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Nortel Networks Symposium Call Center Server

Symposium, M1, and Voice Processing Guide

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Publication history

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This document is the first standard release of the *Nortel Networks Symposium Call Center Server Symposium, M1, and Voice Processing Guide* for Release 3.0.

Contents

Chapter 1

Getting started

In this chapter

Overview

Introduction

The *Nortel Networks Symposium Call Center Server Symposium, M1, and Voice Processing Guide* provides descriptive information and instructions on how to set up and configure the Meridian 1, Meridian Link Services, Meridian Mail, Meridian Integrated RAN (MIRAN), and voice processing for use with Symposium Call Center Server.

Assumptions

This guide is based on the following assumptions:

- Symposium Call Center Server has been correctly installed and is operational.

  If Symposium Call Center Server has not been installed, then you should install it. For instructions, refer to the *Software Installation and Upgrade Guide.*
- The Meridian 1 has been correctly installed and is operational with all current Product Enhancement Packages (PEPs) applied. For information on which PEPs to install on the Meridian 1, contact your Nortel Networks customer support representative.
- The Meridian 1 is running switch release X11R24.25 or greater.

Skills you need

Introduction

This guide is intended for individuals responsible for configuring, administering, and maintaining the Meridian 1, Meridian Mail, or a third-party voice processing system.

This section describes the skills and knowledge you need to use this guide effectively. This guide is not directed at the novice and does not detail the basics of Meridian 1, Meridian Mail, or Symposium Call Center Server operation, features, or administration.

The examples in this document are based on (with most or all packages equipped)

- Meridian 1 Release 22.37
- Meridian Mail Release 11
- Symposium Call Center Server 3.0

Prompts, menus, and windows can look different for different releases of the subsystems, or if the subsystem is equipped with different versions of the packages.

Nortel Networks product knowledge

Knowledge of, or experience with, the following Nortel Networks products can be of assistance when administering Symposium Call Center Server:

- Meridian 1 Options 11, 51C, 61C, 81, or 81C
- Option 11 Release 22 and greater
- Meridian Mail Release 8 and greater

PC experience or knowledge

Knowledge of, or experience with, the following PC products can be assistance when administering Symposium Call Center Server:

- Microsoft Windows 95/98 or Windows NT
- client/server architecture
- Internet Protocol (IP)

Other experience or knowledge

Other types of experience or knowledge that might be of use include

- database management
- flowcharting
- programming

What's new in this version

Introduction

This section gives a brief description of the new features and configuration issues in Release 3.0 of the *Symposium, M1, and Voice Processing Guide.*

Network Skill-Based Routing

The Network Skill-Based Routing (NSBR) feature enables queuing of calls from one Symposium Call Center Server site to another. The Meridian 1 configuration procedures have not changed from Symposium Call Center Server Release 1.5 version. To enable the NSBR feature, you must, however, enter new parameters and edit existing parameters when configuring the Meridian 1 switch. These new parameters enable communication (call routing) between Symposium Call Center Server sites.

Subsystems overview

Introduction

The following subsystems work together to provide processing for a call:

- the Meridian 1 switch
- Symposium Call Center Server
- the voice processing system (Meridian Mail or a third-party voice processing system)
- Meridian Link Services

This chapter describes each of the subsystems.

Communication between Symposium Call Center Server subsystems

Meridian 1 switch

The Meridian 1 switch provides a speech path for a call between its source, usually a trunk, and its destination (a RAN trunk, voice port, or agent). Two connections to the switch interact with voice processing systems: voice paths and signaling links.

Voice paths

Voice paths are connections that carry speech (phone calls). They are configured as terminal numbers (TNs) on the switch. The following table shows the types of voice paths used for different voice processing systems:

Type	Voice processing system
virtual ACD Agent SL1 sets	Meridian Mail
2500 set TNs	usually third-party voice processing systems
2500 set ACD agent TNs	usually third-party voice processing systems
T1 TNs	usually third-party voice processing systems
E1 TNs	usually third-party voice processing systems

Signaling links

Signaling links are connections that carry auxiliary information (such as treatment DNs) between the switch and a voice processing system. Signaling links are optional, but they allow greater cooperation and control between the switch and the voice processing system.

Symposium Call Center Server

The server communicates with the Meridian 1 switch and the voice processing system.

Communicating with the switch

The server executes scripts and instructs the Meridian 1 switch to set up the speech paths necessary to connect calls to voice ports, agents, or RAN trunks, and to provide tone treatments (such as ringback and busy) to calls. The server communicates with the switch over the ELAN using a messaging protocol.

Communicating with Meridian Mail

Symposium Call Center Server also communicates with Meridian Mail to instruct it to play prompts, collect digits input from callers, or both. These instructions and the resulting caller input travel over a messaging protocol across the Meridian Mail ACCESS Link.

The prompts and script commands offer different ways of playing a message to a caller when a call is connected to a voice port.

Meridian Mail

The Meridian Mail voice channels connect to the Meridian 1 switch by means of a special network loop card. On the switch side, this card is configured as an SL1 set TN (Virtual Agents).

Meridian Mail can be accessed by Symposium Call Center Server through the following commands:

- GIVE IVR
- GIVE CONTROLLED BROADCAST
- OPEN VOICE SESSION

When the Give IVR script command is used, Symposium Call Center Server sends the command to the switch with (optionally) a treatment DN. The switch passes the treatment DN to Meridian Mail.

When the Give Controlled Broadcast Announcement or Open Voice Sessions script commands are used, the server uses the ACCESS link to communicate the Play Prompt segment list (specified in the Play Prompt script command) to Meridian Mail. For the Open Voice Session script command, the server can also instruct Meridian Mail to collect digits (using the Collect Digits command).

Meridian Link

Meridian Link is an interface used for communication between a host application and the Meridian 1 switch. The interface facilitates the integration of the computer and the switch. In this integrated environment, the host processor interacts with the Meridian 1 switch by exchanging application layer messages.

You can develop Meridian Link applications, which allow you to use information taken from the switch (such as Caller ID), connect to another application to retrieve a matching record from a database, and then provide a screen pop to help agents prepare for the call.

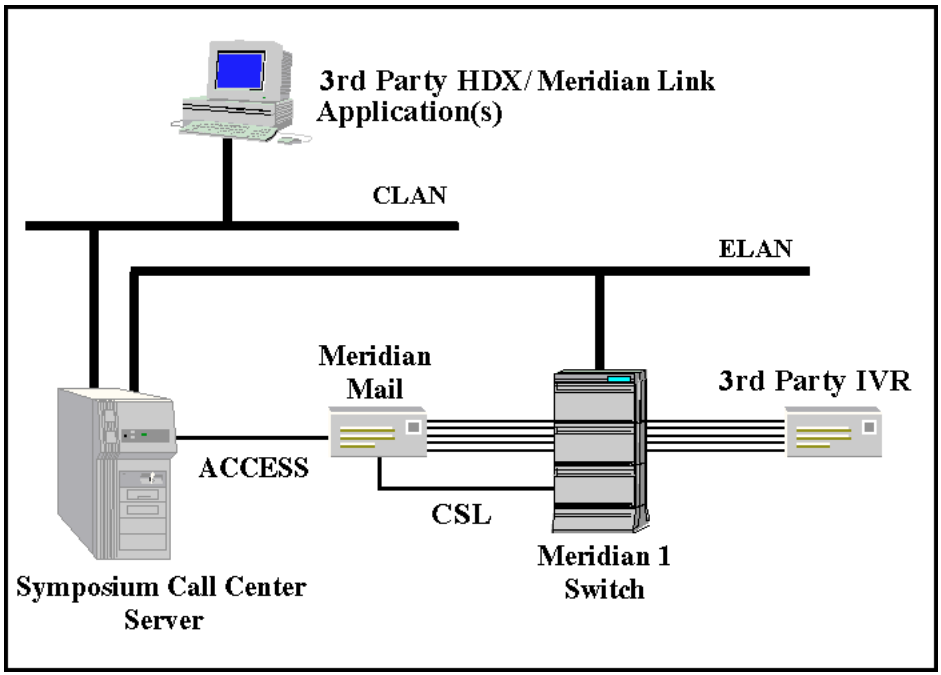
Meridian Link Services

Meridian Link Services (MLS) is a process running on Symposium Call Center Server that gives the server access to the Meridian Link interface. Through MLS, the server can connect to Meridian Link applications over the CLAN.

External applications register with MLS to obtain access to application layer messages. MLS commands that result in call processing requests are sent over the ELAN to the Meridian 1 switch. Examples of external applications that can register with MLS include Host Data Exchange (HDX), Open Database Connectivity (ODBC), and Real-Time Data API.

Connections between the subsystems

The subsystems communicate across local area networks (LANs) and serial links. The following illustration shows the connections:



Local area networks

The subsystems require the following types of LANs for communication:

- **ELAN** (Embedded LAN): The ELAN enables Symposium Call Center Server to communicate with the Meridian 1. The ELAN carries call processing traffic and should be private.
- **CLAN** (Customer LAN): The CLAN enables Symposium Call Center Server to communicate with Supervisor and Administrator workstations and application servers that use Meridian Link, Real Time API, or Host Data Exchange API.

Serial links

The subsystems communicate using the following serial links:

- **ACCESS Link:** This is an RS-232 asynchronous connection between Meridian Mail and ACCESS applications such as Symposium Call Center Server.

- **AML / CSL link:** The connection between the Meridian 1 and Meridian Mail is an RS-232 synchronous link.

Chapter 2

Configuration overview

In this chapter

Overview

Introduction

Upon arrival of a call, control of the call passes from one subsystem to another. Each subsystem provides a specific set of features or services.

To use these features, you must configure the following subsystems correctly:

- the links between Symposium Call Center Server, the Meridian 1 switch, and Meridian Mail (if applicable)
- the Meridian 1 switch
- the voice processing engine (Meridian Mail or a third-party voice processing engine)
- Symposium Call Center Server

Note: If you are using Meridian Link Services (MLS), you must also configure the switch for MLS.

As with any interlinked configuration, many parameters must be configured consistently across these subsystems.

This chapter lists the entities that must be configured in each of the subsystems. It provides a recommended configuration sequence to ensure that each entity is correctly configured. It also provides checklists to use during configuration.

Subsystem configuration reference information

Configuring subsystems

The following table shows the entities that must be configured on the different subsystems. When configuring an entity on more than one subsystem, you must ensure that the parameters are consistent across the subsystems.

Configuration element	Meridian 1	Meridian Mail (voice processing engine)	Symposium Call Center Server
ACD-DNs	x	x	
CDNs	x		x
IVR ACD-DNs	x		x
Mailboxes and passwords		x	x
Phonesets	x	x	x
Routes and queues	x		
Scripts			x
Treatment DNs	x		x
Voice channels		x	
Voice files and segments		x	x
Voice ports (Virtual Agent TNs)	x	x	x
Voice service DNs		x	

To configure these entities on the Meridian 1 switch, see Chapter 4, “Meridian 1 subsystem configuration.” To configure these entities in Meridian Mail, see Chapter 6, “Meridian Mail subsystem configuration.” To configure these entities on Symposium Call Center Server, see Chapter 7, “Symposium Call Center Server subsystem configuration.”

Recommended sequence of configuration

Configuration sequence

Nortel Networks recommends that you complete the configuration in the following sequence:

1. Complete the installation of all subsystems.
2. Ensure that the links between subsystems are configured, connected, and functioning (see Chapter 3, “Subsystem connections configuration”).
3. Configure the Meridian 1 switch (see Chapter 4, “Meridian 1 subsystem configuration”).
4. (Optional) Configure the switch for MLS if you require an interface to other applications (see Chapter 5, “Meridian Link Services configuration”).
5. Configure Meridian Mail or the third-party voice processing application (see Chapter 6, “Meridian Mail subsystem configuration”).
6. Configure Symposium Call Center Server (see Chapter 7, “Symposium Call Center Server subsystem configuration”).
7. Install and configure third-party applications.

Configuration checklists

Checklists (configuration and setup)

The checklists in this section assume an understanding of the configuration elements and how to check their status on the Meridian 1 switch, Meridian Mail, and Symposium Call Center Server. If you do not know how to check the status of a particular element, consult the appropriate section in this guide.

General checklist

Before proceeding to the command-specific setups, complete the following checklist:

	Symposium Call Center Server is up and running correctly.
	The ELAN connection between the server and switch is functioning.
	Voice port TNs on the switch are enabled and logged on.
	Ports on the voice processing engine are enabled.
	If you are using Meridian Mail as the voice processing engine, the CSL connection between the switch and Meridian Mail is functioning.

Give IVR checklist

Using Meridian Mail

Complete the following checklist if you use the Give IVR command with Meridian Mail:

	The general checklist is complete.
	The CDN is acquired on Symposium Call Center Server.
	The correct script is activated on Symposium Call Center Server.

	The IVR ACD-DN is acquired on Symposium Call Center Server.
	Voice ports are acquired with a TN only (the channel number field is blank).
	All of the TNs in the Meridian 1 switch ACD-DN are acquired as voice ports. **Note:** You must use dedicated voice ports.
	The script command refers to the acquired IVR ACD-DN.
	A treatment DN is specified in one of the following ways: • A treatment DN is explicitly specified in the script (for example, GIVE IVR 7000 with treatment 7002). • The Default Treatment DN for the IVR ACD-DN is set correctly on the switch.
	The treatment DN is • configured as a VSDN entry on Meridian Mail and associated with the desired voice application (for example, a voice menu).
	Meridian Mail ports' ACD-DN = Meridian 1 switch TNs ACD-DN = acquired IVR ACD-DN
	Meridian Mail ports' TNs = Meridian 1 switch TNs = acquired TNs.
	Acquired voice ports are Idle on Meridian Mail.

Using a third-party voice processing engine

If you use the Give IVR command with a third-party voice processing engine, ensure the tasks in the following checklist are completed in the order listed.

Note: The Give IVR script statement can only use voice ports defined on the switch as ACD agents. If the voice processing engine connects to the switch as any other type of terminal, only the Route Call command can be used in a script to hand off the call to the voice processing engine.

1	The CDN is acquired on Symposium Call Center Server.

2	The correct script is activated on Symposium Call Center Server.
3	The ACD-DN owning the ports on the Meridian 1 switch = the IVR ACD-DN acquired by the server. • The ACD-DN on the Meridian 1 must have the IVR property set equal to YES.
4	TNs for the ports on the Meridian 1 switch = TNs acquired by the server.
5	The IVR ACD-DN is acquired on Symposium Call Center Server.
6	Voice ports are acquired with a TN only (the channel number field *must* be blank).
7	*All* the TNs in the Meridian 1 switch ACD-DN are acquired by the server as voice ports.
8	The script command refers to the IVR ACD-DN that is acquired.
9	If an APL link exists, a treatment DN is specified in one of the following ways: • A treatment DN is explicitly specified in the script. • The Default Treatment DN for the IVR ACD-DN is set correctly on the switch.
10	Voice ports behind the IVR ACD-DN acquired by the server are indicated by the Meridian 1 switch as in Logged in state.

Controlled Broadcast and Voice Sessions checklist

If you use the Give Controlled Broadcast Announcement or Voice Sessions commands, ensure that the tasks in the following checklist are completed in the order listed.

Note: The Give Controlled Broadcast Announcement and Voice Sessions commands apply only to the Meridian Mail voice processing system.

1	The ACCESS link between Symposium Call Center Server and Meridian Mail is functioning correctly.
2	The CDN is acquired on Symposium Call Center Server.
3	The correct script is activated on Symposium Call Center Server.
4	The IVR ACD-DN is acquired on Symposium Call Center Server.
5	Port Channel Numbers on Server = the Class Numbers for ports on Meridian Mail.
6	Voice ports are acquired with both TNs and channel numbers.
7	*All* of the TNs in the Meridian 1 switch ACD-DN are acquired as voice ports.
8	The script command refers to the acquired IVR ACD-DN.
9	A treatment DN is specified in one of the following ways: • A treatment DN is explicitly specified in the script. • Default Treatment DN is set to the appropriate ACCESS VSDN.
10	The treatment DN is defined in Meridian Mail as an ACCESS VSDN.
11	Meridian Mail ports' ACD-DN = Meridian 1 switch TNs ACD-DN = acquired IVR ACD-DN
12	Meridian Mail ports' TNs = Meridian 1 switch TNs = acquired TNs.
13	Acquired voice ports are Active on Meridian Mail.
14	*Voice segment*, *Number*, or *Numberbydigit* constants and variables refer to the files and voice segments known to Meridian Mail, and the contents are recorded.

15	No non-ACCESS calls arrive at these voice ports.
16	The Maximum Ports Used For Broadcast parameter on Symposium Call Center Server is set if Broadcast and Voice Session calls share the voice ports.
17	The broadcast voice port wait timer has been set appropriately.
18	The Meridian Mail mailbox and password set on Symposium Call Center Server = Meridian Mail mailbox containing the voice files.

Chapter 3

Subsystem connections configuration

In this chapter

Overview

Introduction

To enable the features of Symposium Call Center Server to operate correctly, you must configure the following links between the subsystems:

- the IP address on the Meridian 1 switch. For information about how to configure an IP address for the Meridian 1 switch, refer to the Meridian 1 switch documentation.
- the Embedded LAN (ELAN) between the Meridian 1 and Symposium Call Center Server
- the Meridian ACCESS link between Symposium Call Center Server and Meridian Mail (if you are using Meridian Mail)

Configuring the ELAN

Introduction

To configure and verify the ELAN, you must perform the following tasks:

1. Define the Embedded LAN (ELAN) and a Value Added Server (VAS).
2. Check the ELAN link.
3. Check the AML link.

These tasks associate the link (ELAN) with a VAS ID to allow message transmission.

Note: These procedures assume that Meridian Mail and the Meridian 1 switch are installed and communicating correctly.

Defining the ELAN with LD 17

To define the ELAN and VSID with LD 17, use these prompts and responses.

Note: For prompts that are not specified in the following table, press Enter.

Prompt	Response	Description
REQ	CHG	Change
TYPE	CFN	Configuration Record
ADAN	NEW ELAN 16, CHG ELAN 16, OUT ELAN 16	Add/change/remove I/O device type ELAN 16 (AML over Ethernet).
CTYP	ELAN	Card type
VAS	NEW	Add a value added server.
VSID	16	VAS identifier

Prompt	Response	Description
ELAN	16 to 31	Associate VAS ID with the ELAN.
SECU	Yes	Turn on security for Meridian Link Services applications.
CSQO	255 maximum	Number of call registers linked to input queue
CSQI	255 maximum	Number of call registers linked to output queue
REQ	END	Exit from overlay.

Checking the ELAN with LD 48

Once you configure the VSID and power up Symposium Call Center Server, the ELAN link comes into service.

To check the ELAN link

1. At the M1 administration terminal, load LD 48.
2. Type the command **stat elan.**
3. Ensure that, under your Symposium Call Center Server ELAN IP-address, LYR7 and APPL are active. Note the ELAN ID.

   Example:

   ELAN #: 16 DES: the application
   APPL_IP_ID: 47.152.163.68 LYR7: ACTIVE EMPTY APPL ACTIVE
4. If the ELAN is not active, check the ELAN connection by pinging the Meridian 1 switch IP address from the the application. To do so, open a DOS prompt. Type **ping *nnn.nnn.nnn.nnn*** where *nnn.nnn.nnn.nnn* is the switch IP address, and press Enter.

Configuring the Meridian ACCESS link

Introduction

If you want to include Open Voice Session or Give Controlled Broadcast commands in your scripts, you must

- configure the Meridian ACCESS link between Symposium Call Center Server and Meridian Mail. For more information about configuring the Meridian ACCESS link, see page 28.
- configure IVR ACD-DNs on the Meridian 1. For more information about configuring IVR ACD-DNS on the Meridian 1, see "Configuring IVR ACD-DNs" on page 41.
- configure voice ports on the Meridian 1. For more information about configuring voice ports on the Meridian 1, see "Configuring voice ports" on page 44.
- configure Meridian ACCESS voice ports on Meridian Mail. For more information about configuring Meridian ACCESS voice ports on Meridian Mail, see "Configuring ACCESS voice ports" on page 94.

Note: For a detailed explanation of how voice processing functions in Symposium Call Center Server, refer to Chapter 6, "Meridian Mail subsystem configuration."

To determine whether ACCESS is enabled on Meridian Mail

1. At the Meridian Mail administration terminal, ensure that Meridian Mail is equipped with the Meridian ACCESS feature. You can check this by going to TOOLS level → Display system record.
2. If Meridian Mail does not have the Meridian ACCESS feature, you must do a comprehensive upgrade to include this feature.

To establish a physical connection between Meridian Mail and Symposium Call Center Server

You must establish a physical connection between the Meridian ACCESS voice port on Meridian Mail and the COM2 port on Symposium Call Center Server. The physical cable varies depending on which Meridian Mail platform you are using. The physical connection can be one of the following Meridian Mail platforms:

- On a Modular Option platform, an NT4R20AA fan-out cable is connected to the I/O panel connector labeled RSM. From the defined fan-out port, connect a Null modem cable DB-25 male connector to a DB-9 female connector.
- On the Modular Option EC platform, an NT6P0109 fan-out (five DB-25 connectors) cable is connected to the I/O panel connector labeled 5RS232. From the defined fan-out port, connect a Null modem cable DB-25 male connector to a DB9 female connector.
- On the Option 11c Meridian Mail Platform, you must configure an RSM interface card and a fan-out cable.

To configure the Meridian ACCESS link between Symposium Call Center Server and Meridian Mail

Symposium Call Center Server has the COM2 port automatically configured as the ACCESS Link set at 9600 bps. The physical port used for the ACCESS link on Meridian Mail must be configured for ACCESS, and its baud rate must match what is configured in Symposium Call Center Server.

To configure the Meridian ACCESS link on Meridian Mail, you must perform the following steps:

1 From the Meridian Mail Integrated Communication Link Window, click the Add Link button.

Result: The Add/Modify Link Window appears.

2 In the Link Name box, enter a name for the link. The name of the link can

- be 1-19 alphanumeric characters long
- include single spaces, except at the beginning and end of the name

- not be named “SysOps” or any lowercase or uppercase version of the word

3 From the Users Port drop-down menu, select an appropriate port.

4 Click OK.

Result: If you are creating the first link for a COM port, the Modify Connection Window appears.

5 From the Protocol drop down menu, select MMLink.

6 From the Baud Rate drop down menu, select 9600. The baud rate must match the baud rate configured on Meridian Mail.

7 Click OK.

If you have entered all of the parameters correctly, the ACCESS link appears. You can ensure that the ACCESS link is synchronized through the Meridian Mail Tools ACCESS diagnostics.

Chapter 4

Meridian 1 subsystem configuration

In this chapter

Overview

Introduction

You must configure the following elements on the Meridian 1 (M1) subsystem:

- agent ACD queues
- CDNs
- networked ACD-DNs (NACD-DNs)
- IVR ACD-DNs
- voice ports (virtual agent TNs)
- agent phonesets (TNs)
- routes

This chapter also explains how to initialize the switch and change resources on the switch without causing problems on Symposium Call Center Server.

For a complete list of the X11 overlays, refer to the *X11 Administration Programs Guide* provided with the Meridian 1.

Before you begin

Before you configure elements on the switch, ensure

- you have the packages required for voice processing in Symposium Call Center Server. The packages include 3.11 and 32c.
- you have X11 software Release 24 or greater
- all current Product Enhancement Packages (PEPs) are applied. For information on which PEPs to install on the Meridian 1, contact your Nortel Networks customer support representative.
- all current Product Enhancement Packages (PEPs) are applied if you have purchased the Network Skill-Based Routing option and you want to use network ACD (NACD) routing. For information on which PEPs to install on the Meridian 1, contact your Nortel Networks customer support representative.

Note: Symposium Call Center Server cannot share Meridian 1 resources with other applications such as Meridian Max, Link, and CCR. These include CDNs, ACD-DNs, and TNs.

Changing switch resources

If you need to change any switch resources, you must first deacquire the resources on Symposium Call Center Server.

Configuring CDNs

Introduction

CDNs are specialized ACD-DNs or queues on the switch. A CDN is the entry point of a call into Symposium Call Center Server call processing. You must configure CDNs on the Meridian 1 switch and on Symposium Call Center Server (see Chapter 7, “Symposium Call Center Server subsystem configuration”).

Modes of operation

CDNs have two modes of operation—default and controlled—as shown by the CNTL prompt. Symposium Call Center Server requires the CDN to be configured in default mode. Once Symposium Call Center Server acquires the CDN, the following events occur:

- The CDN is automatically switched to control mode.
- The parameter ASID appears next to the CDN block.
- CNTL and AACQ automatically change to YesPlease confirm the prompts.

If CNTL = No (that is, if the CDN is not acquired), or if no treatment is given in four seconds, calls are directed to the default DN.

Note: Ensure that RPRT = NO.

Assumptions

The following assumptions are made:

- You know the user ID and password to log on to the switch and Meridian Mail administration terminals.
- You are familiar with Meridian 1 Change and Diagnostics overlays.
- You have a listing or printout of available CDNs (overlay program 23).

VSID definitions

Do not enter a VSID definition on CDNs.

Configuring a CDN with LD 23

To configure CDNs in LD 23, use these prompts and responses.

Note: For prompts that are not specified in the following table, press Enter.

Prompt	Response	Description
REQ	NEW, CHG	Add or change a CDN.
TYPE	CDN	Control DN data block
CUST	0–99	Customer number
CDN	xxxx	Control DN number
RPRT	No	Deactivate the report control option.
DFDN		Default ACD-DN where call defaults if there is a problem on Symposium Call Center Server.
CNTL	No	Control DN is in controlled mode.
REQ	END	Exit from overlay.

Verifying that the CDN is acquired

After you acquire the CDN on Symposium Call Center Server, the CDN printout appears as follows:

```
>ld 23
ACD000
MEM AVAIL: (U/P): 3591770    USED: 405925    TOT: 3997695
DISK RECS AVAIL: 2682
ACD DNS  AVAIL: 23758    USED:   242    TOT: 24000
REQ  prt
TYPE cdn
CUST 0
CDN  2003
```

```
TYPE CDN
CUST 0
CDN  2003
FRRT
SRRT
FROA NO
MURT
DFDN 7700
CEIL 2047
OVFL NO
TDNS NO
RPRT NO
AACQ YES
ASID 16
SFNB 1  2  3  4  5  6  9  10  11  12  13  15  16  17  18  19
USFB 1  2  3  4  5  6  7  9  10  11  12  13  14  15
CALB 0  1  2  3  4  5  6  7  8  9  11
CNTL YES
VSID
HSID
CWTH 1
BYTH 0
OVTH 2047
STIO
TSFT 20
```

Changing CDNs on the Meridian 1

Introduction

If you need to make changes to CDNs on the Meridian 1 switch, you must follow specific steps to avoid causing service breaks in Symposium Call Center Server. For example, if you remove a CDN that is currently acquired by Symposium Call Center Server, you can cause some services to stop processing. This prevents calls from being handled by Symposium Call Center Server.

To change the sequence of CDNs on the switch

1. Deacquire the CDN from Symposium Call Center Server.
2. Delete, move, add, or make changes to the CDN as necessary on the Meridian 1.
3. Acquire the CDN on Symposium Call Center Server.

Configuring NACD-DNs

Introduction

The NACD-DN is a local dummy ACD-DN (that is, there are no agent positions assigned to it) with a Night Routing table. NACD-DNs enable you to route calls to other Symposium Call Center Server sites. You can use NACD routing as a backup in the event that a Symposium Call Center Server network routing command fails.

Notes:

1. You must configure NACD-DNs if you have purchased the Network Skill-Based Routing option.
2. To enable NACD routing, the Meridian 1 switch must be connected via a Meridian Customer Defined Network (MCDN). For more information about configuring the Meridian 1 on a MCDN, refer to the documentation that came with the Meridian 1 switch.

When to configure NACD-DNs

Configure NACD-DNs when you want the application to communicate with the Meridian 1 NACD software. This enables you to route overflow calls to other Symposium Call Center Server sites. Use the QUEUE TO NACD script command to access the NACD-DNs.

If you are configuring an NACD-DN, ensure that there are no TNs or positions associated with the ACD-DN that are configured as the NACD routing DN. If there are TNs or positions assigned, reassign them to another ACD-DN.

Assumptions

The following assumptions are made:

- You know the user ID and password to log on to the Meridian 1 administration terminal.
- You are familiar with Meridian 1 Change and Diagnostics overlays.

Configuring a new NACD-DN with LD 23

To configure NACD-DNs in LD 23, use these prompts and responses.

Prompt	Response	Description
REQ	NEW, CHG	Add or change an NACD-DN.
TYPE	ACD	ACD data block
CUST	xx	Customer number
ACDN	xxxx	The existing ACD-DN to be changed
MAXP	1	The number of voice channels allocated to the service
NCFW	x	Enter an "x" to delete night call forward. **Note:** NCFW must be blank to allow the configuration of an NACD Night Routing table.

Configuring an existing ACD-DN as an NACD-DN with LD 23

If you are using an existing ACD-DN, ensure that there are no TNs or positions associated with the ACD-DN that are configured as the NACD routing DN. If there are TNs or positions assigned, reassign them to another ACD-DN. Then use LD 23 to configure the NACD-DN as described in the previous procedure.

Configuring the Night Routing table with LD 23

To configure the Night Routing table, enter the specified values for the following prompts in LD 23.

Prompt	Response	Description
REQ	NEW, CHG	Add or change
TYPE	NACD	Network ACD

Prompt	Response	Description
ACDN	xxx...x	ACD directory number being used
TABL	N	Night Table
TRGT	xxxx tttt	Remote target ACD-DN (xxxx) and the timer (0–1800) in seconds Press Return to add another target. You can add a maximum of 20 targets to the table. Press Return twice to stop adding targets.
REQ	END	Exit from overlay.

Note: Local targets are ignored when using Symposium Call Center Server.

Configuring IVR ACD-DNs

Introduction

An IVR ACD-DN is a DN that is assigned to voice ports that provide voice processing services. You program voice ports as ACD agents belonging to IVR ACD-DNs. Symposium Call Center Server then must acquire the IVR ACD-DNs. You configure IVR ACD-DNs on the Meridian 1 switch in Meridian Mail (see Chapter 6, “Meridian Mail subsystem configuration”), and on Symposium Call Center Server (see Chapter 7, “Symposium Call Center Server subsystem configuration”).

When to configure IVR ACD-DNs

Configure IVR ACD-DNs if you use Meridian Mail or another third-party IVR system to play messages to callers. These messages are stored on Meridian Mail or the third-party IVR system and can be announcements or voice menus. If your system only uses a MIRAN card to provide messages, you do not need to configure IVR ACD-DNs.

Configure an IVR ACD-DN for each group of voice ports (for example, one for ACCESS voice ports, one for non-ACCESS voice ports, and one for Voice Messaging [the Voice Messaging ACD-DN are not acquired by Symposium Call Center Server]).

For more information on voice port partitioning, see “Configuring voice ports” on page 44.

Assumptions

In the configuration descriptions, Meridian Mail is shown as the IVR source. As well, the following assumptions are made:

- You know the user ID and password to log on to the Meridian 1 administration terminal.
- You are familiar with Meridian 1 Change and Diagnostics overlays.
- You have a listing or printout of available IVR ACD-DNs.

- You are familiar with Meridian Mail.
- You can log on to Meridian Mail.

For more information on Meridian Mail, refer to the Meridian Mail documentation.

Configuring IVR ACD-DNs in LD 23

To configure IVR ACD-DNs in LD 23, use these prompts and responses.

Note: You must enter the responses to the prompts listed in the following table. These are not optional.

Prompt	Response	Description
REQ	NEW	Create a new queue.
TYPE	ACD	ACD data blocks
CUST	0–99	Customer number
ACDN	XXXX	The DN of the ACD queue. This is the IVR ACD-DN acquired from Symposium Call Center Server.
MWC	YES	Indicates that this is a message center and that the queue has agents.
CMS	YES	Command and Status Link Application Protocol is used.
IMA	YES	Enables IMS attendant.
MS	YES	Integrated voice messaging. This creates a message center from which messages can be retrieved.
VSID	XX	Do not enter a VAS ID.

MAXP	XX	Indicates the maximum number of ACD agents for this queue.
ALOG	YES	ACD agents are automatically logged on when Meridian Mail is powered on.
IVR	YES	Indicates that the queue can be used with the GIVE IVR command defined in scripts.
TRDN	XXXX	Default treatment DN is used if treatment is not specified in the script. **Note:** Use treatment DNs to select the treatment that the call receives from the voice processing system. You can also use them with Meridian Mail or any voice processing system that connects to the M1 by means of the APL link.
REQ	END	Exit from overlay.

After you finish

You must configure the voice ports as virtual agents. See “Configuring voice ports” on page 44.

Configuring voice ports

Introduction

Voice ports carry speech to Meridian Mail or an IVR system. You must configure voice ports when the ports are Meridian Mail or third-party IVR system ports used to play announcements or voice menus. You must configure voice ports on the Meridian 1 switch in Meridian Mail (see Chapter 6, “Meridian Mail subsystem configuration”), and on Symposium Call Center Server (see Chapter 7, “Symposium Call Center Server subsystem configuration”).

Enter Meridian Mail voice ports or third-party IVR system ports as virtual agent TNs. The class of service must be IMA and VMA to ensure that they are virtual agents.

Note: For Meridian Mail voice ports, ensure that the key layout matches the configuration of keys in Meridian Mail. This matching enables Meridian Mail to answer, disconnect, originate, transfer, and conference calls.

ATTENTION

Some services and applications that handle calls outside of Symposium Call Center Server control can share voice ports, while calls under Symposium Call Center Server control require dedicated voice ports to operate correctly. For more information see,“Voice port partitioning rules” on page 86.

Adding a Meridian Mail voice port with LD 11

To configure voice ports with Overlay 11, use these prompts and responses.

Prompt	Response	Description
REQ	NEW, CHG	Add or change a voice port.
TYPE	2008	
TN	10 (0-15)	Enter the TN of the agent.

KEY	0 ACD XXXX ZZZ NNNN	Define 0 as an ACD agent key. XXXX is the ACD DN of agents to voice mail. ZZZ is the CLID entry number. NNN is the position ID
KEY	1 SCN PPPP	Define key 1 as a single-call non-ringing DN. PPPP is the SCN DN of the SCN. The DN must match the DN on the Channel Allocation Table.
KEY	2 MSB	Define key 2 as a Make Set Busy key.
KEY	3 NRD	Define key 3 as a Not Ready key.
KEY	4 RLS	Define key 9 as a Release key.
KEY	6 TRN	Define key 6 as a Transfer key.
KEY	7 AO3 (letter 'O')	Define key 7 as a Conference key.
REQ	END	Exit from overlay.

Configuring agent and supervisor phonesets

Introduction

If you want a user to log on to a phoneset to receive or monitor incoming calls, you need to configure phonesets and users' phoneset attributes (TNs). You must configure agent and supervisor phonesets on the Meridian 1 switch and on Symposium Call Center Server (see Chapter 7, "Symposium Call Center Server subsystem configuration").

For complete information about features and how to configure agent and supervisor phonesets, refer to the *Administrator's Guide*.

Note: While agent and supervisor phonesets require no special configuration for Symposium Call Center Server, they must belong to an ACD-DN. Normally, call queueing is controlled by Symposium Call Center Server and is not affected by the ACD-DN.

The ACD-DN controls call routing if the Symposium Call Center Server CDN is in default mode or if incoming network ACD calls target the ACD-DN.

VSID definitions

Do not enter a VSID definition on agent or supervisor phonesets.

Assumptions

The following assumptions are made:

- You know the user ID and password to log on to the Meridian 1 administration terminal.
- You are familiar with Meridian 1 Change and Diagnostics overlays.

Example of defining a digital ACD phoneset

To define digital ACD phonesets, use LD 11 with the following prompts.

Note: For prompts that are not included in the following table, use the default value.

Prompt	Response	Description
REQ	NEW	Add new telephones.
TYPE	aaa	Enter telephone type as appropriate.
TN	l s c u	Terminal number
DES	ACD	
CUST	0–99	Customer number
KLS	1-7	Number of key/lamp strips attached
KEY 0	SRC XXXX YYYY	Where XXXX = ACD-DN, and YYYY = Agent Position ID
KEY 1	MSB	Make set busy.
KEY 2	NRD	Not ready
KEY 3	TRN	Transfer
KEY 4	AO6	6-party conference
KEY 7	SRC XXXX	XXXX=IDN
KEY 9	RLS	Release if type is = SL1.
REQ	NEW, END	Either define another multiline ACD phoneset, or exit the overlay saving all of the changes entered.

You program DWC functionality for Symposium Call Center Server in the same way as ACD, but the functionality is not the same. See Chapter 9, “Agent phoneset keys,” for information on the DWC key and Symposium Call Center Server.

If you want to enable the agent phoneset for Computer Telephony Integration (CTI), set the AST prompt to reflect the incalls key and the secondary DN key. For example, if you have a personal DN on key 7, set AST to 00 07 to CTI enable key 0 (the incalls key) and key 7.

Based on what is being acquired, ACQ AS can show any or all of the following values:

- TN for TN
- AST for position ID indicating that a Meridian Link application has registered for the ID
- AST for DN indicating that a Meridian Link application has registered for the secondary DN key

Verifying the TN configuration after being acquired

After you configure a phoneset in the system using LD11 and Symposium Call Center Server acquires it, the printout appears as follows:

```
>ld 11
REQ  prt
TYPE tnb
TN 4 0 4 2
DATE
PAGE
DES
DES agtset
TN 004 0 04 02
TYPE 2616
CDEN 8D
CUST 0
CDN  2003
AOM 0
FDN
TGAR 1
LDN no
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
CLS  CTD FBD WTA LPR MTD FND HTD ADD
     MWD AAD IMD DOS XHD IRD NID OLD VCE DRG1
     POD DSX VMD CMSD CCSD SWD LND CNDA
     CFTD SFD MRD DDV CNID
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDD CFXD ARHD FITD CNTD CLTD ASCD
     CPFA CPTA HSPD ABDD CFHD FICD NAID
     DDGA NAMA
     USMD USRD ULAD RTDD PGND OCBD FLXD
CPND_LANG ENG
HUNT
PLEV 02
AST
IAPG 0
AACS YES
ACQ AS: TN
ASID 16
SFNB 2 5 6 9 10 11 12 13 15 16 17 18 19
```

```
SFRB 1 2 15
USFB 1 2 3 4 5 6 7 9 10 12 13 14 15
CALB 1 3 4 5 6 8 9 11
FCTB
ITNA NO
DGRP
PRI 01
MLWU_LANG 0
DNDR 0
KEY 00 ACD 2001 0 2012
    SPV
  01 NRD
  02 A06
  03 MSB
  04 TRN
  05
  06
  07
  08 SCR 4702 0 MARP
  09 RAG
  10 AAG
  AA AMG
  12 DWC 2001
  13 ACNT
  14
15
```

Note: AACS=YES indicates that the phoneset has been acquired by an application. ACQ AS=TN indicates that the TN has been acquired but no CTI application has registered for the phoneset. ASID=16 indicates that the application on AML 16 has acquired the phoneset. SFNB, SFRB, USFB, and CALB are bitmaps that control what messages are sent to Symposium Call Center Server and are not user-definable.

Defining single-line ACD phonesets

To define single-line ACD phonesets, use LD 10 overlay and follow these prompts:

Prompt	Response	Description
REQ	NEW	Add new telephones.
TYPE	500	Enter telephone type as appropriate.
TN	l s c u	Terminal number
CUST	0–99	Customer number
DN	XXXX	DN for the telephone number

Prompt	Response	Description
CLS	AGTA	Class of service-ACD agent assignment
	THFA	Switchhook flash allowed.
	UND	Unrestricted access
	WTD, (WTA)	Warning Tone Denied (Allowed)
SPID	xxxx	Supervisor’s position ID number
PRI	(1)-48	Priority level for agent
AACD	YES	Associated set for ACD agent (X11 Release 17 or later software)
FTR	ACD XXXX YYYY	ACD feature allowed, where XXXX=the ACD-DN YYYY=the ACD Position (POS-ID)
REQ	NEW,END	Either define another single-line ACD phoneset or exit the overlay saving all of the changes entered.

Configuring routes

Introduction

A route defines a group of trunks. Each trunk carries incoming and outgoing calls to and from the switch. You must configure the routes on the switch. To use the Give RAN and Give MUSIC commands in your scripts, you must configure RAN and MUS routes.

Configure routes according to the Meridian 1 documentation. No special programming is required to work with Symposium Call Center Server. Use Overlay 16 to configure the routes and Overlay 14 to associate the routes with TNs.

Notes:

1. Only RAN, MUS, FGDT, TIE, DID, COT, FEX, and WATS route types are supported by the the application.
2. If you want to generate reports on routes, you must also configure them on Symposium Call Center Server.

Assumptions

The following assumptions are made:

- You know the user ID and password to log on to the Meridian 1 administration terminal.
- You are familiar with Meridian 1 Change and Diagnostics overlays.
- You have obtained a listing of routes using overlay program 21 (REQ=prt, TYPE=RDB).
- If you are not using Meridian Mail to perform voice processing, you do not need to configure any RAN routes.
- At the switch, you have ensured that physical routes are connected to the music source.

Configuring a RAN route with LD 16

To configure a RAN route using LD 16, use these prompts and responses:

Prompt	Response	Description
REQ	NEW, CHG	Add or change a route.
TYPE	RDB	Route data block
CUST	NN	Customer number
TKTP	RAN	Recorded Announcement trunk data block requires package 7.
REQ	END	Exit from overlay.

Configuring a MUS route with LD 16

To configure a MUS route using LD 16, use these prompts and responses:

Prompt	Response	Description
REQ	NEW, CHG	Add or change a route.
TYPE	RDB	Route data block
CUST	NN	Customer number
ROUTE	0-511	Route number
DES	X...X	Designator field for trunk
TKTP	MUS	MUSIC trunk data block requires Music package 44.
REQ	END	Exit from overlay.

Associating the route with TNs in LD 14

To associate a route with a TN in LD 14, use the following prompts and responses:

Prompt	Response	Description
REQ	AAA	Action request
TYPE	AAA	Route data block
REQ	END	Exit from overlay.

Configuring Multiple Queue Assignments

Introduction

If you want to use Multiple Queue Assignments (MQA) with Meridian MAX and you also have Symposium Call Center Server, you must ensure that MQA and Symposium Call Center Server can coexist.

To allow MQA and Symposium Call Center Server to coexist

1. Move all Symposium Call Center Server agents to a temporary ACD-DN from a source ACD-DN.
2. Change RPRT to NO in LD 23 for the source ACD-DN.
3. Move all Symposium Call Center Server agents back to the ACD-DN.
4. Repeat the steps for all ACD-DNs used by Symposium Call Center Server agents.

Note: If you change RPRT to NO for an ACD-DN that has associated agents, you get error sch6643.

Initializing or sys-loading the switch

Introduction

Symposium Call Center Server can only process calls if the Meridian 1 switch is operational. If you need to initialize the switch for any reason, shut down Symposium Call Center Server first.

To initialize the switch

1. On each Symposium Call Center Server client PC, log off SMI Workbench.
2. Shut down Symposium Call Center Server.
3. Install any needed software upgrades on the switch.
4. Restart the switch.
5. Restart Symposium Call Center Server.

Chapter 5

Meridian Link Services configuration

In this chapter

Overview

Introduction

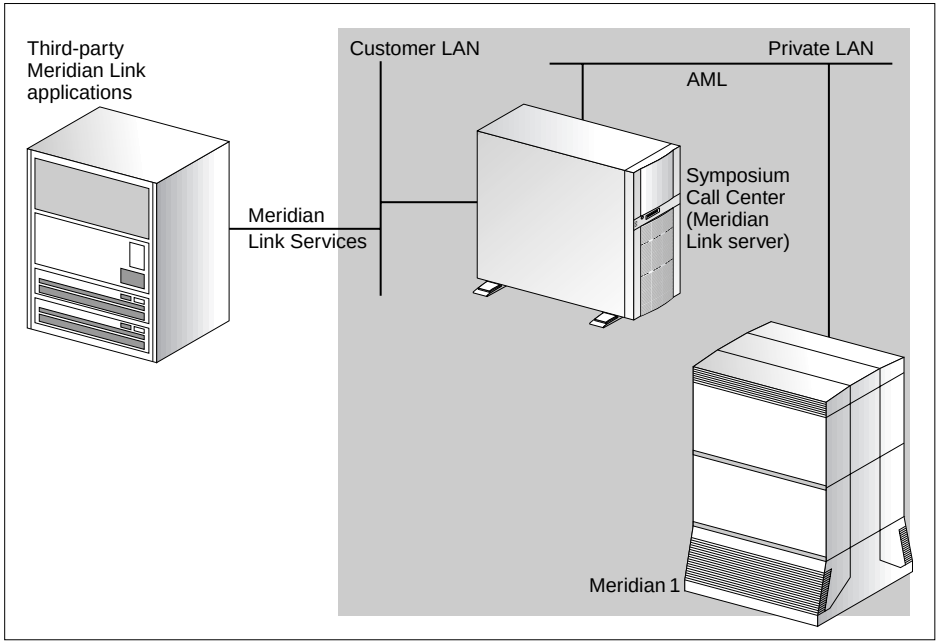
This chapter provides the references and procedures required to set up Meridian Link Services (MLS) for operation with Symposium Call Center Server. MLS is a communications facility that provides an interface between a host application and the Meridian 1. A host can be any computer on which the third-party application runs.

This interface facilitates the integration of the computer and the Private Branch Exchange (PBX). In this integrated environment, the host processor interacts with the Meridian 1 switch by exchanging application layer messages.

Use MLS to develop applications that allow you to use information taken from the switch (such as Caller ID), connect to another application to get that customer's data, and then provide a screen pop to help agents prepare for the call.

If you want to use CTI functionality for your Symposium Call Center Server agents, you must use MLS. You cannot use legacy Meridian Link.

The following diagram shows the relationships between the applications and Symposium Call Center Server.



Meridian Link

With the introduction of Symposium Call Center Server, Meridian Link was rewritten for Windows NT and renamed Meridian Link Services (MLS). MLS runs as a separate process on the Symposium Call Center Server. MLS can provide CTI features for Symposium Call Center Server agents, traditional non-Symposium Call Center Server ACD agents, and non-ACD phonesets. It also provides support for host-enhanced routing and host-enhanced voice processing.

Meridian Link Services features

MLS, wherever possible, preserves the functionality of Meridian Link 5. The *Meridian Link Interface Specification Addendum* describes in detail the differences in implementation between Symposium Call Center Server with MLS and the Meridian Link 5 interface specification.

Installing and configuring Meridian Link Services

Introduction

The configuration procedures on the Meridian 1 switch for both Symposium Call Center Server or stand-alone Meridian Link Services (MLS) are identical. You configure the Meridian 1 switch using X11 software overlays. Through the use of selected overlays, you can set up various resources and configurations to configure the switch properly for MLS.

Prerequisites

Before you configure MLS, you must install it. The software is included on the Symposium Call Center Server Release 3.0 Server Application CD-ROM. Refer to the *Software Installation and Upgrade Guide* for information on installing MLS.

You must also install the Meridian 1 software. For more information on installing the Meridian 1 software, refer to the installation documents that came with the Meridian 1 switch.

Enable Meridian Link Services on the Meridian 1

To enable MLS to function on the Meridian 1, you must

1. allow CTI operations on the ELAN (LD 17)
2. configure phonesets for CTI (LD 11 and LD 10)
3. configure CDNs for Host-Enhanced Routing (LD 23)

Once you configure the Meridian 1, you must connect the host application to the customer LAN (CLAN).

Allowing CTI operations on the ELAN

Introduction

You configure Overlay 17 to allow computer telephony integration with third-party applications. Specifically, where the VAS connection for Symposium Call Center Server is defined as SECU, enter YES as the prompt response. This allows third-party applications to control phoneset functions such as answering or initiating a call.

Completing the basic Meridian Link Services configuration

To complete the basic MLS configuration, use LD 17 with these prompts.:

Prompt	Response	Description
REQ	CHG	Change data in the database.
TYPE	VAS	Value Added Service
_VAS	NEW	When migrating from Meridian Link, define the VAS connection as YES.
_VSID	xx	Try to associate link and VASID so that the messages can be sent.
_ELAN	yy	Associate VASID xx with ELAN yy.
_SECU	YES	If the same ELAN link is used for the Meridian Link application
_INTL	1 – 12*	Time interval for checking Meridian Link for overload in 5-second increments
_MCNT	5 – 100000*	Message count threshold for number of Meridian Link messages per time interval

Defining phonesets for CTI

Introduction

To enable a phoneset or voice processing CTI, the phoneset must be configured as an associated set (AST). This is done using Overlay 11. This includes ASTs defined using LD 10 for single-line phonesets, and LD 11 for multiline phonesets.

To enable a phoneset or voice port CTI, the phoneset must be configured as an AST. This is done using Overlay 11 for digital (multiline) phonesets and Overlay 10 for analog (single-line) phonesets.

Symposium Call Center Server controls Status Change message filtering in MLS during resource acquisition. IAPG groups configured using Overlay 10 and Overlay 11 have no impact on message filtering for Symposium Call Center Server controlled resources.

The configuration of Symposium Call Center Server acquired sets is identical, but you can use Meridian Link Services for

- ACD sets where the routing is done on the Meridian 1
- ACD sets acquired by Symposium Call Center Server and used for skill-based routing
- non-ACD sets that are not acquired by Symposium Call Center Server but can be monitored by Meridian Link Services

These sets can be physical phones or voice ports (including line side E1 and line side T1) used by Symposium OPEN IVR or third-party interactive voice response applications.

Defining multiline ACD phonesets as associated sets

To define multiline ACD phonesets as associated sets, use LD 11. For more information about using LD 11, and creating and defining phonesets, refer to “Configuring agent and supervisor phonesets” on page 46.

Defining single-line ACD phonesets

To define single-line ACD phonesets as associated sets, use LD 10. Ensure that AST = YES. For more information about using LD 10 and creating single-line ACD phonesets, refer to “Configuring agent and supervisor phonesets” on page 46.

Defining non-ACD phonesets as ASTs

Introduction

To configure a DN as an AST telephone, you must make changes to the configuration record using LD 10 for single-line phonesets and LD 11 for multiline phonesets.

Defining non-ACD multiline phonesets

To define non-ACD multiline phonesets as associated sets, use LD 11 with these prompts.

Note: For prompts that do not appear in the following table, use the default value.

Prompt	Response	Description
REQ	NEW	Add new phoneset.
TYPE	aaa	Enter telephone type as appropriate.
TN	l s c u	Terminal number
CDEN	SD, (DD), 4D	Card density
CUST	0–99	Customer number
KLS	1-7	Number of key/lamp strips attached
		Press Enter until the AST prompt appears.
AST	00	DN key with AST telephone assignment (host controllable; up to two DN keys can be assigned as AST). This example shows that key 0 is an AST DN.
IAPG	Return	AML link status message group, defined in LD 15, is not used for MLS.
KEY 0	SCR XXXX	XXXX = DN
KEY 1	TRN	Transfer

Prompt	Response	Description
KEY 2	AO6	6-party conference
KEY 9	RLS	Release if type = SL1
		Press [Return] until the REQ prompt appears.
REQ	NEW, END	Either define another single-line ACD phoneset as an AST, or exit the overlay saving all of the changes entered.

Defining a non-ACD single-line phoneset

To define a non-ACD single-line phoneset as an associated set, use the LD 10 Overlay with these prompts.

Note: For prompts that do not appear in the following table, use the default value.

Prompt	Response	Description
REQ	NEW	Add new phoneset.
TYPE	500	Single-line telephone type
TN	l s c u	Terminal number
CDEN		
DES		
CUST	0–99	Customer number
DN	xxxx	DN for the telephone number
AST	YES	Phoneset is designated as an associated set.
IAPG	Return	AML link status message group, defined in LD 15, is not used for MLS.
		Press Enter until the REQ prompt appears.
REQ	NEW, END	Either define another single-line ACD phoneset as an AST, or exit the overlay saving all of the changes entered.

Configuring CDNs for host-enhanced routing

Introduction

Host-enhanced routing is a Meridian Link Services (MLS) feature that allows a third-party application to control calls that are waiting at a Controlled DN (CDN). A CDN is a specialized ACD queue that has no agents. A CDN in controlled mode offers control of calls to applications.

Note: MLS cannot use a CDN that is already acquired (controlled) by Symposium Call Center Server for host-enhanced routing. However, an application can register for a CDN so that it can receive messages about calls being handled at the CDN.

To set up a CDN for host-enhanced routing, follow the steps detailed in “Configuring CDNs” on page 34. These steps are identical to setting up a CDN that is acquired by Symposium Call Center Server.

Defining a controlled DN

To define a controlled DN for host enhanced routing, use LD 23. Refer to “Configuring CDNs” on page 34 for more information.

Connecting the host application

Introduction

The host application is connected on the Customer LAN (CLAN). The host application must point at the CLAN IP address. Refer to the *Meridian Link Addendum* for more information.

Running Meridian Link traces

Introduction

Symposium Call Center Server provides a tool for saving Meridian Link messages into a text file. You can use this tracing tool to debug problems with a third-party application using Meridian Link Services.

The executable for the tool is ml_trace.exe and is located in the NORTEL\ICCM\bin directory.

To enable message tracing

1. From the Windows Start menu, select Command prompt.
2. At the command prompt, type

   c:\nortel\iccm\bin\m1_trace.exe

   where c is the location of the nortel directory.

   Result: The ICCM Meridian Link Services Manager Trace Tool window appears.
3. Click Configuration and select Trace.

   Result: A dialog box appears.
4. From the dialog box, select the association ID of the application or applications to be traced.
5. Click On.
6. Click Activate.
7. Click Done.

   Result: The dialog box disappears.

To disable message tracing

1. From the Windows Start menu, select Command prompt.

   At the command prompt, type

   c:\nortel\iccm\bin\ml_trace.exe

where c is the location of the nortel directory.

Result: The ICCM Meridian Link Services Manager Trace Tool window appears.

2 Click Configuration and select Trace.

Result: A dialog box appears.

3 From the dialog box, select the association ID of the application or applications.

4 Click Off.

5 Click Done.

Result: The dialog box disappears.

Viewing the trace file

View the file MLSMTraceFile.txt in the directory NORTEL\ICCM\bin directory.

Chapter 6

Meridian Mail subsystem configuration

In this chapter

Overview

Introduction

This chapter provides information on how to set up and configure Symposium Call Center Server voice processing so that calls receive the appropriate treatments.

The first section in this chapter provides a high-level feature summary and background information on voice processing with Symposium Call Center Server. The sections that follow provide specific instructions on configuring Meridian Mail for voice processing. The last section gives a high-level overview of using an external IVR system to provide voice processing.

Section A: Voice processing

In this section

Overview of voice processing

Introduction

Voice processing provides automatic interaction with a caller. You can classify interactions in the following ways:

- **Passive:** playing prerecorded messages to a caller
- **Interactive:** collecting input from a caller, usually with DTMF

Different ways to do voice processing

Symposium Call Center Server supports voice processing using the following methods:

- **Script language commands:** Calls terminate on CDNs and enter the Symposium Call Center Server script. Script statements can direct a call to be connected to a voice port or RAN trunk so that voice processing interaction can take place.
- **Front-end IVR:** Calls terminate directly on a Meridian Mail Voice Menu or IVR system and are not controlled by Symposium Call Center Server until the voice processing engine transfers the call to a CDN. This method of voice processing is largely transparent to Symposium Call Center Server. Symposium Call Center Server must not acquire the voice ports used for front-end IVR.

Different ways to interact with callers

You can interact with callers in the following ways:

Play message to a caller

You can use any of the script language commands listed on the following pages to play a message to a caller. You can use the Give Controlled Broadcast command to play a recorded announcement to a caller in either start/stop mode (where the caller hears the entire message from start to finish), or continuous mode (where the caller can enter and exit at any point in the message). Other announcements do not use these specific modes.

Broadcast announcements to multiple callers

You can use Meridian Mail in stop/start mode to provide the same announcement to multiple callers.

Interact with an external voice system

Interaction with an external voice system enables Symposium Call Center Server to control communication with the caller through commands and treatments placed in the scripts. You can use the Open Voice Session and End Voice Session commands to interact with a caller directly. To use these commands, you must use Meridian Mail.

Interact with a caller indirectly

Communication with the caller is controlled by the voice processing system. You can interact with a caller indirectly by using the Give IVR command to connect him or her to a voice port controlled by Meridian Mail or an external IVR system.

Script language commands

Symposium Call Center Server supports the following voice processing commands:

- **Give RAN:** Use this command to play announcements using a MIRAN card or an announcement machine connected to a RAN trunk. With X11 release 23, it is possible to connect multiple callers to the same RAN port using the RAN Broadcast feature. As the call is connected to a RAN trunk rather than a voice port, this is not strictly a voice processing command. However, it does allow you to play a message to a caller.
- **Give IVR:** Use this command to play an announcement or IVR session using a Meridian Mail Voice Menu or an external IVR system. The voice processing engine controls the treatment that the call receives. The treatment can be based on the IVR ACD-DN or the treatment DN. Digits can be collected from the caller, but they cannot be accessed from the script unless Host Data Exchange (HDX) is used.
- **Give Controlled Broadcast:** Use this command to play a message to multiple callers, using the same voice port. It requires Meridian Mail.

- **Open/End Voice Session:** Use this set of commands to provide an interactive voice session in which you can play prompts and collect digits. It requires Meridian Mail.

Note: During script execution, all voice processing commands, as well as the Give RAN command, suspend the script until the command completes. Once the command completes, the next statement in the script is executed.

Typical uses of voice processing commands

Give RAN

Use Give RAN

- when messages must be spoken to callers
- when legacy RAN equipment already exists
- if RAN equipment is less expensive than a voice processing engine, and other voice processing functionality is not required

Give IVR

Use Give IVR

- when Meridian Mail is used to play announcements or give voice menus to callers and you do not want to use an ACCESS link. You usually use this when migrating from Meridian CCR and you do not want to rerecord your announcements or voice menus.
- when you are using Meridian Mail and you want to give the caller the option to leave a message in a mailbox
- when using a third-party voice processing engine for announcements or voice menus while the call is being controlled by a Symposium Call Center Server script. Usually, if an external IVR system is used, the call is directed to it before it enters the Symposium Call Center Server script (front-end IVR), and Give IVR is not used.

Give Controlled Broadcast Announcement

Use Give Controlled Broadcast Announcement

- when Meridian Mail is used and the same message must be given to multiple callers simultaneously (the traditional RAN scenario, “all agents are busy...”). Its port use is more efficient than when using RAN.

Open/End Voice Session

Use Open/End Voice Session commands

- when Meridian Mail is used and customized messages must be given (for example, a caller's expected wait time)
- when Meridian Mail is used and caller interaction (collect digits) is required

For more information on using these script commands in scripts, refer to the *Scripting Guide*.

Operation modes for voice processing commands

Listen only or interactive

If callers listen to the recorded message only, you can use the following voice processing commands:

- Give RAN
- Give IVR
- Give Controlled Broadcast Announcement
- Open/End Voice Session, Play Prompt

If callers can interact, you can use the following voice processing commands:

- Open/End Voice Session, Collect Digits (Meridian Mail)
- Give IVR (Meridian Mail Voice Menus or external IVR)

Single connection or broadcast

If callers, or a large number of callers, must hear the same announcement, you can use any of the listen-only modes from the previous section. However, port use is more efficient if the broadcast type of command is used, rather than the one-call-to-one-port commands. With broadcast, you can sustain much higher call rates with fewer ports.

Use the one-call-per-port commands if customized messages must be given to callers (for example, Expected Wait Time), if caller input is collected, or if you have a third-party voice processing engine.

The following commands connect multiple calls per port:

- Give Controlled Broadcast Announcement
- Give RAN (if RAN Broadcast is being used)

The following commands connect one call per port:

- Give IVR

- Open/End Voice Session

Start/stop or continuous

You can choose whether callers must hear an entire message, or whether they can enter and exit a message at any point.

Start/stop mode

Start/stop mode means that the caller hears the message from beginning to end. These voice processing commands can operate in start/stop mode:

- Give RAN
- Give IVR
- Give Controlled Broadcast Announcement
- Open/End Voice Session

Continuous operation

Continuous operation means that the message repeats all of the time, and a caller enters anywhere in the message. The following commands can support the continuous mode:

- Give RAN
- Give Controlled Broadcast Announcement

Note: Controlled Broadcast in continuous mode connects the caller immediately upon arrival and continues the script only after one full cycle of the message is heard. RAN operation has not changed with the introduction of Symposium Call Center Server.

Interruptible or non-interruptible

In Symposium Call Center Server Release 3.0, only the Give IVR command supports both interruptible (option) and non-interruptible (default) operation. Both the Controlled Broadcast and Voice Sessions commands support interruptible (default) operation only.

Interruptible operation

Interruptible operation means that if the call is queued *before* the voice processing statement is executed, the voice processing is discontinued if an agent becomes available, and the call is immediately presented to the agent instead.

Use this mode when the message played is for the caller's information (for example, "all agents are busy...") or amusement (for example, advertising), and it is important to get a call to an agent as quickly as possible.

Non-interruptible operation

Non-interruptible operation means that if the call is queued *before* the voice processing command started, the call does not qualify to be answered until the voice processing session reaches its logical end. The call, however, retains its place in queue during the voice processing session.

If an agent becomes available during the voice processing session, the next call that can be answered is presented instead, and when the call in the voice processing session finishes, it then goes to the next agent available.

This mode is useful when it is important that callers listen to a full cycle of a message before speaking to an agent, or when interactive menus are presented to the caller, and the input must be completed before it makes sense to speak to an agent.

If a call is *not* queued before a voice processing session, the interruptible versus non-interruptible operation has no effect; the call always operates in a non-interruptible fashion.

Resource acquisition

Voice processing resource acquisition summary

The following table summarizes the resources that Symposium Call Center Server must acquire for the different voice processing commands.

Command	IVR ACD-DN	Voice Port TN	Voice Port channel
Give IVR	Yes	Yes	No
Controlled broadcast	Yes	Yes	Yes
Voice Sessions	Yes	Yes	Yes
Front-end IVR	No	No	No

Section B: Meridian Mail as a voice processing engine

In this section

Overview

Introduction

You can configure Meridian Mail ports used by Symposium Call Center Server in the following ways:

ACCESS ports

Symposium Call Center Server controls these ports over the ACCESS link (the serial link that connects Meridian Mail and Symposium Call Center Server).

You use the following ports for Give Controlled Broadcast and Voice Sessions:

- voice ports (virtual agent TNs)

  For information on configuring voice ports, see “Configuring voice ports” on page 44.
- Meridian Mail mailboxes

  For information on creating Meridian Mail mailboxes, see “Creating Meridian Mail mailboxes” on page 96.
- voice segments

  For information on creating voice segments, see “Creating voice segments” on page 98.
- Channel Allocation Table

  For information on the Channel Allocation table, see “The Channel Allocation Table” on page 88.
- Voice Service DN Table

  For information on the Voice Service DN Table, see “The Voice Service DN table” on page 90.

Non-ACCESS ports

Meridian Mail controls these ports. The treatment that the caller receives is determined by the treatment DN or IVR ACD-DN.

You use these ports for Give IVR.

Based on your port types, you must configure some or all of the following entities in Meridian Mail:

- voice ports (virtual agent TNs)

  For information on configuring voice ports, see “Configuring voice ports” on page 44.

- treatment DNs (VSDNs)

  For information on configuring treatment DNs, see “Configuring VSDN entries (treatment DNs)” on page 105.

- Voice Menu service or Announcement Service

  For information on configuring treatment DNs, see “Creating announcements and voice menus” on page 103.

- Channel Allocation Table

  For information on the Channel Allocation table, see “The Channel Allocation Table” on page 88.

This section provides procedures for configuring each of these entities. It also provides port-sharing considerations, and describes the Channel Allocation Table and the Voice Service DN table where you configure these entities.

Voice port partitioning rules

Introduction

You can use a single Meridian Mail system to provide auto-attendant, voice menus, fax, voice mail, and applications services. The applications that can use Meridian Mail include Meridian Link, Symposium Link, Meridian (Integrated) IVR, CCR, and Symposium Call Center Server. Some of these services and applications can share ports, while others require a dedicated set of ports behind a dedicated ACD-DN to operate correctly.

Voice port partitioning

To allow for proper operation of Symposium Call Center Server and Meridian Mail or a third-party voice processing system, the voice ports used by Symposium Call Center Server must be dedicated to the Symposium Call Center Server application. You can achieve this by putting the voice ports into different ACD-DNs as follows:

Symposium Call Center Server Give IVR Ports

Configure the Meridian 1 and Meridian Mail to ensure that only Symposium Call Center Server Give IVR calls arrive at this ACD-DN. If you require other Meridian Mail services, such as Call Answering or Voice Menus, set up a separate queue for them (see “General Meridian Mail Services” on page 87).

The Give IVR ports must be configured as standard Meridian Mail or third-party IVR ports. The TNs for the ports must be acquired by Symposium Call Center Server.

Symposium Call Center Server ACCESS Voice Ports

Use these voice ports for Controlled Broadcast Announcement and Open Voice Session. Set up these voice ports as ACCESS Meridian Mail ports. Each port must be given a unique ACCESS class (channel number). The TNs and Mail channel numbers for these ports must be acquired by Symposium Call Center Server.

Set the Maximum Number of Broadcast Ports parameter (see “IVR ACD-DN global settings” on page 117) to limit the number of ports used by broadcast announcements.

Note: Once this limit is reached, calls receive silence instead of any voice processing treatments (for example, a recorded announcement).

General Meridian Mail Services

If you front-end a Symposium Call Center Server CDN with a voice menu, the voice menu should be accessed via this ACD-DN. Symposium Call Center Server does not acquire these ports.

These service include Call Answering, Express Messaging, Voice Menus, Fax on Demand, and voice ports used by CCR for Give IVR. The voice ports for all of these services can belong to the same ACD-DN. By default, this ACD-DN is 7000, which is also used to access call answering.

The Channel Allocation Table

Introduction

The Channel Allocation Table (CAT) allows you to

- view how channels are currently allocated across different queues and services
- see the distribution of port types (that is, the number of basic, full-voice, and multimedia ports)
- move agents from one queue to another, to dedicate a port (Voice ports are relocated on the Meridian 1 switch. The Channel Allocation Table is used to notify Symposium Call Center Server of the new location.)

Agents and DSP ports

The CAT determines how agents on the switch are associated with DSP ports on Meridian Mail. Each DSP port must be associated with an ACD agent defined on the Meridian 1 switch. Agents are identified by a terminal number (TN), an ACD directory number (DN), and a single call non-ringing (SCN) DN. This association enables both the queuing of calls coming in to Meridian Mail, and dial-out features such as Remote Notification, Delivery to Non-Users, and the Voice Prompt Editor.

When Meridian Mail is installed, the CAT is populated automatically by the installation technician. This is also true when you perform a channel expansion (to add new voice ports).

When to modify the CAT table

You modify the Channel Allocation Table when you need to

- move voice ports from one queue to another (to dedicate them to a particular service)
- program the ACCESS class (channel number) for ACCESS voice ports

Note: For non-ACCESS, use the default datafill settings for voice ports in the Channel Allocation Table.

The Voice Service DN table

Introduction

Voice service directory numbers (VSDNs) are defined for every Meridian Mail service that you want to make accessible to callers. These directory numbers (DNs) are entered in the Voice Service DN (VSDN) table, which maps DNs to services.

You define a VSDN for each Meridian Mail service that is accessed through a Treatment DN (for example, Express Messaging), and for Voice Menus and Announcement Services that are accessed via the Symposium Call Center Server Give IVR command. The VSDN entries used by Symposium Call Center Server are referred to as Treatment DNs.

Meridian Mail uses Treatment DNs that are accessed from the VSDN table. Treatment DNs accessed from the VSDN table have a default value associated with an ACD. However, Symposium Call Center Server can override the default value by placing a call to the ACD. The value used by Meridian Mail (either the default value or the Symposium Call Center Server value) determines which Meridian Mail service to start (for example, a recorded announcement).

You also need a VSDN entry for the IVR ACD-DN for the ACCESS voice ports.

Tip: Ensure that access DNs in your VSDN definitions do not duplicate mailbox numbers or Meridian 1 trunk route access codes.

Nightly audits

Meridian Mail performs an audit every morning at 3:30 a.m. This audit can take anywhere from 10 minutes (if the system has not been modified since the last audit) to 2 hours (if many changes have been made, such as adding or modifying users or services).

Note: If you are using Meridian Mail Call Path Diagnostics (CPD), you might receive error events indicating that calls arriving on voice ports are not under the control of Symposium Call Center Server.

Do not add, modify, or delete VSDNs during the nightly audit.

Section C: Configuring Meridian Mail for ACCESS voice ports

In this section

Overview

Introduction

If you want to use ACCESS voice ports (Controlled Broadcast or Voice Session), configure the following on Meridian Mail:

- ACCESS link to Symposium Call Center Server (see Chapter 3, “Subsystem connections configuration”)
- ACCESS IVR ACD-DN and ACCESS Treatment DN entry in the VSDN table
- voice ports in Channel Allocation Table
- mailbox for storing voice prompts (voice segments)
- voice segments

Configuring the ACCESS IVR ACD-DN and the ACCESS Treatment DN

Introduction

The ACCESS voice ports need to belong to a dedicated ACD-DN called the ACCESS IVR DN. You must add an ACCESS Treatment DN to the Meridian Mail VSDN table.

As with all of the IVR ACD-DNs, you must configure this DN on the Meridian 1 in Overlay 23 with IVR = YES (see Chapter 4, "Meridian 1 subsystem configuration"), and Symposium Call Center Server must acquire it (see Chapter 7, "Symposium Call Center Server subsystem configuration").

Note: The ACCESS IVR ACD-DN is the DN that you use in your scripts (for example, Open Voice Session 7001 or Give Controlled Broadcast 7001 [where 7001 is the ACCESS IVR ACD-DN]).

To add a VSDN entry for the ACCESS Treatment DN

1. From the Meridian Mail Main menu, go to the Voice Service-DN table.
   a. Select Voice Administration.
   b. Select Voice Services Administration.
2. From the VSDN table, press Add.
3. Enter the Access DN.
4. Type **ACC** in the Service field.
5. Type **0** as the ACCESS class.
6. Do not enter a Revert DN.
7. Type an optional comment in the Comment field, such as *Symposium ACCESS DN*.
8. Press Save.

   Result: You return to the VSDN table. If you pressed Save, there is now a new entry in the table for this DN.

Configuring ACCESS voice ports

Introduction

You must configure voice ports in the Channel Allocation Table. You must also configure voice ports as virtual agent TNs on the Meridian 1 switch (see Chapter 4, “Meridian 1 subsystem configuration”), and you must configure voice ports on Symposium Call Center Server (see Chapter 7, “Symposium Call Center Server subsystem configuration”).

Prerequisites

Before you configure a voice port in Meridian Mail, you must configure the voice port on the Meridian 1 switch using Overlay 11. The voice port must belong to the ACCESS IVR ACD-DN.

To configure voice ports in the Channel Allocation Table

Note: These steps can vary slightly on different releases of Meridian Mail. Refer to your Meridian Mail documentation.

1. From the System Status and Maintenance menu, disable the DSP port or ports that you want to configure.

   Tip: If you must disable multiple ports, it is quicker to change to range mode.

2. Select Channel Allocation Table from the System Status and Maintenance menu.

   a. If you have a single-site system, the Channel Allocation Table displays. Go to step 4.

   b. If you have a multisite system, go to step 3.

3. Enter the number of the site on which the port resides, and then press Return.

4. Modify the port. For each disabled port, you can change the values in the following fields:

   ACD DN: The ACCESS IVR ACD-DN.

SCN DN: This must match the Meridian 1 configuration.

Capability: Basic. The Meridian Mail keycode determines the types of ports that are available. Basic ports are less expensive to purchase and provide the capability required for ACCESS ports. If the keycode allows it, you can set the capability to Full.

Service: ACC for ACCESS. When you are prompted for an ACC Class, enter a unique number to identify the channel. The ACC Class is the channel number that is used when you acquire the voice port from Symposium Call Center Server. See Chapter 7, “Symposium Call Center Server subsystem configuration.’’

Note: You can only view ports that are not disabled. For disabled ports, the port capability (Full or Basic) is highlighted, and the ACD-DN, SCN, and Outbound fields are underlined.

5 Press Save.

Result: On a single-site system, you return to the System Status and Maintenance menu. On a multisite system, you are prompted for another site. If you have to reallocate ports on another site, return to step 3. Otherwise, press Cancel to return to the System Status and Maintenance menu.

6 Reenable any DSP ports that you put out of service before configuring the voice ports.

Creating Meridian Mail mailboxes

Introduction

You must define a Meridian Mail mailbox to hold the voice files and segments used by Symposium Call Center Server. Assign a password to the mailbox, and tell Symposium Call Center Server what the mailbox and password are in the IVR ACD-DNs Global Settings dialog box (see Chapter 7, “Symposium Call Center Server subsystem configuration”).

Creating Meridian Mail mailbox passwords

When you first create the password, Nortel Networks recommends that you create a password that is the maximum length. Change the password periodically, but only when the system is not in use.

ATTENTION

Do not allow individual Meridian Mail passwords to expire. If a Meridian Mail password expires, all voice processing in Symposium Call Center Server stops. Nortel Networks recommends that the administrator configure all passwords to never expire or, alternatively, to change individual passwords based on the system-defined frequency.

Changing Meridian Mail mailbox passwords

Use the Meridian Mail MMI on the Administration Console to change passwords. Do not use the phoneset logon interface. This mailbox is actively logged on for the ACCESS voice sessions, and voice processing could be affected while changing the password.

If the password expires and you cannot access the VPE, on the Modify Local User screen, change the Meridian Mail logon status field from disabled to enabled, and reenter your password in the New Password and Confirm Password fields. (Do not change your password. Enter the password you used before you were unable to access the VPE.)

If you enter a new password, you must also change the mailbox password on Symposium Call Center Server in the IVR ACD-DN Global Settings dialog box.

Number of allowed invalid logon attempts

You must configure the number of invalid logon attempts the user is allowed when logging on to Meridian Mail. The number of invalid logon attempts should be the same as or greater than the number of voice channels configured for the system.

Creating voice segments

Introduction

Use the Voice Prompt Editor (VPE) to create and delete voice files and to record voice segments. You can use voice segments as building blocks to create powerful, flexible voice applications. The maximum length of a voice segment is 120 seconds.

Note: Voice segment file names are case-sensitive. Voice segments included in scripts must be entered exactly as they appear in the Voice Prompt Editor.

Meridian Mail assigns sequential numbers to each segment created in a voice file using the VPE. The administrator must keep track of which file or segment number corresponds to which spoken prompt.

Note: To use the Voice Prompt Editor, the Channel Allocation Table on Meridian Mail must include at least one channel that has an outbound service value of “all.”

ATTENTION

Do not delete voice segments. This renumbers the segments and puts the configuration of Meridian Mail and the voice segment variables on Symposium Call Center Server out of sync. If segments are deleted, you must manually update the Symposium Call Center Server’s variables.

ATTENTION

Nortel Networks recommends that you limit the number of voice files to two.

When to configure voice files

If you want to include a PLAY PROMPT element in your script, you must configure voice prompts.

Prerequisites

The ACCESS link between Meridian Mail and Symposium Call Center Server has already been set up.

To configure voice prompts

1. At Meridian Mail, create a mailbox for prompt storage. Make sure that the mailbox is empty and note the password.
2. At the application, at the IVR ACD-DN Global Settings dialog box, enter the mailbox number and password.
3. Choose one of the following actions:
   a. If you have a prompt tape, go to step 4.
   b. If you want to create your own voice prompts, go to step 5.

   Note: Nortel Networks does not provide a voice prompt tape as part of the standard product. You can create a tape by creating the prompts manually and then using a Meridian Mail tool to transfer them to tape (refer to the Meridian Mail documentation for instructions on how to create a voice prompt tape). Create the tape at one call center and use this facility to copy the prompts to other call centers in the network.
4. If you have a Symposium Call Center Server prompt tape, load the prompt files into the Meridian Mail mailbox by following these steps:
   a. At Meridian Mail, go to the TOOLS level.
   b. Select Others, and then select Transfer voice prompts.
   c. Follow the Meridian Mail screen steps. All system-provided Symposium Call Center Server script variables should already have mapping to the file and segments previously created and loaded from the tape to the Meridian Mail mailbox.
5. To create your own prompts, from the SMI window in Symposium Call Center Server, follow these steps.
   a. Start the Voice Prompt Editor and create the appropriate prompts.
   b. From the Script Variables window, create new voice prompt variables and associate the file name and segment name into the variables.
6. Once you create all of the prompts, write the the application script, which includes the PLAY Prompt command. For more information on scripts and script variables, refer to the *Scripting Guide*.

Section D: Configuring Meridian Mail for Give IVR

In this section

Overview

Introduction

If you are using Meridian Mail for Give IVR, use non-ACCESS voice ports. Configure the following elements on Meridian Mail:

- voice menus or announcement services, or both
- VSDN entries (treatment DNs) for each voice menu or announcement

You can use announcement services to give in-queue announcements (“Your call is in queue and will be answered shortly”). Prior to Meridian Mail 11, the announcement service did not have the Silent Disconnect option and the announcement was repeated twice. If your Meridian Mail is earlier than Release 11, use voice menus instead.

You can use voice menus to give the caller a choice (“Press 1 for Sales or press 2 for Support”). You can also use voice menus without any options as an alternative to an announcement service.

Prerequisites

The voice ports must belong to a dedicated IVR ACD-DN, and the Channel Allocation Table must match the Meridian 1 configuration. The dedicated IVR ACD-DN is the DN that you use in your scripts (for example, GIVE IVR 7002 where 7002 is the dedicated IVR ACD-DN).

You can configure the voice ports as Basic or Full Service in the Channel Allocation Table.

Creating announcements and voice menus

Announcement services

Follow the instructions in the *Meridian Mail Voice Services Application Guide* to create your announcement service.

Nortel Networks recommends that you select the following options when creating your announcement service:

- Silent Disconnect (so that the caller does not hear “Good-bye” at the end of an announcement)
- Number of times to play announcement should be 1
- Do not set an ACCESS password
- The access to the announcement is via the Give IVR command (see the next section)

Voice menu service

If you want to give the customer a simple voice menu (“Press 1 for Sales and press 2 for Support”), then Nortel Networks recommends that you use ACCESS ports with the Voice Session Play Prompt/Collect Digits Command. This is easier to configure and is more flexible.

If you do want to use a Meridian Mail voice menu, follow the instructions in the *Meridian Mail Voice Services Application Guid*e to create your voice menu service.

Voice menu as an announcement

If you are using a voice menu to provide an announcement service, Nortel Networks recommends that you select the following options:

- Silent Disconnect (so that the caller does not hear “Good-bye” at the end of an announcement)
- Do not set an ACCESS password
- Record the announcement you want the callers to hear as the Greeting
- Do not record the Menu Choices

- Do not assign any action to each key
- Initial No Response DS (disconnect)
- Delayed Response DS (disconnect)

Note: Disconnect (DS) means that Meridian Mail drops out of the call and control returns to the Symposium Call Center Server script. It does not mean that the caller is disconnected.

Voice menu to offer choices to a caller

If you are using a voice menu to offer choices to a caller, Nortel Networks recommends that you select the following options:

- Silent Disconnect (so that the caller does not hear “Good-bye” at the end of an announcement)
- Do not set an access password
- Record the announcement you want the callers to hear as the Greeting
- Record the Menu Choices
- Assign an appropriate action to each key

  Typically, this is a call (CL) with an ACD-DN. For more information, see “Caller hears voice prompts but is never presented to an agent” on page 144.
- Initial No Response DS (disconnect) or RP (Repeat Menu Choices) as appropriate
- Delayed Response DS (disconnect) or RP (Repeat Menu Choices) as appropriate

Notes:

1. The DNs in the actions for each key can be a CDN controlled by Symposium Call Center Server or a Phantom DN that forwards the call to a Symposium Call Center Server CDN. In these cases, the call returns to the Master_Script as a new call and must be treated appropriately.
2. If you use Meridian Mail voice menus to transfer a call to a CDN, the call is pegged as a new call.

Configuring VSDN entries (treatment DNs)

Introduction

To be able to access the voice menu or announcement services, or both, create an entry for each in the VSDN table.

These entries are known as treatment DNs and are used in Symposium Call Center Server scripts to specify which treatment the callers receives (for example, Give IVR 7002 With Treatment 1001, where 1001 is a treatment DN in the VSDN table that points to a specific announcement).

The treatment DN is passed by the the application to the Meridian 1 switch which, in turn, relays it to Meridian Mail. The switch does not interpret a treatment DN, and it does not need to appear anywhere in its configuration.

Note: The Give IVR with Treatment command apply only to the Meridian Mail voice processing system.

Defaults

You can specify an explicit treatment DN in the script for all Give IVR commands. If a treatment DN is not specified, the default treatment DN given on the switch in Overlay 23 (TRDN prompt) for the IVR ACD-DN is used. A null value is sent from the server to the switch, prompting the switch to insert its default before relaying it to Meridian Mail.

To create a VSDN entry

1. From the Meridian Mail Main menu, go to the Voice Service-DN table.
   - **a.** Select Voice Administration.
   - **b.** Select Voice Services Administration.
   - **c.** Select Voice Services-DN table.
2. Press ADD to add a new entry.
   - **a.** The access DN is the treatment DN.

b. Select AS for Announcement Service or MS for the Voice Menu Service.

c. Enter the Voice Menu or Announcement Service ID that was assigned when you created it.

d. For a Voice menu, select Basic as the session profile.

Note: If you have Basic ports defined in the CAT table and you select anything other than Basic in the session profile, the Give IVR command fails on these ports and the caller receives an announcement similar to "Your session cannot be completed at this time."

e. Add an optional comment (for example, Symposium Announcement).

f. Press Save.

3 Repeat for each Announcement or Voice Menu Service.

Section E: Third-party voice processing engines

In this section

Overview of third-party voice processing engines

Introduction

External IVR systems can connect their voice channels to the Meridian 1 in a variety of ways, including as 2500 (analog) set TNs, 2500 (analog) set ACD TNs, and T1, DPNSS, or ISDN channels.

Note: In this section, an external IVR system refers to any device not already discussed that can provide voice services. This includes third-party IVR systems, Symposium OPEN IVR, and third-party voice mail systems.

External IVR systems not accessed from Symposium Call Center Server scripts

Usually, if an external IVR provides a voice menu or other caller interaction, the calls terminate on it directly and are transferred to the Symposium Call Center Server CDN at the end of the IVR session (front-end IVR).

If the call is already under the control of a Symposium Call Center Server script, it can be handed off to the IVR system by using the Route Call script command.

IVR systems used in this manner do not require any special configuration on Symposium Call Center Server. The voice ports and ACD-DN do not need to be acquired.

Using external IVR systems with Give IVR

You can use IVR systems that connect as ACD sets (including Line Side T1 and Line Side E1 connections, which the Meridian 1 treats as analog ACD agents) to service the Symposium Call Center Server Give IVR script command. You can use IVR systems to give announcements or offer voice menus while the call is under the control of the script and is waiting in a queue.

Treatment DNs

Most external IVR systems do not support the WITH TREATMENT part of the Give IVR command. This means that each IVR ACD-DN can only offer one type of treatment.

Some IVR systems support an APL link or Meridian Link that can be used to deliver the treatment DN to the IVR system to determine the message played.

Meridian 1 and Symposium Call Center Server configuration

To use an external IVR system with the Give IVR command, the required configuration is similar to using Meridian Mail for Give IVR:

- The voice ports must belong to an IVR ACD-DN (with IVR = YES).

  As with Meridian Mail, configure the Meridian 1 so that only Symposium Call Center Server Give IVR calls terminate on this ACD-DN.
- Symposium Call Center Server must acquire the IVR ACD-DN.
- Symposium Call Center Server must acquire the voice port TNs.

Configuration of the external IVR

This varies from one IVR system to another and is beyond the scope of this document.

Chapter 7

Symposium Call Center Server subsystem configuration

In this chapter

Overview

Introduction

The Symposium Call Center Server subsystem requires the following major configuration elements for call processing:

- CDNs
- IVR ACD-DNs
- agent phonesets (TNs)
- voice ports (virtual agents)
- IVR ACD-DN Global Settings
  - the default ACCESS treatment DN
  - the Meridian Mail mailbox and password
  - the maximum number of broadcast IVR ports reserved for broadcasts
  - the broadcast voice port wait timer
- voice segment variables
- the application scripts

Configure resources such as agent phonesets, CDNs, and voice ports on the Meridian 1 switch and Meridian Mail before acquiring them from Symposium Call Center Server.

For detailed information on how to configure switch resources for the application, refer to the *Administrator's Guide*.

For detailed information on how to plan and write scripts, refer to the *Scripting Guide*.

CDNs

Introduction

You must configure and acquire all CDNs referenced by scripts and on which calls for Symposium Call Center Server arrive. These CDNs must match those that are configured on the Meridian 1. For more information on referencing CDNs in scripts, refer to the *Scripting Guide*.

To configure and acquire a CDN

1. From the SMI window, open Switch Administration → CDNs.

   Result: The CDNs window appears.

2. Choose one of the following actions:

   a. If you want to configure a new CDN, choose File → New.

   b. If you want to configure an existing CDN, select an existing CDN and choose File → Properties.

   Result: The CDN Properties page appears.

3. Enter the name, number, and call type assigned to the CDN.

4. Click OK.

   Result: The CDN is added to the CDNs window with the status Not Acquired.

5. Select the CDN that you want to acquire and choose File → Acquire.

   Result: The status of the CDN changes to Acquire Pending.

6. To display the results of attempting to acquire the CDN, select View → Refresh.

Note: If you intend to reconfigure a CDN, you must first deacquire the CDN, edit the configuration parameters, and then reacquire the CDN.

IVR ACD-DNs

Introduction

Configure and acquire every Meridian 1 ACD-DN used in voice processing (an ACD-DN behind which the voice ports are defined) by Symposium Call Center Server.

To configure and acquire an IVR ACD-DN

1. From the SMI window, open Switch Administration → IVR ACD-DNs.

   Result: The IVR ACD-DNs window appears.

2. Choose one of the following actions:

   a. If you want to configure a new IVR ACD-DN, choose File → New.

   b. If you want to configure an existing IVR ACD-DN, select an existing IVR ACD-DN and choose File → Properties.

   Result: The IVR ACD-DNs Properties page appears.

3. Enter the IVR ACD-DN name and number, and select the threshold class.

4. Click OK.

   Result: The IVR ACD-DN is added to the IVR ACD-DNs window with the status Not Acquired.

5. Select the IVR ACD-DN to acquire and choose File → Acquire.

   Result: The status of the IVR ACD-DN changes to Acquire Pending.

6. To display the results of attempting to acquire the ACD-DN, select View → Refresh.

Note: If you intend to reconfigure an ACD-DN, you must first deacquire the ACD-DN, edit the configuration parameters, and then reacquire the ACD-DN.

Configuring agent phonesets (TNs)

Introduction

Configure and acquire phonesets, which are also referred to as TNs, for all voice ports being used by Symposium Call Center Server.

The Meridian 1 system types use the following TN formats:

- On the Meridian 1, the format is “loop unit” (for example, 8-5).
- For Option 11 Meridian 1 systems, the Symposium Call Center Server TN format is “loop-0-0-unit” (for example, 8-0-0-5).
- All Meridian 1 system types, except Option 11, use the format “loop-shelf-card-unit” (for example, 24-0-4-5).

To configure and acquire a phoneset

1. From the SMI window, open Switch Administration → Phonesets.

   Result: The Phonesets window appears.

2. Choose File → New.

   Result: The Phoneset Properties property page appears.

3. Enter the terminal name and telephony or port address. Ensure that the Add Voice Port check box is checked.

4. Click OK.

   Result: The phoneset is added to the Phonesets window.

5. Select the phoneset to acquire and choose File → Acquire.

   Result: The status of the phoneset changes to Acquire Pending.

6. To display the results of attempting to acquire the phoneset, select View → Refresh.

Note: If you intend to reconfigure a phoneset, you must first deacquire the phoneset, edit the configuration parameters, and then reacquire the phoneset.

Voice ports (virtual agents)

Introduction

Configure and acquire the voice ports and channel numbers being used by Symposium Call Center Server.

To add a phoneset as a voice port

1. Configure a new phoneset in the Phonesets application to create a virtual TN. See “Configuring agent phonesets (TNs)” on page 115.
2. Click the Add Voice Port box on the Phonesets property page when you create the phoneset. Once you configure the phoneset as a voice port, it appears in the Voice Ports window.
3. From the Voice Ports application, select the phoneset that you added as a voice port.
   a. If it is a non-ACCESS voice port, choose Acquire from the File menu to acquire the port.
   b. If it is an ACCESS voice port, enter the Channel Number (this is the ACCESS class number configured on Meridian Mail for the voice port in the Channel Allocation Table) and choose Acquire from the File menu to acquire the port.

Note: If you intend to reconfigure a voice port, you must first deacquire the voice port, edit the configuration parameters, and then reacquire the voice port.

IVR ACD-DN global settings

Introduction

To support voice processing in Symposium Call Center Server, you must configure a number of items on the IVR ACD-DN Global Settings dialog box. These include

- the number of voice ports that can be used for broadcast announcements
- the wait time for a start/stop broadcast announcement (the amount of time between the arrival of the first call for the start/stop broadcast announcement and when the announcement actually starts)
- the default Meridian ACCESS treatment DN used in the controlled broadcast and voice session script commands if a treatment DN is not explicitly specified in the command within the script
- a Meridian Mail mailbox and password for the mailbox containing the voice files and segments used by controlled broadcast and voice session script commands

Maximum number of broadcast ports

Configure the number of ports that can be used for Controlled Broadcast at any given time. Up to 50 calls can be attached simultaneously to a single voice port on a broadcast announcement. The fifty-first call for an announcement is connected to a new port, as long as the maximum number of broadcast ports is not exceeded. Once the maximum is exceeded, new calls do not receive a Broadcast announcement (new call receive silence.)

If the voice ports are partitioned so that Broadcast calls are directed to a dedicated IVR ACD-DN (that is, it does not share ports with Voice Sessions), then the setting of this parameter is not important as long as it is set equal to or larger than the number of ports in this IVR ACD-DN.

However, if voice ports are shared between Broadcast announcements and Voice Sessions, it might be important to limit the number of ports that can be used by broadcasts so as not to starve the Voice Sessions calls from getting a port.

You can roughly calculate the number of Broadcast ports needed using the call arrival rate, the length of the announcement, and, if start/stop operation is used, the Broadcast Wait Timer. The goal is to minimize the number of ports used by broadcast and to maximize the number of ports used by Voice Sessions. (Since Voice Sessions needs a one-call-to-one-port arrangement, its port use for the same call traffic is generally higher.)

Broadcast voice port wait timer

The value of this timer determines how many calls have a chance to be connected to the same port. The timer matters only if Broadcast voice processing is used in start/stop mode. Continuous mode connects calls immediately upon arrival.

A longer timer allows more calls to connect to the same voice port. Conversely, a shorter timer allows call to get into the announcement more quickly, but, on average, fewer calls use a single port (that is, less efficient use of ports). The default setting is ten seconds and the appropriate setting for this parameter can vary widely from one call center to the next.

Default ACCESS treatment DN

If you do not explicitly specify a treatment DN in the Open Voice Session or Give Controlled Broadcast Announcement command within a script, then the default ACCESS treatment DN is used.

Meridian Mail mailbox and password

To use the Give Controlled Broadcast Announcement and Open/End Voice Session commands, you must configure the Meridian Mail mailbox and password containing the voice files. Only one mailbox is configurable for the application system.

Note: Nortel Networks recommends that you only use the Meridian Administration terminal to change the Meridian Mail mailbox password on Meridian Mail. Do not use the phoneset to change the password as the mailbox could be in use by voice processing and this could interrupt service.

Prerequisites

Before you configure global settings for IVR ACD-DNs in the application, perform the following tasks:

- Ensure that the Meridian Mail mailbox and password are defined on the Meridian Mail subsystem.
- Ensure that the IVR ACD is defined on the Meridian 1.
- Ensure that the Treatment DN is defined on Meridian Mail.

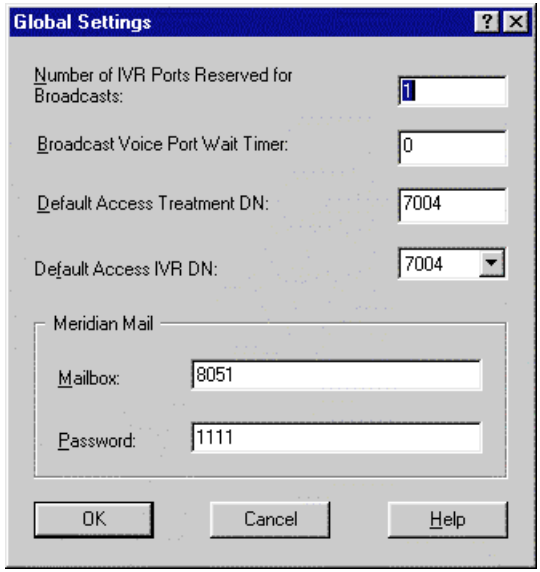
To configure the global settings for IVR ACD-DNs

1 From the SMI window, open Switch Administration → IVR ACD-DNs.

Result: The IVR ACD-DNs window appears.

2 Choose File → Global Settings.

Result: The Global Settings dialog box appears.



3 Enter the number of IVR ports reserved for broadcast.

4 Enter the broadcast voice port wait timer.

5 Enter the default ACCESS treatment DN.

6 Enter the default ACCESS IVR DN.

7 Enter the DN of the Meridian Mail mailbox.

8 Enter the password required to access the Meridian Mail mailbox.

9 Click OK.

Voice segment variables

Introduction

Symposium Call Center Server scripts reference voice segments on Meridian Mail by using voice segment variables. Voice segments can contain one or more voice segment variables. These variables refer to specific words or phrases that are datafilled in Meridian Mail. Each voice segment variable has a name, number, and value that indicates the language used to record the segment.

There are two types of voice segments:

- user defined
- system predefined

User-defined voice segments

You record the voice segments using the Voice Prompt Editor (VPE) on the Symposium Call Center Server client. Then you define the variables using the Script Variables dialog box. A voice segment variable has the type VOICE SEGMENT and can be either a global or call variable. You can define any number of variables.

You can define the variables on Symposium Call Center Server at any time. Neither Symposium Call Center Server nor Meridian Mail checks for the existence of the segments on the other platform except at run time. When Symposium Call Center Server instructs Meridian Mail to play a specific voice segment from a specific file, the referenced segment must exist.

Note: Voice segment file names are case-sensitive. Voice segments included in scripts must be entered exactly as they appear in the Voice Prompt Editor.

For more information on creating voice segments, refer to the *Administrator's Guide.*

System-predefined phrases

To generate spoken numbers, Symposium Call Center Server provides a number of predefined voice segment variables representing spoken numbers. Symposium Call Center Server strings the variables together automatically to create the ability to speak numbers 0 – 999 999 999 999 999.

Predefined voice segment variables have file and segment number "placeholders" when the Symposium Call Center Server system is installed. Record the file and segment numbers referenced by these variables.

Script example

In the following example, a voice session begins in which a caller hears a message prompting him or her to enter an identification number by pressing the phoneset keys. The seven digits entered are collected into a variable named "vardigit_cv". The caller then hears a second message in which the numbers entered are spoken back.

```
OPEN VOICE SESSION 2299
      PLAY PROMPT VOICE SEGMENT enter_ID_number_vs
      COLLECT 7 DIGITS INTO vardigit_cv
      PLAY PROMPT NUMBERBYDIGIT vardigit_cv
END VOICE SESSION
```

Scripts

Introduction

You use script commands to direct calls to voice processing. The services that a particular caller hears depends on the path the call follows through the master script and any secondary scripts. Information about the voice processing treatment that a call receives by Symposium Call Center Server is pegged in the database. This allows you to run reports showing details about voice processing and its effects in your call center.

For more information on scripts, refer to the *Scripting Guide*.

CDNs

Ensure that the script references a CDN that is configured and acquired on Symposium Call Center Server.

NACD ACD-DNs

To route calls to a remote ACD-DN, the Symposium Call Center Server script must contain the following command:

```
QUEUE TO NACD acd-dn [WITH PRIORITY  priority]
```

The script can contain other commands to control the wait time or to change the priority.

IVR ACD-DNs and treatment DNs

All voice processing script commands need and use both an IVR ACD-DN, as well as a treatment DN. Both parameters are optional in the script statement and the defaults are drawn from different places.

The IVR ACD-DN on the voice processing script statement specifies the Meridian 1 ACD-DN to which the voice port TNs belong in the Meridian 1 configuration. Symposium Call Center Server directs voice processing calls to the IVR ACD-DN, and the Meridian 1 ACD software distributes the calls over the voice port (that is, the Meridian 1 selects the actual voice port, not the Symposium Call Center Server software).

Note: If TRDN is not configured in Meridian 1 IVR ACD-DN, then you must include “with treatment” in the Give IVR script element.

Symposium Call Center Server must acquire the IVR ACD-DN for voice processing to operate correctly.

Routes

To use Give Music or Give RAN commands, in the Symposium Call Center Server script, you must reference a RAN or MUS route.

If you want to generate all trunk busy (ATB) reports, then you must configure the routes on Symposium Call Center Server, and you must acquire the routes.

You do not need to acquire music or RAN routes.

Chapter 8

Troubleshooting

In this chapter

Overview

Introduction

This section provides information on how and where to check for the status of the various configuration elements and parameters mentioned in the checklists.

For detailed information on how to configure the elements in each of the subsystems used with the Meridian 1 switch, Meridian Link Services, and voice processing for use with Symposium Call Center Server, refer to Chapter 2, “Configuration overview.”

Subsystem link problems

Server is booted and running

- Server: Status of all of the components is shown as UP by the SMonW utility.
- M1: ELAN connection to Meridian 1 is functioning (see below for specifics).
- Client: Successful logon to server.

ELAN between Server and M1

- M1: LD 48; stat ELAN
- Status must show ACTIVE EMPTY and APPL ENABLE for the ELAN connected to the Server (match the IP addresses if there are multiple ELANs).

Example

```
ld 48
LNK000
.stat elan
SERVER TASK:  ENABLED
ELAN #: 16
   APPL_IP_ID: 47 .82.46.202 LYR7: ACTIVE EMPTY    APPL AC-
TIVE
```

CSL between Meridian 1 and Meridian Mail

- M1: LD 48; stat AML
- Status must show ACTIVE EMPTY for the link number connected to Meridian Mail.

Example

```
ld 48
LNK000
.stat aml
AML: 07 MSDL: 07 PORT: 03 DES: mail
 LYR2: EST AUTO: ON LYR7: ACTIVE EMPTY
```

ACCESS Link between server and Meridian Mail

- Meridian Mail: Log on as tools (and use the password configured for your system).
- Menus:
  - 13 Other
  - 2 ACCESS Diagnostics
  - Status of the link to the Server must show Synchronized.

Example

Links	Description	Location	TKMstat	TCstat	LinkStatus
1	ACCESS	1-6-1	Running	Running	Synchronized

Meridian 1 loop, shelves, and cards

- M1: LD 32; stat n1 n2 n3, where n1 is the loop, n2 is the shelf, and n3 is the card that contains either agents or voice ports.
- Status for real agents shows Logged in or Logged out, based on the state of the agent.
- Status for Meridian Mail voice ports should always show Logged in. If not, disable and enable the port on Meridian Mail to trigger the autologon.

Example

Loop

```
ld 32
NPR000
.stat 24
SUPER LOOP
000 DSBL     038 BUSY
```

Example

Real agents (2500 set agents)

```
.stat 24 0 0
00 = UNIT 00 = IDLE  (L500 LOG IN )
01 = UNIT 01 = IDLE  (L500 LOG IN )
02 = UNIT 02 = IDLE  (L500 LOG IN )
03 = UNIT 03 = IDLE  (L500 LOG IN )
```

```
04 = UNIT 04 = IDLE  (L500 LOG IN )
05 = UNIT 05 = IDLE  (L500 LOG IN )
06 = UNIT 06 = IDLE  (L500 LOG IN )
07 = UNIT 07 = IDLE  (L500 LOG IN )
08 = UNIT 08 = IDLE  (L500 LOG IN )
09 = UNIT 09 = IDLE  (L500 LOG IN )
10 = UNIT 10 = IDLE  (L500 LOG IN )
11 = UNIT 11 = IDLE  (L500 LOG IN )
12 = UNIT 12 = IDLE  (L500 LOG IN )
13 = UNIT 13 = IDLE  (L500 LOG IN )
14 = UNIT 14 = IDLE  (L500 LOG IN )
15 = UNIT 15 = IDLE  (L500 LOG IN )
```

Example

Voice Ports (SL1 sets)

```
.stat 4 0 3
00 = UNIT 00 = IDLE  (BCS  LOG IN )
01 = UNIT 01 = IDLE  (BCS  LOG IN )
02 = UNIT 02 = IDLE  (BCS  LOG IN )
03 = UNIT 03 = IDLE  (BCS  LOG IN )
04 = UNIT 04 = IDLE  (BCS  LOG IN )
05 = UNIT 05 = IDLE  (BCS  LOG IN )
06 = UNIT 06 = IDLE  (BCS  LOG IN )
07 = UNIT 07 = IDLE  (BCS  LOG IN )
```

Meridian Mail ports enabled

- Meridian Mail: Menus
  - 5 System Status and Maintenance
  - 3 DSP Port Status
  - Status for the ports should show Idle (if they are acquired by the TN only from the Server), or Active (if they are acquired by the TN and the channel by the server, if they are busy on a call, or if they are acquired by another ACCESS application).

CDN acquired

- Client: CDNs status must show Acquired.
- M1: LD 23; REQ PRT; TYPE CDN

- The AACQ prompt in the printout should be YES; the ASID prompt should be equal to the ELAN connected to the Symposium Call Center Server; the CNTL prompt should be YES.

Example

```
>ld 23
ACD000
MEM AVAIL: (U/P): 3591770    USED: 405925    TOT: 3997695
DISK RECS AVAIL: 2682
ACD DNS  AVAIL: 23758    USED:   242    TOT: 24000
REQ  prt
TYPE cdn
CUST 0
CDN  2003
TYPE CDN
CUST 0
CDN  2003
FRRT
SRRT
FROA NO
MURT
DFDN 7700
CEIL 2047
OVFL NO
TDNS NO
RPRT YES
AACQ YES
ASID 16
SFNB 1  2  3  4  5  6  9  10  11  12  13  15  16  17  18  19
USFB 1  2  3  4  5  6  7  9  10  11  12  13  14  15
CALB 0  1  2  3  4  5  6  7  8  9  11
CNTL YES
VSID
HSID
CWTH 1
BYTH 0
OVTH 2047
STIO
TSFT 20
```

Correct script activated

- Client: Script must be in Active state.

IVR ACD-DN acquired

- Client: Status should show Acquired.
- M1: LD 23; REQ PRT; TYPE ACD
- The AACQ prompt in the printout is YES; the ASID prompt is equal to the ELAN connected to the Symposium Call Center Server; the IVR prompt is YES; the TRDN prompt is set to an appropriate value if this treatment DN is being used as a default in your system.

Example

```
ld 23
ACD000
MEM AVAIL: (U/P): 3591770    USED: 405925    TOT: 3997695
DISK RECS AVAIL: 2682
ACD DNS  AVAIL: 23758    USED:   242    TOT: 24000
REQ  prt
TYPE acd
CUST 0
ACDN 7725

TYPE ACD
CUST 0
ACDN 7725
MWC  YES
IMS  YES
CMS  YES
IMA  YES
IVMS YES
EES  NO
VSID 7
MAXP 48
SDNB NO
BSCW NO
AACQ YES
ASID 16
SFNB 1  2  3  4  5  6  9  10  11  12  13  15  16  17  18  19
USFB 1  2  3  4  5  6  7  9  10  11  12  13  14  15
CALB 0  1  2  3  4  5  6  7  8  9  11
ALOG YES
RGAI NO
ACAA NO
FRRT
...
CCBA NO
```

```
IVR  YES
TRDN 3600
CWNT NONE
```

Voice ports acquired by TN

- Client: Status shows Acquired Login.
- Meridian Mail: Menus
  - 5 System Status and Maintenance
  - 3 DSP Port Status
  - Status for the ports should show Idle.
- M1: LD 20; REQ TNB; TYPE SL1
- The ACQ AS prompt in the printout is TN; the ASID prompt is equal to the ELAN connected to the Symposium Call Center Server.

Example

```
ld 20
PT0000
REQ: prt
TYPE: tnb
TN   4 0 3 0
DATE
PAGE
DES

DES  MAIL
TN   004 0 03 00
TYPE SL1
CDEN DD
CUST 0
KLS  1
FDN
TGAR 1
LDN  NO
NCOS 0
...
PLEV 02
SPID NONE
AST
IAPG 0
AACS YES
ACQ  AS: TN
```

```
ASID 16
SFNB  1  2  3  4  5  6  9  10  11  12  13  15  16  17  18  19
SFRB
USFB  1  2  3  4  5  6  7  9  10  11  12  13  14  15
CALB  0  1  2  3  4  5  6  7  8  9  11
FCTB
ITNA NO
DGRP
PRI  01
MLWU_LANG 0
DNDR 0
KEY  00 ACD 7725 0  4550
       AGN
   01 SCN 4500 0     MARP
   02 MSB
   03 NRD
   04
   05
   06 TRN
   07 AO3
   08
   09 RLS
```

Voice ports acquired by TN and channel

- Client: TNs show a unique number per port in the Channel column, and status shows Acquired Login.
- Meridian Mail: Menus
  - 5 System Status and Maintenance
  - 3 DSP Port Status
  - Status for the ports shows Active.
- M1: LD 20; REQ TNB; TYPE SL1

Example

Same as the voice ports acquired by TN.

System default Treatment DN

- Client: Default treatment DN in the IVR ACD-DN Global Settings window is set appropriately.

Treatment DN in Meridian Mail VSDN table

See page 105 for a discussion on how treatment DNs are used.

- Meridian Mail: Menus
  - 3 Voice Administration
  - 4 Voice Services Administration
  - 1 Voice Service-DN Table
- The treatment DN defined here as the appropriate service. The example below shows one defined as an Announcement (AS) service.

Example

DN	Service	Comment
3600	AS 2020	Symposium announcement
5555	AS 2021	Symposium announcement
6666	ACC	ACCESS service

- The service number in the above display must be defined to Meridian Mail as an announcement.
- Meridian Mail: Menus
  - 3 Voice Administration
  - 4 Voice Services Administration
  - 3 Announcement Definitions

Example

```
ID          Title
2020        hello
2021        default
```

Matching IVR ACD-DNs on M1, Symposium Call Center Server, and Meridian Mail

- The following appearances of ACD-DNs must match:
  - Meridian Mail Channel Allocation Table
  - Meridian 1 DN

- IVR ACD-DN acquired by the Server
- ACD-DN used in the script

- Meridian Mail: Menus
  - 5 System Status and Maintenance
  - 4 Channel Allocation Table

Example

#	CDP	TN	ACN-DN	SCN	Type	Capability
2 1	2-1-1	004-0- 03-00	7725	4500	Voice	FULL Basic ACC Class: 41

- M1: LD 22; REQ DNB

Example

```
ld 22
PT2000
REQ  prt
TYPE dnb
CUST 0
DN   7725
DATE
PAGE
DES

DN   7725
TYPE MCDN
MCID 4500  TN 004 0 03 00
MCID 4501  TN 004 0 03 01
```

- Client: IVR ACD-DN number status shows Acquired.
- Script:
  - Give Controlled Broadcast 7725
  - Play Prompt voice segment hello

Note: Ensure that you have configured an ACCESS Treatment DN in the Meridian Mail VSDN table.

Matching voice port TNs on M1, Symposium Call Center Server, and Meridian Mail

- The TN configurations belonging to the ACD-DN (see previous section) must match:
  - Meridian Mail Channel Allocation Table
  - Meridian 1 DN
  - IVR ACD-DN acquired by Symposium Call Center Server
- Meridian Mail: Menus
  - 5 System Status and Maintenance
  - 4 Channel Allocation Table

Example

#	CDP	TN	ACN-DN	SCN	Type	Capability
2 1	2-1-1	004-0- 03-00	7725	4500	Voice	FULL Basic ACC Class: 41

- M1: LD 22; REQ DNB

Example

```
ld 22
PT2000
REQ  prt
TYPE dnb
CUST 0
DN   7725
DATE
PAGE
DES

DN   7725
TYPE MCDN
MCID 4500  TN 004 0 03 00
MCID 4501  TN 004 0 03 01
```

- Client: TNs show a unique number per port in the Channel column, and status shows Acquired Login.

Matching voice port channels on the server and Meridian Mail

- Channel number (the number shown after the “Class” keyword in the Meridian Mail Channel Allocation Table) for a specific TN on Meridian Mail must match the channel number for the same TN in the client display.
- Meridian Mail: Menus
  - 5 System Status and Maintenance
  - 4 Channel Allocation Table

Example

#	CDP	TN	ACN-DN	SCN	Type	Capability
2 1	2-1-1	004-0- 03-00	7725	4500	Voice	FULL Basic ACC Class: 41

- Client: TNs show a unique number per port in the Channel column, and status shows Acquired Login.

Non-Symposium Call Center Server calls terminating on voice ports

Introduction

Calls other than those directed by the Symposium Call Center Server script can arrive at the same voice port. There are numerous ways in which to check for this. Explore those listed below, although they comprise only a partial list. Assign a new IVR ACD-DN when you set up Symposium Call Center Server.

Note: This is important only for voice ports acquired by Symposium Call Center Server as both TN and channel.

Checking for non-Symposium Call Center Server calls

- Examine Meridian 1 IDC tables for directing incoming calls to the ACD-DN owning the voice ports.
- Examine Meridian 1 trunk auto-terminate destinations for directing incoming calls to the ACD-DN owning the voice ports.
- Examine Meridian 1 ACD-DN Night Call Forward destinations for directing incoming calls to the ACD-DN owning the voice ports.
- Examine Meridian 1, set Call Forward (No Answer, Busy, and All Calls) destinations for directing incoming calls to the ACD-DN owning the voice ports.
- Examine Meridian 1 CDN Default DN destinations for directing incoming calls to the ACD-DN owning the voice ports.
- Examine CCR scripts for directing calls to the ACD-DN owning the voice ports.
- Examine Meridian Link applications for directing calls to the ACD-DN owning the voice ports.
- Examine Symposium Call Center Server scripts and look for Give IVR commands directing calls to the same ACD-DN set up for Controlled Broadcast and Voice Sessions.

Notes:

1. If you are using Meridian Mail Call Path Diagnostics (CPD), you might receive error events indicating that call arriving on voice ports are not under the control of Symposium Call Center Server.

2. Refer to Symposium Call Center Server error events to determine the cause of non-Symposium Call Center Server calls arriving at a voice port. Use the originating DN and ACD-DN of the call for this purpose.

Call routing or treatment problems

Script skips over voice processing commands

When the script executes, it ignores the voice processing commands and continues to execute after the voice processing commands.

For the Give IVR script command, ensure the following elements are in place:

- The IVR ACD-DN is configured and successfully acquired.
- The IVR ACD-DN is configured on Meridian 1 with IVR = YES.
- There are voice ports logged on and idle (check Meridian 1 and Meridian Mail) for that IVR ACD-DN.

For the Controlled Broadcast and Voice Sessions script commands, ensure the following elements are in place:

- The IVR ACD-DN is configured and acquired.
- Phonesets and voice ports are configured and acquired.
- The Meridian Mail mailbox and password are configured and match the Meridian Mail configuration (check the password by changing it to VM class of service and dial in to the mailbox directly).
- Voice segment numbers and variables are valid and recorded.
- The Meridian Mail DSP Port Status menu shows the channels in use by Symposium Call Center Server in the Active state (this means that Meridian Mail thinks they are acquired).
- Meridian 1 shows the TNs used as voice channels as logged on and idle.
- The ACCESS link is up.

Voice processing commands do not execute consistently

When using the Controlled Broadcast or Voice Sessions script commands, some callers hear the prompts, and others do not.

Ensure the following elements are in place:

- Meridian 1 and Symposium Call Center Server configurations match exactly: the TNs configured for the IVR ACD-DN on Meridian 1 should be the same as the Voice ports configured on Symposium Call Center Server. There should be no extra TNs. If extra TNs exist, then disable them on Meridian Mail, or move them to a different ACD-DN on Meridian 1 and Meridian Mail.
- Check whether non-Symposium Call Center Server calls are arriving at this IVR ACD-DN.

Callers hear silence instead of voice processing treatments

The script executes the voice processing commands, but callers hear only silence instead of the prompts and announcements specified in the script.

For the Give IVR script commands, ensure the following elements are in place:

- The announcement on Meridian Mail indicated by the treatment DN exists and is recorded.
- The voice port where the call ended up is in Idle state on Meridian Mail.

For the Controlled Broadcast and Voice Sessions script commands, ensure the following elements are in place:

- The Meridian Mail DSP Port Status menu shows all channels in use by Symposium Call Center Server in the Active state (this means that Meridian Mail thinks they are acquired).
- There are no extra TNs on the Meridian 1 side in the same ACD-DN as the channels acquired by Symposium Call Center Server.
- The IVR ACD-DN used is the correct one, and the correct ports belong to that ACD-DN on Meridian 1 and Meridian Mail.
- They ACCESS link is up.
- Voice segment numbers and variables used by the script are valid and recorded.

Callers hear a message different from the voice processing treatment specified in the script

Callers hear a message different from the one indicated by the voice processing command.

For the Give IVR script command, ensure the following elements are in place:

- The treatment and recording on Meridian Mail indicated by the treatment DN contains the correct recording.
- If the script statement does not contain a treatment DN, check the Meridian 1 default treatment DN and the treatment that is mapped to Meridian Mail's VSDN Table (and what is recorded for that).

For the Controlled Broadcast and Voice Sessions script commands, ensure that the mapping of Symposium Call Center Server's voice segment variables (the file names and segment numbers in their values) to those recorded on Meridian Mail is correct (you can use the Voice Prompt Editor to play the voice segments).

Callers hear only one of multiple voice processing treatments specified in the script

At first, callers hear the correct voice processing treatment. Then, after some traffic, the script seems to skip the script statements and no longer executes the voice processing commands.

For the Give IVR script command, ensure that no ACCESS application (Symposium Call Center Server or otherwise) is trying to acquire the TNs (on Meridian Mail) used as voice ports by Symposium Call Center Server.

For the Controlled Broadcast and Voice Sessions script commands, ensure that no non-ACCESS calls are ending up at the voice ports used by Symposium Call Center Server. These can come either from a different script using the Give IVR script statement to the same IVR ACD-DN, or via a non-Symposium Call Center Server call routing (trunk auto-termination, call transfer scenarios, call forward scenarios, Meridian Link, or CCR applications).

Calls suspend in the script at the voice processing statement

Scripts advance to the voice processing commands, but never continue beyond that. They remain there until the caller hangs up, or an agent (if the call was previously queued) answers the call.

For the Give IVR script command, ensure that the TNs used as voice ports are acquired by Symposium Call Center Server.

Callers hear broadcast announcement too many times

The script statement specifies a Broadcast Announcement to be repeated x number of times, and the caller hears it between x and 2x times.

For the Controlled Broadcast (continuous mode) script command, calls are connected as soon as they arrive. Callers continue to listen to the announcement until *one full cycle* of the message is played. With small periods of silence at the start and end of the announcement, as well as the small time frame between the time that a call is connected to a port and when it starts hearing the announcement, the application software calculates that the call has *not* heard the announcement all the way through on the first cycle. The human ear, however, might sense that it has heard it. The software plays another cycle of the message before continuing the script.

If the script commands specify multiple repeats of the announcement (for example, four repeats), then the call can hear up to eight messages (or partial messages) as each script command imposes the calculation referred to above.

Callers wait too long to hear voice processing

The caller hears too many cycles of ringback before the message is played.

For the Give IVR and Voice Sessions script commands, ensure that there are enough voice ports available for the traffic.

For the Controlled Broadcast (start/stop mode), ensure that the Broadcast Wait Timer setting on the Server is not too long; the default setting is ten seconds. When a single call is made, the software waits ten seconds before connecting the call to a voice port for the announcement.

Callers hear “Your voice session cannot be completed” message

The caller hears the message “Your Voice Session cannot be completed” instead of the prompts specified in the script.

Ensure the following elements are in place:

- The ACCESS link is up.
- The Mailbox specified on Symposium Call Center Server is the correct one.
- The Mailbox password on Symposium Call Center Server is correct (and has not expired).
- The voice segment being requested exists (recorded).
- The voice ports are acquired correctly (and *all* of the ports in the IVR ACD-DN are acquired).
- The treatment DN in the Controlled Broadcast or Voice Sessions script statements maps to an ACCESS DN in Meridian Mail’s VSDN Table.

Caller hears voice prompts but is never presented to an agent

This occurs if a transfer fails after a Give IVR command to a third-party voice processing system.

If the Give IVR script command is used with a third-party IVR application or a Meridian Mail voice menu to transfer a call to a DN that is busy, Symposium Call Center Server loses control of the call. When using Give IVR, always transfer the call to an ACD-DN.

Channel number field in the voice port window appears dimmed

The configuration guidelines say that the voice ports must be acquired by the TN, and they are currently acquired by the TN and the channel. When you double-click to delete the channel number, you cannot delete it because the field appears dimmed. The same problem applies when voice ports are acquired by a TN, and they must be changed to be acquired by the TN and channel.

You cannot change the acquire status (TN or TN-and-channel) while the port is acquired. If you must change the configuration, then first deacquire the port, double-click on it to change the configuration (now the field is open to be edited), and then reacquire the port.

Network call-by-call reports display inaccurate time stamps

In a call center with multiple sites using the NSBR feature, all Meridian 1 switches are synchronized with identical time stamps. If the network call-by-call reports from your site indicate a reporting inaccuracy, the time stamp for your local Meridian 1 switch could be out of synchronization with the Meridian 1 switches at other sites.

Check the time stamp of your local Meridian 1 at the beginning of each shift. This ensures that network call-by-call reports reflect the accurate time.

To ensure the time stamp is accurate

1. Log on to the Meridian 1 console.
2. Access a Command Prompt and type **id 2**, and then press the Enter key.
3. Type **ttad** at the prompt, and then press the Enter key.

   Result: The current date and time appear in the format DD-MM-YYYY 00:00.
4. Type **stad**.
5. Enter the correct date and time.
6. Log off the Meridian 1 console.

Symposium Call Center Server and the Meridian 1 are not able to communicate

If Symposium Call Center Server and the Meridian 1 switch are not exchanging information, the network traffic volume over the ELAN might be excessive. You must ensure that the network is designed with an appropriate number of devices installed on the ELAN. For information about proper network design, refer to the *Planning and Engineering Guide*.

Frequently asked questions

Introduction

The following is a list of frequently asked questions about voice processing operation.

What is the difference between the Play Prompt command on the Controlled Broadcast and the Open/End Voice Sessions commands?

There are three differences:

- The Controlled Broadcast Play Prompt has a *continuous* option, which is not applicable to the Open/End Voice Session statement.
- The Open/End Voice Session Play Prompt has a *no-type-ahead* option, which is not applicable to the Controlled Broadcast statement.
- The start/stop timer for starting the announcement applies only to Broadcast operation. When the Open/End Voice Session statement is used, every call is connected to its own voice port and the message starts immediately.

Other than the above exceptions, the statements' syntax are the same and they operate in the same way.

Which voice processing commands use Treatment DNs, and what are they used for?

See "Configuring VSDN entries (treatment DNs)" on page 105, which explains how treatment DNs are used and determined.

What is the difference between "front-ending an IVR to Symposium" and "Symposium Voice Processing"?

- Front-ending an IVR means incoming calls from the trunks are directly terminated on a voice processing engine. Based on the caller's choice from the menus, the call can be transferred to a CDN that enters Symposium Call Center Server. This operation is largely transparent to Symposium Call Center Server.

- When you use Symposium Call Center Server voice processing, the calls from the trunks enter a CDN first. Then, as part of the script processing, the call is directed to a voice processing engine. The server is in full control of this interaction.

When do you need channels set up? What are channels?

- Channels refer to a "voice channel" as seen from an ACCESS application's point of view—a voice resource through which what a voice call hears can be controlled—similar to the concept of a speech path on the M1.
- On Meridian Mail, a channel is a port attribute (a TN configured with ACCESS class of service and a Class number assigned to it). On Meridian 1, a channel is a voice port TN.
- Meridian Mail voice ports configured as channels, and Symposium Call Center Server having acquired these ports with channel numbers assigned, are required for the Controlled Broadcast and the Voice Session script commands to work correctly.

Why is the Default operation of Give IVR non-interruptible, while the default for Controlled Broadcast and Voice Sessions is interruptible?

- Interruptible operation as the default makes the most sense for all of the voice processing commands. However, it was decided to keep Give IVR's operation the same as it is in CCR to minimize confusion to customers who are familiar with its operation.

What are the Default IVR ACD-DN and Default Treatment DNs used for? What should they be set to?

See Chapter 4, "Meridian 1 subsystem configuration" which explains how IVR ACD-DN and Treatment DNs are used.

Can Meridian Link and CCR use the same Meridian Mail system as Symposium Call Center Server?

Yes, they can. However, they cannot share voice ports or IVR ACD queues with the server.

What is the difference between the *number* and *numberbydigit* clauses on the Play Prompt script statement?

- The *numberbydigit* clause speaks each number separately (for example, a "Play Prompt numberbydigit 1234" statement reads: one-two-three-four).
- The *number* clause speaks the number as a single entity (for example, a "Play Prompt number 1234" statement reads: one-thousand-two-hundred-thirty-four).

Chapter 9

Agent phoneset keys

In this appendix

Overview

Introduction

This appendix describes how the Display Waiting Calls (DWC) key and lamp function on Symposium Call Center Server. This appendix also lists the agent phoneset keys that Symposium Call Center Server does not support.

The Display Waiting Calls key/lamp

Introduction

Symposium Call Center Server supports the Display Waiting Calls (DWC) key. This feature displays skillset information when the DWC key on an agent's phoneset is pressed.

ATTENTION

The information displayed is different from the DWC feature used in the Meridian 1 ACD environment.

Agent phoneset display

The following information displays on a Symposium Call Center Server agent's phoneset (the "DWC agent") when the agent presses the DWC key:

AAA BBB CCC, where

- AAA is the sum of the numbers of calls waiting in each skillset that the DWC agent is currently logged on to. A call is counted more than once if it is queued to more than one of the skillsets that the DWC agent is logged on to.

Note: If a call is queued to a specific Agent ID (using the Queue to Agent statement in the Symposium Call Center Server scripts), it is not included in the number of calls waiting for the DWC agent. Only calls waiting in the skillsets that the DWC agent is logged on to are reflected.

- BBB is the sum of the number of agents logged on to each skillset that the DWC agent is currently logged on to. An agent is counted more than once if logged on to more than one of the skillsets that the DWC agent is logged on to.
- CCC is the waiting time, in seconds, of the oldest call in all of the skillsets that the DWC agent is logged on to.

Supervisor phoneset display

The DWC key and associated lamp configured on a supervisor's phoneset do not support the display of any Symposium Call Center Server skillset information. If you press the DWC key on a supervisor's phoneset, it shows ACD queue information for that supervisor, as it currently does. The lamp also responds to ACD queue loading and activity for that supervisor, as determined by the Meridian 1 configuration. Calls are not normally queued to ACD queues for Symposium Call Center Server; therefore, the primary uses of this feature for Symposium Call Center Server supervisors are when the call center is handling Network ACD calls or operating in default mode, and the Meridian 1 ACD features are routing the calls.

Skillset information

Skillset information display is only available on phonesets that have numeric display capabilities. Phonesets without numeric displays cannot get skillset information by any other means (for example, no audible tones).

Display format

The information displays with spaces between the fields. Three digits display data for the smallest phoneset display type of 1×12. For phoneset displays larger than 1×12, four digits display the data. The maximum displayable number of calls in queue is 9999 and the maximum number of agents that Symposium Call Center Server currently supports is 40. The maximum displayable amount of time that a call can be in queue is 9999 seconds or 2.78 hours. The following table summarizes the display types and field width for phonesets that display DWC key information.

DWC key phoneset display type and field width

Display type	AAA	BBB	CCC
1 × 12	3 digits	3 digits	3 digits
1 × 16	4 digits	4 digits	4 digits
1 × 40	4 digits	4 digits	4 digits
2 × 24	4 digits	4 digits	4 digits

Sample phoneset displays

The displays illustrated in this section indicate the lengths and positions of the various fields for each supported display configuration.

Notes:

- No more than four digits display per field.
- “n” illustrates the full width of a field.
- Leading zeros display as blanks.

1 × 12 character displays

	1	2	3	4	5	6	7	8	9	10	11	12
1	n	2	3		n	1	7		1	6	5	

1 × 16 character displays

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
1	n	n	2	3		n	n	1	7		n	1	6	5		

1 × 40 character displays

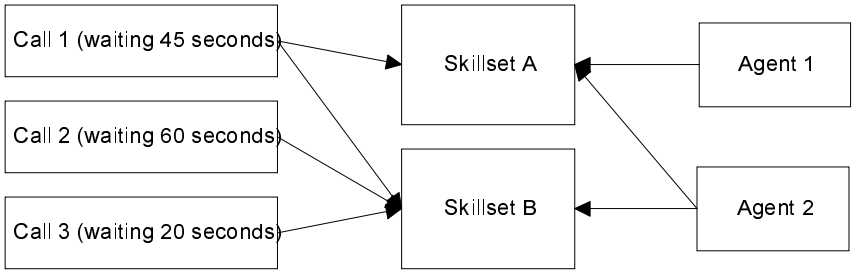
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	→	39	40
1	n	n	2	3		n	n	1	7		n	1	6	5			→		

2 × 24 character displays

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	→	24
1	W	A	I	T	I	N	G		M	A	N	D		L	W	A	I	T	→	
2				n	n	2	3		n	n	1	7			n	1	6	5	→	

DWC examples for agent phonesets

Consider the following diagram with two agents logged on to two skillsets. Three calls are queued to the two skillsets.



The following display results when Agent 1 presses the DWC key:

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	…	24
1	W	A	I	T	I	N	G		M	A	N	D		L	W	A	I	T	…	
2							1					2					4	5	…	

The following display results when Agent 2 presses the DWC key:

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	…	24
1	W	A	I	T	I	N	G		M	A	N	D		L	W	A	I	T	…	
2							4					3					6	0	…	

DWC key lamp

The DWC key lamp on a Symposium Call Center Server agent phoneset does not respond to calls in skillsets; it always remains dark as far as skillset loading and activity are concerned. However, the lamp continues to respond to the call loading and activity in any ACD queues that the agent is logged on to, as determined by the configuration on the Meridian 1. Calls are not normally queued to ACD queues for Symposium Call Center Server; therefore, the primary use of this feature for agents is when the call center is handling Network ACD calls or operating in default mode, and the Meridian 1 ACD features are routing the calls. When the agent presses the DWC key, the agent phoneset display shows Symposium Call Center Server skillset information as detailed in the previous section.

ACD Waiting Calls key/lamp

The ACD Waiting Calls (AWC) key/lamp is not supported in Symposium Call Center Server to indicate skillset information. Any AWC key/lamp defined on an agent's or supervisor's phoneset indicates information on the ACD-DN for the phoneset, as configured on the Meridian 1.

Unsupported agent phoneset keys

Symposium Call Center Server does not support the following keys or report on them:

- Hotline
- Private line
- Voice call
- Dial Intercom

Glossary

A

accelerator key

A key on a phoneset that an agent can use to place a call quickly. When an agent presses an accelerator key, the system places the call to the configured number associated with the key. For example, if an agent presses the Emergency key, the system places a call to the agent's supervisor.

access class

A collection of access levels that defines the actions a member of the access class can perform within the system. For example, a member of the Administrator access class might be given a collection of Read/Write access levels.

access level

A level of access or permission given to a particular user for a particular application or function. For example, a user might be given View Only access to historical reports.

ACCESS link

A communication channel between the Symposium Call Center Server and Meridian Mail.

ACCESS voice port

A Meridian Mail voice port that is controlled by the ACCESS link.

ACD call

See Automatic call distribution call.

ACD-DN

See Automatic call distribution directory number.

ACD routing table

See Automatic call distribution routing table.

acquired resource
A resource configured on the switch that is under the control of the Symposium Call Center Server. Resources must be configured with matching values on both the switch and the Symposium Call Center Server.

activated script
A script that is processing calls or is ready to process calls. Before you can activate a script, you must first validate it.

activity code
A number that an agent enters on his or her phoneset during a call. Activity codes provide a way of tracking the time agents spend on various types of incoming calls. For example, the activity code 720 might be used to track sales calls. Agents can then enter 720 on their phonesets during sales calls, and this information can be generated in an Activity Code report.

administrator
A user who is responsible for maintaining the Symposium Call Center Server.

agent
A user who is responsible for handling customer calls.

agent login ID
A unique identification number assigned to a particular agent. The agent uses this number when logging on. The agent ID is not associated with any particular phoneset.

agent to skillset assignment
A matrix that, when you run it, sets the priority of one or more agents for a skillset. Agent to skillset assignments can be scheduled.

agent to supervisor assignment
A definition that, when you run it, assigns one or more agents to specific supervisors. Agent to supervisor assignments can be scheduled.

application
1. A logical entity that represents a Symposium Call Center Server script for reporting purposes. The master script and each primary script have an associated application.The application has the same name as the script it represents. 2. A program that runs on a computer.

application program interface
A set of routines, protocols, and tools that programmers use to develop software applications. APIs simplify the development process by providing commonly used programming procedures.

associated supervisor
A supervisor who is available for an agent if the agent's reporting supervisor is unavailable. *See also* reporting supervisor.

Automatic call distribution call
A call to an ACD-DN. ACD calls are distributed to agents in an ACD group based on the ACD routing table on the switch.

Automatic call distribution directory number
DNs associated with an ACD group. Calls made to these DNs are distributed to agents belonging to the group, based on the ACD routing table on the switch.

Automatic call distribution routing table
A table configured on the switch that contains a list of ACD-DNs used to define routes for incoming calls. This ensures that incoming calls not processed by Symposium Call Center Server will be queued to ACD groups and handled by available agents.

C

call age
The amount of time a call was waiting in the system before being answered by an agent.

call destination
The site to which an outgoing network call is sent. *See also* call source.

call intrinsic

A script element that stores call-related information assigned when a call enters the Symposium Call Center Server. *See also* intrinsic, skillset intrinsic, time intrinsic, and traffic intrinsic.

call presentation class

A collection of preferences that determines how calls are presented to an agent. A call presentation class specifies whether a break time between calls is allowed, whether an agent can put DN calls on hold for incoming ACD calls, and whether an agent phoneset displays that the agent is reserved for a network call.

call priority

A numerical value assigned in a script that defines the relative importance of a call. If two calls are in the queue when an agent becomes available, and one call is queued with a higher priority than the other, the agent receives the higher priority call first. *See also* skillset priority.

call source

The site from which an incoming network call originates. *See also* call destination.

call treatment

A script element that enables you to provide handling to a call while it is waiting to be answered by a call center agent. For example, a caller can hear a recorded announcement or music while waiting for an agent.

call variable

A script variable that applies to a specific call. A call variable follows the call through the system and is passed from one script to another with the call. *See also* global variable, variable.

Calling Line Identification

This is an optional service that identifies the telephone number of the caller. This information can then be used to route the call to the appropriate agent or skillset. The CLID can also be displayed on an agent’s phoneset.

CDN

See controlled directory number.

CLAN
See Customer local area network.

CLID
See Calling Line Identification.

client
The part of Symposium Call Center Server that runs on a personal computer or workstation and relies on the server to perform some operations. *See also* server.

command
A building block used with expressions, variables, and intrinsics to create scripts. Commands perform distinct functions, such as routing a call to a specific destination, playing music to a caller, or disconnecting a caller.

controlled directory number
A special directory number that allows calls arriving at the switch to be queued when the CDN is controlled by an application such as Symposium Call Center Server. When a call arrives at this number, the switch notifies the application and waits for routing instructions, which are performed by scripts in Symposium Call Center Server.

Customer local area network
The LAN to which your corporate services and resources connect. The Symposium Call Center Server and client both connect to the CLAN. Third-party applications that interface with the server also connect to this LAN.

D

DBMS
Database Management System

deactivated script
A script that does not process any new calls. If a script is in use when it is deactivated, calls continue to be processed by the script until they are completed.

default activity code

The activity code that is assigned to a call if an agent does not enter an activity code manually, or when an agent presses the activity code button twice on his or her phoneset. Each skillset has a defined default activity code.

default skillset

The skillset to which calls are queued if they have not been queued to a skillset or a specific agent by the end of a script.

desktop user

A configured user who can log on to the Symposium Call Center Server from a client PC.

destination site

The site to which an outgoing network call is sent. *See also* source site.

DHCP

See dynamic host configuration protocol.

Dial-Up Networking

See Remote Access Services.

Dialed Number Identification Service

An optional service that allows Symposium Call Center Server to identify the phone number dialed by the incoming caller.

An agent can receive calls from customers calling in on different DNISs and, if the DNIS is displayed on the phoneset, can prepare a response according to the DNIS.

directory number

The number that identifies a phoneset on a switch. The directory number (DN) can be a local extension (local DN), a public network telephone number, or an automatic call distribution directory number (ACD-DN).

directory number call

A call that is presented to the DN key on an agent’s phoneset.

display threshold
A threshold used in real-time displays to highlight a value below or above the normal range.

DN
See directory number.

DN call
See directory number call.

DNIS
See Dialed Number Identification Service.

dynamic host configuration protocol
A protocol for dynamically assigning IP addresses to devices on a network.

dynamic link library
A library of executable functions or data that can be used by a Windows application. Typically, a DLL provides one or more particular functions and a program accesses the functions by creating either a static or dynamic link to the DLL. A DLL can be used by several applications at the same time.

E

ELAN
See embedded local area network.

embedded local area network
A dedicated Ethernet TCP/IP LAN that connects the Symposium Call Center Server and the switch.

Emergency key
A key on an agent's phoneset that, when pressed by the agent, automatically calls his or her supervisor to notify the supervisor of a problem with a caller.

event

1. An occurrence or action on the Symposium Call Center Server, such as the sending or receiving of a message, the opening or closing of an application, or the reporting of an error. Some events are for information only, while others can indicate a problem. Events are categorized by severity: information, minor, major, and critical. 2. An action generated by a script command, such as queuing a call to a skillset or playing music.

expression

A building block used in scripts to test for conditions, perform calculations, or compare values within scripts. *See also* logical expression, mathematical expression, and relational expression.

F

filter timer

The length of time after the system unsuccessfully attempts to route calls to a destination site, before that site is filtered out of a routing table.

first-level threshold

The value that represents the lowest value of the normal range for a statistic in a threshold class. The system tracks how often the value for the statistic falls outside this value.

G

global settings

Settings that apply to all skillsets or IVR ACD-DNs that are configured on your system.

global variable

A variable that contains values that can be used by any script on the system. The value of a global variable can only be changed in the Script Variable Properties sheet. It cannot be changed in a script. *See also* call variable, variable.

I

Incalls key

The key on an agent phoneset to which incoming ACD and Symposium Call Center Server calls are presented.

Interactive voice response

An application that allows telephone callers to interact with a host computer using prerecorded messages and prompts.

Interactive voice response ACD-DN

A directory number that routes a caller to a specific IVR application. An IVR ACD-DN must be acquired for non-integrated IVR systems.

Interactive voice response event

A voice port login or logout. An IVR event is pegged in the database when a call acquires or deacquires a voice port.

Internet Protocol address

An identifier for a computer or device on a TCP/IP network. Networks use the TCP/IP protocol to route messages based on the IP address of the destination. For customers using NSBR, site IP addresses must be unique and correct. The format of an IP address is a 32-bit numeric address written as four values separated by periods. Each value can be 0-255. For example, 1.160.10.240 could be an IP address.

intrinsic

A word or phrase used in a script to gain access to system information about skillsets, agents, time, and call traffic that can then be used in formulas and decision-making statements. *See also* call intrinsic, skillset intrinsic, time intrinsic, and traffic intrinsic.

IP address

See Internet Protocol address.

IVR

See Interactive voice response.

IVR ACD-DN

See Interactive voice response ACD-DN.

IVR event

See Interactive voice response event.

IVR port
See voice port.

L

LAN
See Local area network.

Local area network
A computer network that spans a relatively small area. Most LANs connect workstations and personal computers and are confined to a single building or group of buildings.

local call
A call that originates at the local site. *See also* network call.

local skillset
A skillset that can be used at the local site only. *See also* network skillset, skillset.

logical expression
A symbol used in scripts to test for different conditions. Logical expressions are AND, OR, and NOT. *See also* expression, mathematical expression, and relational expression.

M

M1
Meridian 1 switch

master script
The first script executed when a call arrives at the Symposium Call Center Server. A default master script is provided with Symposium Call Center Server, but it can be customized by an authorized user. It can be deactivated but not deleted. *See also* network script, primary script, script, and secondary script.

mathematical expression
An expression used in scripts to add, subtract, multiply, and divide values. Mathematical expressions are addition (+), subtraction (-), division (/), and multiplication (*). *See also* expression, logical expression, and relational expression.

Meridian Link Services
A communications facility that provides an interface between the switch and a third-party host application.

Meridian Mail
A Nortel Networks product that provides voice messaging and other voice and fax services.

Meridian MAX
A Nortel Networks product that provides call processing based on ACD routing.

MLS
See Meridian Link Services.

MM
See Meridian Mail.

music route
A resource installed on the switch that provides music to callers while they wait for an agent.

N

NACD call
A call that arrives at the server from a network ACD-DN.

NCC
See Network Control Center.

network call
A call that originates at another site in the network. *See also* local call.

Network Control Center

The server on a Symposium Call Center Server system where NSBR is configured and where communication between servers is managed.

network script

The script that is executed to handle error conditions for Symposium Call Center Server calls forwarded from one site to another, for customers using NSBR. The network script is a system-defined script provided with Symposium Call Center Server, but it can be customized by an authorized user. It can be deactivated but not deleted. *See also* master script, primary script, script, and secondary script.

Network Skill-Based Routing

An optional feature with Symposium Call Center Server that provides skill-based routing to multiple networked sites.

network skillset

A skillset that is common to every site on the network. Network skillsets must be created at the Network Control Center (NCC).

night mode

A skillset state in which the server does not queue incoming calls to the skillset, and in which all queued calls are given night treatment. A skillset goes into night mode automatically when the last agent logs off, or the administrator can put it into night mode manually. *See also* out-of-service mode, transition mode.

NPA

See Number Plan Area.

NSBR

See Network Skill-Based Routing.

Number Plan Area

Area code

O

object linking and embedding

A compound document standard that enables you to create objects with one application and then link or embed them in a second application.

ODBC
See Open Database Connectivity.

OEM
Original equipment manufacturer

OLE
See object linking and embedding.

Open Database Connectivity
A Microsoft-defined database application program interface (API) standard.

out-of-service mode
A skillset state in which the skillset does not take calls. A skillset is out of service if there are no agents logged on or if the supervisor puts the skillset into out-of-service mode manually. *See also* night mode, transition mode.

out-of-service skillset
A skillset that is not taking any new calls. While a skillset is out of service, incoming calls cannot be queued to the skillset. *See also* local skillset, network skillset, and skillset.

P

PBX
See private branch exchange.

pegging
The action of incrementing statistical counters to track and report on system events.

pegging threshold
A threshold used to define a cut-off value for statistics such as short call and service level. Pegging thresholds are used in reports.

PEP
See Performance Enhancement Package.

Performance Enhancement Package

A Symposium Call Center Server supplementary software application that enhances the functionality of previously released software by improving performance, adding functionality, or correcting a problem discovered since the original release.

personal directory number

A DN on which an agent can be reached directly, usually for private calls.

phoneset

The physical device, connected to the switch, to which calls are presented. Each agent and supervisor must have a phoneset.

phoneset display

The display area on an agent's phoneset where information about incoming calls can be communicated.

Position ID

A unique identifier for a phoneset, which is used by the switch to route calls to the phoneset.

primary script

A script that is executed or referenced by the master script. A primary script can route calls to skillsets, or it can transfer routing control to a secondary script. *See also* master script, network script, script, and secondary script.

private branch exchange

A telephone switch, typically used by a business to service its internal telephone needs. A PBX usually offers more advanced features than are generally available on the public network.

R

RAN

recorded announcement

RAN route

See recorded announcement route.

RAS
See Remote Access Services.

recorded announcement route
A resource installed on the switch that offers a recorded announcement to callers.

relational expression
An expression used in scripts to test for different conditions. Relational expressions are less than (<), greater than (>), less than or equal to (< =), greater than or equal to (> =), and not equal to (< >). *See also* expression, logical expression, and mathematical expression.

Remote Access Services
A feature built into Windows NT and Windows 95 that enables users to log on to an NT-based LAN using a modem, X.25 connection, or WAN link. This feature is also known as Dial-Up Networking.

reporting supervisor
The supervisor who has primary responsibility for an agent. When an agent presses the Emergency key on the phoneset, the emergency call is presented to the agent's reporting supervisor. *See also* associated supervisor.

round robin routing table
A routing table that queues the first call to the first three sites in the routing table, then the second three sites, then the third three sites, and so on, until an agent is reserved at one of the sites. *See also* sequential routing table.

route
A group of trunks. Each trunk carries either incoming or outgoing calls to the switch. *See also* music route, RAN route.

routing table
A table that defines how calls are routed to the sites on the network. *See also* round robin routing table, sequential routing table.

S

sample script
A script that is installed with the Symposium Call Center Server client. Sample scripts are stored as text files in a special folder on the client. The contents of these scripts can be imported or copied into user scripts to create scripts for typical call center scenarios.

SCM
See Service Control Manager.

script
A set of instructions that relates to a particular type of call, caller, or set of conditions, such as time of day or day of week. *See also* master script, network script, primary script, and secondary script.

script variable
See variable.

second-level threshold
The value used in display thresholds that represents the highest value of the normal range for a given statistic. The system tracks how often the value for the statistic falls outside this value.

secondary script
Any script (other than a master, network, or primary script) that is referenced from a primary script or any other secondary script. There is no pegging of statistics for actions occurring during a secondary script. *See also* master script, network script, primary script, and script.

sequential routing table
A routing table method that always queues a call to the first three active sites in the routing table. *See also* round robin routing table.

server
A computer or device on a network that manages network resources. Examples of servers include file servers, print servers, network servers, and database servers. The Symposium Call Center Server is used to configure the operations of the call center. *See also* client.

service
A process that adheres to a Windows NT structure and requirements. A service provides system functionality.

Service Control Manager
A Windows NT process that manages the different services on the PC.

service level
The percentage of incoming calls answered within a configured number of seconds.

service level threshold
A parameter that defines the number of seconds within which incoming calls should be answered.

Simple Network Management Protocol
A set of protocols for managing complex networks. SNMP works by sending messages, called protocol data units (PDUs), to different parts of a network and then analyzing the responses.

site
1. A system using Symposium Call Center Server that can be accessed using SMI. 2. A system using Symposium Call Center Server and participating in Network Skill-Based Routing.

skillset
A group of capabilities or knowledge required to answer a specific type of call. *See also* local skillset, network skillset.

skillset intrinsic
A script element that inserts information about a skillset in a script. Skillset intrinsics return values such as skillsets, integers, and agent IDs. These values are then used in queuing commands. *See also* call intrinsic, intrinsic, time intrinsic, and traffic intrinsic.

skillset priority
An attribute of a skillset assignment that determines the order in which calls from different skillsets are presented to an agent. When an agent becomes available, calls might be waiting for several of the skillsets to which the agent belongs. The server presents the call queued for the skillset for which the agent has the highest priority.

source site
The site from which an incoming network call originates. *See also* destination site.

standby
In skillset assignments, a property that grants an agent membership in a skillset, but makes the agent inactive for that skillset.

supervisor
A user who manages a group of agents. *See also* associated supervisor, reporting supervisor.

switch
The hardware that receives incoming calls and routes them to their destination.

switch resource
A device that is configured on the switch. For example, a CDN is configured on the switch, and then is used as a resource with Symposium Call Center Server. *See also* acquired resource.

Symposium Call Center Server call
A call to a CDN that is controlled by the Symposium Call Center Server. The call is presented to the Incalls key on an agent's phoneset.

system-defined scripts
The Master_Script and the Network_Script (if NSBR is enabled).These scripts can be customized or deactivated by a user, but cannot be deleted. These scripts are the first scripts executed for every local or network call arriving at the call center.

T

target site
See destination site.

TCP/IP
See Transport Control Protocol/Internet Protocol.

telephony
The science of translating sound into electrical signals, transmitting them, and then converting them back to sound. The term is used frequently to refer to computer hardware and software that perform functions traditionally performed by telephone equipment.

threshold
A value for a statistic at which system handling of the statistic changes.

threshold class
A set of options that specifies how statistics are treated in reports and real-time displays. *See also* display threshold, pegging threshold.

time intrinsic
A script element that stores information about system time, including time of day, day of week, and week of year. *See also* call intrinsic, intrinsic, skillset intrinsic, and traffic intrinsic.

Token Ring
A PC network protocol developed by IBM. A Token Ring network is a type of computer network in which computers are arranged schematically in a circle.

traffic intrinsic
An intrinsic that inserts information about system-level traffic in a script. *See also* call intrinsic, intrinsic, skillset intrinsic, and time intrinsic.

transition mode
A skillset state in which the server presents already queued calls to a skillset. New calls queued to the skillset are given out-of-service treatment. *See also* night mode, out-of-service mode.

Transport Control Protocol/Internet Protocol

The communication protocol used to connect devices on the Internet. TCP/IP is the standard protocol for transmitting data over networks.

treatment

See call treatment.

trunk

A communications link between a PBX and the public central office, or between PBXs. Various trunk types provide services such as Direct Inward Dialing (DID trunks), ISDN, and Central Office connectivity.

U

user-created script

A script that is created by an authorized user on the Symposium Call Center Server system. Primary and secondary scripts are user-created scripts.

user-defined script

A script that is modified by an authorized user on the Symposium Call Center Server system.

utility

A program that performs a specific task, usually related to managing system resources. Operating systems contain a number of utilities for managing disk drives, printers, and other devices.

V

validation

The process of checking a script to ensure that all the syntax and semantics are correct. A script must be validated before it can be activated.

variable

A placeholder for values calculated within a script, such as CLID. Variables are defined in the Script Variable Properties sheet and can be used in multiple scripts to determine treatment and routing of calls entering the Symposium Call Center Server. *See also* call variable, global variable.

voice port

A connection from a telephony port on the switch to a port on the IVR system.

W

WAN

See Wide area network.

Wide area network

A computer network that spans a relatively large geographical area. Typically, a WAN consists of two or more local area networks (LANs). The largest WAN in existence is the Internet.

workload scenarios

Sets of configuration values defined for typical patterns of system operations. Five typical workload scenarios (entry, small, medium, large, and upper end) are used in the Capacity Assessment Tool for capacity analysis for the Symposium Call Center Server.

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Nortel Networks Symposium Call Center Server Product release 3.0
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