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Nortel Symposium Call Center Server

Symposium, M1/Succession 1000, and Voice Processing Guide

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Nortel Symposium Call Center Server

Symposium, M1/Succession 1000, and Voice Processing Guide

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The Standard 2.0 version of the *Nortel Symposium Call Center Server Symposium, M1/Succession 1000, and Voice Processing Guide*, Release 5.0, is released.

April 2004

The Standard 1.0 version of the *Nortel Symposium Call Center Server Symposium, M1/Succession 1000, and Voice Processing Guide,* Release 5.0, is released.

Contents

Chapter 1

Getting started

In this chapter

Overview

Introduction

The *Nortel Symposium Call Center Server Symposium, M1/Succession 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 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 Succession 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, check the following web site:

  https://transportvo.nortelnetworks.com/mpl/mpl

- The switch is running X11 Release 25 or Succession 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 25 or Succession 1000 Release 1.1 or greater
- CallPilot 2.0, or Meridian Mail Release 11 or greater
- Symposium Call Center Server Release 5.0

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

Nortel product knowledge

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

- the appropriate switch type:
  - Meridian 1 Options 11, 11C Mini, 51C, 61C, 81, or 81C (X11 Release 25 or greater)
  - Succession 1000 Release 1.1 or greater
- CallPilot Release 2.0 or greater
- Meridian Mail Release 11 or 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 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

Subsystems overview

Introduction

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

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

  Note: Meridian Mail and CallPilot can coexist in the same environment. For example, you can choose to use a legacy Meridian Mail for voice mailboxes or to provide IVR services, and use CallPilot for voice processing. However, Symposium Call Center Server cannot use voice processing services from both Meridian Mail and CallPilot.
- 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 phonesets	CallPilot or Meridian Mail
2500 phoneset TNs	usually third-party voice processing systems
2500 phoneset ACD agent TNs	usually third-party voice processing systems
T1 TNs	usually third-party voice processing systems
E1 TNs	usually third-party voice processing systems

Signaling links

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

Symposium Call Center Server

The server communicates with the 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 subnet using the AML messaging protocol.

Communicating with CallPilot

Symposium Call Center Server also communicates with CallPilot to instruct it to play prompts, collect digits input by callers, or both.

For basic voice processing (Give IVR), the server communicates with CallPilot over the Nortel server subnet, using the Meridian Link interface.

For advanced voice processing (controlled broadcast and voice session commands), the server tells CallPilot which prompts to play using the ACCESS protocol over the ELAN subnet. Messages from the switch to CallPilot (call arrival messages) travel on the ELAN subnet to Symposium Call Center Server, which then sends them over the Nortel server subnet (using the MLS protocol) to CallPilot.

Communicating with Meridian Mail

Symposium Call Center Server also communicates with Meridian Mail to instruct it to play prompts, collect digits input by callers, or both.

For basic voice processing (Give IVR), the server communicates with Meridian Mail over the ELAN subnet.

For advanced voice processing (controlled broadcast and voice session commands), the server tells Meridian Mail which prompts to play using the ACCESS link. Meridian Mail and the switch communicate using the CSL link.

CallPilot

The CallPilot voice channels connect to the switch by a DS30 cable. On the switch side, this card is configured as an SL1 phoneset 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, the switch alerts CallPilot using Symposium Call Center Server. The alert is sent over the Nortel server subnet, using the Meridian Link interface.

When the Give Controlled Broadcast Announcement or Open Voice Sessions script commands are used, Symposium Call Center Server sends the command and the ACD-DN over the AML (ELAN) connection to the switch. When the call arrives at a CallPilot voice port, the switch alerts CallPilot using Symposium Call Center Server. The alert is sent over the Nortel server subnet, using the Meridian Link interface.

Meridian Mail

Note: The Meridian Mail voice processing system is not available on the Succession 1000 switch.

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 phoneset TN (virtual agent) or, for the Option 11, a 2008 phoneset 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 subnet. 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 (MLink) 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.

Meridian Link Services

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

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 subnet to the switch. Examples of external applications that can register with MLS include Telephony API (TAPI) Service Provider and Computer Telephony Integration (CTI).

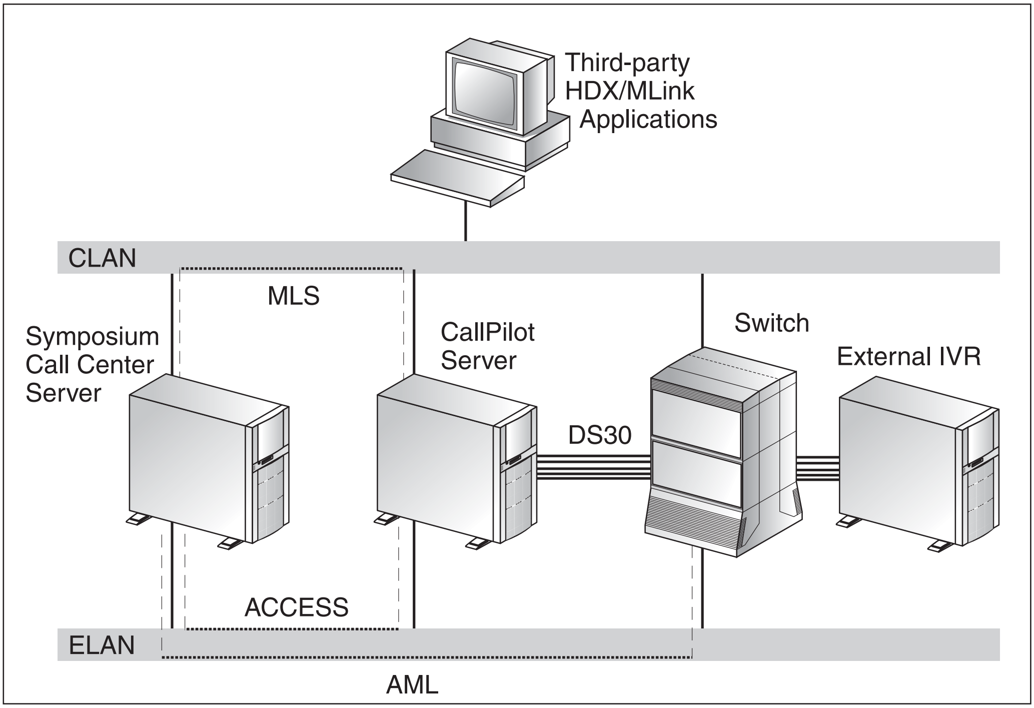
Note: The Give IVR command with Symposium Voice Services on CallPilot requires Meridian Link.

Connections between the subsystems

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

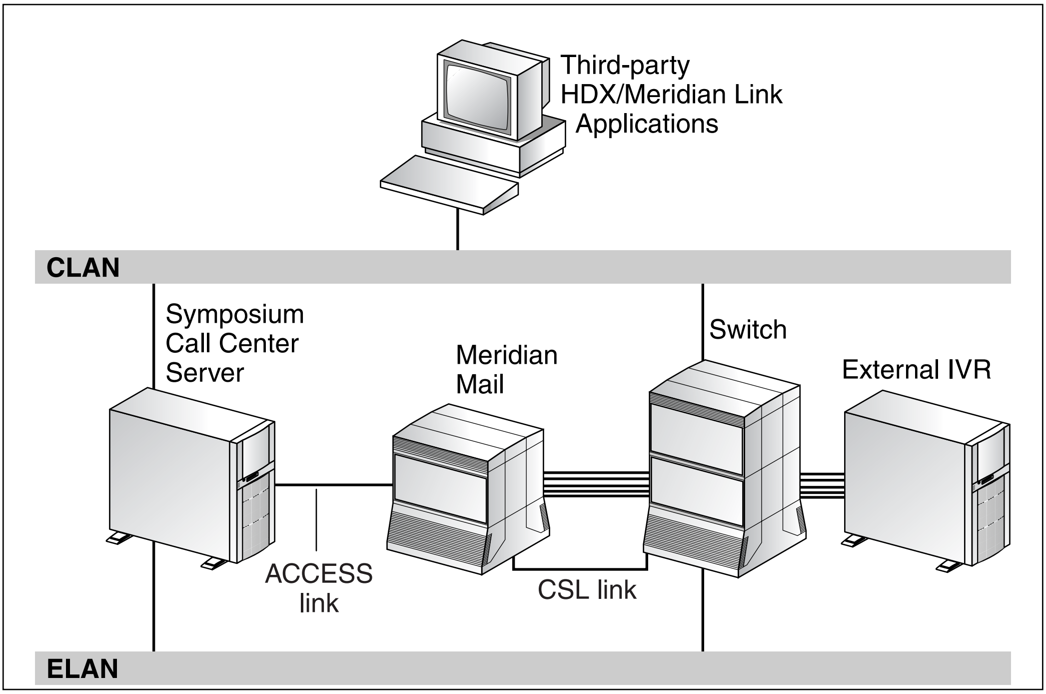
CallPilot

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) subnet:** The ELAN subnet enables Symposium Call Center Server to communicate with the switch, CallPilot, and other servers. The ELAN subnet carries call processing traffic and must be private.
- **Nortel server subnet:** The Nortel server subnet 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 Nortel server subnet 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 (Command and Status Link):** The connection between the switch and Meridian Mail is an RS-232 synchronous link.

What's new in Release 5.0

Multiple servers on a switch

In Symposium Call Center Server Release 5.0, one switch can support up to three Symposium Call Center Servers systems.

The three servers cannot use the same switch resources simultaneously.

Configuring servers

When you configure multiple servers on a switch, you configure each server normally, following the instructions in this guide. All of the servers connect to the same ELAN subnet, although each requires its own port. You must configure the switch to import multiple AML applications.

Servers can belong to the same customer group, or to different customer groups. They need not be in a separate workgroup, but they must have unique computer names.

Using multiple servers with other Nortel products

The following table shows how Nortel products and features interact with multiple servers configured on the same switch.

Product/ Feature	Interaction
Symposium Web Client	You can use a single Symposium Web Client to manage all of the servers installed on your switch. For each server, in the Configuration component, choose Server > Add Server. Then enter the server information.
Classic Client	You can use a single Classic Client to manage multiple servers. Therefore, you can use one Classic Client to manage all of the servers installed on your switch. In the SMI Workbench, add each server as a system (see the *Administrator's Guide*). You can only log on to one server at a time.

Product/ Feature	Interaction
Network Skill-Based Routing (NSBR)	Networking of servers on the same switch is not supported.
CallPilot	If you are using CallPilot to provide front-end IVR, the same CallPilot server can support all three Symposium Call Center Server systems. If you are using Symposium Voice Services on CallPilot—that is, if CallPilot is providing Give IVR or ACCESS voice services (Open/Close Voice Session, Collect Digits, and Give Controlled Broadcast)—CallPilot can serve only one Symposium Call Center Server system.
Meridian Mail	If you are using Meridian Mail to provide front-end IVR, the same Meridian Mail system can support all three Symposium Call Center Server systems. If you are using Symposium Voice Services on Meridian Mail to provide IVR services (that is, with the Give IVR command), the same Meridian Mail can support all three Symposium Call Center Server systems. However, the following restrictions apply: ■ You must allocate the Meridian Mail IVR ports between three IVR queues, and dedicate a queue to each server. ■ All of the servers must belong to the same customer group. (Therefore, you cannot network the servers together.) If you are using Symposium Voice Services on Meridian Mail to provide ACCESS voice services (Open/Close Voice Session, Collect Digits, and Give Controlled Broadcast), Meridian Mail can serve only one Symposium Call Center Server system.
HDX	Multiple servers can access the same database if your HDX application allows all of the servers to register.

Product/ Feature	Interaction
IPML	If you are using the Integration Package for Meridian Link (IPML), an IPML server can support only one Symposium Call Center Server. Each Symposium Call Center Server must have its own IPML server.
TAPI	If you are using the Telephony Application Program Interface (TAPI), a TAPI server can support only one Symposium Call Center Server. Each Symposium Call Center Server must have its own TAPI server.
Agent Greeting/ Remote Observe	Since these features are implemented on the switch, with no dependency on Symposium Call Center Server, multiple servers can share Agent Greeting/Remote Observe.

Chapter 2

Configuration overview

In this chapter

Overview

Subsystems

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

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	
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	x			
Scripts				x
Treatment DNs	x (optional)	x	x	x

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

Creating a system design

Before configuring the subsystems, create a system design based on your call center requirements. Use the Symposium Call Center Server Capacity Assessment Tool (CapTool) to determine the capacity requirements for your call center.

Your design should take the following information into consideration:

- network design (ELAN subnet, Nortel server subnet, and RAS IP requirements)
- projected peak call traffic
- number of agents, shifts, days of operation
- projected call flow/scripting requirements
- voice messaging port requirements (number of ports required for GIVE IVR and ACCESS services (such as voice sessions, controlled broadcasts, and collect digits sessions)

For more information, see the *Planning and Engineering Guide*.

Configuration sequence

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

1. Install 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.

Notes:

- You cannot network multiple servers connected to the same switch.
- Configure networking after each of the individual servers in your network is configured and operational.
- Release 1.1 of the Succession 1000 switch only supports IP line and trunk networking. IP peer networking is supported on Release 2.0.

To configure networking, you must configure the following components:

- **the switch**—Install and configure the NACD package, and create the CDNs to which networked calls for each server will be routed.

ATTENTION

If you are using Network Skill-Based Routing, you must define a Home Location Code (HLOC) in Overlay 15 (for detailed instructions, see your switch documentation). Symposium Call Center Server uses this code to generate network call IDs for networked calls.

- **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*.

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 or Succession 1000 switches 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 Succession 1000).

General checklist

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

Subsystem	Description	✔
Switch	Meridian 1/Succession 1000 switch is up and running correctly.	
Switch	Required Meridian 1/Succession 1000 software packages are available (use LD 22 to verify the packages). (See “Before you begin” on page 70 for more information.)	

Subsystem	Description	✔
Switch	The following additional switch requirements are available: ■ Meridian 1 dependency PEPs for Symposium Call Center Server Release 5.0 ■ Meridian 1 dependency PEPs for the specific voice processing subsystem you are using ■ provisioning of sufficient Call Registers	
Switch	ELAN subnet connection between Symposium Call Center Server and Meridian 1/Succession 1000 switch is functioning. Note the current ELAN subnet/VAS ID number using LD 48. (This can be useful for troubleshooting.)	
Switch	Symposium Call Center Server CDNs are configured on the switch. (See “Configuring CDNs on the switch” on page 73.)	
Switch	Agent ACD queues are configured in LD 23. (See the switch documentation for detailed instructions.)	
Switch	Agent and supervisor TNs are configured on the switch. (See “Configuring agent and supervisor TNs on the switch” on page 89.)	
Switch	Trunk, music, and RAN routes are configured on the switch. (See “Configuring routes on the switch” on page 95.)	

Give IVR checklists

Using CallPilot

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

Subsystem	Description	✔
	The general checklist is complete.	
CallPilot	CallPilot is up and running for voice messaging.	
Switch and CallPilot	CallPilot is in communication with the switch.	
Switch	The link between the switch and the CallPilot server is enabled for CTI operations.	
Switch	The IVR ACD-DN (for IVR voice ports) is configured on the switch (see "Configuring IVR ACD-DNs on the switch" on page 81). **Note:** When you configure the IVR ACD-DN on the switch, set the IVR and ALOG prompts to YES.	
Switch	IVR voice ports are configured on the switch as virtual agents (see "Configuring voice ports on the switch" on page 85). **Note:** Voice ports must be dedicated to Symposium Call Center Server GIVE IVR voice service.	
Symposium Call Center Server	CallPilot connection parameters are configured for TCP voice connection, as follows: ■ CallPilot connection IP (ELAN network interface IP address) is set. ■ CallPilot connection port is set (TCP port should be 10008). See "Configuring the voice connection" on page 159 for more information.	

Subsystem	Description	✔
CallPilot	The CallPilot server configuration is updated and integration is enabled. In the CallPilot Configuration Wizard, review all CallPilot configuration information up to and including the Switch Information page. On the Switch Information page: ■ Enable Symposium Call Center Server Integration is selected. ■ The customer number is entered in the Switch Customer Number box. ■ The Nortel server subnet address of Symposium Call Center Server is entered in the Symposium Call Center Server CLAN IP Address field.	
CallPilot	Voice ports (IVR voice channels) are configured in the CallPilot server configuration. 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 127 for more information.) **Notes:** ■ TN of the voice channel in CallPilot 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 = ACD-DN on the switch = IVR ACD-DN on Symposium Call Center Server.	
CallPilot	The SDN table contains the CallPilot Primary CDN. It has an application name of Voice Messaging and a media type of Voice.	

Subsystem	**Description**	✔
CallPilot	The SDN table contains Symposium Call Center Server IVR ACD-DNs. IVR ACD-DNs are configured as follows: ■ Application name is Symposium Voice Services. ■ Media type is Voice.	
CallPilot	Symposium Call Center Server Treatment DNs are defined as Service DN entries in the SDN table. They must have an application name, and their media type must be Voice. (Use meaningful application names because CallPilot can store a large number of applications.) **Notes:** ■ 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.	
Symposium Call Center Server	IVR ACD-DN is configured and acquired on Symposium Call Center Server (see “Configuring IVR ACD-DNs on the server” on page 165).	
Symposium Call Center Server	In Symposium Call Center Server, VSM and MLSM services are running. (Use the VSM Status window if it is configured.)	

<table>
<tr><th>Subsystem</th><th>Description</th><th>✔</th></tr>
<tr><td>Symposium Call Center Server</td><td>Voice ports are configured and acquired on Symposium Call Center Server (see “Configuring voice ports on the server” on page 167).
Notes:
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.</td><td></td></tr>
<tr><td>CallPilot</td><td>The acquired voice ports are Idle in CallPilot.
Note: Use the Channel Monitor in CallPilot Manager to check voice port status. If the voice ports are not initialized, then restart CallPilot.</td><td></td></tr>
<tr><td>Symposium Call Center Server</td><td>GIVE IVR test script is implemented and activated before activating the rest of the voice processing scripts.
The voice processing script command uses both the IVR ACD-DN and a Treatment DN, as in the following examples:
<code>GIVE IVR ivr_queue WITH TREATMENT welcome_msg</code>
<code>GIVE IVR 6000 WITH TREATMENT 1001</code>
(See “Defining scripts” on page 174 for more information.)
Note: 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 SDN table in CallPilot for proper operation.</td><td></td></tr>
</table>

Subsystem	Description	✔
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: ▪ Other scripts are verified and any additional voice processing scripts are implemented. ▪ SDN table is updated as new treatments are required. Test script is not working: ▪ Start troubleshooting procedures.	

Using Meridian Mail

Note: Meridian Mail is not available on the Succession 1000 switch.

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

Subsystem	Description	✔
	The general checklist is complete.	
Meridian Mail	Meridian Mail is up and running for voice messaging.	
Switch and Meridian Mail	Meridian Mail is in communication with the switch.	
Switch	The IVR ACD-DN (for IVR voice ports) is configured on the switch (see "Configuring IVR ACD-DNs on the switch" on page 81). **Note:** When you configure the IVR ACD-DN on the switch, set the IVR and ALOG prompts to YES.	

Subsystem	**Description**	✔
Switch	IVR voice ports are configured on the switch as virtual agents (see “Configuring voice ports on the switch” on page 85). **Note:** Voice ports must be dedicated to Symposium Call Center Server GIVE IVR voice service.	
Meridian Mail	Symposium Call Center Server IVR ACD-DNs are configured in the VSDN table. They must be associated with a Voice Messaging service type.	
Meridian Mail	Symposium Call Center Server Treatment DNs are configured in the VSDN table. **Notes:** ■ 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 147 for more information.	
Meridian Mail	IVR ACD-DN and IVR ports are defined in the Channel Allocation Table (see “Configuring IVR voice ports” on page 151). **Notes:** ■ TN of the voice channel in 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 Meridian Mail = ACD-DN of the TN on the switch = number of IVR ACD-DN on Symposium Call Center Server.	

Subsystem	**Description**	✔
Symposium Call Center Server	IVR ACD-DN is configured and acquired on the Symposium Call Center Server (see “Configuring IVR ACD-DNs on the server” on page 165).	
Symposium Call Center Server	In Symposium Call Center Server, the VSM service is running. (Use the VSM Status window if it is configured.)	
Symposium Call Center Server	Voice ports are configured and acquired on Symposium Call Center Server (see “Configuring voice ports on the server” on page 167). **Notes:** ▪ 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.	
Meridian Mail	The acquired voice ports are Idle in Meridian Mail. **Note:** Use the DSP Port Status (in the Meridian Mail Administration Terminal) to check voice port status.	

Subsystem	Description	✔

Symposium Call Center Server	GIVE IVR test script is implemented and activated before activating the rest of the voice processing scripts.
	The voice processing script command uses both the IVR ACD-DN and a Treatment DN, as in the following examples:
	`GIVE IVR ivr_queue WITH TREATMENT welcome_msg`
	`GIVE IVR 6000 WITH TREATMENT 1001`
	(See “Defining scripts” on page 174 for more information.)
	Note: 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 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: ▪ Other scripts are verified and any additional voice processing scripts are implemented. ▪ The VSDN table is updated as new treatments are required. Test script is not working: ▪ 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 device, 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.	
Switch	The IVR ACD-DN is configured on the switch (see the documentation provided with your IVR system). **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 analog agents (see the documentation provided with your IVR system). **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 165).	

Subsystem	Description	✔
Symposium Call Center Server	Voice ports are configured and acquired on the server (see "Configuring voice ports on the server" on page 167). **Notes:** ▪ 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. ▪ *All* 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	If IPML exists, the treatment DN is specified in the script. **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.	
Symposium Call Center Server	Voice ports behind the IVR ACD-DN acquired by the server are indicated by the switch and show their state as either Logged In or Logged Out. Ports must be Logged In for the switch to connect calls to the port for voice processing.	

Controlled Broadcast and Voice Sessions checklist

Using CallPilot

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

Subsystem	Description	✔
	The general checklist is complete.	
CallPilot	CallPilot is up and running for voice messaging.	
Switch and CallPilot	CallPilot is in communication with the switch.	
Switch	Link between the switch and the CallPilot server is enabled for CTI operations.	
Switch	The ACCESS ACD-DN (for ACCESS voice ports) is configured on the switch (see “Configuring IVR ACD-DNs on the switch” on page 81). **Note:** When you configure the ACD-DN on the switch, set the IVR and ALOG prompts to YES.	
Switch	ACCESS voice ports are configured on the switch as virtual agents (see “Configuring voice ports on the switch” on page 85). Voice ports must be dedicated to Symposium Call Center Server ACCESS voice service.	
Symposium Call Center Server	CallPilot connection parameters are configured for TCP voice connection, as follows: ▪ CallPilot connection IP (ELAN network interface IP address) is set. ▪ CallPilot connection port is set (TCP port should be 10008). See “Configuring the voice connection” on page 159 for more information.	

Subsystem	Description	✔
Meridian Mail	If you are changing voice services from Meridian Mail to CallPilot, ensure that the following conditions are met: ■ Meridian Mail resources are deacquired on Symposium Call Center Server (IVR and ACCESS ACD-DNs, TNs, voice ports). ■ The ACCESS cable is physically disconnected from the Symposium Call Center Server COM port. The other end of the cable is disconnected from the Meridian Mail utility card.	
CallPilot	The CallPilot server configuration is updated and integration is enabled. In the CallPilot Configuration Wizard, review all CallPilot configuration information up to and including the Switch Information page. On the Switch Information page: ■ Enable Symposium Call Center Server Integration is selected. ■ The customer number is entered in the Switch Customer Number box. ■ The Nortel server subnet address of Symposium Call Center Server is entered in the Symposium Call Center Server CLAN IP Address field.	

<table>
<tr><th>Subsystem</th><th>Description</th><th>✔</th></tr>
<tr><td>CallPilot</td><td>Voice ports (ACCESS channels) are configured in the CallPilot server configuration.
In the CallPilot Configuration Wizard, identify and configure the channels that provide ACCESS services to Symposium Call Center Server. (See “ACCESS voice ports” on page 126 for more information.)
Notes:
Voice port channel number on Symposium Call Center Server = Class ID of the TN on CallPilot.
TN of voice channel in CallPilot = 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 = ACD-DN on switch = ACCESS ACD-DN on Symposium Call Center Server.</td><td></td></tr>
<tr><td>CallPilot</td><td>The SDN table contains the CallPilot Primary CDN. It has an application name of Voice Messaging and a media type of Voice.</td><td></td></tr>
<tr><td>CallPilot</td><td>SDN table in CallPilot is updated with Symposium Call Center Server ACCESS ACD-DNs. ACD-DNs are configured as follows:
Application name is Symposium Voice Services.
Media type is Voice.</td><td></td></tr>
<tr><td>Symposium Call Center Server</td><td>ACCESS ACD-DN is configured and acquired on the Symposium Call Center Server (see “Configuring the ACCESS IVR ACD-DN” on page 140).</td><td></td></tr>
</table>

Subsystem	Description	✔
Symposium Call Center Server	IVR ACD-DN global settings are configured on Symposium Call Center Server (see “Configuring IVR ACD-DN global settings” on page 168) to specify the following: ▪ default ACCESS Treatment DN and ACCESS ACD-DN (these have the same value) ▪ number of ACCESS ports reserved for Broadcasts ▪ Broadcast Voice Port Wait Timer ▪ the mailbox number and password set to non-zero values	
Symposium Call Center Server	Voice ports are configured and acquired on Symposium Call Center Server (see “Configuring voice ports on the server” on page 167). **Notes:** ▪ 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	ACCESS link between Symposium Call Center Server and CallPilot is functioning correctly. In Symposium Call Center Server, VSM and MLSM services are running. (Use the VSM Status window if it is configured.)	
CallPilot	The acquired voice ports are Idle. Use the Channel Monitor in CallPilot Manager to check voice port status. If the voice ports are not initialized, then restart CallPilot.	

Subsystem	Description	✔
Symposium Call Center Server, CallPilot	Voice segments are defined, as follows: ▪ 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. **Notes:** ▪ Use Application Builder to define voice items (referred to in Symposium Call Center Server as voice segments or prompts). In Symposium Call Center Server, the variables are defined as <application name> : <segment ID> (For more information, see “Test scripts” on page 199.) Application names are case-sensitive. ▪ Voice items or voice prompts must first be created in Application Builder before they can be defined in voice segment variables used in Symposium Call Center Server scripts. For more information, see ▪ CallPilot Application Builder Guide ▪ “Defining voice segment variables” on page 172	

<table>
<tr><th>Subsystem</th><th>Description</th><th>✔</th></tr>
<tr><td>Symposium Call Center Server</td><td>ACCESS test script is implemented and activated before activating the rest of the voice processing scripts.

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.

For example:

<code>GIVE CONTROLLED BROADCAST ANNOUNCEMENT PLAY PROMPT VOICE SEGMENT closed_message_vs</code>

<code>OPEN VOICE SESSION PLAY PROMPT VOICE SEGMENT hold_option_vs</code>

<code>COLLECT 1 DIGITS INTO hold_choice_cv</code>

<code>END VOICE SESSION</code>

(See “Defining scripts” on page 174 for more information.)</td><td></td></tr>
<tr><td>Symposium Call Center Server</td><td>Master script is activated.

Note: Scripts containing voice processing instructions should be activated after all relevant subsystem configurations.</td><td></td></tr>
<tr><td>Symposium Call Center Server</td><td>Controlled Broadcast and Open Voice Session test script is working.

■ Other scripts are verified and any additional voice processing scripts are implemented.
■ Voice segment variables are updated as required.

Test script is not working.

■ Start troubleshooting procedures.</td><td></td></tr>
</table>

Using Meridian Mail

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	✔
	The general checklist is complete.	
Meridian Mail	Meridian Mail is up and running for voice messaging.	
Switch and Meridian Mail	Meridian Mail is in communication with the switch.	
Switch	The ACCESS ACD-DN (for ACCESS voice ports) is configured on the switch (see “Configuring IVR ACD-DNs on the switch” on page 81). **Note:** When you configure the ACD-DN on the switch, set the IVR and ALOG prompts to YES.	
Switch	ACCESS voice ports are configured on the switch as virtual agents (see “Configuring voice ports on the switch” on page 85). Voice ports must be dedicated to Symposium Call Center Server ACCESS voice service.	
Symposium Call Center Server	Voice connection type (on Symposium Call Center Server client) is set to serial. See “Configuring the voice connection” on page 159 for more information.	
Meridian Mail	Symposium Call Center Server ACCESS ACD-DNs are configured in the VSDN table. They must be associated with an ACC service type.	

Subsystem	Description	✔
Meridian Mail	ACCESS ACD-DN and ACCESS ports are defined in the Channel Allocation Table (see “Configuring ACCESS voice ports” on page 141). **Notes:** ■ TN of the voice channel in 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 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 the Symposium Call Center Server (see “Configuring the ACCESS IVR ACD-DN” on page 140).	
Symposium Call Center Server and Meridian Mail	Meridian Mail mailbox and password are configured. ■ An appropriate Meridian Mail mailbox password policy is defined for Symposium Call Center Server. ■ 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. **Note:** If the password expires, you cannot access the Voice Prompt Editor and callers cannot hear the prompts because Symposium Call Center Server cannot open the mailbox.	

Subsystem	Description	✔
Symposium Call Center Server	IVR ACD-DN global settings are configured on Symposium Call Center Server (see “Configuring IVR ACD-DN global settings” on page 168) to specify the following: ▪ default ACCESS Treatment DN and ACCESS ACD-DN (these have the same value) ▪ number of ACCESS ports reserved for Broadcasts ▪ Broadcast Voice Port Wait Timer ▪ the ACCESS mailbox number and password configured in Meridian Mail for voice prompts	
Symposium Call Center Server	Voice ports are configured and acquired on the Symposium Call Center Server (see “Configuring voice ports on the server” on page 167). **Notes:** ▪ 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	ACCESS link between Symposium Call Center Server and Meridian Mail is functioning correctly. In Symposium Call Center Server, the MLSM service is running.	
Meridian Mail	The acquired voice ports are Active in Meridian Mail. **Note:** Use the DSP Port Status (in the Meridian Mail Administration Terminal) to check voice port status.	

<table>
<tr><th>Subsystem</th><th>Description</th><th>✔</th></tr>
<tr><td>Symposium Call Center Server, Meridian Mail</td><td>Voice segments are defined, as follows:
■ 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.
Notes:
■ Use the Voice Prompt Editor (VPE) to create or delete voice files and record voice segments.
Voice segments created with VPE are stored in Meridian Mail. The variables are defined as
<code><voice file name> : <segment number></code>
Voice file names are case-sensitive.
■ Voice items or voice prompts must first be created in VPE before they can be defined in voice segment variables used in Symposium Call Center Server scripts.
For more information, see
■ “Configuring Meridian Mail for ACCESS” on page 140
■ “Defining voice segment variables” on page 172</td><td></td></tr>
</table>

<table>
<tr><th>Subsystem</th><th>Description</th><th>✔</th></tr>
<tr><td>Symposium Call Center Server</td><td>ACCESS test script is implemented and activated before activating the rest of the voice processing scripts.

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.

For example:

<code>GIVE CONTROLLED BROADCAST ANNOUNCEMENT PLAY PROMPT VOICE SEGMENT closed_message_vs</code>

<code>OPEN VOICE SESSION PLAY PROMPT VOICE SEGMENT hold_option_vs</code>

<code>COLLECT 1 DIGITS INTO hold_choice_cv</code>

<code>END VOICE SESSION</code>

(See “Defining scripts” on page 174 for more information.)</td><td></td></tr>
<tr><td>Symposium Call Center Server</td><td>Master script is activated.

Note: Scripts containing voice processing instructions should be activated after all relevant subsystem configurations.</td><td></td></tr>
<tr><td>Symposium Call Center Server</td><td>Controlled Broadcast and Open Voice Session test script is working.
▪ Other scripts are verified and any additional voice processing scripts are implemented.
▪ Voice segment variables are updated as required.

Test script is not working.
▪ Start troubleshooting procedures.</td><td></td></tr>
</table>

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 (for detailed instructions, refer to your switch documentation):

Subsystem	Description	✔
Switch	The NACD package is installed on the switch.	
Switch	The Release ID for the D-Channels is defined in LD 17.	
Switch	For a Uniform Dialing Plan (UDP), the following are configured: ■ ISDN Route Data Block with INAC = YES and PNI (Personal Number Identification) of the Target Node to which these routes are connected (LD 16) ■ AC1 or AC2 in the ESN Data Block (LD 86) ■ Route List Index (RLI) with no Digit Manipulation Index (DMI) for the customer (LD 86) ■ DMI table that deletes 3 for the customer (LD 86) ■ HLOC with DMI table defined in LD 86 (LD 90) ■ LOC with RLI for each node (LD 90) ■ HLOC for each customer (LD 15). If AC2 is used within the NACD Routing tables, then put LOC after AC2. If AC1 is used within the NACD Routing tables, then do not put LOC after AC2. ■ PNI that is a unique identifier (LD 15)	

<table>
<tr><th>Subsystem</th><th>Description</th><th>✔</th></tr>
<tr><td>Switch</td><td>For a Coordinated Dialing Plan (CDP), the following are configured:

ISDN Route Data Block with the PNI (Personal Number Identification) of the Target Node to which these routes are connected (LD 16)
Route List Index (RLI) with no Digit Manipulation Index (DMI) for the customer (LD 86)
Local Steering Code (LSC) that deletes 3 digits (if using 7-digit CDP, and DNs are 4 digits) (LD 87)
Distant Steering Code (DSC) with RLI for each node (LD 87)
LSC for each customer (LD 15)
HLOC for each customer (LD 15)
PNI that is a unique identifier (LD 15)

ATTENTION
If you are using Network Skill-Based Routing, you must define a Home Location Code (HLOC) for a CDP. Symposium Call Center Server uses this code to generate network call IDs for networked calls. Nortel suggests that you set the HLOC equal to the LSC or, if you are not using a 7-digit CDP and there is no LSC, set it equal to the NXX or NPA of the local calling area.</td><td></td></tr>
</table>

Subsystem	Description	✔
Switch	An ACD-DN is configured for NACD with the following (LD 23): ■ 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) For more information, refer to the NACD documentation.	
Switch	(for the Queue to NACD command only) An ACD-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 77.)	

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

Chapter 3

Subsystem connections configuration

In this chapter

Overview

Introduction

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

- the IP address on the switch

  For information about how to configure an IP address for the switch, refer to the switch documentation.
- the ELAN subnet between the switch, Symposium Call Center Server, and CallPilot
- (if you are using Symposium Voice Services on CallPilot or Meridian Mail to provide voice processing services) 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 64.

Configuring the ELAN subnet

Introduction

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

- Define the Embedded LAN (ELAN) and a Value Added Server (VAS).
- Enable the ELAN subnet link.
- Check the ELAN subnet link.
- Check the AML link.

These tasks associate the ELAN subnet link 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 subnet 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 subnet port number. Use a generic name because the ELAN subnet 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 subnet.
SECU	Yes	Turn on security for MLS applications.
CSQO	**Succession 1000 Release 3.0 or later** 25% of NCR maximum	Number of call registers linked to output queue. NCR = total number of call registers
	Succession 1000 Release 1.1, 2.0, X11 Release 25 2040 maximum	Switch patch MPLR12423 is required. This patch multiplies the values by 8. For example, a setting of 255 becomes 255 x 8 = 2040.
CSQI	**Succession 1000 Release 3.0 or later** 25% of NCR maximum	Number of call registers linked to output queue. NCR = total number of call registers
	Succession 1000 Release 1.1, 2.0, X11 Release 25 2040 maximum	Switch patch MPLR12423 is required. This patch multiplies the values by 8. For example, a setting of 255 becomes 255 x 8 = 2040.
REQ	END	Exit from overlay.

To enable the ELAN subnet link

1 At the switch administration terminal, load LD 48.

2 Type the command **enl elan 16**.

Checking the ELAN subnet with LD 48

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

To check the ELAN subnet 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 subnet is not active, check the ELAN subnet connection by pinging the switch IP address from the application.

a. To do so, open a DOS prompt.

b. Type **ping *nnn.nnn.nnn.nnn***, where *nnn.nnn.nnn.nnn* is the switch IP address.

c. 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**.

3 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 that provides these services 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.

  Note: CallPilot installs with ACCESS enabled.
- 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 65.

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 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 subnet.

From Meridian Mail

You must establish a physical connection between the 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 Nortel server subnet IP address of Symposium Call Center Server in the CallPilot Configuration Wizard. For detailed instructions, see “To update the Configuration Wizard” on page 128.

When you configure Symposium Call Center Server, you must specify the ELAN network interface IP address of the CallPilot server on Symposium Call Center Server. For detailed instructions, see “To configure the CallPilot connection” on page 160.

From Meridian Mail

On Symposium Call Center Server, the COM2 port is automatically configured for the ACCESS Link, 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 ACCESS link on Meridian Mail

1 In 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 “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 81.
- configure voice ports on the switch. For more information, see "Configuring voice ports on the switch" on page 85.
- 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 128. For more information about configuring ACCESS voice ports on Meridian Mail, see "Configuring Meridian Mail for ACCESS" on page 140.

Chapter 4

Switch subsystem configuration

In this chapter

Overview

Introduction

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

- CDNs (see “Configuring CDNs on the switch” on page 73)
- networked ACD-DNs (see “Configuring NACD-DNs to enable the Queue to NACD command” on page 77)
- IVR ACD-DNs (see “Configuring IVR ACD-DNs on the switch” on page 81)
- voice ports (see “Configuring voice ports on the switch” on page 85)
- agent ACD queues (see “Configuring agent and supervisor TNs on the switch” on page 89)
- agent phonesets (see “Configuring agent and supervisor TNs on the switch” on page 89)
- routes (see “Configuring routes on the switch” on page 95)

This chapter explains how to configure most of 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 or Meridian Mail, 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 customer support representative.
- your switch is running one of the following software releases:
    - X11 software Release 25 or greater
    - Succession 1000 Release 1.1 or Release 2.0
    - Succession 1000M Release 3.0
- all current Product Enhancement Packages (PEPs) are applied. For information on which PEPs to install on the switch, contact your Nortel 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, Meridian Link, and CCR.

Changing switch resources

You can use the CHG command in Overlay 11 to change any non-ACD properties of switch resources without deacquiring the resources, so long as you have the appropriate software version, as follows:

- Succession 1000M Release 3.0
- X11 Release 25.40 or Succession 1000 Release 2.02 or earlier with patch MPLR17921 installed.

CAUTION

Risk of system corruption

If you change acquired resources on a system without this patch, linked list corruption and status mismatches may occur.

Notes:

- Do not change ACD properties (position ID, queue ID, supervisor ID, and so on) on acquired resources.
- Nortel recommends that you take whatever steps are appropriate to insure that the resource being changed is not in use during the change. This not only reduces the risks of any unexpected operations, but also avoids having a user become disconnected when the service change is completed. The best way to avoid such problems is to disable a unit prior to service change.

In Succession 1000M Release 3.0 and supported X11 releases, you can also use the DES command in Overlay 85 to change acquired switch resources, as long as you have patch MPLR18387.

In X11 Release 25 and Succession 1000M Release 3.0 and earlier, you can change the DES of an M3900 set in Overlay 85 as long as you have patch MPLR16833 installed.

Moving and deleting switch resources

Before moving (MOV) or deleting (OUT) a switch resource, you must ensure that it is deacquired. You cannot MOV or OUT an acquired resource.

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	Add 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	xxx(xxxx)	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
SFNB 1  2  3  4  5  6  9  10  11  12  13  15  16  17  18  19
USFB 1  2  3  4  5  6  7  9  10  11  12  13  14  15
CALB 0  1  2  3  4  5  6  7  8  9  11
CNTL YES
VSID
HSID
CWTH 1
BYTH 0
OVTH 2047
STIO
TSFT 20
```

Changing CDNs on the switch

Introduction

If you need to change 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, 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.

Notes:

- 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.
- To ensure that Symposium Call Center Server calls are handled in a timely fashion, the switch alternates presentation of NACD and Symposium Call Center Server calls. After an agent finishes an NACD call, the switch provides a small time window to allow a Symposium Call Center Server call to be presented. If no Symposium Call Center Server call is available, the switch presents the next NACD call.

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 is 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	Add an NACD-DN.
TYPE	ACD	ACD data block
CUST	xx	Customer number
ACDN	xxxx	The ACD-DN to be added
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	Add
TYPE	NACD	Network ACD
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.

Night Service (NSVC)

With NACD, there are three Night Service treatments that can be defined:

- Night RAN Route with Night Tables

  Callers receive Night RAN, while the call is monitored for the timers defined for Targets nodes.
- Night Tables only

  No Night RAN is given, while the call is monitored for the timers defined for Targets.
- Night Tables with Delay Night RAN Treatment (DNRT)

  With Active entries in Night Tables (DNRT) on, callers will get Day treatment. The FRT must be 4 seconds greater than the timer value of the last entry in the Night Table for this to work. When all entries in the Night Table are inactive, then a Night RAN is returned to inform the caller that the network is closed.

> **Note:** When a caller accesses the queue in Night Service and used the NACD Night Table (the Night Table has open Targets), the callers hears first and second RAN.

Transition Mode:
When the Source ACD DN goes into Night Service via the NSVC key (dialing T [8]), calls already in queue still access the Day table, but all new calls access the Night Table. If there is no Night Table defined, traditional Night Treatment is given.

Night Mode:
The Source ACD DN goes into the Night mode using the NSVC key (dialing N [6]), when all agents log out. Then all calls access the Night Table, unless they have outstanding Call Requests from the Day Table.

- Source Node in Night Service
  - Transition mode:

    - new calls access the Night Table

    - existing calls access the Day Table
  - Night Mode:

    - new calls access the Night Table

    - existing calls that have pending Call Requests from the Day Table are honored, but there is no more searching of the Day Table
- Target goes into Night Service
  - Transition and Night mode:
    - new Call Requests are denied
    - existing Call Requests are canceled

When the Source ACD DN comes out of Night Service, only current Call Requests accessing the Night Table still apply. All new calls access the Day Table. Only calls without outstanding Call Requests can access the Day Table.

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, configure 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 85.

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 DNs 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.

Prompt	Response	Description
ALOG	YES	ACD agents are automatically logged on. **Note:** Only CallPilot and Meridian Mail TNs are automatically logged on; analog TNs, such as those used for third-party IVR systems, are not. To log on analog TNs, you must write an application on the IVR system to log on the ports.
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. **Note:** Only Meridian Mail and CallPilot TNs are automatically logged on; analog TNs, such as those used for third-party IVR systems, are not. To log on analog TNs, you must write an application on the IVR system to log on the ports.
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 DNs 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 85.

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 or CallPilot. (For third-party IVR systems, the agent TNs are analog 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 123.

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	Add a voice port.
TYPE	2008	

Prompt	Response	Description
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	FLXA (units 16–31) VCE WTA CTD MMA	Flexible voice/data allowed. Voice Terminal Warning Tone Allowed Conditionally Toll Denied Multimedia Agent **Notes:** ■ 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. ■ CTD is optional, but prevents outbound long distance calls from a voice port.
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.

Prompt	Response	Description
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	Add 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.

Prompt	Response	Description
KEY	7 AO3 (letter ‘O’)	Define key 7 as a Conference key.
KEY	9 RLS	(For SL1 phonesets) 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 phonesets (TNs). 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 ACD queues

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 reason 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 a new phoneset.
TYPE	aaa	Enter phoneset 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	ACD 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	SCR 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 10, “Agent phonesets,” 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 a new phoneset.
TYPE	500	Enter phoneset type as appropriate.
TN	l s c u	Terminal number
CUST	0–99	Customer number
DN	xxxx	DN for the phoneset
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 phonesets.

Prompt	Response	Description
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.

Notes:

- Only RAN, MUS, FGDT, TIE, DID, COT, FEX, and WATS route types are supported by Symposium Call Center Server.
- If you are not using Symposium Voice Services on CallPilot or Meridian Mail, you should configure a MIRAN card and RAN routes to supply messages to callers waiting in queue.

Configuring trunk routes

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

Note: If you want to generate reports on trunk routes, you must also configure the trunk routes on Symposium Call Center Server.

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 21 (REQ=prt, TYPE=RDB).

- At the switch, you have ensured that physical trunks are defined for the routes.

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	Add 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	Add a route.
TYPE	RDB	Route data block
CUST	nn	Customer number
ROUTE	0–511	Route number

Prompt	Response	Description
DES	x...x	Designator field for trunk. Enter a description.
TKTP	MUS	MUSIC trunk data block requires Music package 44.
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 continue to use Multiple Queue Assignments (MQA) with Meridian MAX and you also have Symposium Call Center Server, you must ensure that your MQA agent ACD-DNs are kept separate from Symposium Call Center Server agent queues.

To allow MQA and Symposium Call Center Server to coexist

1. Create new ACD-DNs for the Symposium Call Center Server agents.
2. Move the Symposium Call Center Server agents into the new ACD-DNs.

   Note: You can use the Meridian MAX configuration control feature to move the agents. Perform this step during a maintenance window when all agents are logged off.
3. Change RPRT to NO in LD 23 for the source ACD-DN.

   Note: This ensures that MAX does not report on this queue or the agent positions. Since MAX does not understand skillsets, and it does not know which agents are staffing the agent phonesets, reporting information would be inaccurate.
4. Ensure that the Symposium Call Center Server CDN has the appropriate DFDN defined so that calls are handled properly in default mode.

Notes:

- Symposium Call Center Server can acquire MQA agents. When it does, only one queue is assigned to the phoneset—the first queue the agent logs on to. If Symposium Call Center Server goes down, then calls will be presented to agents in this queue. If agents log on to multiple queues while Symposium Call Center Server is down, then when Symposium Call Center Server comes back up and acquires the agent phonesets, the phonesets revert to one queue.
- Make sure that all Change Orders moving Agent positions from one queue to another are removed from MAX. If you want to continue to use MAX to move agents to a different queue and you do not have Succession 1000M Release 3.0 or the patch MPLR17921, then deacquire the agent phoneset in

Symposium Call Center Server, make the change in MAX, ensure that the switch has performed the change, and then acquire the phoneset in Symposium Call Center Server. If you have Succession 1000M Release 3.0, then you can make the change without deacquiring the agent position.

Switch maintenance

Symposium Call Center Server can only process calls if the switch is operational. Before performing upgrades or maintenance on the switch, you must do the following:

- Shut down CallPilot.
- Shut down Symposium Call Center Server.

Chapter 5

Meridian Link Services configuration

In this chapter

Overview

Introduction

This chapter provides the references and procedures required to set up Meridian Link Services (MLS) for operation with Symposium Call Center Server. MLS is a communications facility that provides an interface between a host application and the 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.

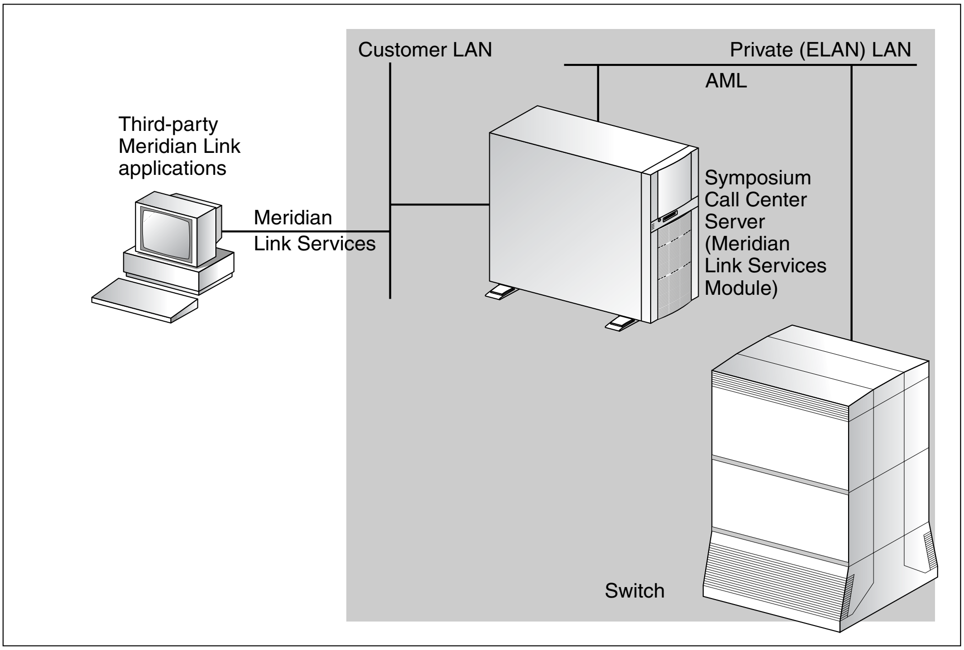
If you are using CallPilot for your Symposium Voice Services, CallPilot communicates with the switch using MLS, so you must configure the switch for Meridian Link. Once you configure MLS on the switch, your connection to the host application is through the Nortel server subnet and ELAN subnet connection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

Notes:

- If you want to use CTI functionality for your Symposium Call Center Server agents, you must use MLS. You cannot use legacy Meridian Link.
- In an environment with multiple servers on the same switch, if you are using TAPI to provide CTI functionality, each server must have its own TAPI server.

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



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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 Symposium Call Center Server. (MLS is installed as part of the Symposium Call Center Server software.) Refer to the *Installation and Maintenance Guide* for installation instructions.

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:

- Allow CTI operations on the ELAN subnet (LD 17).
- Configure phonesets for CTI (LD 11 and LD 10).
- Configure CDNs for Host-Enhanced Routing (LD 23).

Once you configure the switch, you must connect the host application to Nortel server subnet.

Allowing CTI operations on the ELAN subnet

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	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 subnet 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 Nortel IVR or third-party IVR applications.
- CallPilot voice ports are configured as digital voice ports. Nortel 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 89.

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 89.

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.

Note: Key layout varies depending on customer requirements. The key configuration in this table is provided as an example only.

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

Prompt	Response	Description
KEY 2	AO6	6-party conference
KEY 9	RLS	Release if type = SL1
		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 73. 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 73 for more information.

Connecting the host application

The host application is connected on the Nortel server subnet. The host application must point at the Nortel server subnet IP address. Refer to the *Symposium Meridian Link Services Interface Specification* for more information.

Chapter 6

Voice processing subsystem configuration

In this chapter

Overview

Introduction

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

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

Section A: Voice processing

In this section

Overview of voice processing

Introduction

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

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

Different ways to do voice processing

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

- **Script 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 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 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 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 Give IVR.

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, voice 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 voice 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 voice port:

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

The following commands connect one call per voice 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

Notes:

- 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 5.0, only the Give IVR command supports both interruptible (option) and non-interruptible (default) operation. Both the Controlled Broadcast and Voice Sessions commands support interruptible (default) operation only.

Interruptible operation

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

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

Non-interruptible operation

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

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

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

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

Resource acquisition

Voice processing resource acquisition summary

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

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

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, Meridian Link Services, Meridian (Integrated) IVR, CCR, and Symposium Call Center Server

While some of these services and applications can share voice ports, the Symposium Call Center Server requires a dedicated set of voice 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 124).

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 (Give IVR or ACCESS).

Symposium Call Center Server Give IVR voice ports

Give IVR voice 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 Give IVR voice ports.

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

Symposium Call Center Server ACCESS voice ports

ACCESS voice 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 voice ports, giving each voice port a unique ACCESS class (channel number). The TNs and Class IDs for these voice 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 168) to limit the number of voice 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 voice 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

Introduction

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

- Define the ELAN subnet 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.

ATTENTION

The channels that provide ACCESS and IVR services to Symposium Call Center Server must be configured as dedicated Voice channels. Do not use Fax or Speech Recognition channels.

ACCESS voice ports

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

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

To use ACCESS voice 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 128.
- Service Directory Numbers (SDNs)

For information on creating SDNs, see “Updating the SDN table” on page 130.

- voice segments

  For information on creating voice segments (voice items), see the *CallPilot Application Builder Guide*.

Non-ACCESS ports

CallPilot controls these voice ports. The treatment that the caller receives is determined by the treatment DN or IVR ACD-DN. You use these voice ports for Give IVR.

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

- voice ports (virtual agent TNs)

  For information on configuring Give IVR voice ports in CallPilot, see “Updating the CallPilot configuration” on page 128.

- Service Directory Numbers (SDNs)

  For information on creating SDNs, see “Updating the SDN table” on page 130.

- announcements, menus, and so on

  For an example procedure, see “To create an announcement” on page 181. For more information on creating announcements and menus, see the *CallPilot Application Builder Guide*.

CallPilot and multiple servers on the same switch

If you are using CallPilot to provide front-end IVR, the same CallPilot server can support all three Symposium Call Center Server systems.

If you are using Symposium Voice Services on CallPilot—that is, if CallPilot is providing Give IVR or ACCESS voice services (Open/Close Voice Session, Collect Digits, and Give Controlled Broadcast)—CallPilot can serve only one Symposium Call Center Server system. Therefore, each Symposium Call Center Server system must be connected to a separate CallPilot.

Updating the CallPilot configuration

Introduction

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

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

To update the Configuration Wizard

1 On the CallPilot server, start the Configuration Wizard.

2 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.)

3 In the **Symposium Call Center Server CLAN IP Address** field, enter the CLAN address of Symposium Call Center Server.

4 On the left side of the page, click the link for the set of channels you want to configure.

Result: 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.

Result: 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, 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**.

Updating the SDN table

Introduction

You must define Service DNs for use by Symposium Voice Services. Define the Symposium Call Center Server IVR and ACCESS ACD-DNs, as well as any treatment DNs.

To define IVR and ACCESS ACD-DNs

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 In the **Media Type** box, select **Voice**.

7 Click **Save**.

To define treatment DNs

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 treatment DN.

5 In the **Application Name** box, enter the Application Builder application name.

6 In the **Media Type** box, select **Voice**.

7 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

Introduction

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

ACCESS voice ports

Symposium Call Center Server controls these voice ports over the ACCESS link (the serial link that connects Meridian Mail and Symposium Call Center Server). You use these voice ports for Give Controlled Broadcast and Voice Sessions commands. Meridian Mail supports 96 voice ports.

To use ACCESS voice 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 136).
- Meridian Mail mailboxes

  For information on creating Meridian Mail mailboxes, see “Creating Meridian Mail mailboxes” on page 143.
- voice segments

  For information on creating voice segments, see “Creating voice segments” on page 144.

Give IVR voice 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 136.

- treatment DNs (VSDNs)

  For information on configuring treatment DNs, see “Configuring VSDN entries (treatment DNs)” on page 149.

- Voice Menu service or Announcement Service

  For information on configuring treatment DNs, see “Creating announcements and voice menus” on page 147.

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.

Meridian Mail and multiple servers on a switch

If you are using Meridian Mail to provide front-end IVR, the same Meridian Mail can support all three Symposium Call Center Server systems.

If you are using Symposium Voice Services on Meridian Mail to provide IVR services (that is, with the Give IVR command), the same Meridian Mail can support all three Symposium Call Center Server systems. However, the following restrictions apply:

- You must allocate the Meridian Mail IVR ports between three IVR queues, and dedicate a queue to each server.
- All of the servers must belong to the same customer group. (Therefore, you cannot network the servers together.)

If you are using Symposium Voice Services on Meridian Mail to provide ACCESS voice services (Open/Close Voice Session, Collect Digits, and Give Controlled Broadcast), Meridian Mail can serve only one Symposium Call Center Server system.

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 voice port types (that is, the number of basic, full-voice, and multimedia ports)
- move agents from one queue to another, to dedicate a voice 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 voice 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 DNs 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 64)
- 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 81), and Symposium Call Center Server must acquire it (see "Configuring IVR ACD-DNs on the server" on page 165).

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, navigate 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.

Note: Do not enter a Revert DN.

6 Type an optional comment in the **Comment** box, such as *Symposium ACCESS DN*.

7 Click **Save**.

Result: You return to the VSDN table. There is 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.

Rules for configuring voice ports

The following rules for voice processing commands should be used when configuring voice ports in Meridian Mail.

Voice processing command	Type of port required
Give IVR	Full service
Give Controlled Broadcast Announcement	Basic service with ACCESS enabled

Voice processing command	Type of port required
Open-End Voice Session Play Prompt Collect Digits	Basic service with ACCESS enabled
Play Expected Wait Time (or other intrinsics)	Basic service with ACCESS enabled

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.

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

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

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

ACD DN: The ACCESS IVR ACD-DN.

SCN DN: This must match the switch configuration.

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

Note: You cannot use Basic ports for Voice Messaging or Express Messaging.

Service: ACC for ACCESS. When you are prompted for an ACC Class, enter a unique number to identify the channel. The ACC Class is the channel number that you specify when creating an ACCESS voice port on Symposium Call Center Server. See “Configuring voice ports on the server” on page 167.

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 Press **Save**.

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

6 Re-enable any DSP ports that you put out of service before configuring the voice ports.

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 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 recommends that the administrator configure all passwords to never expire or, alternatively, 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 allow you to use the Voice Prompt Editor, the Channel Allocation Table on Meridian Mail must include at least one channel that has an outbound service value of “all.”

ATTENTION

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

ATTENTION

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

Configuring voice prompts

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 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. 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 DNs) 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 provides the following recommendations for creation of the announcement service:

- Enable Silent Disconnect (so that the caller does not hear "Good-bye" at the end of an announcement).
- Set Number of times to play announcement to 1.
- Do not set an ACCESS password.
- The access to the announcement is through the Give IVR command (see the next section).

Voice menu services

If you want to give the customer a simple voice menu (“Press 1 for Sales and press 2 for Support”), then Nortel 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 provides the following recommendations for creation of the announcement service:

- Enable 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.
- Enable Initial No Response DS (disconnect).
- Enable 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 has the following recommendations:

- Enable 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 234.
- Enable Initial No Response DS (disconnect) or RP (Repeat Menu Choices) as appropriate.
- Enable Delayed Response DS (disconnect) or RP (Repeat Menu Choices) as appropriate.

Notes:

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

Configuring VSDN entries (treatment DNs)

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

These entries are known as treatment DNs and are used in Symposium Call Center Server scripts to specify which treatment the 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’s 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 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

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, Nortel IVR (an integrated 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's IVR CTI application uses Meridian Link to deliver the treatment DN to Nortel 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.

Note: You can configure IVR ACD agents as agents in a skillset if you require Symposium Call Center Server real-time and historical reporting and call routing control. For more information, see "Configuring IVR ACD-DNs on the server" on page 165, and "Configuring agent phonesets on the server" on page 166.

Configuration of the external IVR

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

Chapter 7

Symposium Call Center Server subsystem configuration

In this chapter

Overview

Introduction

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

- voice connection
- 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*.

Note: You can also use the Symposium Web Client to configure these resources.

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

Configuring the voice connection

Introduction

If you are using Symposium Voice Services on CallPilot or Symposium Voice Services on Meridian Mail, you must configure the voice connection. 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.

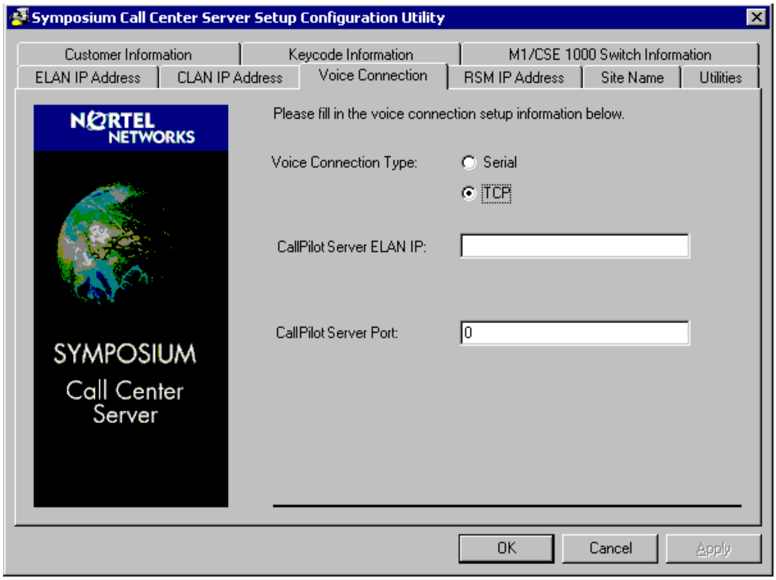
3. If any services are still running, use the Services control panel to shut them down manually (from the Windows **Start** menu, choose **Control Panel**, and double-click **Services**). Then click **Recheck** to update the status log.
4. Click **Accept** to exit the utility.

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 network interface 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**.

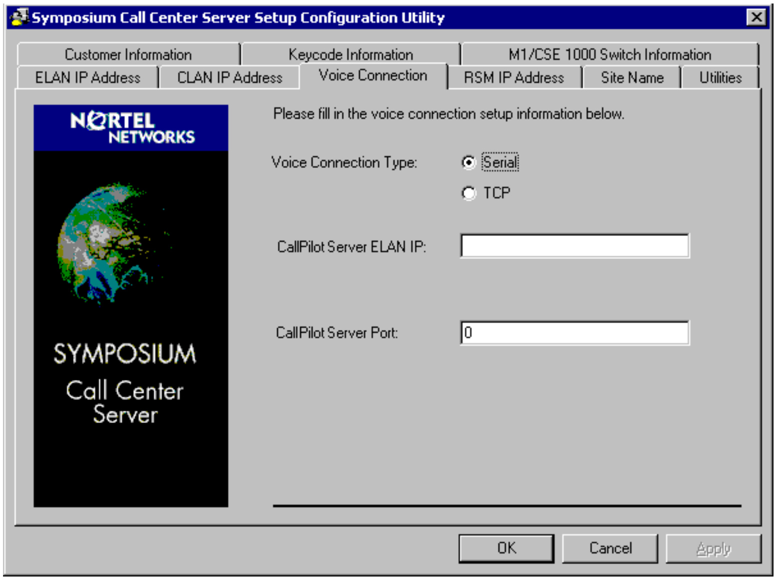
Result: The server restarts.

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.

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. In 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. In 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 IVR 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 want 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 1000, the TN format is "loop-shelf-card-unit" (for example, 24-0-4-5).

To configure and acquire a phoneset

1. In the **SMI** window, open **Switch Administration > Phonesets**.

   Result: The Phonesets window appears.

2. Choose **File > New**.

   Result: The Phonesets Properties page appears.

3. Enter the terminal name and the 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 want 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 166.

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 Type 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 (This is the ACCESS ACD-DN.)
- (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 voice ports that can be used for Controlled Broadcast at any given time. Up to 50 calls can be attached simultaneously to a single voice port on a broadcast announcement. The fifty-first call for an announcement is connected to a new voice port, as long as the maximum number of broadcast voice 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 voice 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 voice ports in this IVR ACD-DN. This allows new calls to be queued to the broadcast voice ports.

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

You can roughly calculate the number of Broadcast voice 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 voice ports used by broadcast and to maximize the number of voice ports used by Voice Sessions. (Since Voice Sessions needs a one-call-to-one-port arrangement, its voice 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 voice 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 voice port (that is, less efficient use of voice 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 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 may be in use by voice processing, and this can 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.

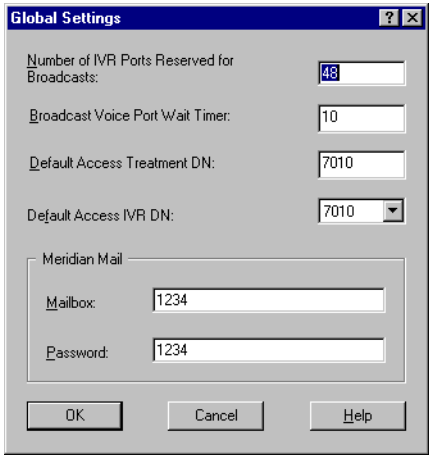
To configure the global settings for IVR ACD-DNs

1 In the **SMI** window, open **Switch Administration > IVR ACD-DNs**.

Result: The IVR ACD-DNs window appears.

2 Choose **File > Global Settings**.

Result: The Global Settings dialog box appears.



3 Enter the number of IVR ports reserved for broadcast.

4 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.

Chapter 8

Testing integration with Symposium Voice Services on CallPilot

In this chapter

Overview

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 basic voice services (Give IVR). Script CP_Test2 tests advanced voice services (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 voice services

On the switch, you must configure the following:

- an ACD-DN with the following attributes:
  - IVR = YES
  - ALOG = YES
- agents in this ACD-DN with the following attributes:
  - CLS = MMA and FLXA
  - AST on keys 0 and 1

In CallPilot you must configure the following:

- a Service Directory Number (SDN) entry for the IVR ACD-DN. The Application Name must be Symposium Voice Services, and Media Type must be Voice.
- an SDN entry for the Application Builder application. The Application Name must equal the application name, as defined in Application Builder, and Media Type must be Voice.

ACCESS voice services

On the switch, you must configure the following:

- an ACD-DN with the following attributes:
  - IVR = YES
  - ALOG = YES
- agents in this ACD-DN with the following attributes:
  - CLS = MMA and FLXA
  - AST on keys 0 and 1

In CallPilot you must configure the following

- a Service Directory Number (SDN) entry for the IVR ACD-DN. The Application Name must be Symposium Voice Services, and Media Type must be Voice.
- an SDN entry for the Application Builder application. The Application Name must equal the application name, as defined in Application Builder, and Media Type must be Voice.

Notes:

- Prompts can be “filed” as in Meridian Mail.
- CallPilot does not require unique SDN entries for voice prompts. Voice prompts can be defined under one or more Application Builder application.

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 (enforced) **Note:** 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.

Testing basic voice services (Give IVR)

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

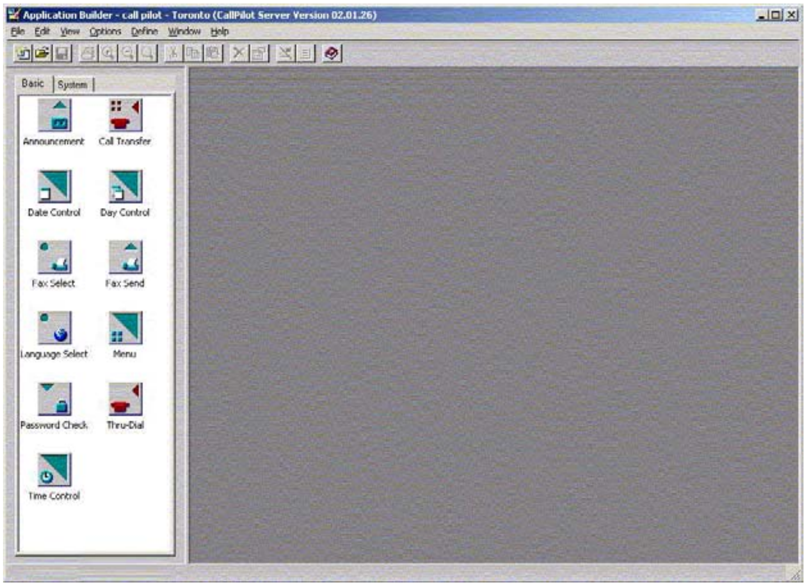
- Create an announcement application using CallPilot Application Builder, and record a prompt.
- Create an SDN numerical entry point to the application name.
- Create the script variables in Symposium Call Center Server.
- Write a test script, or import the CP_Test1 script from \Program Files\Nortel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

When these steps are completed, you can test basic voice services.

To create an announcement

1 In CallPilot, launch Application Builder.

Result: Application Builder appears.



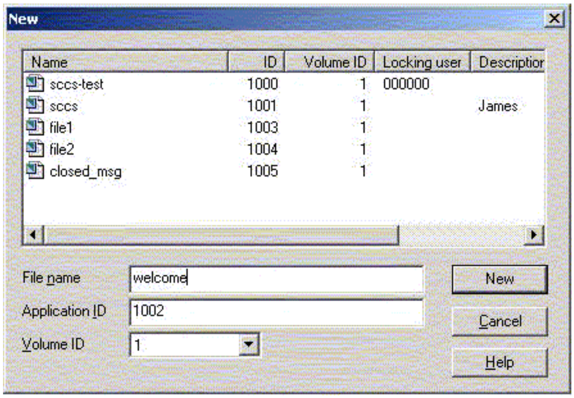
2 Select **File > New**.

Result: The New dialog box 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.

Result: 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.

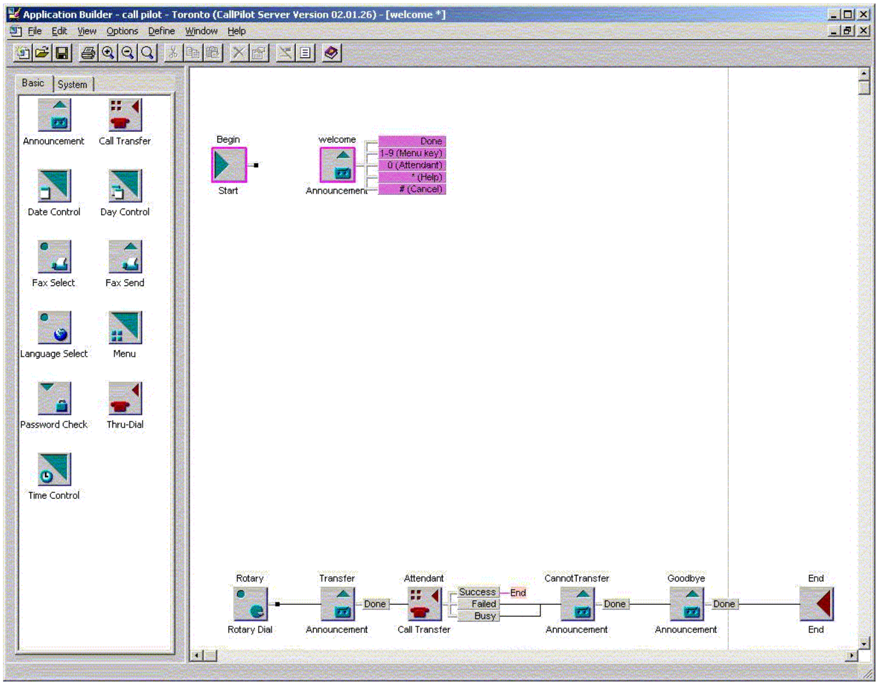
Result: The Add Announcement Block dialog box appears.

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**.

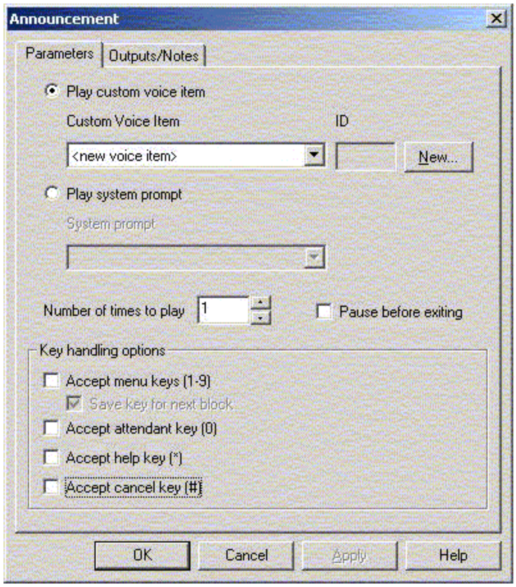
7 Click **OK**.

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



8 Double-click the Announcement block.

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



9 Select **Play custom voice item**, and then click **New**.

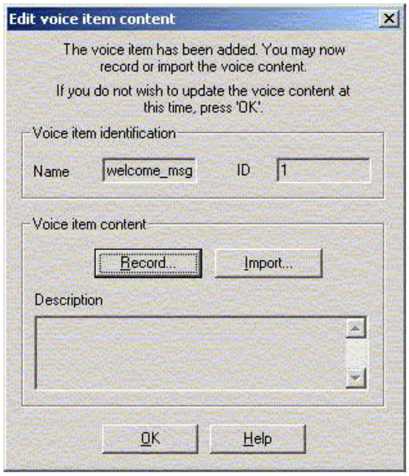
Result: The Add voice item dialog box appears.

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 dialog box 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 dialog box 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 dialog box 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 dialog box 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**.

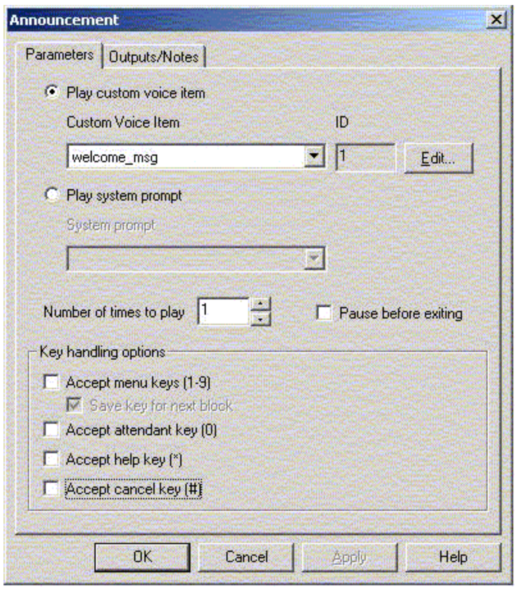
19 Click **Play** to review the recorded voice item. When you are satisfied, hang up the phoneset.

20 Click **Save**.

21 Click **Close**.

22 In the Announcement property sheet (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 property sheet 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.

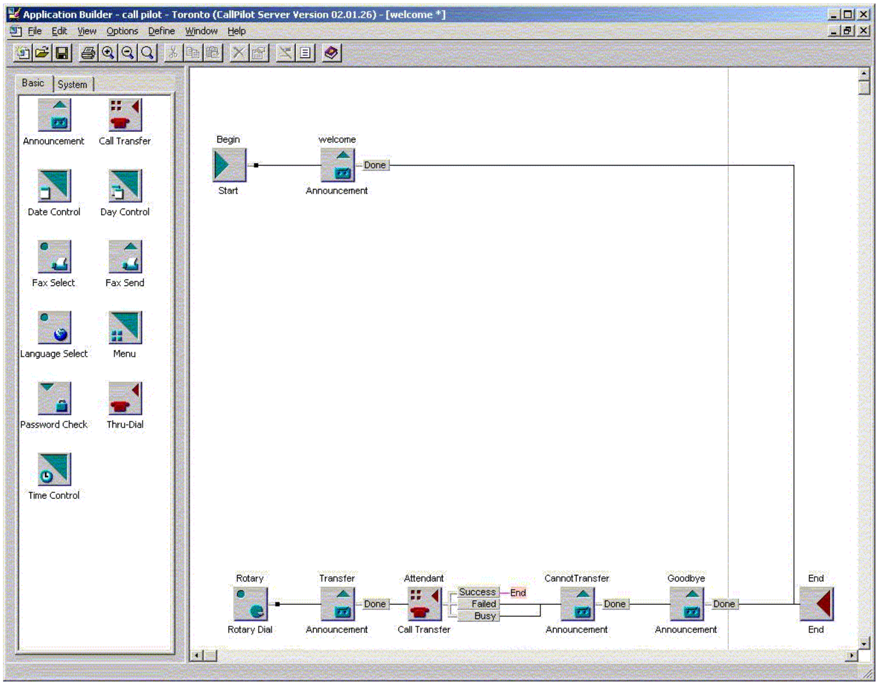
Result: A connection line appears from the Begin block to the Announcement block.

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

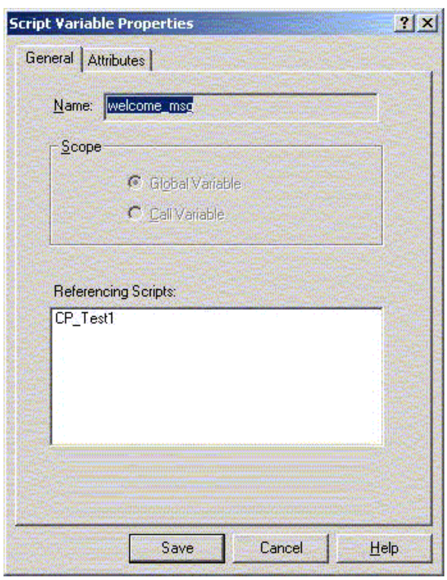
1 On the Symposium Call Center Server client, in the **SMI** window, select **Script Variables**.

Result: The Script Variables window appears.

2 Choose **File > New**.

Result: The Script Variable Properties property sheet appears.

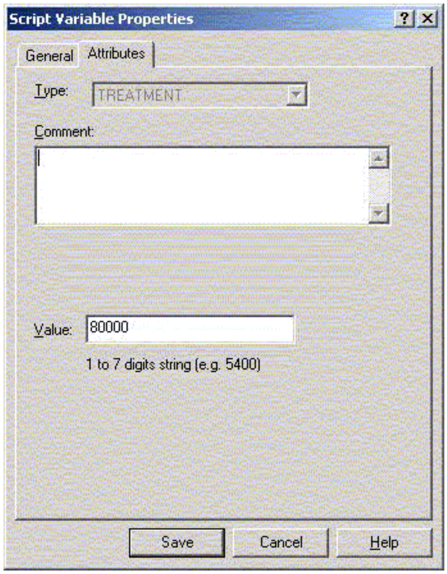
3 On the **General** page, in the **Name** box, type the variable name. (For this example, use the announcement name **welcome_msg**.)



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 200.

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.

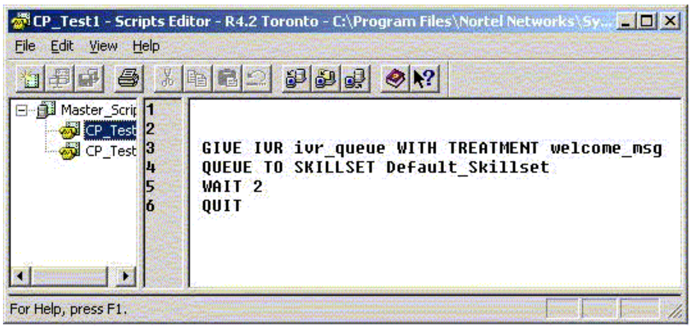
2 Choose **File > New**.

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 199.)
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.

Testing advanced voice services (ACCESS)

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

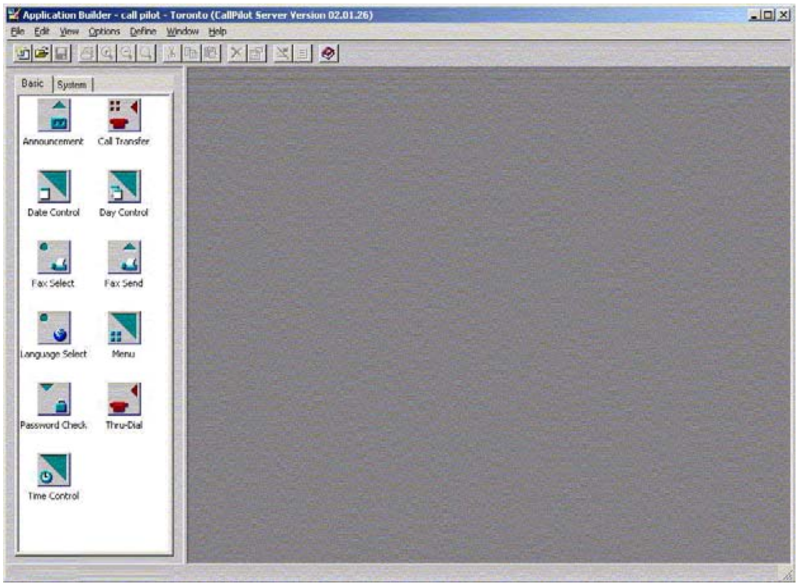
- Create an application using CallPilot Application Builder.
- Create new applications (folders) to hold the voice prompts used for Open Voice Session and Controlled Broadcast Announcement.
- Record the prompts.
- Create the script variables in Symposium Call Center Server.
- Write a test script, or 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 the test script.

When these steps are completed, you can test advanced voice services.

To define ACCESS voice prompts

1 In CallPilot, launch Application Builder.

Result: Application Builder appears.



2 Select **File > New**.

Result: The New dialog box 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**.

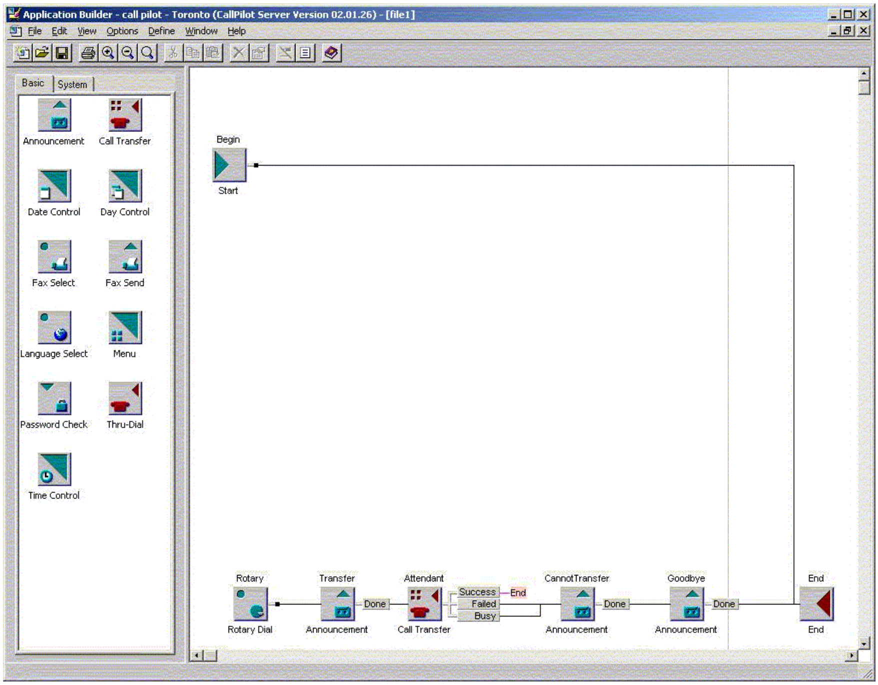
Result: The application is created.

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 dialog box appears.

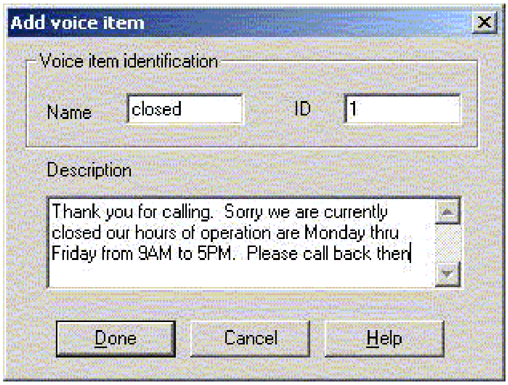
9 To add a voice item, click **Add**.

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 dialog box 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 185.

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 193. 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**.

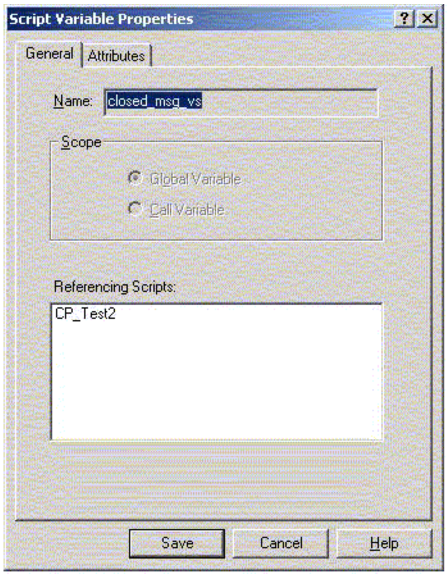
1 On the Symposium Call Center Server client, in the **SMI** window, select **Script Variables**.

Result: The Script Variables window appears.

2 Choose **File > New**.

Result: The Script Variable Properties property sheet appears.

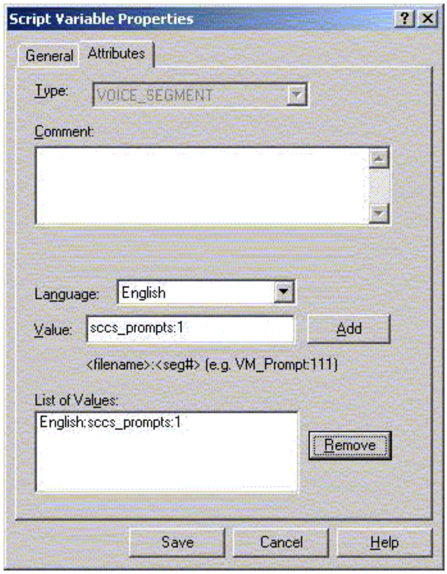
3 On the **General** page, in the **Name** box, type the variable name. (Use the name of the voice item. For this example, type **closed_msg_vs**.)



4 In the **Scope** box, select **Global Variable**.

5 Click the **Attributes** tab.

6 In the **Type** drop-down list box, 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 box, 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 201.

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.

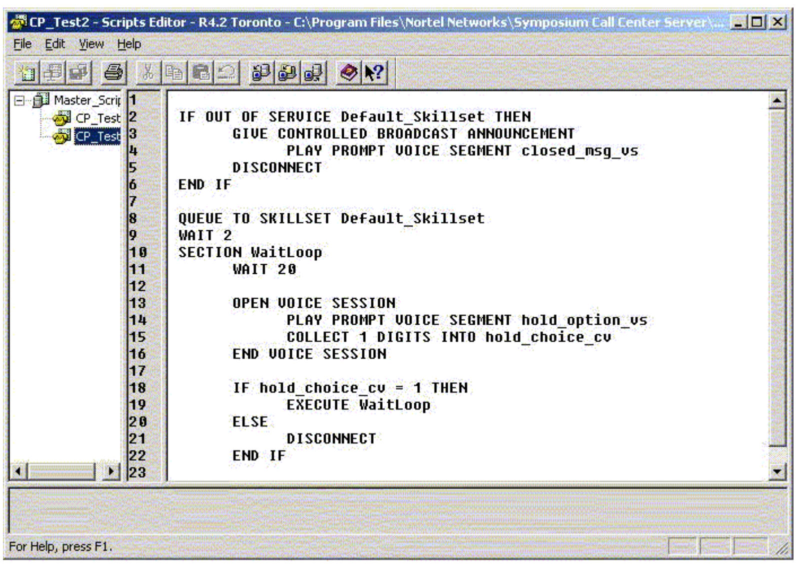
2. Choose **File** > **New**.

   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 Center Server\Client\en\script\samples.

   Result: The test script appears.



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 199.)
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.

Test scripts

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 use a global variable. The value of the variable must be **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. This treatment number must correspond to a CallPilot Service DN (SDN) entry that you defined on CallPilot. The CallPilot SDN links to the Application Builder application that you created. You may define the SDN 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

```
*/

/* Tests the GIVE IVR functionality, use CP_Test2 to test
advanced voice services */

IF CDN = test_cdn THEN
   EXECUTE SCRIPT CP_Test1

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)

Variable name	Class	Type	Value	Comment
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)

```
Variable name       Class   Type      Value   Comment

closed_message_vs   Global  Voice             Application-
                            segment           name:voice
                                              item-ID needs
                                              to be created
                                              and recorded
                                              in CallPilot

hold_option_vs      Global  Voice             Application-
                            segment           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 9

Troubleshooting

In this chapter

Overview

Introduction

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

For detailed information on how to configure the elements in each of the subsystems used with the 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

Make sure the server is up

- **On the server**: In the SMonW utility, all components must have the status UP.
- **On the switch:** ELAN subnet connection to the switch is functioning (see the following section for detailed instructions).
- **On the client**: You can successfully log on to the server.

Check the ELAN subnet connection between the server and switch

On the switch, in Overlay 48, use the following command:

stat ELAN

The status for the ELAN subnet connected to the server must be ACTIVE EMPTY and APPL ACTIVE. (If there are multiple ELAN subnets, match the IP addresses).

Example

```
>ld 48
LNK000
.stat elan
SERVER TASK:  ENABLED
ELAN #: 16
   APPL_IP_ID: 47.166.111.14
   LYR7: ACTIVE  EMPTY   APPL ACTIVE
ELAN #: 17
   APPL_IP_ID: 47.166.111.13
   LYR7: ACTIVE  EMPTY   APPL ACTIVE
```

Check the CSL connection between the switch and Meridian Mail

On the switch, in Overlay 48, use the following command:

stat AML

The status for the link number connected to Meridian Mail must be ACTIVE EMPTY.

Example

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

Check the ACCESS Link between the server and Meridian Mail

In Meridian Mail, log on as tools (using the password configured for your system). Then navigate through the following menus:

- 13Other
- 2ACCESS Diagnostics

LinkStatus for the link to the server must be Synchronized.

Example

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

Check the ACCESS Link between the 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.

Check the switch loop, shelves, and cards

On the switch, in Overlay 32, use the following command:

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.

The status for real agents must be LOG IN or LOG OUT, depending on the state of the agent.

The status for Meridian Mail or CallPilot voice ports must always be LOG IN. If it is 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 )
```

Make sure Meridian Mail ports are enabled

In Meridian Mail, navigate through the following menus:

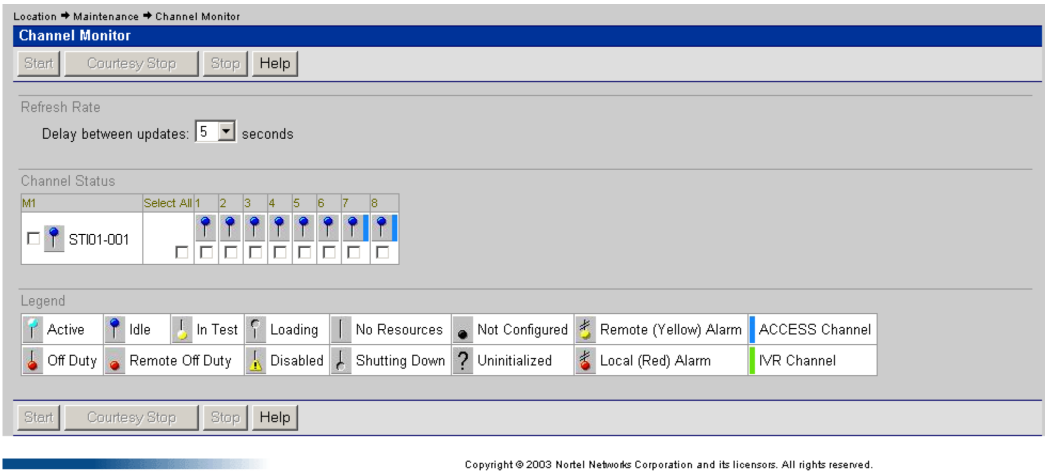
- 5System Status and Maintenance
- 3DSP Port Status

The status for the ports must be one of the following:

Idle	if they are acquired by the TN only from the server
Active	■ if they are acquired by the TN and the channel by the server ■ if they are busy on a call ■ if they are acquired by another ACCESS application

Make sure CallPilot ports are enabled

On the CallPilot client, in the CallPilot Manager, select Channel Monitor link. Channels should be in Idle state.



Note: On this screen, ACCESS channels are blue, and Give IVR channels are green.

Make sure the CDN is acquired

- **On the client:** Display the CDNs window. The CDN status must be Acquired.
- **On the switch:** In Overlay 23, use the following commands:
  - **REQ PRT**
  - **TYPE CDN**

  The following values should appear on the printout:
  - AACQ = YES
  - ASID = ELAN connected to Symposium Call Center Server
  - CNTL = 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
```

Make sure the correct script is activated

On the client, ensure that the script is in Active state.

Make sure that the IVR ACD-DN is acquired

- **On the client:** Display the IVR ACD-DNs window. The IVR ACD-DN status must be Acquired.
- **On the switch:** In Overlay 23, use the following commands:
  - **REQ PRT**

- **TYPE ACD**

The following values should appear on the printout:

- AACQ = YES
- ASID = ELAN connected to Symposium Call Center Server
- IVR = YES
- TRDN = default treatment DN, if any

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
```

Make sure Give IVR voice ports are acquired by TN

- **On the client:** Display the Voice Ports window. The voice port status must be Acquired Login.
- **In CallPilot:** On the CallPilot client, in the CallPilot Manager, select Channel Monitor link. Make sure that the Give IVR (green) channels are in Idle state.
- **In Meridian Mail:** Navigate the following menus:
  - 5System Status and Maintenance
  - 3DSP Port Status

  Make sure that the status for the ports is Idle.
- **On the switch:** In Overlay 20, use the following commands:
  - **REQ TNB**
  - for Meridian Mail with all switches except Meridian 1 Option 11:

    TYPE SL1
  - for CallPilot and for Meridian Mail with Meridian 1 Option 11:

    TYPE 2008

  The following values should appear on the printout:
  - ACQ AS = TN
  - ASID = 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
```

Make sure ACCESS voice ports are acquired by TN and CallPilot class ID or channel

- **On the client:** Display the Voice Ports window. The Channel column must contain a unique number, and the voice port status must be Acquired Login.
- **In CallPilot:** On the CallPilot client, in the CallPilot Manager, select Channel Monitor link. Make sure that the ACCESS (blue) channels are in Idle state.
- **In Meridian Mail:** Navigate through the following menus:
  - 5System Status and Maintenance
  - 3DSP Port Status

  Make sure the voice ports are in Active state.
- **On the switch:** In Overlay 20, use the following commands:
  - **REQ TNB**
  - for Meridian Mail with all switches except Meridian 1 Option 11:

    TYPE SL1
  - for CallPilot and for Meridian Mail with Meridian 1 Option 11:

    TYPE 2008

  The following values should appear on the printout:
  - ACQ AS = TN
  - ASID = ELAN connected to Symposium Call Center Server

Example

See the examples in “Make sure Give IVR voice ports are acquired by TN” on page 214.

Verify that the system default Treatment DN is configured correctly

On the client, ensure that the default treatment DN specified in the IVR ACD-DN Global Settings window is set appropriately.

Make sure treatment DNs are defined in the CallPilot SDN table

In CallPilot, check the SDN table. (From the Configuration Manager, choose System > Service Directory Number.) The table must contain an entry for each treatment DN, where the Application Name is the name of the application created in Application Builder.

Make sure treatment DNs are defined in the Meridian Mail VSDN table

Note: See page 149 for a discussion on how treatment DNs are used with Meridian Mail.

1. In Meridian Mail, navigate through the following menus:
   - 3Voice Administration
   - 4Voice Services Administration
   - 1Voice Service-DN Table

   Verify that the treatment DN is defined here as the appropriate service. The following example 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

2. Navigate through the following menus:
   - 3 Voice Administration
   - 4 Voice Services Administration
   - 3 Announcement Definitions
3. Verify that the service number displayed in the Voice Service DN table is defined as an announcement.

   Example:

```
ID          Title
2020        hello
```

```
        2021        default
```

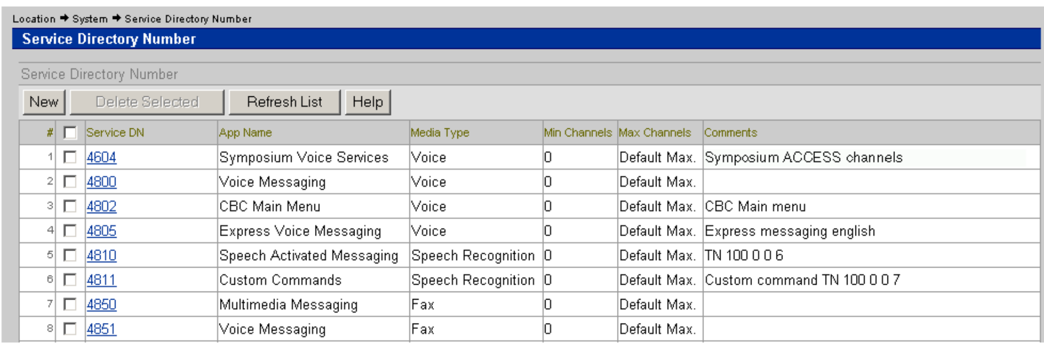
Make sure IVR ACD-DNs match on the switch, Symposium Call Center Server, and voice processing system

The ACD-DNs must match in the following locations:

- Channel Information page in CallPilot Manager
- Meridian Mail Channel Allocation Table
- switch DN
- Symposium Call Center Server script
- IVR ACD-DNs window in Symposium Call Center Server

To verify that they match, follow these steps:

- **In CallPilot:** In CallPilot Manager, choose System > Service Directory Number. Check the value specified in the Service DN field.



#		Service DN	App Name	Media Type	Min Channels	Max Channels	Comments
1		4604	Symposium Voice Services	Voice	0	Default Max.	Symposium ACCESS channels
2		4800	Voice Messaging	Voice	0	Default Max.	
3		4802	CBC Main Menu	Voice	0	Default Max.	CBC Main menu
4		4805	Express Voice Messaging	Voice	0	Default Max.	Express messaging english
5		4810	Speech Activated Messaging	Speech Recognition	0	Default Max.	TN 100 0 0 6
6		4811	Custom Commands	Speech Recognition	0	Default Max.	Custom command TN 100 0 0 7
7		4850	Multimedia Messaging	Fax	0	Default Max.	
8		4851	Voice Messaging	Fax	0	Default Max.	

- **In Meridian Mail:** Navigate through the following 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	4604	4500	Voice	FULL Basic ACC Class: 41

- **On the switch:** In Overlay 22, use the following command:
  - **REQ DNB**

  Example:

```
ld 22
PT2000
REQ  prt
TYPE dnb
CUST 0
DN   4604
DATE
PAGE
DES

DN   4604
TYPE MCDN
MCID 4500  TN 004 0 03 00
```

- **On the client:**
  - The script command should specify the DN defined in the CallPilot SDN table or the Meridian Mail Channel Allocation table:

    ```
    Give Controlled Broadcast 4604
    ```

  - In the IVR ACD-DNs window, the IVR ACD-DN number matches the ACD-DN defined on the switch and in the CallPilot SDN table or the Meridian Mail Channel Allocation table. The status for the IVR ACD-DN must be Acquired.

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

Make sure voice port TNs match on the switch, Symposium Call Center Server, and CallPilot or Meridian Mail

The configurations of the TNs belonging to the ACD-DN (see previous section) must match in the following locations:

- Channel Information page in the CallPilot Manager
- Meridian Mail Channel Allocation Table
- switch DN
- IVR ACD-DN acquired by Symposium Call Center Server

To verify that they match, follow these steps:

- **In CallPilot:** Choose Configuration Wizard > CallPilot Manager > Channel Information page. Check the value in the TN column.
  Example:

STI Board 200i			Board ID 67731456			
Link STI01-001						
#	Channel Name	TN	Key0	Key1	Channel Allocation	Class ID
1	STI01-001-001	8.0.0.0	4300	4400	Multimedia	
2	STI01-001-002	8.0.0.1	4301	4401	Multimedia	
3	STI01-001-003	8.0.0.2	4302	4402	Multimedia	
4	STI01-001-004	8.0.0.3	4303	4403	Multimedia	
5	STI01-001-005	8.0.0.4	4304	4404	Multimedia	
6	STI01-001-006	8.0.0.5	4305	4405	Multimedia	
7	STI01-001-007	8.0.0.6	4306	4406	Access	1
8	STI01-001-008	8.0.0.7	4307	4407	Access	2
9	STI01-001-009					

- **In Meridian Mail:** Navigate through the following 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

- **On the switch:** In Overlay 22, use the following command:
  - **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
```

- **On the client:** In the Voice Ports window, the Channel column for the voice port contains a unique number. The voice port status is Acquired Login.

Make sure channels for ACCESS voice ports match on the server and CallPilot or Meridian Mail

Channel number (the number shown in the Class ID column in the CallPilot Channel Monitor or after the “Class” keyword in the Meridian Mail Channel Allocation Table) for a specific TN must match the channel number for the same TN in the Voice Ports window on Symposium Call Center Server.

- **In CallPilot:** Choose Configuration Wizard > CallPilot Manager > Channel Information page. Check the value in the Class ID column.
  Example:

STI Board 200i | Board ID 67731456

Link STI01-001

#	Channel Name	TN	Key0	Key1	Channel Allocation	Class ID
1	STI01-001-001	8.0.0.0	4300	4400	Multimedia	
2	STI01-001-002	8.0.0.1	4301	4401	Multimedia	
3	STI01-001-003	8.0.0.2	4302	4402	Multimedia	
4	STI01-001-004	8.0.0.3	4303	4403	Multimedia	
5	STI01-001-005	8.0.0.4	4304	4404	Multimedia	
6	STI01-001-006	8.0.0.5	4305	4405	Multimedia	
7	STI01-001-007	8.0.0.6	4306	4406	Access	1
8	STI01-001-008	8.0.0.7	4307	4407	Access	2
9	STI01-001-009					

- **In Meridian Mail:** Navigate through the following 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

- **On the client:** Display the Voice Ports window. Ensure that each TN has a unique number in the Channel column. Ensure that the voice port status is Acquired Login.

Checklist for troubleshooting configuration problems with CallPilot

ELAN subnet		✔
Switch	All VSIDs have SECU YES (see “Configuring the ELAN subnet” on page 61).	
MLS		✔
CallPilot	Symposium Call Center Server Nortel server subnet IP address is configured correctly on Switch Information page (see “Updating the CallPilot configuration” on page 128).	
Switch	Voice ports are configured with AST (Associated Set Assignment) on key 0 and key 1 (see “To create a CallPilot voice port with LD 11” on page 85).	
ACCESS link		
Symposium Call Center Server	The connection to CallPilot is configured correctly on the Voice Connection page of the Server Setup Configuration utility (see “Configuring the voice connection” on page 159).	
ACCESS ACD-DNs		
CallPilot	The ACD-DN is configured in the SDN table with Application Name “Symposium Voice Services” and Media Type “Voice” (see “Updating the SDN table” on page 130).	
Symposium Call Center Server	The ACCESS ACD-DN is specified as Default IVR Access DN and Default Access Treatment DN in the IVR ACD-DN Global Settings (see “Configuring IVR ACD-DN global settings” on page 168).	

Switch	The ACD-DN is configured with IVR YES and ALOG YES (see “To create a CallPilot ACD-DN in LD 23” on page 82).	
ACCESS channels		
CallPilot	The channel has a Class ID. The TN assigned to the channel matches the TN on the switch.	
Symposium Call Center Server	The channel defined for the voice port equals the class ID assigned to the channel in CallPilot (see “Configuring voice ports on the server” on page 167). The TN assigned to the channel matches the TN on the switch.	
Switch	Voice ports are configured with AST 00 01 and CLS MMA FLXA (see “To create a CallPilot voice port with LD 11” on page 85).	
Give IVR ACD-DNs		
CallPilot	The ACD-DN is configured in the SDN table with Application Name “Symposium Voice Services” and Media Type “Voice” (see “Updating the SDN table” on page 130).	
Switch	The ACD-DN is configured with IVR YES and ALOG YES (see “To create a CallPilot ACD-DN in LD 23” on page 82).	
Give IVR channels		
CallPilot	The TN assigned to the channel matches the TN on the switch.	
Symposium Call Center Server	The TN assigned to the channel matches the TN on the switch.	

Switch	Voice ports are configured with AST 00 01 and CLS MMA FLXA (see “To create a CallPilot voice port with LD 11” on page 85).	

Problems with voice ports

Voice ports become deacquired

Note: To avoid this problem, assign new IVR ACD-DNs to Symposium Call Center Server.

This problem can occur if Symposium Call Center Server shares ACCESS voice ports (voice ports acquired as both a TN and channel) with other applications. If non-Symposium Call Center Server calls terminate on ACCESS voice ports used by Symposium Call Center Server, those voice ports can become deacquired. To troubleshoot the problem, perform the checks in the following list:

To check for non-Symposium Call Center Server calls

- Examine switch IDC tables to see if they are directing incoming calls to the ACD-DN owning the voice ports.
- Examine switch trunk auto-terminate destinations to see if they are directing incoming calls to the ACD-DN owning the voice ports.
- Examine switch ACD-DN Night Call Forward destinations to see if they are directing incoming calls to the ACD-DN owning the voice ports.
- Examine switch, set Call Forward (No Answer, Busy, and All Calls) destinations to see if they are directing incoming calls to the ACD-DN owning the voice ports.
- Examine switch CDN Default DN destinations to see if they are directing incoming calls to the ACD-DN owning the voice ports.
- Examine CCR scripts to see if they are directing calls to the ACD-DN owning the voice ports.
- Examine Meridian Link applications to see if they are directing calls to the ACD-DN owning the voice ports.
- Examine Symposium Call Center Server scripts and look for Give IVR commands that are directing calls to an 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.

Channel number field in the Voice Ports 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.

Voice ports are Remote Out-Of-Service in CallPilot

To correct this problem, on the switch, verify that the voice ports are configured with AST (Associated Set Assignment) on key 0 and key 1 (see “To create a CallPilot voice port with LD 11” on page 85).

Call 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.
- Voice ports for that IVR ACD-DN are logged on and idle (check switch and CallPilot or Meridian Mail).

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.
- For Symposium Voice Services on CallPilot
  - Voice items and variables are valid and recorded.
  - In CallPilot, on the Channel Monitor page, the ACCESS voice ports are idle, not active.
- For Symposium Voice Services on Meridian Mail
  - 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).
  - Voice segment numbers and variables are valid and recorded.
  - The DSP Port Status menu in Meridian Mail shows the channels in use by Symposium Call Center Server in the Active state (this means that Meridian Mail thinks they are acquired).
- The switch shows the TNs used as voice channels to be logged on and idle.
- The ACCESS link is up.

Calls suspend in the script at the voice processing command

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.

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. To resolve this problem, perform these tasks:

- Ensure that the switch and Symposium Call Center Server configurations match exactly: the TNs configured for the IVR ACD-DN on the 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 CallPilot or Meridian Mail, or move them to a different ACD-DN on the switch and CallPilot or Meridian Mail.
- Check whether non-Symposium Call Center Server calls are arriving at this IVR ACD-DN. For more information, see “Problems with voice ports” on page 226.

Callers hear ringing, but the call is not answered

This problem occurs if the call is not routed to CallPilot, or if the call is routed to CallPilot, but CallPilot does not answer.

Calls are not routed to CallPilot

If calls are not being routed to CallPilot, follow these steps:

1. In Symposium Call Center Server, check the Services Monitor to determine whether the Meridian Link Services Manager (MLSM) is running.If it is running, continue with the next step. If it is not running, use the Windows Services control panel to restart it. If it does not start, restart the server.
2. Make sure that ACCESS is operational on both Symposium Call Center Server and CallPilot.

- In Symposium Call Center Server, look for event 40305, “The ACCESS Link is up.” A subsequent event 40309, “ACCESS Link time-out” indicates a problem with the link.
- In CallPilot, look for event 60921, “ACCESS link is up.” A subsequent event 60920, “ACCESS link is down” indicates a problem with the link.

If there is a problem with the ACCESS link, check the configuration on the switch, CallPilot, and Symposium Call Center Server (see “Subsystem link problems” on page 207). After making the required changes, restart the VSM service from the Windows Services control panel on the server in Symposium Call Center Server.

3 If ACCESS is operational on both servers, MLS may not be running on the CallPilot server. Restart the CallPilot server.

CallPilot is not answering

If calls are being routed to CallPilot, but CallPilot is not responding, follow these steps:

1 Check the CallPilot SDN table to verify that it contains an SDN for Symposium Voice Services (see “Updating the SDN table” on page 130).

2 Check the voice port status in CallPilot. If it is Remote Out-of-Service, the voice ports are configured incorrectly on the switch. Make sure that voice ports are configured with AST (Associated Set Assignment) on key 0 and key 1 (see “To create a CallPilot voice port with LD 11” on page 85).

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 controlled broadcast in 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.

Call does not receive voice treatment

The script executes the voice processing commands, but callers are not connected to the appropriate voice port.

To resolve this problem, perform these steps:

- In Symposium Call Center Server, make sure that the ACD-DN is configured and acquired.
- Make sure all voice ports that are members of this ACD-DN are configured appropriately on the switch, in Symposium Call Center Server, and in CallPilot. Make sure that the TNs and ACD-DN are the same in all three subsystems. Make sure that the position ID defined on the switch matches key 0, as defined in CallPilot, and that the SCN DN defined on the switch matches key 1, as defined in CallPilot.
- In Symposium Call Center Server, make sure that the voice ports are acquired.

For ACCESS voice ports, make sure that the Voice Port Channel defined in Symposium Call Center Server matches the CallPilot class ID.

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. This problem occurs if there is no application running on the voice port.

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

- The announcement on CallPilot or Meridian Mail indicated by the treatment DN exists and is recorded.
- The voice port where the call ended up is in Idle state in CallPilot or Meridian Mail.
- The voice port is in Acquired Login state in Symposium Call Center Server.

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

- In CallPilot, on the Channel Monitor page, the ACCESS voice ports are idle, not active.
- In Meridian Mail, the DSP Port Status menu shows the channels in the Active state.
- 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 CallPilot or Meridian Mail.
- The ACCESS link is up.
- In CallPilot, on the Channel Monitor page, ensure that the ACCESS voice ports are idle, not active.
- 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).

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 CallPilot or Meridian Mail indicated by the treatment DN contains the correct recording.
- In CallPilot, the default treatment DN on the switch should be listed in the SDN table.
- 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).

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

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

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

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

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

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 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 (is recorded).
- The voice ports are acquired correctly (and *all* of the ports in the IVR ACD-DN are acquired).

Voice prompts do not play

If CallPilot comes up before Meridian Link Services Manager (MLSM) is running, all Give IVR and ACCESS channels remain in uninitialized state. To avoid this problem, always start Symposium Call Center Server before CallPilot.

To ensure that CallPilot always starts after Symposium Call Center Server, even if both servers are powered up at the same time, defer the CallPilot boot-up by 5 minutes by making the following change to the Windows settings on the CallPilot server:

1. From the Windows control panel, open **System**.
2. Click the **Startup/Shutdown** tab.
3. In the **System Startup** box, set **Show list for** to **300 seconds**.

Caller hears voice prompts but is never presented to an agent

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

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

Other problems

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 subnet may be excessive. You must ensure that the network is designed to support the number of devices installed on the ELAN subnet. For information about proper network design, refer to the *Planning and Engineering Guide*.

Symposium Call Center Server calls are not being presented to agents

When agents become idle, they are presented with ACD calls rather than with waiting Symposium Call Center Server calls.

If an ACD call is waiting when an agent becomes idle, the switch ACD software routes the ACD call to the agent, and the switch notifies Symposium Call Center Server that the agent is available. By the time that Symposium Call Center Server receives the notification, the agent is already handling the ACD call, and is no longer available to handle the skillset call.

To prevent this problem, do not use the ACD-DN for customer calls.

Notes:

- If no ACD calls are waiting when the agent becomes idle, a Symposium Call Center Server call is presented.
- This problem does not occur with NACD calls. When an agent finishes handling an NACD call, the switch provides a small time window to allow a Symposium Call Center Server call to be presented. If no Symposium Call Center Server call is available, the switch presents the next NACD call.

Network call-by-call reports display inaccurate timestamps

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

Check the timestamp 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.

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 statement 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 command.
2. The Open/End Voice Session Play Prompt has a *no-type-ahead* option, which is not applicable to the Controlled Broadcast command.
3. The start/stop timer for starting the announcement applies only to Broadcast operation. When the Open/End Voice Session command 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 149, which explains how treatment DNs are used and determined.

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

- 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 services, 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 CallPilot or Meridian Mail, 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.
- CallPilot or Meridian Mail voice ports configured as channels, and Symposium Call Center Server having acquired these ports with channel numbers assigned, are required for the Controlled Broadcast and the Voice Session script commands to work correctly.

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

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

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

See Chapter 4, "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 10

Agent phonesets

In this chapter

Supported phonesets

Introduction

This section describes the phonesets supported with Symposium Call Center Server. For more information, refer to the Partner Information Center publications.

Call center phonesets

Phonesets that are designed specifically for call centers (the 2216 and 3905) are recommended for use with Symposium Call Center Server.

Meridian 1 ACD phonesets

Symposium Call Center Server also supports all phonesets that can be configured for use with Meridian 1 ACD, including the i2004 and i2050 phonesets.

Note: Because these phonesets are not designed for a call center environment, some call center features are not available when you use them.

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 Succession 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

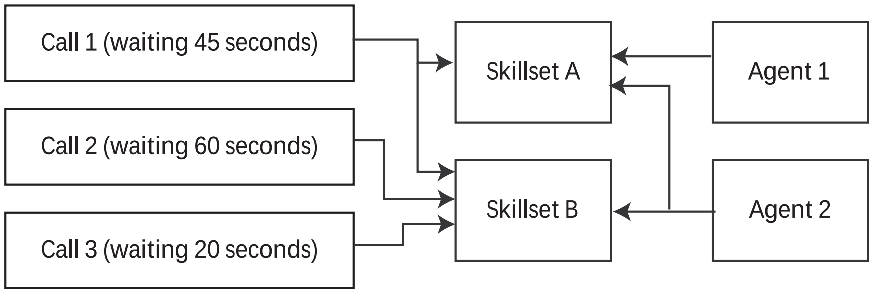
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	→	39	40
1	n	n	2	3		n	n	1	7		n	1	6	5			→		

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.

Unsupported phoneset keys

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.

Other 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

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 265 is documented in detail in the appendix, "Migrating Symposium Call Center Server Voice Services," in the *Meridian Mail to CallPilot Migration Utility Guide.*

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 Nortel server subnet.
Voice services control	Meridian Mail uses the serial ACCESS link.	CallPilot uses the TCP/IP and ACCESS protocols over the ELAN subnet.

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

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 260). 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**
	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.) For example: ▪ 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 DNs are constant numbers, consider using variables of type Treatment DN.
	CallPilot Release 4.0 is installed and operational. CallPilot ELAN subnet IP address: ______________________
	Meridian Mail to CallPilot Migration package (order number NTUB24AC/A0886190) is available for installation. The package consists of the following components: ▪ CP 2.0 Documentation CD-ROM Package (NTRG19AF/ A0820778) ▪ Meridian Mail to CallPilot Migration Tape (NTUB25AA/ A0992892)
	Meridian Mail hardware platform meets the requirements. See the *Meridian Mail to CallPilot Migration Utility Guide*.
	Meridian Mail 11 or greater is installed and operational.
	Meridian Mail tape drive hardware and media meet the requirements. See “Tape drive hardware and media requirements” on page 258.

✔	**Description**
	CallPilot Application Builder is installed and operational in the network.
	Symposium Call Center Server Release 5.0 is installed and operational. Nortel server subnet IP address: ____________________
	Data that cannot be migrated has been identified. In the *Meridian Mail to CallPilot Migration Utility Guide*, review these sections: ▪ Section A: “Understanding what can be migrated to CallPilot” ▪ Section B: “Migration strategies”
	ACCESS mailboxes are checked to ensure that ▪ the mailbox DNs are 3 or more digits in length ▪ the DNs are unique across all mailboxes
	Meridian Mail Class of Service (COS) names are unique across all classes of service. **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 *Meridian Mail to CallPilot Migration Utility Guide* and the *Meridian Mail System Administration Guide*.
	Symposium Call Center Server prompt file names are unique. **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.

✔	Description
	The required number of blank tapes are available to store the data collected from Meridian Mail. **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. For more information, see *Meridian Mail to CallPilot Migration Utility Guide*, Section B: “Migration strategies.”
	A maintenance period is arranged for Meridian Mail, CallPilot, and Symposium Call Center Server, so that system performance is not impacted by the migration. **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.
	The distribution level password for the CallPilot 2.0 support tools is available.

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:

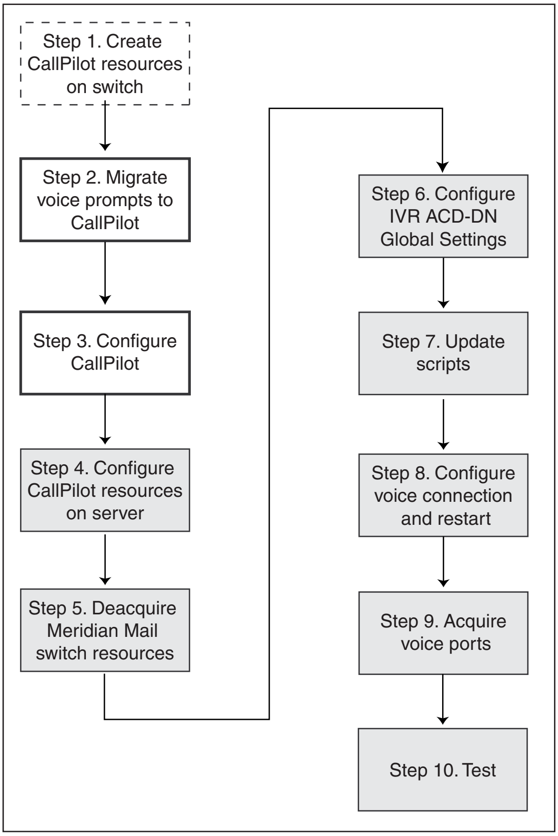
- SLR4 - 2.5 Gbyte tapes
- SLR5 - 2.5 Gbyte tapes
- SLR32 - 2.5 Gbyte tapes
- SLR50 - 2.5 Gbyte tapes

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.

Note: The procedures in this guide use Classic Client. You can also perform these procedures with the Web Client.

✔	**Description**
	Complete the pre-migration checklist on page 255.
	Step 1. Create CallPilot ACD-DNs and voice ports on the switch For instructions, see page 262.
	Step 2. Migrate Meridian Mail voice prompts and voice segments For instructions, see page 265.
	Step 3. Configure CallPilot For instructions, see page 266.
	Step 4. Configure CallPilot switch resources For instructions, see page 269.
	Step 5. Deacquire Meridian Mail voice ports and ACD-DNs For instructions, see page 273.
	Step 6. Configure Symposium Call Center Server connection to CallPilot For instructions, see page 274.
	Step 7. Configure the IVR ACD-DN Global Settings For instructions, see page 276.

✔	**Description**
	Step 8. Acquire CallPilot resources For instructions, see page 278.
	Step 9. Update your scripts For instructions, see page 279.
	Step 10. Test the migration. For instructions, see page 280.

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

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.

Prompt	Response	Description
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	FLXA (units 16–31) VCE WTA CTD MMA	Flexible voice/data allowed. Voice Terminal Warning Tone Allowed Conditionally Toll Denied Multimedia Agent **Note:** CTD is optional, but prevents outbound long distance calls from a voice port.
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).

Prompt	Response	Description
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 the appendix, “Migrating Symposium Call Center Server Voice Services,” in the *Meridian Mail to CallPilot Migration Utility Guide* (NTP 555-7101-801).

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*.

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 a Classic Client or Web Client PC.

Section C: “Comparison of VPE and Application Builder,” on page 293 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 Nortel server subnet IP 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 **Nex**t until the **Switch Information** 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 Step 4, "Configure CallPilot switch resources"), 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**.

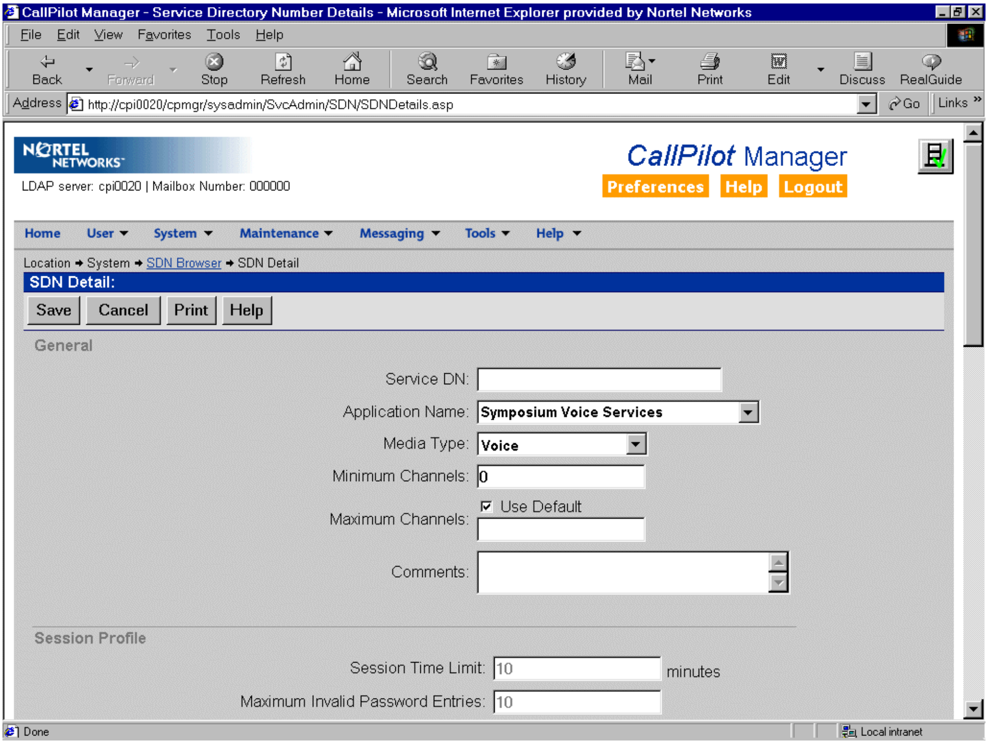
To configure the ACCESS and Give IVR channels in the SDN table

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.



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

Introduction

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.

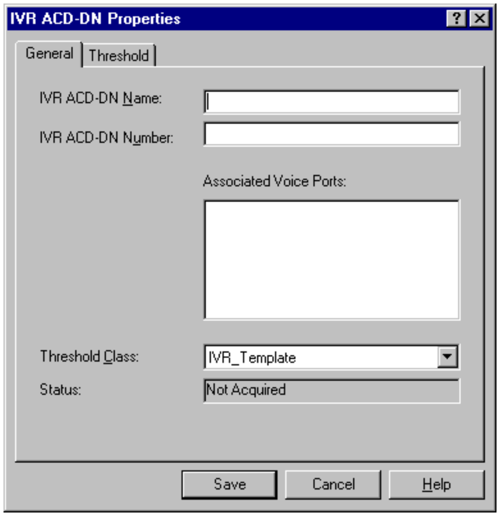
This procedure assumes that the class IDs of your CallPilot TNs are different than your Meridian Mail channel numbers. If they are not, you must either delete the channel number from the Meridian Mail ACCESS ports, or assign different class IDs to your CallPilot TNs.

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 In the **IVR ACD-DNs** window, choose **File** > **New**.

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



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**.

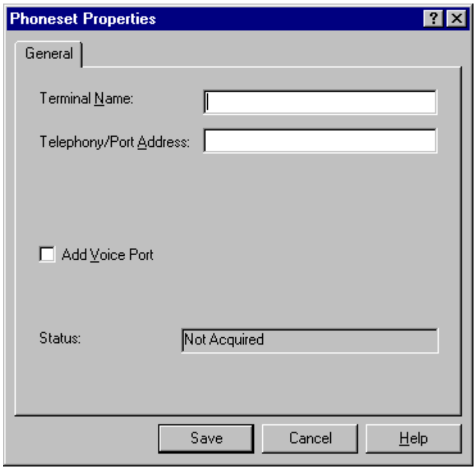
Result: The IVR ACD-DN is added to the list in the IVR ACD-DNs window. It has the status Not Acquired.

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

To create a voice port

1. In 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.

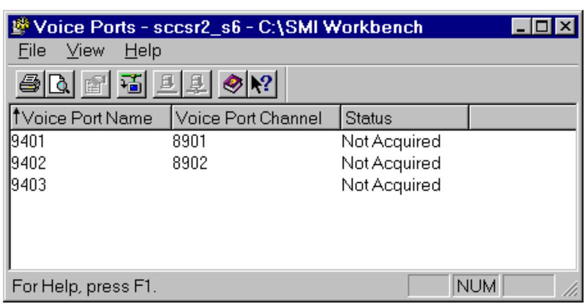
   Telephony/Port Address: The address of the voice port on the switch.

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. In the **SMI** window, choose **Switch Administration** > **Voice Ports**.

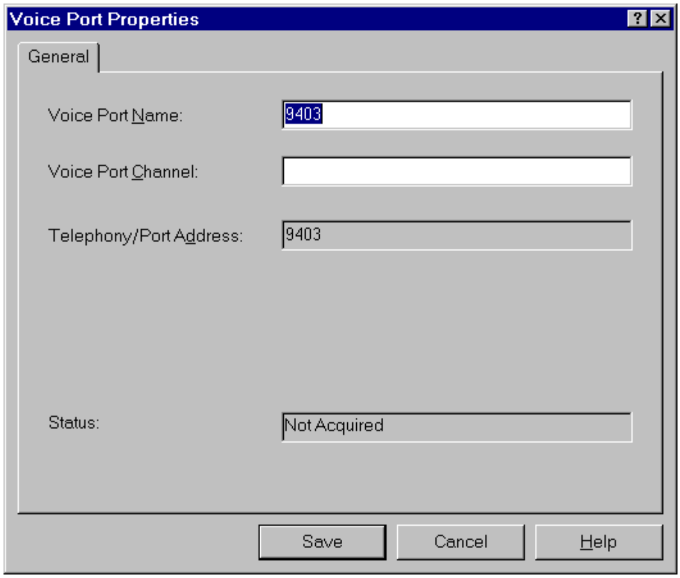
   Result: The Voice Ports window appears.



6 In the **Voice Ports** window, select the phoneset that you added.

7 Choose **File** > **Properties**.

Result: The Voice Port Properties property sheet appears.



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 In 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** > **Deacquire**.

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 In 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 Symposium Call Center Server connection to CallPilot

Introduction

To use CallPilot for voice processing, install Symposium Call Center Server Release 5.0, and configure the Symposium Call Center Server connection to CallPilot. Then restart the Symposium Call Center Server application.

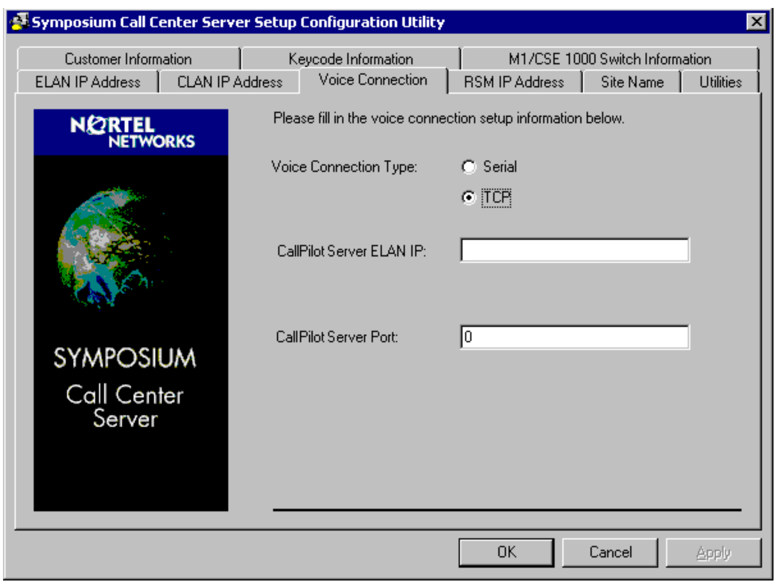
Note: Nortel recommends that you retain the physical cable connections between Meridian Mail and Symposium Call Center Server to facilitate backout, if required.

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 subnet 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.

Step 7. Configure the IVR ACD-DN Global Settings

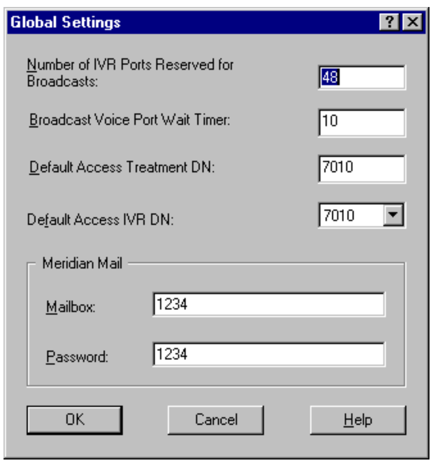
Introduction

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

To update IVR ACD-DN Global Settings

1 In the **IVR ACD-DNs** window, choose **File** > **Global Settings**.

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



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 8. 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. In 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. In 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 9. 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 recommends that you use variables to represent ACD-DNs in your scripts.

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

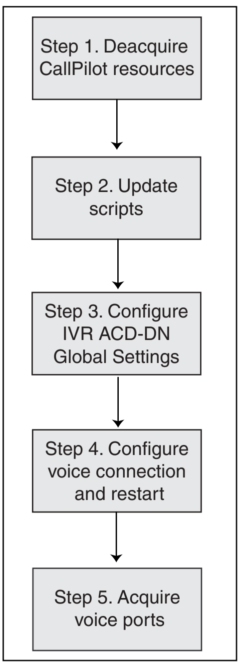
Introduction

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.

Note: The procedures in this guide use Classic Client. You can also perform these procedures with the Web Client.

✔	Description
	Step 1. Deacquire CallPilot ACD-DNs and voice ports For instructions, see page 284.
	Step 2. Configure the connection to Meridian Mail For instructions, see page 285.
	Step 3. Configure IVR ACD-DN Global Settings For instructions, see page 288.
	Step 4. Acquire voice ports For instructions, see page 290.
	Step 5. Update your scripts For instructions, see page 291.

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. In 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. In 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 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 W**indows 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.

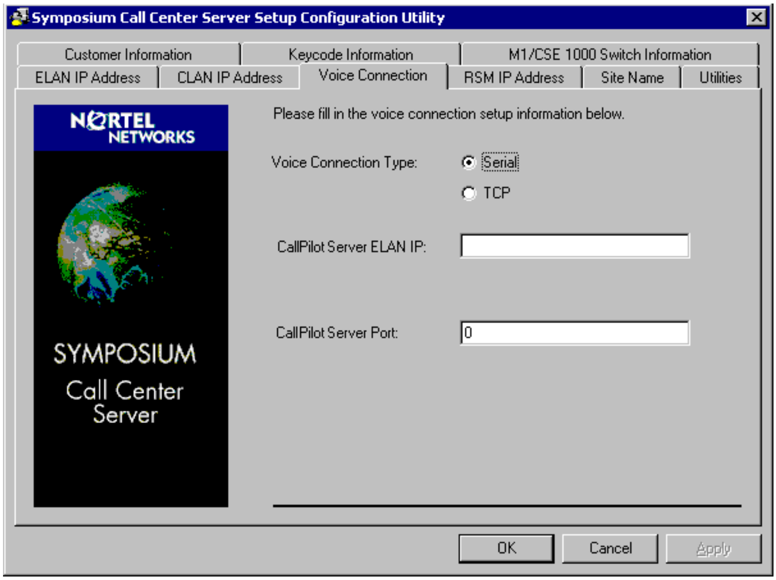
4. Click **Accept** to exit the utility.

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.



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.

Step 3. Configure IVR ACD-DN Global Settings

Introduction

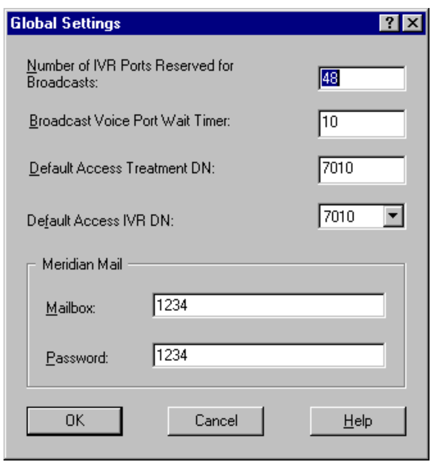
You must update the IVR ACD-DN Global Settings with the Meridian Mail ACCESS IVR and Treatment DNs, and verify that the Meridian Mail mailbox and password are correctly specified.

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 In the **IVR ACD-DNs** window, Choose **File > Global Settings**.

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



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. This number is the same as the ACCESS ACD-DN. Do not explicitly specify a treatment DN in a script.

Default Access IVR DN: This field is the ACCESS ACD-DN as defined on the switch and in Meridian Mail.

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 4. 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 In 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 In 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**.

Step 5. 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 recommends that you use variables to represent ACD-DNs in your scripts.

Section C: Comparison of VPE and Application Builder

In this section

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 userid and password. 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
Undeleting voice files	Undelete menu command	Not available

Feature	VPE	Application Builder
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
Prompt size	Up to 2 minutes	Up to 10 minutes

Feature	VPE	Application Builder
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 overriden by administrator.
Description length	Maximum 2048 characters	Maximum 240 characters

Appendix B

Migrating from Meridian MAX to Symposium Call Center Server

In this appendix

Premigration steps

This section provides a migration plan for a migration from Meridian MAX MQA to Symposium Call Center Server with a *staged* cutover.

Before beginning the migration, verify the following items:

1. In MAX, make sure no Change Orders have been created or scheduled.
2. Check Meridian Mail or CallPilot announcements, menus, and treatments that are used in CCR or as entry points; ensure that they do not route calls to old ACD-DNs or CDNs that are no longer used. Symposium Call Center Server needs its own Meridian Mail ACD-DN for voice processing, and cannot share CCR's IVR ACD-DN.
3. Check the CCR Variable table to ensure that the ACD-DNs have been removed.
4. Check any link programming (LD 17, 15, 10, 11, 23) to ensure that ACD-DNs and CDNs that use CTI are associated with the ELAN subnet link ID and not the AML.
5. Make sure routes are not auto-terminating on old ACD_DNs or CDNs.
6. Check the IDC tables for old ACD-DNs and CDNs.

Migration steps

To perform the migration with a staged cutover:

1. Create a new ACD-DN with enough MAXP assigned.
2. Set RPRT = YES on this new ACD-DN.
3. Instruct call center agents to use single standard login prior to leaving on the night of the cutover. (Alternatively, if acceptable, Sysload the switch. The only ACD-DN maintained is the ACD-DN defined in the service change.)
4. Verify that agents are assigned to one queue in one of the following ways:
   - Run the Configuration Update in MAX, and then print the Configuration Report, which shows all ACD-DNs and the positions assigned to them.
   - View the Real Time Display - Agent by ACD-DN.
5. Build and submit a Meridian MAX Change Order to move Agents to a new ACD-DN.

   Note: Nortel recommends that you execute the Change Order at night when the agents are not present.
6. Verify on the switch and in Meridian MAX that no agent positions are left in old ACD-DNs.
7. Disassociate the CCR script. (The CDN should show CNTL = NO.)
8. For the CDN, change the old default ACD to a new ACD-DN, either in Meridian MAX or Overlay 23. (Optionally, remove the VAS ID, although this is not necessary.) When Symposium Call Center Server acquires the CDN, the ASID appears.
9. Test the CDN and default queue routing. (Calls should be presented to agents in the new ACD-DN.)
10. Create Symposium Call Center Server scripts.

    Note: The Master script must route calls to the CDN's Default DN if the CDN has not been acquired or if it has no script assigned. Therefore, you can add CDNs to the Master script and comment them out until the script has been tested. After testing, you can acquire the CDN and remove the comments. Following is an example Master script:

```
IF TRANSFERRED THEN
   GIVE RINGBACK
   WAIT 6
ELSE
   GIVE RINGBACK /* As long as busy isn't given later */
END IF
IF test_cdn THEN
   EXECUTE SCRIPT Test_1
END IF
/*
WHERE CDN EQUALS
   VALUE xxxx: EXECUTE SCRIPT xxxx
   VALUE xxxy: EXECUTE SCRIPT xxxy
   VALUE xxxz: EXECUTE SCRIPT xxxz
END WHERE

*/
ROUTE CALL DEFAULT DN
```

11 In Symposium Call Center Server, acquire phonesets.

12 In Symposium Call Center Server, acquire a test CDN, and associate a script with this CDN.

13 Place test calls receiving night treatment.

14 Change the hours in the script to offer day treatment.

15 Place test calls receiving day treatment.

16 Change the hours in the script back to night treatment.

17 Acquire CDNs and assign validated/tested scripts to live CDNs.

Note: As part of the migration plan you must consider all Meridian MAX third-party devices such as wallboards, forecasting packages, adherence, quality monitoring, and so on.

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

An internal protocol used by Symposium Call Center Server to directly control some of the voice services available on the CallPilot or Meridian Mail platform.

access class

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

access level

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

ACCESS link

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

ACCESS voice port

A 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 telephony 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.

active server

In a system with a Replication Server, the server that is providing call processing and administration services.

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 might be used to track sales calls. Agents can then enter 720 on their phonesets during sales calls, and this information can be generated in an Activity Code report.

administrator

A user who is responsible for setting up and maintaining the 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.

AML

See Application Module Link.

API

See application program interface.

application

1. A logical entity that represents a Symposium Call Center 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 Module Link

An internal protocol used by Symposium Call Center Server to communicate with the switch.

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.

application server

The server on which the Symposium Web Client software is installed. This server acts as the middle layer that communicates with Symposium Call Center Server and makes information available to the client PCs.

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 telephony 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 telephony switch.

Automatic call distribution routing table

A table configured on the telephony 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, 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.

CallPilot

A multimedia messaging system you can use to manage many types of information, including voice messages, fax messages, e-mail messages, telephone calls (including conferencing), calendars, and directories.

CDN

See controlled directory number.

CLAN subnet

See Customer local area network, enterprise IP network.

Classic Client

The Windows-based client component for Symposium Call Center Server.

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. Two types of client are available: Classic Client and Symposium Web Client. *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.

Contivity VPN Switch

A Nortel product that provides routing, firewall, bandwidth management, encryption, authentication, and data integrity for secure tunneling across managed IP networks and the Internet.

controlled directory number

A special directory number that allows calls arriving at the telephony 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.

CTI

Computer Telephony Integration

Customer local area network

The LAN to which your corporate services and resources connect. The server in 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 is displayed on the phoneset, can prepare a response according to the DNIS.

directory number

The number that identifies a phoneset on a telephony 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.

DMS

Digital Multiplex Switch

DN

See directory number.

DN call

See directory number call.

DNIS

See Dialed Number Identification Service.

dongle

The attachment plugged into the parallel port of a server connected to a DMS/SL-100 switch that authenticates the serial number required at the time of server installation.

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 subnet

See embedded local area network subnet.

embedded local area network subnet

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

enterprise IP network

Your entire IP network, including the ELAN subnet and the Nortel server subnet.

event

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

expression

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

H

HDX

See Host Data Exchange

Host Data Exchange

A rich scripting language provided with Symposium Call Center Server to control treatment of calls.

I

Incalls key

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

Interactive voice response

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

Interactive voice response ACD-DN

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

Interactive voice response event

A voice port 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, 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, relational expression.

M

M1

Meridian 1 switch

M1IE

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/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 telephony switch and a third-party host application.

Meridian Mail

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

Meridian MAX

A Nortel 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 telephony 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 Network Skill-Based Routing (NSBR) is configured and where communication between servers is managed.

network interface card

An expansion board that enables a PC to be connected to a local area network (LAN).

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 Network Skill-Based Routing (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 Sever 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.

Optivity Telephony Manager

A Nortel application used for telephony switch management. It provides management simplicity and flexible control.

OTM

See Optivity Telephony Manager.

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 telephony 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 telephony 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 telephony 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.

Real-time Statistics Multicast

An interface that provides real-time information to third-party applications in either multicast or unicast format.

recorded announcement route

A resource installed on the telephony 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 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.

Replication Server

A server that backs up the active server to the standby server in real time.

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 telephony 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.

RSM

See Real-time Statistics Multicast.

S

sample script

A script that is installed with 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.

SEI

See Symposium Event Interface.

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 (NSBR).

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.

SL-100

Stored Logic 100 switch

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.

standby server

A server that contains an up-to-date version of the database, for use when the active server becomes unavailable.

supervisor

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

SWCP

See Symposium Web Center Portal.

switch

See telephony switch.

switch resource

A device that is configured on the telephony 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 Agent

An agent productivity tool that enables contact center agents to provide intelligent and personalized customer care. Agents use a personal computer to access the agent telephony functions.

Symposium Call Center Server

A client/server contact center solution for varied and changing business requirements. It offers a suite of applications that includes call processing and agent handling, management and reporting, networking, and third-party application interfaces.

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.

Symposium Event Interface

An interface that provides third-party vendors with the information they need to create complementary applications by providing call progress and resource events.

Symposium Standby Server

The server that contains an up-to-date backup version of the Symposium Call Center Server database, for use if the active server fails. The database is kept up-to-date by the Replication Server.

Symposium Web Center Portal

A client/server contact center application that expands contact center e-mail capabilities to allow agents to view, respond to, and track requests over the Internet.

Symposium Web Client

A browser-based tool for call center administrators and supervisors used for managing and configuring a contact center and its users, defining access to data, and viewing real-time and historical reports. The Symposium Web Client software is installed on an application server. *See also* application server.

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

TAPI

See Telephony Application Program Interface.

target site

See destination site.

TCP/IP

See Transmission Control Protocol/Internet Protocol.

TDM

See Time-Division Multiplex.

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.

telephony switch

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

Telephony Application Program Interface

An interface between the telephony switch and an application that allows the application to control the telephone on a user's desktop.

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-Division Multiplex
A method of transmission in which a signal is separated into multiple segments at the transmission source, and then reassembled at the receiving end.

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.

Virtual Private Network

A private network that is configured within a public network to take advantage of the economies of scale and management facilities of large networks.

voice port

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

VPN

See Virtual Private Network.

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 the Symposium Call Center Server.

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Reader Response Form

Nortel Symposium Call Center Server
Product release 5.0
Symposium, M1/Succession 1000, and Voice Processing Guide

Tell us about yourself:

Name: ______________________________

Company: ______________________________

Address: ______________________________

____

Occupation: ______________________ **Phone:** ______________

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2. How do you use this book?

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   ____

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6. How could we improve this book?

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Reader Response Form

Nortel Symposium Call Center Server

Symposium, M1/Succession 1000, and Voice Processing Guide

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