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Nortel Networks Symposium Call Center Server

Symposium, M1/CSE 1000, and Voice Processing Guide

Product release 4.2

Standard 2.0

January 2003

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Contents

1	Getting started	11
	Overview	12
	Skills you need	13
	What's new in this version	15
	Subsystems overview	16
	Communication between Symposium Call Center Server subsystems	17
2	Configuration overview	25
	Overview	26
	Subsystem configuration reference information	27
	Recommended sequence of configuration	29
	Configuration checklists	31
3	Subsystem connections configuration	51
	Overview	52
	Configuring the ELAN	53
	Configuring the ACCESS link	56
4	Switch subsystem configuration	61
	Overview	62
	Configuring CDNs on the switch	64
	Changing CDNs on the switch	67
	Configuring NACD-DNs to enable the QUEUE to NACD command.	68
	Configuring IVR ACD-DNs on the switch.	71
	Configuring voice ports on the switch	75
	Configuring agent and supervisor TNs on the switch.	78
	Configuring routes on the switch	83
	Configuring Multiple Queue Assignments.	86
	Initializing or sys-loading the switch	87

5	Meridian Link Services configuration	89
	Overview	90
	Installing and configuring Meridian Link Services	92
	Allowing CTI operations on the ELAN	93
	Configuring phonesets for CTI	94
	Configuring CDNs for host-enhanced routing	97
	Connecting the host application	98
6	Voice processing subsystem configuration	99
	Overview	100
	Section A: Voice processing	101
	Overview of voice processing	102
	Typical uses of voice processing commands	105
	Operation modes for voice processing commands	107
	Resource acquisition	110
	Voice port partitioning rules	111
	Section B: CallPilot	113
	Overview	114
	Updating the CallPilot configuration	116
	Updating the SDN table	117
	Creating voice segments	118
	Section C: Meridian Mail	119
	Overview	120
	Channel Allocation Table	122
	Voice Services DN table	123
	Configuring Meridian Mail for ACCESS	125
	Configuring Meridian Mail for Give IVR	132
	Section D: Third-party voice processing systems	139
	Overview of third-party voice processing systems	140
7	Symposium Call Center Server subsystem configuration	143
	Overview	144
	Voice connection	145
	Configuring CDNs on the server	150
	Configuring IVR ACD-DNs on the server	151
	Configuring agent phonesets on the server	152
	Configuring voice ports on the server	153

	Configuring IVR ACD-DN global settings	154
	Defining voice segment variables.	158
	Defining scripts.	160
	Testing voice services integration when using CallPilot	162
8	Troubleshooting	187
	Overview.	188
	Running Meridian Link traces	189
	Subsystem link problems	191
	Ports become deaquired	206
	Call routing or treatment problems.	208
	Frequently asked questions.	215
9	Agent phoneset keys	219
	Overview.	220
	Display Waiting Calls key/lamp.	221
A	Migrating from Meridian Mail to CallPilot	227
	Overview.	228
	Section A: Migrating from Meridian Mail to CallPilot	231
	Overview of migration	232
	Pre-migration checklist.	233
	Migration checklist.	238
	Migration procedure	240
	Section B: Backing out of the migration	263
	Overview of backout.	264
	Backout procedure	266
	Section C: Comparison of VPE and Application Builder	275
	Application Builder compared to Voice Prompt Editor	276
B	Migrating prompts and segments from Meridian Mail to CallPilot	279
	Overview.	280
	Understanding the migration process	281
	Performing the Symposium Voice Services migration.	285
	Post migration activities	299

Glossary	303
Index	325

Chapter 1

Getting started

In this chapter

Overview	12
Skills you need	13
What's new in this version	15
Subsystems overview	16
Communication between Symposium Call Center Server subsystems	17

Overview

Introduction

The *Nortel Networks Symposium Call Center Server Symposium, M1/CSE 1000, and Voice Processing Guide* provides descriptive information and instructions on how to set up and configure the following components for use with Symposium Call Center Server:

- Meridian 1 or Succession Communication Server for Enterprise (CSE) 1000 switches
Note: Unless otherwise specified, references in this guide to the Meridian 1 switch are also applicable to the Meridian 1 Internet Enabled switch.
- Meridian Link Services
- CallPilot, Meridian Mail, or a third-party voice processing system
Note: The Meridian Mail voice processing system is not available on the CSE 1000 switch.
- Meridian Integrated RAN (MIRAN)

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 *Installation and Maintenance Guide*.
- The switch has been correctly installed and is operational with all current Product Enhancement Packages (PEPs) applied. For information on which PEPs to install on the switch, contact your Nortel Networks customer support representative.
- The switch is running X11 Release 24 or CSE 1000 Release 1.1 or greater.

Skills you need

Introduction

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

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

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

- X11 Release 24 or CSE 1000 Release 1.1
- CallPilot Release 2.0 or Meridian Mail Release 11
- Symposium Call Center Server 4.2

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:

- the appropriate switch release:
 - Meridian 1 Options 11, 11C Mini, 51C, 61C, 81, or 81C (Release 24 or greater)
 - Meridian 1 IE
 - Succession 1000 CSE, Release 1.1
- CallPilot Release 2.0
- Meridian Mail Release 8 and greater

PC experience or knowledge

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

- Microsoft Windows 2000 Server or Windows 2000 Advanced Server
- Windows 98, Windows NT 4.0 Workstation, Windows 2000 Professional, or Windows XP
- client/server architecture
- Internet Protocol (IP)

Other experience or knowledge

Other types of experience or knowledge that may 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 4.2 of the *Symposium, M1/CSE 1000, and Voice Processing Guide*.

New features in Release 4.2

- You can install the client on a PC running Windows XP.
- Symposium Call Center Server supports the Windows 2000 Advanced Server operating system, as well as Windows 2000 Server.
Note: Other versions of the Windows 2000 Server operating system, such as Windows 2000 Professional and Windows 2000 Datacenter, are not supported.
- Symposium Call Center Server can connect to the following switch types:
 - Succession CSE 1000
 - Meridian 1 Internet Enabled
 - Option 11C Mini
- Symposium Voice Services on CallPilot allows Symposium Call Center Server to use CallPilot as a voice processing system.

Subsystems overview

Introduction

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

- the Meridian 1 or CSE 1000 switch
- Symposium Call Center Server
- the voice processing system (CallPilot, 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

Switch

The 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 sets	CallPilot or 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 DN) 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 switch and the voice processing system.

Communicating with the switch

The server executes scripts and instructs the 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 CallPilot or Meridian Mail

Symposium Call Center Server also communicates with the voice processing system to instruct it to play prompts, collect digits input from callers, or both.

IVR messages between the server and voice processing system travel on the Customer LAN (CLAN), using the Meridian Link interface.

Voice processing messages between the server and CallPilot or Meridian Mail use the Application Module Link (AML) protocol. For CallPilot, these messages travel over the Embedded LAN (ELAN). For Meridian Mail, these messages travel over the serial 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.

CallPilot

The CallPilot voice channels connect to the Meridian 1 switch by a DS30 cable. On the switch side, this card is configured as an SL1 set TN (virtual agents).

Symposium Call Center Server can access the voice services provided by CallPilot 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, the ACD DN, and a treatment DN (if specified) over the AML (ELAN) connection to the switch. When the call arrives at a CallPilot voice port, Meridian 1 alerts CallPilot using Symposium Call Center Server and the MLink (CLAN).

When the Give Controlled Broadcast Announcement or Open Voice Sessions script commands are used, Symposium Call Center Server sends the command, the ACD DN, and a treatment DN (if specified) over the AML (ELAN) connection to the switch. When the call arrives at a CallPilot voice port, Meridian 1 alerts CallPilot using Symposium Call Center Server and the MLink (CLAN).

Meridian Mail

The Meridian Mail voice channels connect to the switch by means of a special network loop card. On the switch side, this card is configured as either an SL1 set TN (virtual agent) or, for the Option 11, a 2008 set TN.

Symposium Call Center Server can access the voice services provided by Meridian Mail 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 and a treatment DN (if specified) to the switch over the ELAN. 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 switch. The interface facilitates the integration of the computer and the switch. In this integrated environment, the host processor interacts with the 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.

Note: The Give IVR command with Symposium Voice Services on CallPilot requires Meridian Link.

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.

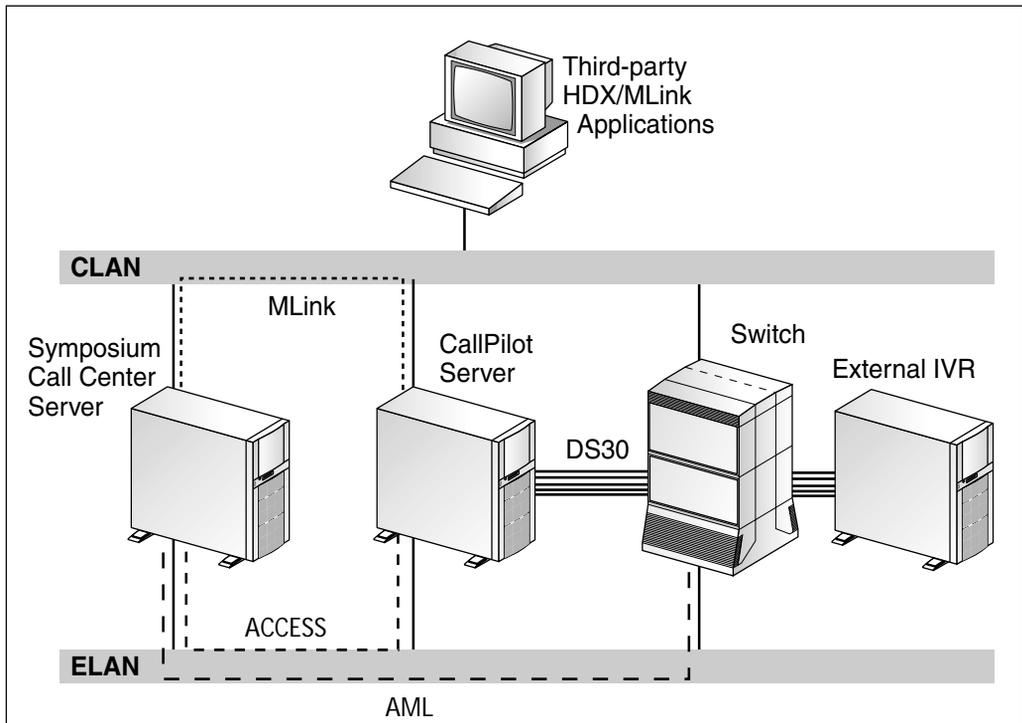
External applications register with MLS to obtain access to application layer messages. MLS commands that result in call processing requests are sent over the ELAN to the switch. Examples of external applications that can register with MLS include Telephony API (TAPI) and Computer Telephony Integration (CTI).

Connections between the subsystems

The subsystems communicate across local area networks (LANs) and serial links.

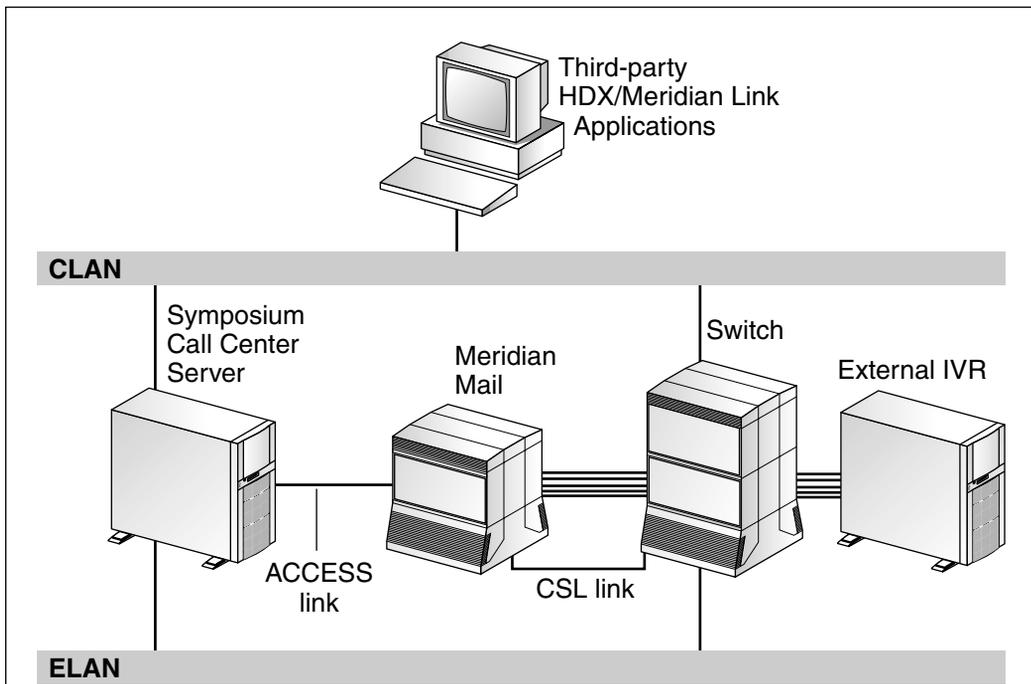
CallPilot

The following illustration shows the connections in a system using Symposium Voice Services on CallPilot:



Meridian Mail

The following illustration shows the connections in a system using Symposium Voice Services on Meridian Mail:



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 switch, CallPilot, and other servers. The ELAN carries call processing traffic and must 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. The CLAN also enables CallPilot to communicate with the switch.

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.
- **CSL link:** Command and Status Link. The connection between the switch and Meridian Mail is an RS-232 synchronous link.

Chapter 2

Configuration overview

In this chapter

Overview	26
Subsystem configuration reference information	27
Recommended sequence of configuration	29
Configuration checklists	31

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 switch, and the voice processing system (if applicable)
- the Meridian 1 or CSE 1000 switch
- the voice processing system (CallPilot, Meridian Mail, or a third-party voice processing system)
- Symposium Call Center Server

Note: If any of the following conditions apply, you must also configure the switch for Meridian Link Services (MLS):

- You are using Symposium Voice Services on CallPilot.
- You are using other Meridian Link Services.

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	Switch	CallPilot voice processing system	Meridian Mail voice processing system	Symposium Call Center Server
ACD-DNs	x	x	x	x
Symposium Call Center Server CDNs	x			x
CallPilot Voice Messaging Primary CDN	x	x		
IVR ACD-DNs (ACCESS and IVR)	x	x	x	x
Mailboxes and passwords (Meridian Mail)			x	x
Phonesets	x			x
Routes and queues	x			
Scripts				x
Treatment DNs	x (optional)	x	x	x
Voice files and segments		x	x	x

Configuration element	Switch	CallPilot voice processing system	Meridian Mail voice processing system	Symposium Call Center Server
Voice ports (Virtual Agent TNs) / voice channels	x	x	x	x
Voice service DNs		x	x	

Note: Meridian Mail refers to voice ports as voice channels.

To configure these entities on the switch, see Chapter 4, “Switch subsystem configuration.” To configure these entities in CallPilot or Meridian Mail, see Chapter 6, “Voice processing 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 create a system design based on your call center requirements. Before you start the configuration of the subsystems, use the *Planning and Engineering Guide* to optimize the integration of the subsystems. Use the Symposium Call Center Server Capacity Tool to determine requirements.

Create a system design based on your call center requirements taking into consideration such elements as

- network design (ELAN, CLAN, and RAS IP requirements)
- projected peak call traffic
- number of agents, shifts, days of operation
- projected call flow/scripting requirements
- voice messaging port requirements (Symposium Call Center Server voice service additional port requirements based on scripting usage of GIVE IVR, voice sessions, broadcasts in call flow, and scripts)

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

1. Complete the installation of all subsystems using the associated documentation.
2. Ensure that the links between subsystems are configured, connected, and functioning (see Chapter 3, “Subsystem connections configuration”).
3. Configure the switch (see Chapter 4, “Switch subsystem configuration”).
4. Configure the switch for MLS if any of the following conditions apply (see Chapter 5, “Meridian Link Services configuration”):
 - You are using CallPilot as your voice processing system.
 - You are using other Meridian Link Services.
5. Configure CallPilot, Meridian Mail, or the third-party voice processing system (see Chapter 6, “Voice processing subsystem configuration”).

6. Configure Symposium Call Center Server (see Chapter 7, “Symposium Call Center Server subsystem configuration”).
7. Install and configure third-party applications.

Configuring networking

If you are using Network Skill-Based Routing (NSBR), you must also configure networking. (Configure networking once each of the individual servers in your network is configured and operational.) To configure networking, you must configure the following:

- **the switch**—Install and configure the NACD package, and create the CDN to which networked calls for each server will be routed.
- **the Network Control Center**—Install and configure the NCC software.
- **the servers**—Define the network CDNs, configure the communication parameters, assign agents to network skillsets, and modify the scripts to use the network skill-based routing feature.

For more information, see the *Network Control Center Administrator's Guide*.

Note: Release 1.1 of the CSE 1000 switch only supports networking over ISDN trunks. IP networking is supported on Release 2.0. For more information, refer to the CSE 1000 documentation.

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 switch, CallPilot or 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 or the appropriate guide for the subsystem.

Note: In the following checklists, the subsystem Switch refers to both the Meridian 1 and the Succession CSE 1000 switch unless otherwise indicated. Also, references to the CallPilot documentation, *CallPilot Installation and Configuration Part 3: Switch and CallPilot Server Configuration*, indicate that you should refer to the CallPilot documentation appropriate for your switch (Meridian 1 or CSE 1000).

General checklist

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

Subsystem	Description	✓
Switch	Meridian 1/CSE 1000 switch is up and running correctly.	
Switch	Required Meridian 1/CSE 1000 software packages are available (use LD 22 to verify the packages). (See “Before you begin” on page 62 for more information.)	

Switch	<p>The following additional switch requirements are available:</p> <ul style="list-style-type: none"> ■ Meridian 1 dependency PEPs for Symposium Call Center Server Release 4.2 ■ Meridian 1 dependency PEPs for the specific voice processing subsystem you are using ■ provisioning of sufficient Call Registers ■ verification of the configuration of SCB or ADS using LD 23 (The preferred method is ADS.) 	
Switch	<p>ELAN connection between Symposium Call Center Server and Meridian 1/CSE 1000 switch is functioning.</p> <p>Note the current ELAN/VAS ID number using LD 48 (this can be useful when troubleshooting).</p>	
Switch	<p>Symposium Call Center Server CDNs are configured on the switch. (See “Changing CDNs on the switch” on page 67.)</p>	
Switch and voice processing	<p>Voice processing subsystem (CallPilot or Meridian Mail) is up and running for voice messaging.</p> <p>Voice processing subsystem is in communication with the switch.</p>	
Switch (CallPilot only)	<p>Link between the switch and the CallPilot server is enabled for CTI operations.</p>	

Give IVR checklist

Using CallPilot or Meridian Mail

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

Subsystem	Description	✓
Switch	<p>The IVR ACD-DN (for IVR voice ports) is configured on the switch (see “Configuring IVR ACD-DNs on the switch” on page 71).</p> <p>Note: When you configure the IVR ACD-DN on the switch, set the IVR prompt and ALOG prompt to YES.</p>	
Switch	<p>IVR voice ports are configured on the switch as virtual agents (see “Configuring voice ports on the switch” on page 75).</p> <p>Voice ports must be dedicated to Symposium Call Center Server GIVE IVR voice service.</p>	
CallPilot	<p>CallPilot server configuration is updated.</p> <p>In the CallPilot Configuration Wizard, review all CallPilot configuration information up to the Switch Information page.</p>	
CallPilot	<p>Integration is enabled.</p> <p>In the CallPilot Configuration Wizard, on the Meridian 1 Switch Information page, the</p> <ul style="list-style-type: none"> ■ Enable Symposium Call Center Server Integration field is selected ■ customer number is entered in the Switch Customer Number field ■ CLAN address of Symposium Call Center Server is entered in the Symposium Call Center Server CLAN IP Address field 	

CallPilot	<p>Voice ports (IVR channels) are configured in the CallPilot server configuration.</p> <p>In the CallPilot Configuration Wizard, identify and configure the channels that provide IVR services to Symposium Call Center Server. (See “Non-ACCESS ports” on page 115 for more information.)</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ TN of the voice channel in CallPilot or Meridian Mail = TN of the virtual agent on the switch = Telephony/Port Address of the phoneset on. ■ ACD-DN defined for the voice channel in CallPilot or Meridian Mail = ACD-DN on the switch = IVR ACD-DN on Symposium Call Center Server. 	
CallPilot	<p>SDN table in CallPilot is updated with Symposium Call Center Server IVR ACD-DNs.</p> <ul style="list-style-type: none"> ■ Application name is Symposium Voice Services. ■ Media type is Voice. ■ CallPilot Primary CDN is in the SDN table with an application name of Voice Messaging and a media type of Voice. 	

CallPilot	<p>IVR ACD-DNs and Symposium Call Center Server Treatment DNs are configured in the CallPilot SDN table.</p> <ul style="list-style-type: none"> ■ In the SDN table, an IVR ACD-DN should use the predefined application name Symposium Voice Services. The media type is Voice. ■ A Treatment DN must correspond to a CallPilot Service DN entry in CallPilot Manager. It must have an application name, and the media type is Voice. ■ Use meaningful application names because CallPilot can store a large number of applications. ■ Applications with media type of Voice contain voice items that are actually announcements and menus played to callers (GIVE IVR script commands). ■ Use CallPilot Application Builder to create, record, and manage voice items. ■ Applications must be complete before they can be selected in the CallPilot SDN table. 	
Meridian Mail	<p>VSDN table is updated with Symposium Call Center Server IVR ACD-DN (it must be associated with a Voice Messaging service type).</p>	
Meridian Mail	<p>IVR ACD-DN and IVR ports are defined in the Channel Allocation Table (see “Configuring IVR voice ports” on page 136).</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ TN of the voice channel in CallPilot or Meridian Mail = TN of the virtual agent on the switch = Telephony/Port Address of the phoneset on. ■ ACD-DN defined for the voice channel in CallPilot or Meridian Mail = ACD-DN of the TN on the switch = number of IVR ACD-DN on Symposium Call Center Server. 	

Meridian Mail	<p>IVR ACD-DNs and Symposium Call Center Server Treatment DNs are configured in the Meridian Mail VSDN table.</p> <ul style="list-style-type: none"> ■ IVR ACD-DN and Treatment DNs are configured as VSDN entries. The IVR ACD-DN is associated with a Voice Messaging type of application. Treatment DNs are associated with the desired voice application (for example, a voice menu or an announcement). ■ Voice segments (menus or announcement services) must be created before being associated with a Treatment DN entry in the VSDN table. ■ See “Configuring Meridian Mail for Give IVR” on page 132 for more information. 	
Symposium Call Center Server	<p>IVR ACD-DN is configured and acquired on Symposium Call Center Server (see “Configuring IVR ACD-DNs on the server” on page 151).</p>	
Symposium Call Center Server	<p>Voice ports are configured and acquired on Symposium Call Center Server (see “Configuring voice ports on the server” on page 153).</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ Voice ports must be dedicated to Symposium Call Center Server. ■ IVR voice ports are defined with a TN with no channel number. ■ All TNs in the switch belonging to the IVR ACD-DNs in the switch are acquired by Symposium Call Center Server as voice ports. 	

Symposium Call Center Server, Voice Processing	<p>In Symposium Call Center Server</p> <ul style="list-style-type: none"> ■ VSM and MLSM services (MLSM services for CallPilot integration only) are running. (Use the VSM Status window if it is configured.) <p>For voice port status (IVR), the acquired voice ports are Idle in CallPilot or Meridian Mail.</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ For Meridian Mail, use the DSP Port Status (in the Meridian Mail Administration Terminal). ■ For CallPilot, use the Channel Monitor in CallPilot Manager. If the voice ports are not initialized, then restart CallPilot. 	
Symposium Call Center Server	<p>GIVE IVR test script is implemented and activated before activating the rest of the voice processing scripts.</p> <p>The voice processing script command uses both the IVR ACD-DN and a Treatment DN such as</p> <pre style="text-align: center;">GIVE IVR ivr_queue WITH TREATMENT welcome_msg</pre> <p style="text-align: center;">or</p> <pre style="text-align: center;">GIVE IVR 6000 WITH TREATMENT 1001</pre> <p>(See “Defining scripts” on page 160 for more information.)</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ Voice processing commands use the same syntax in CallPilot and Meridian Mail. ■ If the Treatment DN is not specified in the script, the server uses the default Treatment DN (TRDN) defined for the IVR ACD-DN on the switch. Ensure that the default treatment is also configured in the VSDN table (in Meridian Mail) or the SDN table (in CallPilot) for proper operation. 	

Symposium Call Center Server	Master script is activated. Note: Scripts containing voice processing instructions should be activated after all relevant subsystem configurations.	
Symposium Call Center Server	GIVE IVR test script is working: <ul style="list-style-type: none">■ Other scripts are verified and any additional voice processing scripts are implemented.■ SDN or VSDN tables are updated as new treatments as required. Test script is not working: <ul style="list-style-type: none">■ Start troubleshooting procedures.	

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.

Subsystem	Description	✓
	The general checklist is complete.	
	The IVR ACD-DN is configured on the switch (see “Configuring IVR ACD-DNs on the switch” on page 71). Note: When you configure the IVR ACD-DN on the switch, set the IVR prompt to YES.	
Switch	Voice ports are configured on the switch as virtual agents (see “Configuring voice ports on the switch” on page 75). Note: You must use dedicated voice ports.	
Switch	The third-party voice processing system is up and communicating properly with the switch.	
Voice processing	Voice ports are configured in the third-party voice processing system.	
Symposium Call Center Server	The IVR ACD-DN is configured and acquired on the server (see “Configuring IVR ACD-DNs on the server” on page 151).	

Symposium Call Center Server	<p>Voice ports are configured and acquired on the server (see “Configuring voice ports on the server” on page 153).</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ Voice ports are defined with a TN only (the channel number field is blank). ■ TN of virtual agent on switch = Telephony/Port Address of phoneset on server. ■ <i>All</i> TNs in the switch ACD-DN are acquired by the server as voice ports. 	
Symposium Call Center Server	The script command refers to the IVR ACD-DN that is acquired.	
Symposium Call Center Server	<p>If IPML exists, the treatment DN is specified in the script.</p> <p>Note: If the treatment DN is not specified in the script, the server uses the default treatment DN defined for the IVR ACD-DN on the switch.</p>	
Symposium Call Center Server	Voice ports behind the IVR ACD-DN acquired by the server are indicated by the switch in Logged in state.	

Controlled Broadcast and Voice Sessions checklist

If you use the Give Controlled Broadcast Announcement or Voice Sessions commands with Symposium Voice Services on CallPilot or Symposium Voice Services on Meridian Mail, ensure that the tasks in the following checklist are completed in the order listed:

Subsystem	Description	✓
Switch	<p>The IVR ACD-DN (for IVR voice ports) is configured on the switch (see “Configuring IVR ACD-DNs on the switch” on page 71).</p> <p>Note: When you configure the IVR ACD-DN on the switch, set the IVR prompt and ALOG prompt to YES.</p>	
Switch	<p>ACCESS voice ports are configured on the switch as virtual agents (see “Configuring voice ports on the switch” on page 75).</p> <p>Voice ports must be dedicated to Symposium Call Center Server ACCESS voice service.</p>	

Symposium Call Center Server	<p>Voice connection type is set on the Voice Connection tab in the Server Configuration Utility (found on Symposium Call Center Server Update NS040206SU07S).</p> <ul style="list-style-type: none"> ■ Serial Default option for connection to Meridian Mail. ■ TCP Enables the integration of CallPilot as the voice processing subsystem. <p>See “Voice connection” on page 145 for more information on voice connection.</p> <p>If you are changing voice services from Meridian Mail to CallPilot, ensure that the</p> <ul style="list-style-type: none"> ■ Meridian Mail resources are deaquired on Symposium Call Center Server (IVR and ACCESS ACD-DNs, TNs, voice ports) ■ ACCESS cable is physically disconnected from the Symposium Call Center Server COM port ■ other end of the cable is disconnected from the Meridian Mail utility card 	
Symposium Call Center Server (CallPilot integration only)	<p>CallPilot connection parameters are configured for TCP voice connection.</p> <ul style="list-style-type: none"> ■ CallPilot connection IP (ELAN IP address) is set. ■ CallPilot connection port is set (TCP port should be 10008). 	

Symposium Call Center Server	<p>Voice connection type (on Symposium Call Center Server client) is set. If the voice connection type on the server is set to</p> <ul style="list-style-type: none"> ■ Serial: the Symposium Call Center Server client system configuration object displays Voice Prompt Editor ■ TCP: the Symposium Call Center Server client system configuration object does not display Voice Prompt Editor <p>Note: Use CallPilot Application Builder to edit, record, and manage voice items used in Symposium Call Center Server scripts.</p>	
CallPilot	<p>CallPilot server configuration is updated.</p> <p>In the CallPilot Configuration Wizard, review all CallPilot configuration information up to the Switch Information page.</p>	
CallPilot	<p>Integration is enabled.</p> <p>In the CallPilot Configuration Wizard, on the Switch Information page, the</p> <ul style="list-style-type: none"> ■ Enable Symposium Call Center Server Integration field is selected ■ customer number is entered in the Switch Customer Number field ■ CLAN address of Symposium Call Center Server is entered in the Symposium Call Center Server CLAN IP Address field 	

CallPilot	<p>Voice ports (ACCESS channels) are configured in the CallPilot server configuration.</p> <p>In the CallPilot Configuration Wizard, identify and configure the channels that provide ACCESS services to Symposium Call Center Server. (See “ACCESS ports” on page 114 for more information.)</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ Voice port channel number on Symposium Call Center Server = class number of the voice channel on Meridian Mail or the Class ID of the TN on CallPilot. ■ TN of voice channel in CallPilot or Meridian Mail = TN of virtual agent on switch = Telephony/Port Address of phoneset on Symposium Call Center Server (with voice port option selected). ■ ACD-DN defined for the voice channel in CallPilot or Meridian Mail = ACD-DN on switch = ACCESS ACD-DN on Symposium Call Center Server. 	
CallPilot	<p>SDN table in CallPilot is updated with Symposium Call Center Server ACCESS ACD-DNs.</p> <ul style="list-style-type: none"> ■ Application name is Symposium Voice Services. ■ Media type is Voice. ■ CallPilot Primary CDN is in the SDN table with the application name of Voice Messaging and a media type of Voice. 	
Meridian Mail	<p>VSDN table is updated with Symposium Call Center Server ACCESS ACD-DN (it must be associated with an ACC service type).</p>	

Meridian Mail	<p>ACCESS ACD-DN and ACCESS ports are defined in the Channel Allocation Table (see “Configuring ACCESS voice ports” on page 126).</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ TN of the voice channel in CallPilot or Meridian Mail = TN of the virtual agent on the switch = Telephony/Port Address of the phoneset on Symposium Call Center Server. ■ ACD-DN defined for the voice channel in CallPilot or Meridian Mail = ACD-DN on the switch = IVR ACD-DN on Symposium Call Center Server. 	
Symposium Call Center Server	ACCESS ACD-DN is configured and acquired on Symposium Call Center Server (see “Configuring the ACCESS IVR ACD-DN” on page 125).	
Symposium Call Center Server and Meridian Mail	<p>Meridian Mail mailbox and password are configured.</p> <ul style="list-style-type: none"> ■ An appropriate Meridian Mail mailbox password policy for Symposium Call Center Server is defined. ■ Meridian Mail mailbox and password are created, updated, or checked. ■ Symposium Call Center Server IVR ACD-DN global settings are updated with the mailbox and password. <p>Note: If the password expires, you cannot access the VPE and callers cannot hear the prompts because Symposium Call Center Server cannot open the mailbox.</p>	

Symposium Call Center Server	<p>IVR ACD-DN global settings are configured on Symposium Call Center Server (see “Configuring IVR ACD-DN global settings” on page 154) to specify the following:</p> <ul style="list-style-type: none"> ■ default ACCESS Treatment DN and ACCESS ACD-DN ■ number of ACCESS ports reserved for Broadcasts ■ Broadcast Voice Port Wait Timer ■ for CallPilot, the mailbox number and password set to non-zero values ■ for Meridian Mail, the ACCESS mailbox number and password configured in Meridian Mail for voice prompts 	
Symposium Call Center Server	<p>Voice ports are configured and acquired on Symposium Call Center Server (see “Configuring voice ports on the server” on page 153).</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ ACCESS voice ports are defined with both a TN and a channel. ■ All TNs in the switch belonging to the ACCESS ACD-DNs in the switch are acquired by Symposium Call Center Server as voice ports. 	

Symposium Call Center Server, Voice Processing	<p>ACCESS link between Symposium Call Center Server and CallPilot or Meridian Mail is functioning correctly.</p> <p>In Symposium Call Center Server</p> <ul style="list-style-type: none">■ VSM and MLSM services (MLSM services for CallPilot integration only) are running. (Use VSM Status window if it is configured.) <p>For voice port status (ACCESS), the acquired voice ports are Idle in CallPilot. In Meridian Mail, the ACCESS channels appear as active when acquired by Symposium Call Center Server.</p> <p>Notes:</p> <ul style="list-style-type: none">■ For Meridian Mail, use the DSP Port Status (in the Meridian Mail Administration Terminal).■ For CallPilot, use the Channel Monitor in CallPilot Manager. If the voice ports are not initialized, then restart CallPilot.	
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Symposium Call Center Server, Voice Processing	<p>Voice segment definitions in Symposium Call Center Server are configured.</p> <ul style="list-style-type: none"> ■ System predefined voice segments (file 1, which contains all the number prompts) is updated. (Make a list of the segment IDs referenced in the Symposium Call Center Server variable table.) ■ New user voice segments are defined. ■ In Meridian Mail, use the Symposium Call Center Server Voice Prompt Editor (VPE) to create or delete voice files and record voice segments. <ul style="list-style-type: none"> Voice segments created with VPE are stored in Meridian Mail. The variables are defined as <voice file name> : <segment number> Voice file names are case-sensitive. ■ In CallPilot, use CallPilot Application Builder to define voice items referred to in Symposium Call Center Server as voice segments or prompts. <ul style="list-style-type: none"> In Symposium Call Center Server, the variables are defined as <application name> : <voice item ID> Application names are case-sensitive. Voice items or voice prompts must first be created in VPE or Application Builder before they can be defined in voice segment variables being used in Symposium Call Center Server scripts. <p>For more information, see</p> <ul style="list-style-type: none"> ■ “Configuring Meridian Mail for ACCESS” on page 125 ■ <i>CallPilot Application Builder Guide</i> ■ “Defining voice segment variables” on page 158 	
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Symposium Call Center Server	<p>ACCESS test script is implemented and activated before activating the rest of the voice processing scripts.</p> <p>The voice processing script command uses both the ACCESS ACD-DN (optional) and voice segment variables that refer to the voice prompts that the caller hears.</p> <p>For example</p> <pre>GIVE CONTROLLED BROADCAST ANNOUNCEMENT PLAY PROMPT VOICE SEGMENT closed_message_vs OPEN VOICE SESSION PLAY PROMPT VOICE SEGMENT hold_option_vs COLLECT 1 DIGITS INTO hold_choice_cv END VOICE SESSION</pre> <p>(See “Defining scripts” on page 160 for more information.)</p> <p>Notes:</p> <ul style="list-style-type: none"> ■ Voice processing commands use the same syntax in CallPilot and Meridian Mail. 	
Symposium Call Center Server	<p>Master script is activated.</p> <p>Note: Scripts containing voice processing instructions should be activated after all relevant subsystem configurations.</p>	
Symposium Call Center Server	<p>CONTROLLED BROADCAST and OPEN VOICE SESSION test script is working.</p> <ul style="list-style-type: none"> ■ Other scripts are verified and any additional voice processing scripts are implemented. ■ Voice segment variables are updated as required. <p>Test script is not working.</p> <ul style="list-style-type: none"> ■ Start troubleshooting procedures. 	

NSBR/Queue to NACD command checklist

If you are using Network Skill-Based Routing or the Queue to NACD command, ensure that NACD is configured correctly on the switch. This entails the tasks in the following checklist:

Subsystem	Description	✓
Switch	The NACD package is installed on the switch.	
Switch	<p>To configure an ACD-DN for NACD, you must set up the following in LD 23:</p> <ul style="list-style-type: none"> ■ Day and Night Routing Tables ■ Call Answering Algorithms—Oldest Call in Network (OCN) and High Priority Queue (HPQ) ■ NACD timers, such as RAGT (Reserve Agent Timer), CRQS (Call Request Queue), and FCTH (Flow Control Threshold) <p>For more information, refer to the NACD documentation.</p>	
Switch	<p>(for the Queue to NACD command only)</p> <p>An NACD-DN is configured on the switch, with an associated Night Routing Table containing remote targets and time values. This ACD-DN has no positions assigned, and, therefore, is always in night mode. It enables the QUEUE TO NACD command. (See “Configuring NACD-DNs to enable the QUEUE to NACD command” on page 68.)</p>	

Note: The switch supports call routing through tandem nodes. To route calls through tandem nodes, ensure that you have provisioned your CLAN/WAN to allow the transmission of messages from one Symposium Call Center Server to another.

Chapter 3

Subsystem connections configuration

In this chapter

Overview	52
Configuring the ELAN	53
Configuring the ACCESS link	56

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 switch
For information about how to configure an IP address for the switch, refer to the switch documentation.
- the Embedded LAN (ELAN) between the switch, Symposium Call Center Server, and CallPilot
- if you are using Symposium Voice Services on CallPilot or Meridian Mail to provide broadcast or voice session services, you must configure the ACCESS link between Symposium Call Center Server and the voice processing system. For more information on configuring the ACCESS link, see “Configuring the ACCESS link” on page 56.

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. Enable the ELAN link.
3. Check the ELAN link.
4. Check the AML link.

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

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

To define the ELAN with LD 17

Use these prompts and responses in Overlay 17.

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

Prompt	Response	Description
DES	NAME	Enter a name for the ELAN port number. Use a generic name because the ELAN port is not dedicated to a specific application.
VAS	NEW	Add a value added server.
VSID	16	VAS identifier
ELAN	16–31	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 output queue
CSQI	255 maximum	Number of call registers linked to input queue
REQ	END	Exit from overlay.

To enable the ELAN link

- 1 At the switch administration terminal, load LD 48.
- 2 Type the command **enl elan 16**.

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 switch 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 (for example, elan16)
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 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.

Checking the AML link

If you are using Meridian Mail for your voice services, you can check the AML link.

To check the AML link

- 1 At the switch administration terminal, load LD 48.
- 2 Type the command **stat aml**.

Result: Ensure that LYR2 is connected and LYR7 is active.

Example

```
>ld 48
.stat aml
AML: 09 ESDI: 09 DES: mail\
LYR2: CONNECTED LYR7: ACTIVE EMPTY
```

Configuring the ACCESS link

Introduction

If you are using Symposium Voice Services on CallPilot or Meridian Mail, you can include Open Voice Session or Give Controlled Broadcast commands in your scripts. Symposium Call Center Server communicates with the voice processing system using the ACCESS link. You must configure the link between the server and the voice processing system by performing these tasks:

- Enable ACCESS on the voice processing system.
- Set up the physical connection from Symposium Call Center Server to the voice processing system.
- Configure the ACCESS link between Symposium Call Center Server and the voice processing system. For more information about configuring the ACCESS link, see “Configuring the ACCESS link to Symposium Call Center Server” on page 57.

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

To determine whether ACCESS is enabled

On CallPilot

On the CallPilot server, ACCESS is a keycode-activated feature. To find out whether this feature is installed on your server, follow these steps.

- 1 From Start → Programs → CallPilot, choose System Utilities → System Monitor, and go the System Information page.
- 2 On the System Information page, verify that the value of the Access Channels field is Yes. If it is not, contact your Nortel Networks authorized distributor.

On Meridian Mail

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.

If Meridian Mail does not have the ACCESS feature, you must do a comprehensive upgrade to include this feature.

Establishing a physical connection to Symposium Call Center Server

From CallPilot

CallPilot communicates with Symposium Call Center Server over the Embedded LAN (ELAN). Ensure that the CallPilot server is physically connected to the ELAN.

From Meridian Mail

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:

- 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 DB-9 female connector.
- On the Option 11C Meridian Mail platform, you must configure an RSM interface card and a fan-out cable.

Configuring the ACCESS link to Symposium Call Center Server

From CallPilot

When you configure CallPilot, you must specify the CLAN address of Symposium Call Center Server in the CallPilot Configuration Wizard. For detailed instructions, see “To update the Configuration Wizard” on page 116.

When you configure Symposium Call Center Server, you must specify the ELAN address of the CallPilot server on Symposium Call Center Server. For detailed instructions, see “To configure the CallPilot connection” on page 146.

From 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 that configured in Symposium Call Center Server.

To configure the Meridian ACCESS link on Meridian Mail

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

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. If the number of ACCESS ports is greater than 48, then the Baud rate should be set to 19.2 in Symposium Call Center Server and 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.

Where to go from here

To complete configuration of voice processing, you must also

- configure IVR ACD-DNs on the switch. For more information, see “Configuring IVR ACD-DNs on the switch” on page 71.
- configure voice ports on the switch. For more information, see “Configuring voice ports on the switch” on page 75.
- configure ACCESS voice ports on the voice processing system. For more information about configuring ACCESS voice ports on CallPilot, see “Updating the CallPilot configuration” on page 116. For more information about configuring ACCESS voice ports on Meridian Mail, see “Configuring Meridian Mail for ACCESS” on page 125.

Chapter 4

Switch subsystem configuration

In this chapter

Overview	62
Configuring CDNs on the switch	64
Changing CDNs on the switch	67
Configuring NACD-DNs to enable the QUEUE to NACD command	68
Configuring IVR ACD-DNs on the switch	71
Configuring voice ports on the switch	75
Configuring agent and supervisor TNs on the switch	78
Configuring routes on the switch	83
Configuring Multiple Queue Assignments	86
Initializing or sys-loading the switch	87

Overview

Introduction

You must configure the following elements on the Meridian 1 or CSE 1000 switch :

- agent ACD queues (refer to your switch documentation for more information)
- CDNs (see “Configuring CDNs on the switch” on page 64)
- networked ACD-DNs (see “Configuring NACD-DNs to enable the QUEUE to NACD command” on page 68)
- IVR ACD-DNs (see “Configuring IVR ACD-DNs on the switch” on page 71)
- voice ports (see “Configuring voice ports on the switch” on page 75)
- agent phonesets (see “Configuring agent and supervisor TNs on the switch” on page 78)
- routes (see “Configuring routes on the switch” on page 83)

This chapter explains how to configure these elements. It 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 switch.

Before you begin

Before you configure elements on the switch, make sure

- you have the packages required for voice processing in Symposium Call Center Server, for voice processing in CallPilot, and for any other desired switch features:
 - The Symposium Call Center Server packages include 311 (NGCC) and 324 (NGEN).

- The Meridian 1 software package list for CallPilot 2.0 is found in the CallPilot 2.0 General Release Bulletin (GRB).
- For more information, contact your Nortel Networks customer support representative.
- your switch is running the appropriate software release:
 - Meridian 1: X11 software Release 24 or greater
 - CSE 1000: Release 1.1 or Release 2.0
- all current Product Enhancement Packages (PEPs) are applied. For information on which PEPs to install on the switch, contact your Nortel Networks customer support representative.
- if you have purchased the Network Skill-Based Routing option, the NACD package is installed on the switch

Note: Symposium Call Center Server cannot share switch resources (CDNs, ACD-DNs, TNs, and so on) with other applications such as Meridian Max, Link, and CCR.

Changing switch resources

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

Configuring CDNs on the switch

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 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 controlled mode.
- The parameter ASID appears next to the CDN block.
- CNTL and AACQ automatically change to Yes.

If CNTL = No (that is, if the CDN is not acquired), or if no treatment is given in 4 seconds, calls are directed to the default DN. (Do not set CNTL to Yes manually.)

Note: Ensure that RPRT = NO.

Assumptions

The following assumptions are made:

- You know the user ID and password to log on to the switch administration terminal.
- You are familiar with switch 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.

To configure a CDN with LD 23

Use these prompts and responses in Overlay 23. 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 Directory 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. (Do not set CNTL to Yes manually.)
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
SPNB 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 switch

Introduction

If you need to make changes to CDNs on the 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 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 switch.
3. Acquire the CDN on Symposium Call Center Server.

Configuring NACD-DNs to enable the QUEUE to NACD command

Introduction

The networked ACD-DN (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.

Note: To enable NACD routing, the switch must be connected using a Meridian Customer Defined Network (MCDN). For more information about configuring the switch on an MCDN, refer to the documentation that came with the switch.

When to configure NACD-DNs

Configure NACD-DNs when you want the application to communicate with the switch 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 switch administration terminal.
- You are familiar with switch Change and Diagnostics overlays.

To configure a new NACD-DN with LD 23

Use these prompts and responses in Overlay 23. For prompts that are not specified in the following table, press Enter.

Prompt	Response	Description
REQ	NEW, CHG	Add or change an NACD-DN.
TYPE	ACD	ACD data block
CUST	xx	Customer number
ACDN	xxxx	The 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 to be 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.

To configure the Night Routing table with LD 23

Use these prompts and responses in Overlay 23. For prompts that are not specified in the following table, press Enter.

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 Enter to add another target. You can add a maximum of 20 targets to the table. Press Enter 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 on the switch

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 switch, in your voice processing system (see Chapter 6, “Voice processing subsystem configuration”), and on Symposium Call Center Server (see Chapter 7, “Symposium Call Center Server subsystem configuration”).

This section describes how to configure IVR ACD-DNs if you are using Symposium Voice Services on CallPilot or Meridian Mail. If you are using a third-party IVR system, refer to the documentation provided with the system.

When to configure IVR ACD-DNs

Configure IVR ACD-DNs if you use CallPilot, Meridian Mail, or a third-party IVR system to play messages to callers. These messages are stored on the voice processing system and can be announcements or voice menus.

Note: 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 [Voice Messaging ACD-DNs are not acquired by Symposium Call Center Server]).

For more information on voice port partitioning, see “Configuring voice ports on the switch” on page 75.

Assumptions

The following procedures assume that

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

To create a CallPilot ACD-DN in LD 23

Use the following prompts and responses in Overlay 23. For prompts not listed in the table, press Enter to accept the default.

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	NO	Indicates that this is not a message center.
MAXP	xx	Indicates the maximum number of ACD agents for this queue.
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. Use treatment DN's to select the treatment that the call receives from the voice processing system. You can also use them with CallPilot, Meridian Mail, or any voice processing system that connects to the switch by means of the AML link.
ALOG	YES	ACD agents are automatically logged on.
REQ	END	Exit from overlay.

To create a Meridian Mail IVR ACD-DN in LD 23

Use the following prompts and responses in Overlay 23. For prompts that are not specified in the following table, press Enter.

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.
IVMS	YES	Integrated voice messaging. This creates a message center from which messages can be retrieved.

Prompt	Response	Description
VSID		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 used if treatment is not specified for a Give IVR command in a script. Use treatment DN's to select the treatment that the call receives from the voice processing system. You can also use them with CallPilot, Meridian Mail, or any voice processing system that connects to the switch by means of the AML link.
REQ	END	Exit from overlay.

After you finish

You must configure the voice ports as virtual agents. See “Configuring voice ports on the switch” on page 75.

Configuring voice ports on the switch

Introduction

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

Configure voice ports as virtual agent TNs. For Meridian Mail, the class of service must be IMA and VMA.

Note: For the voice ports, ensure that the key layout matches the configuration of keys in CallPilot or Meridian Mail. This matching enables CallPilot and 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 111.

To create a CallPilot voice port with LD 11

Use the following prompts and responses in Overlay 11. For prompts not listed in the table, press Enter to accept the default.

Prompt	Response	Description
REQ	NEW, CHG	Add or change a voice port.
TYPE	2008	

TN	l s c u	Terminal Number of the voice port, where l is the loop, s is the shelf, c is the card, and u is the unit. (For the Option 11C, TN is cu only.)
AST	00 01	Associated set assignment on key 0 and key 1 (required for MLSM messages)
CLS	MMA VCE WTA FLXA (units 16–31)	Multimedia agent Voice terminal Warning time allowed. (When changing from CLS DAT to CLS VCE, CLS WTA should be added to avoid conflict with CLS CPTA. CLS CPTA is the default for VCE TNs.) Flexible voice/data allowed.
KEY	0 ACD xxxx zzz nnnn	Define 0 as an ACD key. xxxx is the ACD DN of agents to voice mail. zzz is the CLID entry number. nnnn is the position ID. In CallPilot, the position ID must match the CallPilot Key 0 value.
KEY	1 SCN xxxx	Define key 1 as a single-call non-ringing DN. xxxx is the SCN DN of the SCN. The DN must match the Key 1 value on the CallPilot Meridian 1 Switch Information screen.
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 TRN	Define key 4 as a Transfer key.
KEY	5 AO3 (letter 'O')	Define key 5 as a Conference key.
REQ	END	Exit from overlay.

To create a Meridian Mail voice port with LD 11

Use the following prompts and responses in Overlay 11. For prompts not listed in the table, press Enter to accept the default.

Prompt	Response	Description
REQ	NEW, CHG	Add or change a voice port.
TYPE	2008 or SL1	Use 2008 for Meridian 1 Option 11, and SL1 for all other switch types.
TN	10 (0–15)	Enter the TN of the agent.
CLS	IMA VMA	Integrated messaging service attendant allowed. Server voice messaging allowed.
KEY	0 ACD xxxx zzz nnnn	Define 0 as an ACD agent key. xxxx is the ACD DN of voice agents in voice mail. zzz is the CLID entry number. nnnn is the position ID.
KEY	1 SCN xxx	Define key 1 as a single-call non-ringing DN. xxx 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	6 TRN	Define key 6 as a Transfer key.
KEY	7 AO3 (letter 'O')	Define key 7 as a Conference key.
KEY	9 RLS	(For SL1 sets) define key 9 as a Release key.
REQ	END	Exit from overlay.

Configuring agent and supervisor TNs on the switch

Introduction

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

Note: While agent and supervisor phonesets require no special configuration for Symposium Call Center Server, they must belong to an ACD-DN. Normally, call routing 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.

Configuring the ACD-DN

You use LD 23 to configure the ACD-DN with which a phoneset is to be associated. (For detailed instructions, refer to the switch documentation.) If you want the agent using the phoneset to be able to enter Not Ready reason codes, you must ensure that NRAC for the ACD-DN is set to YES.

Note: To deactivate Not Ready activity codes, set the NRAC to NO.

VSID definitions

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

Assumptions

The following procedures make these assumptions:

- You know the user ID and password to log on to the switch administration terminal.

- You are familiar with switch 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	Name	Enter a name for the phoneset.
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	NRD	Not ready
KEY 2	A06	6-party conference
KEY 3	MSB	Make set busy
KEY 4	TRN	Transfer
KEY 8	SRC xxxx	xxxx=IDN
KEY 13	ACNT	Activity key
REQ	NEW, END	Either define another multiline ACD phoneset, or exit the overlay saving all of the changes entered.

Notes:

- To use Not Ready reason codes, you must program a key as an activity key (ACNT).

- 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 enable the Incalls key and the secondary DN key for CTI. For example, if you have a personal DN on key 7, set AST to 00 07 to 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 after 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
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). Only for AST sets.
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 on the switch

Introduction

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

Configure routes according to the switch 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:

- Only RAN, MUS, FGDT, TIE, DID, COT, FEX, and WATS route types are supported by Symposium Call Center Server.
- If you want to generate reports on trunk routes, you must also configure the trunk routes on Symposium Call Center Server.
- If you are not using Symposium Voice Services on CallPilot or Meridian Mail, you do not need to configure RAN routes.

Assumptions

The following procedures make these assumptions:

- You know the user ID and password to log on to the switch administration terminal.
- You are familiar with switch Change and Diagnostics overlays.
- You have obtained a listing of routes using overlay program 21 (REQ=prt, TYPE=RDB).
- 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 Overlay 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
ROUT	nn	Route number
DES	x...x	Enter a description.
TKTP	RAN	Recorded Announcement trunk data block requires package 7.
ASUP	YES	Answer supervisor
ACOD	nnn	Access code
REQ	END	Exit from overlay.

Configuring a MUS route with LD 16

To configure a MUS route using Overlay 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. Enter a description.
TKTP	MUS	MUSIC trunk data block requires Music package 44.

Prompt	Response	Description
ICOG	OGT	Incoming and/or outgoing trunk
ACOD	nnn	Access code
REQ	END	Exit from overlay.

Associating a MUS route with TNs in LD 14

To associate a MUS route with a TN in Overlay 14, use these prompts and responses:

Prompt	Response	Description
REQ	NEW	Action request
TN	l s c u	Terminal number
TYPE	MUS	music route
CUST	0	Customer number
RTMB	0–127 1–254	Route and member number
CFLP	0–159	Conference loop
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.

Note: MQA agents and Symposium Call Center Server agents cannot be the same. They must be totally separate.

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 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 the voice processing services system.
- 3 Shut down Symposium Call Center Server.
- 4 Install any needed software upgrades on the switch.
- 5 Restart the switch.
- 6 Restart the voice processing services system.
- 7 Restart Symposium Call Center Server.

Chapter 5

Meridian Link Services configuration

In this chapter

Overview	90
Installing and configuring Meridian Link Services	92
Allowing CTI operations on the ELAN	93
Configuring phonesets for CTI	94
Configuring CDNs for host-enhanced routing	97
Connecting the host application	98

Overview

Introduction

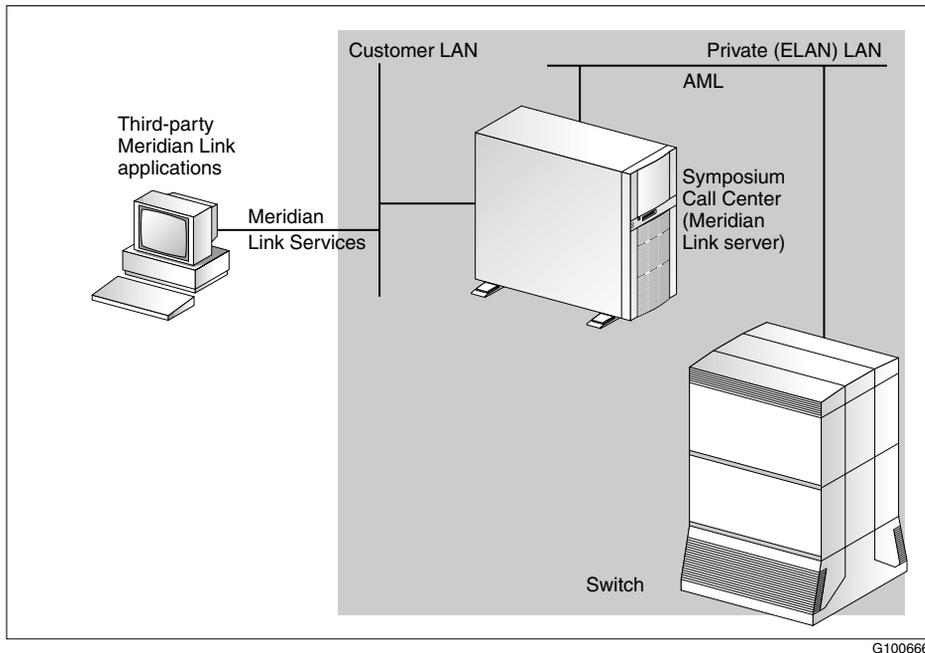
This chapter provides the references and procedures required to set up Meridian Link Services (MLS) for operation with Symposium Call Center Server. MLS is a communications facility that provides an interface between a host application and the switch. (A host is any computer on which the third-party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

If you are using CallPilot for your Symposium Voice Services, CallPilot communicates with the switch using MLS, so you must configure the switch for Meridian Link. Once you configure MLS on the switch, your connection to the host application is through the CLAN and ELAN connection points.

You can also 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.

Note: 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 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 *Symposium Meridian Link Services Interface Specification* 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 switch for both Symposium Call Center Server and stand-alone Meridian Link Services (MLS) are identical. You configure the switch using X11 software overlays.

Prerequisites

Before you configure MLS, you must install it. The software is included on the Symposium Call Center Server Release 4.2 Server Application CD-ROM. Refer to the *Installation and Maintenance Guide* for information on installing MLS.

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

Enable Meridian Link Services on the switch

To enable MLS to function on the switch, you must do the following:

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 switch, 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.

To complete the basic Meridian Link Services configuration

Use these prompts and responses in LD 17.

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

Configuring phonesets for CTI

Introduction

To enable a phoneset for CTI messages, you must configure it as an associated set (AST). This is done using Overlay 11 for digital (multiline) phonesets and Overlay 10 for analog (single-line) phonesets.

Notes:

- Phonesets can be physical phones or voice ports (including line-side E1 and line-side T1) used by Peri IVR or third-party IVR applications.
- CallPilot voice ports are configured as digital voice ports. Peri IVR voice ports are configured as analog voice ports.
- 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.

Types of phonesets

You can use Meridian Link Services for

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

Configuration of ACD phonesets is identical to configuration of Symposium Call Center Server-acquired phonesets.

To define multiline ACD phonesets as associated sets

Use LD 11. For more information about using LD 11, and creating and defining phonesets, see “Configuring agent and supervisor TNs on the switch” on page 78.

To define single-line ACD phonesets as associated sets

Use LD 10. Ensure that AST = YES and AACD = YES. For more information about using LD 10 and creating single-line ACD phonesets, see “Configuring agent and supervisor TNs on the switch” on page 78.

To define a non-ACD multiline phoneset as an associated set

Use these prompts and responses in LD 11. 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	x	AML link status message group, defined in LD 15. Set the message group to 1 to send messages to the CTI application. Set the message group to 0 if you do not want to send messages.
KEY 0	SCR xxxx	where xxxx = DN
KEY 1	TRN	Transfer
KEY 2	AO6	6-party conference
KEY 9	RLS	Release if type = SL1

Prompt	Response	Description
		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.

To define a non-ACD single-line phoneset as an associated set

Use the following prompts and responses in LD 10. 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	SD, (DD), 4D	Card density
DES	x...x	Description
CUST	0–99	Customer number
DN	xxxx	DN for the telephone number
AST	YES	Phoneset is designated as an associated set.
IAPG	Enter	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 the switch” on page 64. 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. See “Configuring CDNs on the switch” on page 64 for more information.

Connecting the host application

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

Chapter 6

Voice processing subsystem configuration

In this chapter

Overview	100
Section A: Voice processing	101
Section B: CallPilot	113
Section C: Meridian Mail	119
Section D: Third-party voice processing systems	139

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 CallPilot or Meridian Mail, or an external IVR system to provide voice processing.

Section A: Voice processing

In this section

Overview of voice processing	102
Typical uses of voice processing commands	105
Operation modes for voice processing commands	107
Resource acquisition	110
Voice port partitioning rules	111

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 CallPilot or Meridian Mail voice menu or an IVR system and are not controlled by Symposium Call Center Server until the voice processing system 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 a message to a caller

You can use any of the script language commands listed in the following section 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 CallPilot or 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 Symposium Voice Services on CallPilot or 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 CallPilot, 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. The RAN Broadcast feature in X11 release 23 allows you to connect multiple callers to the same RAN port. 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 CallPilot or Meridian Mail voice menu or an external IVR system. The voice processing system 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 Symposium Voice Services on CallPilot or 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 Symposium Voice Services on CallPilot or 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 system, and other voice processing functionality is not required

Give IVR

Use Give IVR when

- you are using CallPilot or Meridian Mail to play announcements or give voice menus to callers, and you do not want to use an ACCESS link. You usually use this command if you are migrating from Meridian CCR and you do not want to rerecord your announcements or voice menus.
- you are using CallPilot or Meridian Mail and you want to give the caller the option to leave a message in a mailbox
- you are using a third-party voice processing system 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 unless you are using Nortel Networks integrated IVR CTI applications.

Give Controlled Broadcast Announcement

Use Give Controlled Broadcast Announcement when you are using Symposium Voice Services on CallPilot or Meridian Mail, 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

- you are using Symposium Voice Services on CallPilot or Meridian Mail and customized messages must be given (for example, a caller's expected wait time)
- you are using Symposium Voice Services on CallPilot or Meridian Mail 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 (Symposium Voice Services on CallPilot or Meridian Mail)
- Give IVR (CallPilot or 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 commands specified in the previous section. However, port use is more efficient if you use the broadcast type of command, 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 system.

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 4.2, 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

Voice port partitioning rules

Introduction

Your voice processing system can provide a variety of services, including

- auto-attendant
- voice menus
- fax
- voice mail
- application services to Meridian Link, Symposium Link, Meridian (Integrated) IVR, CCR, and Symposium Call Center Server

While some of these services and applications can share ports, the Symposium Call Center Server requires a dedicated set of ports behind a dedicated ACD-DN to operate correctly. If you require other voice processing services, such as Call Answering or Voice Menus, set up a separate queue for them (see “General Meridian Mail Services” on page 112).

Voice port partitioning

Symposium Call Center Server uses two types of voice ports: IVR and ACCESS ports. To allow for proper operation of Symposium Call Center Server and the voice processing system, each ACD-DN can only contain voice ports of one type (IVR or ACCESS).

Symposium Call Center Server Give IVR Ports

IVR ports are used by the Give IVR script command. Make sure that only Symposium Call Center Server Give IVR calls arrive at an ACD-DN containing IVR ports.

Configure Give IVR ports as standard IVR ports. The TNs for the ports must be acquired by Symposium Call Center Server.

Symposium Call Center Server ACCESS Voice Ports

ACCESS ports are used by the Controlled Broadcast Announcement and Open Voice Session commands. They are available only for integrated voice processing systems, such as CallPilot or Meridian Mail.

Configure these ports as ACCESS ports, giving each port a unique ACCESS class (channel number). The TNs and Class IDs for these ports must be acquired by Symposium Call Center Server.

Set the Maximum Number of Broadcast Ports parameter (see “Configuring IVR ACD-DN global settings” on page 154) to limit the number of ports used by broadcast announcements.

Note: Once this limit is reached, Symposium Call Center Server skips this command and executes the next command in the script.

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 services 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.

Section B: CallPilot

In this section

Overview	114
Updating the CallPilot configuration	116
Updating the SDN table	117
Creating voice segments	118

Overview

Introduction

To use CallPilot as your voice processing system, you must update the CallPilot server configuration:

- Define the ELAN address of Symposium Call Center Server in the Symposium Call Center Server server ELAN IP Address field.
- Identify channels that are used to provide ACCESS and IVR services to the server in Symposium Call Center Server.
- Define SDNs for use by Symposium Voice Services.

ACCESS ports

Symposium Call Center Server controls these ports over the ACCESS link (the IP link that connects CallPilot and Symposium Call Center Server). You use these ports for Give Controlled Broadcast and Voice Sessions.

Meridian Mail supports 96 ports. CallPilot also supports 96 ports in Release 2.0. One port must be reserved for messaging; therefore, only 95 ports are available for voice services.

To use ACCESS ports, you must configure some or all of the following entities in CallPilot:

- voice ports (virtual agent TNs)
For information on configuring ACCESS voice ports in CallPilot, see “Updating the CallPilot configuration” on page 116.
- Service Directory Numbers (SDNs)
For information on creating SDNs, see “Updating the SDN table” on page 117.
- voice segments
For information on creating voice segments (voice items), see the *CallPilot Application Builder Guide*.

Non-ACCESS ports

CallPilot 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.

To use IVR ports, you must configure some or all of the following entities in CallPilot:

- voice ports (virtual agent TNs)
For information on configuring IVR voice ports in CallPilot, see “Updating the CallPilot configuration” on page 116.
- Service Directory Numbers (SDNs)
For information on creating SDNs, see “Updating the SDN table” on page 117.
- voice segments
For information on creating voice segments (voice items), see the *CallPilot Application Builder Guide*.

Updating the CallPilot configuration

Introduction

In the CallPilot Configuration Wizard, you must do the following:

- Define the CLAN address of Symposium Call Center Server in the Symposium Call Center Server server CLAN IP Address field.
- Identify channels that are used to provide ACCESS and IVR services to Symposium Call Center Server.

To update the Configuration Wizard

- 1 On the CallPilot server, start the Configuration Wizard and advance to the Switch Information page. (For detailed instructions, see “Part 3: Switch and CallPilot Server Configuration,” in the *CallPilot Installation and Configuration Guide* for your switch type.)
- 2 In Symposium Call Center Server CLAN IP Address field, enter the CLAN address of Symposium Call Center Server.
- 3 On the left side of the page, click the link for the set of channels you want to configure. The Channel Name column displays the channels on the selected link.
- 4 Click the first channel that you want to configure in the Channel Name column. The Channel Detail Information page appears.
- 5 For each TN used to provide IVR services to Symposium Call Center Server, select the IVR check box.
- 6 For each TN used to provide ACCESS services, select the ACCESS check box, and specify a class ID. The class ID is used for communication between the server and CallPilot over the ACCESS link.
Note: When you define the TN as a voice port on Symposium Call Center Server, make sure that the channel number you assign to the voice port matches the class ID for the TN.
- 7 Click Fill.
- 8 Click OK.

Updating the SDN table

Introduction

You must define Service DNs for use by Symposium Voice Services.

To update the SDN table

- 1 Start CallPilot Manager (for detailed instructions, see the *CallPilot Administrator's Guide* [NTP 555-7101-391]).
- 2 Choose System → Service Directory Number.
- 3 Click New.
- 4 In the Service DN box, enter the Symposium Call Center Server IVR and ACCESS ACD-DN numbers, as defined on the switch.
- 5 In the Application Name box, select Symposium Voice Services.
- 6 Click Save.

Creating voice segments

Introduction

You can use voice segments as building blocks to create powerful, flexible voice applications. If you are using voice processing commands (specifically, the Play Prompt element), you must define voice segments.

Creating voice segments

Use the Application Builder to create, record, and manage voice segments. For detailed instructions, refer to the *CallPilot Application Builder Guide*.

Section C: Meridian Mail

In this section

Overview	120
Channel Allocation Table	122
Voice Services DN table	123
Configuring Meridian Mail for ACCESS	125
Configuring Meridian Mail for Give IVR	132

Overview

Introduction

Meridian Mail uses two types of ports to provide voice processing services for the Symposium Call Center Server— ACCESS ports and IVR ports.

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 these ports for Give Controlled Broadcast and Voice Sessions commands.

To use ACCESS ports, you must configure some or all of the following entities in Meridian Mail:

- voice ports (virtual agent TNs)
You configure voice ports in the Channel Allocation Table (see “Channel Allocation Table” on page 122).
- Meridian Mail mailboxes
For information on creating Meridian Mail mailboxes, see “Creating Meridian Mail mailboxes” on page 128.
- voice segments
For information on creating voice segments, see “Creating voice segments” on page 129.

IVR 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.

To use IVR ports, 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 “Channel Allocation Table” on page 122.
- treatment DNs (VSDNs)
For information on configuring treatment DNs, see “Configuring VSDN entries (treatment DNs)” on page 134.
- Voice Menu service or Announcement Service
For information on configuring treatment DNs, see “Creating announcements and voice menus” on page 132.

This section provides procedures for configuring each of these entities. It also describes the Channel Allocation Table and the Voice Services DN table where you configure these entities.

Channel Allocation Table

Introduction

The Channel Allocation Table (CAT) allows you to

- view how channels (voice ports) 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 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 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 voice ports, use the default datafill settings for voice ports in the Channel Allocation Table.

Voice Services 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 Services DN (VSDN) table, which maps DN to services.

You define a VSDN for

- Meridian Mail services that are accessed through a Treatment DN (for example, Express Messaging)
- 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 switch 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 may 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.

Configuring Meridian Mail for ACCESS

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 “Configuring the ACCESS link” on page 56)
- ACCESS IVR ACD-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

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

As with all of the IVR ACD-DNs, you must configure this DN on the switch in Overlay 23 with IVR = YES (see “Configuring IVR ACD-DNs on the switch” on page 71), and Symposium Call Center Server must acquire it (see “Configuring IVR ACD-DNs on the server” on page 151).

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 Services DN table.
 - a. Select Voice Administration.
 - b. Select Voice Services Administration.
 - c. Select Voice Services-DN table.
- 2 From the VSDN table, press Add.

- 3 Enter the ACCESS DN.
- 4 Type **ACC** in the Service box.
- 5 Type **0** as the ACCESS class.
- 6 Do not enter a Revert DN.
- 7 Type an optional comment in the Comment box, 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

You must configure voice ports in the Channel Allocation Table. You must also configure voice ports as virtual agent TNs on the switch (see Chapter 4, “Switch 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 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 in 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 From the System Status and Maintenance menu, select Channel Allocation Table.
 - a. If you have a single-site system, the Channel Allocation Table appears. Go to step 4.

Creating Meridian Mail mailboxes

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 can 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

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 rennumbers 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 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 On the Symposium Call Center Server client, in 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 application script, which includes the PLAY Prompt command. For more information on scripts and script variables, refer to the *Scripting Guide*.

Configuring Meridian Mail for Give IVR

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 DN) for each voice menu or announcement
- voice ports in Channel Allocation Table

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.

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 through 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 want to use a Meridian Mail voice menu, follow the instructions in the *Meridian Mail Voice Services Application Guide* 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 212.
- 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 DN 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 DN)

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 DN and are used in Symposium Call Center Server scripts to specify which treatment the caller 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 application to the 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 applies only to the Meridian Mail voice processing system or Nortel Networks' integrated IVR CTI application.

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 Services-DN table.
 - a. Select Voice Administration.
 - b. Select Voice Services Administration.
 - c. Select Voice Services-DN table.
- 2 Click ADD to add a new entry.
 - a. Enter the treatment DN used by Symposium Call Center Server in the script command.
 - 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. Click Save.
- 3 Repeat for each Announcement or Voice Menu Service.

Configuring IVR voice ports

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

The voice ports must belong to a dedicated IVR ACD-DN, and the Channel Allocation Table must match the switch 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).

Prerequisites

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

Note: An IVR ACD-DN cannot contain both Basic and Full Service ports. If you use both types of ports, you must have at least two IVR ACD-DNs.

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 appears. Go to step 4.
 - b. If you have a multisite system, go to step 3.
- 3 Type the number of the site on which the port resides, and then press Enter.

- 4 Modify the port. For each disabled port, you can change the values in the following fields:
 - ACD DN:** The IVR ACD-DN.
 - SCN DN:** This must match the switch configuration.
 - Capability:** Full. 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.
 - Note:** You can only modify ports that have been disabled. For disabled ports, the port capability (Full or Basic) is highlighted, and the ACD-DN, SCN, and Outbound fields are underlined.
- 5 Click 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, click 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.

Section D: Third-party voice processing systems

In this section

Overview of third-party voice processing systems

140

Overview of third-party voice processing systems

Introduction

External IVR systems can connect their voice channels to the switch 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, Peri IVR, and Nortel Networks' integrated IVR CTI application, 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 switch 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. Nortel Networks' IVR CTI application uses Meridian Link to deliver the treatment DN to Peri IVR.

Switch 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 switch 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	144
Voice connection	145
Configuring CDNs on the server	150
Configuring IVR ACD-DNs on the server	151
Configuring agent phonesets on the server	152
Configuring voice ports on the server	153
Configuring IVR ACD-DN global settings	154
Defining voice segment variables	158
Defining scripts	160
Testing voice services integration when using CallPilot	162

Overview

Introduction

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

- voice connection (if you are using Symposium Call Center Server Service Update NS040206SU07S or the Symposium Call Center Server Release 4.2 Revision 5 CD)
- CDNs (for Symposium Call Center Server only)
- 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 switch and in the voice processing system (if applicable) 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*.

Voice connection

Introduction

Symposium Call Center Server Service Update NS040206SU07S and Symposium Call Center Server Release 4.2 Revision 5 CD installs Symposium Voice Services on CallPilot on your server. (For installation instructions, see the *Installation and Maintenance Guide*.) This feature allows you to use CallPilot Release 2.0 as your voice processing system. If you have installed this Service Update on your server, you must configure the voice connection to CallPilot. If you want to revert to using Meridian Mail, you must configure the connection to Meridian Mail.

To configure the voice connection, you must perform these tasks:

1. Shut down Symposium Call Center Server (if it is running).
2. Configure the CallPilot or Meridian Mail connection.

To shut down the server

- 1 From the Windows Start menu, choose Programs → Symposium Call Center Server → Shutdown.

Result: The Symposium Call Center Server Shutdown dialog box appears.

- 2 Click OK.

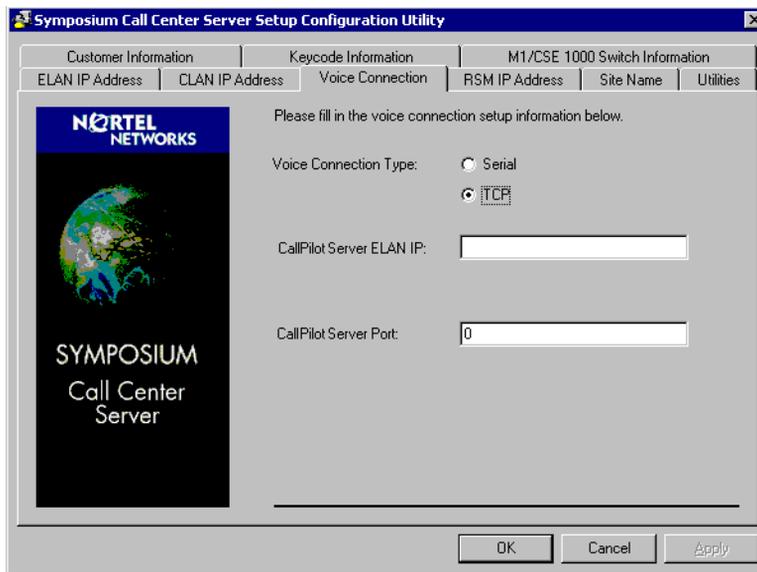
Result: The utility shuts down all services, and then the Service Status Log dialog box appears. This log displays any services that failed to shut down. Click Recheck to refresh the service statuses.

- 3 If any services are still running, use the Services control panel to shut them down manually (from the Windows Start menu, choose Control Panel, and double-click Services). Then click Recheck to update the status log.
- 4 Click Accept to exit the utility.

To configure the CallPilot connection

- 1 On Symposium Call Center Server, choose Start → Programs → Symposium Call Center Server → Server Setup Configuration.
- 2 Click the Voice Connection tab.

Result: The Voice Connection property page appears.



- 3 For Voice Connection Type, choose TCP.
- 4 Enter information into the following boxes:
CallPilot Server ELAN IP: The ELAN IP address of the CallPilot server.
CallPilot Server Port: Enter **10008**.
- 5 Click OK.

Result: The utility prompts you to verify your keycode information.

- 6 Click Yes.

Result: The utility shuts down the services and updates the Symposium Call Center Server database. This takes several minutes. Then, the utility displays the message Symposium Call Center Server Setup Configuration is completed successfully.

7 Click OK.

Result: The utility prompts you to make a Platform Recovery disk.

8 Insert a disk in the floppy drive.

9 Click Create disk.

Result: The utility displays the message `Setup Configuration has been exported and saved successfully.`

10 Click OK.

11 Click Cancel to exit the recovery disk window.

Result: The system prompts `You must reboot now to commit changes. Press OK to reboot or Cancel to stop.`

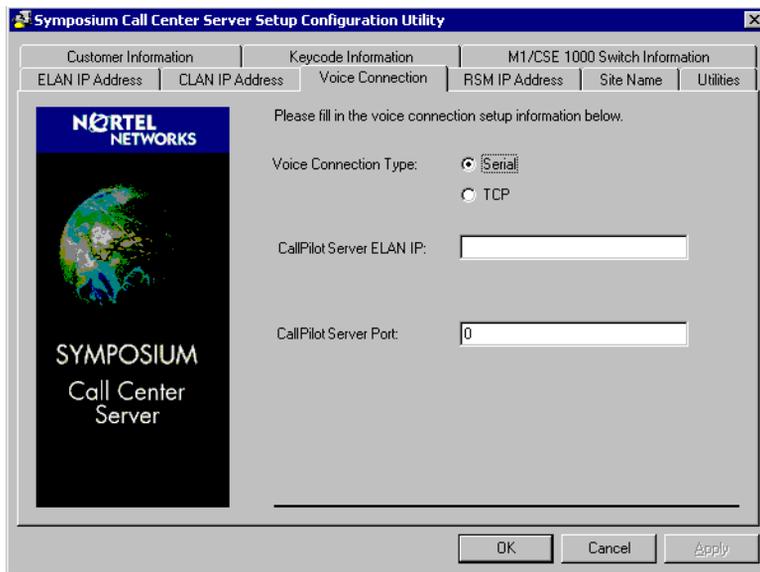
12 Click OK.

Result: The server restarts. When Symposium Call Center Server has come up, continue with the following procedures to acquire the CallPilot ACD-DNs and voice ports.

To configure the Meridian Mail connection

- 1 On Symposium Call Center Server, choose Start → Programs → Symposium Call Center Server → Server Setup Configuration.
- 2 Click the Voice Connection tab.

Result: The Voice Connection property page appears.



- 3 For Voice Connection Type, choose Serial.

Note: The CallPilot Server ELAN IP box clears, and the CallPilot Server Port box is set to 0. Do not change these values.

- 4 Click OK.

Result: The utility prompts you to verify your keycode information.

- 5 Click Yes.

Result: The utility shuts down the services and updates the Symposium Call Center Server database. This takes several minutes. Then, the utility displays the message, Symposium Call Center Server Setup Configuration is completed successfully.

- 6 Click OK.

Result: The utility prompts you to make a Platform Recovery disk.

7 Insert a disk in the floppy drive.

8 Click Create disk.

Result: The utility displays the message Setup Configuration has been exported and saved successfully.

9 Click OK.

10 Click Cancel to exit the recovery disk window.

Result: The system prompts You must reboot now to commit changes. Press OK to reboot or Cancel to stop.

11 Click OK.

Result: The server restarts. When Symposium Call Center Server has come up, continue with the following procedures to acquire the Meridian Mail ACD-DNs and voice ports.

Configuring CDNs on the server

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 switch. 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 To configure a new CDN, choose File → New.
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.
- 7 Repeat steps 1 to 6 for each CDN that you want to configure and acquire.

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

Configuring IVR ACD-DNs on the server

Introduction

Configure and acquire every switch 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 To configure a new IVR ACD-DN, choose File → New.
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.
- 7 Repeat steps 1 to 6 for each IVR ACD-DN that you want to configure and acquire.

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 on the server

Introduction

Configure and acquire phonesets (TNs), for all agents and supervisors.

The switch system types use the following TN formats:

- For Option 11 Meridian 1 systems, the TN format is “loop-0-0-unit” (for example, 8-0-0-5).
- For all other Meridian 1 system types, and the Succession CSE 1000, the TN format is “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 unchecked.
- 4 Click OK.
Result: The phoneset is added to the Phonesets window.
- 5 Select the phoneset to acquire, and then 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.
- 7 Repeat steps 1 to 6 for each phoneset that you want to configure and acquire.

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

Configuring voice ports on the server

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 on the server” on page 152.

Note: 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.

- 2 From the Voice Ports application, select the phoneset that you added as a voice port. Then continue with one of the following procedures.
- 3 Repeat steps 1 and 2 for each phoneset that you want to add as a voice port.

If the voice port is a non-ACCESS port

Choose File → Acquire to acquire the voice port.

If the voice port is an ACCESS port

- 1 Display the voice port properties (File → Properties).
- 2 Enter the Channel Number. For Symposium Voice Services on Meridian Mail, this is the ACCESS class number configured for the voice port in the Channel Allocation Table. For Symposium Voice Services on CallPilot, this is the voice application class number used when communicating with CallPilot using the AML protocol.
- 3 Click OK.
- 4 Choose File → Acquire to acquire the port.
- 5 Repeat steps 1 to 4 for each voice port you want to configure.

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.

Configuring 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 ACCESS treatment DN for the controlled broadcast and voice session script commands
- (if you are using Symposium Voice Services on Meridian Mail) the DN and password for the mailbox containing the voice files and segments used by controlled broadcast and voice session script commands

If you are using CallPilot, you must enter any value (even though the fields are not used).

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 51st 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 calls skip past the prompt and execute the next command in the script.)

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 to greater than the number of ports in this IVR ACD-DN. This allows new calls to be queued to the broadcast ports.

However, if voice ports are shared between Broadcast announcements and Voice Sessions, it may 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 calls 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 10 seconds and the appropriate setting for this parameter can vary widely from one call center to the next.

Default ACCESS treatment DN

Do not explicitly specify a treatment DN in the Open Voice Session or Give Controlled Broadcast Announcement command within a script. Use the default ACCESS treatment DN instead.

Note: The default ACCESS treatment DN must be the same as the Access IVR DN.

Meridian Mail mailbox and password

To use the Give Controlled Broadcast Announcement and Open/End Voice Session commands with Meridian Mail, 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:

- If you are using Symposium Voice Services on Meridian Mail, ensure that the Meridian Mail mailbox and password are defined on the Meridian Mail subsystem.
- Ensure that the IVR ACD-DN is defined on the switch.
- Ensure that the Treatment DN is defined on CallPilot or 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.

The screenshot shows a dialog box titled "Global Settings". It contains the following fields and values:

- Number of IVR Ports Reserved for Broadcasts: 48
- Broadcast Voice Port Wait Timer: 10
- Default Access Treatment DN: 7010
- Default Access IVR DN: 7010 (dropdown menu)
- Meridian Mail section:
 - Mailbox: 1234
 - Password: 1234

Buttons at the bottom: OK, Cancel, Help.

- 3** Enter the number of IVR ports reserved for broadcast.
- 4** In the Broadcast Voice Port Wait Timer box, specify the number of seconds that the system waits before giving a broadcast.
- 5** For Meridian Mail, in the Mailbox box, enter the DN of the Meridian Mail mailbox.
In CallPilot, this field is not used, but you must enter any two digits.
- 6** For Meridian Mail, in the Password box, enter the password required to access the Meridian Mail mailbox.
In CallPilot, this field is not used, but you must enter any four digits or use the default.
- 7** Click OK.

Defining voice segment variables

Introduction

Symposium Call Center Server scripts reference voice segments on CallPilot or Meridian Mail by using voice segment variables. Voice segment variables can contain one or more voice segments. These voice segments contain specific words or phrases recorded in the Voice Prompt Editor or Application Builder. 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

Record user-defined voice segments for CallPilot or Meridian Mail. For CallPilot, you record the voice segments using Application Builder. For Meridian Mail, 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 is a global 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 CallPilot or Meridian Mail checks for the existence of the segments on the other platform except at run time. When Symposium Call Center Server instructs CallPilot or 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 Application Builder or 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 segments representing spoken numbers. Symposium Call Center Server strings the segments 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
```

Defining scripts

Introduction

You use script commands to determine how calls are handled. The services that a particular caller hears depend 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 and 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 switch ACD-DN to which the voice port TNs belong in the switch configuration. Symposium Call Center Server directs voice processing calls to the IVR ACD-DN, and the switch ACD software distributes the calls over the voice port (that is, the switch selects the actual voice port, not the Symposium Call Center Server software).

Note: If TRDN is not configured in the switch 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

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

To use Give Music or Give RAN commands, in the Symposium Call Center Server script, you must reference a RAN or MUS route. You do not need to acquire music or RAN routes.

Testing voice services integration when using CallPilot

Introduction

If you are using CallPilot for voice services, you can use the test scripts supplied in Symposium Call Center Server to test the integration. Script CP_Test1 tests the basic voice services of GIVE IVR. Script CP_Test2 tests the advanced voice services of ACCESS. Before using the test scripts, you must create test voice items using CallPilot Application Builder and script variables in Symposium Call Center Server.

Configuration information

The following configuration information applies to the GIVE IVR and ACCESS commands, and CallPilot configuration limits:

GIVE IVR ACD-DN

- CallPilot
 - requires corresponding SDN entry, Symposium Voice Services, and Voice
 - SDN entry “Application Name = <ApplicationBuilder application>”, Media Type = Voice
- Switch
 - ACD-DN has IVR = YES, ALOG = YES
 - Agents in ACD-DN have CLS = MMA and FLXA, as well as AST on keys 0 and 1.

ACCESS ACD-DN

- CallPilot
 - does not require unique SDN entries
 - Prompts can be “filed” as in Meridian Mail.
 - SDN entry “Application Name = Symposium Voice Services”, Media Type = Voice

- Switch
 - ACD-DN has IVR = YES, ALOG = YES.
 - Agents in ACD-DN have CLS = MMA and FLXA, as well as AST on keys 0 and 1.

CallPilot limits

- Applications: 2500 maximum (500 on drive D, 1000 on drive E, and 1000 on drive F)
- Prompts per application: 3000
- SDN entries: 1500 maximum (but not enforced). Each entry requires a unique number that does not have to be a Meridian 1 dialable number if used as a Treatment DN for Symposium Call Center Server.

Preparing to test basic voice services (GIVE IVR)

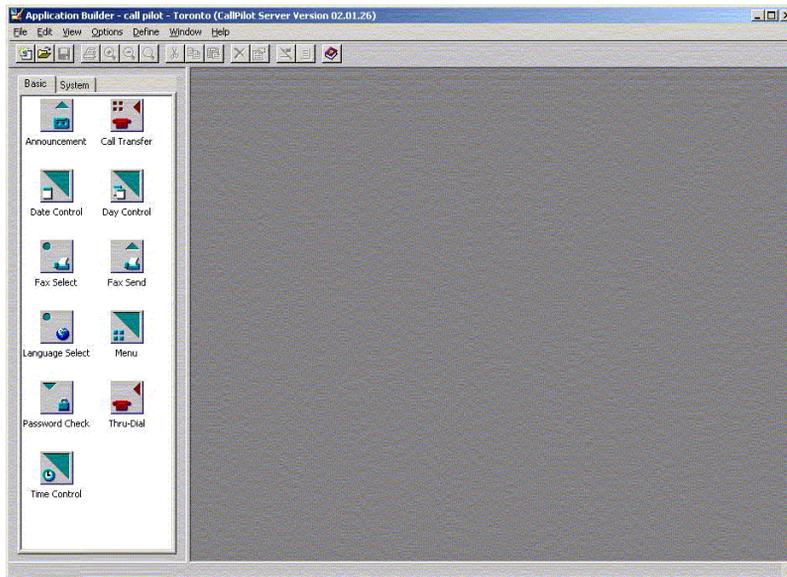
Before using test script CP_Test1 to test the GIVE IVR command, you must do the following:

- Create an announcement application using CallPilot Application Builder, and record a prompt.
- Create an SDN numerical entry point to the application name.
- Create the script variables in Symposium Call Center Server.
- Write a test script, or import the CP_Test1 script from
\\Program Files\Nortel Networks\Symposium Call Center Server\Client\en\script\samples.
- Associate the script with the test CDN in the Master Script.
- Activate and test.

To create an announcement

- 1 In CallPilot, launch Application Builder.

Result: Application Builder appears.



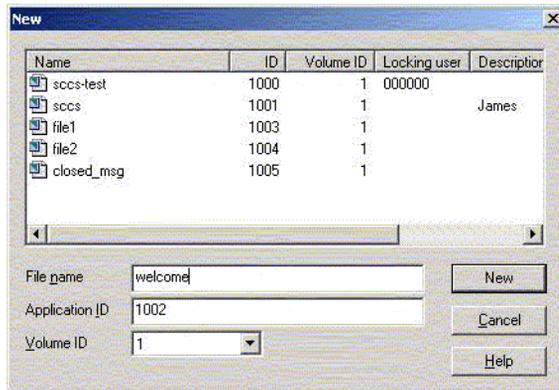
- 2 Select File → New.

Result: The New window appears and displays a list of existing folders.

- 3 In the File name box, enter a name for the application. (For CP_Test1, use **welcome**.)

Note: This is the name that appears as an option in the Application Name pull-down menu in the CallPilot Manager SDN table.

The system assigns the next available Application ID. You can change the ID by typing an ID that is not in use.



- 4 Click New.

Result: The application is created.

- 5 In Application Builder, click the Announcement Block in the pallet, and then drag the block onto the drawing board.

Result: The Add Announcement Block window appears.

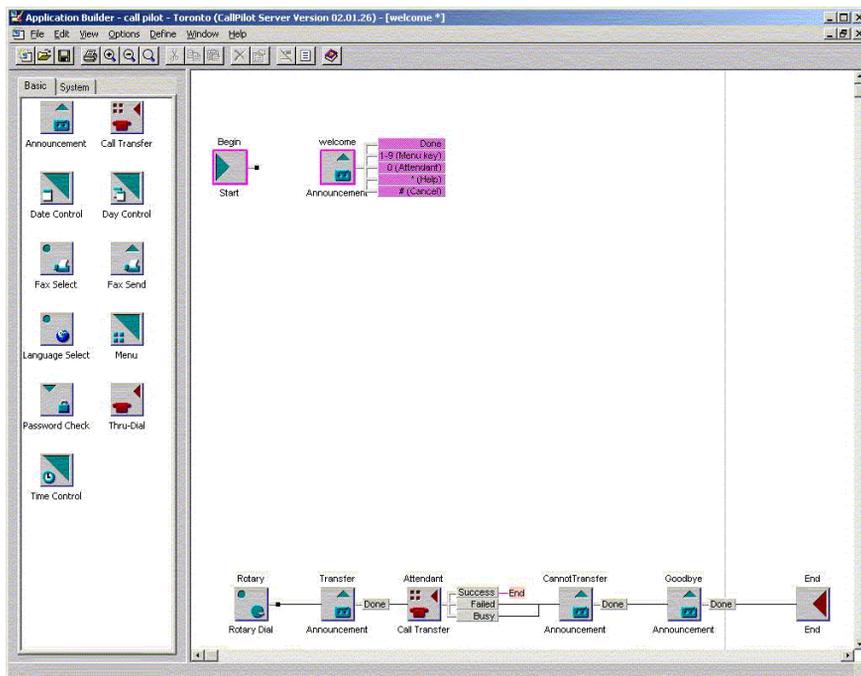
- 6 In the Enter block name box, type a descriptive name for the announcement. The name can be up to 30 characters.

Note: Create a name based on what the message will say to the caller. For CP_Test1, name the welcome message **welcome**.



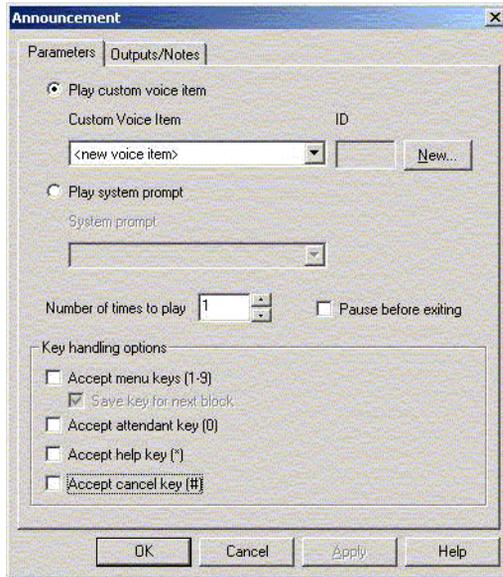
7 Click OK.

Result: The new Announcement block appears on the Application Builder drawing board.



- 8 Double-click the Announcement block.

Result: The Announcement window appears where you can define properties for the new announcement.



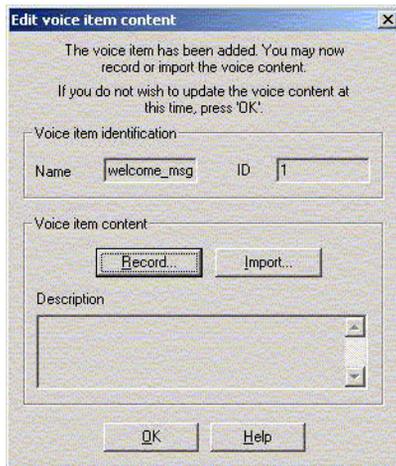
- 9 Select the Play custom voice item button, and then click New.

Result: The Add voice item window appears.

- 10 In the Name box, type a name for the voice item. (For CP_Test1, use the name **welcome_msg**.) The system assigns an ID.
- 11 In the Description box, type a description for the voice item.

12 Click Done.

Result: The Edit voice item content window appears.

**13** You can import an existing voice item, or record a new voice item.

To import an existing voice item, click Import.

Result: The Import voice item window appears, in which you can locate and select a .WAV file as a voice item.

14 To record a new voice item, click Record.

Result: The Specify Phoneset window appears.

Note: If you want to record a new item, you must have the CallPilot Desktop Player installed or an error message appears. To get the CallPilot Desktop Player, select Download Player from the right side of the main CallPilot Manager menu.

15 Type the DN of the phoneset from which you want to record.**16** Click OK.

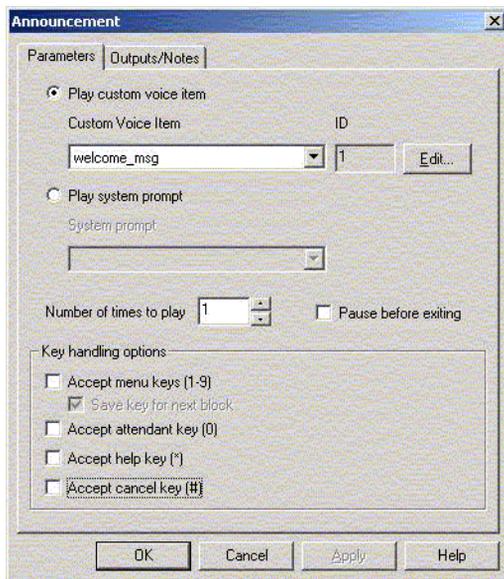
Result: The Application Builder Player window appears.

17 Click Record (red button).

Result: The phoneset rings.

- 18 Pick up the receiver, record the voice item after the tone, and then click Stop.
- 19 Click Play to review the recorded voice item. When you are satisfied, hang up the phoneset.
- 20 Click Save.
- 21 Click Close.
- 22 In the Announcement window (which is still open on your desktop), select the number of times you want the announcement to play in the Number of times to play box.

Deselect the Pause before exiting check box and all of the Key handling option check boxes.



- 23 Click OK.
- Result:** The Announcement window closes.
- 24 In the Application Builder drawing board, connect the Begin block to the Announcement block, and the Announcement block to the End block:

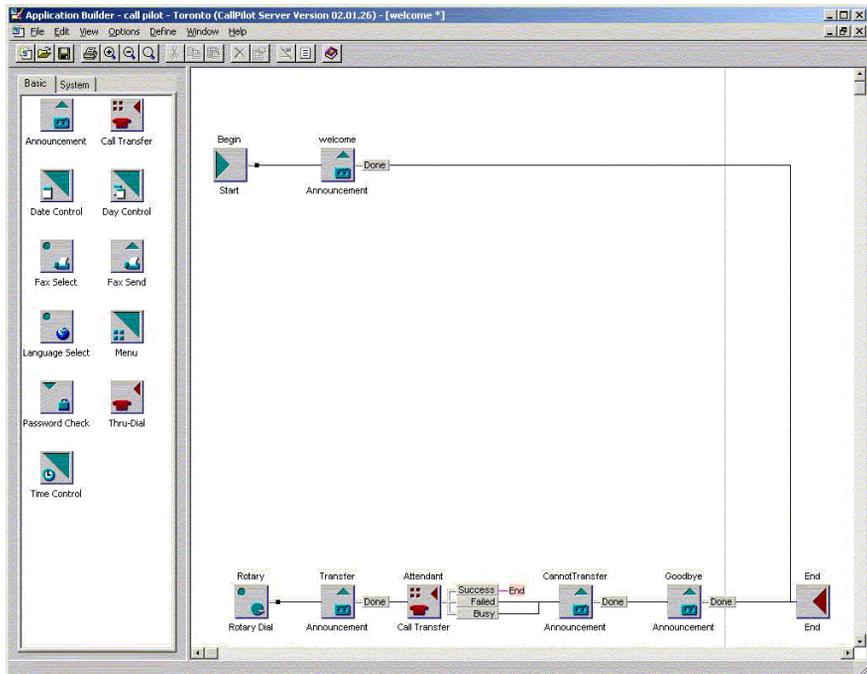
- a. Click the tip of the Begin connector (it turns red). Then, click the Announcement block.

Result: A connection line appears from the Begin block to the Announcement block.

- b. Click the Done box of the Announcement block.
- c. Click the End block.

Result: A connection line appears from the Done box of the Announcement block to the End block.

Note: When the connections are properly completed, the blocks turn from pink to black, and you can save the application.



- 25 Click Save and close Application Builder.

To create an SDN numerical entry point

- 1 In the CallPilot Manager browser window, select System → Service Directory Number.

Result: The Service Directory Number window appears.

- 2 Click New.

Result: The SDN Details window appears.

- 3 In the Service DN box, type a numerical value for the SDN (for this example, type **80000**).

Note: This value is used as the Treatment DN in the Symposium Call Center Server script.

- 4 In the Application name pull-down menu, select the name of the application (for this example, use the application name **welcome**). Leave all other fields at the default settings.

Note: The Media Type must be defined as Voice.

- 5 Click Save.

To create the script variables in Symposium Call Center Server

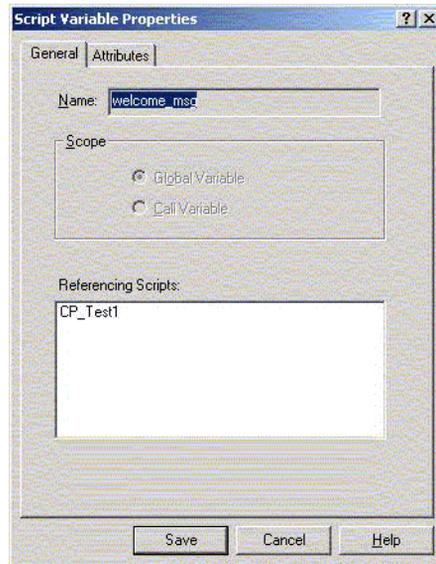
- 1 On the Symposium Call Center Server client, in the SMI window, select Script Variables.

Result: The Script Variables window appears.

- 2 Choose File → New.

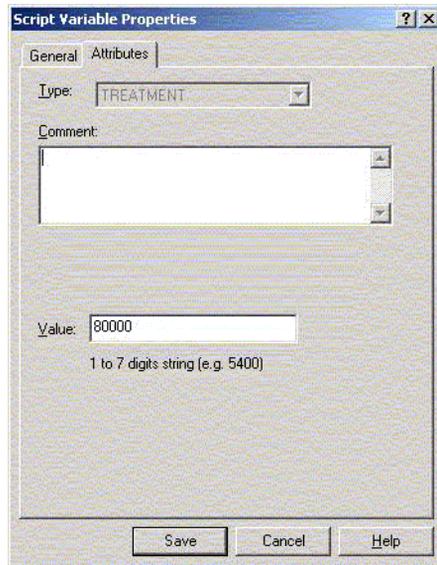
Result: The Script Variables Properties window appears.

- 3 On the General page, in the Name box, type the variable name. (For this example, use the announcement name **welcome_msg**.)



- 4 In the Scope box, select Global Variable.
- 5 Click the Attributes tab.

- 6 In the Type drop-down list, select TREATMENT.



- 7 (Optional) In the Comment box, type a comment.

Note: It is a useful practice to put the text of the prompt in the Comment field for future reference.

- 8 In the Value box, type the corresponding SDN numerical value (the SDN numerical value that you created for the announcement, which for this example is **80000**).
- 9 Click Save.

Result: The variable is defined.

To import the test script

Next, you create a test script, or import test script CP_Test1 provided by Symposium Call Center Server. For more information on test script CP_Test1, see "CP_Test1" on page 183.

For more information on scripts, see the *Scripting Guide*.

- 1 To import test script CP_Test1, in the SMI window, select Call Flow Administration → Scripts.

Result: The Scripts Manager appears.

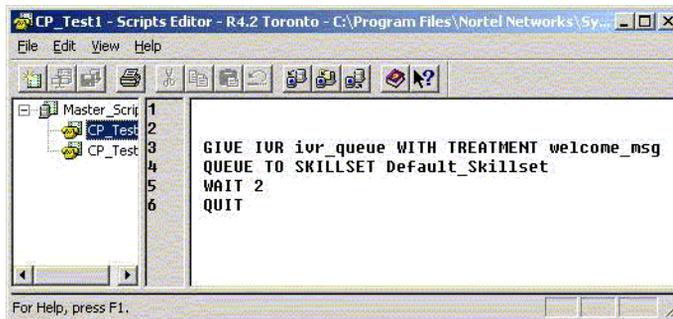
- 2 Choose File → New.

Result: The Scripts Editor appears.

- 3 Choose File → Import, and browse to find CP_Test1.

Note: CP_Test1 is located in
\\Program Files\Nortel Networks\Symposium Call Center Server\Client\en\script\samples.

Result: The test script appears.



Testing the GIVE IVR application

To complete the test of GIVE IVR, do the following:

1. Validate and activate the test script.
2. Associate the script with the test CDN in the Master Script.
3. Activate the Master Script. (See “Master Script” on page 183.)
4. Call the test CDN.
5. Verify that you hear the test message.

If you do not hear the test message, verify that the configuration is correct.

Preparing to test advanced voice services (ACCESS)

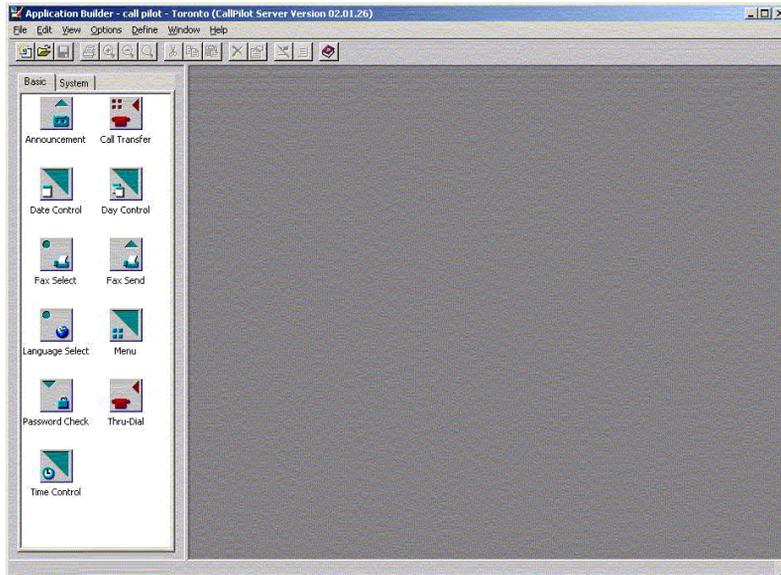
Before using test script CP_Test2 to test the ACCESS command, you must do the following:

- Create an application using CallPilot Application Builder.
- Create new applications (folders) to hold the voice prompts used for OPEN VOICE SESSION and CONTROLLED BROADCAST ANNOUNCEMENT.
- Record the prompts.
- Create the script variables in Symposium Call Center Server.
- Write a test script, or import the CP_Test2 script from \Program Files\Nortel Networks\Symposium Call Center Server\Client\en\script\samples.
- Associate the script with the test CDN in the Master Script.
- Activate and test.

To define ACCESS prompts

- 1 In CallPilot, launch Application Builder.

Result: Application Builder appears.



- 2 Select File → New.

Result: The New window appears and displays a list of existing applications (folders).

- 3 In the File name box, enter a unique name for the application. (For CP_Test2, use **sccs_prompts**).

Note: The system assigns the next available Application ID. You can change the ID by typing an ID that is not in use.

- 4 Click New.

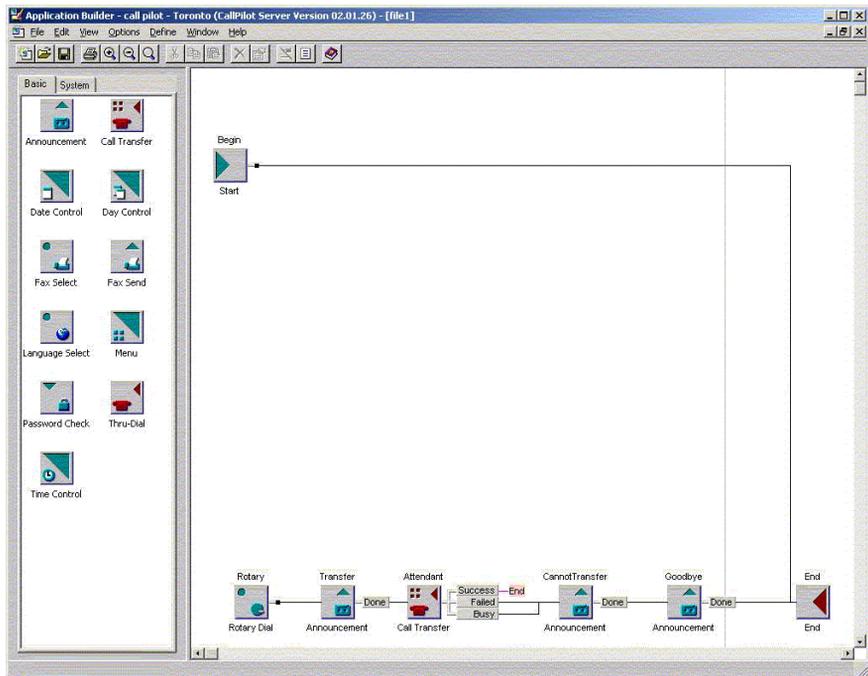
Result: The application is created.

- 5 On the Application Builder drawing board, click the tip of the BEGIN connector.

Result: The connector turns red.

- 6 Click the END block.

Result: A line is drawn from the BEGIN block to the END block. All of the lines turn black.



- 7 Click Save.

- 8 Select Define → Voice Items.

Result: The Define voice items window appears.

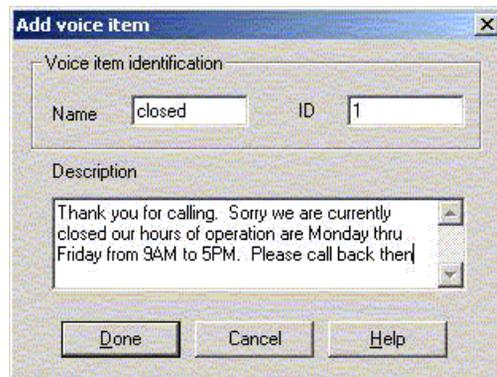
- 9 To add a voice item, click Add.

Result: The Add voice item window appears.

- 10 In the Name box, type a name for the voice item (for this example, type **closed_message_vs**).

The system assigns an ID number.

- 11 In the Description box, type a description of the voice item.



- 12 Click Done.

Result: The Edit voice item content window appears.

- 13 You can record or import voice content for each voice item as you define it, or you can continue defining all of the voice items and then record or import the voice content later.

To record the content for a voice item, click Record and follow the same steps you used in the GIVE IVR procedure, starting at Step 14 on page 168.

- 14 When you are finished recording or importing the voice item, click Done.
- 15 Repeat the procedure for adding and recording a voice item, starting at Step 8 on page 177. For this example, add and record voice item **hold_option_vs**.
- 16 In Application Builder, click Save and close Application Builder.

To define voice segment variables in Symposium Call Center Server

All voice prompts must be stored in voice segment variables. To use test script CP_Test2, you must define two voice segment variables, **closed_message_vs** and **hold_option_vs**.

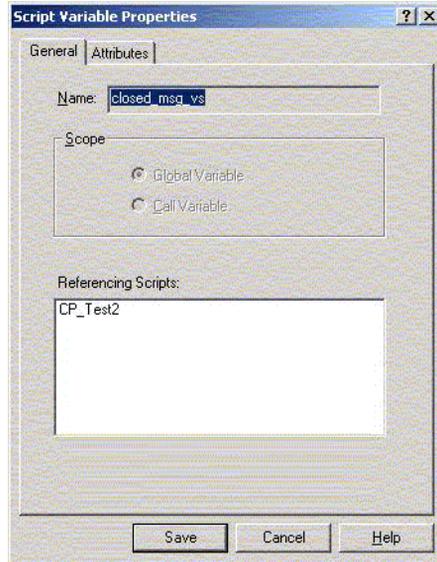
- 1 On the Symposium Call Center Server client, in the SMI window, select Script Variables.

Result: The Script Variables window appears.

- 2 Choose File → New.

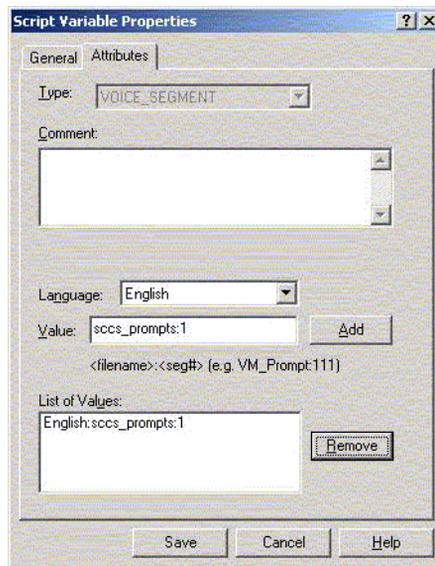
Result: The Script Variables Properties window appears.

- 3 On the General page, in the Name box, type the variable name. (Use the name of the voice item. For this example, type **closed_msg_vs.**)



- 4 In the Scope box, select Global Variable.
- 5 Click the Attributes tab.

- 6 In the Type drop-down list, select VOICE_SEGMENT.



- 7 (Optional) In the Comment box, type a comment.
Note: It is a useful practice to put the text of the prompt in the Comment field for future reference.
- 8 From the Language drop-down list, select the required language.
- 9 In the Value box, type the application name, a colon for the separator, and the ID. (Use the application name you created in Application Builder and the assigned ID, which for this example is **sccs_prompts:1**.)
- 10 Click Add.
Result: The value is added to the List of Values box.
- 11 Click Save.
Result: The variable is defined.
- 12 To define the second variable required to run CP_Test2, repeat the procedure and create **hold_option_vs**.

To import the test script

Next, you create a test script or import test script CP_Test2 provided by Symposium Call Center Server. For more information on script CP_Test2, see “CP_Test2” on page 184.

For more information on scripts, see the *Scripting Guide*.

- 1 To import test script CP_Test2, in the SMI window, select Call Flow Administration → Scripts.

Result: The Scripts Manager appears.

- 2 Choose File → New.

Result: The Scripts Editor appears.

- 3 Choose File → Import, and browse to find CP_Test2.

Note: CP_Test2 is located in
\\Program Files\\Nortel Networks\\Symposium Call Centre Server\\Client\\en\\script\\samples.

Result: The test script appears.

```
1 IF OUT OF SERVICE Default_Skillset THEN
2   GIVE CONTROLLED BROADCAST ANNOUNCEMENT
3   PLAY PROMPT VOICE SEGMENT closed_msg_vs
4   DISCONNECT
5 END IF
6
7
8 QUEUE TO SKILLSET Default_Skillset
9 WAIT 2
10 SECTION WaitLoop
11   WAIT 20
12
13 OPEN VOICE SESSION
14   PLAY PROMPT VOICE SEGMENT hold_option_vs
15   COLLECT 1 DIGITS INTO hold_choice_cv
16 END VOICE SESSION
17
18 IF hold_choice_cv = 1 THEN
19   EXECUTE WaitLoop
20 ELSE
21   DISCONNECT
22 END IF
23
```

For Help, press F1.

Testing the ACCESS application

To complete the test of advanced voice services ACCESS, do the following:

1. Select the default ACD-DN queue for ACCESS treatment in the Global Settings located in the IVR ACD-DN application in the system tree. (File → Global Settings)
2. Validate and activate the test script.
3. Associate the script with the test CDN in the Master Script.
4. Activate the Master Script. (See “Master Script” on page 183.)
5. Call the test CDN.
6. Verify that you hear the test message.

If you do not hear the test message, verify that the configuration is correct.

Scripting information

The following information explains how prompts and applications are referenced in the test scripts:

- When you create an Application Builder application, you define a filename/application name. Application Builder assigns an application-ID for the application and a segment-ID for each segment flow that you create in your application.
- In Symposium Call Center Server scripts, when you want to access a voice segment (over a Symposium Call Center Server ACCESS port), you do so through a global variable and specify the value as **application-name:segment-ID**, as defined in Application Builder. (In CallPilot, the application name is the same as the filename when using Meridian Mail and Voice Prompts. The application name is case-sensitive.)
- For the Symposium Call Center Server IVR port, you reference the Application Builder application (either directly or through a global variable) using a treatment number that corresponds to a CallPilot SDN service-DN entry that you defined on CallPilot. The CallPilot service-DN links to the Application Builder application that you created. You may define the service-DN to be the same as the Application ID assigned by Application Builder as long as there is no numbering conflict.

Sample scripts

Master Script

```
/* Add the test CDN to the existing MASTER SCRIPT. To test
Basic Voice Services using the GIVE IVR command, use
CP_Test1. To test advanced voice services using the Open
Voice Session and Controlled Broadcast Announcements, use
CP_Test2. All configuration must be complete on Meridian 1,
CallPilot, and Symposium Call Center Server before testing.
Variables must be created and Scripts must be imported,
saved, and validated. Then, add the commands below to the
Master Script and activate.
```

Variable name	Class	Type	Value	Comment
test_cdn	Global	CDN	xxxx	CDN value used for testing

```
*/
IF CDN = test_cdn THEN
EXECUTE SCRIPT CP_Test1 /* Tests the GIVE IVR
functionality, use CP_Test2 to test advanced voice services
*/
END IF
```

CP_Test1

```
/* This simple script tests the GIVE IVR command using
CallPilot. For the Symposium Call Center Server IVR port
you reference the Application Builder application (either
directly or through a global variable) using a treatment
number that corresponds to a CallPilot SDN service-DN entry
that you defined on CallPilot. The CallPilot service-DN
links in the Application Builder application that you
created. By convention you define the service-DN to be the
same as the Application ID assigned by Application Builder
(just for consistency). Assign a test agent to the default
skillset and log the agent in. At the GIVE IVR command, you
```

should hear the IVR speak a simple welcome greeting and then be queued to the default skillset. Answer the call.

Variable name	Class	Type	Value	Comment
ivr_queue	Global	ACD	xxxx	Symposium Voice Services DN in CallPilot (dedicated ACD-DN for GIVE IVR)
welcome_msg	Global	Treatment DN	xxxx	Announcement/voice menu DN in CallPilot SDN table

*/

```
GIVE IVR ivr_queue WITH TREATMENT welcome_msg
QUEUE TO SKILLSET Default_Skillset
WAIT 2
QUIT
```

CP_Test2

```
/* If the desired skillset has no agents logged in, the caller receives a closed broadcast message and is disconnected. If the skillset is staffed, the caller is queued to the default skillset. After the call has waited for 20 seconds, the caller is asked to press 1 if the caller wants to keep on holding. If the caller selects 1, the caller remains queued, otherwise the caller is disconnected. Make sure that the default queue is set to the Symposium Voice Services DN in CallPilot (dedicated ACDN for ACCESS). To access a voice segment (over a Symposium Call Center Server ACCESS Port), you do it through a global voice segment variable that has as its value the application-name:segment-ID, as defined in Application Builder
```

*/

Global Variable List:

Variable name	Class	Type	Value	Comment
access_queue	Global	ACD	xxxx	Symposium Voice Services DN in CallPilot (dedicated ACD-DN for ACCESS)
closed_message_vs	Global	Voice segment		Application-name:voice item-ID needs to be created and recorded in CallPilot
hold_option_vs	Global	Voice segment		Application-name:voice item-ID needs to be created and recorded in CallPilot
hold_choice_cv	Call	DN	0	

```
*/
```

```
IF OUT OF SERVICE Default_Skillset THEN
```

```
GIVE CONTROLLED BROADCAST ANNOUNCEMENT
```

```
PLAY PROMPT VOICE SEGMENT closed_message_vs
```

```
DISCONNECT
```

```
END IF
```

```
QUEUE TO SKILLSET Default_Skillset
```

```
WAIT 2 /* Allow time in case an agent is available */
```

```
SECTION WaitLoop
```

```
WAIT 20
```

```
OPEN VOICE SESSION
PLAY PROMPT VOICE SEGMENT hold_option_vs
COLLECT 1 DIGITS INTO hold_choice_cv
END VOICE SESSION
IF hold_choice_cv = 1 THEN
EXECUTE WaitLoop
ELSE
DISCONNECT
END IF
```

Chapter 8

Troubleshooting

In this chapter

Overview	188
Running Meridian Link traces	189
Subsystem link problems	191
Ports become deaquired	206
Call routing or treatment problems	208
Frequently asked questions	215

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 switch, Meridian Link Services, and voice processing for use with Symposium Call Center Server, refer to Chapter 2, “Configuration overview.”

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 called `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\ml_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 then 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 then 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 `..\Nortel\ICCM\bin` directory.

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 the switch 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 .166 .111 .14 LXR7: ACTIVE  EMP-
TY  APPL ACTIVE
ELAN #: 17
  APPL_IP_ID: 47 .166 .111 .13 LXR7: ACTIVE  EMP-
TY  APPL ACTIVE
```

CSL between the switch 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

ACCESS Link between server and CallPilot

- CallPilot: From System Utilities → Support Tools → CallPilot Processing Utilities → Trace Viewer <nbtview>, in Trace Control, select
 - MLink_Trace for messages on MLink
 - NBAPE for messages on ACCESS Link
- Symposium Call Center Server: From Start/Run, enter **tsm_oam**, and then select option 3.
 - For VSM and MLSM session traces:
 - From the OAM menu, select option 2, and then enter 0 at the prompt.
 - Note the Session ID for VSM_Service, MLink SP (CallPilot Application). Press Return to go back to the OAM menu.
 - Select option 5, enter the Session ID, and then respond to the prompts as appropriate.
 - For AML trace:
 - From the OAM menu, select option 7.
 - From the AML Trace menu, select option 4.

- For Access Protocol trace:
From the OAM menu, select option 9. Select option 3 to enable the trace.
- For Access Protocol Debug trace:
From the OAM menu, select option 10. Select option 3 to enable the trace.

Switch loop, shelves, and cards

- Switch: 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 or CallPilot voice ports should always show Logged in. If not, disable and enable the port on Meridian Mail or CallPilot 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).

CallPilot ports enabled

- CallPilot
 - On the CallPilot client, in the CallPilot Manager, select Channel Monitor link.

CDN acquired

- Client: CDNs status must show Acquired.
- Switch: 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 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.
- Switch: LD 23; REQ PRT; TYPE ACD
- The AACQ prompt in the printout is YES; the ASID prompt is equal to the ELAN connected to 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.
- CallPilot
 - On the CallPilot client, in the CallPilot Manager, select Channel Monitor link.
- Switch: LD 20; REQ TNB; TYPE SL1; (for CallPilot) TYPE=2008
- The ACQ AS prompt in the printout is TN; the ASID prompt is equal to the ELAN connected to Symposium Call Center Server.

Example (Meridian Mail)

```

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
    
```

Example (CallPilot and Symposium Call Center Server)

```

DES CLPLT
TN 024 1 13 26
TYPE 2008
CDEN 8D
CTYP XDLC
CUST 0
FDN
TGAR 1
LDN NO
NCOS 3
RNPG 0
SCI 0
SSU
XLST
SCPW
    
```

```

CLS CDT ...
CPND_LANG ENG
HUNT
SPID NONE
AST 00 01
IAPG 0
AACS YES
ACQ AS: TN,AST-DN,AST-POSID
ASID 16
SFNB 1 2 3 4 5 6 11 12 13 18 22
SFRB
USFB 1 2 3 4 5 6 7 9 10 11 12 13 14 15
CALB 0 1 2 3 4 5 6 8 9 10 11 12
FCTB
ITNA NO
DGRP
PRI 01
DNDR 0
DTMK
KEY 00 ACD 5990 0 5356
      AGN
      01 SCN 5386 0      MARP
      CPND
          NAME CallPilot
          XPLN 27
          DISPLAY_FMT FIRST, LAST
02 MSB
03 NRD
04 TRN
05 AO3
06
07

```

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.
- CallPilot

- On the CallPilot client, in the CallPilot Manager, select Channel Monitor link.
- Switch: LD 20; REQ TNB; TYPE SL1; (for CallPilot) TYPE=2008

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 134 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 is 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

Treatment DN in CallPilot

The treatment DN maps to the name of an application created in Application Builder. The information appears on the SDN page of the CallPilot Manager.

Matching IVR ACD-DNs on switch, Symposium Call Center Server, and voice processing system

- The following appearances of ACD-DNs must match:
 - Meridian Mail Channel Allocation Table
 - CallPilot Channel Information page on the CallPilot Manager
 - switch 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

- Script:
 - Give Controlled Broadcast 7725

#	CDP	TN	ACN-DN	SCN	Type	Capability
21	2-1-1	004-0-03-00	7725	4500	Voice	FULL Basic ACC Class: 41

- Switch: 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
```

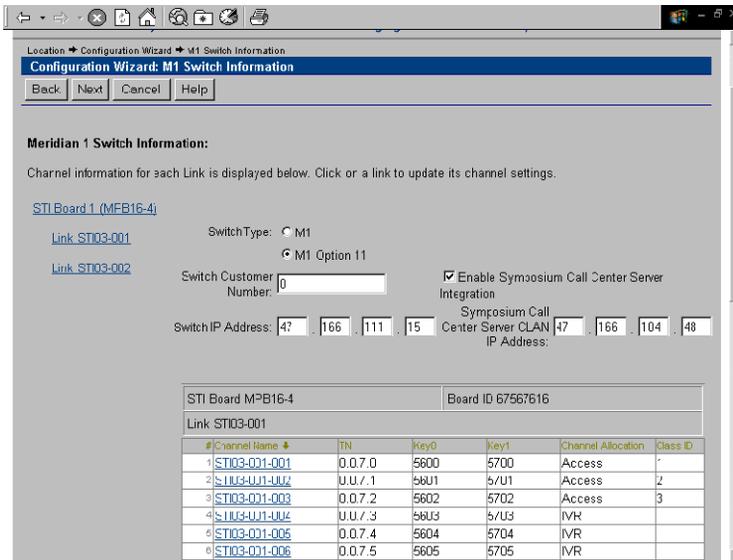
- Client: IVR ACD-DN number status shows Acquired.

Note: In Meridian Mail, ensure that you have configured the ACCESS IVR ACD-DN in the VSDN table. In CallPilot, ensure that you have configured the ACCESS IVR ACD-DN in the Service DN table on the CallPilot Manager.

Matching voice port TNs on switch, Symposium Call Center Server, and Meridian Mail or CallPilot

- The TN configurations belonging to the ACD-DN (see previous section) must match:
 - Meridian Mail Channel Allocation Table
 - CallPilot Channel Information page on the CallPilot Manager
 - switch DN
 - IVR ACD-DN acquired by Symposium Call Center Server
- CallPilot: Configuration Wizard → CallPilot Manager → Channel Information page → TN column

Example



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

Example

#	CDP	TN	ACN-DN	SCN	Type	Capability
21	2-1-1	004-0-03-00	7725	4500	Voice	FULL Basic ACC Class: 41

- Switch: LD 22; REQ DNB

Example

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

DES

```

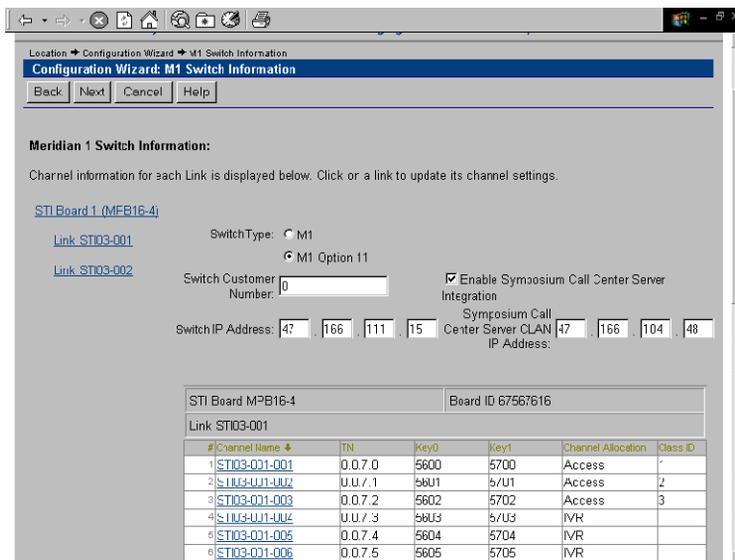
DN      7725
TYPE   MCDN
MCID   4500  TN 004 0 03 00
MCID   4501  TN 004 0 03 01
    
```

- 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 or CallPilot

- Channel number (the number shown after the “Class” keyword in the Meridian Mail Channel Allocation Table) for a specific TN on Meridian Mail or CallPilot must match the channel number for the same TN in the client display.
- CallPilot: Configuration Wizard → CallPilot Manager → Channel Information page → Class ID column

Example



- Meridian Mail: Menus

- 5 System Status and Maintenance
- 4 Channel Allocation Table

Example

#	CDP	TN	ACN-DN	SCN	Type	Capability
21	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.

Ports become deaquired

Introduction

Calls other than those directed by the Symposium Call Center Server script can arrive at the same voice port. If non-Symposium Call Center Server calls are terminating on shared voice ports, use the following list of suggestions to check. 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 switch IDC tables for directing incoming calls to the ACD-DN owning the voice ports.
- Examine switch trunk auto-terminate destinations for directing incoming calls to the ACD-DN owning the voice ports.
- Examine switch ACD-DN Night Call Forward destinations for directing incoming calls to the ACD-DN owning the voice ports.
- Examine switch, set Call Forward (No Answer, Busy, and All Calls) destinations for directing incoming calls to the ACD-DN owning the voice ports.
- Examine switch 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 may receive error events indicating that calls 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 the switch with IVR = YES.
- There are voice ports logged on and idle (check switch and Meridian Mail or CallPilot) 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 Symposium Call Center Server IVR ACD-DN global settings configuration (check the password by changing it to VM class of service, and dial in to the mailbox directly).
- In Meridian Mail and Symposium Call Center Server integration, voice segment numbers and variables are valid and recorded.

In CallPilot and Symposium Call Center Server integration, voice items and variables are valid and recorded.

- In Meridian Mail, the 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).

In CallPilot, on the Channel Monitor page, ensure that the ACCESS voice ports are idle, not active.

- The switch 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 that the following elements are in place:

- The switch and Symposium Call Center Server configurations match exactly: the TNs configured for the IVR ACD-DN on switch 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 CallPilot, or move them to a different ACD-DN on the switch and Meridian Mail or CallPilot.
- Check whether non-Symposium Call Center Server calls are arriving at this IVR ACD-DN. For more information, see “Ports become deaquired” on page 206.

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

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

- In Meridian Mail, the DSP Port Status menu shows the channels in the Active state.
In CallPilot, on the Channel Monitor page, ensure that the ACCESS voice ports are idle, not active.
- There are no extra TNs on the switch 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 the switch and Meridian Mail or CallPilot.

- The ACCESS link is up.
- In Meridian Mail, the 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).
In CallPilot, on the Channel Monitor page, ensure that the ACCESS voice ports are idle, not active.

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 or CallPilot indicated by the treatment DN contains the correct recording.
- In Meridian Mail, if the script statement does not contain a treatment DN, check the switch default treatment DN and the treatment that is mapped to Meridian Mail's VSDN Table (and what is recorded for that).
In CallPilot, the default treatment DN on the switch should be listed in the SDN table.

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 or CallPilot is correct. In Meridian Mail, you can use the Voice Prompt Editor to play the voice segments. In CallPilot, check the file and voice items on Application Builder.

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 or CallPilot) 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, may 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 caller 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 10 seconds. When a single call is made, the software waits 10 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.
- In Meridian Mail, the Mailbox specified on Symposium Call Center Server is the correct one. (In CallPilot, you can have any input.)
- In Meridian Mail, the Mailbox password on Symposium Call Center Server is correct and has not expired. (In CallPilot, you can have any input.)
- 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).

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 or CallPilot 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 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 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 switch may be out of synchronization with the switches at other sites.

Check the time stamp of your local switch 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 switch console.
- 2 Open a Command Prompt and type **ld 2**, and then press Enter.
- 3 Type **ttad** at the prompt, and then press Enter.

Result: The current date and time appear in the format DD-MM-YYYY 00:00.

- 4 To change the type, follow these steps:
 - a. Type **stad**.
 - b. Enter the correct date and time.
- 5 Log off the switch console.

Symposium Call Center Server and the switch are not communicating

If Symposium Call Center Server and the switch are not exchanging information, the network traffic volume over the ELAN may be excessive. You must ensure that the network is designed to support the 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:

1. The Controlled Broadcast Play Prompt has a *continuous* option, which is not applicable to the Open/End Voice Session statement.
2. The Open/End Voice Session Play Prompt has a *no-type-ahead* option, which is not applicable to the Controlled Broadcast statement.
3. 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 134, 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 you can control what a voice call hears—similar to the concept of a speech path on the switch.
- On Meridian Mail and Call Pilot, a channel is a port attribute (a TN configured with ACCESS class of service and a Class number assigned to it). On the switch, a channel is a voice port TN.
- Meridian Mail or CallPilot 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, “Switch 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 chapter

Overview	220
Display Waiting Calls key/lamp	221

Overview

Introduction

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

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 or CSE 1000 ACD environments.

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

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 switch 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 switch 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 (such as 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

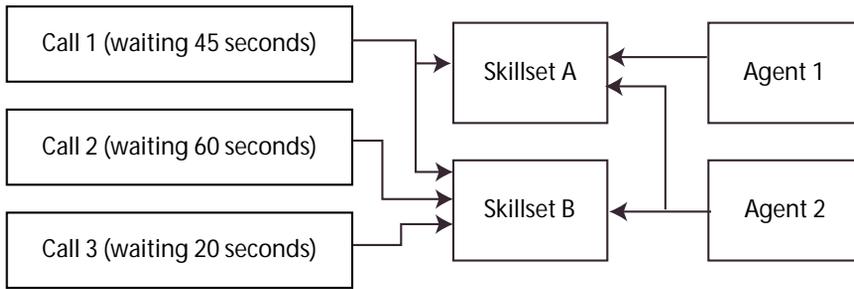
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	>	39	40
1	n	n	2	3		n	n	1	7		n	1	6	5			>		

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 switch. 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 switch 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 switch.

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

Appendix A

Migrating from Meridian Mail to CallPilot

In this appendix

Overview	228
Section A: Migrating from Meridian Mail to CallPilot	231
Section B: Backing out of the migration	263
Section C: Comparison of VPE and Application Builder	275

Overview

Introduction

This appendix contains the following information:

- procedures for migrating from Symposium Voice Services on Meridian Mail to Symposium Voice Services on CallPilot
- procedures for backing out of the migration (reverting from Symposium Voice Services on CallPilot to Symposium Voice Services on Meridian Mail), if the migration is unsuccessful
- description of the differences between the Voice Prompt Editor and Application Builder

As you work through the voice prompt migration steps, “Step 2. Migrate Meridian Mail voice prompts and voice segments” on page 244 is documented in detail in Appendix B, “Migrating prompts and segments from Meridian Mail to CallPilot” on page 279.

Comparison of CallPilot and Meridian Mail voice services

Functionality of Symposium Voice Services on CallPilot is the same as Symposium Voice Services on Meridian Mail, except as shown in the following table:

Feature	Meridian Mail	CallPilot
Call processing control	Meridian Mail uses the serial X.25 AML link.	CallPilot uses the TCP/IP and MLS protocols on the CLAN.
Voice services control	Meridian Mail uses the serial ACCESS link.	CallPilot uses the TCP/IP and ACCESS protocols over the ELAN.

Feature	Meridian Mail	CallPilot
Managing voice prompts	You use the integrated Voice Prompt Editor to administer and edit voice prompts.	You use CallPilot Application Builder to administer and edit voice prompts. Application Builder is shipped with CallPilot, but you must install it separately. Note: To edit segment length, you must use a third-party application.
Voice segment storage	Voice segments are stored in a mailbox; access is controlled with a password.	Voice segments are stored in a folder; access is controlled by the Application Builder logon.
Voice segment length	Voice segments cannot exceed 2 minutes.	Voice segments cannot exceed 10 minutes.
Voice segment deletion	When a segment is deleted, the IDs of all subsequent segments are renumbered consecutively.	Segment IDs do not change when segments are deleted.
Voice prompt migration	Not applicable	When you migrate voice segments to CallPilot from Meridian Mail, the segment name is preserved. The title is the item that is concatenated with the description and appears as '[Title] Description'.
Front-end IVR robustness	Meridian Mail ACD-DN night call forward (NCFW) to Symposium Call Center Server CDN.	CallPilot default ACD-DN NCFW to Symposium Call Center Server CDN.

Feature	Meridian Mail	CallPilot
Maximum capacity	Meridian Mail supports 96 ports. (One port must be reserved for messaging. Therefore, only 95 ports are available for voice services.)	CallPilot supports 96 ports in Release 2.0. (One port must be reserved for messaging. Therefore, only 95 ports are available for voice services.)

Section A: Migrating from Meridian Mail to CallPilot

In this section

Overview of migration	232
Pre-migration checklist	233
Migration checklist	238
Migration procedure	240

Overview of migration

Introduction

If you are currently using Symposium Voice Services on Meridian Mail as your voice processing system, and you want to use Symposium Voice Services on CallPilot, use the procedures in this section to migrate from Meridian Mail to CallPilot.



CAUTION

Risk of migration failure. Use the Migration checklist.

To complete the migration successfully, you must follow the migration checklist (see page 238). This checklist lists the steps you must perform, and indicates the order in which you must perform them.

Gradual migration

Rather than migrating all of your voice ports at one time, in some cases you can choose to do a staged migration. The following are the migration conditions:

- If you have a combination of ACCESS and IVR ports, you can migrate your ACCESS ports to CallPilot, and continue to use Meridian Mail to service your IVR ports.
Note: Symposium Call Center Server supports only one ACCESS link; therefore, you must migrate all of your ACCESS ports at the same time.
- If all of your ports are IVR, you can migrate some ports to CallPilot, and continue to use Meridian Mail to service the other ports.
- If all of your ports are ACCESS, you must migrate all ports to CallPilot at the same time (no staged migration).

Pre-migration checklist

Before you begin the migration, complete the following checklist.

✓	Description
	<p>Review all the voice processing scripts and check that the voice processing commands are using variables instead of constants. (This will be helpful if you have to back out of the migration process.)</p> <p>For example:</p> <ul style="list-style-type: none"> ■ If ACCESS scripts use a constant number as the ACCESS ACD-DN, consider using a global variable that can be modified to represent a CallPilot ACCESS ACD-DN or a Meridian Mail ACCESS ACD-DN. ■ If treatment DN's are constant numbers, consider using variables of type Treatment DN.
	<p>CallPilot Release 2.0 is installed and operational. CallPilot ELAN IP address: _____</p>
	<p>Meridian Mail to CallPilot Migration package (order number NTUB24AC/A0886190) is available for installation. The package consists of the following components:</p> <ul style="list-style-type: none"> ■ CP 2.0 Documentation CD-ROM Package (NTRG19AF/A0820778) ■ Meridian Mail to CallPilot Migration Tape (NTUB25AA/A0992892)
	<p>Meridian Mail hardware platform meets the requirements. See "Preparing Meridian Mail for migration" on page 285.</p>
	<p>Meridian Mail 11 or greater is installed and operational.</p>
	<p>Meridian Mail tape drive hardware and media meet the requirements. See "Tape drive hardware and media requirements" on page 236.</p>

	CallPilot Application Builder is installed and operational in the network.
	Symposium Call Center Server Release 4.2 is installed and operational. Server CLAN IP address: _____
	Symposium Call Center Server Service Update NS040206SU07S or Symposium Call Center Server Release 4.2 Revision 5 CD is available for installation.
	<p>Data that cannot be migrated has been identified. See “Data that cannot be migrated” on page 283.</p> <p>In the <i>Meridian Mail to CallPilot Migration Utility Guide</i>, review the sections</p> <ul style="list-style-type: none"> ■ Section A: “Understanding what can be migrated to CallPilot” ■ Section B: “Migration strategies”
	<p>ACCESS mailboxes are checked to ensure that</p> <ul style="list-style-type: none"> ■ the mailbox DNs are 3 or more digits in length ■ the DNs are unique across all mailboxes
	<p>Meridian Mail Class of Service (COS) names are unique across all classes of service.</p> <p>Note: Ensure that the mailbox that contains the ACCESS prompts does not have personal COS. If you are migrating users with personal COS, refer to the <i>Meridian Mail to CallPilot Migration Utility Guide</i> and the <i>Meridian Mail System Administration Guide</i>.</p>
	<p>Symposium Call Center Server prompt file names are unique.</p> <p>Note: If Symposium Call Center Server prompts exist in more than one mailbox on Meridian Mail, the data collection utility appends the mailbox number to the end of the file name. If the prompt file name and mailbox combination result in file names that are not unique, the prompts in the duplicate files are collected by the Meridian Mail data collection utility but not migrated to the CallPilot server.</p>

	<p>The required number of blank tapes are available to store the data collected from Meridian Mail.</p> <p>Note: The number of tapes required is based on the size of Meridian Mail. If you are migrating Symposium Call Center Server Voice Services on an average system, you will probably need two tapes—one for the system data, which includes voice prompts (menus and announcements), and one tape for the ACCESS mailbox content.</p> <p>For more information, see <i>Meridian Mail to CallPilot Migration Utility Guide</i>, Section B: “Migration strategies.”</p>
	<p>A maintenance period is arranged for Meridian Mail, CallPilot, and Symposium Call Center Server, so that system performance is not impacted by the migration.</p> <p>Note: You should also freeze all modifications to the Meridian Mail system (such as adding additional voice prompts) during the migration. If you do not, these changes may not be migrated.</p>
	<p>The distribution level password for the CallPilot 2.0 support tools is available.</p>

Migration limitations

To migrate voice prompts from Meridian Mail to CallPilot Release 2.0, you must have one of the following Meridian Mail platforms:

- Card Option
- Option EC 11
- Modular Option
- Modular Option EC

Note: If your Meridian Mail system is an Entopia system, you must be running Meridian Mail Release 12 or greater to migrate voice prompts to CallPilot Release 2.0.

For more information, see the *Meridian Mail to CallPilot Migration Utility Guide*.

Tape drive hardware and media requirements

Use the following information to make sure that the tape you use to collect the Meridian Mail data is compatible with your CallPilot tape drive.

Based on your Meridian Mail software release and hardware platform, your Meridian Mail system uses one of the following tape drives and tapes:

- Archive Viper tape drive - 250 Mbyte tapes
- Tandberg SLR4 tape drive (TDC4220) - 2.5 Gbyte tapes
- Tandberg SLR5 tape drive - 2.5 Gbyte tapes

Based on your CallPilot software and hardware platform, your CallPilot system uses one of the following tape drives and tapes:

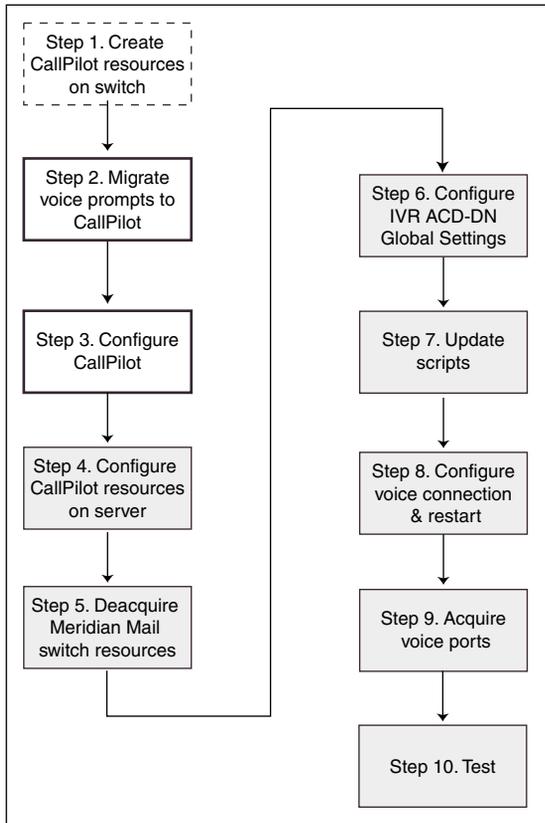
- SLR4 - 2.5 Gbyte tapes
- SLR5 - 2.5 Gbyte tapes
- SLR32 - 2.5 Gbyte tapes
- SLR50 - 2.5 Gbyte tapes

The CallPilot server tape drives SLR32 and SLR50 cannot read the 250 Mbyte tapes created on the Archive Viper tape drive. If the Meridian Mail system has an Archive Viper tape drive, and the CallPilot server to which you are migrating has an SLR32 or SLR50 tape drive, you must upgrade the Meridian Mail system to an SLR4 or SLR5 tape drive prior to performing the migration.

For more information, see the tape drive hardware and media requirements in the *Meridian Mail to CallPilot Migration Utility Guide*.

Migration process

The following flowchart illustrates the migration process:



Note:

- Tasks in boxes with dashed borders are performed on the switch.
- Tasks in boxes with dark borders are performed in CallPilot.
- Tasks in gray boxes (light borders) are performed on Symposium Call Center Server.

Migration checklist

To ensure a successful migration, use the following checklist. Copy or print the checklist, and review it carefully before you begin. The checklist contains each step you must perform during the migration. The order in which you complete certain steps is critical, so follow the order in the checklists. During the migration procedure, check off each item in the checklist as you complete it.

Description	✓
Complete the pre-migration checklist on page 233.	
Step 1. Create CallPilot ACD-DNs and voice ports on the switch For instructions, see page 241.	
Step 2. Migrate Meridian Mail voice prompts and voice segments For instructions, see page 244.	
Step 3. Configure CallPilot For instructions, see page 245.	
Step 4. Configure CallPilot switch resources For instructions, see page 248.	
Step 5. Deacquire Meridian Mail voice ports and ACD-DNs For instructions, see page 252.	
Step 6. Configure the IVR ACD-DN Global Settings For instructions, see page 253.	
Step 7. Update your scripts For instructions, see page 255.	
Step 8. Configure Symposium Call Center Server connection to CallPilot For instructions, see page 256.	

Step 9. Acquire CallPilot resources For instructions, see page 260.	
Step 10. Test the migration. For instructions, see page 261.	

Note: Nortel Networks recommends that you leave your Meridian Mail configuration intact, in case you need to back out of the migration.

Migration procedure

Perform each step in the migration procedure to ensure a successful migration. The order in which you complete the steps is critical.

Step 1. Create CallPilot ACD-DNs and voice ports on the switch

Introduction

You must create the CallPilot ACD-DNs (IVR and ACCESS) and voice ports on the switch. CallPilot ACD-DNs and voice ports have different properties than Meridian Mail ACD-DNs.

Note: You must create separate ACD-DNs for your IVR and ACCESS voice ports. (The configuration for IVR and ACCESS ACD-DNs is identical.)

To create an ACD-DN in LD 23

Use the following prompts and responses in Overlay 23. For prompts not listed in the table, press Enter to accept the default.

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	NO	Indicates that this is not a message center.
MAXP	xx	Indicates the maximum number of ACD agents for this queue.
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.

ALOG	YES	ACD agents are automatically logged on.
REQ	END	Exit from overlay.

To create a voice port in LD 11

Use the following prompts and responses. For prompts not listed in the table, press Enter to accept the default.

Prompt	Response	Description
REQ	NEW	Add a voice port.
TYPE	2008	
TN	l s c u	Terminal Number of the voice port, where l is the loop, s is the shelf, c is the card, and u is the unit. (For the Option 11C, TN is cu only.)
AST	00 01	Associated set assignment on key 0 and key 1.
CLS	MMA FLXA (units 16-31)	Multimedia Agent Flexible voice/data allowed.
KEY	0 ACD xxxx zzz nnnn	Define 0 as an ACD key. xxxx is the ACD-DN. zzz is the CLID entry number. nnnn is the position ID. The DN must match the DN on the Channel Allocation Table (in Meridian Mail) or the Channel Information page (in CallPilot).
KEY	1 SCN xxxx	Define key 1 as a single-call non-ringing DN. xxxx is the SCN DN of the SCN. The DN must match the DN on the Channel Allocation Table (in Meridian Mail) or the Channel Information page (in CallPilot).
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 TRN	Define key 4 as a Transfer key.
KEY	5 AO3 (letter 'O')	Define key 5 as a Conference key.
REQ	END	Exit from overlay.

Step 2. Migrate Meridian Mail voice prompts and voice segments

Migrating voice prompts

Unless you plan to recreate your voice prompts in CallPilot, you must migrate your Meridian Mail voice prompts and voice segments (announcements and menus) to CallPilot.

The Meridian Mail voice prompt and voice segment migration process is documented in detail in “Appendix B, “Migrating prompts and segments from Meridian Mail to CallPilot,” on page 279.

The Meridian Mail to CallPilot Migration Utility helps you collect the Meridian Mail data and migrate it to CallPilot. For more information about this utility, and for detailed instructions, refer to the chapter, “Collecting Data from Meridian Mail,” in the *Meridian Mail to CallPilot Migration Utility Guide* (NTP 555-7101-801).

ATTENTION

The migration moves the voice prompts only. The menu and announcement structures are not moved.

Managing voice prompts

Once the voice prompts are in CallPilot, you manage them with Application Builder. Application Builder is not installed as part of the CallPilot installation. You must ensure that it is installed on a PC in your network. For more information about installing and using Application Builder, see the *CallPilot Application Builder Guide* (NTP 555-7171-325).

Note: You can install Application Builder on the Symposium Call Center Server client PC.

Section C: “Comparison of VPE and Application Builder” on page 275 describes the differences between Application Builder and the Voice Prompt Editor, which you used with Meridian Mail.

Step 3. Configure CallPilot

Introduction

You must perform the following tasks to configure CallPilot for Symposium Call Center Server integration:

- In the Configuration Wizard, on the Switch Identification page, define the CLAN address of Symposium Call Center Server in the server CLAN IP Address field.
- In the Configuration Wizard, on the Channel Detail Information page, define a channel for each voice port that is used to provide ACCESS or IVR services to Symposium Call Center Server.
- Define Service DNs for Symposium Voice Services.

To update the Configuration Wizard

ATTENTION

To minimize downtime, and to make back-out easier, make sure that the CallPilot class IDs are different than the channel numbers of your Meridian Mail voice ports. For example, if your Meridian Mail voice ports use channel numbers 1 to 48, then assign class IDs 49 to 96 to your CallPilot TNs.

This procedure assumes that you use new class IDs. If you do not, then you must delete the channel number from your Meridian Mail voice ports before creating your CallPilot voice ports.

- 1 On the CallPilot server, start the Configuration Wizard (for detailed instructions, see “Part 3: Switch and CallPilot Server Configuration,” in the *CallPilot Installation and Configuration Guide* for your switch type).
- 2 Click Next until the Switch Identification page appears.
- 3 In the Symposium Call Center Server CLAN IP Address box, enter the CLAN address of the Symposium Call Center Server.

- 4 On the left side of the page, click the link for the set of channels you want to configure. The Channel Name column displays the channels on the selected link.
- 5 Click the first channel that you want to configure in the Channel Name column. The Channel Detail Information page appears.
- 6 For each TN used to provide IVR services to Symposium Call Center Server, select the IVR check box.
- 7 For each TN used to provide ACCESS services, select the ACCESS check box, and specify a class ID. The class ID is used for communication between the server and CallPilot over the ACCESS link.
Note: When you define the TN as a voice port on Symposium Call Center Server (see “Configure CallPilot switch resources” on page 248), make sure that the channel number you assign to the voice port matches the class ID for the TN.
- 8 Click Fill.
- 9 Click OK.

To configure the ACCESS and IVR channels in the SDN table

- 1 Start CallPilot Manager (for detailed instructions, see the *CallPilot Administrator's Guide* [NTP 555-7101-301]).
- 2 Choose System → Service Directory Number.
- 3 Click New.

Result: The SDN Detail window appears.

The screenshot shows the CallPilot Manager web interface in Microsoft Internet Explorer. The browser title is "CallPilot Manager - Service Directory Number Details - Microsoft Internet Explorer provided by Nortel Networks". The address bar shows the URL: <http://cpi0020/cpmgr/sysadmin/SvcAdmin/SDN/SDNDetails.asp>. The page header includes the Nortel Networks logo and "CallPilot Manager" with links for "Preferences", "Help", and "Logout". Below the header is a navigation menu with "Home", "User", "System", "Maintenance", "Messaging", "Tools", and "Help". The breadcrumb trail is "Location > System > SDN Browser > SDN Detail". The main content area is titled "SDN Detail:" and contains a "Save" button, "Cancel", "Print", and "Help" buttons. The "General" section has the following fields:

- Service DN: [text input]
- Application Name: [dropdown menu with "Symposium Voice Services" selected]
- Media Type: [dropdown menu with "Voice" selected]
- Minimum Channels: [text input with "0"]
- Maximum Channels: [checkbox "Use Default" checked] and [text input]
- Comments: [text area]

The "Session Profile" section has:

- Session Time Limit: [text input with "10"] minutes
- Maximum Invalid Password Entries: [text input with "10"]

The status bar at the bottom shows "Done" and "Local intranet".

- 4 In the Service DN box, enter the CallPilot ACD-DN number, as defined on the switch.
- 5 In the Application Name box, select Symposium Voice Services from the drop-down list.
- 6 Click Save.

Step 4. Configure CallPilot switch resources

You must update the configuration of switch resources on Symposium Call Center Server. This involves the following tasks:

- Create the CallPilot ACD-DNs.
- Create the CallPilot voice ports.

This procedure assumes that the class IDs of your CallPilot TNs are different than your Meridian Mail channel numbers. If they are not, you must either delete the channel number from the Meridian Mail ACCESS ports, or assign different class IDs to your CallPilot TNs.

Tip: To minimize backout effort, do not delete your Meridian Mail voice ports at this time. You can delete them after you confirm that the migration was successful.

To create a new IVR ACD-DN

- 1 From the IVR ACD-DNs window, choose File→New.

Result: The IVR ACD-DN Properties property sheet appears.

The screenshot shows a window titled "IVR ACD-DN Properties" with a blue title bar. It has two tabs: "General" and "Threshold". The "General" tab is selected. The window contains the following fields and controls:

- "IVR ACD-DN Name:" followed by a text input field.
- "IVR ACD-DN Number:" followed by a text input field.
- "Associated Voice Ports:" followed by a large empty rectangular area.
- "Threshold Class:" followed by a dropdown menu showing "IVR_Template".
- "Status:" followed by a text input field showing "Not Acquired".
- At the bottom, there are three buttons: "Save", "Cancel", and "Help".

- 2 Enter information into the following boxes:

IVR ACD-DN Name: The name of the IVR ACD-DN as it appears on reports.

IVR ACD-DN Number: The DN number, as defined on the switch.

Threshold Class: The threshold class for the IVR ACD-DN.

- 3 Click Save.

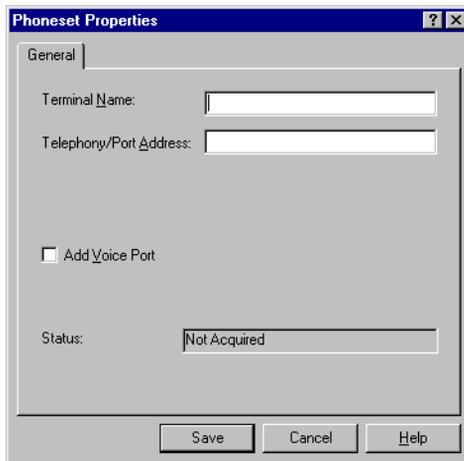
Result: The IVR ACD-DN is added to the list in the IVR ACD-DNs window. It has the status Not Acquired.

Repeat steps 1 to 3 for each CallPilot ACD-DN (ACCESS and IVR) you want to add.

To create a voice port

- 1 From the Phonesets window choose File → New.

Result: The Phoneset Properties property page appears.



- 2 Enter information into the following boxes:

Terminal Name: The name of the phoneset as it appears on reports.

Telephony/Port Address: The address of the voice port on the switch.

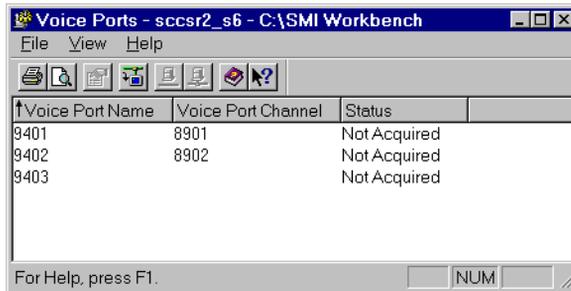
- 3 Ensure that the Add Voice Port box is checked.

- 4 Click Save.

Result: The phoneset is added to the list in the Phonesets window.

- 5 From the SMI window, choose Switch Administration → Voice Ports.

Result: The Voice Ports window appears.



The screenshot shows a window titled "Voice Ports - sccsr2_s6 - C:\SMI Workbench". It has a menu bar with "File", "View", and "Help". Below the menu bar is a toolbar with icons for file operations and help. The main area contains a table with three columns: "Voice Port Name", "Voice Port Channel", and "Status".

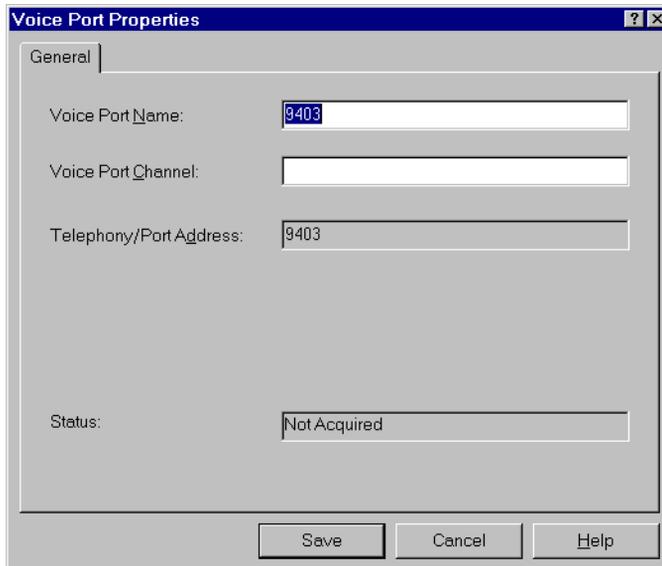
Voice Port Name	Voice Port Channel	Status
9401	8901	Not Acquired
9402	8902	Not Acquired
9403		Not Acquired

At the bottom of the window, there is a text box that says "For Help, press F1." and a "NUM" button.

- 6 From the Voice Ports window, select the phoneset that you added.

- 7 Choose File → Properties.

Result: The Voice Port Properties property sheet appears.



The screenshot shows a dialog box titled "Voice Port Properties" with a "General" tab. It contains several input fields:

- "Voice Port Name": 9403
- "Voice Port Channel": (empty)
- "Telephony/Port Address": 9403
- "Status": Not Acquired

At the bottom of the dialog box, there are three buttons: "Save", "Cancel", and "Help".

- 8 Enter information into the following boxes:

Voice Port Name: The name of the voice port as it appears on reports, if it is different from the phoneset name.

Voice Port Channel: (ACCESS ports only) For Symposium Voice Services on Meridian Mail, this is the ACCESS class number configured for the voice port in the Channel Allocation Table. For Symposium Voice Services on CallPilot, this is the class ID in the Channel Information screen on CallPilot Manager.

9 Click Save.

Result: The voice port channel number is added to the list in the Voice Ports window.

Note: If the channel number is being used by a Meridian Mail voice port, you cannot add the new voice port. You must assign a different channel number. (You must also assign a different class ID in CallPilot.)

Repeat steps 1 to 9 for each CallPilot voice port you want to add.

Step 5. Deacquire Meridian Mail voice ports and ACD-DNs

Introduction

Before you make changes to resources on the switch, you must deacquire those resources from Symposium Call Center Server. You must deacquire the Meridian Mail voice ports first, and then the ACD-DNs (both IVR and ACCESS).

To deacquire a voice port

- 1 From the SMI window, choose Switch Administration → Voice Ports.

Result: The Voice Ports window appears.

- 2 Select the voice port you want to deacquire.
- 3 Choose File → De-acquire.

Repeat steps 1 to 3 for each Meridian Mail voice port that you want to deacquire.

- 4 To refresh the voice port status on the display, choose View → Refresh.

To deacquire an ACD-DN

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

Result: The IVR ACD-DNs window appears.

- 2 Select the ACD-DN you want to deacquire.
- 3 Choose File → De-acquire.

Repeat steps 1 to 3 for each Meridian Mail ACD-DN that you want to deacquire.

- 4 To refresh the ACD-DN status on the display, choose View → Refresh.

Step 6. Configure the IVR ACD-DN Global Settings

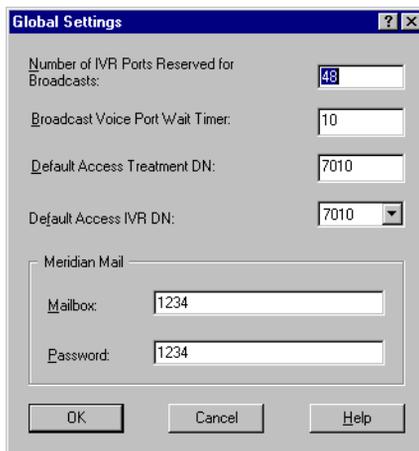
Introduction

You must update the IVR ACD-DN Global Settings with the CallPilot ACCESS DN's.

To update IVR ACD-DN Global Settings

- 1 From the IVR ACD-DNs window, choose File → Global Settings.

Result: The IVR ACD-DNs Global Settings dialog box appears.



The screenshot shows a dialog box titled "Global Settings" with the following fields and values:

- Number of IVR Ports Reserved for Broadcasts: 48
- Broadcast Voice Port Wait Timer: 10
- Default Access Treatment DN: 7010
- Default Access IVR DN: 7010 (dropdown menu)
- Meridian Mail section:
 - Mailbox: 1234
 - Password: 1234

Buttons at the bottom: OK, Cancel, Help.

- 2 Make the desired changes to the properties. You can change the following properties:

Number of IVR Ports Reserved for Broadcasts: The total number of IVR ports that can be user-controlled for broadcast at any time.

Broadcast Voice Port Wait Timer: The number of seconds the system will wait for a voice port to become available.

Default Access Treatment DN: The treatment DN used for the Open Voice Session or Give Controlled Broadcast command. Do not explicitly specify a treatment DN in a script. (This DN is the ACCESS ACD-DN as defined in the switch.)

Default Access IVR DN: This field must contain the same value as the Default Access Treatment DN field.

Mailbox: This field is not used by CallPilot, but it cannot be left blank, so leave the existing value.

Password: This field is not used by CallPilot, but it cannot be left blank, so leave the existing value.

3 Click OK.

Result: You return to the IVR ACD-DNs window.

Step 7. Update your scripts

Introduction

You must update all of your scripts to use the new ACD-DNs.

Hard-coded ACD-DNs

If you hard-coded the ACD-DNs in your scripts, then you must change and reactivate your scripts. For more information about changing scripts, see the *Scripting Guide*.

Global variables

If you used global variables to represent ACD-DNs in your scripts, then you must assign new values to the script variables. For more information about changing variables, see the *Scripting Guide*.

Note: Nortel Networks recommends that you use variables to represent ACD-DNs in your scripts.

Step 8. Configure Symposium Call Center Server connection to CallPilot

Introduction

To use CallPilot for voice processing, install the Symposium Call Center Server Service Update NS040206SU07S or the Symposium Call Center Server Release 4.2 Revision 5 CD, and configure the Symposium Call Center Server connection to CallPilot. Then, restart the Symposium Call Center Server application.

If you are installing using the Symposium Call Center Server Service Update NS040206SU07S, continue with “To install the server service update” on page 256. If you are installing using the Symposium Call Center Server Release 4.2 Revision 5 CD, continue with “To configure the CallPilot connection” on page 258.

Note: Nortel Networks recommends that you retain the physical cable connections between Meridian Mail and Symposium Call Center Server to facilitate backout, if required.

To install the server service update

- 1 Use the PEP Viewer utility to ensure that NS040206SU07S has not already been installed. For more information, refer to the *Installation and Maintenance Guide*.
- 2 Close all application windows.
- 3 If you are installing the server service update from a CD-ROM, insert the Supplementary CD into the CD-ROM drive.
- 4 Locate the directory on the CD-ROM, or the directory into which you downloaded the server service update from the Web.
- 5 Double-click the NS040206SU07S.msi file associated with the server service update.

Result: The program prompts Do you wish to view the README file?

- 6 Click Yes.

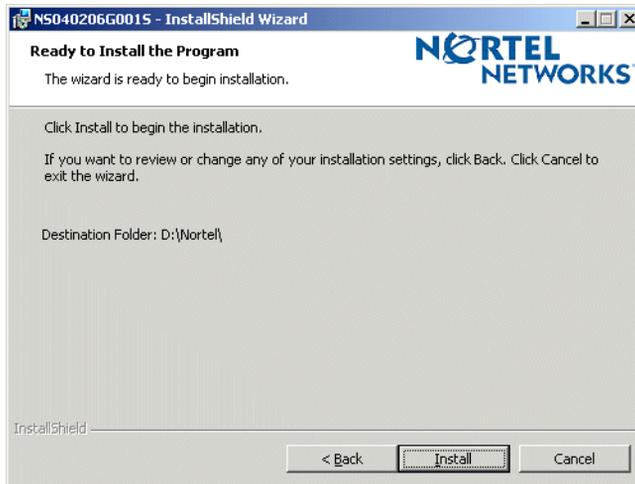
Note: If you do not want to read the ReadMe file, click No and go to step 8.

Result: The contents of the ReadMe file appears.

- 7 When you are finished reading, close the ReadMe file window.

- 8 Click Next to proceed.

Result: The following window appears:



- 9 Click Install.

Result: The system displays messages as it brings down each of the server's services. It then installs the server service update on the server. When installation is complete, the system may prompt *You must restart your system for the changes to take effect. In this case, do not restart, but click Cancel.* The system displays the InstallShield Wizard Completed screen.

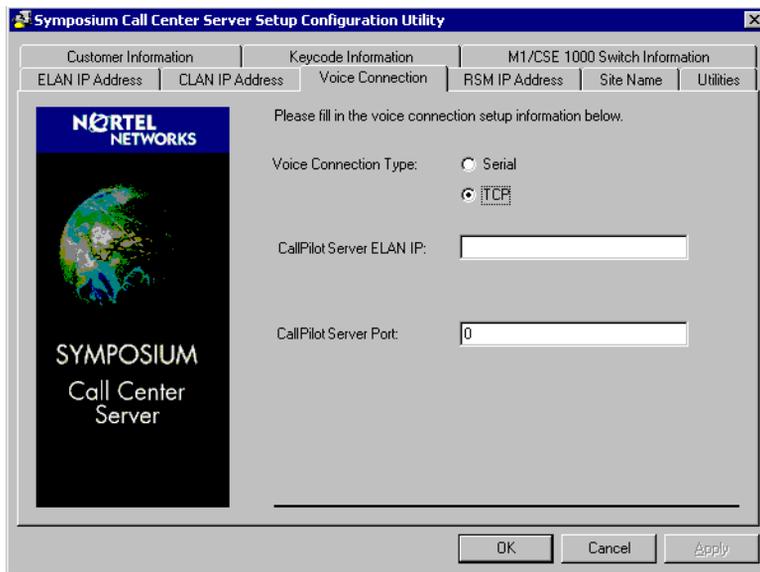
- 10 Click Finish.

- 11 You can use the PEP Viewer utility to confirm that the server service update has been installed on the server. For more information, refer to the *Installation and Maintenance Guide*.

To configure the CallPilot connection

- 1 On Symposium Call Center Server, choose Start → Programs → Symposium Call Center Server → Server Setup Configuration.
- 2 Click the Voice Connection tab.

Result: The Voice Connection property page appears.



- 3 For Voice Connection Type, choose TCP.
- 4 Enter information into the following boxes:
CallPilot Server ELAN IP: The ELAN IP address of the CallPilot server.
CallPilot Server Port: Enter **10008**.
- 5 Click OK.

Result: The utility prompts you to verify your keycode information.

- 6 Click Yes.

Result: The utility shuts down the services and updates the Symposium Call Center Server database. This takes several minutes. Then, the utility displays the message Symposium Call Center Server Setup Configuration is completed successfully.

7 Click OK.

Result: The utility prompts you to make a Platform Recovery disk.

8 Insert a disk in the floppy drive.

9 Click Create disk.

Result: The utility displays the message `Setup Configuration has been exported and saved successfully.`

10 Click OK.

11 Click Cancel to exit the recovery disk window.

Result: The system prompts `You must reboot now to commit changes. Press OK to reboot or Cancel to stop.`

12 Click OK.

Result: The server restarts. When Symposium Call Center Server has come up, continue with the following procedures to acquire the CallPilot ACD-DNs and voice ports.

Step 9. Acquire CallPilot resources

Introduction

After the Symposium Call Center Server application comes up, you must acquire the CallPilot resources to allow voice processing to begin.

To acquire an ACD-DN

- 1 From the SMI window, choose Switch Administration → IVR ACD-DNs.
Result: The IVR ACD-DNs window appears.
- 2 Select the ACD-DN you want to acquire.
- 3 Choose File → Acquire.
Repeat steps 1 to 3 for each CallPilot ACD-DN that you want to acquire.
- 4 To refresh the ACD-DN status on the display, choose View → Refresh.

To acquire a voice port

- 1 From the SMI window, choose Switch Administration → Voice Ports.
Result: The Voice Ports window appears.
- 2 Select the voice port you want to acquire.
- 3 Choose File → Acquire.
Repeat steps 1 to 3 for each CallPilot voice port that you want to acquire.
- 4 To refresh the voice port status on the display, choose View → Refresh.

Step 10. Test the migration

Introduction

After you complete the migration, you must test your system to ensure that it functions as expected. Test each path in the call scripts to ensure that calls are routed properly, and that they receive the appropriate treatment.

Section B: Backing out of the migration

In this section

Overview of backout	264
Backout procedure	266

Overview of backout

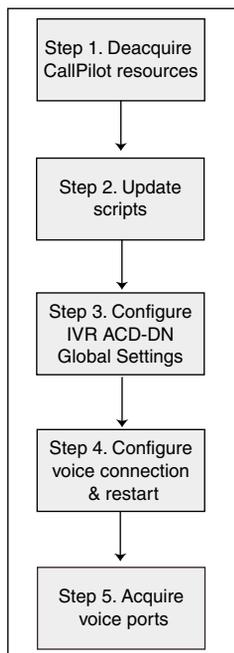
Introduction

If the migration to CallPilot is not successful, you can use the procedure in this section to back out of the migration.

Note: This procedure assumes that you have not deleted your Meridian Mail resources from the switch. If you have deleted them, then you must create them before you continue. For more information about creating Meridian Mail resources, see Chapter 4, “Switch subsystem configuration.”

Backout process

The following flowchart illustrates the backout process:



Backout checklist

Use the following checklist to perform the backout:

Description	✓
Step 1. Deacquire CallPilot ACD-DNs and voice ports For instructions, see page 267.	
Step 2. Configure IVR ACD-DN Global Settings For instructions, see page 268.	
Step 3. Update your scripts For instructions, see page 270.	
Step 4. Configure the connection to Meridian Mail For instructions, see page 271.	
Step 5. Acquire voice ports For instructions, see page 274.	

Backout procedure

If the migration to CallPilot is not successful, use the procedure in this section to back out of the migration.

This procedure assumes that you have not deleted your Meridian Mail resources from the switch. If you have deleted them, then you must create them before you continue.

Step 1. Deacquire CallPilot ACD-DNs and voice ports

Introduction

You must deacquire the CallPilot voice ports and ACD-DNs on Symposium Call Center Server.

To deacquire a voice port

- 1 From the SMI window, choose Switch Administration → Voice Ports.
Result: The Voice Ports window appears.
- 2 Select the voice port you want to deacquire.
- 3 Choose File → De-acquire.
Repeat steps 1 to 3 for each CallPilot voice port that you want to deacquire.
- 4 To refresh the voice port status on the display, choose View → Refresh.

To deacquire an ACD-DN

- 1 From the SMI window, choose Switch Administration → IVR ACD-DNs.
Result: The IVR ACD-DNs window appears.
- 2 Select the ACD-DN you want to deacquire.
- 3 Choose File → De-acquire.
Repeat steps 1 to 3 for each CallPilot ACD-DN that you want to deacquire.
- 4 To refresh the ACD-DN status on the display, choose View → Refresh.

Step 2. Configure IVR ACD-DN Global Settings

Introduction

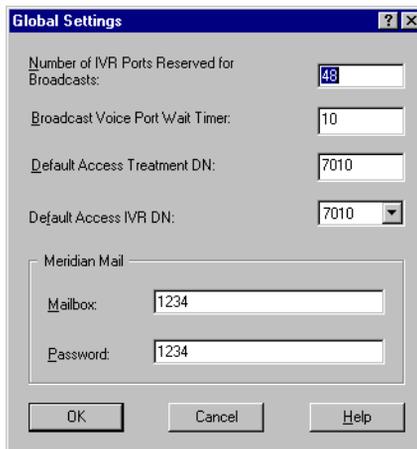
You must update the IVR ACD-DN Global Settings with the Meridian Mail ACCESS IVR and Treatment DNs, and verify that the Meridian Mail mailbox and password are correctly specified.

Note: This procedure assumes that your Meridian Mail ACD-DNs and voice ports still exist on the server. If they do not exist, you must create them. For detailed instructions, see Chapter 7, “Symposium Call Center Server subsystem configuration.”

To update IVR ACD-DN Global Settings

- 1 From the IVR ACD-DNs window, Choose File → Global Settings.

Result: The IVR ACD-DNs Global Settings dialog box appears.



The screenshot shows a dialog box titled "Global Settings". It contains the following fields and values:

- Number of IVR Ports Reserved for Broadcasts: 48
- Broadcast Voice Port Wait Timer: 10
- Default Access Treatment DN: 7010
- Default Access IVR DN: 7010 (dropdown menu)
- Meridian Mail section:
 - Mailbox: 1234
 - Password: 1234

Buttons at the bottom: OK, Cancel, Help.

- 2 Make the desired changes to the properties. You can change the following properties:

Number of IVR Ports Reserved for Broadcasts: The total number of IVR ports that can be user-controlled for broadcast at any time.

Broadcast Voice Port Wait Timer: The number of seconds the system will wait for a voice port to become available.

Default Access Treatment DN: The treatment DN used for the Open Voice Session or Give Controlled Broadcast command. Do not explicitly specify a treatment DN in a script.

Default Access IVR DN: This field must contain the same value as the Default Access Treatment DN field.

Mailbox: The DN of the Meridian Mail mailbox.

Password: The password required to access the Meridian Mail mailbox.

3 Click OK.

Result: You return to the IVR ACD-DNs window.

Step 3. Update your scripts

Introduction

If you are using different ACD-DNs for CallPilot and Meridian Mail, then you must update all of your scripts to reference the Meridian Mail ACD-DNs.

Hard-coded ACD-DNs

If you hard-coded the ACD-DNs in your scripts, then you must change and reactivate your scripts. For more information about changing scripts, see the *Scripting Guide*.

Global variables

If you used global variables to represent ACD-DNs in your scripts, then you must assign new values to the script variables. For more information about changing variables, see the *Scripting Guide*.

Note: Nortel Networks recommends that you use variables to represent ACD-DNs in your scripts.

Step 4. Configure the connection to Meridian Mail

Introduction

You must now shut down the Symposium Call Center Server application, configure the Symposium Call Center Server's connection to Meridian Mail, and restart the Symposium Call Center Server application.

Note: If you removed the serial cable connection between Symposium Call Center Server and Meridian Mail, you must reinstall it now.

To shut down the Symposium Call Center Server application

- 1 From the Windows Start menu, choose Programs → Symposium Call Center Server → Shutdown.

Result: The Symposium Call Center Server Shutdown dialog box appears.

- 2 Click OK.

Result: The utility shuts down all services, and then the Service Status Log dialog box appears. This log displays any services that failed to shut down. Click Recheck to refresh the service statuses.

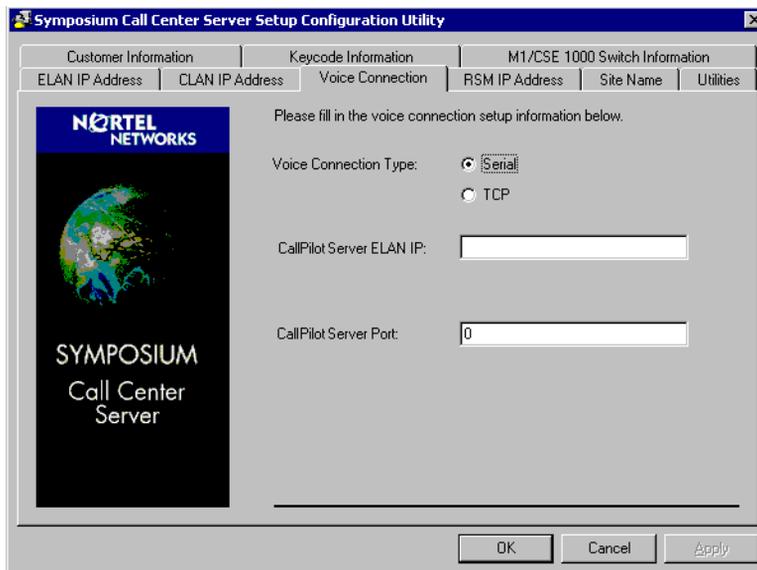
- 3 If any services are still running, use the Services control panel to shut them down manually (from the Windows Start menu, choose Control Panel, and double-click Services). Then click Recheck to update the status log.

- 4 Click Accept to exit the utility.

To configure the Meridian Mail connection

- 1 On Symposium Call Center Server, choose Start → Programs → Symposium Call Center Server → Server Setup Configuration.
- 2 Click the Voice Connection tab.

Result: The Voice Connection property page appears.



The screenshot shows a window titled "Symposium Call Center Server Setup Configuration Utility". It has several tabs: "Customer Information", "Keycode Information", and "M1/CSE 1000 Switch Information". Under "Customer Information", there are sub-tabs for "ELAN IP Address", "CLAN IP Address", and "Voice Connection". Under "M1/CSE 1000 Switch Information", there are sub-tabs for "RSM IP Address", "Site Name", and "Utilities". The "Voice Connection" tab is active. The dialog contains a Nortel Networks logo and a globe image with the text "SYMPOSIUM Call Center Server". The main area says "Please fill in the voice connection setup information below." and has three fields: "Voice Connection Type:" with radio buttons for "Serial" (selected) and "TCP"; "CallPilot Server ELAN IP:" with an empty text box; and "CallPilot Server Port:" with a text box containing "0". At the bottom are "OK", "Cancel", and "Apply" buttons.

- 3 For Voice Connection Type, choose Serial.

Note: The CallPilot Server ELAN IP box clears and the CallPilot Server Port box is set to 0. Do not change these values.

- 4 Click OK.

Result: The utility prompts you to verify your keycode information.

- 5 Click Yes.

Result: The utility shuts down the services and updates the Symposium Call Center Server database. This takes several minutes. Then, the utility displays the message Symposium Call Center Server Setup Configuration is completed successfully.

- 6 Click OK.

Result: The utility prompts you to make a Platform Recovery disk.

7 Insert a disk in the floppy drive.

8 Click Create disk.

Result: The utility displays the message Setup Configuration has been exported and saved successfully.

9 Click OK.

10 Click Cancel to exit the recovery disk window.

Result: The system prompts You must reboot now to commit changes. Press OK to reboot or Cancel to stop.

11 Click OK.

Result: The server restarts. When Symposium Call Center Server has come up, continue with the following procedures to acquire the Meridian Mail ACD-DNs and voice ports.

Step 5. Acquire voice ports

Introduction

After the server comes up, you must acquire the Meridian Mail resources to allow voice processing to begin.

To acquire an ACD-DN

- 1 From the SMI window, choose Switch Administration → IVR ACD-DNs.
Result: The IVR ACD-DNs window appears.
- 2 Select the ACD-DN that you want to acquire.
- 3 Choose File → Acquire.
Repeat steps 1 to 3 for each Meridian Mail ACD-DN that you want to acquire.
- 4 To refresh the ACD-DN status on the display, choose View → Refresh.

To acquire a voice port

- 1 From the SMI window, choose Switch Administration → Voice Ports.
Result: The Voice Ports window appears.
- 2 Select the voice port you want to acquire.
- 3 Choose File → Acquire.
Repeat steps 1 to 3 for each Meridian Mail voice port that you want to acquire.
- 4 To refresh the voice port status on the display, choose View → Refresh.

Section C: Comparison of VPE and Application Builder

In this section

Application Builder compared to Voice Prompt Editor

276

Application Builder compared to Voice Prompt Editor

Introduction

After you migrate to CallPilot, you use the Application Builder to create, record, and manage your voice prompts. (For detailed instructions, see the *CallPilot Application Builder Guide*.) Application Builder provides much of the same functionality as the Voice Prompt Editor. This section lists some of the differences between the two programs.

Differences

Feature	VPE	Application Builder
Logging on	Enter Mailbox, password, and telephone number.	Enter Mailbox, password, and location. The program prompts for the telephone number only if you are recording prompts.
Creating voice files	Description is entered at the time of creation.	Description can be added later.
Opening voice files	You can only open one voice file at a time.	You can open multiple files (applications) simultaneously.
Reverting to previous file version	Menu command	Close the file without saving changes.
Copying voice files	Copy menu command	Save As menu command
Renaming voice files	Rename menu command	Save As menu command; delete previous version

Feature	VPE	Application Builder
Undeleting voice files	Undelete menu command	Not available
Copying voice segments between applications	Unavailable	Supported
Listing voice segments	List shows segment duration. Each segment has a name, title, and description.	Segment duration is not available. Each segment has a name and description.
Playing the next segment	Play Next Segment button.	To play the next segment, select the next segment and click Play.
Editing segment length	Length button	Use a third-party application.
Searching for voice segments	Search button	Sort the segment list by ID or name.
Undeleting voice segments	Undelete menu command	Close and reopen the file.
Deleting voice segments	Deletion causes renumbering of IDs in segment list, and may affect scripts	Deletion does not cause renumbering.
Playing a group of recorded prompts	Supported	You can build an application to play prompt groups.
Telset application for recording prompts	Not available	Voice Item Maintenance

Feature	VPE	Application Builder
Prompt size	Up to 2 minutes	Up to 10 minutes
Maximum number of segments per file	1000	3000
Import/Export .WAV file	Not available	Supported
Connection to server	VPE uses a serial ACCESS link.	Application Builder uses a LAN link, resulting in faster file/segment operations.
Assignment of segment IDs	Automatically assigned, starting at 1.	Default provided, but can be overridden by administrator.
Description length	Maximum 2048 characters	Maximum 240 characters

Appendix B

Migrating prompts and segments from Meridian Mail to CallPilot

In this appendix

Overview	280
Understanding the migration process	281
Performing the Symposium Voice Services migration	285
Post migration activities	299

Overview

Introduction

Unless you plan to recreate your voice prompts in CallPilot, you must migrate your Meridian Mail voice prompts and voice segments (announcements and menus) to CallPilot. The Meridian Mail to CallPilot Migration Utility helps you collect the Meridian Mail data and migrate it to CallPilot.

This appendix contains the following:

- information on what data can and cannot be migrated
- procedures for migrating from Meridian Mail to CallPilot, including hardware requirements, loading the migration utility, and collecting and migrating voice segments data and voice prompts
- post-migration activities

Understanding the migration process

Overview

Before migrating prompts, you need to know what you want to migrate, what cannot be migrated, and how the migration process works.

The following documents should be available to assist with the migration:

- *Meridian Mail System Administration Guide* (NTP 555-7001-301) for Meridian Mail administration tasks if they are needed
- *Meridian Mail to CallPilot Migration Utility Guide* (NTP 555-7101-801)
- *CallPilot 2.0 Application Builder Guide* (NTP 555-7101-325)

Note: In the Meridian Mail to CallPilot migration procedures, the terms “voice prompt” and “voice segment” may be used interchangeably and differently from how you normally use them in Symposium Call Center Server. Each procedure should make clear the voice items being migrated.

Migrating the voice processing engine

This migration can be a lengthy operation, and a strategy should be created using the information in the *Meridian Mail to CallPilot Migration Utility Guide*.

You can migrate the following categories of data:

- **System Data:** system and customer profiles, system names and greetings, restriction/permission lists (RPLs), networking data, classes of service (COS), and messaging settings
- **User data:** classes of service (COS), local voice users, personal greetings (internal, external, and temporary), personal distribution lists (PDLs), user core and mailbox properties, user voice messages
- **Networking data:** See the *Meridian Mail to CallPilot Migration Utility Guide* for more details.
- **Corporate directory entries:** See the *Meridian Mail to CallPilot Migration Utility Guide* for more details.
- **Voice services:** voice segments in menus, announcements, fax segments

- **Messages:** voice messages in mailboxes
- **Symposium Call Center Server prompts:** prompts in mailboxes
Note: You can migrate Symposium Call Center Server prompts to CallPilot 2.0 only. Symposium Call Center Server prompts are not supported on CallPilot 1.7.

The CallPilot 2.0 migration utility automatically creates the Application Builder applications during the prompt migration. The information specific to Symposium Voice Services is menu segments, announcements, and prompts. As a guideline, it is recommended that you migrate the system data first, and then migrate the voice messages and Symposium prompts.

Note: Meridian Mail users must be redefined in CallPilot before migrating their voice messages.

For a detailed procedure, see the *Meridian Mail to CallPilot Migration Utility Guide*.

When you have completed the voice prompts and segments migration, continue to “Step 3. Configure CallPilot” on page 245. You must complete all the steps in the process to test the success of the migration.

Migrating Symposium Voice Services only

The data for migration is determined by the voice services implemented on your existing Meridian Mail system, and whether you want to maintain the same services on CallPilot:

- For GIVE IVR voice services, migrate the voice prompts (menus and announcements). These elements can be identified in the Meridian Mail VSDN table.
- For ACCESS voice services, migrate only the Symposium Call Center Server prompts. These elements are stored in specific mailboxes. In Meridian Mail, only one mailbox can be used at a time.
- For GIVE IVR and ACCESS voice services, migrate voice prompts (menus and announcements) and Symposium Call Center Server prompts. Migrate the voice prompts (menus and announcements) first, and then migrate the active Meridian Mail mailboxes, which contain the voice files for ACCESS.

Note: If Meridian Mail is also used as a front-end IVR subsystem, the migration should be treated as a whole voice processing engine migration.

Data that cannot be migrated

Some data that directly impacts Symposium Call Center Server voice services cannot be automatically migrated. The following is an example of the data that is not automatically migrated. For a complete list, see the *Meridian Mail to CallPilot Migration Utility Guide*.

Example of system data that cannot be migrated

- voice forms, voice menu structure (voice link information), VSDN table, and any other voice service information
- hardware information (such as the channel allocation table, SMDI link information, T1 or E1 link information)
- multi-customers: CallPilot currently supports only one Meridian 1 customer. Mailboxes that have the same mailbox numbers as ones used in other customer groups are not migrated.

Examples of user data that cannot be migrated

- user mailboxes that are less than 3 digits in length
- duplicate DNs (Meridian Mail users with duplicate primary and secondary DN entries)
- users with personal COS

How to handle data that cannot be migrated

For GIVE IVR voice services, the information in the Meridian Mail VSDN table and the voice menu structure is critical. As this data cannot be automatically migrated, you must use Application Builder to

- recreate or rebuild the menu or announcement structure extracted from the VSDN table. (See “How to use voice items created for Meridian Mail Voice Services” in the *CallPilot 2.0 Application Builder Guide*.)
- manage the applications and the control blocks (save and complete them)
- publish applications in the CallPilot 2.0 service DN table

Note: If you migrated voice segments (menus and announcements) to CallPilot, you must run the Application Builder Data Integrity and Repair tool before you attempt to use Application Builder to create the application. For more information, see “Correcting and moving Application Builder data” in the *Meridian Mail to CallPilot Migration Utility Guide*.

Performing the Symposium Voice Services migration

Overview

The procedure for migrating Symposium Voice Services consists of

- preparing Meridian Mail for migration
- preparing CallPilot for migration
- (if necessary) loading the migration utility (data preparation software) onto the Meridian Mail system
- collecting and migrating voice segments data

If you are migrating GIVE IVR voice services, which consist of menu and announcement segments, collect the data containing the voice segment items.
- creating the CallPilot applications for GIVE IVR voice services
- collecting and migrating user data

This includes the migration of the ACCESS mailboxes (containing voice prompts). This step involves an additional tape.
- validating the migrated data, and performing any additional configuration that may be required to make the CallPilot configuration consistent with the Meridian Mail configuration

Preparing Meridian Mail for migration

Before beginning the migration, do the following:

- Review the migration process and complete the pre-migration checklist.
- Review the Meridian Mail System Event and Error Reports (SEERs) to
 - ensure that the data you want to collect is clean and consistent
 - verify that there are no reported problems with the system or the files

Class 11, 31, and 66 SEERs indicate format errors or disk corruption. If these SEER classes have been reported, report the errors to the Nortel Networks CallPilot customer support group to confirm that the migration can still take place.

Preparing CallPilot for migration

CallPilot 2.0 tower or rackmount hardware requirements

The tower or rackmount server must be equipped with an internal tape drive. Based on the CallPilot server software release and the server model, the internal tape drive can be either the Tandberg SLR32 or the Tandberg SLR50.

If an internal tape drive is not installed in the tower or rackmount server, then see Part 5 of the *CallPilot Installation and Configuration Guide* for your server model for instructions.

Notes:

1. The CallPilot server may already have an internal tape drive installed. If this is the case, then you may not need to install a tape drive or tape drive driver.
2. The CallPilot server tape drives SLR32 and SLR50 cannot read the 250 Mbyte tapes created on the Archive Viper tape drive. If the Meridian Mail system has an Archive Viper tape drive, and the CallPilot server to which you are migrating has an SLR32 or SLR50 tape drive, you must upgrade the Meridian Mail system to an SLR4 or SLR5 tape drive prior to performing the migration.
3. The Archive Viper tape drive cannot write on 2.5 Gbyte tapes.

CallPilot 2.0 IPE server hardware requirements

An external tape drive must be connected to the IPE servers. Both servers have a built-in SCSI connector on the faceplate.

The IPE servers are supplied with the Tandberg SLR5 (NTRH9038) tape drive. This tape drive can read the 2.5 Gbyte Meridian Mail tapes that are created using the Tandberg SLR4 (TDC4220) tape drive.

CallPilot server preparation

Review the following preparations:

- No users have been added to the CallPilot system.
- All CallPilot RPL names are unique from Meridian Mail RPL names.
- All existing CallPilot mailbox classes are renamed.

Note: If a duplicate class of service (COS) is found during the migration, the Meridian Mail COS is renamed to COSname_MMail COSID and then migrated to CallPilot. This may cause migrated users to be assigned to the wrong mailbox class.

- The tape device connection is correct.
 - If the CallPilot server is a 200i or 201i server, check to make sure that an external tape drive is connected to the server. If an external tape drive is not connected, you need to connect one.

For instructions on connecting the tape drive, see Part 2, “Server Hardware Install,” in the *CallPilot Installation and Configuration Guide* for your server model.

- If the server is a tower or rackmount server, make sure that an internal tape drive is installed in the server. If an internal tape drive is not installed in the tower or rackmount server, see Part 5 of the *CallPilot Installation and Configuration Guide* for your server model for instructions.
- Event throttling is disabled on CallPilot.

For CallPilot Release 2.0 instructions, see the *CallPilot Administrator’s Guide* (NTP 555-7101-301) and the CallPilot Manager online Help.
- No client applications are running on CallPilot while the migration is in progress. This includes CallPilot Manager and all other client software.

Loading the migration utility

You must install the latest Data Collection Utility from the supplied tape. Make sure you have this tape available. This tape initializes the Meridian Mail system for data collection.

Note: Even though the data collection utility is included in systems running Meridian Mail Release 13.14, you must install the latest Data Collection Utility from the supplied tape.



CAUTION

Risk of reduced system performance

Do not run the data collection utility while Meridian Mail is online. Courtesy down the Meridian Mail system before you prepare it for data collection. For instruction, see Meridian Mail documentation.

On a Card Option platform, you should disable all channels; otherwise, you may not have enough memory.

- 1 Restart the Meridian Mail system with the Meridian Mail to CallPilot migration tape in the tape drive.
- 2 Run the data collection utility.

Result: The CallPilot Data Collection Utility Preparation Menu appears.

```
CallPilot Data Collection Utility Preparation Menu
-----
1 Preparation for MM11 System
2 Preparation for MM12 System
3 Preparation for MM13 System
4 Preparation for MM13,14 System
5 EXIT to support level

Please enter the operation number: █
```

- 3 Select the software release for your Meridian Mail system.

Result: The utility asks for confirmation.

- 4 Select Yes.

Result: A series of messages appears. The system begins to copy the files. The message `Starting RW100 server and copy file utility files from the Tape to the Meridian Mail hard disk appears.`

- 5 When the copying is finished, the message `Please remove CallPilot Data Collection Preparation tape and reboot system into full service and continue data migration by logging into tools level appears.`

Remove the tape and restart Meridian Mail.

Note: Meridian Mail must be brought back into full service before continuing the data migration.

Collect and migrate voice segments data

In this procedure, you collect and migrate voice segments using selective data collection.

Note: In this procedure, voice segments refer to IVR voice items.

Selective voice segments data collection from Meridian Mail

Note: The SLR32 and SLR50 tape drives in CallPilot servers cannot read tapes created with the Archive Viper tape drive (250 Mbyte tapes). Data must be collected on a 2.5 Gbyte QIC tape.

To collect selective voice segments from Meridian Mail

- 1 On the Meridian Mail administration terminal, log on to the Tools menu.

Result: The main Tools menu appears.

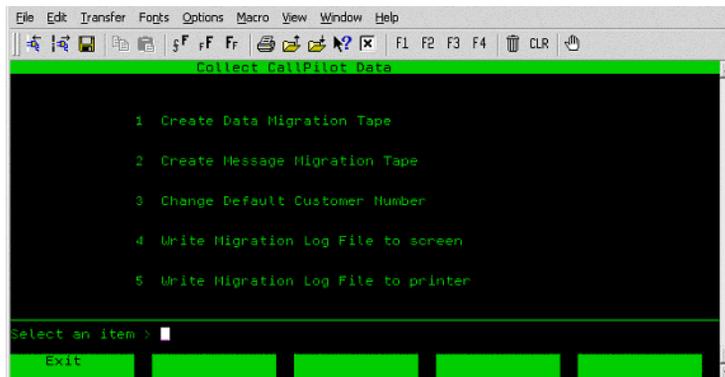
Note: On a Meridian Mail 13 system, use the Tools user ID and the Tools level password. The screens on the Meridian Mail systems can differ from those shown in this guide, depending on the Meridian Mail release number and the type and number of features installed.

- 2 Select Other (Option 13 or 14 in the main Tools menu).

Result: The System / Feature Dependent Tools menu appears.

- 3 Select Collect CallPilot Data (Option 6 or 7 in the System / Feature Tools menu).

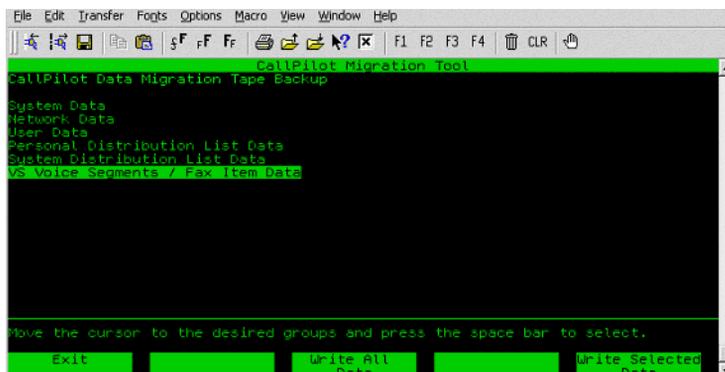
Result: The Collect CallPilot Data menu appears.



- 4 For selective data collection, select Create Data Migration Tape.

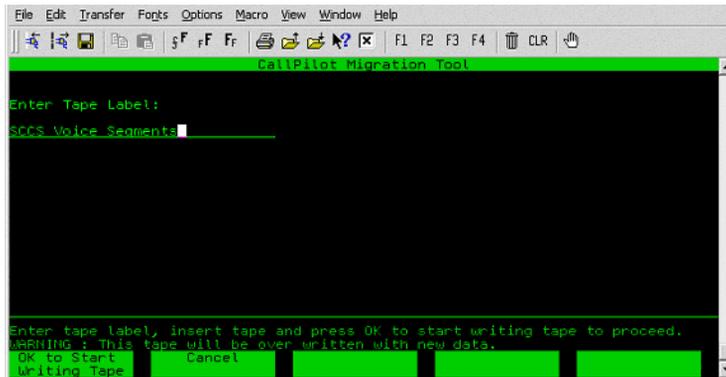
Result: The Collect CallPilot Data Migration Tape Backup window appears.

- 5 To back up the Symposium Call Center Server voice segments, select VS Voice Segments / Fax Item Data. (Use the up and down arrow to move to the desired line, and then press the spacebar to highlight VS Voice Segments / Fax Item Data.)



- 6 Select Write Selected Data (the softkey in the bottom right corner of the screen).

Result: The Tape Label entry window appears.



- 7 On your blank tape, label the tape as Symposium Call Center Server Voice Segments (announcements and menus).
- 8 Insert the tape into the Meridian Mail tape drive, and then select the OK to Start Writing Tape softkey.

Note: The new data overwrites any existing data on the tape.

Result: The system receives tape descriptor data, and then rewinds the tape and writes the segment data to the tape. A message indicates that you can check the log file from the main menu for error messages, and then you are prompted to continue.

- 9 Press Return to continue.

Result: You return to the previous menu.

- 10 Select the Exit softkey.

Result: You return to the Collect CallPilot Data menu.

- 11 Select the Write Migration Log File to screen softkey.

Result: The Meridian Mail Migration log appears.

Note: The log window displays all migration operation logs from the oldest operation to the latest operation. Identify the logs corresponding to your specific data collection start time.

- 12 Check the log file for

- tape label

- data collection for CallPilot begin date and time
 - any specific errors
 - total number of voice services (such as Symposium Call Center Server voice segments), which must match the segments defined in the Voice Service DN table
- 13** Press Return until you get back to the Collect CallPilot Data menu.
- If you are satisfied with the content of the migration log file, proceed with the CallPilot data migration.

Selective voice segments data migration to CallPilot

If you performed a selective data collection from Meridian Mail, you must perform a selective data migration to CallPilot.

Whole voice processing engine migration is not detailed in this guide. See the *Meridian Mail to CallPilot Migration Utility Guide* (NTP 555-7101-801) for more information on whole voice processing engine migration.

Review the following information about the CallPilot 2.0 migration utility and process before starting the migration:

- You can select which data set you want to migrate to CallPilot in a particular migration session.
- The utility offers an option that allows you to copy files to the CallPilot server before the migration begins. The files are copied to a folder referred to as the staging area.
- Do not remove the tape during the migration. If the files are not present on the disk for any reason, the data is migrated from the files on the tape.
- When the files are copied to the CallPilot server, they are copied to the D:\nortel\MPCX\Migration\MigrationFiles folder.
- The migration utility offers an option that allows you to delete the files from the CallPilot server when the migration is finished. Nortel Networks recommends that you do not delete the staging files until you are certain that the migration is successful.

To migrate selective voice segments to CallPilot

- 1 Locate the migration utility file "migrate.exe" in the D:\nortel\MPCX\Migration folder.
- 2 Open a command prompt window, go to the migration folder, and launch the migration tool.
Result: The Meridian Mail to CallPilot Migration Utility window appears with the CI> prompt.
- 3 Insert the Symposium Call Center Server Voice Segments data tape in the tape drive.
- 4 Type **ReadTapeLabel** to verify that you have the correct tape.
- 5 At the CI> prompt, type **migrate**, and then press Enter.
Result: The following prompt appears:

```
Enter Data Set to: All System Data
```
- 6 To restore Symposium Call Center Server Voice Segments, select ApplServices (using the up and down arrow keys).
Result: More prompts appear.
- 7 Respond to the prompts as follows:
 - Do you wish to copy the Meridian Mail data files now - **Yes**
 - Do you wish to delete the staging files after migration is complete - **No**
- 8 Insert the Meridian Mail Voice Segments data tape in the tape drive, and then press Enter.
Result: The migration utility reads the tape and writes the data onto the CallPilot server disk. A log appears on the screen at the end of the operation.
- 9 Check the MigTransaction.log.
Note: Pay particular attention to the Total Number of Services created/updated successfully.
- 10 If you are ready to exit the migration, at the CI> prompt, type **quit**, and then press Enter.

If you are satisfied with the content of the migration log file, proceed to "Post migration activities," on page 299.

Collect and migrate Symposium Call Center Server voice prompts

In this procedure, you collect and migrate ACCESS voice prompts.

Note: In this procedure, voice prompts refer to ACCESS voice items.

Symposium Call Center Server voice prompts collection from Meridian Mail

Note: The SLR32 and SLR50 tape drives in CallPilot servers cannot read tapes created with the Archive Viper tape drive (250 Mbyte tapes). Data must be collected on a 2.5 Gbyte QIC tape.

To collect voice prompts from Meridian Mail

- 1 On the Meridian Mail administration terminal, log on to the Tools menu.

Result: The main Tools menu appears.

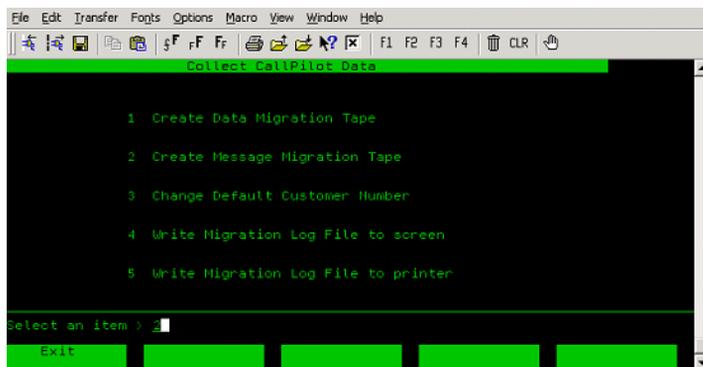
Note: On a Meridian Mail 13 system, use the Tools user ID and the Tools level password. The screens on the Meridian Mail systems can differ from those shown in this guide, depending on the Meridian Mail release number and the type and number of features installed.

- 2 Select Other (Option 13 or 14 in the main Tools menu).

Result: The System / Feature Dependent Tools menu appears.

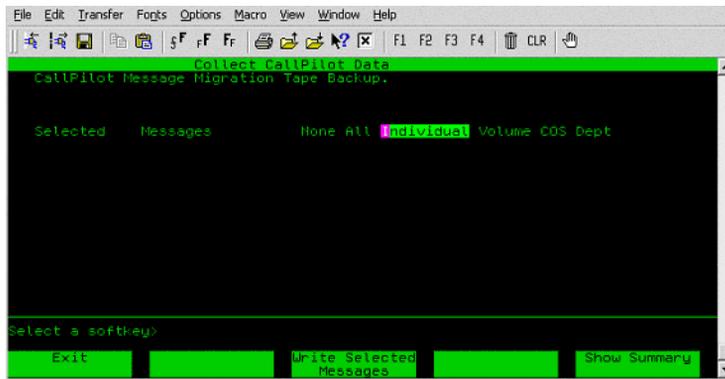
- 3 Select Collect CallPilot Data (Option 6 or 7 in the System / Feature Tools menu).

Result: The Collect CallPilot Data menu appears.



- 4 For voice prompt collection, select Create Message Migration Tape.

Result: The CallPilot Message Migration Tape Backup window appears.



- 5 To create a message and Symposium Call Center Server prompt migration for the ACCESS mailbox, select Individual. (Use the up and down arrow or spacebar to move the cursor.)

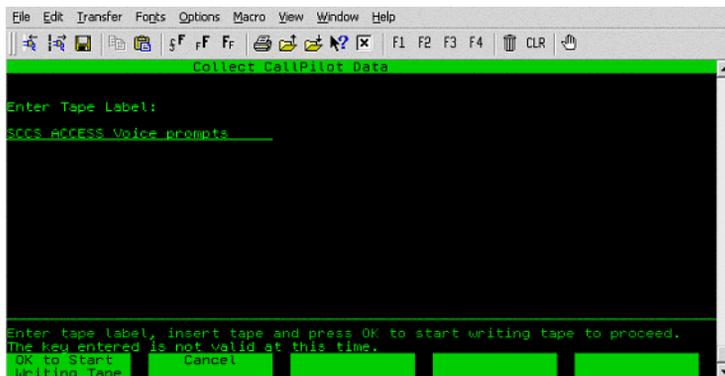
- 6 Press Enter.

Result: You are allowed to enter your selected mailboxes.

- 7 Move to the grid lines and enter the ACCESS mailbox number. (This is the mailbox number used to log on to the Symposium Call Center Server Voice Prompt Editor.)

- 8 Select the Write Selected Messages softkey.

Result: The system prompts you to enter a tape label.



- 9 On your blank tape, label the tape as Symposium Call Center Server ACCESS Voice Prompts.
- 10 Insert the tape into the Meridian Mail tape drive, and then select the OK to Start Writing Tape softkey.
Note: The new data overwrites any existing data on the tape.
Result: The system receives tape descriptor data, then rewinds the tape and writes the prompt data to the tape. A message indicates that you can check the log file from the main menu for error messages, and then you are prompted to continue.
- 11 Press Return to continue.
Result: You return to the previous menu.
- 12 Select the Exit softkey.
Result: You return to the Collect CallPilot Data menu.
- 13 Select the Write Migration Log File to screen softkey.
Result: The Meridian Mail Migration log appears.
Note: The log window displays all migration operation logs from the oldest operation to the latest operation. Identify the logs corresponding to your specific data collection start time.
- 14 Check the log file for
 - tape label
 - data collection for CallPilot begin date and time
 - any specific errors
 - total number of voice services (such as Symposium Call Center Server ACCESS prompts)
- 15 Press Return until you get back to the Collect CallPilot Data menu.
If you are satisfied with the content of the migration log file, proceed with the CallPilot data migration.

Symposium Call Center Server ACCESS prompts migration to CallPilot

The migration process of the message/Symposium Call Center Server prompt tape to CallPilot is based on how you collected the data from Meridian Mail.

- If you collected all messages and Symposium Call Center Server prompts on one tape, then all the messages and Symposium Call Center Server prompts on the tape are migrated to CallPilot in a single session.
- If you collected messages and Symposium Call Center Server prompts on multiple tapes (that is, performed a selective message migration), messages are migrated to CallPilot in multiple sessions, one tape per session, in the order in which the tapes were created.
- The CallPilot migration utility allows you to select the data set that you want to migrate to CallPilot in a particular migration session.

Note: Do not remove the tape during the migration.

To start the migration operation

1 Locate the migration utility file “migrate.exe” in the D:\nortel\MPCX\Migration folder.

2 Open a command prompt window, go to the migration folder, and launch the migration tool.

Result: The Meridian Mail to CallPilot Migration Utility window appears with the C|> prompt.

3 Insert the Symposium Call Center Server ACCESS Voice Prompts data tape in the tape drive.

4 Type **ReadTapeLabel** to verify that you have the correct tape.

5 At the C|> prompt, type **migrate**, and then press Enter.

Result: The following prompt appears:

```
Enter Data Set to: All System Data
```

6 To restore Symposium Call Center Server Voice Segments, select SCCSPromptsOnly (using the up and down arrow keys).

7 Insert the Symposium Call Center Server ACCESS Voice Prompts data tape in the tape drive, and then press Enter.

Result: The migration utility reads the tape and writes the data onto the CallPilot server disk. A log appears on the screen at the end of the operation.

8 Check the MigTransaction.log.

Note: Pay particular attention to the Total Number of Services created/updated successfully.

- 9 If you are ready to exit the migration, at the `Cl>` prompt, type **quit**, and then press Enter.

If you are satisfied with the content of the migration log file, proceed to “Post migration activities,” on page 299.

Post migration activities

After the migration of voice services from Meridian Mail to CallPilot is finished, review and complete the following post migration activities.

Disconnecting the tape drive

If you used an external tape drive to perform the migration, you must disconnect it when you are finished. Do not disconnect the tape drive until you are certain that the migration is successful.

ATTENTION

You must power down the server before you disconnect the external tape drive.

If the server has an internal tape drive installed, do not remove it.

CallPilot configuration for Symposium Call Center Server integration

- If you intend to use CallPilot to provide messaging services in a Symposium Call Center Server environment, then see “Using CallPilot 2.0 Application Builder for application completion and publication,” on page 301 for more information. You should also review “Configuration checklists,” on page 31.

You must also configure items that are not migrated. For the integration, you need to configure the Service Directory Numbers (if they have not already been configured in CallPilot Manager).

- If you migrated voice segments (voice menus, voice services, or announcements) for GIVE IVR, you must run the Application Builder Data Integrity and Repair tool before you attempt to use Application Builder to create the application. See the next section, “Correcting and moving Application Builder data,” on page 300.

Correcting and moving Application Builder data

If you migrated voice segments (voice menus, voice services, or announcements), you must run the Application Builder Data Integrity and Repair tool before attempting to use Application Builder to create the application. If you do not run the tool first, the voice segments are not visible in Application Builder.

Notes:

- You do not need to run the Application Builder Data Integrity and Repair tool if you migrated Symposium Call Center Server ACCESS voice prompts from Meridian Mail to CallPilot 2.0. The CallPilot 2.0 migration utility automatically creates the applications during the Symposium Call Center Server prompt migration.
- The Application Builder Data Integrity and Repair tool is a specialist tool. You must have the distribution level password for the CallPilot 2.0 support tools available before using the tool.

To run the Application Builder Data Integrity and Repair tool

1 From the Start menu, select Programs → CallPilot → System Utilities → Support Tools.

2 Type the distribution level user name and password.

Result: The Main menu appears.

3 Select option 8, Application Builder Tools.

Result: The Application Builder Tools menu appears.

4 Select option 3, Application Data Integrity and Repair Tools < srepair>.

Result: A prompt appears.

5 Press Enter.

Result: The Application Data Integrity and Repair window appears. (The window may not come to the forefront of the screen, so you may have to locate it.)

6 Click Start.

Result: The Analysis Results window appears.

The Analysis Results window has an upper and lower section. The upper section is called Consistent data found, and the lower section is called Inconsistent data found. The lower section contains the newly migrated prompts.

- 7 Click Repair all.

Result: The process begins. Allow the process to finish. You may view the log for errors.

- 8 Click Close.
- 9 To close the Support Tools window
 - a. Press Enter.
 - b. Select 1 to exit, and then press Enter.
 - c. Select Y, and then press Enter.

The migrated voice segments (voice menus, voice services, and announcements) are repaired and can now be accessed using Application Builder.

Using CallPilot 2.0 Application Builder for application completion and publication

For ACCESS voice services

You do not need to complete applications containing migrated Symposium Call Center Server ACCESS voice prompts. The CallPilot 2.0 migration utility preserves Meridian Mail voice file names and segment numbers. After the prompt migration, voice files are transformed into application names, and prompt segment numbers are transformed into Application Builder voice item IDs.

Normally, Symposium Call Center Server variable definitions should not be changed. The only thing that should change is the ACCESS ACD-DN. You must specify the change to the ACCESS ACD-DN in the Symposium Call Center Server Global Settings window.

For IVR voice services

For GIVE IVR voice service, the information in the Meridian Mail VSDN table is critical, as well as the voice menu structure.

As this data cannot be migrated automatically, you must use CallPilot Application Builder to

- recreate or rebuild the menu or announcement structure extracted from the Meridian Mail VSDN table. Information in the VSDN table can be printed beforehand to assist with the recreation or rebuilding tasks in CallPilot Application Builder.
- manage the applications and the control blocks, and save and complete them
- publish applications in the CallPilot Service DN table

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 may 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 may be given View Only access to historical reports.

ACCESS link

A communication channel between 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 Symposium Call Center Server. Resources must be configured with matching values on both the switch and 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. They are also known as Line of Business (LOB) codes. For example, the activity code 720 may 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 setting up and maintaining Symposium Call Center Server.

agent

A user who is responsible for handling customer calls.

agent logon 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

A means of automatically distributing an organization's incoming calls among a number of answering positions (ACD agents). Automatic call distribution is useful in operations where callers want a service rather than a specific person. Calls are serviced in the order they arrive and are distributed so that the workload at each answering position is approximately equal.

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. *See also* Automatic call distribution directory number.

Automatic call distribution directory number

A DN associated with an ACD group. Calls made to an automatic call distribution directory number 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 are 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 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, script variable.

Calling Line Identification

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.

CSE 1000 switch

Succession Communication Server for Enterprise 1000 switch

Customer local area network

The LAN to which your corporate services and resources connect. Symposium Call Center Server and the 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 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 appears 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. Several applications can use a DLL at the same time.

E**ELAN**

See embedded local area network.

embedded local area network

A dedicated Ethernet TCP/IP LAN that connects the server in Symposium Call Center Server and the switch.

Emergency key

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

event

1. An occurrence or action on 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 below 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. You can only change the value of a global variable in the Script Variable Properties sheet. You cannot change it in a script. *See also* call variable, variable.

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 logon or logoff. An IVR event is pegged in the database when a call acquires or de-acquires 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 to 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

M1 IE

Meridian 1 Internet Enabled switch

Management Information Base

A data structure that describes the collection of all possible objects in a network. Each managed node maintains one or more variables (objects) that describe its state. Symposium Call Center Server Management Information Bases (MIBs) contribute to the overall network MIB by

- identifying Nortel Networks/Meridian/Symposium Call Center Server nodes within the network
- identifying significant events (SNMP traps), such as alarms reporting
- specifying formats of alarms

Master script

The first script executed when a call arrives at 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, 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.

MIB

See Management Information Base.

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, 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, 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, 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, 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, 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, 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, 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. 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 systematic way of monitoring and managing a computer network. The SNMP model consists of four components:

- managed nodes, which are any device, such as hosts, routers, and printers, capable of communicating status to the outside world via an SNMP management process called an SNMP Agent
- management stations, which are computers running special network management software that interact with the Agents for status
- management information, which is conveyed through exact specifications and format of status specified by the MIB
- Management Protocol or SNMP, which sends messages called protocol data units (PDUs)

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.

SNMP

See Simple Network Management Protocol.

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 and 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 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 Transmission 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, traffic intrinsic.

Token Ring

A PC network protocol developed by IBM. A Token Ring network is a type of computer network in which all the 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, 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.

Transmission 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 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 also 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 Symposium Call Center Server.

Index

Numerics

2008 set TN, 19

A

ACCESS

- configuring Meridian Mail for, 125–131
- default treatment DN, 155
- determining if enabled, 56

ACCESS channels, 116

ACCESS IVR ACD-DNs

- adding VSDN entries for, 125
- configuring in Meridian Mail, 125

ACCESS link, 23

- configuring, 56–59

ACCESS voice ports, 112, 120

- configuring in CallPilot, 114
- configuring in Meridian Mail, 126
- configuring IVR ACD-DNs for, 71
- prerequisites for configuring, 126

ACD phonesets

- configuring as associated sets, 94, 95
- digital, 79
- single-line, 82

ACD Waiting Calls (AWC) key/lamp, 226

ACD-DNs, 27

- configuring on the switch, 62

ACNT key, 79

acquiring

- agent phonesets, 152
- CDNs, 150
- IVR ACD-DNs, 151
- resources for voice processing, 110
- TNs, 152

activity codes, deactivating Not ready, 78

Activity key, 79

adding

- announcements, 132

Meridian Mail mailbox passwords, 128

Meridian Mail mailboxes, 128

phonesets as voice ports, 153

voice menus, 132

voice segments, 118, 129

VSDNs for ACCESS IVR ACD-DNs, 125

VSDNs for Give IVR ACD-DNs, 135

VSDNs for non-ACCESS IVR ACD-DNs,
135

agent phoneset display, 221

examples, 224

agent phonesets

acquiring, 152

configuring on the server, 152

configuring on the switch, 78–82

agent TNs, virtual, 28

agents and DSP ports, 122

announcement services, 132

announcements

creating, 132

using voice menus as, 133

Application Builder

Application Builder Data Integrity and Repair
tool, 300

comparison with Voice Prompt Editor, 275

using for application completion and
publication, 301

associated sets

configuring digital ACD phonesets as, 94

configuring single-line ACD phonesets as, 95

associating routes with TNs, 85

audits, and the Voice Service DN table, 123

B

broadcast messages, 107

broadcast ports

maximum number of, 154

wait timer, 155

Broadcast Voice Port Wait Timer box, 253, 269
broadcast voice ports. *See* broadcast ports

C

CallPilot, 17, 18, 20

communication with Symposium Call Center Server, 18

configuring ACCESS link on, 57

configuring as a voice processing engine, 113–118

connecting to server, 57

connection to switch, 18

connections, 21

creating IVR ACD-DNs for, 72

creating voice ports for, 75

determining whether ACCESS is enabled on, 56

testing voice services, 162

CallPilot Configuration Wizard, 116

CDNs, 27

acquiring, 150

changing on the switch, 67

configuring for host-enhanced routing, 97

configuring on the server, 150

configuring on the switch, 64–65

scripts and, 160

verifying, 65

changing

CDN configurations on the switch, 67

Meridian Mail mailbox passwords, 128

checking

ELAN links, 54

for non-Symposium Call Center Server calls, 206

checklists

for backing out of the voice service migration, 265

for configuration, 31

for general setup, 31

for Give Controlled Broadcast, 41

for GIVE IVR with Meridian Mail, 33

for GIVE IVR with third-party voice processing systems, 39

for voice services migration, 238

for voice services pre-migration, 233

for voice sessions, 41

CLAN, 22

CLAN address of Symposium Call Center Server, 116

CLAN IP Address field, 116

Command and Status Link, 23

Communication Server for Enterprise 1000 switch, 15

Computer Telephony Integration. *See* CTI configuration

checklists, 31

overview, 26

recommended sequence, 29

Configuration Wizard, 116

configuring

ACCESS IVR ACD-DNs in Meridian Mail, 125

ACCESS link, 56–59

ACCESS voice ports in Meridian Mail, 126

agent phonesets on the server, 152

CallPilot as a voice processing engine, 113–118

CDNs for host-enhanced routing, 97

CDNs on the server, 150

CDNs on the switch, 64–65

ELAN, 53–55

Give IVR ACD-DNs in Meridian Mail, 134

Give IVR voice ports in Meridian Mail, 136

IVR ACD-DNs on the server, 151

IVR ACD-DNs on the switch, 71–74, 242

Meridian Mail as a voice processing system, 119–124

Meridian Mail for ACCESS, 125–131

Meridian Mail for Give IVR, 132–135

MLS, 89–190

music routes on the switch, 84

Night Routing table on the switch, 68–70

non-ACCESS IVR ACD-DNs in Meridian Mail, 134

non-ACCESS voice ports in Meridian Mail, 136

phonesets for CTI, 94

phonesets on the switch, 78–82

RAN routes on the switch, 84

routes on the switch, 83–85

- subsystems, 27, 52
- TNs on the server, 152
- treatment DN, 134
- voice ports on the switch, 75–77, 243
- voice prompts, 130
- connecting voice processing system and server, 57
- connectivity
 - and switch, 17
 - and Symposium subsystems, 17
- continuous mode, 108
- Controlled Broadcast Announcements,
 - maximum number of ports, 154
- COT trunk types, 83
- creating
 - announcements, 132
 - Meridian Mail mailbox passwords, 128
 - Meridian Mail mailboxes, 128
 - voice menus, 132
 - voice segments, 118, 129
- CSE 1000 switch, 15
- CSL link, 23
- CTI, 80, 93
 - configuring phonesets for, 94
- Customer LAN, 22

D

- deactivating Not Ready activity codes, 78
- default
 - ACCESS treatment DN, 155
 - treatment DN, 135
- Default Access IVR DN box, 254, 269
- Default Access Treatment DN box, 254, 269
- determining whether ACCESS is enabled, 56
- DID trunk types, 83
- digital ACD phonesets, 79
 - configuring as associated sets, 94
- digital non-ACD phonesets, configuring as
 - associated sets, 95
- Display Waiting Calls key, 80, 221
 - and agent phoneset displays, 221
 - and skillset information, 222
 - and supervisor phoneset displays, 222
 - display format, 222

- Display Waiting Calls lamp, 221, 225
- DSP ports and agents, 122
- DWC key. *See* Display Waiting Calls key

E

- ELAN, 22
 - checking links, 54
 - configuring, 53–55
 - enabling CTI operations on, 93
- embedded LAN. *See* ELAN
- enabling CTI, 80, 93
- End Voice Session commands, 106
 - using for voice processing, 104
- external IVR systems
 - and scripts, 140
 - using with Give IVR, 140

F

- FEX trunk types, 83
- FGDT trunk types, 83
- frequently asked questions, 215–217
- front-end IVR, and voice processing, 102

G

- general Meridian Mail services, 112
- Give Controlled Broadcast Announcement
 - command
 - and CallPilot, 19
 - and mailbox passwords, 155
 - and Meridian Mail, 19
 - checklist for configuration, 41
 - typical uses of, 105
 - using for voice processing, 103
- Give IVR ACD-DNs
 - adding VSDNs for, 135
 - configuring in Meridian Mail, 134
- Give IVR command
 - and CallPilot, 19
 - and Meridian Link, 20
 - and Meridian Mail, 19
 - configuration checklist, 33

- configuring Meridian Mail for, 132–135
- typical uses of, 105
- using external IVR systems with, 140
- using for voice processing, 103
- using third-party IVR systems with, 140

Give IVR voice ports, 111, 120

- configuring in CallPilot, 115
- configuring in Meridian Mail, 136
- prerequisites for configuring in Meridian Mail, 136

Give RAN command

- typical uses of, 105
- using for voice processing, 103

global settings, for IVR ACD-DNs, 154, 156–157

H

host-enhanced routing, 91, 97

host-enhanced voice processing, 91

I

initializing the switch, 87

interacting with callers

- directly using script commands, 103
- indirectly using script commands, 103

interactive voice processing mode, 102

intermittent voice processing treatment, 209

interruptible operation, 109

IVR ACD-DN Name box, 249

IVR ACD-DN Number box, 249

IVR ACD-DNs, 27

- acquiring, 151
- configuring on the server, 151
- configuring on the switch, 71–74, 242
- global settings, 154, 156–157
- scripts and, 160

IVR channels, 116

L

listen only mode, 107

M

Mailbox box, 254, 269

mailboxes, 27

- adding, 128
- creating, 128
- passwords and, 155

maximum number of broadcast ports, 154

Meridian 1 Internet Enabled switch, 15

Meridian 1 Option 11C Mini switch, 15

Meridian Customer Defined Network, 68

Meridian Link, 20, 91

Meridian Link Services. *See* MLS

Meridian Link traces, 189

Meridian Mail, 17, 19

- changing mailbox passwords, 128
- communication with Symposium Call Center Server, 18
- configuring ACCESS IVR ACD-DNs in, 125
- configuring ACCESS link on, 58
- configuring ACCESS voice ports in, 126
- configuring as a voice processing system, 119–124
- configuring for ACCESS, 125–131
- configuring for Give IVR, 132–135
- configuring Give IVR ACD-DNs in, 134
- configuring Give IVR voice ports in, 136
- configuring non-ACCESS IVR ACD-DNs in, 134
- configuring non-ACCESS voice ports in, 136
- connecting to server, 57
- connection to switch, 19
- connections, 22
- creating IVR ACD-DNs for, 73
- creating mailbox passwords, 128
- creating voice ports for, 77
- determining whether ACCESS is enabled on, 57
- migrating voice prompts, 227
- tape drive and media, 236

Meridian Mail mailboxes, 155

- creating, 128

Meridian MAX, 86

messages

- callers hearing incorrect, 210
- playing to callers, using script commands, 102

methods of voice processing, 102
 front-end IVR, 102
 script language commands, 102

migration
 utility preparation tape, 288
 voice services, 232
 voice services post migration activities, 299

MIRAN cards, 71

MLS, 20
 configuring, 89–190

MQA, 86
 agents, 86

multi-line ACD phonesets. *See* digital ACD phonesets

Multiple Queue Assignments. *See* MQA

music routes, configuring on the switch, 84

N

NACD ACD-DNs and scripts, 160

NACD routing, 68

NACD-DNs, configuring, 68–70

Night Routing table, configuring on the switch, 68–70

nightly audits, and the Voice Service DN table, 123

non-ACCESS IVR ACD-DNs
 adding VSDN entries for, 135
 configuring in Meridian Mail, 134

non-ACCESS voice ports, 71, 111, 120
 configuring in CallPilot, 115
 configuring in Meridian Mail, 136
 prerequisites for configuring in Meridian Mail, 136

non-ACD phonesets, configuring as associated sets, 95, 96

non-interruptible operation, 109

non-Symposium calls terminating on voice ports, 206

Not Ready reason codes, 79
 deactivating, 78

Number of IVR Ports Reserved for Broadcasts box, 253, 268

O

Open Voice Session commands, 106
 and CallPilot, 19
 Meridian Mail and, 19
 using for voice processing, 104

operation modes, 109
 broadcast, 107
 continuous, 108
 for voice processing commands, 107–109
 interruptible, 109
 listen only, 107
 non-interruptible, 109
 single connection, 107
 start/stop, 108

Option 11C Mini switch, 15

overview
 of configuration, 26
 of subsystems, 16
 of subsystems configuration, 52

P

partitioning voice ports, 111

passive voice processing mode, 102

Password box, 254, 269

passwords, 27
 and Meridian Mail mailboxes, 155
 changing for Meridian Mail mailboxes, 128
 creating for Meridian Mail mailboxes, 128

phoneset keys, unsupported, 226

phonesets, 27
 adding as virtual agents, 153
 adding as voice ports, 153
 configuring for CTI, 94
 configuring on the switch, 78–82

physical connection between voice processing system and server, establishing, 57

playing messages to callers using script commands, 102

predefined voice segments, 159

prerecorded voice segments, 159

prerequisites for configuring IVR ACD-DNs
 global setting, 156

problems with subsystem links, 191–205

Q

queues, 27

R

RAN routes, configuring on the switch, 84
resources for voice processing, acquiring, 110
routes, 27
 associating with TNs, 85
 configuring on the switch, 83–85
 scripts and, 161

S

sample phoneset displays, 223
script example, using voice segments, 159
script language commands
 for voice processing, 102
 Give Controlled Broadcast Announcement, 103
 Give IVR, 103
 Give RAN, 103
 Open/End Voice Session, 104
scripts, 27
 and third-party IVR systems, 140
 CDNs and, 160
 external IVR systems and, 140
 IVR ACD-DNs and, 160
 NACD ACD-DNs and, 160
 routes and, 161
 skipping voice processing commands, 208
 testing CallPilot integration, 162
 treatment DNs and, 160
 voice processing and, 160
SDN table, 117
sequence of configuration, 29
server, connecting voice processing system to, 57
Service Directory Number table, 117
setup checklists, 31
signaling links, 17
silence, callers hearing, 209
single connection, 107
single-line ACD phonesets, 82

 configuring as associated sets, 95
single-line non-ACD phonesets, configuring as
 associated sets, 96
SL1 set TN, 18, 19
start/stop mode, 108
subsystems
 and communication with Symposium Call
 Center Server, 18
 configuration overview, 52
 configuration reference, 27
 link problems, 191–205
 overview, 16
Succession CSE 1000 switch, 15
supervisor phonesets, configuring on the switch,
 78–82
switch
 and connectivity, 17
 CallPilot connection to, 18
 changing CDNs on, 67
 communication with Symposium Call Center
 Server, 18
 configuring CDNs on, 64–65
 configuring IVR ACD-DNs on, 71–74
 configuring music routes on, 84
 configuring Night Routing table on, 68–70
 configuring phonesets on, 78–82
 configuring RAN routes on, 84
 configuring routes on, 83–85
 configuring voice ports on, 75–77
 initializing, 87
 Meridian Mail connection to, 19
 verifying acquisition of CDNs on, 65
Symposium Call Center Server
 ACCESS voice ports, 112
 communication with subsystems, 18
 Give IVR Ports, 111
Symposium subsystems and connectivity, 17
Symposium Voice Services on CallPilot
 creating IVR ACD-DNs for, 72
 creating voice ports for, 75
Symposium Voice Services on Meridian Mail
 creating IVR ACD-DNs for, 73
 creating voice ports for, 77
sys-loading the switch, 87
system-predefined phrases, 159

T

- tape drives, 236
 - server connection, 287
- Telephony/Port Address box, 249
- Terminal Name box, 249
- third-party IVR systems
 - scripts and, 140
 - using with Give IVR, 140
- third-party voice processing systems, 17, 140
 - checklist for configuration, 39
- Threshold Class box, 249
- TIE trunk types, 83
- TNs
 - acquiring, 152
 - and voice processing systems, 17
 - associating routes with, 85
 - configuring on the server, 152
- Trace Tool, 189
- traces, Meridian Link, 189
- treatment DNs, 27
 - configuring, 134
 - default, 135, 155
 - external IVR systems and, 141
 - scripts and, 160
 - third-party IVR systems and, 141
- troubleshooting, 187–217
- trunk routes, configuring on the switch, 83–85
- typical uses of, 106

U

- unsupported keys, 226
- user-defined voice segments, 158
- using
 - external IVR systems with Give IVR, 140
 - Give Controlled Broadcast Announcement, 105
 - Give IVR, 105
 - Give RAN, 105
 - Open/End Voice Session, 106
 - third-party IVR systems with Give IVR, 140
 - voice menus as announcements, 133
 - voice segments, script example, 159

V

- variables, voice segments, 158
- virtual agent TNs, 28, 75
- virtual agents. *See* voice ports
- voice files, 27
 - when to configure, 130
- voice menu services, 133
- voice menus
 - creating, 132
 - using as announcements, 133
 - using to offer choices to callers, 133
- voice paths
 - about, 17
 - and voice processing systems, 17
 - types of, 17
- Voice Port Channel box, 251
- Voice Port Name box, 251
- voice ports, 28
 - ACCESS, 71, 112
 - ACCESS, configuring in CallPilot, 114
 - ACCESS, configuring in Meridian Mail, 126
 - configuring on the server, 153
 - configuring on the switch, 75–77, 243
 - Give IVR, 111
 - Give IVR, configuring in CallPilot, 115
 - Give IVR, configuring in Meridian Mail, 136
 - non-ACCESS, 71, 111
 - non-ACCESS, configuring in CallPilot, 115
 - non-ACCESS, configuring in Meridian Mail, 136
 - partitioning rules, 111
- voice processing
 - acquiring resources for, 110
 - and script language commands, 102
 - and scripts, 160
 - configuring CallPilot for, 113–118
 - configuring Meridian Mail for, 119–124
 - interactive, 102
 - intermittent treatment, 209
 - methods, 102
 - operation modes, 107–109
 - passive, 102
 - types of, 102
 - using front-end IVR, 102

- using Give Controlled Broadcast Announcement, 103
 - using Give IVR, 103
 - using Give RAN, 103
 - using Open/End Voice Session, 104
 - voice processing commands, scripts skipping, 208
 - voice processing system
 - configuring ACCESS link on, 57
 - connecting to server, 57
 - TNs and, 17
 - voice prompts
 - configuring, 130
 - migrating to CallPilot, 227
 - voice segments, 27
 - adding, 118, 129
 - creating, 118, 129
 - predefined, 159
 - prerequisites for configuring, 130
 - script example using, 159
 - user-defined, 158
 - Voice Service DN table, 123
 - and nightly audits, 123
 - voice service DN, 28
 - voice services
 - backing out of the migration, 263
 - comparison of CallPilot and Meridian Mail, 228
 - comparison of VPE and Application Builder, 275
 - data that cannot be migrated, 283
 - gradual migration, 232
 - migrating Symposium Voice Services only, 282
 - migrating the voice processing engine, 281
 - migrating to CallPilot, 227
 - migrating voice prompts, 244
 - migrating voice prompts and voice segments, 279
 - migrating voice segments, 244
 - migration checklist, 238
 - migration limitations, 235
 - migration overview, 232
 - migration utility, 288
 - performing the migration, 285
 - post migration activities, 299
 - pre-migration checklist, 233
 - testing for CallPilot, 162
 - using Application Builder, 301
 - voice session commands and mailbox
 - passwords, 155
 - voice sessions, checklist for configuration, 41
 - VSDN table
 - configuring ACCESS IVR ACD-DNs in, 125
 - configuring Give IVR ACD-DNs in, 134
 - configuring non-ACCESS IVR ACD-DNs in, 134
 - VSDNs
 - adding for ACCESS IVR ACD-DNs, 125
 - adding for Give IVR ACD-DNs, 135
 - adding for non-ACCESS IVR ACD-DNs, 135
- ## W
- WATS trunk types, 83
 - when to configure voice files, 130
 - when to modify the CAT table, 122
 - Windows 2000 Professional, 15
 - Windows 2000 Server, 15



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