trunks to handle your average traffic load or your busiest traffic load. Assess the impact on your business if an incoming or outgoing call is lost because of insufficient trunks during your busiest times.
- You must also decide what percentage of blocked calls your users or callers will tolerate. You will require more trunks if you want to guarantee a maximum of 2% blockage compared to 5% blockage.

  Decide what blockage is acceptable on each trunk route. You may find that 5% blockage is acceptable on a TIE trunk route used only for on-net calls but you may want a maximum of 2% blockage on Central Office trunks used by customers.

Network Planning

Traffic data you need

In order to provision your trunks properly, you need data on the total usage times of all calls on each route during the average busy hour of a week or the busiest hour of a week.To provision trunks properly, unanswered and very short calls should be part of your total usage data. These calls contribute to the load on your trunks and you will need to provision your trunks with this usage included.

If the records are not sorted automatically for you, you may have to do some manual calculations, totalling up unsorted records.

- Find out if all calls that seize a trunk are part of your CDR and traffic study data.
  - Sometimes CDR downstream programs eliminate very short, uncompleted calls from reports since they are not billed. When you are using CDR for provisioning purposes, you need to stop eliminating these records.
  - You may have answer supervision activated on your system and only answered calls are printing CDR records. When you are using CDR for provisioning purposes, you need to track unanswered calls as well.
  - Traffic studies normally include only calls that are considered to be established (those that seize a trunk for longer than the End-of dialing timer). When the Trunk Seizure option is enabled in the configuration record of your system, the traffic study accumulates data for trunk usage upon trunk seizure, not just for established calls. This option was introduced in Release 21. Enabling this option allows you to do a better job of provisioning.
- Schedule a traffic study at each site for an average busy week or a peak busy week. You will use the trunk group traffic study data to provision the proper number of trunks.

  Plan to run studies regularly during the year, to monitor trunk usage and to optimize your network on an ongoing basis.

- If there is no recent traffic study data, you can use Call Detail Records (CDR).

  However, CDR is usually only run on trunk routes that handle long distance or chargeable types of calls. You need CDR records for local, non-chargeable calls as well, if you are to provision your trunk routes properly. You may need to activate CDR on all outgoing calls for a typical week or month in order to get the data you need.

- If you run studies or collect data, the network administrator at each site will probably be responsible for supervising those tasks. Ask them to check CDR devices and traffic study printers frequently to ensure they are in good working order. Try to prevent data from going missing or a ruined study due to lack of paper in a printer or a malfunctioning device. If no one notices for several days, the study will have to be re-scheduled. Assign someone to retrieving the hard copy printouts daily, to ensure they are stored safely.

- Traffic study analysis must be tied in with tariff and cost analysis. Network planning can become the art of deciding when to replace one kind of trunk route with one that makes more sense from a utility and cost point of view.

  For example, it is possible that the public network calls routed from a remote site through a node could decrease and that using TIE trunks to the node would no longer be cost justified for off-net calls. At some point, it will make more sense to remove the TIE trunks used for that purpose and install public network trunks at the remote site instead. A planner should then also consider installing software (such as BARS) at the remote site to handle calls effectively there.

  Analysing the point at which these kinds of changes should be made may form the major part of the planner’s job.

- If your network costs and traffic patterns fluctuate frequently, it may not be cost-effective to react to every fluctuation. React only to major trends.

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Billing system

- Give some thought to your present or planned network billing system.
- Choose the billing strategy that best suits your company. Here are some examples of how to bill users:
  - bill each end user for all calls, except for non-chargeable Central Office trunk calls. If you are billing them for the use of fixed rate trunks such as TIE and Foreign Exchange, a flat rate per call must be applied. You can use CDR and traffic study data to calculate that rate and to keep it accurate.
  - bill users for their long distance calls only, not TIE trunk calls to other locations on the company network. The charges for TIE trunks may be paid as a lump sum by each site or paid for entirely by head office as a corporate expense.
  - bill the users in such a way that the money received is greater than the expenses incurred. In so doing you can generate enough revenue to pay for regular system and network improvements.
- If you intend to bill end users for their calls, you must install and administer some kind of Call Detail Recording system. There are many different kinds. Consult your system supplier about which ones they recommend.
- If you do not want to install CDR systems at each site in a network, and you still want to bill users for every chargeable call, you must route the chargeable calls from remote switches through sites that do have CDR systems. Additionally, you must implement software and hardware that allows the user's identity (usually a unique Directory Number) to travel through the network with their calls. This Directory Number appears in the record that is made for each call. If you do this, the system administrators for the CDR-equipped systems will be able to sort the call records based on the caller's DN and send the bills to the proper users at the remote sites. (For more information on this, refer to the *CLID and Name Display options* section.)

- If you decide against having CDR systems installed at each site, weigh the expense of additional TIE trunks that may be required to route the calls through CDR-equipped nodes, the cost of the software and hardware that allows the caller's DN to be sent between switches and the CDR administration costs.
- You will benefit in other ways when you install software and hardware that allows the users' DNs to travel with their calls. You may find that these additional benefits help to cost justify the solution. Refer to the sections in this book on ISDN networks, for more information on the types of networks where this kind of software and hardware can be used.

Job functions

As a network planner you must understand the job functions that users perform at each site. Job functions have an impact on the use of the network. For example, call centers receive a large volume of incoming calls. You may want to configure your system so certain incoming trunk routes are dedicated to the call center groups. In this way, incoming calls to other user groups at the same site are not affected by the call center traffic. Similarly, telemarketing and sales groups can have a high impact on outgoing trunk groups.

Dedicating trunk groups to certain user groups may be a good idea for control and study purposes. However, segmenting trunks into separate groups reduces their efficiency. For example, fifty trunks in one trunk group are more efficient (handle more call traffic) than five trunk groups with ten trunks in each group.

Existing system hardware and architecture

- You will implement network solutions more effectively if you understand your existing systems from these points of view:
  - physical hardware and architecture
  - what the components are
  - how the components work together
  - capacity

- what the rules and limits are for system growth
- how your system is configured (used slots and spare slots)

◆ performance

- what the guidelines are for system operation
- how your system is performing compared to the guidelines

◆ To ensure that you understand the hardware-related costs that will result from the network changes that you are considering, become fully acquainted with the system architecture, capacity, and performance and the way the systems are configured at each site.

◆ Running traffic studies at each site will help you analyse the existing system performance. Your network solution may not be as effective as it is meant to be if you do not address other system-related issues. For example, if the processor in the existing system is already running out of real time at peak periods, implementing ISDN without a processor upgrade will not go very smoothly. Adding additional trunks to a system that already needs more loops will result in a poorer grade of service for everyone on the system.

You can learn more about traffic studies in

- the section called *Network Traffic studies* in this book
- the *Basic Telecom Management Guide* (basic study information)
- the *Traffic* section of the NTPs (detailed study information)

Existing network-related problems

◆ Contact the network administrator at each site. The network administrator may be able to provide you with information that will help you, especially if you do not have enough time to run studies before you attempt to reconfigure the network. The information they have about network-related problems will help you focus on the most serious issues first.

- The historical data that administrators have can be useful as well. You may be able to identify trends or find out about solutions that were tried in the past. If previous solutions did not address problems, you need to know about them before you implement your own solutions.
- Here are some suggested questions about existing network problems that you can ask the administrators;
- Are there any trunks groups that are busy frequently? (Do users report blockage, or does the attendant notice the Trunk Group Busy lamps on the console flash frequently, or do traffic studies indicate blockage?)
- Are trunk groups busy because some trunks are out of service or because there are insufficient trunks? (Have technicians reported problems? Are there error messages printing out on maintenance terminals? Do traffic studies show trunks out of service?)
- Is there little or no usage on some trunks or trunk groups? (Look at CDR records, error messages on maintenance terminals, and monthly bills.)
- Is system equipment affecting network performance? (Does the existing system processor need to be upgraded? Are there any problems with cards or shelves? Are there any power problems? Any troubles caused by outside equipment?)
- Are there trouble reports from users? (Are there any training issues, directory issues, dialing concerns or common questions?)
- Is there a process in place to handle trouble reports?
- Is anyone else on site involved with network issues?

◆ If each site does not have a network administrator, you may treat this as a network-related problem. If your network improvement plan will require someone on site to implement and administer it properly, you will need to make adjustments before you proceed. Either choose a solution that does not require as much administration or consider assigning someone the task of

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administration. It is usually true that when there is one contact person or department at each site dealing with network-related issues, network performance is improved.

Suggested changes

- Ask on-site administrators for their opinions and suggestions for changes. This information can be especially useful if you are inexperienced at network planning or if you are unfamiliar with the particular network you are managing.

  Solicit the opinions of the administrators when you are considering more than one alternative solution. They may be able to help you decide. They know the local site well enough to be able to point out any factors that might affect the solutions you are considering. For example, if your proposed change to the network will require a great deal of user training, you will need to know if the training can be done and whether it can be done in the given amount of time. If the users' jobs will prevent them from being trained quickly or at all, you should modify your plan accordingly.
- If you are proposing changes to the processes used in billing for network expenses, you will need to coordinate this with the on-site administrators and train them well.

Future growth

- Plans for company-wide growth or growth at individual sites will have a major impact on your network planning. Network planners who communicate with corporate planners and site administrators can prepare the network properly for growth and changes.
- You may need a great deal of advance warning before you can install new trunks, especially if additional equipment, system expansion or an upgrade is required to accommodate the growth.
- Plans to expand or reduce staff, or to change calling patterns must be communicated to network planners so the network can be adapted to support the changing needs of the company.
- If growth is imminent, network and system planners include that consideration when they plan to install, upgrade, or expand a system.

Future plans

- As a network planner, you will need to be aware of short-term and long-term plans that may be in place at each site. These plans can include information about increased or decreased staff, changes to job functions, new ways of doing business, new customers or suppliers, and changes affecting on-site network administration personnel.
- If you have an overall network focus, you may be in a good position to notice that people in different locations share the same job functions or problems. There are many network applications that allow people or systems to be shared across your network. These applications can save you money, improve your customer service, improve the efficiency of your people and your network and centralize your network management.
- Encourage people to keep you informed about changes they are considering in the ways they do business. You may be able to use the network to support their objectives. Also, if there will be fewer users or lower traffic, you need to be kept informed so the network can be kept in line with the decreased needs of the users.

Learn about networking solutions

- Once you have gathered the information you require from each site on your existing network, you can learn about the network solutions that are offered by Nortel Networks.

  Some suggested sources of information are:

  - your system supplier representatives
  - user (SL-1 User) or distributor (Distributor Alliance Council) groups
  - NTP manuals
  - sales and marketing literature
  - telecommunications magazines

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- tradeshows
- courses offered by your system supplier or Nortel Networks

- New solutions are introduced regularly, so it is important for you to remain informed. It is also important for you to keep your system supplier informed about changes that may be occurring in your network. This helps your supplier keep you up-to-date when new software and hardware is introduced that could be of benefit to you.
- Before you read the *Network configuration* sections of this book, you should find out which network protocols, features and hardware are supported in your market region. Then you will know which features and applications you can implement and where gateways between different types of networks will be required. Refer to the information in the sections called *About this guide* and *Introduction to ISDN* to find out what is available in the different market regions.
- A major goal for many network planners and administrators is network-wide uniformity. Most planners prefer to implement the same network signaling and features at all sites, if possible. This makes network management and training much easier. Choose your solutions with this factor in mind.
- You may find that several sites require the same improvements. Often, network-wide solutions can address problems at individual sites. When you make proposals for changes across the network, it will be easier for you to justify them if you can prove that several sites share the same issues.

Investigate other solution requirements

- When you look at reorganizing a multi-site network, it often means that you have to reconfigure a system that was set up to handle calls for users at that site alone and transform it into a node, handling calls for users at connected remote sites as well.

  If you do this, there will be increased incoming call traffic on the TIE trunks to the node from each remote site. There may also be additional calls on the public network trunks at the node. You will probably need to install extra trunks. Therefore, you will probably need additional trunk cards installed at the node and at the remote sites.

You will need to plan for the extra cards that will be required. Some typical examples of the impact on your system are:

- Analog trunk cards must be installed in spare slots on peripheral equipment shelves. Digital trunk cards require space in network equipment shelves (on systems other than the Option 11.)
- If the extra analog trunks will overload the existing loops on the system, it is possible that additional network (loop) cards may be required to handle the extra load. The network cards sit in slots in the network equipment shelves. A digital trunk card must be connected to its own dedicated loop on a network card. You may need additional loop cards to dedicate to this. Therefore you will need to provision space for these cards on your system.
- Upgrading a network from ESN operation to ISDN operation requires the addition of D-channel Handler Interface (DCHI) cards. These cards occupy space in network shelves. (There is more information on D-channels in the sections on ISDN, later in this book.) You may need to install additional network shelves or network groups to accommodate them.

◆ When there are major additions to a system you may need to upgrade from one model to another or it may require system expansion or system reconfiguration.

  Your system supplier can help you understand what hardware is required for the network solutions you choose and the impact this will have on your systems.

◆ Sometimes the features you are considering will require you to upgrade to a more recent release of software. Or, you may need a faster CPU or more memory in your system in order for your solution to work.

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Remember that upgrading your system and software can give your users other benefits too. Investigate what those are before you implement your network solution.

For more information on system architecture and upgrades refer to:

– your system supplier

– the Basic Telecom Management Guide

– courses offered at Nortel Networks Training Centers

– NTP binders: Installation and Maintenance and Upgrades

- In a network, it is important for you to consider than when you change or upgrade one switch it often means that you must make the same change to any switch it is connected to on the network. When one switch signals the other with information, the other one can only handle the information properly if it has been upgraded too.

- If your network incorporates different network protocols and signaling, familiarize yourself with the gateways and interworking capabilities that are required between them.

- You must also investigate if there are external factors that may influence the solutions you can choose. Contractual agreements, laws, telco issues and regulations all influence what choices you can make.

- You may benefit from asking local network administrators or system suppliers at each site what network-related issues are present at each site. There may be everything from chronic trunk repair problems to the need for sophisticated new technology to support the users at each site. The solution you are proposing may not be implemented smoothly unless you deal with these other issues first.

- If your solution is going to require a great deal of assistance from site administrators or other staff, plan for that well in advance. If you will not receive the assistance you need, assess your solution accordingly.

Calculate the cost of the solution

Pricing

- When getting quotes for the equipment and software required for your network solution, consider coordinating this work with other telecommunications changes you may need to make.
- Stay in touch with your supplier and with your SL-1 User group (if there is one in your area) so you will know about special promotions when they happen.
- Arrange to have training costs included in the cost of the solution. You will need training for the network planning and administrative staff and for the end users.

Phased implementation

- If the price of a solution is beyond your budget, ask your supplier to discuss a phased approach with you. Breaking the implementation into phases will not only make it more affordable; phased installations often make the most sense from a strategy point of view. Some of the more complex solutions require a phased implementation, for technical reasons. Also, implementation and training may be easier.

Include cost savings

When you assess the cost of a network solution, remember to include the cost savings or other economic benefits of your proposed solution.

Sharing resources

For example, one of the main advantages offered by networking and the applications you can build into a network is the sharing of resources that you can do as a result. Here are some examples of what sharing resources can do for you:

- users at remote switches can share the node’s trunks and save the company money in trunk charges and lower administration costs

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- incoming calls can be sent to a shared group of users who are dispersed across the network. This may result in the need for fewer employees, increased business, lower 1-800 bills, more calls and more business.
- equipment such as Meridian Mail can be installed at a node and used by remote users as well
- CDR devices can be installed at nodes and track calls that originate in remote sites
- attendants can work at nodes and answer calls for remote sites also
- Message Centers can be set up at nodes and the people who staff them can take messages for users at the node and remote sites

The following two examples show you how networking can be used to share resources.

- If you propose Network Automatic Call Distribution (NACD) as a network solution, point out that it allows people in call centers at different network sites to handle incoming calls as if they are all part of the same call center. This results in better customer service (which may result in increased business), shorter holding times (lower 1-800 bills and more calls), and fewer staff than if each site handles only its own calls. If you can calculate the value that these benefits will bring to your organization, you can subtract that amount from the cost of the ISDN and NACD software packages and any related hardware that the solution requires.
- In North America, there are three services that allow B-channels in a PRI route to be used as required by many different service routes.
  - ISDN PRI ISA service to DMS (Release 12)
  - ISDN PRI CBC service to AT&T#4 (Release 16)
  - NI-2 TR-1270 PRI Call by Call Service Selection (Release 23)

With these three services, a B-channel acts as a trunk in the appropriate route as required for each call. The next time that same B-channel is used, it is assigned a service route number, as required.

This capability can reduce the number of channels you need to provision in the PRI route, especially if the peak busy hours of the service routes occur at different times of the day. Instead of having dedicated channels in some routes sitting idle at times when other routes are overloaded, all channels can be used as required for the route that needs them. If you have traffic study data on the public network routes at a site, it can help you decide whether this capability will save you money by reducing the number of channels you need.

There is information about these features and services in this book.

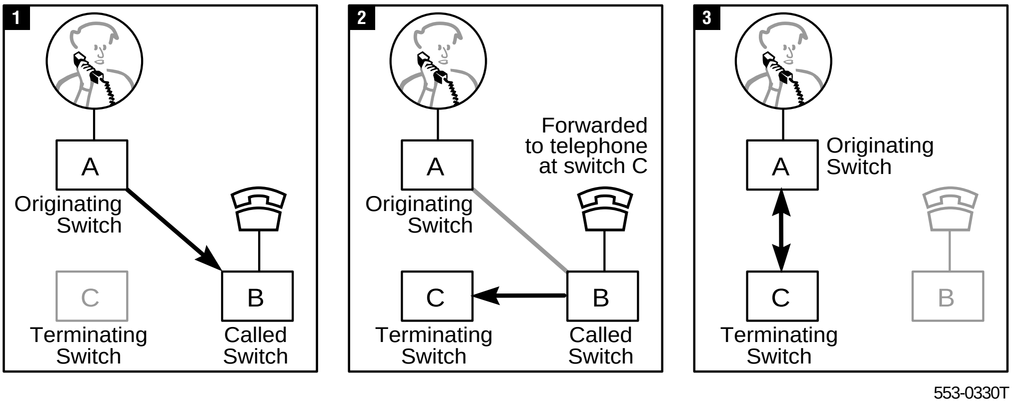
Network efficiency

Some applications can provide you with greater efficiencies in the use of your network. This helps to reduce your network expenses. Subtract the reduced expense from the cost of adding the application.

One such application is Trunk Route Optimization (TRO). (Refer to the section called *Optimizing trunks)*. Briefly, it works as follows: when a call is placed from one switch to another to a telephone that is forwarded, it can be diverted to another switch by the forwarded telephone. In the illustration that follows, the caller is at switch A and the telephone called is at switch B. This telephone is forwarded to a telephone at switch C. The TRO application finds the most direct route for the diverted call. It drops inefficient paths that the call may have taken because it was diverted along the way. The most direct route is the route between switch A and C. The TRO feature uses this one direct route trunk and releases the other two trunks. These two trunks can be used for other calls.

Network Planning

Figure 4 - Trunk Route Optimization



Without the capability to optimize, inefficient paths would continue to be set up and extra trunks would be used as a result.

With optimization in place, it is possible you will need fewer trunks with the added benefit that the quality of your network transmission will improve. Use these benefits to help justify the cost of the proposed solution.

Increased control

The control offered by networking software can reduce your network-related costs immensely. Some of the ways you can improve the control you have of your users and your network are:

- increasing use of inexpensive routes
- restricting use of expensive routes
- restricting certain numbers from being dialed
- determining what destinations users can call
- decreasing administration time required to monitor for abuse
- using TIE trunks to other locations for free calls only
- implementing time of day restrictions on trunk routes and types of calls
- controlling the calls made at nodes by incoming TIE trunk users

The increased control results in cost savings that help you justify the cost of the associated hardware and software.

Consider alternatives

Analog alternatives

If a digital networking solution does not fit your needs or your budget, consider an analog version of the same solution.

For example, if you do not have the call volumes or the budget to justify Primary Rate Interface (PRI), consider the ISDN Signaling Link (ISL) alternative. ISL offers many of the same functionalities as PRI, where voice grade networking is concerned, and has some benefits that you should consider.

- Analog trunks are usually not as expensive as digital trunks so the analog solution can be less expensive on a monthly basis.
- The analog alternative can provide certain capabilities (such as Backup D-channel) in a very cost-effective way.

The same argument applies to implementing the Analog Private Network Signaling System (APNSS), instead of the Digital Private Network Signaling System No. 1 (DPNSS1).

Other alternatives

The functionality that you are looking for may be offered by several different network configurations and applications. When you investigate the different ways to offer certain functionality on your network, look for products and services that do not require your network to be configured in ways that you cannot afford.

One example is Virtual Network Services. In order to provide ISDN Primary Rate Interface or ISDN Signaling Link features and functions across a network of Meridian 1 systems, TIE trunks are required between the systems. For networks with limited TIE trunks or even those with no TIE trunks between certain Meridian sites, VNS offers an affordable alternative. This feature uses ISDN D-channel signaling between two Meridian 1 systems but TIE trunks are not required as B-channels. You can use public network trunks. Additionally, VNS can be used as an inexpensive overflow alternative if you already have PRI or ISL but you do not have sufficient TIE trunks for peak periods.

Select the network solution that fits

Given your immediate and long-term requirements, several network solutions may apply. You can use the following factors, weighted in a manner most appropriate for your situation, to analyse which solution fits best:

- company goals
- cost
- compatibility/consistency issues
- reliability/robustness
- compliance with standards
- flexibility
- supplier support required
- effects of installation phase(s) on your business
- post-installation administration requirements
- training requirements

Corporate goals

Network planners who can support company goals with network solutions, often find that their recommendations for network improvements are approved more often than network planners who do not operate this way. If executives treat the network solely as an expense item and if they expect planners to focus strictly on saving money, they often miss out on opportunities that network enhancements can provide.

With the assistance of your company executives, prioritize your company goals and implement the solution that addresses most of these goals or the highest priority goals.

Network economics

Economics are often a key factor in determining which solution you can choose. However, the least expensive solution may not always be the best in the long run. Consider the benefits that the more expensive solutions can provide for your business. A simple example of this happens when a network administrator removes trunks to save money and then finds that there is increased call blockage. When there are insufficient trunks for customers and employees to use, the level of service that a business can offer will decline. This kind of result can happen when costs are given too much weight in evaluating a proposal.

Similarly, if a company's goal is to be perceived to be on the leading edge of technology, there will be some expense involved in getting there and staying there. If the end result is increased business, then the expense has been worth it.

If the best solution is also the most expensive one and if you do not intend to implement it for cost reasons, make sure other people involved with the network understand what the limits of the affordable solution are. They must have reasonable expectations of what will be accomplished by the solution you plan to implement.

Understand your business

Understanding the type of employees in your company and the callers who deal with your company will help you choose the most appropriate network solution. Allow your system supplier to gain a good understanding of your needs and your business. The solutions that they recommend will be the best possible for you.

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Administration

- You should also evaluate proposed network solutions from an administration point of view. If the network solution will require people with training and experience to install and maintain it and you do not have these people available to you, then the solution may not be the best one for you.
- Also, consider the solution from a training point of view. How well will end users and administrators adjust to it?
- After installation, plan to follow up with users and outside callers, to evaluate whether the solution is meeting the requirements that it was supposed to meet. Consider putting a contingency plan into place, in case the solution falls short of your expectations.

Plan your implementation strategy

Pre-installation approval

- Investigate who must sign any agreements before you purchase and install the software and hardware.
- Find out how much detailed knowledge you and the other people will require in order to adequately understand and approve the proposed solution. If training will be required before you sign any agreements, arrange to get it.

Roles and process

- Confirm what your role will be before, during and after the implementation of the solution.
- Make sure everyone on the implementation team understands each other’s roles.
- Ensure there is a process in place to keep everyone informed as the project progresses.

- Let the team know, early in the process, if you want to be very involved during the implementation (signing off programming sheets before any programming is done, for example) or if you would be satisfied with overviews or reports as the implementation progresses.
- If you intend to hand over the implementation part of the process to administrative people, ensure that they understand the rationale and goals for the solution. Give the implementers as much background information as possible to help them support the project.
  - If there is an overall design, such as for a numbering plan, that the team is to adhere to, lay that out in your instructions. Make the original design clear to the implementation team or the solution may not be installed in accordance with the proposal. When implementers do not have enough information, they may make decisions that go against the original plan for the solution. This is especially true as time goes on. It will help network planners and administrators, who were not on the implementation team, do a better job of administering and maintaining a network if there are records of what the original plan was and what the solution will require from them.
- Confirm the approvals process for each step of the plan.
- Confirm implementation schedules, especially if you plan to implement the solution in phases.
- Set up contingency plans to be used if there are problems. Establish rules of accountability for potential problems.
- Confirm how long after the implementation your system supplier representatives will be dedicated to supporting you. On major network changes their support after the change can be crucial to a smooth implementation.

Phases

- Many planners implement network solutions in phases for some of the following reasons:
  - costs can be spread out over time

  - benefits can be gained for some parts of the solution earlier than others
  - people can be given new information a little bit at a time, making it easier for them to adjust to major changes
  - solutions that build on other applications can be implemented in layers
  - it is easier to manage the changes
  - it is easier to implement with fewer administrative staff
  - it is easier to handle post-implementation questions and issues

- If the operation of your business will change as a result of each phase of the project, inform users and callers about what impact each phase will have on them. When you are implementing sophisticated, complex network solutions, the software must be programmed in layers and stages. This often means that there is a gradual transition from the old way of operating to a new dialing plan and then to a new set of features and functions. These features may go into effect in stages as well. Explain this to people so they know what to do while the transition is happening. They may need to use temporary procedures for short periods of time. To make the transition period smoother, keep the number of different procedures to a minimum and keep them as simple as possible. Let people know who to contact with questions during each phase.

Training

- Assess the training needs of your planning and administrative staff. Discuss the expertise of the system supplier implementers as frankly as possible with them.
- If the solution you plan to implement is very new, complex or unique, it may be worthwhile to arrange training for everyone on the implementation team together. You may be able to conduct it as a workshop, using the real network as a focus of the training. That way the team can address issues, concerns and questions that come up early in the implementation process together. The trainer may be able to act as a technology consultant and advise the team about the best ways to approach the implementation. Also when

questions are raised, everyone gets the same answer and understands why certain decisions are made when they are all learning together.

- The earlier you plan for this training the better. Courses offered by Nortel Networks are often booked well in advance so it is best to reserve early.

  End user training may require a great deal of planning and scheduling. The users will have to arrange their schedules around your training dates. Discuss how the training will be conducted and by whom. It is wise to have people on your staff who are trained as experts to act as a resource during and after the implementation. These experts may be able to conduct some of the training.

- If you plan to have support staff on site during the installation to answer questions and handle trouble reports, these people will need time and training to become very familiar with the implementation. Console attendants also need proper training if there will be major changes to the way they operate.

- Your system supplier may be able to put you in touch with other network planners or implementers who have already installed the solution you want to install. They may be able to give you valuable advice. If you belong to a user group, you may be able to contact experienced people through that organization.

Selecting sites

- Some implementers use a small system in their network as a test site to begin the implementation of a multi-site application. Since the site acts as a training ground for the implementation team, the people at the test site are told to expect some glitches. That way, if glitches do happen, the users will deal with them better. There is a problem-reporting process in place and the process is well understood by users at the test site.

  You might also decide to begin your implementation at a site where:

  – the users are able to deal with change easily

- the solution is needed most
- the business will be affected least, if problems do occur

◆ It is a good idea to conduct a review after the implementation has been finished at the first site. The review should result in an improved process to be used in the next phases of the project.

◆ Investigate the external factors that may interfere with the implementation. These factors could include:

- events such as audits. These are stressful enough without being compounded it with changes to the system and the network. Try to schedule telecommunications changes for a time when the users are not distracted with other events.
- having customers or visitors at the site. These are times when your staff usually wants to look their best. Your systems and network should be performing well during these times too. Making major changes at this time is not recommended.
- market events such as sales campaigns. If there will be predictable times when the network will be used heavily in the course of doing business, do not choose these times to make system or network changes. The last thing you want to do is cause an interruption in business because of your implementation.
- seasonal issues. For example, if summer is a slow time for your business, you might think it is a good time to choose for a change to the network, from a business perspective. However, if users are away on holidays, they will not be available for training so this could pose problems for a smooth implementation

◆ If your solution will be implemented simultaneously at more than one site, set up a process and schedule for the personnel who will be needed at each site for implementation and testing. If this will have to be done outside regular business hours, investigate what impact this will have on issues such as overtime costs.

◆ Keep the network administrators at all sites informed of the changes that are being made to the other sites in the network and the planned schedule. These changes may have an effect on their sites. Give them sufficient time to warn their own users and to tell them what to expect during these times. For example, when Network Attendant Service (NAS) is installed, attendants can be removed from the remote sites. The attendants at the node will answer incoming calls for users at the remote site. Users at the remote site will need information in advance about this and how it will affect them.

Miscellaneous

Get directories printed

- If the network solution you are implementing will change the dialing plan or the directory numbers in any way, plan to get directories printed early. That way you can familiarize users with the new instructions during training sessions.
- Attendants usually want these new instructions well in advance so they can become very comfortable with the new numbers or dialing plan. As a result, they are as efficient after the solution is implemented as they were beforehand.

Temporary procedures

If you can predict that there will be times during the implementation when calls will not be handled properly, you can make plans to do one or more of the following things:

- reroute calls to other locations at the same site or other sites
- answer calls with a recorded message if that is appropriate for the types of calls you expect
- have the attendants or message center staff handle additional calls temporarily
- give the callers overflow tone, if you think that is appropriate. Tell these people in advance, if possible, that this will happen for a given amount of time
- give callers an alternate number to call
- give callers an alternate way of doing business during the transition time

Plan the administration of the solution

If the solution you implement is to perform at peak efficiency, you must ensure that it is well administered after it is installed.

- Communicate clearly with the network administrators so they know what is expected of the systems and people involved with the solution. They must also understand what you expect of them.
- Involve administrators during the implementation. They will have a better appreciation for what they will need to do than if they are involved after installation only.
- Decide whether you need on site administrators or if the network can be managed from a central location. If you prefer centralization, plan for regular visits to the site or meetings with on site people to keep in touch with what is happening locally.
- If you have implemented a new billing system or a change to an existing one, plan to provide the administrators with the training and post-installation support they will need initially. Plan to check the recording or billing system for problems as quickly as possible so the effect of these problems can be minimized. For example, do not wait a month for long distance bills to arrive to discover that your recording system was not set up properly.
- Plan to run traffic studies after installation to verify that the programming was done properly and to monitor the effectiveness of the solution. Inform the people who will be responsible for these studies about their duties. Set up a process for handling the study data and analysing it.
- Set up a process for running traffic studies at least once a year or when major changes have been made at a site. You can use the data from these studies in many ways:
  - to monitor the network and the system for problems
  - to optimize the network and the system
  - to aid in your analysis before making changes

- Prepare a set of records for the network administrators and another set for technicians to use. These records should give them clear instructions on what they are expected to do to keep the solution running efficiently.

  Technicians who are unfamiliar with the site and new network administrators should be able to find out what they need from these records.

  The records can include:

  - the overall design for the solution you implemented
  - numbering plan constraints or rules
  - dialing plan considerations
  - programming instructions
  - how to interpret reports
  - information on internal processes
  - training information for new users
  - information about the network
  - contact names

- Implement a process for users at each site to use when they want to report network-related problems or if they have suggestions. Plan to use this information to make changes that may be required.

- There should be procedures at each site for programmers to use when making changes and when installing new telephones and trunks.

  - For example, when people program new telephones, they should have access to clear instructions regarding what restrictions and class of service to assign. You might put manuals in the switch room with the guidelines explained. Outline how you want telephones for people with different job functions to be programmed. Assigning default class of service and restriction values to telephones and trunks can lead to abuse of the network or problems that can be difficult to trace.

– The same is true when new trunks are installed. When adding a new TIE trunk to an existing route, it should be programmed the same way that existing ones in the same route are programmed, assuming there are no problems resulting from the existing programming. If a new trunk route is being installed, programmers should know who to contact for correct programming information.

- Work out network-wide policies and procedures for site administrators to use when planning to make changes to systems and the network. If administrators at individual sites implement changes without considering the effects the changes may have on other sites, this can have dramatic effects on a network and a company as a result. If administrators consider network-wide concerns before they make changes, they are more likely to coordinate their activities and to consider solutions that will address the needs of more than one site.
- Set up a way of tracking what improvements resulted from the implementation of the solution. Prepare to present these results to the people who were involved with the original proposal to implement the solution. Prepare a method for evaluating the results against the expected results.
- Develop a training package for new employees at each site.
- Consider doing refresher training. Plan to follow up with users who were trained before the installation. Plan to retrain users who require more training.

  Consider providing additional training to enhance the users' knowledge after they have had a chance to adjust to the new way of operating. Include time in the process for you to obtain feedback from users. This will help you evaluate the solution from their point of view and make modifications that are warranted.
- Encourage ongoing communication between the network administrators, the administrators and the planners, the system supplier and the planners and the planners and the executives.

Plan the maintenance of the solution

- Clarify what maintenance will be required to keep your solution running at peak efficiency.
- Clarify the responsibilities of your system supplier. Include these agreements in the contract you sign related to the solution.
- If you or your staff will be involved in performing maintenance tasks, work out agreements on roles before installation.
- Evaluate what training may be required for people to properly maintain the systems.
- Ensure training will be done before installation.
- If equipment will be needed in order to maintain the systems, ensure it will be available before installation. Investigate if there will be a charge associated with this equipment or the use of it.
- If maintenance will involve reading reports or printouts, ensure the proper people understand the data or have the time required for training.
- Make sure you understand what the symptoms are when the systems require maintenance. Explain the symptoms and the solutions to the network administrators. Give on-site technicians written instructions about what to do if different maintenance situations arise.
- If regular maintenance is to be performed, keep a log book at each site to record these activities.

Network Planning

Implement the solution

- Follow the processes that you have put in place.
- Distribute a master plan and master schedule to the implementation team members.
- Do regular evaluations of where you are in the installation process compared to the master schedule.
- When the unexpected happens, arrange to communicate new plans to the group. Make sure everyone understands how any changes affect them.
- Support the users, administrators and callers during the transition period.
- Obtain approvals that you need as the process continues.
- Stay in touch with on site people or be on site yourself for the installation.
- Evaluate the process as you use it. Modify your overall plan if required.
- When the installation is complete, obtain the final approval you require from the users, administrators, planners, and executives.
- Conduct a meeting to end the implementation phase and move into the administration and maintenance phase.
- Thank the implementation team for their efforts.

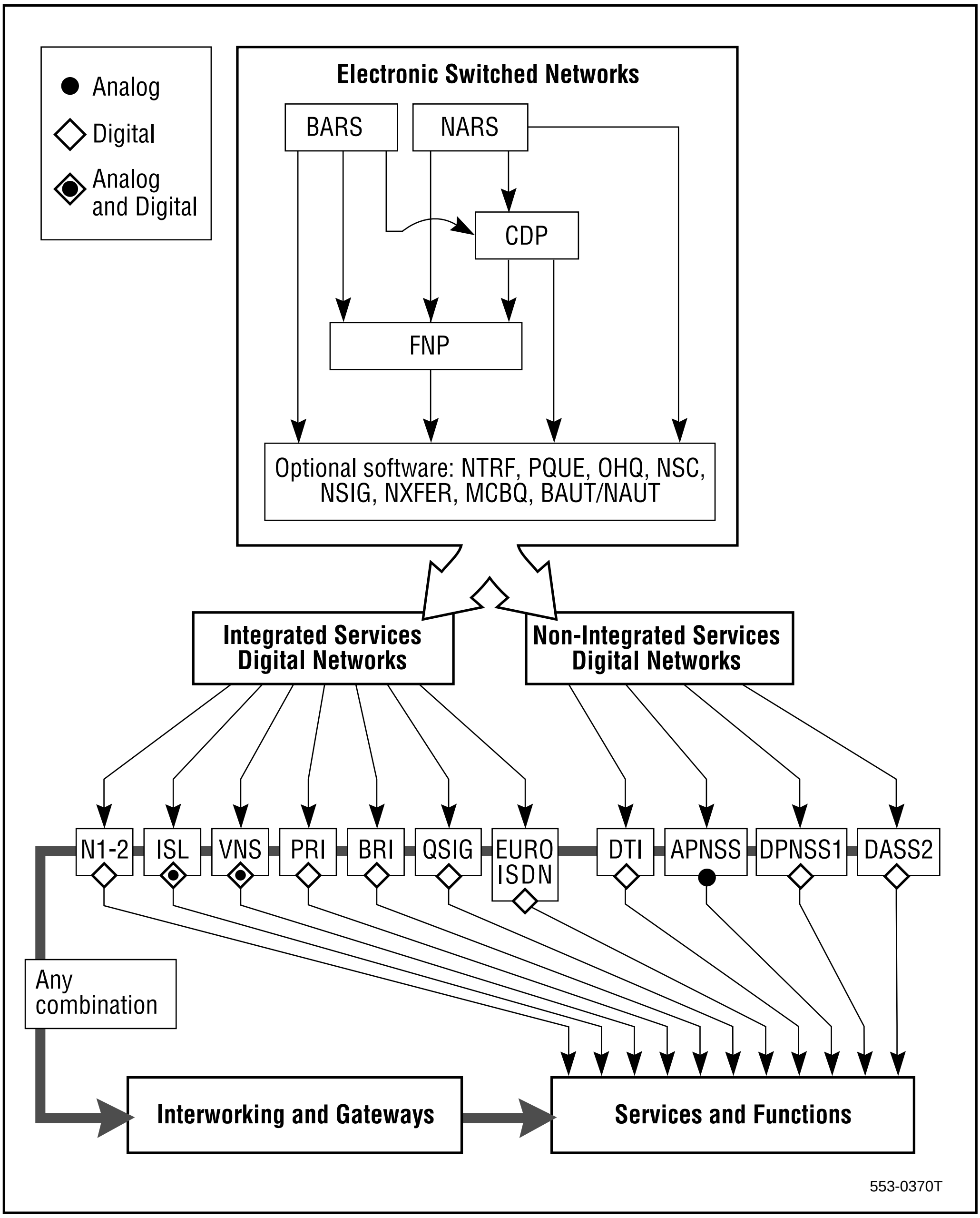
Implementation diagram

The illustration on the next page gives you an indication of the layers that build on each other in a network as it moves from a collection of sites to an ESN configuration, to an ISDN network with features and applications and possibly gateways.

Keep this layered approach in mind as you read the sections of this book.

If you read the sections in the order that they appear in the flow chart for the network solution you are considering, you will understand the material much better.

Implementation diagram



Transmission guidelines

Digit transmission

In an ESN network, digits of the called number are transmitted between pairs of switches in one of two modes: Dial Pulse (DIP) or Dual Tone Multifrequency (DTMF).

In Dial pulse, each digit is represented by a string of pulses. The digit zero is represented by ten pulses in North America. Each other digit is represented by the corresponding number of pulses. The pulses are transmitted as interruptions of direct current.

In DTMF, digits are transmitted over the speech path by a tone code. Digit transmission takes place at a higher rate than dial pulse (typically two to ten times faster).

Modern PBX switches are compatible with either mode. Older equipment is only compatible with dial pulse. It is preferable to use DTMF wherever practical, to take advantage of the higher transmission speed. However, dial pulse is sometimes required. Furthermore, there are some transmission impairments associated with DTMF. These impairments are normally only important when internal hardware is still connected after conversation begins to take place.

Modes of operation

A PBX switch can operate in one of two modes when routing a call over a TIE trunk. These two modes are:

- cut-through mode
- senderized mode.

Cut-through operation

With cut-through operation, a trunk is accessed immediately following an access code. Subsequent digits are forwarded to the trunk as dialed. The telephone user monitors call progress tones from

Transmission guidelines

connected switches. The user may be required to pause in dialing to monitor for dial tone, or may be required to abandon prior to completing dialing due to blocking tone.

Pure cut-through operation provides the greatest flexibility for providing compatible operation for calls originated at main and tributary switches.

- The number of digits transmitted to the node can be flexible.
- The node can prompt for additional digits, when required, for features such as the use of authorization codes.

A variation on cut-through operation is to provide DTMF to dial pulse conversion. The user dials DTMF digits, but dial pulses are transmitted to the trunk. The converter may block transmission in the caller to called party direction while waiting for digits.

Timing is often used to determine the end of dialing.When the switch has completed outpulsing, it waits a specified time for additional digits. If there are not any digits received in that interval, conversion is disabled. The timing may not properly distinguish a pause in dialing from a last digit, prematurely cancelling the outpulsing of digits.

Extending the end-of-dialing timer may cause transmission impairments during conversation if the called party answers quickly.

Senderized operation

With senderized operation, all digits of the called number are collected before an outgoing trunk is accessed. The transmitted digits need not be the same as those dialed (digit manipulation may change what was dialed in order to outpulse different digits). The user does not receive call progress tones until all digits have been transmitted. No tones are provided during dialing, other than a locally generated dial tone following the trunk access code.

Senderized operation limits flexibility. The remote PBX must be programmed to determine how many digits to collect before forwarding those digits to the node.

Transmission guidelines

The Meridian 1 ESN nodes combine cut-through and senderized modes of operation. The switch collects the access code and enough digits to select a trunk. During this interval, the operation is very close to a senderized mode. The trunk is accessed and a string of digits outpulsed (not necessarily the same as those dialed). Subsequent digits are forwarded to the trunk as dialed, in a receive and resend mode, which is closer to cut-through operation.

Start arrangements

Outpulsing control

After a TIE trunk is seized, digits of the called number will normally be transmitted. (The only exception is a manual trunk, which rings a designated telephone automatically when seized.)

Most terminating equipment requires a variable time interval to prepare for reception of digits. This time interval often depends on the call-processing load. Therefore, a fixed delay before sending digits may be unreliable. There are four commonly used ways of handling start dial control.

Immediate start

Applies in those cases where a short fixed delay is required for the switch to prepare to receive digits.

Delay for dial tone

A dial tone is provided when the switch is ready to receive digits. The delay for dial tone is used when a user controls digit sending, as in a TIE Trunk Tandem Network (TTTN).

Wink start

A momentary off-hook signal is sent when the terminating equipment is ready to receive digits.

Delay dial

An off-hook signal is sent to signify that the switch is not ready to receive digits. An on-hook signal is sent when the switch is ready to receive digits.

Transmission guidelines

The Meridian 1 switches in an ESN network are able to work with any of the start dial signals. Certain network applications require a specific form of start dial control. These are noted in the relevant sections of this book.

ESN transmission

Echo

All voice connections between telephones require two directions of transmission for a conversation to take place. When the signal transmitted in one direction is reflected over the other directional path, the caller hears his/her own voice with a slight delay.

Depending on the delay, the effect is perceived as sidetone, rain barrel effect or echo.

Two-wire facilities require matching impedances in order to prevent reflections. Four wire facilities do not generate reflections themselves, but they do not eliminate the reflection problem in built-up connections, since there are, in most cases, 2-wire connections to the telephones.

The objection to echo increases with the echo delay. The Via Net Loss (VNL) plan provides an increasing loss, depending on delay. However, the loss also reduces the received volume. Limits are placed on the amount of loss used to suppress echo. When these limits are exceeded, echo suppressor devices can be used instead.

Loss

The provision of good transmission requires the following compromises:

- the need for sufficiently low (one-way) loss in each direction to provide satisfactorily high received volumes
- minimum contrast in received volumes on different calls
- the need for sufficiently high round-trip losses to ensure adequate performance from the standpoint of suppressing talker echo, noise, and near-singing

Transmission guidelines

The following loss plan has been developed for ESN. The network is partitioned into node-to-node connections, node-to-main (remote switch) connections and main-to-satellite or tributary connections. The plan requires that:

- node-to-node trunks have a maximum loss of 3.5 dB (decibels)
- node-to-node tandem connections have a maximum loss of 4.1 dB
- node-to-main trunks have a maximum loss of 2.5 dB

These loss objectives are met by installing echo suppressors and reducing the loss to 0 dB on trunks when the objective loss is exceeded with VNL alone.

Your system supplier is responsible for ensuring a good quality of transmission on your trunks.

Refer to *ESN transmission guidelines* (NTP 309-3001-181) for further information. You can also refer to the *Software Feature Guide* for information on loss-related features such as: Alternative Loss Plan, Alternative Loss Plan for China, B34 Codec Static Loss Plan Downloading, B34 Dynamic Loss Switching, and China - Toll Call Loss Plan.

Voice quality performance

The quality of voice connection made over tandem trunks is a function of the composite characteristics of the trunks. Each trunk added to the connection degrades the overall transmission performance. Thus, some limits must be placed on the number of trunks permitted in tandem, as well as which trunks may be connected.

To maintain adequate voice quality and at the same time keep the routing restrictions from becoming too complex, ESN is partitioned into two basic connection categories, each with its own set of requirements. The connection categories are:

- node-to-node
- node-to-main, satellite or tributary

Transmission guidelines

Node-to-node connections

The restrictions on node-to-node connections are:

- No trunks have a loss exceeding 3.5 dB.
- A combination of trunks used for a valid connection not having a loss exceeding 4.1 dB.

Split echo suppressors are provided at each end of each trunk equipped with echo suppressors. Each echo suppressor is enabled or disabled by the switch at its end.

Generally, no more than three TIE trunks should be connected in tandem. A limit of four is imposed between echo suppressor controlled trunks. Software is arranged to disable echo suppressors when it tandems a call from an echo suppressor controlled trunk to another such trunk. It does not disable echo suppressors on other connections.

Tandem connections of echo-suppressor controlled trunks are permitted, provided the intermediate echo suppressors are disabled. When a direct connection is made between two echo-suppressor controlled trunks, the switch disables the echo suppressors it controls, thus meeting this requirement. Paths with one (or more) intermediate non-echo-suppressor controlled trunks between echo-suppressor controlled trunks are not allowed because the intermediate echo suppressor is not disabled.

Node-to-main, satellite or tributary connections

Restrictions on these TIE trunks are described below.

The node-to main trunk is normally a land circuit not exceeding 250 miles (400 km). If this objective cannot be met, the main PBX is treated as a node for transmission planning.

The node-to-main TIE trunk, if less than 400 km, must have a loss not exceeding 2.5 dB.

Main-to-satellite and main-to-tributary trunks must have a loss not exceeding 2 dB.

Transmission guidelines

Transmission performance

In some cases, the telco may only be able to provide non-VNL trunks which have a loss exceeding these objectives. If the loss is significantly higher than VNL, the switchable pad must be in the “pad-out” mode for connection to these trunks.

Switches can route calls from private network facilities to public network trunks. The public network has a designed loss which does not take into account the added loss of extending the call over private network facilities.

ESN, as any other private network, provides a lower quality performance on these connections than if the call were routed directly to the public network. The amount of degradation must be kept small enough that the connection is acceptable to most users. This loss is restricted as follows:

Off-network long distance connections are only to be established from trunks terminating on nodes. The private network loss added to the public network loss is limited to:

- 4.1 dB for calls originating at node stations
- 6.6 dB for calls originating at main stations.

Off-network local connections can exit at the node and main PBX. For nodes, the loss is as above. For mains, the loss is limited to:

- 6.6 dB for calls originating at node stations
- 9.1 dB for calls originating at main stations.

Supervision

Supervision is a binary signal associated with each direction of transmission on a trunk facility. The two states are on-hook and off-hook. This is analogous to an on-hook (hung up) or off-hook (in use) condition of a telephone.

Transmission guidelines

Each switch connected to a trunk sends a supervision signal to, and receives a supervision signal from the connected switch. Thus, the trunk has four supervision states. They are:

- The trunk is idle when both directions are on-hook.
- Off-hook is sent when a call is initiated on an idle trunk. This action is called “seizing” the trunk. The distant switch receives the off-hook signal, and prepares to receive digits.
- A momentary off-hook condition returned from the destination switch may occur during call set up, but a steady off-hook is not transmitted until the called telephone answers. This off-hook signal is called “answer supervision”.
- The supervision changes to on-hook when the called telephone hangs up or is disconnected.

When a switch serves as an intermediate (tandem) connection between trunks, it normally sends the supervision signal it receives from each incoming trunk to the connected outgoing trunk. It also monitors for a disconnect signal so the trunks can be returned to an idle state and be ready for new calls.

In some cases, there is no answer supervision signal returned from the destination when the called telephone answers.

Under this condition, an off-hook signal can be returned from an intermediate switch. This off-hook signal is called “substitute answer supervision”. This signal is provided to remove transmission impairments associated with the on-hook condition on some trunk facilities and to distinguish a call which has been blocked from one which might have reached a destination party.

A connection by a public network trunk is a typical case where a called telephone answer indication is not returned. Even if provisions can be made to provide an indication of answer, it normally is not. The PBX switch which connects the call to the public network trunk should be configured to provide an off-hook signal to the incoming trunk after sending digits to the public network trunk.

Transmission guidelines

Called Party Disconnect Control

With this feature activated on a trunk route, the Meridian 1 controls the disconnection of calls. When an incoming caller goes on-hook, the call is not released until the Meridian 1 end goes on-hook. This allows calls to be traced in emergency situations.

The routes can be any of the following types:

- Central Office
- Foreign Exchange
- Common Control Switching Arrangement
- Direct Inward Dialing
- TIE
- Wide Area Telephone Service
- modem
- Centralized Automatic Message Accounting

Transmission guidelines

Call Detail Recording

Purpose

With the Call Detail Recording option (CDR) implemented on a Meridian 1 system, you can track users' calls for billing purposes or restriction purposes.

Only information relevant to networking is presented in this section. If you want information on all the types of call records that are available, refer to the International *Software System Management Book 1 of 2*, or the North American *X11 System Management Overview, Applications, and Security* binder.

Setting up

The system generates raw data in Call Detail Records. You can have these printed on a TTY or a tape, or have them sent to a polling device or computer for processing.

You can program your system to give priority to CDR but this is not recommended.

If you give priority to CDR, and your printer fails, the system will not have any call registers for new calls or features at some point. The system uses the call registers to keep CDR records in memory until they can be printed.

Call Detail Recording

The minimum information provided on the call records:

- customer group number
- calling-trunk identification (trunk group number and member number of trunk) or internal-party DN
- terminating-trunk identification or internal DN
- date and time of call
- call duration
- digits dialed

As an option, the Terminal Number (TN) of the originating terminal can be included.

CDR activation involves several steps in programming:

- activate it in the Customer Data Block (LD 15)
- activate CDR for each trunk group for which you want to print call records.

Each trunk group can be programmed independently to show CDR records for:

- all outgoing calls, or
- all outgoing toll calls and/or
- all incoming calls

Types of basic call records

The call records discussed here are the most common ones. They are mentioned throughout this book.

Normal, N-records

These print out as each two-party basic call is completed. The record is identified with the letter N as the first field in the record.

Table 1
N-record

N	001	00	DN4999	T006001	06/28	10:15	00:00:20	98289124 0

The fields are:

1. record type (N in this case)
2. call record number (001)
3. Customer Group number (00)
4. DN of originating telephone (4999)
5. route and member number of the trunk used (Route 6 member 1)
6. date the call was made (June 28)
7. time the call was made (10:15 am)
8. duration of the call (20 seconds)
9. digits field (982891240)

All telephone key-pad input can be included in the record. It also can include the things listed below.

- If the asterisk (*) is stored as a pause-for-dial-tone symbol in a Speed Call number, it appears in the call record.
- If the user presses octothorpe (#) when dialing a call, the digits, up to and including the #, print out. The remaining digits dialed after # do not show up on the CDR record. If you do not prevent this, users who know about dialing the # key can dial calls which the CDR does not track.

Call Detail Recording

For example, a user may dial a trunk access code and then # and then the digits in a toll call. The CDR record shows the trunk access code and the # only. When you receive the bill for the call, there will be no CDR record to match with the bill.

There is a software patch supplied by Nortel Networks to prevent users from dialing # for outgoing calls.

Start, S-records and End, E-records

- When a user activates Call Transfer on an established call, a Start record is generated instead of a Normal record. The record is identified with the letter S as the first field in the record.

  When the transfer is completed, the Start record prints out and shows the two parties involved immediately before the transfer feature was activated. One of the parties can be a trunk.

  When the call is disconnected, an End record is generated showing the final two parties in the call. The record is identified with the letter E as the first field in the record. The End record shows the trunk as the originating terminal and the Directory Number (DN) of the telephone user as the terminating terminal.

  When a call is transferred more than once, start records are not generated for intermediate stations. If you want a print out of the intermediate parties, there is an enhancement available in Release 20 to do this. Refer to the information on CDR Transfer Enhancement later in this section.

- When a user activates the Call Forward All Calls feature and if this results in a call for that telephone, originating from a trunk, going out of the system on a trunk, a consecutive pair of Start records is generated, as well as an End record.

  The first S-record indicates the incoming trunk as the originating terminal and the forwarded DN as the terminating terminal.

  The second S-record indicates the forwarded DN as the originating terminal and the outgoing trunk as the terminating terminal. Both records indicate the same timestamps and duration data. An E-record is generated at the end of the call.

When a user activates the Call Forward All Calls feature at a telephone which results in a call from an incoming TIE trunk going out on an outgoing TIE trunk, two Normal records are generated, one is for the incoming TIE trunk to the telephone and the other is for the telephone to the outgoing TIE trunk.

Table 2
S-record

S	003	00	T000004	DN5064	06/28	10:15		

Table 3
E-record

E	005	00	T000004	DN5055	06/28	10:16		

Interactions with other features

Multi-Tenant software interacts with CDR

With Multi-Tenant software package 86, the telephones are assigned to a Tenant group within the customer group. The tenant numbers of the telephones are included in the call records when users make calls.

ESN software packages interact with CDR

The field of data showing the digits dialed or outpulsed on the CDR record may be preceded by an “A” or an “E.” These letters indicate that route-selection software chose the route for the user. The route selection software can be either Basic Automatic Route Selection (BARS), Network Alternate Route Selection (NARS), Coordinated Dialing Plan (CDP), or Route Selection Automatic Number Identification (RS/ANI).

Table 4
A appears on the record
N-record

N	001	00	DN4999	A00000907.1.02.1	06/28	10:15	00:00:20	A98289124 0

Call Detail Recording

In addition to that, the “E” indicates that Expensive Route Warning Tone was given to the caller and the call was routed on a trunk route programmed as expensive.

Table 5
E appears on the record
N-record

N	001	00	DN4999	A00000907.1.02.1	06/28	10:15	00:00:20	E98289124 0

Refer to the *Control tips* in this section, if you have a route selection software in place, but you still find call records without an “A” or an “E” preceding the digits in the call.

NARS, BARS and BARS/NARS CDR

BARS CDR format is different from NARS CDR format. When BARS and NARS software packages are both present on a system, the CDR prints out in the BARS format.

For the examples below, the user dialed an on-net call (6-655-2315) that stayed on-net. The call was handled by a TIE trunk in a route with access code 457.

Table 6
BARS call record
N-record

N	001	00	DN4999	A00000907.1.02.1	06/28	10:15	00:00:20	A4576552315

Table 7
NARS call record
N-record

N	001	00	DN4999	A00000907.1.02.1	06/28	10:15	00:00:20	A66552315

Note that with BARS, the trunk route access code prints out in the digits field. With NARS, you must look at the trunk route and member number to find out which trunk was used.

Call Detail Recording

CDP and CDR

As of Release 10, you can choose to print out the trunk route access code and the steering code plus the remaining digits that the user dialed in the CDR record. Without this enabled, the trunk route access code for the route used replaces the steering code. The steering code is not printed.

Enabling the option is very useful at remote sites where all calls are sent to a node on a TIE trunk route. You would want to print out the steering code for each call, not just the TIE route access code, so you can identify which location the user dialed.

Authorization Codes and CDR

If you are using Basic or Network Authorization Code software, a separate call record prints out when the user dials an authcode.

The first call record is an A-record. In this example, the user dialed auth code 123456. You will use this code to bill the user for calls made instead of using the DN. The call may have been made from another user’s telephone.

Table 8
A-record

A	003	00	DN3456	T000004	06/28	10:15	123456	

The second call record prints out some time later when the call is finished. You can match the two records together for billing purposes using the DN and the trunk route and member number.

Table 9
BARS call record
N-record

N	019	00	DN3456	T000004	06/28	10:15	00:01:20	A4576552315

Call Detail Recording

ISDN software packages interact with CDR

Calling Line ID in CDR

Table 10
Software requirements

Release required	Software package(s) required
12	118 – Calling Line Identification in CDR (CCDR)

If ISDN is not installed and a user at a remote switch makes a call through a node using one of the node's trunks, the CDR at the node identifies the originator of the call as the TN of the incoming TIE trunk. This makes billing remote users for their calls very difficult.

When ISDN is installed, a CLID is signaled from one switch to another with every call. With CCDR software installed, the user's CLID prints out in the CDR at the node, identifying the user who is calling in on an ISDN trunk and out on one of the trunks at the node.

The format of the CLID changes depending on how calls are dialed. It is best if the CDR collection software or processing you use can be configured for all CLID possibilities for the users.

Improving performance

Timing

Table 11
Software requirements

Release required	Software package(s) required
9.30A	97 – Japan Central Office Trunk (JPN)

Normally, call duration for CDR records is measured in two second increments, but with this package the CDR timing can be configured with half-second increments for greater accuracy.

Call Detail Recording

CDR Expansion

Table 12
Software requirements

Release required	Software package(s) required
9.30A	151 – CDR Expansion (CDRE)

If you have the DN Expansion software package equipped, the DNs at your switch can be longer than four digits and less than, or equal to, seven digits.

If you want complete call records, the CDR Expansion package is required in order to capture the full DN in the call records. If DN Expansion is used without CDR Expansion, only the last four digits of the DNs print in the CDR.

Outpulsed digits

When ESN packages like Basic Automatic Route Selection (BARS) or Network Alternate Route Selection (NARS) software are programmed on a system, the outpulsed-digits option can be very useful. The typical ESN dialing plan has users dialing digits which do not necessarily correspond to what is actually outpulsed on each trunk route.

As of Release 12, the outpulsed digits, rather than the dialed digits, can appear in the CDR records. This helps you to match up your bills with the CDR records.

This option is activated on a per-route basis. The system must be initialized for this to take effect.

The outpulsed-digits option is not available with the ESN package called Coordinated Dialing Plan.

Call Detail Recording

Example:

A typical Private network call using the ESN Dialing Plan on a system equipped with BARS and NARS:

6 + Location code (343) + Directory Number (2214)

Outpulsed digits:

16139672214

Table 13
Outpulsed digits option activated on BARS-type CDR

N	001	00	DN4999	T006001	06/28	10:15	00:10:20	A916139672214

End-to-End Signaling (EES)

End-to-End Signaling is used after a call is established and the user dials more digits. For example, users may access external Interactive Voice Response systems that require them to dial digits to make choices in a queue.

Beginning with Release 14.46E (International software) and then in Release 19 (North America), the printing of these digits is suppressed by default in the Customer Data Block. If you want to see the extra digits in the CDR records, the EES option can be activated.

Be careful about who has access to the CDR printouts, if you print EES digits. Users may be dialing passwords or auth codes for external systems that they do not want other people to know about.

Toll Calls Only Option (OTL)

Previous to Release 8, the CDR could only recognize toll calls if the users actually dialed the digit “1” or “0” as the first or second digit following a trunk access code. If users did not dial a “0” or “1” to place a toll call, in order to have records of the toll calls, *all calls* had to be printed on the CDR records. This was a problem on systems where the dialing plan did not include 1+ dialing or where toll calls were not dialed with the digits 1 or 0. A lot of paper was wasted, or processing time and expense was involved, with extracting the records of the toll calls from the printouts of all calls.

Call Detail Recording

As of Release 5.31 and Release 8 software (not Release 7), selection of the Toll Calls Only option on a trunk route is sufficient to print toll calls only, even if the digits “1” or “0” are not dialed by the user. If these digits are inserted by digit manipulation tables and are outpulsed on the trunk, then these calls appear as toll calls on the CDR output.

Flexible Definition of Toll

In Release 13, Flexible Definition of Toll provided an option on a trunk-route basis, to define single digits following the trunk access code which indicate a toll call for CDR purposes. However, the digits defined are not used for restriction purposes.

Answer Supervision

If Answer Supervision is allowed in the programming of TIE trunks, CDR measures call duration for calls placed over the TIE trunks from the moment a call is answered.

For outgoing calls on other kinds or trunks, all calls seizing a trunk in the trunk group are recorded from the time a trunk is seized. If you want records for answered calls only, you can enable this option.

On North American based ground-start and loop-start and loop-start XFCOT-type trunks, CDR Answer Supervision detects an answer condition when the polarity on the trunk is reversed by the Central Office (CO).

The Answer Supervision option can be enabled in the Route Data Block (LD 16) for each trunk group. When enabled, and for an answered call with supervision, the record shows an “A” in the terminating ID field for the trunk.

With this enabled, the timing for a CDR record does not start when a trunk is seized, but only after the call is answered.

Call Detail Recording

Before the Answer Supervision option was introduced, the terminating ID field was always preceded by the letter “T”. After Answer supervision was introduced, if the Answer Supervision option is enabled for a trunk route, but no supervision is returned on a call, the terminating ID field is still preceded by a T.

- Release 14 - loop-start Answer Supervision was introduced.
- Release 18 - with double-density or quad-density trunk cards, ground-start Answer Supervision can be detected.
- Release 19 - loop-start Answer Supervision capability for trunks connected to Intelligent Peripheral Equipment trunk cards was introduced.

The CO must provide Answer Supervision for this feature to work and the trunk group must be programmed for Answer Supervision.

Format CDR

Table 14
Software requirements

Release required	Software package(s) required
18.20H	234 – New Format CDR (FCDR)

On systems without this software package, or if this feature is not enabled, the fields of data in CDR printouts are output in variable locations, depending on which software packages are equipped. This makes CDR processing difficult, especially if the call records in a network are in different formats at each site.

This software package allows you to have the individual fields of data in CDR records print in fixed locations in every call record, no matter which optional features affect each call.

CDR Transfer Enhancement

Table 15
Software requirements

Release required	Software package(s) required
20	259 – Enhanced Call Detail Recording (CDRX) 234 – Format CDR (FCDR)

On systems without this software, there is only an S-record for the initial phase of the call and an E-record showing the final two parties in the call. Intermediate parties are not shown in the CDR records.

With this software, if a call is transferred, an X-record is printed which identifies the new DN involved with the call. If there are multiple transfers for one call, many X-records print out in sequence as the call is transferred.

Flexible CDR Digit Suppression

This capability was introduced in non-North American markets. It allows you to suppress (choose not to print) a specific number of dialed digits in the digits field of the CDR record.

On a route by route basis, you program how many of the first digits dialed are to be printed. You can also program how many of the last digits dialed are not to be printed.

If the user dialed digits after the call was established and End-to-End Signaling (EES) software outpulsed these digits for the user, they are included in the digit count. If the customer group has the CDR End-to-End Signaling option chosen and the route has the Outpulsed digits option chosen, the Digit Suppression affects only the initially outpulsed digits, not the EES digits.

Periodic Pulse Metering (PPM)

PPM records the number of accumulated pulses on a call. Using that number, the charge for the call can be calculated by multiplying the number of pulses times the unit charge per pulse.

Call Detail Recording

If both the CDR and Message Registration features are equipped, the PPM pulse counts are recorded on the CDR printouts for metered calls. If the charge information is allowed, the charge is calculated and recorded also.

Call Charge Keeping

If the Meridian 1 with Periodic Pulse Metering and ISDN software is connected to a 1TR6 Central Office, call charge information is received from the network as part of the connect data. This information is temporarily stored by the Meridian 1. Further charge information from the network is added to the information being stored. When the call is completed, the information is used to add call charges to the user’s meter. The accumulated call charging information for each call can be printed in CDR records. A Meridian 1 proprietary telephone user can display the charge information using a Message Registration (MRK) key.

CDR 100 Hour Call

Table 16
Software requirements

Release required	Software package(s) required
22	234 – New Format CDR (FCDR)

A field appears on the third line of Fixed Format CDR records to indicate when a call has a duration of 100 hours or longer. This three digit field indicates call duration in hundreds and thousands of hours as follows:

A call lasting more than 100 hours but less than 200 hours is represented by a field showing 001. A call lasting more than 1800 hours but less than 1900 hours is represented by a field showing 018.

Numbering Plan Identification (NPI) and Type of Number (TON) in CDR tickets

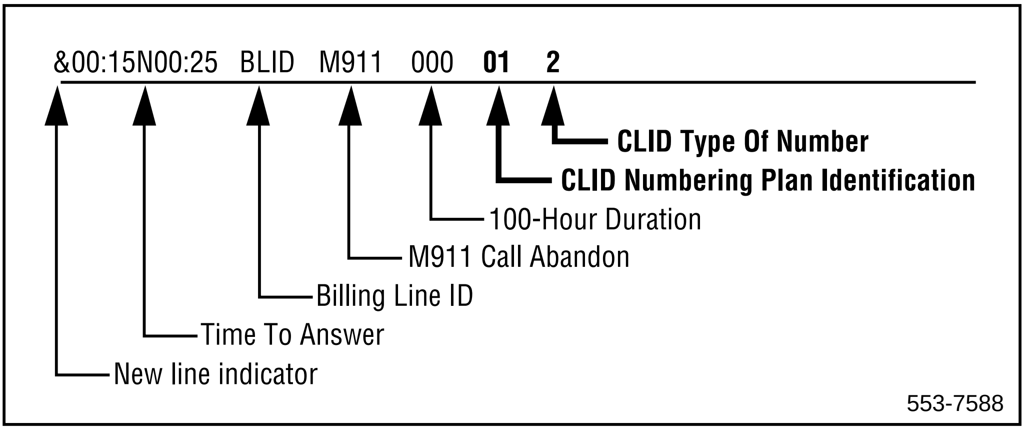
For incoming calls on ISDN trunks, as an option, you can configure new fields to print out in CDR records on line three. The information provided is related to the CLID of the call originator. It is useful if you want to bill calls to call originators. NPI information occupies fields 44 and 45. TON information occupies field 47.

Table 17
Software requirements

Release required	Software package(s) required
23	234 – New Format CDR (FCDR) 118 – Calling Line Identification in CDR (CCDR)

The illustration that follows shows you where the NPI and TON information appears in the call record.

Figure 5
Third line format for CDR record



Call Detail Recording

When an incoming call arrives on the Meridian 1, NPI and TON are sent in the calling party Information Element (IE) and are mapped into internal values. The correspondence between the Meridian 1 values and the values given in the specifications are described in Tables 18 to 27 on the following pages.

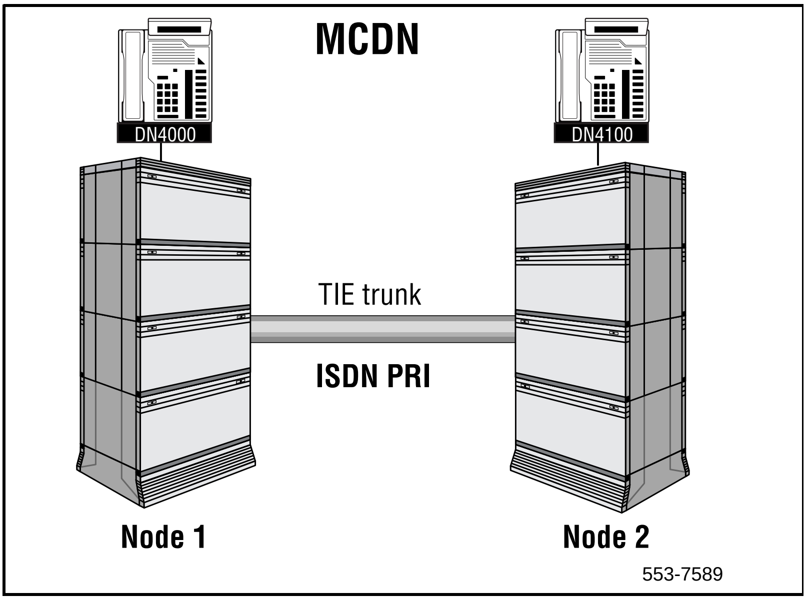
Tables 18 to 27 show the information printed in the CDR ticket, depending on the incoming trunk protocol. As shown in these tables, not all combinations of NPI and TON exist.

In the TON tables, only ISDN/Telephony numbering plan (Rec. E.164/E.163) and private numbering plans are detailed. For all other supported NPI values, TON has the value of "unknown number".

Figure 6 shows you a scenario in which DN 4000 (on Node 1) places a call to DN 4100 (on Node 2) over a Meridian Customer Defined Network (MCDN), using a Coordinated Dialing Plan (CDP) Distant Steering Code (DSC). The call arrives at Node 2 on Route 201 Member 4. A CDR "N" record is produced when the call is disconnected.

Call Detail Recording

Figure 6
Scenario involving a call over
a Meridian Customer Defined Network



The CDR "N" record produced would look like this:

```
N 001 02 T201004 DN4100 02/10 18:07:15:00:02:24.0
 & 4000XXXXXXXXXXXX
 &00:15N00:25 BLIDXXXXXXXXXXXX         000 09 6
```

On line 3 of the CDR record shown above, the NPI value of "09" represents a private numbering plan. The TON value of "6" represents an Electronic Switched Network (ESN) Customer Dialing Plan (CDP). Refer to Tables 18 and 19 for the NPI and TON information for an MCDN incoming trunk.

Call Detail Recording

Table 18
NPI information printed in the CDR record for an MCDN incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	000 - unknown numbering plan
01	0001 - ISDN/Telephony numbering plan (Rec. E.164)
02	not used
03	not used
04	not used
08	not used
09	1001 - private numbering plan

Table 19
TON information printed in the CDR record for an MCDN incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	0000 - unknown number	0000 - unknown number
1	0001 - international number	not used
2	0010 - national number	not used
3	not used	0011 - ESN SPN
4	0100 - local number	not used
5	not used	0101 - ESN LOC
6	not used	0110 - ESN CDP

Table 20
NPI information printed in the CDR record for a EuroISDN incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	0000 - unknown
01	0001 - ISDN/Telephony numbering plan (Rec.E.164/E.163)
02	not used
03	0011 - data numbering plan (Rec.X.121)
04	0100 - telex numbering plan (Rec.F.69)
08	1000 - national standard numbering plan
09	1001 - private numbering plan

Table 21
TON information printed in the CDR record for a EuroISDN incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	000 - unknown or 110 - abbreviated number	000 - unknown or 110 - abbreviated number or 001 - level 2 regional number
1	001 - international number	cannot be mapped
2	010 - national number	010 - level 1 regional number
3	011 - network specific number	011 - network specific number
4	100 - subscriber number	100 - subscriber number
5	not used	not used
6	cannot be mapped	cannot be mapped

Call Detail Recording

Table 22
NPI information printed in the CDR ticket for a QSIG incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	0000 - unknown
01	0001 - ISDN/Telephony numbering plan (Rec.E.164/E.163)
02	not used
03	0011 - data numbering plan (Rec.X.121)
04	0100 - telex numbering plan (Rec. F.69)
08	1000 - national standard numbering plan
09	1001 - private numbering plan

Note: QSIG refers to ISO QSIG and ETSI QSIG.

Table 23
TON information printed in the CDR ticket for a QSIG incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	000 - unknown or 110 - abbreviated number	000 - unknown or 110 - abbreviated number or 001 - level 2 regional number or 101 - level3 regional number
1	001 - international number	cannot be mapped
2	010 - national number	010 - level 1 regional number
3	011 - network specific number	011 - PTN specific number
4	100 - subscriber number	100 - local number
5	not used	cannot be mapped
6	cannot be mapped	cannot be mapped

Note: QSIG refers to ISO QSIG and ETSI QSIG.

Table 24
NPI information printed in the CDR ticket for a non-UIPE and non-MCDN incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	0000 - unknown numbering plan
01	0001 - Rec. E.164
02	0010 - Rec. E.163
03	0011 - Rec. X.121
04	0100 - Telex numbering plan
08	1000 - national numbering plan
09	1001 - private numbering plan

Note: Non-UIPE refers to the 1TR6, AXE-10 for Australia and Sweden, Swissnet 2, Numeris VN4, SYS-12, and D70 connectivities.

Table 25
TON information printed in the CDR ticket for a non-UIPE and non-MCDN incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	0000 - unknown number[1]	0000 - unknown number[1]
1	0001 - international number[2]	not used
2	0010 - national number[2]	not used
3	not used	0011 - network specific number[2]
4	0100 - subscriber number[2]	not used
5	not used	not used
6	not used	0110 - abbreviated number[2]

1. For SYS-12, AXE-10 for Australia and Sweden, Swissnet, Numeris VN4, and D70 interfaces, all received values are mapped into unknown code.

2. For all interfaces not mentioned in 1.

Call Detail Recording

Table 26
NPI information printed in the CDR ticket for an NI-2 incoming trunk

NPI code in CDR	Corresponding value of NPI in specification
00	0000 - unknown numbering plan
01	0001 - ISDN/Telephony numbering plan (Rec. E.164)
02	unused
03	unused
04	unused
08	unused
09	1001 - private numbering plan

Table 27
TON information printed in the CDR ticket for an NI-2 incoming trunk

TON code in CDR	Corresponding value of TON in specification	
	NPI = ISDN/Telephony numbering plan (Rec. E.164)	NPI = private numbering plan
0	not used	not used
1	001 - international number	not used
2	010 - national number	not used
3	not used	not used
4	100 - local number	100 - subscriber number
5	not used	not used
6	not used	not used

Bearer Capability in CDR (BCAP)

In X11 Release 24, you can find out the type of trunk (bearer channel) used for an incoming call by printing a new two-digit field in the CDR. This feature applies to calls received from ISDN or DTI and DTI2 trunks. The use made of the bearer channel can determine how the call is billed.

Table 28 lists the two-digit codes that appear in the CDR and their meaning.

Table 28
Bearer Capability codes and their meaning

Code	Meaning
01	Circuit mode speech
02	Circuit mode 3.1 kHz audio
03	Circuit mode unrestricted 64 kbps digital information transfer
04	Circuit mode unrestricted 64 kbps digital information transfer rate adapted from 56 kbps
05	Packet mode unrestricted digital information transfer
06	Circuit mode 7 kHz audio or videotelephony
07	Circuit mode restricted 64 kbps digital information transfer
08	Circuit mode video
99	Unknown or non-existent

Note 1: This table is based on ETSI specifications. The Meridian 1 does not support all these values (for example, packet mode and information transfer rates different from 64 kbps are not supported for ISDN protocols)

Note 2: For DTI2 and DTI trunks, the programming of the Route Data Block determines the code output in the CDR: 01 for voice channels, 03 for data channels, 99 for voice and data channels (DTI2 only)

The information appears on line three of the CDR, in fields 49 and 50.

If a call comes in on a trunk that is not supported for this capability, the code “99” is included in the CDR. If the Bearer Capability information presented in an incoming call does not match the values known by the software, 99 is output.

Call Detail Recording

If both the originating and terminating sides of a call are trunks, the Bearer Capability code is taken from the incoming trunk.

If a call involves only telephones, or the BCAP feature is not activated, then the fields are left blank.

Incoming calls on the following types of trunks contain the necessary information to send to the CDR:

- EuroISDN
- APISDN
- Numeris, 1TR6, Swissnet, Sys12, D70, AXE10, (Sweden and Australia)
- DASS2
- DPNSS1
- QSIG
- MCDN
- North American ISDN
- TIE, COT or DID trunks on DTI or DTI2
- CIS DTI2

If Basic Call Service is supported on Primary Rate and Basic Rate on a particular trunk interface, then the Bearer Capability feature is supported for both as well.

Control tips

- Look for Call Records that indicate features like Call Forward All Calls and Call Transfer are being abused to set up unauthorized trunk to trunk connections. This type of activity on your system can result in high telecommunications bills. If you want that activity prevented, implement the Call Forward External Deny feature on the telephones of the offending users. You can stop trunk-to-trunk transfers by removing supervision programming from your trunk groups. Discuss this with your system supplier.

- On systems with ESN software programmed, the absence of the letter A or E preceding the dialed or outpulsed digits field in the CDR means the user dialed a direct trunk access code to place the call, instead of a BARS or NARS access code.

  This indicates users are bypassing BARS or NARS and not taking advantage of the cost savings and features these software packages can provide. If you find this is happening on your system, implement TGAR codes to prevent direct trunk access.

- If you use Direct Inward System Access (DISA) ports, it is imperative that you monitor CDR frequently. Unauthorized callers who use your DISA ports can be caught if you pay attention to the call records that print out. Talk to your system supplier about ways to implement a security routine that includes regular inspection of raw CDR records to prevent security breaches on your system.
- If you are using Authorization Codes on your system, be careful who sees the CDR records. The Authorization codes print out as part of the records.

Administration tips

- Decide how often you wish to check for unusual and unauthorized calls using the CDR printouts. Check for such things as:
  - long-duration calls
  - calls in and out after normal working hours
  - calls from publicly accessible telephones

Call Detail Recording

- calls from meeting rooms or empty office telephones
- personal calls
- incoming trunk calls forwarded out on a trunk
- incoming trunk calls from other network locations calling out on trunks at your location
- calls made with direct trunk-access codes, if BARS or NARS is supposed to route calls
- S- and E-records on systems that cannot print X-records for transferred calls. Identify these and, where appropriate, bill the originator of the call, instead of the final DN to which the call was transferred.

◆ It is often true that if users and managers do not receive the bills and an itemized accounting of calls they made, they have very little interest in reducing the expenses associated with their calls. For example, if users know how much their calls cost when they go out on expensive routes compared to less expensive choices, they are usually more willing to help reduce expenses.

◆ It is wise to provide a secondary device for CDR printing, in case your primary CDR device experiences a problem.

Training tips

◆ Include information in training sessions for your system users regarding the monitoring you will be doing using CDR. If users know that calls are being monitored, your telecommunications expenses will stay close to the minimum.

Network Traffic studies

Purpose

It is recommended that you install the Network Traffic software package on each system where you are installing any other ESN software package(s). Talk to your system supplier about it. They may install the software on systems, automatically.

You can use traffic study data to monitor a number of things after your ESN software has been installed. The data provides you with information about:

- the amount of call traffic on each choice in each route list
- the number of calls going out on expensive routes in each route list
- queuing activity (Off-Hook Queuing and Callback Queuing) and the length of time users queue, on average
- users' behaviour - whether they use ESN features or if they require more training
- the calls being made by each NCOS group
- the number of calls that were blocked - by route list and by NCOS group
- the queuing that is being offered by the node to remote switches

Network Traffic studies

Setting up

The overlay program for scheduling and selecting the study options is overlay program 2 (LD 2). Refer to the International *Software System Management Guide* for more information on overlay program 2. In North America, refer to the *X11 System Management Overview, Applications, and Security* binder.

In order to receive traffic study data you must ensure the following things are done:

- the studies must be scheduled. Network Traffic data is associated with a Customer group. Therefore, you will need to program a schedule for Customer Traffic in order to get Network Traffic data.
- the particular study options that you want to run must be selected. You will need to select one or more Network Traffic options. There are three:
  - TFN001 – Routing
  - TFN001 – Network Class of Service
  - TFN003 – Incoming Trunk Group (Queuing)
- the system must be programmed with instructions as to where to print out the traffic study data. Your system supplier can configure a Serial Data Interface (SDI) port on your system to output the traffic study data to a printer or PC set up for this purpose. This is done in the Configuration Record, LD 17 (which is beyond the scope of this book).

It is useful if you run traffic studies on your trunk routes during the same time you are running Network Traffic Studies since you may want to use that data to analyze your trunk routes. You must select Customer option number 2 (TFC002) to get the trunk route data. This study is a part of the normal traffic routines, that are provided with every system. There is information provided in this section about the data that study provides.

Network Traffic studies

Procedures

- Check your maintenance agreement with your system supplier before you attempt to set up a traffic study.
- If your system supplier agrees that you may run studies, they can train you to schedule the studies properly and choose the appropriate study options.
- *Print the existing traffic study schedules and the options which are already selected before you make changes. If you do not do this, you might accidently change a schedule that someone else has set up. This can affect a study already in progress or one planned for the near future.*

  Tell other people who set up studies to print any existing traffic study schedules before they set a new schedule.
- *Notify other people who are involved with your system when you are running a study. The technician needs to know, for example, so that when study data prints out, it will not be discarded accidentally.*
- Print the schedules and options after you have finished inputting to verify that you entered the settings correctly.
- Check the printer during the first scheduled output time to be sure data prints out with no problems.
- Check your printer often during the study period to ensure that you are getting all the data you should be getting and that the printer is in good working order.

Network Traffic studies

Printout formats

The beginning of a study is labelled with the header message **TFS000** followed by the date and time of the printout.

The end of the study is labelled with a footer message **TFS999**.

Be careful when tearing off the printer paper. Make sure you see both the header message and the footer message. If you miss these, you may also miss the *important warning messages and threshold violations which print at the beginning of the study* or you will miss parts of the last study option printout.

Some of these warnings might be telling you to ignore the data for various reasons. For example, if the system initialized during the previous study period, the traffic registers were cleared out. There is a warning message for that situation. If this occurred, there is no point in using the data since it is incomplete.

Invoking data

If you check the printer and you find that a problem of some kind prevented the data from printing out, you can still retrieve the data.

However, you can only retrieve the data from the most recent scheduled study period. Retrieving this data is called *invoking* the data.

The data from the most recent study period is held in memory while the data from the next study period is being collected. When the system is scheduled to print the new data, the old data is removed from memory and replaced with the new data. If you do not invoke and print the old data quickly enough, it is replaced with new data and no longer available to you.

You must retrieve old data before the next printout is scheduled or it will be erased.

Ask your system supplier to train you on this procedure. You can read about the commands for this in the International *Software System Management Guide.* In North America, refer to the *X11 System Management Overview, Applications, and Security* binder.

Using the data

Grade of service objectives

You must decide what service level (grade of service) objectives you have for your system before you analyze any traffic study data.

System suppliers provision systems to meet the grade of service objectives as shown in the following table.

You can use more stringent objectives than these, if you wish. If you do so, you might need additional equipment to meet your objectives.

You can use less stringent objectives, if you wish, but you will sacrifice service. For example, poor service can result in a blocked incoming call, delayed dial tone for your users, or a feature which did not operate when needed. Assess the impact of poorer service on your business before you choose reduced service levels.

Table 29
Nortel Networks grade of service guidelines

Type of service	Maximum blockage objective (%)
incoming calls	1
outgoing calls	1
intracustomer calls	4
tandem (trunk to trunk) calls	1
less than 3 second wait for dial tone	1.5

The guidelines are objectives against which you can measure your system performance.

Network Traffic studies

You and your system supplier can use the data from the study period to evaluate your system performance against the grade of service objectives.

Discuss with your system supplier what projected traffic load was used when they originally configured your system. If your system was configured based on your projected busy hour traffic load, the traffic study data should be analyzed based on your busy hour statistics.

If you find your actual busy hour traffic is consistently different from what was projected, you may have to reprovision your system in order to meet your grade of service objectives.

When to run a study

Running these studies after installation can help you find programming errors that your testing did not find. It can help you monitor the way the software is dealing with every call being made. You can use the data, along with trunk route traffic data, to provision your trunks properly. You can also find out about user-related problems that you may need to fix quickly to prevent long-term problems.

You can also run these studies at regular intervals during the year to ensure that your programming keeps pace with the inevitable changes that will happen to your network.

- Determine what weeks of the year are slow times for your organization and what weeks are the busiest (Busy Season).

  During the busiest times, your system is handling its greatest call volumes.

  If you do not understand the traffic patterns on your system well enough to determine your Busy Season, ask people who might know. Some people you might ask are the attendant(s), executives, sales people, and secretarial staff.

Network Traffic studies

- Decide what call volume your system is expected to handle and still meet the grade of service objectives. Choose one of the following types of *Busy Hours*:
  - the busiest hour during the busiest week (also called the Peak Busy Season Busy Hour)
  - the average of your five busiest hours, one busiest hour from each day during a busy week (also called the Busy Season Average Busy Hour)
  - the busiest hour during an average busy week (also called an Average Season Busy Hour)

  If you provision your system to handle the traffic load during your absolute busiest times, excellent service for incoming and outgoing calls is guaranteed. Internal users and external callers will not encounter blocking, even during peak traffic periods.

  If you provision based on a study which is run during an *average* busy time, there might be peak busy times when the your grade of service objectives will not be met. Evaluate the potential costs and other negative results of blocked calls before you decide to do this.

- Give your system supplier sufficient time and information to set up a traffic study and analyze the data, if they are conducting the traffic study for you. They need to know:
  - if you have a deadline you are trying to meet
  - if you are preparing a budget for possible equipment purchases based on the study results
  - if you are expecting future increases in call traffic

  This information affects the recommendations your supplier will make to you.

Network Traffic studies

- Decide how often you want to run a study.

  It is a good policy to run a minimum of one study annually.

  If your organization is changing rapidly and this is impacting your calling patterns, your system should be monitored more frequently. Your system can be configured in advance to handle predictable changes to your volume of calls.

  If your system supplier is running the studies for you, there may be a charge associated with more than one annual study.

  If you are considering doing your own traffic study analysis, evaluate the time and expense involved in doing the study against the benefits you will achieve from doing your own.

  You will need training on how to analyze the data if you plan to do your own studies.

- Discuss setting up traffic threshold levels with your system supplier. Instead of running complete studies, the system can be programmed to print out messages only at times when these traffic-related thresholds are violated. Along with the threshold violation message, it prints out enough traffic-related data to help you analyze the source of the problem. You and your supplier can coordinate a procedure for using this method to monitor the traffic on your system.

There is more information on these threshold settings and traffic studies in general in the International *Software System Management Guide*. In North America, refer to the *X11 System Management Overview, Applications, and Security* binder.

Network Traffic studies

Terms you should know

Peg count

Many of the traffic study options are designed to keep a tally of how often certain events occur. *Peg count* is another word for tally.

CCS (Centa Call Seconds)

One CCS is equal to 100 seconds.

- If the duration of a conversation was 230 seconds, you can say it lasted for 2.3 ccs.
- One minute is equal to 60 seconds or.6 ccs.
- One hour is equal to 3600 seconds or 36 ccs.

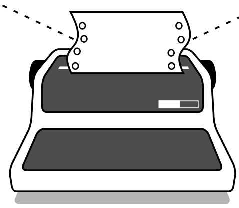
Network Traffic studies

TFC002 – Trunks

Sample data

System ID	**TFC002**
Customer number	
Route number	**Trunk type**
Trunks equipped	**Trunks working**
Incoming usage	**Incoming peg count**
Outgoing usage	**Outgoing peg count**
Outgoing overflow	**All trunks busy**
Toll peg count	
Incoming ISA peg count	**Outgoing ISA peg count**
200	TFC002
007	
004	COT
00008	00007
0000088	00046
0000114	00052
00001	00002
00006	
00000	00000

The headings shown in this example do not appear in the printout.



533-0300T TTY

This study prints out the usage data in units of CCS.

Network Traffic studies

Purposes of TFC002 study

The data in study option TFC002 is mainly used for provisioning the correct number of trunks in each trunk group. Based on the usage you actually have on each group of trunks during your busy hour(s), you and your supplier can use trunk provisioning tables or computerized tools to calculate how many trunks should be in each trunk group to provide the level of service you are expecting.

Grade of service

You must decide what grade of service, or in other words what maximum level of blockage, you can tolerate. Each trunk group can be configured individually for a separate grade of service.

For example, you might want to provision public network Central Office trunks at 2% blockage since your customers use them to call in to you. You might provision your private network TIE trunks with 5% blockage as a maximum since these trunks might have a higher monthly cost than Central Office trunks. Also, since the TIE trunks handle calls only from your own private network users, you can train them to use the Ring Again feature to queue for the trunks when these are busy, or they can try the call at a later time after the blockage has cleared.

The higher the acceptable blockage, the fewer trunks you need for the given amount of traffic.

You must assess what impact an *all trunks busy* condition might have on the type of caller who uses the trunks, and the resulting impact on your business before you choose your grade of service objective.

Provisioning tables

You and your system supplier must discuss the kind of provisioning tables to use. Three of the most common ones are called:

- Poisson
- Erlang B
- Erlang C

Network Traffic studies

Poisson and Erlang B statistical tables provision almost the same number of trunks when there are low levels of traffic on a trunk group. However, as the traffic levels increase, the Poisson tables provision more trunks than the Erlang B tables.

Use the Poisson table if:

- you want to provision a buffer for periods of peak traffic
- the trunk group you are provisioning is a last choice trunk group on a system which uses automatic route selection
- you want to have enough trunks for all calls that attempt to seize a trunk (even the calls that never get established and occupy trunks for short periods of time)

Use the Erlang B tables if you are provisioning first choice trunk groups in route lists. They provision exactly enough trunks for the grade of service you requested. You can expect that overflowed calls during short-term peaks in traffic will go to the last choice trunk group, if the first choices are busy.

Use Erlang C tables only if you expect your users to queue during busy times when all trunks in that group are busy. *Do not provision using these tables if your users will not queue or if your business cannot tolerate queuing.* These tables provision low numbers of trunks since these tables assume that queuing will occur.

Other information

Other fields of data in this study show you the following additional information:

- the number of trunks equipped and the number of trunks working

  This data is one way to monitor each trunk group to ensure there are no disabled trunks. If there are any, be sure to enable them and run a new study before you assess the traffic data.

  Your system maintainer is probably running maintenance diagnostics on your trunks periodically in order to keep your trunks in good working order.

Network Traffic studies

There are also maintenance messages which print out on maintenance printers, when there are trunk problems.

Your attendant can also check each trunk in each trunk group on a regular basis from the console. Instructions on how to do that are in the Console User Guide.

- how many times during the study period there were no available trunks in that trunk group and a call intended for that group of trunks was blocked or sent to a second trunk choice, if one exists. These are referred to as overflowed calls.

  Overflows are not necessarily bad, especially if the overflowed calls go out on a second choice trunk group and the cost for these overflowed calls is lower than the cost of installing additional trunks in the trunk group which overflowed them.

- how many times during the study period the last available trunk in that trunk group was used by a call

  A high number is not necessarily bad unless it is accompanied by a high number of overflows as well. Then the same argument stated in the previous item applies.

  There is a row (or optionally two rows) of keys on the attendant console for Trunk Group Busy indicators. Your attendant can monitor trunk groups by noticing how often these key lamps flash. A flashing lamp means all the trunks in that group are busy.

  Ask the attendant to inform you whenever certain trunk groups appear to be busy frequently, and to note the times of day when this is happening.

- the number of calls which were dialed with a 0 or 1 following the trunk group access code

  This pegs only for Central Office and Foreign Exchange trunk groups. If users are supposed to be restricted, you might use this as a quick way of checking if the necessary restrictions are in place. Your Call Detail Records will give you more detailed information on what calls are being made and what telephones are being used to make them.

Network Traffic studies

- the last two fields of data apply to ISDN PRI trunks. In North America, with Integrated Services Access (ISA), you configure service routes on a PRI master route. These two fields of this study will show an incoming or outgoing peg count when a channel becomes idle after being used for a call.

NI-2 CBC Service

With Release 23 and the introduction of NI-2 Call By Call Service Selection, the fields have been modified slightly.

Trunk Type

For an NI-2 CBC master route, 'CBCT' is output in this field. For an NI-2 CBC service route, one of the following is output in the field, based on the trunk type of the route:

- FEX
- TIE
- CO
- DID
- WATS

Trunks Equipped

This field indicates the total number of trunks configured for the NI-2 CBC master route. This field would always be 0 for service routes.

Trunks Working

This field indicates the total number of trunks enabled (working) for the NI-2 CBC master rout. This field would always be 0 for service routes.

Incoming usage

This field indicates the total usage of incoming routes, for both the NI-2 CBC master route and service routes.

Outgoing usage

This field indicates the total usage of outgoing routes, for both the NI-2 CBC master route and service routes.

Outgoing overflow

This field indicates the number of outgoing overflowed calls for the NI-2 CBC master route. This field would always be 0 for service routes.

Toll peg count

This field indicates the toll call peg count for the NI-2 CBC master route. This field would always be 0 for service routes.

Incoming peg count

This field indicates the total number of incoming calls on the NI-2 CBC master route. This field would always be 0 for service routes.

Outgoing peg count

This field indicates the total number of outgoing calls on the NI-2 CBC master route. This field would always be 0 for service routes.

All Trunk Busy

This field indicates the increment when the last trunk on the NI-2 CBC master route becomes busy. For a service route, this field increments when the number of calls (both incoming and outgoing) for the service route has reached its maximum value.

Incoming CBC peg count

This field indicates the total number of incoming calls on an NI-2 CBC service route.

Outgoing CBC peg count

This field indicates the total number of outgoing calls on an NI-2 CBC service route.

Threshold TFC104

There is an All Trunks Busy threshold which you can program to automatically monitor the trunk groups on your system. The threshold violation message indicates that the last trunk in an identified group was seized more than the allowed percentage of the time. Whenever a trunk group exceeds the percentage you program, the threshold violation message prints out on the Traffic printer along with TFC002 traffic study data, to help you analyze the situation.

A suggested threshold is 5 percent, initially.

Network Traffic studies

Trunk Traffic Reporting Enhancement (Release 21)

There are two options that are part of the enhancements:

- Traffic Period Option (TPO)
- Trunk Seizure Option (TSO)

Traffic Period Option (TPO)

Normally, when a call is in progress at the time a TFC002 study is scheduled to print out, the duration and peg count for that call will not be included in that printout. The data for that call only prints out at the next scheduled print out time, after the call ends.

When the TPO option is activated in the Configuration Record (LD 17), TFC002 trunk usage data in each printout will include all duration data even though some calls are still in progress. When calls are disconnected, the next scheduled printout after the disconnect shows the duration data of the calls for that reporting period and a peg count for the calls.

Trunk Seizure Option (TSO)

Under normal conditions, trunk usage data begins to accumulate for the TFC002 study option only after a call is considered to be established.

A call is considered to be established when:

- the End-of-Dialing timer expires after the last digit is dialed
- octothorpe (#) is dialed
- answer supervision is received from the other end

Network Traffic studies

Some calls that users make are not answered. Data will still accumulate if this option has been activated on your system. The TSO option allows the data to be accumulated beginning with trunk seizure, and not only after the call is established. You can have this option activated in the Configuration Record.

However, if the time between trunk seizure and call disconnect is too small (less than 4 seconds), the usage and peg count will not be accumulated.

Activating this option can allow you to provision your trunks to include the calls that never get established but occupy trunks for short periods.

Network Traffic studies

TFN001

System ID		TFN001							
Customer number									
RLST	XXX	RLI requests	RLI requests served without delay	expensive route acceptance	RLI requests standard blocking	not defined	not defined		
	RT	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use
		RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use
		RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use
		RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use
		RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use
		RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use
		RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use
		RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use	RLI entry use
		TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls
		TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls
		TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls
		TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls
		TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls
		TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls
		TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls
		TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls	TD calls
	OHQ	OHQ calls	time in OHQ	abandoned calls					
	CBQ	CBQ calls	avg. time in CBQ	CBQ offerings	CBQ user cancellation				
	RVQ	RVQ calls	avg. time in RVQ	RVQ offerings	RVQ user cancellation				

Network Traffic studies

The information in this section will make more sense to you if you are familiar with NARS. Refer to the *NARS and BARS* section for clarification of terms and concepts you do not understand.

TFN001 Routing traffic measurements

The TFN001 routing measurements provide data related to route list utilization. The measurements show how often a route list was accessed, which entries in the list were used, and whether the call was established or blocked.

The routing traffic measurements contain the following statistics for each defined route list:

Route list requests

This measurement identifies the total number of call attempts for which translations identified this route list for call completion. Calls do not have to be completed to be included in this count. You can find out about the call patterns of your users. You will know what destinations the users call frequently or not at all.

Route list requests served without delay

This measurement reflects the total number of network calls that were routed without encountering blocking or queuing. Expensive route acceptances are included.

Expensive route acceptances

This measurement identifies the number of calls that were allowed to complete over an expensive trunk route after the Expensive Route Warning Tone (ERWT) was given. If there is a high number you may need to do some refresher training or bill for expensive calls.

Route list requests standard blocking

This measurement identifies the number of call attempts that could not be served because a route or queuing process was not available to a user. A call may have been blocked due to the low FRL of the user, queuing eligibility tests failed when trunks were busy, or the user timed out in the Off-Hook Queue before a trunk became available. The blocked call may have been routed to overflow tone, a recorded announcement, or the attendant.

Network Traffic studies

The data in this study is very useful when you first install NARS or BARS or CDP software. It helps you identify route lists with incorrect FRL programming or NCOSs with incorrect FRL programming. You may not have known that certain users need to call a certain destination and you have inadvertently blocked them. On the other hand you may have blocked callers who were not supposed to have been dialing those destinations in the first place and this proves you have prevented them. You will be able to see what NCOS group was affected if you look at the TFN002 data.

Route list entry usage count

This measurement identifies the number of calls that were routed successfully over a particular route (entry) in a route list. Calls will be counted if they go out normally or after some form of queuing. You may find that the number of requests for a route list does not match the total of calls handled by the entries since some users hang up before calls go out on a trunk.

If you have Tone Detectors activated, calls going out on Tone Detector routes will show up in the TD calls fields.

Use this data to determine if alternate routing is working properly. If you see usage of routes that are supposed to be turned off or vice versa, you should check your programming. If there is no overflow you may have too many trunks in your first choice route. If one entry is supposed to screen for certain calls only, but all calls go out there, you may have an FCAS or FSNS table problem.

Quantity of calls placed in OHQ

This measurement identifies the number of calls that attempted to use a route in the route list. Because facilities were not immediately available, the call was permitted to remain off hook to wait for facilities. If network-wide queuing is enabled, these queued calls are included. You must decide what percentage of calls you will accept for queuing. When the number becomes higher than that, you should look at your trunks and do some maintenance or additions. You may have limited some users to an overly limited set of route choices.

Network Traffic studies

Average time in OHQ

This measurement identifies the average duration that calls remained in the OHQ until a route became available. The value (expressed in units of 0.1 second) represents the average time in the queue. A value of 500 means 50.0 seconds. Calls that timed out in the queue before a route was selected or calls that abandoned are also included in the average.

If users are queuing for prolonged periods, you may need to do some maintenance or additions. You may want to allow them to access more trunk choices.

Quantity of calls abandoned from OHQ

This measurement identifies the number of calls that were placed in the OHQ but were abandoned. For example, the caller went on hook before a route became available or the time limit was reached. This indicates that users are frustrated. Look at your other data to find out why this is happening.

Quantity of CBQ calls

This measurement identifies the number of calls that were offered CBQ and accepted the offer. If there is a great difference between the amount of queuing offered and that accepted, you may need to do some refresher training. After installation, users sometimes forget what overflow tone means and how to activate queuing.

Average time in CBQ

This measurement identifies the average duration (in units of 0.1 second) calls remained in the CBQ. Calls that were cancelled and calls that were served are included in this measurement.

If users are queuing for prolonged periods, you may need to do some maintenance or additions. You may want to allow access to more trunk choices.

Network Traffic studies

Quantity of CBQ offerings

This measurement is a count of the number of calls that were offered CBQ, regardless of whether or not the offer was accepted. You must decide what percentage of calls you will accept for queuing. When the number is higher than that, you should look at your trunks and do some maintenance or additions. You may have limited some users to an overly limited set of route choices.

Quantity of CBQ user cancellations

This measurement identifies the number of calls that were removed from the CBQ because the user cancelled the Ring Again feature.

Users should cancel Ring Again if they are not going to be able to answer a callback. Callback calls occupy trunks. It is not necessarily bad if you see cancellations. However, it can indicate that queuing is taking too long and that users are frustrated.

Quantity of RVQ calls

This measurement identifies the number of calls that were offered RVQ and accepted the offer.

Average time in RVQ

This measurement identifies the average duration (in units of 0.1 second) calls remained in the RVQ. Calls that were cancelled and calls that were served are included in this measurement.

Quantity of RVQ offerings

This measurement is a count of the number of calls that were offered RVQ callbacks, regardless of whether the user answered the callback.

Quantity of RVQ user cancellations

This measurement identifies the number of calls that were removed from the RVQ because the user cancelled the Ring Again feature.

Network Traffic studies

TFN002

System ID		TFN002					
Customer number							
NCOS	NCOS group	calls attempted	routing requests served without delay	expensive route acceptances	network call standard blocking	not defined	calls refusing expensive routes
	OHQ	OHQ calls	avg. time in OHQ				
	CBQ	CBQ calls	avg. time in CBQ				
	RVQ	RVQ calls	avg. time in RVQ				

Network Traffic studies

TFN002 NCOS measurements

Traffic measurements are collected for each defined NCOS group to indicate the grade of service, in terms of blocking and queuing delay, being provided by the system.If a network administrator deems the level of service to be inappropriate for users in a particular NCOS group, then the network administrator can either reassign the users to another NCOS group, redefine the characteristics of the existing NCOS group, or change the routing parameters.

Quantity of calls attempted

This measurement identifies the total number of call attempts generated by users in an NCOS group.

Routing requests served without delay

This measurement identifies the number of call attempts that were routed without encountering blocking or queuing.

If you see a great difference between the number of call attempts and the requests served without delay, look at the data in TFN001 to find out which route lists indicate a high level of blockage or queuing. You may need to adjust the route list programming or the NCOS programming to reduce the blockage. It may verify that users are being blocked from calls they are not supposed to make.

Expensive route acceptance

This is a count of the number of callers who accepted an expensive route to complete a call. This data helps you focus your training on the proper users if you find too many users are accepting expensive routes.

Network Call Standard Blocking

This measurement identifies the number of call attempts that could not be completed because a route or queuing process was not available to a user. The blocked call may have been routed to overflow tone, a recorded announcement, or the attendant. You may need to adjust the route list programming or the NCOS programming to reduce the blockage. It may verify that users are being blocked from calls they are not supposed to make.

Network Traffic studies

Quantity of calls refusing expensive routes

This measurement identifies the number of calls that were given ERWT and elected not to use the expensive route. As users become more familiar with a system this number usually increases.

Quantity of calls placed in OHQ

This measurement identifies the number of calls that were placed in the OHQ. You must decide what percentage of calls you will accept for queuing. When the number becomes higher, you should look at your trunks and do some maintenance or additions. You may have limited some users to an overly limited set of route choices.

Average time in OHQ

This measurement identifies the average duration that calls remained in the OHQ until a route became available. The value (in units of 0.1 second) represents the average time that calls were in the queue. Calls that timed out in the queue before a route was selected are also included in the average. If users are queuing for prolonged periods, you may need to do some maintenance or additions. You may want to allow them to access more trunk choices.

Quantity of CBQ calls

This measurement identifies the number of calls that were offered CBQ and accepted the offer. When the number becomes higher you should look at your trunks and do some maintenance or additions. You may have limited some users to an overly limited set of route choices.

Average time in CBQ

This measurement identifies the average time that calls waited in the CBQ for a route to become available. It includes calls that requested a cancellation, calls that were served, and direct Ring Again against trunks. The average time is expressed in units of 0.1 second. If users are queuing for prolonged periods, you may need to do some maintenance or additions. You may want to allow them to access more trunk choices.

Network Traffic studies

TFN003

System ID		TFN003				
Customer number						
TRKG	Incoming trunk group					
	OHQ	calls placed in OHQ	avg. time in OHQ			
	CBQ	incoming calls offered CBQ, CCBQ, CBQCM	calls accepting CBQ, CCBQ, CBQCM	avg. time in CBQ, CCBQ, CBQCM	blocked CBQ, CCBQ, CBQCM callbacks	callback attempts not answered or cancelled
	RVQ	incoming calls offered RVQ, RVQCM	calls accepting RVQ, RVQCM	avg. time in RVQ, RVQCM	blocked RVQ, RVQCM callbacks	callback attempts not answered or cancelled

TFN003 Incoming trunk group measurements

TFN003 data provide an indication of the incremental traffic that was imposed on incoming trunk groups by the network queuing features. Data are provided for each incoming or two-way trunk group that is offered OHQ, CCBQ, or CBQCM.

The information in this section will make more sense to you if you are familiar with Queuing. Refer to the *Queuing* section for clarification of terms and concepts you do not understand.

The following measurements are accumulated for each incoming (or two-way) trunk group:

Quantity of calls placed in OHQ

This measurement identifies the number of incoming trunk calls that were placed in the OHQ for possible connection to another trunk group at the node. When OHQ is offered to a user at a remote site, the TIE trunk is held up during the queueing process. If you multiply the number of queued calls times the average time in queue you can calculate how much queuing time is occupying your incoming trunks. You can calculate whether it would save you money to install additional trunks at the node or allow the user to have access to more trunk choices at the node instead of queuing on the TIE trunks. You may be able to remove some TIE trunks if they are no longer needed for queuing.

Average time in OHQ

This measurement reflects the average time (in units of 0.1 second) that calls waited in the OHQ for a trunk to become available. The average time includes those calls that were removed from the OHQ by caller abandonment or were removed from the queue after expiration of the OHQ time limit.

Quantity of incoming calls offered CCBQ or CBQCM

This measurement identifies the number of incoming trunk calls that were blocked at the ESN node and for which the user was given the option of accepting an ESN node-initiated call back when facilities would become available. The measurement relates to use of the CBQ feature by users at an ESN main (Coordinated Call-Back Queuing) or Conventional main (Call-Back Queuing to Conventional Mains).

Network Traffic studies

Quantity of calls accepting CCBQ or CBQCM

This measurement identifies the number of incoming trunk calls that were blocked at the ESN node, were offered CBQ, and accepted the offer. The count relates to CBQ acceptances by users at an ESN main or Conventional main. If the offers greatly exceed the acceptances you may need to train users or make some changes to reduce the amount of queuing offered.

Average time in CBQ

This measurement (expressed in units of 0.1 second) reflects the average time that users at an ESN main or Conventional main remained in the CBQ (at the ESN node) for a facility to become available.

> ***Note:*** When a CCBQ call back is offered to a busy station at an ESN main, the call is removed from the queue for 5 minutes, then reinserted in the queue. This process occurs only once.
> The additional queuing time is added to the average time.
> The 5-minute suspension time is not included in the average time, nor is its reinsertion into the queue pegged as another CBQ call.

> ***Note:*** When a CBQCM call back is offered to a station at a Conventional main that is busy or fails to answer the call back, the call is removed from the queue and reinserted into the queue as specified in Note 1.

Quantity of calls blocked in call back

This measurement identifies the number of CBQ call backs (CCBQ or CBQCM) initiated by the ESN node that could not be completed because an outgoing trunk group (to the ESN main or Conventional main) was not available. If this happens frequently, you should look at adding more trunks. If callbacks take too long, users will become frustrated.

Callback Attempts No Answer and cancellation

This measurement identifies the number of callback attempts that were not successful because the caller failed to answer the callback. CBQ callbacks to a station at an ESN main that has previously cancelled CBQ are treated as Call-back Attempts No Answer.

Network Traffic studies

If users are not answering callbacks, they waste trunks at both the node and the remote switch. Train users to answer callbacks or not to activate queuing.

TFN101

OHQ threshold violation measurement

The OHQ overflow threshold measurement (TFN101) provides an indication that more than the desired percentage of users are timing out in the OHQ. This means that OHQ is offered and accepted, but an inexpensive trunk does not become available before the service-changeable OHQ time limit expires, more than the preset threshold percentage of occurrences. This could result from trunks being out of service, an OHQ timer set too low, overly limited trunk choices for users or temporary traffic overload.

Control tips

- If you tell users you are running a traffic study, they might alter their habits when using the telephone. This is especially true of attendants who may think you are doing an analysis of them for job performance purposes. If you want to capture the normal activity levels, do not tell users about the study.
- Tell the system maintainer that you are running a study so they can avoid doing work and maintenance routines which have an impact on the data. For example, doing a manual initialization clears out the traffic data in memory; avoid doing this while a study is running.
- If cards are moved to different loops or new cards are installed ask the system maintainer to let you know the date and time of this work, so you can include that information in your analysis.
- Ask the system maintainer to keep track of warning messages which might print out concerning, for example, the loops, Superloops, timeslots, and trunks. If there are problems which the system identifies, these warnings should be included in the traffic analysis as well.

Network Traffic studies

Administration tips

- If the TTY device which your system maintainer uses to program and maintain your system is also configured to receive traffic study data, your system maintainer may find this annoying.

  When the traffic study data is printing, it interrupts the programmer until all the data from the study has printed out. Once it has printed, the programmer can resume where he or she left off, but it may take some time for the data to print.

  Configure a separate printer for Traffic studies, if you can.

- Note the speed at which the study prints out. If the traffic study data prints and then stops and then prints again, and this continues, it is an indication that your system CPU is working too hard at that time.

  Traffic study printing is a low priority task for the CPU. If there are many other tasks to do, the study printouts slow down. If you are running TFS004, pay attention to the CPU real-time analysis, to find out if your CPU is overloaded.

Training tips

- The data from study option TFC005 can have a major impact on your training programs. Once you learn about the patterns of feature use and non-use on your system, you can use the data to focus your training efforts effectively.
- The data in study options TFC003 and TFC004 can have a major effect on the training you do with the attendants. Use the data in conjunction with your observations about their performance and the goals of your organization for efficient call answering.

NARS and BARS

There are two cases where automatic route selection is desirable at a switch. One is for a Meridian 1 that has multiple public network route choices, such as FEX, COT (or WATS, in the U.S.) but no TIE trunks to other private network systems. The other case is for a Meridian 1 that has one or more TIE trunk routes connecting it to other private network switches as well as one or more public network route choice(s). The two types of automatic route selection software called Basic Automatic Route Selection (BARS) and Network Alternate Route Selection (NARS) were developed with these two cases in mind.

NARS

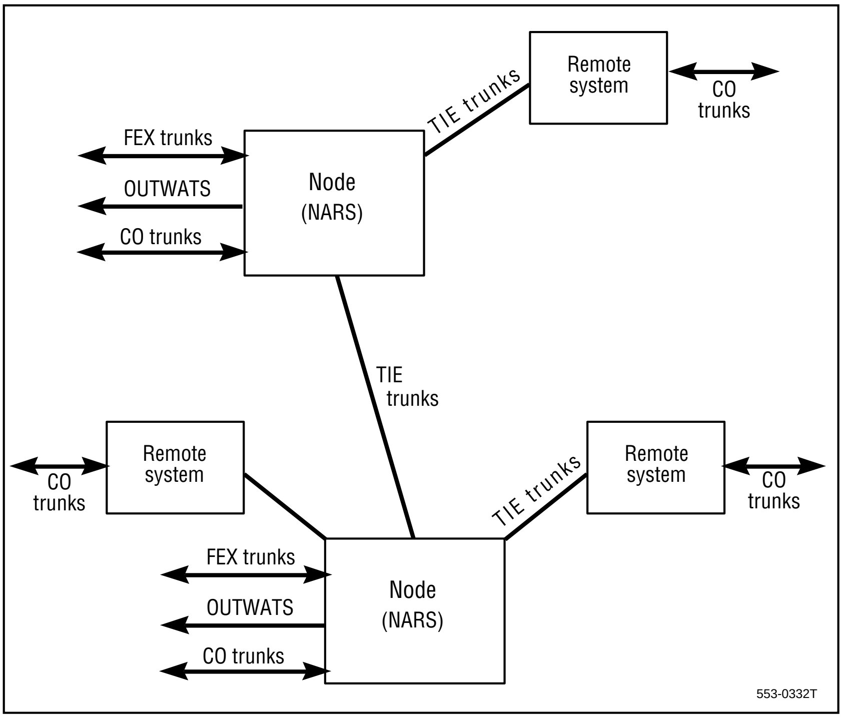
Electronic Switched Networks (ESNs) are usually formed around centralized nodes to which other switches are connected with TIE trunks. Nortel Networks recommends that you equip the node switches with NARS software.

NARS software was designed to:

- handle outgoing calls cost-effectively
- provide an easy, consistent dialing plan for users at the switch where the software is installed and at the switches connected to it with TIE trunks
- offer tools for good network management by
  - providing many ways of restricting users to reduce long distance costs
  - preventing unauthorized usage of trunks
  - increasing security

A typical network where NARS would be installed looks like this:

Figure 7-Typical application for NARS software



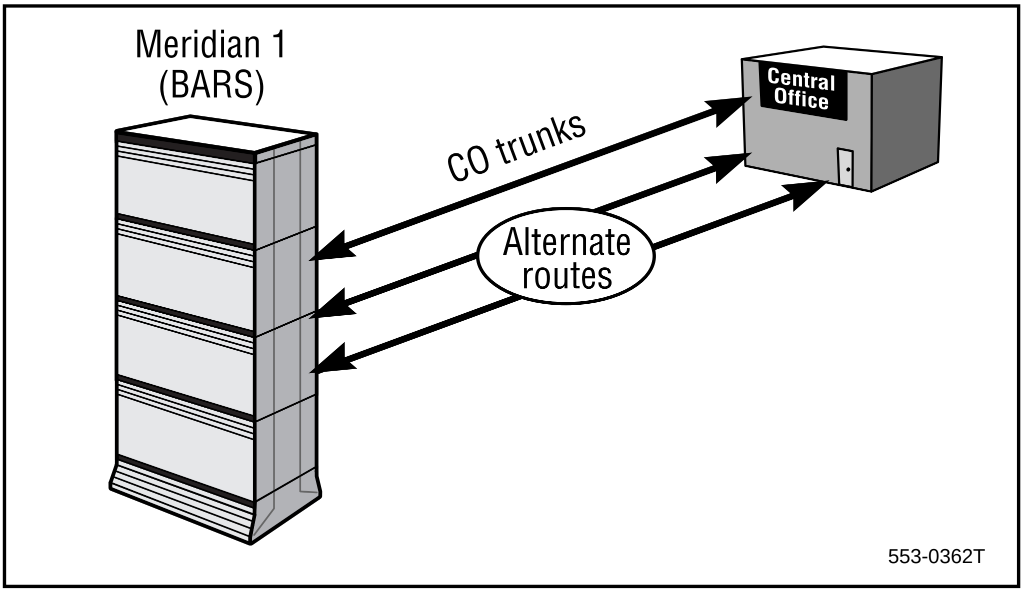
BARS

BARS was originally designed to perform automatic routing of calls for users at a switch connected to the public network only. In other words, BARS was not designed for switches connected to other private network switches.

However, BARS may be used at remote switches that have only one TIE trunk route to a node and one or more public network trunk route.

NARS and BARS

Figure 8- Typical application for BARS software



It is true that there are ESN networks where BARS has been installed at a switch performing as a node. This is not a supported configuration. This becomes especially important when the ESN network is being upgraded to ISDN functionality. The BARS software must be upgraded to NARS (or you can also choose BARS/NARS. Refer to the note marked ** under Table 30 that follows).

Table 30 compares the capabilities of BARS and NARS, and points out the differences. If you are just beginning to learn about BARS and NARS, the information contained in the table will make better sense after you have read the entire *NARS and BARS* section.

NARS and BARS

Table 30
BARS compared to NARS

NARS	BARS	Notes
AC1 and AC2 translation tables	Only an AC1 translation table	More flexibility with NARS. Less chance of conflict between translated codes. If there is a conflict, put one code in the AC1 table and the other in the AC2 table.
Location codes can be translated for on-net calls	No location codes	If you translate an on-net code as an SPN, the calls do not get processed until the NIT timer expires or the user dials #. If you translate an on-net code as an NXX, the DNs must be 4 digits long.
Location code translation includes conversion (on-net to off-net automatic overflow)	No conversion feature	Users with NARS who dial a DN that is out of range can reach an LDN at the other end. With BARS, if you want out-of-range calls to be processed, you must translate the NXX or SPN plus extra digits. Send these calls to a route list where you use DMIs to outpulse the LDN.
Home location code can be translated	No home location code.	Use local recognition with BARS. If there is an SPN or NXX code that represents the local switch, recognize the first digits of the internal DNs as Local DID or DDD DNs in the translation table.
When an HNPA code is translated, the system deletes it and inserts AC2 in front of the remaining digits.	When an HNPA code is translated, the system deletes it and inserts AC1 in front of the remaining digits.	In either case, you must translate the codes that will follow the automatically inserted AC2 or AC1 for the calls to be processed.

Note: ** The CDR feature prints call records differently with BARS than it does with NARS. In some countries you can order both BARS and NARS (referred to as BARS/NARS). If you are upgrading a BARS switch to NARS, and you want the CDR to remain in the BARS format, order BARS/NARS. If you do not mind the CDR format changing, order NARS. Refer to the CDR section, page 112, for more information.

NARS and BARS

Basic configuration

One of the things that NARS is designed to do is allow uniform, easy dialing between users at the different switches. These calls are referred to as *on-net calls*. (Refer to the section on the Uniform Dialing Plan on page 182.)

The software allows an outgoing call to be routed on the best available trunk route choice for that user at the time they are making the call.

Mains

NARS can handle the outgoing calls for users at the node and for users at the connected remote switches. The remote switches are referred to as mains. There are two kinds of mains:

- ESN Main
- Conventional Main

ESN Mains have NSIG (Network Signaling) software. They can also have one or more of: BARS, CDP, Network Call Transfer and Queuing software packages, to enhance the way they operate. Refer to the section on Network Signaling for more information on its functionality.

Conventional Mains do not have NSIG software. They can be any kind of system that can connect to the Meridian 1 node with TIE trunks. The node can provide a Conventional Main with access to many ESN features and functions. These are described in the sections of this book where relevant.

BARS

The main reason that BARS is not the best choice for a node is that it does not provide a Uniform Dialing Plan in the way that NARS does. You must do work-around programming to implement an on-net dialing plan. (Refer to the information on Translation of Location codes on page 174).

As a result, BARS it is not supported at a node in a multi-site private network, NARS is recommended instead. Properly configuring the systems in your network is important when you want to upgrade from an ESN network to a more sophisticated network using ISDN, NI-2 or QSIG, for example. (Refer to the *Administration tips* in this section).

Elements of NARS

Prime elements of the NARS feature are:

- Translation of digits
  - simple network access codes
  - translation of area codes, exchange codes, special numbers, location codes, home area codes, home location codes
  - Automatic on-network (on-net) to off-network (off-net) overflow (conversion)
  - Multiple DID Office Code screening
  - Supplemental Digit Restriction
  - Incoming Trunk Group Exclusion
  - Off-net number recognition
  - Local recognition
  - Flexible ESN “0” Routing
  - Interchangeable Numbering Plan Area
  - Carrier Access Codes
- Uniform Dialing Plan (UDP)
- Automatic least-cost routing
- Time of Day (TOD) routing
- Free Calling Area Screening (FCAS)
- Digit Manipulation Indexes (DMI)

- Network controls
  - Network Class of Service (NCOS)
  - Facility Restriction Levels (FRL)
  - Expensive Route Warning Tone (ERWT)
  - Routing Controls
- Queuing parameters

Translation

AC1 and AC2

Translation occurs when a user dials a call beginning with one of the two NARS access codes (AC1 and AC2). The NARS software must translate the digits that follow the access code in order to determine what to do with the call being dialed.

You do not have to implement a dialing plan using both access codes. You may want users to dial all calls beginning with the same access code, for simplicity.

However, some network planners prefer to associate long distance calls with AC1 and local non-chargeable (or less expensive) calls with AC2. This sometimes results in cost savings since users are more aware of when they are dialing expensive calls.

Conditionally Toll Denied Class of Service

If you are going to leave AC2 unassigned and users will dial local calls using a direct trunk access code (such as 9), it is very important that you assign a restricted class of service to telephones. This will prevent users from dialing 9+1 or 9+0 to make long distance calls, bypassing NARS (or BARS).

The class of service designed to control ESN calls is Conditionally Toll Denied (CTD).

Users with this class of service are:

- allowed to receive calls from the exchange network
- allowed access to WATS trunks for toll calls using direct trunk access codes unless New Flexible Code Restriction (NFCR) is programmed to deny certain digits
- denied from calls on COT/FEX trunks where 0 or 1 is dialed as a first or second digit following a direct trunk access code (special numbers such as 411, 611, 911 excepted). New Flexible Code Restriction tables can be used to deny or allow certain calls on these routes.
- allowed access to toll calls on COT/FEX/WATS trunks placed using BARS or NARS or CDP access codes. The FRL of the NCOS determines which toll calls are allowed. NFCR tables, if programmed on the routes are ignored for CTD users dialing ESN access codes.
- allowed toll calls and special number calls on TIE trunks unless NFCR tables specifically deny certain digits. Direct trunk access is permitted as well as BARS or NARS access. NFCR tables only deny calls for these users if direct TIE trunk access codes are used.

Translation tables

NARS has two translation tables, one for each access code. The user dials a call beginning with one of the access codes. One of two things can happen:

- If the digits dialed have been programmed into the translation table for that access code, NARS will handle the call.
- If the user dialed digits that are not part of the translation table, then intercept treatment is given. Refer to page 182 for more information on Translation-related Intercept treatments. Intercept can happen if:
  - the user has misdialed a call because of a lack of understanding of the dialing plan
  - the digits dialed are missing from the translation table

Functions of translation tables

The translation tables are a critical part of the operation of NARS. It is very important that this overlay program is done accurately and completely when NARS is installed. It must be well maintained after installation to ensure good network performance.

Translation tables contain programming to do the following:

- identify what type of code is being translated
- point the call to a certain route list index (trunk routes that are appropriate for this call)
- deny certain digits that follow the translated code (Supplemental Digit Restriction)
- allow only certain digits that follow the translated code
- identify an alternate route list index to be used when certain digits are dialed following a translated Special number code (Alternate Routing Remote Number with Flexible Numbering Plan)
- recognize certain digits following a translated code as a remote DID number (Remote DID Recognition)
- recognize certain digits following a translated code as a remote attendant number (Remote DDD Recognition)
- recognize certain digits following a translated code as a local DID number (Local DID Recognition)
- recognize certain digits following a translated code as a local DN number (Local DDD Recognition)
- deny certain digits that follow the translated code when the digits come into the node from one or more specified incoming trunk groups (Incoming Trunk Group Exclusion)

Translatable codes

Within one translation table there cannot be overlap between any two codes. This rule is called the “leftwise unique rule”.

NARS and BARS

For example, you cannot translate code 2 and code 22 in the AC1 translation table. However, you can translate code 2 in the AC1 translation table and code 22 in the AC2 translation table. For information on how this affects you, refer to the Uniform Dialing Plan on page 182.

There are two exceptions to this rule:

- Flexible ESN "0" Routing. Refer to page 178**.**
- Allowed digits. You can enter overlapping numbers when you are programming specifically allowed digits that follow a translated code.

The NARS software is designed to identify the digits that follow AC1 or AC2 as a certain type of code.

Table 31
Types of translatable codes

Type of code	Mnemonic	Example
Area code (North America)	NPA	**514**-555-1212
Exchange code	NXX	**555**-1212
Special Number	SPN	**911**
Location code	LOC	**333**-0000
Home Location code	HLOC	**846**-0000
Home area code	HNPA	**506**-555-1212

Once NARS has identified the type of code it is translating it can be prepared for the number of digits that will be dialed after the code (with area codes and exchange codes) or perform special operations on the call (with home area codes, location codes and home location codes).

Area codes (NPA)

Area codes are used in North America to identify regions. Area codes are three digits long. If the user is supposed to dial the digit 1 or 0 in front of these codes when they make calls, the area code must be programmed in the translation table as a four digit number to include the 1 or 0 at the beginning. The software expects 7 digits to follow a code identified and translated as an area code. It immediately routes the call when that seventh digit is dialed.

Until 1995, area codes had to have a 0 or 1 as the middle digit of the three digits. Now however, the middle digit can be any digit form 0 to 9, increasing the available codes to 640. These new format NPAs are called Interchangeable NPAs.

The introduction of Interchangeable NPAs meant that an area code (NPA) could be identical to a Central Office exchange code or a private network location code (LOC). This caused some conflicts with existing numbering plans.

> ***Note:*** Most network planners today avoid conflicts by implementing a “One plus” dialing plan where users will dial the digit 1 (or 0) ahead of all NPAs and no 1 (or 0) ahead of other codes. This prevents programming conflicts.
> Another alternative is to translate the two identical codes in different translation tables. The users can dial AC1 plus the NPA and AC2 plus the conflicting LOC or NXX. There is no conflict if you do that.

Exchange codes (NXX)

There are many Central Offices in each region or area code. Each Central Office can have many exchange codes. The exchange codes are three digits long. If the user is supposed to dial the digit 1 or 0 in front of these codes when they make calls, the exchange code must be programmed in the translation table as a four digit number to include the 1 or 0 at the beginning. The software expects 4 digits to follow a code identified and translated as an exchange code. It immediately routes the call when that fourth digit is dialed.

If the format of public exchange numbers in your region of the world does not match the requirements of the NXX code in the NARS translation table you can program Special numbers or use Flexible Numbering Plan software. Refer to page 178 and page 186 respectively for more information.

NARS and BARS

Location codes (LOC)

Each switch in a private Electronic Switched Network has a location code assigned to it. A user on the network dials this code when calling a user at another site.

The location codes are three digits long. The software expects 4 digits to follow a code identified and translated as a location code. It immediately routes the call when that fourth digit is dialed.

LOC codes are usually programmed in the AC1 translation table. You will not be able to program a LOC code into the database if it conflicts with any other numbers in the same translation table. You must assign LOC codes that are different from any other possible codes following AC1 in your dialing plan.

An easy way to ensure there will be no conflicts is to enforce a “one-plus” dialing plan. This means that users will dial 1+NPA for off-net long distance calls. If your LOC codes do not begin with “1” then there will be no conflict.

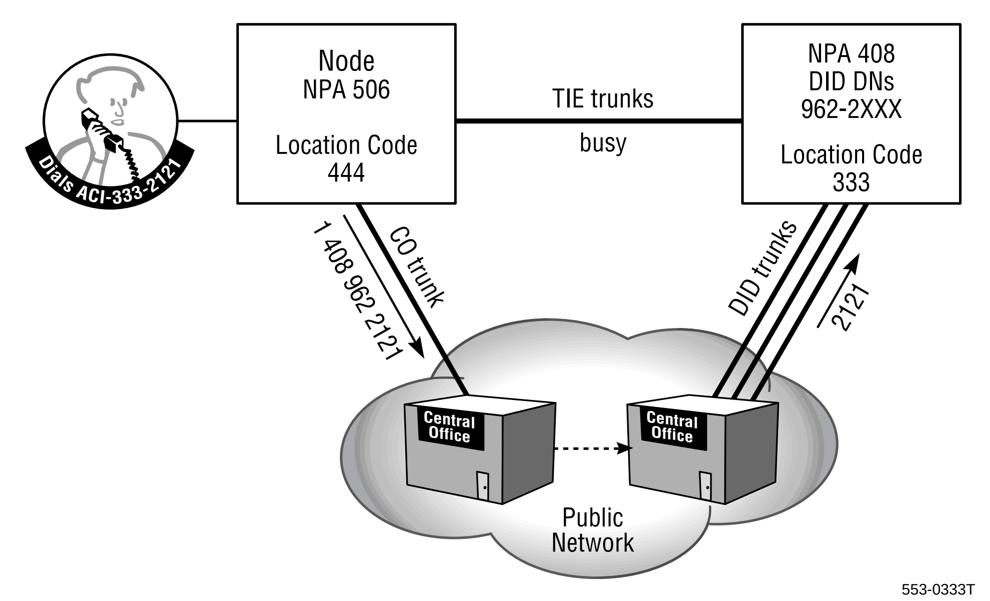
When you program a location code into the NARS translation database at a node, you also store other information about the distant site:

- whether there is more than one exchange code (NXX) for DID numbers assigned to the distant site
- the DN range that is valid for each DID exchange code
- the main listed number (to be used if someone dials a call that is outside the valid range of DNs for that location)

This information about the off-net numbering for the site is used if the location code (on-net) call is being routed on a public network trunk (off-net) from the node. This information will be used for the capability called *on-net to off-net automatic overflow* or *conversion.*

The user in the illustration that follows has dialed a location code and a DN (333-2121). The call is going to be sent out on a Central Office Trunk at the node because the TIE trunks to the remote site are busy at the time of the call.

Figure 9 - On-net to off-net automatic overflow



The NARS system uses the information in the translation table to assess whether the DN dialed (2121 in this example) is valid for that location code. If it is valid, it inserts the correct NXX code in front of the DID DN. It uses the main listed number information in the table to determine what area code, if any, must be inserted ahead of the NXX. All of this is done automatically by the conversion feature. You turn on the conversion feature in the programming of the Central Office trunk route choice in the route list index for calls to that location code.

You can program a digit manipulation index to insert the digit 1at the beginning of the digits to make it a long distance call. Digit manipulation occurs after conversion when both features operate on the same call.

NARS and BARS

BARS does not have the capability to translate LOC codes. If you attempt to use BARS at a node in a multi-switch network you would have to translate on-net codes as SPNs or NXXs. Refer to NXXs on page 173 and SPNs. on page 178.

- Translating the codes as SPNs makes the call processing quite slow. The system is not expecting a particular number of digits to follow an SPN. It waits for a timer to expire before processing the call. Users can press the # key to speed up call processing, when they have finished dialing. If you have DNs that are not always four digits long or if they vary in length, SPNs can be ideal for you.
- Translating the codes as NXXs. This is a better choice, from a call processing point of view, if you have four digit DNs following the on-net code. NXXs are expected to have four digits following.

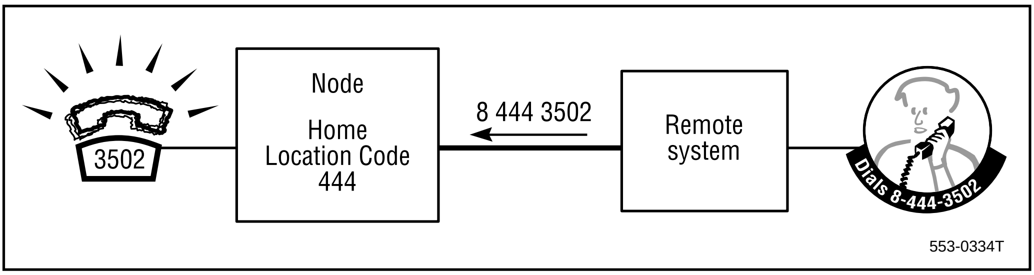
You cannot have any conflicting codes. Meeting this requirement can be difficult since BARS only has one translation table (AC1).

Home location codes (HLOC)

Each NARS switch must be able to identify its own location code when these digits are received from other switches. It must be able to handle these calls properly.

When a code is translated as a Home location code, the translation table also points to a digit manipulation index. This index will delete digits and or insert digits, as needed, so the incoming call will be sent to the appropriate DN at the node.

Figure 10 - Node translates Home location code



In the preceding illustration, a user at the remote switch is trying to call a user at the node. The user dialed 84443502. The call from the remote site comes in on a TIE trunk at the node. The node translates the three digits (444) that follow AC1 (8) as a HLOC. The AC1 code is automatically deleted by the NARS software when it translates the call. The digit manipulation index assigned to this home location code is programmed to delete three digits. Therefore, the system is left with a call to 3502, which is a DN at the node. The call is processed properly.

BARS nodes would have to perform a workaround using the Local Recognition feature, described in the *Improving feature performance* part of this section to deal with incoming calls to its own on-net code (translated as an NXX or SPN).

Home area codes (HNPA)

You can translate the NPA in which the switch is located in one of two ways:

- as an NPA
- as a Home NPA (HNPA)

If you translate a code as an HNPA:

- the system deletes the code after it translates it
- it automatically inserts
  - AC2 in front of the remaining digits with NARS
  - AC1 in front of the remaining digits with BARS
- the call is translated again as AC2 (or AC1) + NXX-XXXX

If you program an HNPA with NARS, you are forced to assign AC2. Users do not have to use it in the dialing plan but you must enter it in programming so it can be used for this translation. You must remember to program every NXX that exists in the area code into the AC2 translation table and route each one properly. (Refer to *Administration tips, Translating an HNPA with NARS* for more information on this.)

Special numbers (SPN)

Program numbers as Special numbers when they do not fit the format of the codes described previously.

When you translate the code as a SPN, the system does not expect any particular number of digits to follow. It outpulses the call when the end-of-dialing timer expires.

As a result, SPN calls can seem slow to users. Users can press the # button, when they are finished dialing, to make the timer expire manually. This makes call processing faster.

For example, calls to numbers such as 911 (a North American public network emergency number) cannot be handled as NPA or NXX or LOC calls since no digits will be dialed following the 911 code.

> ***Note:*** It is very important, in North America, that your system is programmed to handle 911 calls in such a way that they will always be processed, regardless of which telephone is used. Also, if you have both AC1 and AC2 programmed, translate 911 in both translation tables. This is because in an emergency, users may be confused and dial the call as AC1 + 911 or AC2 + 911. If AC2 is 9, it is also a good idea to program 11 as an SPN in the AC2 translation table. This allows users to simply dial 9+ 11. With Release 23 software, the Emergency Services Access feature improves access to emergency numbers. Refer to page 224 for more information.

Another example is calls that start with the digit 0. Overseas calls start with 00 (international operator), 01 (calling card, collect, or other operator-assisted international calls) or 011 (station-to-station international calls). Calls to the local operator begin with the digit 0.

The *Flexible ESN "0" Routing* capability allows you to translate the codes 0, 00, 01, and 011 in the same translation table, even though they overlap. For these four codes, the system does not apply the leftwise unique rule.

Because of this capability, you can program your system to allow everyone to make calls to the operator (0) but only your highest level users to call overseas (011), for example.

Carrier Access Codes (CACs) give a user access to any interexchange carrier or Operator Service Provider (OSP). In the United States, FCC regulations require that Call Aggregators, such as hotels, motels, hospitals, universities, airports, gas stations, and pay telephone owners, provide multi-carrier access to the public. Callers must be able to dial the CAC to reach their desired carrier or OSP before dialing the telephone number.

Aggregators, although they must allow callers access to any long distance carrier, are permitted to block calls selectively. Selective equal access lets aggregators choose to block direct-dialed calls that result in charges to the originating telephone. Aggregators cannot block operator-assisted calls.

Nortel Networks provided an up-issue of X11 Release 14 in 1992 to conform to FCC Equal Access requirements. Beginning with X11 release 17, all software releases support Equal Access. (X11 Releases 15 and 16 do not support Equal Access.) Support for expanded codes, as described in the following paragraph, is available beginning with X11 Release 19.

The CAC includes a “10” identifying prefix followed by a three-digit Carrier Identification Code (CIC) for a total of five digits. Additional FCC regulations, reflected in X11 Release 19, require that the CAC expand to seven digits: a “101” identifying prefix followed by a four-digit CIC. The regulations required that both the old five-digit format and the newer seven-digit format were supported for an 18 month permissive period during 1995 and 1996. After this date, only the longer format is supported.

Meridian 1 allows the following operator-assisted North American and international dialing sequences:

- CAC+ 0
- CAC + 0 + NPA + NXX + XXXX
- CAC + 01 + Country Code + National Number
- CAC + 1 + NPA + NXX + XXXX
- CAC + 011 + Country Code + National Number

NARS and BARS

where:

CAC= Carrier Access Code (10XXX or 101XXXX)

For complete information on implementation and configuration, refer to “Equal Access Compliance” in the *X11 Software Features Guide.*

E.164/ESN Numbering Plan Expansion feature was introduced in X11 Release 22. It provides capabilities to meet the International Telegraph and Telephone Consultative Committee (CCITT) recommendation E.164 for Integrated Services Digital Network (ISDN) and Public Switching Telephone Network (PSTN) dialing.

Generally, the following enhancements are offered by this feature:

- the Meridian 1 base features, such as Autodial, can store up to 31 digits
- the addition of “allow” (ALOW) as a Supplemental Digit Restriction and Recognition (SDRR) entry in the translation table.
- previously when you programmed digits to be recognized or restricted following a translated code, you could not have overlapping numbers. They had to be leftwise unique. With this feature that is no longer true. For example, if you deny the digits 555 following NPA 514, the system will also accept 55 as a recognized number or 5551212 as allowed. The system waits for all digits to be dialed until a decision is made on how the call is to be handled. The longest matched sequence between the dialed digits and the programmed instructions determines what the system will do with the call.
- the Special Number (SPN) translation screening digits scheme is changed from an 11 digit maximum to a 19 digit maximum

The specific enhancements delivered by the E.164/ESN Numbering Plan Expansion feature are provided in *Appendix 2*.

Restricted numbers and recognized numbers

In the translation tables you can input digits that are to be treated uniquely when they follow the translated code. Using this capability you can treat calls in the following ways:

- restrict certain digits for the entire customer group (Supplemental Digit Restriction - refer to page 230)
- allow specific digits after the translated code for the entire customer group (refer to the note on page 222)
- send SPN calls with certain digits to an alternate route list index (refer to page 288)
- recognize that certain digits are assigned to a main answering position at a remote switch (refer to note 1 and page 220)
- recognize that certain digits are assigned to DNs at a remote switch (refer to note 1 and page 220)
- recognize that certain digits are assigned to a main answering position at the local switch (refer to note 2 and page 220)
- recognize that certain digits are assigned to DNs at the local switch (refer to note 2 and page 220)
- recognize that certain digits are to be blocked when they come into the node on an incoming TIE trunk (Incoming Trunk Group Exclusion - refer to note 3 and page 217)

It is beyond the scope of this book to discuss the actual programming of these different forms of restriction and recognition. However, it is important that you find out if there are numbers following each translated code (NPA, NXX, SPN, LOC, HNPA, HLOC) at each node that are to be treated in any of the ways listed in the preceding list. Your system supplier identifies these digits with the proper mnemonic in the translation tables so the proper treatment occurs.

> ***Note 1:*** For remote recognition you must program the TIE trunk to the remote switch as a conventional switch (CNVT) in the Route Data Block. Then you specify the special Digit Manipulation Index (DDMI) that deletes and inserts digits as

required (to outpulse a remote DN) or you input the attendant DN in the Route Data Block that should be outpulsed when a remote attendant number has been recognized.

Note 2: For local recognition you must assign a Digit Manipulation Index (DMI) in the translation table that inserts and deletes digits so calls are terminated properly to the internal DN or attendant position.

Note 3: To block certain calls from incoming trunk groups, you assign an index number to the excluded digits in the translation table. Then you apply this index to the correct incoming route(s).

Translation-related Intercept treatments

Intercept treatments are programmed in the Customer Data Block. They tell the system what to do when users in that Customer Group dial illegal or invalid numbers. The three choices of treatments are:

- give the user overflow tone
- connect the user with the attendant
- connect the user with a recorded announcement

The treatments that apply to ESN translation are:

- Invalid translation intercept treatment - given to the user if the digits dialed have not been programmed into the translation table for the access code used
- NARS/BARS restricted call intercept treatment - given to a user who dials a number that is programmed as restricted

Refer to *Administration tips* and *Maintenance tips* for more information about how to use intercept treatments effectively.

Uniform Dialing Plan (UDP)

When network planners implement networks, their goals usually include:

- simplifying the dialing plan at all sites
- making the dialing plan uniform at all sites

The Uniform Dialing Plan provided by NARS addresses these requirements and provides the following benefits:

- end user training is easier
- users travelling from site to site, will know how to make calls, no matter where they are on the network
- network maintenance is easier

By installing NARS at nodes, any remote sites connected directly to the nodes with TIE trunks can use the UDP, *no matter what type of system or what software is present at the remote site.*

Users do not have to be concerned with how the call is going to be routed. They do not have to think about which route to use and remember the correct access code for each route. They follow the NARS dialing plan and the NARS software takes care of it.

A typical UDP looks like this:

Table 32
Sample UDP

Type of call	Dialing plan
Off-net, long distance calls	AC1 + 1 + NPA + NXXXXXX
On-net calls	AC1 + LOC + XXXX
Off-net, local calls	AC2 + NXXXXXX

Planning your UDP

Choosing access codes (AC1 and AC2) for NARS may be difficult. You must try to find two numbers that are available at all network sites.

- If the remote site does not have any alternate routing software (CDP, BARS or NARS) the users will dial the access code for the TIE trunk route to the node when they make long distance calls. The calls will be handled by the node and sent out on its trunks. If you want the users at the remote site to have a UDP, the same as the node, the access code to the TIE trunk route at the remote site will have to be the same as the NARS AC1 at the node. The access code for local public network calls at the remote site will have to be the same as the NARS AC2.

- If the remote site has CDP, the Trunk Steering Code should be the same as AC1 at the node. Refer to the section on CDP in this book for more information on steering codes.
- If the remote site has BARS or NARS, the AC1 at the remote site should be the same as the AC1 at the node.

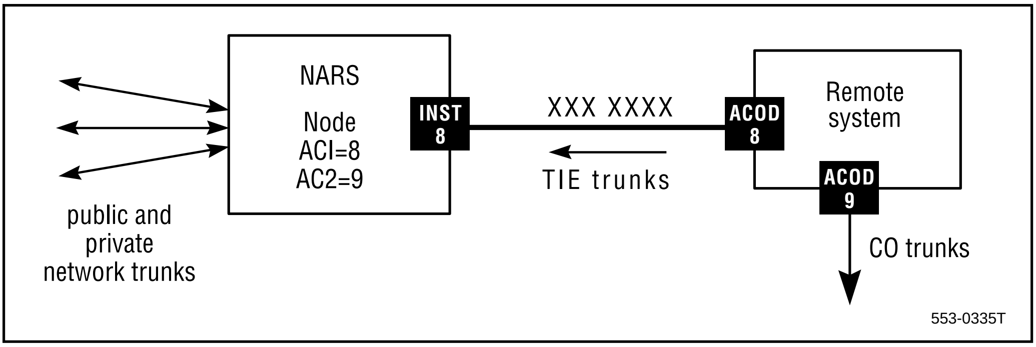
You may have to change existing programming at some sites in order to make the codes you are planning to implement available for the UDP dialing plan at all sites.

For example, you may want AC1 to be 8 and AC2 to be 9 at the node. You may find that one of the remote sites connected to it already has 81 assigned as a TIE trunk route access code (in the Route Data Block) and 82 is already assigned as an FEX trunk route access code (in the Route Data Block). You must change the TIE and FEX access codes to numbers that do not start with 8. That way you can assign 8 as the access code to the TIE trunk route in order to make the dialing plan consistent at the node and the remote site. Make sure that 9 is assigned as the Central Office trunk route access code at the remote site.

Using the INST prompt

In order to make this dialing plan work, you must program the incoming TIE trunk route at the node to insert the digit 8 before the rest of the digits in calls coming from the remote site. (Do this in the Route Data Block using the INST prompt). In this way, the node will have the digit 8 (AC1) at the beginning of all calls from the remote site. As a result, NARS will translate the calls.

Figure 11- Uniform Dialing Plan and the INST prompt



Use the INST prompt only if the following conditions apply:

- if BARS, NARS or CDP software is *not* programmed at the remote site. With one of these programmed, the remote site can use digit manipulation indexes to outpulse the correct digits to the node. Therefore, there is no need to insert a digit at the node.
- if you are *not* using ISDN networking. With ISDN you would use the INAC prompt when programming the TIE trunk in the Route Data Block. With INAC activated, the system inserts AC1 or AC2 ahead of the digits it receives, *depending on what digits they are*. In the Customer Data Block at the node, you list the codes that should have AC2 put in front of them. The unspecified codes will have AC1 put in front of them by the INAC software, automatically,

After implementing the INST capability, the users at the node and at the remote site in the preceding illustration dial calls this way:

Table 33
Dialing Plan

Type of call	Digits dialed
Off-net, long distance calls	8 + 1 + NPA + NXXXXXX
On-net calls	8 + LOC + XXXX
Off-net, local calls	9 + NXXXXXX

Flexible Numbering Plan (FNP)

With X11 Release 20, FNP was offered globally. Previous to that release it was only available in non-North American markets. NARS is a pre-requisite for FNP.

FNP allows variability in the length of the on-net DNs within one network. It allows up to 7 digits for location codes and up to 10 digits for on-net calls. The total number of digits to be dialed can vary depending on the telephone called.

FNP also interacts with Coordinated Dialing Plan (CDP) software. Refer to the section on CDP or refer to the section on FNP in this book for more information.

Route List Indexes (RLIs)

Translation and route list indexes

Translation tables point each translated code to the correct route list index. *There is no limit to the codes that can point to the same route list index.* When you are planning to translate a new code, evaluate your existing route list indexes to find out if there is already one programmed that is suitable. Do not add a new route list index for every new code you translate. You will save a lot of memory if several codes point to the same route list index.

Route list related Intercept treatment

If the route list index (RLI) that a translated code is pointed to has been removed from the programming, the "Invalid NARS/BARS call" Intercept treatment will be given to any user who dials the translated code. This Intercept treatment (attendant, overflow tone or RAN) is programmed in the Customer Data Block.

First choices, last choices

The route list index (RLI) is a series of trunk route choices that are listed in the order in which the network planner wants them to be scanned. The routes are scanned in that order, each time a call is made and sent to that RLI.

Planners usually list the least expensive routes first, to save the most money.

> ***Note:*** Studies have shown that when software is programmed to select the best trunk for a user, monthly savings can be as high as 20%.

When users make calls using direct trunk access codes they often have difficulty deciding what route to choose for each call they make. This results in under-utilized trunk routes and trunk routes that are used improperly.

Also, if a first choice route is busy, many users will immediately try the most expensive trunk route next. Some impatient users end up using only the most expensive trunk, all the time, since the less expensive ones were blocked when they tried them initially.

When calls are routed automatically, the system can scan many appropriate trunk choices in a pre-programmed sequence.This produces substantial cost savings.

Planners also put flat rate trunk route choices first (such as TIE trunks or FEX trunks) so they get the highest amount of traffic on these. Putting large numbers of calls on flat rate trunks brings the cost of each call down. The last choice(s) in the route list index can be the most expensive trunk routes or those on which you want the least traffic.

NARS and BARS

Initial set (Iset)

There is a marker that must be programmed for each route list index. It is called the Initial set (Iset).

- The Iset is a count of the number of routes in the route list index that are to be considered inexpensive. If you program a route as expensive but include it in the Iset of a RLI, the system will not treat it as expensive. In other words, the Iset marker takes precedence over the expensive/non-expensive designation of a given route.
- The Iset routes are also the routes that everyone queues for at first when they activate Callback Queuing (the NARS version of Ring Again). The shortest amount of time users can queue for these routes only before they can also queue for the expensive routes is 30 seconds. Refer to the section on *Queuing* in this book for more information.

There is more information on route list indexes in the Basic and Network Alternate Route Selection section of the Networking NTPs. Refer to "Automatic Least Cost Routing."

The trunk routes that are listed in a RLI do not all have to be different routes. You can list the same trunk route more than once and use it for different kinds of calls by different users if you want.

For example, if you have TIE trunks to a system in a different area code, you may program the TIE trunk route first in the route list, to be used for free calls to that area code, accessible by all users. You may program it a second time in the same route list index to be used for long distance calls to that area code, accessible by high level users only. You can set up the programming on each route entry to:

- screen the calls for the exchanges dialed. Allow only free ones on the first appearance of the TIE trunk route. Only long distance ones would be allowed on the second appearance of the TIE trunk route in the route list index.
- control which users can access each entry when dialing either of these two kinds of exchanges

Each choice in the RLI has the following parameters programmed for it:

- an entry number (0 to 63). The system always scans route list indexes starting at entry 0.
- trunk route number. Each trunk route must have a route number assigned before it can be programmed as an entry in a RLI.
- Time of Day schedules (TOD) when the route is on or off in that route list. Refer to page 212.
- instructions on whether to use on-net to off-net automatic overflow instructions from the translation table when outpulsing a call on this route. Refer to page 192.
- instructions on whether to treat the call as an internal one. Refer to page 192.
- Digit Manipulation Index (DMI). Refer to page 193.
- a minimum level (Facility Restriction Level - FRL) that users must be in order to access the route when it is used in this RLI. Each entry in a route list has a minimum FRL required for access. All network users are assigned an FRL through their Network Class of Service (NCOS). You can restrict the type of calls and route choices allowed to users. Refer to page 196 for more information on NCOS.
- Free Calling Area Screening Index (FCI). Refer to page 207.
- whether it is to be treated as expensive/non-expensive. Refer to page 198 and refer to *Control tips*.
- whether Queuing options are activated or not (Callback Queuing, Off-Hook Queuing, ISDN Off-Hook Queuing, ISDN Drop Back Busy, Step Back on Congestion). Refer to the *Queuing* section in this book.
- Free Special Number Screening Index (FSNS). (Refer to the *Flexible Numbering Plan* section for more information).

NARS and BARS

- instructions on whether to skip conventional signaling or not (refer to the *ISDN* section on Revert to Conventional Signaling)
- Digit Manipulation Index (ISDM) to use if ISDN Signaling Link (ISL) is installed and the D-channel is not working temporarily (Refer to the *ISDN Signaling Link (ISL)* section - Revert to Conventional Signaling feature.)
- instructions on whether the route is to be used for Virtual Network Services or not (refer to the *Virtual Network Services (VNS)* section)
- instructions on whether to use Tone Detectors or not

The entire RLI has:

- an Iset marker
- the minimum level (MFRL) users have to be to begin to scan the RLI (only used with Network Authorization Codes software)
- the number of digits that must be dialed, after the ESN access code, before the SETUP message is sent (ISDN) or outpulsing begins

The decisions you need to make concerning these route list parameters are listed in the *What to have ready* table in this section.

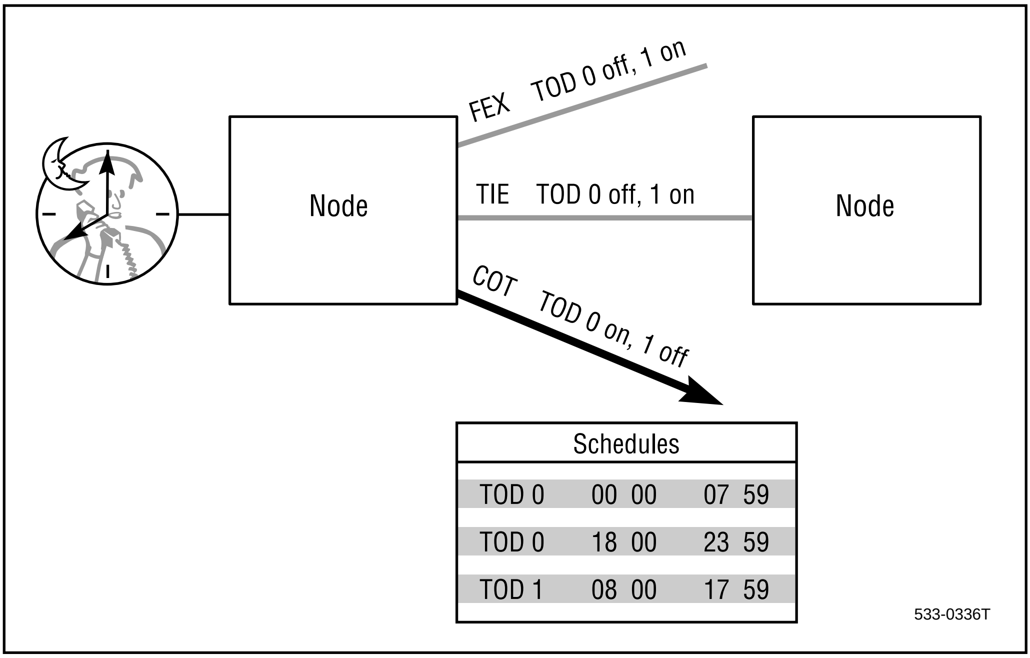
Time-of-day schedules

You can break the 24 hour day into a maximum of eight parts or schedules. You use the schedules to turn routes on and off at certain times in certain route lists. Each entry in the route list has all eight time-of-day schedules applied.

The entry is on all day by default. You turn the schedules off for the hours of the day when you do not want the entry used.

As you can see in the illustration that follows, the user is making a call at 8:00 pm. The FEX and the TIE trunk routes in the route list being used are turned off during TOD 0 (6 pm to 8 am). The COT trunk route is turned on during TOD 0. Therefore, the call goes out on the COT trunk route.

Figure 12 - Route list indexes and Time-of-day schedules



If you want routes on during the work day only, use time-of-day (TOD) schedules to turn off all routes in all route lists at a certain time every day. (You can also use the Routing Control feature to do this. Refer to page 212.)

Each schedule can have more than one period in it. For example, schedule 1 could be 6:00 am to 8:00 am and 4:00 pm to 7:00 pm. You would program a schedule this way if you treat routes the same way during both of these time periods. In other words if you turn a route off at 6:00 am to 8:00 am it will also be off between 4:00 pm and 7:00 pm, if both time periods are part of the same schedule.

Most planners use the TOD schedules to turn off routes when they are not cost effective or to force calls to overflow to other choices.

NARS and BARS

Automatic on-net to off-net overflow (conversion)

This feature applies to location code calls only. Location codes are translated and pointed to route lists that may include entries for TIE trunks (on-net trunks) and public network trunks such as FEX and COT trunks (off-net trunks).

- If the call is going to be sent out on a TIE trunk you do not need to convert the number to an off-net format, so the conversion feature is left off (default).
- If the call is going to be sent out on a public network trunk, then conversion is needed. The off-net number information from the translation table for that location code will be used to outpulse the correct off-net number to the public network. The public network will send the call to the other site on public network trunks, usually DID trunks. Conversion puts the correct exchange code and possibly area code in front of the DID DN dialed.

Local Termination

You may have a case where you want calls sent to an internal number if all the other entries in the route list are busy. You can do this if you program an entry in the route list at the end of all the other entries that has local termination active.

The entry can refer to a digit manipulation table that will process the digits dialed so you are left with the internal DN you want. This internal DN can be:

- a single call or multiple call DN on one or more telephones
- a voice mail DN (where an announcement service could give the caller instructions or the user could leave a message)
- an ACD DN
- an attendant DN

You might want to use this capability to signal a network administrator’s DN. These calls would tell that person that all routes in a certain route list are busy. If they started to get a large volume of calls they could investigate the problem.

Instead of an internal DN you can also use digit manipulation to end up with AC1or AC2 plus more digits so that translation begins all over again. Another route list with different routes could be tried. In this way you can daisy-chain route lists.

Do not use local termination on an entry unless it is the last one in the route list. The system will not overflow calls to entries after one with local termination activated.

Digit manipulation indexes (DMI)

NARS uses digit manipulation indexes to modify the digits that the user dialed to make them appropriate for the trunk being used for the call.

You must know what digits the system at the other end of the trunk requires in order to process the call in the way you want it to process the call. You can control the way the other system will handle each call you send to it by sending digits that will force it to handle the call a certain way. For example, if you want the call to go out on a particular route at the other site, send a trunk route access code as leading digits. If you want the other site to use alternate routing, send it AC1 or AC2 or CDP steering codes, as required.

For example, if you have a remote site with BARS, you can send calls to it on TIE trunks with AC1 at the beginning. This will allow the remote switch to use BARS to route the call effectively. On the other hand, you could send the call to the remote site with a certain trunk route access code at the beginning to force the remote site to send the call out on a certain route.

Digit manipulation indexes delete digits from the beginning of the string of dialed digits. You don't specify what specific digits are being deleted, you program the number of digits that are to be deleted. After deletion occurs, then insertion of specific digits at the beginning of the remaining digits occurs.

AC1 and AC2 do not have to be deleted manually, they are deleted automatically by the system once the call gets to the correct translation table.

You can use DMIs to insert pauses. An asterisk (*) indicates a three second pause before the remaining digits are outpulsed. You do not need to insert pauses for dial tone on Meridian 1 systems. You may need them if you send calls to other systems that require a pause in order to find a trunk and send the call out.

If you do not want any digits deleted or inserted before the call is sent out on an entry in a route list, assign DMI 0. This index has no deletion and insertion, by default.

Network Class of Service (NCOS)

You can assign up to 100 NCOS groups per customer group.

It is best to keep the number of NCOS groups you use to a minimum. The fewer groups you define, the easier it is for network administrators to maintain the network.

Do not make the differences between your NCOS groups so minor that it will be difficult for programmers to decide what NCOS to assign when they install new telephones.

Evaluate the network requirements of your users. Group them together based on those needs, not necessarily based on their job functions. For example, if executives and managers have the same network requirements put them together in the same NCOS group.

You can assign an NCOS to:

- telephones (a call placed from any DN assigned to the telephone is controlled by the NCOS assigned to the telephone)
- attendant consoles (usually you assign the least restricted NCOS)
- DISA DNs (externally-originated calls placed using this port are controlled by this NCOS)

- incoming TIE trunks, and the effect of this is:
  - Users at remote sites who make calls to the node on a TIE trunk are controlled by the NCOS assigned to the TIE trunk they are using. The node restricts the caller to placing calls that are allowed for the NCOS of the TIE trunk.
  - If Network Signaling software (NSIG) is programmed at the remote site and at the node, then the NCOS of the caller will travel on a TIE trunk as extra signaling digits. This travelling NCOS overrides the NCOS of the TIE trunk. The node deals with each caller based on the caller's NCOS, not the NCOS of the TIE trunk. Refer to the *Network Signaling* section in this book for more information on NSIG.
- ACD DNs
- Meridian Mail ACD DNs. Refer to *Control Tips* for more information.
- Authorization Codes (optional software)
- Network Speed Call lists (optional software)

If you have NSIG, CDP, BARS, or NARS software programmed at a switch you will be able to program NCOS groups.

If you have a network of switches you should program an identical set of NCOS groups at all sites. This makes network administration and maintenance easier.

Be careful when you are installing anything new that requires an NCOS, such as TIE trunks and telephones. *NCOS 0 is assigned by default*. Refer to *Maintenance tips* and *Control tips* for the impact this can have on your system.

Make sure you program the correct NCOS value. Assigning incorrect or default NCOSs can be a source of many problems in a network. As a network planner you should find ways to clearly communicate the NCOS programming you have in mind to the administrators at all sites.

NARS and BARS

Attributes of an NCOS

The attributes affiliated with an NCOS are:

- Facility Restriction Level (FRL)
- Expensive Route Warning Tone activation
- Queuing parameters
- Network Speed Call feature and list access
- Automatic Redial parameters
- Maximum Precedence Level (Autovon)
- Equal Access

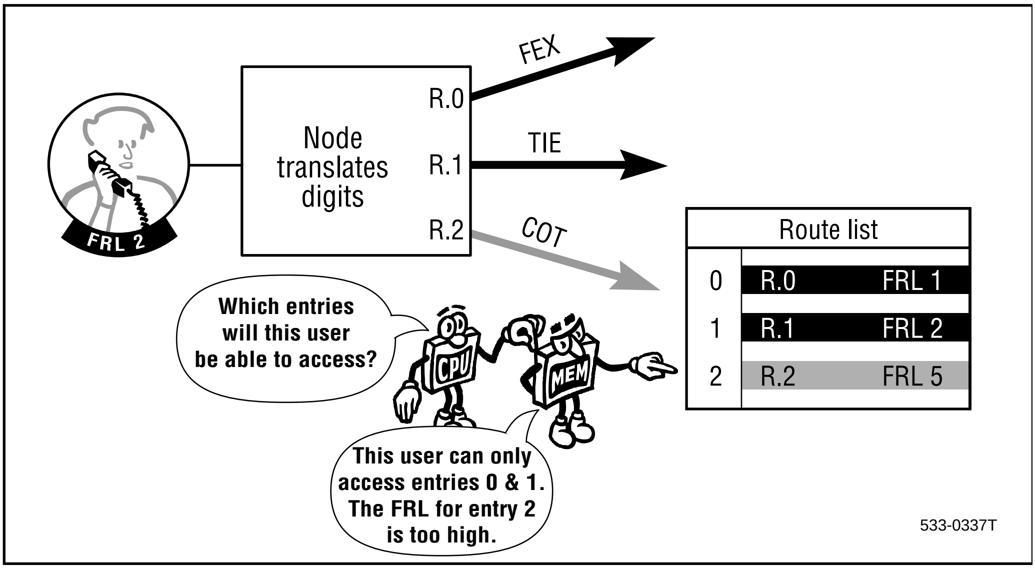
NCOS – Facility Restriction Level (FRL)

You assign the minimum level that a caller must have in order to access each entry in a route list. The level is called a Facility Restriction Level since it is a way of restricting access to facilities (trunks).

The user's level is determined by the Network Class of Service that is assigned to them. Each NCOS you program has an FRL assigned.

If a user's FRL is equal to or greater than the required FRL on the entry in the route list, they can access it.

Figure 13 - Route list indexes and FRLs



FRL-related Intercept treatment

When a user dials a call that requires a higher FRL than they have, the "NARS/BARS blocked call" Intercept treatment will be given.

How to use FRLs

FRLs can range from 0 (most restricted) to 7 (least restricted).

If you do not want low level users to access expensive trunks, assign a low level FRL to their NCOSs. Program a higher FRL on the expensive route entries in the route lists.

The same route can appear as an entry in many different route list indexes. Each time the route appears as an entry it can have a different FRL. The FRL you program depends on what types of calls the route is handling in that route list. You decide for each route entry in each route list the minimum FRL a user must be in order to use it.

For example, COT trunks in route 8, used in a route list for overseas calls, might require an FRL of 7 but the same COT trunks in route 8 in a route list for local calls might only require an FRL of 1.

NARS and BARS

Also, if long distance calls are less expensive at night, you may want to restrict the users who access the COT trunks during the day and allow more users to call out on these trunks at night. On the route lists for long distance calls, the entry for the COT trunks in the daytime might require an FRL of 4 when the calls are expensive but the same COT trunk route in the same route list for night time long distance calls might only require an FRL of 1. In this example, the same route appears twice in one route list. With different time-of-day schedules active to control when the route is on and off, you can use FRLs to control who accesses the route at different times of day.

911 emergency calls

In North America, you will probably have more than one route list for 911 emergency calls:

- calls to AC1 plus 911 and AC2 plus 911 can use the same route list. Make sure that the routes in the route list have FRL 0 required.
- calls to AC2 plus 11 (if AC2 is 9) will be handled by a different route list where a digit manipulation table inserts the digit 9 before the call is outpulsed. Make sure that the routes in the route list have FRL 0 required.

NCOS – Expensive Route Warning Tone

Users who are able to scan all routes in route lists when they make calls can be given Expensive Route Warning Tone (ERWT). Refer to the information that follows on how Call back Queuing parameters affect what routes a user will scan when they access route lists.

This tone is given when there is no inexpensive trunk available in the route list and when they are about to access a trunk in a route that is programmed as expensive.

If you want users in a certain NOS group to hear the ERWT, activate it in the programming of the NCOS group. You must also designate certain routes as expensive, exclude these routes from the Initial sets of the route lists and turn on the ERWT for the Customer Group.

When the tone is given, the expensive trunk is in a seized state, waiting for the decision the user will make. The user can decide to:

- hang on. The call will go out on the expensive trunk, after the Expensive Route Delay Timer expires.
- hang up and try to make the call later. Hopefully, it will go out on an inexpensive trunk then.
- activate Callback Queuing (the NARS version of the Ring Again feature). They will queue for the inexpensive trunk routes.

Refer to *Training tips* and *Control tips* for more information on the implementation of this tone.

NCOS – Queuing parameters

A user may be offered queuing when trunks or DNs are busy. Some queuing is offered when a DN is not answered.

There are several forms of queuing offered in ESN and MCDN ISDN networks.

A user's access to these features can often be controlled in the NCOS programming and in the telephone set programming as well.

The types of queuing are:

ESN and ISDN

- Ring Again - part of basic system
- Callback Queuing - part of BARS, NARS, CDP
- Priority Queuing - optional software
- Off-Hook Queuing - optional software

ESN only

- Coordinated Callback Queuing - optional software
- Callback Queuing to/from a Conventional Main - optional software

NARS and BARS

ISDN only

- Remote Virtual Queuing - optional software
- Automatic Redial
- Network Drop Back Busy

Refer to the section called *Queuing* in this book for more information on the many types of queuing offered in all types of networks.

To queue or not to queue

You will have to assess whether you want to implement queuing features on your network. You must evaluate the effect that delaying calls will have on your users and on your business.

You may find that it is worth it to you to install the necessary trunks for your peak traffic times rather than to provision fewer trunks so that users will be expected to queue at busy times.

When you expect users to queue, at times, you will need to set up a training program to ensure they understand how to queue. You should also monitor the situation afterwards to investigate:

- how much queuing is occurring
- what routes and route lists are blocked
- the amount of time users spend queuing

You can monitor the situation using Network Traffic software to run studies. Refer to the section called *Network Traffic studies* for further information.

You can survey users to get feedback on what type of calls get blocked. Also, attendants can tell you if there are blocked trunks by looking for Trunk Group Busy keys on the console that flash frequently.

Be ready to perform refresher training, especially if you find that trunks are blocked at times but there is very little queuing being done.

Refer to *Improving feature performance* and *tips* in this section for further information.

Ring Again vs. Callback Queuing

Callback Queuing

This form of queuing is a basic capability of NARS (and also BARS and CDP). It allows a user who encounters busy trunks to activate queuing in the same way they would activate Ring Again. They dial a SPRE code plus 1 on an analog (500/2500 type) set, or they dial a Flexible Feature Code, or they press a key on a Meridian1 proprietary set.

The differences between Ring Again and Callback Queuing are summarized in the following chart.

Ring Again	Callback Queuing
The user dials ACOD for the route; if all trunks in that route are busy, congestion tone is given.	The user dials AC1 or AC2 plus all remaining digits; if all routes in the route list for that call are busy, congestion tone is given.
The user queues for one trunk route.	The user queues for Iset routes in the route list initially until the Extended Route Advance Timer expires. If the timer expires before a trunk is found, the user queues for all routes in the route list.
When a trunk is found, the user is given callback. The ACOD does not have to be redialed but the rest of the digits must be dialed.	When a trunk is found, the user is given callback. The digits in the call do not have to be redialed.
A trunk is seized when the user is being given callback	A trunk is seized when the user is being given callback. Digits are slowly outpulsed until the user answers or the callback timer expires.
The user has 6 seconds to answer the Ring Again callback on analog (500/2500) sets and 10 seconds from Meridian 1 proprietary sets.	The user has 6 seconds to answer the CBQ callback on analog (500/2500) sets and a programmable number of seconds (10 - 30) from Meridian 1 proprietary sets.
Ring Again is assigned in the Class of Service or on a key.	CBQ must pass several eligibility tests and Ring Again is assigned in the Class of Service or on a key.

NARS and BARS

Callback Queuing type affects the routes scanned

Callback Queuing parameters assigned in a user's NCOS not only affect the user after they queue but also when they scan routes in route lists when they make calls.

There are two kinds of CBQ that can be assigned in a user's NCOS:

- CBQ type I (default)
  - CBQ type I users only scan Initial set (Iset) routes in route lists when they make calls
- CBQ type A
  - CBQ type A users scan all routes (Initial set and extended set) in route lists

Higher level users should be given CBQ type A in their NCOS.

If you want a user group to be given Expensive Route Warning Tone, they must be assigned CBQ type A so they can scan expensive routes.

When users encounter blocked trunks and they activate CBQ, *everyone starts off queuing for the Iset (inexpensive) routes*. This is the design intent of Callback Queuing to save the business as much money as possible.

The Extended Route Advance Timer programmed in the NCOS, determines how long a user will queue for Iset routes only before they can queue for all routes in a route list. You can program this timer in 30 second increments. The lowest possible setting for this timer is 30 seconds. This means that after 30 seconds of queuing for Iset routes only, a user can queue for all routes in a route list. If you do not want the timer to expire (to keep the user queuing on Iset routes) set the timer at 0 (default).

You may decide to keep certain low level users queuing for the inexpensive routes only, until a trunk becomes available. You may decide that high level users should begin to queue for all the routes in a route list after a certain number of seconds.

When a trunk becomes available the system still checks the FRL required on the route against the user's FRL. If the user's FRL is not high enough the trunk will not be offered.

If the call ends up going out on an expensive trunk after the user has already queued, there will not be an Expensive Route Warning Tone given.

NCOS – Network Speed Call feature and list access

Network Speed Call (NSC) gives users across an ESN network access to System Speed Call lists programmed at the node.

To do this, an access code for each list must be programmed into the AC1 or AC2 translation tables at the node. These access codes cannot conflict with any other code in the table. This is part of the UDP offered by a NARS node when it has the optional Network Speed Call software package equipped.

The user will dial AC1 or AC2 plus one, two or three digits (that do not conflict with any other codes) followed by a one, two or three digit speed call entry code that identifies the entry on the System Speed Call list that the user wishes to dial.

If you have Network Speed Call software programmed on your system, you will be able to specify:

- the NCOS groups that have access to the feature
- the list(s) each NCOS group can use

An NCOS group can access all System Speed Call lists that have Network Speed Call access codes programmed in the translation table or one specific list.

Each Network Speed Call list has an assigned NCOS that will override the user's NCOS, if the NCOS of the list has a higher FRL than the user's FRL. In other words, use of Network Speed Call will never lower a user's capabilities.

Users at connected remote switches can access Network Speed Call. Assign an NCOS to the TIE trunk that will give them access to the NSC list(s) the users at the remote site need.

NARS and BARS

NCOS – Automatic Redial parameters

If a user dials a public network number and hears a busy signal, Automatic Redial allows Ring Again to be used to redial the call. The system will redial repeatedly until it receives an answer or until it reaches a programmed limit of retry attempts.

The NCOS determines whether the user’s retries can use Iset routes only or if any route in the route list can be used.

You may decide that if a low level user can only scan Iset routes when they first make a call that the retries will also be restricted to Iset routes. Or you could decide that retries can scan all routes to make it less possible that route blockage would delay the retries even further.

NCOS – Maximum Precedence Level

This capability is only used in Autovon (proprietary military) networks. If you use Autovon software, refer to the documentation that you have for further details.

NCOS – Equal Access

When a system with NARS is to have Equal Access to Common Carriers (U.S. only) you may want to restrict certain NCOS groups from using the Equal Access capability.

Users who dial AC1or AC2 plus a Carrier Access Code 10 or 101 (refer to Carrier Access Codes on page 179) will be routed to a route list that has an entry for the Carrier route. If you want a user group to be restricted, allow Equal Access Restrictions in the NCOS. The default setting is Equal Access Restrictions denied.

NARS and BARS

Example of an NCOS Summary

It is a good idea to have a written description of your NCOS groups kept on site, to be used for reference purposes. A written summary helps network administrators and programmers understand the overall capabilities of each NCOS quickly. They don't have to print out the NCOS programming and the route list programming to figure out the initial design.

Refer to the following example of an NCOS summary that could be used at a site that has NARS and Network Speed Call software.

NARS and BARS

NCOS SUMMARY

NCOS 1

- ALLOWED LOCAL CALLS AND LOCATION CODE CALLS ONLY
- SCAN INEXPENSIVE ROUTES ONLY
- QUEUE FOR INEXPENSIVE ROUTES ONLY
- EXPENSIVE ROUTE WARNING TONE OFF
- NO EQUAL ACCESS
- NO NETWORK SPEED CALL ACCESS

NCOS 3

- ALLOWED LOCAL CALLS , LOCATION CODE CALLS, AND LONG DISTANCE CALLS TO CUSTOMERS ONLY
- SCAN INEXPENSIVE ROUTES ONLY
- •QUEUE FOR INEXPENSIVE ROUTES ONLY
- EXPENSIVE ROUTE WARNING TONE OFF
- EQUAL ACCESS
- NETWORK SPEED CALL ACCESS

NCOS 5

- ALLOWED LOCAL CALLS , LOCATION CODE CALLS, AND LONG DISTANCE CALLS WITHIN NORTH AMERICA ONLY
- SCAN INEXPENSIVE ROUTES ONLY
- QUEUE FOR INEXPENSIVE ROUTES FOR 30 SECONDS AND THEN ALL ROUTES
- EXPENSIVE ROUTE WARNING TONE OFF
- EQUAL ACCESS
- NETWORK SPEED CALL ACCESS

NCOS 7

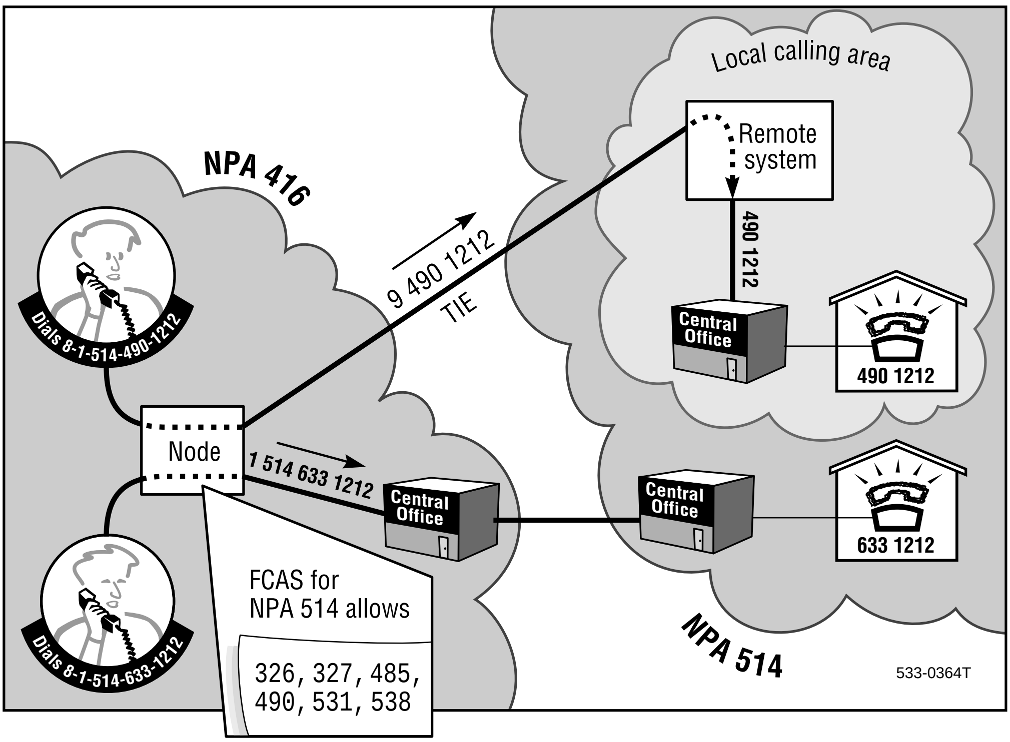
- ALLOWED LOCAL CALLS , LOCATION CODE CALLS, AND LONG DISTANCE CALLS ANYWHERE
- SCAN ALL ROUTES
- QUEUE FOR INEXPENSIVE ROUTES FOR 30 SECONDS AND THEN ALL ROUTES
- EXPENSIVE ROUTE WARNING TONE OFF
- EQUAL ACCESS
- NETWORK SPEED CALL ACCESS

Free Calling Area Screening

NARS translation tables are programmed to point each area code to the route list that will handle the calls properly. In North America, when you dial an area code (NPA), it is always followed by an exchange code.

You may want the system to only allow certain exchange codes to go out on a certain route in the route list for that NPA. Or you may want to deny certain exchange codes from going out on a certain route. You use Free Calling Area Screening Indexes (FCIs) to do this.

Figure 14 - Free Calling Area Screening



As in the preceding illustration, you may have a switch in your network located in a certain NPA. You may have TIE trunks from a node to that switch. When you program NARS at the node, if you have enough TIE trunks to carry the traffic, you may want to send calls to that NPA on the TIE trunks.

However, you may only want to send calls with exchange codes that are in the free calling area of that switch. To do this, you would program an FCI for the NPA in question, and program up to 800 exchange codes that are free, as allowed. You would then apply that FCI to the TIE trunk route entry in the route list for that NPA. Only those codes would be allowed to go out on the TIE trunk route. Other exchange codes would be sent to other routes in the route list.

If you are using Flexible Numbering Plan software you will be able to screen special numbers for allowed and denied digits that follow. This capability is very useful in non-North American markets where there are no NPAs. Refer to the *Flexible Numbering Plan* section (Free Special Number Screening) for further information.

Customer-wide NARS (BARS) parameters

There is programming that must be done that will affect the overall NARS (or BARS) operation for a particular Customer Group. It involves the following:

- setting aside memory for such things as:
  - route lists
  - digit manipulation indexes
  - Free Calling Area Screening Indexes
  - location codes
  - Supplemental Digit Restriction blocks
  - Incoming Trunk Group Exclusion tables
  - CDP steering codes (if CDP is installed)
  - Special Common Carrier entries
  - Free Special Number Screening tables

  Note: Choose maximum numbers that will accommodate future growth so you do not get warning messages about parameters being exceeded when you want to add new route lists and codes to be translated later on.

- programming AC1
- programming AC2 (optional)
- activating dialtone to be heard after AC1/AC2
- activating Expensive Route Warning Tone and the Expensive Route Delay timer
- programming the time-of-day schedules
- enabling the Routing Control feature – programming the map of NCOSs and the extended time-of-day schedule when Routing Control is to be used
- activating the ESN software to look at a user's TGAR (as well as the FRL) and use it to allow/restrict ESN calls on routes in route lists. Refer to the information on *NARS bypass control and TGAR* that follows.

ESN bypass control and TGAR

A system *without* BARS, NARS or CDP installed uses the Trunk Group Access Restriction (TGAR) code programmed for the calling station to determine if a user can access a route.

Users dial trunk route access codes in order to make calls on these non-ESN systems.

The way TGAR and TARG work is: when a user dials a trunk route access code, if the TGAR code programmed for the caller's telephone matches one of the Trunk Access Restriction Group (TARG) codes programmed on the trunk route being accessed, the call is blocked.

As of Release 22 the default for both TGAR and TARG is 1. Therefore users are blocked from direct access to routes by default.

The default class of service is Conditionally Toll Denied (CTD). This will block the user from dialing toll calls (calls beginning with 1 or 0) on direct access calls. It will allow ESN toll calls that the user's NCOS permits.

> ***Note:*** There is an Intercept treatment for “Calls from a restricted station”. It is given to a user who dials any long distance call from a toll denied (TLD) telephone or when a user dials a toll call using a trunk route access code from a telephone that is Conditionally Toll Denied (CTD)

Previous to Release 22 the default for TGAR was 0 and routes had no TARG programmed by default. TGAR 0 is unrestricted, by definition. This meant that users had access to all routes by default.

The default class of service was Unrestricted (UNR). This allows all calls.

A system *with* BARS, NARS or CDP installed uses the FRL programmed in the NCOS of the calling station to determine if the user can access a route when the user dials AC1 or AC2 or CDP steering codes in order to make calls.

With ESN, the digits are translated and sent to route lists. The FRL for each route is compared to the FRL of the caller. If the caller’s FRL is equal to or greater then the FRL required on the route, then the call can be sent out on that route.

However, users can bypass the route lists by dialing direct trunk route access codes. They will be able to call out on the trunk routes of their choice unless they are stopped by the TGAR and TARG restriction. If you do not implement this restriction, it will probably cost you money.

> ***Note:*** Users will see the trunk route access codes on the displays of their telephones. Some people will try to dial these codes when they make calls, if they are not prevented.

Preventing bypass

It is strongly recommended that you prevent users from direct access calls, thereby bypassing ESN and all the controls it offers. Use TGARs and TARGs to prevent bypass.

Make sure the caller’s TGAR matches the TARG on routes to which you do not want them to have direct access.

Do not forget that TIE trunks and other non-telephone terminals (such as Authorization Codes) can have TGARs assigned. Include them in your TGAR programming plan.

If you are not going to program AC2 and if users will dial a trunk route access code for the COT route, make sure that there is no TARG programmed on the COT route. Otherwise people will not be able to make local calls from the switch.

You may want selected users such as executives, network administrators and planners, and technicians to have direct access to routes for testing purposes or to place calls in emergencies. Some examples are:

Attendants always have direct access to routes since you cannot assign a TGAR code to a console.

Activating TGAR for ESN calls

Optionally, you can have an ESN system look at the user's TGAR, in addition to the FRL, for ESN calls.

This capability was designed for networks where there are two different departments of users within one Customer Group. Each department requires their own special service trunk route (FEX, for example). Each department wishes to pay for their own route themselves and ensure that only their users call out on it. They want to put both FEX routes in the same route lists, so users can use the UDP to make calls.

It was not possible to restrict users to their routes only using FRL only.

Instead, if you activate TGAR on ESN calls, and you assign the same FRL to both groups but a TGAR that matches the TARG of the other group's FEX, it will ensure that calls do not overflow to the wrong FEX.

The drawback to using TGAR with ESN calls is that if you use TGAR to block certain ESN calls, you would have to leave most routes without TARGs so ESN calls could be made. Therefore, you cannot also use TGAR to prevent bypass.This is a substantial security and control measure to give up.

NARS and BARS

Expensive Route Warning Tone

If any NCOS group is going to have the tone activated, you must activate it for your Customer Group first.

The customer-wide Expensive Route Delay Timer determines how long the system will wait for the users to decide whether they will hang up or queue after hearing the tone.

If users are going to hang on most of the time and go out on the expensive route anyway, you may not want to activate the tone in the first place, since this will slow down call processing.

If you still want them to hear the tone because you will bill them extra for expensive calls, and if they will queue sometimes, or hang up sometimes, keep the timer short. The shortest timer is 2 seconds. The longest you can make it is 10 seconds.

Network Traffic studies and user surveys can help you get the feedback you need to adjust the timer after installation.

Time-of-day schedules

You can plan for up to eight schedules.They are used in the route lists to turn routes on and off. This way routes are only available to users when they are the most effective and when you want them to be used.

Routing Control (RTC)

The Routing Control feature provides a mechanism for changing a user’s NCOS. You program a “map” of instructions that tells the system what each NCOS becomes when the feature is active. You can have an NCOS become a higher or lower number or stay the same. You must specify what is to happen to each one. If you do not specify, an NCOS will become NCOS 0 by default when RTC is active. It is a good idea to print out the programmed data, to verify the mapping that is programmed.

RTC can be activated three different ways:

- when time-of-day schedule 7 is in effect
- during certain, specified days of the week (called an extended time of day)
- or when the attendant presses a Routing Control (RTC) key, one of the programmable feature keys of the console

TODS 7

If you want user's NCOSs to change during a certain period every day, program the time period in schedule 7. For example, if users are supposed to leave the building at 5:00 pm every day and not return until 7:00 am, you can change the NCOSs to 0 during that period every day. The route lists must be programmed so that NCOS 0 will not have a high enough FRL to access any routes. Make sure however, that NCOS 0 is still a high enough level for users to reach emergency numbers.

Extended TOD (ETOD)

If you do not want users to call out on weekends, specify those days as active RTC days. All day (24 hours) Saturday and Sunday, the feature will automatically be activated.

Routing Control (RTC) key

If you cannot be certain of the times when you will want the feature to be activated automatically, then assign an RTC key to the console. Someone can activate/deactivate the feature when it is appropriate.

> ***Note:*** Authorization Codes can be used to override the restrictions imposed through Routing Control. A user with an authorization code can go to any telephone and make a call. If a user enters a valid authorization code while Routing Control is in effect, the NCOS number associated with the auth code is applied for the duration of the call. The NCOS assigned to the auth code is not affected by the Routing Control feature (this is the design intent). Calls can be billed to the auth code dialed, if you are using Call Detail Recording (CDR).

Using Routing Control as emergency bypass

This feature can provide a way of re-routing calls when certain trunks are not working temporarily.

For example, if the TIE trunks to remote sites are not working, the attendant can press the Trunk Group Busy key(s) associated with the route(s) to make them appear busy for outgoing calls. Then, if Routing Control is programmed to raise the NCOSs of the users, the RTC key can be pressed. This allows users to access alternate routes temporarily, now that they have a higher NCOS. When the emergency is over, the TGB key(s) can be pressed to make the trunks idle again, and the RTC key can be pressed to return the NCOSs to normal.

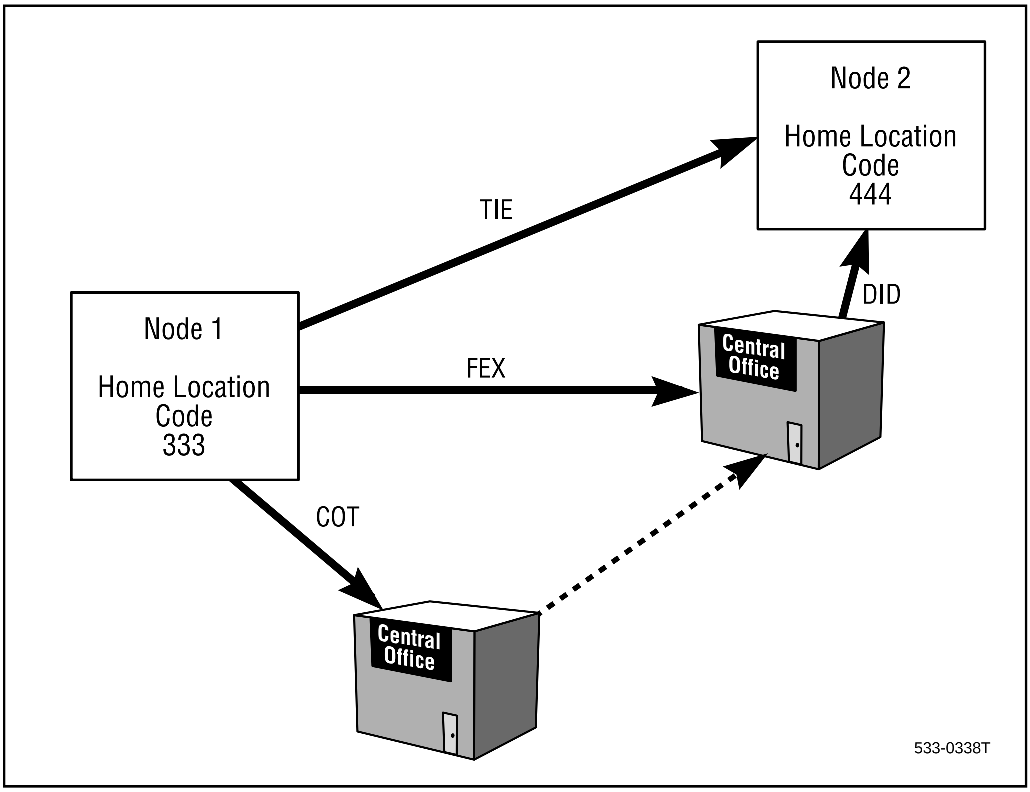
Flexibility of NARS and BARS

The information on the previous pages in this section covers the basic configuration of NARS. You have read about some of the decisions and the planning involved in implementing NARS as effectively as possible for your needs.

Refer to *Improving feature performance* for information on some of the more advanced aspects to NARS that have not been presented so far in this section.

A simple example follows to further illustrate, as a review, the kinds of decisions you can make before and after you install NARS (or BARS). These decisions will have an impact on the programming at the site.

Figure 15 - Two node network



If you look at the previous illustration you can see a simple example of a two switch network. For discussion purposes we'll say that the network planner decided to use the routes at node 1 in the following way for on-net (location code) calls to node 2:

- TIE trunks – used all day, all users can access, not expensive
- FEX trunks – used all day, all users can access, not expensive
- COT trunks – used all day, all users can access, not expensive

The planner also decided that all users can scan all route choices when they make calls to node 2.

NARS and BARS

Here are some examples of what another network planner could do differently:

- remove the FEX trunks as a choice in this route list.

  If there are only enough FEX trunks for off-net calls to that area code, allowing on-net calls to use the FEX route would block off-net calls.

- turn off the TIE trunk route entries in all route lists at the node during peak busy hours.

  That way users at node 2 will not be blocked when they try to make calls into the node at those times. The users at the node have alternate route choices to use for overflow at the peak busy hours.

  Doing this can save money since the cost of allowing overflow at node 1 to the FEX and COT trunks during busy hours might not be as much as the cost of installing additional TIE trunks to handle the peak traffic loads.

- remove the COT route choice. They do not want to use the COT trunks for on-net calls.

  Long distance bills will be reduced.

  There would have to be enough TIEs and FEXs for the expected traffic. If not, users will get blocked or they will need to queue when trunks are busy.

- allow only certain users to access certain choices. For example, only users who are FRL 3 or higher can use the FEX trunk route for these calls. Only users who are FRL 5 or higher can use the COT trunk route for these calls.

  Long distance bills will be reduced and fewer FEX and COT trunks will be needed.

- program the COT trunk route in this route list as expensive. Users with CBQ type “A” and Expensive Route Warning Tone activated in their NCOS will hear the tone when they are going to access a COT trunk.

  Users can decide to hang up and try the call later or to queue for the cheaper routes. This can save money.

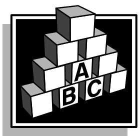
- program the FEX and COT trunk routes in this route list as expensive. Users with CBQ type “A” and Expensive Route Warning Tone activated in their NCOS will hear Expensive Route Warning Tone when they are going to access an FEX or COT trunk.

  Users may hang up and try the call later or queue for the cheaper route. This can save money.

- restrict which trunk routes are scanned when calls are made. Program the Iset as 1. CBQ type “I” users will only scan the TIE trunks when they make location code calls. FRL 3 and 5 users will scan all routes.

  Long distance bills will be reduced and fewer FEX and COT trunks will be needed.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

How to optimize the use of trunks (Supplemental Digit Recognition)

Incoming Trunk Group Exclusion (ITGE)

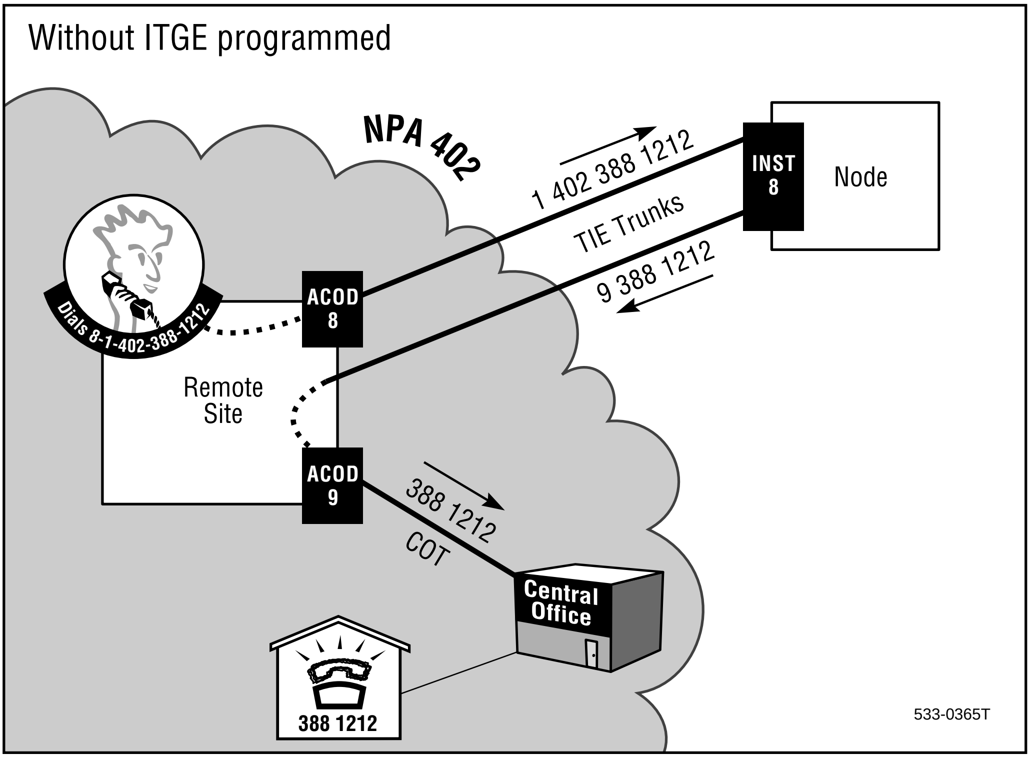
If the user at a remote switch dials digits that have been denied following a translated code, the node will block the call. The denied digits are referenced to one or more incoming trunk routes. If the call originates from one of these routes, ITGE denial will occur. As a result:

- A caller cannot loopback through the node to the home switch wasting TIE trunks. For example, a user at a remote switch might try to call a local exchange in his own area code by dialing the call as: AC1 + 1NPA + NXXXXXX

  Refer to the illustration that follows.

NARS and BARS

Figure 16 - Example of the need for ITGE

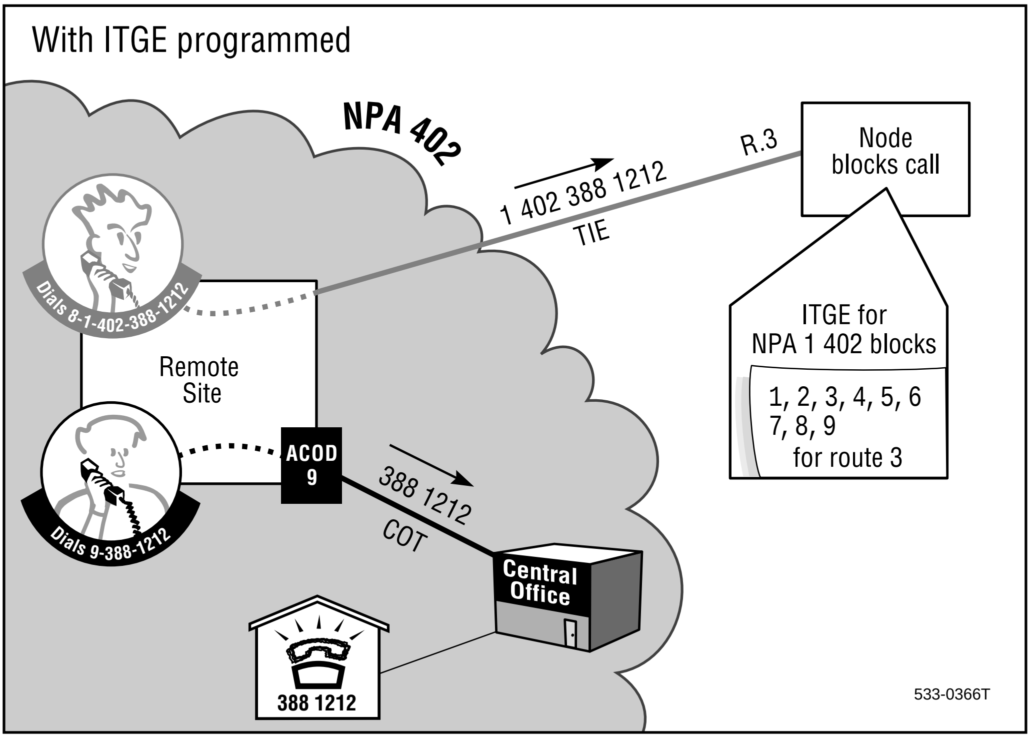


Since the access code to the TIE route is the same as AC1 at the node, the call goes to the node.

Without ITGE programmed, the node would route the NPA call back to the remote switch on a second TIE trunk and out on a COT trunk at the remote switch. The call has tied up two TIE trunks and a COT trunk. The user should have called out directly on a COT trunk at his own switch.

When ITGE is implemented, the user will encounter blockage when he dials his call (all digits after that NPA will be blocked for that TIE trunk route) and he will learn to dial properly (9-NXX-XXXX). TIE trunks will not be used for these improperly dialed calls any more.

Figure 17 - Example of ITGE working



- A user at a remote switch might try to call their own location code by dialing the call in the following format:

  TIE trunk route access code + LOC + XXXX

  If AC1 is inserted by the node on the incoming TIE trunk, NARS at the node translates the call.

  Without ITGE programmed, the node sends the location code call back to the remote switch on a second TIE trunk. The DN at the remote switch rings. Two TIE trunks are used needlessly.

  When ITGE is implemented, the user encounters blockage from the node when they dial the call (that LOC code is identified as excluded for that incoming TIE trunk route). The users learn to dial internal calls properly (DN digits only). TIE trunks are not used for these improperly dialed calls any more.

  Refer to page 181 for more information.

NARS and BARS

You can use Incoming Trunk Group Exclusion in the following ways:

- to block users at remote switches from making calls through the node to NPAs, NXXs, SPNs, or LOCs that they are not permitted to call from their own switch.
- to block users at remote switches from making calls through the node to NPAs, NXXs, SPNs, or LOCs that they should have dialed using trunks at their own switch.

  It is important to train users at remote switches to dial certain calls using their own trunks, if it is less expensive than using trunks at the node. If they cannot learn to dial calls properly, consider installing BARS or NARS at the remote site to route the calls effectively for them.

Off-Net Number Recognition (Supplemental Digit Recognition)

Network users often dial on-net calls in an off-net format.

In other words, they dial:

AC1 + NPA + NXX- XXXX

instead of:

AC1 + LOC +XXXX

This is usually a result of lack of training on how to use the UDP on-net dialing plan.

Make sure nodes have recognition programmed to deal with these misdialed calls. If you do not implement Recognition, it can result in the need for extra trunks or excess blockage on trunks.

If a node is the last intelligent (NARS or BARS or CDP-equipped) switch as a call travels through a network it should be programmed to recognize two different types of misdialed calls:

- calls that are meant for its own users (Local Recognition)
- calls that are meant for users at the next switch in the network (Remote Recognition)

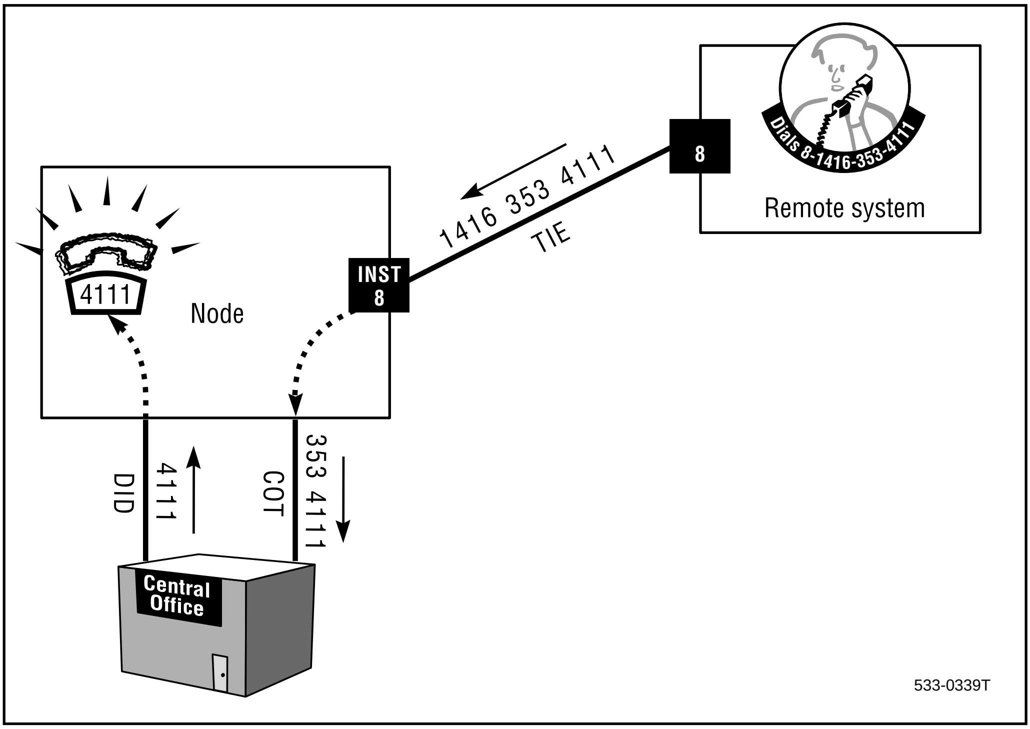
Local and Remote Recognition can identify two different types of DNs at the local or remote switch: DID and non-DID (called DDD in programming). Tell your programmers what the DID and non-DID DN ranges are at all the locations in your network so they can deal with recognized calls properly.

What recognition does

A local recognized DID call does not go out of the node and back in on a DID trunk. The call will be directed to an internal DN immediately.

A local recognized DDD call does not go out of the node and back in on a COT trunk to the attendant. The call will be directed to an attendant or internal DN immediately.

Figure 18- Example of the need for Local Recognition

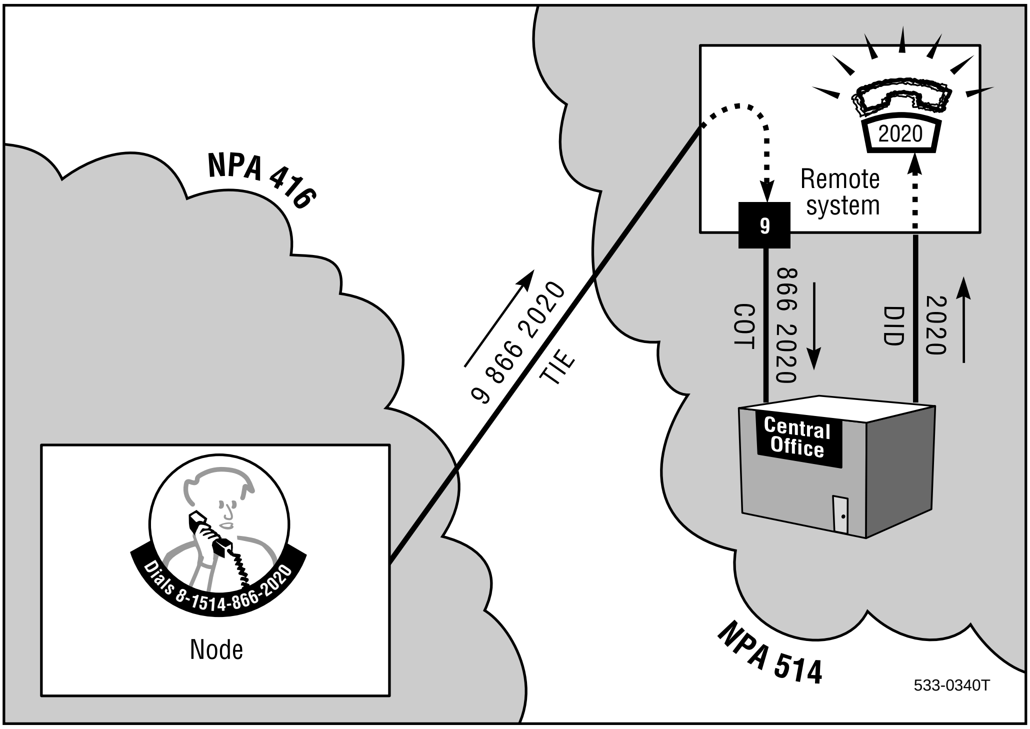


NARS and BARS

A remote recognized DID call does not go out of the remote switch and back in on a DID trunk. The call will be directed to an internal DN at the remote switch immediately.

A remote recognized DDD call does not go out of the remote switch and back in on a COT trunk to the attendant. The call will be directed to an attendant or internal DN at the remote switch immediately.

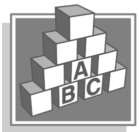
Figure 19 - Example of Remote Recognition working



Note: There is a limit to the number of entries you can have for recognized and restricted numbers for each translated code. This is important if you do not have all the DID numbers in a particular range at a given site. Input, as allowed codes, the numbers that do not belong to your DID range. Calls to these numbers are routed normally. Then you program only the first digits of the recognized DID numbers and they are handled differently. That way you do not have to program many entries for all the DID numbers around the numbers that do not belong to your site. In this way you can save programming time and memory.

How to make call processing faster

Fast Tone and Digit Switch (FTDS)



The FTDS software and hardware reduces call set up time. The software can load up to 32 digits into the buffer at a time and can outpulse digits at a maximum rate of 10 digits per second. The length of the Dual-tone Multifrequency (DTMF) bursts can be set at 100 ms or 50 ms.

NARS inter-digit timer

The NARS inter-digit timer (NIT) is programmed on a Customer Group basis. It determines the length of time the system will wait after the user dials the last digit in the call before it will send it out on a trunk.

The timer is set at 8 seconds by default. You can shorten it, if you want calls to be processed faster. The minimum setting is 2 seconds.

Be aware that if your users pause while making calls, the system will assume they are finished dialing, too soon, if the timer is set too short.

Dialing octothorpe (#)

Users may learn that calls go out faster if they dial a # (octothorpe) at the end of the call. This manually forces the NIT timer to expire.

There is a software patch on most systems installed today that prevents users from dialing a # before all the digits in a call have been dialed. Network managers were complaining that CDR records were incomplete when users dialed # midway through a call.This made correct billing impossible.

The patch intercepts calls if they are not completely dialed before # is dialed.

If you find CDR records that show partial numbers ended with #, ask your system supplier to install the patch on your system.

NARS and BARS

Flexible Numbering Plan

If you use FNP in North America you need to know that SPNs 0, 00, 01, 011, 411, 611, 911, 800, and 1800 are affected by a prompt (INPL) in the translation tables. The default response for this prompt is NO.

A NO response means that call termination occurs immediately after the SPN is entered. This is good for 911 calls.

No further digits are outpulsed, even if they are dialed. This would be bad for overseas (011) calls.

Arrange to have the response programmed as YES if more digits will be dialed after the SPN and you want them to be outpulsed after the number is dialed or when the NIT times out.

How to provide access to emergency services

The Emergency Services Access (ESA) feature allows the Meridian 1 to provide FCC compliant Private 911 service to users. The following functions are part of this service:

- ESA Call Recognition – when users dial a defined internal ESA DN, the system recognizes the call as an emergency call. You usually choose the same number for the DN that people dial at home in an emergency (in North America, this number is usually 911). The DN can be a maximum of four digits in length. Ensure there is no conflict with other numbers on the system.

  If the dialing plan you are planning has AC1 or AC2 preceding the ESA DN, you must translate the DN as an SPN code that is handled by a route list index entry programmed as a Local Terminating entry. Refer to Local Termination on page 192.

- ESA Calling Terminal Identification – you program a CLID for each telephone that will be sent out with the emergency call, even though ISDN software may not be present. Non-emergency calls will not send a CLID unless ISDN is set up. If the call must travel across a trunk to a Meridian 1 system before it goes to the emergency number, the calling line ID is based on that assigned to the incoming trunk route.

- ESA Call Routing – emergency calls are routed to a specially designated route. Calls can overflow to other routes, if STEP programming is done on the route. Restrictions applied to the telephone are ignored when ESA calls are made.

  If the route is a CAMA route (recommended by the FCC), special digit manipulation may be used, if required, to outpulse the correct digits for the user.

  If the originator is calling across a trunk into the Meridian 1 and ESA_SUPP software is not equipped, the ESA calling number is the one composed for the incoming trunk route. If ESA_SUPP is equipped, and a valid ANI/CLID number for the caller is passed in on the incoming trunk, the ESA feature sends the ANI/CLID number on the outgoing trunk. If there is no ANI/CLID number on the incoming trunk or the number is invalid, the ESA number programmed for the incoming trunk route is sent out.

- On Site Notification – when an emergency call is being made, notification can be given on-site to a Meridian Modular telephone and/or a TTY device. On the TTY, an OSN call record prints out for people to use to identify who is calling. The call comes into a particular DN on the OSN telephone and the caller can be identified on the display of the telephone. The DN must be a single appearance DN.

The Emergency Services Access (ESA) software package provides ESA Call Recognition, ESA Call Termination Identification, and ESA Call Routing. It also provides basic number translation, such as the addition of NXX and NPA (area and office) codes, but not non-DID to DID translation.

The ESA_SUPP package provides supplementary capabilities, such as networking capabilities (for example, incoming ISDN to CAMA number conversions), and On Site Notification.

The ESA_CLMP package provides translation of non-DID to DID numbers.

Discuss your needs with your system supplier. It is important that you tailor your software and programming to your particular configuration.

NARS and BARS

How to optimize queuing

For users

Users who encounter busy trunks may not understand that overflow tone indicates that trunks are busy. If you have enabled Callback Queuing they can use it at that point.

Many Network Traffic studies show that even though users were offered queuing they did not use it, especially when NARS or BARS or CDP are first installed.

Proper training can help users take advantage of queuing instead of repeatedly dialing the same call.

You can replace overflow tone with a Recorded Announcement. The announcement can tell users what to do. This cuts down on the training you will need to do, user complaints and possibly the need for more trunks if users become comfortable with queuing.

Set up your network so that queuing times are guaranteed to be short. Users will accept queuing if they know a trunk will be offered to them very quickly and if it is easy to do.

You can do this by:

- monitoring the routes and route lists (do Network Traffic studies) to find out where blockage is occurring
- finding out if there are trunks that need repair
- adding more trunks to maintain an acceptable grade of service
- allowing users to access more routes when they make calls or when they queue
- implementing Off-Hook Queuing since it is more user friendly than Callback Queuing

For the network

When a trunk has been found for a user by the Callback Queuing feature, digits are slowly outpulsed on the trunk to indicate that the trunk is in use.

These calls will show up in CDR records, even if the user does not answer the callback call.

Users ignore callbacks if it has taken too long for the system to find a trunk. Train users to activate queuing only if they intend to answer the callback when a trunk is found. Do what you can to help them by ensuring short queuing times.

Unanswered callbacks add needless additional usage to trunks. To minimize the extra trunk usage, keep the callback timer short. (Two rings should be sufficient. That takes 12 seconds in North America.) The maximum timer is 30 seconds. That is probably longer than necessary for most networks.

Control tips

There are many system and customer-related features and capabilities that you can use to improve the control you have of your users and the network.

Non-ESN control-related features

TGAR and TARG

- Refer to page 209 for the background information you will need regarding the use of TGAR and TARG**.**

Class of service

- You can assign restricted classes of service to users just as you would on a standalone system. Refer to page 169 for more information on the CTD class of service recommended for most users on ESN networks. You can also use more restricted classes of service. Refer to the section of the *Software Features Guide* called Access Restrictions for more information.

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Call Forward External Deny

- Users sometimes Call Forward their telephones to trunk route access codes or ESN numbers before they go home at the end of a shift. From home they can dial into their telephone (using the DID number). The telephone forwards them to a trunk or a destination. They may finish dialing a call from home if the office telephone is only forwarded to an access code.

  The call adds extra usage to your trunks.

  The bottom line is that the long distance charges are billed to you, not the user doing the dialing.

  These issues can pose big problems, especially if the DID number is given to other people to use.

  The Call Forward External Deny feature prevents this form of abuse.

Timed Forced Disconnect Timer

- If you want to set a time limit to long conversations, you can implement a Timed Forced Disconnect timer on each route. Any conversation that reaches that timer threshold will be disconnected instantly.

Control of incoming calls on TIE trunks

- Controlling calls coming in from TIE trunks can be one of the biggest issues that network planners and administrators face. It is difficult to trace calls that are being made from remote sites since the CDR records at the node only indicate that the TIE trunk was the originating party. The remote site CDR records indicate the DN of the caller and the TIE trunk that was used to call the node. The problem is that the timer stamps from the two switches are rarely synchronized. It is next to impossible to match the call records to find out who made each call.

NARS and BARS

To address this, consider implementing one of the following:

– ISDN and CLID in CDR (CCDR) software. A major benefit of ISDN is that the user's Calling Line ID is sent with each call on the TIE trunk. If you have CCDR software, this will print out in the CDR at the node. You can bill the user for the calls and discuss the user's calling habits.

– programming TGAR, Class of Service and NCOS restrictions on the incoming TIE trunks that allow the remote users to make only the calls that you want then to make

– implementing Incoming Trunk Group Exclusion Indexes to block certain numbers or entire blocks of numbers

Controlled Class of Service and Electronic Lock

◆ If you do not want to use the Routing Control feature to raise or lower the NCOS groups on your system, you can use the Controlled Class of Service or Electronic Lock features instead.

Controlled Class of Service allows a user to change the Class of Service (COS) of a DN to a different pre-programmed customer-wide level.

Electronic Lock allows users to change their own Class of Service to prevent other people from making unauthorized calls from their telephones when they are not there. It is a password protected feature. To release the restriction a password must be dialed.

Scheduled Access Restrictions

This software allows you to define TGAR, COS, and NCOS restrictions for different hours and days. Each combination of TGAR, COS and NCOS is given a SAR group number. You assign a SAR group number to each telephone and route. At different times and days, the restrictions can change based on the needs of the users, giving you better control. Refer to the NTPs that cover regular system features for more information.

New Flexible Code Restriction

◆ New Flexible Code Restriction (NFCR) can be used to block calls that users make using direct trunk access codes. If you do not intend to program AC2 and users will dial local calls using the

NARS and BARS

direct COT trunk route access code, you might want to use NFCR tables to restrict certain users from certain calls. Refer to the NTPs for more information.

ESN control-related features

Translation

- Users, especially untrained ones, will try to dial calls many different ways. Decide how tolerant you want your database to be of different dialing choices.

  For example, will you permit users to dial AC1 or AC2 for area code calls? If so, you must program all area codes into both translation tables. You may have a higher implementation cost because of this but it may reduce the training you will need to do and it may make users happier.

  If you want users to be able to dial calls without access codes you will need to install Coordinated Dialing Plan (CDP) software and translate steering codes. Refer to the *Coordinated Dialing Plan (CDP)* section in this book for more information.

- Do not translate numbers that you do not want users to dial (1900 calls in North America, for example). Be aware that users who are Unrestricted and not TGAR-restricted will be able to reach these numbers using direct trunk access codes.

Supplemental Digit Restriction (SDR)

When you translate digits you can block certain digits that follow the translated code. For example, if you do not want your employees calling a competitor, you will translate the area code or special number or NXX at the beginning of the number but deny the specific digits that belong to the competitor's telephone number. Users can call every number in the area code, special number or NXX except that one.

You can use SDR to deny certain blocks of numbers. For example, you can deny a particular NXX, or part thereof, in a translated area code. Calls to every other number that does not contain the denied digits will be allowed.

Users cannot bypass this restriction, if they dial calls using AC1 or AC2. Bypass can occur only if a user has no TGAR and class of service restrictions and therefore has direct, unlimited access to trunks.

NCOS

- Restrict your lowest level users to inexpensive routes, even when they queue, by assigning a low FRL, CBQ type I, and an Extended Route Advance Timer set to 0.
- Everyone queues for the Iset routes initially (minimum 30 seconds). To control network costs, lengthen the time users spend queuing on the Iset to the maximum they will tolerate before you let them queue for extended set routes.
- NCOS 0 is default. If you want to make sure that any new telephone is restricted, when installed with the default NCOS, leave NCOS 0 programmed with an FRL 0. If you want these telephones to have some access to the network, program NCOS 0 with an FRL that will give them the required access.
- Never make assumptions about an NCOS until you have printed out the NCOS programming to be sure. NCOS 0 may seem to be the lowest level but it can have FRL 7. Also, you must check the route lists since FRL 0 might be high enough to call everywhere.
- You can control the calls that users are able to make when they are calling out from a Meridian Mail mailbox using the Thru-dialing feature. The Meridian Mail system is connected to the Meridian 1 using ACD queues. Each queue has an assigned NCOS that will control the calls that can be made when a call comes back to the Meridian 1 from Meridian Mail.

Route lists

- Expensive routes - It has become less common to implement Expensive Route Warning Tone today than it was in the past.You do not have to designate any routes as expensive routes. The tone is only marginally effective in many cases. Network administrators have found that users who are untrained or impatient hang on after

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hearing the tone. As a result, their call goes out on the expensive route and hearing the tone only served to delay the call for a few seconds.

If you plan to implement ERWT as a way of controlling expenses, plan to do proper user training and follow up.

Extra billing for expensive calls encourages users to pay attention to the tone. CDR records show an “E” before the digits dialed to indicate that the user heard the tone and went out on an expensive trunk.

Routing Control

- You can change the NCOSs at certain times of the day, on certain days of the week or when a key on the attendant console is pressed. You can use this feature to prevent users from making certain types of calls or using certain routes during off-hours or at times when the calls are expensive. Refer to the information on Routing Control, starting on page212.

Administration tips

- If you publish network-wide directories and a network map, this helps users understand the Uniform Dialing Plan. This will save you money since you can publish one directory for all sites. All users will be able to look up numbers in that directory to call all other network users.
- As a planner you may be tempted to install BARS software at nodes since it may be less expensive than NARS software.

  BARS was designed to perform automatic, least-cost routing for a stand alone switch, not to operate at a node in a multi-switch network. Therefore, BARS is not supported in a node configuration.

  This is an important issue when you want to upgrade an ESN network to an ISDN network. BARS may have supported your ESN needs but you will have to upgrade your BARS switch to a NARS node. Some ESN programming changes will be required. For example, it will work best if you translate location codes and a home location code. This relates to features such as Calling Line ID. For more information, refer to the section called *CLID and Name Display options*.

When you upgrade from BARS to NARS your CDR format will change. The digits dialed field will appear in a different format. If you do not want to adjust to a new format, ask your supplier to install both BARS and NARS software on your system. Your CDR records will remain as they were with BARS. (In Canada that software combination has a special name. It is called BARS/NARS.)

- Thoroughly test the NARS programming before you allow users to make calls on route lists. Use temporary AC1/AC2 codes for testing purposes. You should do this testing after business hours because you will have to make routes busy and this will interfere with users' calls. Do not tell users about these testing access codes.
  - Try calls to all route lists on all possible route choices. Use the Trunk Group Busy keys on the attendant console to make routes busy to prove alternate routing works.
  - Use test telephones with different NCOS values to verify that FRL restrictions work.
  - Change the system time to verify time-of-day programming on trunk routes work properly.
  - Dial calls with allowed and denied numbers to prove screening and restriction is working properly.

  Note: You will not be able to test queuing for trunk routes that have been made busy with Trunk Group Busy keys. Make the routes busy with real calls on hold, if possible.

There is more information on how to test your programming in Appendix 6.

When ready, reassign the AC1/AC2 codes to the proper ones you have chosen for your dialing plan. Inform users when they can begin to dial calls using these access codes.

Translation

- It is possible to use a two digit AC1 and AC2, but it is not recommended, unless you are running out of numbers or if you cannot find one digit numbers that are available at all sites.
- Verify with your system supplier how they plan to keep your translation tables and Screening tables up to date. Discuss how long it will take to add new codes to your database. If they will allow someone on your staff to make these programming changes, arrange training for that person.
- In North America contact the Center for Communications Management Information (1-800-929-4824 extension 2835) to subscribe to services they offer to keep you up-to date on new exchange codes to be added to the public network.
- At installation, it is a good idea to program the Intercept treatment for invalid numbers to send calls to the attendant instead of overflow tone. This way the attendant can report any numbers that are missing from translation tables. Prepare the attendants for this role beforehand.

  You may also consider changing the intercept to a Recorded Announcement (RAN) later on. The recorded announcement can tell users to contact their network administrator if they dial a number that the system cannot translate.
- If you want to study the traffic that goes to a certain destination, make sure the code points it to its own route list index in the translation tables. The Network Traffic study TFN001 will print data about the calls that were routed to that route list.

Translating an HNPA with NARS

1. If you decide to program the area code in which the node sits as a Home NPA:

- you must activate AC2, even though users may never dial it in your particular dialing plan (because the system automatically removes the HNPA from the digits string and inserts the AC2 code in front of the remaining digits). If users will not dial AC2, assign a two or more digit, non-conflicting code, used for translation purposes only. This helps to save one digit codes for other things in your numbering plan.

- you must translate all NXX codes that follow that inserted AC2 code and send them to the correct route lists. Some of the NXXs are local ones; others are long distance. Send each NXX to the proper route list index.
- doing this can add extra programming to your work at installation, compared to programming the area code as a normal NPA. Translating an HNPA requires ongoing administration to keep the NXX translation tables up-to-date, if new NXXs are added in that NPA.

2. If you choose to program the area code as an NPA instead of an HNPA, the programmer has two choices for dealing with the local and long distance exchanges in that area code:

a translate the area code followed by each local exchange code in that area code as a six digit NPA and send each one to a local call route list. Translate the long distance exchanges the same way and send them to one or more suitable route lists.

For example, consider a node that sits in area code 506. If 685, 686 and 687 are local exchanges in the 506 area code, you can translate the codes: 506 685, and 506 686, and 506 687 as NPAs. Send these calls to a route list for local calls.

If 434, 435, and 436 are long distance exchanges, then translate 506 434, and 506 435, and 506 436 as NPAs and send these codes to a route list suitable for long distance calls.

b translate 506 as an NPA and send all calls to the same route list. Assign a Free Calling Area Screening (FCAS) table to the first choice route which is designed to allow only local calls. This screening table must contain a list of all local exchanges. It looks for a match between the dialed exchange and an exchange allowed in the FCAS table assigned to the route list entry. If the exchange dialed matches one of the allowed exchanges in the FCAS table, the call goes out as a local call. If the exchange dialed is a long distance one, it is not allowed

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on the first choice route. The FCAS table does not include the long distance exchanges. You must set up another route list entry in that route list for long distance calls.

The advantage of this second NPA alternative is that you only have to keep track of the local exchanges. For the first NPA alternative and the HNPA alternative you need to keep track of every NXX in the NPA. Calls will be intercepted as invalid when they cannot be translated.

3. If you assign 9 as a COT trunk access code for local calls, instead of using AC2, this reduces the amount of translation required. When users' class of service is CTD (default as of Release 22), they cannot make long distance calls on the COT route. This gives you the control you need.

Translating an HNPA with BARS

1. If you decide to program the area code in which the node sits as a Home NPA:

- the system automatically removes the HNPA from the digits string and inserts the AC1 code in front of the remaining digits.
- you must translate all NXX codes that follow that inserted AC1 code and send them to the correct route lists. Some of the NXXs are local ones; others are long distance. Send each one to the proper route list index.
- doing this adds extra programming to your work at installation, compared to programming the NPA as a normal NPA. Users would not normally dial long distance exchange codes following AC1, but an area code instead. If you translate your own area code as an HNPA you are forced to translate the long distance exchange codes in the AC1 translation table.
- Translating an HNPA also requires ongoing administration to keep the NXX translation tables up-to-date if new NXXs are added in that NPA.

2. If you choose to program the area code as an NPA instead of an HNPA, the programmer has two choices for dealing with the local and long distance exchanges in that area code. Refer to the discussion in item 2. under *Translating an HNPA with NARS* on page 234.

3. If you assign 9 as a COT trunk access code for local calls, and use AC1 for long distance calls only, this reduces the amount of translation required. When users’ class of service is CTD (default as of Release 22), they cannot make long distance calls on the COT route. This gives you the control you need.

Upgrading BARS with HNPA to NARS

If you have a system with BARS software where you translated an HNPA and you are upgrading it with NARS software, you will have to plan for the introduction of an AC2 translation table. Even though your dialing plan may not have the users dialing AC2, you must translate NXX codes in the AC2 translation table. You may decide to remove the HNPA from the translation table, during the upgrade and change it to a regular NPA. Refer to the information in the previous section on *Translating an HNPA with NARS*.

Conversion

- The on-net to off-net overflow capability of NARS and location codes will send misdialed (out of range) calls to the main listed number programmed in the translation table for the location code. If that location does not have attendants or you do not have enough attendants for this call traffic, program another number as a main listed number instead. The number you program must have the correct NPA and NXX to be inserted in front of DNs in a single NXX site however.

  If you do not want misdialed calls to be sent from a node where they will occupy trunks and tie up attendants at the other end, translate the location code along with the valid leading digits of the DNs in the translation table. For example, if there is a location code 333 and the valid DNs start with 22, then translate the

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location code as 333 22 in the translation table. Any misdialed DNs, beginning with anything other than 22 will not be translated (they will be intercepted) and will not be sent to a route list.

Before you install new DID ranges at any sites in your network, ensure that programmers at all nodes are informed, in time to update their location code translation tables.

NCOS

- When setting up NCOS groups, many network planners find administration is easier if they match the NCOS number and the FRL number.

  For example, NCOS 1, is assigned an FRL 1; NCOS 3, is assigned an FRL 3 and so on. Or NCOS 20 is assigned an FRL2, NCOS 30 is assigned an FRL 3 and so on.

Route list worksheets

Ask your system supplier if they have a user-friendly worksheet that they use to summarize the route lists you are using. Worksheets may be easier to interpret than data printouts. If they do not have a standard work sheet, you may want to design one of your own.

Keep a set of these worksheets at each site and in the files you have for each NARS, BARS or CDP-equipped site.

Call Detail Recording

- Make sure that all users' calls produce call records where the letter A precedes the dialed digits, (except those users you have authorized for direct trunk access). This indicates that BARS, NARS, RS-ANI or CDP was used to make the call. If an unauthorized person's calls produce call records without the "A" then TGAR and TARG are not programmed properly to block direct trunk access. Investigate.

- You can program each trunk route to print toll calls only. The early systems considered toll calls to be those where the user dialed 0 or 1 as the first or second digit after the access code.

  With Rls 5.31 and Rls 8 and above, if the user did not dial the 1 or 0 but digit manipulation tables inserted the digits, the CDR toll calls only option would print these calls.

  With Rls 13 a Flexible Toll Definition option was introduced to allow non-North American customers to specify other digits that indicate toll calls, on a per route basis.

- Monitor CDR regularly. If you have CDR reports printed automatically, ask for automated exception reports or use manual methods to look for possible problem calls.

  Look for extra long calls, calls made before or after work hours, many unanswered callback queue calls, calls with a few digits ended with #.

- It is important to be prepared for the fact that users who make unauthorized calls often use telephones other than their own. It is important to make every telephone as secure as possible or to implement controls on a customer-wide or system-wide basis.

- Refer to the *Call Detail Recording (CDR)* section of this book and the *Basic Telecom Management Guide* for further information on CDR.

Network Traffic studies

- After you install NARS, follow up with a Network Traffic study a short time later. Some of the things you can find are:
  - translation problems
  - route list problems
  - trunk provisioning problems
  - whether Expensive Route Warning Tone is being used properly. NTRF studies can study the use of expensive routes, how often ERWT tone was given and how often the expensive routes were refused.

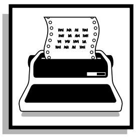
- which NCOS groups need training. If there is high blockage but little queuing by all groups or certain user groups, you should plan to do some training and/or order more trunks.
- whether or not user complaints about long queuing times are justified

Refer to the section called *Network Traffic studies* in this book for more information.

Training tips

- Make sure that users know who to call with a trouble report or questions related to the network.
- Plan to do ongoing refresher training especially about tones, billing procedures and dialing, especially if there are dialing plan changes or additions to the features.
- If you implement Expensive Route Warning Tone, follow up with Network Traffic (NTRF) studies to find out if it is really saving you money. Users may need more training after installation to help them understand what to do when they hear the tone. Surveys will tell you how users are accepting this tone. Tell users about the “E” that prints out on CDR records when they accept an expensive route. It is important that users understand that you have a way of identifying expensive calls and charging extra for these calls.

Maintenance tips



- A common source of problems is the use of defaults when assigning NCOS, Class of Service and TGAR to new telephones and trunks. Programmers should have clear instructions when installing about what to assign. You could decide to define NCOS 0 (the default) with higher capabilities so users will not be blocked by default but, if you do that, you lose some control.
- Remind network administrators when they encounter user reported problems, that simply printing out the NCOS of a telephone is not enough. They must also print the NCOS programming at that site to find out how that NCOS was set up. For example, they may assume that NCOS 1 was a fairly restricted NCOS but it could have the highest FRL (7) assigned. No one should make assumptions about a user's network-related capabilities based solely on knowing the NCOS assigned.
- If Callback Queuing fails at times there may not be sufficient call registers. Look at regular traffic study data (system option TFS004) to verify if there are call register overflows.
  If you do not have traffic study data, you can ask your system supplier to increase the number of call registers in 10% increments until the problem disappears.
- Scan CDR records for calls that are not being processed on the most efficient route. This is a sign that FCAS tables may need correcting or updating or that the wrong FCAS table has been programmed in the route list. It can also mean there are not enough trunks and the calls are overflowing to more expensive choices.
- Refer to *Appendix 7* on Trouble Reports for more information.

NARS and BARS

What to have ready

The following checklist summarizes the steps you should take before implementing NARS.

Table 34
NARS planning checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Decide on the UDP for your network. Choose AC1 and AC2 (optional).
✔		Decide if any users will have direct trunk access. Assign TGARs accordingly.
✔		Decide if any users will not be conditionally toll denied.
✔		Decide on the number of NCOS groups you need for the entire network.
✔		Decide if you want to use Expensive Route Warning tone.
✔		Decide which NCOS groups will be CBQ type I or type A.
✔		Decide on NCOSs for incoming TIE trunks, NSC lists, Auth codes etc.
	✔	Decide if you are going to use Routing Control. Plan your NCOS map and days and times when you want to use it. Assign a key to the console (optional).
✔		Gather all NPAs, NXXs, SPNs, LOCs, HLOCs, for your dialing plan. Decide whether you want to program an HNPA.
✔		Find out the valid DID ranges for all sites with location codes.
✔		Gather the main listed numbers and DID numbers for all sites for local and remote recognition.

— continued —

Table 34
NARS planning checklist (Continued)

Basic	Optional	Preparation
✔		Consider 911 calls or other emergency numbers. Decide how you want to configure ESA, in North America.
	✔	Predict everything your users may dial accidentally. Translate any codes that you want to permit.
✔		Decide if any numbers are to be restricted or left out of translation tables (in North America, 1-900 for example).
✔		Confirm numbers that incoming trunk routes should be blocked from calling.
	✔	Find a source of information to keep your translation tables up to date.
✔		Confirm the sequence in which you want routes in each route list scanned.
✔		Confirm (for each route list) what route choices you consider inexpensive or expensive (if any).
✔		Confirm any trunk route choices that you want turned off at certain times of day.
✔		Decide if you want to use TIE trunk routes to call free calling areas only. Confirm the NXXs in the free calling areas (N.A. only).
✔		Decide if you want FEXs used for free calls only or long distance calls also.
✔		Confirm for each route list what FRL can access each entry.
✔		Decide whether you want to activate queuing and for which users, on what routes.
✔		Verify the Iset for each route list.
✔		Decide if the last choice for any route lists should route calls to an internal number.
✔		Decide what you want for the network-related Intercept treatments.
— continued —		

NARS and BARS

Table 34
NARS planning checklist (Continued)

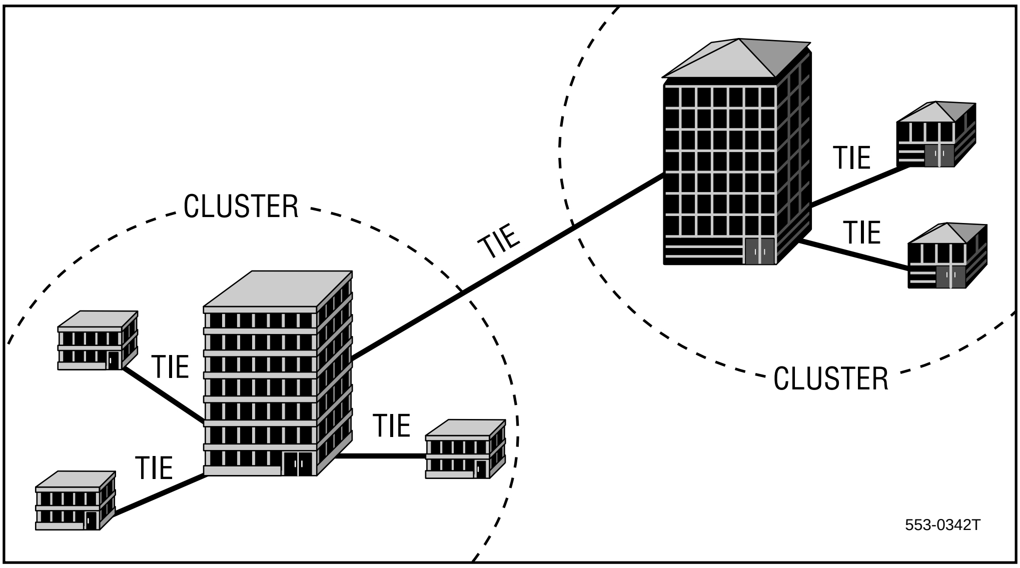
Basic	Optional	Preparation
✔		Prepare end user training programs. Prepare training for attendants.
✔		Prepare procedures to use for trouble reporting.
✔		Publish a network-wide directory.
✔		Train network administrators.

Coordinated Dialing Plan (CDP)

Coordinated Dialing Plan (CDP) software was designed to:

- provide an easy on-net dialing plan (simplified dialing)
- handle outgoing calls cost-effectively

A typical network where CDP is useful looks like this:



Coordinated Dialing Plan (CDP)

Basic feature configuration

Coordinated Dialing Plan (CDP) software lets you dial other users on the network without dialing access codes to trunks.

It can provide an easier dialing plan than the Uniform Dialing Plan (UDP) offered by Network Alternate Route Selection (NARS). Refer to the section on *NARS and BARS* in this book for more information on the UDP.

A CDP Dialing Plan for on-net calls could be as simple as:

XXXX

A UDP Dialing Plan with NARS requires the user to dial:

AC1 + LOC + XXXX

A telephone user can call any other telephone in the network by dialing a unique three to seven digit CDP DN assigned to the telephone. If the switch is equipped with the Directory Number Expansion (DNXP) package, the number assigned to each telephone can have a maximum of ten digits.

CDP DNs *must be a consistent length throughout the CDP network.* If you require variable length CDP DNs in your network, you should also install Flexible Numbering Plan (FNP) software. FNP software was designed for the European market where DNs of a consistent length are not always possible. However, FNP can be used globally. For more information on FNP, refer to page 283.

Each switch with only CDP equipped, is programmed to expect a certain number of digits. That way when users at that switch dial a CDP DN or if a CDP DN comes in on a trunk, the switch will process the call after receiving that number of digits. The system considers any call beginning with a Distant Steering Code to be a CDP DN. There is more information on Steering Codes later in this section.

Coordinated Dialing Plan (CDP)

A CDP-equipped switch can act in some ways like a node for other switches. It can provide a centralized public exchange network access capability that directs calls to and from the public network for the switches in the CDP group.

A common way to centralize trunks is to install DID trunks at one CDP-equipped switch. That switch receives the DID DNs from the Central Office and uses CDP to direct calls to CDP DNs at remote sites in the cluster. There is an option in the Customer Data Block that allows calls to overflow to public network trunks if the TIE trunks to the remote sites are busy. You need fewer DID trunks if you centralize them rather than installing separate DID trunks at each site.

NARS is the best software to configure at a node when you are centralizing trunks for several switches to share for outgoing calls. CDP lacks some of the features offered by NARS. Refer to page 248 for information on what CDP lacks compared to NARS.

CDP DNs dialed can be terminated locally or they can be routed to a remote switch in the CDP group, following translation, route selection, and digit deletion and/or insertion.

The major benefits of CDP

- simplified dialing plan
- alternate routing for on-net and off-net calls
- control of users by Network Class of Service
- time-of-day controls can be used on routes
- queuing can be implemented
- TIE trunks are not required between switches
- switches without CDP installed can be dialed using CDP DNs

Elements of CDP

Prime elements of the CDP feature are:

- Translation of digits
  - Steering Codes
- Automatic least-cost routing
- Time of Day (TOD) routing
- Digit Manipulation Indexes (DMI)
- Network controls
  - Network Class of Service (NCOS)
  - Facility Restriction Levels (FRL)
  - Routing Controls
- Queuing parameters

CDP compared to NARS

NARS was designed for nodes, CDP was designed for simplified on-net dialing. As a result, CDP software does not have some of the capabilities of NARS.

CDP does not provide:

- AC1 or AC2
- translation of NPAs, NXXs, LOCs, SPNs, HNPAs, HLOCs
- Iset marker on Route List Indexes
- Expensive Route Warning Tone
- CBQ type (I or A) in NCOS groups
- Free Calling Area Screening Indexes
- Supplemental Digit Restriction and Recognition

If CDP resides with NARS or BARS on a system, then the elements in the previous list are provided by the NARS or BARS software.

Coordinated Dialing Plan (CDP)

Translation

Functions of translation tables

Translation tables for CDP contain programming to do the following:

- identify what type of steering code is being translated
- point the call to a certain route list index (trunk routes that are appropriate for this call)

Steering codes

There are three kinds of steering codes that can be translated.

- Distant Steering Codes
- Trunk Steering Codes
- Local Steering Codes

Each one has a specific design intent.

Within one translation table there cannot be overlap between any two codes. This rule is called the “leftwise unique rule.”

For example, you cannot translate Distant Steering Code 2 and Trunk Steering Code 22 in the translation table. Steering codes cannot conflict with each other and they cannot conflict with the other numbers already assigned in the numbering plan. The numbering plan contains the attendant DN, Special Prefix (SPRE) Code, trunk route access codes, BARS AC1 or NARS AC1 and AC2, and internal DNs.

Distant Steering Codes (DSCs) are designed for on-net calls to other switches. The system treats any call that begins with a Distant Steering Code as a call to a CDP DN on the network. All CDP-equipped systems within one network must be programmed to expect the same number of digits in a CDP DN.

When a user dials another site, the leading digit(s) in the CDP DN dialed identify the site. Therefore, each switch in a CDP network must have unique leading digits in its CDP DNs. The calls are directed to the proper route list, based on the digits in the Distant Steering Code.

Coordinated Dialing Plan (CDP)

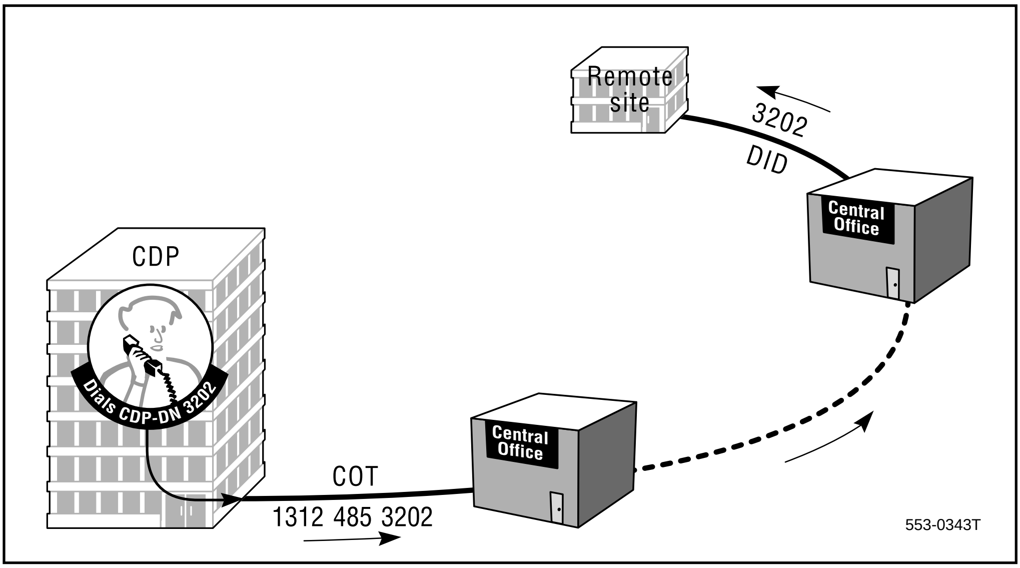
The destination dialed:

- does not have to be directly connected to the calling switch with TIE trunks
- does not have to have CDP equipped
- can be connected to the calling switch through other switches

The calling switch can outpulse whatever digits are required to reach the remote switch. Digit manipulation indexes change the CDP DN that the user dialed into the proper sequence of digits required.

The three examples listed above are illustrated in Figure 20, Figure 21 and Figure 22.

Figure 20- CDP user calls remote switch with no TIE trunks



Coordinated Dialing Plan (CDP)

Figure 21 - CDP user calls remote switch with no CDP

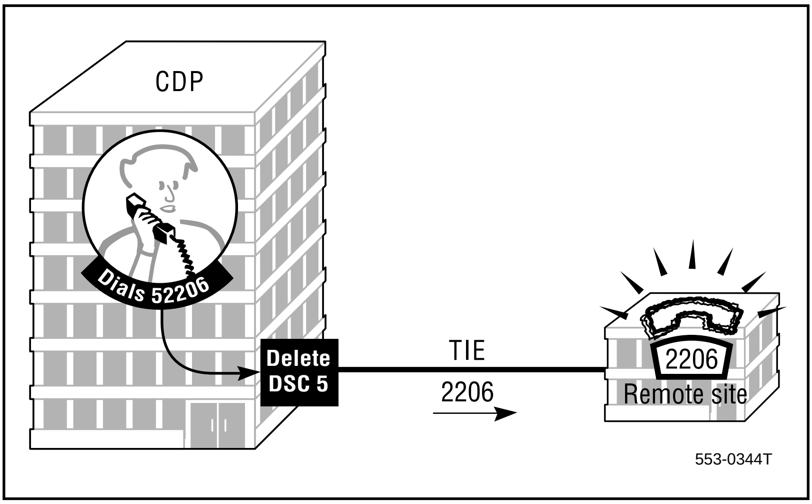
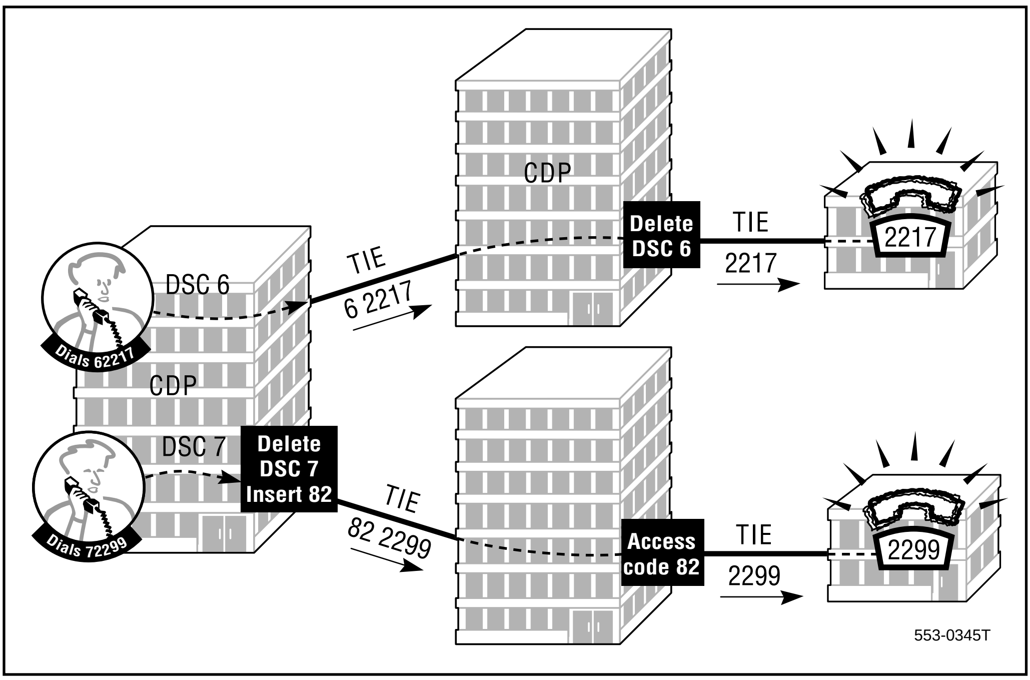


Figure 22 - CDP user tandems to reach remote switch



Coordinated Dialing Plan (CDP)

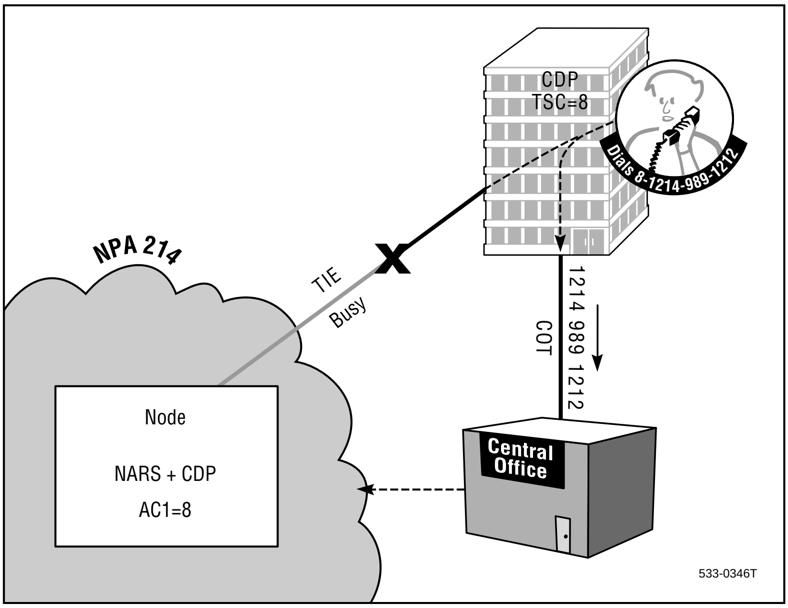
Trunk Steering Codes (TSCs) can be used for off-net calls. However, you do not have to program any Trunk Steering Codes. Users can make off-net calls using direct trunk access codes, if you prefer. However, if you want users at a CDP site to access route lists when they make public network calls, implement trunk steering codes instead.

When users make calls on route lists, they have access to alternate routes. These routes are scanned in the order you planned, and there are controls on the users and the routes to maximize your network efficiency.

A common application where TSCs are used is at a remote site connected to a node. If the node has CDP and NARS equipped and the remote site has CDP, then the user can make long distance calls by dialing a TSC, instead of the direct trunk route access code for the TIE trunks to the node. In order to maintain the UDP provided by the NARS node, the TSC should match whatever has been assigned for AC1 at the node.

The advantage of doing this is that if the TIE trunks to the node are busy, the users at the remote site can overflow to other trunks at their own site. As a network planner, you can decide which route(s) to use for the overflow at the remote site and which users can overflow.

Figure 23- Using Trunk Steering Codes at a remote site



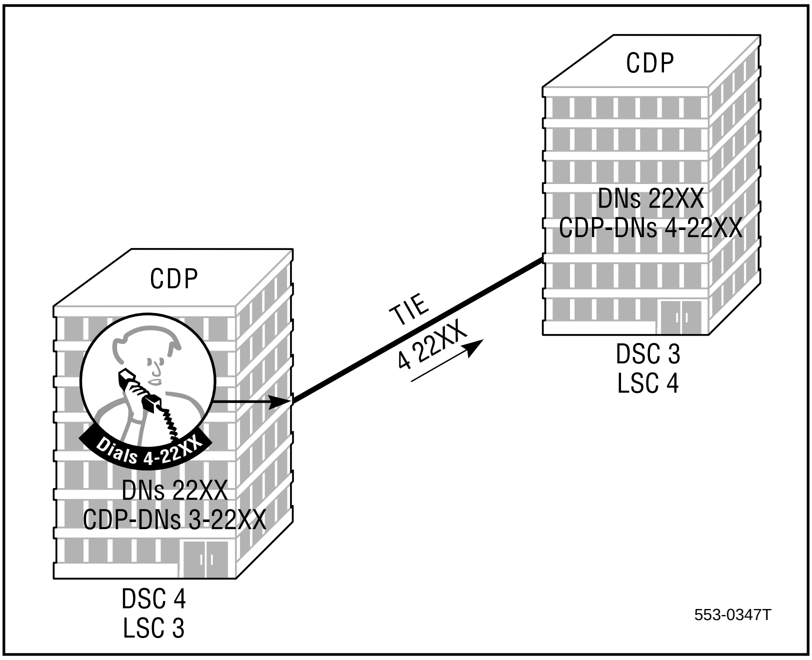
Local Steering Codes (LSCs) are designed to indicate the leading digit(s) of local DNs at a CDP switch. You may not need these if the local CDP DNs that a switch will receive are the same as the local DNs themselves.

For example, if you have a four digit CDP dialing plan and the local DNs are four digits long and /or the CDP DNs which the local site will receive from the remote site(s) match the digits in the local DNs exactly, the switch will have no trouble handling the calls using normal translation capabilities. No Local Steering Codes are needed.

However, if the local DNs are four digits long but users will dial five digit CDP DNs, then the leading digit must be translated as a Local Steering Code (LSC).

Coordinated Dialing Plan (CDP)

Figure 24- Use of Local Steering Codes



As you can see in the previous illustration, the DNs at the two switches are in the same number range. If neither switch administrator is prepared to change the DNs at their site, then the CDP dialing plan will have to be adjusted to make the CDP DNs unique at each site.

The network planner decides which extra leading digit to assign to each site. These leading digits make the CDP DNs unique at each site.

When a user at a remote site calls a user at the node, the five digit CDP DN can be sent into the node. The node will translate the leading digit as a Local Steering Code. In this case, the leading digit must be deleted so the call can be sent to the correct four digit DN. You can use a digit manipulation index to delete and insert digits after the LSC code is translated.

Coordinated Dialing Plan (CDP)

Alternatively, if the remote site has CDP programmed, the leading digit of the CDP DN at the node can be deleted at the remote site before the four digit DN is sent to the node on a TIE trunk.

> ***Note:*** It is a general rule to allow the last intelligent switch that will handle a call deal with as much translation as possible. Intelligent switches are those with BARS, NARS or CDP programmed. A switch can send digits, untreated, to an intelligent switch and let the intelligent switch deal with calls based on its programming.

You can decide for your own network how you will handle call translation and digit transmission between switches. Make sure you are consistent. Make sure that the digits you send to a site can be translated effectively by that site.

Capacities

- Steering Codes can be composed of one to four digits. If DN Expansion (DNXP) software is equipped, the steering codes can have up to seven digits.
- CDP supports up to 10,000 steering codes (32,000 if you have Flexible Numbering Plan software also). You may need that capacity if your users require transportable DNs (users at several switches in a cluster want to take their DNs with them when they move from one switch to another.)

Transportable DNs

The illustration below shows you a CDP network, after installation. The numbering plan is organized in such a way that the DNs at each switch have unique leading digits. When users move from site to site, taking their DN(s) with them, the numbering plan becomes less organized. The second illustration shows you the effect three moves have on the CDP programming.

Coordinated Dialing Plan (CDP)

Figure 25 - Transportable DNs - Before the move

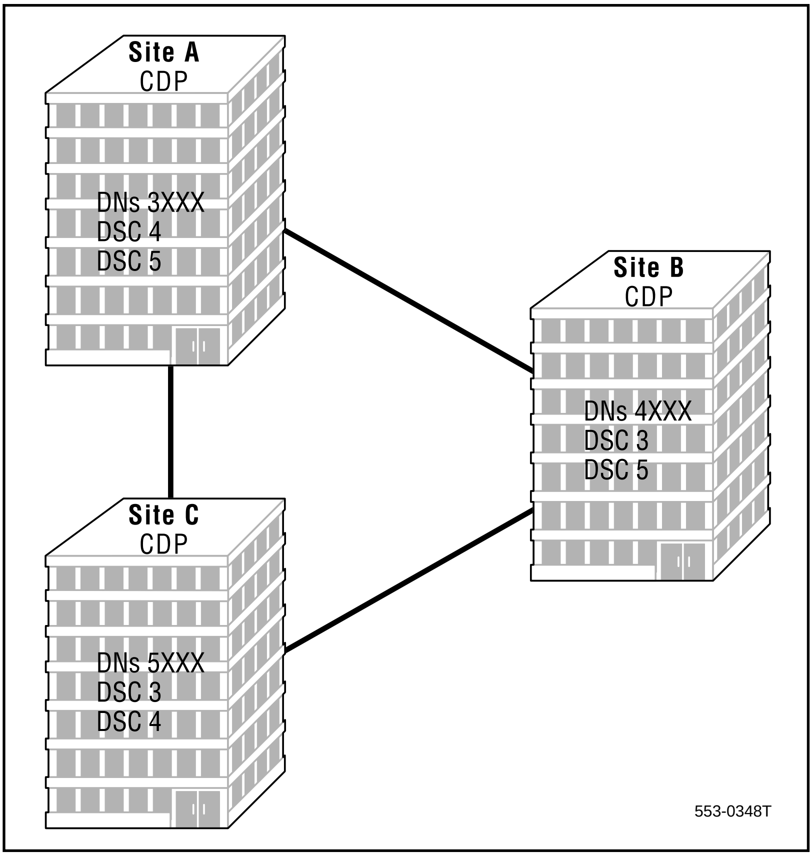
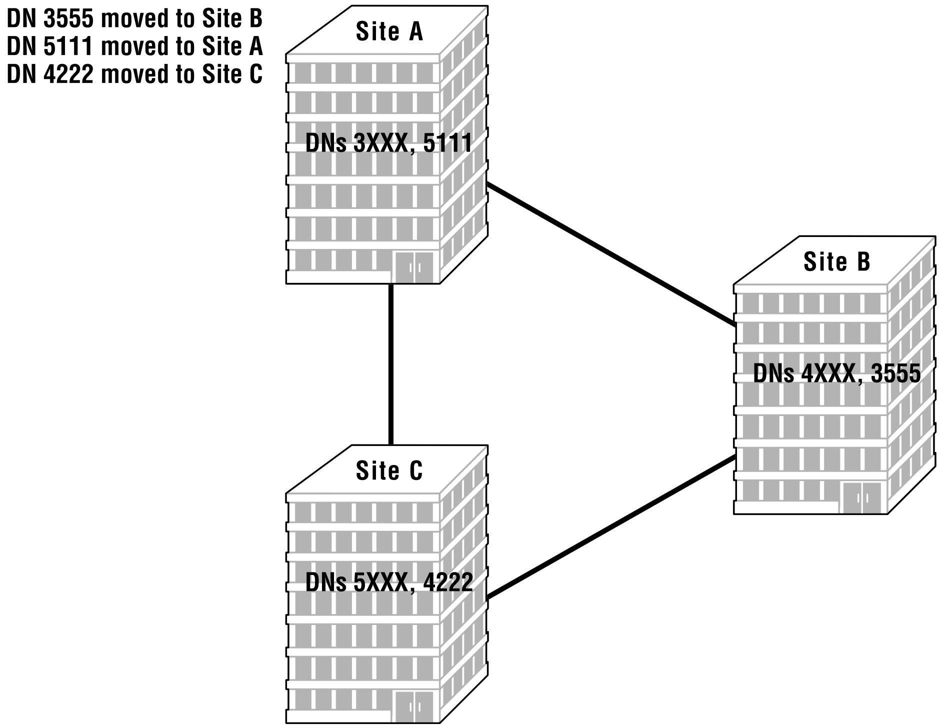


Figure 26 - Transportable DNs - After the move



DCSs		
Site A	**Site B**	**Site C**
3555, 40, 41, 420, 421, 4220, 4221, 4223, 4224, 4225, 4226, 4227, 4228, 4229, 4222, 423, 424, 425, 426, 427, 428, 429, 50, 510, 5110, 5112, 5113, 5114, 5115, 5116, 5117, 5118, 5119, 52, 53, 54, 55, 56, 57, 58, 59	30, 31, 32, 33, 34, 350, 351, 352, 353, 354, 3550, 3551, 3552, 3553, 3554, 3556, 3557, 3558, 3559, 36, 37, 38, 39, 4222, 510, 5110, 5111, 5112, 5113, 5114, 5115, 5116, 5117, 5118, 5119, 50, 52, 53, 54, 55, 56, 57, 58, 59	30, 31, 32, 33, 34, 350, 351, 352, 353, 354, 3550, 3551, 3552, 3553, 3554, 3555, 3556, 3557, 3558, 3559, 36, 37, 38, 39, 40, 41, 420, 421, 4220, 4221, 4223, 4224, 4225, 4226, 4227, 4228, 4229, 423, 424, 425, 426, 427, 428, 429, 43, 44, 45, 46, 47, 48, 49, 5111

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Coordinated Dialing Plan (CDP)

If you want users to have transportable DNs, at installation, you may want to program every four digit DN at the other sites into the database of the site you are programming. Enter the remote DNs into the database at each site calling them DSCs. When a DN moves to the site you can remove that DSC. If a DN leaves that site, you can add it as a new DSC.

Refer to information in the *Flexible Numbering Plan* section for the way the Vacant Number Routing feature can be configured to handle transportable DNs in an even simpler way.

Translation-related Intercept treatments

Intercept treatments are programmed in the Customer Data Block. They tell the system what to do when users in that Customer Group dial illegal or invalid numbers. The three choices of treatments are:

- give the user overflow tone
- connect the user with the attendant
- connect the user with a recorded announcement

The treatment that applies to CDP translation is Invalid translation intercept treatment. This is given to the user if the steering code digits dialed have not been programmed into the translation table.

Refer to the *Administration tips* in this section for more information about how to use intercept treatments effectively.

Group Dialing Plan (UDP and CDP)

When a node has NARS and CDP programmed and the remote sites all have CDP programmed, the users will be able to use both the Uniform Dialing Plan and the Coordinated Dialing Plan. This is called a Group Dialing Plan.

For off-net calls, users at the remote site will dial a Trunk Steering Code (TSC) which is the same as the AC1 at the node. The TSC directs the calls to a route list.

For on-net calls, users will be able to dial CDP DNs that start with Distant Steering codes or they can dial a TSC followed by a location code.

Coordinated Dialing Plan (CDP)

In order to properly translate on-net and off-net calls that both begin with a TSC, you would have to translate TSCs that have more than one digit. For example, if AC1 at the node is 8 and there are location codes 333, 444, and 555 programmed at the node, you would translate 8333, 8444, 8555 as TSCs at the remote site. These calls would go to a route list (containing the TIE trunk route only, possibly). You would translate 81 as a TSC for long distance off-net calls that go to a route list (containing the TIE trunk route and the COT trunk route, possibly).

When you have CDP software programmed at the remote site, you cannot implement a UDP by allowing users to have direct trunk route access to the TIE trunk route. If you did this, you would have to insert the AC1 digit at the node on the incoming TIE route. This digit insertion would interfere with the CDP DNs that are coming in on the same route from the remote site.

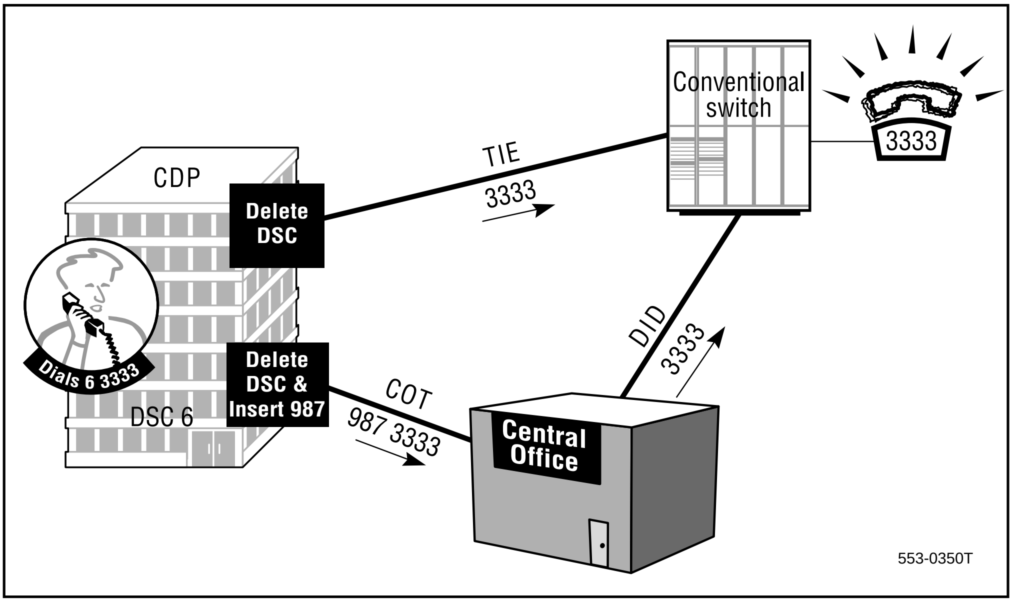
Conventional switch access

If your network has a switch without CDP software installed, refer to it as a conventional switch. Users at CDP-equipped sites can use DSCs to call the conventional switch users but the users at the conventional site cannot use DSCs.

The CDP-equipped sites can use digit manipulation indexes to send the correct digits to the conventional switch to allow the call to go through.

Coordinated Dialing Plan (CDP)

Figure 27 - CDP user calls conventional switch



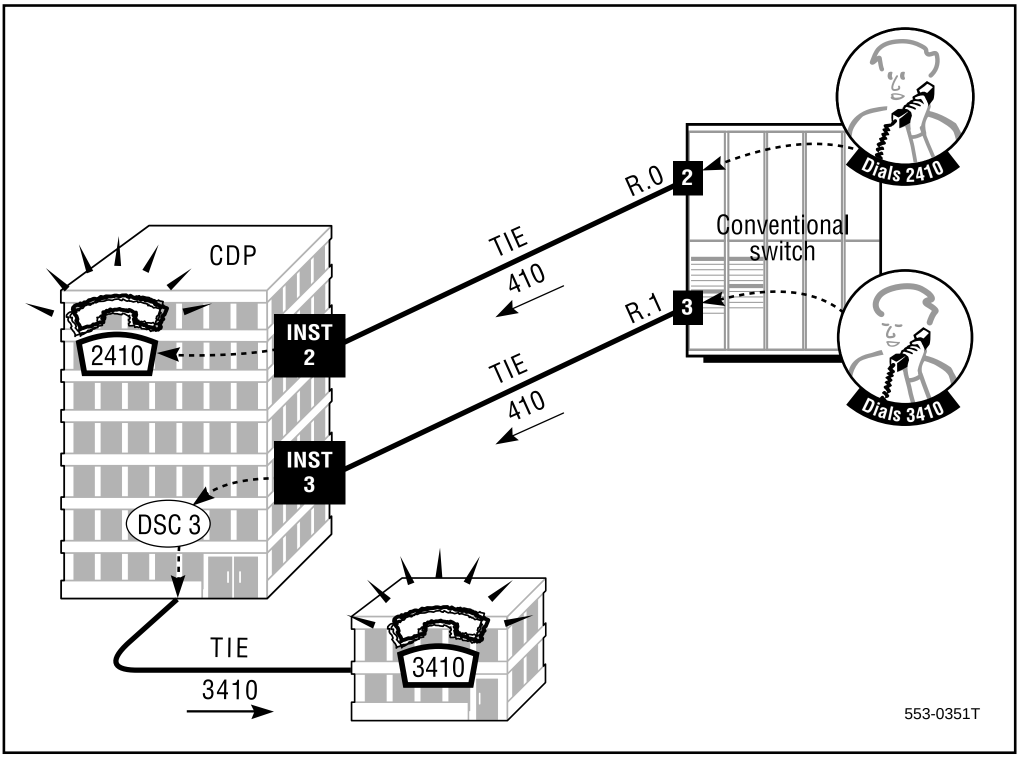
The users at the conventional switch will probably dial direct trunk access codes to make their calls. Train the users at the conventional switch to dial the TIE trunk access code followed by the CDP DN of the user they are trying to reach. If the conventional switch is connected to a node, then users should dial location codes instead of CDP DNs.

> ***Note:*** If the conventional switch is a non-Nortel Networks system, investigate whether you can install software similar to CDP in order to allow users to be part of the simplified dialing plan without having to use direct trunk access codes.

The closest you can come to a work around solution in a CDP network is to install several TIE trunk routes between the conventional switch and the nearest CDP-equipped switch. Make the access code to the first TIE route the same as the first digit of the CDP DNs at a CDP site. Program the incoming TIE trunk route at the CDP-equipped switch to insert the same digit. As a result, the call will have the proper DSC leading digit for the CDP-equipped switch to translate the call properly.

Coordinated Dialing Plan (CDP)

Figure 28 - Conventional switch workaround for CDP-type dialing



The cost of the TIE trunk routes required for this work around solution may be greater than the cost of implementing CDP software. Having several TIE trunk routes to the same destination is a very inefficient configuration. You will need fewer TIE trunks if they are collected together into one route as they would be if CDP software is installed.

Flexible Numbering Plan (FNP) and CDP

With X11 Release 20, FNP was offered globally. Previous to that release, it was only available in non-North American markets.

NARS is a pre-requisite for FNP. When CDP is installed as well, it allows flexibility in the CDP dialing plan. Normally CDP DNs must be a consistent length for an entire CDP network. FNP allows for variability in the length of the on-net DNs in the same network. The total number of digits to be dialed can vary depending on the telephone called.

Coordinated Dialing Plan (CDP)

A Flexible Length parameter is defined for each DSC and TSC in the translation table. This indicates the maximum number of digits that will be dialed for calls beginning with that steering code. When users dial that number of digits, the call is immediately processed. When they dial fewer digits, the call is processed after the NARS Interdigit Timer (NIT) timer expires or the user presses the octothorpe (#) key. For more information on the NIT, refer to page 223.

Refer to the section called *Flexible Numbering Plan,* if you want more information on how FNP works and the effect it can have on your dialing plan.

CDP routing

Translation tables point each translated code to the correct route list index. There is no limit to the codes that can point to the same route list index. When you are planning to translate a new code, evaluate your existing route list indexes to find out if one that is already programmed is suitable. Do not add a new route list index for every new code you translate. You will save a lot of memory if you direct many steering codes to the same route list.

The Invalid NARS/BARS call intercept treatment is given to any user who dials a translated code for which the route list index (RLI) that the translation table points to has been removed. This intercept treatment is programmed in the Customer Data Block.

Route list indexes (RLIs)

First choices, last choices

The route list index (RLI) is a series of trunk route choices that are listed in the order in which the network planner wants them to be scanned. The routes are scanned in that order, each time a call is made and sent to that RLI.

To save the most money, planners usually list the least expensive routes first.

Planners also put flat-rate trunk route choices first (such as TIE trunks or FEX trunks) so they get the highest amount of traffic on these. Putting large numbers of calls on flat rate trunks brings the cost of

Coordinated Dialing Plan (CDP)

each call down. The last choice(s) in the route list index can be the most expensive trunk routes or those on which you want the least traffic.

Route choices in a route list are called route list entries. There can be up to seven route list entries associated with each route list. If a switch is equipped with NARS (or BARS) software in addition to CDP software, route lists can be shared by both NARS and CDP calls. The capacities on such a switch become the same as the capacities for NARS or BARS.

Initial set (Iset)

On a switch that has only CDP and no NARS or BARS software, there is no Iset marker to be programmed for each route list index. For more information on the Iset marker refer to page 188 in the NARS section.

The absence of an Iset marker means that when routes in a route list are busy, users who activate Callback Queuing are queuing for all routes in the route list and not just the Iset routes initially.

Also, there is no Callback Queue type (I or A) to be programmed in the NCOS groups. However, you must still activate the Callback Queuing feature in the NCOS programming. All users can scan all routes in the route list and access any routes their FRL allows.

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Each choice in the RLI has the following parameters programmed for it:

- an entry number (0 to 6). The system always scans route list indexes starting at entry 0.
- trunk route number. Each trunk route must have a route number assigned before it can be programmed as an entry in a RLI.
- Time of Day schedules (TOD) when the route is on or off in that route list. Refer to page 212.
- instructions on whether to treat the call as an internal one. Refer to page 192.
- Digit Manipulation Index (DMI). Refer to page 193.
- a minimum level (Facility Restriction Level - FRL) that users must be in order to access the route when it is used in this RLI. Each entry in a route list has a minimum FRL required for access. All network users are assigned an FRL through their Network Class of Service (NCOS). You can restrict the type of calls and route choices allowed to users. Refer to page 196 for more information on NCOS.
- whether it is to be treated as expensive/non-expensive. Refer to page 212 and refer to *Control tips*.
- whether Queuing options are activated or not (Callback Queuing, Off-Hook Queuing, ISDN Off-Hook Queuing, ISDN Drop Back Busy, Step Back on Congestion). Refer to the *Queuing* section in this book.
- instructions on whether to skip conventional signaling or not. (Refer to the *ISDN* section on Revert to Conventional Signaling.)
- Digit Manipulation Index (DMI) to use if ISDN Signaling Link (ISL) is installed and the D-channel is not working temporarily (ISL - Revert to Conventional Signaling feature)
- instructions on whether the route is to be used for Virtual Network Services or not. (Refer to the *VNS* section.)
- instructions on whether to use Tone Detectors or not

Coordinated Dialing Plan (CDP)

The entire RLI has:

- the minimum level (MFRL) users have to be to begin to scan the RLI (only operates with Network Authorization Codes software)

The decisions you need to make concerning these route list parameters are listed in the *What to have ready* table in this section.

Time-of-day schedules

CDP software allows you to break the 24 hour day into a maximum of three parts or schedules. You use the schedules to turn routes on and off at certain times in certain route lists. Each entry in the route list has all three time-of-day schedules applied. The entry is on all day by default. You turn the schedules off for the hours of the day when you do not want the entry used.

If you want routes on during the work day only, use time-of-day (TOD) schedules to turn off all routes in all route lists at a certain time every day. (You can also use the Routing Control feature to do this. Refer to page 212 in the NARS section.)

Each schedule can have more than one period. For example, schedule 1 could be 6:00 am to 8:00 am and 4:00 pm to 7:00 pm. You would program a schedule this way if you treat routes the same way during both of these time periods. In other words if you turn a route off at 6:00 am to 8:00 am it will also be off between 4:00 pm and 7:00 pm, if both time periods are part of the same schedule.

Most planners use the TOD schedules to turn off routes when they are not cost effective or to force calls to overflow to other choices.

Local Termination

You may have a case where you want calls sent to an internal number if all the other entries in the route list are busy. You can do this if you program an entry in the route list at the end of all the other entries that has local termination active.

Coordinated Dialing Plan (CDP)

The entry can refer to a digit manipulation table that will process the digits dialed so you are left with the internal DN you want. This internal DN can be:

- a single call or multiple call DN on one or more telephones
- a voice mail DN (where an announcement service could give the caller instructions or the user could leave a message)
- an ACD DN
- an attendant DN

You might want to use this capability to signal a network administrator’s DN. These calls would tell the administrator that all routes in a certain route list are busy. If they started to get a large volume of calls, they could investigate the problem.

Instead of an internal DN, you can also use digit manipulation to end up with digits so that translation begins all over again. Another route list with different routes could be tried. In this way you can daisy-chain route lists.

Do not use local termination on an entry unless it is the last one in the route list. The system will not overflow calls to entries after one with local termination activated.

Digit manipulation indexes (DMI)

CDP uses digit manipulation indexes to modify the digits that the user dialed to make them appropriate for the trunk being used for the call.

You must know what digits the system at the other end of the trunk requires in order to process the call in the way you want it to process the call. You can control the way the other system will handle each call you send to it by sending digits that will force it to handle the call a certain way. For example, if you want the call to go out on a particular route at the other site, send a trunk route access code as leading digits. If you want the other site to use alternate routing, send it AC1 or AC2 or CDP steering codes, as required.

For example, if you have a remote site with BARS, you can send calls to it on TIE trunks with AC1 at the beginning. This will allow the remote switch to use BARS to route the call effectively. On the other

Coordinated Dialing Plan (CDP)

hand, you could send the call to the remote site with a certain trunk route access code at the beginning to force the remote site to send the call out on a certain route.

Digit manipulation indexes delete digits from the beginning of the string of dialed digits. You don't specify what specific digits are being deleted, you program the number of digits that are to be deleted. After deletion occurs, then insertion of specific digits at the beginning of the remaining digits occurs.

Unlike the AC1 and AC2 codes with NARS that are deleted by the software automatically, Distant Steering Codes and Trunk Steering Codes must be deleted using digit manipulation indexes (DMIs) applied in the route lists, if you do not want these digits outpulsed. Local Steering Code calls are handled by a digit manipulation index programmed in the translation table.

You can use DMIs to insert pauses. An asterisk (*) indicates a three second pause before the remaining digits are outpulsed. You do not need to insert pauses for dial tone on Meridian 1 systems. You may need them if you send calls to other systems that require a pause in order to find a trunk before sending the call out.

If you do not want any digits deleted or inserted before the call is sent out on an entry in a route list, assign DMI 0. This index has no deletion and insertion, by default.

Network Class of Service (NCOS)

There are differences between the NCOS programming for a NARS system and that of a CDP system. Refer to the introductory information on Network Class of Service in the *NARS and BARS* section on page 194. Then read the information on the next few pages applying NCOS to CDP.

Coordinated Dialing Plan (CDP)

Attributes of an NCOS

The attributes affiliated with an NCOS are:

- Facility Restriction Level (FRL)
- Queuing parameters
- Network Speed Call feature and list access
- Automatic Redial parameters
- Maximum Precedence Level (Autovon)
- Equal Access

NCOS – Facility Restriction Level (FRL)

You assign the minimum level that a caller must have in order to access each entry in a route list. The level is called a Facility Restriction Level since it is a way of restricting access to facilities (trunks).

The user's level is determined by the Network Class of Service that is assigned to them. Each NCOS you program has an FRL assigned.

If a user's FRL is equal to or greater than the required FRL on the entry in the route list, they can access it.

FRL-related Intercept treatment

When a user dials a call that requires a higher FRL than they have, "NARS/BARS blocked call" Intercept treatment will be given.

How to use FRLs

FRLs can range from 0 (most restricted) to 7 (least restricted).

If you do not want low-level users to access expensive trunks, assign a low-level FRL to those users' NCOSs. Program a higher FRL on the expensive route entries in the route lists.

The same route can appear as an entry in many different route list indexes. Each time the route appears as an entry, it can have a different FRL. The FRL you program depends on what types of calls the route is handling in that route list. You decide for each route entry in each route list the minimum FRL a user must be in order to use it.

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For example, COT trunks in route 8, used in a route list for calls to a site overseas, might require an FRL of 7 but the same COT trunks in route 8 in a route list for calls to a site in the same city might only require an FRL of 1.

Also, if long distance calls are less expensive at night, you may want to control the users who access the COT trunks during the day and allow more users to call out on these trunks at night. You will have a route list for long distance calls. The entry for the COT trunks in the daytime might require an FRL of 4 when the calls are expensive but the same COT trunk route in the same route list for night time long distance calls might only require an FRL of 1.The same route appears twice in one route list in this example. With different time-of-day schedules active to control when the route is on and off you can use FRLs to control who accesses the route at different times of day.

911 emergency calls

In North America, you will have to decide how you are going to translate 911 calls.

If your system has CDP only, you can translate 911 as a Trunk Steering Code. That way users do not have to dial 9911 as they would if 9 is the COT trunk route access code. TSCs are not deleted by software and so 911 will be outpulsed with no digit manipulation required. You will have to translate Trunk Steering Codes for the other public network calls (codes such as 912, 913, ..., 919, 92, 93, ..., 99) until you have all the public network calls translated.

NCOS – Expensive Route Warning Tone

There is no Expensive Route Warning Tone on systems equipped with Coordinated Dialing Plan software only.

NCOS – Queuing parameters

There are several forms of queuing offered in ESN and ISDN networks. A user’s access to these features can be controlled in the NCOS programming and often in the telephone set programming as well. Refer to the information on Callback Queuing on page 201 in the *NARS and BARS* section.

Coordinated Dialing Plan (CDP)

Since there is no Iset marker in the route lists you program on a CDP system, there is no Callback Queuing type (I or A) in the NCOS.

Users scan all routes in the route lists when they make calls and they queue for all routes, if they are busy. When a trunk becomes available, the system still checks the FRL required on the route against the user's FRL. If the user's FRL is not high enough, the trunk will not be offered.

NCOS – Automatic Redial parameters

Refer to the *NARS and BARS* section.

NCOS – Maximum Precedence Level

Refer to the *NARS and BARS* section.

NCOS – Equal Access

Refer to the *NARS and BARS* section.

Queuing

The Ring Again feature allows users to queue for busy internal DNs and trunk routes (when direct trunk access codes are dialed).

For calls directed to a remote switch, Callback Queuing can be applied if all local outgoing trunk routes to the remote CDP switch are busy or blocked.

You must install ISDN and the Network Ring Again feature if you want users to queue for busy DNs at other CDP-equipped switches. Refer to information on Network Ring Again in the *Queuing* section of this book.

Customer-wide CDP parameters

There is programming that must be done that will affect the overall CDP operation for a particular Customer Group. It involves the following:

- setting aside memory for such things as:
  - route lists

– digit manipulation indexes

– CDP steering codes

Note: Choose maximum numbers that will accommodate future growth.

- programming the time-of-day schedules
- enabling the Routing Control feature – programming the map of NCOSs and the time-of-day schedule or extended time-of-day schedule (day(s) of the week) when Routing Control is to be used
- activating the ESN software to look at a user's TGAR (as well as the FRL) and use it to allow/restrict ESN calls on routes in route lists. Refer to the information on *ESN bypass control and TGAR* that follows.

ESN bypass control and TGAR

A system *without* BARS, NARS or CDP installed uses the Trunk Group Access Restriction (TGAR) code programmed for the calling station to determine if the user can access a route.

Users dial trunk route access codes in order to make calls on these non-ESN systems.

The way TGAR and TARG work is: by blocking the call when a user dials a trunk route access code, where the TGAR code programmed for the caller matches one of the Trunk Access Restriction Group (TARG) codes programmed on the route.

As of Release 22 the default TGAR is 1 and the default TARG is 1. Therefore, users are blocked from direct access to routes by default.

The default class of service is Conditionally Toll Denied (CTD). This will block the user from dialing toll calls (calls beginning with 1 or 0) on direct access calls. It will allow ESN toll calls that the user's NCOS permits.

> ***Note:*** There is an Intercept treatment for “Calls from a restricted station”. It is given to a user who dials any long distance call from a toll denied (TLD) telephone or when a user dials a toll call using a trunk route access code from a telephone that is Conditionally Toll Denied (CTD).

Previous to Release 22, the default for TGAR was 0 and routes had no TARG programmed by default. TGAR 0 is unrestricted, by definition. This meant that users had access to all routes by default.

The default class of service was Unrestricted (UNR). UNR allows all calls.

A system *with* BARS, NARS or CDP installed uses the FRL programmed in the NCOS of the calling station to determine if the user can access a route when the user dials AC1 or AC2 or CDP steering codes in order to make calls.

With ESN, the digits are translated and sent to route lists. The FRL for each route is compared to the FRL of the caller. If the caller’s FRL is equal to or greater then the FRL required on the route, then the call can be sent out on that route.

However, if users bypass ESN route lists by dialing trunk route access codes when they make calls, they will be able to call out unless they are controlled by the TGAR and TARG interaction.

> ***Note:*** Users will see the trunk route access codes on the displays of their telephones. Some people will try to dial these codes when they make calls, if they are not prevented.

Preventing bypass

It is strongly recommended that you prevent users from direct access calls that bypass ESN and all the controls it offers. Use TGARs and TARGs to prevent bypass.

Make sure the caller’s TGAR matches the TARG on routes to which you do not want them to have direct access.

Coordinated Dialing Plan (CDP)

Do not forget that TIE trunks and other non-telephone terminals (such as Authorization Codes) can have TGARs assigned. Include them in your TGAR programming plan.

If 9 is going to remain a trunk route access code for the COT route, make sure that there is no TARG programmed on the COT route. Otherwise people will not be able to make local calls from the switch.

You may want selected users such as executives, network administrators and planners, and technicians to have direct access to routes for testing purposes or to place calls in emergencies.

Attendants always have direct access to routes since you cannot assign a TGAR code to a console.

Activating TGAR for ESN calls

Optionally, you can have an ESN system look at the user's TGAR, in addition to the FRL, for ESN calls.

This capability was designed for networks where there are two different departments of users within one Customer Group. If each department requires:

- its own special service trunk route (FEX, for example)
- each department to pay for its own route
- to ensure that only its users call out on its own route
- both FEX routes in the same route lists, so all users can use the UDP to make calls

It is not possible to restrict users to their routes only using FRL only.

Instead, if you activate TGAR on ESN calls, and you assign the same FRL to both groups but a TGAR that matches the TARG of the other group's FEX, it will ensure that calls do not overflow to the wrong FEX.

Coordinated Dialing Plan (CDP)

The drawback to using TGAR with ESN calls is that if you use TGAR to block certain ESN calls, you would have to leave most routes without TARGs so ESN calls could be made. Therefore, you cannot also use TGAR to prevent bypass.This is a substantial security and control measure to give up.

Routing Control (RTC)

The Routing Control feature provides a mechanism for changing a user's NCOS. You program a "map" of instructions that tells the system what each NCOS becomes when the feature is active. You can have an NCOS become a higher number or lower number or stay the same. You must specify what is to happen to each one. If you do not specify, an NCOS will become NCOS 0 by default when RTC is active. It is a good idea to print out the programmed data, to be sure of the map programmed.

RTC can be activated three different ways:

- when time-of-day schedule 2 is in effect, when only CDP software is equipped
- during certain, specified days of the week (called an extended time of day)
- or when the attendant presses a Routing Control (RTC) key, one of the programmable feature keys of the console

TODS 2

If you want user's NCOSs to change during a certain period every day, program the time period in schedule 2. For example, if users are supposed to leave the building at 5:00 pm every day and not return until 7:00 am, you can change the NCOSs to 0 during that period every day. The route lists must be programmed so that NCOS 0 will not have a high enough FRL to access any routes. Make sure however, that NCOS 0 is still a high enough level for users to reach emergency numbers.

Extended TOD (ETOD)

If you do not want users to call out on weekends, specify those days as active RTC days. All day (24 hours) Saturday and Sunday, the feature will automatically be activated.

Coordinated Dialing Plan (CDP)

Routing Control (RTC) key

If you cannot be certain of the times when you will want the feature to be activated automatically, then assign an RTC key to the console. Someone can activate/deactivate the feature when it is appropriate.

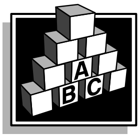
> ***Note:*** Authorization Codes can be used to override the restrictions imposed through Routing Control. A user with an auth code can go to any telephone and make a call. If a user enters a valid authorization code while Routing Control is in effect, the NCOS number associated with the auth code is applied for the duration of the call. The NCOS assigned to the auth code is not affected by the Routing Control feature (design intent). Calls made will be billed to the auth code dialed, if you have CDR.

Using Routing Control as emergency bypass

This feature can provide a way of re-routing calls when certain trunks are not working temporarily.

For example, if the TIE trunks to remote sites are not working, the attendant can press the Trunk Group Busy key(s) associated with the route(s) to make them appear busy for outgoing calls. Then, if Routing Control is programmed to raise the NCOSs of the users, the RTC key can be pressed. This allows users to access alternate routes temporarily, now that they have a higher NCOS. When the emergency is over, the TGB key(s) can be pressed to make the trunks idle again, and the RTC key can be pressed to return the NCOSs to normal.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Telephone displays

When a user receives a call from an external CDP DN, the display shows the access code for the incoming trunk route and the member number of the trunk being used for the call. The caller's DN does not display, unless ISDN hardware and software is equipped. The caller sees the CDP DN of the user dialed.

If you upgrade the network so Calling Line IDs (CLIDs) travel, users will see the CLID of the caller, preceded by an H. The H identifies a private network call.

DID call routing to other switches

When DID trunks terminate at one central switch, CDP can be used to route calls to telephones at other switches. The DID DN comes into the central location on a DID trunk, CDP translates the leading digits as DSC digits and sends the call to a route list and out to the remote switch.

You can allow the DID call to overflow to public network trunks when the TIE trunks between switches are busy if you activate an option (RTA) in the Customer Data Block. You can use digit manipulation indexes to insert and delete the necessary digits to outpulse the correct public network number.

You can use this capability to reduce the number of DID trunks you need when you install them all at one switch instead of spreading them out at several switches.

Coordinated Dialing Plan (CDP)

Control tips

There are many system and customer-related features and capabilities that you can use to improve the control you have of your users and the network.

- Users sometimes Call Forward their telephones to trunk route access codes or ESN numbers before they go home at the end of a shift. From home, they can dial into their telephone (using the DID number). The telephone forwards them to a trunk or a destination. They may finish dialing a call from home if the office telephone is only forwarded to an access code.

  The bottom line is that the long distance charges are billed to you, not the user doing the dialing.

  The call also adds extra usage to your trunks.

  These issues can pose big problems especially if they give the DID number to other people to use.

  Prevent this by activating the Call Forward External Deny feature on the telephones.

NCOS

- When setting up NCOS groups, many network planners find administration is easier if they match the NCOS number and the FRL number.

  For example, NCOS 1, is assigned an FRL 1; NCOS 3, is assigned an FRL 3 and so on. Or NCOS 20 is assigned an FRL2, NCOS 30 is assigned an FRL 3 and so on.

Coordinated Dialing Plan (CDP)

Route list worksheets

- Ask your system supplier if they have a user-friendly worksheet that they use to summarize the route lists you are using. Worksheets may be easier to interpret than data printouts. If they do not have a standard work sheet, you may want to design one of your own.

  Keep a set of these worksheets at each site and in the files you have for each NARS, BARS or CDP site.

Call Detail Recording

- Make sure that all users’ calls produce call records where the letter A precedes the dialed digits, (except those users you have authorized for direct trunk access). This indicates that BARS, NARS, RS-ANI or CDP was used to make the call. If an unauthorized person’s calls produce call records without the “A” then TGAR and TARG are not programmed properly to block direct trunk access. Investigate.

Network Traffic studies

- After you install CDP, follow up with a Network Traffic study a short time later. Some of the things you will find are:
  - translation problems
  - route list problems
  - trunk provisioning problems
  - which NCOS groups need training. If there is high blockage but little queueing by all groups or certain user groups, you should plan to do some training and/or order more trunks.
  - whether or not user complaints about long queuing times are justified

Refer to the *Network Traffic studies* section of this book for more information.

Coordinated Dialing Plan (CDP)

Administration tips

- Without ISDN software, you cannot program a CDP DN as a HUNT DN or a Call Forward No Answer DN for a local telephone. However, people can use the Call Forward All Calls feature to redirect calls to a CDP DN at another site. Attendants can post trunks to CDP DNs using Flexible Night Service.

Call Detail Recording

- CDR printouts show the internal DN of the user who made the call, not the CDP DN (including the Local Steering Code).
- A standard CDP call record shows the trunk route access code for the route that handled a call instead of the steering code dialed by the user. If you need to know what the user dialed for billing purposes, you must activate an option in the Customer Data Block that allows both the steering code and the trunk route access code to print out.

Training tips

- If users must dial extra leading digits in your CDP network, label the keys of their telephones with their full CDP DN, not their real internal DN. This will help to confirm the idea that users are supposed to dial the same number of digits to reach an internal or an external user.
- CDP calls can take longer when using public network trunks compared to TIE trunks. Tell users that if a call seems to take longer than normal, it is probably because it went on the public network. If you let users know about this, you will not have as many trouble reports when users experience these delays.

Maintenance tips

- When you first install CDP, plan to use Intercept treatments effectively to help you identify programming errors. Instead of overflow tone, give users a recorded announcement (with instructions) or send calls to the attendant so the user can speak to someone.

  If you are using transportable DNs, you might want to use the attendant for Intercept treatment on an ongoing basis so you will know immediately if users experience problems reaching a user who has moved. This will help you keep your translation tables up to date.

What to have ready

The following checklist summarizes the steps you should take before implementing CDP.

Table 35
CDP planning checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Decide on the CDP DNs for your network. Choose the DSCs (and LSCs and TSCs - optional).
✔		Decide if any users will have direct trunk access. Assign TGARs accordingly.
✔		Decide if any users will not be conditionally toll denied.
✔		Decide on NCOS groups you need for the entire network.
✔		Decide on NCOSs for incoming TIE trunks, Auth codes etc.
— continued —		

Coordinated Dialing Plan (CDP)

Table 35
CDP planning checklist (Continued)

Basic	Optional	Preparation
	✔	Decide if you are going to use Routing Control. Plan your NCOS map and days and times when you want to use it. Assign a key to the console (optional).
✔		Consider 911 calls or other emergency numbers.
	✔	Predict everything your users may dial accidentally. Translate any codes that you want to permit.
✔		Decide if any numbers are to be left out of translation tables (1900 for example).
	✔	Decide if you want incoming DID calls to overflow to the public network if outgoing TIE trunks are busy.
✔		Confirm the sequence in which you want routes in each route list scanned.
✔		Confirm any trunk route choices that you want turned off at certain times of day.
✔		Confirm for each route list what FRL can access each entry.
✔		Decide whether you want to activate queuing and for which users, on what routes.
✔		Decide if the last choice for any route lists should route calls to an internal number.
✔		Decide what you want for the network-related Intercept treatments.
✔		Decide if you want your CDR records to print the trunk route access code only or the steering code as well.
✔		Prepare end user training programs. Prepare training for attendants.
— continued —		

Coordinated Dialing Plan (CDP)

Table 35
CDP planning checklist (Continued)

Basic	Optional	Preparation
✔		Prepare procedures to use for trouble reporting.
✔		Publish a network-wide directory.
✔		Train network administrators.

Flexible Numbering Plan

ESN software packages such as Network Alternate Route Selection (NARS), Basic Automatic Route Selection (BARS) and Coordinated Dialing Plan (CDP) were originally designed to support North American numbering plans. Flexible Numbering Plan (FNP) software was designed for markets outside of North America, to support the variable numbering systems that exist globally. FNP software can be used globally, wherever it is required.

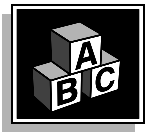
NARS software is a pre-requisite for FNP. You will understand FNP better if you read the section called *NARS and BARS* in this book first. If you will be using simplified dialing as well, you should read the section called *Coordinated Dialing Plan* in this book before you read about FNP.

FNP offers:

- 32000 steering codes per customer group
- NARS access codes AC1 and AC2 can be up to four digits long
- Flexible length parameter (FLEN)
- Vacant Number Routing
- Transferrable DNs (TNDNs)
- Universal Numbering Plan (UNP)
- Group Dialing Plan
- Free Special Number Screening (FSNS)
- Alternate Routing Remote Number (ARRN) with SPNs

Flexible Numbering Plan

Basic configuration



A network with CDP (and no FNP) requires all CDP-DNs to be the same length. CDP-DNs are on-net calls starting with a Distant Steering Code (DSC). You program the number of digits that will be in CDP-DNs and all outgoing and incoming CDP on-net calls are expected to meet this requirement. All switches in a CDP network must have the same setting for this parameter if CDP is to work.

NARS without FNP uses on-net location code calls that are of a fixed length. The total digits is expected to be seven.

With FNP, the total number of digits that are to be dialed for on-net calls can vary, depending on the station dialed.

Flexible Length (FLEN)

Using the FLEN prompt, you can tell the system when to process a call. When the FLEN number of digits are dialed, the call is processed. You can set a different FLEN parameter for each code you translate. This allows flexibility in the dialing plan. The user can dial more digits than the number programmed for FLEN, but the call will be processed when the FLEN number of digits has been dialed.

If the user dials fewer digits than the FLEN programmed for the call, the call will be processed when either of the following happens:

- the user dials # (octothorpe)
- the NIT timer expires

When FNP software is installed, you can program a location code in the translation table and program up to ten digits for FLEN.

You can program CDP Trunk Steering Codes and Distant Steering code calls with a FLEN as well. The maximum FLEN for DSCs is 10 digits. As of Release 22, TSCs can have a maximum FLEN of 24 digits.

With FNP, steering codes you translate can be up to seven digits long. You can have up to 32000 steering codes per customer group.

Flexible Numbering Plan

Vacant Number Routing (VNR)

Using the VNR capability, you can program an FNP-equipped switch to direct calls to a designated route list index if a user dials a CDP or NARS number that cannot be translated. You can use this capability to send calls to a central location where all translation for an entire network will be done. As a result, you will not need to translate CDP and NARS calls at all sites, only the central site.

Transferrable DNs

VNR can be very useful in networks where users move frequently. You will only need to make programming changes at one site when a user moves, if you use VNR. With CDP and no FNP software installed, every time a user moves from one site to another, and keeps their DN with them, you must make programming changes to all other CDP-equipped sites.

When you use VNR, you must program the maximum number of digits that will be dialed for CDP and NARS calls. The maximum for both types of calls is ten digits.

> When you use VNR, be careful to avoid scenarios that may route calls back and forth inefficiently when the calls are sent from one switch to another. If you want maximum control of how calls are routed, it is best to translate digits at each site.
>
> Features such as Network Message Services and Network Ring Again are not compatible with Vacant Number Routing.

Network dialing plans

Flexible numbering can be used with one of the following plans to create a variety of different network configurations:

Coordinated Dialing Plan (CDP)

Stations on any node have unique 1 to 10-digit numbers. One to seven digit numbers can be transferrable. Within the CDP group, a caller can reach a station directly without having to input an access code or pause for a dial tone.

Flexible Numbering Plan

Universal Numbering Plan (UNP)

Three to seven digit numbers, transferrable across the network, are provided. If a caller dials a vacant number on any node, the call reroutes to a common location, where a list is consulted to determine if there is an alternate location for the called Directory Number. If so, routing occurs. Otherwise, the caller receives vacant-number treatment. The transferrable DN cannot be any longer than 7 digits.

Group Dialing Plan

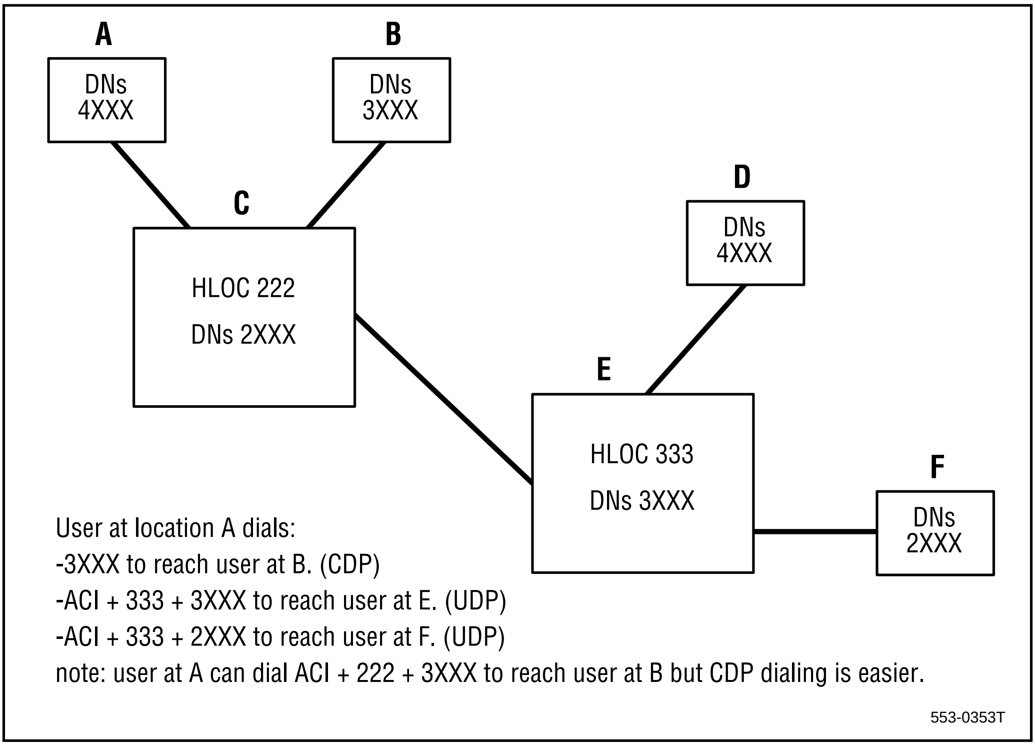
CDP groups or clusters can exist independently within a network. With GDP the number used to dial a station varies depending on the location of the calling station.

- A caller within one cluster of CDP-equipped switches dials a station on another switch within the same cluster by dialing a CDP DN beginning with a Distant Steering Code.
- Callers outside the CDP cluster must dial the location code followed by the Directory Number. Each switch in the cluster can have its own location code or all switches on one cluster can share the same location code.

  When the location code is translated as a home location code (HLOC) by the node in the cluster, it is deleted and the call is sent to the proper switch using DN translation (internal call) or CDP translation (for Local and Distant Steering Code calls). The maximum number of digits for either the location code and Directory Number, the Local Steering Code and Directory Number, or the Distant Steering Code and Directory Number cannot exceed 10.

Flexible Numbering Plan

Figure 29 - Group Dialing Plan



Special number treatments

When FNP is enabled, an outgoing trunk is not seized immediately upon completion of dialing.

Trunk seizure occurs when the expected number of digits have been dialed, the Fast Connect (octothorpe or #) key has been pressed, or the interdigit timer expires.

FLEN maximum for SPNs is 24 digits as of Release 22.

A Free Special Number Screening (FSNS) capability screens a 1- to 11-digit Special Number (SPN) allowing or denying 3-digit codes that follow the SPN. An FSNS table is applied to a route in a route list index to allow or deny calls to that 3-digit code on the route.

FSNS screening is increased to a maximum of 22 digits in Release 22.

Flexible Numbering Plan

Alternate Routing Remote Number

You translate Special Numbers and direct them to route lists. If you want certain digits that users dial *after* the SPN to be treated differently, you can route them in a different way from the other calls to that SPN.

Example:

Assume there is a network outside North America where there are TIE trunks between switches A and B in two different cities. Users at switch A call a public network number in the other city by dialing AC1 + SPN + the remaining digits. There is a digit manipulation index (DMI) assigned to the TIE trunks that is appropriate for these public network calls. The DMI manipulates the dialed digits to ensure the call is outpulsed to the other switch, beginning with the access code for the COT trunks at the other switch. The other switch receives the call, accesses a COT trunk and sends the call to the public network.

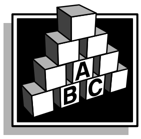
A problem arises when users at switch A dial calls in the public network format when they call users at switch B. (They should have dialed AC1 + LOC + XXXX.) Therefore, the DMI assigned to the TIE trunk route in the route list is not appropriate for these particular calls. The call is sent out to the public network from switch B and ends up coming back into switch B as an incoming call. This call occupies two public network trunks at switch B, needlessly. For the most efficient use of trunks at switch B, it is best if switch A outpulses extension numbers only on the TIE trunks for these calls, not the public network numbers.

With the Alternate Routing Remote Number capability you can translate the SPN at switch A and send the switch B public network numbers that follow the SPN to a different route list index (an Alternate Routing Route List). When these particular calls are directed to an appropriate route list, the appropriate DMI manipulates the digits to outpulse only extension numbers on the TIE trunks. This results in more efficient use of trunks at switch B.

The Alternate Routing Remote Number capability only applies to SPNs. You cannot use ARRN when you translate any other type of code.

There are many ways to use the ARRN capability. For another example, refer to the *Networking* NTPs in the Flexible Numbering Plan section.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

- With FNP the maximum number of route-list blocks and digit-manipulation tables allowed is 1000 to support Global Networking and Universal Numbering Plan requirements. You can have up to 100 entries in a route list index when FNP is installed.
- Choose a length for FLEN equal to the length of the translated code plus the greatest number of digits that will be recognized or restricted following that code. That way call processing will not begin until all the digits that should be analyzed are dialed first.
- If you use FNP in North America, beware that SPNs 0, 00, 01, 011, 411, 611, 911, 800, and 1800 are affected by a prompt (INPL) in the translation table that is NO by default.

  INPL = NO means that call termination occurs immediately after the SPN is entered but no further digits are outpulsed if they are dialed. This is good for 911 calls that should be processed as quickly as possible.

  Digits dialed in excess of the FLEN are outpulsed (unless the INPL prompt is NO in the translation table. This applies to certain SPNs (0, 01, etc.)

  Program the response as YES where more digits will be dialed after the SPN and you want them to be outpulsed after # is dialed or when the NIT timer expires. Do this for 011 calls overseas.

Flexible Numbering Plan

- For Trunk Steering Code calls there is a FLEN setting. If you do not want calls with fewer digits than the FLEN parameter to be processed you can activate the Inhibit Time out Handling feature. This will prevent calls with fewer digits than FLEN from being processed when the NIT timer expires.
- When you translate steering codes in ISDN networks, you can program whether you want to send the LSC or the LOC programmed in the Customer Data Block to the far end on the D-channel or just a DN. Decide for each DSC what will be best for the users at the far end to see on their displays. If they do not have NARS, there is no point in sending a LOC.

Tips

Vacant number routing

Some network planners prefer not to route calls by default when the system cannot translate them. If you use VNR you may end up with calls that are not routed as efficiently as possible, users making calls that you do not want them to make, and improper routes being used. Some planners prefer to translate every authorized call instead of using VNR. They give Intercept treatment to users who dial untranslated codes.

ESN and ISDN and VNR

For nodes using in-band, non-ISDN signaling, the VNR setting in the terminating node’s Customer Data Block determines whether or not to use VNR. For nodes connected by trunks using ISDN out-of-band signaling, the VNR setting in the originating node’s Customer Data Block determines whether or not to use VNR.

FNP Package Enhancement

With Release 23 (and upissues to Release 21 and 22), systems with the FNP software package can be configured with the FNP option turned on or off on a customer group by customer group basis.

On an existing system with FNP programming already configured, if the option is being turned off, all FNP -related data must be removed prior to turning off the FNP option for the entire customer group.

Features such as Vacant Number Routing (VNR) must be turned off when FNP is turned off. However, the FLEN settings do not have to be removed when FNP is turned off. If the FNP is reactivated later, the FLEN settings are reactivated again, automatically.

When removing FNP data from a customer group:

- Set VNR to NO in that customer group.
- Set Free Special Number Screening Indexes to 0 in the Route List blocks for calls to Special Numbers.
- Ensure the maximum Route List and Digit Manipulation index numbers do not exceed 256, the maximum for NARS.
- Ensure AC1 and AC2 are not longer than two digits.
- Ensure the maximum for the Free Special Number Screening Indexes is 0.

What to have ready

The following checklist summarizes the steps you should take before implementing the basic FNP feature and/or the optional related features associated with the basic feature.

Table 36
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Decide if you want flexible NARS- or CDP-type dialing plans or a combination of both. If not, turn off the FNP option for the customer group.
✔		Read the information on NARS and CDP in this book and the NTPs.
✔		Decide on your FLEN parameters.

Flexible Numbering Plan

Table 36
Checklist (Continued)

Basic	Optional	Preparation
✔		Decide if you will use VNR. What system will do the routing? What RLI will that system use? Is that RLI appropriate for all vacant number calls, and all users?
✔		Will you allow transferrable DNs? Will you use VNR for the programming?
✔		Will you assign location codes to all sites or will switches in a cluster share one location code?
✔		Are there certain SPN calls that require an ARRN?
		— continued —
✔		Are there certain codes you want to allow or deny on certain routes?
	✔	In North America, do you want extra digits outpulsed after each of the nine SPNs specified?
	✔	Do you want to Inhibit Time out Handling when fewer than the FLEN digits are dialed for TSC calls?
✔		Configure VNR in the proper Customer Data Block if you have ISDN or non-ISDN signaling.

R2MFC

Multifrequency Compelled (MFC) signaling uses the R2MFC standard developed by the international committee formerly known as the CCITT. Standard CCITT R2 protocols recommended for CO/PSTN operation are slightly modified for DID/TIE trunk operation.

A Meridian 1 uses the MFC signaling protocol to allow a Meridian 1 to exchange information with:

- another Meridian 1 on a TIE trunk
- a Central Office on a DID trunk

Called address digits are transmitted using MFC signaling in addition to the status and category of the calling and called parties.

The basic R2MFC protocol used by early Meridian 1 software releases was modified to provide R2MFC services in China, India and the Commonwealth of Independent States. The features developed for these countries are called:

- China Number 1 Signaling
- R2MFC signaling for India
- Multifrequency Shuttle for Commonwealth of Independent States

There is information on these three capabilities later in this section.

Basic configuration

An optional software package and special MFC sender/receiver circuit cards are required on a Meridian 1 using this protocol.

Multifrequency Compelled Signaling is based on Forward and Backward signals.

Forward signals consist of signals transmitted from the originating end to the terminating end. These are the dialed address digits identifying the called party and signals identifying the category of the calling party. For example, the calling party category may indicate the caller is restricted. Each Forward signal is steadily maintained until acknowledged by a Backward signal.

Backward signals consist of signals transmitted from the terminating end to the originating end. Backward signals respond to Forward signals and identify the status of the called party which may be busy, for example. When a backward signal is received, the Forward signal is removed, which in turn forces the Backward signal to be removed as well. This “compelled” sequence is repeated until the protocol is complete and the call is established. Backward signals can also be sent in pulse form without the prior reception of a Forward signal.

R2MFC signals are programmable in terms of frequency combinations. Each MFC route is associated with the R2MFC signal functions that it requires to operate properly with the system at the other end.

MFC incoming trunk connected to MFC outgoing trunk

When a Meridian 1 acts as a tandem switch between two MFC routes, as soon as a call is received, an MFC Sender/Receiver card is attached to the incoming call and a Backward signal is outpulsed to the originating end. When the trunk route access code is recognized and an outgoing trunk is successfully seized, one of two treatments will occur:

- End-to-End Signaling – This applies if both routes are using the same R2MFC data block or different data blocks with the same End-to-End signaling code. A Send Next Digit signal is returned to the originating Meridian 1, the MFC Sender/Receiver is released, and a speech path is set up between the outgoing and incoming trunks. End-to-End Signaling proceeds through this speech path.
- Buffered Signaling – This applies if the R2MFC data blocks are different for the two routes. Forward signals received from the originating switch must be translated into the signals the terminating end requires. Two MFC Sender/Receivers are required for this type of tandem connection.

R2 Modification

If you want to suppress the sending of a Send Next Digit signal, you can set a parameter that suppresses it after a programmable number of digits have been sent. This is useful after the digits in a complete call have been sent. This allows the COMPLETE signal to be sent instead.

Calling Number Identification (CNI)

The CNI capability allows the Meridian 1 to request and receive the Calling Party Number from the originating end (CO/PSTN). This applies to DID and incoming MFC trunks. Refer to *Improving feature performance*, China Number 1 Signaling Enhancements on page 313.

Semi-compelled MFC and Calling Number Identification Changes

In Release 18, there were two changes made to the way R2MFC signaling operated on the Meridian 1. They were:

- reduction in signaling times because of MFC Backward signals that are sent for 150 milliseconds instead of being compelled to terminate because of a Forward signal. This is very useful when transmitting with satellite connections.
- an option to end the Calling Number Identification transmission when a NEXT signal is received. The system switches to called number transmission.

Backward signal suppression

This feature allows the suppression of Backward signals under error conditions. If this feature is active, the Meridian 1 will not abort the call under error conditions. The trunk will be held up until the CO/PSTN switch times out and disconnects.

Alternative configurations

Multifrequency Signaling for Socotel

This signaling protocol allows a Meridian 1 to communicate with a Public Exchange switch using Socotel signaling on DID/DOD trunks. The DOD numbering plan is divided into two parts:

- calls to other locations which use a six-digit plan
- calls to special services which use a two digit plan

The switch uses the first digit to distinguish between the numbers. The digits are outpulsed as soon as the user dials them. If a very short time is required between digits, use the Delay Digit Outpulsing feature.

R2 MFC Signaling DID/DTMF DOD

In some countries the Central Offices do not provide MFC signaling on DOD trunks. On the Meridian 1, prior to Release 21, you need two trunks for every DID/DOD trunk pair where the DID trunk uses MFC signaling and the DOD trunk uses Dual Tone Multifrequency signaling. With Release 21, the Meridian 1 can handle incoming DID calls with R2MFC signaling and outgoing DOD calls with DTMF register signaling. Because of this capability you need fewer trunks. Only analog and DTI2 trunks are supported.

R2 MFC Signaling 1.5 Mbps Digital Trunk Interface

With Release 21, R2MFC signaling is supported on I.5 Mbps DTI trunks. Prior to Release 21, the only trunks where MFC signaling could be used were analog or 2.0Mbps DTI trunks. The R2MFC DID/Dual tone Multifrequency (DTMF) DOD feature is not supported.

China Number 1 Signaling

The features associated with China Number 1 Signaling have been designed to interface with the special needs of the Chinese Public Network, however, many of them can be used elsewhere, particularly those that do not involve an external operator.

The trunks must be fully supervised analog trunks or 2.0Mbps digital DID/DOD trunks.

Calling Number Identification (CNI) on Outgoing Multifrequency Compelled Signaling

CNI can be applied to outgoing DOD trunks. Category codes can be assigned to telephones. Both the category and CNI digits are sent by the Meridian 1 to the PSTN switch upon its request.

Called Party Control

This provides the PSTN operator with disconnect control when a user dials a Special Service number identified in programming on the Meridian 1. After the caller talks to the operator, and the user goes on-hook, the telephone is placed on hold for a designated period of time (programmed on a trunk route basis). During this time period the operator can signal the telephone by sending a special operator signal instead of redialing the entire number. Analog telephones must have Permanent Hold allowed in order for this to work.

Called Party Control is not supported on calls involving more than one trunk.

R2MFC

Calling Party Control

Disconnect control can be given to the Calling Party. If a user places a call which has been extended by an external operator and the call has been answered, the call is put on hold for a programmable period of time, if the called party goes on-hook. If the called party goes off-hook during this time period, a speechpath between the called party and trunk is re-established.

For incoming calls, if the calling party goes on-hook, the call is disconnected as normal.

Calling Party Control is not supported on calls involving more than one trunk.

Flexible Feature Codes (FFCs)

The following features are accessible using FFCs to satisfy the Ministry of Electronic Industry requirement in China:

- Autodial
- Call Waiting
- Make Set Busy
- Multiple Wake-up

Autodial

A user can dial a number automatically from an analog (500/2500type) telephone by leaving the handset off-hook for a predefined amount of time. The Autodial number can be programmed in the system by a technician or it can be entered by the user from the telephone.

Call Waiting

A user can allow or deny the Call Waiting feature in the Class of Service of the telephone to meet changing needs through the day. There is a Flexible Feature Code for activating and another one for deactivating the feature. Printouts indicate the present status of the telephone.

Make Set Busy

A user may need to make a telephone appear to be busy to incoming calls. There is an FFC to activate the Make Set Busy feature and a code to deactivate it. Calls to the busy telephone are affected by features such as Hunting during the time the telephone has been made busy by the user.

Multiple Wake-up

A user may enter up to four times of the day when the telephone is to ring with a wake-up call. This feature can be used as a reminder service. There is a code for entering the times and a code for removing times that are no longer needed.

Active Feature Dial Tone

A user hears a distinctive dial tone when lifting the telephone handset if one of the following features is active:

- Do Not Disturb – attendant activated
- Make Set Busy – user activated

Flexible Timers

You can define customer-wide time-outs for the following:

- Digit Pause Timer
  - dial tone timeout – if no digits are dialed after the telephone is off-hook
  - inter-digit pause timeout between the first and second digits
  - inter-digit pause timeout after the second digit
- Delayed Answer Timer – if a telephone rings for longer than this timer, the ringing and ringback stop and the call is disconnected

Off-hook (Howler) Tone

The attendant can signal an analog (500/2500 type) telephone when it is off-hook and in line-lockout mode. The Howler Tone has a timeout setting.

R2MFC

KE Multifrequency Compelled Tandem Signaling

You can configure the Meridian 1 to send an H MFC Forward signal to the CO. The H indicates that the call is to be tandemed through it to the next CO. If the CO is a DMS-100, the H is not sent.

Malicious Call Trace Enhancement

A Meridian 1 can have Called Party Control on incoming calls when the user activates the Malicious Call Trace feature or when the MFC Idle Call Trace signal is sent.

If the MFC calling number identification digits were received, they will appear on the CDR record for the call.

Audible Alarm

An alarm sounds when an emergency call is dialed or when an incoming call is traced by a telephone user. Up to 100 numbers can be identified as emergency numbers.

Toll Call Identification

You can identify toll calls for MFC routes in the programming of a Meridian 1. You program Special Service numbers to be marked as toll calls within a Special Service List which is then assigned to the MFC route(s).

Toll Operator Call Back

When a call has been established by an external operator and the user goes on-hook, the call can be put on hold for a programmable amount of time waiting for the operator to signal the telephone. For supervised trunks, the Meridian 1 is programmed to assume that all Special Service calls require the Special Operator Signal to Call Back a telephone.

Toll Operator Call Back Enhancement

The operator can have access to Call Waiting or Camp-on when signaling a telephone on a Called Party Control call or Calling Party Control call on 2.0Mbps digital DID or DOD trunks. The operator can also ring the telephone continuously while the call back key is pressed.

Toll Operator Break-in

The toll operator can use this feature to break-in on a conversation and extend a call to one of the parties. The operator can call back the telephone when it goes on-hook without having to redial the number.

Vacant Number Announcement

An announcement can be given to a caller when:

- a number that does not exist (a vacant number) has been dialed
- a vacant office has been dialed
- congestion is encountered

Multifrequency Shuttle for Commonwealth of Independent States (CIS)

This feature set is also referred to as the CIS Multifrequency (MF) Shuttle. The Multifrequency Shuttle Signaling protocol works on the CIS 2.0 Mbps Digital Trunk Interface. Release 23 software is required.

The CIS MF Shuttle uses a combination of two of six tones for Multifrequency Signaling between exchange connections. This reduces call set up time compared to the earlier Dial Pulse CIS DTI2 configuration.

CIS MF Shuttle supports the following features:

- Multifrequency Shuttle signaling
- Dial Tone to the calling party, after the Meridian 1 seizes an outgoing CIS trunk
- Buffered Dial Pulse signaling
- Dial Pulse signaling digit collection
- Downloadable Dial Pulse speed and Make/Break ratio
- CIS digital trunks signaling (outgoing, incoming toll, and incoming local calls)
- Automatic Number Information (ANI) transmission on request from the Central Office (CO)

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- Cyclic Redundancy Check (CRC) multiframe format (allowed optionally)
- A-law/µ-law conversion
- Expansion of the call types recognition mechanism based on the Special Service List (SSL) with the addition of a call type - Special Service Unanswered Calls
- Unanswered free special service calls
- Special disconnect procedure (two-way release) providing Malicious Call Trace in CIS telephone network
- CIS transmission plan
- Downloading of the required firmware mode per loop-limited ordinary DTI2 or CIS DTI2
- Periodic Pulse Metering (PPM) when working in non-CIS mode
- Continuous Pulsing Detection (CPD) when working in non-CIS mode

The existing limitations of the CIS-specific Digital Trunk Interface cards still apply to CIS MF Shuttle. Therefore:

- data in Automatic Number Identification (ANI) always refers to the first originator of the call. Thus, when a call is transferred the information in the ANI message does not correspond with the DN and ANI category of the telephone to which the call is transferred.
- on outgoing local calls, there is a 700 ms delay in the “Answer” signal recognition before the call is established. This delay is in addition to standard Meridian 1 answer validation timing.
- data calls are not supported.
- Toll Operator Break-In/Trunk Offer abilities are not supported.
- Toll Operator Manual Ringing capability is not supported.
- Overlap signaling is not supported on outgoing MF Shuttle calls.
- Dial Tone for incoming trunks is not supported.
- CIS MF Shuttle cannot be used for Virtual Network Services.

Sending and receiving digits

The functionality of Special Service List (SSL) tables has been expanded with the CIS MF Shuttle feature. The Number of Digits entry in the SSL table determines the number of digits which should be collected before the seizure of an outgoing CIS MFS trunk or received by an incoming CIS MFS trunk. This entry in the SSL table is used in the configuration of ENBLOCK or Flexible DN Size SSL tables. These tables are used as follows:

- When the outgoing trunk is idle, CIS MF Shuttle uses the ENBLOCK SSL table for ENBLOCK dialing.
- For incoming calls, the Flexible DN Size SSL table is used.

ENBLOCK dialing

For outgoing CIS MF Shuttle trunks, the ENBLOCK Special Service List (SSL) table is used to collect digits for optimal dialing operation. The ENBLOCK SSL table uses the number of digits dialed to determine the End of Dialing. It is recommended that *all* the possible numbers that can be dialed through the outgoing CIS MF Shuttle trunk be defined in the ENBLOCK SSL table. If an extra digit is dialed that has not been defined, the CIS MF Shuttle protocol considers the call a failure.

Outgoing toll calls should be defined in the SSL table. ENBLOCK signaling for outgoing toll calls remains active until all the toll access digits are dialed. The Special Service Digits field of the ENBLOCK SSL table should *not* include the outgoing CIS MF Shuttle trunk access code.

Fixed or Flexible DN size

For incoming CIS MF Shuttle trunks, the DN size is defined as either fixed or flexible. The DN size is an important aspect of CIS MF Shuttle configuration because it determines the number of digits that are expected from the CIS CO for incoming CIS MF Shuttle routes. Therefore, entering the DN size is obligatory for all calls which originate from an incoming CIS MF Shuttle trunk.

When the number of digits expected from the CIS CO is constant, and does not vary between different call types, then the DN size for the incoming CIS MF Shuttle route is fixed. Fixed DN size is defined for each route. For example, if it is known that for each incoming CIS MF Shuttle call the CIS CO dials four digits, then the fixed DN size would be defined as four in the Route Data Block(s).

When the number of digits expected from the CIS CO varies from call to call, the DN size for the incoming CIS MF Shuttle route is flexible. If this is the case, it is necessary to define a separate DN size for each tandem direction. The flexible DN size is defined in relation to a DN prefix of up to four digits of the DN dialed by the CIS CO. Flexible DNs are defined using the Flexible DN size SSL table. The table contains information on the number of DNs associated with each DN prefix. A DN prefix should be as short as possible while being explicit about the DN size. For example, if there is a tandem outgoing route with the access code “966” and it is the only DN that begins with the digit “9,” then the DN prefix should be the single digit “9.” If there is more than one DN type beginning with the same digit, one of the DNs should be defined using the single digit as a prefix and all the others should be defined using longer DN prefixes that allow for distinction between DN types.

CIS MF Shuttle interacts with NARS

Network Alternate Route Selection (NARS) has its own ENBLOCK dialing processing. Outgoing local CIS MF Shuttle trunks also require and perform ENBLOCK dialing. Thus, for outgoing NARS calls through CIS MF Shuttle trunks, the following conditions must be met:

- NARS and FNP must be configured with the Flexible Length value (FLEN) equal to the maximum possible length of dialed numbers.
- Inhibit Timeout Handling (ITOH) must be set to NO to allow a call to be attempted after the NARS Interdigit Number (NIT) timer has expired (even if fewer digits than the FLEN value have been dialed).
- The ENBLOCK SSL table should include only the DNs which will be sent to the CIS Central Office (CO), DNs which are outpulsed after the NARS translation of the dialed number.

For incoming CIS MF Shuttle calls, the NARS call processing is activated only after all the dialed digits are received from the CIS CO. Include inserted digits in the SSL table for an incoming route that is programmed for digit insertion in the Route Data Block.

When the Fixed DN size is used to define the expected number of digits, there is no interaction with NARS.

When the Flexible DN size feature is used to define the expected number of digits, the Flexible DN size SSL table must be defined to include DNs which are received from CIS CO. Do not include DNs which are generated after the NARS translation in the Flexible DN size SSL table.

CIS ANI Reception

The Commonwealth of Independent States (CIS) Automatic Number Identification (ANI) Reception feature in X11 Release 24 allows the Meridian 1 to receive the Automatic Number Information (ANI) from a calling party connected to a CIS Central Office (CO) on incoming local calls. The CIS Public Telephone Network does not provide ANI information on incoming toll calls.

The ANI digits received from the CIS CO are used by the Meridian 1 in the same way as the R2MFC Calling Number Identification (CNI) digits in the following ways:

- they are tandemed as the Calling Line Identification (CLID) to the Integrated Services Digital Network (ISDN) gateway, or the Originating Line Identifier (OLI) to the Digital Private Network Signalling System (DPNSS1) gateway, and to the Basic Rate Interface (BRI) gateway
- they are mapped into the R2MFC CNI as follows: all the ANI digits, except the ANI Calling Party Category Code (CAC), are used to compose the CNI. The ANI CAC is converted to the MFC CNI CAC according to the CAC Conversion Tables described in the section on the CIS Gateway.
- they appear on Meridian 1 proprietary telephone displays and on attendant console displays
- they are stored in Call Detail Records
- they are sent through the Meridian Link and the ICCM link using the fields dedicated for the R2MFC CNI digits

ANI Reception is performed in one of two ways:

- an ANI request is issued automatically by the incoming local CIS DTI2 trunk during the call setup
- an ANI request is issued on the incoming local CIS DTI2 trunk upon a manual request from a Meridian 1 proprietary telephone with a display or from the attendant console. The request to receive the ANI information is made by pressing a Display key or by pressing the Malicious Call Trace (MCT) key. An ANI request can also be made from analog 500/2500 telephones using the Flexible Feature Code (FFC) assigned for Malicious Call Trace. An MCT record prints out on the TTY configured for MCT records.

Automatic ANI request

The automatic ANI request is sent by the CDTI2 card to the CIS CO before the incoming local call is answered. If the incoming trunk operates in the decadic, or Dial Pulse (DP) mode, the ANI request is sent to the CIS CO after all dialed digits have been collected from the CIS CO. If the trunk operates in the MF Shuttle mode, the ANI request is sent after the end of the MF Shuttle dialing. The ANI digits are uploaded to the Meridian 1.

The Automatic ANI request option may be used only in conjunction with the DN Size Feature. DN Size Flexible (use the SSL tables) or Fixed should be defined for the incoming CIS DTI2 DID route before setting the automatic ANI option to “Yes”.

The translation of the dialed number which is received from the CIS CO is postponed until the CDTI2 card informs Meridian 1 that the ANI digits have been received.

If the ANI reception report does not arrive from the card, the call is treated after the ANI timer expires. The treatment for a call that failed to provide the automatic ANI is configured in the Route Data Block. The choices are:

- drop the call
- route the call
- intercept the call to a DN

CAC display

The CAC digit is separated from the ANI digits by a minus sign. In the Route Data Block you configure the position of the CAC in the display. The options are:

- CAC appears before the ANI digits
- CAC appears after the ANI digits
- CAC does not appear

ANI Digits in CDR

The ANI digits are placed in the CDR at the place intended for the R2MFC CNI digits. The CAC can be stored in the CDR together with the ANI digits. The presentation of the CAC in the CDR is configured in the Route Data Block as follows:

- placed before the ANI digits
- placed after the ANI digits
- not placed in the CDR

CIS Gateways Enhancements

This feature, introduced in X11 Release 24, enhances gateways involving CIS trunks. The CIS trunks include: CIS three-wire analog trunks and CIS digital trunk interface (DP and MF Shuttle).

The enhancements are split into four parts:

- mapping the R2MFC CNI to the ANI of CIS trunks
  - mapping between the MFC CAC and the CIS CAC in both directions
- mapping the CLID (ISDN) and the OLI (DPNSS1) to the ANI of CIS trunks
- mapping ANI to ANI in CIS/CIS gateway
- solving the ambiguity of the CAC configuration on telephones for systems using both MFC trunks and CIS trunks. The telephones can have a CAC code for calls on MFC trunks and a different one for the CIS trunks, if required.

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CIS ANI digits manipulation

CIS ANI Digits Manipulation feature, in X11 Release 24, allows the ANI to be built in a more flexible way when the call is originated from a telephone and from a trunk route.

ANI Definition

The Automatic Number Identification (ANI) information is a string of digits sent to the Central Office (CO), which it uses to identify the calling subscriber for billing purposes, Malicious Call Trace (MCT) purposes, and for immediate information about the subscriber when reaching some vital service such as fire brigade, emergency medical care, or law enforcement officials. The ANI information is sent over the speech path whenever the CO requests it.

On a Meridian 1, the ANI is sent on the following types of CIS trunks:

- CIS three-wire analog trunk
- CIS digital trunk interface Dial Pulse (DP) and Multi-Frequency Shuttle (MFS)

Before Release 24, the ANI is a sequence of eight digits composed of:

- the number to be used for billing purposes. It consists of three digits for the CO local exchange code (LEC) to which the PBX is connected plus four digits for the subscriber number (ANI DN)
- the 1-digit subscriber category (CAC), which gives the level of services the user can access

Pre-Release 24 operation

The ANI DN is:

- the primary DN, if the originator is a set with Class of Service DNAA
- the LDN0 programmed in the Customer Data Block, if the originator is an attendant with Class of Service DNAA
- the ANDN configured on the outgoing route, if the originator is a telephone or an attendant with Class of Service DNAD

- part of CLID/OLI determined by Remote DN Length (RDNL) (least significant digits), if the originator is an ISDN route (MCDN, QSIG, DPNSS, BRIT) and if RDNL is not set to 0.
- the ANDN configured on the incoming ISDN route, if the originator is an ISDN route and if RDNL is set to 0 but ANDN is configured
- the ANDN configured on the incoming non-ISDN route (ANDN is configured)
- the ANDN configured on the outgoing route, if the originator is an ISDN route with RDNL set to 0 and ANDN is not configured
- the ANDN configured on the outgoing route, if the originator is a non-ISDN route with ANDN not configured

The ANI DN, together with the local exchange code (LEC) shall always comprise 7 digits. If it is less, additional digit(s), defined in the Route Data Block for the outgoing trunk route, are inserted between the subscriber category and the least significant digit of the extension number. The LEC sent to the CO is always the LEC programmed for the CIS outgoing route.

If ANI DN + LEC together comprise more than seven digits, then the least significant ANI DN digit(s) are not used and are omitted.

ANI Digits Manipulation feature functionality (Release 24)

The ANI may be built in two ways with the enhancements introduced by the CIS ANI Digits Manipulation feature as described in the list below:

1. The ANI may be built the same manner as it was before Release 24, with some modifications listed below:

   – The length of the ANI information built by the software is configurable on a per route basis and may reach 15 digits (for LEC+ANI DN).

   – The part of the ANI DN to be truncated (in case the truncation cannot be avoided) is the beginning (that is, the most significant digits).

- The system has the option to work without LEC. You do not have to program an LEC in the Route Data Block anymore.
- In the case of the LEC+ANI DN length being shorter than the required length, additional digit(s) can be added at the beginning of the ANI DN (between the ANI DN and the LEC), in compliance with the CIS standards. You can insert more than one digit.

2. Optionally, the ANI data may be retrieved from entries configured in the Customer Data Block. This is the same type of enhancement provided for the ISDN CLID by the Release 22 feature ISDN CLID enhancements. It provides much more flexibility in building the ANI. An ANI entry number can be assigned to each PBX telephone, BRI telephone and proprietary telephone DN key.

CIS Dial Tone Detection

The Commonwealth of Independent States (CIS) Toll Dial Tone Detection feature, in X11 Release 24, allows the Meridian 1 to detect dial tone from a CIS Toll Central Office (CO) on outgoing toll calls. When received, the tone indicates that the CIS CO is prepared to collect dial pulse (decadic) digits from the Meridian 1 for outgoing toll calls. This prevents the Meridian 1 from outpulsing digits before the Toll CO is ready to receive them. The feature is implemented only for CIS DTI2 trunks.

The CIS DTD feature introduces CIS toll outpulsing criteria, which define conditions that need to be satisfied to allow the Meridian 1 to start outpulsing the decadic digits on the outgoing Toll CIS DTI2 call.

The criterion is composed as a combination of two events, dial tone detection and ANI interaction. It may include only dial tone detection, only ANI Interaction, dial tone or ANI, dial tone and ANI. The criterion is defined in the Route Data Block. The detection is performed by the CIS DTI2 cards and when the criterion is satisfied, the Meridian 1 receives a report from the card. The Meridian 1 postpones the outpulsing of the digits until it receives the report.

If the report does not arrive before the ANI time-out timer for the route expires, you can choose one of two options:

- continue the outpulsing
- disconnect the call and provide busy tone to the user

Outgoing toll CIS DTI2 calls can be made using a direct connection to the Toll CO or an indirect connection through a local CO.

R2MFC signaling for India

In order to comply with the Department of Telecommunications (DOT) requirements in India, enhancements to basic MFC signaling are offered in Release 23. The capabilities are:

- support India’s R2 modified MFC signaling
- work with both analog and digital (DTI2) interfaces

Indian R2 modification

- Indian COs support only 10 of the 15 signals supported by the Meridian 1.
- The terminating party is allowed to send the Send Category signal while the address is being exchanged.
- To indicate the called party is free from calls, a B.6 signal (Called Line Free with Metering Signal) is sent. The Meridian 1 can be configured to send this signal.
- To make a request for the originating party to restart sending the dialed digits from the beginning, an A.2 signal must be sent.
- The maximum number of CNI digits can be 16 in the MFC Route Data Block. The DOT recommendations suggest 10 digits as a maximum for current requirements.
- When a Meridian 1 is a tandem switch between a PBX and a CO, only buffered tandeming is supported, not End-to-End signaling.

R2MFC Timer Control

When a Meridian 1 communicates with a 2EAX GTE Central Office switch with R2MFC signaling, a timer can be configured in the Route Data Block for inter-digit timing. This prevents the CO from

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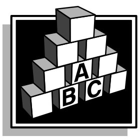
disconnecting the call prematurely when calls with more than seven digits are outpulsed. This timer is used while digits are outpulsing. The timer expires when all of the called address digits are outpulsed. An R2MFC End-of-Dialing signal is sent to the far end switch. This prompts the CO to request the category digits of the calling party. The call is processed effectively.

R2MFC DID/ DTMF DOD trunks

Refer to information on page 765 in the section called Interworking and gateways, for more information on this functionality.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Terminating R2MFC DID/TIE calls on a CDN

With Release 21 software, R2MFC DID or TIE trunk calls can terminate on an Automatic Call Distribution Control DN (CDN). CDNs are used when configuring incoming call Customer Controlled Routing (CCR) queues.

Meridian 1 acts as tandem switch

MFC incoming to Non-MFC outgoing

The incoming MFC digits are buffered by the Meridian 1, to be sent out on the Non-MFC trunk later. Each received digit is immediately acknowledged by a Send Next Digit signal. The outgoing digits are sent in the format programmed for the outgoing route such as Dial Pulse or Digitone. It is recommended that you set the MFC timer on the incoming route at a higher value than the End-of Dialing timer on the outgoing route so the call does not get disconnected if the call is not answered prior to the MFC timer expiring.

Non-MFC incoming to MFC outgoing

This type of call is treated as though the call originated from a telephone at the tandem Meridian 1.

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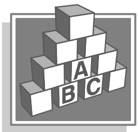
China Number 1 Signaling Enhancements



The following enhancements are offered:

- delayed outpulsing is allowed on outgoing CO or DOD trunks to ensure that when digits begin outpulsing there are no long delays between digits. If the following features are used, Delay Digit Outpulsing is denied:
  - Autodial
  - Last Number Redial
  - Speed Call
  - Stored Number Redial
- a request for Calling Number Identification (CNI) is only allowed if the incoming trunk has CNA programmed in the Class of Service and the telephone receiving the call has Malicious Call Trace allowed
- the tone called Busy Tone to Calling Party is given to the last party still off-hook when one party goes on-hook and there is Calling or Called Party Control in operation

Process Notification for Networked Calls (Release 24)

In an existing R2MFC Meridian 1 network, when an outgoing call hops through more than two R2MFC steps, it experiences a delay. The caller sometimes drops the call, since it sounds as though the call is not proceeding.

The Process Notification for Networked Calls feature informs the user that the call is progressing during this delay, using a recorded announcement (RAN) message or a tone. You determine the number of times the message is to repeat and the interval between the announcements. If you are using the tone option, you determine the tone and cadence by programming the Flexible Tones and Cadences software package.

This feature is enabled for each route supporting MFC signaling. You can implement a tone on some routes and announcements on others. You can implement different delay timers on the routes.

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The Process Notification tone or message is given when the first digit on an MFC trunk is outpulsed, and if the user waits for more than 6 seconds (default configuration) between the digits or after all the digits.

If the user continues to dial, the announcement is stopped and the notification delay timer is restarted. The digit is outpulsed.

Process Notification is abruptly stopped when one of the following events occurs:

- the call is completed successfully
- the call is dropped due to a busy destination telephone
- one of the legs of the R2MFC trunks is busy
- the MFC timed out waiting for a response
- the caller hangs up

If the RAN trunks are busy and unavailable to give a message, then the tone is given.

When an attendant receives a Process Notification announcement and puts the call on hold or presses another loop key, the Process Notification announcement is stopped.

If at any time a special dial tone is given by another feature, the Process Notification treatment will not be provided. During the Process Notification announcement, if a special dial tone has to be given, preference is given to the special dial tone and the Process Notification announcement is removed immediately.

What to have ready

The following checklists summarize the steps you should take before implementing the various forms of R2MFC signaling.

Table 37
Checklist China Number 1 Signaling

Basic	Optional	Preparation
✔		Determine if Calling Party or Called Party Control is required.
✔		Decide which, if any, Flexible Feature Codes you need.
✔		Decide if users should hear a distinctive dial tone when DND or MSB features are active.
✔		Talk to your PSTN representative about Flexible Timers, Tandem Signaling and Delay Digit Outpulsing that may be required.
✔		Decide if you want to implement Howler Tone for telephones in line-lockout.
✔		If your users need Malicious Call Trace, arrange the proper Control arrangements, alarms and CDR process to handle it.
✔		List the Special Service numbers that are to be identified and printed as toll calls.
✔		Determine if the toll operator will place calls for users. Discuss the arrangements you want to make for the operator to extend calls back to users.
✔		Decide if announcements are required for intercept treatments. Decide which treatments will have announcements and what the announcement(s) will say.

R2MFC

Table 38
Checklist for CIS

Basic	Optional	Preparation
✔		Arrange to define *all* the possible numbers that users will dial through the outgoing CIS MF Shuttle trunks in the ENBLOCK SSL table.
✔		Arrange to define the DN size as either fixed or flexible for incoming CIS MF Shuttle trunks.
	✔	If you are using NARS and FNP software: Configure the Flexible Length value (FLEN) value equal to the maximum possible length of dialed numbers. Arrange to set the Inhibit Timeout Handling (ITOH) to NO to allow a call to be attempted after the NARS Interdigit Number (NIT) timer has expired (even if fewer digits than the FLEN value have been dialed).
✔		Decide if you want ANI reception to be automatic or by manual request.
	✔	If users will display the ANI manually, decide if you want them to use the Display key or the MCT key.
	✔	If you are allowing analog 500/2500 telephone users to activate MCT to print out the ANI, assign a FFC.
✔		Decide on the format you prefer for the ANI display on telephones and in the CDR.
✔		Decide what is to happen to an incoming call that has no ANI associated with it.
✔		Decide on your outgoing ANI format requirements.
✔		Find out if your Toll CO requires Dial Tone Detection to process your users' toll calls more efficiently.

R2MFC

Table 39
Checklist for India

Basic	Optional	Preparation
✔		Ensure only 10 of the 15 MFC signals are defined.
✔		If the Meridian1 is a tandem switch, ensure End-to-End signaling is set up to prevent End-to-End tandeming.

Table 40
Checklist for R2MFC

Basic	Optional	Preparation
	✔	If your users experience outgoing call processing delays and hang up before call completion, decide if you want to implement a RAN or tone during the delay.

R2MFC

Automatic Number Identification (ANI)

Purpose

Systems can be configured to use Automatic Number Identification software and Multifrequency (MF) sender cards to transmit the caller’s number along with the called number. As a result, toll calls can be billed to the person making the calls. The network service provider collects the information and sends it to you as part of their billing procedure. The trunks required for this functionality are called CAMA (Central Automatic Message Accounting) trunks.

Basic configuration

Prior to Release 23, the maximum number of digits that could be sent out by the MF sender card was 16. With Release 23, the maximum has increased to 32 digits.

ANI signaling

The supported forms of trunk signaling to the Central Office are:

- E&M
- DX
- loop

Three basic methods are used:

- Bell – interfaces the Meridian 1 to:
  - Bell system TOPS (Traffic Operator Position System)
  - Bell system TSPS (Traffic Service Position System)
  - Bell system CAMA office

Automatic Number Identification (ANI)

  - Strowger Automatic Toll Ticketing (SATT) systems 57, 59, 62, and 70A
  - Stromberg Carlson Ticketing Systems

- NT400 – Nortel NT400 Ticketing System
- NT500 – Nortel NT500 Ticketing System

Discuss the form of trunk signaling and the methods required with your network services provider.

Called number information

The called number always includes the Directory Number (DN) dialed which is usually seven or ten digits. The toll access code (usually 0 or 1) may or may not be outpulsed, depending on the method being used, chosen from the list above. Additional control digits such as preparatory digits and end-of-pulsing digits are also sent using MF signaling.

In Release 12 an enhancement was developed to deal with calls to 00 (overseas operator). These calls may be dialed with 00 alone or with 00 followed by more digits.

Calling number information

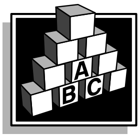
The calling number is sent using MF signaling. It comprises:

- an ANI Listed Directory Number (LDN) that identifies the customer to the toll office
- the DN of the telephone, the ANI attendant number, or the ANI TIE trunk group number
- preparatory signals, end-of-pulsing signals, and other auxiliary signals (for signals such as class marks and category digits)

Automatic Number Identification (ANI)

Improving feature performance

Improving feature performance



Route Selection (RS-ANI)

RS-ANI automatically routes toll calls through specified trunks to toll offices and routes local calls to local exchange offices. Digits following the ANI access code are translated by the ANI software and calls are routed accordingly.

Automatic Number Identification on DTI

- In Release 14, ANI functionality became possible on digital trunks to the Central Office. ISDN Primary Rate Interface trunks could also be configured for ANI functionality. The three basic signaling methods (Bell, NT400 and NT500) are supported. With Release 15, DTMF ANI on DTI is provided in North America by a feature called In-band ANI. This is used in ACD (non-ESN, non-ISDN) environments for incoming call identification.

Control tips

Conditionally Unrestricted Service (CUN)

In order to ensure that users make toll calls on the ANI trunk groups only, and are therefore billed for their calls, assign the CUN access restriction level in their Class of Service.

Conditionally Unrestricted users are allowed:

- to make and receive internal calls
- access to external calls by using ANI (Automatic Number Identification) trunk groups only

Restriction

New Flexible Code Restriction only applies to users with a TLD (Toll Denied) access restriction level. Users with a CUN access restriction level are treated as unrestricted.

Automatic Number Identification (ANI)

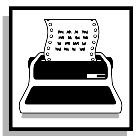
Administration tips

- Decide if any users should be allowed access to non-ANI trunks for their toll calls. Assign them an unrestricted access restriction level. Decide if any users should be toll-denied or even more restricted. The remaining users, probably the majority, will be Conditionally Unrestricted.
- Decide what intercept treatment you want users to experience if they misdial toll calls. You may want to use a recorded announcement initially to tell them how to dial properly.

Training tips

- Ensure users are aware of the proper way to dial an ANI toll call. Train them on the proper access code to use.
- Train users on your billing procedures for toll calls.

Maintenance tips



- At installation good information about your Central Office machine type and signaling methods is required in order to configure your ANI trunks properly.

Automatic Number Identification (ANI)

What to have ready

The following checklist summarizes the steps you should take before implementing ANI.

Table 41
Checklist

Basic	Optional	Preparation
✔		Find out if your network service provider offers CAMA trunks.
✔		Decide if you want to use other services that identify callers for billing purposes instead of ANI such as Call Detail Recording on site.
✔		Familiarize yourself with the billing procedures used by your network service provider.
✔		Discuss your trunk types and the required signaling methods with your network service provider.
✔		Arrange your DNs and ANI LDN properly so a seven-digit number for the caller is always sent.
✔		Decide if you want to route toll calls automatically using RS-ANI.
✔		Determine what access restrictions to assign to telephones.
✔		Decide on the intercept treatment you want to use at installation and afterwards, if there is to be a change.
✔		Arrange training on the ANI dialing plan and billing procedures.

Automatic Number Identification (ANI)

Introduction to ISDN

Integrated Services Digital Networks (ISDN) use software and hardware that comply with standards developed and approved by various telecommunications committees in different regions of the world. The term “standard”, when used in the context of ISDN, does not always mean common.

The International Telecommunication Union - Telecom (ITU-T), formerly the CCITT, is a global body that approves and publishes standards. Within that committee is the International Standards Organization (ISO). In North America there is Bellcore (for the Regional Bell Operating Companies or Telco’s) and users’ forums called ANSI (the American National Standards Institute) and the National ISDN Users Forum (NIUF). In Europe there is the European Telecommunications Standards Institute (ETSI) and the European Computer Manufacturer’s Association (ECMA).

Goals

The goals of ISDN are:

- to provide global communications networks that use interfaces and signaling based on accepted standards
- to provide high quality, rapid, economical and easy transmission of voice, data, video and image
- to provide a wide range of services

Benefits

ISDN provides integrated access to both voice and data circuits and packet switching networks, eliminating the need for separate access for each type of network service. ISDN can provide a broad range of services that were not possible with earlier forms of network-related hardware and software.

Introduction to ISDN

ISDN networks

All forms of ISDN networks build on the two basic standards of ISDN which are:

- Primary Rate Interface (PRI)
- Basic Rate Interface (BRI)

The enhancements to and regional variations of these two basic standards are:

- EuroISDN
- QSIG
- National ISDN (NI-1, NI-2, NI-3)

Each one of the enhancements and variations listed above is presented in a section of its own in this book.

What is MCDN?

ISDN functionality began to evolve on the Meridian 1 when X11 Release 12 software was being developed. The ISDN standards that were laid down at that time were in a very formative state. Nortel Networks introduced Primary Rate Interface (PRI) and Basic Rate Interface (BRI) connectivities using a proprietary interpretation of these early standards (CCITT Q.931). The Meridian 1 interfaces evolved, along with interfaces on other Nortel Networks products under the *Meridian Customer Defined Network (MCDN)* initiative. Cross-product development was coordinated and compatibility testing was done with various kinds of switches including the DMS and SL-100.

Interworking between different private networks or interfacing to non-Nortel Networks PBXs, would have required more development of the proprietary MCDN protocol in order to be compatible with these other networks. In the case of supplementary services, there might not have been any interworking, using the MSDN platform. Therefore, networking software compliant with different, current ISDN standards was introduced.

Introduction to ISDN

Since those early days, several ISDN standards have emerged. Standards known as EuroISDN, NI-2, and QSIG all rolled out after Release 12.

The European Computer Manufacturer's Association (ECMA) defined an ISDN protocol called QSIG that has been adopted by the European Telecommunications Standards Institute (ETSI) and the International Standards Institute (ISO). In order to offer ISDN connectivity and services on switches in QSIG networks, Nortel Networks developed ISDN BRI and PRI connectivity based on the QSIG standard and continues to evolve these connectivities.

However, Nortel Networks also continues to develop software compliant with its proprietary MCDN standards.

In the NTP manuals when the term PRI and BRI are used with no other qualifiers or descriptions such as EuroISDN or QSIG, assume the PRI or BRI being discussed is based on the MCDN standards.

There are also proprietary forms of networking supported by Nortel Networks that offer similar functionality to ISDN. They are:

- Digital Access Signaling System No. 2 (DASS2)
- Digital Private Network Signaling System No. 1 (DPNSS1)
- ISDN Signaling Link (ISL)
- Analog Private Network Signaling System (APNSS)
- Virtual Network Service (VNS)
- ISDN Semi Permanent Connections for Australia (ISPC)
- Information Notification Service for Japan

There is a series of sections in this book called *Services and functions* that deal with services and functions that can be used in many different types of ISDN networks. Features specific to a certain form of ISDN are included in the section for that particular topic.

Introduction to ISDN

This section covers the basic information about ISDN that you need to know before you read the more specific information related to each configuration of ISDN and the information that follows about services and functions.

Basic configuration

Integrated Services Digital Networks (ISDN) do not replace Electronic Switched Networks (ESN). ISDN networks build upon a platform of ESN programming. ISDN features and functionality add to and enhance what was offered by ESN.

The *About networking* section of this book contains an Implementation diagram that will help you understand the layered approach to implementation that is necessary.

ISDN switches must have the instructions that Network Alternate Route Selection (NARS) or Coordinated Dialing Plan (CDP) software provides in order to route calls intelligently. There is more information on the exact pre-requisites required for each form of ISDN later in this book.

Before you install any form of ISDN functionality on your network, it would be wise to familiarize yourself with the information on NARS and CDP in this book and in the NTPs. Refer to the section called *About this guide* for a list of reference sources you can use.

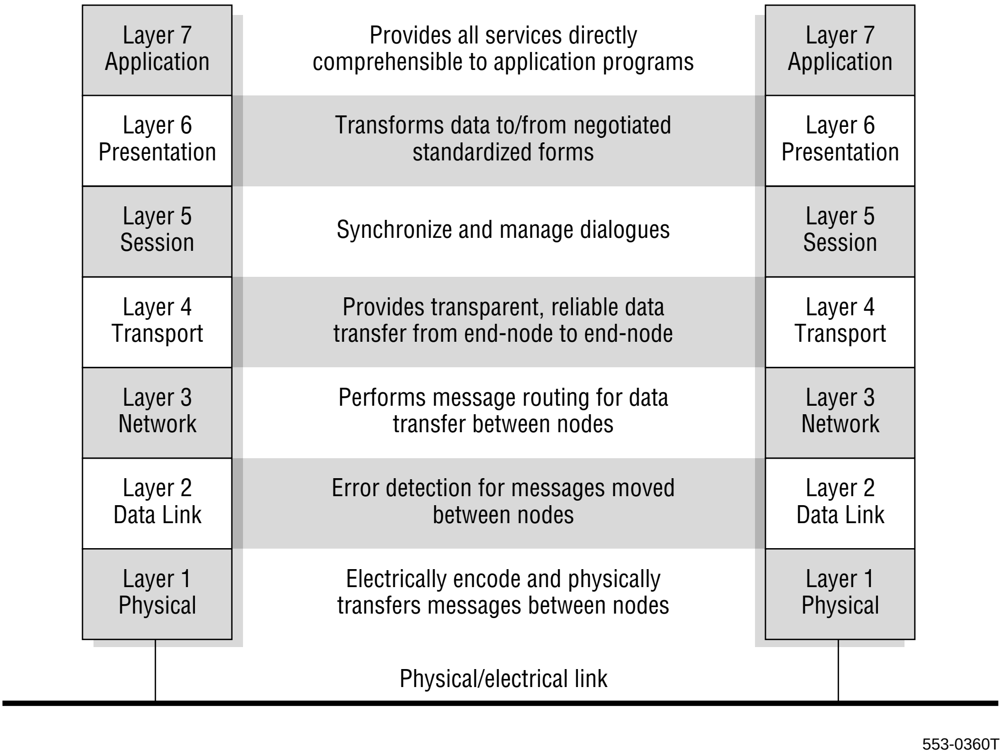
You will achieve the greatest efficiencies from your ISDN network if the pre-requisite ESN programming has been done properly, in advance.

Introduction to ISDN

OSI layers

The International Standards Organization (ISO) developed a seven-layer reference model that was adopted by the CCITT committee. The model lays out a framework for Open System Interconnection (OSI) that can be used for describing, developing and standardizing ISDN protocols.

Figure 30 - OSI layers diagram



Each layer of the protocol builds on the services provided by the layer below it. This layered approach separates the complex protocols into a series of more easily managed parts. Because of this approach, each part of the protocol can be modified without affecting the protocols in other layers.

There is detailed information about the OSI model in the *Networking Feature Document*.

Introduction to ISDN

The power of ISDN

In order to transmit calls and other information between switches, some form of signaling is required. ISDN networks dedicate the signaling functionality to a separate trunk or channel. This is called “out-of-band” signaling. On non-ISDN networks, signaling takes place using the same trunks or channels that the voice or data calls use. This is known as in-band signaling. Out-of-band signaling makes transmission faster and more efficient.

Out-of-band signaling also allows “clear channels” to be used for voice and data. Clear channels are those that only handle voice and data and no signaling. Therefore the entire bandwidth available in a channel can be devoted to the voice or data transmission alone. This allows high speed transmission of voice, data and signaling. As a result, capabilities such as video transmission are better.

The channels used for voice/data connections and signaling have names.

B-channels are the channels carrying (bearing) the voice and data calls. Each one has a data rate of 64 kbps.

D-channels are the channels used for signaling. Each one has a data rate of 64 kbps (if Primary Rate Interface is used) or 16 kbps (if Basic Rate Interface is used). Information related to call set up, call release and feature activation is carried on the D-channel.

Primary Rate Interface (PRI)

This configuration of ISDN uses T1 facilities in North America and E1 facilities in International markets. You must install hardware and software when you implement PRI.

PRI can be used in both public and private network applications.

PRI

T1 provides 24 channels. PRI configures them as 23 B-channels and one D-channel.

Introduction to ISDN

PRI2

E1 provides 32 channels. PRI2 configures them as 30 B-channels and one D-channel. The remaining channel is a framing channel.

Interfaces

Primary Rate Interface links can extend from a Meridian 1 to the following kinds of systems:

- Meridian 1
- host computers
- SL-100
- ISO QSIG
- ETSI QSIG
- DMS-100
- DMS-250
- Magellan and Meridian Passport
- 4ESS
- 5ESS
- SYS-12
- AXE-10
- 1TR6
- INS1500 (Japan D70)
- NUMERIS VN3
- SWISSNET 2
- NEAX-61
- Euro ISDN Central Offices (ETSI NET-5)
- Asia-Pacific Central Offices in Australia, Hong Kong, New Zealand, Singapore, Thailand, Indonesia, China, Japan, Malaysia, India, Taiwan, and Philippines

Introduction to ISDN

The list of connectivities continues to expand with each new release.

Enhanced functionality

Backup D-channel

You can configure your system so that if the primary D-channel fails, there is a secondary D-channel to take over the signaling function. This secondary D-channel can be provided by a second PRI link or a BRI link (in Europe). The D70 Central Office in Japan does not support the backup D-channel functionality.

The T309 timer and D-channel failure

Prior to Release 17, all active calls on a PRI link to the public network were cleared when a D-channel failed and were then re-established. As of Release 17, the timer provides a 30 second (non-programmable) wait interval after a D-channel failure before the B-channels are cleared. When the D-channel re-establishes, and before the timer expires, the two sides can communicate to preserve the B-channels. The T309 timer is not used for Meridian 1 to Meridian 1 connections. Active calls remain established until the users release the calls. If the network side does not have a T309 timer, then calls are cleared when the D-channel is re-established.

nB+D

If you need more than one TI or E1 facility to handle the connections between two switches, you can use the nB+D configuration.

The facilities controlled by one D-channel must extend between the same two switches.

Up to 16 T1 or E1 facilities can be controlled by one primary D-channel. Japan (D70) nB+D is an exception to this. The D70 Central Office will only support 9 links for one D-channel.

With 15 or fewer T1 facilities, the backup D-channel is not required but it is recommended. Using the maximum number of 16 T1 carriers between two switches requires the backup D-channel functionality.

16 T1 facilities, with 24 channels for each T1, provide a maximum of 382 B-channels ([16 x 24] – 2 D-channels) and 480 B-channels for E1 ([16 x 30]). For Japan D70 COs, the maximum is 215 B-channels ([9 x 24] – 1 D-channel).

Without a backup D-channel, 15 T1 facilities provide a maximum of 359 B-channels ([15 x 24] – 1 D-channel).

With X11 Release 24, a trunk route can have a maximum of 510 members. (The previous maximum was 254.) This means you can have a D-channel with 382 B-channels, a Backup D-channel and a second D-channel with 128 B-channels operating over one trunk route.

Integrated Trunk Access (ITA)

This Primary Rate Interface capability allows a T1 facility to include channels that have out-of-band signaling and others that have in-band signaling (A+B bit signaling). This capability is only provided in North America.

Integrated Services Access (ISA)

The Meridian 1 can send and understand call set up messages that specify the type of route a channel will be used for, on a call by call basis. ISA is sometimes called Call by Call Service and Dynamic Channel Assignment.

ISA eliminates the need for dedicated B-channels for each service route. The channels are assigned to routes on a call-by-call basis and used as needed. However, you can continue to have dedicated channels in routes, if you need them. This capability is only provided in North America.

Network Time Synchronization

This feature is used in countries outside of North America to ensure that all time stamps in an MCDN network are synchronized from one source. One switch is designated as the Master for this purpose. One switch requests Time Synchronization messages from another switch using the D-channel. There is a way to compensate for time zone differences where it will be important for features such as Automatic Wake-up and Centralized CDR.

Introduction to ISDN

B-Channel Overload Control

If a system experiences periods of extremely high incoming call traffic on PRI routes, you can implement B-Channel Overload Control. As of Release 23, on North American systems, you can program a delay timer on each route. The timer determines how many milliseconds the system will wait before releasing a call to a busy telephone or ACD queue. This delays a new call from being presented so quickly on the same B-channel as the previous call that encountered the busy. This can prevent the system from experiencing dial tone delays on systems where there is a very high volume of incoming calls to busy telephones or to ACD queues that become busy.

You can implement this feature on routes that interface to the following North American Central Offices:

- DMS 100
- DMS 250
- SL-100
- 4ESS
- 5ESS
- NI-2
- QSIG

MCDN Alternate Routing (when network is busy)

When a caller tries to establish a call across an MCDN network and all of the routing alternatives at one of the nodes are congested (busy) or there are other reasons that a node is congested, this feature allows a signal to be sent back from the congested node to the preceding node. The preceding node can reroute the call to the final destination, bypassing the congested node. The bypass occurs by sending the call out of the preceding node on an alternate route in the route list index for calls to the dialed destination.

Transit nodes can reroute calls or send a message back to a preceding switch. Originating switches always attempt to reroute calls.

Introduction to ISDN

The signal sent back to the preceding node from a congested switch is called a Call Clearing Message. When a transit node receives an incoming Call Clearing Message from the outgoing trunk to a congested switch, it routes the call based on the programming associated with that route in the route list.

Alternate Routing programming options on the route list entry can be one of the following:

- NRR: No rerouting, pass the Call Clearing Message back to the previous node, if at a transit node; give the user a congestion tone, if at an originating node
- RRO: Reroute, pass the Call Clearing Message back to the previous node, if the node is a transit switch; if the node is an originating switch, the system looks for an alternate route
- RRA: Reroute the call at a transit node or originating node. If no available trunk is found at the transit node, the Call Clearing Message is sent to the preceding node. If no trunk is found at the originating switch, congestion tone is sent to the user.

When the transit node tries alternate routing, the channel to the congested node is released and a new call is generated on the alternate route. If the first alternate route is busy, the next one in the route list index for the dialed destination is tried until an available one is found. If no trunk is available, a Call Clearing Message with congestion information is sent to the preceding node.

Congestion tone is given to the user by the originating switch, if no alternate route can be found to bypass the congestion.

The Cause Information Element carries information about the cause of congestion. In an MCDN network, MCDN Alternate Routing works if the first Call Clearing Message contains a received cause value programmed to activate the feature.

Introduction to ISDN

Acceptable cause values include:

- cause 3 *No route to destination*
- cause 27 *Destination is out of service*
- cause 34 *No channel/circuit available*
- cause 38 *Network out of order*
- cause 41 *Temporary Failure*
- cause 42 *Congestion*

MCDN Alternate Routing is not supported on MCDN BRI trunks.

UDP or CDP dialing plans are required for MCDN Alternate Routing to work. Calls made using trunk route access codes are not offered this functionality.

NARS and CDP software normally retry calls (not due to MCDN Alternate Routing) when the following cause values are received:

- 6 *Channel unacceptable*
- 44 *Requested circuit/channel not available*
- 82 *Identified channel does not exist*

Alternate routing of calls using this feature disregards the Iset on the route lists. All routes are scanned in the route list when MCDN Alternate Routing is attempting to route a call. If the call is routed on an expensive route at a transit node, the Expensive Route Warning Tone is not given. The user's NCOS and Facility Restriction Level (FRL) still control the trunk routes upon which they can make calls.

In a network where there could be blockage at several nodes and several attempts to alternately route calls, be careful not to allow so much alternate routing that the call set up timers expire before the call is established. Consider choosing the RRO or NRR option at some of the nodes, instead of RRA at all of them.

The Calling Line ID for the alternately routed call is the same as for the original call.

MCDN Alternate Routing takes precedence over the Network Drop Back Busy and Off Hook Queuing feature and the Remote Virtual Queuing feature. However, regular Off Hook Queuing takes precedence over MCDN Alternate Routing.

Basic Rate Interface (BRI)

This form of ISDN can be used to connect terminals (lines) to a system. There are BRI telephones for example. BRI can also be used in a trunk configuration to connect one switch to another (not supported in North America).

BRI connections have 2B-channels and one D-channel. Each B-channel has a rate of 64 kbps and the D-channel carries signals at a rate of 16 kbps. The D-channel can also carry packet data.

There is a EuroISDN BRI standard called ETSI NET-3 and a standard for Asia-Pacific called INS NET-64. This Asia-Pacific standard applies to Japan D70 BRI as well.

Variations on PRI or BRI

Euro ISDN

This form of ISDN is based on a memorandum of understanding signed by 20 European countries. It is a common European ETSI standard. There are National Applications Documents for country by country deployment. It covers PRI and BRI Central Office access and BRI terminal support.

QSIG

Q Reference Signaling Point Basic complies with the OSI model layer 3 requirement for support of circuit switched call control at the “Q” reference point between Private Telecommunications Network Exchanges (PTNXs) connected with the Private Telecommunications Network (PTN). The protocol has been adopted by ETSI and the ISO. This standard applies to both PRI and BRI (trunk application). The switches involved are peers (such as two PBXs, two Centrex systems, or a PBX and a Centrex).

Introduction to ISDN

National ISDN (NI-1, NI-2, NI-3)

In North America, Bellcore developed standards based on the National ISDN Users Forum (NIUF) Recommendations. The National Institute of Standards and Technology sponsored the NIUF.

Bellcore's goals for National ISDN were:

- standard interface configurations
- full complement of standard services and applications
- stable platform for development
- service uniformity
- portability of terminals

NI-1 deals with BRI; NI-2 deals with PRI standard interfaces; NI-3 deals with PRI services.

Proprietary non-ISDN configurations

Digital Access Signaling System No. 2 (DASS2)

British Telecom's DASS2 is the signaling protocol defined for PBX access to the public ISDN, in the United Kingdom. DASS2 is defined in terms of the OSI model developed by the ISO. The protocols in layers 1, 2 and 3 are defined.

E1 facilities are used. Each E1 link must be connected to an ISDN local public exchange (Central Office). Some channels on the link may be configured as TIE trunks. This will allow a DPNSS1 connection to be established between PBXs through the public network.

Digital Private Network Signaling System No. 1 (DPNSS1)

British Telecom's DPNSS1 is the signaling protocol defined for PBX private network connections. The Meridian 1 is compliant.

E1 facilities are used in DPNSS1 networks. E1 trunks are not required for the proprietary form of ISDN called APNSS. There is information later in this section about APNSS.

Introduction to ISDN

ISDN Signaling Link (ISL)

ISL is a Nortel Networks proprietary configuration for providing ISDN-type functionality in a Meridian 1 private network.

ISL can be set up when you have one or more of the following conditions:

- digital TIE trunks and Digital Trunk Interface (DTI or DTI2) cards that you are not upgrading to PRI cards
- analog TIE trunks between two switches in a private network
- digital TIE trunks and Primary Rate Interface (PRI or PRI2) cards that you program as if they were not PRI because you want to use some of the capabilities that ISL provides

ISL can provide all of the functionality of PRI (except for 64 kbps clear channels on systems using T1 carriers that require the Revert to Conventional Signaling feature).

Analog Private Network Signaling System (APNSS)

APNSS supports only PBX to PBX connectivity. The two PBXs do not have to be the same type of system.

APNSS is configured on a route basis. Each trunk in the route is associated with a D-channel that carries DPNSS1 messages. This D-channel cannot also be used for DPNSS B-channels. It must be dedicated to the APNSS trunks or channels. It can control up to 30 APNSS B-channels.

The B-channels for APNSS are normally carried over analog two or four wire E&M trunk circuits, or AC15 trunks. However, digital (DTI2) TIE B-channels can also be used for APNSS.

The D-channel may be carried over a 64 kbps digital channel, or an analog trunk using modem equipment. Normally, the D-channel is run using leased-line modems, but may also be connected using dial-up modems, a 500 line card and any trunk circuit.

Virtual channels are required for some supplementary services and are programmed on an unused loop within the Meridian 1.

Introduction to ISDN

Virtual Network Services (VNS)

VNS is an enhancement of the ISL interface. The enhancement allows the voice and the signaling in one call to take two different paths. The two PBXs are connected together with a D-channel that controls public network PRI channels as if they are private network TIE trunks. Network applications can be implemented as if the PBXs are connected with PRI TIE trunks. The public network channels can also be used for public network calls.

ISDN Semi Permanent Connections for Australia (ISPC)

In Australia, fractional PRI is not available. ISPC allows subscription on a B-channel basis. You can replace up to 30 leased lines to different sites with individual nailed-up public network B-channels that are configured as if they are private network TIE trunks. The nailed-up B-channels between Meridian 1 switches are used by the ISL feature as if they are TIE trunks. MCDN features can be supported on these trunks as if they were private ISL trunks.

An ISL call involves two trunks at each switch. The physical trunk is a COT trunk at the originating end and a DID trunk at the terminating end. The logical trunk links the two Meridian 1 switches together using a TIE route with phantom trunks (on a phantom DTI2 loop).

The phantom DTI2 loop uses up configurable TNs on the system. If your system is close to capacity, assess the impact this will have on your system.

The nailed-up B-channels and the associated D-channel must extend between the same two Meridian 1 switches.

A channel to the Central Office may be configured as a D-channel for ISPC operation.

ISPC B-channels cannot be used for the VNS feature.

BRI trunks cannot be used. AXE-10 does not support this capability.

> ***Note:*** Refer to the sections on PRI and BRI in this book for more information about country-specific ISDN functionality in Australia.

Introduction to ISDN

Information Notification Service for Japan (Release 24)

The Information Notification Service for Japan (INS-J) feature allows a local exchange in Japan to extract the calling line identification information received on Japanese COT or DID analog trunks and to deliver it to subscribers' terminals or trunks that have display capability and customer oriented applications.

The INS-J feature introduces a new circuit card, (the NT5D39 DXUT-J card). The DXUT-J is a Digital Signaling Processor-based Extended Universal Trunk card for the Japanese market. The DXUT-J collects the Frequency Shift Key (FSK)-format INS-J information sent by the CO and sends it to the Meridian 1 software. The DXUT-J also supports the Busy Tone Detection for Japan that is available on the previously available EXUT-J card.

On an incoming call with INS-J, the Meridian 1 extracts information such as: Calling Party Number, Calling Party Name, Called Party Number, Date and Time, and, if applicable, Reason for absence of Calling Party Number/Calling Party Name. This information is passed on to the terminating party, which can be:

- a trunk
- a terminal or
- an application.

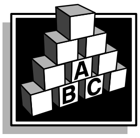
The Meridian 1 software extracts the Calling Party Number, Called Party Number, Calling Party Name, and Date and Time information. The call termination follows the existing procedure. For example, if the call is from an incoming CO trunk, it terminates at the attendant or where designated by the system's database; if the call is a DID call, the Meridian 1 software extracts the information from the INS-J messages and terminates the call accordingly.

Gateways

A gateway is a means of connecting two different signaling schemes. The Meridian 1 supports a number of gateways for networks using a variety of protocols. Refer to the section on *Interworking and gateways* in this book for a complete list of what is supported.

Introduction to ISDN

Improving feature performance



The information that follows may make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

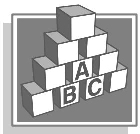
Traffic study

System Traffic Study Option 9 (TFS009) can be a useful tool in improving the performance of your ISDN network. The data that is produced by this study can tell you:

- the number of times when no request output message buffer was available. You may want to reconfigure your buffers.
- the number of PRI layer-3 protocol errors since the last traffic report. This can help you explain why ISDN-related features may have experienced problems during that time.
- the number of times the D-channel was down and the length of time it was down. You will know when a D-channel fails, without the traffic data, but the study provides data you will need in discussions with maintenance people and your system supplier.
- if you have an MSDL card for your D-channel the data will indicate
  - average incoming link usage as a percentage of the link capacity
  - average outgoing link usage as a percentage of the link capacity
  - peak incoming link usage as a percentage of the link capacity over a 5 second period
  - peak outgoing link usage as a percentage of the link capacity over a 5 second period

You should also run studies on your CPU to find out if the system real time capacity can handle the extra load that ISDN signaling will put onto the CPU. Run traffic study TFS004 to look at that data.

Introduction to ISDN



If you will be adding more DTMF incoming trunks, look at studies TFS002 and TFS003 to find out if you need to order more Tone and Digit Switches and Digitone Receiver cards to handle the DTMF tones that your switch will be receiving.

Run these studies before and after cutover.

Recorded Announcement (RAN) for Calls Diverted to External Trunks

This feature provides a recorded announcement to a caller when a call is being forwarded to an external AXE-10 or EuroISDN Public Exchange on a DTI, DTI2, PRI, PRI2, analog or BRI trunk. It can be used to warn the caller that the call set up time may be longer than normal. DPNSS1 networks cannot support this feature and it is not offered in North America.

The outgoing trunk must be COT, not incoming and not used for Radio Paging or data only. The trunk can also be a DID trunk configured for outgoing calls or incoming and outgoing calls.

If the caller attempts to dial extra digits as the RAN is being given, the digits will be lost.

If the outgoing route has Expensive Route Warning tone associated with it in the route list, the tone will be given before the RAN is given.

Introduction to ISDN

Channel negotiation

This feature operates on connections between the Meridian 1 and Central Offices that conform to the following protocols:

- AXE-10
- SYS-12
- 1TR6
- QSIG
- Japan D70 (INS NET-64)
- NEAX-61
- Numeris
- EuroISDN (in some countries)
- NI-2

Channel negotiation allows call setup to proceed even when a B-channel chosen by one end is unacceptable to the receiving switch. A search for an alternative channel that is acceptable to both ends can take place. Call clearing takes place if the channel is unacceptable to the receiving switch and Channel Negotiation is not enabled. In order for the feature to work, the B-channels must not be shared between customer groups.

Control tips

- ISA can reduce the number of channels you need and help you control your expenses, due to the following factors:
  – A common pool of channels is used dynamically. You take advantage of economies of scale. Large groups of trunks are more efficient than several smaller groups with the same total number of trunks.

Introduction to ISDN

- Call processing is done at 64 kbps.
- If busy hours on different trunk routes at your site occur at different times of the day, you will not need as many trunks when channels are shared. Without ISA, dedicated channels can sit idle while other routes are overloaded.

◆ Before you decide how you want calls to alternately route during network congestion, think about the following factors:

- your billing system for calls that travel across your network. If calls from one switch use expensive routes at other switches when they alternately route, find out if your billing systems and processes support that.
- find out if you have provisioned the trunk routes at all locations to support alternately routed calls from other switches

Administration tips

◆ You cannot have non-ISDN and ISDN trunks in the same route.

◆ The Integrated Trunk Access (ITA) capability allows you to program channels in a T1 link as trunk types that are not supported by PRI yet. These non-ISDN, A+B bit channels can be in the same T1 link with channels controlled by a D-channel.

◆ Your network will only be as functional as its least capable switch. When you implement ISDN, you must upgrade each switch to a level that allows it to support the network-wide features and services you require. Each switch needs to have the software and hardware that allow it to deal with the messages it will be receiving from other switches in the network.

◆ Good coordination and planning is required in an ISDN network. Some examples follow.

- The D-channel(s) at each site must be configured for the types of messages to be received and the release of software of the switch at the other end.

Introduction to ISDN

- You must consider the impact on ESN parameters and programming when you upgrade or make changes to one system. Consider such things as: travelling NCOSs; digits to be outpulsed; the INAC prompt (so the correct NARS AC1 or AC2 code is inserted in front of incoming digits); special digit manipulation tables (used if the ISL link reverts to conventional signaling).
- You must decide which switch will act as a master for clocking. Public network switches are usually masters. The others connected to it will be slaves.
- If a Multi-Purpose Serial Data Link (MSDL) card is being used for the D-channel and if there is an MSDL card at the other end, you must specify that in programming.
- The method of line coding (B8ZS or AMI) must be compatible with the other end.
- Yellow alarm parameters for maintenance of the D-channel (DG2 or FDL) must be compatible.
- The Recovery to Prime D-channel capability must be activated at both ends in order to work. If both ends are programmed as NO for this capability, or not matched, the backup D-channel continues to be used even after the prime D-channel is working again. The prime D-channel does not recover. This forces a technician to manually switch back to the primary D-channel when it is stable.
- System IDs (SIDs), Private Network Identifiers (PNIs), Channel IDs (CHIDs) must be coordinated in the programming at any two connected switches. The people who program your system coordinate these numbers.
- You will need to know when the Central Office (local exchange) is upgraded, so you can adjust your programming. You may be able to take advantage of new public network features and capabilities. However, you may need to upgrade your system to do so.
- You must configure your D-channel for the type of Central Office it is connected to when there is a PRI connection.

– The INAC (Insert Network Access Code) prompt is used by a switch to be able to insert the correct NARS access code (AC1 or AC2) in front of digits in an incoming ISDN call. Insertion is based on the Call Type (CTYP) identifier it received in the D-channel signal preceding the call digits. You can change the CTYP you send to the other end. You may do this if you have a CDP switch connected to a NARS switch. You may want the CTYP to indicate a location code call is coming, instead of a CDP call, so the NARS switch will insert AC1 in front of the location code digits. There is more information on this in the section called *CLID and Name Display options*.

– Programmers should refer to the Administration section of the Networking NTPs for information on Connection parameters and Correlation tables for more information on the coordinated programming that is required.

◆ If you are using packet data transmission, you must dedicate channels on the PRI link to that function. Bear that in mind if you program ISA for the remaining channels.

◆ If you reserve a minimum number of channels for each service route on a PRI link configured for ISA, your Network Traffic study data will show those channels as in use for the entire study period.

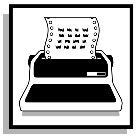
◆ If you would like to install a backup D-channel for PRI trunks to a DMS Central Office, your Meridian 1 must be equipped with Release 17 or later software and the DMS Central Office will need a minimum of BCS 31. You may also want to consider installing analog trunks to act as backup trunks in the event that the PRI trunks fail. Enter these trunks in route lists, to be used at all times for overflow or only for emergencies (with the Routing Control feature active.)

Introduction to ISDN

Training tips

- In training sessions, tell Meridian Mail users that the system will not be able to give the caller’s DN, if the call was not sent in on ISDN links. If a call was tandemed across many switches, there must be ISDN signaling all the way through in order for the user to hear the caller’s ID number in the envelope when they play a message or to use the Call Sender feature.

Maintenance tips



- Recovery to Primary D-channel – When the system is using its backup D-channel because the primary D-channel failed, you can program the system to switch back, automatically, to the primary D-channel when it returns to good working order. However, Nortel Networks does not recommend that you activate that capability. If you force a manual recovery to the primary D-channel instead, a technician can ensure that the primary D-channel is stable before the system uses it. If you have a volatile primary D-channel, your system could be switching back and forth from primary to secondary D-channels frequently.
- The Network Call Trace capability can be used by technicians to print records of calls as they are being set up on the network to find where troubles are occurring. Refer to the Appendixes for more information about testing and troubleshooting.
- Service messages – maintenance messages concerning the D-channel and the ISDN link between two switches can be activated. Talk to your system supplier about whether they recommend and support activating this capability for the switch connectivities you are using on your network.

What to have ready

The following checklist summarizes the steps you should take before implementing basic ISDN.

Table 42
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Ensure you have NARS or CDP (or both) and any other ESN packages you need programmed to your satisfaction.
✔		Find out what standards you must comply with in your region.
✔		Decide what features and applications you want to implement. Then choose the ISDN hardware or software that supports your requirements.
✔		Plan changes to your trunking so you achieve your goals with ISDN.
✔		Learn about the cards you will need for your trunks, D-channels and clocking requirements. Find out about any modems you will need.
✔		Discuss the impact the new hardware and software will have on your system.
✔		Plan to train users on new capabilities they will have.

Introduction to ISDN

Primary Rate Interface (PRI)

Primary Rate Interface (PRI) is configured using T1 facilities in North America and E1 facilities in International markets. Where T1 carriers are used, it is called PRI and where E1 is used, it is called PRI2. You must install software and the proper hardware when you implement either type of Primary Rate Interface.

PRI

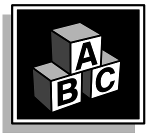
T1 carriers provide 24 digital channels. ISDN PRI configures them as 23 B-channels and one D-channel.

PRI2

E1 carriers provide 32 channels. PRI2 configures them as 30 B-channels and one D-channel. The remaining channel is a framing channel.

Primary Rate Interface (PRI)

Basic configuration



Refer to the section called *Introduction to ISDN* in this guide for basic information about ISDN, Primary Rate Interface and MCDN networks.

Primary Rate Interface can be configured in private and public network applications. Refer to the *Introduction to ISDN* section in this guide for a complete list of supported connectivities. For detailed information about Central Office connectivities outside North America refer to the section called *ISDN PRI Central Office Connectivity* in the Networking NTPs.

PRI uses “out-of-band” signaling. Out-of-band signaling makes transmission faster and more efficient. Out-of-band signaling also allows “clear channels” to be used for voice and data. Clear channels are those that only handle voice and data and no signaling. Therefore the entire bandwidth available in a channel can be devoted to the voice or data transmission alone. This allows high speed transmission of voice, data and signaling. Capabilities such as video transmission are better as a result.

The channels used for voice/data connections and signaling have names.

B-channels are the channels carrying (bearing) the voice and data calls. Each one has a data rate of 64 kbps.

D-channels are the channels used for signaling. Each one has a data rate of 64 kbps. Information related to call set up, call release and feature activation is carried on the D-channel.

Additional capabilities listed below are also possible with PRI:

- Backup D-channel
- nB+D
- Integrated Trunk Access (ITA)
- Integrated Services Access (ISA)
- Network Time Synchronization
- Channel negotiation
- MCDN Alternate Routing

Refer to the section called *Introduction to ISDN* for more information.

Country-specific PRI configurations

Japan

PRI connectivity is supported to Japan D70 interfaces. The standard configuration is 23B+D.

- The INS-NET64 specification states that Overlap sending and receiving states are not provided.
- If the network sends a call with a message stating “no channel is available or no channel is indicated”, the Meridian 1 will clear the call. A channel must be indicated for an incoming call to be processed by the Meridian 1.
- nB+D. D70 Central Offices only support up to 9 carriers for a single D-channel, so the maximum B-channels per D-channel is 215.
- Backup D-channels are not supported by the D70 interface.
- Only circuit mode connection is supported, not packet mode.
- The following features and capabilities are not supported at this time:
  - ISA (Call by Call Service)
  - FEX, WATS call types

- Network Ring Again
- Network Calling Party Name Display
- CLID Enhancements (Redirecting Number)
- Network Call Redirection, Network Call Forwarding, Network Call Forwarding No Answer
- Trunk Optimization/Anti-Tromboning
- ISDN Signaling Link (ISL)
- ESN Signaling on PRI (NCOS)

Australia and New Zealand

Connectivity to the NEAX-61 exchange office switch is done using PRI2 (30 B+D) E1 links. This PRI2 connectivity is the only one offered for Australia and New Zealand for X11 prior to Release 21. The following features are supported:

- Basic Call Service
- circuit mode and packet data on “nailed up” B-channel
- Calling Line Identification (public and private)
  - public: display of a CLID received from the public network
  - private: sending the CLID from the private network to the public network
- Overlap sending
- COT, DID, DOD and TIE trunk call types
- 64 kbps clear capability
- channel negotiation
- Flexible Numbering Plan
- nB+D
- PSTN 3-Party conferencing
- Malicious Call Trace

> ***Note:*** the NEAX-61 does not support private network dialing plans such as UDP location codes and CDP private numbering plans. ESN access code insertion can be performed.
>
> ***Note:*** refer to the information on APISDN below for the Release 21 PRI capabilities offered in New Zealand.

The following capabilities are unsupported:

- packet data on the D-channel
- partitioning the 30 B-channels into more than one route
- more than one type of trunk route in a PRA Group (an nB+D group of four PRI2 links to the CO). If the channels are configured as DID trunks in that PRA Group, then all channels must receive the same number of digits from the CO.
- CDR records when a conferee leaves a conference. Records are only produced when the conference is completely torn down, not when each party leaves the conference.

Asia Pacific (APISDN) (Australia, Hong Kong, Singapore, Thailand, New Zealand)

This protocol, supported as of Release 21, involves both PRI and BRI interfaces. PRI2 (2.0 Mbps) is used in all countries except Hong Kong, where PRI (1.544 Mbps) is used.

The following features are supported:

- Basic Call Service
- circuit switched voice and data on the B-channel
- Calling Line Identification and Restriction (public and private)
  - public: display of a CLID received from the public network
  - private: sending the CLID from the private network to the public network

Primary Rate Interface (PRI)

- COT, DID, DOD, and TIE trunk call types
- 64 kbps clear capability
- channel negotiation
- Flexible Numbering Plan

Australian APISDN supports Advice of Charge and Malicious Call Trace features.

New Zealand NEAX-61 APISDN supports Three Party Conference and Malicious Call Trace.

Overlap sending is supported for:

- Australia
- Hong Kong
- New Zealand
- Singapore
- Thailand

Overlap receiving is supported for Thailand.

nB+D connectivity is supported for the New Zealand interface. Up to 120 B-channels can be configured.

Backup D-channel is not supported.

BRI Trunk Access can be used as an overflow when PRI trunks are busy.

Asia Pacific (APISDN) (Indonesia, China, Japan, and Malaysia)

This protocol, supported as of Release 23, involves both PRI and BRI interfaces. PRI2 is used in all countries.

The following features are supported:

- Basic Call Service
- circuit switched voice and data on the B-channel - Only point-to-point data link connection is supported.
- Calling Line Identification Presentation (CLIP) and Restriction (CLIR) (public and private)
  - public: display of a CLID received from the public network
  - private: sending the CLID from the private network to the public network
- COT, DID, DOD trunk call types
- TIE trunk call types (Japan and Malaysia only)
- 64 kbps clear capability
- channel negotiation
- Enbloc Signaling using CDP steering code access

Overlap Sending is supported on Indonesian, Chinese, and Malaysian interfaces. Overlap Receiving is supported on the Indonesian and Chinese interfaces as optional. It is supported on the Malaysian interface.

Indonesia/China: Subaddressing - no checking is done by the M1 of either the Called Party Subaddress IE or the Calling Party Subaddress IE. These are passed along by a Meridian 1 at a tandem node from an Indonesia/China interface.

Indonesia: The supplementary services supported are: Connected Line Identification Presentation (COLP) and Connected Line Identification Restriction (COLR). Refer to page 630 for more information.

Malaysia: nB+D is supported for up to four PRI2 loops (120 B-channels).

In Japan the interface for Advice of Charge (at the end of the call) is supported.

Asia Pacific (APISDN) Connectivity Phase III – India, Taiwan, Philippines, Hong Kong, Singapore

In X11 Release 24, PRI2 and BRI connectivities are supported in India and the Philippines according to country-specific specifications. In India these specifications are a subset of the ITU-T Q.931 (White Book) standard. In the Philippines these specifications are a subset of the CCITT Q.931 (Blue Book) standard.

1.5 Mbps PRI and also BRI connectivities are supported in Taiwan. In Taiwan these specifications are a subset of the ITU-T Q.931 (White Book) standard.

Only circuit-switched calls are supported, not packet-switched calls.

A CDP call can access an Asia Pacific trunk. Neither the private plan nor the CDP numbering plan is supported by APISDN. These calls are converted to a plan and type supported by the public network. This applies to both the calling and called number plan and type.

Alternate routes are not possible for an APISDN Central Office connectivity, therefore NARS is not supported.

The following basic and supplementary features are supported in India, Taiwan, and the Philippines:

- Basic Call Service
- overlap sending and receiving for India
- overlap sending for Taiwan
- COT, DID, DOD, and TIE trunk call types
- 64 kbps clear bearer capability
- Calling Line Identification Presentation (CLIP)

- Calling Line Identification Restriction (CLIR)
- Connected Line Identification Presentation (COLP)
- Connected Line Identification Restriction (COLR)
- sub-addressing (supported at tandem)

Also, the earlier APISDN connectivities in Hong Kong and Singapore are enhanced. The updated connectivity in Singapore complies with the updated national ISDN specification issued by TAS. In Singapore the standard refers to the ITU-T Q.931 (White Book) standard. Hong Kong Telecom issued a supplementary specification which is a subset of the CCITT Q.931 (Blue Book) standard. The Release 24 connectivity complies with this specification.

The following supplementary features are supported in Hong Kong:

- non-associated (nB+D) signaling
- Backup D-channel
- 64 kbps unrestricted data call
- CLIP on attendant console and Meridian 1 CT2 Mobility Option
- CLIR message display (ANONYMOUS, OUT OF AREA)
- BRI voice capability
- PRI data capability
- sub-addressing for data call

The following changes are incorporated into the Singapore connectivity to comply with the country specifications:

- channel negotiation is no longer supported
- messages with Global Call Reference are supported only for PRI
- PROGRESS message is supported only for BRI

Primary Rate Interface (PRI)

Taiwan R1 Modified Signaling

With X11 Release 24, the Meridian 1 can support Taiwan R1 Modified Signaling (TWR1 MS) over DTI trunks, in compliance with the specifications for Digital Switching Equipment in Taiwan. With this signaling, the capability to carry CLID across the network is supported for local and national calls.

The TWR1 MS feature supports Taiwan R1 Modified Signaling over Taiwan R1 Direct Inward Dial (DID) or Direct Outward Dial (DOD) DTI trunks.

In a networking environment, the CLID is supported only for TWR1 calls that are tandemed over the following kinds of trunks:

- MCDN
- Taiwan ISDN
- QSIG

The TWR1 MS feature can operate in a stand-alone or network environment. In a stand-alone environment, the Meridian 1 connects directly to a Taiwan Public Exchange over Taiwan R1 Modified Signaling trunks. In a networking environment, the tandem Meridian 1 connects to a Taiwan Public Exchange over Taiwan R1 or Taiwan ISDN trunks, and to one or more other Meridian 1systems over MCDN, QSIG, Dual Tone Multi Frequency (DTMF), or Dial Pulse (DP) trunks.

Connectivity to the Private Exchange in the following ways is not supported:

- over Taiwan R1 Modified Signaling trunks
- over Digital Private Networking Signaling System No. 1 (DPNSS1) trunks
- over Level 2 Multi Frequency Compelled Signaling (R2 MFC) trunks

The Taiwan R1 trunk is a one-way trunk, either outgoing only or incoming only.

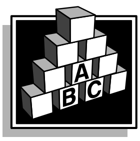
Taiwan R1 trunks use Multi Frequency (MF) signaling only. Dual Tone Multi Frequency (DTMF) signaling and Dial Pulse (DP) signaling are not supported.

Taiwan R1 trunks support Loop Dial Repeating Signaling (LDR) only.

Only DID/DOD and DTI Taiwan R1 trunks are supported for this feature.

Both NARS and trunk route access code dialing are supported for making TWR1 outgoing international calls. If NARS dialing is used for making outgoing international calls, a Special Number (SPN) of "002", (the Taiwan international call access code) must be translated.

Improving feature performance



The information that follows may make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Recovery to Primary D-channel

When the system is using its backup D-channel because the primary D-channel failed, you can program the system to switch back, automatically, to the primary D-channel when it returns to good working order. However, Nortel Networks does not recommend that you activate that capability. If you force a manual recovery to the primary D-channel instead, a technician can ensure that the primary D-channel is stable before the system uses it. If you have a volatile primary D-channel, your system could be switching back and forth from primary to secondary D-channels frequently.

Call Connection Restriction

With Release 14, restriction capabilities were introduced that allow you to control connections which would degrade transmission or network performance.

Primary Rate Interface (PRI)

When ISDN software is installed, you can place the following conditions on network call connections:

- no more than one trunk without disconnect supervision can be used in a call connection. This prevents trunk lock-up.
- controlled number of tandem nodes per call. The range is zero to thirty-one.
- limited PSTN connections. You can permit only a single connection or an unlimited number of PSTN connections.
- controlled number of μ-Law to A-Law conversions per call. The range is zero to thirty-one.
- limited number of satellite delays. The range is zero to five. If Network Signaling software is configured, this capability overrides the Satellite Link Control feature by making the number of satellite connections programmable.

Control tips

ISA can reduce the number of channels you need and help you control your expenses, due to the following factors:

- A common pool of channels is used dynamically. You take advantage of economies of scale. Large groups of trunks are more efficient than several smaller groups with the same total number of trunks.
- Call processing is done at 64 kbps.
- If busy hours on different trunk routes occur at different times of the day, you will not need as many trunks when channels are shared. Without ISA, dedicated channels can sit idle while other routes are overloaded.

Primary Rate Interface (PRI)

Administration tips

- You cannot have non-ISDN and ISDN trunks in the same route.
- The Integrated Trunk Access (ITA) capability allows you to program channels in an E1 or T1 link as trunk types that are not supported by PRI yet. These non-ISDN, A+B bit channels can be in the same T1 or E1 link with channels controlled by a D-channel.
- Your network will only be as functional as its least capable switch. When you implement ISDN you can no longer upgrade switches independently of the others. Each switch needs to have the software and hardware that allow it to deal with the messages it will be receiving from other switches in the network. Upgrade each switch to a level that allows it to support the network-wide features and services you require.
- Good coordination and planning is required in an ISDN network. Here are some examples:
  - The D-channel(s) at each site must be configured for the types of messages to be received and the release of software of the switch at the other end.
  - You must consider the impact on ESN parameters and programming when you upgrade or make changes to one system. Consider such things as: travelling NCOSs; digits to be outpulsed; the INAC prompt (so the correct NARS AC1 or AC2 code is inserted in front of incoming digits); special digit manipulation tables (used if the ISL link reverts to conventional signaling).
  - You must decide which switch will act as a master for clocking. Public network switches are usually masters. The others connected to it will be slaves.
  - If a Multi-Purpose SDI Link (MSDL) card is being used for the D-channel and if there is an MSDL card at the other end, you must specify that in programming.
  - The method of line coding (B8ZS or AMI) must be compatible with the other end.

Primary Rate Interface (PRI)

– Yellow alarm parameters for maintenance of the D-channel (DG2 or FDL) must be compatible.

– The Recovery to Prime D-channel capability must be activated at both ends in order to work. If both ends are programmed as NO for this capability, or not matched, the backup D-channel continues to be used even after the prime D-channel is working again. The prime D-channel is not recovered. This forces a technician to manually switch back to the primary D-channel when it is stable.

– System IDs (SIDs), Private Network Identifiers (PNIs), Channel IDs (CHIDs) must be coordinated in the programming at any two connected switches. The people who program your system coordinate these numbers.

– You will need to know when the Central Office (local exchange) is upgraded, so you can adjust your programming. You may be able to take advantage of new public network features and capabilities. However, you may need to upgrade your system to do so.

– You must configure your D-channel for the type of Central Office it is connected to when there is a PRI connection.

– The INAC (Insert Network Access Code) prompt is used by a switch to be able to insert the correct NARS access code (AC1 or AC2) in front of digits in an incoming ISDN call. Insertion is based on the Call Type (CTYP) identifier it received in the D-channel signal preceding the call digits. You can change the CTYP you send to the other end. You may do this if you have a CDP switch connected to a NARS switch. You may want the CTYP to indicate a location code call is coming, instead of a CDP call, so the NARS switch will insert AC1 in front of the location code digits.

– Programmers should refer to the Administration section of the Networking NTPs for information on Connection parameters and Correlation tables for more information on the coordinated programming that is required.

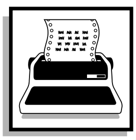
Primary Rate Interface (PRI)

- If you are using packet data transmission, you must dedicate channels on the PRI link to that function. Bear that in mind if you program ISA for the remaining channels.
- If you would like to install a backup D-channel for PRI trunks to a DMS Central Office, your Meridian 1 must be equipped with Release 17 software and the DMS Central Office will need BCS 31. You may also want to consider installing analog trunks to act as backup trunks in the event that the PRI trunks fail. Enter these trunks in route lists, to be used at all times for overflow or only for emergencies (with the Routing Control feature active.)

Training tips

- In training sessions, tell Meridian Mail users that the system will not be able to give the caller’s DN, if the call was not sent in on ISDN links. If a call was tandemed across many switches, there must be ISDN signaling all the way through in order for the user to hear the caller’s ID number in the envelope when they play a message or to use the Call Sender feature.

Maintenance tips



- The Network Call Trace feature can be used by technicians to monitor calls on the network as they are being set up to find where troubles are occurring. Refer to the Appendixes on testing and troubleshooting.

Primary Rate Interface (PRI)

What to have ready

The following checklist summarizes the steps you should take before implementing Primary Rate Interface.

Table 43
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Ensure you have NARS or CDP (or both) and any other ESN packages you need programmed to your satisfaction.
✔		Find out what standards you must comply with in your region.
✔		Decide what features and applications you want to implement before you install any ISDN hardware or software.
✔		Decide if you want to use a back-up D-channel.
✔		Decide if you want manual or automatic recovery to primary D-channel configured.
✔		Plan changes to your trunking so you will accomplish what you want to with ISDN.
✔		Learn about the cards you will need for your trunks, D-channels and clocking requirements.
✔		Discuss the impact the new hardware and software will have on your system.
✔		Plan to train users on new capabilities they will have.

ISDN Signaling Link (ISL)

ISDN Signaling Link (ISL) is a Nortel Networks proprietary configuration for providing ISDN-type functionality in a Meridian 1 private network.

ISL can be set up when you have:

- digital TIE trunks and Digital Trunk Interface (DTI or DTI2) cards that you are not upgrading to PRI cards
- analog TIE trunks between two switches in a private network
- digital TIE trunks and Primary Rate Interface (PRI or PRI2) cards that you program as if they were not PRI because of the ISL advantages discussed later in this section

Basic configuration

The ISL interface can operate under two basic modes of operation: shared and dedicated.

ISL Shared mode requires a 23B+D, 30B+D link between the originating and terminating switches. The D-channel from that link provides out-of-band signaling for the B-channels on that link and can provide signaling for additional digital links or analog trunks between the two systems.

ISL Dedicated mode is appropriate when no E1 or T1 link exists between originating and terminating switches or when you do not wish to share an ISL D-channel as described above. In this mode of operation, a dedicated D-channel is established between the originating and terminating switches. The ISDN signaling messages for the analog or digital trunks between the two switches are transported by way of this channel. You will need modems or data modules for the D-channel connection. Discuss the configuration that suits your needs best with your system supplier.

ISDN Signaling Link (ISL)

Advantages of ISL

There are some advantages that ISL provides, compared to PRI, that you should consider. They are:

- An inexpensive backup D-channel. You do not need a second T1 or E1 carrier or a BRI link (in Europe). You can use a single TIE trunk or leased line.
- Individual trunks can be added to an ISL T1 or E1 link, for more traffic-carrying capacity. You do not need entire T1 or E1 carriers, as for nB+D.
- Revert to Conventional Signaling feature (refer to *Improving feature performance* later in this section)
- Established calls on ISL links remain established if the D-channel fails; they are not cleared. They can be released, using conventional signaling, when the conversation ends.

Consider using E1 or T1 links configured as ISL shared-mode channels to take advantage of the benefits that ISL offers.

Improving feature performance

The information that follows may make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Revert to Conventional Signaling feature

If the D-channel fails, ISL links can be used as if they are conventional non-ISDN trunks. Connections can still be set up between switches without the assistance of the D-channel.

You should have special digit manipulation (ISDM) tables set up in the route lists so that if the signaling does revert, your switch can send the proper digits to the other end. When the D-channel is working, the other end interprets the D-channel message that tells it what type of call is coming. If the response to the INAC prompt for the incoming route is YES, it uses the responses to the AC2 prompt in the Customer Data Block to insert the correct access code (AC1 or AC2) in front of the digits and process the call properly. When the D-channel is not

ISDN Signaling Link (ISL)

working you must compensate by sending the correct digits from the originating end so the terminating switch will process the calls properly. Special digit manipulation indexes accomplish this. They are only used when conventional signaling is being used.

When using T1 carriers, if the PRI D-channel is programmed with a data rate of 56 kbps, then the link can revert to conventional signaling, using the B-channels as A+B bit channels, if the D-channel fails.

> ***Note:*** ISDN features will not work when conventional signaling is being used. If you want the ISDN features to continue to work and you have digital trunks between the two sites, it is better to use a back-up D-channel configuration.

Where E1 carriers are used, the ISL D-channel rate can be up to 64 kbps. The Revert to Conventional Signaling capability cannot be provided on PRI2 cards, only DTI2 cards.

For the ISL configuration on DTI2 links, the D-channel is normally carried on one of the 30 B-channels (leaving 29 B-channels). Modems are used on the channel being used as the D-channel. With DTI2 cards, timeslot 16 is idle normally until the Revert to Conventional Signaling feature activates it for conventional ABCD bit signaling. With PRI2 cards, timeslot 16 is dedicated to carrying the D-channel messages. Therefore, Revert to Conventional Signaling is not possible with this card.

ISDN Signaling Link (ISL)

What to have ready

The following checklist summarizes the steps you should take before implementing an ISDN Signaling Link.

Table 44
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Ensure you have NARS or CDP (or both), and any other ESN packages you need, programmed to your satisfaction.
✔		Decide if you want the Revert to Conventional Signaling capability. If so, program the switches with special digit manipulation indexes.
✔		Decide what features and applications you want to implement before you install any ISDN hardware or software.
✔		Decide on the speed you want for your D-channel, if you are using a leased line.
✔		Discuss your D-channel modem or data module requirements with your system supplier.
✔		Learn about the cards you will need for your trunks, D-channels and clocking requirements.
✔		Discuss the impact the new hardware and software will have on your system.
✔		Plan to train users on new capabilities they will have.

Basic Rate Interface (BRI)

Since this book is focused on networking, this section provides information on Basic Rate Interface Trunk Access only. BRI Trunk Access is offered in countries outside of North America.

ISDN Basic Rate Interface (BRI) is a digital connection that provides three digital channels. There are two B-channels, each with a bandwidth of 64 kbps. Each B-channel acts as a trunk. There is one D-channel with a bandwidth of 16 kbps. The D-channel is used for signaling. This 2B+D connection is called a Digital Subscriber Loop (DSL).

ISDN Basic Rate Interface connections can be configured to provide line access, trunk access or packet data transmission.

Basic Rate Interface (BRI)

Basic configuration

BRI Trunk Access

The most important capabilities of ISDN BRI Trunk Access are:

- ISDN private connections without the need for PRI
- ISDN public connections without the need for PRI
- backup D-channel for PRI D-channel
- overflow trunks for PRI connections if they are busy or they fail

Trunk Access provides:

- Meridian 1 to Meridian 1 Meridian Customer Defined Networking TIE trunk connectivity
- QSIG ISDN BRI trunk connectivity
- CO/DID trunk connectivity to local exchanges that support:
  - Numeris VN3
  - 1TR6
  - ETSI NET-3 (Euro ISDN)
  - INS NET-64 (including Japan D70)
  - Asia-Pacific protocols

BRI trunk access is not supported in North America.

Basic Rate Interface (BRI)

BRI as partial PRI

You can implement BRI Trunk Access as if it is partial PRI. In other words, when you do not have the traffic or the budget for PRI with the E1 trunks that it requires, you may be able to implement BRI instead.

ISDN Signaling Link (ISL) is also a less expensive alternative to PRI but BRI has these advantages:

- BRI is a standard protocol. ISL is proprietary.
- the BRI D-channel has a data rate of 16 kbps. In North America, ISL can use trunks ranging in data rates from 4800 baud up to 56 kbps (and still have the Revert to Conventional signaling capability). However, the higher the speed, the greater the monthly charge for the D-channel trunk.

  Note: Internationally, where E1 carriers are used, the ISL D-channel rate can be up to 64 kbps. (However, the Revert to Conventional Signaling capability cannot be provided on PRI2 cards, only DTI2 cards).

Connectivities

- Meridian 1 to Meridian 1 MCDN TIE trunk connectivity.
  - You need either an S/T interface (SILC card) or U interface (UILC card) for every 8 DSLs. The U interface is implemented as an ANSI standard interface for North America only.
  - You need a MISP card for every 4 SILC and/or UILC cards.
  - The BRI TIE trunks extend between two Meridian 1s through a passive local exchange that supports Numeris VN3, 1TR6, ETSI NET-3 (EuroISDN), INS NET-64 (including Japan D70), and Asia-Pacific protocols, or they can extend directly between two Meridian 1s. Refer to the BRI NTPs for further information and illustrations.
- QSIG ISDN BRI trunk connectivity.

Basic Rate Interface (BRI)

You will need:

– S/T interface (SILC card) or U interface (UILC card) for every 8 DSLs

– MISP card(s) for every 4 SILC or UILC cards

◆ CO/DID trunk connectivity to local exchanges that support Numeris VN3, 1TR6, ETSI NET-3 (EuroISDN), INS NET-64 (including Japan D70), and Asia-Pacific protocols. You need:

– S/T interface (SILC card) for every 8 DSLs

– MISP card(s) for every 4 SILC or UILC cards

SILC cards can be used for CO trunk or TIE trunk connectivity; UILC cards are used for TIE trunks only.

SILC cards can be used for BRI trunks and lines. You can put BRI telephones and terminals and BRI trunks on the same card.

Each MISP card uses one network equipment slot. It uses an even numbered loop on the network bus to provide 32 time slots for D-channel signaling and packet data for SILCs and UILCs. The odd numbered loop is unconfigurable because it is used to communicate with the CPU.

> ***Note:*** Using Basic Rate Signaling Concentrators (BRSCs) reduces or eliminates the number of MISPs needed. Each BRSC supports 15 SILCs and/or UILCs and each SILC and UILC supports 8 DSLs.
>
> One MISP can serve up to 8 BRSCs and two line cards. Therefore, this increases DSL capacity for the MISP from 32 to 996 (8 x 120 DSLs plus 16 DSLs). MISP cards sit in NE slots, BRSCs sit in IPE slots. BRSCs can be used when you have BRI lines or for packet data transmission from BRI terminals to an external DPN-100 using PRI links.
>
> *BRSCs cannot be used for BRI Trunk Access.*

Basic Rate Interface (BRI)

Distance limitations

There are distance limitations (the maximum distance being 1 km) when you extend BRI trunks between:

- SILC cards and NT1 adapters (which connect to a UILC card in another Meridian 1 up to 5.5 km away)
- two SILC cards in two Meridian 1s.

Trunks and routes

Each DSL can have a different protocol.

Both B-channels on one DSL must belong to the same route.

If channel negotiation fails on one DSL, the system will not attempt to negotiate channels on another DSL even though it may have channels in the same route.

BRI trunks can be accessed by non-BRI terminals.

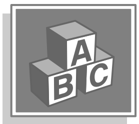
If a BRI TIE trunk is:

- between two Meridian 1 switches with a passive exchange office between them (use an SILC card at each end and a NT1 adapter at each end, supplied by the PTT). There is no distance limitation.
- directly between Meridian 1 switches, use one NT1 adapter, one SILC card and one UILC card. The UILC end can run in free run clocking mode, no clock controller is needed. The distance limitation (without a signal amplification device) is 6.5 km.
- directly between two Meridian 1 switches, use one SILC card at each end. One end can run in free run clocking mode. The distance limitation is 1 km. Use this application if you have many buildings on one property. There are no protection devices with this configuration however.

BRI to PRI connectivity

You must configure the D-channel for connectivity to a PRI link at the other end if the interface is a Meridian 1, D70, 4ESS, or 5ESS switch.

Basic Rate Interface (BRI)



Clocking

The BRI clock source must be a slave to the network clock. BRI trunks can get their clock from other PRI2 or DTI2 connections on the same system. They do not have to be connected to the clock source directly. The BRI trunk can also act as the clock source for other digital trunks.

Features support

When you install BRI Trunk Access you should know about the capabilities it does and does not provide, before you start.

Refer to the BRI NTPs for the features that are not supported when BRI trunks are used.

The most notable features that are not supported, or have limited support are:

- backup D-channel. The D-channel from one BRI DSL cannot act as backup for the D-channel on another DSL.
- ISA. Both B-channels must belong to the same route.
- ISL. Therefore there is no Revert to Conventional Signaling capability.
- nB+D. Non-associated signaling channels or "shared mode" is not supported.
- Trunk Anti-Tromboning (TAT) works only when BRI trunks are used for Virtual Network Services.
- Trunk Route Optimization (TRO) only works when BRI trunks are connecting two Meridian 1s.
- Virtual Network Services (VNS). A D-channel of a BRI TIE trunk cannot be a VNS D-channel between switches. A Public BRI B-channel can act as a VNS voice channel.
- The most notable Supplementary features with limited or no support are:
  - Autodial Tandem Transfer
  - Automatic Redial

- CDR Enhancement
- Call Park Network Wide
- DPNSS
- FCC compliance for Equal Access – CAC expansion
- Network Call Pick-up and TAFAS
- NRAG
- Ring Again on no Answer

Since BRI cannot be used for the features listed above, you will need to install an alternative such as PRI or ISL, if you want to implement them.

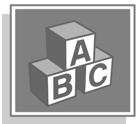
Country-specific BRI configurations

Japan

The following features are supported:

- Basic Call Service
- Circuit switched voice and data on the B-channel
- Calling Line Identification (public and private)
- Advice of Charge at the end of the call
- COT, DID, DOD, and TIE trunk call types
- 64 kbps clear bearer capability
- channel negotiation within one DSL
- Flexible Numbering Plan
- PRI backup (overflow when PRI trunks are busy)
- BRI/PRI interworking for IEs supported by BRI

Basic Rate Interface (BRI)



The following features and capabilities are not supported at this time:

- ISA (Call by Call Service)
- FEX, WATS call types
- Network Ring Again
- Network Calling Party Name Display
- CLID Enhancements (Redirecting Number)
- Network Call Redirection, Network Call Forwarding, Network Call Forwarding No Answer
- Trunk Optimization/Anti-Tromboning
- ISDN Signaling Link (ISL)
- ESN Signaling on PRI (NCOS)

Asia Pacific (APISDN) (Australia, Hong Kong, Singapore, Thailand, New Zealand)

This protocol involves both PRI and BRI interfaces.

The features that are supported for a BRI interface are:

- Basic Call Service
- circuit switched voice and data on the B-channel
- Calling Line Identification and Restriction (public and private)
  - public: display of a CLID received from the public network
  - private: sending the CLID from the private network to the public network
- COT, DID, DOD, and TIE trunk call types
- 64 kbps clear capability
- channel negotiation
- Flexible Numbering Plan

Australian APISDN supports Advice of Charge and Malicious Call Trace features.

New Zealand NEAX-61 APISDN supports Three Party Conference and Malicious Call Trace.

Overlap sending is supported for:

- Australia
- Hong Kong
- New Zealand
- Singapore
- Thailand

Overlap receiving is supported for Thailand.

BRI in Hong Kong supports only data calls.

Backup D-channel is not supported.

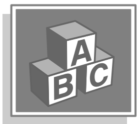
BRI Trunk Access can be used as an overflow when PRI trunks are busy.

Asia Pacific (APISDN) (Indonesia, China, Japan, and Malaysia)

With Release 23, BRIT the following capabilities are supported:

- Basic Call Service
- circuit switched voice and data on the B-channel
- Calling Line Identification Presentation (CLIP) and Restriction (CLIR) (public and private)
  - public: display of a CLID received from the public network
  - private: sending the CLID from the private network to the public network
- COT, DID, DOD trunk call types
- TIE trunk call types (Japan and Malaysia only)
- 64 kbps clear capability

Basic Rate Interface (BRI)



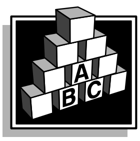
- channel negotiation
- Enbloc Signaling using CDP steering code access

Overlap Sending is supported on Indonesian, Chinese, and Malaysian interfaces. Overlap Receiving is supported on the Indonesian and Chinese interfaces as optional. It is supported on the Malaysian interface.

Indonesia - country specific specifications do not clearly state whether an outgoing call on BRI supports the “any channel acceptable” option for B-channel selection. It is not currently implemented by Meridian 1.

Only circuit switched calls are supported. Packet switched calls are not supported.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

BRI as backup for PRI

- You can program BRI trunk routes to act as overflow for busy PRI trunks. Program the BRI route(s) in the NARS or CDP route lists.
- You could use a BRI trunk route as an emergency alternate route if you experience problems with the PRI link. Program the BRI route(s) in route lists and assign a very high FRL. When problems occur with the PRI link, you could use the Routing Control Feature to raise certain NCOS groups to the required FRL so certain users can call out on the BRI trunks during the emergency.
- You can install BRI to act as a backup D-channel for PRI. The reverse is not true however. PRI D-channels cannot backup BRI D-channels. Also, BRI cannot backup other BRI channels.

Basic Rate Interface (BRI)

Traffic studies

There are several traffic studies you can run on the BRI links. They are System Traffic Study Options 11, 12, 13, and 15.

These studies help technicians determine if there are hardware related problems with any of the cards or software-related problems with the messages travelling on the D-channel.

- TFS011 looks at the number of calls and messages handled by the MISP card(s).
- TFS012 checks for communication problems between the MISP and the terminals. This study does not apply to BRI Trunk Access.
- TFS013 looks at MISP messages to check if messages are within specified lengths.
- TFS015 looks at the Meridian Packet Handler.

If you are experiencing BRI-related network problems, use traffic study data in discussions with your system supplier and maintenance people to help resolve the problems.

Administration tips

- When configuring the two B-channels on the DSL as trunks, do not make them consecutive members of the route. The impact of a faulty DSL will not be as great, while you are fixing it, if the trunks are spread out as members in a route.

Basic Rate Interface (BRI)

What to have ready

The following checklist summarizes the steps you should take before implementing the BRI feature.

Table 45
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Ensure you have NARS or CDP (or both) and any other ESN packages you need programmed to your satisfaction.
✔		If you will use BRI routes as overflow for PRI, program your route lists accordingly.
	✔	Decide if you will use a BRI D-channel to backup PRI.
✔		Find out what standards you must comply with in your region.
✔		Decide what features and applications you want to implement. Then choose the ISDN hardware or software that supports your requirements.
✔		Plan changes to your trunking so you will accomplish what you want to with BRI.
✔		Learn about the cards you will need for your trunks, D-channels and clocking requirements.
✔		Discuss the impact the new hardware and software will have on your system with your supplier.
✔		Plan to train users on new capabilities they will have.

DPNSS1

APNSS

There is a variation of DPNSS1 known as the Analog Private Network Signaling System (APNSS). It can be connected between switches through a dumb modem using a dedicated analog or digital signaling path. There is more information on APNSS in this section and in the *Introduction to ISDN* section.

DPNSS1

DPNSS1 is a common channel signaling system and an open signaling protocol standard for intelligent *private network* digital connections. DPNSS1 provides the signaling capability to establish simple telephony and data calls, as well as supplementary services (features).

In the United Kingdom (UK) British Telecom's Digital Private Network Signaling System No. 1 (DPNSS1) is the prevalent intelligent private network signaling system. DPNSS1 is being used outside the UK as well.

DPNSS1 is unique in its allowance of intelligent networking between different-vendor PBXs.

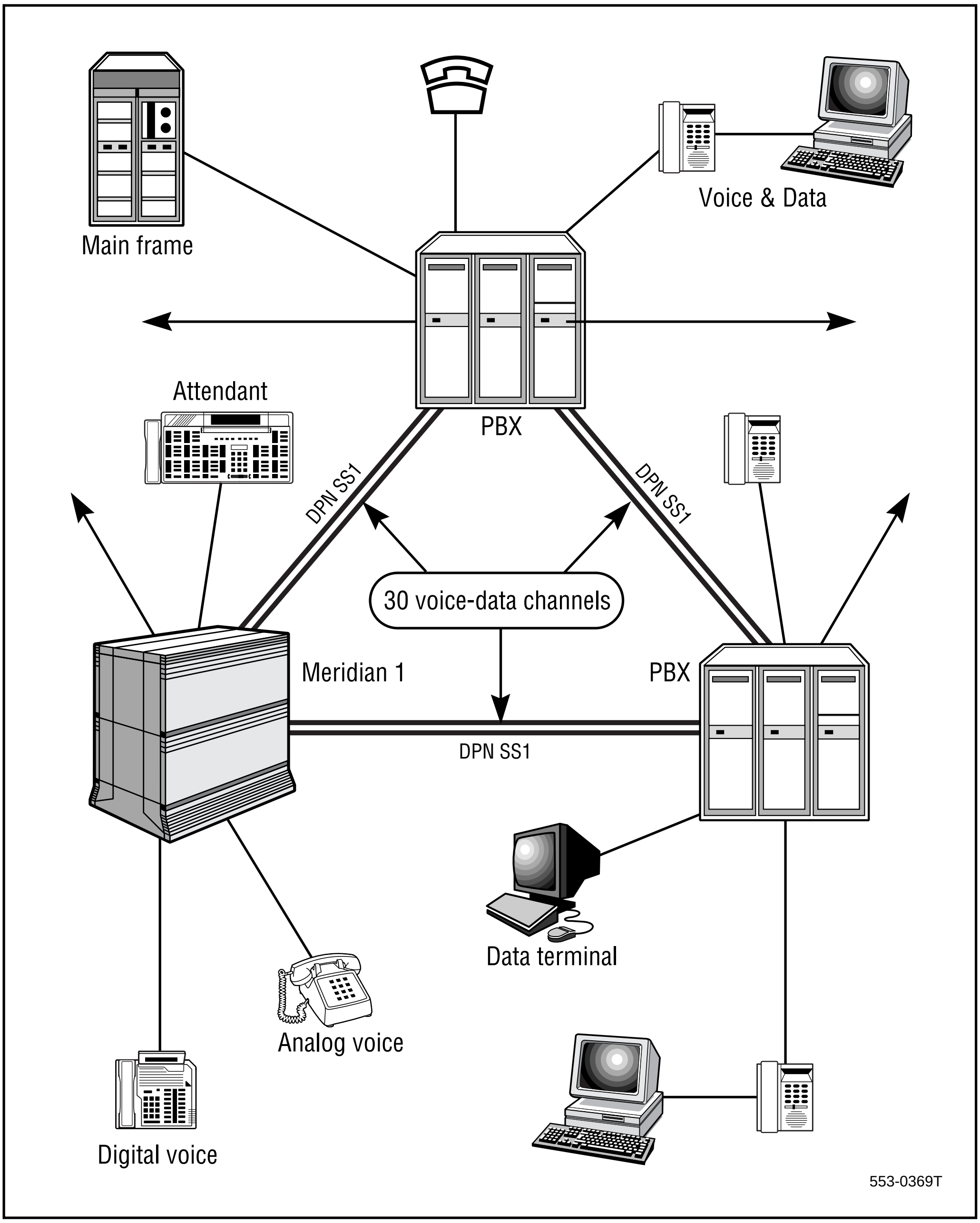
The Meridian 1 uses unique hardware and software elements to provide the DPNSS1 functionality. These are implemented in compliance with BTNR 188 sections. Refer to the *DPNSS1 Product Information Guide*, 553-3921-100, for details about compliance with BTNR 188.

Transmission system

The 2.048 Mbps digital transmission carrier is divided into 32 timeslots, numbered 0-31. Timeslots 1-15 and 17-31 provide 30 voice/data traffic channels. Timeslot 0 is used as a synchronization channel. DPNSS1 is a message-based signaling system that uses a common signaling channel in timeslot 16.

DPNSS1

Figure 31- DPNSS1 configured in a network



Link designation

The ends of each inter-PBX link are labelled arbitrarily A and B, and the ends of each DPNSS1 channel are designated X and Y. If both ends attempt to use the channel at the same time, the X end has priority.

End PBXs, gateways and transits

A PBX that connects a DPNSS1 channel to or from a non-DPNSS1 device is termed an end PBX. If that device is a trunk, then the PBX is termed a gateway. A PBX connecting two DPNSS1 channels is a transit.

Basic configuration

For more information about the technical details related to DPNSS1, refer to the *DPNSS1 Product Information Guide*. Introductory information about DPNSS1 is presented in the section of this book called *Introduction to ISDN*. For information about DASS2, refer to the section on DASS2 in this book.

Configuration of trunks

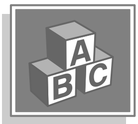
D-channels

This out-of-band signaling channel may be on the same or a different carrying medium as the B-channels it supports. One D-channel can support up to 30 B-channels.

Virtual channels

These channels are used for supplementary services such as Call Back When Free. Virtual channels do not require an actual B-channel. Up to 30 virtual channels may be configured on a DPNSS1 link. These channels are associated with routes that have real B-channels.

DPNSS1



DPNSS1 trunks are configured using the same route and member method used for other trunks. Other characteristics of DPNSS1 trunks are:

- any number of routes may be associated with the same link
- a route may be associated with any number of links
- each route member must be assigned to one channel
- not all channels need to be associated with members. These non-associated channels cannot, however, be used for calls
- members and channels must be numbered separately
- members are screened for outgoing calls using a linear search (Sequential Line) or round robin (Cyclic Line). For DPNSS1 links, a linear search should be used.
- each route may be configured only for incoming calls, only for outgoing calls, or for both
- each route must be configured with DPNSS1 channels only

Numbering plans

The numbering plans supported on DPNSS1 networks are:

- Coordinated Dialing Plan (CDP)
- Flexible Numbering Plan (FNP)
- Group Dialing Plan (GDP)
- Uniform Dialing Plan (UDP)

There is more information on these dialing plans in this book. Please refer to the Table of Contents.

Previous to Release 21, incoming calls on DPNSS1 trunks going out to DPNSS1 trunks could only be translated as CDP Distant Steering Code or Trunk Steering Code calls.

DPNSS1

The digits that come into a switch are called the Destination Address (DA). With CDP DPNSS1, the DA had to pass through the DPNSS1 switch unchanged. If digit insertion is being performed on incoming routes, the same digits should be deleted before the call goes out on a DPNSS1 trunk.

In Release 21, NARS (UDP) functionality was added.

- You can translate up to 10 digits for LOC calls and 16 digits for SPN calls. This allows greater flexibility in digit translation and restriction/recognition.
- Also when interworking with Q.931 (MCDN PRI) networks, that commonly had NARS, the mix of CDP and NARS networks did not support features such as Network Ring Again. It is best if all switches involved have the same ESN software packages (either CDP or NARS or both, not a mixture of these software packages).

Incoming DASS2 (public network) calls can be translated by NARS if they are in the format: AC + LOC/SPN + X...X.

Features supported

The features supported by the original CDP DPNSS1 include:

- DPNSS1 Three Party Service
- DPNSS1 Loop Avoidance
- DPNSS1 Route Optimization
- DPNSS1 Step Back On Busy Congestion
- DPNSS1 Diversion
- DPNSS1 Call Back When Free and Call Back When Next Used
- DPNSS1 Redirection
- DPNSS1 Attendant Call Offer
- DPNSS1 Executive Intrusion

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UDP DPNSS1 supports:

- Call Back When Free and Call Back When Next Used
- Executive Intrusion
- Loop Avoidance
- Three-Party Service
- Call Offer
- Redirection
- Step Back on Congestion
- Route Optimization

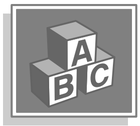
Satellite connections

As of Release 21, all systems except the Option 11 can be configured to use DPNSS1 signaling to satellites. You will need the appropriate D-channel card with the correct firmware for this capability.

Note that DPNSS features and services are available only in DPNSS1/DASS2 network environments, and may not be available in your area. For more information, contact your system supplier.

Features not supported

- BARS (without NARS)
- travelling NCOS
- nB+D
- backup D-channel
- Off-Hook Queuing and Callback Queuing
- incoming calls with NPA/HNPA/NXX codes – they are cleared (these codes are not applicable to the UK) – use SPNs instead
- calls received on DPNSS1 routes will not be screened by FCAS or FSNS tables – use the Alternate Route Number (ARRN) feature instead
- calls made with a Trunk Route Access Code followed by a DN coming in on a DPNSS1 route with INAC set to YES



INAC

DPNSS1 networks do not send a call type with calls. This means that one switch does not signal another about whether the call is UDP-type or CDP-type. You can use INAC programming on each route to get around this limitation. There is more information about INAC on page 613.

- If INAC is NO, then calls must be received as non-UDP numbers or with AC1 or AC2 already present.
- If INAC is YES, the DA is treated as a UDP number (LOC/HLOCor SPN).
  - if the RDB contains SPN YES, then the system searches for a valid SPN first
  - if the RDB contains SPN NO, then the system searches for a valid LOC/HLOC first. If a valid HLOC is found, the digits that follow the HLOC can be CDP numbers.

The Customer Data Block gives instructions about what AC code to insert in front of the SPN or LOC in the call.

Hardware

MSDL cards do not support DPNSS1 signaling.

DPNSS1 Three Party Service

This feature allows a controlling party to place an established call on hold and make an enquiry call to a third party. The controlling party may then transfer the held party to the enquired-to-party, or form a three-party conference.

DPNSS1 Loop Avoidance

The DPNSS1 Loop Avoidance feature prevents a DPNSS1 call from being looped through a network, due to errors in configuration, by placing a limit on the number of channels that a call may use.

The call is treated as if clearing has occurred due to congestion. If configured, alternative routing using Step Back on Congestion is attempted at the originating end only, if all of the available routes for

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the call have not been used. If alternative routing using Step Back on Congestion is not available, the treatment that the call receives depends on the originating party.

If the originating party is a non-ISDN trunk, the originating party receives congestion treatment as defined in the Customer Data Block. This treatment may be busy tone or overflow tone. If the call was routed due to Network Alternate Route Selection (NARS), NARS call blocking intercept treatment is given (either overflow tone, busy tone, recorded announcement, or route to attendant). If the originating party is an ISDN trunk, the originating party receives congestion treatment as defined in the Customer Data Block (busy tone or overflow tone).

If the originating party is a local set, treatment depends on the customer-defined congestion treatment (either busy tone or overflow tone) or NARS call blocking intercept treatment (either overflow tone, busy tone, recorded announcement, or route to attendant). If the originating party is a local attendant, busy indication is given. At this point, the DPNSS1 Attendant Camp-on feature may not be used.

Intercept

The Intercept treatment for Network Alternate Route Selection calls that are blocked, configured in the Customer Data Block, in response to the INTR prompt, should be the same as for calls receiving Loop Avoidance call-back treatment, configured in the Customer Data Block, in response to the CONG prompt.

Plan to configure the following parameters:

Implementation of Loop Avoidance

Enter the Loop Avoidance Limit for DPNSS1 calls or the Tandem Threshold Limit for ISDN calls.

- If the ISDN and ISDN SUPP packages are configured, you can choose a value from 0-31 to define the Tandem Threshold Limit for ISDN calls; 15 is the default setting.
- If the ISDN and ISDN SUPP packages are not configured, you can choose a value from 0-25 to define the Loop Avoidance Limit for DPNSS1 calls; 15 is the default setting.

DPNSS1 Route Optimization

The DPNSS1 Route Optimization feature has been developed to optimize trunk usage within a DPNSS1 network by replacing non-optimum call paths through a DPNSS1 private network with optimum paths. An optimum path is the path that uses only the first choice routes to link two PBXs across the network. The first choice is determined by the programming of the route list indexes at each PBX. This optimization applies to established simple voice calls which were routed during set-up, or transferred or attendant-extended to another party.

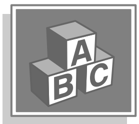
Route optimization is initiated by the originating PBX, after recognizing that a DPNSS1 call may have been set up over a non-optimum path due to alternative routing or call modification.

If the call is ringing, the originating PBX waits for an answer signal before initiating optimization. If the call has been transferred, on answer, or attendant-extended to a another party, then the transfer or extension signaling sequence initiates the optimization.

If the Route Optimization request call set-up successfully gets back to the originating PBX, a conference is established at the originating node between the originating party, the original path still carrying the speech, and the silent new path. A message of acknowledgment is returned to the terminating PBX on the new path. Upon receiving this acknowledgment, the terminating PBX replaces the old path with the new (optimized) path, and sends a connect indication across the new path to the originating PBX. The old path is silenced. Upon receiving the connect indication, the originating PBX terminates the conference, connects the originating party to the optimized path, and clears the original path.

If the Route Optimization request call set-up fails, the originating PBX receives a notification message that the Route Optimization request was not successful. The originating PBX may then attempt Route Optimization again, at 60 second intervals. During this interval, the Meridian 1 may initiate Route Optimization requests for other DPNSS1 calls.

DPNSS1



You can define the following Route Optimization options in the Customer Data Block:

- NRO (No Route Optimization). Route optimization is inhibited for all calls. This option would typically be used on PBXs having high levels of call traffic where TRO would add extra real time pressures to an already busy system.
- ROA (Route Optimization for Alternately Routed calls). Route Optimization is initiated for calls which have undergone alternative routing, such as by Step Back on Congestion. A call is considered to be alternatively routed if it originated over a route which was not the first choice route, or if alternative routing indication is sent in the Routing Information (RTI) of a Network Indication Message (NIM).
- ROX (Route Optimization for Transferred calls). Route Optimization is initiated for transferred or attended-extended calls.
- RAX (Route Optimization for transferred or alternatively routed calls).

Feature interactions

DPNSS1 calls in the ringing state are optimized immediately upon being answered.

If a Route Optimization call setup encounters any redirection features, these features are ignored.

The condition for a diverted call to have Route Optimization after connection is the same as a simple DPNSS1 call. Route Optimization starts if the diverted call is routed through a non-first choice route or when a call transfer involving the diverted call is completed.

A call that has been call-forwarded may be optimized when answered only if it has undergone alternative routing. If the forwarded call was not alternatively routed, it may use a non-optimum path.

A call that has been picked up or that has undergone hunting may be optimized when answered only if it has undergone alternative routing.

A Ring Again call may be optimized only if it has undergone alternative routing.

A call transferred to another party may be optimized only after the call transfer has been completed. A call transferred to a ringing set may be optimized only after being answered.

A call that has been re-routed due to Step Back on Congestion may be optimized after it is answered.

During a group hunt, a call to a Pilot DN which has been defined as a trunk access code may be optimized upon being answered only if it has undergone alternative routing.

A call which is camped-on or call-waiting to a telephone may not be optimized until the call is answered on the telephone.

Route optimization may be applied to a call that is being overridden only after it becomes a simple call.

Route Optimization cannot be applied to the following calls:

- data calls
- conference calls (however, Route Optimization may be applied when the conference call reverts to a normal two-party connection)
- calls on hold
- attendant-originated calls

It is possible to configure Trunk Barring (TBAR) to prevent trunk-to-trunk connections on a local node.The call will not be optimized if the TBAR configuration restricts the local connection.

DPNSS1 Step Back on Congestion

The Step Back on Congestion (SBOC) feature operates during periods of high traffic when DPNSS1 calls may encounter congestion. The call may be passed back or re-routed using the next free alternative route.

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If a congestion message is received at a DPNSS1/ISDN gateway node, the ISDN Drop Back Busy options are checked to determine whether the call is to be dropped back. If not, the DPNSS1 Step Back on Congestion feature is invoked.

If an originating node receives either a congestion message or a Network Termination and a Loop Avoidance Supplementary string message, the call may be routed using the next free alternative route, or receive call blocking treatment if no re-routing is configured or if no alternative route is available.

Feature requirements

This feature uses the ESN Coordinated Dialing Plan, Network Alternate Route Selection (NARS) or Basic Alternate Route Selection (BARS) to re-route a congested call.

Re-routing is not attempted if a trunk access code was used to originate the call.

Feature interactions

A call that is blocked due to the DPNSS1 Loop Avoidance feature may be re-routed at the originating node, but not at a transit node.

DPNSS1 route optimized calls that encounter congestion are not re-routed, since Route Optimization only uses first choice routes.

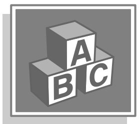
Intercept

The Intercept treatment applied due to network blocking is defined in the Customer Data Block.

Feature implementation

The types of re-routing you can choose are:

- no re-routing of calls
- re-route calls if at an originating node, or to step back if at a transit node
- re-route calls at all nodes



In the route lists you must define the alternative route sets for ISDN Drop Back Busy. You can choose either:

- all routes may be used to route calls
- only Iset routes may be used to route calls

In the Customer Data Block you must define the network blocking treatment for calls stepped back to the originating node.

DPNSS1 Diversion

DPNSS1 Diversion is a British Telecom Network Requirement (BTNR) supplementary service that provides full DPNSS1 Diversion signaling on DPNSS1 links, when one of the redirection features listed below is invoked:

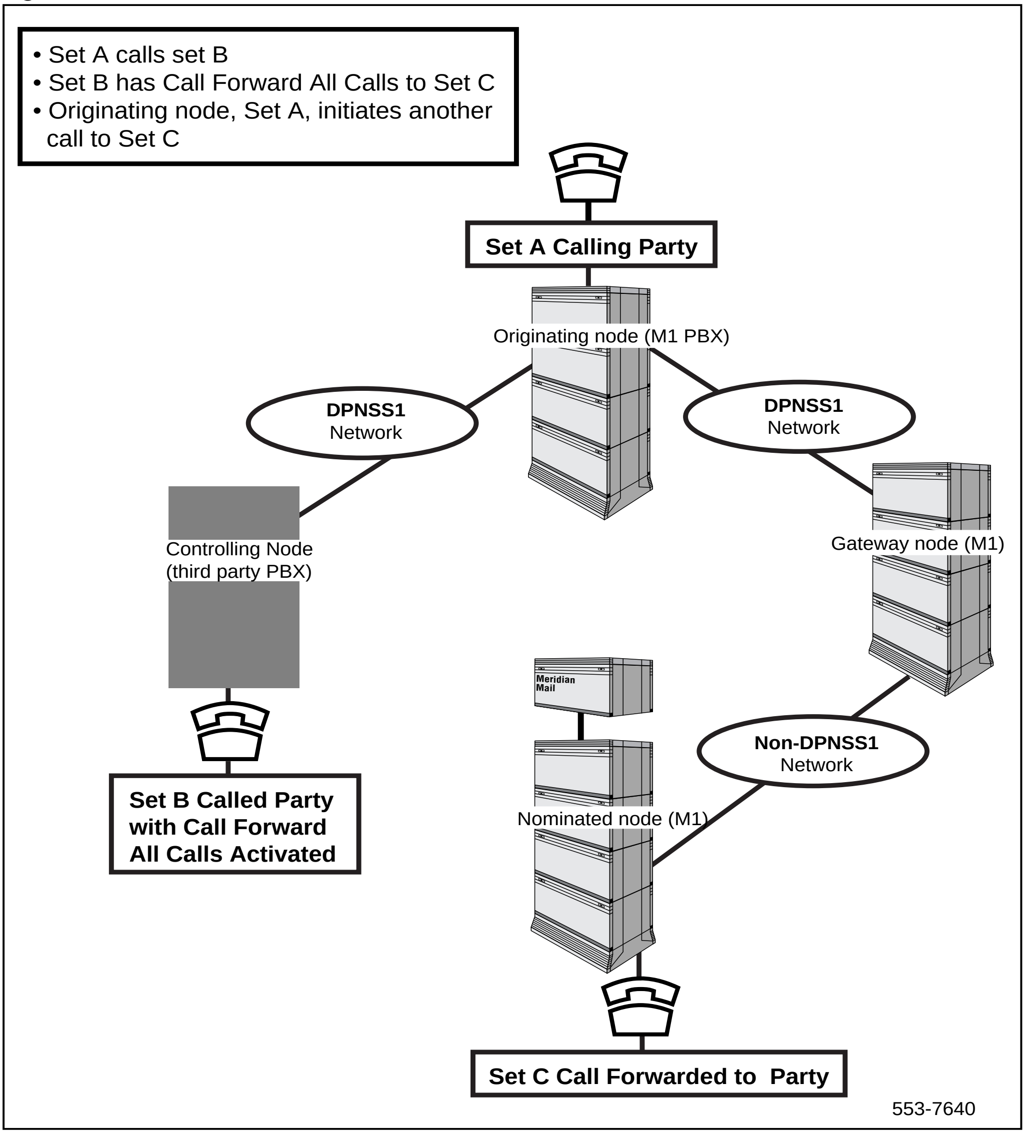
- Call Forward All Calls
- Call Forward No Answer
- Call Forward by Call Type
- Call Forward Busy
- Hunting/Group Hunting
- Intercept Computer Call Forward All Calls
- Call Forward Internal Calls
- Meridian Customer Defined Network Call Redirection
- Call Party Name Display

When a user activates a redirection feature such as Call Forward All Calls, DPNSS1 signaling informs the call originating node that the call is being forwarded to another telephone. If the forwarded party is located on another node, the call originating node is requested to initiate a new call. When the forwarded to party is reached by the call originator via DPNSS1, the forwarded to party is notified that the incoming call has been forwarded.

DPNSS1

Call Diversion functions on Meridian 1 nodes that are linked to third party Private Branch Exchanges (PBXs) within a DPNSS1 network. Meridian 1 gateway nodes provide links to other Meridian 1 nodes by Meridian Customer Defined Network (MCDN) and to Meridian 1 and third party PBXs through DPNSS1.

Figure 32- Diversion Environment



The following capabilities are provided as part of the DPNSS1 Diversion: Diversion Validation, Diversion Cancellation, Diversion Follow-Me, Diversion By-Pass, Diversion Immediate, Diversion On Busy and Diversion On No Reply. These capabilities are described as follows.

Diversion Validation

DPNSS1 Diversion must operate on Meridian 1 nodes that are linked to third party PBXs, within a full DPNSS1 environment. Validation is performed on forward-to DNs, for example.

Meridian 1 gateway nodes are linked with another Meridian 1 node through a Meridian 1 Customer Defined Network (MCDN) and the other Meridian 1 or other third party PBX via a DPNSS1 network.

Diversion Cancellation

Diversion Cancellation allows the forwarded to party to remotely deactivate call diversion initiated by the forwarding party. Meridian 1 DNs cannot originate Diversion Cancellation requests; however, Meridian 1 PBXs can process Diversion Cancellation requests.

The sequence for Diversion Cancellation is as follows:

- Telephone A has activated Call Forward All Calls to Telephone B.
- Telephone B requests either Diversion Immediate or Diversion-All Cancellation to Telephone A.
- Upon receipt of the cancellation request, Telephone A’s node determines that Telephone B is currently Call Forward All Calls activate to Telephone A’s DN.
- If the DN is confirmed, then the CFAC feature is deactivated
- Telephone B is notified that the cancellation request is successful

If the Diversion Cancellation request encounters any gateway, the gateway responds with a “Service Unavailable” notification.

Diversion Follow-Me

Diversion Follow-Me allows the forwarding party to remotely request and change the forward-to DN. As an example, Telephone A has activated Call Forward All Calls to Telephone B, in a full DPNSS1 environment.Telephone A then decides to change the forwarded -to party, to Telephone C. When Diversion Follow-Me is activated, Telephone A's node uses Diversion Validation to confirm that the new forwarded-to DN is valid.

If a Diversion Follow-Me request encounters any gateway, the gateway responds with a "Service Unavailable" notification.
A Follow-Me request is always rejected when routed through a gateway.

Meridian 1 PBXs can process Diversion Follow-Me requests but cannot initiate any requests.

Diversion By-Pass

Diversion By-Pass allows the calling party to ignore the diversion assigned by the party that activated call redirection. Meridian 1 DNs cannot originate Diversion By-Pass requests, but can process requests.

Diversion Immediate

With Diversion Immediate, the calling party, Telephone A, dials Telephone B that has activated Call Forward All Calls to Telephone C. Upon receipt of the call, Telephone B's node instructs Telephone A's node to Divert-Immediate to Telephone C.

When instructed to divert, Telephone A's node clears the old call and initiates a new call to Telephone C. Telephone A's display is updated with diversion information, when the call is established with Telephone C.

Diversion On Busy

The sequence for Diversion On Busy via Separate Channel is similar to Diversion-Immediate. The differences occur with message contents and the reason for diversion, if Call Party Name Display is activated.

Diversion On No Reply

Call Diversion on No Reply ensures that a Call Forward No Answer call is not disconnected until the new diversion call is successful.

The following is the sequence for Diversion On No Reply functionality. Telephone A, the calling party, dials Telephone B. Telephone B rings and has Call Forward No Answer activated to Telephone C. When requested by Telephone B’s node to Divert the call on No Reply, Telephone A’s node initiates a new call to Telephone C. When Telephone C answers the diverted call, the original call between Telephone A and Telephone B is disconnected.

DPNSS1 Call Back When Free and Call Back When Next Used

A user can use these features to:

- queue for a busy telephone across a DPNSS1 network
- queue for a telephone that did not answer, waiting for the next time the telephone is used
- place a call automatically, after congestion clears between two systems

Users access the Ring Again feature for Call Back When Free and for Call Back When Next Used.

The system alerts the queuing telephone with six ring cycles. The user answers the call back by lifting the handset or accessing the DN being rung on a proprietary telephone.

The features cannot be used to queue for attendant consoles or ACD queues. Other users nearby cannot use the Call Pickup feature to answer the call back.

Attendant-related capabilities

DPNSS1 Redirection

The DPNSS1 Redirection feature allows a DPNSS1 call that is extended by an attendant and not answered after a defined period of time, to be recalled to an attendant. This attendant may be the one who originally extended the call, or another attendant on the same or different node within the network.

> ***Note:*** The DPNSS1 Redirection feature is required for DPNSS1 networks using a Centralized Operator Service, if the network nodes on which operator consoles are located use DPNSS1 Redirection to provide timed operator recall functionality.
> If operator consoles are located on a Meridian 1 PBX, timed operator recall is provided by the DPNSS1 Timed Recall feature described in this section.

When an attendant extends a call to a destination, and the destination does not answer before the attendant releases the call, information is passed to the originating DPNSS1 node to initiate recall timing.

- If the information indicates that the destination is busy, then the camp-on recall timer is started. For camp-on timing, if the destination party becomes free before the camp-on timer expires, then the destination party receives ringing. The camp-on timer is cancelled, and the slow answer recall timer is started.
- If the information indicates that the destination is free, then the slow answer recall timer is started.
  - If the destination answers the call extension before the recall timer expires, the recall timer is cancelled and the source and destination are connected.
  - If the recall timer expires before the call extension is answered, a new call is initiated to the local attendant.

    If the local attendant is unavailable, Network Attendant Service (NAS) routes the call to another node. If the call reaches a state of attendant receiving buzzing, attendant receiving ringing, or queued to attendant, then the originating party is connected to the new call. The original call is dropped. If a new call cannot be established, a Clear Request Message is sent to the originating node and the original call remains connected. If the original call, while in call waiting or camp-on, is answered by the destination party before ringing state is attained, a Call Connected Message is sent to the originating node. The new call is cleared and the original call remains connected.

DPNSS1 Attendant Call Offer

The DPNSS1 Attendant Call Offer feature allows attendant-extended calls routed over DPNSS1 to be camped-on to a remote busy extension. This Call Offer functionality is provided over a DPNSS1 network or over a DPNSS1 to ISDN gateway.

After being offered the camp-on, the destination party has the option of either accepting the offer, or not. During the camp-on offer, the destination party receives camp-on tone, heard over the conversation. The destination party accepts the call offer by clearing the established call (the offered call may not be accepted by simply placing the established call on hold). The destination party rejects the call offer by not answering it.

If the busy party goes on hook, allowing the offered call to ring the telephone, the recall timer for the call is reset to the value programmed for ringing calls. If the call remains unanswered when this timer expires, the offered call is recalled to the attendant queue. If the call is accepted, the originating party receives ringback until the destination party goes off hook to answer the call.

If the call is not accepted, the camp-on is recalled to the attendant after the camp-on timer times out. Timing for camp-on recall begins as soon as the attendant presses the Release key to extend the camp-on to the destination party. The destination party may still answer the camp-on as long as the call is still on the attendant console (that is, while the attendant is talking to the source.) The attendant may clear the camp-on by releasing the destination.

DPNSS1 Night Service

The DPNSS1 Night Service allows a Meridian 1 to treat a third-party PBX's request to divert a call queued to an attendant that is in Night Service mode, back to the local attendant queue of the originating DPNSS1 node.

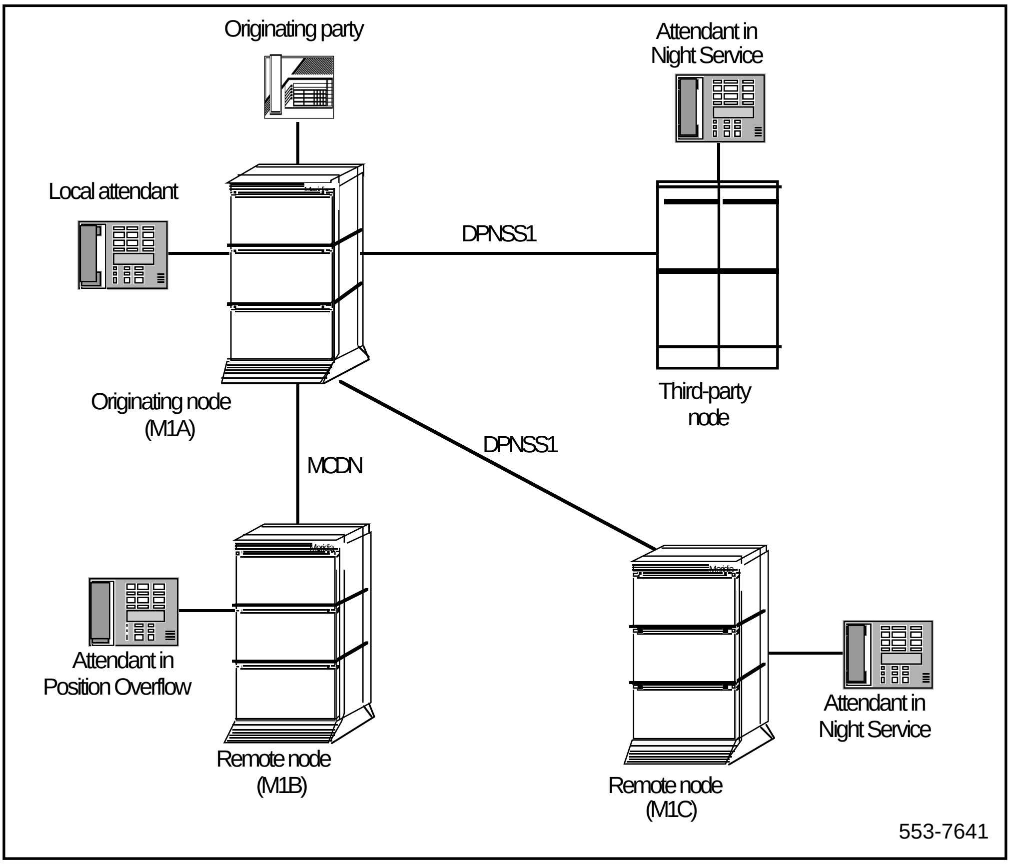
A DPNSS1 call from the originating Meridian 1 node (A) terminates to the attendant on a third-party PBX. The attendant is in Night Service. The third-party PBX signals the Meridian 1 to initiate Night Service Diversion. The call is then diverted back to the originating node, where a new call is initiated to the queue of the local attendant.

DPNSS1

Note: This diversion is the functionality that has been introduced by the DPNSS1 Night Service feature. The call processing which follows is part of the standard Network Attendant Service (NAS) functionality.

At this point, the call is treated as a standard call to the local attendant. If the local attendant is also in Night Service, Network Attendant Service (NAS) routing is applied. The call is routed to a remote attendant (B). Since this attendant is in Position Overflow, it cannot take the call and clears it. The next alternative in the NAS routing table is tried. This is for the originating Meridian 1 to route the call to the remote attendant (C). Here, the attendant is also in Night Service and clears the call. Eventually, the Night DN is tried successfully. The new call from the originating Meridian 1 to the NIGHT DN is kept, and the old call to the third-party PBX is released.

Figure 33- Example of DPNSS1 Night Service Diversion



Feature interactions

DPNSS1 Redirection

A redirected call may undergo Night Service Diversion, if a new call is attempted to an attendant on a third-party PBX that initiates Night Service Diversion.

DPNSS1 Route Optimization

Route Optimization is applied if a non-optimum path has been taken by a call answered by either the third-party PBX on which the target operator, the local attendant, remote attendant, or the Night DN is located.

DPNSS1

DPNSS1 Step Back on Congestion

If a call to the remote attendant encounters congestion, Step Back on Congestion is initiated and attempted at any node.

DPNSS1 Extension Three Party Service

An enquiry call reaching an attendant in Night Service will undergo Night Service diversion, if available.

Diversion

A diverted call reaching an attendant in Night Service will undergo Night Service diversion, if available.

Attendant Incoming Call Indicators

When a Night Service call is diverted to an attendant, the Incoming Call Indicator is the number of the incoming route.This is the same as for a NAS MCDN call routed to an attendant.

Call Waiting

If a call is diverted to a third-party operator Night DN that is busy, Call Waiting may be activated (if equipped). The call to the third-party operator PBX is released.

DPNSS1 Executive Intrusion

An attendant console operator can break in on an established call involving at least one party at a switch connected to the attendant's switch with a DPNSS1 link. The Meridian 1 will accept an Executive Intrusion activation initiated by a regular telephone at a third-party PBX.

The attendant uses the Break-in key on the console for the Executive Intrusion. The Intrusion Capability Level of the console must be higher than the Intrusion Protection Levels of the parties involved in the established call for the intrusion to be allowed.

Alternative configurations

APNSS

Refer to the introductory information provided in the section called *Introduction to ISDN*.

A dedicated D-channel for APNSS signaling is used exclusively for analog bearers, and cannot be used to support DPNSS1 digital bearers. One D-channel can support a maximum of 30 B-channels.

The B-channels for APNSS are normally carried over analog two or four wire E&M trunk circuits, or AC15 trunks. However, digital (DTI2) TIE B-channels can also be used for APNSS.

The D-channel may be carried over a 64 kbps digital link, or an analog link using modem equipment. Normally, the D-channel is run using leased-line modems, but may also be connected using dial-up modems, a 500 line card and any trunk circuit.

APNSS is configured on a route basis, with each trunk on that route being associated with a D-channel number and a trunk identifier to identify the signaling channel for the trunk. Call set-up, establishment, and tear-down are controlled by the DPNSS1 signaling messages and call states.

Virtual channels for APNSS are programmed on an unused loop within the Meridian 1.

Certain limitations apply to APNSS. APNSS supports only PBX to PBX (similar or different) connectivity; with APNSS there is no check for B-channel speech transmission.

DPNSS1 Message Waiting Indication functionality can be configured on APNSS networks. Refer to page 412 for more information.

DPNSS1

DPNSS1 to MCDN Trunk Interworking

If a call is travelling across a UDP DPNSS1-MCDN gateway, enbloc or overlap sending and receiving can be used.

Supplementary services supported are:

- Callback When Free
- Callback When Next Used
- Loop Avoidance

Interworking of Call Diversion with MCDN Trunk Route Optimization (TRO) is supported.

The DPNSS1 Attendant Call Offer feature allows attendant-extended calls routed over DPNSS1 to be camped-on to a remote busy extension. This Call Offer functionality is provided over a DPNSS1 network or over a DPNSS1 to ISDN gateway.

If an incoming call is handled for Network Attendant Services routing towards DPNSS1, no diversion signaling is sent back to the calling party.

If a Meridian 1 system is connected to a voice messaging system on either another vendor's PBX or another Meridian 1 using a series of interfaces consisting of DPNSS and MCDN trunks, the DPNSS-MCDN Message Waiting Indication gateway allows them to have their Message Waiting lamp lit or darkened by the remote message system by providing the means to pass a Message Waiting Indicator message across a DPNSS-MCDN (or MCDN-DPNSS) gateway.

Virtual Network Services in the UK with DASS2/DPNSS1 Bearers

Virtual Network Services (VNS) provides ISDN features to customers when no ISDN Primary Rate Interface (PRI) or ISDN Signalling Link (ISL) Bearer Channels are available between two Meridian 1 switches.

The Virtual Network Services with DASS2/DPNSS1 Bearers feature introduces VNS in the UK using Digital Private Network Signalling System No.1 (DPNSS1) or Digital Access Signalling System No.2 (DASS2) trunks as VNS Bearer trunks.

All of the operating parameters that pertain to the Basic VNS feature also apply to the Virtual Network Services with DASS2/DPNSS1 Bearers feature. The following parameters also apply.

- Analog Private Networking Signalling System (APNSS) trunks cannot function as VNS Bearer trunks.
- No DPNSS1 Supplementary Service is provided when DPNSS1 trunks are used as a VNS Bearer trunk. ISDN features are provided instead. If any of the DPNSS1 Supplementary Service features requires a DPNSS1 route, it cannot use a VNS route.
- If ESN is configured, a route list entry with both VNS and DPNSS1 is not chosen.
- For DPNSS1/VNS gateway nodes in mixed DASS2/DPNSS1 and VNS networks, the gateway nodes are subject to the same feature support and limitations as the standard DPNSS1/ISDN gateway without VNS. If there is no DPNSS1/ISDN gateway, the feature will be stopped at the DPNSS1/VNS node.

DPNSS1 Attendant Call Offer

DPNSS1 Attendant Call Offer is not supported over VNS Bearer trunks. Standard ISDN Camp-on may be provided instead, if NAS is configured over the VNS Bearer trunks.

DPNSS1 Attendant Timed Reminder Recall and Attendant Three-Party Service

DPNSS1 Attendant Timed Reminder Recall and Attendant Three-Party Service are not supported over VNS Bearer trunks. If NAS is configured over the VNS Bearer trunks, NAS call extension and Attendant Recall will be offered instead.

DPNSS1

DPNSS1 Call Back When Free and Call Back When Next Used

DPNSS1 Call Back When Free and Call Back When Next Used are not supported over VNS Bearer trunks. Network Ring Again or Network Ring Again on No Answer may be provided instead, if Network Ring Again or Network Ring Again on No Answer are configured over the VNS Bearer trunks.

DPNSS1 Diversion

DPNSS1 Diversion is not supported over VNS Bearer trunks. Network Call Redirection and Trunk Route Optimization can be provided instead, if configured over the VNS D-Channel.

DPNSS1 Extension Three-Party Service

DPNSS1 Extension Three-Party Service is not supported over VNS Bearer trunks. Network Call Redirection and Trunk Route Optimization can be provided instead, if configured over the VNS D-Channels.

DPNSS1 Loop Avoidance

DPNSS1 Loop Avoidance is not supported over VNS Bearer trunks The ISDN Call Connection Limitation is provided, if it is configured over the VNS D-Channel.

DPNSS1 Route Optimization

DPNSS1 Route Optimization is not supported over VNS Bearer trunks.

DPNSS1 Route Optimization/ISDN Trunk Anti-Tromboning Interworking

ISDN Trunk Anti-Tromboning may be applied to the VNS part of the call, if configured on the VNS D-Channel.

DPNSS1 Step Back On Congestion

DPNSS1 Step Back On Congestion handles high traffic situations when congestion is encountered by DPNSS1 trunks. Some scenarios follow that you can encounter for interworking with VNS.

Homogeneous Networks

DPNSS1 Step Back On Congestion is supported over VNS Bearer trunks, if all the transit nodes within the DPNSS1 network used for VNS are configured accordingly:

— if the SBOC (Step Back On Congestion) prompt is set to NRR (No Reroute) or RRO (Reroute Originator), then it would be sufficient that the VNS originating node be configured with either RRO (Reroute Originator) or RRA (Reroute All).

— if the SBOC (Step Back On Congestion) prompt is set to RRA (Reroute All) for a transit node, then the different alternative routes at this node must be configured with VNS and must be configured as VNS Bearers.

Hybrid Networks

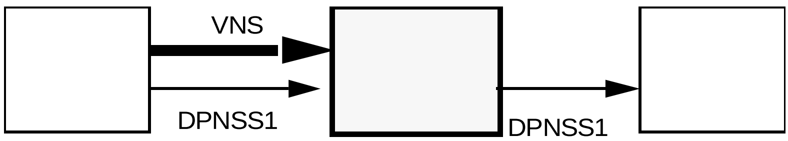
MCDN/VNS with DPNSS1 node

— If congestion is encountered inside the VNS portion of the path, the node behaves as an MCDN/MCDN tandem. The ISDN Drop Back Busy (IDBB) and ISDN Off-Hook Queuing (IOHQ) are transmitted, so that they may applied further along the VNS portion of the path, or at the tandem node.

— If congestion is encountered within the DPNSS1 network, the VNS portion of the call is cleared and the disconnection is propagated back to the originating side of the MCDN path. Neither Drop Back Busy nor Off-Hook Queuing is activated at the tandem node, even if IDBB or IOHQ are activated.

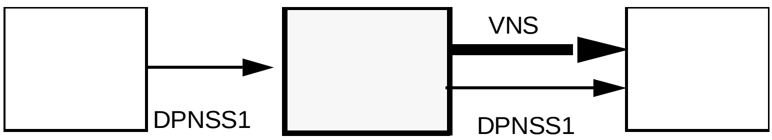
DPNSS1

VNS with DPNSS1/DPNSS1 node



This scenario is considered as an MCDN/DPNSS1 gateway. The functionality is the same as for the Step Back on Congestion feature.

DPNSS1/VNS with DPNSS1 node



— If congestion is encountered inside the VNS portion of the path, the VNS portion of the call is cleared and the disconnection is propagated back to the originating DPNSS1 side. The Step Back on Congestion feature is invoked, if it is configured.

— If congestion is encountered the within the DPNSS1 portion of the path, with the DPNSS1 trunk being used as a VNS Bearer, the VNS portion of the call is cleared and a normal disconnection is propagated back to the originating DPNSS1 side. The Step Back on Congestion feature is not invoked, even if it is configured.

DPNSS1 Executive Intrusion

DPNSS1 Extension Three-Party Service is not supported over VNS Bearer trunks. Attendant Break-in may be provided instead, if NAS is configured over the VNS Bearer trunks.

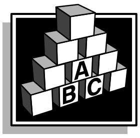
Standalone Meridian Mail

Standalone Meridian Mail is not supported over VNS Bearer trunks. A mailbox user may access Meridian Mail, if the ISDN Network Message Services is configured.

DPNSS1/DASS2 to ISDN PRI Gateway

A VNS call over a DPNSS1 or DASS2 Bearer trunk of an DPNSS1/DASS2 to ISDN PRI Gateway acts as the ISDN leg of the Gateway.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

DPNSS1 Route Optimization/MCDN Trunk Anti-Tromboning Interworking

The DPNSS1 Route Optimization (RO)/Meridian Customer Defined Networking (MCDN) Trunk Anti-Tromboning (TAT) Interworking feature provides RO and TAT interworking at DPNSS1/MCDN gateway nodes.

There is more detail about this capability provided in the section called *Interworking* in this book.

For detailed information on the Trunk Anti-Tromboning feature, refer to the *International ISDN PRI features description and administration* (for 2.0 Mbit applications), or the *ISDN Primary Rate Interface description and administration* (for 1.5 Mbit applications.)

DASS2/DPNSS1 INIT Call Cut Off

In the UK, prior to the introduction of X11 Release 23,when the Meridian 1 is connected with DPNSS1 to an SX2000 PBX, all calls, including established calls, are cleared if the system initializes. With the introduction of Release 23 software and new firmware on the Dual D-Channel Daughterboard, established calls remain active during and after the initialization.

Networks using APNSS do not support this functionality. This functionality is not offered on the Option 11C.

Standalone Meridian Mail

Standalone Meridian Mail provides access for non-Nortel Networks switches to a Meridian 1 with Meridian Mail so it can act as the voice mail for the non-Nortel Networks system users. The interface is done using DPNSS1 signaling.

Compatible switches are:

- Plessey iSDX
- Ericsson MD-110
- GPT BTeX
- Mitel SX-2000

Only the Plessey iSDX can have a Message Waiting Lamp light up when the user has a message.

Access forwarding to Meridian Mail via a full DPNSS1 environment operates from X11 Release 20 and later Meridian 1 nodes. Access forwarding to Meridian Mail via full MCDN/DPNSS1 gateway operates, if all the gateways between the calling party and the Meridian Mail host node, are X11 Release 22 Meridian 1 nodes.

DPNSS1 Message Waiting Indication

This feature allows telephones on a Meridian 1 to respond to a message waiting indication or cancellation message received from another vendor's PBX across a DPNSS1trunk. Release 23 software is required.

DPNSS1 Call Diversion interworks with this feature to make it work. UDP and CDP dialing plans are supported. Visual and audible message waiting indications are supported. The feature can also be configured on APNSS networks.

Any number of DPNSS1 trunks can be between the Voice Messaging Server and the telephone. Refer to the information on DPNSS1 to MCDN Trunk Interworking for a gateway functionality that is offered.

Control tips

- Call Detail Recording (CDR) records are not printed at the originating or terminating PBX, during Route Optimization. CDR records are printed at tandem nodes when the non-optimum path is released. The CDR records print out information as if the call had occurred on the new path at the time that the original trunks were seized.

  The cost of the call (that is, the Periodic Pulse Metering information) that has been optimized is the sum of the cost before Route Optimization plus the cost after optimization. The originator of the original call is shown as the originator of the new call, at the originating PBX. The terminator of the call is shown as the terminator of the new call, at the terminating PBX. At transit PBXs, normal information is printed, showing original tandem connections being released as if for calls being cleared at the time of Route Optimization.

Administration tips

- On Meridian 1 nodes, M3000 Meridian 1 proprietary sets are not supported when using DPNSS1 signaling.
- The forwarded to telephone display is updated in full DPNSS1 or mixed DPNSS1/MCDN routes. The forwarded-to party can be Meridian Mail.

Training tips

- The features Call Back When Free and Call Back When Next Used are only available if the call did not get routed over the public network at any point. The DASS2 networks do not support that functionality. Train users to understand why this feature may not work at times.
- During a Route Optimization attempt, any key operation from a telephone involved in the call is ignored, except the release or onhook function.

  If a telephone that is not involved in a call is configured in a single call multiple appearance DN arrangement with a telephone involved in a Route Optimization attempt, then any key operation that interferes with the Route Optimization attempt is ignored. Therefore, the telephone is inhibited from joining the call during the Route Optimization attempt.

  If your network will have Route Optimization implemented, train users to expect this at times.

What to have ready

The following checklist summarizes the steps you should take before implementing DPNSS1 and/or the optional related features associated with the basic feature.

Table 46
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Ascertain if DPNSS1 is available in your area.
✔		Decide if you want DPNSS1 or APNSS1.
✔		Set up your numbering plan.
✔		Program the NARS and/or CDP database.
✔		Assess the features supported. Determine which of these you want to use.
✔		Decide on the Loop Avoidance Intercept treatment you want. Match it to the Congestion tone.
✔		Decide if all nodes will be able to re-route calls for SBOC. Decide if re-routing will be on Iset routes only or all routes.
✔		Make sure you understand the interworking and gateway operation and features supported if you interface your DPNSS1 network with other kinds of networks.
✔		Decide if you need established calls to remain active if your system initializes.

— continued —

DPNSS1

Table 46
Checklist (Continued)

Basic	Optional	Preparation
	✔	Decide if you will allow users at non-Nortel Networks systems to use the Meridian Mail at a Meridian 1 in the DPNSS1 network.
	✔	Find out if telephones on the Meridian 1 are to be supported by a Voice Messaging Server at another location, connected with DPNSS1 trunks and/or MCDN trunks.
✔		Train users on call operation and tones that will affect them.
✔		Train attendants on the operation of the DPNSS1 features that will affect them.

DASS2

Digital Access Signaling System No. 2 (DASS2) was developed in the UK for PBX access to British Telecom’s public ISDN.

DASS2 provides:

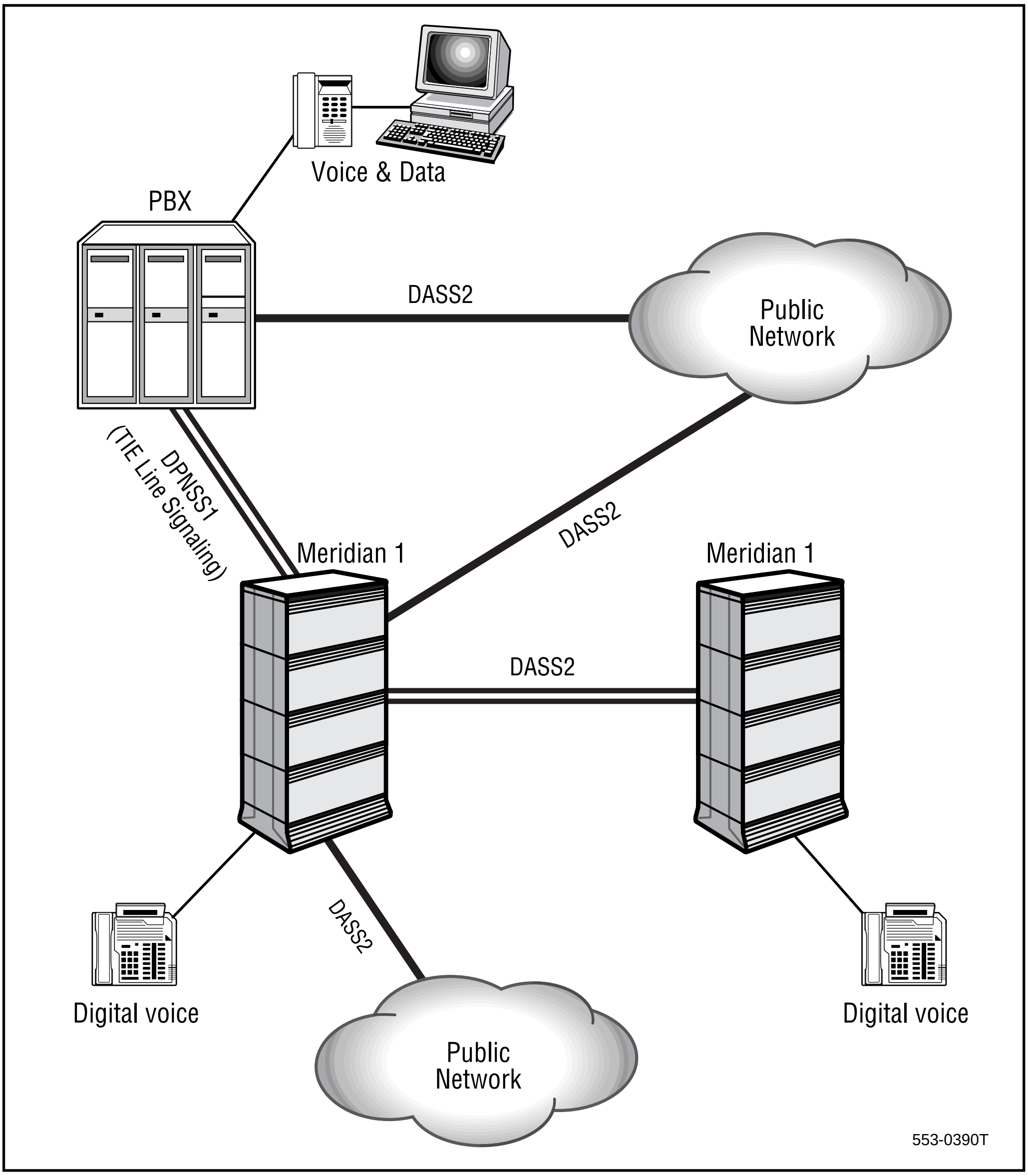
- digital access to the public ISDN for voice and data calls
- supplementary facilities, such as Call Charge Indication and Calling Line Identity
- TIE trunk user-to-user signaling facility, allowing semi-permanent private networking connections to be established between PBXs using the ISDN

Transmission system

The 2.048 Mbps digital transmission carrier is divided into 32 timeslots, numbered 0-31. Timeslots 1-15 and 17-31 provide 30 voice/data traffic channels. Timeslot 0 is used as a synchronization channel. DASS2 is a message-based signaling system that uses a common signaling channel in timeslot 16.

DASS2

Figure 34 - DASS2 configured in a network



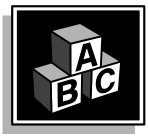
Link designation

The ends of each inter-PBX link are labelled arbitrarily A and B, and the ends of each DASS2 channel are designated X and Y. If both ends attempt to use the channel at the same time, the X end has priority.

End PBXs, gateways and transits

A PBX that connects a DASS2 channel to or from a non-DASS2 device is termed an end PBX. If that device is a trunk, then the PBX is termed a gateway. A PBX connecting two DASS2 channels is a transit.

Basic configuration



For more information about the technical details related to DASS2, refer to the *DASS2 and DPNSS1 (UK) NTP*. Refer to the *Introduction to ISDN* section for introductory information about DASS2 and DPNSS1.

Configuration of trunks

D-channels

This out-of-band, 64 kbps signaling channel may be on the same or a different carrying medium as the B-channels it supports. One D-channel can support up to 30 B-channels.

B-channels

Voice and data transmission over DASS2 links is done at a rate of 64 kbps.

DASS2

DASS2 trunks are configured using the same route and member method used for other trunks. Other characteristics of DASS2 trunks are:

- any number of routes may be associated with the same link
- a route may be associated with any number of links
- each route member must be assigned to one channel
- not all channels need to be associated with members. These non-associated channels cannot, however, be used for calls
- members and channels must be numbered separately
- members are screened for outgoing calls using a linear search (Sequential Line) preferably
- each route may be configured only for incoming calls, only for outgoing calls, or for both
- each route must be configured with DASS2 channels only

Numbering plans

With the exception of DPNSS1 Trunk Identities, there is very little difference between DASS2 routes and standard Direct Dial In (DDI)/Exchange routes where numbering is concerned. Incoming DASS2 (public network) calls can be translated by NARS if they are in the format: AC-LOC/SPN- X...X. Incoming DDI DASS2 calls may be routed through the private network on DPNSS1 or non-IDA trunks. The received digits must be converted into the appropriate CDP steering code or NARS and BARS Special Number.

For outgoing calls, DASS2 routes may be accessed by:

- Coordinated Dialing Plan (CDP) Trunk Steering Code
- direct access code
- NARS and BARS Special Numbers
- pre-translation resulting in any of the above

There is more information on these dialing plans in this book. Please refer to the Table of Contents.

DASS2 Line Identities

Any ISDN Line Identities to be displayed on the Meridian 1 must be preceded by the “ISDN Access Code”. You define this code in the customer data block. It must be defined at each network node which has access to the ISDN, either locally or through the DPNSS1 network.

If a non-DDI extension makes an outgoing call to the ISDN, no line identity will be supplied by the private network. The ISDN must provide a default line identity.

Trunk Identities

When used with DPNSS1, a “trunk identity” will be used for originating or called line identity exchange and display purposes, for DPNSS1 calls.

Improving feature performance

The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Interworking

Full interworking is provided between DASS2 and the following signaling interfaces:

- Q.931 MCDN
- Q.931 public ISDN access
- DPNSS1 private network

Note that DPNSS features and services are available only in DPNSS1/DASS2 network environments, and may not be available in your area. For more information, contact your system supplier.

DASS2

DASS2/DPNSS1 INIT Call Cut Off

In the UK, prior to the introduction of X11 Release 23,when the Meridian 1 is connected with DASS2 to a System Y exchange, all calls, including established calls, are cleared if the system initializes. With the introduction of Release 23 software and firmware on the Dual D-Channel Daughterboard, established calls remain active during and after the initialization. This functionality is not offered on the Option 11C.

Virtual Network Services in the UK with DASS2/DPNSS1 Bearers

Virtual Network Services (VNS) provides ISDN features to customers when no ISDN Primary Rate Interface (PRI) or ISDN Signalling Link (ISL) Bearer Channels are available between two Meridian 1 switches.

The Virtual Network Services with DASS2/DPNSS1 Bearers feature introduces VNS in the UK using Digital Private Network Signalling System No.1 (DPNSS1) or Digital Access Signalling System No.2 (DASS2) trunks as VNS Bearer trunks.

All of the operating parameters that pertain to the Basic VNS feature also apply to the Virtual Network Services with DASS2/DPNSS1 Bearers feature. The following parameters also apply.

- Analog Private Networking Signalling System (APNSS) trunks cannot function as VNS Bearer trunks.
- No DPNSS1 Supplementary Service is provided when DPNSS1 trunks are used as a VNS Bearer trunk. ISDN features are provided instead. If any of the DPNSS1 Supplementary Service features requires a DPNSS1 route, it cannot use a VNS route.
- If ESN is configured, a route list entry with both VNS and DPNSS1 is not chosen.
- For DPNSS1/VNS gateway nodes in mixed DASS2/DPNSS1 and VNS networks, the gateway nodes are subject to the same feature support and limitations as the standard DPNSS1/ISDN gateway without VNS. If there is no DPNSS1/ISDN gateway, the feature will be stopped at the DPNSS1/VNS node.

DPNSS1 Attendant Call Offer

DPNSS1 Attendant Call Offer is not supported over VNS Bearer trunks. Standard ISDN Camp-on may be provided instead, if NAS is configured over the VNS Bearer trunks.

DPNSS1 Attendant Timed Reminder Recall and Attendant Three-Party Service

DPNSS1 Attendant Timed Reminder Recall and Attendant Three-Party Service are not supported over VNS Bearer trunks. If NAS is configured over the VNS Bearer trunks, NAS call extension and Attendant Recall will be offered instead.

DPNSS1 Call Back When Free and Call Back When Next Used

DPNSS1 Call Back When Free and Call Back When Next Used are not supported over VNS Bearer trunks. Network Ring Again or Network Ring Again on No Answer may be provided instead, if Network Ring Again or Network Ring Again on No Answer are configured over the VNS Bearer trunks.

DPNSS1 Diversion

DPNSS1 Diversion is not supported over VNS Bearer trunks. Network Call Redirection and Trunk Route Optimization can be provided instead, if configured over the VNS D-Channel.

DPNSS1 Extension Three-Party Service

DPNSS1 Extension Three-Party Service is not supported over VNS Bearer trunks. Network Call Redirection and Trunk Route Optimization can be provided instead, if configured over the VNS D-Channels.

DPNSS1 Loop Avoidance

DPNSS1 Loop Avoidance is not supported over VNS Bearer trunks The ISDN Call Connection Limitation is provided, if it is configured over the VNS D-Channel.

DPNSS1 Route Optimization

DPNSS1 Route Optimization is not supported over VNS Bearer trunks.

DASS2

DPNSS1 Route Optimization/ISDN Trunk Anti-Tromboning Interworking

ISDN Trunk Anti-Tromboning may be applied to the VNS part of the call, if configured on the VNS D-Channel.

DPNSS1 Step Back On Congestion

DPNSS1 Step Back On Congestion handles high traffic situations when congestion is encountered by DPNSS1 trunks. Some scenarios follow that you can encounter for interworking with VNS.

Homogeneous Networks

DPNSS1 Step Back On Congestion is supported over VNS Bearer trunks, if all the transit nodes within the DPNSS1 network used for VNS are configured accordingly:

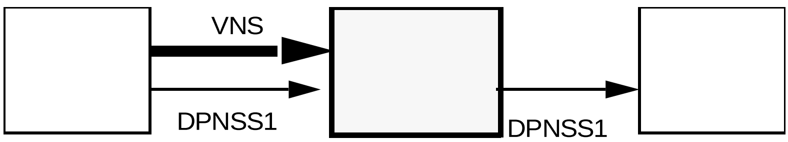
- if the SBOC (Step Back On Congestion) prompt is set to NRR (No Reroute) or RRO (Reroute Originator), then it would be sufficient that the VNS originating node be configured with either RRO (Reroute Originator) or RRA (Reroute All).
- if the SBOC (Step Back On Congestion) prompt is set to RRA (Reroute All) for a transit node, then the different alternative routes at this node must be configured with VNS and must be configured as VNS Bearers.

Hybrid Networks

MCDN/VNS with DPNSS1 node

- If congestion is encountered inside the VNS portion of the path, the node behaves as an MCDN/MCDN tandem. The ISDN Drop Back Busy (IDBB) and ISDN Off-Hook Queuing (IOHQ) are transmitted, so that they may applied further along the VNS portion of the path, or at the tandem node.
- If congestion is encountered within the DPNSS1 network, the VNS portion of the call is cleared and the disconnection is propagated back to the originating side of the MCDN path. Neither Drop Back Busy nor Off-Hook Queuing is activated at the tandem node, even if IDBB or IOHQ are activated.

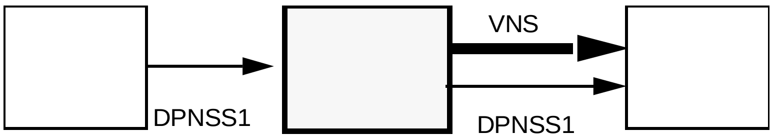
VNS with DPNSS1/DPNSS1 node



This scenario is considered as an MCDN/DPNSS1 gateway.
The functionality is the same as for the Step Back on Congestion feature.

DASS2

DPNSS1/VNS with DPNSS1 node



- If congestion is encountered inside the VNS portion of the path, the VNS portion of the call is cleared and the disconnection is propagated back to the originating DPNSS1 side. The Step Back on Congestion feature is invoked, if it is configured.
- If congestion is encountered the within the DPNSS1 portion of the path, with the DPNSS1 trunk being used as a VNS Bearer, the VNS portion of the call is cleared and a normal disconnection is propagated back to the originating DPNSS1 side. The Step Back on Congestion feature is not invoked, even if it is configured.

DPNSS1 Executive Intrusion

DPNSS1 Extension Three-Party Service is not supported over VNS Bearer trunks. Attendant Break-in may be provided instead, if NAS is configured over the VNS Bearer trunks.

Standalone Meridian Mail

Standalone Meridian Mail is not supported over VNS Bearer trunks. A mailbox user may access Meridian Mail, if the ISDN Network Message Services is configured.

DPNSS1/DASS2 to ISDN PRI Gateway

A VNS call over a DPNSS1 or DASS2 Bearer trunk of an DPNSS1/DASS2 to ISDN PRI Gateway acts as the ISDN leg of the Gateway.

Control tips

- The numbering plan and call routing should be set up in such a way that private network calls are not routed on DASS2 routes and thus do not incur call charges.

Administration tips

- You cannot combine DASS2 and non-DASS2 trunks in one route.

Training tips

- Make sure users understand how to dial internal DNs, private network DNs and public network calls.

DASS2

What to have ready

The following checklist summarizes the steps you should take before implementing DASS2.

Table 47
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Find out if DASS2 is available in your area.
✔		Set up your numbering plan.
✔		Program the BARS or NARS and/or CDP database.
✔		Make sure you understand the interworking and gateway operation and features supported if you interface your DASS2 network with other kinds of networks.
✔		Decide if you need established calls to remain active if your system initializes.
✔		Train users on the dialing plan.

EuroISDN

This feature provides ISDN Primary and Basic Rate Interfaces to COs that comply with European Telecom Standards Institute (ETSI) specifications. The specifications continue to evolve and new document numbers are assigned. EuroISDN connectivity continues to evolve also, remaining compliant with the latest standards.

Basic configuration

EuroISDN (Release 20)

The specification that X11 Release 20 software complies with is ETS 300 102 for layer 3 messages. This EuroISDN connectivity is in compliance with layers 1 and 2, as required in the ETSI standard. The interfaces introduced in Release 20 also comply with country-specific Application Documents for:

- Austria
- Denmark
- Finland
- Germany
- Holland
- Ireland
- Italy
- Norway
- Portugal
- Sweden
- Switzerland (Basic Rate only)

EuroISDN

The supplementary services Advice of Charge, Calling Line Identification Presentation, Calling Line Identification Restriction, Connected Line Identification Presentation and Connected Line Identification Restriction are provided for the countries listed above where application documents for those services were available when the feature was being developed.

If you have a system in a country not specifically mentioned in the previous list, the EuroISDN interface may be compatible if:

- the EuroISDN offered is compliant with the ETS 300 102 standard
- it does not include conditions which are not supported by this feature
- it does not require anything considered to be limitations of this feature at this time. Limitations are such things as:
  - The country-specific application document may have changed since the interface was developed. Check for recent compatibility tests in your country.
  - There is no Low Layer or High Layer Compatibility checking. This Information Element (IE) can be tandemed by the Meridian 1 but it will not be used to perform any checking.
  - There is no Transit network selection IE in the SETUP message. This was not required by any country when the feature was developed.
  - Extension for symmetric call operation for private networks is not supported. The QSIG protocol is used on Meridian 1 instead.
  - Network specific facility selection procedures are not supported.
  - D-channel backup procedures described in ETS 300 102-1, Annex F are not supported.
  - Message segmentation procedures as described in ETS 300 102-1, Annex K are not supported. These procedures have been implemented in Italy and Norway. Check with your supplier, if your system is in one of these two countries.

- The USER INFORMATION message is not implemented.
- For the CLIP supplementary service, the calling user cannot insert the calling line identity at call request, on a per call basis, as the Meridian 1 configures CLIP using a Class of Service on a telephone basis.
- For the COLP supplementary service, the connected user cannot insert the connected line identity on call acceptance, on a per call basis, as the Meridian 1 configures COLP using a Class of Service on a telephone basis.
- The same Class of Service is used to control both CLIR and COLR supplementary services. If a user has presentation restricted, its number is not presented to the other party for both incoming and outgoing calls of this user.
- Supplementary services requiring temporary signaling connections are not supported.

For Connected Line Identification Presentation (COLP) supplementary service, sending or restricting connected line identification on a per call basis is not supported on ISDN BRI telephones.

Refer to the *CLID and Name Display options* section for more information about COLP, CLIR, COLR and CLIP supplementary services.

Hardware

2.0 Mbps PRI is used for EuroISDN connectivity. This type of facility is not available in North America.

Large systems must use an MSDL card for the PRI2 D-channel or a downloadable D-channel daughterboard on a Dual DTI/PRI2 (DDP2) card is required for the D-channel. EuroISDN is not supported on non-downloadable D-channel cards.

EuroISDN

Features supported

As new releases of software are introduced, the list of supported features will expand. As of Release 20, the supported features are:

- Basic Call Service
- 64 kbps bearer capability
- DID, DOD, COT call types
- Channel negotiation
- Overlap sending and Overlap receiving
- Flexible Numbering Plan
- Calling Line Identification Presentation and Calling Line Identification Restriction (refer to the *CLID and Name Display options* section of this book)
- Connected Line Presentation (COLP) and Connected Line Restriction (COLR) (refer to the *CLID and Name Display options* section of this book)

  Note: In Austria, EuroISDN COLP and COLR are not supported.

- Advice of Charge (some countries)

TIE trunks

The transfer of an unanswered EuroISDN call to a remote ringing telephone requires disconnect supervision from a TIE trunk. If disconnect supervision is not available when an external user goes on-hook, the trunk could be locked out. Therefore, it is important to ensure that trunks used for a EuroISDN call have disconnect supervision.

Advice of Charge (AOC) for EuroISDN

This feature can be applied to EuroISDN networks where billing is required for each call made to the Central Office. It replaces the older Periodic Pulse Metering (PPM).

AOC for EuroISDN complies with the European Telecom Standard Institute (ETSI) standard specifications ETS 300 102.

EuroISDN

The capability was introduced in Release 20. It is supported on ISDN PRI2 and ISDN BRI trunks. In Release 20, the following countries supported AOC on EuroISDN: Austria, Finland, Germany, Holland, Italy, Norway and Portugal.

For EuroISDN, the feature introduced AOC:

- at the start of the call (AOC-S)
- during the call (AOC-D) on a Real Time basis
- at the end of the call (AOC-E).

Advice of Charge display requires a Meridian 1 proprietary telephone. With an analog telephone the charge can be presented to CDR records and Background Terminals but there is no display to the user. You will need a TTY to test both Call Detail Recording and the Background Terminal.

Real Time Display, during the call, is supported only on Meridian Modular Telephones. Some countries do not support this feature.

The display is under the control of the Central Office. Users cannot request the information when they want.

Reverse Charging is not supported.

Functional Protocol and Keypad Protocol

The following is the distinction between Functional Protocol and Keypad Protocol.

- Functional Protocol incorporates functional Information Elements (IEs) which require a degree of intelligent processing by a terminal. No user operation is required. Functional Protocol may operate end-to-end.
- In general, Keypad Protocol has only local significance. Keypad Protocol incorporates stimulus IEs, which are generated by a single event at the user/terminal interface, or by a basic instruction from the network to be executed by the user. The user device does not have to have a knowledge of the supplementary service.

EuroISDN

The Functional AOC protocol is not country dependent and is available for all interfaces for which EuroISDN is supported, including Austria, Belgium, Denmark, Finland, Germany, Holland, Italy, Ireland, Portugal, Spain, Switzerland, Sweden and the United Kingdom. It is useful in countries where Keypad Protocol is still offered by operators. AOC is considered to evolve to comply with Functional Protocol.

With Release 22, the following countries support Keypad Protocol on EuroISDN: Belgium, Holland, Italy, Portugal, Spain and Switzerland. Keypad Protocol for the ETSI type of interface is not supported because it is country dependent.

ISDN QSIG/EuroISDN Call Completion (Release 22)

The ISDN QSIG/EuroISDN Call Completion feature expands Ring Again functionalities on QSIG and EuroISDN interfaces. This feature uses the ETSI Generic Functional (GF) as a transport platform.

This feature provides the following supplementary services:

- Call Completion to Busy Subscriber (CCBS)
- Call Completion on No Response (CCNR)

Call Completion to Busy Subscriber (CCBS) supplementary service allows the calling party to apply a Ring Again request when encountering a busy Directory Number (DN). The Meridian 1 alerts the calling party when the occupied DN is available to receive a call. CCBS is supported on QSIG and EuroISDN signaling protocols.

A PINX (Private Integrated Services Network Exchange) DN must be configured for the Customer Group. The PINX DN is used for routing free notification on the MCDN or DPNSS1 network if the EuroISDN network does not provide a calling number for the service. The DN configured should be consistent with the numbering plan used at the site. The PINX DN is also used for incoming DID calls in the same way the Listed Directory Number 0 (LDN0) is used for a basic call.

Call Completion on No Response is provided on QSIG networks only, not on EuroISDN networks.

Existing limitations, applicable to stand alone or network wide Ring Again operation, apply to CCBS on EuroISDN.

ISDN QSIG/EuroISDN Call Completion supports Uniform Dialing, Customer Dialing and Group Dialing plans.

Gateways between EuroISDN CCBS, MCDN Network Ring Again on Busy and DPNSS1 Call Back When Free are supported.

Call Completion to Busy Subscriber is not supported on ISDN BRI telephones.

ISDN QSIG/EuroISDN Call Completion Enhancement

In X11 Release 24, the following enhancements are available:

- the Operation Coding method for the CCBS and CCNR Supplementary Services is configurable on the D-channel for PRI routes and on the route for BRI trunks. The two methods are:
  - Object Identifier – used for ETSI interfaces
  - Integer Value – used for ISO interfaces
- the CCBS and CCNR features are supported over ETSI/QSIG to MCDN and DPNSS1 gateways
- the Call Completion services can be supported in all EuroISDN countries

CCBS is available only on EuroISDN interfaces supporting the ETSI QSIG GF transport protocol.

There are two methods used for call completion:

- Connection Retention. (The signaling connection remains established until call completion occurs or the call completion attempt is cancelled.)
- Connection Release. (The signaling connection is cleared after each phase of call signaling. A new signaling connection is established for each subsequent phase of call-independent signaling.)

EuroISDN

On the originating side, the Meridian 1 supports only Connection Retention. This matches the call completion method performed by EuroISDN. On the terminating side the Meridian 1 also supports Connection Release. With Connection Release you must have a symmetrical dialing plan. In other words, all switches must be equipped with NARS, if the CLIDs are in the UDP format; all switches must be equipped with CDP if the CLIDs are in the CDP format.

There are two methods used for call completion recall:

- Non-path Reservation. (A B-channel connection between the originating and terminating sides is established after the originating party answers the recall. The recall attempt is cancelled if there is network congestion.)
- Path Reservation. (A B-channel between the originating and terminating parties is established before presenting the recall to the originating party, in order to avoid network congestion.)

On the originating side, the Meridian 1 supports only the Non-path Reservation method for call completion recall. This matches the call completion recall method performed by EuroISDN and MCDN and the Path Reservation method is not supported over QSIG to MCDN or DPNSS1 gateways. On the terminating side the Meridian 1 also supports Path Reservation.

If the called party is busy again after the originating party responds to a call completion recall, there are two methods possible to handle the situation:

- Request Retention. (The call completion request remains in place at both the originating and terminating sides. The terminating side begins monitoring the terminating party.)
- Service Cancellation. (The call completion request is cancelled at both the originating and terminating sides.)

The Meridian 1 supports the Service Cancellation method.

EuroISDN Continuation (Release 22)

In Release 22, EuroISDN connectivity expanded to include application support for Switzerland (PRI connectivity added), Spain, Belgium and the United Kingdom.

This development also provides a subset of the ETSI Generic functional protocol for the support of supplementary services. Implementation relies on the Generic Functional (GF) Transport platform introduced in the X11 Release 22. The ETSI GF protocol is implemented on the ETSI, Swiss, German and Danish EuroISDN interfaces.

Layer 1 and Layer 2 compliance with ETSI requirements are also supported.

In addition, the following functionalities are provided by the EuroISDN Continuation feature:

- Intercept treatment upon reception of an invalid or incomplete called party number
- interception to an attendant for EuroISDN voice calls terminating on a data device
- the capability to listen to tones and announcements provided by the Central Office on call clearing (T306 supported)
- the capability of transferring outgoing EuroISDN calls after completion of dialing (Italy only)
- Calling Line Identification and Connected Line Identification transparency to or from EuroISDN to or from a Basic Rate Interface (BRI) telephone
- the configuration of Connected Line Presentation (COLP) on a per D-channel basis (remote capability)
- 3.1 KHz bearer capability for outgoing fax calls, based on a Class of Service assigned to the analog (500/2500 type) TN
- the capability of defining the bearer as “Voice” or “3.1 KHz” on a system basis

- the addition and display of a national or international prefix in front of the received Calling/Connected Line Identification on incoming/outgoing EuroISDN calls
- flexible national and local prefixes in addition to the Calling/Connected Line Identification on incoming/outgoing calls based on the route configuration
- User-to-User information transparency in call control messages
- capability of using different options to build the Calling Line Identification (CLID)
- called party number size increase to 31 octets

EuroISDN Continuation Phase III (Release 23)

In Release 23, EuroISDN connectivity expanded to include application support for France, Russia and the Ukraine.

In addition to providing the capabilities supported by the earlier EuroISDN developments, improvements have been made to call treatments for BRI calls. When a user of a BRI telephone calls a busy or unassigned DN, an in-band tone or announcement is heard and the display indicates the cause of the rejected call. The display capability is the feature added in Release 23. This display shows up for internal calls and external calls using EuroISDN or MCDN networks. Another improvement is to pass a RELEASE message to a BRI terminal immediately, without providing a tone, when a BRI telephone originates a data call that is rejected.

Additional functionalities affecting all EuroISDN interfaces are also part of Release 23. These functionalities are:

- Optional sending of last forwarding DN as CLID
- Trunk Route Optimization - before Answer applied to incoming EuroISDN trunks
- Numbering Plan Identification (NPI) and Type of Number (TON) included in CDR tickets for EuroISDN calls

EuroISDN

Optional sending of last forwarding DN as CLID

If a call redirection occurs at the gateway node, the last forwarding DN is sent as a CLID to the terminating telephone. Redirections supported are Call Forward All Calls, Call Forward No Answer, Hunting, ACD Night Call Forward (NCFW) and ACD Interflow.

If the last redirection is due to NCFW or Interflow, and if that redirection is preceded by one or more redirections on the same node, the called number on that node is sent as the CLID rather than the ACD DN.

When the redirection does not occur at the gateway node for calls redirected to a public network Central Office or a private exchange over a standard ISDN connectivity, the outgoing CLID from the gateway switch is that of the first redirecting DN on the last redirecting node.

Trunk Route Optimization - before Answer on EuroISDN trunks

If a call involves a EuroISDN trunk and one or more MCDN trunks between several switches, when a redundant loop of MCDN trunks results from a call redirection, the redundant trunks are dropped and a direct connection remains established between the caller and the final destination telephone. Redirections supported are Call Forward All Calls, Call Forward No Answer, Call Forward Busy and Hunting.

Numbering Plan Identification (NPI) and Type of Number (TON) in CDR tickets for EuroISDN calls

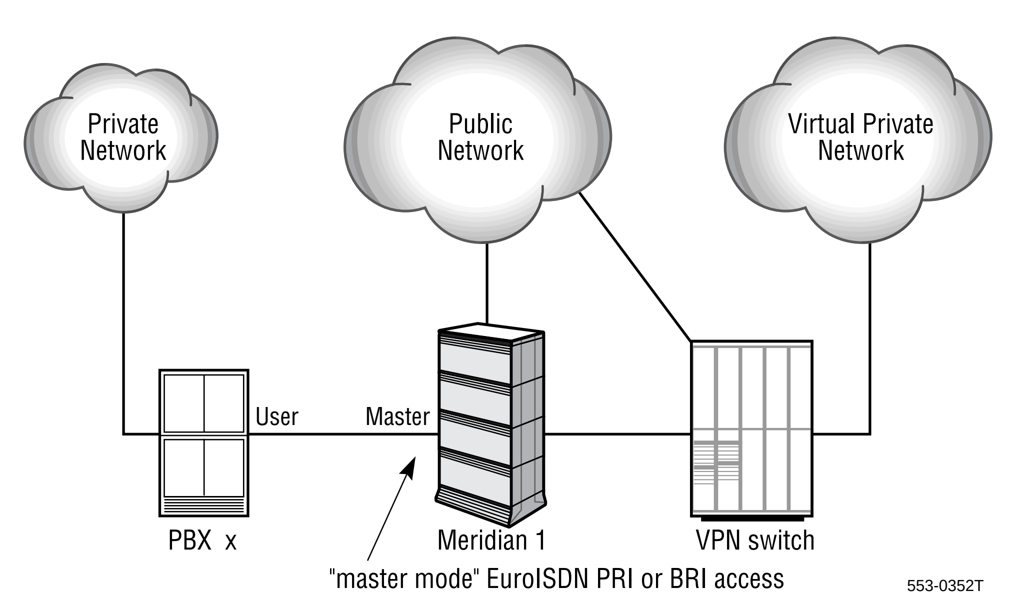
This functionality applies to incoming calls over ISDN trunks allowing calls to be billed to the call originator. Refer to the *Call Detail Recording* section of this book for more detailed information.

EuroISDN Trunk - Network Side

The EuroISDN interface delivered with Release 20 has the behavior of a terminal interface (generally referred to as the “user side”. EuroISDN Trunk - Network Side refers to the behavior of EuroISDN as a network interface (generally referred to as “network side”).

EuroISDN

Figure 35 - EuroISDN Trunk - Network Side configured in a network



Virtual Private Networks offer network users reduced telephone bills by using their VPN network instead of the public network. The Meridian 1 may be used as a "remote point of presence" (RPOP) in a VPN, closest to the customer premises. It supports many different trunk connectivities and it has a lower cost than a public network switch.

As part of the EuroISDN Trunk - Network Side feature, the following capabilities are provided:

- ETSI EuroISDN network side compliance to go along with the ETSI EuroISDN user side compliance delivered by Generic X11 Release 20 software. The ETSI EuroISDN network side interface provides the capability to connect the ETSI EuroISDN user side interface of any system to the Meridian 1. Another arrangement could be the connection of key systems to a Meridian 1, to form part of the private network.

EuroISDN

Similarly, this product also offers a trunk interface for the connection of terminal equipment, such as fax servers, routers and multiplexers, which would normally connect to the public network, but which now can be connected to the Meridian 1 to achieve a greater public network connection efficiency. Such access to the Meridian 1 may be via ISDN Primary Rate Interface (ISDN PRI) or ISDN Basic Rate Interface (ISDN BRI) trunks.

The network side interface provides all feature operations, interactions and gateways that are supported by the user side. It supports the following EuroISDN capabilities:

- Basic call
- Direct Inward Dialing
- Sub-addressing
- Calling Line Identification Presentation (CLIP)/Calling Line Identification Restriction (CLIR)
- Connected Line Identification Presentation (COLP)/Connected Line Identification Restriction(COLR), and CCBS
- User-to-User Signaling 1 (UUS1) information exchange transparency, which allows a calling S0 terminal and a called S0 terminal to exchange small amounts of data over an ISDN PRI or ISDN BRI trunk’s D-Channel. This data is contained in the USER_USER Information Element (IE) within the call control messages. The terminals cannot be BRI terminals. The User-to-User transparency capability is only supported between EuroISDN trunks.
- Bearer Capability-Based Routing, which allows outgoing calls to be selectively routed (over Central Office, Direct Inward Dialing, or TIE trunks for ISDN routes, and Integrated Digital Access trunks for DPNSS1 and DASS2 routes) based on the its Bearer Capability. Any ISDN PRI or ISDN BRI route can be configured to be dedicated to handle voice calls only, data calls only, voice with 3.1 KHz, data with 3.1 KHz, or both voice and data with 3.1 KHz.

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The following protocols are supported by Bearer Capability-Based Routing:

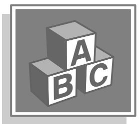
– EuroISDN
– Asia Pacific
– Numeris
– 1TR6
– SWISSNET2
– AXE10 (for Sweden and Australia)
– SYS12
– D70
– QSIG
– DPNSS1
– DASS2
– MCDN
– All North American ISDN connectivities (Meridian 1 to Meridian 1, Meridian 1 to SL-100, Meridian 1 to DMS-100/250, Meridian 1 to AT&T 4ESS, Meridian 1 to AT&T 4ESS/5ESS).

EuroISDN Malicious Call Identification

Sometimes a telephone user wants to have an incoming call traced to find out the identity of the caller, usually for security purposes. With Release 23, if a call comes from the public network on a EuroISDN trunk, the user can activate the Malicious Call Trace feature from the telephone thereby generating a call record at the network exchange office. This record includes the CLID of the caller along with the Numbering Plan Identifier and Type of Number information which can be configured in Release 23 CDR records. Refer to the section on *Call Detail Recording* in this book for further information.

This functionality is offered on PRI trunks and BRI trunks.

EuroISDN



The trace can be activated during the call or for a short time after the call is finished. Only one call can be traced from a telephone at a time. Analog (500/2500) type telephone users use the switch-hook to put the call on hold briefly and then they dial a feature access code. Proprietary telephones have a key assigned.

The Meridian 1 can be configured to activate alarms and recorder trunks before the trace request is sent to the switch in the public network.

For the situation where the caller disconnects, there is a configurable timer you can program on routes to delay the actual disconnection of the call for up to 30 seconds so the call can be traced. Routes between switches can be programmed to delay disconnection if calls from a remote Meridian 1, not directly connected to the public network, may need to be traced.

Refer to the *Interworking and gateways* section of this book for information on networks using MCDN and EuroISDN connectivity.

EuroISDN ETS 300-403 Compliance Update (Release 24)

With X11 Release 24, EuroISDN Primary Rate and Basic Rate Interfaces to Central Offices comply with the ETS 300 403-1 ETSI standard.

The main enhanced functionalities are:

- signaling procedures for bearer capability and High Layer Capability are supported. These changes aim at providing higher quality bearer services or teleservices with alternate bearer capability or High Layer Compatibility in case of fallback. The Fallback mechanism occurs when a called user, receiving two choices of teleservices in the set up message, accepts the alternate one offered. For example, if the called user does not support the 7kHz teleservice, it accepts the alternate teleservice provided (3.1 kHz).

  Note: Teleservices and bearer services are part of basic telecommunications services. Bearer services are services such as: unrestricted 64 kbps for data calls, 3.1 kHz audio for speech and 3.1 kHz bandwidth of audio information transfer. These are

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information transfer services using only layers 1 to 3 of the ISO layer structures. If the provided service is extended to all the layers (1 to 7) of the ISO structure, then such a service is called a teleservice. Teleservices are services such as: Group 4 Facsimile, videotex and videotelephony.

- Basic telecommunication service identification is supported. Each basic telecommunication service has a bearer capability IE and, if applicable, the required High Layer Compatibility IE encoding defined for that service. The requested teleservice is identified by taking the presented bearer capability and High Layer Compatibility IEs in all combinations. If there is no valid combination, the presented bearer capability IE is considered in order to identify a bearer service. One of the features supported by this capability is known as *EuroISDN 7kHz Videotelephony teleservices*.

The interfaces provided by this feature comply with the country specific Application Documents for the following countries:

- Switzerland
- Austria
- Denmark
- Finland
- Germany
- Italy
- Norway
- Portugal
- Holland
- Ireland
- Sweden
- Spain
- Belgium
- UK

- France
- CIS

You can configure an interface to behave like an extended ETS 300 102 compliant interface, if you do not have a connectivity requiring compliance with ETS 300 403. That is, in addition to the existing ETS 300 102 capabilities, the interface can have the bearer capability and High Layer Compatibility selection procedures (used in the fall-back mechanism), and the basic telecommunication service identification. Use these to take advantage of teleservices being offered in Release 24, such as 7kHz telephony and Videotelephony. You can also program the interface to operate as it did prior to X11 Release 24, (compliant with ETS 300 102 with no additional capabilities).

Existing EuroISDN functionalities supported by the EuroISDN ETS 300 403 Compliance Update feature

The ETSI GF subset provides the following two types of functions to the supplementary services control entities:

- Bearer-related transport with a point-to-point transport mechanism. This service is used for the transport of supplementary service protocol information in association with a basic call.
- Bearer-independent transport with a point-to-point facility. This service is used for the transport of supplementary service protocol information which is entirely independent of any existing basic calls.

  Note: The ETSI GF does not by itself control any supplementary service but rather provides a generic transport platform that will support ETSI compliance supplementary services. The ETSI Generic Functional protocol is implemented for all ETSI interfaces.

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The following call services are supported by the Meridian 1 on Release 24 EuroISDN connectivities:

- Basic call service (3.1 kHz, speech, unrestricted digital information)
- 64 kbps bearer capability
- DID, DOD, COT and TIE call types
- Channel negotiation
- Enbloc dialing
- Overlap sending and Overlap receiving
- Flexible Numbering Plan
- Calling Line Identification Presentation and Calling Line Identification Restriction
- Connected Line Presentation and Connected Line Restriction
- Malicious Call Identification (MCID)
- Call Completion to a Busy Subscriber (CCBS)
- 7 kHZ telephony (with fallback to 3.1 kHz)
- Videotelephony
- Advice of Charge

The following functionalities are available for EuroISDN interfaces. When not specified, these functionalities are networkwide:

- Intercept Treatment on reception of an invalid or incomplete called party number
- Interception to an attendant for incoming EuroISDN voice calls terminating on a data telephone
- Capability to listen to Tones and Announcements provided by the Central Office on call clearing
- Capability of transferring outgoing EuroISDN calls after completion of dialing (applies only for Italy)

- Calling Line Identification and Connected Line Identification transparency to and from EuroISDN, to and from a BRI telephone
- Connected Line Presentation service, configurable for each D-Channel, as a remote capability
- Capability of providing 3.1KHz audio bearer capability for outgoing fax calls, based on a Class of Service assigned to a 500/2500 TN
- Capability of defining bearer capability “Speech” or “3.1KHz” on a system basis. (This functionality is also provided on ISDN interfaces.)
- Addition and display of a National or International Prefix on top of the received Calling or Connected Line Identification on incoming or outgoing EuroISDN calls
- Addition of the Flexible National and Local prefixes in the Calling or Connected Line Identification for outgoing or incoming calls, based on the route configuration
- User to User information transparency in a call control message (only for EuroISDN trunk interfaces)
- Called party number size increases to 31 octets
- Capability of mapping a PROGRESS message or a Progress Indicator in the CALL PROCEEDING message, into an ALERT or a CONNECT message on an individual configuration basis
- Capability of using different options to build the CLID
- Capability of sending a RELEASE message to a BRI telephone, with cause and Progress Indicator (PI) Number 8. The user is provided with a display of cause in addition to a tone, for the following scenarios:
  - an internal BRI telephone originating a call to a busy or invalid DN
  - when receiving a disconnect message from the Central Office by way of EuroISDN (if a Progress Indicator (PI) Number 8 is not present in the disconnect message, the tone is provided locally. Otherwise, an inband tone is remotely provided).

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- when receiving a disconnect message over an MCDN network

- ◆ Capability of immediately releasing rejected BRI data calls without providing tone provision to the BRI terminal
- ◆ Capability, on an individual configuration basis, of sending the last forwarding DN as CLID information, for the following types of calls:
  - incoming EuroISDN calls being forwarded back to the public network by way of EuroISDN
  - incoming MCDN/QSIG calls forwarded to the public network by way of EuroISDN
  - internal calls (stand-alone case) being forwarded to the public network by way of EuroISDN
- ◆ Trunk optimization before call establishment (TRO) is enhanced for incoming EuroISDN calls
- ◆ The Type Of Number (TON) and Numbering Plan Identification (NPI) are included in CDR tickets, when the configuration record parameter CLID is set to YES and when the configuration record parameter FCDR is set to NEW. This allows incoming calls to be billed to different accounts according to the originator of the call. The TON and NPI are required, since the CLID provides a sequence of digits that could be the same for a local, national or international number.
- ◆ The EuroISDN interface also provides interworking with other ISDN or non-ISDN interfaces, such as existing ISDN Central Office connectivities (AXE10, 1TR6, SYS12, Numeris and Swiss Net 2), MCDN (1.5 and 2.0 Mbps), QSIG, DPNSS1, DASS2, R2MFC, MFE, DTI, DTI2 and analog trunk interfaces.
- ◆ Layer 1 and layer 2 compliance with ETSI requirements are also supported

EuroISDN

The following points describe other ways in which the EuroISDN interface has been implemented on the Meridian 1:

- If more than one Channel Identification IE is received in a SETUP message, only the first one is used by the Meridian 1. If it is not available, the call is processed using the channel negotiation configuration. The call is released if no negotiation is allowed.
- The transfer of a unanswered EuroISDN call to a remote ringing telephone requires disconnect supervision on the TIE trunk. If the disconnect supervision is not available, after the external user hangs up, the trunk can be locked out. It is the system maintainer’s responsibility to ensure that the trunks used for this type of call provide disconnect supervision.

Operating parameters pertaining to ETS 300 102

- The user-to-user compatibility checking, by the means of the Low Layer Compatibility IE and/or the High Layer Compatibility IE, is not supported. The LLC IE and the HLC IE are tandemed by the Meridian 1, but this information is not used to perform any checking on a Meridian 1 node.
- Transit network selection is not supported. This IE is normally used by the user to identify a selected transit network in the SET-UP message. As no European country specifies the coding to use this IE, this service is not supported by the Meridian 1. As a result, this IE is never sent by the Meridian1.
- Extension for symmetric call operation is not supported. This is normally used to implement a private network application.
- Network Specific Facility selection procedures are not supported.
- D-Channel backup procedures are not supported by EuroISDN.
- Message segmentation procedures are not supported. These are normally used to split messages that are too long.
- Low Layer Information coding principle is not checked by the Meridian 1. No LLC is generated by the Meridian 1, but this information is tandemed, if received (from an ISDN BRI telephone, for example).

EuroISDN

- Low Layer Compatibility negotiation procedures are not supported.
- The USER INFORMATION message is not implemented in the Meridian 1 software.

Operating parameters pertaining to ETSI GF

- ETSI GF gateways to and from other signalling systems, such as DPNSS1, QSIG, and MCDN, are not supported.
- The following ETSI GF procedures are not implemented:
  - control of supplementary services using the separate message approach (HOLD/RETRIEVE)
  - bearer-related broadcast transport mechanism (multipoint configuration)
  - bearer-independent point-to-point connectionless transport mechanism
  - bearer-independent broadcast transport mechanism
  - generic notification procedures
  - network-side channel reservation function
  - generic procedures for supplementary service management
  - generic status request procedure
  - support of the Extended Facility Information Element
- DN address translation requires the association with a customer number. For an ETSI basic call establishment, the customer number association is found through the B-Channel identified in the channel ID IE. For DN address translation that is not associated with a basic call, the customer number association must be determined through other methods.

EuroISDN

For a BRI trunk DSL interface, there is a customer number association with the D-Channel. For a PRI interface, there is a prompt in the programming for an ETSI D-Channel configuration to create a customer number association with a given D-Channel. This implies that bearer independent messages on a primary rate D-Channel are associated with a single customer.

For example, in a multi-customer configuration, if every customer on the switch wishes to use the bearer-independent transport service over ETSI PRI interfaces, then each customer requires a separate D-Channel.

- The Facility Information Element (FIE) is a repeatable IE, and its length is application dependent. However, due to system capacity considerations, such as call register usage and real time usage, there are two types of limitations enforced by the ETSI GF transport platform:
  - the Meridian 1 ETSI GF platform supports up to a maximum of eight ROSE components in one message. The eight components may be included in one Facility Information Element (FIE), or multiple FIEs. In addition, the inclusion of components in a message is also limited by the “available message length”. The available message length is the difference between the maximum message length (260 octets), and the maximum message length taken up by other mandatory and optional IEs supported in the given message.
  - when a supplementary service requests the ISDN ETSI GF transport to send a component which exceeds the available message length or the number of components supported, the supplementary service is notified.

EuroISDN

Operating parameters pertaining to ETS 300 403

- The operating parameters pertaining to ETS 300 102 are applicable to ETS 300 403, except for the user-to-user compatibility checking, which is partially supported in some cases, as in the case of the EuroISDN 7 kHz/Videotelephony teleservices.
- Codeset 4, which is reserved for use by the ISO/IEC standards, has been added to the ETS 300 403 standard. This codeset is currently not used by the Meridian 1. Therefore, it is not supported.
- The fallback capability for multirate services is not supported.

EuroISDN 7kHz and Videotelephony Teleservices

This X11 Release 24 feature allows the Meridian 1 to support the establishment of calls using the telephony 7kHz teleservice or the videotelephony teleservice on EuroISDN links and with ETSI BRI telephones.

The 7 kHz teleservice offers better voice quality. The Telephony 7kHz teleservice is a real-time teleservice in which voice, using one circuit-mode 64 kbps connection with 7 kHz bandwidth, conforms to CCITT recommendation G.722 “7 kHz audio-coding within 64 kbps”. There is a Telephony 3.1 kHz teleservice which is a base service for voice call, defined by “3.1 kHz audio” or “speech” as the Information Transfer Capability.

The Videotelephony teleservice is a real-time audiovisual teleservice in which voice and moving pictures are interchanged by means of one or two 64 kbps circuit mode ISDN connections.

This feature allows the Meridian 1 to support these teleservices on PRI2 and BRI trunk EuroISDN interfaces and on ETSI BRI telephones. The establishment of calls and the fallback procedures are supported.

Fallback

Fallback allows a request for the telephony 7 kHz teleservice or for the videotelephony teleservice, to use the alternate teleservice. In the case of the 7 kHz teleservice, the alternative is the 3.1 kHz teleservice. In the case of the Videotelephony teleservice, the alternative is the 7 kHz teleservice (if supported), or the 3.1 kHz teleservice.

- If fallback is not allowed, the call is either established with the requested teleservice, or rejected.

  If the call request reaches a non-EuroISDN link, or if the requested telephone is a non-ETSI BRI telephone, the call is rejected. For analog (500/2500) telephones and Meridian proprietary telephones, the call is rejected. The call is also rejected for calls over a EuroISDN link and for ETSI BRI telephones that are not configured to support the 7 kHz/Videotelephony teleservices.

- If fallback is allowed, the call is established with the preferred teleservice or with the alternate teleservice.

  If the call request reaches a non-EuroISDN link, or if the destination telephone does not support the requested teleservice, the call is established with the alternate teleservice (3.1 kHz teleservice). The originator is notified that fallback has occurred.

When a call is established with the videotelephony teleservice, a second connection can be established to provide a better video quality. Only the calling party can initiate a request for a second connection, and only after the first connection is established.

This Release 24 feature supports the following communications:

- videotelephone terminal to videotelephone terminal
- 7 kHz terminal to 7 kHz terminal
- videotelephone terminal to 7 kHz terminal or 3.1 kHz terminal, if fallback is allowed
- 7 kHz terminal to 3.1 kHz terminal, if fallback is allowed

Originating and terminating terminals can be local or remote, but calls using one of these teleservices with at least one remote terminal must be made over a EuroISDN PRI2 or EuroISDN BRI link.

EuroISDN

When a user originates a call requesting a 7 kHz or videotelephony teleservice, the user can choose to allow or deny fallback. That is, they can choose whether or not the call is to be established with the alternative teleservice, if the requested teleservice is not available. The user can make this request either from a local ETSI BRI telephone supporting the teleservice, or from a remote telephone whose request is received over a EuroISDN link. The terminating party can accept the call, if the teleservice is supported end to end, or reject the call, if the requested teleservice is not supported.

> ***Note:*** A videotelephone terminal sometimes does not support the telephony 7 kHz teleservice.

ETSI Australian ISDN

With X11 Release 24, the user side of the ETSI Australia connectivity is provided. The protocol is based on ITU-T Q.931, as modified in Helsinki in 1993, with clarifications based on ETSI specification ETS 300 403-1 (November 1995).

The following services are supported:

- Basic Call Service on PRI2 and BRI
- Circuit-mode bearer capabilities (speech, 3.1 kHz audio, 64 kbps digital and adapted 56 kbps to 64 kbps digital)
- DID, DOD, COT, and TIE call types
- Enbloc sending
- Overlap sending
- Channel negotiation

The following supplementary services are supported:

- Calling Line Identification Presentation (CLIP)
- Calling Line Identification Restriction (CLIR)
- Connected Line Identification Presentation (COLP)
- Connected Line Identification Restriction (COLR)
- Malicious Call Trace (MCT)
- Advice of Charge (AOC), for all calls, not individual calls
- Direct Dial In (DDI)
- Subaddress (SUB)

Note that Backup D-channel and nB+D functionality is not supported.

Tandeming of AOC information across a Meridian 1 network is not supported.

BARS is not supported; NARS is recommended. CDP can be used however, the private plan and the CDP numbering type are not supported by the ETSI Australia functionality. The calling and called number plan and type are converted to unknown type and plan.

Hardware

On large systems, the MSDL card is required for PRI2 access to Central Offices. On the small systems (Option 11 family), the modified DDCH card is required.

On large and small systems, the MISP card is required to support Basic Rate Interfaces to Central Offices.

EuroISDN

Business Networking Express (BNE) features

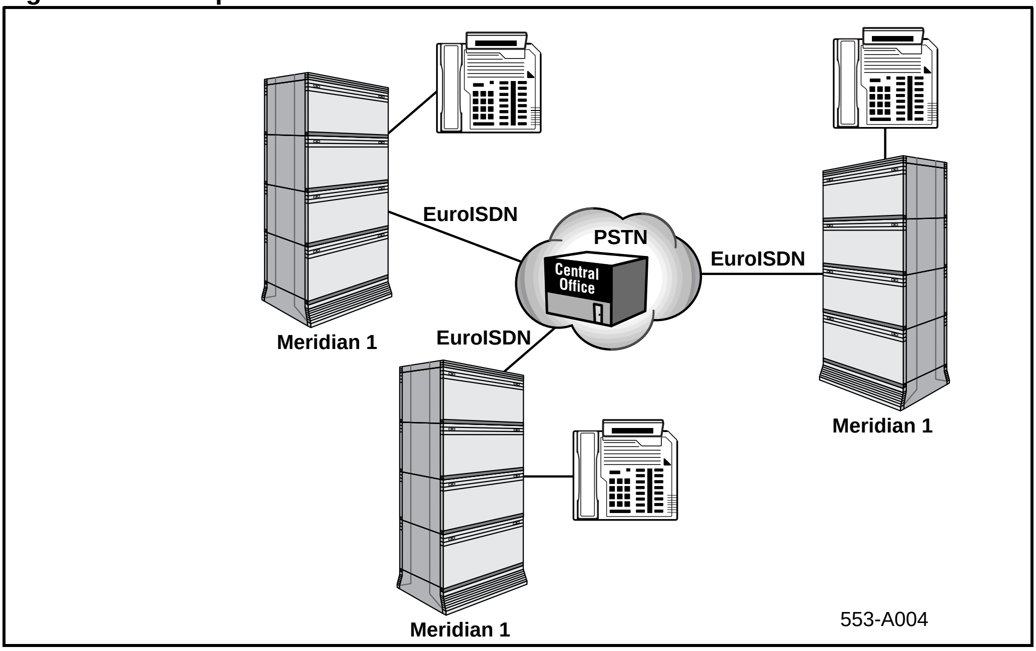
Business Networking Express (BNE) is a term that refers to a group of different EuroISDN network functionalities.

Introduced in Release 25, BNE provides a Virtual Private Network (VPN) solution for Meridian 1 systems through the EuroISDN public network. BNE is appropriate for companies that require a network that operates as if it is a private network, and has an affordable start-up cost. The Virtual Network Services (VNS) solution is similar to BNE, and provides more features than BNE; however, VNS requires a leased line for the D-channel between the Meridian 1 systems.

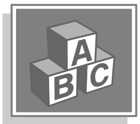
The BNE capabilities provide Meridian 1 systems that are connected on a EuroISDN public network with the following functionalities:

- EuroISDN Name and Private Number Display
- EuroISDN Call Diversion
- EuroISDN Explicit Call Transfer
- EuroISDN Call Completion

Figure 36 - Example of a network where BNE is useful



BNE Name and Private Number Display



With BNE Name and Private Number Display, when a user dials a private network number to reach a user at another Meridian 1 site through the public network, the ESN software causes the dialed number to be outpulsed as a public number. The BNE software inserts the Calling Name and the Private CLID in the SETUP message.

At the destination switch, the Private CLID is displayed, along with the Calling Name, on the alerted telephone. The name associated with the alerted telephone is delivered to the calling user and displayed on the calling telephone.

When the call is answered, the Connected Name and the private Connected Number is provided to the calling user telephone.

Consistent with MCDN and QSIG networking, the letter H is displayed in front of the private number.

You can implement restrictions on displaying the name and number of the calling, called, or connected party.

The information that follows deals in more detail with the Name and Private Number Display parts of the BNE package.

Name Display on EuroISDN

This functionality is based on the existing MCDN and QSIG Call Party Name Display (CPND) and Display of Calling Party Denied (DPD) features. The following three services are supported.

1. Calling Name Identification Presentation (CNIP)

CNIP is a supplementary service which provides the called user with the calling user’s name. This service is permanent and based on the Class of Service of the telephone originating the call.

CNIP does not deliver the calling user’s name to the called user if:

- the Calling Party Name is not available. This occurs when a name is not configured in the CPND data block for the calling party DN, or in the case of interworking.
- presentation is restricted for the telephone originating the call, as controlled by the CNIR service.

2. Connected Name Identification Presentation (CONP)

This is a service offered to the calling user. CONP provides the calling user with the alerted/connected user’s name. CONP service also delivers:

- the name of the alerted user to the calling user whenever the called user’s telephone starts ringing
- the name associated with the telephone that answers the call

CONP does not deliver the called user’s name to the calling user if:

- the Called Party Name is not available. This occurs when a name is not configured in the CPND data block for the called party DN, or in the case of interworking.
- presentation is restricted for the terminating telephone as controlled by the CNIR service.

3. Calling/Connected Name Identification Restriction (CNIR)

This service prevents the user’s name from being presented to another user. This service is activated in two ways:

- **for all calls.** It is based on the Display of Calling Party Denied feature. The Calling/Connected/Called/Alerting Name is denied or allowed using the Class of Service.
- **for each call** (Class of Service NAMA and the user dials the Calling Party Privacy Flexible Feature Code when initiating a call). The Calling Number and Name is restricted when the user dials the CPP code. Attendants can dial the CPP code for CNIR.

Display of restricted name

If the Calling Name information is received with a “presentation restricted” setting, then Xs are displayed on the called user’s display, if it is able and authorized to receive the Calling Name information. If the called user’s name information is received in the ALERTING message and its presentation is restricted, then Xs are displayed on the calling user’s display, if it is able and authorized to receive name information. If the connected user’s name information is received in the CONNECT message and its presentation is restricted, then Xs are displayed on the calling user’s display, if it is able and authorized to receive name information.

Private Calling Number on EuroISDN

EuroISDN public networks can support the same private Calling Number capabilities as QSIG and MCDN networks, with the BNE/Name and Private Number Display feature implemented on the Meridian 1 systems.

This functionality delivers a Calling Party Number in a private format (based on a Coordinated Dialing Plan or Uniform Dialing Plan numbering plan) in addition to the public-format Calling Party Number.

The private format depends on the numbering plan the caller used to dial the call.

The private calling number is constructed based on the CLID Enhancement feature. It contains the following information:

- numbering plan field (private)
- type of number field (CDP or LOC or unknown)
- the DN digits of the calling telephone prefixed by an LSC (CDP) or HLOC (UDP), if configured
- presentation flag to allow or deny the display on the called user's telephone

The following two services are supported:

1. Calling Line Identification Presentation (CLIP)

CLIP provides the called party with the identification of the calling party in a form that allows the called party to return the call, if desired, using the VPN network built on the public EuroISDN connections.

EuroISDN

2. Calling Line Identification Restriction (CLIR)

This service enables the calling party to prevent presentation of the calling number on the called user's telephone. There are two options for implementation:

- **presentation restricted for all calls:** define DDGD in the Class of Service of the telephone. CLIR is not supported for attendant consoles.
- **presentation restricted for individual calls:** the user dials the Calling Party Privacy (CPP) Flexible Feature Code. Define DDGA in the Class of Service of the telephone.

Class of Service CLBA/CLBD (Calling Party Number and Name per-line blocking allowed or denied): On a permanent basis, the Calling Number and Name can be restricted using the CLBA Class of Service (not applicable to BRI telephones). If you program CLBD, the user can dial the CPP code for blocking of name and number for individual calls. Users of BRI telephones cannot dial the CPP code to block name and number; they must use a presentation soft key.

Private Connected Number on EuroISDN

EuroISDN public networks can support the same private Connected Number capabilities as QSIG and MCDN networks, with the BNE/Name and Private Number Display feature equipped on the Meridian 1 systems.

This functionality delivers a Connected Number in a private format (CDP or UDP numbering plan) in addition to the public-format Connected Number. The BNE software is responsible for delivering the private Connected Number to the calling party. The Connected Number is provided by the Central Office in a public format but the private Connected Number is displayed on the calling user's telephone.

The private Connected Number is delivered to the calling user only if a private Calling Number was provided from the calling user. The format depends on the numbering plan of the received private CLID.

EuroISDN

The private Connected Number contains the following information:

- numbering plan field (private) - depends on the NPI of the received CLID
- type of number (TON) field (CDP or LOC or unknown) - depends on the TON of the received CLID
- the DN digits of the connected telephone prefixed by an LSC (CDP) or HLOC (UDP), if configured
- presentation flag to allow or deny the display on the calling user's telephone

Two different services are supported:

1. Connected Line Identification Presentation (COLP)

This service allows the calling party to receive identification of the connected party. The Connected Number replaces the dialed number on the display of the calling telephone. If the called party has presentation restriction, using the COLR supplementary service, the private Connected Number field is empty or presented with the presentation restriction flag on (to users with an override category). The attendant DN is sent when the call is answered by the attendant.

BRI telephones and attendant consoles can have an override key.

2. Connected Line Identification Restriction (COLR)

This service enables the connected party to prevent presentation of its number on the calling user's telephone. There are two options for implementation:

- presentation allowed: the calling user is presented with the Connected Number.
- presentation restricted: the Connected Number is always provided to the network. If the calling user has an "override" category, the network passes this Connected Number to it. If not, the Connected Number is not available to the calling user. The same rules are used for the public Connected Number, if no private Connected Number is received.

EuroISDN

Operating parameters for BNE Name and Private Number Display

For BNE functionality to work, the public network must support User-to-User service 1 implicit procedures. The Meridian 1 node at the terminating end must support the BNE/Name and Private Number Display feature. Configure PSTN routes in Route List Indexes to these destinations with the BNE feature activated (BNE = YES). For calls to Meridian 1 nodes that do not support the feature, use the default setting (BNE = NO) on PSTN routes in the Route List Indexes.

CDP or UDP numbering plans must be used. Trunk route access codes are not supported. CDP or BARS or NARS software must be equipped.

The maximum length of names carried by the BNE feature is 27 characters (maximum length allowed by the CPND feature). Other factors that can affect the number of characters displayed are the size of the display on the telephone and the display of the charges. Names are truncated if their length exceeds 18 characters.

Basic Rate Interface (BRI) telephones cannot have names displayed but they can send a name to a called telephone.

If the called telephone is busy, BNE/Name and Private Number Display does not operate.

When the Call Transfer and Conference features are used, the BNE feature does not provide to the caller the name and number associated with the remote telephone.

The BNE/Name and Private Number Display feature does not introduce any new Classes of Service or configurable data related to the telephone. The Classes of Service are used by the BNE feature in the same way they are used on EuroISDN or QSIG networks for Calling Number or Name display information.

EuroISDN

Some BRI telephones cannot handle the presentation flag in the Calling Number. You can remove the digits in the CLID sent to the BRI telephone when the presentation is restricted. For Calling Line Identification Restriction (CLIR), if the BRI telephone provides a presentation indication in the CLID information, the PRES option is not used. In all other cases, the presentation flag is set based on the PRES configuration. If no CLID is provided by the BRI telephone, the default DN of the Terminal Service Profile (TSP) is used. The same rules are used for the public Calling Number, if no private Calling Number is received.

BNE Call Diversion

BNE Call Diversion offers the Call Diversion supplementary services that are compliant with EuroISDN standard EN 300 207-1. They include the following:

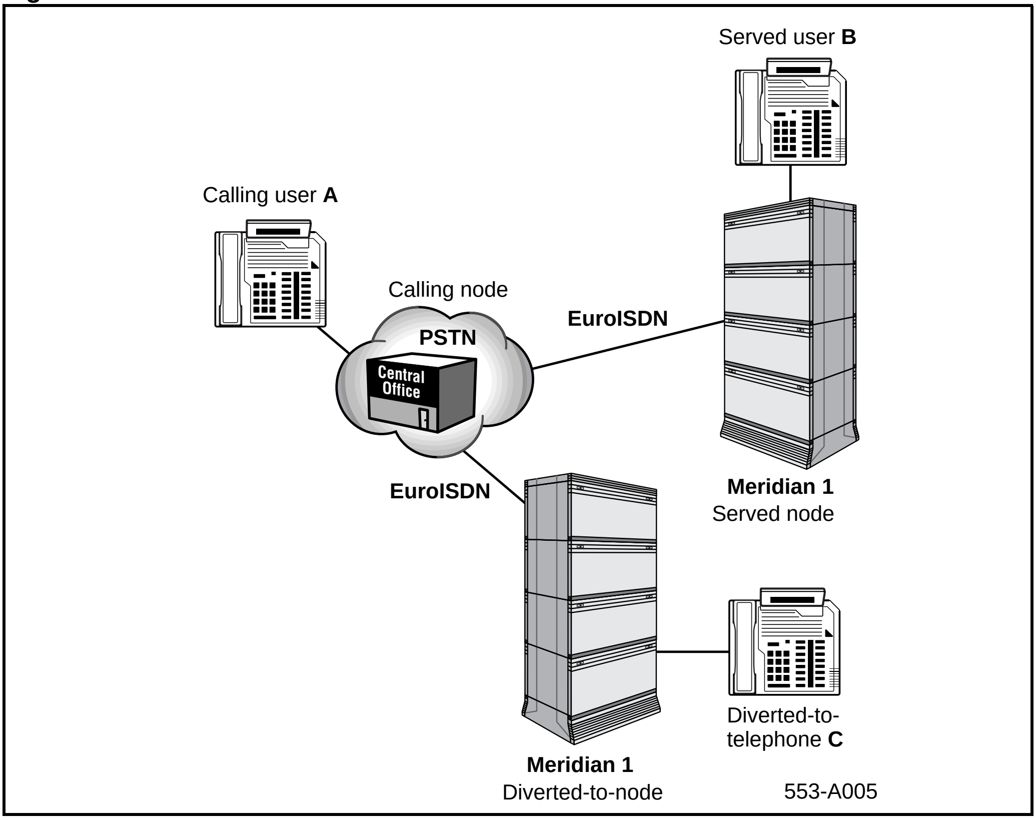
- Call Forwarding Unconditional (CFU) - known as Call Forward All Calls on the Meridian 1
- Call Forwarding Busy (CFB) - known as Hunting on the Meridian 1
- Call Forwarding No Reply (CFNR) - known as Call Forward No Answer on the Meridian 1

Refer to Table 51 on page 474 for a complete list of the equivalent features on the Meridian 1 supported by this feature.

This section uses the terms *served user* or *served telephone*. These terms refer to the telephone that is diverting calls to another telephone in the network. Figure 37 shows the component parts and terms used in the Call Diversion environment.

EuroISDN

Figure 37 - Call Diversion environment



Redirection services

The CFU service enables the network to redirect all calls, addressed to the ISDN number of a served telephone, to another telephone in the network. The CFU supplementary service does not affect the served user's ability to originate calls. The network forwards calls, independent of the status of the served telephone, when the CFU supplementary service is active.

The CFB service enables the network to redirect calls, addressed to the busy ISDN number of a served telephone, to another telephone in the network. The CFB service does not affect the served user's ability to originate calls.

When a call is not answered (for a defined period of time) at an ISDN number of a served telephone, the CFNR service enables the network to redirect calls to another telephone in the network. The CFNR supplementary service does not affect the served user's ability to originate calls.

Rerouting

The public EuroISDN network performs Call Rerouting. Rerouting performs call diversion by replacing the connection from user A's node (located in the public ISDN) to user B's node (located in a private ISDN), with another connection from user A's node to user C's node (located in the public ISDN). The new connection is established in the public ISDN by joining together the original connection from user A's node to the public ISDN gateway node and a second, new connection from the public ISDN gateway node to user C's node.

The Meridian 1 sends a Rerouting request to tell the network that it must reroute the call. This feature controls only the EuroISDN user side (the Meridian 1 side).

The following EuroISDN interfaces are supported:

- ETS 300 102 compliant
- EN 300 403 compliant

EuroISDN

Class of Service

The Class of Service (CLS) of the served telephone affects the notification that the calling telephone and the final destination telephone receive. Tables 48 and 50 shows a summary of the relationships.

Table 48
Relationship between the served user's CLS and the calling user's notification

Served user's Class of Service	Calling user's notification
DN01	No notification
DN02	Notification without diverted-to-number
DN03	Notification with diverted-to-number (default)

The diverted-to-number displays on the calling user's telephone under the following conditions:

- if the information received indicates to allow presentation
- if the served user's CLS (received within the Diversion Notification Information from the served node) allows presentation.

Table 49 gives you examples of the display on the calling user's telephone under different conditions.

Table 49
Examples of calling user's display related to the served user's CLS.

	Calling user's display	
Class of service of served telephone: "Calling user receives notification that call has been diverted"	**after receipt of served telephone's diversion notification information**	**after receipt of diverted-to-telephone's diversion notification information**
No	0164665000	0164665000
Yes without diverted-to-number	0164665000 F	0164665000 F
Yes with diverted-to-number	00164665000 F	0164666000 F

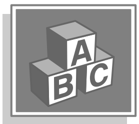
Table 50
Relationship between the served user's Class of Service and the notification on the final destination (diverted-to) telephone

Served user's Class of Service	Diverted-to-telephone receives
DNDN	Served telephone number not shown
DNDY	Served telephone number shown (default)

Diversion reason codes appear on the calling user's telephone and the diverted-to-telephone, if:

- they are programmed at their correct nodes; and
- if the CLS of the served telephone allows it.

EuroISDN

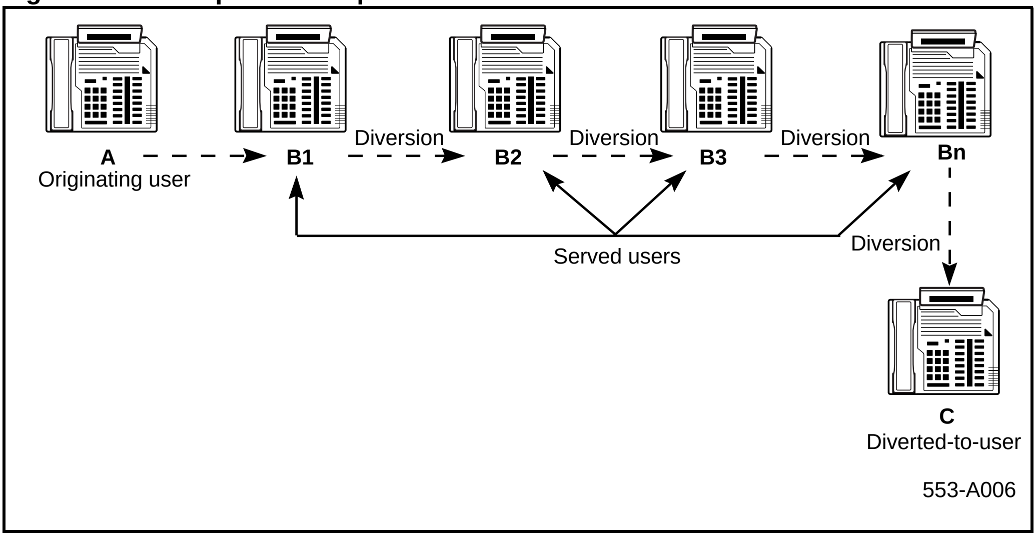


The redirection code displays when the telephone receives the Diversion Notification Information from the served node.

Multiple Diversions

For the purpose of discussion, assume that the following events occur: Originating user A calls B1. B1 has activated CFU, CFB or CFNR to B2. B2 has activated diversion to B3. B3 has activated diversion to the next telephone. Call diversion continues until telephone Bn activates diversion to telephone C. The user at telephone C answers.

Figure 38 - Example of multiple diversions



Identification of the diverted-to-user to the calling user

Diversion Reason Notification rules:

The last diversion reason replaces the previous one.

Diverted-to-Number Notification rules:

When diversion first occurs, and for each diversion following, the Meridian 1 receives the following information from the served node (public ISDN):

- the CLS setting related to if “notify the calling user of diversion”
- the reason for redirection code
- the diverted-to-number

EuroISDN

The Meridian 1 presents the diverted-to-number to the calling user if both the following conditions exist:

- any CLS information related to “notify the calling user of diversion” received contains the value “Yes with diverted-to number”
- any diverted-to-number information received allows the display of the diverted-to-number

The last diverted-to-number replaces the previous one.

Notification at the diverted-to-user

Diversion Reason Notification rules:

The rules are identical to that of a single diversion case.

Served Number Notification rules:

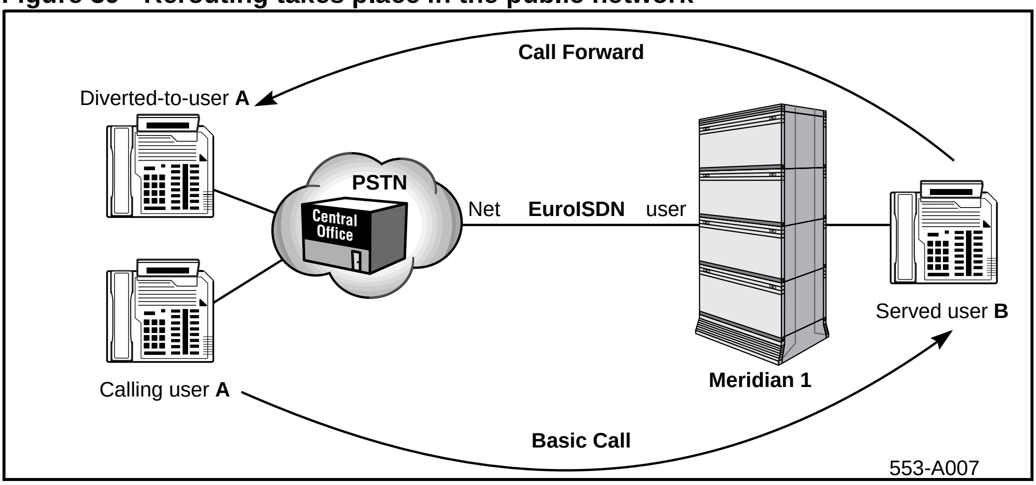
No served user number displays because of the digital telephone display limits.

Procedures for interworking with private ISDNs

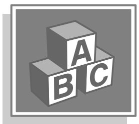
A call from the public ISDN is diverted by rerouting

Figure 39 shows an example of interworking. In the figure, calling user A in the public network makes a call through a EuroISDN link to the served user B on a Meridian 1. The call is forwarded through a EuroISDN link to the diverted-to-user C in the public network.

Figure 39 - Rerouting takes place in the public network



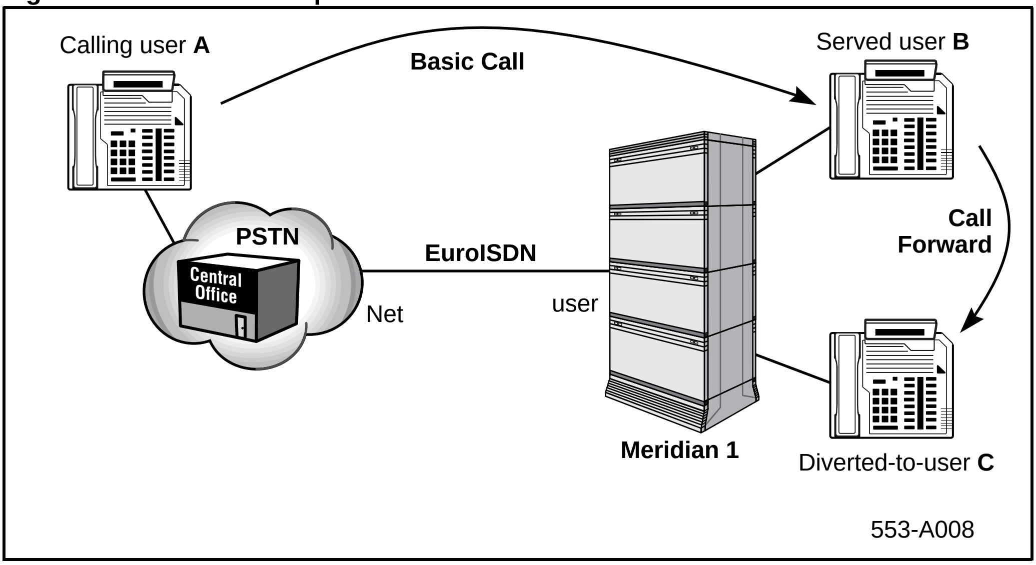
EuroISDN

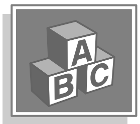


A call from the public ISDN is diverted within the Meridian 1

Figure 40 shows an example of interworking. Calling user A, in the public network, calls through a EuroISDN link to the served user B on a Meridian 1. The call forwards to the diverted-to-user C on the same Meridian 1.

Figure 40 - Call from the public ISDN is diverted within the Meridian 1

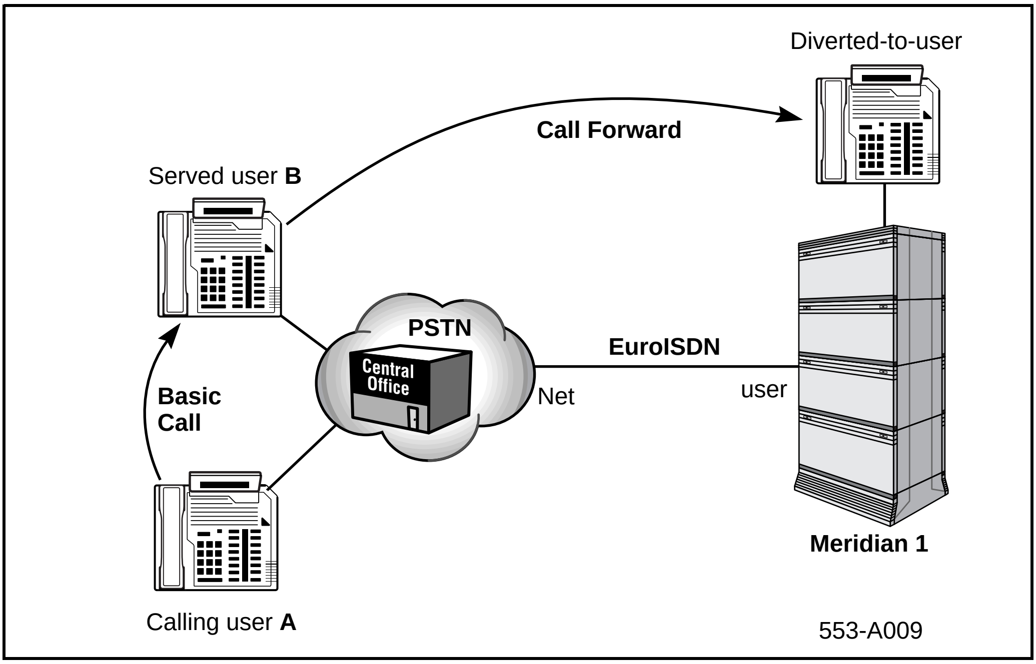




Presentation of a call that is diverted within the public ISDN to the Meridian 1

Figure 41 shows an example of interworking. In this example, the calling user A in the public network makes a call to the served user B who is also in the public network. The call forwards to the diverted-to-user C on a Meridian 1 through the EuroISDN link.

Figure 41 - The calling and served users are in the public network and the diverted-to-user is on the Meridian 1

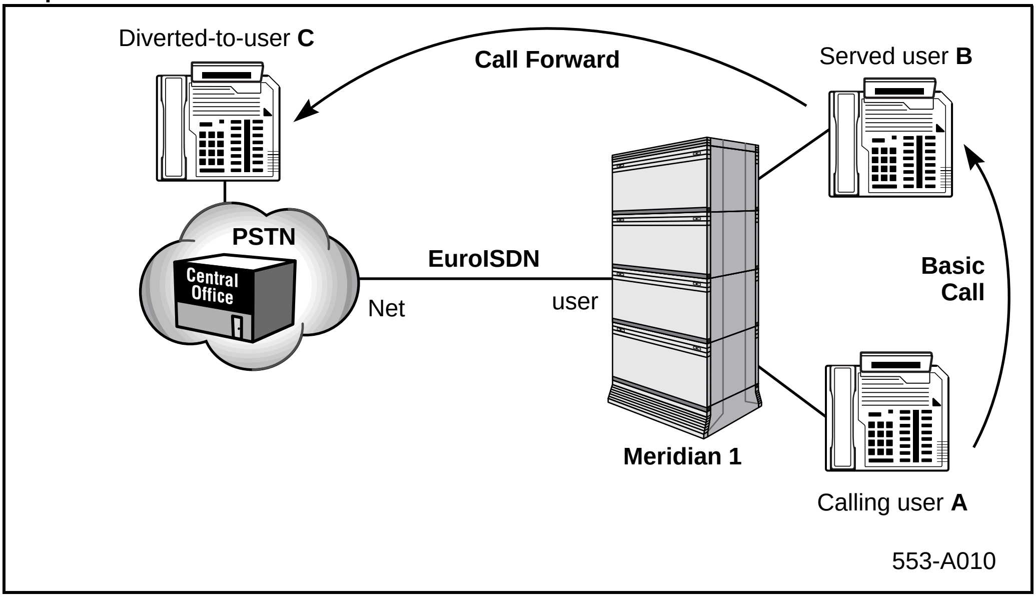


EuroISDN

Presentation of a diverted call from the Meridian 1 to the public ISDN

Figure 42 illustrates an example of interworking. In this illustration, the calling user A on a Meridian 1 makes a call to the served user B on the same Meridian 1. The call is then forwarded through a EuroISDN link to the diverted-to-user C in the public network.

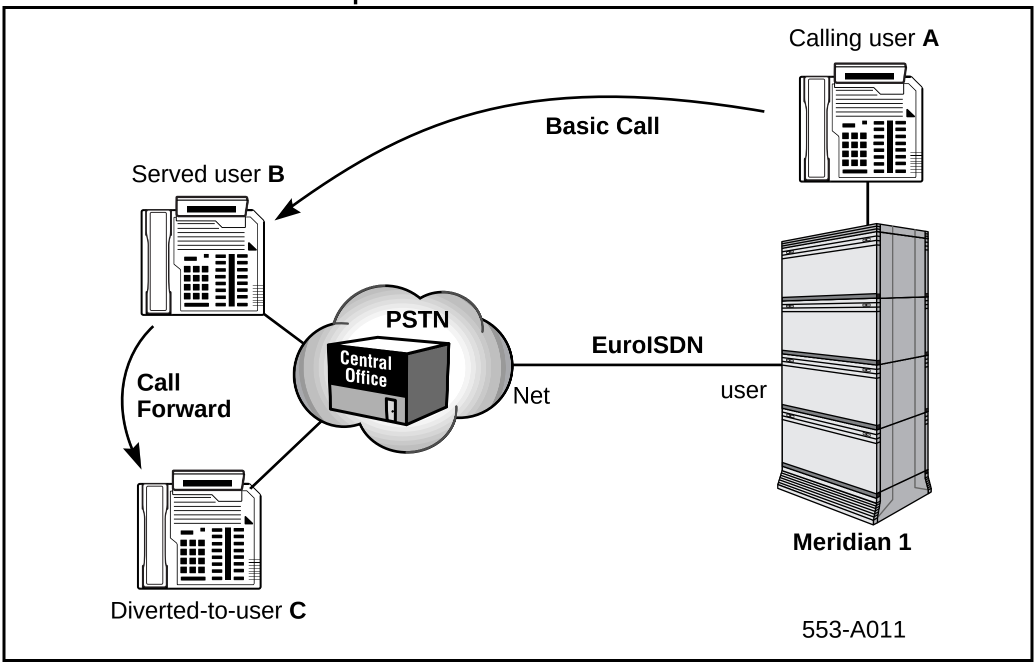
Figure 42 - The calling and served users are on the Meridian 1 and the diverted-to-user is in the public network



A call from the Meridian 1 is diverted within the public ISDN

Figure 43 illustrates an example of interworking. In this illustration, the calling user A, on a Meridian 1, makes a call through a EuroISDN link to the served user B in the public network. The call forwards to the diverted-to-user C who is also in the public network. The calling user displays the diversion reason and the diverted-to-number.

Figure 43 - The calling user is on the Meridian 1 and the served user and the diverted-to-users are in the public network



EuroISDN

Operating parameters for BNE Call Diversion

Table 51 summarizes the correspondence between the ETSI reason for diversion supported by this feature, and the Meridian 1 equivalent features.

Table 51
Correspondence between the ETSI reason for diversion names and the Meridian 1 features

ETSI reason for diversion	Meridian 1 features
Call Forwarding Unconditional (CFU)	Call Forward All Calls
	Internal Call Forward
	BRI ETSI Call Forward Unconditional
	ICP Call Forward
Call Forwarding Busy (CFB)	Hunting
	Hunt by Call Type
	BRI Special Hunt
	Call Forward Busy
Call Forwarding No Reply	Call Forward No Answer
	Flexible Call Forward No Answer
	Second-Level Call Forward No Answer
	BRI Special Call Forward No Answer
	Call Forward No Answer by Call Type
	Attendant Forward No Answer
	Timed Reminder Recall (all types)
	Call Waiting Redirection

EuroISDN

The following services included in EN 300 207-1 are not part of this feature:

- Call Deflection (CD)
- Selective Call Forwarding Busy (SCFB)
- Selective Call Forwarding No Reply (SCFNR)
- Selective Call Forwarding Unconditional (SCFU)

When implementing the BNE/EuroISDN Call Diversion feature, you must be aware of the following:

- The served user does not receive an indication when a call is diverted.
- The calling user is notified each time a redirection occurs, if information is provided by the network. This means that:
  - the last received notification replaces the previous diversion notification
  - if a redirection occurs and no diversion information is provided by the network, the previous notification (if any) remains unchanged
- A user on the Meridian 1 cannot activate, deactivate, or interrogate EuroISDN Diversion on any switch remotely through the EuroISDN network. A user on another switch cannot activate, deactivate or interrogate EuroISDN Diversion on the Meridian 1 remotely through the EuroISDN network.
- Verification of the validity of the diverted-to-number is not supported.
- The EuroISDN Call Diversion supplementary service is not supported for BRI lines. This feature supports EuroISDN Call Diversion over PRI2 and BRI trunks. Any procedure that is specific to the BRI telephone is beyond the scope of this feature.

EuroISDN

- The calling user can receive notification that a call has been diverted. There are three possible values:
  - No
  - Yes without diverted-to-number
  - Yes with diverted-to-number (when available).

  Due to the limitation of the number of digits that a digital telephone can display, only the diverted-to number displays, when available, on the calling telephone. The served user number does not display.

  Table 52 summarizes the effect of these Class of Service settings on the calling user's display. These examples only show the information that is displayed on a telephone which is related to this feature. Each terminal has its own way of presenting the information and this feature does not change that. For the purposes of this discussion, consider only the information type in the examples, not the information location.

  For this example assume:
  - Served user's ISDN number: 0164665000
  - Diverted-to user's ISDN number: 0164666000
  - The reason for redirection code for Call Forward All Calls is "F.

Table 52
Relationship between the calling user's display and the served user's Class of Service

Served user's Class of Service related to the calling user's display	**Calling user's display once diverted-to-telephone rings**
No	0164665000
Yes without diverted-to number	0164665000 F
Yes with diverted-to number when available	0164666000 F

EuroISDN

- The served user can release their number to the diverted-to-user. There are two possible values: No, Yes. Due to the limitation of the number of digits that a digital telephone can display, only the calling number displays on the diverted-to-telephone, independent of the served user's Class of Service.
- Table 53 summarizes the effect of these Class of Service settings on the display of the diverted-to-telephone. These examples only show the information that is displayed on a telephone which is related to this feature. Each terminal has its own way of presenting the information and this feature does not change that. For the purposes of this discussion, consider only the information type in the examples, not the information location.

  For this example assume:

  Calling user's ISDN number is: 0164664000

  The reason for redirection code for Call Forward All Calls is "F".

Table 53
Relationship between the display on the diverted-to-telephone and the served user's Class of Service.

Served user's Class of Service related to the diverted-to-telephone display	Display of the diverted-to-telephone
No	0164664000 F
Yes	0164664000 F

EuroISDN

When both the calling and served telephones are on the same node, there is no change introduced with this feature. In particular, the served telephone’s Class of Service has no impact on the notification to the calling user.

If the Central Office (CO) rejects the call rerouting request, the Meridian 1 does nothing. It remains in the same basic call state it was in before it sent the call rerouting request. The Meridian 1 waits for the CO to disconnect the call.

If a call from the public ISDN is diverted within the Meridian 1 and a reject component is received from the CO, the Meridian 1 accepts this information and continues to establish the call.

If a diverted call is presented from a public ISDN to the Meridian 1 and a reject component is received from the CO, the Meridian 1 accepts this information and continues to establish the call.

BNE Explicit Call Transfer

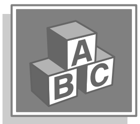
BNE/EuroISDN Explicit Call Transfer extends the functionality of the private network. It notifies the public network that a transfer has occurred within the private network. Also, it optimizes the call by requesting the public network to perform the transfer.

BNE/EuroISDN Explicit Call Transfer example

Telephone A connects to telephone B. Telephone A transfers telephone B to telephone C. Telephone A is on the served node, telephones B and C are on the remote nodes.

At least one call, telephone A to B, or telephone A to C, is over a EuroISDN link. The Meridian 1 supports the functionality Explicit Call Transfer at the served node (the node receiving the original call) and at the remote nodes.

EuroISDN



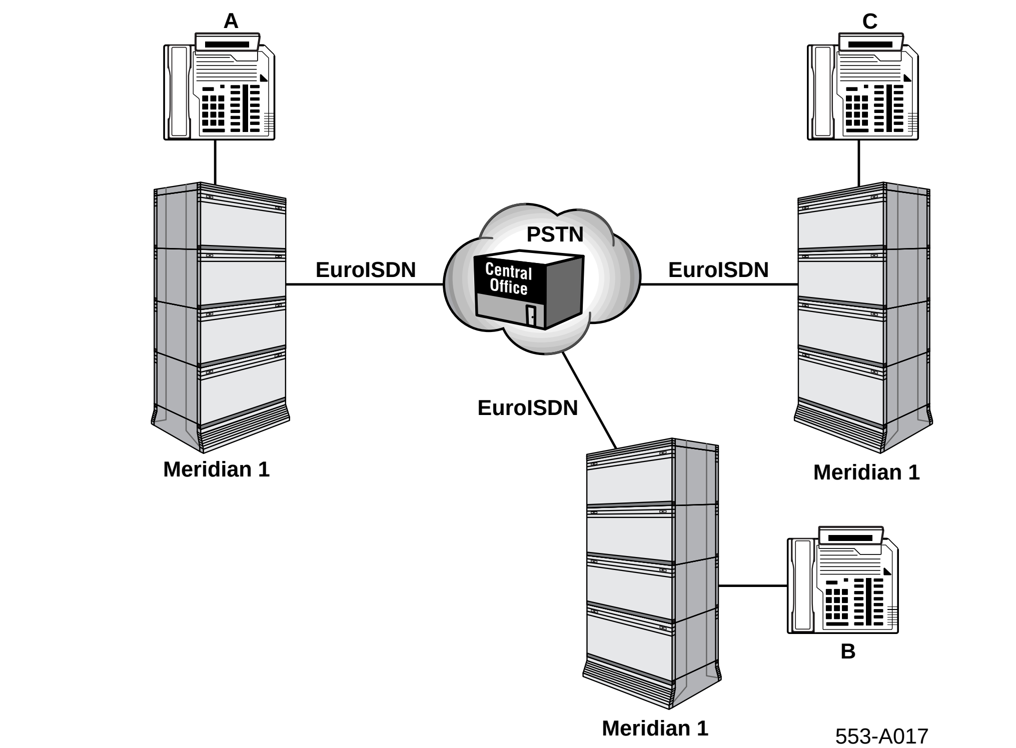
With this feature, the Meridian 1 can, depending on network configuration:

- send transfer notifications on a EuroISDN link
- receive transfer notifications on a EuroISDN link
- activate Call Transfer within the public network on a EuroISDN link

There are three types of network configuration:

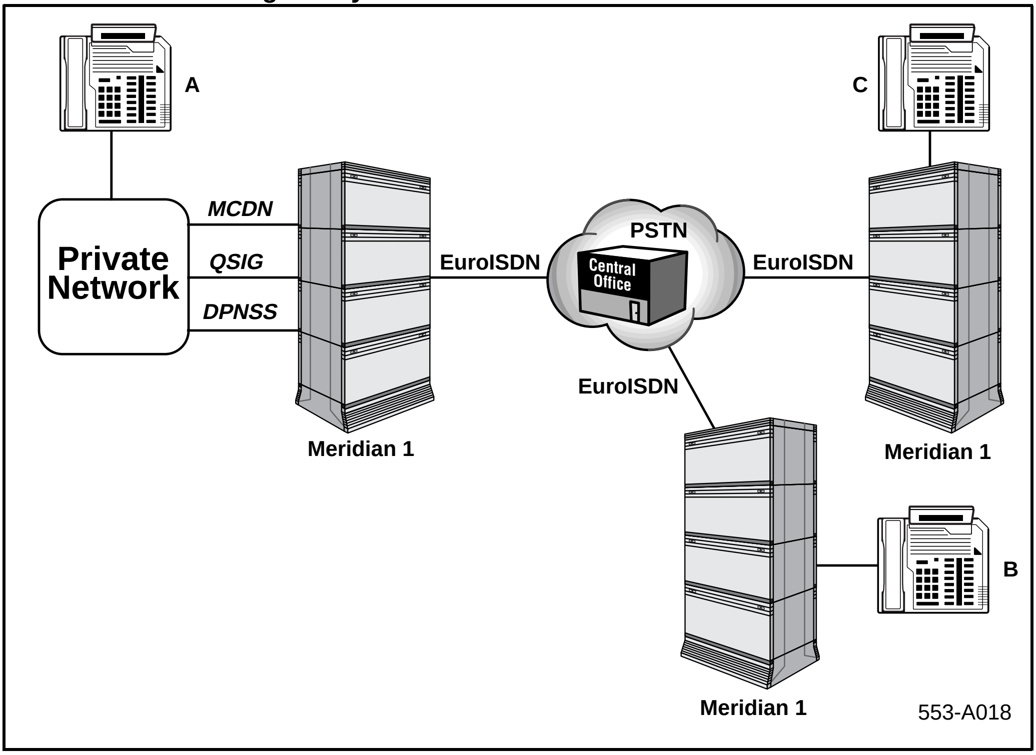
1. Only the Served User (A) is in the Meridian 1 private network (or stand alone Meridian 1). The Meridian 1 sends transfer notifications to, or activates Call Transfer within PSTN. Refer to Figure 44 and Figure 45.

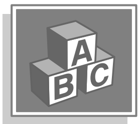
Figure 44 - Served User A is the only one in the Meridian 1 private network (or stand-alone Meridian 1)



EuroISDN

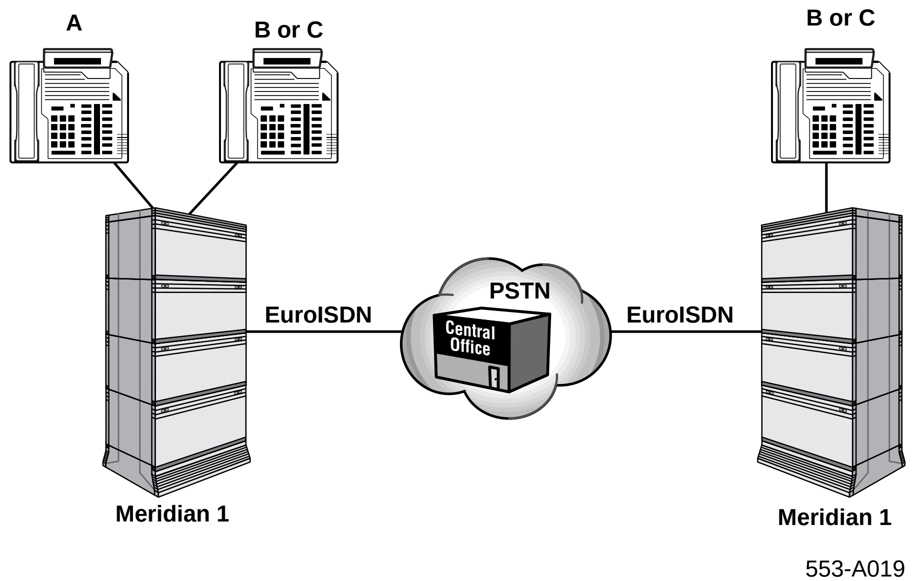
Figure 45 - Served user A is the only one in the Meridian 1 private network. The Meridian 1 acts as a gateway





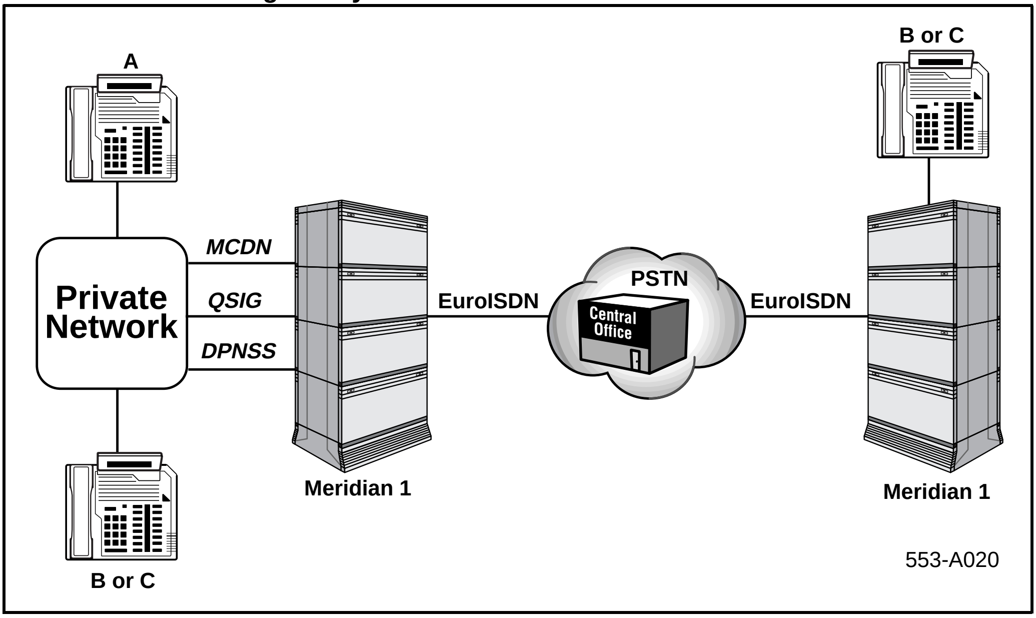
2. The served user (A) and one of the remote users (B or C) are in the Meridian 1 private network. The Meridian 1 sends transfer notifications. Refer to Figures 46 to 48. In a gateway connection, the Meridian 1 can be the gateway node (Figure 47), or the gateway node and the served node together (Figure 48).

Figure 46 - User A and user B or C are in the Meridian 1 private network



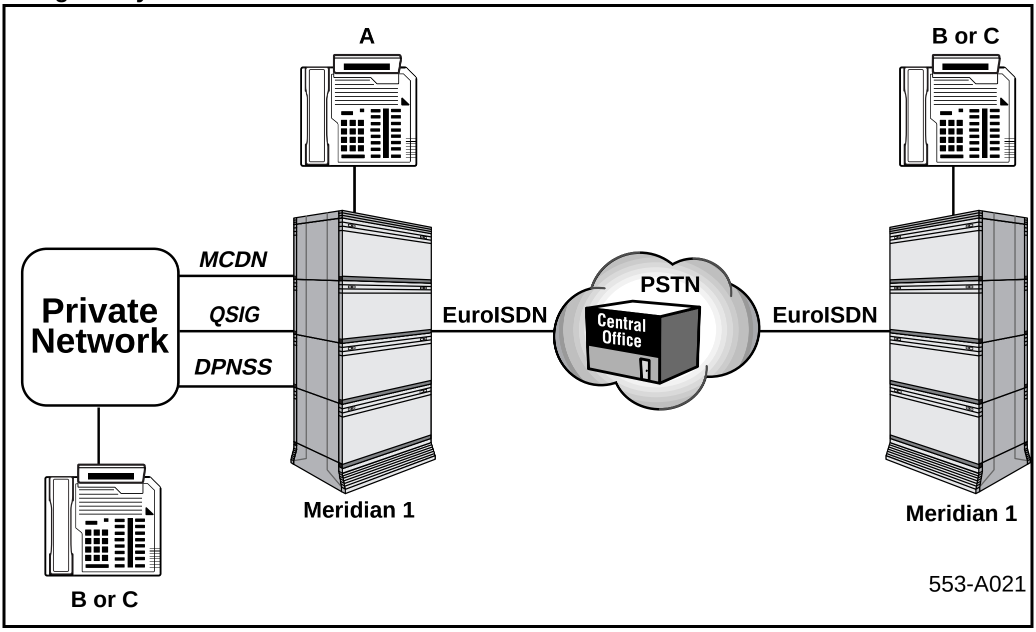
EuroISDN

Figure 47 - User A and user B or C are in the Meridian 1 private network. The Meridian 1 acts as a gateway

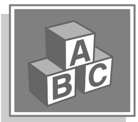


EuroISDN

Figure 48 - User A and user B or C are in the Meridian 1 private network. The Meridian 1 acts as a gateway node and a served node



EuroISDN



3. Only remote user(s) B or C, or B and C, are in the Meridian 1 private network. The Meridian 1 receives transfer notifications. Refer to Figures 49 and 50.

Figure 49 - Remote user B or C is in a Meridian 1 private network

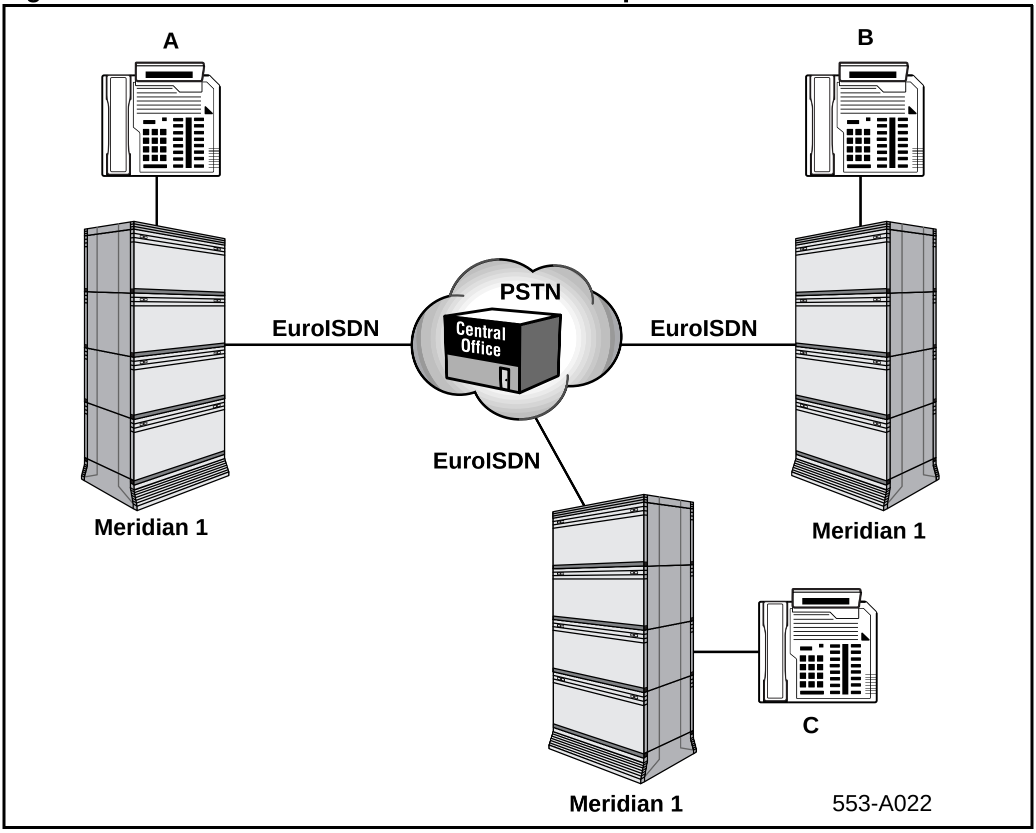
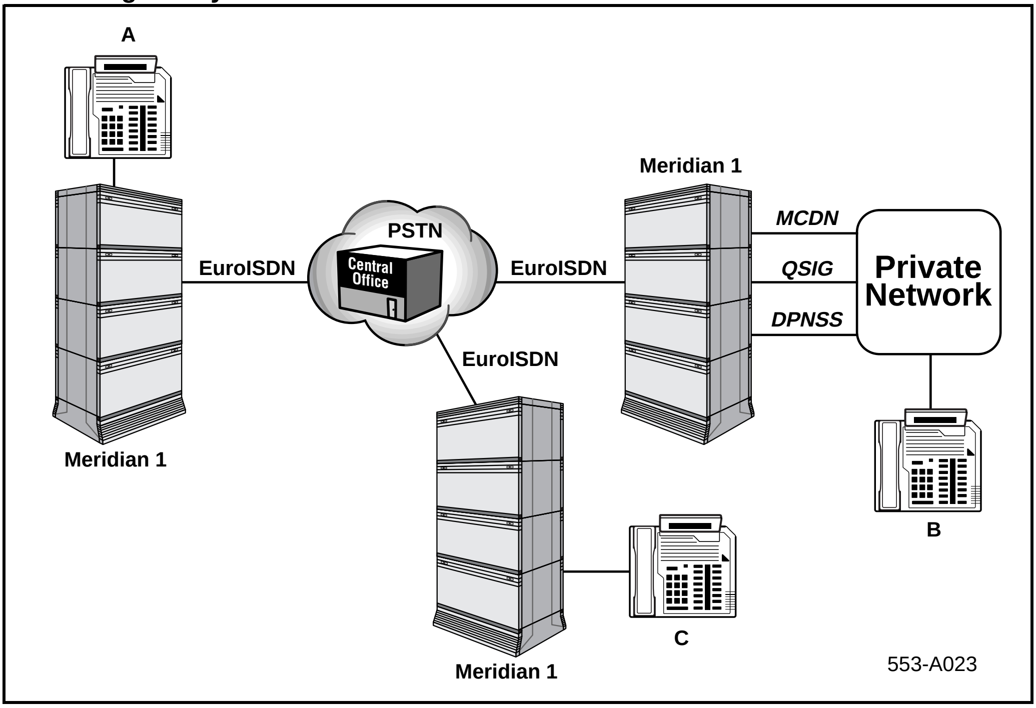


Figure 50 - Remote user B or C is in the Meridian 1 private network. The Meridian 1 acts as a gateway



EuroISDN

Call Transfer through the PSTN

When all of the following conditions are met, Call Transfer through the PSTN is possible:

- Only the served user, A, is in the Meridian 1 private network (refer to Figure 44).
- Both D-Channels required in the transfer have the Remote Capability for Call Transfer notification and invocation (ECTO) configured.

  Note: In most situations, D-Channels 1 and 2 are the same.

- Both calls are in the established call state.

The Meridian 1, working as the served node, invokes the Call Transfer in the public network. This can optimize trunk usage by suppressing tromboning between the Meridian 1 and the PSTN. Trunk optimization occurs only when both calls involved in the transfer are in the established call state.

With a supervised transfer, the transfer (by join) is first completed on the Meridian 1, and notifications are sent to the PSTN. If the conditions are met, the Meridian 1 invokes Explicit Call Transfer to the PSTN.

With an Unsupervised transfer, the transfer (by join) is first completed on the Meridian 1, and notifications are sent to the PSTN. The Meridian 1 waits until both calls are in an established call state. When both calls are established, and all the conditions are met, the Meridian 1 invokes Explicit Call Transfer in the PSTN.

Call Transfer Notification Display

If the network provides the information, the originating caller is notified, on the display of the telephone, when a transfer occurs. This means that:

- if a previous Call Transfer notification was provided, it is replaced by the last received notification.
- if a transfer occurs, with no Call Transfer information provided by the network, and a previous notification was provided, the notification remains unchanged.

The following scenario is considered to be a standard Call Transfer situation (refer to Figures 51, 52, 53, and 54):

1. User B calls user A
2. User A answers the call
3. User A presses the transfer key, and calls user C.
4. User A presses the Call Transfer key again to complete the transfer (the transfer can be completed when the secondary call is alerting or established).

Figure 51 - User B calls user A. User A transfers call to user C

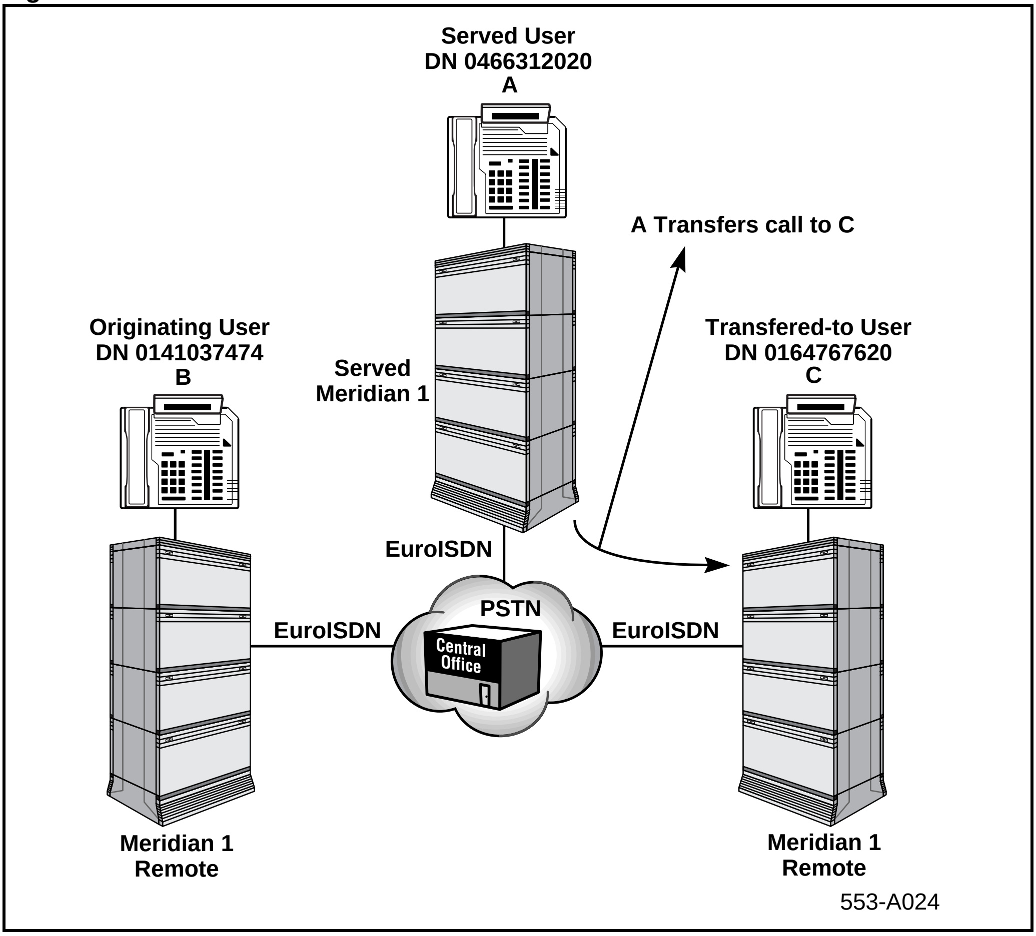


Figure 52 - Display of established call between user A and user B.

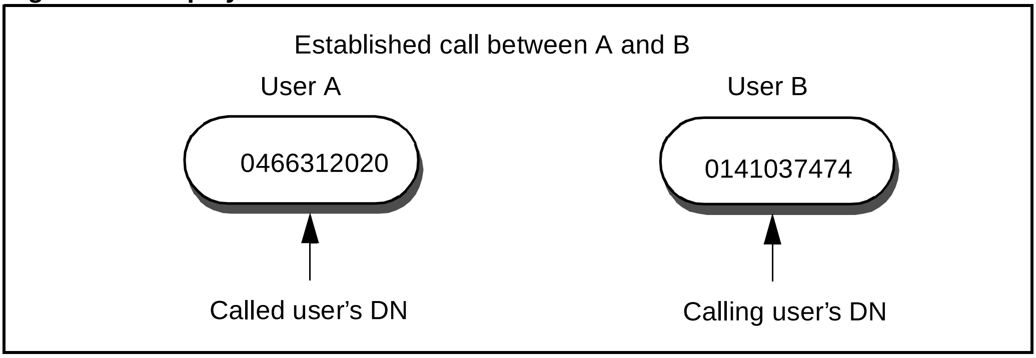


Figure 53 - User A presses Call Transfer key and calls user C.

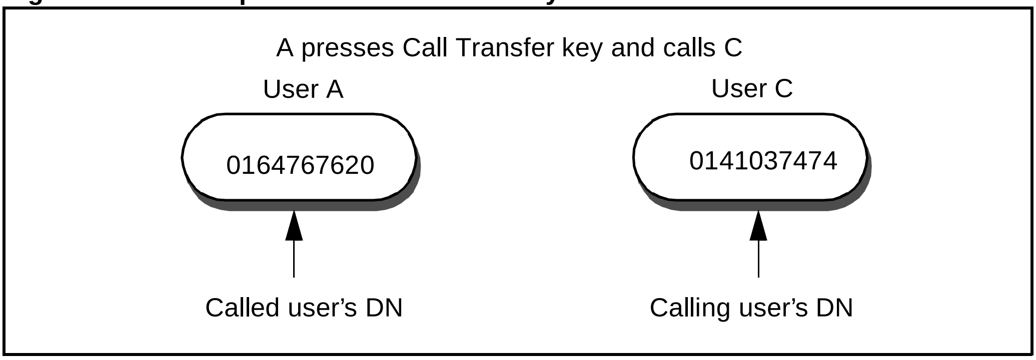
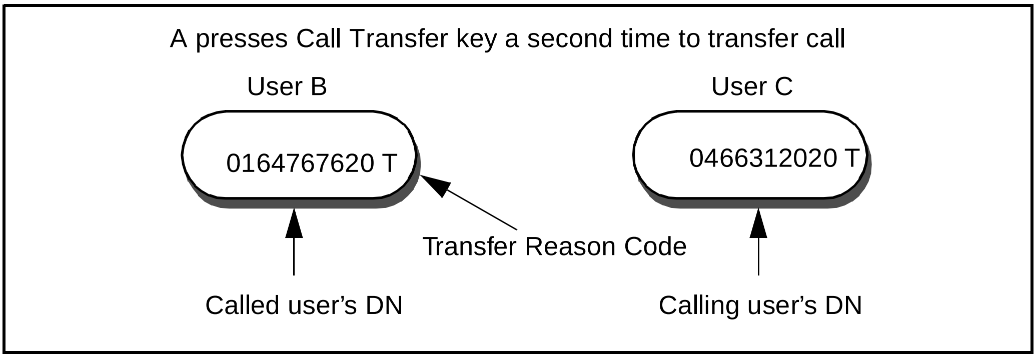


Figure 54 - User A presses Call Transfer key a second time to transfer the call to user C.



Call Transfer Notifications Display Rules

The Call Transfer Notifications Display Rules are similar to Transfer Notifications rules on a Meridian 1 private link, and depend on the:

- Class of Service configured on the telephones
- Presentation Indicator of the Redirection Number received

Class of service definitions:

- DDGA – DN Display on other telephone Allowed
- DDGD – DN Display on other telephone Denied
- CNDA – Calling Name Display Allowed
- CNDD – Calling Name Display Denied.

The reason is displayed on a telephone only when the CNDA Class of Service is configured.

Transferred and Transferred-to users' notification rules

Transfer Reason Notification rules:

No redirection reason is displayed on a telephone if the CNDD Class of Service is configured.

Redirection Number Notification rules:

The redirection number displays on the user's telephone if the received presentation information indicates that presentation is allowed. Otherwise, the telephone displays the trunk route access code and trunk route member number (instead of the redirection number).

If a remote user has the DDGD Class of Service defined, the telephone sends its number in a redirection number with the Presentation Indicator set to Presentation Restricted.

If a remote user has the DDGA Class of Service defined, the telephone sends its number in a redirection number with Presentation Indicator set to Presentation Allowed.

Table 54 on page 490 identifies the originating user's notification according to the originating and transferred-to users' configuration options.

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Table 54
Originating and Transferred-to users' notification in a Meridian 1 environment.

CLS *Originating user* telephone B DN 0141037474	**CLS** *Transferred-to user* telephone C DN 0164767620	Originating user's display after receipt of Transferred-to user's transfer notification information	Transferred-to user's display after receipt of originating user's transfer notification information
CNDA DDGA	CNDA DDGA	0164767620 T	0141037474 T
CNDD DDGA	CNDD DDGA	0164767620	0141037474
CNDA DDGD	CNDA DDGA	0164767620 T	211-4 T
CNDD DDGD	CNDD DDGA	0164767620	211-4
CNDA DDGD	CNDA DDGD	312-6 T	211-4 T
CNDD DDGD	CNDD DDGD	312-6	211-4

Feature interactions

The following information describes the interactions that the various capabilities of EuroISDN have with other features and capabilities. You should investigate these to find out if any of them apply to the solution you want to implement.

Advice of Charge for EuroISDN

Feature interactions

Redirections

Calls charged with Advice of Charge that are either transferred, extended or redirected to another telephone (with Call Forward All Calls, Call Forward No Answer, Hunting or Call Pickup) are charged against the last telephone that answers the call and the controlling telephone releases. Additionally, the last party that transfers or forwards a call to an ISDN Central Office trunk is charged for both connections.

Call Park

When a telephone parks a call charged with Advice of Charge, the calling party continues to be charged until the call is answered by a telephone.

Call Transfer

When a telephone is connected to an ISDN CO trunk conveying AOC charging information, the received call charging information is stored against this telephone.

If the user transfers the call while the dialed telephone is still ringing, call charging information is stored against the transferring telephone until the call is either answered or abandoned by the external party. If the user consults with the dialed transfer telephone, charging information is stored against the transferring telephone until the call is either answered or abandoned.

EuroISDN

If the transferred call is redirected by a call redirection feature, the call is charged against the transferring telephone until the call transfer is completed and the call is answered.

In all instances, if the call is answered, new call charging information is stored against the telephone receiving the transferred call.

Call Waiting

When an Advice of Charge call is transferred to a busy telephone with Call Waiting Allowed, the transferring station is charged until the call is answered.

Call Detail Recording

For Advice of Charge at start of the call (AOC-S) and during the call (AOC-D), the Periodic Pulse Metering (PPM) count and call charge fields in the Call Detail Record (CDR) are updated when a CDR S, X and E record is output on the TTY.

For Advice of Charge at end of the call (AOC-E), the PPM count and call charge fields are only updated when a CDR W record is output on the TTY.

Conference

If a telephone is participating in a conference, no charge is displayed for that telephone. Whenever an ISDN CO trunk that provides AOC is added to a conference, the call charging information, received from the network, accumulates against the telephone that initiates the call.

Once the last telephone involved in a conference call disconnects, a search is made of all trunks remaining in the conference call to determine which trunk has been established in the call for the longest period of time. This trunk becomes the chargeable TN. Once this trunk disconnects, the process is repeated so a new chargeable TN can be located.

Periodic Pulse Metering

Advice of Charge has the following interactions with Periodic Pulse Metering (PPM): recording of accumulated call charging information for each call on the CDR record, calculating the total charge for each call based on the assigned unit cost and the accumulated information received from the network, allowing the attendant to read the number of call charge units on a per call basis and allowing a telephone with a MRK key to access Message Registration information.

Virtual Network Services

For a Virtual Network Service (VNS) simple call connectivity, received charging information is stored against the Message Registration meter of the calling party's telephone.

On an outgoing trunk call on VNS connectivity, charging information is processed based on the Trunk to Trunk Connection feature.

ISDN QSIG/EuroISDN Call Completion

Feature interactions

Access Restrictions
Call Restriction
Trunk Group Access Restrictions

ISDN QSIG/EuroISDN Call Completion does not override Access, Call Restriction or Trunk Group Access restrictions. When Call Completion is activated, the second call has the same restrictions as the initial call that received either no answer or a busy indication.

Advice of Charge for EuroISDN

Advice of Charge applies to the initial and call completion (second call). The initial call receives the same charging information as a normal busy call. The call completion receives the same charging information as the first call.

Attendant Consoles

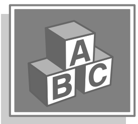
Call Completion requests cannot be directed to or from an attendant console.

Automatic Call Distribution

Call Completion requests cannot be directed to or from Automatic Call Distribution (ACD) Directory Numbers (DNs).

An ACD telephone that uses a normal DN key (not the ACD key) can activate the Ring Again key when encountering a busy or no answer situation.

EuroISDN



Call Detail Recording

A Call Detail Recording (CDR) is not produced for Call Completion signaling. The second call receives a CDR record as for a normal call.

Call Forward All Calls

When the Call Forward All Calls feature is activated on a local basis and an incoming Call Completion request is received, the Call Completion request is registered against the forwarded DN.

Call Transfer

Call Completion notification is only presented to the Call Completion originating telephone. This notification cannot be transferred to another station. Once the second call is completed, the call can be transferred.

If a user encounters a busy during a transfer operation, Call Completion can be activated.

Call Waiting

On an Analog (500/2500 type) telephone, Call Completion notification waits until the telephone has finished an active call.

If Call Waiting is configured on a telephone, notification is presented after the Call Waiting call. If an additional call is queued while Ring Again free notification is waiting on a telephone, the waiting call takes precedence over the Call Completion notification. An established Call Completion call is also queued if the original calling telephone has the Call Waiting feature equipped and is occupied on another call.

Conference

A Call Completion request cannot be made on a conference call attempt.

Direct Inward System Access

Call Completion on Busy Subscriber (CCBS) is not supported on Direct Inward System Access (DISA) calls when the call destination is busy.

Do Not Disturb

An incoming notification overrides a telephone with Do Not Disturb (DND) activated. Call Completion requests can be applied to telephones with the DND feature activated. However, this request does not advance until the DND feature is deactivated.

Euro ISDN Trunk - Network Side

Call Completion to Busy Subscriber is supported on a EuroISDN Trunk - Network Side connectivity interface.

Flexible Feature Codes

Analog (500/2500 type) telephone can use Flexible Feature Codes (FFCs) to activate Call Completion to Busy Subscriber requests.

Group Call

Call Completion cannot be applied to a Group Call.

Group Hunt

Call Completion to Busy Subscriber cannot be applied to a Pilot DN when no idle telephone is located during a Group Hunt call.

Hot Line

Call Completion cannot be used in conjunction with the Hot Line feature.

Maintenance Busy

Call Completion on Busy Subscriber is not accepted against a telephone in a Maintenance Busy state.

Make Set Busy

Telephones that have Make Set Busy (MSB) activated can request Call Completion to another DN, as the free notification overrides the MSB feature. Incoming Call Completion to Busy Subscriber (CCBS) requests do not override the MSB feature. A telephone is considered busy while MSB is active. A CCBS request is registered against a busy telephone, but only advances when the MSB feature is deactivated and the telephone remains free.

Permanent Hold

Analog (500/2500 type) sets with Permanent Hold cannot use the Ring Again functionalities.

EuroISDN

Ring Again
Ring Again No Answer

Analog (500/2500 type) sets can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 proprietary sets can make Ring Again requests based on the number of Ring Again keys programmed on a telephone.

EuroISDN Continuation

Feature interactions

Auto terminate Call

This feature is not supported.

Call Back Queuing and Off-hook Queuing

This feature is not supported with Overlap Signaling.

Call Completion Supplementary Service

Call Completion interacts with Call Transfer Over EuroISDN SN3. Call Completion Free Notification can only be presented to the Call Completion originating station and cannot be transferred to any other station. However, once the Call Completion call is established, it behaves as a normal call with respect to Call Transfer.

End-to-End Signaling

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Flexible Hotline

This feature is not supported with Overlap Signaling.

Incoming Digit Conversion Enhancement

The Incoming Digit Conversion Enhancement (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the Meridian 1.

EuroISDN

ISDN Basic Rate Interface Connected Line Presentation/Restriction

The EuroISDN Continuation capability adds National and Local prefixes to the connected number being sent. This is programmed on a route basis and is applicable to connected numbers received from a ISDN BRI terminal and sent over an ISDN trunk.

ISDN Calling Line Identification Enhancements

The EuroISDN Continuation feature allows Home National Numbers and Home Local Numbers to be configured on a route. When an ISDN call is made from a telephone to a EuroISDN interface, the Calling Line Identification (CLID) constructed by EuroISDN, based on the outgoing route, takes precedence over the CLID constructed for the calling station.

KD3 Signaling

Interworking with KD3 signaling is not supported.

Integrated Service Access (ISA)

Integrated Service Access is not supported.

Special Dial Tone after Dialed Numbers

Special dial tones after dialed numbers are not supported for incoming calls.

Transfer of Unanswered Call

The EuroISDN Continuation feature supports the transfer of a call from a telephone dialing an external number, before the external telephone answers, to both local and remote telephones upon ringing.

Virtual Network Services

A EuroISDN link can be used as a B-channel for the Virtual Network Services feature.

R2MFC Calling Number Identification/Call Detail Recording Enhancements

The outgoing Calling Line Identification (CLID) element of the EuroISDN Continuation feature is mutually exclusive with the R2MFC CNI/CDR Enhancements feature.

EuroISDN

If the CLID is to be composed from the EuroISDN Continuation feature, it will not contain the Calling Number Identification (CNI). If the CLID is to be composed from the CNI, no prefixes will be added to the number.

EuroISDN Continuation Phase III

Feature interactions

Intercept Treatment and Partial Dialing

The Intercept feature allows calls with dialing irregularities to be routed to a Recorded Announcement, an attendant, Overflow Tone, or busy tone. You can configure separate treatments for DID and TIE trunks, for CDP/UDP calls or for non-CDP and non-UDP calls.

The Partial Dial Timing feature (PRDL) allows incoming incomplete non-ISDN DID calls to route to the attendant, after a configurable amount of time, if the received digits cannot be translated.

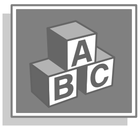
The Vacant Number Routing feature (VNR) allows a call to a vacant number to be routed to another location where the call is either treated as a vacant number, routed or given intercept treatment.

The PRDL feature is enhanced with EuroISDN Continuation Phase III. Once the PRDL feature is activated, instead of routing the call to an attendant automatically, the call is handled by the VNR and Intercept features. When PRDL is applied to EuroISDN trunks, the End-of-Dialing Timer used as the PRDL timer must be at least two seconds shorter than the ISDN interdigit timer for the VNR or Intercept treatments to occur. Otherwise the call is rejected.

Call Transfer

Users and attendants can transfer EuroISDN calls to MCDN or QSIG trunks without waiting for an answer. This is called the Transfer of an Unanswered Call feature. NAS software must be equipped on an MCDN network for this to work. Slow Answer Recall must be configured on QSIG networks. The EuroISDN trunk must be a COT, DID, TIE, FEX, or WATT. If the trunk is not one of these, the transfer can only occur after the destination party answers. If a call is transferred when the telephone is still ringing, a CDR record will only print out after the telephone is answered.

EuroISDN Trunk - Network Side



Feature interactions

Call Completion to Busy Subscriber

This feature is supported on a EuroISDN Trunk - Network Side connectivity interface.

Calling Line Identification Presentation (CLIP)/Calling Line Identification Restriction (CLIR)

The EuroISDN Trunk - Network Side connectivity can generate, tandem or receive a Calling Line Identification (CLID) with presentation allowed (CLIP) or restricted (CLIR.) A CLID that is generated is constructed in the same manner as the EuroISDN user mode connectivity. A CLID that is received is displayed on the called user’s display, if call presentation is allowed.

Even though the EuroISDN Trunk - Network Side connectivity acts as the network side of the Central Office connectivity, it does not provide the network functions (screening and validation) for the Calling Line Identification service.

Connected Line Identification Presentation (COLP)/Connected Line Identification Restriction (COLR)

The EuroISDN Trunk - Network Side connectivity can generate, tandem or receive a Connected Line Identification with presentation allowed (COLP) or restricted (COLR). A Connected Line Identification that is generated or tandemed is constructed in the same manner as the EuroISDN user mode connectivity. A Connected Line Identification that is received is displayed on the called user’s display, if call presentation is allowed.

Even though the EuroISDN Trunk - Network Side connectivity acts as the network side of the Central Office connectivity, it does not provide the network functions (screening and validation) for the Connected Line Identification service.

Calling Party Privacy

If a number presentation for a call is blocked by the Calling Party Privacy feature, the CLID, sent over a EuroISDN Trunk - Network Side connectivity, will have the presentation flagged as restricted.

EuroISDN

End-to-End Signaling

End-to-End Signaling, which allows in-band dialing to be performed on ISDN trunks before and after the call has been answered, is supported on the EuroISDN Trunk - Network Side connectivity.

In the case of tandem with ISDN trunks, the necessary information to allow the End-to-End Signaling feature is tandemed to the ISDN trunk. At this point, it becomes the responsibility of the end user switch to provide the End-to-End Signaling service.

Incoming Digit Conversion (IDC) Enhancement

This feature is supported on the incoming EuroISDN Trunk - Network Side connectivity DID routes. If IDC is equipped, digits received as a called party number are converted, and digit analysis is then performed on the converted digits.

ISDN CLID Enhancements

The EuroISDN Trunk - Network Side connectivity supports all of the user side ISDN CLID enhancements introduced by Generic X11 Release 22 software.

Integrated Services Access (ISA)

ISA is not supported on a EuroISDN Trunk - Network Side connectivity. ISA is currently a feature implemented for the North American ISDN interfaces only.

Network Alternate Route Selection (NARS)/Basic Alternate Route Selection (BARS)/Coordinated Dialing Plan (CDP)

For NARS (and BARS) the Numbering Plan Area (NPA) code and Central Office Code (NXX) cannot be used on the ETSI network side interface, since the codes are not supported by the European public network. Special Numbers (SPNs) get converted to “unknown”, since SPNs are used only in North America. Also, when using Location Codes (LOCs), the networking features do not accept a Digit Manipulation Index (DMI) used to insert an ESN access code. Therefore, the trunks are treated as though they were non-ISDN.

A CDP call can access a trunk on the EuroISDN Network Side. However, since neither the private number nor a CDP number is supported by the ETSI EuroISDN Trunk - Network Side, they get converted to a type that is supported by the public network.

This applies to both the called and calling number plan and type. Also, normal usage of steering codes with Distant Steering Codes (DSCs) and Trunk Steering Codes (TSCs) is supported, as is the use of Digit Manipulation Indexes (DMIs).

Network Attendant Service

Network Attendant Service signaling is not supported on a EuroISDN Trunk - Network Side connectivity. However, NAS will interwork with an incoming call from the EuroISDN Trunk - Network Side (routing and call handling).

Network Call Redirection (Call Forward, Call Forward No Answer, Hunt) and Network Call Modification (Conference, Transfer)

It is possible to have a telephone Call Forward, Call Forward No Answer or Hunt to an external number over a EuroISDN Trunk - Network Side connectivity ISDN PRI or ISDN BRI trunk. It is also possible to transfer or conference a call to an external number over a EuroISDN Trunk - Network Side connectivity ISDN PRI or ISDN BRI trunk. Access restrictions may block some transfers from being completed.

Notices of call redirection or call modification are not transmitted over a EuroISDN Trunk - Network Side connectivity.

Network Call Party Name Display

This feature is not supported on a EuroISDN Trunk - Network Side connectivity.

Name Display

The transport of the name information, is not supported on a EuroISDN Trunk - Network Side connectivity interface.

Special Dial Tone After Dialed Numbers

This feature is supported for outgoing calls.

Trunk Optimization

Trunk Optimization is not supported across a EuroISDN Trunk - Network Side interface. Trunk Optimization Before Answer is not supported within an MCDN network, if the call originated from a EuroISDN Trunk - Network Side connectivity interface.

EuroISDN

Virtual Network Services

A EuroISDN Trunk - Network Side trunk can be used as a VNS bearer trunk.

EuroISDN Malicious Call Identification

Feature interactions

Call Detail Recording

If the caller hangs up and the call is being traced, there is a delay in the printing of the CDR record equal to the length of the delay timer prolonging disconnect that is programmed on the route.

Business Networking Express (BNE) feature interactions

BNE Name and Private Number Display feature interactions

CNIR and CNIP/CONP

The CNIR supplementary service takes precedence over the CNIP supplementary service.

The CNIR supplementary service takes precedence over the CONP supplementary service.

COLR and COLP

The COLR supplementary service takes precedence over the COLP supplementary service.

CLIR and COLR

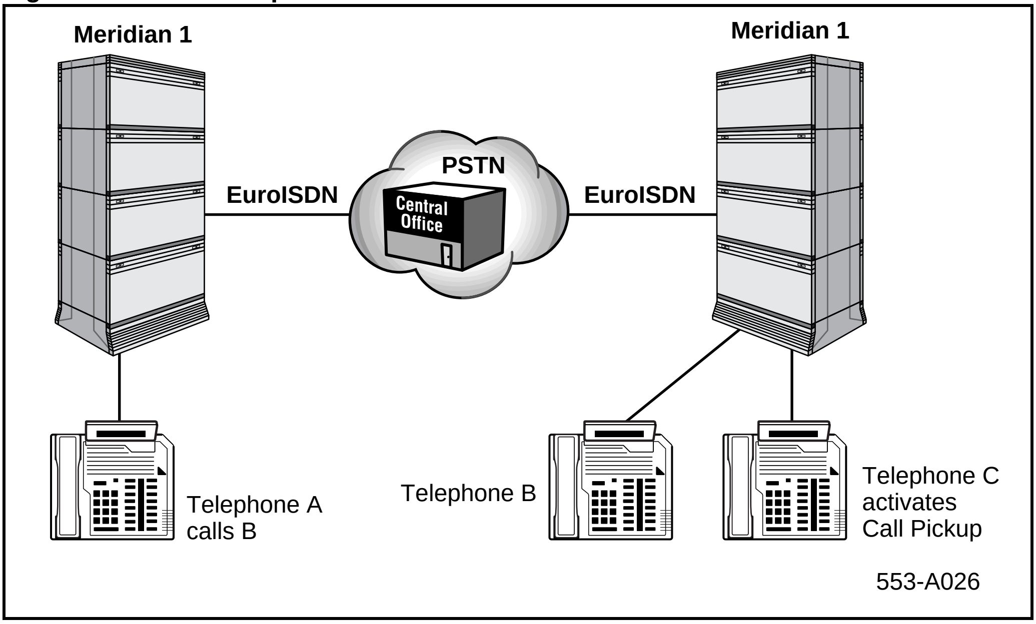
The same Class of Service controls the CLIR and COLR services. If a user has presentation restricted configured, the number is sent to the other party with the presentation flag set to restricted for incoming and outgoing calls.

Call Pickup

Refer to Figure 55. If telephone A at one node calls telephone B at another node but telephone C activates Call Pickup, the name and private number associated with telephone A are displayed on telephone C, according to the presentation programming of telephone A.

The display of telephone A shows the name and private number associated with telephone B while telephone B is ringing, if the presentation is allowed. Telephone A is updated with name and Connected Number information when a user at telephone C answers.

Figure 55 - Call Pickup in a EuroISDN network

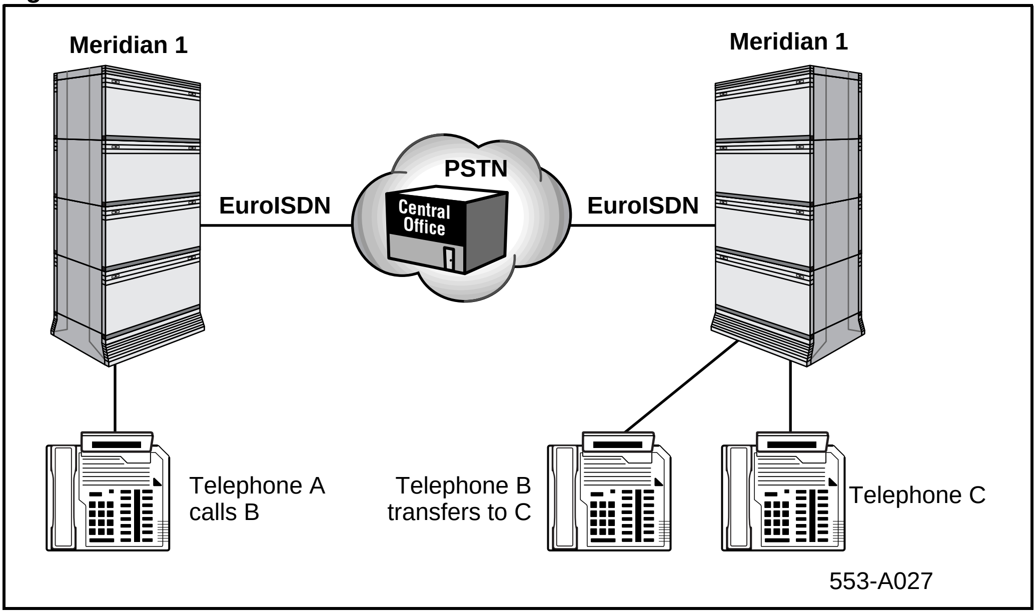


Call Transfer

Refer to Figure 56 for an illustration of a local Call Transfer. Refer to Figure 57 for an illustration of an external Call Transfer. Note that in these illustrations, Explicit Call Transfer is not activated.

EuroISDN

Figure 56 - Local Call Transfer



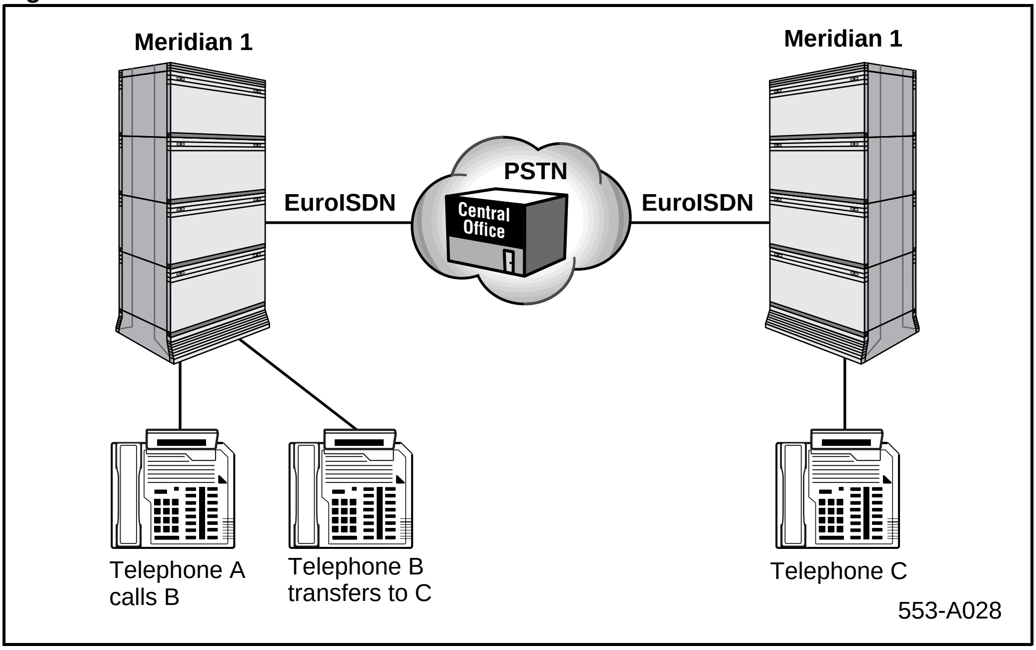
Transfer on ringing (internal)

Figure 56 illustrates a local transfer of an incoming EuroISDN call that has BNE Name information and a private CLID. For this discussion, assume the user transfers the call while telephone C is ringing. When the Call Transfer is completed, the name and private number associated with telephone A, display on telephone C, according to the presentation programming of telephone A. Telephone A shows the name and number associated with telephone B.

Transfer after answer (internal)

Figure 56 illustrates a local transfer of an incoming EuroISDN call that has BNE Name information and a private CLID. For this discussion, assume the user transfers the call after a user at telephone C answers. When the Call Transfer is completed, the name and private number associated with telephone A display on telephone C, according to the presentation programming of telephone A. Telephone A shows the name and number associated with telephone B.

Figure 57 - External Call Transfer



Transfer on ringing (external)

Figure 57 illustrates the transfer of a local call over the EuroISDN network to telephone C. For this discussion, assume the user transfers the call while telephone C is ringing. When the Call Transfer is completed, and while telephone C is ringing, the displays do not change. When the user at telephone C answers, the user’s name and number associated with telephone C display on telephone A, according to the presentation programming of telephone C. Telephone C shows the name and private number of telephone B.

Transfer after answer (external)

Figure 57 illustrates the transfer of a local call over the EuroISDN network to telephone C. For this discussion, assume the user transfers the call after the user at telephone C answers. When the Call Transfer is completed, the displays do not change; the user’s name and number associated with telephone B display on telephone A. Telephone C shows the name and private number associated with telephone B.

EuroISDN

Conference

Figure 58 illustrates a conference call involving parties connected through a EuroISDN network.

Figure 58 - Local Conference

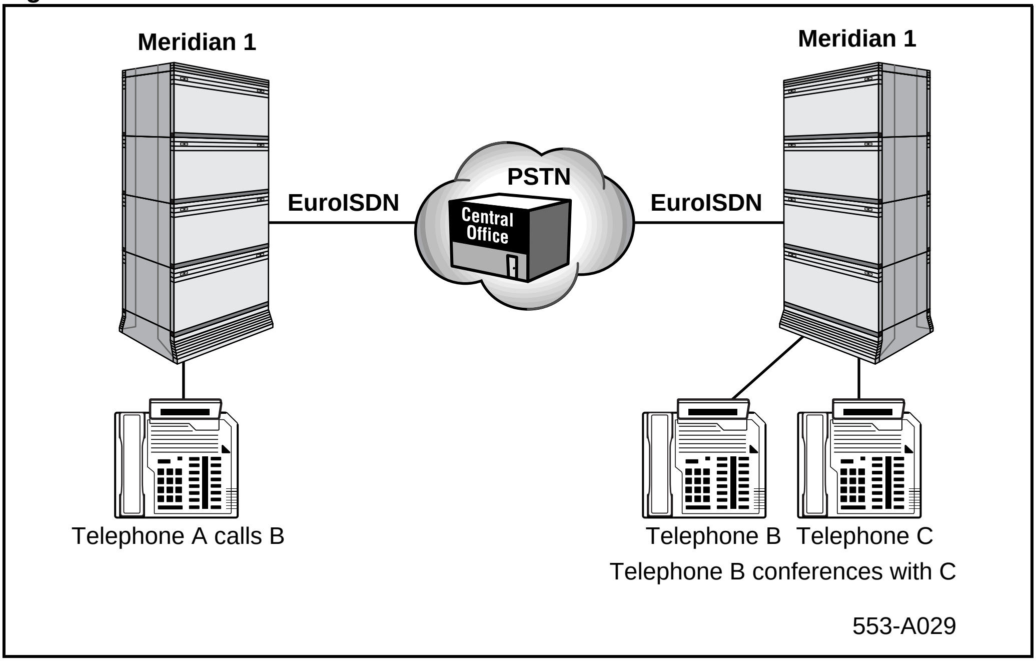
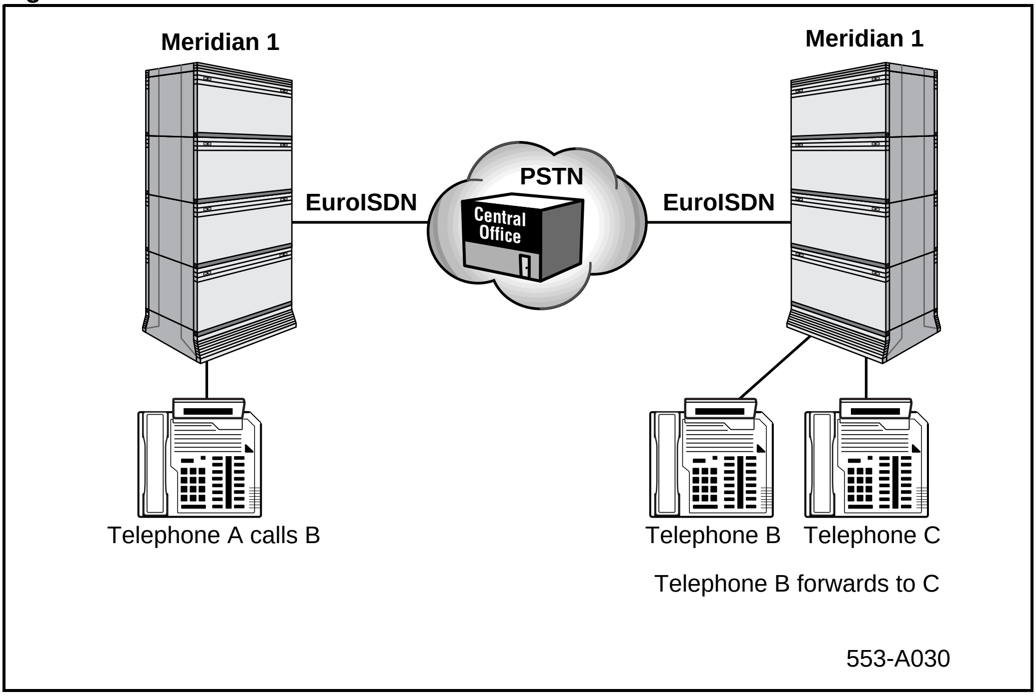


Figure 58 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID which is conferenced locally. If telephone B drops out of the conference call, the user's name and private number associated with telephone A display on telephone C, if the presentation is allowed, but the display on telephone A does not change.

Call Forward

Figure 59 illustrates a local Call Forward situation involving parties connected through a EuroISDN network. Figure 60 illustrates an external Call Forward situation involving parties connected through a EuroISDN network. Note that in this illustration, Explicit Call Transfer is not activated.

Figure 59 - Local Call Forward



Call Forward All Calls (internal)

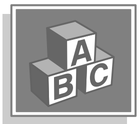
Figure 59 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID which forwards all calls to a local telephone. While telephone C is ringing, telephone A shows the name and number associated with telephone C. The name and number associated with telephone A display on telephone C, according to the presentation programming of telephone A.

Call Forward No Answer (internal)

Figure 59 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID which forwards calls on a no answer condition to a local telephone.

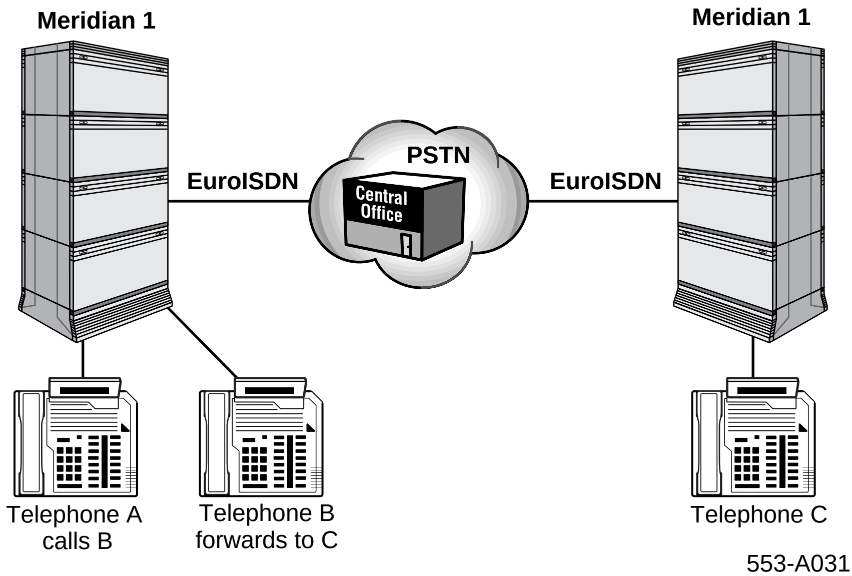
After the call forwards, and while telephone C is ringing, the display on telephone A shows the name and private number associated with telephone B.

EuroISDN



The name and number associated with telephone A display on telephone C, according to the presentation programming of telephone A. When the user at telephone C answers, telephone A shows the name and number associated with telephone C. The display on telephone C does not change.

Figure 60 - External Call Forward



Call Forward All Calls (external)

A local call can forward over a EuroISDN network, as shown in Figure 60. While telephone C is ringing, telephone A shows the name and number associated with telephone C. The name and number associated with telephone A are displayed on telephone C, according to the presentation programming of telephone A.

Call Forward No Answer (external)

A local call can forward unanswered calls over a EuroISDN network as shown in Figure 60. After the forwarding occurs, and while telephone C is ringing, telephone A shows the name and private number associated with telephone B. The name and number associated with telephone A are displayed on telephone C, according to the presentation programming of telephone A.

EuroISDN

When the user at telephone C answers, telephone A shows the name and number associated with telephone C. The display on telephone C does not change.

Hunting/Group Hunt

Figure 61 illustrates a local Hunting situation.

Figure 61 - Local Hunting

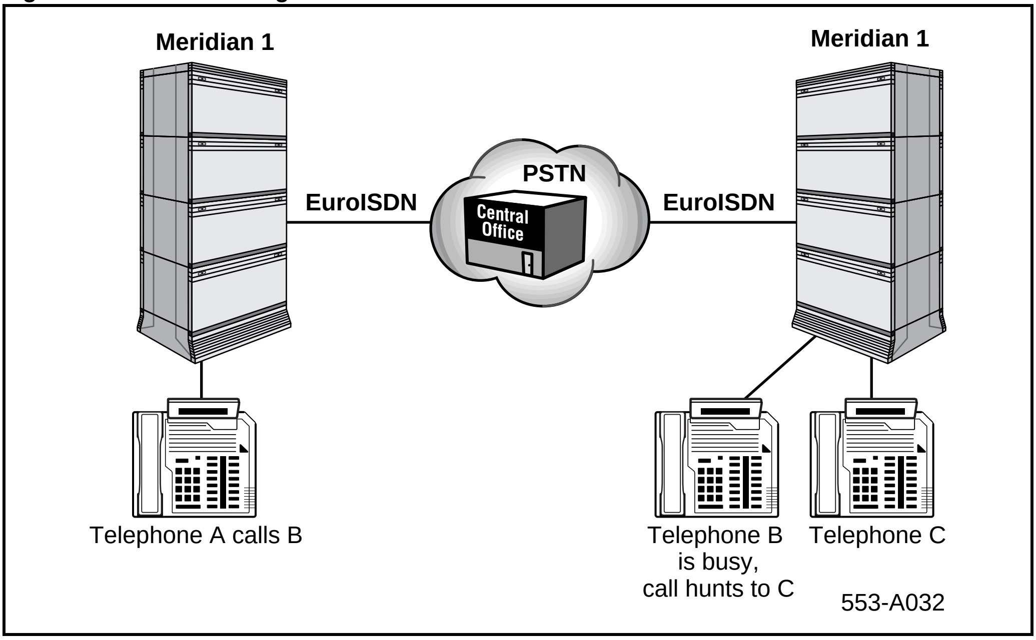
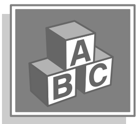


Figure 61 illustrates an incoming EuroISDN call with BNE Name information and a private CLID that redirects to telephone C when telephone B is busy. As soon as telephone C rings, the name and private number associated with telephone A display on telephone C, according to the presentation programming of telephone A. The name and private number associated with telephone C are delivered to telephone A.

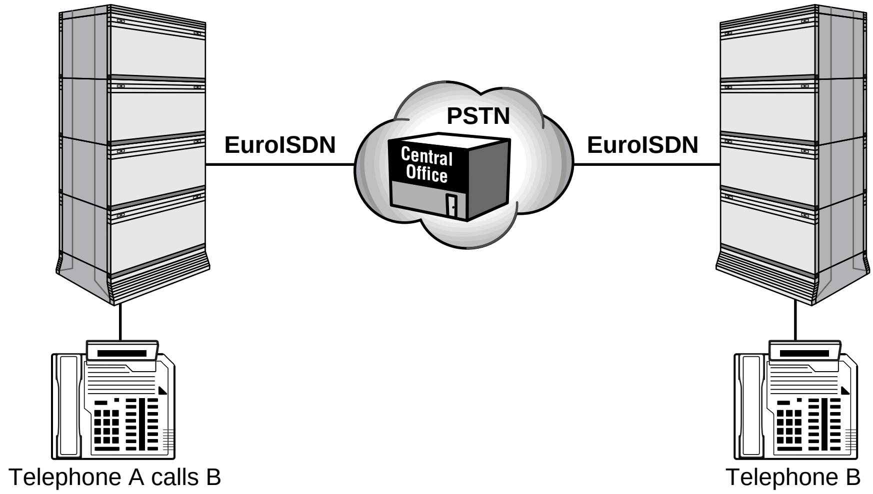
EuroISDN



Advice of Charge (AOC)

Figure 62 illustrates an example of a situation involving the Advice of Charge feature.

Figure 62 - AOC and Charge Display



An outgoing EuroISDN call, carrying BNE signaling, as shown in Figure 62, is charged by the Central Office. If telephone A is a Meridian Modular Digital telephone and AOC Real Time Display is configured, the charge is displayed in the right corner of the first line, when it is received. The display of charge takes precedence over the display of name. The name displayed on telephone A is truncated if there is not enough space to display both the name and the charge.

Display of Access Prefix

The private numbers provided by the BNE feature are displayed with the prefixes configured by the Display of Access Prefix on the CLID feature for a private numbering plan.

Networking feature interactions

Call Diversion (diversion notification sent):

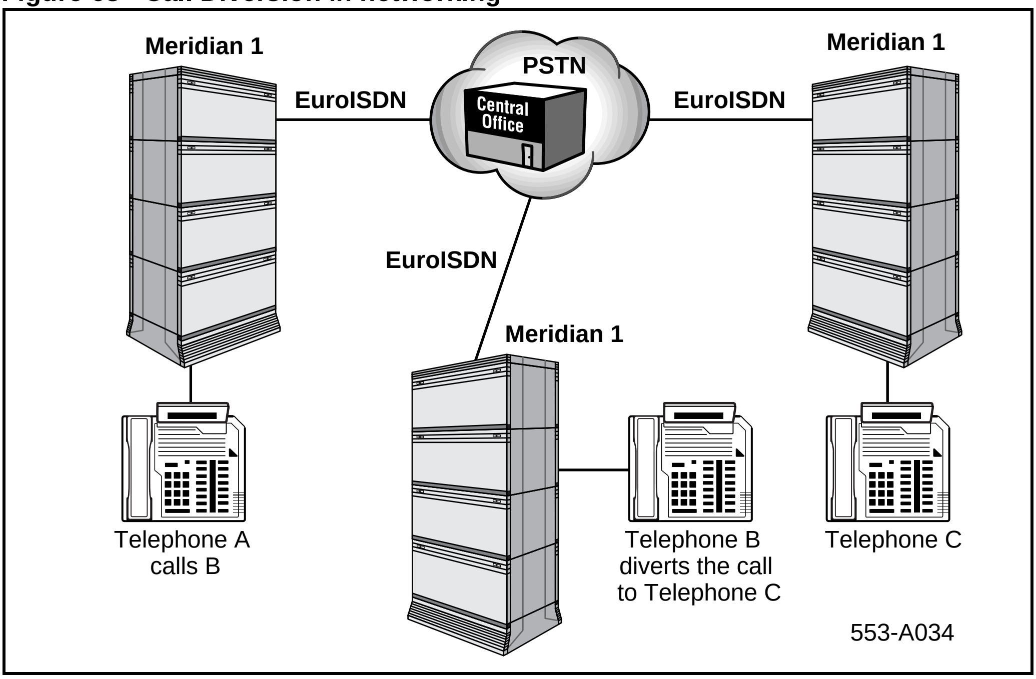
Call Forward All Calls (Call Forward Unconditional)

Figure 63 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID forwarding all calls to telephone C over the EuroISDN network. After the call forwards, the BNE information name and number are replaced by the notification numbers provided by the Call Diversion feature. While telephone C is ringing, telephone A shows the name associated with telephone C.

Call Forward No Answer

Figure 63 illustrates an incoming EuroISDN call that has BNE Name information and a private CLID forwarding unanswered calls to telephone C over the EuroISDN network. After the call forwards, the BNE information name and number are replaced by the notification numbers provided by the Call Diversion feature. Telephone A shows the name associated with telephone C when the call is established.

Figure 63 - Call Diversion in networking

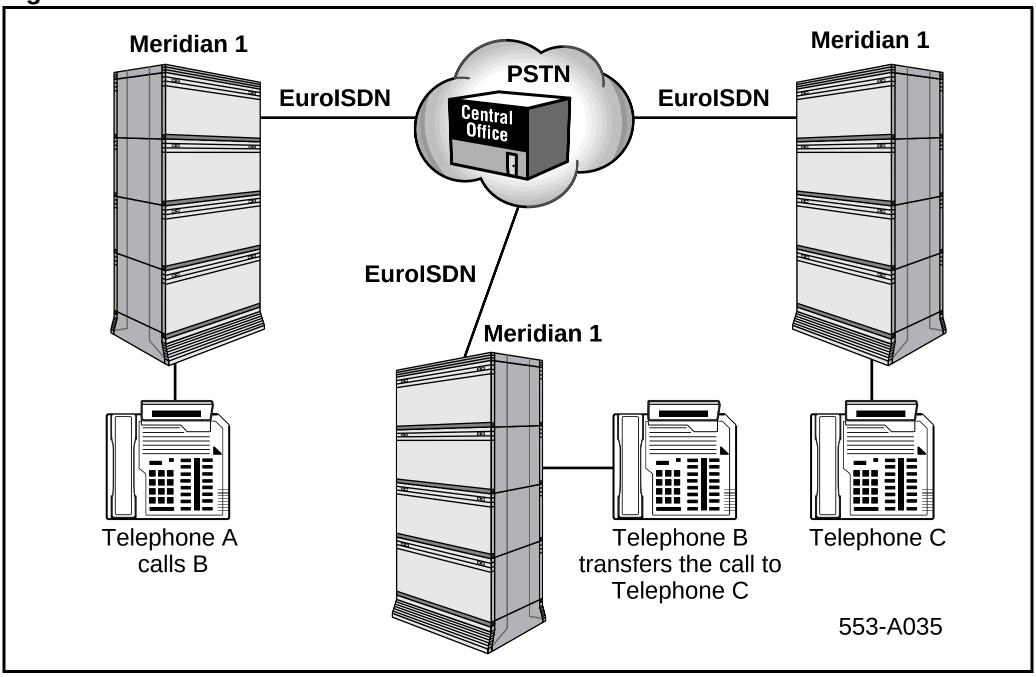


EuroISDN

Explicit Call Transfer (Call Transfer notification sent)

Figure 64 illustrates a Call Transfer across a network. Before the transfer is completed, telephone C shows the name and number associated with telephone B. Telephone A shows the name and number associated with telephone B. After the transfer, the BNE information name and number are replaced by the notification numbers provided by the Call Transfer feature.

Figure 64 - Call Transfer across a network



BNE Call Diversion feature interactions

Call Detail Recording (CDR)

When a call forwards by rerouting, no CDR ticket is generated because no established call takes place and the rerouting operation is done by the CO.

Call Forward by Call Type

This feature redirects internal and external calls differently with both the Call Forward No Answer and Hunting features. Different DNs are programmed for internal calls and external calls.

Call Forward by Call Type is supported by the EuroISDN Call Diversion service and the definition of an internal call is not modified by this feature. In particular, ISDN trunk calls using public numbering are considered external.

> ***Note:*** The system does not attempt to determine the real originating party with EuroISDN; it only looks at the type of numbering plan for the EuroISDN call.

Call Forward Option

The active Class of Service is always the served user's Class of Service. The EuroISDN Call Diversion feature is not affected by the OPT configuration (CFO/CFF) in the served user's Customer Data Block.

Call Waiting Redirection

The Call Waiting Redirection (CWTR) feature allows unanswered calls in the Call Waiting state to be redirected using Call Forward No Answer (CFNA). The waiting call redirects to the active telephone's CFNA DN, after the CFNA timer defined in the Customer Data Block expires. The CFNA DN (which can be a messaging service such as Meridian Mail, Voice Mail, and Message Center) handles this redirected call as an unanswered call.

The EuroISDN Call Diversion service handles this type of call as a usual Call Forward No Answer call.

Phantom TN

When a Phantom TN is Call Forwarded, the EuroISDN Call Diversion feature treats the Phantom TN as a normal TN.

EuroISDN

Flexible Orbiting Prevention Timer

The Flexible Orbiting Prevention Timer feature prevents a call from being diverted off-node by the Call Forward feature at a station for a period of FOPT seconds after a call has already been forwarded off-node by a station. FOPT is defined on a customer group basis.

EuroISDN Call Diversion supports the Flexible Orbiting Prevention Timer feature. Consider using it as a workaround to help prevent Reciprocal Call Forward network-wide. However, while this feature allows you to avoid infinite looping, it also limits the number of diversions that can be performed by a telephone in a specified period of time. Therefore, if you expect frequent use of EuroISDN Call Diversion, consider using Total Redirection Count instead, which limits the number of diversions on a single call.

Call Forward/Hunt Override

The feature allows the use of the Flexible Feature Code for Call Forward/HUNT Override to override Call Forward All Calls, ICP-Call Forward, Call Forward No Answer, Hunting or Make Set Busy at the telephone level and by attendants, in both stand-alone and network (MCDN) applications.

This feature is not supported by EuroISDN Call Diversion. A Meridian 1 user can neither originate nor receive a call by using the FFC for CFHO through EuroISDN.

Access restrictions/trunk group access restrictions

EuroISDN Call Diversion is not performed if the served user is not able to access the route to the diverted-to-node.

Networking feature interactions

User to User (UUS1) services

The Meridian 1 does not support the diversion of UUS1 messages.

Network Automatic Call Distribution (NACD)

If a DID call terminates on an ACD DN, the DID call is linked to the ACD queue. NACD takes precedence over EuroISDN Call Diversion.

BNE Name and Private number display

After an incoming EuroISDN call with BNE Name information and a private CLID forwards through a EuroISDN network, the BNE information disappears from the display and is replaced by the notification numbers provided by the Call Diversion feature.

Auxiliary product interactions

Symposium Call Center Server

The call type is updated in the SCC message if the EuroISDN call is diverted to a CDN.

Meridian Link

Present Call Indication (PCI)

This message contains an IE called "Call Type" which contains diversion information about the incoming call. This field is updated for an incoming diverted EuroISDN call.

Unsolicited Status Message (USM)

When a telephone stops ringing because the Call Forward No Answer feature has sent the call to the EuroISDN network, a USM message is sent to Meridian Link.

Meridian Mail

A call diverted to Meridian Mail through EuroISDN can access Meridian Mail functionalities (such as message reception and mailbox interrogation) in the same way as a simple call to the mailbox.

BNE Explicit Call Transfer feature interactions

Call Detail Recording (CDR)

For invocation of Explicit Call Transfer within the public network, CDR tickets issued do not reflect the complete duration of the call to the transferred-to telephone.

When Call Transfer is completed on an established call, an S (Start) record is generated for each calling party involved at the time Call Transfer was activated. After the call is terminated, an E (End) record is generated showing its final disposition. Start and End records are generated at the Transferring node.

EuroISDN

If more than one transfer occurs, an X (Transfer) record is generated for each transfer when the primary call involved a CDR-X call. If N transfers occurs, (N-1) records are generated in addition to the Start and End records.

When a EuroISDN gateway is used, the BLID field is updated with the Call Transfer Notification information received at the Transferring node.

In a stand-alone situation, when only the served user A is on the Meridian 1, no notification is received. There is always one incoming call, and one outgoing call, because it is not possible to transfer an incoming DID call over an outgoing DID call. When a transferred call is released, the BLID field of the E record is filled with the Redirection number sent on the outgoing side of the transfer.

Networking feature interactions

BNE Name and Private Number display

BNE Name and Private number information can not be carried out in EuroISDN Explicit Call Transfer Notifications. If an incoming EuroISDN call with the BNE name information and the private CLID is being forwarded through EuroISDN, after Call Transfer occurs, the BNE information name and number are replaced on the display and the notification numbers provided by the Explicit Call Transfer feature.

Auxiliary product interactions

Meridian Link

Unsolicited Status Message (USM)

When an ACD agent is transferred over a EuroISDN link, a USM message is sent to the Meridian link.

Meridian Mail

A caller transferred to Meridian Mail through EuroISDN can access Meridian Mail functionalities such as message reception and mailbox interrogation.

What to have ready

The following checklist summarizes the steps you should take before implementing the EuroISDN feature and/or the optional related features associated with the basic feature.

Table 55
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Determine if EuroISDN is available in your area. Find out which ETS specification your interface must comply with.
✔		Decide if you will implement NARS or CDP (including FNP) or both.
✔		Assess the features and capabilities that are not supported. If you need any of these, discuss work around plans with your supplier. Train users accordingly.
✔		Train users to understand the redirection DNs that will appear on telephone displays.
✔		Adjust CDR processing to accommodate NPI and TON fields.
	✔	Find out if users need to trace malicious calls coming from EuroISDN trunks and the public network. Set up a process internally and with the network operator for handling situations where calls are traced.
✔		Find out about hardware you will need. Determine what impact this will have on your system.
— continued —		

EuroISDN

Table 55
Checklist (Continued)

Basic	Optional	Preparation
	✔	Determine if VPN will save you money.
	✔	Determine the costs of installing EuroISDN Trunk- Network side for VPN.
		If you are using VPN, determine if you need any of the Business Networking Express (BNE) features. If implementing any of the BNE features, ensure that you fully recognize the interactions that these features have with other features already installed on your system.
✔		Decide if you want to enable fallback on EuroISDN 7kHz and Videotelephony Teleservices.

QSIG

Within the Nortel Networks ISDN Primary Rate Interface (PRI) product families of Meridian 1, SL-100, DMS-100, and DMS-250, the protocol used is based on the MCDN proprietary protocol specification which defines both the basic call service and supplementary services. To achieve interworking between different private networks of non-Nortel Networks PBXs and Centrex, a regional (European) standard was defined and adopted regionally and globally.

The European Computer Manufacturer's Association (ECMA) has defined an ISDN protocol that specifies the layer 3 signaling requirement for support of circuit switched call control at the "Q" reference point between Private Telecommunications Network Exchanges (PTNXs) connected within a Private Telecommunications Network (PTN). This protocol, referred to as QSIG, has been adopted by the European Telecommunications Standards Institute (ETSI) which represents European interests and the International Standards Institute (ISO) which represents global interests. The National ISDN User's Forum in North America, Japan and Australia are also showing interest in this standard.

QSIG is oriented towards signaling and services that occur between peer-to-peer switches (two PBXs, two Centrex systems, a PBX and a Centrex system). The signaling required to provide services is exchanged across a "Q" reference point.

Both BRI trunks and PRI connectivities are supported based on the QSIG standard.

Basic configuration

Standards

In X11 Release 20, support for the ISDN QSIG interface and the basic call capability was introduced. This version of the interface was based on the ETS 300-172, first edition (1990) document.

In X11 Release 22, the QSIG-GF (Generic Functional) Transport interface was introduced, in compliance with the ETS 300-239 (1993) standard document. The basic call functionality was upgraded to be compliant with the second edition of the ETS 300-172 document. In X11 Release 24, an enhancement makes the QSIG/ETSI GF transport platform compliant with the following documents:

- ISO/IEC DIS 11582
- ETS 300 239
- ETS 300 196-1 and ETS 300 196-1 A1

With X11 Release 24.2x, two European QSIG interfaces are available. One version (called the ISDN QSIG-BC and QSIG-GF Compliance Update) is compliant with latest editions of the standards. The other version is compliant with older (X11 Release 22) versions of the standards.

The latest European standards documents for QSIG are:

- ETS 300-172, fourth edition (1997)
- ETS 300-239, second edition.

Features

The following features are supported by the Meridian 1 on the QSIG interface as of Release 24:

- basic call service
- Basic Rate Trunk Access (BRIT)
- 64 kbps clear bearer capability
- TIE call types

QSIG

- Calling Line Identification Presentation/Restriction
- Connected Line Identification Presentation/Restriction
- Name Display Services
- Name Display Enhancement
- channel negotiation
- Flexible Numbering Plan
- Uniform Dialing Plan
- Coordinated Dialing Plan
- Enbloc dialing
- Overlap sending/receiving
- Loop avoidance (transit count) on ETSI QSIG, not ISO QSIG
- Party Category *partially* supported on ETSI QSIG, not ISO QSIG
- Call Completion for Busy Subscriber
- Call Completion on No Reply
- Call Completion Enhancement
- Call Diversion Notification
- Call Diversion Notification Enhancements
- ANF Path Replacement
- Alternate Routing (during network congestion)
- Call Transfer Notification
- Generic Functional (GF) Transport
- GF Enhancement

Because of the X11 Release 24 MCDN End To End Transparency feature, features such as Network Attendant Service (NAS), Network ACD (NACD) and Network Message Services-Message Center (NMS-MC) and Network Message Services-Meridian Mail (NMS-MM) are supported in QSIG networks and in mixed MCDN/QSIG networks.

QSIG

Note: The NMS-MM features Call Sender and Thru-Dialing are not supported.

Feature limitations

As a network planner you need to know about the current feature limitations of the QSIG protocol. They are:

- message segmentation and reassembly is not supported yet. The maximum length of an ISDN message is 260 bytes.
- only circuit mode connection is supported
- nB+D is not supported. It is not defined in the QSIG specification.
- each customer group on a switch needs it own D-channel for call-independent QSIG transport service (MCDN networks used the Private Network Identifier (PNI) to identify a customer group in the signaling between Meridian 1 switches)
- applications that make use of call-independent transport services may be limited by the amount of system resources, such as call registers, allocated at transit nodes. This is because call registers are tied up at transit nodes that must retain information pertaining to call-independent connections for the time required by an application in order to route subsequent messages associated with those connections properly.

The following features are not supported by Release 24 software. As QSIG matures, features from this list will move to the list of features that are supported.

- Call by Call Service (ISA)
- Backup D-channel
- ISDN Signaling Link
- Network Call Trace
- Remote Virtual Queuing
- Network Distinctive Ringing
- Off-Hook Queuing

- Virtual Network Service (QSIG channel can act as VNS B-channel)
- Preventing Reciprocal Call Forward

MCDN interworking with QSIG

The MCDN features that interwork with QSIG are:

- basic call (Release 20 - Europe only, Release 22 - global)
- Calling Line ID Presentation/Restriction (CLIP/CLIR) (Release 20)
- Connected Line Presentation/Restriction (COLP/COLR) (Release 20)
- Calling Name Identification Presentation (CNIP) (Release 22)
- Connected Name Identification Presentation (CONP) (Release 22)
- Calling/Connected Name Identification Restriction (CNIR) (Release 22)
- Name Display Enhancement (Release 24)
- Call Completion to Busy Subscriber (CCBS) for ETSI QSIG
- Call Completion on No Response (CCNR) for ETSI QSIG
- Call Diversion Notification
- Call Transfer Notification
- Network Attendant Service (NAS)
- Network ACD (NACD)
- Network Message Services-Message Center (NMS-MC)
- Network Message Services-Meridian Mail (NMS-MM)

Alternative configurations

Japan TTC Common Channel Signaling

Japan Telecommunication Technology Committee (JTTC) Common Channel Signaling is the Japanese version of the International Standard Organization (ISO) ISDN QSIG. It specifies the Layer 3 protocol signaling requirement for support of circuit switched call control at the "Q" reference point between a Private Integrated Services Digital Network (PISN) and a Private Telecommunication Network (PTN).

The JTTC Common Channel Signaling feature provides basic call service on Meridian 1 ISDN 1.5 Mbps PRI on TTC connectivity.

The JTTC interface supports the following call services on the Meridian 1:

- Basic Call Service
- 64K Clear Bearer capability
- Calling Line Identification Presentation/Calling Line Identification Restriction (CLIP/CLIR)
- Channel Negotiation
- Enbloc Dialing
- User to User Information Element Transparent Transport
- TIE call types
- Traveling Class Mark (TCM) Transparent Transport

Only circuit mode connection is supported on the JTTC interface.

For Option 11C, ISDN Signaling Link (ISL) is not supported on the JTTC interface.

The 2.0 Mbps JDMI interface is required for ISDN Signaling Link (ISL) functionality.

The Multi-purpose Serial Data Link (MSDL) card is required.

Improving feature performance

The information that follows may make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Name Display features

ISDN QSIG Name Display

This X11 Release 22 feature provides three supplementary services pertaining to the calling or connected party name display. These supplementary services allow the calling/connected party to either present or restrict the display of name identification on an ISDN QSIG PRI or ISDN QSIG BRI trunk. The ISDN QSIG Name Display feature supports the following services:

- Calling Name Identification Presentation (CNIP)
- Connected Name Identification Presentation (CONP)
- Calling/Connected Name Identification Restriction (CNIR)

Calling Name Identification Presentation (CNIP) service is available to the called/connected party. When this service is enabled, the calling party’s name is displayed on the connected party’s telephone. CNIP service is available on a permanent basis only, and requires the calling party to have the Class of Service Name Presentation Allowed.

Connected Name Identification Presentation (CONP) is a service available to the calling party. When this service is enabled, the called/connected party’s name is displayed on the calling party’s set. CONP is provided on a permanent basis only and requires the called/connected party to have the Class of Service Name Presentation Allowed.

Calling/Connected Name Identification Restriction (CNIR) prevents the calling/connected party’s name from being presented to a called/calling party. CNIR is invoked on either a permanent basis, provided the calling/connected party has a Class of Service Name Presentation Denied, or on a per-call basis. When CNIR is activated

QSIG

on a per-call basis, the Class of Service Name Presentation Allowed is configured, and the Calling Party Privacy (CPP) Flexible Feature Code (FFC) is dialed prior to initiating a call. This capability is not supported for BRI telephones. This supplementary service restricts presentation of the calling party's name during a normal call establishment and also when the possibility of name presentation arises again in the same call from the operation of other features, such as Call Transfer, Call Forwarding or Hunting.

QSIG allows a maximum of 50 characters to be displayed. However, there can only be up to 27 characters displayed by the Calling Party Name Display feature. Therefore, if the name string exceeds 27 characters, it is truncated.

Users cannot override the name restriction feature. When name information is restricted, there is no way for a user to display it.

Names associated with trunk groups are also supported.

Name Display Enhancement

In X11 Release 24, there is an option of choosing the method of Operation Coding for the Name Display supplementary services. The choices are:

- Object Identifier - for ETSI interfaces
- Integer Value - for ISO interfaces

The Operation Coding method is set up when programming the Remote Capability of a D-channel.

QSIG links

Name Display operates even when the following features are provided over a QSIG link:

- Call Pickup (caller at different switch from the ringing telephone and the telephone answering the call)
- Call Transfer
- Conference

- Call Forward of the following types:
  - Call Forward All Calls
  - Call Forward Busy
  - Call Forward by Call Type
  - Call Forward External Deny
  - Call Forward, Internal Calls
  - Call Forward No Answer
  - Call Forward No Answer, Second Level
- Hunting
- Incoming Digit Conversion (IDC) - If name information is not received from a DID trunk, IDC is activated and a name is associated with the new digit sequence, this information is passed over a QSIG link to a terminating telephone. If name information is received from a DID trunk, it takes precedence over the IDC trunk name.

QSIG/MCDN links

Name Display operates even when the following features are performed over a QSIG/MCDN link:

- Call Forward of the following types:
  - Call Forward All Calls
  - Call Forward Busy
  - Call Forward by Call Type
  - Call Forward External Deny
  - Call Forward, Internal Calls
  - Call Forward No Answer
  - Call Forward No Answer, Second Level
- Hunting

- Call Pickup Network Wide (caller at one switch, ringing telephone at another switch and answering telephone at a third switch)
- Call Transfer
- Conference
- Incoming Digit Conversion (MCDN/QSIG and QSIG/MCDN)

Choose a Remote Capability of ND3 if you want equivalent name display service between QSIG and MCDN networks.

Business Network Express (BNE) Name and Private Number Display

For detailed information on BNE features, which were introduced in Release 25, refer to the EuroISDN section of this guide.

BNE Name and Private Number Display is offered across a EuroISDN/QSIG gateway.

BNE Name Display

The Name Display functionality is based on the existing MCDN and QSIG Call Party Name Display (CPND) and Display of Calling Party Denied (DPD) features. With BNE Name and Private Number Display, when a user dials a private network number to reach a user at another Meridian 1 site through the public network, the ESN software causes the dialed number to be outpulsed as a public number. The BNE software inserts the Calling Name and the Private CLID to the call.

At the destination switch, the Private CLID is displayed, along with the Calling Name, on the alerted telephone. The name associated with the alerted telephone is delivered to the calling user and displayed on the calling telephone.

When the call is answered, the Connected Name and the private Connected Number is provided to the calling user set. Consistent with MCDN and QSIG networking, the letter H is displayed in front of the private number.

You can implement restrictions on displaying the name and number of the calling, called, or connected party.

BNE Private Number

This functionality delivers a Calling Party Number in a private format (based on a Coordinated Dialing Plan or Uniform Dialing Plan numbering plan) in addition to the public-format Calling Party Number.

The private format depends on the numbering plan the caller used to dial the call.

The private calling number is constructed based on the CLID Enhancement feature. It contains the following information:

- numbering plan field (private)
- type of number field (CDP or LOC or unknown)
- the DN digits of the calling telephone prefixed by an LSC (CDP) or HLOC (UDP), if configured
- presentation flag to allow or deny the display on the called user's telephone

Call Completion/Queuing

ISDN QSIG/EuroISDN Call Completion

This feature expands Ring Again functionalities on QSIG and EuroISDN interfaces in X11 Release 22. This feature uses the QSIG Generic Functional (GF) as a transport platform. This platform is supported on ISO QSIG and ETSI QSIG.

The supplementary services introduced by Call Completion are:

- Call Completion to Busy Subscriber (CCBS)
- Call Completion on No Response (CCNR)

Call Completion to Busy Subscriber (CCBS) supplementary service allows the calling party to apply a Ring Again request when encountering a busy Directory Number (DN). The Meridian 1 alerts the calling party when the occupied DN is available to receive a call. The calling party has the option of completing this call without making a new call attempt. CCBS is supported on QSIG and EuroISDN signaling protocols.

Call Completion on No Response (CCNR) supplementary service allows the calling party to apply a Ring Again request to an unanswered DN. With this service, the Meridian 1 alerts the calling party when the dialed DN becomes idle after a period of activity. The calling party has the option of completing the call without making a new call attempt. CCNR is only supported on the QSIG signaling protocol.

ISDN QSIG/EuroISDN Call Completion supports Uniform Dialing, Customer Dialing and Group Dialing plans.

Gateways between EuroISDN CCBS, MCDN Network Ring Again on Busy and DPNSS1 Call Back When Free are supported.

Call Completion to Busy Subscriber and Call Completion on No Response are not supported on ISDN BRI sets.

ISDN QSIG/EuroISDN Call Completion Enhancement

In X11 Release 24, the following enhancements are available:

- the Operation Coding method for the CCBS and CCNR Supplementary Services is configurable on the D-channel for PRI routes and on the route for BRI trunks. The two methods are:
  - Object Identifier – used for ETSI interfaces
  - Integer Value – used for ISO interfaces
- the CCBS and CCNR features are supported over ETSI/QSIG to MCDN and DPNSS1 gateways

There are two methods used for call completion:

- Connection Retention. (The signaling connection remains established until call completion occurs or the call completion attempt is cancelled.)
- Connection Release. (The signaling connection is cleared after each phase of call signaling. A new signaling connection is established for each subsequent phase of call-independent signaling.)

On the originating side, the Meridian 1 supports only Connection Retention. This matches the call completion method performed by EuroISDN. On the terminating side the Meridian 1 also supports Connection Release. With Connection Release you must have a symmetrical dialing plan. In other words, all switches must be equipped with NARS, if the CLIDs are in the UDP format; all switches must be equipped with CDP if the CLIDs are in the CDP format.

There are two methods used for call completion recall:

- Non-path Reservation. (A B-channel connection between the originating and terminating sides is established after the originating party answers the recall. The recall attempt is cancelled if there is network congestion.)
- Path Reservation. (A B-channel between the originating and terminating parties is established before presenting the recall to the originating party, in order to avoid network congestion.)

On the originating side, the Meridian 1 supports only the Non-path Reservation method for call completion recall. This matches the call completion recall method performed by EuroISDN and MCDN and the Path Reservation method is not supported over QSIG to MCDN or DPNSS1 gateways. On the terminating side, the Meridian 1 also supports Path Reservation.

If the called party is busy again after the originating party responds to a call completion recall, there are two methods possible to handle the situation:

- Request Retention. (The call completion request remains in place at both the originating and terminating sides. The terminating side begins monitoring the terminating party.)
- Service Cancellation. (The call completion request is cancelled at both the originating and terminating sides.)

The Meridian 1 supports the Service Cancellation method.

QSIG

QSIG Alternate Routing (when network is busy)

When a caller tries to establish a call across a QSIG network and all of the routing alternatives at one of the nodes are congested (busy) or there are other reasons that a node is congested, this feature allows a signal to be sent back from the congested node to the preceding node. The preceding node can reroute the call to the final destination, bypassing the congested node. The bypass occurs by sending the call out of the preceding node on an alternate route in the route list index for calls to the dialed destination.

Transit nodes can reroute calls or send a message back to a preceding switch. Originating switches always attempt to reroute calls.

The signal sent back to the preceding node from a congested switch is called a Call Clearing Message. When a transit node receives an incoming Call Clearing Message from the outgoing trunk to a congested switch, it routes the call based on the programming associated with that route in the route list.

Alternate Routing programming options on the route list entry can be one of the following:

- NRR: No rerouting, pass the Call Clearing Message back to the previous node, if at a transit node; give the user a congestion tone, if at an originating node
- RRO: Reroute, pass the Call Clearing Message back to the previous node, if the node is a transit switch; if the node is an originating switch, the system looks for an alternate route
- RRA: Reroute the call at a transit node or originating node

When the transit node tries alternate routing, the channel to the congested node is released and a new call is generated on the alternate route. If the first alternate route is busy, the next one in the route list index for the dialed destination is tried until an available one is found. If no route is available, a Call Clearing Message with congestion information is sent to the preceding node.

Congestion tone is given to the user by the originating switch, if no alternate route can be found to bypass the congestion.

The Cause Information Element carries information about the cause of congestion. In a QSIG network, QSIG Alternate Routing works if the first Call Clearing Message during times of congestion is a DISCONNECT or RELEASE COMPLETE message and the received cause value is programmed to activate the feature. There is a choice between two options in the route list index that determines the set of clearing causes that will activate this feature.

Option 1 cause values include:

- cause 34 *No channel/circuit available*
- cause 38 *Network out of order*
- cause 42 *Congestion*

Option 2 cause values include:

- cause 27 *Destination is out of service*
- cause 34 *No channel/circuit available*
- cause 38 *Network out of order*
- cause 42 *Congestion*

Option 1 is the default option.

If no RELEASE COMPLETE message is received from the congested node, the transit node does not alternately route the call.

UDP or CDP dialing plans are required for QSIG Alternate Routing to work. Calls made using trunk route access codes are not offered this functionality.

Mandatory Information Elements (IEs) are transparently tandemed in the SETUP message when alternate routing occurs.

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The following optional IEs are tandemed in the SETUP message when a call is alternately routed, if they were present in the original call:

- calling number
- called and calling subaddress
- High Layer Compatibility
- Low Layer Compatibility

> ***Note:*** The calling number is modified according to the configuration on the alternate route (insertion of prefixes, home location code or local steering code).

The Generic Functional (GF) Facility IE and the progress IE in the SETUP message are lost when alternate routing occurs at the transit node. Other IEs that are not mandatory or those that are not shown in the list of supported optional IEs are not tandemed by the alternate routing switch. Services such as QSIG Name Display and QSIG Call Diversion are not delivered to the end-user. QSIG Call Transfer is only supported in the call join method (Multi-Party Operations feature).

QSIG Alternate Routing can result in missing progress IEs which can cause a call to fail from a Meridian 1 proprietary telephone to a BRI telephone across a QSIG link.

ANF Path Replacement is rejected, if congestion is encountered.

If there are QSIG-based bearer interfaces supporting Virtual Network Services, QSIG Alternate Routing can function.

Call Diversion

Call Diversion Notification

In Release 23, if calls are diverted across a QSIG PRI2 or BRI trunk network, the display of the calling party telephone can be set up to indicate the reason for redirection and the number (and name, if programmed) of the final destination telephone. The display of the final destination telephone can be set up to indicate the reason for redirection and the number (and name, if programmed) of the diverting telephone or of the calling telephone.

The redirections or diversion-related features that are supported are:

- Call Forward Unconditional (called Call Forward All Calls on the Meridian 1)
- Call Forward to a busy telephone (called Hunting on the Meridian 1)
- Call Forward No Reply (called Call Forward No Answer on the Meridian 1)

Each D-channel must be configured with diversion as a remote capability at the originating, diverting, final destination and rerouting nodes.

Refer to the section in this book called *CLID and Name Display options* for more information about the programming rules that apply to the name and number displays when calls are diverted.

ANF Path Replacement

The Additional Network Feature (ANF) Path Replacement Supplementary Service, offered in X11 Release 23, allows an active connection through a QSIG private network to be replaced by a new, more efficient connection after a call modification. The feature has been developed in compliance with ETSI and ISO standards. The service can operate under two conditions:

- triangulation
- tromboning

Triangulation occurs when a call from one switch to another is diverted to a third switch. There may be a direct connection that can be established between the caller's switch and the third switch. If so, it will be established, using Path Replacement, after the call is answered. The other routes will be dropped, once the optimum connection has been established.

Tromboning occurs when a call from one switch to another is redirected back to the originating switch by call modifications such as transfer or diversion. When the destination telephone answers, the trunks that are no longer needed between the caller's switch and the

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diverting switch are dropped. A new connection is established between the caller and the final destination telephone. For this to work, the trunks must both belong to the same customer group.

QSIG Path Replacement does not directly attempt to reduce the number of trunks in a call to a minimum. Rather, it attempts to replace a non-optimum path with an optimum path. The optimum path is defined as the path which takes the first choice route at all sites involved with the call. The replacement occurs after the final destination telephone answers, not while the telephone is ringing.

There are three different circumstances that act as triggers. The triggers are:

- Call Diversion
- Connected number different from called number
- Congestion

Call Diversion has been explained in the earlier text on triangulation and tromboning. The connected number may be different from the called number in a situation where the caller has dialed a telephone on another switch which cannot indicate redirections to the originating switch. When the call is redirected to a third switch and it sends a connect message to the originating switch, identifying the DN that is answering, the originating switch identifies a mismatch between that number and the called number. This triggers Path Replacement. Congestion can trigger Path Replacement if the first choice trunk route to a particular switch is not available initially to handle a call and the call is routed on an alternate route to another switch in the network. If this switch sends the call to the final destination switch and in the meantime, after the call gets answered, a trunk in the initial choice route at the originating switch becomes available, it can be used for Path Replacement. (Note that Path Replacement does not occur if the congestion occurs at a transit node, not the originating switch).

You can configure each switch that may act as a cooperating switch in a Path Replacement scenario involving other switches in the network to retain its portion of the call connection after the Path Replacement has been invoked at the other locations. This path will be retained, if the connection used the optimum route to reach the rerouted number.

The remote capability must be configured for each route involved with Path Replacement. One or more of the triggers are selected and the retain option must be activated if that functionality is desired. If Path Replacement is chosen but no triggers are configured, Path Replacement activations from a remote switch are processed by the Meridian 1. If Path Replacement is chosen and triggers are configured, Path Replacement activations from a remote switch are processed by the Meridian 1 and activations can be initiated by the Meridian 1.

Call Diversion Notification Enhancements

In X11 Release 24, the Call Diversion Notification functionality has been extended to include a QSIG/MCDN gateway, and a QSIG/DPNSS1 gateway.

Two mechanisms for handling diversion are possible. They are:

- Forward Switching: performs the diversion by joining two circuits. The first circuit is the connection between the originating switch and the switch that is diverting the call. The second circuit is the connection between the switch that is diverting the call and the switch that is receiving the diverted call.
- Rerouting: performs the diversion by replacing the connection from the originating switch to the diverting switch with another new connection from the originating switch to the switch that is receiving the diverted call.

The Meridian 1 supports the Forward Switching method, both in a sending and receiving mode. The Rerouting method is only supported in a receiving mode. The Meridian 1 supports the receipt and treatment of a Rerouting request sent by another vendor switch but does not generate such a request. The notification to the originating and terminating users' telephones is supported with both Rerouting and Forward Switching methods.

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Refer to the section in this book called *CLID and Name Display options* for more information about the programming rules that apply to the name and number displays when calls are diverted.

If there is no answer at the terminating telephone or the terminating telephone is busy after a call has been diverted, the originating user can activate CCNR and CCBS features respectively. The originating switch stores the termination number sent by the diverting switch, to be used to complete the call later.

In optimizing the use of trunks, the QSIG Diversion feature uses diversion by Rerouting to optimize the use of trunks first. The terminating node receives a message stating the optimization method used. QSIG Path Replacement can then be used, if the terminating switch determines that the call has not been fully optimized.

Total Redirection Count is supported on QSIG networks. This is important in cases where there are multiple diversions possible for one call across a network. It is also one way to deal with the lack of support for Preventing Reciprocal Call Forward on QSIG networks.

Business Network Express (BNE) Call Diversion

For detailed information on BNE features, which were introduced in Release 25, refer to the EuroISDN section of this guide.

BNE Call Diversion is offered across a EuroISDN/QSIG gateway.

BNE Call Diversion offers the Call Diversion supplementary services include the following:

- Call Forwarding Unconditional (CFU) - known as Call Forward All Calls on the Meridian 1
- Call Forwarding Busy (CFB) - known as Hunting on the Meridian 1
- Call Forwarding No Reply (CFNR) - known as Call Forward No Answer on the Meridian 1

Call Transfer Notification

In X11 Release 24, when a call transfer occurs over a QSIG PRI2 or BRI trunk link, a notification of the transfer is sent to the originating party and the party to whom the call is transferred. The transfer must be done using the Call Join feature (Multi-Party Operations). ANF Path Replacement takes effect, if applicable, in order to obtain a more efficient connection.

The information appears on the displays of the originating and terminating telephones. It includes:

- the redirection and originating number
- redirection and originating name
- redirection reason mnemonic (a code defined for Call Transfer).

The same notification occurs over a QSIG/MCDN or QSIG/DPNSS1 gateway.

Although Call Transfer by Call Join was supported over a QSIG link as early as Release 20, there was no notification to the users that a transfer had taken place.

The method of Operation Coding is by Object Identifier for ETSI interfaces or Integer Value for ISO interfaces.

QSIG/ETSI GF Enhancement

In X11 Release 24, the enhancement to the GF transport platform comprises the following capabilities:

- modified call control to support call-independent gateways from QSIG to EuroISDN, MCDN and DPNSS1 signaling protocols
- modified call-related Application Protocol Data Unit (APDU) transport to ensure that a call that is being optimized during QSIG Path Replacement is using the optimum path (the first route in the route list index). This modified mechanism is also needed so the Facility Information Elements carried in a SETUP message which has encountered congestion are included in the alternate call, if alternate routing is performed at a transit node. The Call Completion to Busy Subscriber feature and ANF Path Replacement feature have been modified to use this capability.

QSIG

- more accurate calculation of the remaining available length for APDUs in basic call control messages for QSIG GF and ETSI GF transport.

The APDU call-independent transport mechanism is connection oriented. Connectionless APDU call-independent transport is not supported.

When a call-independent SETUP message includes several Facility Information Elements (FIEs) belonging to different Supplementary Services, the gateway is not supported.

MCDN End To End Transparency

Because of the X11 Release 24 MCDN End To End Transparency feature, features such as Network Attendant Service (NAS), Network ACD (NACD) and Network Message Services-Message Center (NMS-MC) and Network Message Services-Meridian Mail (NMS-MM) are supported in QSIG networks and in mixed MCDN/QSIG networks.

> ***Note:*** The NMS-MM features Call Sender and Thru-Dialing are not supported.

Business Network Express (BNE) Explicit Call Transfer

For detailed information on BNE features, refer to the EuroISDN section of this guide.

BNE Explicit Call Transfer is offered across a EuroISDN/QSIG gateway.

BNE Explicit Call Transfer extends the functionality of the private network. It notifies the public network that a transfer has occurred within the private network. Also, it optimizes the call by requesting the public network to perform the transfer.

Feature interactions

ISDN QSIG/EuroISDN Call Completion

Feature interactions

There are some interactions with other features that you should know about before you implement these capabilities. Train users to understand any of the interactions that will affect them in your network.

Attendant Consoles

Call Completion requests cannot be directed to or from an attendant console.

Automatic Call Distribution

Call Completion requests cannot be directed to or from Automatic Call Distribution (ACD) Directory Numbers (DNs).

An ACD set that uses a normal DN key (not the ACD key) can activate the Ring Again key when encountering a busy or no answer situation.

Call Detail Recording

A Call Detail Recording (CDR) is not produced for Call Completion signaling. The second call prints out a CDR record as a normal call.

Call Forward All Calls

When the Call Forward All Calls feature is activated on a local basis and an incoming Call Completion request is received, the Call Completion request is registered against the forwarded DN.

Call Transfer

Call Completion notification is only presented to the Call Completion originating telephone. This notification cannot be transferred to another station. Once the second call is completed, the call can be transferred.

If a user encounters a busy or no answer situation during a transfer operation, Call Completion can be activated.

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Call Waiting

On an Analog (500/2500 type) telephone, Call Completion notification waits until the user has finished an active call. If Call Waiting is configured, notification is presented after the Call Waiting call. If an additional call is queued while Ring Again free notification is waiting, the waiting call takes precedence over the Call Completion notification. An established Call Completion call is also queued if a telephone has Call Waiting feature equipped and is occupied on another call.

Conference

A Call Completion request cannot be made on a conference call attempt.

Direct Inward System Access

Call Completion on Busy Subscriber (CCBS) and Call Completion No Response (CCNR) are not supported on Direct Inward System Access (DISA) calls when the call destination is busy.

Do Not Disturb

An incoming notification overrides a telephone with Do Not Disturb (DND) activated. Call Completion requests can be applied to telephones with the DND feature activated. However, this request does not advance until the DND feature is deactivated.

Euro ISDN Trunk - Network Side

Call Completion to Busy Subscriber is supported on a EuroISDN Trunk - Network Side connectivity interface.

Call Completion on No Reply is supported on QSIG and DPNSS1 (as Call Back When Next Used) interfaces, corresponding to the MCDN Network Ring Again on No Answer feature. It is not supported on a EuroISDN Trunk - Network Side connectivity interface.

Flexible Feature Codes

Analog (500/2500 type) telephone users can use Flexible Feature Codes (FFCs) to activate Call Completion to Busy Subscriber requests.

Group Call

Call Completion cannot be applied to a Group Call.

Group Hunt

Call Completion to Busy Subscriber cannot be applied to the Pilot DN when no idle telephone is located during a Group Hunt call.

Hot Line

Call Completion cannot be used in conjunction with the Hot Line feature.

Maintenance Busy

Call Completion on Busy Subscriber is not accepted against a telephone in a Maintenance Busy state.

Make Set Busy

Telephones that have Make Set Busy (MSB) activated can request Call Completion to another DN. The free notification overrides the MSB feature. Incoming Call Completion to Busy Subscriber (CCBS) requests do not override the MSB feature. A set is considered busy while MSB is active. A CCBS request is registered against a busy telephone, but only advances when the MSB feature is deactivated and the telephone remains free.

Permanent Hold

Analog (500/2500 type) telephones with Permanent Hold cannot use the Ring Again functionalities.

Ring Again
Ring Again No Answer

Analog (500/2500 type) telephones can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 proprietary telephones can make Ring Again requests based on the number of Ring Again keys programmed on a telephone.

Call Diversion Notification

Feature interactions

Basic Rate Interface

If Basic Rate Interface trunks are used, programming must be done to configure diversion as a remote capability. Only nodes that will be originating, diverting, final destination or rerouting nodes need this configured.

QSIG

Call Party Name Display

The maximum allowed size of the name information for QSIG is 50 characters but only 27 characters are supported by the Meridian 1 CPND feature. If names longer than 27 characters are received, the names will be truncated on displays.

ESN numbering plans

The network must have a UDP or CDP numbering plan without digit manipulation for the Call Diversion Notification to work.

Forward Switching

This is a network routing algorithm which performs a call diversion by joining together the first connection from User A’s node to User B’s node and a second, new connection from User B’s node to User C’s node. This method is fully supported by the Meridian 1. When the Meridian 1 activates QSIG Diversion, the Forward Switching method is always used.

Meridian Mail

The only Meridian Mail functionality that is supported on a QSIG network is message reception and mailbox interrogation. There is no other interworking supported at this time.

Rerouting method

This is a network routing algorithm which performs a call diversion by replacing the connection from User A’s node to User B’s node with another connection from User A’s node to User C’s node. This method is partially supported. It is only supported at reception. The Meridian 1 supports the receipt and treatment of a Rerouting request sent by another vendor PBX, but never generates this request.

Interworking

No interworking/gateway with any other connectivity (such as MCDN or DPNSS for example) is supported at this time.

ANF Path Replacement

Feature interactions

Call Transfer

The QSIG Call Transfer Supplementary Service has not been developed as of Release 23, so it does not act as a trigger for Path Replacement.

Call Waiting

Path Replacement does not take place on a waiting call until it becomes connected to the desired party.

Camp on

Path Replacement does not take place on a camp on call until it becomes connected to the desired party.

Conference

Path Replacement is not invoked for a user involved in an established conference.

Data and fax calls

Path Replacement will not be invoked for data or fax calls to prevent corruption or disruption to the transmission during the Path Replacement.

ESN numbering plans

The network must have a UDP or CDP numbering plan, without digit manipulation being performed at transit switches, for the Path Replacement feature to work.

Japan TTC Common Channel Signaling

Feature interactions

ISDN Signaling Link

Integrated Signaling Link (ISL) operation is supported on the JTTC interface on the JDMI 2 Mbps interface only. Only the first 23 TIE trunks of the JDMI loop are configurable for ISL operation.

Network Call Redirection

When a call is terminated on the Meridian 1 and Network Call Redirection is active, Japan TTC Common Channel Signaling can still operate. However, the original called number and the redirection number Information Elements that are used by the Network Call Redirection feature are not sent on the JTTC interface.

Virtual Network Services (VNS)

A JTTC link can be used as a B-Channel for Virtual Network Service (VNS) over a private network. All VNS services are supported normally. JTTC is used only as a speech bearer.

QSIG Alternate Routing

Feature interactions

Traffic measurements

Traffic measurements on trunk groups in Customer Traffic Option 2 (TFC002) increments the All Trunks Busy (ATB) peg count when the trunks are busy at the congested node. The ATB count in TFC002 increments at a transit node when a rerouting attempt fails due to congestion. Refer to the section in this book called *Network Traffic Studies* for more information.

Network Drop Back Busy and Off-Hook Queuing

The MCDN feature called Network Drop Back Busy and Off-Hook Queuing has a similar functionality to QSIG Alternate Routing. With Network Drop Back Busy, network calls that are blocked at a node are dropped back to the originating node (not just the preceding node). The switch encountering blockage is programmed to drop the call back to the originating node. At a gateway node where both the Network Drop Back Busy and Off-Hook Queuing feature and the QSIG Alternate Routing feature are configured, the QSIG Alternate Routing feature takes precedence over the Network Drop Back Busy and Off-Hook Queuing feature.

Call Detail Recording (CDR)

The QSIG Alternate Routing feature can increase the time it takes for a call to reach its destination. The duration of calls increases when rerouting occurs. This is shown in the CDRs.

What to have ready

The following checklist summarizes the steps you should take before implementing the QSIG feature and/or the related features associated with the basic feature.

Table 56
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Determine the standards that your network must comply with.
✔		Assess the features you need. Determine if QSIG can support what you need.
✔		Decide which users must send names with their calls when they are calling or called.
	✔	If some users want to restrict their names from being sent, assign an FFC. Train users.
	✔	If you want to use QSIG Alternate Routing, decide where the alternate routing will occur – all nodes, no nodes or originating nodes only. If at all nodes, implement NCOSs, billing procedures and trunk provisioning to support the alternate routing for your needs. Decide on the cause values you want to trigger the alternate routing.
		If you are using VPN, determine if you need any of the Business Networking Express (BNE) features. If implementing any of the BNE features, ensure that you fully recognize the interactions that these features have with other features already installed on your system.

— continued —

QSIG

Table 56
Checklist (Continued)

Basic	Optional	Preparation
	✔	Determine which telephones will divert calls to other switches. Ensure the redirection DNs are programmed in a UDP or CDP format. Decide on the calling party and destination telephone display formats desired by each diverting telephone user.
	✔	Decide which sites could benefit from Path Replacement. Investigate what kinds of triggers your sites encounter.
✔		Investigate what gateways and interworking you will need on your network. Find out if these gateways are supported.
✔		Determine what hardware and software you will need.
✔		Arrange training for users and attendants.

National ISDN 2 (NI-2)

National ISDN 2 (NI-2) relates to the North American ISDN Users Forum (NIUF) Implementation Agreement NIU.302, *Layer 3 Signalling Specification for the Minimal Set of Circuit Switched Bearer Services for the ISDN Class 2 Primary Rate Interfaces.*

Basic configuration

NI-2 TR-1268 PRI Basic Call Feature

This capability provides an ISDN PRI interface to Central Offices in North America that conform to portions of the Bellcore Technical Reference TR-NWT-001268, *ISDN Primary Rate Interface Call Control Switching and Signalling Generic Requirements for Class II Equipment.* Requirements in this technical reference are consistent with the NIU.302 agreement.

Features supported (Release 21)

The features supported by Release 21 software are listed below:

- Basic Call Service
- Enbloc sending
- Channel negotiation
- In-band Tone and Announcement
- Backup D-channel
- nB+D (maximum 16 DS1s)
- Calling Number Delivery

The Calling Number Delivery is affected by the Class of Service of non-BRI telephones. Presentation allowed/denied are the choices. BRI telephones use the PRES option in programming to allow or deny presentation.

National ISDN 2 (NI-2)

NI-2 TR-1270 PRI Call by Call Service Selection

This capability, introduced in Release 23, provides a standardized version of Call by Call Service (CBC) over an NI-2 TR-1268 interface, as defined in the Bellcore Technical Reference Specification TR-NWT-001270. NI-2 CBC works with Class 5 local exchanges in North America that comply with the NI-2 TR1270 standard.

If you need more information on Call by Call Service, as provided on MCDN PRI links, refer to page 333 under Integrated Services Access.

Call by Call Service works with a pool of B-channels on a master route within which there are service routes. Instead of configuring dedicated channels on each type of route as you would without CBC Service, you configure a maximum number of channels that each service route can access on the master route. The B-channels are programmed as members of a master route. Service routes do not have any B-channels assigned to them. The service route programming determines the characteristics of the call as it is being routed.

The maximum number of channels for each service route is based on a trunk subscription you have arranged with your service provider. The value assigned must match at the Meridian 1 and the Central Office and it is controlled by the Central Office. There is no minimum number of channels to be reserved for each service route. This requirement was not included in the Bellcore Specification. However, the total of the maximums assigned to all the service routes on one master route can add up to more than the total number of channels on the master route. It is best to assign maximums so that there will be one or more channels available at all times for each service route.

When a call is made, the system assigns a call type identifier to it and looks for an available B-channel on the master route. When a call comes into a Meridian 1 from another switch, the Meridian 1 sees the call identifier and understands how to process the call on the B-channel. If there is no call type identifier, then it is treated as a public network call.

National ISDN 2 (NI-2)

Benefits

If the busy hour traffic on your routes does not happen simultaneously at a particular peak hour of the day, you can benefit from CBC. If one route is busy during times that other routes are not heavily utilized, you can provision fewer total trunks than if you dedicate trunks to each route based on the peak traffic requirements of each route.

OUTWATS

- You can configure only one IntraLATA service route for each CBC master route. The band number is 0. The inter-exchange carrier is 0 (for local provider).
- You can configure bands one to nine, one route for each subscribed band. (Bands 1-99 are part of the feature for possible future requirements). The inter-exchange carrier is 0 (for local provider).
- You can configure only one InterLATA service route for each CBC master route. No band number is required. Specify an inter-exchange carrier different from the local service provider.

INWATS

You can configure only one INWATS service route for each CBC master route.

Public network routes

For each master route, there can be only one incoming public network service route (that is, a route with no service route identifier). The service route can be COT or DID type. If you need more than one incoming public network service route, consider using more than one master route. There can be more than one outgoing public network service route.

National ISDN 2 (NI-2)

Hardware

Non-Option 11 systems

You will need MSDL cards or Dual DTI/PRI (DDP) cards with downloadable D-channel daughterboards for the D-channels.

Option 11

You will need downloadable D-channel daughterboards.

Comparison of different CBC options

Table 57 summarizes the three kinds of CBC Service offered by Nortel Networks to date. Use the information in the chart to choose the configuration that suits your needs best.

National ISDN 2 (NI-2)

Table 57
Comparison of the three Nortel Networks CBC options

ISA service to DMS (Release 12)	CBC service to Lucent 5ESS (Release 16)	NI-2 CBC service (Release 23)
Minimum and maximum number of channels are defined on a per service route basis.	Minimum and maximum number of channels are defined on a per service route basis.	Maximum number of channels defined on a per service route basis. No minimum value is allowed by the Bellcore Spec.
TIEs- senderized	N/A	Rls 23 - TIEs senderized only.
Up to 512 service routes ◆ Private ◆ Public ◆ FX ◆ TIE ◆ INWATS ◆ OUTWATS	Only one service route per service type - up to 9 possible services ◆ ACCUNET ◆ SDN ◆ MEGACOM ◆ MEGACOM 800 ◆ WATB ◆ WATM ◆ LDS ◆ IWAT ◆ I800	Up to 512 service routes Standardized services: ◆ TR-1268 Public (CO or DID) ◆ FX ◆ TIE ◆ OUTWATS: Bands (1-99), 1 IntraLATA and 1 InterLATA ◆ 1 INWATS Non-standardized services: ◆ Access to operator ◆ Access to Exchange Carrier Services (call by call services) ◆ Access to Virtual Private Network ◆ MEGACOM/ MEGACOM 800 ◆ ACCUNET switched digital service ◆ International long distance service ◆ International 800 ◆ Electronic Tandem Network ◆ Private Virtual Network ◆ DIAL IT and Lucent MultiQuest
Proprietary (interface specific to DMS-100 and DMS-250).	Proprietary (interface specific to Lucent 5ESS Local Exchange Carriers).	Bellcore-based standard CBC service (independent of switch type).

National ISDN 2 (NI-2)

What to have ready

The following checklist summarizes the steps you should take before implementing NI-2 networking.

Table 58
Checklist

Basic	Optional	Preparation
✔		Verify what kind of Central Office is connected to your system.
✔		Assess what ISDN features you need on your network.
✔		Decide if you want to implement ISDN PRI based on the Q.931 protocol with a fuller feature complement or if you want to implement NI-2 protocol.
✔		Decide if CBC Service Selection would be of benefit to you. Calculate the number of channels you will need in each service route. Make the subscription arrangements with your service provider.
✔		Decide if you will need backup D-channel capabilities.
✔		Assess your hardware. Find out what equipment you will need.
✔		Decide which telephones need a Call Name Delivery presentation denied Class of Service or PRES setting (BRI).

Virtual Network Services

Virtual Network Services (VNS) allows you to design a multi-site network with advanced applications without the use of TIE trunks. VNS allows networks to use proprietary MCDN features over PSTN trunks rather than dedicated TIE trunks. Examples of these MCDN features are ISDN Calling Line ID, Network Call Transfer, and Network Call Redirection. VNS also supports centralized services such as Network ACD and Network Message Services over PSTN trunks.

VNS allows you to save money in the following ways:

- VNS provides MCDN features and services without the need for leased TIE trunks.
- VNS allows you to be billed based on call usage, rather than being billed at a fixed cost associated with leased TIE trunks.
- VNS allows you to design your network to include carriers that offer the least cost per call.
- VNS allows you to take advantage of the free calling area within your PSTN environment.
- For networks that already have TIE trunks, VNS allows calls to overflow to the PSTN trunks when all of the TIE trunks are busy.

The requirement for VNS is that two Meridian 1 switches are linked by a D-channel, and there is an available Public Switched Telephone Network (PSTN) trunk. The D-channel is used for signaling and features support. The PSTN trunk serves as the B-channel for the duration of the call. The public trunk can be COT, DOD, or TIE for outgoing calls. The public trunk can be DID or TIE for incoming calls. The PSTN trunks remain in use for the duration of the call.

VNS emulates MCDN functionality over TIE trunks. Although PSTN trunks handle the call processing, VNS calls are presented and interact with other features as though private ISDN TIE trunks are being used.

Virtual Network Services

VNS is supported over PRI, DTI, and analog trunks.

Basic configuration

The D-channel used for VNS between the two Meridian 1 systems must be a direct, non-tandemed D-channel. It can be configured for VNS use only, or it can be configured in shared mode with existing PRI/DTI/ISL functionalities.

The public network channels can also be used for regular public network (non-VNS) calls.

Disconnect supervision is mandatory on trunks used to carry VNS calls.

You can use VNS as an overflow alternative when your private network TIE trunks are busy.

VNS requires that VNS DNs (VDNs) be configured as part of the customer numbering plan. Consider the following when you configure the VDNs:

- The assigned VDNs must fall within a valid existing range of unassigned DID numbers.
- Two VDNs are used for each VNS call, one at the originating Meridian 1 and one at the target Meridian 1. The VDN at the originating switch is used for feature control. The VDNs at both the originating and terminating switch are used for call control. The two VDNs do not have to match. Both VDNs are used for the duration of the call.
- VDNs are defined in blocks. Within a block, it is possible to define a contiguous range of VDNs as well as individual VDNs. For example, if you need to configure 164 VDNs, you could define the following VDNs.

  7200-7299
  7320-7325
  7355
  7400-7455
  7676

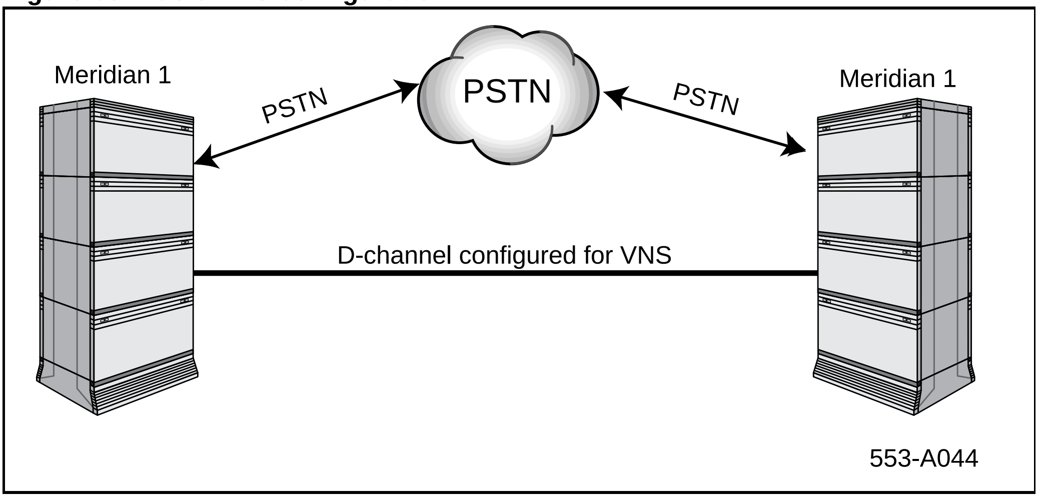
Virtual Network Services

- The total number of VDNs cannot exceed 4000
- If one switch acts as a node, the total number of assigned VDNs at the node must equal the total number of VDNs assigned at all other directly-connected switches

CDR is supported on VNS routes. The CDR output is controlled by the Bearer trunk. That is, the CDR output shows the details of the call made on the Bearer, and not on the VNS D-channel. However, calling party/called party and dialed digit information identifies the actual call participants and not the use of the VNS Virtual DN. The VNS Virtual DN does not appear in the CDR output. The Calling Line Identity is output in the CDR record for incoming calls.

The illustrations that follow depict some typical VNS configurations.

Figure 65- Basic VNS configuration



Virtual Network Services

Figure 66 - VNS can be used to handle overflow from TIE trunks to PSTN trunks

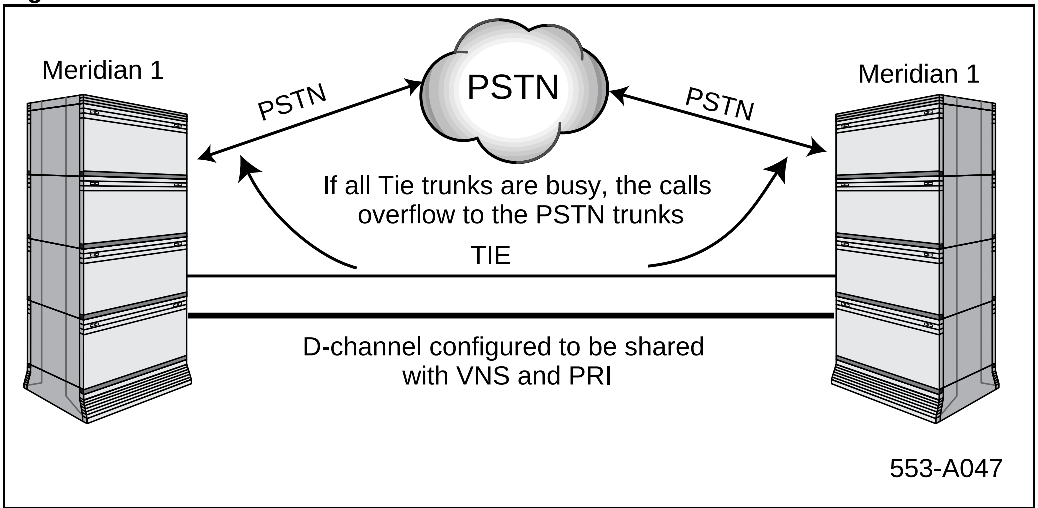
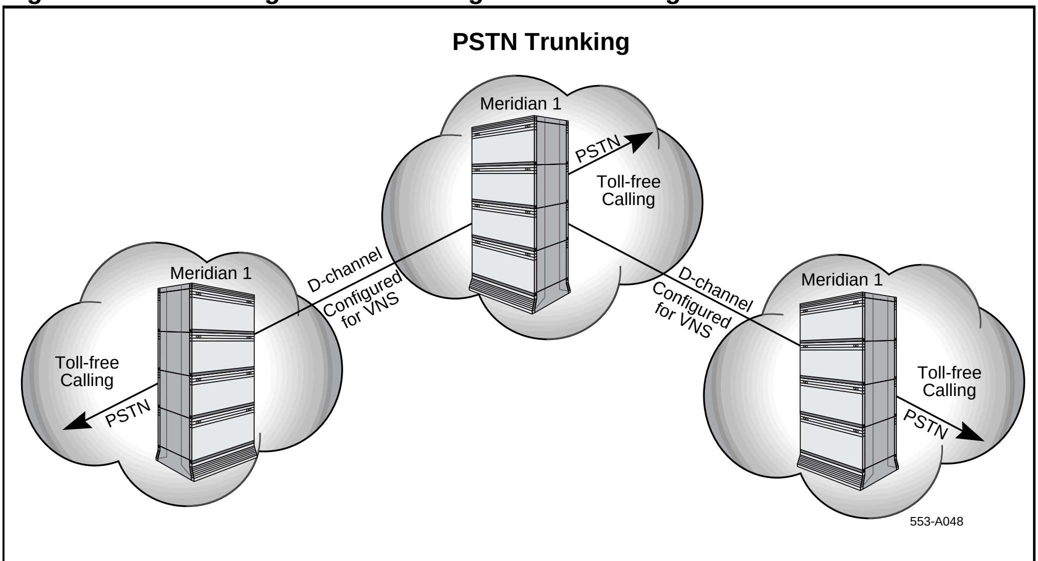


Figure 67- VNS configuration allowing toll-free calling



Basic VNS call processing example

The example that follows provides a basic VNS call processing scenario. In this example, assume the following:

- The D-channel between Meridian 1 A and Meridian 1 B has been configured to support VNS.
- A free VDN is available at each switch, when one is requested by the Meridian 1.
- A free trunk is available to complete the call.

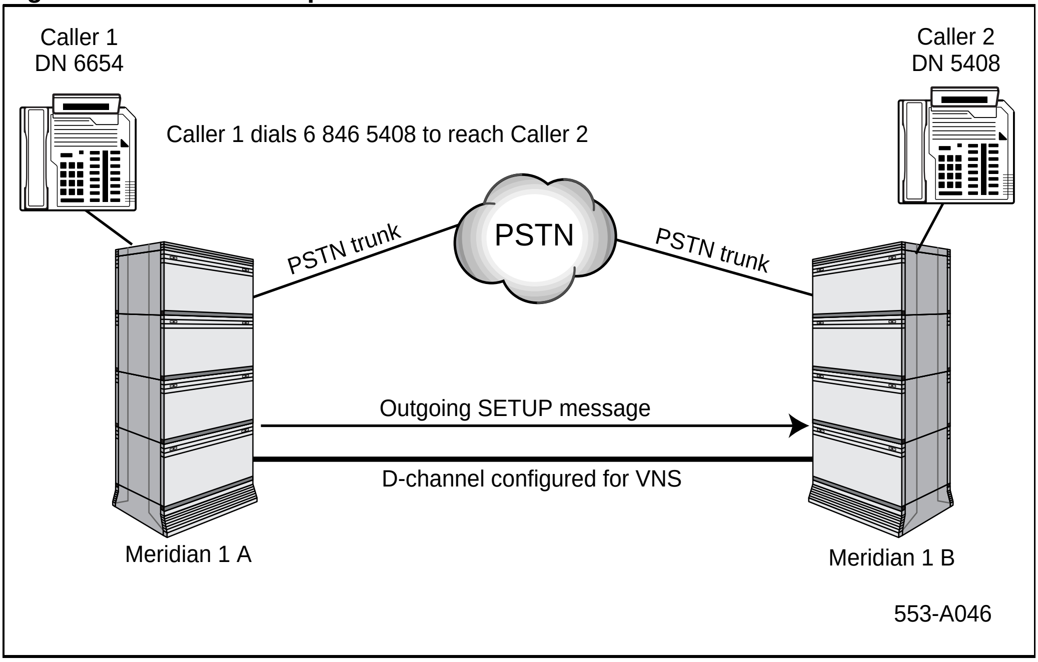
Call flow

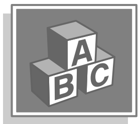
- Caller 1 at the originating Meridian 1 A dials UDP number 6-846-5408 (AC1+LOC+DN) to reach Caller 2 on the terminating Meridian 1 B. Note that a CDP number can also be dialed.
- Meridian 1 A sends an outgoing SETUP message, over the VNS D-channel, to Meridian 1 B. The SETUP message:
  - indicates that an outgoing call is being sent to the telephone with DN 5408
  - requests that a free VDN be assigned at Meridian 1 B
  - requests that this VDN be sent to Meridian 1 A
  - requests that a free VDN be assigned at Meridian 1 A

Refer to Figure 68.

Virtual Network Services

Figure 68- VNS call setup

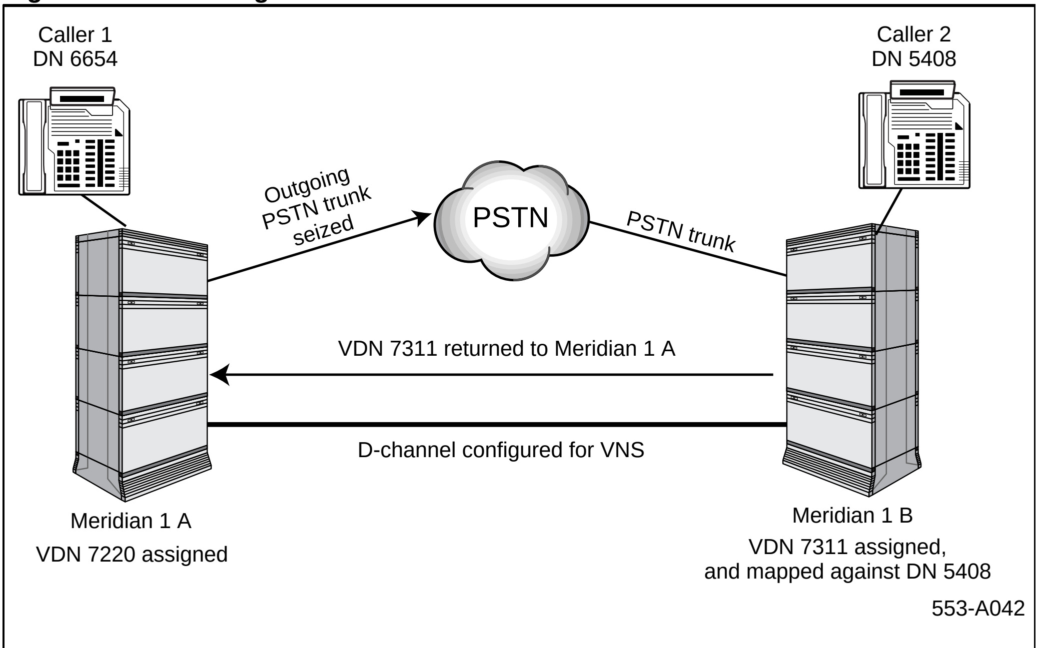




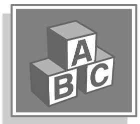
- Meridian 1 B assigns VDN 7311 to the call. This VDN is part of a valid DID group at Meridian 1 B. It is reserved for the duration of the call.
- VDN 7311 is mapped against DN 5408, and stored in memory.
- VDN 7311 is returned to Meridian 1 A, over the VNS D-channel.
- Meridian 1 A is assigned VDN 7220. VDN 7220 is reserved for the duration of the call. This is in case the call is modified (for example, if it is transferred back to Meridian 1 A from Meridian 1 B after the call is presented to Caller B.)
- Meridian 1 A checks the Route List Index for a free outgoing PSTN trunk to the Central Office. If one is found, it is seized. This trunk will remain in use for the duration of the call.

Refer to Figure 69.

Figure 69- VDN assignment



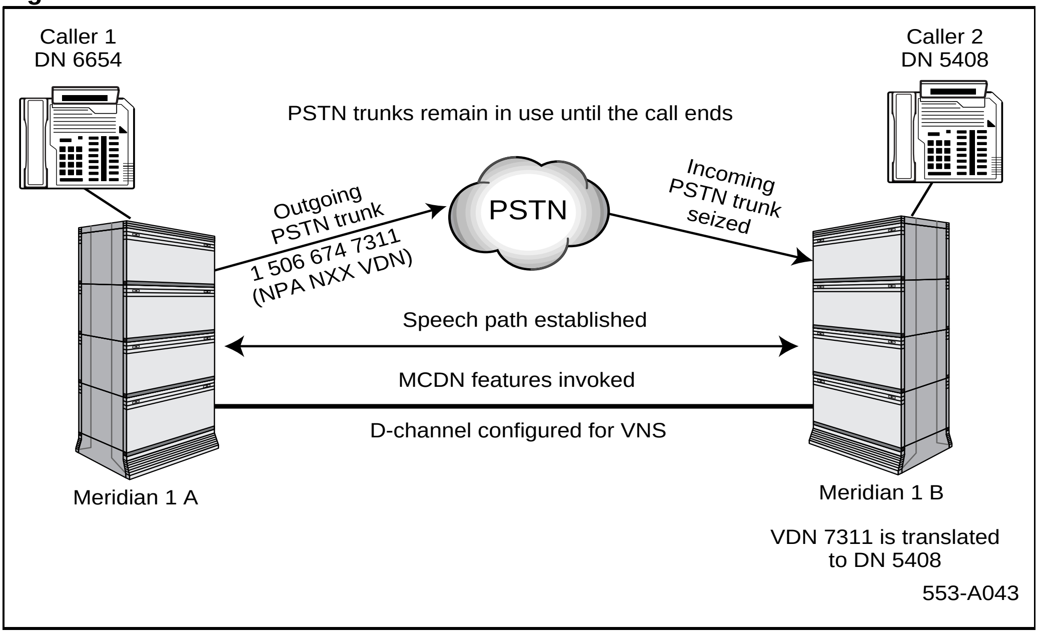
Virtual Network Services



- The appropriate NPA/NXX/VDN is inserted (in the example, 1506-674-7311), and sent out over an outgoing trunk to the Central Office.
- If an incoming PSTN trunk to Meridian 1 B is free, it is seized by the CO. This trunk will remain in use for the duration of the call.
- The CO forwards the call to Meridian 1 B. VDN 7311 is translated to the originally dialed DN (5408), and presented to Caller 2.
- When Caller 2 answers, a speech path is established between Meridian 1 A and Meridian 1 B.
- MCDN features can now be invoked.
- When the call ends, the PSTN trunks are released.

Refer to Figure 70.

Figure 70- Call connection

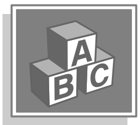


Virtual Network Services

Supported VNS Bearer trunks

- PRI2 connectivities:
  - AXE10 – CO/DID (Sweden, Denmark, Australia)
  - NUMERIS – CO/DID
  - SWISSNET2 – CO/DID
  - SYS12 – CO/DID
  - 1TR6 – CO/DID
  - SL1 – TIE
- PRI connectivities in North America:
  - DMS-100 – CO/DID
  - DMS - 250 – CO/DID
  - AT&T #4 ESS – CO/DID
  - AT&T #5 ESS – CO/DID
  - SL-100 – CO/DID
  - TR1268 – CO/DID
  - SL1 – TIE
- DTI2 connectivities:
  - MFC – CO/DID/TIE
  - DTN – CO/DID/TIE
  - DIP – CO/DID/TIE
- DTI connectivities:
  - both DTMF and DIP signaling
- EuroISDN connectivities:
  - CO/DID including PRI and BRI trunks
- Analog trunks with Disconnect Supervision

Virtual Network Services



Virtual Network Services with DASS2/DPNSS1 Bearers

This feature allows Digital Private Network Signalling System No.1 (DPNSS1) or Digital Access Signalling System No.2 (DASS2) trunks to be used as VNS Bearer trunks. You must use Release 22 or higher software. APNSS trunks cannot be used as bearer trunks. Many MCDN features can be supported with this configuration.

If you want to request a DPNSS1 Supplementary Service (such as Route Optimization, Step Back on Congestion, Call Offer, or Executive Intrusion), you must not configure your route list entries for VNS. You must configure them for DPNSS1 only, since the Meridian 1 will not use any route list entry configured for both DPNSS1 and VNS.

Refer to the *Virtual Network Services* chapter in the *X11 Networking Features and Services* (553-2901-301) NTP for information on the interaction between VNS and DPNSS1 Step Back On Congestion.

Trunks that are not supported

- Autoterminating
- APNSS
- Japan D70
- MFE/MFE Socotel signaling trunks
- CIS DTI2
- KD3 signaling trunks

Virtual Network Services

Supported features

- Calling Line Identification
- Enbloc sending on D-channel; Overlap sending and receiving on B-channels
- Network Attendant Service
- Network ACD
- Network Call Party Name Display
- Network Call Redirection
- Network Distinctive Ringing
- Network Message Service to Meridian Mail
- Network Ring Again
- Network Ring Again No Answer
- Trunk Route Optimization Before Answer
- Trunk Anti-Tromboning
- Network Signaling (Release 23)

Features or capabilities that are not supported

- backup D-channel
- D-channel cannot be a BRI D-channel, (however, the COT trunk B-channel can be a BRI B-channel on NUMERIS, 1TR6 and MCDN networks)

VNS and NARS

The NARS access code AC1 is automatically inserted to the incoming portion of the VNS call on the VNS D-channel, whatever the INAC option is for the bearer route. Therefore, the VNS digit manipulation index used on the D-channel must be performed so the terminating switch translates the call set-up message properly. The B-channel call is sent from the originating switch after the route list entry digit manipulation index has been used so digits are sent that are appropriate for the public network.

Virtual Network Services

When the route to be used for VNS calls is programmed in a route list as an entry:

- the VNS feature must be activated for the route list entry
- the D-channel to be used for signaling is programmed for the route list entry
- two digit manipulation indexes (DMIs) must be associated with it:
  - one (called VDMI) is for the D-channel messages.

    This DMI typically inserts the NPA and NXX of the target Meridian 1, for UDP calls. The VDN number (which is a DID number at the target Meridian 1) is automatically inserted after the DMI digits. If using an equal access carrier, you must add carrier access codes to the VDMI table. The carrier access codes must be inserted ahead of the NPA and NXX.

    If a CDP number is dialed, the call type may need to be changed. NARS will be needed at the other end to translate the digits properly.
  - the other DMI is for the digits in the calls on the B-channel
- set the number of outgoing trunks that will allow simultaneous VNS calls on that route. This setting prevents VNS calls from occupying all the trunks in a route.

If Incoming Digit Conversion is active, you must not include the VDN digits in your conversion table. The VDNs must be forwarded unchanged to be translated to the appropriate DNs.

Provisioning guidelines

Assigning VDNs

The number of VDNs required for a VNS application is a function of call traffic. For each VNS call, one trunk and one VDN is used at the originating Meridian 1, and one trunk and one VDN is used at the terminating Meridian 1. Both VDNs and both trunks are tied up for the duration of the call.

Configuring VNS trunks

VNS Bearer trunks are engineered using the same guidelines as any other trunk.

Outgoing trunks can be DOD, COT, or TIE. Incoming trunks can be DID or TIE.

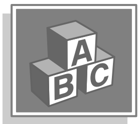
The PSTN trunks configured for VNS are usually configured to be both incoming and outgoing. If VNS is configured to be unidirectional (outgoing only, or incoming only), ensure that call blocking does not occur.

Remember that the PSTN trunks over which VNS calls are made can be shared with other types of PSTN calls. Ensure that you provision the PSTN trunks for VNS to provide a good grade of service. It is important that you properly assess traffic requirements so that you maintain adequate trunks for VNS calls and PSTN calls.

If an outgoing trunk is available, VNS tries to route the call to the target Meridian 1 over the outgoing trunk. If none is available, the VNS call fails.

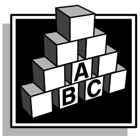
Consider, as an example, a two-node network. The incoming and outgoing VNS traffic between the two Meridian 1s is equal. Traffic projections indicate that ten trunks are required. In this case, each Meridian 1 should have ten VDNs that are valid DID numbers within the PSTN numbering plan. This is required to support two way connectivity, since VDNs are part of a shared pool that are dynamically, non-directly allocated to a PSTN trunk.

Virtual Network Services



Assuming that the outgoing and incoming calls are to be equally balanced 50/50, then five DOD/COT/TIE trunks are required for the outgoing calls on each Meridian 1, and five DID/TIE trunks are required for incoming calls. The VNS D-channel is programmed to support ten VNS calls (the total number VDNs on each Meridian 1) and the RLI entries for each Meridian 1 is set to support five trunks (outgoing). VNS is configured for non blocking treatment. Outgoing/incoming trunks can exist in larger shared trunks groups, that is, DID groups for public incoming calls. Dedicated outgoing/incoming trunk groups can be arranged with the Central Office, if required.

Improving feature performance



The information that follows makes you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Authorization codes

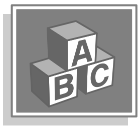
When using AUTH CODES at the Central Office level, the DMI in the RLB can insert the AUTH CODE ahead of the public number. This configuration must be negotiated with Central Office.

Revert to conventional signaling

There is no backup D-channel when the VNS D-channel fails. If the D-channel fails, calls go out on the trunk route using conventional signaling. Make sure you have special digit manipulation indexes programmed for this case in your route lists.

If the D-channel fails, VNS calls will not be processed until the D-channel is restored.

Virtual Network Services



Advanced networking features over VNS routes and the speed of the D-channel

For basic call service, a 9600 baud circuit should be adequate. However, advanced networking features, such as NACD and Meridian Mail, require higher D-channel bandwidth. This means that the VNS D-channel has to be configured at a sufficient speed to handle the projected traffic.

For the number of B-channels that can be supported according to modem baud rate, refer to the *ISL* section in the *MSDL* chapter, in the NTP *Capacity Engineering* (553-3001-149).

Control tips

- If you want users to be restricted when they try to access VNS trunks, use the FRL to do it. Set the FRL on the VNS routes in route lists to be higher than the user's FRL.

Administration tips

- Even though VNS calls come in on DID B-channels, they are treated as if they are TIE trunk calls. Therefore calls are given treatments such as Intercepts as if they came in on TIE trunks.

Training tips

- Users can notice a time delay when a private network call uses VNS trunks compared to private ISDN trunks. Explain to the users that the delay is expected due to the public network routing of the call. Tell them not to report the delay as a problem.

Virtual Network Services

What to have ready

The following checklist summarizes the steps you should take before implementing the basic VNS feature and/or the optional related features associated with the basic feature.

Table 59
Checklist

Basic	Optional	Preparation
✔		Ensure disconnect supervision is programmed on the public network trunks.
✔		Ensure the D-channel is not tandemed.
✔		Make sure you do not require features and capabilities the VNS trunks do not support.
✔		Ensure you have NARS or CDP programmed at both Meridian 1 systems.
✔		Ensure that you have programmed VDNs as part of your numbering plan.
✔		Determine the number of VDNs that you need to configure on your system, keeping in mind that the maximum number is 4000. The number of VDNs required depends on call traffic. Utilize traffic reports when making your analysis.
✔		Decide how many trunks you want used in the route for VNS calls.
✔		Confirm which users are to be restricted from the VNS route. Program the FRL accordingly.
✔		Decide if you need the NSIG functionality on the VNS route.
	✔	If the VNS D-channel is to be shared with the D-channel of a PRI that links two Meridian 1 switches, configure it in shared mode.
	✔	If you are using advanced networking features such as NACD over VNS, ensure that the speed of the D-channel is sufficient.
✔		Prepare training.

Authorization codes

Purpose

There are two authorization code software packages upon which to build further enhancements:

- Basic Authorization Code
- Network Authorization Code

The enhancements are:

- Station Specific Authcode
- Authcode security enhancements
- Direct Private Network Access

Authorization codes are used to identify and control users.

When a user makes a call from another user’s telephone, the access restrictions of that telephone control the user. If you want a user to be able to make calls from any telephone and have individual access restrictions apply, assign an authcode to the user. The user enters the code when making calls and the restrictions associated with the code apply to the calls.

The CDR records identify both the telephone used for the call and the authcode the user entered. You can bill calls to the users who made them, not to the telephones that were used.

Basic feature configuration

The Authorization Code features allow selected users to temporarily override the access restrictions assigned to a station or trunk. A user enters an authorization code (authcode) to access the system facilities based on the assigned Network Class of Service (NCOS), Class of Service (CLS), and Trunk Group Access Restriction (TGAR) of the authcode.

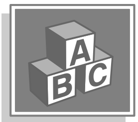
Typically, if you plan to enforce the use of authcodes for all calls, you set up the telephones with very restricted capabilities. Only through the use of an authcode will restrictions be removed, to the degree that is appropriate for that user. As an exception, you may assign unrestricted capabilities to executives' telephones or telephones in locked offices so these users will not be forced to dial authcodes.

Authcode validation

The system validates the authcode dialed. If a user dials an invalid code or does not dial a code, there is a pause equal to the interdigit time-out before 15 seconds of overflow tone is given; the call is then force disconnected. This is designed so the user does not know if the code dialed was of an incorrect length or if the digits themselves were invalid. This is a security feature built into the design of the authcode software.

Authorization codes

Basic Authorization Codes (BAUT)



This software package was designed for stand alone switch applications. It was not designed for network applications.

The Basic Authorization Code (BAUT) package provides up to 4096 authcodes of 1 to 14 digits. Users can enter an authcode after dialing the Special Service Prefix code (SPRE) and the digit “6,” before dialing any call, including a NARS or BARS call. With the BAUT package, an authcode can be entered when:

- originating a call from a local station
- initiating a call transfer or conference from a local station
- originating a call by means of the Direct Inward System Access (DISA) feature

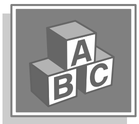
If a user calling into a BAUT switch on a TIE trunk attempts to use a BAUT authcode, the Uniform Dialing Plan no longer works. The user would have to dial: the TIE trunk route access code, followed by the SPRE + 6 + authcode + access code for the call at the node + the remaining digits in the call.

Network Authorization Codes (NAUT)

The Network Authorization Codes (NAUT) package was designed to be used in a network application. It provides for up to 20,000 authcodes of 1 to 14 digits. The NAUT package incorporates all the features of the Basic Authorization Code (BAUT) package and adds:

- the attendant is allowed to enter an authcode
- a “conditionally last” option for entering an authcode after dialing a NARS call
- network access from remote sites, maintaining the Uniform Dialing Plan
- automatic generation of codes at installation

Authorization codes



Authcode conditionally last option

With NAUT, users dial NARS calls normally. If the NCOS of the telephone does not have a high enough FRL, the system will prompt the user to input an authcode (without the SPRE code). If the user’s authcode has an NCOS with a high enough FRL, the call is allowed to proceed.

Users are also allowed to dial the authcode (with a SPRE code) first.

Prompt for authcode

This tone is composed of ten short bursts of dial tone followed by steady dial tone. Optionally you can implement a recorded announcement (RAN) that will play ahead of the tone.

Authcode conditionally last (ACCL) Enhancements

In X11 Release 24, you can configure tones of your choice for the prompt for authcode. You can determine the number of tones and their cadence. These tones still precede steady dial tone. This enhancement also allows you to input a code at any time during the recorded announcement (RAN). The user does not have to wait until the RAN is finished before inputting the code.

How it works

In a network application, the user at the remote switch dials a TIE trunk route access code followed by the remaining digits in the NARS call.

You set up the TIE trunk route so the NARS access code is automatically inserted for incoming calls. If the FRL of the TIE trunk (or the user, if you have travelling NCOSs implemented) is not high enough, the NARS node gives the user the conditionally last treatment. This forces the user to input an authcode to raise the FRL high enough, hopefully, to make the call.

This authcode will also print out on CDR records at the node, identifying the user who is making the call.

You must activate the authcode feature on the TIE trunk route at the node for users to be able to dial authcodes conditionally last.

Minimum FRL

The route lists at a node equipped with NAUT software will have a minimum FRL programmed. This is the lowest FRL that a user must have in order to access any route entry in the route list. This is the first parameter that the system with NAUT will check before it scans any entries in the route list. If the user's FRL is not high enough, the prompt tone will be given.

Attendant input of authcode

Normally, because an attendant is not restricted from accessing any system resource, the attendant does not need to have an authcode. The Network Authorization Code package enables the attendant to enter an authcode for other callers. For example, the attendant can enter an authcode, after dialing the Special Service Prefix code (SPRE) and the digit "6," and complete a long distance call for a local user who is very restricted. If the Call Detail Recording (CDR) of authcodes is defined for the customer, the local user's authcode digits appear in the CDR records for billing purposes.

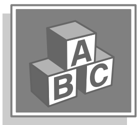
Attendants are normally assigned an NCOS having a high FRL so that they can make any type of call, including NARS calls. However, an attendant can be prompted for an authcode entry if the FRL required to access a route list for a NARS call is greater than the FRL of the attendant's NCOS.

Authcode administration

With the NAUT and BAUT packages a "classcode" structure is used as part of authcode administration. Each authcode is associated with a classcode.

A classcode is a definition of a combination of a CLS, Trunk Group Access Restriction level (TGAR), and an NCOS value. There can be up to 116 (0 to 115) classcodes defined in the Authcode Data Block, each having a different combination of CLS, TGAR, and NCOS codes. Authcodes that have the same combination of CLS, TGAR, and NCOS codes should be assigned the same classcode.

Authorization codes



Automatic generation

With the NAUT package, authcodes can be defined individually or generated automatically by the system. Before there are any authcodes in the system database you can select the number of codes you need in each classcode and the system will automatically print them out. This can save you a great deal of time at installation. Every system that generates the same number of codes, generates identical codes.

Exempting an authcode

When an authcode is to be removed from use, a facility exists make the authcode inactive. This is accomplished through an “exemptcode.”

When an authcode is exempted, an exemptcode is assigned to the authcode in place of the classcode. The exemptcode is the month, (for example, JAN, FEB), taken from the system clock.

You can put an exempted code back into service by assigning it a regular classcode.

You can also decide to remove an authcode from the database completely. You can use it again in the future if you wish to reassign it.

Call Detail Recording

If the Call Detail Recording (CDR) of authcodes is activated, then each time an authcode is entered, a record is generated on the CDR device. The record is passed to CDR only if one of the following occurs:

- the call becomes established, for example, a trunk is seized or a local telephone answers
- the call cannot be completed, for example, no trunks were available and the authcode was input by means of a TIE trunk

If your network is equipped with ISDN and CCDR (CLID in CDR) you can use the CLID that is transmitted with calls to identify the user. If users at remote sites do not always use their own telephones you may still want to use authcodes.

Authorization codes

Station Specific Authcode

With this feature, telephones can be programmed not to allow any authcode to be entered or to allow all authcodes to be entered or to allow only specific authcodes (up to six codes) to be entered.

With this last capability activated, you can specify the specific authcodes that will be accepted at a telephone when you program it.

Authcode security enhancements

When a user attempts to use an invalid authcode, this enhancement causes an alarm to print out on the TTY. The alarm shows the terminal being used. If alarms are being monitored on your system you can catch users in the act of attempting to make these fraudulent calls.

Direct Inward System Access

Direct Inward System Access (DISA) allows an external caller using a Digitone telephone to access a Meridian 1 and make calls on the system as if using an internal telephone. The user dials the telephone number assigned to the trunk configured as a DISA port and then dials the rest of the digits in the call.

If a caller makes a NARS (or BARS or CDP) call using a DISA port, the NCOS associated with the DISA Terminal Number (TN) is used for NARS route selection. If the FRL of this NCOS is too low to access the route list that NARS has selected for the call, the caller will be prompted for an authcode entry, unless an authcode was entered previously, on systems equipped with NAUT software.

Alternately, you can configure the DISA port to expect an authcode for every call, without the user dialing SPRE code first. The first digits dialed before the called number digits are then automatically treated as authcode digits.

Direct Private Number Access (DPNA)

DPNA addresses the application where the Meridian 1 is used as a front end concentrator for the resale of long distance service to International clients. The subscriber dials a DISA number and then calls out on the long distance trunks installed at the Meridian 1. The client calls out at a reduced rate compared to the long distance service they could install at their own site. The reseller is taking advantage of volume discounts they get by aggregating many clients' calls. They pass on some of the discount they receive to their clients.

DPNA offers three related functionalities:

- DISA Digit Insertion
- DISA RAN
- Authcode-last – Retry

DISA Digit Insertion

1-31 digits are automatically inserted after the DISA DN is accessed. This can be used so the subscriber does not have to dial the DISA password, if any, and an authcode themselves. It can also insert common carrier access codes or even the digits in a call.

DISA RAN

The caller can be greeted with a recorded announcement when accessing the DISA service.

Authcode-last–Retry

The caller is reprompted for an authcode if the first authcode entered was invalid. If this option is activated for the Customer Group, then all authcode -last attempts will be affected, not just the DISA attempts.

Improving feature performance

The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Network requirements

Users on a PBX or Centrex system connected by TIE trunks to an ESN node can use the Authcode Conditionally Last feature, provided that these systems transmit or repeat all digits dialed by the users in response to the authcode request. This feature cannot be used by certain systems that operate in senderized mode. Correct operation may require adjustment of EOD timeout on systems that employ

Authorization codes

simulated cut-through operation. End-to-End Signaling software is used by a Meridian 1 main to outpulse Authorization Code digits to a node, once the call is established or in response to a prompt for authcode.

In an ESN network consisting of multiple switches equipped with the NAUT package, authcodes should be requested only once on a given call. This requires careful engineering of:

- the TIE trunk group option for authcode prompting
- the FRL values assigned to route lists and incoming trunk routes and users (if travelling NCOS has been installed)

Speed Call and Autodial

Authcodes can be stored as Speed Call or Autodial entries. When this is done, the stored number (entry) must contain the SPRE code plus 6 and authcode digits first, before the rest of the digits in the call.

You can also store SPRE plus 6 plus an authcode with no other digits. This presents a security risk since other users can make calls using the stored authcode on the telephone.

All digits in the entry after the access code are interpreted as authcode digits. In the case of Authcode Conditionally Last, authcodes can be stored as Autodial entries, but not Speed Call entries. If necessary, the caller can continue to enter more authcode digits after operation of the Autodial or Speed Call key. However, for security reasons, authcodes should not be stored as Autodial or Speed Call entries.

Authcodes and conference calls

In X11 Release 23 (issue 37 and later), there is an enhancement that allows an authcode prompt to be given to a user for each leg of a conference call. With this feature enabled in the Customer Data Block, each external call made using the conference feature is assessed for authcode requirements. The user must input an authcode to set up each leg of the conference, if the prompt tone is given. One CDR A-record is produced each time an authcode is input. With the feature deactivated, the user is only prompted for the first leg of the conference call. Only one A-record is produced for the conference call.

Authorization codes

Control tips

- If you intend to use DISA ports, try to do everything possible to maintain a high level of security on the port.
  - change the password frequently at irregular intervals
  - have passwords of 6 or more digits
  - reassign authcodes occasionally
  - assign a very restricted NCOS, TGAR and CLS
  - monitor CDR closely
  - pay attention to very long calls and investigate
- When codes are automatically generated, the same codes are produced on every switch when the same number of codes is requested. Users from other systems that had authcodes may attempt to use a code they are familiar with from the other system. It is important that you take unused codes out of service and pay attention to user complaints about fraudulent use of codes. Refer to the *Maintenance tips* on handling unused codes.

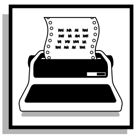
Administration tips

- When a user enters an authcode, the TGAR, class of service and NCOS assigned to the authcode apply to the call even if these restrictions reduce the capabilities of the telephone. For example, if a user makes a call from a telephone that has a TGAR 0, class of service UNR and NCOS 7, if the authcode is TGAR 1, class of service CTD and NCOS 0, the telephone will become more restricted based on the programming associated with the authcode.
- Be sure to generate enough codes at installation to handle the number of users you will have using the codes over the next few years. Exempt the codes you have not assigned yet to prevent unauthorized users from using them.

Training tips

- The standard prompt for authcode is ten short bursts of dial tone followed by steady dial tone. Tell users that it sounds almost the same as overflow tone at first except overflow tone does not end with steady dial tone. Allow users to hear the tone during training sessions to familiarize them with it. If users have difficulty identifying the tone, consider using X11 Release 24 software to configure tones that your users will understand better.

Maintenance tips



- Take unused authcodes out of service by removing them completely or exempting them temporarily. Users often experiment and try to make calls using authcodes that have not been assigned or codes that have been assigned to other users. Monitor CDR records carefully for signs of abuse, especially when the originator is a DISA port.

What to have ready

The following checklist summarizes the steps you should take before implementing the authcode-related features.

Table 60
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Determine if users need to make calls from telephones other than their own.
✔		Do you bill individual users for calls?
✔		Decide if you need to control the calls that each user can make, no matter which telephone they use.
— continued —		

Authorization codes

Table 60
Checklist (Continued)

Basic	Optional	Preparation
✔		Decide on BAUT (single site) or NAUT (single site or network).
✔		Decide what restrictions to place on telephones.
	✔	Decide if you want to implement a RAN as the prompt for authcode.
✔		Decide how many classcodes you need for authcodes.
✔		Verify if the MFRLs on the route lists give you the control you want. If not, adjust the FRLs on the route lists or the telephones so the proper call controls are in place.
✔		Decide if you want to use Station Specific Authcodes.
	✔	Decide if you want to configure a unique prompt for authcode tone and have an interruptible RAN.
✔		Make sure your incoming network trunks are properly configured to control calls from remote sites.
	✔	Decide if you need DISA and how you want to use authcodes for DISA calls.
✔		Prepare your CDR and your plans to monitor CDR regularly.
✔		Prepare training sessions.

Network Signaling (NSIG)

When the Network Signaling (NSIG) feature is equipped on a Meridian 1, you can define the signaling arrangements between that Meridian 1 and any other switch connected to it by means of TIE trunks. These signaling arrangements define what call information is to be transmitted to a connected switch. If the connected switch is also equipped with NSIG software, you can prepare your switch for information that will be received.

Installing NSIG at a Conventional main enhances it. It then becomes an ESN main. When callers at an ESN main place calls through a node with NSIG, their NCOS or TCOS travel with the call and are interpreted at other NSIG equipped switches. The TIE trunk settings determine and control the operation of this feature.

In order for NSIG and the features that it supports to work, the software must be equipped at both connected switches.

The signaling arrangement selected depends on the type of connected switch (ESN node, ESN main, Conventional main, ETN switch) and the features required.

Features supported by NSIG are:

- traveling NCOS
- TCOS (traveling FRL)
- TCM (traveling class mark)
- Network Call Transfer
- Coordinated Callback Queuing
- Satellite Link Control
- Transparent Data Networking

Network Signaling (NSIG)

Basic feature configuration

NSIG software can be installed as the only ESN software package at a Meridian 1 or it can be installed along with other packages that require it in order to work. NSIG can be supported on ISDN networks, not just ESN networks.

Signaling options (SIGO)

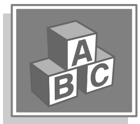
The signaling options (SIGO) you can program on the TIE trunk route are STD (standard), ESN, ESN2, ESN3, ESN5, ETN (Electronic Tie Network), and EN19.

- STD arranges a TIE trunk group for transmission/reception of the called number only between switches.
- ESN (X11 Release 2 only) arranges a TIE trunk group for transmission/reception of the call type, NCOS/TCOS, and called number between switches, and is required on systems equipped with the CCBQ/CBQCM feature. It sends call type, NCOS or TCOS, and dialed digits.
- ESN2 (X11 Release 3 and later) arranges a TIE trunk group as described for ESN in X11 Release 2. It is used unless the switch has NXFER or Satellite Link Control (SAT).
- ESN3 (X11 Release 3) arranges a TIE trunk group as described for ESN2 and is required on systems equipped with the NXFER or Satellite Link Control features.
- ESN5 (X11 Release 5 and later) arranges a TIE trunk group as described for ESN3; it is needed with Digital Trunk Interface.
- ETN arranges a TIE trunk group for transmission/reception of the called number and a Travelling Class Mark (TCM) between switches. ETN is an AT&T specific protocol. It is used when a Meridian 1 is connected to an ETN switch. It sends outpulsed digits and a TCM.

  Note: Traveling Class Mark (TCM) is another name for traveling class of service (TCOS) or traveling FRL.

- EN19 arranges a TIE trunk for Transparent Data Networking

Network Signaling (NSIG)



Travelling Class of Service (TCOS)

Network Control (NCTL) at an ESN node can provide a Traveling Class of Service (TCOS) mechanism that controls route access and Off-Hook Queuing (OHQ) eligibility for calls placed to or through another ESN node or an associated ESN main. It enables the ESN node to interface with switches that are part of an Electronic Tandem Network (ETN), provided that the SIGO setting on the TIE trunk is set for ETN at both ends.

The traveling class of service (TCOS) is, in effect, the FRL of a user's assigned NCOS. When a user at an ESN node initiates a call to another ESN node or an ESN main, the TCOS (the FRL of the user's assigned NCOS) is transmitted to the other ESN node. At the receiving ESN node, the TCOS (0 to 7) replaces the FRL of the NCOS assigned to the incoming trunk group.

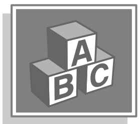
Route access and OHQ eligibility for the call are, therefore, based on the NCOS of the incoming trunk group with the modified FRL (the TCOS).

ETN switch compatibility

The TCOS is equivalent to the Traveling Class Mark (TCM) used at Electronic Tandem Network (ETN) switches. When a ten-digit UDP call or Distance Steering Code (DSC) Coordinated Dialing Plan (CDP) call is made from an ESN node to an Electronic Tandem Network (ETN) switch, the dialed digits, together with the TCOS number (0 to 7), are sent to the connected ETN switch. At the ETN switch, the TCOS number received from the ESN node is used as a Traveling Class Mark (TCM) to determine route access and OHQ eligibility at the ETN switch.

Similarly, when a call is made from an ETN switch to an ESN node, the dialed digits, together with the Traveling Class Mark (TCM) number (0 to 7), are sent to the connected ESN node. The ESN node interprets the received TCM number as a TCOS number. The received TCM (for example, TCOS) replaces the FRL of the NCOS assigned to the incoming trunk group from the ETN switch. This new FRL (for example, TCM) is then used to determine route access and OHQ eligibility for the call.

Network Signaling (NSIG)



Transparent Data Networking (TDN)

This feature, introduced in Release 19, provides a transparent data channel across a network for data modules to perform end-to-end protocol exchange.

Providing a transparent data channel is essential to users who make data calls using Meridian Modular telephone data units. TDN also allows PSDS (Public Switched Data Service) calls to tandem across a private network before terminating on the public network. In this way, a data call uses the private network as much as possible, usually the most economical route, before going onto the public network for a PSDS call.

ESN19 signalling contains a call type that identifies a Transparent Data Networking (TDN) call.

Prior to Release 19, the Meridian 1 X11 software restricts all data calls to be established in a point-to-point protocol exchange. Before the introduction of MCA (Meridian Communications Adapter), MCU (Meridian Communications Unit), and BRI data modules, all the data modules on the Meridian 1 communicated using DM-DM protocol, a proprietary NT data protocol (with the exception of the QMT21-C which could communicate in PSDS). The data modules on other Nortel Netorks switches and those of other vendors use T-LINK or PSDS.

TDN accommodates two types of data calls from modules which use protocols other than DM-DM:

- a data call remaining within the private network is treated as a TDN call type. This call type travels across TIE routes using the ESN19 signaling and protocol converters are turned off.
- a non-DM-DM data call, coming in from the public network or one that hops off a private network to the public network may use TDN routes. These TDN routes are DID trunks which use the standard signaling but allow data calls to go through without protocol converters getting in the way.

Network requirements

You can connect an ESN main to only one ESN node. Both Meridian 1 switches must have NSIG to support NSIG-related features.

You must arrange TIE trunks between ESN nodes and ESN mains with answer supervision, Dual Tone Multi-Frequency (DTMF) sending/receiving, and wink-start operation. If the signaling option is ETN or STD, the TIE route can be digitone or DIP.

ESN node compatibility with ETN switches is limited to 7-digit on-network, 10-digit off-network, and Distance Steering Code (DSC) CDP calls.

Network configurations using NSIG

NSIG only

If two connected switches have only NSIG equipped, there will be a traveling NCOS capability. This means that the NCOS assigned to the user will travel as extra signaling digits to the next switch. The receiving switch will handle these digits as required if the signaling arrangements have been properly programmed on the TIE trunk route.

Example 1

If SIGO is STD and the TIES are DTI (programmed for voice or data) a leading digit is outpulsed before each call to identify to the other end whether this is a data call or voice call (1 = voice call, 2 = data call).

> ***Note:*** This is why you install NSIG at both switches, so the other end is expecting the extra digit.

Example:

Caller at the main dials: 83332202

With no NSIG installed, the main outpulses: 83332202

With NSIG installed, the main outpulses: 183332202 (voice call)

Network Signaling (NSIG)

NSIG and Satellite Link Control (SAT)

Tandem trunk calls, when connected through more than one communications satellite trunk, are subject to transmission distortion due to propagation to and from communications satellites. The Satellite Link Control (SAT) feature ensures that the configuration of a call does not include more than one communications satellite trunk.

This feature applies to ESN network calls (NARS/BARS/CDP) only.

You must program any routes that receive digits from satellites or send digits to satellites as SATELLITE routes.

NSIG and TCOS

Example 2

If SIGO is ETN and the TIES are programmed for voice only, then a trailing digit is outpulsed after each call that identifies to the other end the TCOS of the caller.

This is why you install NSIG at both ends and the response to the SIGO prompt should be the same at both ends or the trailing digit will be ignored.

Example:

Caller at the main dials: 83332202

With no NSIG, the main outpulses: 2202

With NSIG, and if the caller’s NCOS is 0 (and if NCOS 0 has FRL 0 at the main), the main outpulses: 22020

Example 3

If SIGO is ETN and the TIES are DTI (programmed for voice or data):

- a leading digit is outpulsed before each call to identify to the other end whether this is a data call or voice call
- a trailing digit is outpulsed after each call to identify the TCOS of the caller to the other end

Data terminal dials: 83332202

With no NSIG, the main outpulses: 2202

With NSIG, if the calling data terminal is NCOS 3 (and if NCOS 3 has FRL 3), the main outpulses: 222023

NSIG and Coordinated Callback Queuing (CCBQ)

There is more information on CCBQ in the *Queuing* section of this book.

If SIGO is ESN2, then traveling NCOS, TCOS and CCBQ are supported. Different digits are outpulsed, depending on the circumstances of the call. The leading digits can be:

- 1XX
  - the digit 1 tells the other switch that the next two digits represent a traveling NCOS
  - the XX digits can be 00-99
- 2X
  - the digit 2 tells the other switch that the next digit represents a traveling FRL or TCOS
  - the X digit can be 0-7
- 31XX
  - the digit 3 tells the other switch that CBQ can be offered, if blocked trunks are encountered at the other end

Network Signaling (NSIG)

– the digit 1 tells the other switch that the next two digits represent a traveling NCOS
– XX digits can be 00-99

Example 4

Conditions: The TIE trunk route has been programmed for ESN2 and the MCBQ package (for CCBQ and CBQCM) is not installed or the programming to activate CBQ has not been done.

The caller dials: 84446024

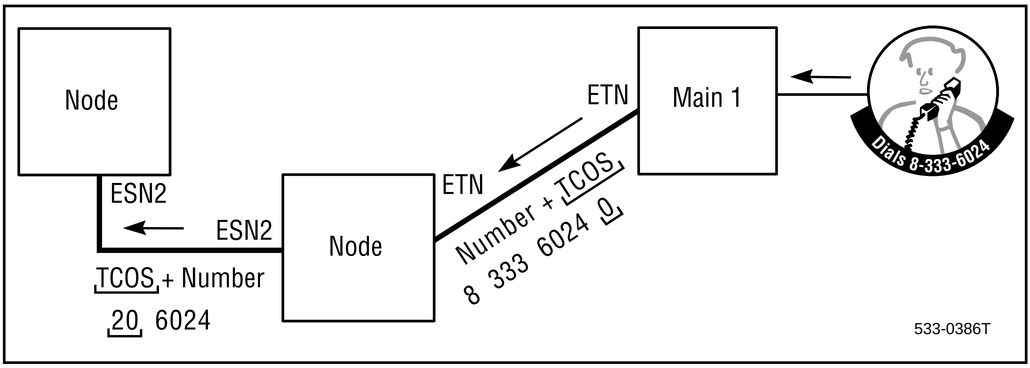
The caller's NCOS is 0

With no NSIG, the main outpulses: 84446024

With NSIG, the main outpulses: 1008446024

Example 5

Figure 71- ETN and ESN2 interaction



As illustrated in Figure 71, there are three switches in the scenario:

Main 1 is possibly a non-Nortel Networks switch capable of sending TCOS or TCM to the node. The TIE route is programmed for ETN at Main 1 and the node.

The node-to-node-TIE route is programmed for ESN2 at both ends.

The caller at Main 1 tries to call the second node through the first node. The user dials: 83336024.

Network Signaling (NSIG)

The caller’s NCOS is 0 (and FRL or TCM = 0)

With no NSIG, Main 1 outpulses: 84446024

With NSIG, Main 1 outpulses: 833360240 to the node

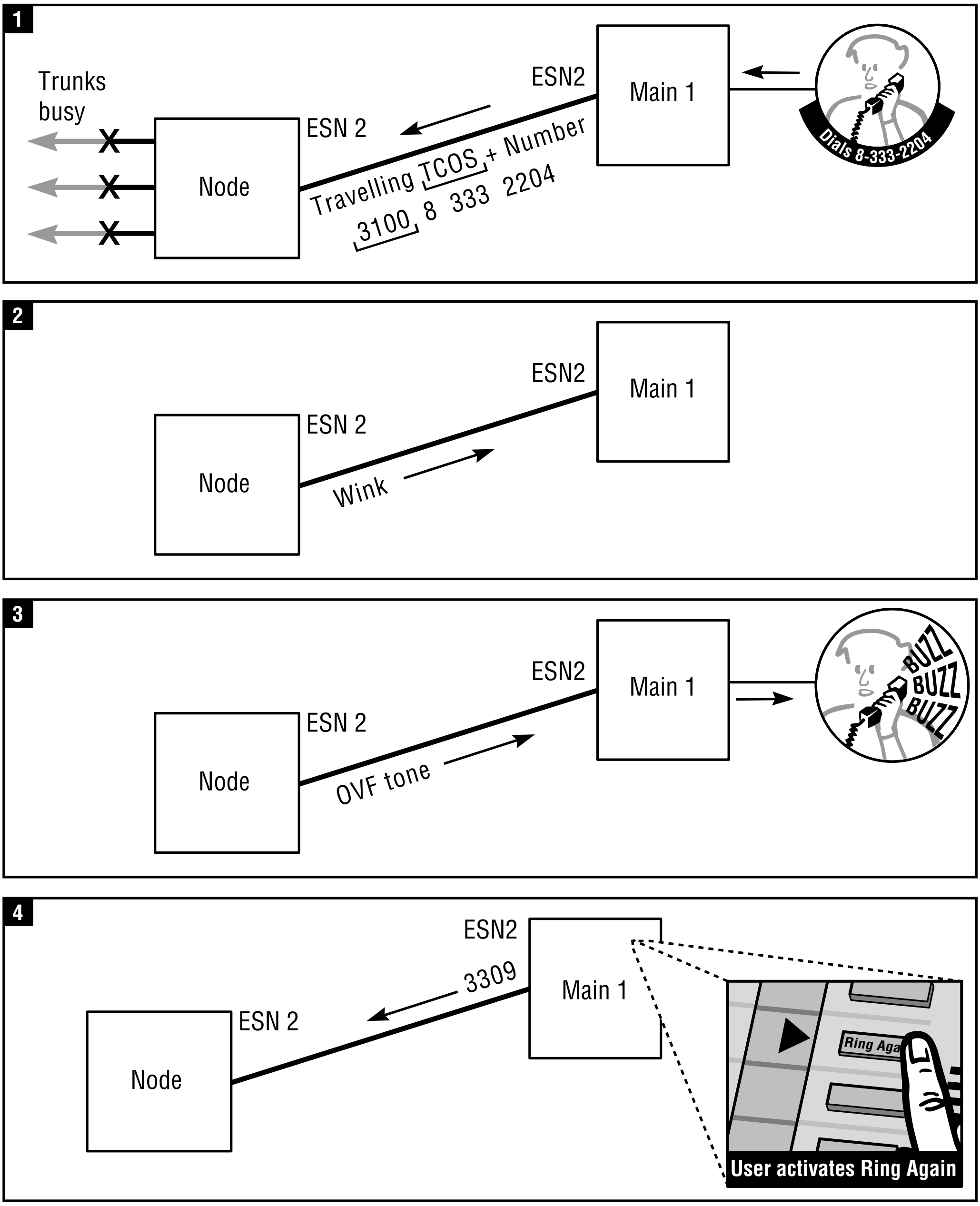
The node translates the call.

With no NSIG, the node outpulses: 6024

With NSIG, the node outpulses: 206024 to the next node - the leading digit 2 tells the second node that the next digit is the traveling FRL of the caller (0, in this case).

Network Signaling (NSIG)

Figure 72- Example 6 - Coordinated Callback Queuing



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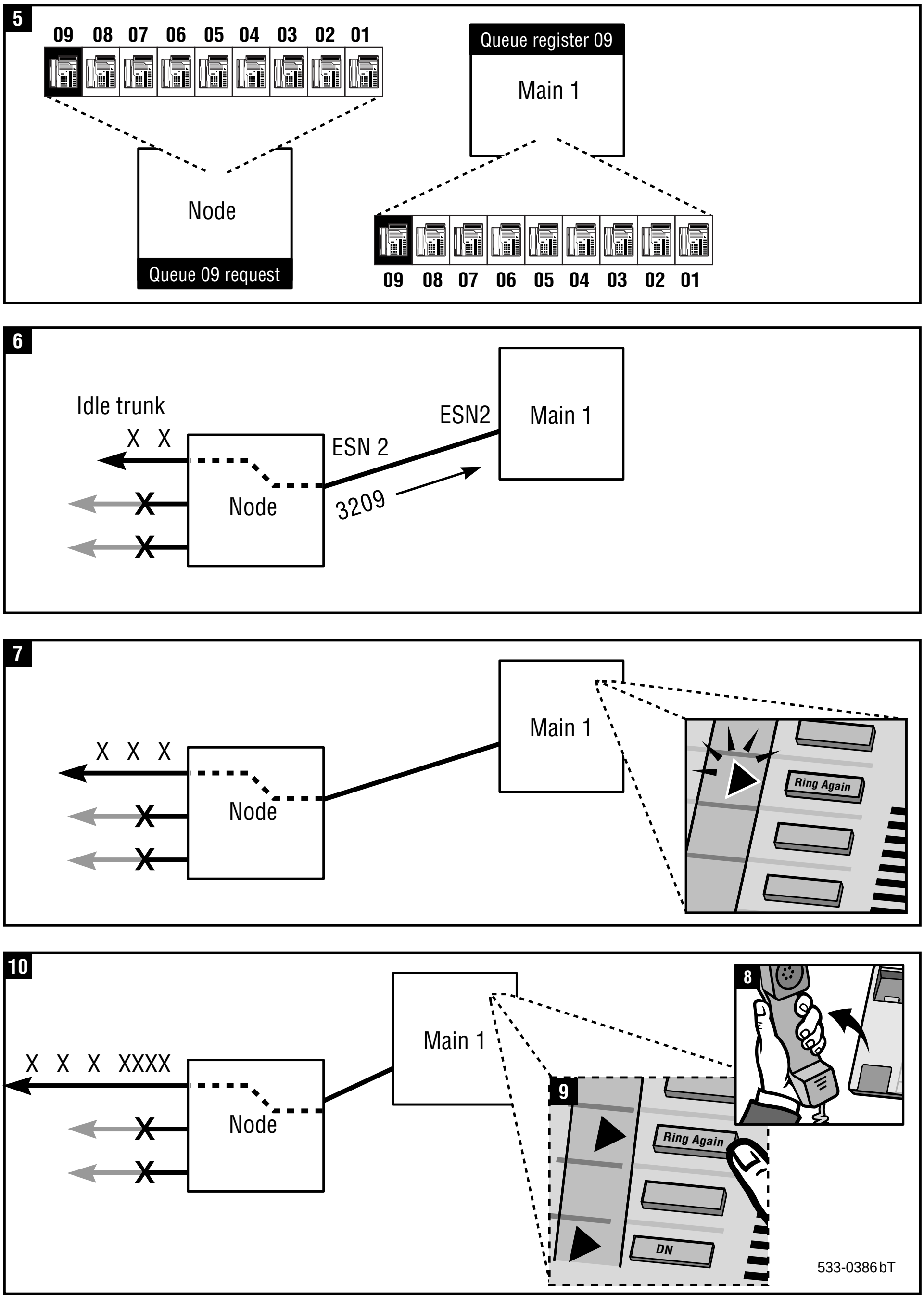
Network Signaling (NSIG)

Figure 73 - Example 6 - Coordinated Callback Queuing (continued)



Network Signaling (NSIG)

Example 6 (continued)

Conditions: SIGO = ESN2. The main and node both have MCBQ packages equipped.

The caller dials: 83332204

With no NSIG, the main outpulses: 83332204

With NSIG, the main outpulses: 310083332204

- 3 – the leading digit 3 alerts the node that queuing can be offered to the caller at the main, if the node's trunks are busy
- 1 – the digit 1 tells the node that the two following digits represent the user's NCOS (00-99)
- 00 – caller's NCOS

If the trunks at the node are blocked, the node sends a wink signal on the TIE trunk to the main. The node gives the caller overflow tone.

The main outpulses 33XX.

- 33 – represents a queue request
- XX – digits that represent a queue ID number assigned by the main (for this example, 09)

The node puts the queue request in its trunk queue.

> ***Note:*** If the user's NCOS has CBQ = NO programmed, then the main does not send 33+ QID to the node, so the node does not queue the caller.

When a trunk suitable for the queued call at the node is found, the node outpulses the digits 32XX to the main.

- 32 – digits 32 tell the main that this is a CCBQ callback.
- XX – two digits that identify the QID assigned by the main. The main matches this QID for the callback with the call information in its queue register (09 in the example above).

Using the queue information in its memory, the main gives CBQ callback treatment to the proper caller for the queued call.

Network Signaling (NSIG)

Meanwhile, the node is slowly outpulsing the digits in the queued call on the trunk it found. It is waiting for the DN at the main to answer.

> ***Note:*** The TIES must be programmed as wink start, digitone and have answer supervision for CCBQ to work.

When the user at the main answers, the remaining digits are outpulsed by the node at normal speed and the call is completed.

Example 7

Conditions: SIGO=ESN3/ESN5 and MCBQ and NXFR packages are installed.

Satellite calls:

If a call is received from a COT trunk programmed as SAT YES and then the call is routed out on a route programmed for SIGO = ESN3 or ESN5:

41XX is outpulsed before the remaining digits in the call.

- 4 – means the call was received from a satellite non-TIE route. This call will not be routed out on a satellite route by any other switches equipped with NSIG and routes programmed with SIGO = ESN3 or ESN5
- 1 – means the next two digits are a traveling NCOS (00-99)

NSIG, Network Call Transfer (NXFER) and CCBQ

There is more information on Network Call Transfer in the *Optimizing trunks* section of this book.

Network Signaling (NSIG)

Example 8

The caller dials:84446022

With NSIG, the main outpulses: 510011384446022

- 5 – means voice call-type; queuing and Network Call Transfer are allowed, if the proper circumstances arise
- 1 – indicates the calling telephone has Ring Again allowed and the next two digits are the user's traveling NCOS
- 00 – represent traveling NCOS 00
- 113 – indicates the trunk member number at the main that is being used for the call. This information is used if the call gets transferred back to the main.

A. Node outpulses: 520011384446022

- 5 – means voice call-type; queuing and Network Call Transfer are allowed, if the proper circumstances arise
- 2 – indicates the call came from a TIE route programmed SAT YES that does not have Ring Again allowed. The call will not be routed over another SAT route. The digit 2 also indicates the next two digits are the user's traveling NCOS.
- 00 – represent traveling NCOS 00
- 113 – indicates the trunk member number at the main that was used for the call

B. Node outpulses: 550011384446022

- 5 – means voice call-type; queuing and Network Call Transfer are allowed, if the proper circumstances arise
- 5 – indicates the call came from a TIE route programmed SAT NO that does not have Ring Again allowed. The digit 5 also indicates the next two digits are the user's traveling NCOS.
- 00 – represent traveling NCOS 00
- 113 – indicates the trunk member number at the main that was used for the call.

Network Signaling (NSIG)

C. Node outpulses: 56011384446022

- 5 – means voice call-type; queuing and Network Call Transfer are allowed, if the proper circumstances arise
- 6 – indicates the call came from a TIE route programmed SAT YES that does not have Ring Again allowed. The call will not be routed over another SAT route. The digit 6 also indicates the next digit is the user's traveling FRL or TCOS (TCM). The incoming TIE route was programmed for ETN and therefore the FRL of the caller travels.
- 0 – traveling FRL 0
- 113 – indicates the incoming trunk member at the main that was used for the call

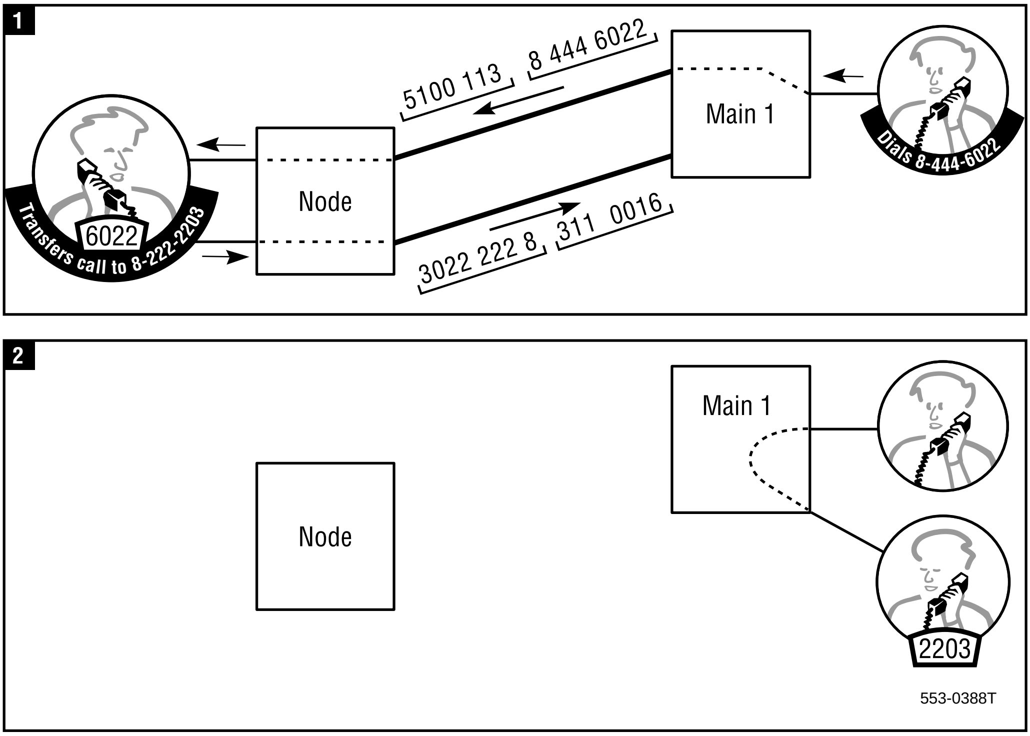
D. Node outpulses: 610011382222203 (refer to Figure 74)

- 6 – means an incoming call from a TIE trunk is being transferred back to its originating switch
- 1 – the telephone being used for the transfer is a Meridian 1proprietary telephone with the Ring Again feature
- 00 – NCOS of the transferring telephone
- 113 – indicates the trunk member at the main that was used for the original call to the node

The TIE trunks are dropped when the main receives a call preceded by the digit 6 and the trunk member number matches the one used for the outgoing call from the main to the node. When the main receives these digits, it completes the call to the DN (2203), drops the TIE trunk being used for the incoming call and also the second trunk (member 113), which was the TIE trunk used for the original call to the node.

Network Signaling (NSIG)

Figure 74 - Network Call Transfer (Example 8-D)

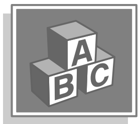


E. Node outpulses: 650011382222203

- 6 – means an incoming call from a TIE trunk is being transferred back to its originating switch
- 5 – the telephone being used for the transfer is an analog (500/2500 type) telephone or a Meridian 1 proprietary telephone without the Ring Again feature
- 00 – NCOS of the transferring telephone
- 113 – indicates the trunk member at the main that was used for the call to the node

The TIE trunks are dropped, if the trunk member number in the outpulsed digits from the node to the main is the same as the trunk member number used for the outgoing call from the main to the node.

Network Signaling (NSIG)



When the main receives these digits, it completes the call to the DN (2203), drops the TIE trunk being used for the incoming call and also thw second trunk (member 113) which was the TIE trunk used for the original call to the node.

Other examples

SIGO = ESN5 (required on DTI trunks). If the DTI trunks are programmed for VOD (voice or data) and a data call is made, the following outpulsing can occur:

- 7100XXX84443305

  7 = data call, 1= telephone has Ring Again, 00= NCOS of caller, XXX = trunk member number, 84443305 = called number

- 7200XXX84443305

  7 = data call, 2 = telephone has no Ring Again, 00 = NCOS of caller, XXX = trunk member number, 84443305 = called number

- 73084443305

  7 = data call, 3 = TCOS to follow, 0 =TCOS of caller, 84443305 = called number

- 750384443305

  7 = data call, 5 = SAT YES on incoming route, 03 = NCOS of caller, 84443305 = called number

Control tips

- Each NSIG-equipped switch has NCOS group programming. It is very important that you program consistent NCOS groups across the entire network.
- You must take the traveling NCOSs and TCOSs into account when you set up the route lists at each switch. If you do not want remote users to make certain calls, you can block the calls using the FRLs on the route list entries.

Network Signaling (NSIG)

Administration tips

- When a conventional main is connected to a node equipped with NSIG and DTI trunks, program the DTI trunks for VCE only or DTA only, not voice or data (VOD). The leading digits 1 or 2 will interfere with call processing between the node and the conventional main.
- For NSIG stand-alone, NXFER, and CCBQ applications trunk signaling can be: DX2 (2-wire Duplex), EAM (2-wire E&M), DX4 (4-wire Duplex), EM4 (4-wire E&M). LDR (Loop Dial Repeating) trunks only support SAT, TCOS, and traveling NCOS. Talk to your system supplier about the type of signaling on your trunks and any changes you must make for the applications you want to use.
- Interaction with CDR: The extra outpulsed digits will appear in the CDR records, if the CDR Outpulsed Digits option is enabled. Adjust your downstream processing accordingly.

Training tips

- If you are implementing Network Call Transfer to have the redundant TIE trunks drop, train users to perform blind transfers (transfer without waiting for an answer).

Network Signaling (NSIG)

What to have ready

The following checklist summarizes the steps you should take before implementing Network Signaling.

Table 61
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Find out the types of switches that will be involved in the NSIG related features.
✔		Set up a network-wide NCOS plan that will be programmed identically at all switches.
✔		Ensure the TIE trunks are programmed correctly to support the features.
✔		Ensure your BARS, NARS and CDP programming at all sites will accommodate appropriately the NCOSs and TCOSs that are traveling.
✔		Prepare training sessions.

Network Signaling (NSIG)

Basic Call Service

Purpose

The ISDN Basic Call Service permits the transmission of the ISDN call. The Basic Call Service consists of call-progress signaling and voice and data transmission.

Basic Call Service signaling is done on the out-of-band D-channel. It involves:

- call set up
- call tear down
- feature activation
- busy and reorder tone activation

Voice and data are transmitted over B-channels. The following modes are available:

- 64 kbps circuit-switched voice and data
- 64 kbps packet data

Basic configuration

Basic Call Service is a capability of PRI. Nortel Networks first introduced PRI using MCDN protocols and then later using other protocols such as DPNSS1 and QSIG. Refer to the sections on the different forms of signaling for more details specific to each one.

Messages to generate tones are transmitted on the out-of-band D-channel, based on a signal generated from the other end.

Both out-of-band messages and in-band tones are provided for ringback. If a call is established and then transferred to a ringing telephone, the user will hear ringback as an in-band tone.

B-channels used for voice and data are assigned on a per call basis.

You can use the PRI link to transmit data at 56 kbps on T1 carriers, if you program the link properly. E1 can support 64 kbps.

Four numbering plans are supported:

- Coordinated Dialing Plan (CDP)
- North American 10-digit numbering plan
- Uniform Dialing Plan (UDP) - private network numbering
- Flexible Numbering Plan (FNP)

BRI and ISL also have Basic Call Service capabilities but their characteristics are different from PRI.

BRI D-channel signaling is done at 16 kbps. ISL can use a D-channel rate up to 56 kbps on T1 (and still have the Revert to Conventional Signaling capability) or 64 kbps on E1 or as low as 2400 baud, depending on the facility being used for the D-channel.

PRI Basic Call Service is supported for:

- Meridian 1 to Meridian 1
- Meridian 1 to SL-100
- Meridian 1 to DMS-100

- Meridian 1 to DMS-250
- Meridian 1 to DMS Centrex
- Meridian 1 to Lucent ESS#4
- Meridian 1 to Lucent ESS#5
- Meridian 1 to AXE-10
- Meridian 1 to Numeris VN2 and VN3
- Meridian 1 to 1TR6
- Meridian 1 to Japan D70
- Meridian 1 to SwissNet
- Meridian 1 to NEAX-61
- Meridian 1 to SYS-12

What to have ready

The following checklist summarizes the steps you should take before implementing the Basic Call Service feature.

Table 62
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Decide if you will use NARS or CDP or both.
✔		Plan the trunking you will require.
✔		Order the ISDN hardware required.
✔		Arrange the information you will need to program the NARS and/or CDP.
✔		Arrange to program the ISDN parameters for basic PRI.

Basic Call Service

ESN on ISDN

Purpose

ISDN basic calls can be performed without ESN software packages installed at either end of the ISDN connection. Users can use direct trunk access codes to make a basic call. However, the services and features offered by ISDN often require signalling between switches, before, during or after the call. One switch must be able to reach another switch automatically, using the information sent to it as part of the D-channel signaling message.

Therefore, Nortel Networks only supports ISDN configurations that build upon an ESN platform. This platform can be based on Network Alternate Route Selection (NARS) or Coordinated Dialing Plan (CDP) or both.

ISDN networks therefore support alternate least cost routing provided by the ESN pre-requisite packages.

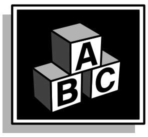
ISDN networks also support a travelling NCOS if NSIG software is installed at all sites where you want this capability. Refer to the section on *Network Signaling (NSIG)* in this book for more information on what NSIG software does.

You can allow users to access Callback Queuing when trunks are busy at the local switch. You can also allow Off-Hook Queuing. OHQ is accessible by local users or remote users coming in on ISDN trunks when local trunks are busy.

ISDN networks do not support Network Call Transfer and Coordinated Callback Queuing functionality that NSIG helps to provide on ESN networks. There are applications, namely Trunk Route Optimization Before Answer and Remote Virtual Queuing, that have been developed for ISDN that replace and enhance these two capabilities.

ESN on ISDN

Basic configuration



You must be aware of the needs of the terminating end for call routing and features every time a call is sent to it. Carefully plan the information that will be sent out to it on the D-channel so one switch can communicate effectively with the other.

You must decide which numbering plans and dialing plans you will use before you can decide what software package(s) best suit(s) your needs. Use:

- NARS, if you require the Uniform Dialing Plan
- CDP, if you want simplified dialing
- FNP, if you want flexibility with NARS or CDP numbering plans, especially in non-North American markets

Network applications are easier to install and maintain if the same ESN software packages are equipped at all sites that must communicate.

Some ISDN applications will tolerate different ESN packages at each site (NARS at one and CDP at the other). Once you understand how the ISDN applications use the Call Type and Calling Line ID in order to function, you will be able to decide if the same packages are required at all sites. You can program the systems in creative ways to make a CDP-equipped system send information to a NARS switch that allows them to work together. For more information on this refer to the *CLID and Name Display options* section in this book.

Control tips

- When you send NCOS values with calls, you will have to include the incoming NCOS values in your restriction plans. For example, if you do not want remote users to dial certain calls or use certain routes, you must set up the route lists to block the calls appropriately.

Administration tips

- If you are using NSIG to allow users’ NCOS values to travel through the network with their calls, it will be best if you define NCOS parameters consistently at all sites.

Training tips

- If users are using a feature such as Coordinated Callback Queuing on an ESN network and you upgrade to ISDN without installing Remote Virtual Queuing, you will have to tell users that they can no longer queue when trunks are busy at distant nodes.

ESN on ISDN

What to have ready

The following checklist summarizes the steps you should take before implementing ISDN and/or the optional related features associated with ISDN.

Table 63
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Find out what ISDN applications you need and learn about how they work.
✔		Plan the ESN and ISDN software for each site that makes the applications work with the least amount of ‘tweaking”. If you must use minimal software for cost reasons, be sure that your solution will not cause installation or maintenance problems.
	✔	If you want NCOSs to travel, plan your NCOS structure from a network-wide perspective.
✔		Train users, if the change to ISDN will affect them.

CLID and Name Display options

Purpose

When a user makes a call using ISDN trunks, a signal is sent on the D-channel that starts with a call type, is followed by the user's CLID and finishes with the digits in the call.

The call type is used by the originating and receiving switches.

- the originating switch uses it to determine what format to use for the CLID
- the receiving switch uses it to determine which NARS access code (AC1 or AC2) to insert in front of the digits in the call

The CLID must be sent in a format that is suitable for the terminating switch and its users. The switch will be able to use the CLID for applications that are supposed to operate on the network. The users see the caller's CLID on the displays of their telephones. The CLID they see should be in the same format they would use to dial that caller. In other words, it is important that one switch is programmed to send a "meaningful CLID" to the other switch.

CLID and Name Display options

Basic configuration

Call type and CLID

The default call type is determined by the way the user dialed the call and the trunk route the call is taking.

Table 64
Relationship between dialing method, trunk type, call type and format of CLID

Dialing method	Outgoing trunk type	Call type	CLID Number of digits	
CDP number (DSC/TSC)	COT	National	10 digit public	
CDP number (DSC/TSC)	TIE	CDP	LSC +DN	Note
UDP LOC +DN	COT	National	10 digit public	
UDP LOC +DN	TIE	UDP	HLOC + DN	Note
AC1 or AC2 +NPA+XXX-XXXX	COT	National	10 digit public	
AC1 or AC2 +NPA+XXX-XXXX	TIE	National	10 digit public	Note
AC1 or AC2 +NXX XXXX	COT	Subscriber	7 digit public	
AC1 or AC2 +NXX XXXX	TIE	Subscriber	7 digit public	Note
AC1 or AC2 +SPN	COT	National International	10 digit public variable number of digits	
AC1 or AC2 +SPN	TIE	SPN	HLOC + DN	Note

Note: You can change the default call type using digit manipulation indexes on TIE trunks.

CLID and Name Display options

The terminating switch uses the call type to determine which NARS access code, if any, to insert in front of the digits in the call. If the call type is UKWN (Unknown) or CDP, then no access code will be inserted. Any other call types will be affected by the operation of the INAC prompt.

INAC

INAC (Insert Network Access Code) is programmed on incoming ISDN TIE trunks in MCDN or QSIG networks. As calls come in, it will reference information programmed in the Customer Data Block to determine whether to put AC1 or AC2 in front of the digits in the incoming call. In the Customer Data Block the types of calls that are to have AC2 inserted in front are programmed. The call types not included there will have AC1 inserted.

> ***Note:*** on ESN networks you may be outpulsing the access code from the originating end or using the insert capability on the incoming TIE route. Neither of these is recommended for ISDN networks. If you are upgrading an ESN network to ISDN functionality, you will need to make modifications in order to use INAC.

Using call type and INAC

- If you have a remote switch equipped with CDP and a node equipped with NARS, users will be dialing Trunk Steering Codes for calls to the node that are supposed to go out on the public network.

  If a CDP call type is sent to the node on a TIE trunk then INAC will not operate and no access code will be inserted in front of the digits in the call. However, if you use digit manipulation so that a call type of National or Subscriber is sent, the node will be able to use INAC properly. The call will be routed to the public network.

- If a user at a CDP switch dials a TSC followed by a seven digit on-net call and it is sent to a NARS node on a TIE trunk, you will need to insert a call type of UDP. That way the NARS switch uses INAC to translate the call as an on-net LOC call and not a CDP call.

CLID and Name Display options

Format of CLID

If the call was dialed as a NARS or CDP on-net call and it is being routed on a private trunk (TIE), then the CLID sent will be the Prime DN of the caller with a HLOC or LSC inserted in front. The HLOC and LSC are programmed in the Customer Data Block of the originating switch. If your CDP- DNs do not have extra leading digits, you do not have to send any LSC, leave it blank when you program.

If the caller has dialed a public number and the call is being routed on a public trunk, then the CLID will be composed based on a customer-wide option and a telephone class of service.

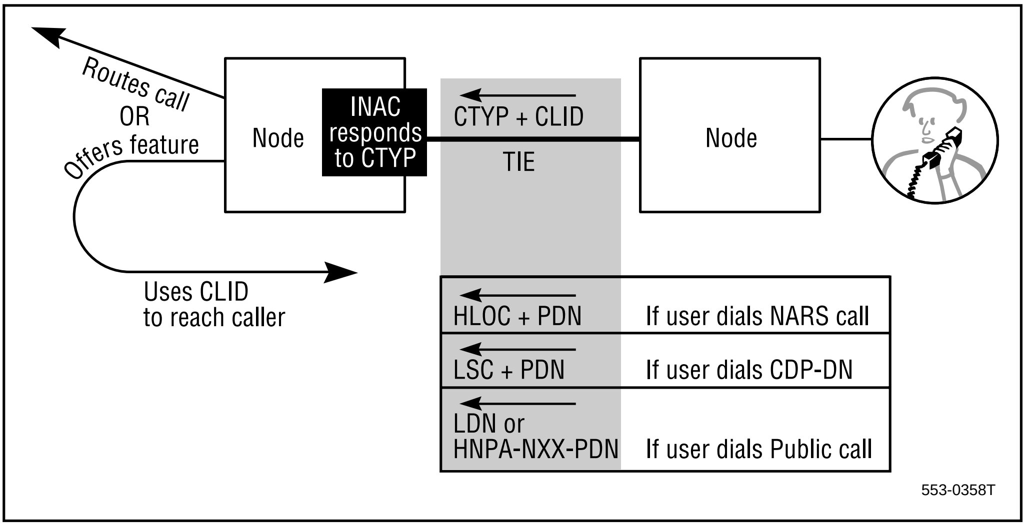
If the customer option is LDN (Listed Directory Number), then the LDN for the customer group is sent. This is appropriate if the user does not have a DID telephone number. The called party will only be able to call back to the LDN of the customer group to reach the caller another time.

If the customer option is PDN (Prime DN) then the class of service assigned to the telephone determines what CLID will be sent. If the telephone has the default setting (PDN) then the DN assigned to key 0 (the prime DN) is sent, with the NXX (and possibly the NPA) of the LDN inserted in front. If the telephone has a class of service of LDN, then the customer LDN number is sent.

The CLID for the attendant: If the call type is private, the attendant DN is used (if the attendant DN is 0 then the called party sees H0 on the display). If the call type is public, LDN0 is used.

CLID and Name Display options

Figure 75 - Format of CLID



Improving feature performance

The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

ISDN CLID Enhancements (Release 22)

The capabilities introduced by the ISDN Calling Line Identification (CLID) Enhancements greatly increase the functionality of CLID within any ISDN network.

These enhancements replace the previous method of constructing CLID, and allow much more flexibility when programming CLID for any telephone on a Meridian 1.

CLID and Name Display options

Prior to Release 22, a CLID was defined as either a Listed Directory Number (LDN) or Primary Directory Number (PDN). The contents of these data fields were used to generate the CLID for a call. The CLID could only be built from key 0 of a telephone. This meant that regardless of what key was used to make a call, it was the CLID for key 0 that was sent. Also, only one office code (NXX) and one location code (LOC) was assignable for a customer group to be used for all outgoing calls from that group.

As of Release 22, CLID is table-driven. Virtually any of the information contained in the fields of the CLID table can be programmed against any DN or DN key, on a per telephone basis. The CLID that is sent from a telephone is based on what is in the CLID table, rather than the LDN or PDN.

That is, a CLID for any key is built by taking the information contained in a particular field in the CLID table and adding that information to the key's DN. A multi-line telephone can have DN keys that each has its own CLID. Or, the CLID of any one key could be programmed to use the CLID of any other key on the telephone.

The Meridian 1 can support multiple office codes, location codes and steering codes in CLID. This means that any telephone on one Meridian 1 can send a CLID that will have calls returned to another Meridian 1. This type of configuration would typically be used in cases where a customer would want calls to be returned to only one central location.

> ***Note:*** Since the system does not perform verification of CLID entries that are defined in a CLID table, it is your responsibility to ensure that the programming is suitable for your needs.

How a CLID table is built

North American Numbering Plan

For users of a North American Numbering Plan, the Meridian 1 can support multiple Home Central Office Codes (HNXXs), Home Numbering Plan Area (HNPA) codes, Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For a North American Numbering Plan, each CLID entry can contain the following:

- a one-six digit national code for a home national number (HNTN)
- a one-12 digit local code for a home local number (HLCL), or a one-12 digit Listed Directory Number for a switchboard
- a one-seven digit Home Location Code (HLOC)
- a one-seven digit Local Steering Code (LSC)

International Numbering Plan

For users of an International Numbering Plan, the Meridian 1 can support multiple Prefix 1 (PFX1) and Prefix 2 (PFX2) contents, and multiple Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For an International Numbering Plan, each CLID entry can contain the following:

- a one-six digit national code for a home national number (HNTN), which is the equivalent of PFX1
- a one-12 digit local code for a home local number (HLCL), which is the equivalent of PFX2, or a one-12 digit Listed Directory Number for a switchboard
- a one-seven digit Home Location Code (HLOC)
- a one-seven digit Local Steering Code (LSC)

Another capability, which applies to both the North American Numbering Plan and the International Numbering Plan, pertains to how the HLCL is constructed.

A prompt, DIDN (which signifies “use DN as a DID number”) allows the HLCL to be built either using the digits in the HLCL plus the digits of the active key, or only the digits in the HLCL, or based on a search on the DN keys, beginning from key 0, to find the CLID to be used.

You can allow the HLCL to be built using the digits in the HLCL plus the digits of the active key, or only the digits in the HLCL, or based on a search on the DN keys, beginning from key 0.

CLID and Name Display options

How CLID entries are assigned to a telephone

Once the CLID table has been built for a customer, any CLID entry can be assigned to any telephone, on a per DN or DN key basis. This means that for a multi-line telephone, DN key 0 may send one CLID (the information contained in one CLID entry), DN key 1 may send a different CLID (the information contained in another CLID entry), and DN key 2 may send yet another CLID (the information contained in a third CLID entry). Therefore, a customer can send different CLIDs from the same telephone.

When configuring the CLID for a DN key on a Meridian proprietary telephone, the value entered may be a CLID table entry number (0-3999), which corresponds to a CLID entry in the CLID table or ‘D’. If ‘D’ is entered, the Meridian 1 initiates a search on the telephone for a DN key, starting from key 0. The first found CLID is then sent as the CLID of the active DN key. This means that a call can be made on one key, and the CLID of another key is sent. This configuration is typically used in an ACD or Hotline DN key application, where you do not want the CLID for a particular key to be sent.

Prior to the CLID enhancement, for feature keys, such as the Call Transfer key (TRN), the Conference key (AO3 and AO6) and the No-hold Conference key (NHC), the DN of key 0 was used as the CLID DN when a call transfer or conference occurred over ISDN trunks. With the CLID enhancement, the CLID DN is associated with the active DN key, rather than key 0.

Refer to the NTPs on *Networking, Book 1 of 2, (ISDN Calling Line Identification Enhancements)* for more information and examples of setting up the CLID based on the Release 22 capabilities.

Operating parameters

In order for CLID to be properly delivered, a CLID entry must be defined in the CLID table for a customer. If a CLID entry or table is not defined, the active DN is sent as CLID.

The ISDN Calling Line Identification (CLID) Enhancements feature only pertains to calls that are made over ISDN routes.

CLID and Name Display options

The CLID that is sent from the Meridian 1 may be subject to any restrictions which may be imposed by the serving Telco.

This feature does not change the operation of CLID for attendant consoles. If the call type is private, the attendant DN is used. If the call type is public, LDN0 is used.

If a CLID entry is deleted from the CLID table and the CLID entry remains assigned to a DN key, the active DN is sent as CLID.

The maximum number of digits for a CLID in an ISDN message is 16.

Constructing a seven-digit local number or a ten-digit national number, for the North American Numbering Plan, remains the restriction. Also, a seven-digit maximum is maintained for an ESN Uniform Dialing Plan number, if the Flexible Numbering Plan package is not equipped. The seven and 10 digit restriction for a public number does not apply to International ISDN interfaces.

Feature interactions

The ACD DN is sent as the CLID for a call made by an ACD agent using a DN key on a key other than Key 0. With the feature enhancement, the CLID is constructed using the CLID entry associated with the active DN key. The ACD agent ID is not designed to be sent as the CLID.

If an ACD agent has an active call on Key 0 and if a call transfer or conference is initiated by the ACD agent, the CLID entry associated with the ACD DN Key 0 is used as the CLID (for remote calls only.)

There is no CLID entry for an analog (500/2500 type) ACD DN. The CLID associated with the analog (500/2500 type) DN is used when a call transfer or conference is initiated by an ACD agent on an analog (500/2500 type) telephone.

Call Detail Recording

The CLID in the CDR records, including the X records, contains the DN of the key from which the call is made, not the DN of Key 0.

CLID and Name Display options

Call Pickup Network Wide

The Private Integrated Services Network Exchange (PINX) DN in the Customer Data Block is used for Call Pickup Network Wide. The network DN for the PINX DN is constructed using the existing Home Location Code (HLOC) or Local Steering Code (LSC) in LD 15. The DN of the originating party is constructed using the CLID associated with the active DN key. The DN of the originally called (ringing) party is constructed using the existing HLOC or LSC in LD 15.

Calling Party Name Display

If a call transfer or conference is initiated on a multiple appearance DN programmed on a key other than Key 0, the Call Party Name Display associated with the DN of the active key is used, rather than the Call Party Name Display for Key 0.

CLID for an ISDN BRI telephone

For an internal call terminating on an ISDN BRI telephone, the calling telephone’s Dialed Digits Denied (DDGD)/Dialed Digits Allowed (DDGA) Class of Service is used to determine whether to send Calling Party Number to the terminating ISDN BRI telephone for display purposes.

Connected Number

When a call is modified, by a call transfer for example, this feature enhancement will try to use the CLID entry associated with the active DN key if available, otherwise the connected number will be constructed using CLID entry 0.

EuroISDN Continuation for UK/Spain/Belgium/SN3

The EuroISDN Continuation feature allows Home National Numbers and Home Local Numbers to be configured on a route. When an ISDN call is made from a telephone to a EuroISDN interface, the CLID constructed by EuroISDN, based on the outgoing route, takes precedence over the CLID constructed for the calling telephone.

EuroISDN Trunk - Network Side

The EuroISDN Trunk - Network Side connectivity supports all of the user side ISDN CLID enhancements introduced by Generic X11 Release 22 software.

CLID and Name Display options

Network Attendant Service

If Network Attendant Service is equipped, CLID entry 0 is used for incoming trunks.

Network Call Redirection

Network Call Redirection constructs Redirecting Number and Redirection Number. The feature enhancement does not change the construction of the Redirecting Number. However, the Redirection Number of the Notify message is constructed using the CLID entry 0 of the table in the Customer Data Block.

Network Message Services
Message Waiting Indication with DMS

When a user leaves a voice message, from a multiple appearance DN on a key other than Key 0 (such as Key 1), the caller’s recorded number will be the multiple appearance DN on Key 1, rather than the primary DN of Key 0. This means that when the user returns the call, he/she will ring the DN of Key 1 on all the sets that have the appearance of the DN.

When a user retrieves messages using a multiple appearance DN key other than Key 0, the user retrieves the messages associated with the non-prime DN key.

In general, with the enhancement of sending the DN associated with an active key to make a call to Meridian Mail, the secondary key’s DN should be included in the Meridian Mail User’s definition (for either a local or remote user).

Network Ring Again

The Network Ring Again feature remains operational when a user uses a multiple appearance DN on a key other than Key 0 to activate Network Ring Again, since Network Ring Again saves the Terminal Number of the telephone that initiates Network Ring Again.

Remote Virtual Queuing

The Remote Virtual Queuing (RVQ) feature remains operational when a user uses a multiple appearance DN on a key other than Key 0 to activate RVQ.

CLID and Name Display options

Incoming Trunk Programmable CLID

British Telecom in Australia requested this feature for analog TIE trunks connected to a PSTN.

A billing number of 1 to 16 digits can be assigned to incoming trunk routes. It is passed along as a CLID, if the incoming trunk terminates on a PRI/PRI2/BRI trunk connected to a Central Office. If an incoming route supports CLID, you can replace it with the billing number, if you wish.

This billing number can be used for display purposes as well.

It is not supported on outgoing interfaces that are programmed as SL1 or DPNSS1.

CLID Suppression

M5317 (BRI) telephone users can suppress sending a CLID on a per call basis. Other telephone users are affected by the Calling Party Privacy feature.

Network Call Party Name Display/Network Name Delivery

This feature provides a network-wide visual display of names and telephone numbers to both parties involved in a call.

The capability to send and receive names is configured on a trunk route basis.

DNs are assigned names in the programming of the originating switch. CPND software is required.

> ***Note:*** There is a class of service you can activate related to redirected calls (Dialed Name Display allowed/denied). You can choose between allowing the terminating telephone user to see the caller's name or the redirecting telephone user's name when calls forward or hunt before they are answered. Secretaries often prefer to see the name of the user for whom they are answering, instead of the caller's name.

CLID and Name Display options

There are two formats for names on MCDN networks:

- ND1 was introduced in Release 13 for Meridian 1 to Meridian 1 connections only
- ND2 was introduced in Release 17 for Meridian 1 to SL-100 and DMS connections

Table 65
ND1 and ND2

ND1	ND2
called party name is displayed on the caller's display only after the call is answered unless the call is being redirected	called party name is displayed on the caller's display during the ringing phase
maximum number of characters in name is 24	maximum number of characters in name is 15

> ***Note:*** The ISDN QSIG Name Display feature introduced the ND3 capability for the transport of names on QSIG networks. There is information on the ND3 format later in this section.

You can also name the reasons for redirection. This helps users to understand why a call is being redirected to the specific telephone. The reasons can be: Hunting, Call Forward All Calls, Call Forward No Answer, Call Transfer and Call Pickup.

Calling Party Privacy (Release 21)

This feature allows the Meridian 1 to support the “Calling Party Number and Name per-call blocking” requirement of the Federal Communications Commission (FCC) in the United States and the Canadian Radio and Television Commission (CRTC) in Canada.

Users can have a Class of Service that activates or deactivates the blocking all the time. Also, if the blocking is deactivated, the user can dial a Flexible Feature code before making a call to block the number and name delivery for that call.

CLID and Name Display options

The feature works on non-ISDN trunks as well as ISDN trunks. Only COT, FEX, WAT and DID non-ISDN trunks are supported at present. Non-ISDN trunks may tandem to ISDN trunks in the network. If the user dials the CPP feature code, a Privacy Indicator is transmitted with the call on a non-ISDN trunk. The switch that receives that code will transmit it to the next switch in the network to indicate the call is to be treated as private. As a result, the terminating telephone will not display a name or number for the caller. If the terminating telephone is on a Meridian 1, it displays the trunk route access code and member number instead.

The Privacy Indicator can be up to four digits. It can never include #. On dial pulse trunks, * is not allowed.

Interfaces supported are:

- DMS-100
- DMS-250
- SL100
- Lucent #4 ESS
- Lucent #5 ESS
- TR-1268 (NI-2)
- MCDN private networks
- QSIG
- EuroISDN
- BRI Trunk Access

Display of Calling Party Denied (Release 21)

Analog (500/2500 type) telephone users and Meridian 1 proprietary telephone users can use this feature to allow or deny the display of their name and number on the display of the telephone being called.

This feature operates internally and on MCDN ISDN networks.

CLID and Name Display options

Two class of service options are involved with this feature:

- Display Digits allowed/denied (DDGA/DDGD)
- Name Display allowed/denied (NAMA/NAMD)

The caller always sees the digits dialed, no matter what class of service is programmed on the terminating telephone.

When digits are suppressed, they are replaced with a dash for each suppressed digit. Where a name is suppressed four Xs (XXXX) will display instead.

Refer to the *Networking* NTPs (Display of Calling Party Denied) if you would like to see examples of the display possibilities related to the class of service possibilities.

If you want attendants on an ISDN PRI network to see the name and number of callers, despite what is programmed on the caller's telephone, you will need to install Network Attendant (NAS) software on your switch.

Add a Prefix (Release 23)

This feature adds fixed prefix digits to the display of the CLID or Connected Number.

> ***Note:*** This feature was part of the EuroISDN Continuation Phase III development.

The added prefix digits assist the user in calling back to the national or international number. Enable the feature on incoming and outgoing trunk routes.

The addition of the prefix is allowed on the following types of interfaces:

- EuroISDN
- JTTC
- MCDN
- QSIG

CLID and Name Display options

If incoming calls are from these types of interfaces, the prefix is allowed, provided the Route Data Block prompt ADDP is programmed with a YES response.

Then, if the Type of Number (TON) field of the incoming call is National, the prefix inserted is 0. If the TON is International, 00 is inserted.

CLASS: Calling Number and Name Delivery (Release 23)

This feature allows a Meridian 1 system, in North America, to deliver the calling number and/or the calling name, along with the calling date and time, to a non-proprietary analog telephone that complies with the Custom Local Area Signaling Service (CLASS) Bellcore CND standard. These types of telephones are often installed in homes, hotels, hospitals, and schools.

A CLASS telephone is, by definition, any non-proprietary analog telephone with an integrated display and a built-in Frequency Shift Key (FSK) modem receiver. However, the CLASS feature will work with an analog telephone which is connected to a display adjunct and an FSK modem as well. The CLASS telephones are configured on the Meridian 1 as analog (500/2500 type) telephones and are supported by the existing 500/2500-type line cards.

The calling number and/or calling name data is delivered by the Meridian 1 using FSK signaling using a CLASS modem (CMOD) unit. The modems are supported by an Extended CLASS Modem (XCMC) line card. When a call is to be presented to a CLASS telephone, an available CMOD is automatically allocated.

The CND information is delivered to the CLASS telephone after ringing has been applied for a new call, in the first silent interval that is greater than two seconds.

Up to 10 digits can be sent as the calling number. Up to 15 characters can be sent as the calling name. Once the Meridian 1 delivers the CLASS CND information, it is completely up to the CLASS telephone to determine how the information is to be displayed. Some CLASS telephones may not display all the information presented to them.

CLID and Name Display options

When the calling number or name is unknown, an indicator in the form of the ASCII character “O” is delivered to the CLASS telephone. When the calling number or name is suppressed for privacy reasons, an indicator in the form of the ASCII character P” is delivered, to the CLASS telephone, instead of the calling number. The CLASS telephones present this information on the display in their own way.

Meridian 1 systems do not have to be configured for ISDN to deliver this functionality but they all require the CLID Number and CLID Name software packages (whether ISDN is equipped or not).

Calling Party Privacy Override (Release 24)

The Calling Party Privacy feature, introduced in X11 Release 21, allowed you to permanently block the Calling Party Number and Name from being sent from a telephone. It also allowed a user not to send this information on an individual call by dialing a Flexible Feature Code before placing a call.

The Release 24 Calling Party Privacy Override (CPPO) feature is targeted to the North American market to comply with an FCC requirement. It can be used globally.

It allows you to permit a user with a Class of Service that blocks the sending of Calling Party Number and Name to unblock this on an individual call. The user must dial a Flexible Feature Code (FFC) before placing a call in order to activate the unblocking.

You must activate this feature on the outgoing trunk routes of your choice as well. This feature does not apply to internal calls.

ISDN or non-ISDN DID, COT, WAT, and FEX trunk types support this feature; non-ISDN TIE trunks do not. The following connectivities are supported:

- DMS100/250
- SL100
- Lucent #4 ESS (ESS4)
- Lucent #5 ESS (ESS5)

CLID and Name Display options

- TR-1268 (NI-2)
- MCDN private networks
- EuroISDN
- QSIG
- BRI trunks

The FFC digits are not stored in the call register. However, if the trunk route has the feature activated, there is a CPPO indicator that is sent to the other end, allowing it to display the Calling Party Number and Name on the terminating telephone. For non-ISDN routes, confirm that the switch at the other end also supports CPPO. Program digits for the Privacy Override Indicator that the other switch will recognize and handle properly.

The CPPO feature takes precedence over the Display of Calling Party Denied feature.

CDR continues to include the Calling Party Number, even when the user blocks it.

For tandem calls:

- incoming ISDN presentation indicators are passed out to ISDN outgoing trunks
- incoming ISDN trunk calls with both CLID and CPND indicators set to “allowed” are treated as CPPO calls when tandeming to a non-ISDN trunk
- incoming non-ISDN calls are tandemed according to the programming of the TCPP prompt on the outgoing route. If TCPP is NO, then the call is sent out as a CPPO call.

Display of Access Prefix on CLID (Release 24)

This feature enhances the earlier “Add a Prefix” capability introduced in X11 Release 23. It is designed for use by Orion telephone applications such as Call Log. (This feature is also useful with Symposium Call Manager and Visit.) Call Log stores numbers which were calling or called numbers to and from that telephone. It stores the

CLID and Name Display options

numbers directly from the display. The feature is also supported on Modular Digital telephones, mobile telephones and Attendant Consoles.

As with the previous "Add a Prefix" capability, the Access Prefix is inserted in front of the CLID or Connected Number. The prefix is obtained from a table with values for all the allowed Numbering Plan Identifier (NPI) and Type of Number (TON) combinations. The prefix you program for each NPI and TON combination in the table can be up to four digits long. In a recall situation, the prefix digits are handled as if they were dialed from the keypad.

The digits inserted must be able to route the call properly using trunk route access codes, NARS access codes or CDP steering codes. Default table number 0 adds prefix digits the same way it did with the Release 23 "Add a Prefix" capability.

You must do the following programming for the feature to work:

- set up different tables in the Customer Data Block, to be used on different trunk routes
- enable this feature in the Route Data Block and assign the appropriate table from the Customer Data Block to the route
- allow the feature in the Class of Service of applicable telephones

This feature only applies to ISDN routes. The following interfaces are supported:

- MCDN (BRI and PRI)
- QSIG
- EuroISDN
- JTTC

The CLID field in the CDR is not affected by this feature.

CLID and Name Display options

ISDN QSIG and EuroISDN Calling/connected number display options

Calling Line Identification Presentation (CLIP)

This supplementary service is offered to the called user. It provides the called user with the possibility of receiving identification of the calling party. When the system is connected to a network that can provide CLIP, the incoming SETUP message contains the Calling Party Information Element (IE).

Calling Line Identification Restriction (CLIR)

This supplementary service is offered to the calling user. It provides the calling user with the capability of preventing presentation of the ISDN number to the called party.

The Meridian 1 uses the Display of Calling Party Denied feature to accomplish this on a telephone basis. Called parties with an override capability can still be presented with the number. Basic Rate Interface telephones can present or restrict a CLID on a per call basis, if the CLIR service is defined for the telephone.

The restricted mode can be provided on a permanent basis for the646

Meridian 1 by the network in some countries.

Connected Line Identification Presentation (COLP)

This supplementary service is offered to the calling user. It provides the calling user with the capability of receiving identification of the connected party (that is, the telephone that received the call, possibly after some form of redirection occurred). When the network is able to provide COLP service, the CONNECT message contains a Connected Number Information Element. It can be implemented on BRI telephones as of Release 22.

The connected party number appears on the display beside the called party number.

CLID and Name Display options

Connected Line Identification Restriction (COLR)

This supplementary service is offered to the connected user. It allows the connected party to prevent presentation of its ISDN number to the calling party. The Meridian 1 uses the Display of Calling Party Denied feature to accomplish this on a telephone basis. It can be implemented on BRI telephones as of Release 22.

In some countries, COLR can be implemented by the network in a permanent mode.

If a telephone is programmed as Restricted, a calling party with override capability can still be presented with the connected number.

ISDN QSIG Name Display

There are three supplementary services associated with this feature in X11 Release 22. These services apply to a PRI or BRI link.

The maximum number of characters is 50.

Calling Name Identification Presentation (CNIP)

When this service is enabled the calling party’s name is displayed on the called/connected party’s telephone. The CNIP service is available on a permanent basis only. The calling party must have Name Presentation allowed in the class of service.

Connected Name Identification Presentation (CONP)

When this service is enabled the called/connected party’s name is displayed on the calling party’s telephone. The CONP service is available on a permanent basis only. The called/connected party must have Name Presentation allowed in the class of service.

Calling/Connected Name Identification Restriction (CNIR)

This feature prevents the calling party’s name from being presented to the called party’s telephone. It also prevents the called party’s name from being presented to the calling party’s telephone.

CLID and Name Display options

The feature can be invoked in one of two ways:

- on a permanent basis with a class of service of Name Presentation denied
- on a per call basis. The class of service is Name Presentation allowed but the user dials a Calling Party Privacy Flexible Feature Code to invoke privacy before initiating the call.

QSIG Name Display Enhancement

In X11 Release 24, there is an option of choosing the method of Operation Coding for the Name Display supplementary services. The choices are:

- for ETSI interfaces, the choice can be by Object Identifier
- for ISO interfaces, the choice can be by Integer Value

The Operation Coding method is set up when programming the Remote Capability of a D-channel.

QSIG links

Name Display operates even when the following features are performed over a QSIG link:

- Call Pickup (caller at different switch from the ringing telephone and the telephone answering the call)
- Call Transfer
- Conference
- Call Forward of the following types:
  - Call Forward All Calls
  - Call Forward Busy
  - Call Forward by Call Type
  - Call Forward External Deny
  - Call Forward, Internal Calls

- Call Forward No Answer
- Call Forward No Answer, Second Level

◆ Hunting

◆ Incoming Digit Conversion (IDC) - If name information is not received from a DID trunk, IDC is activated and a name is associated with the new digit sequence, this information is passed over a QSIG link to a terminating telephone. If name information is received from a DID trunk, it takes precedence over the IDC trunk name.

QSIG/MCDN links

Name Display operates even when the following features are performed over a QSIG/MCDN link:

◆ Call Forward of the following types:

- Call Forward All Calls
- Call Forward Busy
- Call Forward by Call Type
- Call Forward External Deny
- Call Forward, Internal Calls
- Call Forward No Answer
- Call Forward No Answer, Second Level

◆ Hunting

◆ Call Pickup Network Wide (caller at one switch, ringing telephone at another switch and answering telephone at a third switch)

◆ Call Transfer

◆ Conference

◆ Incoming Digit Conversion (MCDN/QSIG and QSIG/MCDN)

Choose a Remote Capability of ND3 if you want equivalent name display service between QSIG and MCDN networks.

CLID and Name Display options

QSIG Call Diversion Notification

In Release 23, if calls are diverted across a QSIG PRI2 or BRI trunk network, the display of the calling party telephone can be set up to indicate the reason for redirection and the number (and name, if programmed) of the final destination telephone. The display of the final destination telephone can be set up to indicate the reason for redirection and the number (and name, if programmed) of the diverting telephone or of the calling telephone.

The redirections or diversion-related features that are supported are:

- Call Forward Unconditional (called Call Forward All Calls on the Meridian 1)
- Call Forward Busy (called Hunting on the Meridian 1)
- Call Forward No Reply (called Call Forward No Answer on the Meridian 1)

The functionality centers around the programming associated with the diverting telephone. There are two aspects to this programming. One part controls what will be shown on the display of the calling telephone and the other part controls the display on the final destination telephone.

The three options affecting the display of the calling telephone are:

- show the dialed number only
- show the dialed number and a reason for redirection code
- show the dialed number, the final destination number (and name, if programmed) and a reason for redirection code

The two options affecting the display of the final destination telephone are:

- show the number of the calling telephone and the number (and name, if programmed) of the diverting telephone
- show the number of the calling party

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The Dialed Name Display feature also interacts with the operation of the Call Diversion Notification functionality. The class of service of the final destination telephone can be programmed as Dialed Name Display allowed (DNDA) or denied (DNDD). This feature cannot override the programming of the diverting telephone. In other words, if the diverting telephone is programmed not to send a number (and name), the programming of the final destination telephone cannot force these to be sent. If the diverting telephone is programmed not to send number and name, the final telephone cannot display the information. If the final telephone is programmed for Dialed Name Display denied, the name of the calling user is displayed instead, if it is sent with the call.

An example follows which shows the different options available. The following information applies to the example:

Caller: DN 4000Name: Arnold

Diverting telephone: DN 5000 Name: Bob

Final destination telephone: DN 6000 Name: Car

CLID and Name Display options

Table 66
Example of programming involved in QSIG Call Diversion Notification

Diverting telephone option concerning calling user	Calling user's display after final destination telephone rings
NO	5000
YES (without final destination telephone number and name, when available)	5000 F
YES (with final destination telephone number and name, when available)	5000 H6000 F CARL

Diverting telephone's option concerning final destination telephone display	Final destination telephone display related to Dialed Name Display class of service	
	DNDD	DNDA
NO	H4000 H---- F ARNOLD or H4000 F ARNOLD	H4000 H---- F or H4000 F
YES	H4000 H5000 F ARNOLD	H4000 H5000 F BOB

Each D-channel must be configured with diversion as a remote capability at the originating, diverting, final destination and rerouting nodes.

QSIG Call Diversion Notification Enhancements

In X11 Release 24, the Call Diversion Notification functionality has been extended to include a QSIG/MCDN gateway, and a QSIG/DPNSS1 gateway.

Two mechanisms for handling diversion are possible. They are:

- Forward Switching: performs the diversion by joining two circuits. The first circuit is the connection between the originating switch and the switch that is diverting the call. The second circuit is the connection between the switch that is diverting the call and the switch that is receiving the diverted call.
- Rerouting: performs the diversion by replacing the connection from the originating switch to the diverting switch with another new connection from the originating switch to the switch that is receiving the diverted call.

The Meridian 1 supports the Forward Switching method, both in a sending and receiving mode. The Rerouting method is only supported in a receiving mode. The Meridian 1 supports the receipt and treatment of a Rerouting request sent by another vendor switch but does not generate such a request. The notification to the originating and terminating users' telephones is supported with both Rerouting and Forward Switching methods.

Single Diversion Notification rules

This section describes the originating and terminating user's notification rules, and the reason code notification rules for single diversions in a pure QSIG environment.

The notifications are detailed for the following scenario: "Station A (originating telephone) calls Station B (diverting telephone). One of the following features occurs at Station B: Call Forward Unconditional (CFU), Call Forward Busy (CFB) or Call Forward No Reply (CFNR). The call is diverted to Station C. Station C (terminating telephone) answers".

Originating user's notification rules

Diversion Reason Code Notification rules:

- This reason code is displayed or not displayed on the originating user's telephone (as soon as the Diversion Notification information is received from the node with the diverting telephone) according to the subscription option programmed for the diverting user's node (see Table 67 on page 638).

sums up the originating user's Notification, according to the diverting user's subscription options. Note that, in the case presented in the example, the originating user's name is Abby, with a DN of 4000. The diverting user's name is Bobby, with a DN of 5000. The terminating user's name is Cathy, with a DN of 6000.

Table 67
Originating user's notification with respect to the diverting user's subscription option

	Originating user's display	
Diverting user's subscription option: "Calling user receives notification that the call has been diverted"	after receipt of diverting user's Diversion Notification information	after receipt of terminating user's Diversion Notification information
No	5000	5000
Yes without diverted-to number and name	5000 F	5000 F
Yes with diverted-to number and name when available	5000 F	5000 H6000 F Cathy

CLID and Name Display options

Terminating Number Notification rules:

- The terminating number (received in the Diversion Notification information delivered by the diverting telephone's node) is displayed on the originating user's telephone:
  - if the received presentation information (received in the Diversion Notification information issued from the terminating node) indicates that presentation is allowed.
  - if the diverting telephone's subscription option (received within the Diversion Notification information from the diverting telephone's node) allows it (see Table 67).

Terminating Name Notification rules:

- The terminating name, when available (optionally received as part of the Diversion Notification information delivered by the terminating node), is displayed on the originating user's telephone:
  - if the intrinsic name presentation (received in the Diversion Notification information issued from the terminating node) indicates that presentation is allowed.
  - if the diverting user's subscription option (received within Diversion Notification information from the diverting node) allows it (see Table 67 on page 638).

  Note that Dialed Name Display allowed (DNDA) or denied (DNDD) functionality has no impact on the originating user's notification.

When both originating and served users are on the same node, the existing Meridian 1 treatment is still applicable, which is that the diverting user's subscription options have no impact on the originating user notification.

When both originating and terminating users are on the same Meridian 1 node, and diversion is performed by the Rerouting method, then the diverting user's subscription options are effective for the originating user notification only before the terminating user answers, but no name is provided. As soon as the terminating user

CLID and Name Display options

answers, then the originating user receives full notification (reason, terminating user's number and name), according to the diverting user's subscription option.

Terminating user's notification rules

Diversion Reason Code Notification rules:

- The reason is displayed on the terminating user's telephone (as soon as Diversion Notification Information is received from the diverting node).

Diverting telephone number Notification rules:

- The diverting user's number (the originally-called number) is displayed or not displayed on the terminating user's telephone, according to the diverting user's subscription option (see Table 68 on page 642). If the diverting user's subscription option is to not release the calling party name/number to the terminating user, the diverting user's number may be either displayed with dashes (if received with a restricted presentation) or not displayed at all (if no diverting user's number is received).

Diverting telephone's name Notification rules:

- The Dialed Name Display allowed (DNDA) or denied (DNDD) functionality allows the terminating user to choose the name to be displayed after diversion has taken place:
  - if the terminating telephone has a DNDA class of service, then the terminating user's telephone displays one of the following:

    – the original called name, when available (as optionally received from the diverting user's node, depending on the previous diverting telephone's subscription option)

CLID and Name Display options

– if the original called name is not available, the redirecting name, when available (as optionally received from the diverting user's node, depending on the diverting user's subscription option)

– nothing, if neither of the two previous names is available.

– if the terminating telephone has a DNDD class of service, then the calling user's name is displayed, if available, otherwise nothing is displayed.

When both diverting and terminating users are on the same node, the present Meridian 1 treatment applies, that is, the diverting user's subscription options have no impact on the terminating user notification.

When both the originating and terminating users are on the same Meridian 1 node and diversion is performed by the Rerouting method, then the diverting user's subscription options have no impact on the terminating user notification. In this case, DNDA functionality is not supported.

Table 68 sums up the terminating user's Notification rules, according to the diverting user's subscription options and the terminating user's class of service (DNDA/DNDD).

Note that, in the case presented in the example, the originating user's name is Abby, with a DN of 4000. The diverting user's name is Bobby, with a DN of 5000. The terminating user's name is Cathy, with a DN of 6000

CLID and Name Display options

Table 68
Terminating user’s notification according to the diverting user’s subscription option and terminating user’s class of service for single diversion

Diverting user’s subscription option: “Diverting user releases the number/name to the terminating user”	Terminating user’s display (after receipt of diverting user’s Diversion Notification information) depending on the terminating user’s class of service:	
	DNDD	DNDA
No	H4000 H---- F Abby OR H4000 F Abby	H4000 H---- F OR H4000 F
Yes	H4000 H5000 F Abby	H4000 H5000 F Bobby

Multiple diversion Notification rules

This section describes originating and terminating user’s notification rules, for multiple diversions in a pure QSIG environment.

Originating and terminating user’s Notifications are detailed for the following scenario: “Station A calls Station B. Station B diverts the call to Station C using one of the features: CFU, CFB or CFNR. Station C has activated diversion to Station D, which itself has activated diversion to Station E. Station E has activated diversion to Station F. Station F answers”.

CLID and Name Display options

Originating user's notification rules

Diversion Reason Code Notification rules:

- The same rules apply as for the single diversion case (refer to page 637). The last diversion reason will replace the previous one.

Terminating Number Notification rules:

The terminating number (as received in the last Diversion Notification information message issued from the last diverting node) is presented to the calling user, if *all* of the following conditions apply:

- all previous Diversion Notification information messages received at the originating node contain a subscription option with a value of "Yes with diverted-to number/name"
- any previous Diversion Notification information message issued from the terminating node contains a presentation indicator that allows presentation

The final termination number replaces a previous one.

Terminating Name Notification rules:

The terminating name, when available (as optionally received within the Diversion Notification information message delivered by the terminating node), is presented to the calling user if *all* of the following conditions apply:

- all previous Diversion Notifications information messages received at the originating node include a subscription option with a value of "Yes with diverted-to number/name"
- the intrinsic name presentation is allowed

The last termination name will replace the previous one. Note that DNDA/DNDD functionality has no impact on the originating user's notification.

Terminating user's notification rules

Diversion Reason Code Notification rules:

- The same rules apply as for the single diversion case (refer to page 640).

CLID and Name Display options

Diverting Number Notification rules:

- Either the originally-called number (as soon as a Diversion Notification information message is received from the last diverting node) is displayed or not displayed on the terminating user's telephone, according to the first diverting user's subscription option.

Diverting Name Notification rules:

- The same rules, including the DNDA/DNDD functionality, apply as for the single diversion case (refer to page 640).

Table 69 sums up a multiple diversion case considering the different parties' subscription options, and the terminating user's class of service.

Note that, in the case presented in the example, the originating user's name is Abby, with a DN of 4000. The first diverting user's name is Bobby, with a DN of 5001. The second diverting user's name is Billy, with a DN of 5002. The terminating user's name is Cathy, with a DN of 6000.

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Table 69
Terminating user's notification with respect to the diverting user's subscription option and terminating user's class of service for multiple diversions

Diverting user's subscription option: "Diverting user releases the number/name to the terminating user"		Terminating user's display (after receipt of Diversion Notification information) depending on the terminating user's class of service:	
Bobby	Billy	DNDD	DNDA
No	No	H4000 H---- F Abby OR H4000 F Abby	H4000 H---- F OR H4000 F
Yes	No	H4000 H5001 F Abby	H4000 H5001 F Bobby
No	Yes	H4000 H---- F Abby OR H4000 F Abby	H4000 H---- F Billy OR H4000 F Billy
Yes	Yes	H4000 H5001 F Abby	H4000 H5001 F Bobby

CLID and Name Display options

QSIG Call Transfer Notification

On QSIG links when calls are transferred, names and numbers and a reason for redirection code is displayed on the originating and terminating telephones. Refer to the *QSIG* section "Call Transfer Notification" on page 539 for more information about the support of this feature in X11 Release 24.

EuroISDN - Optional sending of last forwarding DN as CLID (Release 23)

Refer to the *EuroISDN* section, page 439 for more information about this CLID-related feature supported on EuroISDN networks.

ANI features

Refer to the R2MFC section of this book for more information on the ANI-related features provided on country-specific connectivities.

Control tips

- Use the CLID Enhancements capability carefully to provide a meaningful CLID to called parties. This CLID should be a number that the called party can dial to reach a number that is the appropriate one for their calls. You can control the DNs that will receive calls if you manipulate the CLIDs to best suit your needs.

Administration tips

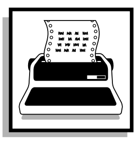
- Do not get the ISDN CLID-related prompts in the Customer Data Block confused with those in the translation table. The HNPA in the Customer Data Block is used for ISDN CLID purposes, whereas the HNPA in the translation table is used for call translation and processing purposes. You can have something entered in the Customer Data Block that is not entered in the translation table and vice versa.

- CLID in CDR (CCDR) is a very valuable software package when you administer a node. If you install this CDR package, the calls made by remote users on the trunks at the node will be printed in CDR records at the node.
- If you program the same reason for redirection codes (with Call Party Name Display software) at all locations, users who move from site to site will be familiar with the codes they see on their displays wherever they go.

Training tips

- If the caller's CLID was not available because the incoming call was not routed entirely on ISDN trunks, the called party will see the incoming trunk route access code and member number, instead of the CLID.
- The MCDN format for CLID displays the letter "H" in front of UDP or CDP private network calls. Train users to understand what the H means.
- Tell users about the meaning of dashes and XXXX on the display. When they receive calls from users who are blocking names and numbers, their own displays will look like that.

Maintenance tips



- You may want to program some telephones to send department names and not the user's own name. This is easier to maintain since you will not need to do so much programming when employees leave jobs and new employees start. It is also another way to deal with providing the calling party with some privacy.

CLID and Name Display options

What to have ready

The following checklist summarizes the steps you should take before implementing CLID and name delivery features.

Table 70
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Determine the formats of the CLIDs that will make the most sense to called parties on your network.
✔		Assess the ESN software packages equipped at different sites. Decide if you will need to change any default call types with digit manipulation tables.
✔		Work with your programmers to use INAC properly.
✔		Investigate whether you have users who do not want names/numbers to be presented.
✔		Do you have any other reasons to prevent names or numbers from being sent?
✔		Are there users who need to use the CLID enhancement capabilities to change the default CLID set up?
✔		Are there users who should be named with a general department name instead of a personal name?
✔		Decide if you want to use FFCs to block or unblock the sending of names and numbers. Decide what type of name and number restriction suits your users best and determine if you can use it on the type of ISDN network you are using.

— continued —

CLID and Name Display options

Table 70
Checklist (Continued)

Basic	Optional	Preparation
	✔	If redirection features are operating, decide what reason codes you want and whether users want to see the caller’s name or the name associated with the redirecting telephone.
	✔	If your system supports CLASS telephones, implement name and/or number display as each user requires.
	✔	Decide if you want or need a prefix code added to the display of the CLID. Check that you have the proper kind of connectivity to do this.
	✔	If you are in Australia, decide if you want to use billing numbers on incoming trunks.

CLID and Name Display options

Attendant-related features

In an ISDN environment the capabilities of the attendants must be extended across a network of systems. Many functions and features have been developed to ensure that attendants can operate efficiently in an ISDN network.

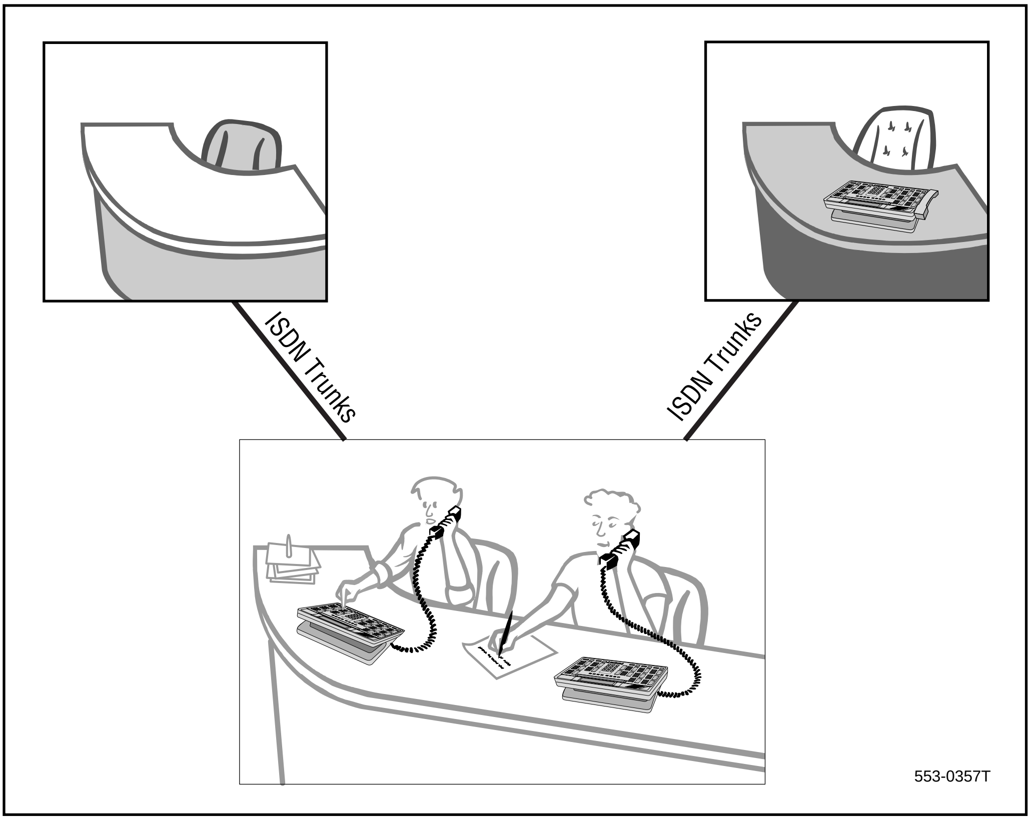
The benefits of having centralized attendants include:

- management of a unified group of operators
- training of a unified group of operators
- fewer staff
- less equipment required
- small sites can benefit from the services of an attendant
- organized night service

Information in this section covers attendant-related features on MCDN networks. There is information on DPNSS1 attendant features in the section called *DPNSS1*.

Attendant-related features

Figure 76 - Network Attendant Service



Attendants at one site can serve sites with no attendants or part-time attendants

Basic configuration

In a non-ISDN network there is an application you can use called Centralized Attendant Service (CAS). Analog or digital trunks programmed as Release Trunks (RLTs) are required for use between the switches. Refer to the *Software Feature Guide* for more information about Centralized Attendant Service. For a comparison between CAS and the ISDN application called Network Attendant Service (NAS), refer to the Network Attendant Service section in the *Networking* NTPs.

Network Attendant Service (NAS)

NAS is an ISDN application that is based on MCDN protocols. It can be supported using PRI and ISL configurations between switches.

Any node in the NAS network may have its attendant services located on any other node in the network on a full-time or part-time basis.

The attendant consoles may be equipped with NAS keys that cause the feature to be activated. Call routing is based on time-of-day schedules and alternate route choices that have been pre-programmed. Refer to the Network Attendant Service section of the NTPs for the details on how these parameters interact.

You will need to program NARS or CDP route list indexes for the alternate routing to remote attendants to work.

Drop Back Busy

If Drop Back Busy is activated at an originating switch and it attempts to route a call to a remote attendant but there is none available at the site, and the remote site is going to use NAS alternate routing or Night Service on the call, the originating switch will select another alternative instead.

Attendant-related features

Off-Hook Queuing

If all routes to a remote site are busy and OHQ is activated at the originating site and on the routes to that alternate site, the call will be queued, until the OHQ timer expires. If no route is found during this time, another site will be tried. If you keep the timer short, the caller will not be delayed too long waiting for an attendant.

Incoming Call Indicators (ICIs)

Incoming Call Indicators tell the attendant the origin of a call. This helps attendants to answer calls more intelligently and the ICIs allow the attendants to give priority to certain types of calls, if they are trained to do so.

There are special ICI key designations for NAS attendants. If the attendant is answering a call, as an alternate attendant, the ICI key shows what kind of trunk the caller is using when calls are presented. The caller is using a trunk that terminates on a different site before being routed across the ISDN trunk to the alternate attendant. The choices are:

- NDID
- NCO
- NTIE
- NFEX
- NWAT

Trunk Group Busy keys (TGB)

TGB keys can only be used to busy out trunks at the site where the attendant physically resides. They indicate the busy/idle status of trunks at the local site.

Break-in

This feature allows an attendant to enter an established connection anywhere in the NAS network. Attendants can use this feature to give important information to one of the parties in the call or to pass an important call to one of the users.

Attendant-related features

You must assign a Break-in key to consoles if the attendants are supposed to have this capability. Users who do not want the attendant to break in on established calls must have the Camp-on feature denied in the class of service of the telephone.

Idle Extension Notification

Instead of breaking in on an existing call, the attendant may want to have the system notify the console when the called extension becomes available. This feature will notify attendants in an MCDN network environment when extensions across the network become idle.

Call extension (Attendant Control)

If NAS Attendant Control is activated in the Customer Data Block, any telephone called by the attendant cannot release the connection, only the attendant can release.

Timed reminder recalls

Recalls occur if calls are not answered after ringing, during Camp-on or during Call Waiting. When an external trunk call is extended by an attendant to a user at a remote site, the trunk to trunk connection stays active even if the trunks are ”tromboned”. This is done so the recall feature can monitor the call, in case it does not get answered. If there is no answer, the call will return to an attendant or the same attendant (if Recall to Same Attendant has been activated). The Recall to Same Attendant feature must be activated in the Customer Data Block of the switch where the attendant who answers the call resides.

You can program anti-tromboning to take effect after the call is answered. This is recommended. Activate it on the trunk routes between the switches where tromboning will occur.

Gateways and Interworking

If your MCDN network interfaces with a DPNSS1 network, and if the anti-tromboning capability of NAS has been activated, the Route Optimization capability of the DPNSS1 network will not occur along with the anti-tromboning of the MCDN connections.

NAS attendant can handle and route calls coming in on EuroISDN -Network Side trunks but NAS signaling is not supported.

Attendant-related features

Recall with Priority during Night Service

Use this feature if you want to place calls in a priority ranking when they are queued to a Night DN when using NAS. The order of priority is:

- recall of an external call
- a new external call
- other calls

Equi-distribution Routing, Network Attendant Service

This feature routes NAS calls using a loading factor algorithm. As routing to alternate attendants is attempted, if they are all busy, this feature reads the load level at each location. The call is routed to the alternate with the lowest load factor.

The load factor is calculated using an efficiency factor programmed in the Customer Data Block multiplied by the number of calls waiting for an attendant divided by the number of attendants in service.

Network-wide Listed Directory Number

DID numbers or DNs can be configured as LDNs on the attendant console Incoming Call Indicator keys. When an external DID call to that number or an internal user dials the DN, the call is presented to the proper ICI key. Network-wide application of this feature expands the available LDNs from four to six keys.

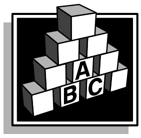
You must configure the same LDNs at more than one site, if you expect the attendants to answer the calls efficiently.

Attendant Blocking of DN

Using this feature, attendants are able to make a telephone busy to incoming calls and unable to be used to make calls. The attendant would do this if the user requested the attendant to make a call and the attendant does not want the telephone to be occupied when the call is established, ready to be extended back to the user.

Telephones that can be affected by this feature can be found anywhere in an MCDN NAS network.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Attendant and Network Wide Remote Call Forward

If you want the attendants to be able to forward telephones to other telephones in the network, use this feature. They will need to dial a Flexible Feature Code and a password to activate and deactivate the Call Forward All Calls feature for telephones.

You will need to ensure that the Private Network Identifiers (PNIs) at all sites are the same.

Call Park Network Wide

This feature is useful if you want attendants to park calls at one site and have users retrieve them from another site in an MCDN network. The opposite is true as well. The attendant can park a call on a Parking DN at another site. The user must be informed of the Parking DN number to dial, in some way, in order to retrieve the parked call. You can use Call Page Network Wide to do this.

The parked call will recall to the attendant who parked it, if it goes unanswered.

Users can retrieve parked calls using MCDN TIE trunks or DID trunks. Calls parked on individual parking DNs cannot be retrieved using an external trunk; only calls parked on System Park DNs can be retrieved this way.

NAS software must be equipped at the originating site and the site where parked calls will be answered. If the call must tandem through other switches, those switches do not need to have NAS installed.

The users and attendants use NARS or CDP to dial the proper parking DN numbers.

Attendant-related features

Call Page Network Wide

This capability allows attendants and telephone users to access paging systems at other switches. You do not require ISDN or NAS software to do this.

TIE trunks and VNS trunks may be used.

You can control which users will be able to access the paging systems to prevent their unauthorized use.

Intercept Computer Dial from Directory

The Intercept Computer (ICP) is offered outside North America only. The ICP works with the attendant console to make the attendant more productive and efficient. Information about the caller or redirecting party is displayed on the ICP screen as calls are presented.

When the attendant wants to extend a call, the name or number requested can be reached by accepting the choice on the screen. The attendant does not need to dial the number.

CDP dialing plans are supported. There is a seven digit limit to the numbers that can be received and stored in the ICP device.

Radio Paging Improvement

Prior to X11 Release 20, the Radio Paging (RPA) feature was supported on stand alone systems only. The attendant recall on unanswered pages directed calls to the local attendant on the node where the RPA system was connected.

As of Release 20, the RPA feature can be used with Network Attendant Service equipped in an MCDN network. The product improvement allows an attendant anywhere in the MCDN NAS network to page and the unanswered pages will recall to the attendant who paged, no matter where the attendant is in the network.

Attendant-related features

Attendant Through Dialing Networkwide

When users are restricted, they sometimes ask the attendant to dial calls for them. Instead of having the attendant dial all the digits in the call, the Through Dialing feature allows the attendant to dial a trunk route access code, CDP steering code or ESN access code followed by a number such as a Special Number and let the user finish dialing the call.

With Release 23 software this feature can be used in a network application. It is not dependent on NAS software.

To use the feature, the user calls the attendant located on another node using a TIE trunk. The attendant can connect the user to any trunk, except a DPNSS1 or VNS trunk. A requirement is that if a Trunk Steering Code or Special Number is dialed by the attendant in the course of extending the call, the trunk must support Overlap Sending. Also, the attendant cannot simply dial AC1 or AC2 and allow the user to dial the remaining digits in a NARS call. The attendant must dial enough digits for the call to be translated (a Special Number for example) before extending the call to the user.

CDR records print out at the user’s node and the attendant’s node. They can be set up to show the user’s DN and the number dialed. Use these records for control and billing purposes.

Control tips

- It is recommended that you implement controls on users to prevent unauthorized access to network-wide paging.
- Investigate the amount of time users will require at each site in order to answer calls from the attendants. Keep the recall timers as short as you can to minimize the time that trunks between the sites may be in a tromboned condition, waiting for an answer.

Attendant-related features

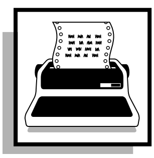
Administration tips

- If you plan to implement Off-Hook Queuing on busy inter-switch trunks, keep the queuing timer short. This will prevent lengthy delays for callers.

Training tips

- Assign ICIs that will help attendants answer calls with as much knowledge about the incoming caller as possible. Train the attendants to use these indicators to answer the calls appropriately or to prioritize calls as you see fit.
- Train attendants to understand what conditions warrant a break-in on an existing call. Survey users periodically to find out if the attendants are using the Break-in feature properly. If not, you may want to implement Idle Extension Notification instead and remove the BKI keys.
- When the user changes the password they must remember to inform the attendants, if you are going to allow attendants to remotely Call Forward telephones.

Maintenance tips



- Train attendants to tell you when Trunk Group Busy keys flash. When the TGB keys flash, it means the trunk group that the key represents is busy. It can help you identify when trunks are not working or when it is time to order more trunks, preferably after doing a traffic study.

What to have ready

The following checklist summarizes the steps you should take before implementing attendant-related features.

Table 71
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Decide which attendants are to act as backup for each other and in what order they are to be scanned. Take time zones into account since Night Service will come into play.
✔		Which NAS ICI keys does each attendant need for the calls that they will receive?
✔		Decide if you want Drop Back Busy/ Remote Virtual Queuing and/or Off-Hook Queuing.
✔		Determine which routes could have tromboned connections based on the alternatives you are planning to use. Activate anti-tromboning on the proper routes.
✔		Find out which users do not want to be broken in on by the attendant. Assign a warning-tone-denied class of service to these telephones.
✔		Decide if you want Attendant Control activated.
✔		Decide how long you want your recall timers to be.
	✔	Decide if you want to activate Recall to the Same Attendant.
	✔	Ensure the PNIs at all sites are the same if you are activating Attendant and Network Wide Remote Call Forward.

— continued —

Attendant-related features

Table 71
Checklist (Continued)

Basic	Optional	Preparation
	✔	Do you want to activate Recall with Priority During Night Service?
	✔	Do you want the system to use equi-distribution when routing NAS calls? If so, discuss an efficiency factor setting for each site with your system supplier.
	✔	Assign Network LDNs at any sites that answer these calls.
	✔	Will the attendants make calls for users? Do you want them to use Attendant Call Blocking?
	✔	Will the attendants place calls for users at other network locations? If so, consider implementing Attendant Through Dialing Networkwide.

Restriction-related features

There are many ways to restrict users or trunks from making certain calls on an ESN or ISDN network. Many of these capabilities are covered in the *NARS and BARS* section of this book. Refer to the information on class of service, TGAR, Network Class of Service, Supplemental Digit Restriction, Incoming Trunk Group Exclusion, and Free Calling Area Screening.

You can also read sections of the *Software Feature Guide* to find out about features such as Scheduled Access Restrictions, Trunk Barring and New Flexible Code Restriction. As of X11 Release 23, the Flexible Trunk to Trunk Connections feature affects trunk connections involved with transfers and conferences. Your system supplier has information on this feature.

The book called *Controlling Access Privileges (P0735064)* is a handy reference book that covers the restriction capabilities provided by Meridian 1 systems.

The information that follows covers two specific restriction-related ISDN features:

- Electronic Lock Network Wide/Electronic Lock on Private Lines
- Call Connection Restriction

Restriction-related features

Basic configuration

Planning and implementing restrictions can be one of the most important parts of managing a network.

You must consider that internal and external users will attempt to make all kinds of calls that you may not want them to make. Unauthorized calls cost you money in extra long distance charges, unproductive employee time, and extra usage on trunks. In order to control calling patterns, gather complete information about where users need to call for business reasons and determine the best possible ways to route these calls.

You must also control the operation of trunks so connections are made with the best transmission and signaling possible.

Electronic Lock Network Wide/Electronic Lock on Private Lines

The Electronic Lock feature was first introduced for stand alone systems. It allows users to change the class of service of a telephone to a customer-wide Controlled Class of Service by dialing a Flexible Feature Code and a password and the DN of the telephone to be locked. Users do this if they want to prevent other people from making calls on the telephone when it is unattended. The user can cancel the Electronic Lock feature and bring it back to its normal class of service with a Flexible Feature Code.

With the Network Wide version of the feature the telephone that is locked takes on a new customer-wide NCOS value. A remote user who is locking the telephone dials a Flexible Feature Code followed by a password and the UDP or CDP DN of the telephone to be locked. Since one switch cannot find out the correct password length programmed at another switch, you must set the same password lengths at all switches, if you want to use this feature network-wide.

Restriction-related features

If you have Network Signaling and the NCOS value is travelling with users' calls from one switch to another, the Electronic Lock NCOS will travel with a locked telephone user's calls. This prevents users from making calls using TIE trunks that are blocked at the home switch.

You can activate Electronic Lock on private lines that terminate on telephones. The Controlled Class of Service (stand alone switch application) or the Electronic Lock NCOS (network-wide application) of other DNs on the telephone will take effect on the private line.

Call Connection Restriction

This feature allows limiting conditions to be placed on call connections across the ISDN network. Call configurations which would degrade transmission integrity or network performance are prevented.

The following conditions can be placed on network connections:

- no more than one trunk without disconnect supervision can be used in a call connection. This prevents trunks from locking up when calls are not released upon disconnection.
- tandem nodes are limited, to prevent potential transmission problems. The maximum number of tandem nodes allowed in one connection is programmable between 1-31.
- the number of PSTN switches permitted in one connection can be limited (one or an unlimited number)
- the number of mu-Law to A-Law conversions per connection can be limited to prevent transmission problems (0-31)
- the number of satellite delays per connection can be limited (0-5)

Restriction-related features

What to have ready

The following checklist summarizes the steps you should take before implementing Electronic Lock Network Wide and Call Connection Restrictions.

Table 72
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		If you want to implement Electronic Lock, program the FFC and passwords required. Choose a customer-wide NCOS for locked telephones.
✔		If you want to implement Call Connection Restrictions discuss settings that your system supplier recommends.
✔		Plan training.

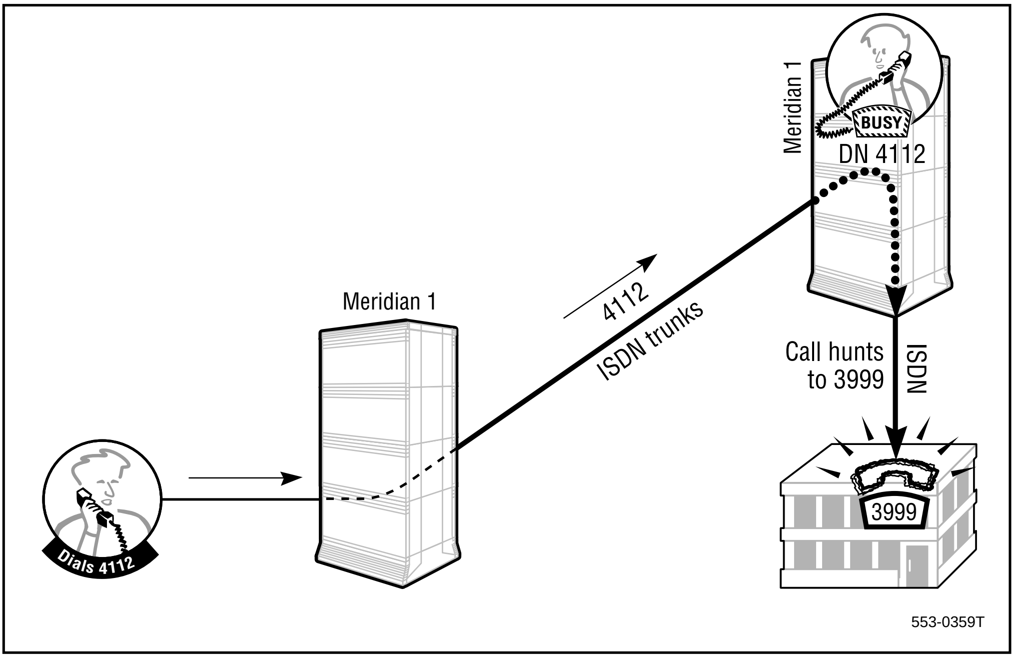
Redirection-related features

Network Call Redirection allows calls to be directed from one Meridian 1 to another Meridian 1 when a called user is not available to answer.

The private network dialing plans supported are:

- Uniform Dialing Plan
- Coordinated Dialing Plan

This capability is supported on Meridian 1 PRI or ISDN Signaling Link (ISL) networks. It can operate when a Meridian 1 is connected to a DMS Centrex on private network TIE trunks also.



Redirection-related features

Basic configuration

On a Meridian 1 system you can program redirection features for each telephone to send calls to other DNs when the user is busy, not answering or forwarded.

The ISDN version of these features allows you to send calls across a network to a UDP (location code) or CDP DN. Users could always use Call Forward All Calls to manually forward calls across a network. With Network Call Redirection the system can automatically do the redirecting when the user is busy or not answering as well. Network Call Redirection also supports Call Pickup and Call Transfer. You can have an ISDN call forward twice on a no answer condition if the telephone that the call initially forwarded to has Second Level Forward No Answer allowed in the class of service.

You should read the preceding sections of this book for the type of network you are planning to implement before you assume you can implement redirection features. Some types of network signaling do not support redirection, at this time.

Network Call Redirection

Redirection counter

You must program each Customer Data Block with a Redirection counter. As a call is going to be redirected to another switch, the system checks the number of times the call has already been redirected against the number of times allowed in the counter. If the numbers are the same, the telephone will continue to ring, if it is already ringing. If the telephone is busy or forwarded, overflow tone will be given to the caller and the call will not be redirected.

For detailed information on CLID and Name Display options, refer to the section in this book on that topic or refer to the NTPs for information on individual features related to those capabilities.

Redirection-related features

Network Call Party Name Display (NCPND)

As of X11 Release 16, names could be sent between Meridian 1 systems, using the Network Call Party Name Display feature. The DN of the calling and called parties must be programmed with a name at the home switch for each. Using an ND1 setting allows up to 24 characters. ND2, introduced in Release 17, only allows 15 characters.

Network Name Delivery (NND)

As of X11 Release 17, names could be transported between Meridian 1 and DMS Centrex switches using the Network Name Delivery feature. Format ND2 was introduced. ND2 only allows up to 15 characters.

If you want names to be transported across the ISDN network, you must activate the feature on the routes between the switches.

Reason for Redirection Codes

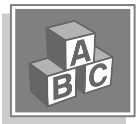
If you want callers to know the reason for the redirection, you can program codes that indicate why the call is going to a different telephone than the one dialed. The codes can be up to four characters long.

For example, if the user dials a busy telephone, the reason code (BUSY) could be displayed on the caller’s telephone.

The redirection codes you can program are listed below with a suggested English language character string for each one:

- Call Forward All Calls (FRWD
- Call Forward No Answer (CFNA)
- Hunt (BUSY)
- Call Transfer (TRAN)
- Call Pickup (PKUP)
  - Directed Call Pickup by Group Number
  - Directed Call Pickup by DN
  - Ringing Number Pickup

Redirection-related features



Note: Some buildings are served by two systems. Some network planners configure their networks in this way for redundancy reasons. If one system fails, half the users will still have telephone service. In this case, it is possible for users sitting next to each other to be connected to two different systems. They may need to answer calls for each other; they may need to belong to the same pickup group and therefore they will use Call Pickup Network Wide.

If you want Reason for Redirection codes to be displayed, you must enable the capability on the ISDN route between switches.

The numbers and names change as the call progresses. For example, as a call is transferred the caller sees the number, and possibly the name of the called person updated to a different number, and possibly a name, once the call is transferred.

Call Forward, Break-in and Hunt Internal or External Network Wide

Call Forward by Call Type can treat internal and external calls differently.

On an ISDN network a network-wide call receives *internal* treatment if one of the following types of information is available:

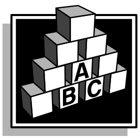
- the Numbering Plan Identifier (NPI) in the CLID is labelled “private”
- if the NPI is not present, the Network Attendant Service (NAS) information will be used, if it is configured
- if the call comes from a QSIG network, the specific data giving information on the far end of the call is used
- if interfacing to a DPNSS1 network, the Calling Line Category is used

If none of these protocols is available to identify the call by type, or the capability has not been activated for the customer group, the Route Class (internal, external) programmed for the incoming trunk route is used.

You can also decide to program certain incoming routes to ignore the network-wide information being sent with calls and always treat all calls coming in on the route as internal or external.

The call type designated by this feature affects Internal Call Forward and Attendant Break-in as well.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Recorded Announcement for Calls Diverted to External Trunks

You can allow callers to hear a Recorded Announcement when their calls are forwarded to external CO trunks on your system. If the forwarding will take some time, the announcement can tell the caller that the call is proceeding and there will be a short delay. The CO trunks can be configured in any of the following ways:

- DTI
- DTI2
- PRI
- PRI2
- analog
- BRI connected to AXE-10 or EuroISDN routes

This feature is supported in network-wide applications, only in an MCDN network.

Call Forward/Hunt Override Via Flexible Feature Code

In MCDN networks, the user can begin a call with a Flexible Feature Code in order to override the Call Forward All Calls, Intercept Computer Call Forward, Call Forward No Answer, Hunting and Make Set Busy features that may be active at a called telephone.

Redirection-related features

Flexible Orbiting Prevention Timer

When telephone A forwards calls to telephone B at another location, endless loops of trunks would be formed if telephone B is forwarded back to telephone A. When you implement this timer, any telephone is prohibited from call forwarding more than one call off-node for the period of time programmed. The range can be from 0-30 seconds.

Attendant and Network Wide Remote Call Forward

There is more information on this feature in the *Attendant-related features* section of this book or in the Networking NTPs.

Non-MCDN networks

QSIG - Call Diversion Notification and Enhancement

In a QSIG-equipped network, if calls are diverted across the network, the displays of the calling party telephone and the final destination telephone can be set up to indicate the reason for redirection and the name and number of the calling telephone or of the diverting telephone. Refer to the QSIG section of this book for more information.

EuroISDN - Optional sending of last forwarding DN as CLID

If a call redirection occurs at the gateway node, the last forwarding DN is sent as a CLID to the terminating telephone. Redirections supported are Call Forward All Calls, Call Forward No Answer, Hunting, ACD Night Call Forward (NCFW) and ACD Interflow.

If the last redirection is due to NCFW or Interflow, and if that redirection is preceded by one or more redirections on the same node, the called number on that node is sent as the CLID rather than the ACD DN.

When the redirection does not occur at the gateway node for calls redirected to a public network Central Office or a private exchange over a standard ISDN connectivity, the outgoing CLID from the gateway switch is that of the first redirecting DN on the last redirecting node.

Redirection-related features

Control tips

CDR Enhancement

When calls are transferred, you will not receive information about intermediate stations involved with a call unless you have the CDR Transfer Enhancement installed. For more information on this feature, refer to the *CDR* section in this book or the Software System Management NTPs.

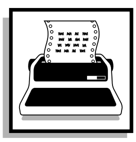
Administration tips

- You can program Night DNs as CDP DNs or UDP numbers if you want telephones at remote switches to answer calls at night.
- Program the same Reason for Redirection Codes at all switches, if possible.

Training tips

- Users on ISL networks may notice a delay in call processing compared to PRI while a call is redirected. Users should be told not to report this as a problem. Call completion time will depend on the speed of the ISL D-channel you are using.

Maintenance tips



- If users report that telephones ring and do not forward it is possible the ISDN Redirection Counter was exceeded and the call could not forward again. It can also mean that the redirection DN programmed was not in the CDP or UDP format (including AC1 or AC2).

Redirection-related features

What to have ready

The following checklist summarizes the steps you should take before implementing redirection-related features.

Table 73
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Find out if the network you are planning or working with can support redirection-related features.
✔		Determine if you want names to display with redirected calls.
✔		Decide on the format (ND1, ND2,or ND3) that you prefer.
✔		Find out which of your users needs to see the name of the called party instead of the name of the caller on redirected calls.
✔		Decide on your reason for redirection codes and make them the same across the network, if possible.
✔		Decide on the Redirection Counter at each switch.
✔		Determine if you want to differentiate internal and external calls. Determine what you will need to do on your network to make it work to suit your needs.
	✔	If you will allow some users to override redirection features, assign a FFC for this feature.
	✔	If you will implement Attendant and Network Wide Remote Call Forward, assign an FFC for the feature and passwords to telephones.
	✔	Decide if you want more CDR information for transferred calls. If you do, plan to install the CDR Enhancement.

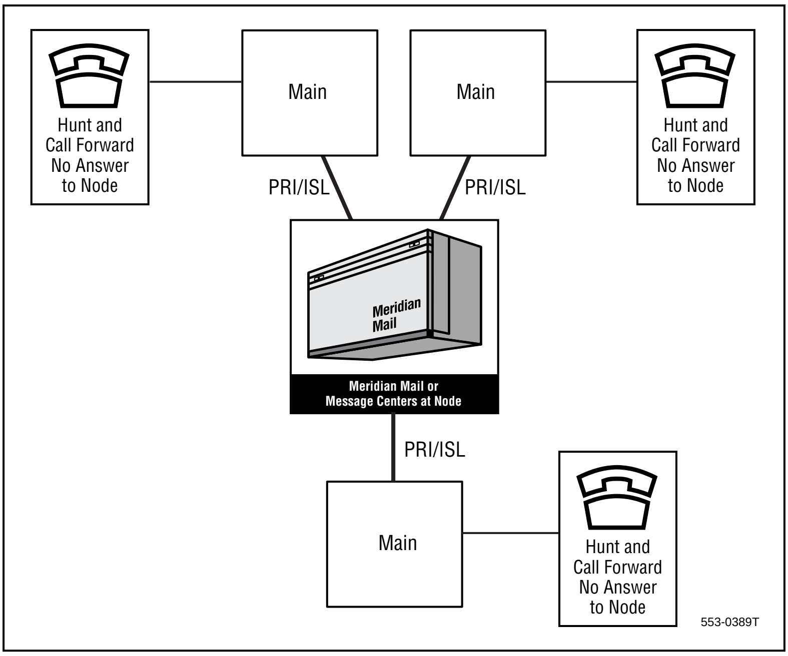
Network applications

Network-wide use of one voice mail system and network-wide Automatic Call Distribution fall into the applications category.

Network Message Services (NMS)

With Network Message Services, messages can be taken at a node for users at several remote sites as well as users at the node where the Meridian Mail or message center is installed.

Because of the ISDN network platform, the voice mail and message centers are able to identify the user for whom the message must be taken. When that user calls in to retrieve messages the ISDN D-channel messages allow proper handling of the calls. Message Waiting indications are activated as a result of the ISDN and Message Center software packages at both sites interworking properly.



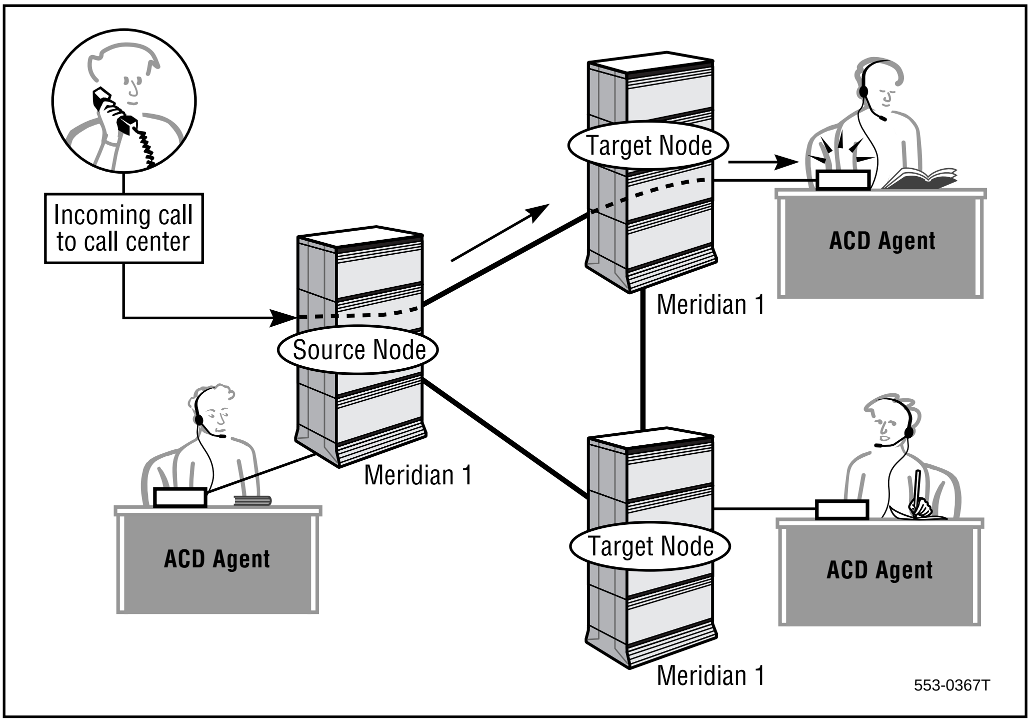
Network applications

Network Automatic Call Distribution (NACD)

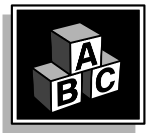
Where there are Automatic Call Distribution (ACD) or Call Center agents working at several sites in a network answering calls from their local calling areas, extending their capabilities network-wide can be a real advantage.

When a group of agents at one site becomes very busy and calls are not being answered as quickly as they should be, NACD uses ISDN signaling to query other sites to find available agents. The agents receive pertinent information about the original source of the call so the call can be answered properly.

In this way, several groups of agents can answer calls as if they belong to one group, taking advantage of the efficiency and the economy of scale.



Basic configuration



The information included in this section is an overview of the services and functions that are part of NMS and NACD. If you want detailed information about the network configuration and software requirements for these applications, refer to the NTPs on *Networking, Meridian Mail,* and *Automatic Call Distribution.*

Network Message Services

There are two applications provided by this software:

- Network Message Services - Message Center
- Network Message Services - Meridian Mail

These applications allow messages to be taken at message center telephones or consoles, and by voice mail, respectively, at one switch in a network of switches.

These applications can result in huge cost savings for you since you do not need message centers or voice mail at each location in the ISDN network, one centralized location can provide the service to all users.

To make this work, the users at the remote locations take advantage of the network redirection features, forwarding and hunting calls to the message center or voice mail DNs at the central location. The Hunt DNs and Call Forward No Answer DNs must be programmed as UDP or CDP DNs.

Message Waiting Indication (MWI) is also provided between the two systems. MWI provides the user with notification that there is a message waiting. This can be a lamp indication on a Meridian 1 proprietary set or an audible tone (interrupted dial tone) on an analog (500/2500 type) set or a proprietary set with no lamp programmed.

MWI deactivates the message waiting indication after all the messages have been retrieved.

Network applications

Message Center

A message center operator sees the CLID of the redirected telephone on the telephone display as the call is answered. The caller's CLID may also be presented if the user is calling from a system that is able to send the CLID through the network.

The message center operator simply presses the message indication key to activate an indication at the user's telephone. Because of the ISDN CLID information, the indication appears at the correct telephone, automatically.

Meridian Mail

If Meridian Mail is answering, the system puts the call into the correct mailbox if the CLID information of the redirected telephone has been pre-programmed into the centralized Meridian Mail system as a mailbox number.

The reason the call has been redirected is announced to the caller, who will be told if the called telephone is busy or not available.

Meridian Mail mimics the operation of a message indication key using software. It sends a message, using the D-channel, to the other system to activate message indication at the proper telephone.

When the user at the remote site dials the UDP or CDP DN of the voice mail queue, the Meridian Mail system identifies the user by the CLID and allows access to the proper mailbox for the user to retrieve messages.

Network applications

Network Automatic Call Distribution (NACD)

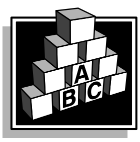
This application allows call centers spread out across an MCDN PRI or ISL network to answer calls coming into any call center node in the network. NACD helps you deal with fluctuating call traffic effectively. As one call center at one location gets too busy, calls are routed to other sites that are less busy.

Using the Timed Overflow routing tables associated with ACD, you specify the ACD-DNs to which calls are to be redirected. The ACD-DNs can be UDP or CDP DNs for call centers at other sites. The numbers must be entered into the routing tables in a CDP or UDP private network format.

Network ACD is cost effective because calls remain in queue at the original source queue while waiting for an agent to become available in the source queue or at one of the target queues located throughout the network. When an agent is guaranteed to be available to answer the call, the call is routed to the site across the ISDN link(s).

The nodes communicate using the D-channel. Switches send messages when agents become available and when queues open or close. The switch with the target queue for the call reserves the idle agent for a call coming from the source queue at another switch.

Improving feature performance



The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Message Waiting Indication Interworking with DMS

If the Meridian 1 is connected to a DMS (Centrex) switch in a private network, message waiting interworking is possible, if the Meridian Mail is on the Meridian 1 or if the voice mail system is on the Centrex.

Network applications

Meridian Mail Trunk Access Restriction

This feature prevents direct or indirect Call Transfer, Conference, Call Join or No Hold Conference connections of external calls with Meridian Mail. External calls are those that originate outside the private network.

When a user with an external call connected tries to complete the feature, the system ignores the user when the Call Transfer or Conference key is pressed the second time. This forces users to remain involved in the connection with Meridian Mail. When the conversation is finished, the internal user controls the disconnect of the call.

Control tips

- Since the centralized voice mail system serves users at several sites, efficient use of memory is an important issue. You will need to manage the use of the Meridian Mail memory closely. Set up rules for users to follow regarding the storage and deletion of messages. Program limits for stored messages that suit their needs without wasting memory. Also, password changes should be managed carefully with security in mind.

Administration tips

- If you implement NACD, you can manage the network queues from a centralized management application called the Network Administration Center (NAC). From this administration center you can monitor and print reports on the operation of all queues. You can even make programming changes to local and remote queues to maintain service levels as situations change. You will need to install ACD MAX hardware and software at all sites you want to monitor using the NAC. Discuss implementation with your system supplier.
- There is a conflict on systems equipped with PRI software and Meridian Mail automated attendant menu services. Attendant Overflow does not overflow calls. Calls remain in the menu services queue, until answered.

Network applications

Training tips

- Message Center operators need training in order to answer calls in the way you want for your company standards. Train the operators to use the display information on their telephones to greet callers properly.
- With NACD, the call center agents at all sites will answer calls as if they belong to the same queue. Train the agents to follow the same procedures throughout your network. Callers should be handled the same way regardless of the location at which the call terminates.

Maintenance tips

- Trunk Route Optimization - Before Answer will not operate when NMS - Meridian Mail handles redirected calls. Calls do not go unanswered. Calls are directed into an ACD queue which answers the calls as they are presented. If the ISDN trunks between sites have been provisioned with TRO factored in, the redirected calls may be blocked due to insufficient trunks, not because the systems or software is experiencing problems.

What to have ready

The following checklist summarizes the steps you should take before implementing NMS or NACD.

Table 74
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Find out if the network you are planning or working with can support redirection-related features. Refer to the redirection-related features section.
✔		NMS - MC: Assess the impact of centralizing message centers. Discuss trunking, telephone set and staff changes with your system supplier.
✔		NMS - MM: Estimate the capacity requirements for your Meridian Mail system, if you centralize it. Discuss the trunk, software and system upgrade requirements, if any.
✔		Arrange to program telephones with the CDP DN or UDP DN of the Message Center or Meridian Mail ACD queue that will answer calls.
✔		Decide if NACD could help you handle fluctuating call volumes across a network. Discuss your service level requirements and timed overflow parameters with your system supplier.
✔		Arrange training.

Optimizing trunks

Features related to redirections and call modifications can sometimes route calls between switches using two or more trunks where fewer trunks would have been possible.

ESN and ISDN networks can be set up to prevent or manage connections for calls that do not use the most efficient routes possible.

Some features work to prevent the inefficient trunk connections before the call reaches its destination and others optimize trunk usage during the call completion phase.

Features related to optimizing the use of trunks are:

- ESN or MCDN networks
  - Network Call Transfer
  - Trunk Optimization (Before Answer)
  - Trunk Anti-Tromboning
- DPNSS1
  - Route Optimization
  - Loop Avoidance
  - Step Back on Congestion
- EuroISDN
  - Trunk Route Optimization - before Answer
- QSIG
  - ANF Path Replacement

Optimizing trunks

Basic feature configuration

MCDN

Network Call Transfer

Network Call Transfer (NXFER) is an ESN feature that improves the operation of the Call Transfer feature between two switches when a call is transferred back to the originating switch using two private network trunks. The regular XFER feature requires two TIE trunks to complete the call. With NXFER, the originating switch connects the calling party directly with the terminating party; the redundant TIE trunks to the transferring user’s switch are then dropped.

NXFER is based on Network Signaling (NSIG). Therefore, NSIG and NXFER software must be equipped at the switches where this kind of transfer may occur. NXFER is supported only on trunks with dual tone multifrequency (DTMF) signaling, a wink start arrangement, and answer supervision.

The benefits of the Network Call Transfer (NXFER) feature include:

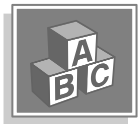
- reduced use of access TIE lines
- improved transmission performance, since TIE lines are not used for the completed connection
- similar operation to the existing Call Transfer feature

> ***Note:*** The user who is transferring the call must *not* wait for an answer before transferring the call or the TIE trunks will not be dropped.

> ***Note:*** This feature is not supported on ISDN networks; other features have been introduced to replace it.

Optimizing trunks

MCDN



Trunk Optimization (Before Answer)

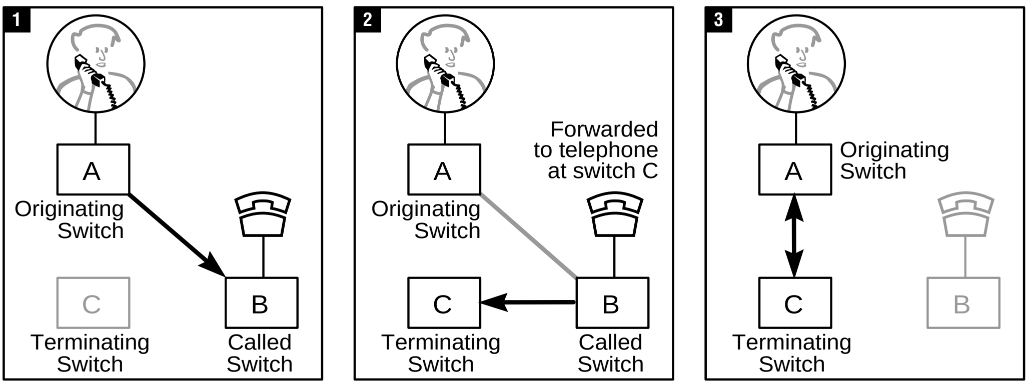
This feature provides the Meridian 1 with the ability to automatically optimize trunks, before the call is answered, for redirected calls. Calls can be redirected by the following features:

- Call Forward All Calls
- Call Forward No Answer
- Hunting

The following example illustrates when Trunk Optimization can be useful.

Example

A telephone user at switch A calls a telephone at switch B. The call is redirected to a telephone at switch C by one of the redirection features. Switch A has direct ISDN connections to switches B and C and the TRO feature has been enabled on those routes.



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Before the call is set up at switch C, messages are sent to switch A from switch B telling it the redirection telephone number, the redirection reason, and the redirection count for the call. If switch A has a trunk available in the first choice route in the route list to switch C, and the redirection counter has not been exceeded, switch A sends

Optimizing trunks

back a message accepting TRO. Switch B sends back a message to release the connection. Switch A confirms the trunk is dropped. Switch A sets up a direct connection to telephone C.

This feature can only operate on PRI or ISL networks. The switches involved must be Meridian 1s (although other systems that can send along TRO messages can act as tandem switches).

The redirection may not occur if the redirecting switch cannot get a response from the originating switch within a fixed 2 second timer.

Mixing UDP and CDP dialing plans is not supported for TRO - Before Answer. Equip all switches with the same software and dialing plans if you want TRO - Before Answer to work.

Trunk Optimization is not supported for the following call types:

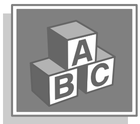
- incoming DID trunk calls at the originating switch
- the call originated from a EuroISDN Trunk-Network Side interface
- the attendant answers the call at the originating switch and releases as the call is extended to a telephone at another switch
- the telephone at the originating switch uses Call Transfer or Conference to set up the call to the telephone at the other switch
- calls to Meridian Mail Auto Attendant, Thru Dialing or Operator Revert services
- ACD Night Call Forward and Interflow Calls because the call is considered to be answered by the ACD queue

Trunks being used by the Meridian Mail Call Sender feature can be optimized by this feature.

MCDN

Trunk Anti-Tromboning (TAT)

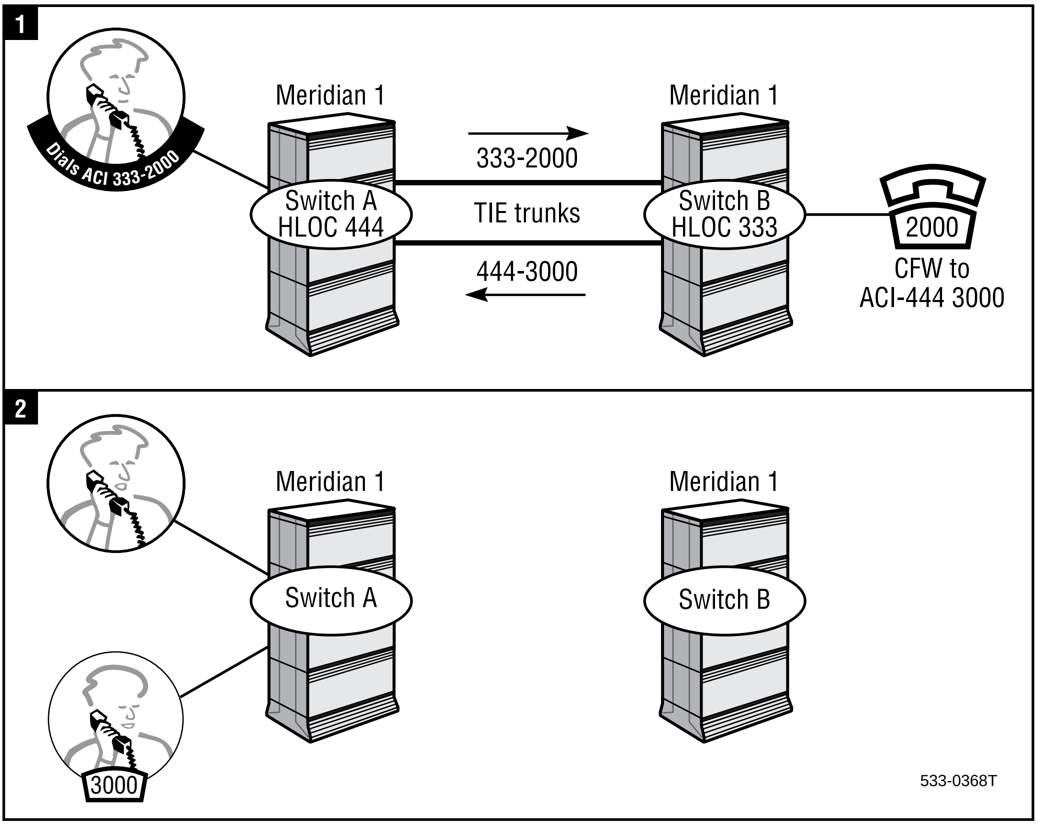
If a caller at switch A calls a user at switch B and user B has activated Call Forward All Calls to another telephone at switch A, a redundant loop of two channels can result. This looping is referred to as “tromboning”.



The caller at switch A does not need the two channels between the switches in order to talk to the terminating telephone at switch A. This feature releases the redundant channels.

The tromboning trunks are released after the terminating telephone is answered.

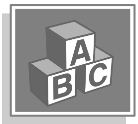
Figure 77 - Trunk Anti-Tromboning



This capability is supported over ISDN PRI, ISL and VNS networks using MCDN protocols. TAT will be performed on VNS BRI trunks but not other configurations of BRI trunks.

Anti-Trombоning will occur between Meridian 1 switches after call redirections (Call Forward All Calls and Hunting for example), and call modifications (Call Transfer, and Conference).

Optimizing trunks



Anti-tromboning will also release redundant private network trunks when a call that has tromboned enters the public network on a COT trunk.

MCDN Release Link trunks provided by an SL-100 or DMS-250 Sprint switch using BCS 32 or higher, or DMS-100 using Release NA007 or higher can use the capability as well. Do not confuse this with Release Trunks (RLTs) used with a non-ISDN service called Centralized Attendant Service (CAS).

The tromboned B-channels must be controlled by the same D-channel. The tromboned PRI B-channels must belong to the same customer group.

TAT will not release channels, if attendants are involved at the two switches.

You must have MSDL cards for the D-channels.

Non-MCDN networksNon-MCDN networksNon-MCDN networksNon-MCDN networksNon-MCDN networksNon-MCDN networksNon-MCDN networks

EuroISDN

Trunk Route Optimization - before Answer

If a call involves a EuroISDN trunk and one or more MCDN trunks between several switches, when a redundant loop of MCDN trunks results from a call redirection, the redundant trunks are dropped and a direct connection remains established between the caller and the final destination telephone. Redirections supported are Call Forward All Calls, Call Forward No Answer, Call Forward Busy and Hunting.

DPNSS1

Route Optimization

Please refer to the *DPNSS1* section of this book.

Loop Avoidance

Please refer to the *DPNSS1* section of this book.

Step Back on Congestion

Please refer to the *DPNSS1* section of this book.

QSIG

ANF Path Replacement

The Additional Network Feature (ANF) Path Replacement Supplementary Service, offered in X11 Release 23, allows an active connection through a QSIG private network to be replaced by a new, more efficient connection after a call modification. The feature has been developed in compliance with ETSI and ISO standards. The service can operate under two conditions:

- triangulation
- tromboning

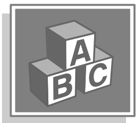
Triangulation occurs when a call from one switch to another is diverted to a third switch. There may be a direct connection that can be established between the caller's switch and the third switch. If so, it will be established, using Path Replacement, after the call is answered. The other routes will be dropped, once the optimum connection has been established.

Tromboning occurs when a call from one switch to another is redirected back to the originating switch by call modifications such as transfer or diversion. When the destination telephone answers, the trunks that are no longer needed between the caller's switch and the diverting switch are dropped. A new connection is established between the caller and the final destination telephone. For this to work, the trunks must both belong to the same customer group.

QSIG Path Replacement does not directly attempt to reduce the number of trunks in a call to a minimum. Rather, it attempts to replace a non-optimum path with an optimum path. The optimum path is

Optimizing trunks

defined as the path which takes the first choice route at all sites involved with the call. The replacement occurs after the final destination telephone answers, not while the telephone is ringing.



There are three different circumstances that act as triggers. The triggers are:

- Call Diversion
- Connected number different from called number
- Congestion

Call Diversion has been explained in the earlier text on triangulation and tromboning. The connected number may be different from the called number in a situation where the caller has dialed a telephone on another switch which cannot indicate redirections to the originating switch. When the call is redirected to a third switch and it sends a connect message to the originating switch, identifying the DN that is answering, the originating switch identifies a mismatch between that number and the called number. This triggers Path Replacement. Congestion can trigger Path Replacement if the first choice trunk route to a particular switch is not available initially to handle a call and the call is routed on an alternate route to another switch in the network. If this switch sends the call to the final destination switch and in the meantime, after the call gets answered, a trunk in the initial choice route at the originating switch becomes available, it can be used for Path Replacement. (Note that Path Replacement does not occur if the congestion occurs at a transit node, not the originating switch).

You can configure each switch that may act as a cooperating switch in a Path Replacement scenario involving other switches in the network to retain its portion of the call connection after the Path Replacement has been invoked at the other locations. This path will be retained, if the connection used the optimum route to reach the rerouted number.

The remote capability must be configured for each route involved with Path Replacement. One or more of the triggers are selected and the retain option must be activated if that functionality is desired. If Path Replacement is chosen but no triggers are configured, Path Replacement activations from a remote switch are processed by the

Meridian 1. If Path Replacement is chosen and triggers are configured, Path Replacement activations from a remote switch are processed by the Meridian 1 and activations can be initiated by the Meridian 1.

Optimizing trunks

Control tips

TRO

Mixed UDP and CDP dialing plans on one network can cause problems for the TRO feature. The redirecting switch will ask the originating switch if it has a direct route to the telephone to which it will redirect the call. The originating switch will check its translation tables based on the number sent in the D-channel message from the redirecting switch. If the dialing plans do not match, there could be translation conflicts. Ensure that you program redirection numbers in the format to which the originating switch will be able to respond.

Administration tips

- When a call is optimized by the TRO feature, the called user's telephone display will not show the dialed name and number.

Training tips

- If you want to use Network Call Transfer, you must train users not to wait for an answer when they transfer calls to other switches.

What to have ready

The following checklist summarizes the steps you should take before implementing the Optimizing trunks features.

Table 75
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Ensure you have the ESN programming done at nodes.
✔		If you want to use NXFER, ensure your TIE trunks are configured for this feature.
✔		If you are implementing NXFER, train users to transfer calls when the destination telephone is ringing.
✔		Ensure the trunk routes will have TRO activated.
✔		Find out if you have a network configuration that will support TAT.
✔		If you have a non-MCDN network, assess the need for the trunk optimization feature(s) offered.

Optimizing trunks

Queuing

When calls encounter a busy trunk group, a busy DN or a DN that is not answered, queuing features can save the users from having to redial until they get through. With many types of queuing the system automatically monitors the trunk group or DN and alerts the user when the trunk or DN is available.

Queuing features can save network administrators money. Users will be able to wait until inexpensive trunks become available and make calls on these trunks rather than on more expensive trunks. You can provision fewer trunks if users are willing to queue occasionally during busy times.

As a network planner, you must calculate whether you want users to wait (queue) or make calls immediately. There are costs associated with both alternatives.

Queuing for busy or unanswered DNs provides assistance to users and is usually a feature that is worthwhile to implement on networks where it can be implemented.

There are types of queuing that occur without users being aware that it is occurring. For example, Step Back on Congestion and Network Drop Back Busy can be used in a network where blockage is encountered and the originating node reroutes calls to avoid the blocked node.

The information in this section covers the various types of queuing offered on the different network platforms. Investigate the ones that are available in your market region. Evaluate their suitability for your network and the users on your network.

MCDN network queuing features

The MCDN network queuing features are listed on the following page.

Queuing

ESN and ISDN

- Ring Again - part of basic system
- Callback Queuing - part of BARS, NARS, CDP
- Priority Queuing - optional software
- Off-Hook Queuing - optional software

ESN only

- Coordinated Callback Queuing - optional software
- Callback Queuing to/from a Conventional Main - optional software

ISDN only

- Network Ring Again - optional software
- Automatic Redial
- Remote Virtual Queuing - optional software
- Network Drop Back Busy - optional software

QSIG and EuroISDN network queuing features

- Call Completion to Busy Subscriber
- Call Completion on No Response

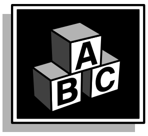
DPNSS1 network queuing features

- Call Back When Free
- Call Back When Next Used
- Attendant Call Offer
- Step Back on Congestion

VNS network queuing features

- Network Ring Again
- Network Ring Again No Answer

Basic feature configuration



To queue or not to queue

You will have to assess whether you want to implement queuing on your network. If the type of queuing you are considering will delay calls, you must evaluate the impact the delay will have on your users and on your business.

You may find that it is worth it to install the necessary trunks for your peak traffic times rather than to provision fewer trunks so that users will be expected to queue at busy times.

If you expect users to queue at times, you will need to set up a good training program to ensure that users understand how to queue. You should also monitor the situation afterwards to assess:

- how much queuing is occurring
- what routes and route lists are blocked
- the amount of time users spend queuing

You can monitor the situation using Network Traffic software to run studies.

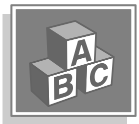
You can survey users to get feedback on what type of calls get blocked. Also, Attendants can tell you if there are blocked trunks by looking for Trunk Group Busy keys on the console that flash frequently.

Be ready to perform refresher training, especially if you find that trunks are blocked at times but there is very little queuing being done.

Refer to *Improving feature performance* and *tips* in this section for further information.

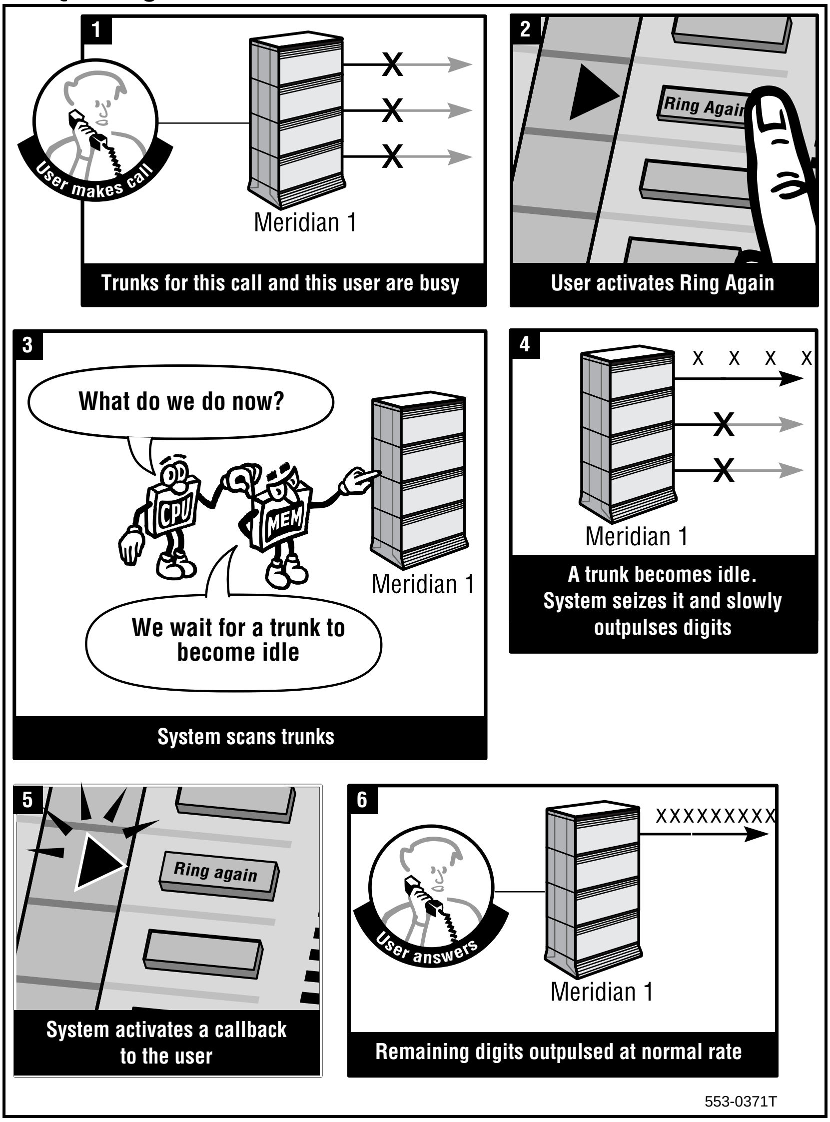
Queuing

Ring Again and Callback Queuing (CBQ)



Callback Queuing is a basic capability of NARS (BARS and CDP). It allows a user who encounters busy trunks to activate queuing in the same way they would activate Ring Again. They dial a SPRE code plus 1 on an analog (500/2500) set, or they dial a Flexible Feature Code, or they press a key on a Meridian1 proprietary set.

Figure 78- Callback Queuing



Ring Again differs from Callback Queuing. Refer to Table 76 for a comparison.

Table 76
Ring Again compared to Callback Queuing

Ring Again	Callback Queuing
The user dials ACOD for the route; if all trunks are busy, congestion tone is given.	The user dials AC1 or AC2 plus all remaining digits; if all *routes* are busy, congestion tone is given.
The user queues for one trunk route.	The user queues for Iset routes in the route list initially until the Extended Route Advance Timer expires.
When a trunk is found, the user is given callback. The ACOD does not have to be redialed but the rest of the digits must be dialed.	When a trunk is found, the user is given callback. The digits in the call do not have to be redialed.
A trunk is seized when the user is being given callback	A trunk is seized when the user is being given callback. Digits are slowly outpulsed until the user answers or the callback timer expires.
The user has 6 seconds to answer the Ring Again callback on analog (500/2500) sets and 10 seconds from Meridian 1 proprietary sets.	The user has 6 seconds to answer the CBQ callback on analog (500/2500) sets and a programmable number of seconds (10 - 30) from Meridian 1 proprietary sets.
Ring Again is assigned in the Class of Service or on a key.	CBQ must pass several eligibility tests and Ring Again is assigned in the Class of Service or on a key.

Queuing

Callback Queuing type affects the routes scanned

Callback Queuing parameters assigned in a user's NCOS not only affect the user after they queue but also when they first scan routes in route lists when they make calls.

There are two kinds of CBQ that can be assigned in a user's NCOS:

- CBQ type I (default)
  - CBQ type I users only scan Iset routes in route lists when they make calls
- CBQ type A
  - CBQ type A users scan all routes in route lists when they make calls

There is more information on the Iset in the NARS section on page 188**.**

Higher level users should be given CBQ type A in their NCOS.

If you want a user group to be given Expensive Route Warning Tone, they must be assigned CBQ type A so they can scan expensive routes.

When users encounter blocked trunks and they activate CBQ, *everyone starts off queuing for the Iset routes*. This is the design intent of Callback Queuing. In this way, queuing can save the business as much money as possible.

The Extended Route Advance Timer (RADT) programmed in the NCOS, determines how long a user will queue for Iset routes only before they can queue for all routes in a route list. You can program this timer in 30 second increments. The lowest possible setting for this timer is 30 seconds. This means that after 30 seconds of queuing for Iset routes, a user can queue for all routes in a route list. If you do not want the timer to expire (to keep the user queuing on Iset routes) set the timer at 0 (default).

Queuing

You may decide to keep certain low level users queuing for the inexpensive routes until a trunk becomes available. You may decide that high level users should begin to queue for all the routes in a route list after a certain number of seconds, if an inexpensive trunk has not become available.

When a trunk becomes available the system still checks the FRL required on the route against the user's FRL. If the user's FRL is not high enough the trunk will not be offered.

If the call ends up going out on an expensive trunk after the user has already queued, there will not be an Expensive Route Warning Tone given.

Callback Queuing Eligibility tests

Before offering CBQ to a call originator, the following eligibility tests are performed by the system:

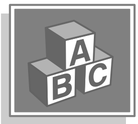
- The CBQ feature must be enabled for the customer group.
- At least one of the routes in the Iset of the route list must be defined as CBQ allowed.
- The user's NCOS must have CBQ allowed, either CBQ(I) or CBQ(A).
- The call cannot be eligible for OHQ. Calls that are eligible for both OHQ and CBQ will be offered OHQ.
- The class of service of the user's telephone has the Ring Again feature allowed and does not have another CBQ or Ring Again call already in the queue.

CBQ and Expensive Route Warning Tone

Users who are able to scan all routes in route lists when they make calls can be given Expensive Route Warning Tone.

This tone is given when there is no inexpensive trunk available in the route list and when they are about to access a trunk in a route that is programmed as expensive.

Queuing



When the tone is given, the expensive trunk is in a seized state, waiting for the decision the user will make. The user can decide to:

- hang on. The call will go out on the expensive trunk, after the Expensive Route Delay Timer expires.
- hang up and try to make the call later. Hopefully, it will go out on an inexpensive trunk then.
- activate Callback Queuing. They will queue for the inexpensive trunk routes.

Callback

When a trunk becomes available for a CBQ call, it is seized to prevent incoming originations during the CBQ call back period. Outpulsing of digits (either those originally dialed by the user or those resulting from digit manipulation) is started at a slow, fixed rate. The system waits 10 seconds before the first digit is outpulsed and 2.56 seconds between subsequent digits.

Network Ring Again (NRAG)

NRAG allows a caller to activate Ring Again against a terminating telephone that is busy at another location on the network.

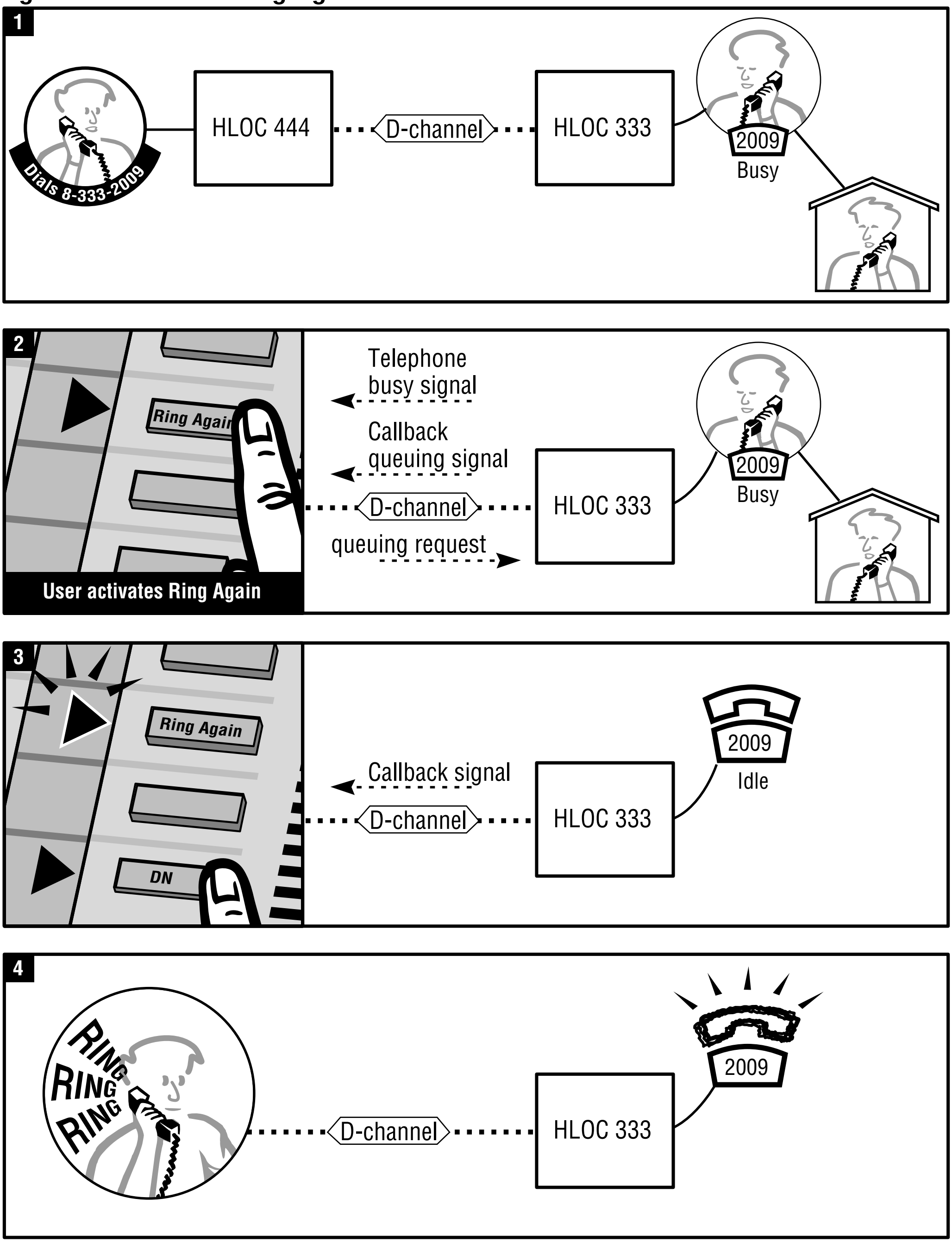
The feature works on MCDN PRI/ISL private networks and in Europe across an ETSI QSIG or DPNSS1 gateway.

When the terminating telephone becomes idle, the terminating system uses the D-channel to signal the originating system to initiate a callback to the originating telephone.

When more than one caller activates queuing against one telephone, the calls are queued on a first-come-first-served basis. When the called telephone becomes idle, the first caller to queue is signaled. The second caller in the queue is signaled after the Queue Advance Timer (set at four seconds) expires.

Queuing

Figure 79 - Network Ring Again



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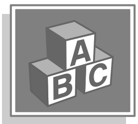
Queuing

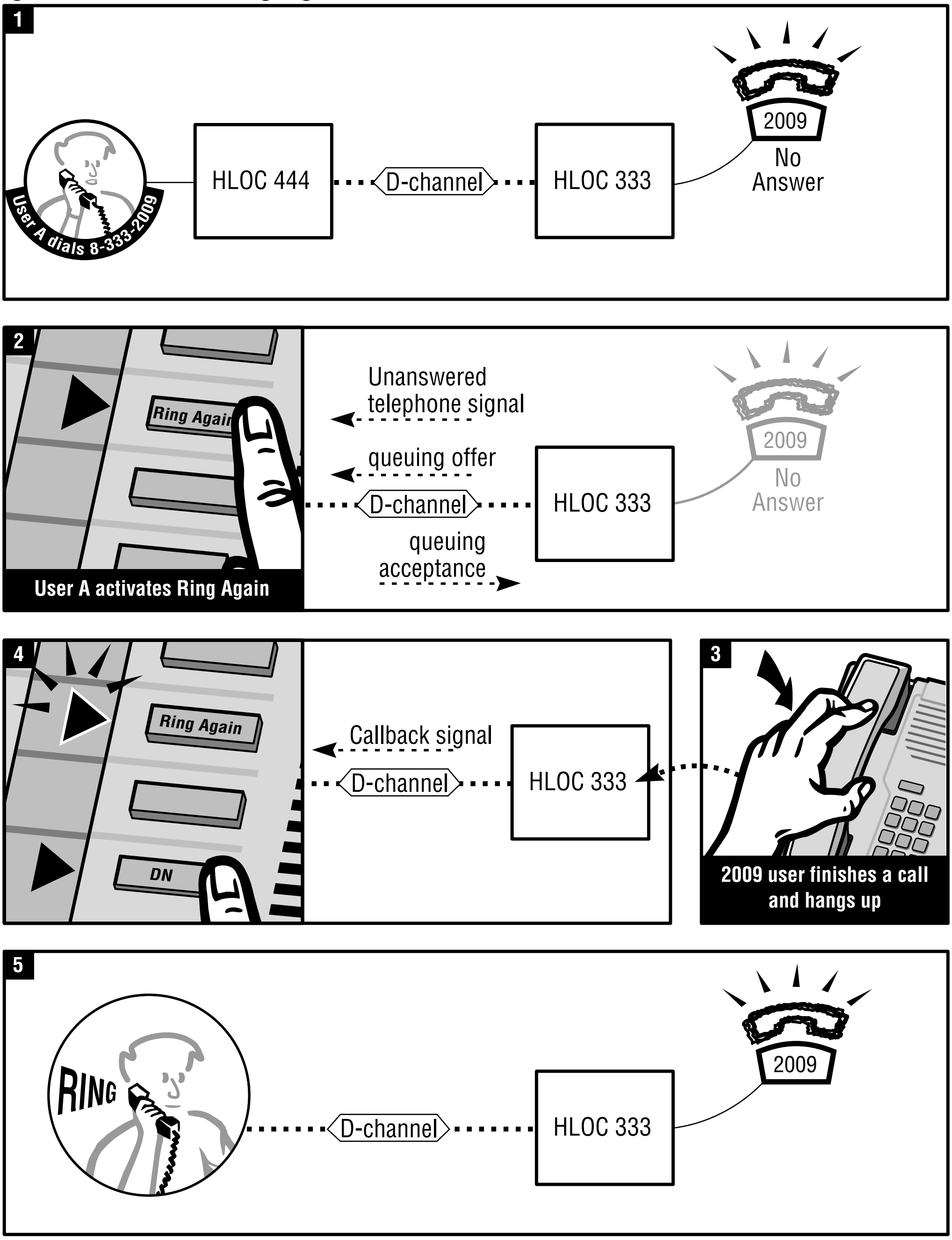
Network Ring Again on No Answer



Network Ring Again on No Answer allows a caller to activate a form of Ring Again when calling a telephone that is not answered. When the unanswered telephone goes off-hook and then on-hook the next time, the caller’s telephone is given a callback.

The feature can be supported globally on ISDN and VNS networks.

Figure 80- Network Ring Again on No Answer



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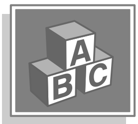
Queuing

Priority Queuing (PQUE)



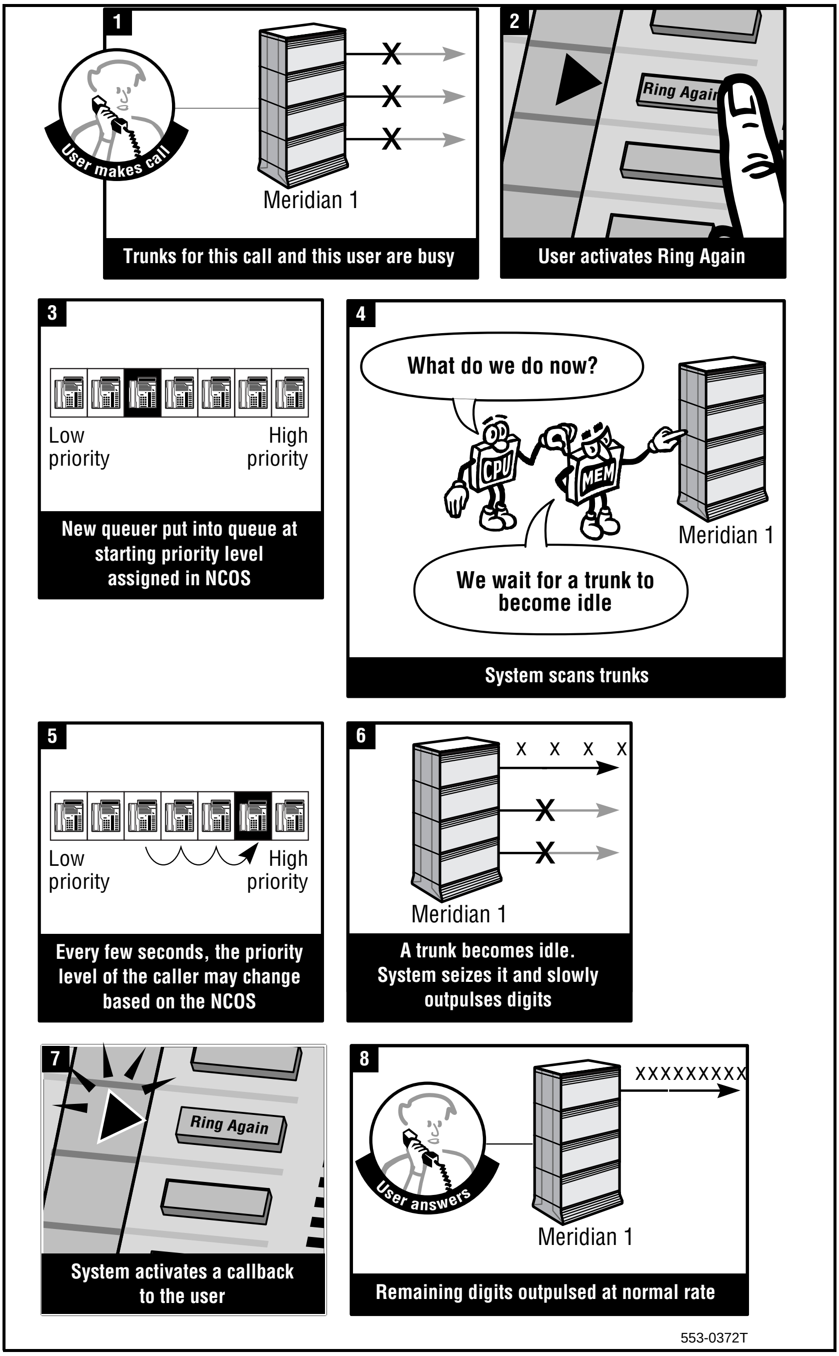
Priority Queuing enhances regular Callback Queuing. If your system has the optional PQUE software package, a maximum priority level (0, 1, 2, or 3) and a starting priority (0, 1, 2, or 3) are assigned to each NCOS group. Zero is the lowest priority level while three is the highest.

Calls are placed in the Callback Queue according to their starting priority level and move forward in the queue (up to their maximum priority level) as their promotion timer allows.

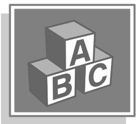
The promotion timer is set in increments of 30 seconds. If you do not want a user group to promote to a higher priority level, leave the promotion timer at zero.

Queuing

Figure 81 - Priority Queuing



Queuing



CBQ interacts with PQUE

CBQ calls are placed in a priority-ordered trunk queue (together with OHQ calls, if any). At the same time, two timers are started: a PQUE promotion timer and a CBQ route advance timer, each with values defined in the originator's NCOS

At intervals defined by the promotion timer, the priority level of the call is incremented until it reaches its maximum priority level.

Each time the call priority is incremented, its position in the queue is advanced. If the route advance timer reaches its maximum value before the call can be terminated on a route in the I set, the extended set of routes is added to the routes that the call is currently queued against.

> ***Note:*** A route advance timer (RADT) set to "0" never expires. The user always queues for I set routes only.

Expensive route warning tone is not given to calls after they have been queued, even if they terminate on expensive facilities.

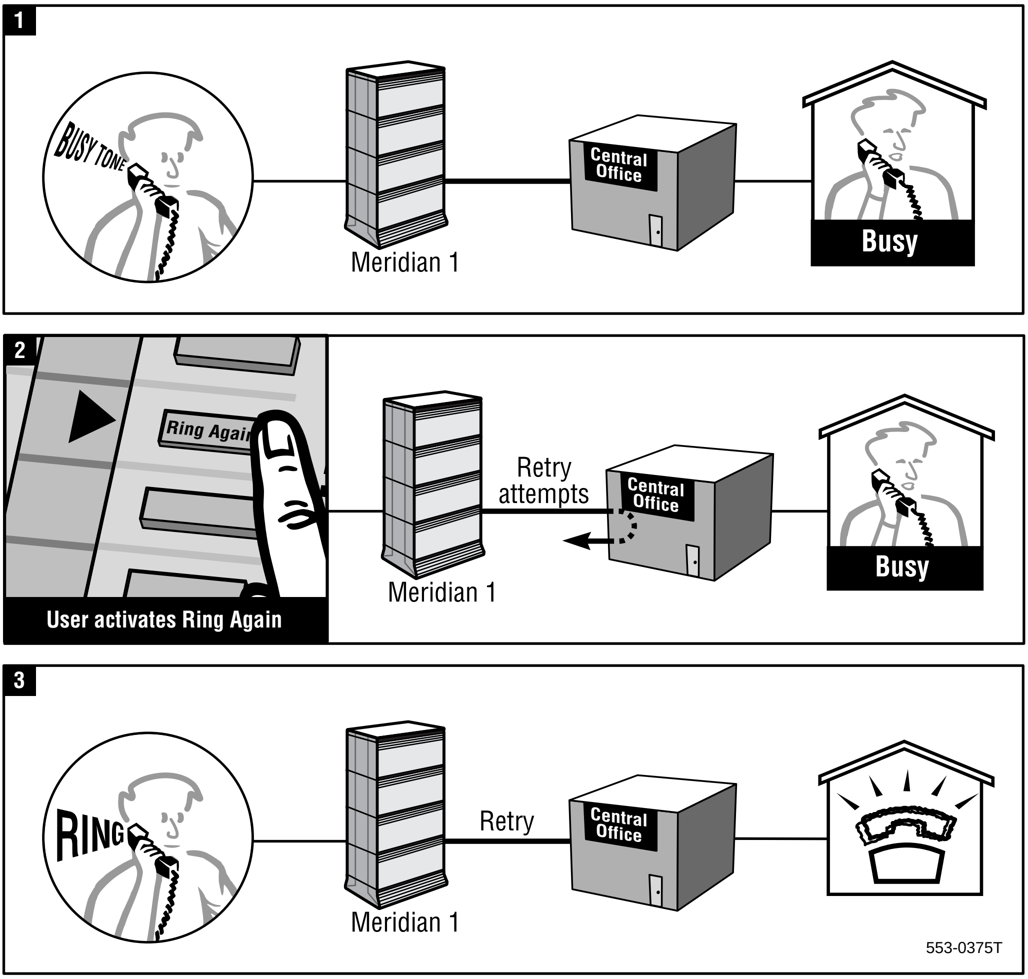
Unless cancelled by the call originator, CBQ calls remain in the queue until they have been offered a trunk; there is no time limit on CBQ calls. Calls can only be routed on routes in the I set or extended set if the FRL in the NCOS is equal to or greater than the FRL assigned to the route in the route list. The priority level does not affect the FRL.

Automatic Redial parameters

If a user dials a public network number and hears a busy signal, Automatic Redial allows Ring Again to be used to redial the call. The system will redial repeatedly until it receives an answer or until it reaches the programmed limit of retry attempts.

Queuing

Figure 82 - Automatic Redial



The NCOS determines whether the user's retries can use Iset routes only or if any route in the route list can be used.

You may decide that if a low level user can only scan Iset routes when they first make a call that the retries will also be restricted to Iset routes. Or you could decide that retries can scan all routes to make it less possible that route blockage would delay the retries even further.

Retries add extra usage to your trunks. Set the number of retries low since the user will not wait a long time for a call to be completed; you probably do not want repeated redial call attempts on your trunks.

Queuing

Coordinated Call-Back Queuing (CCBQ)

The Coordinated Call-Back Queuing (CCBQ) feature enables stations at an ESN Main to be offered CBQ when network calls are blocked at the serving ESN node. This feature requires that the ESN Main and associated ESN node be equipped with the Network Signaling (NSIG) and CCBQ software packages. The node must have NARS or BARS or CDP.

When trunks are busy at the node, the user will be given overflow tone from the node or optionally RAN followed by overflow tone. The user simply activates Ring Again to accept the CBQ offer.

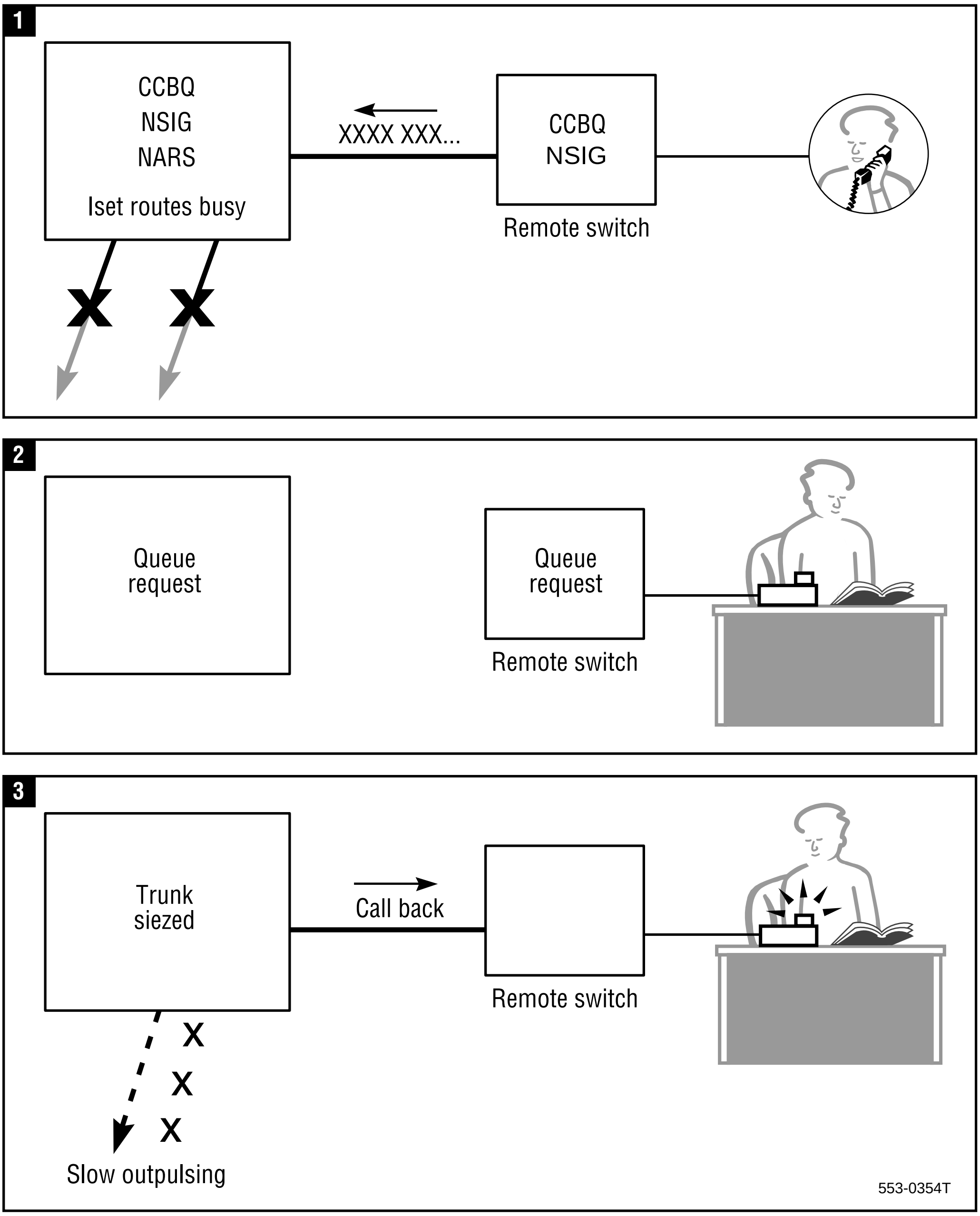
During the queuing time, the TIE trunk to the node is dropped. The node scans routes (Iset or all) based on the NCOS of the caller that was sent with the original call.

When a trunk becomes available at the ESN node, the call originator at the ESN Main is alerted by a call back from the node. The node slowly outpulses digits on the trunk it found until the caller answers the callback. When the caller answers, the remaining digits are outpulsed normally. If the user does not answer within the callback time, the node's trunk and the TIE trunk are dropped. You must install TIE trunks with answer supervision, wink start arrangement and DTMF signaling for CCBQ to work.

> ***Note:*** If no TIE trunks to the ESN Main are idle when the node attempts the callback, the outgoing trunk is released and can be offered to another caller. The CCBQ call retains its position in the queue but is not offered another trunk until a TIE trunk to the ESN Main becomes available.

Queuing

Figure 83- Coordinated Call-Back Queuing



Queuing

Coordinated Call-Back Queuing Against Main

The Coordinated Call-Back Queuing Against Main (CCBQAM) feature is provided by the CCBQ package. It enables stations at nodes to be offered CBQ for network calls that are blocked at a main. When facilities become available at the main, the call originator at the node is alerted by a callback from the main. CCBQAM otherwise functions identically to CCBQ at the node.

CCBQ Eligibility tests

When a telephone at an ESN Main originates a network call through an ESN node, the NCOS of the call originator, the call type, and whether or not the telephone is allowed access to the Ring Again feature is transmitted to the ESN node. NSIG software automatically outpulses these extra digits from the ESN Main to the node along with the call. If an authcode is entered at the ESN Main prior to dialing a network call, the NCOS associated with the authcode is transmitted to the ESN node. When received by the node, this NCOS is used to determine CCBQ eligibility and is used for the duration of the call, unless further modified by the Authcode Conditionally Last feature.

The normal CBQ eligibility tests are performed by the node. In addition, further tests are performed for CCBQ:

- the incoming TIE trunk route at the node must have CBQ activated
- the signaling protocol type programmed on the TIE trunk route must be suitable for queuing and match at both ends
- the NCOS of the caller that travelled to the node must have CBQ allowed

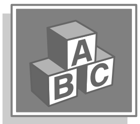
CCBQ is offered to the user at the ESN Main if the eligibility tests are successful. If the tests are unsuccessful, standard call blocking is applied to the call.

Multiple Callback Queue requests are allowed based on the availability of call registers at the node.

CCBQ and ERWT

As for stations at an ESN node, the call originator at an ESN Main can invoke Ring Again upon receipt of ERWT if the originator’s NCOS is defined at the ESN Main and node as CBQ(A) and ERWT eligible.

Queuing



User busy during callback

If the call originator is equipped with an analog (500/2500 type) telephone and is engaged in a call when the ESN node initiates a CCBQ call back, a signal is transmitted from the ESN Main to the ESN node. The ESN node releases the outgoing trunk and places the CCBQ call into a holding queue for five minutes. No attempt is made by the node to seize another outgoing trunk for the call until the holding time expires. This process occurs only once.

If the originating telephone is still busy, the CCBQ is canceled automatically at the ESN node. No indication is given to the call originator of the CCBQ cancellation. To prevent the CCBQ call from remaining indefinitely in the holding queue at the ESN Main, the ESN Main sets a time limit of 1 hour for CCBQ calls. When this time limit expires, the CCBQ call is canceled automatically at the ESN Main.

CCBQ callback to a busy Meridian 1 proprietary telephone is as for normal Ring Again.

CCBQ cancellation

The call originator at the ESN Main can cancel the CCBQ call at any time; however, the ESN node is not aware of the cancellation until the CCBQ callback is attempted.

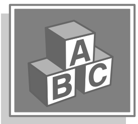
Callback Queuing to a Conventional Main (CBQCM)

Users at a remote switch (regardless of type) connected to a node that has CBQCM software can queue when the trunks at the node are busy. However, the user cannot simply activate Ring Again as with CCBQ. They must be trained to expect interrupted dial tone (or an optional recorded announcement followed by interrupted dial tone) after which the user must enter the extension number of the telephone.

The user can dial this number (dial pulse) or enter it into the telephone keypad (DTMF signaling). When the last digit of the extension number is entered, a confirmation tone (three 256-ms bursts of dial tone) is sent from the ESN node to the call originator. The call is placed in the CBQ at the ESN node when the call originator goes on hook.

The node uses this number to callback the user when a trunk is found. This feature will not work with wink start or delay dial TIE trunks.

Queuing



CBQCM Eligibility tests

When a telephone at a Conventional Main originates a network call through an ESN node, the NCOS assigned to the incoming trunk group is used to determine the Callback Queuing to Conventional Main (CBQCM) eligibility. This NCOS, as well as the incoming trunk group, must be defined as CBQ eligible.

CBQCM Offer not accepted

The CBQCM offer can be refused by going on hook any time before the last digit of the extension number is dialed, or by remaining off hook for longer than 30 seconds after receipt of the offer tone. If the CBQCM is neither accepted nor rejected within 30 seconds, the caller is given overflow tone (from the ESN node) and the call is disconnected.

Callback

When an outgoing trunk becomes available at the ESN node, it is seized; slow outpulsing is started. The ESN node then seizes a TIE trunk to the Conventional Main and outpulses the extension number of the call originator. The call originator must answer the callback before slow outpulsing is completed; otherwise, the call back is canceled and the outgoing trunk is released.

> ***Note:*** If no TIE trunks are currently available to the Conventional Main, the node releases the outgoing trunk. The CBQCM call retains its position in the queue but is not offered another outgoing trunk until a TIE trunk to the Conventional Main becomes available.

When the call originator answers the CBQCM callback (which is ringing the telephone in the same way as a normal call), answer supervision must be transmitted from the Conventional Main to the ESN node. Upon receipt of answer supervision from the Conventional Main, the ESN node transmits a tone (three 256-ms bursts of dial tone) to notify the call originator that the call is a CBQCM callback, and completes the call. You must train users to understand that these tones indicate the call is a CBQCM callback.

Queuing

If the call originator's telephone is busy, the ESN node places the call in a suspended state for 5 minutes. After five minutes, another callback is attempted, if the outgoing trunk is free. If the telephone that originated the call is still busy, the ESN node cancels the callback.

No provision is made for CBQCM cancellation by a call originator at a Conventional Main. Once the CBQCM offer is accepted, the call remains in the queue until the ESN node initiates a callback.

CBQCM requirements

Station users at Conventional Mains cannot activate Ring Again to refuse expensive routes after ERWT is given.

The ESN node must be able to seize a trunk in the same TIE trunk group for the CBQCM callback that was used to initiate CBQCM. Thus, these trunk groups must be two-way (incoming and outgoing).

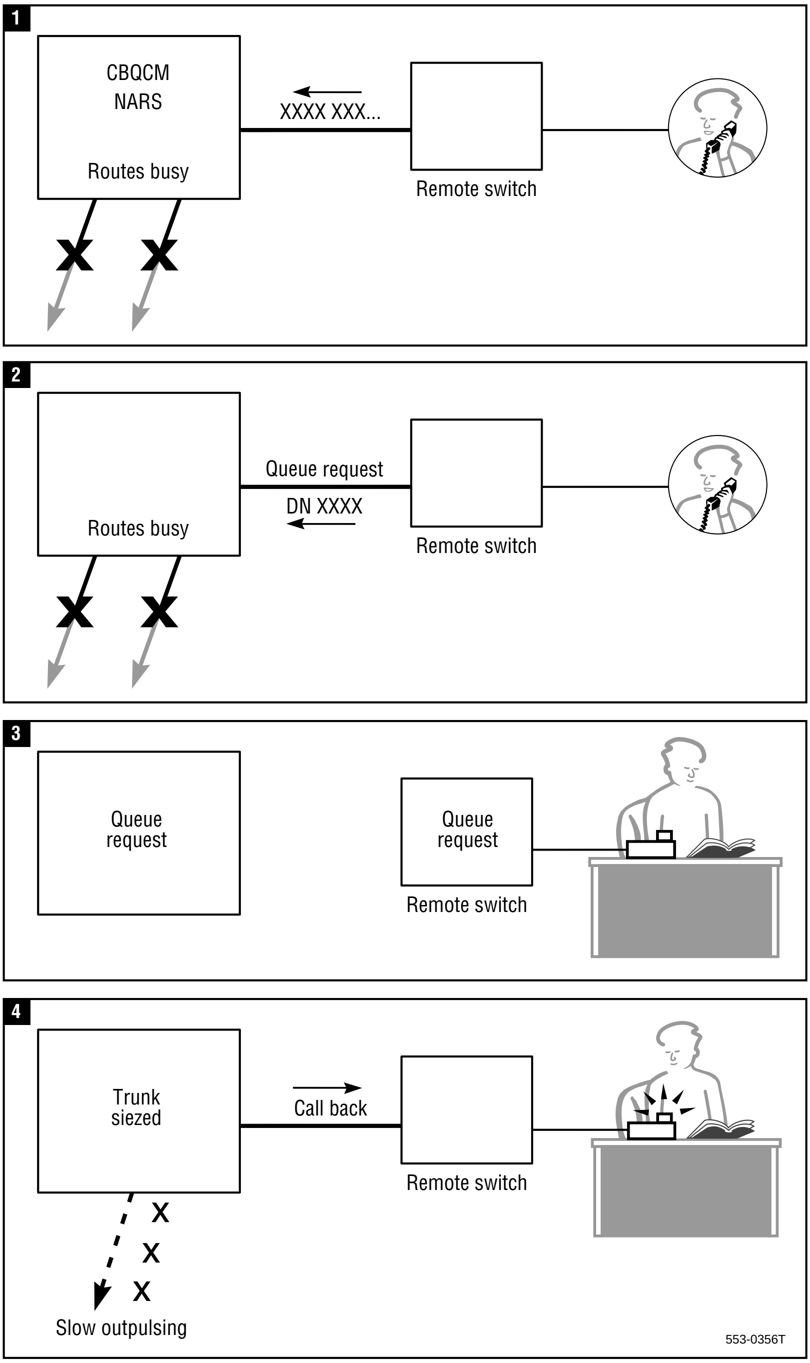
Conventional Mains must provide answer supervision on TIE trunks connected to the ESN node. These switches must also permit transmission or repetition of telephone dial pulses for CBQCM operation. This feature cannot be used with systems that operate in senderized mode. Operation may require adjustment of the interdigit timeout on systems that employ simulated cut-through operation.

Multiple Callback Queue requests are allowed based on the availability of call registers at the node.

The TIE trunk route to the Conventional Main must be programmed with Standard Signaling, if NSIG software is equipped at the node.

Queuing

Figure 84- CBQCM



Remote Virtual Queuing (RVQ)

RVQ provides for ISDN networks the functionality that CCBQ and CBQCM provided on ESN networks. In addition, this feature also allows a user to queue for busy trunks at a distant switch. The RVQ software scans ahead to ensure the entire path is clear before calling a user back.

RVQ is supported on a private MCDN PRI/ISL network only.

An RVQ retry timer is set for each NCOS. The lower the timer, the more often the system checks for a trunk. The maximum amount of time that RVQ will check for a trunk is 30 minutes.

The first caller to activate RVQ is not necessarily the first user given a callback.

The trunks (B-channels) are reserved during the callback period.

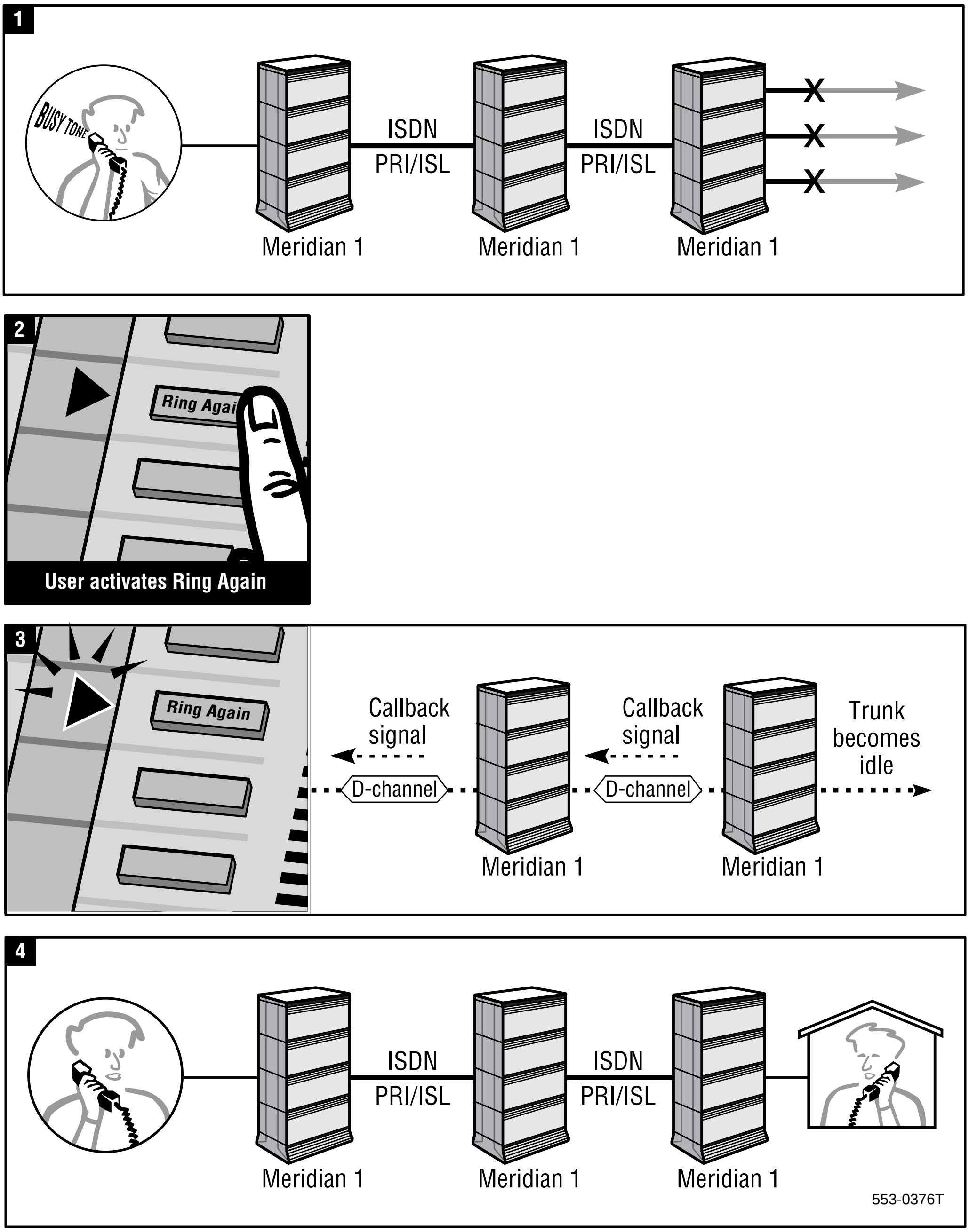
The RVQ feature depends on UDP and CDP dialing plans. If there is a non-MCDN compatible switch in the network, the E.164 public numbering plan is supported to allow it to queue.

RVQ does not allow queuing after Expensive Route Warning tone is given.

RVQ and Drop Back Busy (DBB) capabilities are packaged together with the Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package. Drop Back Busy takes precedence over RVQ, there is information on DBB later in this section. In order to use RVQ, you must disable DBB on the route list entries.

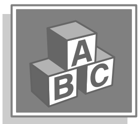
Queuing

Figure 85 - Remote Virtual Queuing



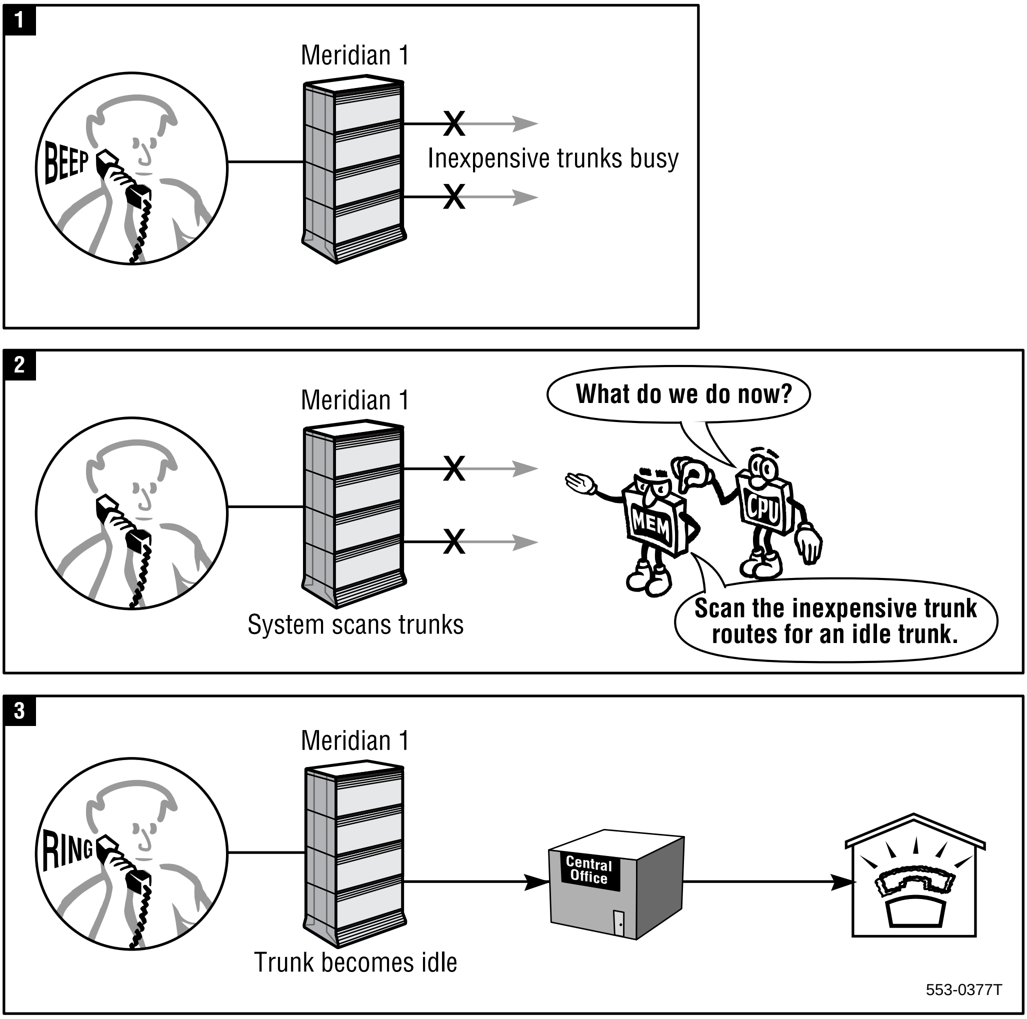
Queuing

Off-Hook Queuing (OHQ)



The feature lets a call originator remain off hook after hearing a one-second burst of tone, for a short time (which you can set) until a trunk becomes available.

Figure 86 - Off-Hook Queuing



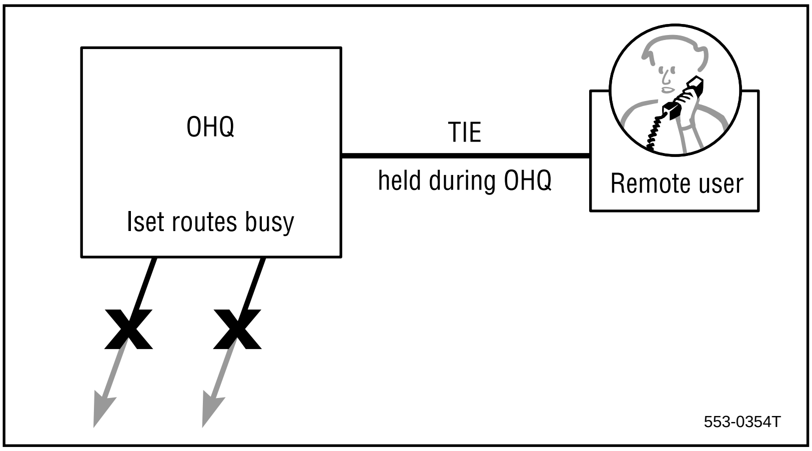
You must select the NCOS groups to be allowed OHQ, as well as setting the customer-wide OHQ timer that determines the maximum amount of time a user will queue offhook.

Queuing

OHQ takes precedence over CBQ. If both OHQ and CBQ are activated for a user, if the eligibility tests for OHQ pass, the user will be offered Off-Hook Queuing.

If you plan to allow users at a remote site (regardless of type) to queue for the node's trunks when they are busy, equip the OHQ software at the node. During the queuing time, the remote user remains off hook and the TIE trunk to the node remains held up. This adds extra usage to the TIE trunk.

Figure 87 - Network-wide use of Off-Hook Queuing



Also, as a call progresses through the network, OHQ can be offered to the call originator from any of the ESN nodes or ESN Mains that are used to process the call. As a result, OHQ can be offered more than once for a given call.

Off-Hook Queuing Eligibility tests

Network calls may be placed in an OHQ if all trunk routes (entries) in the Iset of a route list are busy, and the following criteria are met:

- OHQ has been allowed for that customer group.
- An OHQ timer has been set (up to 60 seconds)
- At least one of the trunk routes in the Iset of the route list is defined as being eligible for OHQ.
- The NCOS of the call originator (at an ESN node or an ESN Main) is defined to permit OHQ.
- The number of calls already queuing for the routes at priority level three is below a specific threshold for at least one of the routes in the Iset. The number of allowed priority level three calls is set on each route (maximum 63). The priority level three queued calls include:
  - Ring Again calls
  - Direct access calls being attempted
  - Priority level three Priority Queued calls
  - Off-Hook Queued calls

If a trunk does not become available during the time the user is queuing on the Iset routes, the system scans the extended set routes once. If there is a trunk available *and the user has a high enough FRL*, the call is processed on the extended set trunk. There is no Expensive Route Warning Tone given.

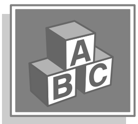
Calls that do not meet the preceding requirements for OHQ eligibility can be offered CBQ at this point.

Network-wide OHQ eligibility tests

- The incoming trunk group at the ESN node or ESN Main is defined in software to permit OHQ for incoming calls or the caller’s NCOS (if you have travelling NCOSs) is allowed OHQ.
- The (Travelling Class Mark) TCM from an ETN switch, or the TCOS from another ESN node or an ESN Main is the same as an FRL that is defined to permit OHQ.

Queuing

Network Drop Back Busy (DBB) and Off-Hook Queuing



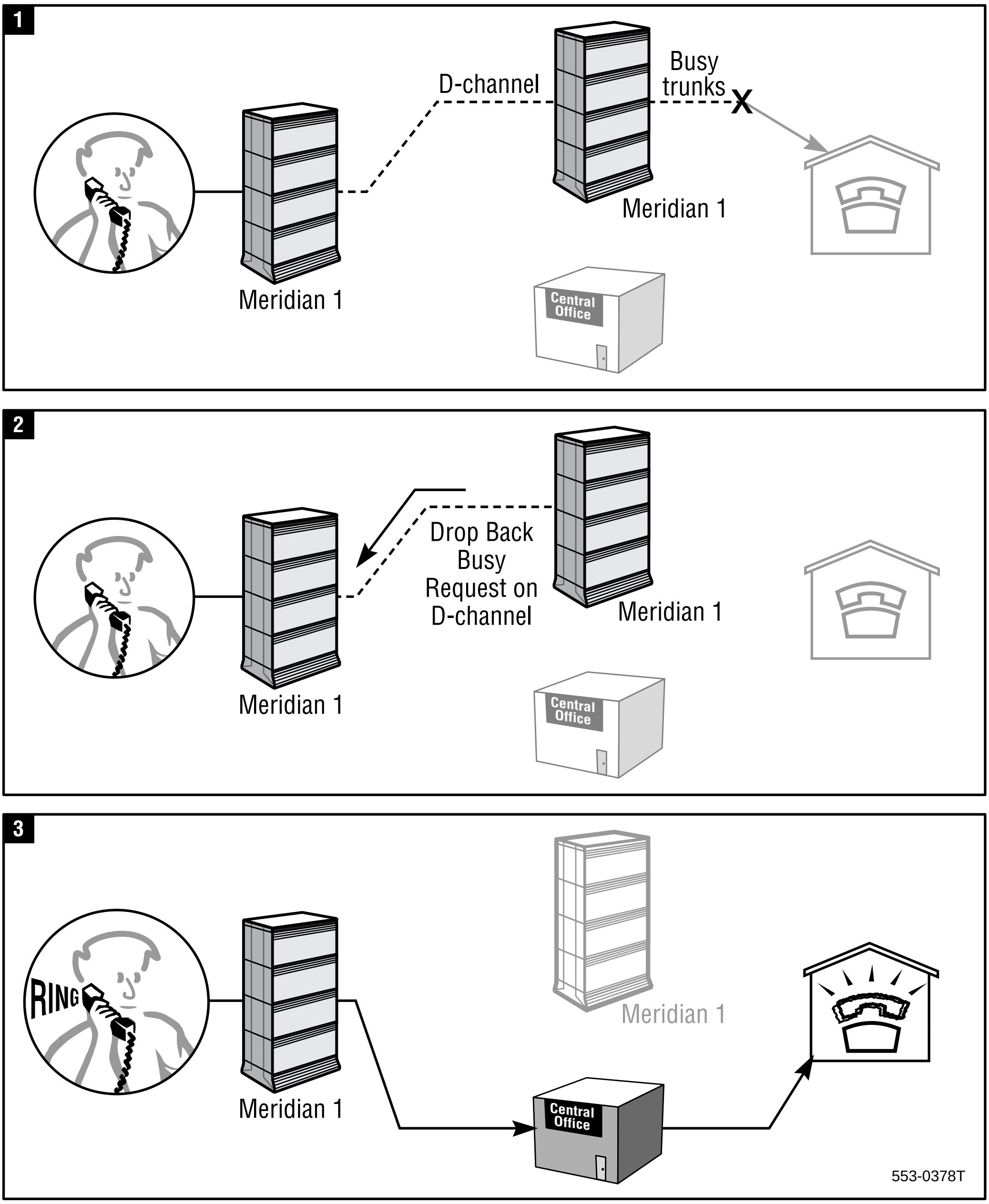
This feature allows the originating node to remain involved in the routing of outgoing network calls. The switches must be equipped with ISDN and NARS or CDP. The DBB and OHQ features are activated on entries in the route lists.

Network Drop Back Busy allows network calls that are blocked at a tandem node to be rerouted (dropped back) to the originating node. The calls are then directed over an alternate route by the node.The switch encountering blockage decides to drop the call back to the originating switch based on the route list associated with the call. You can program route list entries to drop the call back, if the Iset routes are busy or if all routes are busy.

The Network Off-Hook queuing feature allows OHQ to be offered at a tandem node. The call is dropped back if the Iset remains busy after queuing or if the extended set is also busy after queuing. The OHQ package must be equipped at the tandem node(s).

DBB and RVQ capabilities are packaged together with the Originating Routing Control/Remote Virtual Queuing (ORC-RVQ) package. Drop Back Busy takes precedence over RVQ. In order to use RVQ, you must disable DBB on the route list entries.

Figure 88 - Network Drop Back Busy



Queuing

Table 77
DBB and OHQ

Configuration	Condition	Treatment
DBI, IOHQ = NO	All routes in Iset are busy	Drop back to originating node
DBA IOHQ = NO	All routes in Iset and extended set are busy	Drop back to originating node
DBI, IOHQ = YES	All routes in Iset are busy	Off-Hook Queue. If OHQ times out, drop back to originating node.
DBA, IOHQ = YES	All routes in Iset and extended set are busy	Off-Hook Queue. If OHQ times out, attempt routing over extended set. If all routes busy, drop back to originating node.

QSIG and EuroISDN networks

- Call Completion to Busy Subscriber
- Call Completion on No Response

Refer to the *EuroISDN* section for more information on these types of queuing.

DPNSS1 networks

- Call Back When Free
- Call Back When Next Used
- Attendant Call Offer
- Step Back on Congestion

Refer to the *DPNSS1* section for more information on these types of queuing.

Improving feature performance

The information that follows make you aware of issues that could affect implementation. You should resolve these issues before you begin programming. Use the checklist under *What to have ready* to confirm that you have what you need.

Network Traffic Studies

Running traffic studies on your system after you have installed any form of queuing can provide you with very valuable information. Please refer to the *Network Traffic studies* section in this book for more detailed information.

Control tips

OHQ

If your long distance bills are too high on a system where OHQ is installed do one or more of the following:

– lengthen the OHQ timer

– install more inexpensive routes

– remove the expensive choice(s) from the route lists

Usage on your expensive routes should be very low if all users have OHQ programmed. Users only scan Iset routes initially and then, only if the OHQ timer expires, they scan non-Iset routes.

You can restrict users from getting out on the expensive routes when the OHQ timer expires by raising the FRL required on the extended set routes to a higher level than certain NCOS groups have.

Queuing

Administration tips

If you are going to install queuing, it is easiest if you assign the same type of queuing to all users. Your administration and trouble shooting will be much easier and training will also be easier.

CBQ

Issues affecting the callback timer setting:

- bear in mind that you do not want complete calls being outpulsed for unanswered callbacks. There are some sample calculations below:
  - local calls (7 digits) would take 10 seconds + (6x 2.56 seconds) = 25.36 seconds
  - calls to CDP location when TIEs are blocked, with 4 digit CDP DNs would take 10 seconds + (3 x 2.56 seconds) = 17.68 seconds
  - calls to a CDP location when TIEs are blocked and NSIG is outpulsing extra signaling digits (outpulses the CDP DN + TCOS digit) 10 seconds. + (4 x 2.56 seconds) = 20.24 seconds
  - calls to CDP location when TIEs are blocked, with 3 digit CDP DNs would take 10 seconds + (2 x 2.56 seconds) = 15.12 seconds

Setting the timer for less than 15 seconds results in the most efficient use of trunks.

Also, train users to answer callbacks quickly.

PQUE

Try to avoid making NCOSs too complex by assigning minor differences in PQUE promotion timers. The cost savings to be gained from manipulating priority levels will be lost in the analysis and administration time that goes along with a more complicated system. If there are users who should be high priority when they queue; give them priority level three to start with; give everyone else priority level zero.

Queuing

Priority queuing may not be appropriate if you expect a lot of queuing. You should install more trunks instead, otherwise priority level zero users will queue for very long periods of time.

Some network planners give low FRL users a high starting priority level since they do not have access to many routes.

It makes sense to set the Extended Route Advance Timer to expire only after the user has had a chance to reach the highest priority level on the Iset routes first. Program the promotion timer in such a way that this happens.

OHQ and CBQ

If you are activating both forms of queuing for one NCOS group, sometimes, the users may be offered CBQ (overflow tone or RAN) when the eligibility tests for OHQ do not pass. This can be a confusing situation for users who are used to hearing one burst of tone when trunks are busy.

Implement OHQ in such a way that the eligibility tests for OHQ will always pass. You can ensure this if you set the maximum number of priority level three queuers at 63 on every route.

CCBQ and CDR and Traffic studies

While the node is waiting for the user at the main to answer the callback, digits are slowly being outpulsed on one of the node’s trunks. These digits may appear on a CDR record whether or not the user at the main answers. Bear in mind that these unanswered callbacks may add extra usage to the node’s trunks.

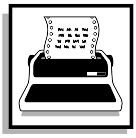
Look at Network Traffic studies to find out if there are frequent unanswered callbacks. A TIE trunk is held up between the node and the main during the callback. You may need to re-train users and tell them the impact of queuing.

Queuing

Training tips

- Make sure users are comfortable with the queuing-related tones they will hear and what they should do when they hear them. It helps if they understand the impact of their actions, from a cost point of view.
- Encouraging users to queue after ERWT requires good training and possibly extra billing if they do not queue for less expensive routes often enough.
- CCBQ and CBQCM - Installing the optional RAN instead of overflow tone reduces the need for training. The recorded announcement tells users what to do when the trunks are blocked.
- CCBQ and CBQCM- tell users of analog (500/2500 type) sets to keep their telephones idle when queuing since the node only tries the telephone twice. If it is busy both times, it removes the information from its memory. From a system point of view this makes sense because two trunks are not occupied with callback attempts that encounter busy signals repeatedly. Tell users not to report this situation as a trouble if they were using the phone during the five minute period after queuing.
- CBQCM requires proper training; the users must know about dialing their own DNs when given the queuing tones. The user must wait for confirmation tone from the node before hanging up. Train the user about the tones they will hear when they answer the callback.

Maintenance tips



NCOS structure

It is wise to match NCOS definitions at mains and nodes when network-wide features like queuing are implemented to make network administration easier.

Call Trace

Your system supplier will use Network Call Trace and the call trace overlay program to follow calls through a network to locate blockage points. If there are queuing problems, you will be able to get printouts to help you identify where the problems are.

Queuing

What to have ready

The following checklist summarizes the steps you should take before implementing queuing.

Table 78
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Analyze the projected cost savings and costs associated with queuing. Evaluate the impact of queuing on your users.
✔		Analyze the calling patterns and trunking equipped at network sites. Decide where queuing should be implemented with the greatest benefit and acceptance.
✔		Ensure you have NARS or CDP installed and programmed at nodes and the other switches where required, for the queuing feature(s) you are implementing.
✔		Plan to implement a consistent NCOS structure across your network, if NSIG is installed.
✔		Decide on timers associated with some types of queuing.
✔		Plan to monitor the operation of the queuing features after implementation.
✔		Prepare training and refresher training sessions.

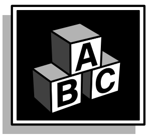
Other network features

The two features in this section extend the capabilities of the corresponding stand alone system features across a network. They are:

- Network and Executive Distinctive Ringing
- Network Intercom (Hot Type D and Hot Type I Enhancements)

Basic configuration

Network and Executive Distinctive Ringing



This feature allows users to know more about the type of call that is coming into a telephone by the way the telephone is ringing. The user does not have to rely on a display. Analog (500/2500 type) sets do not have displays so the ringing can help the user identify types of callers. Also, the user who is answering a call may need to know what priority level or job function the caller has in order to answer the call efficiently. The CLID does not give a user this information.

Executive Distinctive Ringing

You can assign one of five “Executive” classes of service to certain telephones. When the user of an executive telephone makes a call, the terminating telephone in the ISDN network rings distinctively, if the route to the terminating switch has this feature activated. This feature also works internally at a local switch.

The five class of service choices are:

- EXR1
- EXR2
- EXR3
- EXR4
- EXR0 (means the feature is not active)

Other network features

Network Distinctive Ringing

You can assign four different distinctive ringing cadences to incoming ISDN trunk routes. You configure the ringing choices you want using the Flexible Tones and Cadences tables.

Calls that tandem on an ISDN network will ring distinctively if the TIE trunk the call was routed on has distinctive ringing activated and the terminating switch has Network Distinctive Ringing equipped.

Network Intercom

There are two aspects to this feature:

- Hot Line (Type D Enhancement)
- Hot Line Type I

Prior to Release 21, there were two types of Hot Line keys: DN-based Hot Type D and Speed Call List-based Hot Type L. Refer to the *Software Features Guide* for more information.

Release 21 introduced Hot Line - Type I.

Hot Type I, used in a network application, can only be supported in networks that use:

- PRI2
- ISL
- VNS
- BRI Trunk Access

The originating, terminating and tandem switches must all be equipped with Release 21 software for Hot Type I to work.

The network DN dialed by the Hot Line for Hot Type I and Hot Type D must be either a CDP or Universal Numbering Plan number that terminates on a Prime DN of a Meridian digital telephone.

There is an option with Hot Type I to provide a No Answer Indication to a called party if the user was absent when a Hot Line call was made to the telephone. The Hot Line key winks. The unanswered call is handled the way a call to the Prime DN would have been handled if it went unanswered.

Hot Type D provides the ability for Meridian digital telephones to have two-way intercom calls on specially designated keys, not the DN keys. The other telephones can be digital telephones spread across an MCDN network. The call can terminate in one of three modes: voice, ringing and non-ringing. In the voice mode, the call is automatically answered after a short ring.

With Hot Type D the terminating DN can appear on more than one telephone. A No Answer indication can be left at the terminating telephone.

Hot Type D can be supported on DPNSS1 networks and R2MFC networks.

Control tips

- Use the distinctive ringing choices in a way that suits your users best. You may decide that only executives' telephones are to be assigned distinctive ringing so users receiving calls from these people will answer quickly and politely.

Administration tips

- Monitor your users' behavior if you implement Executive Distinctive Ringing. The users may be giving priority to certain types of calls but also they may be ignoring other calls that are not ringing distinctively. This can lead to caller complaints.

Other network features

Training tips

- Do not implement more ringing choices than your users can handle. Make the ringing styles very different from each other so users can easily distinguish the different types of incoming calls.
- Train users to understand the meaning of the winking Hot Line key and what they are to do when they see that indication on a telephone.

What to have ready

The following checklist summarizes the steps you should take before implementing Network and Executive Distinctive Ringing and Network Intercom.

Table 79
Checklist

Basic	Optional	Preparation
✔		Do the planning pre-work as described in the *Network planning* section.
✔		Decide if your users need a way of knowing the type of caller from the way the telephone is ringing.
✔		Determine the number of ringing styles that will suit your needs.
✔		Determine if users need a simple way of reaching another DN that is across the network.
✔		Determine if your network type and dialing plan can support Network Intercom.
✔		Decide if the user(s) need voice, ringing or non-ringing access.
✔		Decide if you will implement the No Answer Indication.
✔		Plan training.

Interworking and gateways

Connecting the portions of your network that use different signaling systems or protocols can be a complex task.

In order to do this effectively, you need to know what features are supported by gateways and what functionalities can interwork. The features that cannot interwork or for which there are no gateways will not operate across the gateway.

You will need to:

- assess your needs, your existing network and network implementation constraints
- confirm what functionalities can be supported
- configure possible work arounds, where appropriate
- redesign your network and train users to compensate for any lost functionality

Networking features gateways

The Meridian 1 provides a number of gateways and converters that make network integration simpler and more effective. A gateway is a means of connecting systems using two different signaling schemes.

Some networking features currently exist on more than one ISDN interface implemented on the Meridian 1. For example, some of these features are listed in Table 80 on page 739. This table illustrates feature interworking with respect to a EuroISDN interface. Any networking feature that does not appear in the table is only supported on one ISDN interface and is rejected by the gateway with an X11 Release 24 EuroISDN interface when the service is requested.

Interworking and gateways

Gateways for features completing calls to busy numbers

Gateways between EuroISDN CCBS, MCDN Network Ring Again on Busy and DPNSS1 Call Back When Free are supported.

A PINX (Private Integrated Services Network Exchange) DN must be configured in the Customer Data Block. The PINX DN is used for routing free notification on the MCDN or DPNSS1, if the EuroISDN network does not provide a calling number for the service. The DN configured should be consistent with the type of number plan used.

Call Completion to Busy Subscriber is supported on a EuroISDN Trunk - Network Side connectivity interface.

Call Completion on No Reply is supported on QSIG and DPNSS1 (as Call Back When Next Used) interfaces, corresponding to the MCDN Network Ring Again on No Answer feature. It is not supported on a EuroISDN Trunk - Network Side connectivity interface.

X08/X11 gateway

The X08 generic software was developed for non-North American markets. It has since been recombined with X11 North American software.

The gateway allows the use of both Generic X08 and Generic X11 software on switches in the same network. The gateway allows X11 nodes to bridge R2MFC signaling and/or L1 signaling with ISDN signaling. The following configurations are allowed by the gateway:

- R2MFC connections to public exchange Central Offices
- R2MFC connections between X08 and X11 nodes
- L1 connections between X08 and X11 nodes
- R2MFC connections between X11 nodes
- X11 tandem connections between L1 routes and (non-R2MFC) CO/TIE routes
- X11 tandem connections between L1 routes and ISDN routes
- X11 tandem connections between R2MFC routes and ISDN routes

Interworking and gateways

There is programming required on the X08 and X11 switches. Refer to the Networking NTPs for details.

1.5/2.0 Mbps Gateway

A Meridian 1 can support both 2.0 Mbps (E1) connections and 1.5 Mbps (T1) connections simultaneously. Therefore, the Meridian 1 can act as a gateway between North American and International networks. The systems can be used in North America (as of X11 Release 24) or in International markets.

The system and the trunks can be programmed as A-law or Mu-law to suit the gateway requirements of your network.

This capability supports the following interfaces:

- SL1
- D100
- SL100
- D250
- ISIG
- ESIG
- ISGF
- ESGF

This feature provides A-law to Mu-law conversion. It does not provide protocol interworking between North American and International features. Only those features supported on both sides of the node are transparent across the gateway.

Interworking and gateways

EuroISDN

EuroISDN can interwork with the following types of interfaces:

- AXE10
- 1TR6
- SYS12
- Numeris
- Swiss Net 2
- MCDN (1.5 and 2.0 Mbit/s)
- QSIG
- DPNSS
- DASS2
- R2MFC
- MFE
- DTI
- DTI2
- analog trunks

Table 80 summarizes the networking features supported at gateways with the EuroISDN connectivity.

Any feature that is not listed in this table is not supported at gateways with EuroISDN connectivity. A “YES” indicates that the gateway is supported. A “NO” indicates that the gateway is not supported.

Interworking and gateways

Table 80
Networking features supported by more than one ISDN interface (Release 24)

	Euro-ISDN	MCDN	QSIG	ETSI BRI sets	DPNSS1	MFC	DTI2	Analog	CIS	KD3
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No	No [5]	No[4]
Connected Number	Yes	Yes	Yes	No	Yes	No	No	No	No [5]	No[4]
Transit Counter	No	Yes[1]	Yes[2]	No	Yes[3]	No	No	No	No [5]	No[4]

Note 1: This is supported using the ICCL tandem count feature existing over the NAS feature.

Note 2: This is only supported for ETSI version of QSIG. For the ISO version, it is discarded.

Note 3: This is supported using the Loop Avoidance supplementary service.

Note 4: The basic call is not supported at the gateway.

Note 5: The gateway between EuroISDN and CIS MF Shuttle (Release 23) is supported.

EuroISDN and BRI

When interworking EuroISDN with BRI there is no Connected Number Information Element handled by the ISDN BRI protocol. The caller will not see the connected number, only the called number, if redirection takes place.

Interworking and gateways

EuroISDN Network-side basic call gateways

The gateways that are supported for the EuroISDN Trunk - Network Side connectivity are the same as the ones supported for the EuroISDN connectivities. The table below lists these gateways and the associated support for basic call functionality.

Table 81
EuroISDN Network-side basic call gateways

Gateway	Basic Call Support
Analog (End-of-Signaling)	Basic Voice Call
DTI2 (End-of-Signaling)	Basic Voice Call Data Call (64K unrestricted)
MFE	Basic Voice Call
KD3	Not supported
R2MFC	Basic Voice Call Calling Line ID transmission (provided by the Tandem Call/CDR feature)
DPNSS1	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
ISDN COs (North American and non-EuroISDN interfaces)	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
EuroISDN and Asia Pacific ISDN	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
NI2	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
— continued —	

Interworking and gateways

Table 81
EuroISDN Network-side basic call gateways

Gateway	Basic Call Support
QSIG	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
MCDN	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
CIS	Basic Voice Call Calling Line ID transmission
Note: For MFE, MFC, and CIS using DTI2, data calls are possible.	

MCDN/EuroISDN Malicious Call Trace gateway

If a Meridian 1 is connected to another Meridian 1 acting as a gateway to the public switched network using a EuroISDN route, NAS signaling must be enabled on the MCDN trunk for a user-activated call trace signal to reach the public network switch. The EuroISDN connection to the public network must have Malicious Call Trace enabled.

Business Networking Express (BNE) EuroISDN gateways

Business Networking Express (BNE) is a term that refers to a group of different EuroISDN network functionalities.

Introduced in Release 25, BNE provides a Virtual Private Network (VPN) solution for Meridian 1 systems through the EuroISDN public network.This section provides gateway information pertaining to the following BNE capabilities:

- EuroISDN Name and Private Number Display
- EuroISDN Call Completion

For complete information on BNE, refer to the BNE section in the EuroISDN chapter.

Interworking and gateways

BNE Name and Private Number Display

Calling Name Identification Presentation (CNIP) — EuroISDN/MCDN Gateway

On reception of a call coming from an MCDN network, with the calling user's name information routed to the PSTN network, the calling name is sent through EuroISDN to the destination node (if the EuroISDN route list block supports BNE).

On reception of a EuroISDN call with the calling user's name information routed to the MCDN network, the gateway node delivers the calling user's name information to the MCDN network.

CNIP — EuroISDN/QSIG Gateway

On reception of a call coming from a QSIG network with the calling user's name information, and routed to the PSTN network, the calling name is sent through EuroISDN to the destination node (if the EuroISDN route list block supports BNE).

On reception of a EuroISDN call, with calling user's name information routed to the QSIG network, the gateway node delivers the calling user's name information to the QSIG network.

CNIP — EuroISDN/DPNSS Gateway

DPNSS does not support name display.

Connected Name Identification Presentation (CONP) — EuroISDN/QSIG Gateway

The QSIG network receives the connected (or alerting) user's name from the BNE feature. The connected (or alerting) user's name provided by the QSIG network is sent over EuroISDN to the originator of the call.

CONP — EuroISDN/MCDN Gateway

The connected (or alerting) user's name provided by the MCDN network is sent over EuroISDN to the originator of the call.

The connected (or alerting) user's name delivered by the BNE feature is sent to the MCDN network.

CONP — EuroISDN/DPNSS Gateway

DPNSS does not support name display.

Interworking and gateways

Calling/Connected Name Identification Restriction (CNIR) — EuroISDN/QSIG Gateway

When a user invokes the CNIR service, the calling, alerting, and connected names are marked as “presentation is restricted”, and this indication is passed to the other network.

CNIR — EuroISDN/MCDN Gateway

When a user invokes the CNIR service, the calling, alerting, and connected names are marked as “presentation is restricted”, and this indication is passed to the other network.

CNIP — EuroISDN/DPNSS Gateway

DPNSS does not support name display.

BNE Private Calling Number on EuroISDN

Calling Line Identification Presentation (CLIP) — EuroISDN/MCDN gateway

On reception of a call coming from MCDN network with a private calling number and routed to the PSTN network, the private calling number is sent through EuroISDN to the destination node by the BNE feature.

On reception of a EuroISDN call with a BNE private calling number routed to the MCDN network, the gateway node uses the calling number delivered by the BNE feature to build the CLID IE sent over MCDN.

CLIP — EuroISDN/QSIG Gateway

On reception of a call coming from a QSIG network, with a private calling number and routed to the PSTN network, the private calling number is sent through EuroISDN to the destination node by the BNE feature.

On reception of a EuroISDN call, with a BNE private calling number routed to the QSIG network, the gateway node uses the calling number delivered by the BNE feature to build the CLID sent over QSIG.

Interworking and gateways

CLIP — EuroISDN/DPNSS Gateway

On reception of a call coming from a DPNSS network, with a private calling number (OLI) and routed to the PSTN network, the private calling number is sent through EuroISDN to the destination node by the BNE feature.

On reception of a EuroISDN call, with a BNE private calling number routed to the DPNSS network, the gateway node uses the calling number delivered by the BNE feature to build the OLI sent over DPNSS.

‘H’ is not displayed in the private number on the DPNSS side, according to the existing DPNSS gateway.

Calling Line Identification Restriction (CLIR) — EuroISDN/QSIG Gateway

When the CLIR service is invoked, the calling number is marked as “presentation is restricted”, and this indication is passed to the other network.

CLIR — EuroISDN/MCDN Gateway

When the CLIR service is invoked, the calling number is marked as “presentation is restricted”, and this indication is passed to the other network.

CLIR — EuroISDN/DPNSS Gateway

The CLIR service is not supported on DPNSS. Upon receiving the calling number from DPNSS, it is marked as "presentation is unrestricted" and then passed to the EuroISDN side.

If a calling number marked as "presentation restricted" is received from the EuroISDN side, it is passed to the DPNSS side without the possibility of indicating "presentation restriction". Therefore, the calling number will display.

Private Connected Number on EuroISDN

Connected Line Identification Presentation (COLP) — EuroISDN/QSIG Gateway

The connected number, delivered by the BNE feature, is sent to the QSIG network. The connected number, provided by the QSIG network, is sent over EuroISDN to the originator of the call.

Interworking and gateways

COLP — EuroISDN/MCDN Gateway

The connected number, delivered by the BNE feature, is sent to the MCDN network. The connected number, provided by the MCDN network, is sent over EuroISDN to the originator of the call.

The connected number is provided by the MCDN network only in the case of call diversion.

COLP — EuroISDN/DPNSS Gateway

The connected number, delivered by the BNE feature, is sent to the DPNSS network. The connected number, provided by the DPNSS network, is sent over EuroISDN to the originator of the call.

‘H’ is not displayed in the private number on the DPNSS side, in accordance with the existing DPNSS gateway.

Connected Line Identification Restriction (COLR) — EuroISDN/QSIG Gateway

When the COLR service is invoked, the connected number is marked as “presentation is restricted”, and this indication is passed to the other network.

COLR — EuroISDN/MCDN Gateway

When the COLR service is invoked, the connected number is marked as “presentation is restricted”, and this indication is passed to the other network.

COLR — EuroISDN/DPNSS Gateway

The COLR service is not supported on DPNSS. Upon receiving the connected number from DPNSS, it is marked as "presentation is unrestricted" and then passed to the EuroISDN side.

If a connected number marked as "presentation restricted" is received from the EuroISDN side, it is passed to the DPNSS side without the possibility of indicating "presentation restriction". Therefore, the connected number will display.

Interworking and gateways

MSIG

The MCDN features that interwork with QSIG are:

- basic call (Release 20 - Europe only, Release 22 - global)
- Calling Line ID Presentation/Restriction (CLIP/CLIR) (Release 20)
- Connected Line Presentation/Restriction (COLP/COLR) (Release 20)
- Calling Name Identification Presentation (CNIP) (Release 22)
- Connected Name Identification Presentation (CONP) (Release 22)
- Calling/Connected Name Identification Restriction (CNIR) (Release 22)
- Name Display Enhancement (Release 24)

  Refer to the *CLID and Name Display options* section of this book, for more information.
- Call Completion to Busy Subscriber (CCBS) for ETSI QSIG
- Call Completion on No Response (CCNR) for ETSI QSIG
- Call Diversion Notification
- Call Transfer Notification

Name length for QSIG is a maximum of 50 characters whereas the maximum is only 27 characters for MCDN networks. If QSIG and MCDN networks are interworking, the name displayed may be truncated. Configure the MCDN side for display choice ND3.

Call Diversion Notification Enhancements

In X11 Release 24, the Call Diversion Notification functionality has been extended to include a QSIG/MCDN gateway. Refer to the QSIG section in this book for more information.

Interworking and gateways

MCDN End To End Transparency

Because of the X11 Release 24 MCDN End To End Transparency feature, features such as Network Attendant Service (NAS), Network ACD (NACD) and Network Message Services-Message Center (NMS-MC) and Network Message Services-Meridian Mail (NMS-MM) are supported in QSIG networks and in mixed MCDN/QSIG networks.

Note: The NMS-MM features Call Sender and Thru-Dialing are not supported.

Call Transfer Notification

In X11 Release 24, when a call transfer occurs over a QSIG PRI2 or BRI trunk link, a notification of the transfer is sent to the originating party and the party to whom the call is transferred. The transfer must be done using the Call Join feature (Multi-Party Operations). ANF Path Replacement takes effect, if applicable, in order to obtain a more efficient connection.

The information appears on the displays of the originating and terminating telephones. It includes:

- the redirection and originating number
- redirection and originating name
- redirection reason mnemonic (a code defined for Call Transfer).

The same notification occurs over a QSIG/MCDN gateway.

Although Call Transfer by Call Join was supported over a QSIG link as early as Release 20, there was no notification to the users that a transfer had taken place.

The method of Operation Coding is by Object Identifier for ETSI interfaces or Integer Value for ISO interfaces.

Interworking and gateways

QSIG/ETSI GF Enhancement

In X11 Release 24, the enhancement to the GF transport platform comprises the following capabilities:

- modified call control to support call-independent gateways from QSIG to EuroISDN, MCDN and DPNSS1 signaling protocols
- modified call-related Application Protocol Data Unit (APDU) transport to ensure that a call that is being optimized during QSIG Path Replacement is using the optimum path (the first route in the route list index). This modified mechanism is also needed so the Facility Information Elements carried in a SETUP message which has encountered congestion are included in the alternate call, if alternate routing is performed at a transit node. The Call Completion to Busy Subscriber feature and ANF Path Replacement feature have been modified to use this new capability.
- more accurate calculation of the remaining available length for APDUs in basic call control messages for QSIG GF and ETSI GF transport.

The APDU call-independent transport mechanism is connection oriented. Connectionless APDU call-independent transport is not supported.

When a call-independent SETUP message includes several Facility Information Elements (FIEs) belonging to different Supplementary Services, the gateway is not supported.

MCDN and Japan TTC Common Channel Signaling

Japan Telecommunication Technology Committee (JTTC) Common Channel Signaling is the Japanese version of the International Standard Organization (ISO) ISDN QSIG. The MCDN/JTTC gateway supports only Basic Call and Calling Line Identification Presentation/Calling Line Identification Restriction (CLIP/CLIR).

Some networking features currently exist on more than one ISDN interface on the Meridian 1 system. These features are listed in the table that follows. The columns list the services and the rows list the interfaces.

Interworking and gateways

Networking features that do not appear in the table are only supported on one ISDN interface and are, therefore, rejected by all JTTC gateways when the service is requested. This is the case for all MCDN features that are not supported over the JTTC interface.

In the table when a service is supported (marked with a Y), the information related to this service is accepted, decoded, and used according to the service description. When a service is not supported (marked with an N), the information related to this service is not sent to the interface. The request for the service is rejected according to the service rejection procedures. When a service is supported on two interfaces, a gateway function exists. This passes the information in order to support the service from one interface to the other.

Table 82
Networking features that exist on more than one ISDN interface implemented on Meridian 1

	JTTC	ETSI QSIG	ISO QSIG	Euro-ISDN	MCDN	Analog	D70
Calling Line ID/Calling Party Subaddress	Y	Y	Y	Y	Y	N	Y
Transit Counter	N	Y	N	N	Y	N	N
Call Charge	N	N	N	Y	N	Y	N

BNE Call Diversion QSIG, MCDN and DPNSS gateways

Introduced in Release 25, the BNE/EuroISDN Call Diversion feature allows notification to occur when the private network is a multi-node network using the following protocols (see figure 89):

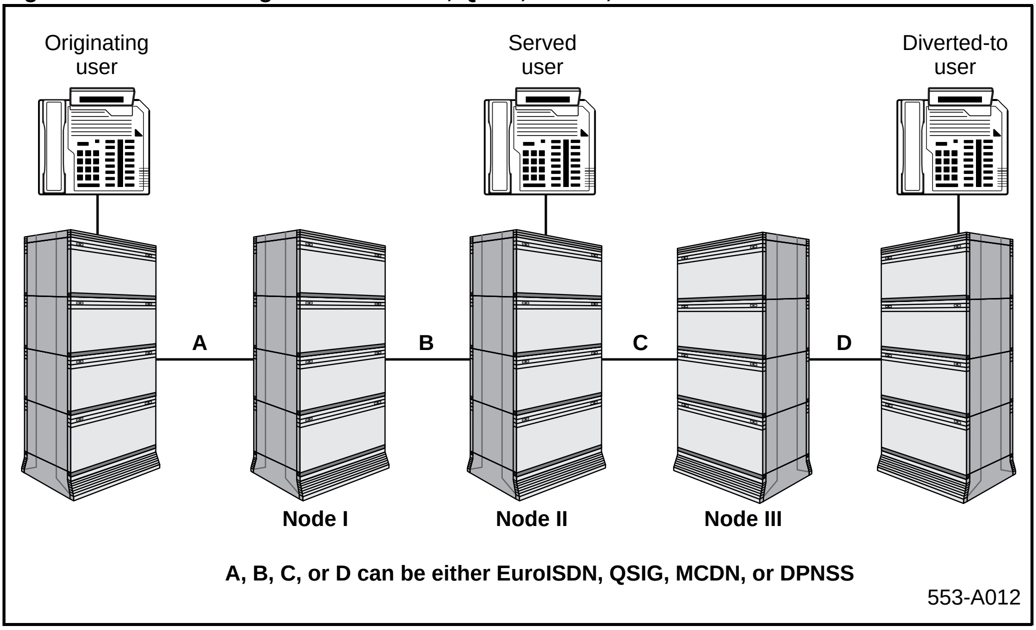
- QSIG
- DPNSS
- MCDN

Interworking and gateways

The notification of the originating, and diverted-to user, depends on the different protocols involved at various stages of the call establishment, and on the diversion specifications for the protocols.

For example, in case of a EuroISDN/DPNSS gateway, the presentation information is not mapped, since it is not supported by the DPNSS protocol.

Figure 89 - Interworking with EuroISDN, QSIG, MCDN, or DPNSS



Node I Gateways (A/B)

The following gateways can exist between the Originating user node and the Served user node (refer to Figure 89):

- QSIG/EuroISDN and EuroISDN/QSIG
- MCDN/EuroISDN
- EuroISDN/MCDN
- DPNSS/EuroISDN
- EuroISDN/DPNSS

Interworking and gateways

Node II Gateways (B/C)

The following gateways can exist at the Served user node (refer to Figure 89):

- QSIG/EuroISDN and EuroISDN/QSIG
- MCDN/EuroISDN
- EuroISDN/MCDN
- DPNSS/EuroISDN
- EuroISDN/DPNSS

Node III Gateways (C/D)

The following gateways can exist between the Served user node and the Diverted-to user node (refer to Figure 89):

- QSIG/EuroISDN and EuroISDN/QSIG
- MCDN/EuroISDN
- EuroISDN/MCDN
- DPNSS/EuroISDN
- EuroISDN/DPNSS

QSIG and R2MFC/MFE

The R2MFC/MFE protocol involves inter-register signaling on trunks that can provide:

- the called party number (an exchange on the terminating end of the call can request the Calling Party Number from the originating exchange). The Calling Party Number is not mapped into the QSIG network because Numbering Plans are different.
- the state of the called party. The possibilities are:
  - station idle
  - station busy
  - congestion (lack of resources)

Interworking and gateways

- station out of order (called party is maintenance busy)
- vacant number (invalid number)
- failure (call failure due to time out)

There is a mapping for these states between the QSIG and R2MFC/MFE protocols.

DASS2/DPNSS1 gateways

DASS2/DPNSS1 to MCDN gateway

On Meridian 1, the preferred method of interconnection between Meridian 1 PBXs and other products in the Meridian family is the Q.931 intelligent private network signaling protocol. MCDN networks use Q.931 signaling.

DASS2

DASS2 on the Meridian 1 offers transparent gateway working with the Q.931 signaling protocols, with the following functions:

- Basic Call Service
- Calling Line Identification
- Called Line Identification
- Display update on call diversion
- Call Charging Indication
- Coordinated Dialing Plan

Interworking and gateways

DPNSS1

DPNSS1 on the Meridian 1 offers a transparent gateway working with the Q.931 (MCDN) signaling protocols, with the following functions:

- Basic Call Service (circuit switched voice calls)
- Calling Line Identification
- Called Line Identification
- Connected Line Identification
- Call Diversion
- Display update on call diversion
- Network Ring Again
- Coordinated Dialing Plan
- Message Waiting Indication (Release 23)

DPNSS-MCDN Message Waiting Indication gateway

There are Meridian 1 systems connected to a voice messaging system on either another vendor's PBX or another Meridian 1 using a series of interfaces consisting of DPNSS and MCDN trunks. The DPNSS-MCDN Message Waiting Indication gateway allows them to have their Message Waiting lamp lit or darkened by the remote message system by providing the means to pass a Message Waiting Indicator message across a DPNSS-MCDN (or MCDN-DPNSS) gateway.

When telephone A (a telephone on the originating switch or on a Meridian 1 node with Message Center users (they can be the same) calls telephone B (a telephone on the Meridian 1 node with Message Center users which is redirected to the Message Center) the DPNSS Diversion feature or the MCDN Trunk Optimization feature (or a combination of the two) drops the old call to telephone B and makes a new call to the Message Center, where telephone A can leave a message for telephone B. When the message is left, the Message Center signals the host PBX (the switch with the voice messaging system) to send a message to telephone B to light its Message Waiting lamp. At the gateway node the message will be mapped to its equivalent for the other signalling system and the new

Interworking and gateways

message sent on to the node with Message Center users, where it will be used to light the MW lamp of telephone B. An Acknowledgment message is sent back, and this too is translated by the gateway node.

After telephone B has called the Message Center and listened to its messages, signalling takes place again, but with messages indicating a Message Waiting Cancellation.

This feature only handles the gateway for the Notification or Cancellation of the Message Waiting Indication. Mapping of other Network Message Service capabilities such as Call Sender or Connection Notification is not supported.

This feature only supports the case when the telephone with the Message Waiting Indication is on a Meridian 1. Providing a Message Waiting Indication to a telephone on a third party PBX is not supported.

This feature only handles the gateway between DPNSS and an MCDN link to another Meridian 1 switch. The connection to a DMS-100 (enabled by the Release 19 ‘MWI Interworking with DMS-100’ feature) is not supported.

This feature builds on the Release 23 ‘DPNSS1 Message Waiting Indication’ feature and the Release 16 ‘Network Message Services - Meridian Mail’ feature.

Feature Packaging

This feature requires the DPNSS MWI (DMWI) package (#325), and the Network Message Center (NMC) package (#175).

All nodes with DPNSS routes need all the IDA and DPNSS prerequisite packages.

All nodes with MCDN routes need all the MCDN prerequisite packages.

All nodes with DPNSS routes need the DPNSS Network Services (DNWK) package (#231), for the DPNSS1 Diversion capabilities.

Interworking and gateways

All Meridian 1 originating nodes and Meridian 1 nodes with Message Center users with MCDN routes need the MCDN Network Services (NTWK) package (#148), for the MCDN Trunk Optimization capabilities.

The Meridian 1 nodes with Message Center users also need the End-to-End Signalling (EES) package (#10) and the Message Waiting Center (MWC) package (#46).

The Flexible Tones and Cadences (FTC) package (#125) is also needed, if an audible MWI is provided to analog telephone sets, just as the Message Intercept (MINT) package (#163) is required if a Message Waiting announcement is provided instead.

DPNSS1 Route Optimization/MCDN Trunk Anti-Tromboning Interworking

The DPNSS1 Route Optimization (RO)/Meridian Customer Defined Networking (MCDN) Trunk Anti-Tromboning (TAT) Interworking feature provides RO and TAT interworking at DPNSS1/MCDN gateway nodes.

RO/TAT interworking scenarios

RO/TAT interworking within a DPNSS1 to MCDN gateway

The following example presents a case where RO/TAT interworking occurs within a DPNSS1 to MCDN gateway.

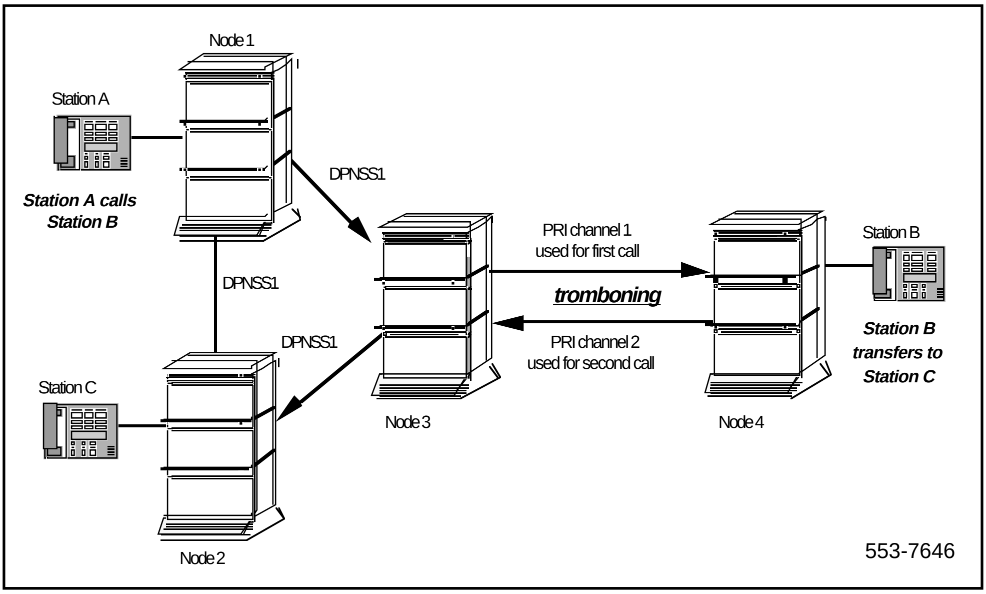
> ***Note:*** In this example, we have used the case where a call has been redirected due to Network Call Transfer. The same functionality would apply if the call had been redirected by Network Call Forward No Answer, and Network Hunting, or modified by Network Call Transfer or Attendant Call Transfer.

Station A, located at Node 1 on the DPNSS1 side of the DPNSS1/MCDN gateway, calls Station B located at Node 4 on the MCDN side of the gateway. Station B activates Network Call Transfer to Station C, located at Node 2 on the DPNSS1 side of the gateway.

Interworking and gateways

Upon activation, the existing call is put on hold and a new call is originated to Station C. Station C Answers. Station B completes the call transfer, leaving A connected to C using two DPNSS1 trunks and two PRI trunks.

Figure 90
DPNSS1/MCDN scenario with Network Call Transfer, before RO/TAT optimization

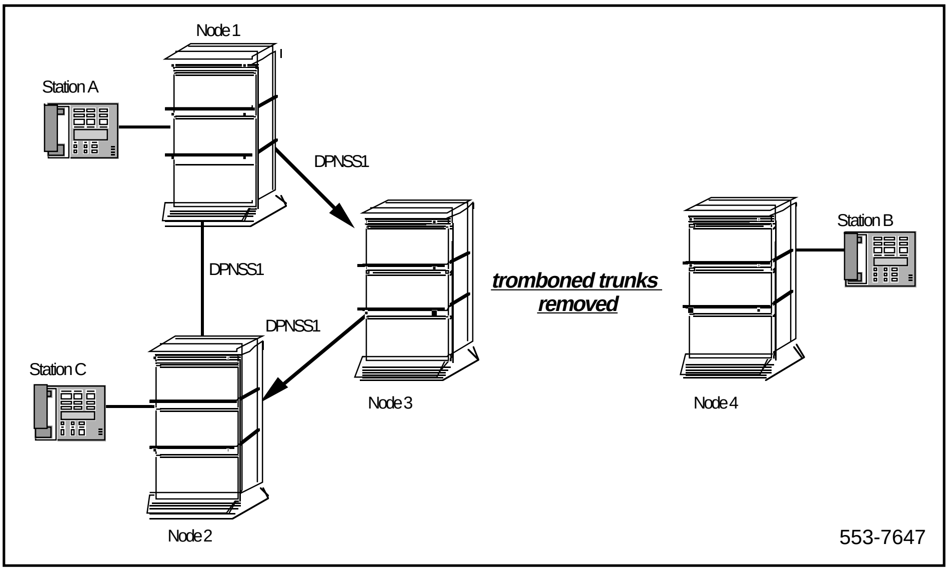


Note: The Network Call Transfer/Three Party Service gateway is not supported at the gateway Node 3. Therefore, RO is not initiated at Node 1, and the non-optimized DPNSS1 trunks remain connected.

On the MCDN side, TAT is initiated at Node 4. The call between A and C is bridged, and the redundant PRI trunks are removed between Node 4 and Node 3. For the meantime, the non-optimized DPNSS1 trunks remain connected.

Interworking and gateways

Figure 91
DPNSS1/MCDN RO/TAT Interworking scenario, after TAT has been applied

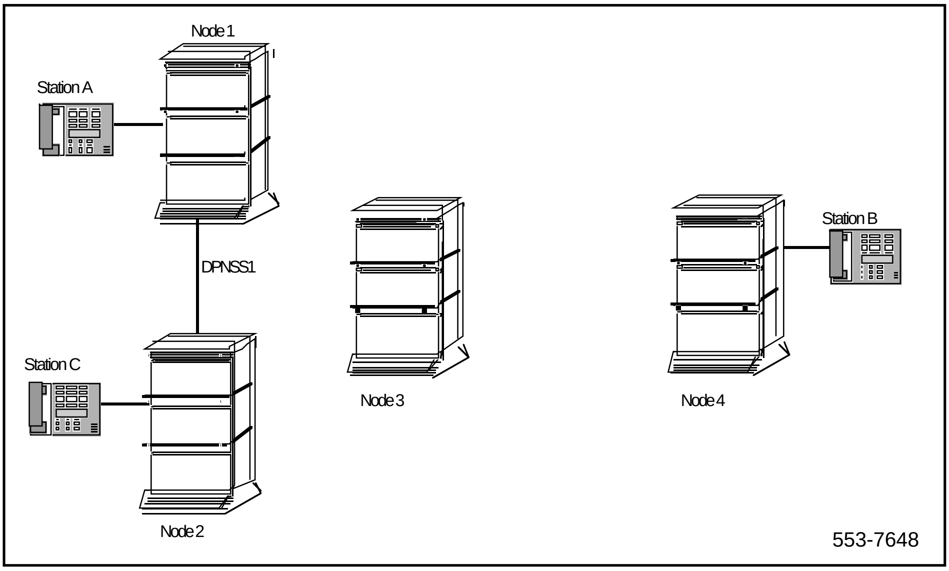


When TAT is completed on the MCDN side, The RO/TAT Interworking feature initiates RO on the DPNSS1 side by simulating a transfer at the gateway Node 3. The Three Party Service feature initiates signaling to update displays. Then, RO is initiated at Node 1, the originating node. The DPNSS1 trunks are dropped between Node 3 and 2 and Node 3 and Node 1, with Station A and Station C being connected over one DPNSS1 trunk.

> ***Note:*** If a non-optimum route is used at the originating node or at any transit node, Route Optimization may start from Node 1 (the normal RO operation for the first call optimization) or Node 3 (the normal RO operation for the second call optimization), before TAT is completed. If TAT invocation is received on Node 3 while RO is being applied between Node 1 and Node 3 or Node 3 and Node 2, the completion of TAT is delayed until RO is finished.

Interworking and gateways

Figure 92
DPNSS1/MCDN RO/TAT Interworking scenario, after RO has been applied



> ***Note:*** If Station A is an attendant, TAT takes place on the MCDN side of the gateway but RO cannot take place on the DPNSS1 side. This is an RO limitation.

RO/TAT interworking within a DPNSS1 to MCDN gateway

The following example presents a case where RO/TAT interworking occurs within an MCDN to DPNSS1 gateway. Here, too, we are using the case of a call being transferred (using the DPNSS1 Three Party Service feature) across the gateway.

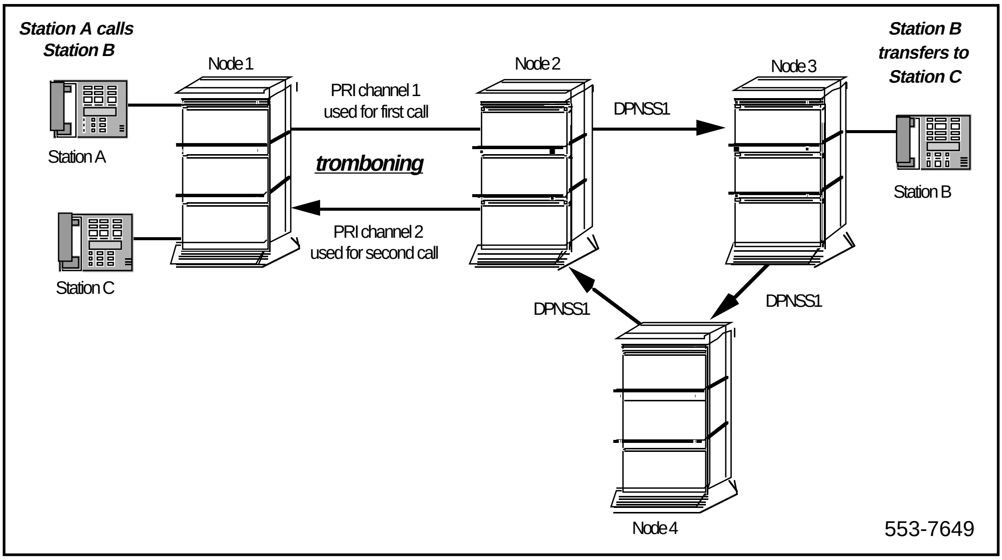
Station A, located at Node 1 on the MCDN side of the MCDN/DPNSS1 gateway, calls Station B located at Node 3 on the DPNSS1 side of the MCDN/DPNSS1 gateway. Station B transfers the call (using the Three Party Service feature) to Station C, also located at Node 1 on the MCDN side of the gateway.

Upon activation, the existing call is put on hold and a new call is originated to Station C.

Interworking and gateways

Station C Answers. Station B completes the call transfer, leaving A connected to C using three DPNSS1 trunks (in the example, the call is routed through Node 4) trunks and two PRI trunks.

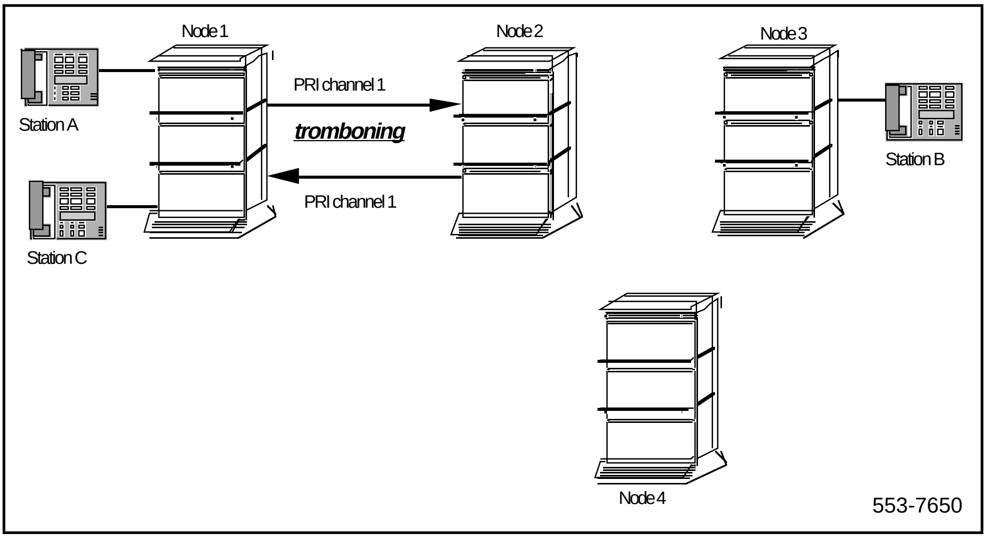
Figure 93
MCDN/DPNSS1 RO/TAT Interworking scenario, before RO has been applied



Once Three Party Service messaging has taken place, Node 2 initiates RO. The initial DPNSS1 routes are cleared. Node 2 becomes a MCDN/MCDN transit node, and the two tromboning PRI routes between Node 2 and Node 1.

Interworking and gateways

Figure 94
MCDN/DPNSS1 RO/TAT Interworking scenario, after RO has been applied



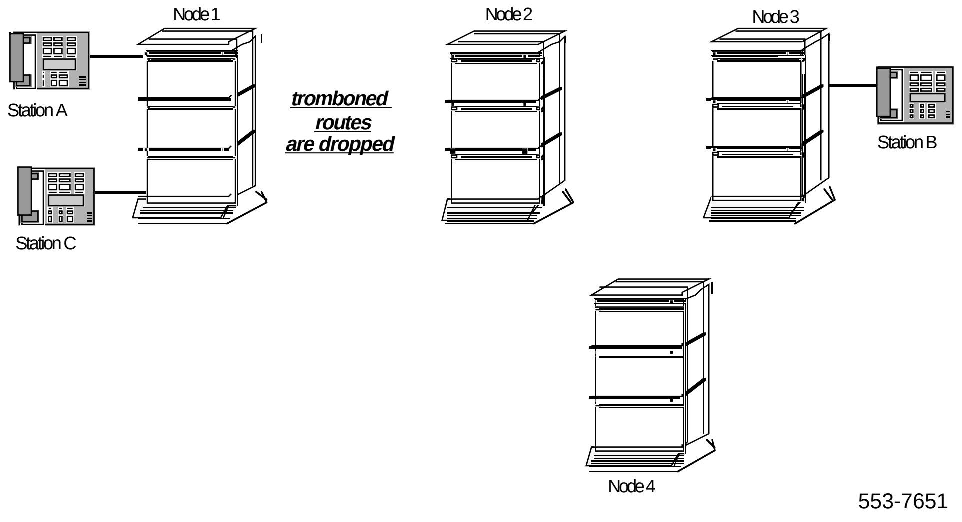
As soon as RO is completed, the RO/TAT initiates TAT at gateway Node 2. After TAT has been completed at Node 1, Node 2 simulates a transfer message to both Station A and Station C. This allows the Network Call Redirection feature to update the displays.

> ***Note:*** If the originating and terminating nodes are one and the same, and if this node is not a tandem node, as is the case for Node 1 in our example, the displays are updated without the notification from the Network Call Redirection feature.

TAT is then completed. The redundant routes are cleared, and Station A and Station C are bridged.

Interworking and gateways

Figure 95
MCDN/DPNSS1 RO/TAT Interworking scenario, after TAT has been applied



Note: If Station A is an attendant, and the Network Attendant Service feature is configured, Station B cannot transfer to Station C, and no optimization can take place. If NAS is not configured, Station B may transfer to Station C, and optimization will take place as described in this example.

Note: In the case of call diversion on the DPNSS1 side (Diversion Immediate, Diversion on Busy, and Diversion on No Reply), there is no interaction with the RO/TAT Interworking feature (the interaction occurs between the Diversion and TAT features). In the case of tromboning on the DPNSS1 side, the Diversion feature clears the DPNSS1 tromboning trunks before Station C answers the call. When C answers, TAT is applied transparently.

Note: Node 1 cannot be a DMS switch for the RO/TAT Interworking feature to operate.

Interworking and gateways

DASS2/DPNSS1 to ISDN BRI, QSIG, and EuroISDN gateway

The following services are provided with the DPNSS1 to ISDN BRI (line and trunk applications), QSIG, and EuroISDN gateways:

- basic call service (3.1 kHz, speech, 64 Kbit/s restricted/unrestricted digital information)
- overlap sending and receiving
- 64 Kbit/s bearer capability

ISDN BRI supports a gateway between IDA (DPNSS version of PRI) and MCDN protocols for basic call features. Ring again-type features are not supported.

Call Diversion Notification Enhancements

In X11 Release 24, the Call Diversion Notification functionality has been extended to include a QSIG/DPNSS1 gateway. Refer to the QSIG section in this book for more information.

Call Transfer Notification

In X11 Release 24, when a call transfer occurs over a QSIG PRI2 or BRI trunk link, a notification of the transfer is sent to the originating party and the party to whom the call is transferred. The transfer must be done using the Call Join feature (Multi-Party Operations). ANF Path Replacement takes effect, if applicable, in order to obtain a more efficient connection.

The information appears on the displays of the originating and terminating telephones. It includes:

- the redirection and originating number
- redirection and originating name
- redirection reason mnemonic (a code defined for Call Transfer).

The same notification occurs over a QSIG/DPNSS1 gateway.

Although Call Transfer by Call Join was supported over a QSIG link as early as Release 20, there was no notification to the users that a transfer had taken place.

Interworking and gateways

The method of Operation Coding is by Object Identifier for ETSI interfaces or Integer Value for ISO interfaces.

QSIG/ETSI GF Enhancement

In X11 Release 24, the enhancement to the GF transport platform comprises the following capabilities:

- modified call control to support call-independent gateways from QSIG to EuroISDN, MCDN and DPNSS1 signaling protocols
- modified call-related Application Protocol Data Unit (APDU) transport to ensure that a call that is being optimized during QSIG Path Replacement is using the optimum path (the first route in the route list index). This modified mechanism is also needed so the Facility Information Elements carried in a SETUP message which has encountered congestion are included in the alternate call, if alternate routing is performed at a transit node. The Call Completion to Busy Subscriber feature and ANF Path Replacement feature have been modified to use this new capability.
- more accurate calculation of the remaining available length for APDUs in basic call control messages for QSIG GF and ETSI GF transport.

The APDU call-independent transport mechanism is connection oriented. Connectionless APDU call-independent transport is not supported.

When a call-independent SETUP message includes several Facility Information Elements (FIEs) belonging to different Supplementary Services, the gateway is not supported.

DPNSS1 (APNSS) to R2MFC gateway

The DPNSS1 to R2MFC interworking provides an interface for R2MFC DID and DOD calls and DPNSS TIE trunks.

Interworking between R2 MFC TIE trunks and DPNSS TIE trunks is not supported.

Interworking and gateways

For R2MFC DID calls routing onto DPNSS TIE trunks, this feature provides the mapping of the DPNSS message to the corresponding R2MFC signal and returns the signal to the CO. It also provides the option of delivering the Calling Number Identification (CNI) on the DPNSS route, if CNI is available for the call.

For DPNSS TIE trunk calls routing onto R2MFC DOD trunks this feature provides the mapping of the R2MFC status signal to the corresponding DPNSS message and returns the message to the originating PBX.

The R2 MFC to DPNSS gateway also provides the following enhancements in order to provide CNI support for R2 MFC DID to DPNSS tandem calls:

- The ability to request CNI for an incoming R2 MFC call immediately after a predetermined number of digits are received. The allowable range for this option is 0 to 7.
- The ability to request CNI for an incoming R2 MFC call immediately after an ESN code is dialed. The ESN codes recognized for this purpose are Distance Steering Codes (DSC), Trunk Steering Codes (TSC), AC1s, and AC2s.

R2MFC to DPNSS gateway interacts with Virtual Network Services

If a call on a DPNSS or R2MFC trunk is tandeming to the R2 MFC or DPNSS trunk on a VNS call, the R2MFC to DPNSS gateway capability does not apply.

If a DPNSS/R2MFC tandem is encountered during the routing of a VNS call, the gateway feature applies.

DPNSS1 and Asia Pacific ISDN interworking

The Information Elements (IEs) supported on the Asia Pacific interfaces which have equivalents in DPNSS1 interworking are translated on an IE by IE basis. IEs which have no matching DPNSS1 element are discarded. Supported IEs with DPNSS1 equivalents pass through with appropriate mapping.

Interworking and gateways

DPNSS1 gateway interworking with other signaling systems

Please be advised that to date, DPNSS1 has only been launched as part of the Meridian 1 product in the United Kingdom, and that the gateway working is only supported between DPNSS1 and certain interfaces. For more information regarding gateway interworking, contact Nortel Product Management.

If DPNSS1 trunks are being used for VNS, the gateway nodes have the same features and limitations as the usual DPNSS1/MCDN gateways without VNS. Any feature for which a DPNSS1/MCDN gateway does not exist will be stopped at the DPNSS1/VNS node.

For a node interworking DPNSS1/VNSS and another connectivity, the features and limitations of an MCDN gateway apply not the features and limitations of an DPNSS1 gateway.

BNE DPNSS gateways

Introduced in Release 25, the BNE/EuroISDN Call Diversion feature allows notification to occur when the private network is a multi-node network using the following protocols:

- QSIG
- DPNSS
- MCDN

Refer to “BNE Call Diversion QSIG, MCDN and DPNSS gateways” on page 749 for information.

R2MFC

R2MFC DID/ DTMF DOD Trunks

This capability will be used in Denmark and Thailand.

This feature allows a single trunk to use the functionality of R2MFC for incoming calls and DTMF signaling for outgoing calls.

Interworking and gateways

The DID trunk route is programmed as Incoming and Outgoing with MFC signaling. Only the incoming MFC table is defined. Each DID trunk has a Class of Service of MFX. If the trunk is seized for an incoming call, the software finds the incoming MFC table. Therefore MFC signaling is used. If the trunk is seized for an outgoing call, the software selects DTMF signaling.

R2MFC on DTI 1.5

R2MFC signaling was permitted first on analog trunks and DTI2 links. Then, in Release 21, R2MFC signaling was allowed on DTI 1.5 links.

This capability was requested for the Caribbean and Latin America markets. However, t he capability can be used for DID or TIE trunks where T-1 and R2MFC signaling is used in the public network or where T-1 links are used with R2MFC signaling in the private network.

The Alternate Loss Plan is not supported for this capability. Therefore, it cannot be used in Australia.

R2MFC to DPNSS gateway

This gateway provides the following enhancements to the R2MFC incoming Calling Number Identification (CNI) request functionalities:

- there is an option to request CNI for an incoming R2MFC call immediately after a pre-determined number of digits are received
- there is an option to request CNI for an incoming R2MFC call immediately after an ESN code is dialed

Interworking and gateways

R2MFC to MCDN gateway

R2MFC CNI/CDR Enhancements

The R2 Multifrequency Compelled Signaling (R2MFC) Calling Number Identification (CNI)/Call Detail Recording (CDR) Enhancements feature provides the following capabilities across an R2MFC/ISDN gateway:

- the mapping of the R2MFC CNI to the ISDN CLID, and vice versa
- the location of the captured CNI in the CDR is an option. The CNI may appear in the digits field, the CLID field in line two or not appear at all in the CDR
- the CNI is provided to the ACD MAX, Network ACD, and Radio Paging equipment, similar to how the CLID has been provided before the introduction of this feature

Examples of CDR formats

The following examples provide cases of the CNI (23008) appearing in the digits field, in the CLID field in line two, and not appearing at all in the CDR.

CDR record with CNI in digits field

N 003 00 T078001 T008001 31/05 1:40:05 05:30.5 C23008
& 0000 0000

CDR record with CNI in CLID field

N 003 00 T078001 T008001 31/05 13:40:05 0:05:30.5 71082317
&23008xxxxxxxxxxx 0000 0000

CDR record with no CNI appearance

N 003 00 T078001 T008001 31/05 13:40:05 0:05:30.5 71082317
& 0000 0000

Operating parameters

The feature is applicable to Meridian 1 Options 11C, 51C, 61C, 81 and 81C systems.

The maximum length of the CLID field in the CDR is 16 digits.

Interworking and gateways

On the ISDN side of the R2MFC/ISDN gateway, the ISDN access may be via ISDN Primary Rate Interface (PRI), ISDN Signaling Link (ISL), Virtual Network Services (VNS) or Basic Rate Interface (BRI) trunks.

This feature does not affect the manner in which the CNI is composed at the originating switch of an R2MFC/ISDN gateway.

If the CNI cannot be composed at a R2MFC/ISDN gateway tandem switch, the CNI DN and the Trunk ID will be sent in the CNI (as was the functionality prior to the introduction of this feature).

Feature interactions

Calling Party Privacy and Display Calling Party Denied

If the CLID is received with presentation denied, it is not mapped to the CNI. Instead, the CNI is composed of the CNI DN and the Trunk ID. Optionally, the CNI request may be set to ECNI (the CNI End-of-CNI R2MFC level 1 forward signal).

EuroISDN Continuation

The outgoing CLID element of the EuroISDN Continuation feature is mutually exclusive with the R2MFC CNI/CDR Enhancements feature. If the CLID is to be composed from the EuroISDN Continuation feature, it will not contain the CNI. If the CLID is to be composed from the CNI, no prefixes will be added to the number.

Feature Group D

Feature Group D trunks (in the U.S. A.) do not support CNI. If a CNI is available in addition to the CLID on a Feature Group D trunk, the CLID of the Feature Group D trunk would be used for the CLID.

In-Band Automatic Number Identification (IANI)

Inband ANI trunks do not support CNI. If a CNI is available in addition to the IANI on an IANI trunk, the IANI would be used for the CLID.

Incoming Trunk Programmable CLID

Incoming Trunk Programmable CLID takes precedence over the R2MFC CNI/CDR Enhancements feature. If the outgoing ISDN trunk is allowed to send a billing number, the billing number is sent out as the CLID, not the CNI from the incoming trunk.

Interworking and gateways

M911

M911 trunks do not support CNI. If a CNI is available on an M911 trunk in addition to the ANI, the ANI would be used for the CLID.

R2MFC to DPNSS1 gateway

The R2MFC CNI/CDR Enhancements feature uses the CNI request enhancement (Upfront CNI) developed for the R2MFC to DPNSS1 gateway.

CIS MF Shuttle

Networking Features

Only B-Free and B-Busy condition transmitting and receiving are supported. CIS MF Shuttle supports B-Free/B-Busy networking interactions with the following signaling protocols:

- R2/MFC
- ISDN (DPNSS, QSIG, EuroISDN, and MCDN)
- CIS MF Shuttle
- CIS Dial Pulse DID and CO

Incoming CIS MF Shuttle trunks only accept the networking information from the trunks which support the Direct Inward Dial (DID) gateways and the CIS Dial Pulse outgoing CO trunks.

Outgoing local CIS MF Shuttle trunks may pass the B-Free/B-Busy networking information to the R2/MFC, ISDN, CIS MF Shuttle and CIS Dial Pulse DID trunks.

Interworking and gateways

ANI Gateways (Release 24)

The ANI digits which are received from the CIS CO calling party as a response to the automatic ANI request are propagated to the Meridian 1 terminating party, if it is capable of receiving the CNI digits.

The ANI digits are propagated to the following terminating types:

- R2MFC trunks - the ANI to R2MFC CNI mapping is performed in the following way: all the ANI digits except for the ANI Calling Party Category Code (CAC) are used for the CNI composition, the ANI CAC is converted to the Multi-frequency Compelled (MFC) CNI CAC according to the CAC conversion tables.
- ISDN trunks (MCDN, EuroISDN, QSIG), and DPNSS1. The ANI to CLID/OLI mapping is based on the R2MFC CNI to CLID mapping.
- CIS trunks - the ANI to ANI mapping is implemented in the framework of the CIS ANI Digits Manipulation and Gateways Enhancements feature described in the R2MFC section. The ANI information that is received from the incoming CIS DTI2 trunks is used by the CIS Gateways Enhancements feature to compose the ANI information to be downloaded to the outgoing CIS trunks.

Asia Pacific (APISDN) Connectivity Phase III

Call Party Name Display

MCDN networks cannot send or receive Calling Party Name Display (CPND) to or from an APISDN network.

Network Call Redirection

Notice of redirection (through features such as Call Forward All Calls, Hunting, and Call Forward No Answer) or call modification (through features such as Conference and Call Transfer) and resulting changes to the CLID are not transmitted from an MCDN to an AP ISDN interface.

Interworking and gateways

India, Taiwan, Philippines

As of X11 Release 24, interworking is provided with the following other ISDN or non-ISDN interfaces:

- ISDN CO connectivities that already exist (Numeris and Swiss Net 2, for example)
- Taiwan R1 Modified Signaling
- private networks (MCDN and QSIG, for example)
- other digital signaling systems (DPNSS1 and DASS2 and 1.5 or 2.0 Mbps DTI, for example)
- analog signaling systems (R2/MFC, Multifrequency Compelled Signaling for Socotel (MFE), dial pulse and Digitone)

Meridian 1 to Meridian 1 to Central Office

Advice of Charge

The AOC information is not transported between Meridian 1 switches. Therefore, if a tandem call is made from one Meridian1 to another and then to a Central Office, the caller’s display is not updated with the charge information. The charge information is printed in the CDR record at the switch that made the connection to the Central Office.

Interworking and gateways

Meridian 1 and Passport interworking

Passport integrates your Meridian voice network and your enterprise data networks. It manages bandwidth, as needed, between voice, data, image and video transmissions. It assigns bandwidth only when traffic is present. This allows you to create custom network solutions that accommodates your changing needs. You make the most of your bandwidth, gain faster response time, cut operating costs and are able to evolve smoothly into an ATM environment, if your multimedia traffic warrants it.

Passport gives you two major advantages:

- **Lower Voice Networking Costs**
  - Passport can cut voice trunking costs by reducing the bandwidth needed to carry voice traffic and by consolidating voice and data traffic into a common WAN. This arrangement can cost less than using dedicated voice links.
- **Multimedia Wide Area Networking**
  - By adding Passport to your Meridian 1 systems, you can build a WAN to handle voice, data, image and video traffic. You can also use Passport to consolidate all multimedia traffic generated at a Meridian 1 location and deliver it to a larger existing WAN.

Interworking and gateways

Passport Overview

Passport allows voice, data, image, and video traffic between nodes to be consolidated onto a highly efficient multimedia wide area network.

Passport's unique architecture supports both cell and frame switching and trunking. This flexibility helps to maximize the traffic volume and throughput over enterprise network facilities. Frame switching is well suited for data since it matches the characteristics of most data traffic and is efficient for handling bursty data traffic. Cell switching is well suited to voice and video applications because of its consistent delay characteristics. Passport's ability to support both cell and frame switching make it ideal for a multimedia enterprise network environment.

Passport interfaces with the Meridian 1 system by connecting T1 or ISDN PRI links from Meridian 1 to Passport DS1V Function Processor cards. Data and video communication systems are interfaced with Passport by connection to V.35 or V.11 Function Processor cards. V.35 and V.11 are internationally recognized standard interfaces for data communications equipment. Voice traffic from the Meridian 1 system and data and video traffic are switched through Passport and multiplexed onto enterprise network facilities. Passport supports Nortel Networks's frame/cell trunking over DS1 or DS3 leased line facilities as well as ATM trunking over private and public ATM networks.

Passport’s multimedia capabilities give you:

- A Simpler Network
  - When you reduce the number of networks and network elements, you reduce costs.
- Dynamic Bandwidth Management
  - Passport logically allots bandwidth to applications only when traffic is present, instead of wasting it by reserving it for specific applications. It also reduces your network's bandwidth requirements by compressing and statistically

Interworking and gateways

multiplexing voice traffic. The result can be an 85 percent savings in bandwidth for voice traffic. You can use this bandwidth for any other service on your network.

- Service Quality Assurance
  - Passport allows you to set service priorities and reserve bandwidth to ensure service availability at the lowest network cost.
- Frame and Cell Switching and Trunking
  - Frame switching works best for bursty data traffic while cell switching is ideal for delay-sensitive voice and video traffic. Passport combines both for flexible, high-quality network handling.
- LAN Interconnection
  - You can seamlessly unite LAN-based workgroups across your corporate WAN by interfacing existing routers to Passport. This costs less than using separate router networks.
- High Performance
  - Passport offers faster response times and bandwidth-on-demand.
- Protects your investment in your Meridian 1 system.
  - It supports existing computing and networking standards. It provides for future growth through scalable, modular, ATM-based architecture.

For more information about how Passport can be integrated into your Meridian 1 network, talk to your system supplier.

Introduction to trunks

Telephone users connected to Private Branch Exchange systems (PBXs), such as the Meridian 1, need to make calls to internal users connected to the same system and to users connected to other systems. External systems can be Central Office switches (public network) or other PBXs and key systems (private network). To allow these calls, PBXs are connected to other systems with pairs of wires.

A minimum of one pair of wires is required for each connection called a trunk. However, some kinds of trunks are comprised of more than one pair of wires.

One analog trunk can handle one conversation at a time. If the trunk is set up for digital transmission, several conversations can be carried on one pair of wires, simultaneously. This is referred to as multiplexing.

Identical trunks are grouped together in trunk routes. If you require ten of a certain kind of trunk for the call volumes you are expecting, you would program a route in the system database first. Then, you would associate all ten of these trunks with that route. As of X11 Release 24 you can have up to 510 trunks in one route. Prior to Release 24, the maximum number of trunks in a route was 254.

The types of trunks included in this Appendix are the generic types that are the most common. There are variations of these types offered in different countries and by other common carriers in North America. Trunk-related tariffs and country-specific regulations may also affect the types of trunks available in a certain market region. In many countries, with the deregulation of telephone companies, other carriers offer their own variations on conventional trunk types. The principles discussed here can be applied to the variations you may encounter in your market area.

Introduction to trunks

It is a network planner's responsibility to become aware of the many trunk choices offered in the regions in which the planner has systems and to make a point of staying informed of changes that may occur in those regions.

Different kinds of trunks are available for the different kinds of calls that users make. The most common types of trunks are:

- Central Office
- Direct-Inward-Dial or Direct-Dial-In
- Foreign Exchange
- TIE
- Wide Area Telephone System (WATS)

Ask representatives from your telephone company (and other common carriers if they exist), for information on the different trunk types that are available in your market region.

Central Office trunks

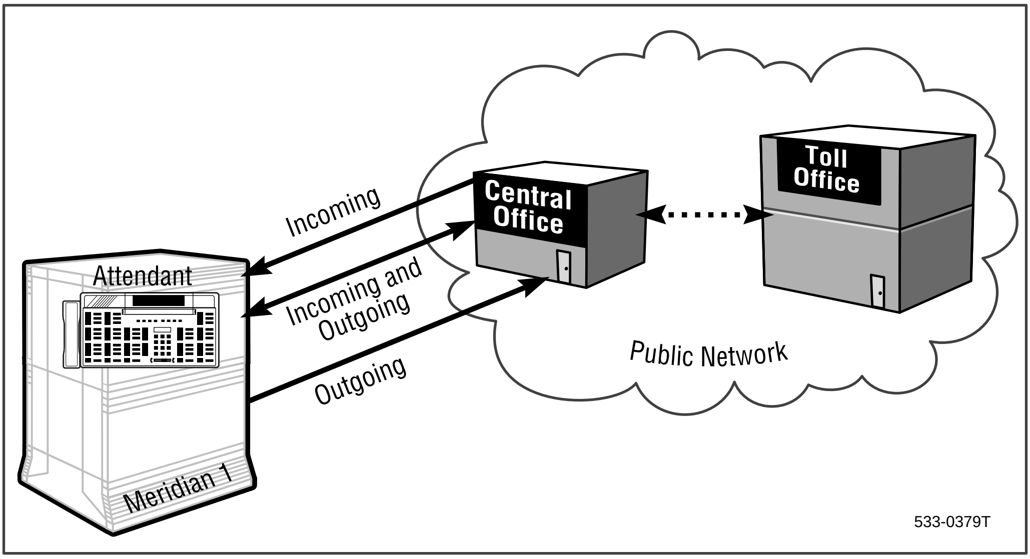
These two-wire connections to the Central Office (or Exchange Office) nearest to the Meridian 1 are used for public network local calls and for long distance, toll calls. In North America they are often called "dial 9" trunks because the most common access code for these trunks is the digit 9. You can program the access code that best suits your requirements, however.

There are three configurations of these trunks to choose from:

- Incoming
- Outgoing
- Incoming and Outgoing

Introduction to trunks

Figure 96 - Central Office trunk configurations



As the name suggests, Incoming Central Office Trunks (COTs) are set up to receive calls from the public network and send them into the users at the Meridian 1. Outgoing calls from the Meridian 1 to the Central Office (CO) will not be handled by these trunks.

Outgoing Central Office Trunks (COTs) are set up to handle calls from the users at the Meridian 1 and send them to the public network. Incoming calls from the Central Office to the Meridian 1 will not be handled by these trunks.

Incoming and Outgoing Central Office Trunks (COTs) handle both incoming and outgoing calls to and from the Meridian 1.

Some planners use two or more of these types of trunks on one system. For example, you may want to install both Incoming COTs and Incoming and Outgoing COTs to ensure that you have incoming trunks available for your customers to call in on that will not be occupied with outgoing calls during busy times.

Proper provisioning is important to ensure there will be sufficient trunks for the call volumes you are expecting and that the amount of blockage, if any, will not exceed the level you find acceptable.

Introduction to trunks

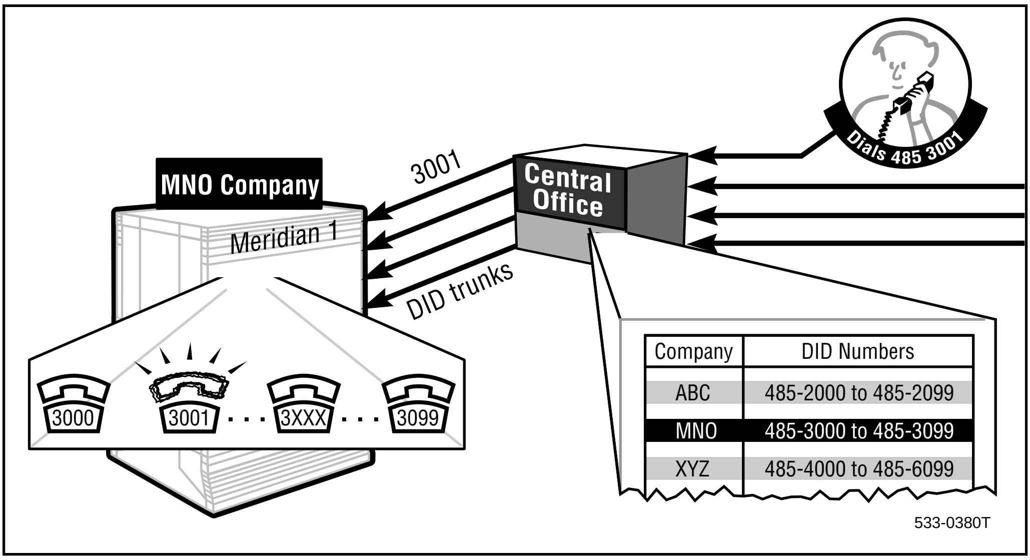
Central Office trunks are answered at main answering positions called attendant consoles. The attendant answers the externally-originated calls and extends them to the internal users. Where there is no console, night service or automatic answering services such as Meridian Mail Automated Attendant can answer instead. The attendant can also answer internally-originated calls.

Direct-Inward-Dial (DID) or Direct-Dial-In (DDI) trunks

These trunks extend from the Central Office to the Meridian 1. They serve internal users who each require a unique telephone number that can be reached directly from an external telephone.

For example, instead of having an attendant handle incoming calls to a group of sales people in your organization, you can assign a DID telephone number to each sales person. Customers can dial into a sales person directly, without being transferred by the attendant first. You can reduce the number of attendants you need when you install DID trunks.

Figure 97 - DID trunk configuration



You usually reserve and use these numbers in blocks. They are most economical when you install them for a large group of users who receive a high volume of calls.

Introduction to trunks

Call processing works in the following manner. Blocks of public network telephone numbers are assigned to you by the service provider. The last few digits in the DID telephone numbers are outpulsed from the CO to the Meridian 1. You request the last three, four, five or more digits to be outpulsed, based on your numbering plan at the Meridian 1. If your Meridian 1 can translate these digits intact as Directory Numbers for your telephones, so much the better. If the digits assigned do not match your internal DN numbers, you can use programming to delete and insert digits as required. Talk to your system supplier about how this can be done, if you require it. You may also decide to renumber your telephone DNs to coordinate with the assigned DID numbers.

> ***Note:*** In X11 Release 24, the Flexible DID feature allows you to assign DID numbers to telephones as you need them. This feature is designed for the hotel environment using a Property Management System (PMS). It helps to reduce the number of DID numbers required, since the DID DNs are not assigned to telephones permanently but only when required.

Provisioning is an especially important issue with these trunks since all the DID users share the trunks in the DID trunk group. You must ensure that you have sufficient trunks installed to handle the expected call traffic to the DID numbers. If you do not have sufficient DID trunks, callers will be blocked at the Central Office, unable to find an available trunk to call the user at your Meridian 1.

Common ratios of users to trunks is shown in the following chart.

Table 83 Common DID trunk provisioning ratios

Traffic levels	User to trunk ratio
Low	8:1
Medium	6:1
High	4:1

Introduction to trunks

If you are unsure of the traffic levels to expect, estimate the trunks you need using the preceding chart and follow up with a traffic study after the trunks are installed. While you wait for the study to be done, survey the DID users to find out if they are receiving customer complaints about blockage problems.

DID users can receive calls from internal users and the attendant, if you have one. If you want some users on your system to receive calls from non-DID trunks, you can have DID and non-DID trunks on one system. If an external caller dials the main number of the company and the call comes in on a Central Office Trunk, the attendant can extend the call to a non-DID or DID telephone user.

Ensure that you program your system features so that unanswered and busy DID telephones handle calls properly. Since the attendant is not involved with call processing, you must rely on system and telephone set features to route calls to backup answering positions in these situations.

Foreign Exchange trunks

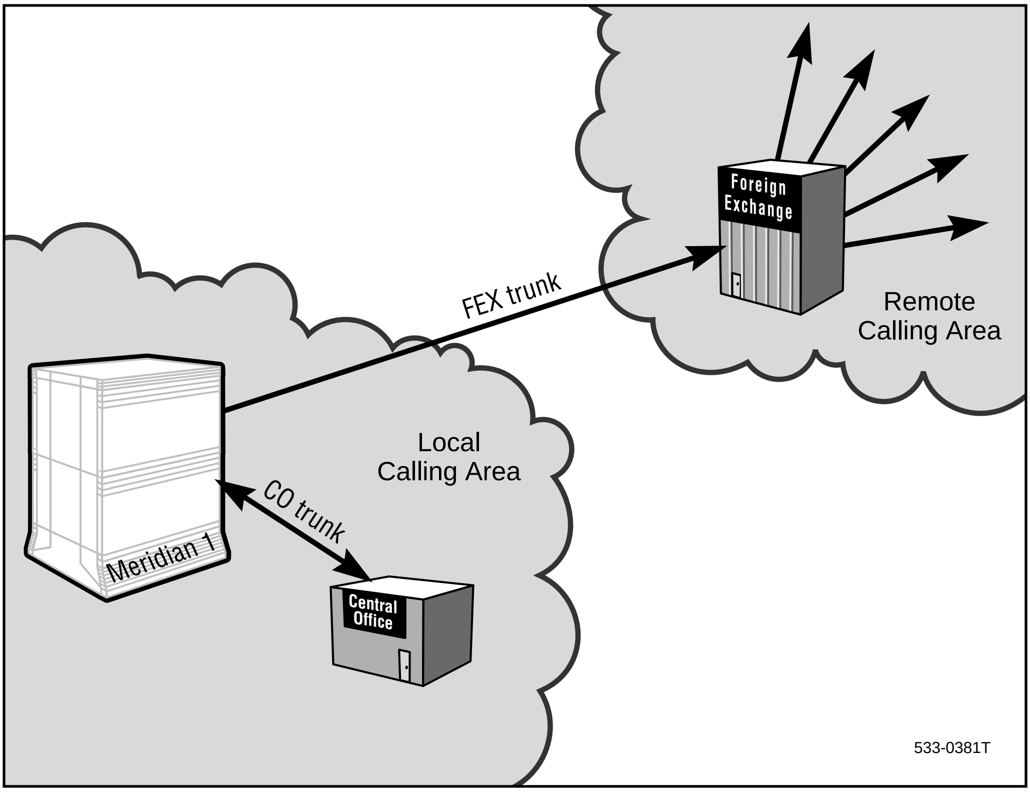
If the users of a system make or receive numerous calls to/from an area served by a Central Office other than the local one, it may be economical to install one or more direct trunk connections to the remote Central Office (CO), otherwise known as a “Foreign Exchange”.

This FEX connection allows users to make calls to the remote calling area served by that CO without having to dial the call as a toll call. FEX trunks are usually billed at a fixed cost per month based on the distance between the Meridian 1 and the remote CO.

You must get data on the call traffic going to that foreign exchange area in order to provision sufficient trunks. If you have the call volume to justify installing at least one FEX trunk, this fixed cost can be substantially lower than the associated toll call costs would be.

Users and programmers must be kept informed of the exchange codes that are local to the Foreign CO so they can use the trunk properly.

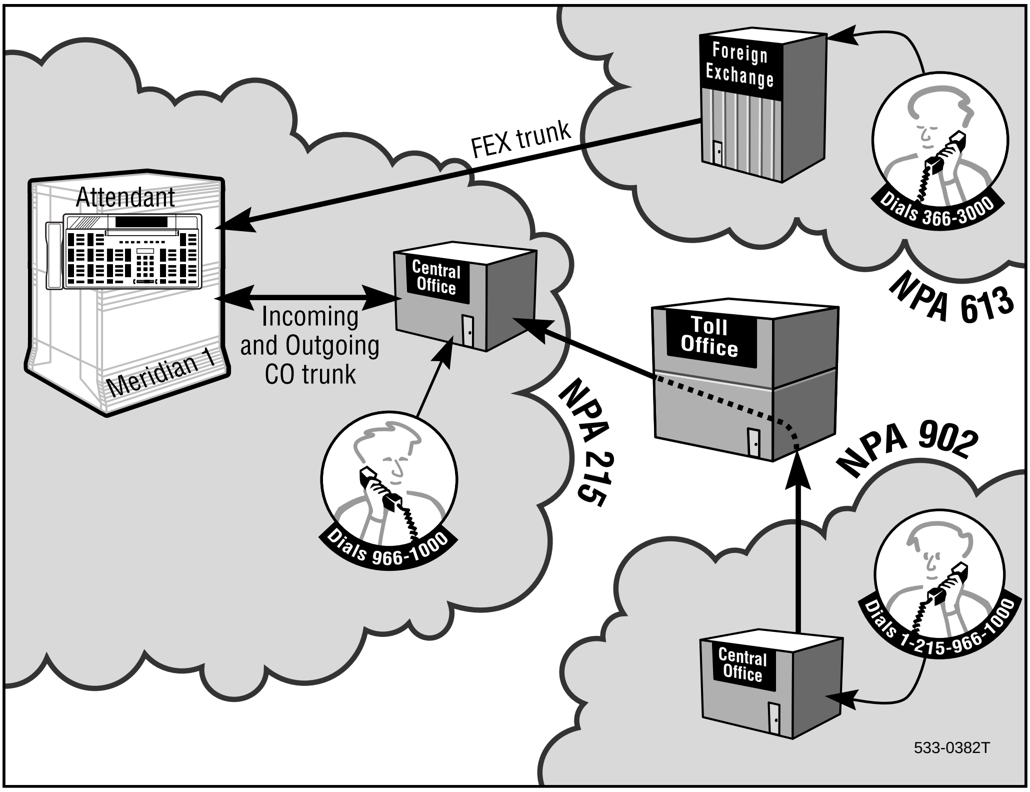
Figure 98- Outgoing usage of an FEX trunk



Introduction to trunks

Callers located in the remote calling area can dial the FEX telephone number, thinking they are dialing a number in the local calling area. The call will terminate on the Meridian 1 system located some distance away. The caller does not incur toll charges.

Figure 99 - Incoming usage of an FEX trunk



Introduction to trunks

TIE trunks

Direct access between private switches in a network can be provided using TIE trunks. These are two or four wire connections that allow users to bypass the public network to call users at another system in their network. The TIE trunk may travel through the CO where transmission may be improved but the CO does not perform switching on the TIE trunks. TIE trunks can also be installed privately between buildings situated close to each other, on hospital or university campuses for example, and not route through COs at all.

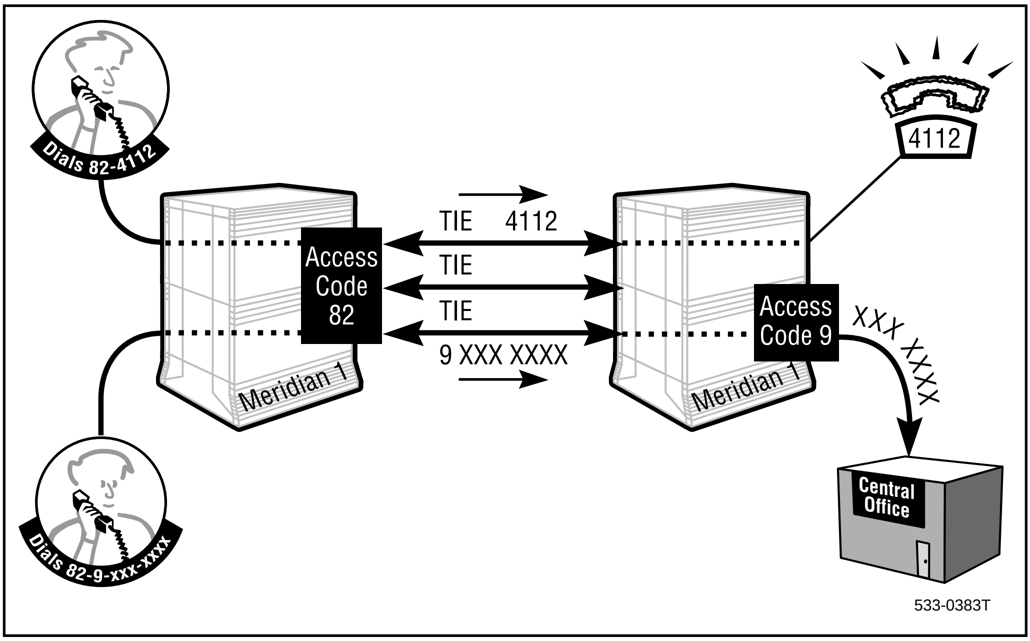
When users dial the access code assigned to the TIE trunk route, the system at the other end will process the digits that follow.

If the digits are those in a DN at the terminating switch the call will be sent to the telephone dialed. This is an on-net call. If the digits are those in a trunk route access code and restrictions do not prevent it, the call will be routed out of the terminating switch on a trunk in that route. This is referred to as “tail end hop-off” or an off-net call.

TIE trunks are billed at a flat rate every month. If calls routed on the public network between the two switches would normally be toll calls, having TIE trunks can save you a substantial amount of money in network costs. Using the TIE trunks to hop-off to the public network can also reduce your toll bills, where this is permitted.

Introduction to trunks

TIE trunks used for on-net and off-net calls



Wide Area Telephone Service (WATS) trunks

This type of trunk is offered in North America only. There are two kinds of WATS trunks:

- In-WATS
- Out-WATS

In-WATS trunks are sometimes called 1-800 trunks or 800 services. These trunks are designed to allow callers to reach the “owner” of the 1-800 trunk by calling toll free, even though the caller may be a great distance away. The caller dials a number beginning with the digits 1800, followed by a seven digit number.

The zone covered by the WATS trunk determines the cost per month. Zones include specific area codes from which calls can be made. For example, if the owner of the trunk pays for a zone 2 WATS trunk then calls from zones 3 and higher are not permitted but may be allowed from the area codes in zone 1 and 2. It may not be economical to allow calls from the area codes in zone 1. You may be able to block those calls at the Central Office. If there may be many calls from zone 1, it may be a good idea to install a separate zone 1 trunk or trunks. The bottom line is that you must have data that tells you how much traffic to expect from each area code so you can order the correct zone and the correct number of WATS trunks to carry the traffic.

Where Out-WATS trunks are offered, they are designed to carry outgoing calls to certain zones at a more economical rate than regular toll calls. It is the planner’s job to compare the rates for calls on CO trunks, WATS trunks, and other common carrier trunks.

Introduction to trunks

Figure 100 - Use of In-WATS

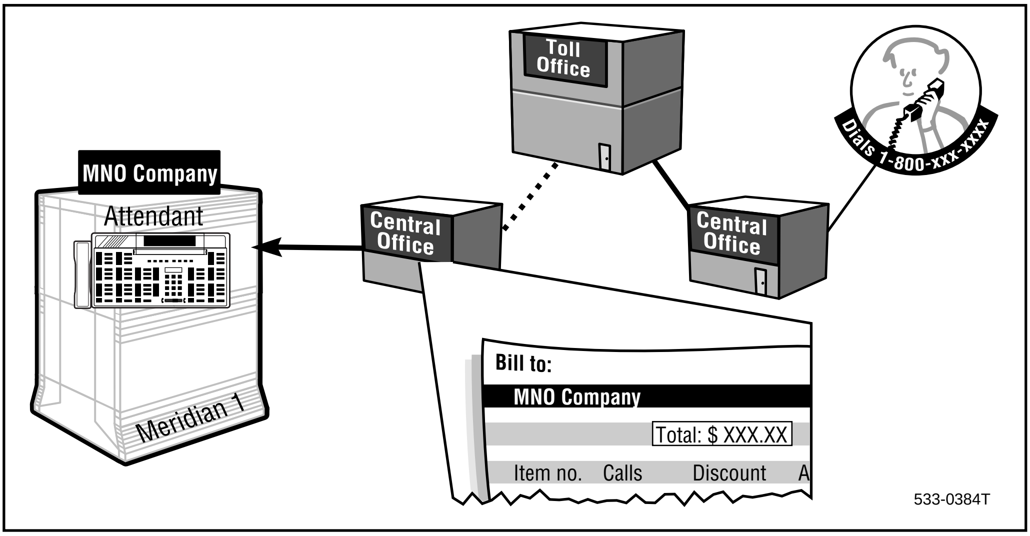
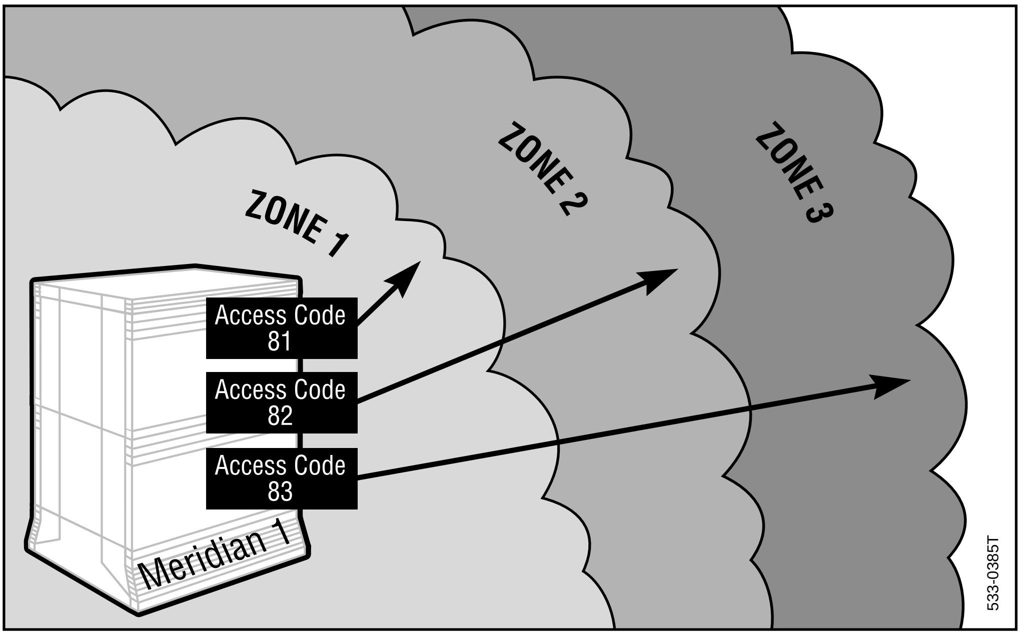


Figure 101 - Use of Out-WATS



Introduction to trunks

Digital trunks

Digital transmission of voice and data between switches can be done in several different ways. Satellite transmission, microwave signaling and transmission using fiber optics are some of them. The oldest method, still in use today, uses two pairs of copper wires, one pair to transmit and one pair to receive.

In North America, the transmission system is called T1. The signal being carried is called DS-1. The data rate of this signal is 1.544 Mbps. The DS-1 signal is comprised of 24 coded channels called DS0s, each with a rate of 64 kbps.

In many countries outside of North America, the transmission system is called E1. The data rate of the signal being carried is 2.048 Mbps. The signal is comprised of 32 coded channels, each with a rate of 64 kbps.

The types of trunks described in the previous pages of this section may be configured on digital channels.

In-band signaling is used for non-ISDN calls, out-of-band signaling is used for ISDN calls. Refer to the information in the section in this guide called *Introduction to ISDN* for further information.

The interface cards for digital trunks connected to a Meridian 1 are called Digital Trunk Interface (DTI) cards. If ISDN and Primary Rate Interface are required, Primary Rate Interface (PRI) cards are installed instead. PRI cards can be used for non-ISDN digital trunks also. There are cards that can handle two DTI and/or PRI facilities. These are known as Dual DTI/PRI (DDP) cards where 1.5 44 Mbps T-i1carriers are used and DDP2 cards where 2.0 Mbps E1 carriers are used.

Introduction to trunks

ESN Expansion feature

This table summarizes the changes made to the ESN functionality with the introduction of this feature in X11 Release 22. Please note that only the components that are affected are listed.

Component	Pre-Release 22	Release 22 and later
Maximum number of digits for a Special Numbering Translation screening.	11	19
Maximum number of Digit Manipulation Index (DMI) deletion digits.	15	19
Maximum number of Digit Manipulation Index (DMI) insertion digits.	24	31
Maximum number of Supplemental Digit Restriction and Recognition (SDRR) tables, with BARS.	256	1500
Maximum number of Supplemental Digit Restriction and Recognition (SDRR) tables, with NARS.	512	1500
Maximum number of words per Supplemental Digit Restriction and Recognition (SDRR) entry.	3	4
Maximum number of digits in each Supplemental Digit Restriction and Recognition (SDRR) entry.	7	10
Maximum length of Flexible Numbering Plan (FNP) Flexible Digit Number Length (FLEN) numbers for Special Number Translation (SPN).	16	24
Maximum length of Flexible Numbering Plan (FNP) Flexible Digit Number Length (FLEN) numbers for Trunk Steering Codes (TSCs).	16	24
Maximum number of digits per Free Special Number Screening (FSNS) Special Number.	11	19
Total number of digits for screening under Free Special Number Screening (FSNS).	14	22
— continued —		

ESN Expansion feature

Component	Pre-Release 22	Release 22 and later
Maximum number of possible Supplemental Digit Restriction and Recognition (SDRR) entry types.	8	9 (‘ALOW’ is added). ALOW is a new entry type in LD 90. This entry allows a call to go through, as if the dialed digits did not match any entry within the Supplemental Digit Restriction and Recognition table.
Restriction imposed on Supplemental Digit Restriction and Recognition (SDRR) entry codes.	Leftwise Unique	None. The leftwise-unique restriction, imposed on SDRR entry codes, is removed. For example, if 555 is an existing entry, a new entry of either 55 or 5551212 can be entered.

ESN software and benefits

Table 84
Possible ESN Node and Main configurations with benefits provided

No.	Meridian 1 Node	Main	Benefits
1	NARS	none	- alternate routing - easy dialing plan - control of users
2	NARS	Norstar	- Norstar can make calls on node’s trunks - fewer trunks required - lower long distance bills - alternate routing - easy dialing plan - control of users
3	NARS	non- Meridian 1	- Main can make calls on node’s trunks - fewer trunks required - lower long distance bills - alternate routing - easy dialing plan - control of users
4	NARS	Meridian 1	- Main can make calls on node’s trunks - fewer trunks required - lower long distance bills - alternate routing - easy dialing plan - control of users
5	NARS + NSIG	Meridian 1 + NSIG	- all of the above plus: - traveling NCOS

— continued —

ESN software and benefits

Table 84
Possible ESN Node and Main configurations with benefits provided

No.	Meridian 1 Node	Main	Benefits
6	NARS	Meridian 1 + CDP	- same as Meridian 1 Main without CDP with the added benefit of alternate routing from the Main - CDP dialing from the Main to the node
7	NARS + CDP	Meridian 1 + CDP	- same as above plus: - CDP dialing plan between node and main
8	NARS + CDP +NSIG	Meridian 1 + CDP + NSIG	- same as above plus: - traveling NCOS
9	NARS + CBQCM	Meridian 1 or non- Meridian 1	- same as 3 and 4 plus: - users can queue when the node's trunks are busy
10	NARS + NSIG + CCBQ	Meridian 1 + NSIG +CCBQ	- same as 3 and 4 plus: - users can queue using Ring Again when the node's trunks are busy
11	NARS + OHQ	Meridian 1 or non- Meridian 1	- same as 9
12	NARS + NSIG + OHQ	Meridian 1 + NSIG	- same as 5 and 11 plus: - can restrict which TCOSs can OHQ at node
13	NARS + CDP + OHQ	Meridian 1 + CDP	- same as 7 and 11 plus: - can OHQ for CDP calls to other switches if Iset routes busy at node
14	NARS	Meridian 1 + BARS	- same as 4 plus: - node has same benefits as Main
15	NARS + NSIG	Meridian 1 + BARS + NSIG	- same as 14 plus: - TCOS or NCOS of caller controls routing of calls at main
16	NARS + NSIG + CCBQ	Meridian 1 + BARS + NSIG + CCBQ	- same as 15 plus: - queuing for each other's trunks - NCOS of caller controls queuing privileges

— continued —

ESN software and benefits

Table 84
Possible ESN Node and Main configurations with benefits provided

No.	Meridian 1 Node	Main	Benefits
17	NARS + OHQ	Meridian 1 + NARS + OHQ	- same as 14 plus: - one call can OHQ at each node
18	NARS + CDP	Meridian 1 + BARS +CDP	- same as 14 plus: - CDP and UDP dialing plan
19	NARS + CDP	Meridian 1 + NARS + CDP	- same as 18
20	NARS + CDP + NSIG	Meridian 1 + BARS +CDP + NSIG	- same as 19 plus: - TCOS or NCOS could control what routes taken to other switches for CDP calls
21	NARS + CDP + NSIG + CCBQ	Meridian 1 + BARS +CDP + NSIG +CCBQ	- same as 20 plus: - users can queue for CDP calls to other switches when routes are busy

ESN software and benefits

Answers to common questions

What other types of switches are compatible with a Meridian 1 ESN network?

Once the NARS Uniform Dialing Plan has been implemented, users at other switches can use the same dialing plan, if the switch they are using is connected to the NARS-equipped switch with TIE trunks.

The switch they are using can be any type of switch. It does not need any networking software and the node does not need anything more than NARS to provide this capability.

Nodes with Off-Hook Queuing software can allow remote switch users to queue off-hook when the trunks at the node are busy.

Nodes with Callback Queuing from a Conventional Main (CBQCM) software can allow remote switch users to callback queue when the trunks at the node are busy.

Nodes with Network Authorization Code software can allow remote switch users to use authcodes to place calls using the trunks at the node.

Nodes with Network Speed Call software can allow remote switch users to access System Speed Call lists at the node to place calls.

Features that require Network Signaling can even be made to work with non-Nortel Networks switches. The travelling class of service (Travelling Class Mark in Lucent terms) is compatible in design with Lucent and IBM systems. This feature allows a user’s NCOS (or FRL) level to travel with calls. It allows other switches in the network to give the user the same privileges they would have at their own switch.

Answers to common questions

When there are two sites in a network, you can either install CDP and ISDN at both sites to tie them together or install Remote Peripheral Equipment (RPE) and serve both sites with one system. Compare the two solutions.

Dialing plan

CDP/ISDN	RPE
3-10 digit dialing	3-7 digit dialing
no duplication of DNs at both sites	duplication of DNs is permitted

Trunking

CDP/ISDN	RPE
one site may have public network trunks used by both sites or each site may have its own trunks	all public network trunks installed at main location
could build in network redundancy for emergencies	if trunks at the main location fail, both sites are affected

Call routing

CDP/ISDN	RPE
calls between locations may use TIE trunks and or public network trunks	calls between locations use loop channels extended off-site. Loop blockage can be an issue. Digital trunking between sites can be expensive.

Answers to common questions

Features

CDP/ISDN	RPE
almost complete feature transparency can be provided if duplicate feature access codes are assigned at both sites.	feature transparency is provided because all users are connected to the same switch

Management

CDP/ISDN	RPE
two switches to be programmed, installed and maintained	on switch to program and maintain. Install RPE equipment at remote site.
two sets of traffic reports	traffic reports at main site only
two sets of CDR reports (unless all network calls go through one switch and CLID in CDR is used)	CDR reports at main site only

Answers to common questions

Can users on an ESN network use the Uniform Dialing Plan to call users at a site that is not connected to the private network?

The user has two options:

1. Dial AC1 followed by a location code and the DN of the user. If there is a location code for that site in the NARS programming, then the call will be directed to a route list. A digit manipulation index can be used to change what the user dialed into the correct public network number to reach the user.
2. Dial AC1 (or AC2) plus the public network number for the user.

The users at the remote site can access the ESN network and the UDP, if they have access to a DISA port at the NARS node. Once into the NARS system using that DISA connection, they can dial calls as if they are a user at the node.

Answers to common questions

What alternatives are there for network administrators who want to bill all network users for calls they make?

There are three solutions to this problem:

Network Authorization Codes (NAUT)

Implement NAUT software at nodes. When a billable call is made by a user at the node or from a remote site, an authcode will be requested. The user dials the authcode at the end of the call.

If the authcode has a high enough FRL level. the call will be placed. The authcode and the call are printed out in CDR records and can be used for billing purposes.

CDR at all sites

You can track all calls made at a site or you can specify the kinds of calls you want to track on CDR. Refer to the *CDR* section of this book for more information.

If you want to track calls that are made at remote sites as they go to the node and then out of the node, you will have to match call records from various sites. Time stamps for the calls will have to be matched. You will need to synchronize the systems' clocks very carefully if you intend to use CDR in this way.

CDR software is provided on all systems. If you intend to print out or download CDR information, you will need printers or polling devices to retrieve the CDR data on a regular basis.

Calling Line ID in CDR (CCDR)

When ISDN is installed, the caller's DN or Calling Line ID travels as a signal on the D-channel when a call is made. This CLID can print out in CDR records if you install the CCDR software at the node. That way, you have a record at the node of the caller's DN for every call printing out.

You do not need to install CDR devices at all sites; you do not need to match CDR records and users do not need to dial authcodes. This solution is the most elegant one of the three and simplest to manage.

Answers to common questions

If I am setting up a Coordinated Dialing Plan and find that two locations have the same DN ranges, what can I do?

If two switches have four digit DNs assigned in the 4000-4999 range this presents a problem for CDP implementation. You have three options:

Change the DNs at one site

Consider the time and expense involved in changing the DNs at one site. If there are DID trunks you may not be able to accomplish this at either site. If you have published these numbers consider the impact of changing them. Consider any retraining that internal users or external callers will need.

Use a five digit CDP-DN

Add an extra digit to the front of the DN to be used for CDP dialing purposes. The telephones do not get re-programmed. You program a Distant Steering Code at each site that identifies the unique leading digit for each other site. The leading digit can be outpulsed to other CDP-equipped sites where it will be handled as a Local Steering Code or you can delete the extra digit at the sending switch.

You can label the DN keys of telephones with the CDP-DN instead of the internal DN. This encourages users to dial a consistent number of digits for all DN-type calls.

Use a fictitious CDP-DN

You can assign a Distant Steering Code that will direct CDP-DN calls to another site. This DSC can be deleted and replaced with another digit as the call is being outpulsed to the other site. In this way users could dial 5XXX to reach another site where the DNs are in the 4XXX range. That way, two sites can continue to have DNs programmed in the same range but they are made to differ for dialing purposes.

Answers to common questions

How can I control calls that come into a node on TIE trunks?

You assign a Class of Service, TGAR and NCOS to a TIE trunk in the same way you assign them to a telephone. These restrictions control what calls will be allowed.

You can program Incoming Trunk Group Restriction at the node to block numbers that you do not want dialed when they originate from TIE trunks.

You can force TIE trunk users to enter authcodes when they make calls using the trunks at the node.

You can assign an NCOS to the TIE trunks with a very low FRL and access to the Network Speed Call feature, if you have NSC software at the node. The remote users can dial AC1 or AC2 followed by a System Speed Call list code and the entry number for the number they wish to dial. Numbers on the list are approved ones only. The FRL of the caller is raised temporarily by the NSC feature and the call is made. Other long distance calls would not be permitted because of the low FRL of the TIE trunks.

Answers to common questions

How can I allow certain lower level users to call a few long distance numbers without removing restrictions completely?

You can program these numbers on a System Speed Call list and then give them access to the list where the numbers are stored. Program their telephones with a Toll Denied access restriction capability. When they use System Speed Call, the feature makes the telephone unrestricted until the call ends.

If you have Network Speed Call software, you can allow users to access System Speed Call by dialing AC1 or AC2 and a code for a particular System Speed Call list. If the NCOS group the telephone belongs to has access to that list, the user will be permitted to do this. When the user accesses the list, the NCOS assigned to the list takes over, if it is a higher FRL than the telephone being used.

Either of these solutions allows you to control the calls users are making, without preventing them from making necessary business calls. You have also made dialing these calls very easy; users will appreciate the simplicity.

Answers to common questions

I have different levels of users at a remote site. How can I give them different levels of access when they place calls through the node?

The TIE trunks from a remote site into a node have an assigned NCOS. You should program all trunks in one route with the same NCOS. If you do this, all users will be treated as the same level when they place calls. You have two options:

1. install additional TIE trunk routes between the remote site and the node. Each route will be designed for certain users at the remote switch to use. Block other users with TGAR and TARG. The trunks in each route will have an NCOS assigned so the node will treats users differently when they use the different incoming routes.
2. Install NSIG software at the node and the remote site. That way, you can program NCOS values at the remote site and assign an NCOS to each telephone. The caller’s NCOS (or FRL) can be programmed to travel on the TIE trunk with each call. The node sees this value and treats the user accordingly. If the remote system is not a Meridian 1, coordinate with the other switch maintainer to see if the switch has the ability to send a caller’s NCOS or FRL with calls.

Answers to common questions

What is MCDN?

ISDN functionality began to evolve on the Meridian 1 when X11 Release 12 software was being developed. The ISDN standards that were laid down at that time were in a very formative state. Nortel Networks introduced Primary Rate Interface (PRI) and Basic Rate Interface (BRI) connectivities using a proprietary interpretation of these early standards (CCITT Q.931). The Meridian 1 interfaces evolved, along with interfaces on other Nortel Networks products under the *Meridian Customer Defined Network (MCDN)* initiative. Cross-product development was coordinated and compatibility testing was done with various kinds of switches including the DMS and SL-100.

Interworking between different private networks or interfacing to non-Nortel Networks PBXs, would have required more development of the proprietary MCDN protocol in order to be compatible with these other networks. In the case of supplementary services, there might not have been any interworking, using the MSDN platform. Therefore, networking software compliant with different, current ISDN standards was introduced.

Since those early days, several ISDN standards have emerged. Standards known as EuroISDN, NI-2, and QSIG all rolled out after Release 12.

The European Computer Manufacturer's Association (ECMA) defined an ISDN protocol called QSIG that has been adopted by the European Telecommunications Standards Institute (ETSI) and the International Standards Institute (ISO). In order to offer ISDN connectivity and services on switches in QSIG networks, Nortel Networks developed ISDN BRI and PRI connectivity based on the QSIG standard and continues to evolve these connectivities.

However, Nortel Networks also continues to develop software compliant with its proprietary MCDN standards.

In the NTP manuals when the term PRI and BRI are used with no other qualifiers or descriptions such as EuroISDN or QSIG, assume the PRI or BRI being discussed is based on the MCDN standards.

Answers to common questions

There are also proprietary forms of networking supported by Nortel Networks that offer similar functionality to ISDN. They are:

- Digital Access Signaling System No. 2 (DASS2)
- Digital Private Network Signaling System No. 1 (DPNSS1)
- ISDN Signaling Link (ISL)
- Analog Private Network Signaling System (APNSS)
- Virtual Network Service (VNS)
- ISDN Semi Permanent Connections for Australia (ISPC)

There is a series of sections in this book called *Services and functions* that deal with services and functions that can be used in many different types of ISDN networks. Features specific to a certain form of ISDN are included in the section for that particular topic.

Answers to common questions

Testing a network

Once BARS, NARS or CDP software has been implemented, test calls should be placed to each route list on the system to ensure that call processing is occurring as you planned. Use your route list index summaries or worksheets as a guide.

Not only should you test that the call is routed with no problems but that alternate routing behaves as expected. Each feature, such as ERWT and CBQ, should also be tested.

Tracing calls

There are two ways to trace calls:

- LD 80 on TTY terminal
- Network Call Trace feature from telephone

LD 80

When using LD 80 and the TTY, at least two people are usually required to conduct the testing. Someone dials calls from a test telephone with a known DN and TN. The other person has the LD 80 TRAC routine loaded and ready at the TTY and everything typed in for the TRAC command except the final <cr>. Just as the last digit in the call is being dialed, the person at the TTY enters the final <cr> in the command. The printout that results shows how the call is being handled and prints out the relevant ESN information that controlled call routing.

The TRAC feature is very sensitive. When activated, (<cr> is entered on the TTY) it takes a “snapshot” of what was occurring on the DN or TN at that instant. If you trace a call too early, you will not get as much information as you would have if you had waited a little longer.

Testing a network

LD 80 commands

Trace calls on a DN

Table 85

Prompt	Response
>	LD 80 <cr>
.	TRAC X Y..Y DEV <cr>

X = customer number (you need a customer number when tracing calls on a DN)

Y..Y = the Directory Number (DN) on the test telephone being used to place calls

Trace calls on a TN

Table 86

Prompt	Response
>	LD 80 <cr>
.	TRAC L S C U ZZ DEV <cr>

L S C U = Terminal Number of the test telephone (Loop, Shelf, Card, Unit number)

ZZ = the key number on the test telephone being used to place calls

> ***Note:***
> If you do not type the key number of the key to be used for the call, irrelevant information about the status of all keys on the test telephone prints out.

Testing a network

Network Call Trace

As of X11 Release 17, you can use a telephone to make and trace network calls on PRI or ISL networks.

If you have programmed the maintenance terminal (TTY) with MTC capability and each D-channel with the remote capability (RCAP) set to NCT (Network Call Trace), information about a call will continue to print out as the call progresses through an ISDN network. Each switch uses the D-channel to send messages back to the originating switch, with information on how it is routing or blocking the call.

If queuing is offered to the user, the call trace function is re- activated when a trunk is offered to the user.

The user of the test telephone initiates the call by dialing the special prefix (SPRE) code followed by 9912. Then, call trace function code 01 or 02 is dialed.

> ***Note:*** Function 01 is most common. Function 02 is used by software designers at Nortel Networks or system suppliers.

The rest of the digits in the call are then dialed.

The telephone must have a class of service that allows Call Trace (CLTA). Both Meridian 1 proprietary sets and analog (500/2500 type) sets can be used to initiate Network Call Trace.

Samples of typical call trace printouts are included in the *Networking* NTPs in the Maintenance section.

Testing a network

Suggested steps for testing

Place calls from different telephones with different NCOSs, changing the time of day of the system, and calling different numbers to fully test the database. If an error is discovered, find out why the call was routed improperly and correct the problem.

Test telephones

- Assign various NCOS values to the test telephones.

  - LD 10 for analog (500/2500 type) sets
  - LD 11 for Meridian 1 proprietary sets

The telephone sets on an actual network would usually be assigned a CTD (conditionally toll denied) class of service to prevent them from making direct access toll calls, bypassing BARS or NARS.

For the purposes of the testing, do not assign a CTD class of service or a TGAR code to the sets. The telephones need access to the trunk groups directly by dialing access codes, in order to make these routes busy, later in the exercise.

Translation verification

- Place calls to all of the route lists which are programmed.

Trace the calls to verify they were handled by the correct route list indexes.

Post the network diagram in front of you to assist you in understanding where calls terminate. Initially, calls will be routed on first choice routes for each route list. As you begin to test alternate routing on different route choices, the network diagram will help you understand where calls terminate.

Schedule people to be available at the different sites in your network during the testing so you will be able to dial their DNs and the calls will be answered. This proves the call was routed to the correct destination. You will not always be able to arrange for every call to be answered, especially public network calls.

Testing a network

Alternate routing verification

- Trunk Group Busy (TGB) keys on the far left side of the console indicate when all the trunks in a route are in use (flashing TGB Indicator).

  The TGB keys can also be used to make a trunk route appear to be busy (to users with TGAR 0 to 7) by pushing the specific key associated with the route (steady lamp when activated). Key 0 corresponds to trunk route 0, key 1 to route 1 etc. (You may want to label these keys).
- Test each alternate route choice in each route list in turn and verify that calls are handled by all appropriate entries in each route list. Use the TGB keys to busy out different trunk routes.
- Trace the calls as you place them from test telephones. You may need help from one or more people during the testing to coordinate the use of the TGB keys.

Remember:

1. The NCOS (FRL) of your test telephones will affect call routing.

2. Calls to certain exchanges or codes may be allowed or denied on some routes.

3. Calls may be handled differently at certain times of day.

4. The CBQ type (I or A) of the test set will affect the number of entries that can be scanned.

If a call does not get routed properly, investigate and correct the problem. Occasionally, a trunk route will be busy with an incoming call from another switch. This will cause unexpected results for your call testing. Check your attendant console TGB keys before making a test call. If the indicator is flashing, the route is currently busy with calls.

Testing a network

Callback Queuing verification

You will have to make trunks busy to test queuing. Two methods can be used. *The console TGB keys cannot be used for this.*

- Method 1: dial the access code to the trunk route(s) you are attempting to make busy. Use the console or telephone sets for this. Put the calls on hold to occupy the trunks. Place enough calls so that the trunk route becomes busy.
- Method 2: Disable the trunks, using a maintenance overlay program (LD 32). Do not forget to re-enable them.

You can verify that the entire route is busy by looking at the TGB keys on the console. If the indicator is flashing, all the trunks in the route are actually busy with active calls or the trunks are disabled.

How to queue

Analog (500/2500 type) set, after hearing overflow tone, the user performs switchhook flash (link) + dials the SPRE code + “1” and hangs up.

When a trunk becomes idle, the set is called back with short burst ringing. The user must answer within 6 seconds from the beginning of the short burst ringing. The call is dialed automatically when the handset is lifted.

Meridian 1 proprietary set, after hearing overflow tone, the user presses the Ring Again feature key and hangs up.

When the set is called back, the set buzzes and the Ring Again key indicator flashes.

To answer, the user must activate a DN and press the Ring Again key. (See telephone set user guides for additional information, if required.)

Steps in testing Callback Queuing

- Make the inexpensive trunk routes busy on each route list. Place calls from a telephone with a low FRL level and CBQ type I. Verify that overflow tone is heard and that Ring Again can be activated.
- Verify that the system calls back when you release one of the held trunks. Trace the call.
- Busy out the inexpensive trunk routes on each route list.
- Make a call from a telephone with a high FRL level, CBQ type A, and Expensive Route Warning Tone (ERWT) assigned in its NCOS.
- Verify that ERWT is heard.
  - Hang on and allow the call to go out on the expensive route. Trace the call.
  - Activate Ring Again after hearing ERWT. Trace the call after Ring Again is activated.
  - Make an inexpensive trunk idle.
  - Verify that the set gets a callback.
  - Trace the call once the set answers the callback. Verify that the call is routed on an inexpensive trunk.
- Keep the Iset trunks busy for longer than the Extended Route Advance Timer (RADT) timer programmed in the NCOS of the test set.
- Trace the call after the set is called back to verify it is going out on an expensive trunk.

Testing a network

Call blocking verification

- Place a call from a set with an FRL that is not high enough for any route in the route list chosen.
  - Verify that overflow tone is heard.
  - Verify that Ring Again does not work in this instance.
  - Trace the call.

Time of day routing verification

- Change the system time of day, if you are using time-of-day routing on any route lists. Use LD 2, the STAD command.
- Verify that calls go out on trunk choices at a different times of day as you programmed.
- Return the system to the correct time before proceeding.

Directory assistance verification

Required in North America only.

- If you have special entries for calls to Directory Assistance (NXX 555) in each area code, place a call to NXX 555 within each test area code.
- Trace the calls.
- Verify that these calls are routed properly.

Free Calling Area Screening verification

Required in North America only.

- Dial an NXX which has been allowed in an FCAS table on a route in a particular route list.
- Verify that the call goes out on the correct entry.
- Dial an NXX which has not been allowed or has specifically been denied on a route choice.
- Verify that the call does not go out on the route where the FCAS table is denying it (or not allowing it).

Free Special Number Screening verification

Required outside North America only.

- Dial a code which has been allowed in an FSNS table on a route in a particular route list.
- Verify that the call goes out on the correct entry.
- Dial a code which has not been allowed or has specifically been denied on a route choice.
- Verify that the call does not go out on the route where the FSNS table is denying it (or not allowing it).

Supplemental Digit Restriction verification

- Dial a number which is denied from the system by Supplemental Digit Restriction (SDRR) programming.
- Verify that the call is blocked.

Supplemental Digit Recognition verification

- Dial a number which is recognized due to *remote* recognition programming.
- Verify that the digits outpulsed on a TIE trunk are different for a recognized number as compared to a non-recognized one.
- Dial a number which is to be recognized due to *local* recognition programming.
- Verify that when numbers are dialed for telephones at your own switch in the off-net format, the call is translated as an internal call at your switch and does not get handled by a route list.

Emergency calls verification

Required in North America only.

- Verify that calls to emergency number 911 are processed for all telephones of all FRL levels. Dial the call as ACI + 911, (also AC2 + 911 with NARS), and AC1 + 11 (if AC1 = 9 and 11 has been programmed as a special number or SPN).

Testing a network

Trouble reports

What is a trouble reporting system?

A trouble reporting system is simply a process that lets end users notify you when something is wrong.

A trouble reporting system may consist of:

- an ESN support line that lets users reach service personnel directly to report a possible Meridian 1 problem
- electronic reporting through forms or electronic mail
- tracking of problem reports. It is a good idea to keep a permanent record of problem reports so you can develop historical data in order to spot trends and track chronic problems.

Your trouble reporting system will depend entirely on your needs and resources.

You may want to integrate the trouble reporting system for the Meridian 1 network with trouble reporting systems used elsewhere in your organization.

How do you prepare a trouble reporting system?

Access to the system

Make the trouble reporting system easy to access and easy to use. While a support hotline or trouble line is a good idea, remember that users may not be able to use their telephone at all. Provide alternate ways for submitting problem reports.

Trouble reports

Information in a trouble ticket

When an end user reports a problem and a service agent creates a trouble ticket (a record or form that identifies the reported problem), the agent should ask the following types of questions:

- Exactly what did you dial when you experienced the problem? (Make sure the user enters the exact digits, since they may have used Autodial or Speed Call to make the call.)
- What did you hear? (Ask them to describe the tone(s) or sounds exactly.)
- At what point in the call did you experience a problem? (If they were blocked before they finished dialing this usually indicates a translation problem. If they were blocked after the whole call was dialed, this could mean all trunks were busy or the user's FRL is too low.)
- What is the DN of the telephone you were using? (They may not have used their own telephone. If the DN is a Multiple Appearance DN, get them to specify which telephone they used.)
- Can you normally call this number? (This will tell you if a recent change has affected them.)
- At what time and on what day did you first have this problem? (Features such as Routing Control can cause unusual call routing effects on weekends and after hours.)
- Do you have this problem only occasionally? (Occasional problems can be caused by such things as trunk blockage and can be addressed with training on queuing and traffic studies.)
- Are you having any other network-related problems? If so, since when? (This might be an opportunity for you to get other information about unrelated problems or issues.)

Some investigations you might want to do to follow up are:

- Printout the NCOS, Class of Service and TGAR of the user's telephone.
- Does the same problem occur at other telephones that are the same as this user's telephone?
- Does the same problem occur at other telephones that are not the same as this user's telephone?
- Have there been recent programming changes made at this site that would cause the problem?

To facilitate information gathering, create forms or templates to help service personnel or customer assistance personnel ask the necessary questions.

Troubleshooting

It often helps troubleshooters to identify problems quickly if they approach trouble situations in the same way that the system deals with each call.

The system translates each call first so troubleshooters should always check translation tables first. *Never assume that translation is still working the way it was originally programmed to work or that it is working the way it is described on worksheets.*

The sequence that the system with NARS uses to process a call is as follows:

1. Translation
2. Restriction/Recognition
3. Route list index
4. User's Network Class of Service
5. Each route entry – FRL, TOD, FCAS, LTER, DMI, Queuing

The only possible results of call processing are: the call is processed on a trunk, the call is blocked due to restrictions, the call is intercepted, or the call is not processed because of busy trunks.

Trouble reports

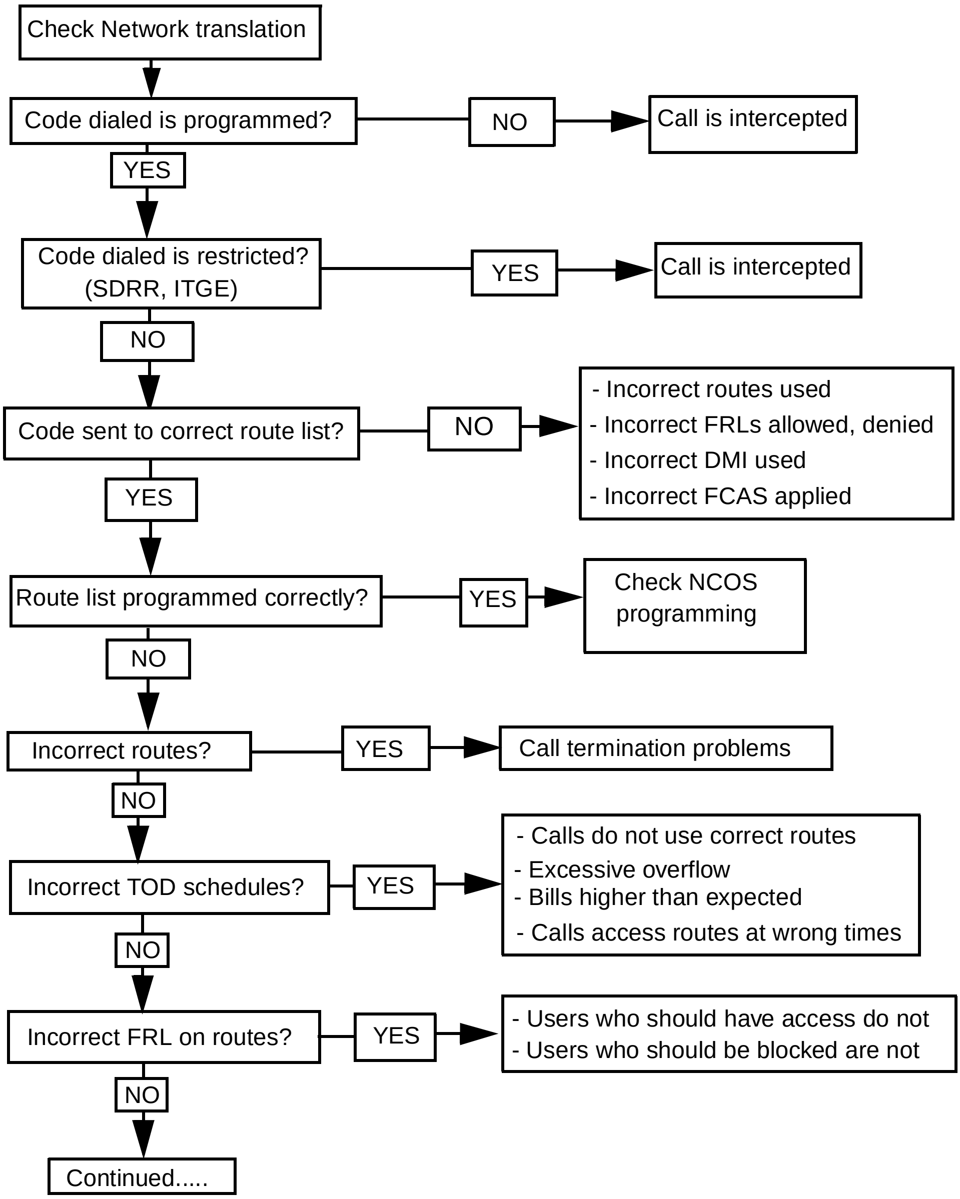
Resolution

Once service personnel believe a particular condition has been cleared or a problem has been solved, it is important that they perform the following steps:

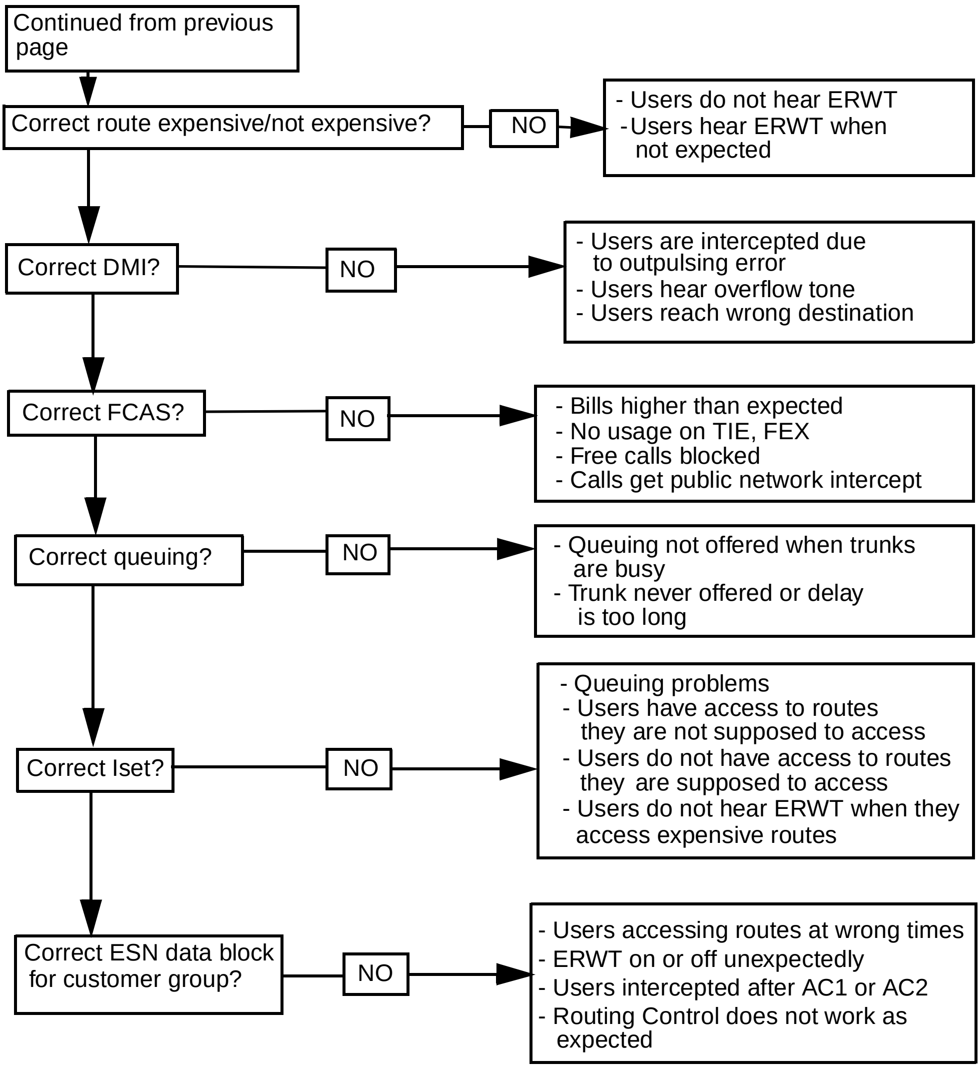
- verify that the problem has been cleared
- inform the end user who reported the problem
- close the trouble ticket in the reporting system so that other personnel do not work on it needlessly
- follow up on any actions taken with other vendors, telephone company personnel, or Nortel Networks

When planning a trouble reporting system, prepare training and job aids for personnel performing the tasks related to troubleshooting.

NARS troubleshooting flow chart



NARS troubleshooting flow chart



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