SpeechAttendant™

Implementation Guide

Version 8.2

Document history

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Preface

SpeechAttendant™ integrates the latest innovations in speech technology and lends itself to a variety of telephony applications, including speech-activated call routing, call dispatching in a call center environment, access to personal contacts, the provision of self-service information and call forward.

This guide explains the procedures for implementing a new SpeechAttendant application.

Who should read this guide

Implementing SpeechAttendant involves the collaboration of several people: the ScanSoft project manager, the distributor representative, the telecommunications technician, the network administrator, the SpeechAttendant system administrator, etc.

This guide is intended for all these people. However, first and foremost, it is intended for specialists duly certified by ScanSoft who are responsible for installing and configuring SpeechAttendant at the client site.

Readers of this guide should:

- Have taken the *Implementing SpeechAttendant* course
- Have a working knowledge of computer systems
- Be familiar with Windows operating systems
- Be familiar with PBXs and other telephone systems

About this guide

This *Implementation Guide* explains how to implement SpeechAttendant and includes the following chapters:

1. **Implementation Process**—Describes all the steps in the implementation process and contains a practical checklist to make sure that you are performing all the necessary tasks.
2. **Getting Started**—Explains the first steps of the installation process, which include computer setup, network connection, and license activation.
3. **PBX/KSU Preparation**—Explains how to configure the PBX/KSU and connect it to SpeechAttendant for a standard analog or digital application.
4. **ACD Environment**—Contains the necessary procedures for integrating SpeechAttendant in a call center environment (Meridian or Avaya).

6 **Business Communications Manager (BCM)**—Explains how to integrate SpeechAttendant with Nortel Networks BCM.

7 **Multi-server application**—Contains all the information pertaining to a multi-server application, including how to configure data replication.

8 **SpeechAttendant Configuration**—Explains how to configure the SpeechAttendant system according to customer needs.

9 **pcAnywhere**—Explains how to prepare the pcAnywhere connection, which makes it possible to manage the application from a remote computer.

10 **Testing**—Explains how to test PBX/KSU integration and SpeechAttendant functions.

An index and a feedback form complete the guide.

What’s new

As with each new version of the product, we have made certain changes to the system.

In the present manual, pay particular attention to the following changes:

- D/42 and D/82 boards are now supported in digital Avaya environments, as they are in digital Meridian and Mitel environments. To know how to connect those systems to the PBX, see page 22.

Conventions

The following conventions are used throughout this guide:

Visual cue	Meaning
Bold face	Is used to highlight the names of windows, dialog boxes, fields and other interface elements.
Italic	Is used for new terms, names given as examples, and text to be replaced with appropriate values.
`Courier font`	Is used for file paths and text that you must type.
Greater-than sign (>)	Indicates a sequence of menu-commands to select. For example, **File** > **Save** means that you must select **File**, then **Save**.
Plus sign (+)	Separates keys that you must press at the same time. For example, Ctrl+C.
	Introduces a note, a reminder or extra information.
	Warns you that an action could cause the system to malfunction.
	Indicates a tip.

Related documentation

In addition to the *Implementation Guide*, the following documentation is also available:

- The *Site Preparation Document* lays the groundwork for the visit by the project manager on customer site. It describes the requirements of system implementation and allows to gather customer preferences.
- The *Installation Handbook* explains how to install the software components of a SpeechAttendant application: OpenSpeech Recognizer, Speechify (if included in the license), the companion programs, and the SpeechAttendant software itself.
- The various *Telephony Integration Documents* are extracted from the present *Implementation Guide*. They explains to the telephony technician how to program the customer's PBX and connect it to SpeechAttendant. There is a document for each type of integration: analog, digital (Avaya, Meridian, Mitel, and NEC), ACD queue (Meridian and Avaya), and Nortel Networks BCM.
- The *Administration Guide* covers all aspects of the day-to-day operation of SpeechAttendant.

About us

ScanSoft offers advanced network technologies through automatic speech recognition (ASR), text-to-speech (TTS), and speaker verification solutions to leading corporations, telecommunications providers and government organizations worldwide. Through the power of ScanSoft's speech recognition products and related technologies, the human voice is all a person needs to access instant information and conduct transactions from any landline or wireless phone, car or other handheld device. Around the world, customer service innovators are realizing returns on ScanSoft applications that consistently delight and serve customers 24 hours a day.

For more information about ScanSoft, visit our Web site at www.scansoft.com.

Support

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Documentation

We appreciate your feedback on our documentation. Please fill out the feedback form at the end of this guide and fax it to the Documentation department at (514) 954-3805.

1

Implementation Process

This chapter outlines the principal steps in implementing SpeechAttendant on the client site. It also contains a checklist that you can use to make sure that you have remembered everything.

Introduction

The SpeechAttendant software is shipped installed on a dedicated computer. The system must now be implemented at the client site. To do so, you must:

1. Set up the SpeechAttendant computer (see page 8).
2. Connect the computer to the network (see page 10).
3. Activate the SpeechAttendant license (see page 11).
4. Program the PBX/KSU and connect it to SpeechAttendant, referring to the appropriate chapter:

Integration type	Reference
Analog	Chapter 3, page 13
Digital (Avaya, Meridian, Mitel, or NEC)	Chapter 3, page 13
Meridian ACD queue	Chapter 4, page 28
Avaya ACD queue	Chapter 4, page 33
Nortel Networks BCM	Chapter 5, page 39

5. Configure the SpeechAttendant system (see Chapter 7, page 73).
6. Prepare the pcAnywhere connection, if desired (see Chapter 8, page 105).
7. Create the menu structure and perform the initial load of the directory (refer to the *Administration Guide*).
8. Test the PBX integration and SpeechAttendant functionality (see Chapter 9, page 115).

In the case of a **multi-server** application, the implementation process is similar, except that you must also program data replication. See Chapter 6, page 57 for specific guidelines.

Checklist

By following along with this checklist, you can make sure you are performing all the necessary tasks for SpeechAttendant to function properly, in the correct order. It is a good idea to photocopy it and refer to it during the implementation

Get started

Action	OK	N/A	Reference
1 Set up the SpeechAttendant computer.	❑	❑	See page 8.
2 Connect the computer to the network.	❑	❑	See page 10.
3 Activate the SpeechAttendant license.	❑	❑	See page 11.

Prepare the PBX/KSU†

Action	OK	N/A	Reference
4 Program the hunting method or the ACD queue.	❑	❑	See page 15, for standard application, or Chapter 4, page 27, for an ACD queue.
5 Program the attributes of the ports dedicated to SpeechAttendant, and assign the appropriate class of service.	❑	❑	See pages 16 to 17.
6 Prepare the modem connection.	❑	❑	See page 17.
7 Establish a fallback plan in case all ports are busy or calls cannot be answered.	❑	❑	See page 18.
8 Test the PBX/KSU.	❑	❑	See page 18.
9 Install the ATAs, if applicable.	❑	❑	See page 19.
10 Connect SpeechAttendant to the PBX/KSU.	❑	❑	See page 20.

† In the case of an integration with Nortel Networks BCM, see Chapter 5, page 39.

Configure the SpeechAttendant system

Action	OK	N/A	Reference
11 In the configuration panel, make all the necessary changes to the settings.	❑	❑	See page 74.
12 Configure Windows parameters.	❑	❑	See page 99.
13 Connect to a printer.	❑	❑	See page 100.
14 Modify the default passwords.	❑	❑	See page 100.
15 Create the menu structure and perform the initial load of the directory. **NOTE** Remember to validate the directory and back up the application.	❑	❑	Refer to the *Administration Guide*.

Prepare pcAnywhere connection

Action	OK	N/A	Reference
16 If applicable, prepare pcAnywhere connection.	❑	❑	See Chapter 8, page 105.

Test the installation

Action	OK	N/A	Reference
17 Test PBX integration:			
◆ Noise level	❑	❑	See page 116.
◆ Barge-in activation	❑	❑	See page 117.
◆ Ring tone detection	❑	❑	See page 117.
◆ Busy tone detection	❑	❑	See page 118.
◆ Hangup detection	❑	❑	See page 119.
◆ DTMF transfers	❑	❑	See page 119.
◆ Call forward capabilities	❑	❑	See page 120.
18 Test SpeechAttendant functions:			
◆ Transfer to the operator	❑	❑	See page 121.
◆ Transfer to directory entries	❑	❑	See page 122.
◆ Transfer to external and voice mail numbers	❑	❑	See page 122.
◆ Homophone resolution	❑	❑	See page 123.
◆ Maintenance mode functions	❑	❑	See page 124.
◆ Personal administration mode (PAM)	❑	❑	See page 124.
◆ Dynamic call routing	❑	❑	See page 125.
◆ Transfer to personal contacts	❑	❑	See page 126.
◆ Private phones	❑	❑	See page 127.
◆ Language toggling	❑	❑	See page 127.
◆ Call forward	❑	❑	See page 128.
◆ Remote connection	❑	❑	See page 128.

The implementation of SpeechAttendant is complete and the application is functional. It is now up to the system administrator to see to the day-to-day management of the application using the *Administration Guide*.

Installed by:	Date:
Approved by:	Date:

Follow-up items:

2

Getting Started

The first steps in implementing SpeechAttendant are:

- Setting up the SpeechAttendant computer
- Connecting to the network
- Activating the license

The instructions to do so are provided in this chapter.

Setting up the SpeechAttendant computer

Before setting up the SpeechAttendant computer, verify all the items shipped against the shipping list. Advise the Customer Service Center if anything is missing.

Required equipment

The following equipment is needed to set up the SpeechAttendant computer(s).

- A Phillips screwdriver
- A static-safe kit (dissipative mat, wrist strap and groundwire)
- One RJ11 cable for each port in the license, plus one for the modem[1]
- One analog terminal adapter (ATA) for each SpeechAttendant port[2], plus one for the modem (for Norstar integrations only)
- A network cable (if a network connection is required)
- Up to three analog or digital telephones, depending on the system (or two analog telephones and a butt set), for testing

1. Required cables are provided with the system.
2. ATAs are provided when required.

Location

The SpeechAttendant computer must be set up in a room equipped with the following:

- Up to five AC outlets, 15 A /120 V (or one outlet plus a power bar) located within approximately three feet of the system
- 50 to 75 degrees Fahrenheit
- 20% to 80% non-condensing humidity
- One network jack and cable
- One phone jack per SpeechAttendant port, within eight feet of where the SpeechAttendant computer will be installed
- A telephone that is connected to the PBX/KSU and located near the SpeechAttendant computer
- A modem line for remote connection

Since it will be used to record names and messages, the computer should be installed in a quiet environment. If this is not possible, you must make another computer available for these recordings. This computer must be equipped with pcAnywhere remote access software, and must be installed in a quiet room (see Chapter 8, page 105, for details about pcAnywhere).

If the system includes the MultiAdmin option, names and messages can be recorded from an administrator workstation. Refer to the *Administration Guide* for details on the MultiAdmin option.

Connecting to the network

Connecting SpeechAttendant to the LAN is highly recommended, and it is mandatory if the system includes (or must allow) the following:

- Statistics (log files) transmittal to ScanSoft
- Access to ScanSoft FTP server
- A multi-server application (see Chapter 6, page 57)
- The corporate directory interface option (refer to the *Administration Guide*)
- SpeechContacts option (refer to the *Administration Guide*)

The SpeechAttendant computer includes a PCI Ethernet card with an RJ45 (100BaseT) jack. You need to provide a Category 5 cable, terminated in a "male" RJ45 connector to connect the SpeechAttendant system to the network.

The network administrator or IT specialist is best positioned to configure the network connection.

When connecting to the network:

- Make sure the `locus` administrator retains its administration rights.
- Do not change the name of the SpeechAttendant computer. If you have done so by mistake, contact the Customer Service Center. To know the name of the computer, click **Help** > **About SpeechAttendant** in the **Menu Editor** window.
- Provide an IP address (a static address is recommended).
- Set the DNS suffix for the Apache server.

The SpeechAttendant computer is not meant to be a part of the customer domain. This provides security for the customer by ensuring ScanSoft can only access the SpeechAttendant computer.

Activating the license

Once you have set up the SpeechAttendant computer and connected it to the network, you are ready to activate the license. The procedure is the same for master and slave servers.

To activate the license:

1. Log on to the SpeechAttendant computer as `locus`.

   You will not find default passwords in this guide. If you do not know these passwords, contact the Customer Service Center.

2. Insert the SpeechAttendant CD-ROM into the disk drive.

3. When prompted to perform a full system backup, click **OK to continue installation**, then **Next**.

4. When informed that SpeechAttendant is already installed, choose **Reconfigure License** then click **Next**.

   The **License Setup** and **License Options** windows appear. They contain information about the system purchased.

5. Verify the information and click **Next**.

   The **First Config Setup** window appears. It allows you to customize global system parameters if required.

6. Verify the information to make sure it corresponds to the telephony environment, paying particular attention to the **PBX**[1] and **Number of ports activated** fields. Make any necessary changes. When you are done, click **Next**.

   For details on all the system's configuration settings, see Chapter 7, page 73.

7. In the **Second Config Setup** window, validate the system configuration. Make any necessary changes, then click **Next**.

1.If this is a digital PBX, choose a model with the suffix **VB3000**, **D42**, or **D82**, depending on the case. If you do not know which PBX/KSU to choose in the list, contact the Customer Service Center.

8 Call the Customer Service Center at:

- ❖ 1 866 434-2564

  or

- ❖ (514) 390-3922

  Say "Technical support," then tell the technician that you need a valid serial ID.

9 In the **Serial ID** window, enter the serial number that was given to you, then click **Validate**.

  All letters in the serial ID are capitalized, and there are never any zeroes.

10 When prompted to stop the SpeechAttendant services, click **Yes**.

11 When asked if the current system is connected to a network, click **Yes** or **No** depending on the case.

12 When informed that the configuration is complete, click **OK**.

13 In the case of a multi-server application, repeat the entire procedure on the slave server.

You are ready to continue with the implementation of SpeechAttendant.

3

PBX/KSU Preparation

For SpeechAttendant to be able to receive and route calls, it must be connected to a PBX/KSU that you have programmed according to the customer's needs (refer to the signature pages of the *Site Preparation Document*).

This chapter contains all the information necessary to integrate SpeechAttendant with a **standard analog or digital environment**.

For any other type of integration, refer to the relevant chapter:

Integration type	Reference
Meridian ACD queue	Chapter 4, page 28
Avaya ACD queue	Chapter 4, page 33
Nortel Networks BCM	Chapter 5, page 39
Multi-server application	Chapter 6, page 57

Introduction

For the PBX/KSU to interact with SpeechAttendant, you must:

1. Program the PBX/KSU (see below).
2. Test the SpeechAttendant ports (see page 18).
3. Install the ATAs (if this is a Norstar integration; see page 19).
4. Connect SpeechAttendant to the PBX/KSU (see page 20).

The requirements for each of these steps are described in the following pages. For details about programming the PBX/KSU, refer to the manufacturer’s documentation.

It is recommended to test the application internally for a few weeks before activating it on the company’s external line.

Programming the PBX/KSU

For SpeechAttendant to be eventually integrated with the PBX/KSU, you must program:

1. The hunting method (see page 15)
2. A phantom number (only in digital environments, see page 15)
3. The SpeechAttendant port attributes (see page 16)
4. The class of service for each port (see page 17)
5. The modem connection (see page 17)
6. The fallback plan, in the event of a system failure or if all SpeechAttendant ports are busy (see page 18)

Hunting method

You must assign extensions to SpeechAttendant in the PBX/KSU (one for each licensed port) and determine the lead number. You must also determine the sequence in which calls will be distributed to those extensions.

The hunting method can be:

- Sequential, for all calls to be directed to the first extension in the group, moving on to the next one only when the first one is busy.

  In this case, the first SpeechAttendant port should receive more calls then any other, while the last one should only be the "last resort."

- Rotary, for the first incoming call to be directed to the first extension in the group, the second call, to the second extension, and so on, going back to the first extension after the last one has received a call.

  This way, all SpeechAttendant ports receive approximately the same amount of calls.

In digital Meridian and Avaya environments, SpeechAttendant ports can also be programmed as ACD agents (see Chapter 4, page 27).

Phantom number

In the case of a digital integration, you must program a virtual or "phantom" number in the PBX, and forward it directly to the SpeechAttendant lead number. This is necessary to allow DNIS-based call routing (see page 125). Ideally, the phantom number should be a DID number to allow for both internal and external call testing. If SpeechAttendant must answer several phone numbers, it is recommended to program a phantom number for each entry point.

The procedure to follow varies depending on the PBX model; for example, the Meridian requires a phantom ACD queue with "night call forward" (NCFW) to SpeechAttendant. Please refer to the manufacturer's documentation for details.

Port attributes

The attributes of SpeechAttendant ports vary depending on the type of integration.

Analog integration

Ports must be programmed with the following attributes:

Attribute	Purpose
Hookswitch transfer	For SpeechAttendant to be able to transfer calls to the PBX/KSU.
DTMF	For SpeechAttendant ports to emit DTMF tones, not rotary tones.
Tone disconnect	For SpeechAttendant to release ports on hang-up.
Dial tone	For testing or troubleshooting.

Avaya digital integration

Ports must be programmed as Model 8434DX phone sets, with default settings.

SpeechAttendant is compatible with 2-wire Avaya line cards. Integration with 4-wire line cards has not been tested.

Meridian digital integration

Ports must be programmed as 2616 phone sets, as indicated below:

1 Define key **0** as the primary DN.

2 Define key **15** as the transfer key (with the **TRN** attribute)

3 At the **CLS** prompt, enter the following command:

- **ADD**, to allow digit display
- **HFD**, for hands free to be denied
- **CNDA**, for caller names to be displayed
- **HTA**, to allow hunting

Mitel digital integration

Ports must be programmed as Superset 430 phone sets, with default settings.

NEC digital integration

Ports must be programmed as DTermIII phone sets, with the following attributes:

- Primary line (extension number) on key 16
- TRF (transfer) on key 94
- SPKR (speaker) on key 96
- Automatic Idle Return, for immediate disconnect when caller hangs up

The following line cards have been successfully tested with SpeechAttendant:

Revision	Firmware	DSE version and hotfix
16ELCJ-BB	SP3514-3A	DSE 3.1 or later
16ELCJ-BC	SP3655-1A	DSE 4.0 HF5
16ELCJ-BD	SP3656-3A	DSE 4.0 HF5
16ELCJ-BE	SP3752-2A	DSE 4.0 HF5

Other line cards may not be supported.

Class of service

Depending on the types of calls the system must be able to route (internal, local, long distance, and/or overseas), you must establish a class of service (that is, the routing restrictions or trunk access) for each SpeechAttendant port.

Modem

For remote support and monitoring purpose, each SpeechAttendant server requires a modem line. Since ScanSoft technicians must be able to access the SpeechAttendant system even if the PBX/KSU is not functioning, an **analog** line is required. Terminate the line in a RJ11 jack.

The modem phone number should have been specified by the customer in the Signature Pages of the *Site Preparation Document*.

Fallback plan

In accordance with the customer preferences, you must program a fallback plan in the PBX/KSU for each of the following situations:

Condition	Proposed fallback plan
Busy (all the SpeechAttendant ports are busy)	Callers may: ◆ Hear a busy signal ◆ Be placed on hold ◆ Be transferred to a live operator, auto-attendant or a voice mail system
Ring/No answer (calls are not answered due to a system failure)	Calls may be transferred to a live operator, auto-attendant or voice mail system.

Testing the PBX/KSU

Before connecting SpeechAttendant to the PBX/KSU, it is a good idea to perform a few tests (especially if you did not install the PBX/KSU yourself).

You will need the following equipment to test the system:

Digital system	Analog system
Three telephones, one of which must be digital	Up to three analog telephones or Two analog phones and a butt set

Perform the following tests:

- ◆ Check for a dial tone on all ports.
- ◆ See if hang-ups are detected.
- ◆ Check the noise level on each port (too much noise can trigger the barge-in function, seriously impacting the system's performance).
- ◆ Test call transferring and hunting.

Installing the ATAs

SpeechAttendant systems that must be integrated into Norstar[1] environments require special adapters. ScanSoft supplies those ATAs, but you must make sure that there are enough outlets to connect them.

You can connect ATAs to a power bar, but keep in mind that each ATA takes up the space of two outlets on a regular power bar. There are power bars especially designed for ATAs.

To install an ATA:

1. Plug the ATA into an electrical outlet.
2. Plug the line from the Norstar KSU into the **wall** jack of the ATA.
3. Plug the digital phone line into the **phone** jack of the ATA.
4. Plug the analog telephone into the **analog device** jack of the ATA.

   The LED (power indicator) on the front of the ATA box flashes quickly while the ATA synchronizes with the Norstar KSU. The LED then turns solid, indicating that it is in **shared** mode.
5. If the LED does not stop flashing, connect a 7310 or 7308 phone set to the ATA and make sure that it is programmed as follows:
   - **Auto Set Relocate** must be **No**.
   - **Handsfree** must be **Off**.
   - **Prime Line** must be **Intercom**.
   - Number of intercom keys must be **2**.

1. Norstar software MICS (version 3.0 and up) and DR5 are compatible with SpeechAttendant. Older versions are not supported.

Then, without picking up the handset, dial **#*#00** on the digital phone connected to the ATA. The LED should stop flashing and remain lit.

6 Without picking up the handset, dial **#*#91** on the digital phone connected to the ATA.

Flash to Feature Y appears on the display of the digital phone, confirming that both **shared** and **Liaison** modes are active.

Repeat steps this procedure for each ATA to be installed.

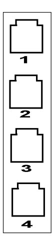
Connecting the PBX/KSU

After you have programmed the PBX/KSU and tested the ports assigned to SpeechAttendant, you are ready to connect the systems to each other (remembering the modem).

Analog integration

In the case of an analog integration, the RJ11 cables provided with SpeechAttendant are connected directly to Dialogic cards.

2, 4, 6, and 8 ports



D/41 card

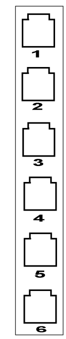
These systems are shipped with one or two D/41 cards. Proceed as follows:

1 Plug a cable into each jack of the D/41 card (line 1 to jack 1, cable 2 to jack 2, and so on).

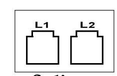
2 Plug the other ends of the cables into the wall jacks.

10 ports and more

These systems are shipped with one, two or three D/120 cards and splitters. Even if they only have six jacks, D/120 cards can accommodate up to twelve lines. Proceed as follows:



D/120 card



Splitter

If the wall jacks are RJ11:

1 Connect a splitter to each jack of the D/120 card, and plug two cables to each splitter.

2 Plug the other ends of the cables into the wall jacks.

If the wall jacks are RJ14, splitters are not necessary, since each of these jacks can accommodate two lines:

1 Connect one cable in each jack of the D/120 card.

2 Plug the other ends of the cables into the wall jacks.

Avaya, Meridian or Mitel digital integration

New SpeechAttendant systems[1] that must be integrated with a digital Avaya, Meridian or Mitel PBX are equipped with D/42 JCT-U or D/82 JCT-U universal boards. A D/42 board can accommodate up to 4 ports, while a D/82 board can accommodate up to 8 ports. Systems with more than 8 ports are delivered with two D/82 boards (or more), or a combination of D/42 and D/82 boards.

For each D/42 or D/82 board, you should receive a 25-pair cable with a **male** amphenol connector and a breakout box.

You can use one of the following methods to connect to the PBX:

- Amphenol to amphenol (see below)
- With the breakout box (mini-patch panel; see page 23)

Connecting amphenol to amphenol

This method is recommended for systems with eight ports or more.

You need to provide a 25-pair cable with a **female** amphenol connector (RJ21X) for each D/42 or D/82 board.

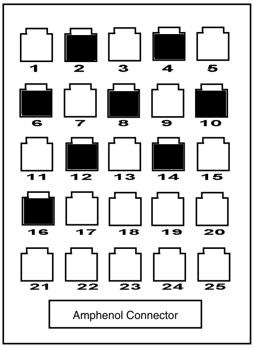
1. Connect the 25-pair cable provided with SpeechAttendant to the D/42 or D/82 board.
2. Connect the male amphenol connector of that cable with your female amphenol connector.
3. Insert the cable's pairs into the bix connector, as indicated below:

Pair > Line	Pair > Line
2 > 1	10 > 5
4 > 2	12 > 6
6 > 3	14 > 7
8 > 4	16 > 8

1. DSE integration boards are still supported on old systems upgraded to version 8.2.

4 If SpeechAttendant has more than 8 ports, connect a second 25-pair cable to a second D/42 or D/82 board, connect amphenols together, then connect pairs 2, 4, 6, 8, etc. to lines 9, 10, 11, 12, etc. on the same bix connector.

Connecting with a breakout box



1 Connect the 25-pair cable provided with SpeechAttendant to the D/42 or D/82 board.

2 Connect the amphenol connector of that cable to the breakout box.

3 Plug the RJ11 cables provided with SpeechAttendant into the jacks of the breakout box (line 1 in jack 2, line 2 in jack 4, line 3 in jack 6, and so on).

4 Plug the other end of each RJ11 cable into a wall jack.

Even if the breakout box has 25 jacks, you cannot plug more lines to it than the number of ports supported by the corresponding board (i.e., 4, for an D/42 board, or 8 for a D/82 board).

NEC digital integration

In the case of a NEC digital integration, SpeechAttendant is delivered with a DSE DL3009 integration board, a 25-pair cable with a **female** amphenol connector, and a breakout box. If the SpeechAttendant system has more than 16 ports, it will be delivered with two DSE boards, two 25-pair cables, and two breakout boxes.

You can use one of the following methods to connect to the PBX:

- Amphenol to amphenol (see below)
- With the breakout box (mini-patch panel; see page 25)

Connecting amphenol to amphenol

This method is recommended for systems with eight ports or more.

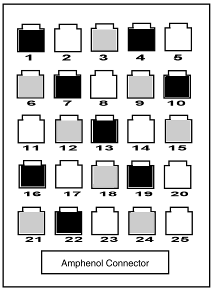
You need to provide a 25-pair cable with a **male** amphenol connector (RJ21X) for each DSE integration board.

1. Connect the 25-pair cable provided with SpeechAttendant to the DSE integration board.
2. Connect the female amphenol connector of that cable with your male amphenol connector.
3. Insert the cable's pairs into the bix connector, as indicated below:

Pair > Line	Pair > Line
1 > 1	3 > 9
4 > 2	6 > 10
7 > 3	9 > 11
10 > 4	12 > 12
13 > 5	15 > 13
16 > 6	18 > 14
19 > 7	21 > 15
22 > 8	24 > 16

4 If SpeechAttendant has more than 16 ports, connect the second 25-pair cable provided with the system to the second DSE integration board, connect the amphenols together, then plug the remaining lines to a second bix connector as indicated in the previous table.

Connecting with a breakout box



1 Connect the 25-pair cable provided with SpeechAttendant to the DSE integration board.

2 Connect the amphenol connector of that cable to the breakout box.

3 Plug the RJ11 cables provided with SpeechAttendant into the jacks of the breakout box as indicated below:

Line > Jack	Line > Jack
1 > 1	9 > 3
2 > 4	10 > 6
3 > 7	11 > 9
4 > 10	12 > 12
5 > 13	13 > 15
6 > 16	14 > 18
7 > 19	15 > 21
8 > 22	16 > 24

4 Plug the other end of each RJ11 cable into a wall jack.

4

ACD Environment

In Meridian and Avaya digital environments, SpeechAttendant ports can be programmed as ACD agents (as opposed to a hunt group; see page 15). This allows the customer to benefit from the advantages of speech-recognition and automatic call distribution, even if SpeechAttendant is not actually installed in a call center.

This chapter contains indications on how to integrate SpeechAttendant in an ACD environment. It also explains how to replace some of the default voice files with other ones that are more suitable for call centers.

Meridian

This section explains how to integrate SpeechAttendant in a digital Meridian ACD environment. It assumes that you are familiar with programming a Meridian 1 PBX. Refer to the Nortel Networks documentation for more information.

Introduction

Integrating SpeechAttendant in a Meridian ACD environment involves the following steps:

In the PBX

1. Identifying the ACD software package included with the PBX (see below)
2. Programming the ACD queue (see page 30)
3. Programming the ports as ACD agents (see page 31)

In the SpeechAttendant configuration panel

4. Configuring the settings related to automatic call distribution (see page 32)

Identifying the ACD software

Meridian 1 PBXs may be shipped with basic ACD, ACD-A, ACD-B, ACD-C, ACD-D, EOVF, and NACD packages.

As the method for programming the ACD queue can vary accordingly, you must first determine which ACD package is installed.

To do so:

1. Load overlay program 22.
2. At the **REQ** prompt, enter `PRT`, then at the **TYPE** prompt, enter `ISS`.

   This will tell you the release and issue of the software package.
3. At the **REQ** prompt, enter `PRT`, then at the **TYPE** prompt, enter `PKG`.

   This will tell you which package is installed.
4. At the **REQ** prompt, enter `PRT`, then at the **TYPE** prompt, enter `SLT`.

   This will tell you the number of licenses purchased for TNs, ACD, DChannel, etc.
5. Load overlay program 23.
6. At the **REQ** prompt, enter `PRT`, then at the **TYPE** prompt, enter `SCB`.

   If you get an SCH0721 code, you know that the “Position ID” mode is used.

   If the “Agent ID” mode is used, you get the lower and upper boundaries for agent IDs (see “Nortel-M1 Agent ID Mode” and “Agent ID -Port *x*” on page 32).

   For the “Agent ID” mode to be available, the PBX must be equipped with the ACD-C or ACD-D package.

Programming the ACD queue

Programming an ACD queue is done in overlay program 23. Enter the responses indicated for the prompts listed below (certain prompts may not appear depending on the package installed). Leave the default value for prompts that do not appear in the list (simply press **Enter** to skip over them).

Prompt	Response
REQ	NEW
TYPE	ACD
ACDN	*Enter up to 4 digits for the ACD queue directory number, for example, 5555.*
MAXP	*Enter the number of ports.*
NCFW	*Enter the directory number that calls should be transferred to when all the ports are logged off or in case of a system failure.*
RTQT	*Specify a number of rings (3 to 5 is recommended) after which ports should be automatically logged off.*
RTQO	MSB
OVTH	*Enter the number of calls waiting that will trigger overflow.*
TOFT	*Enter a number from 2 to 1800. This is the time, in seconds, that the call waiting in queue will go to overflow.*
OVDN	*Enter the overflow directory number.*
IFDN	Enter the interflow directory number.
AENI	YES
HOML	NO, *otherwise the SpeechAttendant port will log off after each hang-up.* *Note that the HOML prompt only appears if a schedule block (SCB) exists.*
RPRT	YES, *if customer wants to capture statistics on Meridian MAX; otherwise,* NO.

Programming ports

Once you have programmed the ACD queue, you are ready to program the ports as ACD agents.

You must program the ports in overlay program 11. Enter the responses indicated for the prompts listed below. Leave the default value for prompts that do not appear in the list (simply press **Enter** to skip over them).

Prompt	Response
TYPE	2616
CUST	0
TGAR	*Enter the trunk group access restriction.*
NCOS	*Enter the NCOS, if required.*
CLS	WTD HPR MTD ADD HFD CPFD CPTD CNDD CNIA DNDD **NOTE** Keep the default values, but make sure that the above values are included among them. Add them if they are not.
SPID	NONE
KEY 00	ACD + *the ACD queue directory number* + 0 + *the agent ID*
KEY 01	NRD
KEY 02	MSB
KEY 12	A06
KEY 15	TRN

Setting ACD parameters in the configuration panel

Once you have programmed the ports, you must configure ACD settings in the configuration panel (about other configuration settings, see Chapter 7, page 73).

1 Access level 2 of the SpeechAttendant configuration panel.

2 Set **ACD Enabled** to **Yes**.

3 If the login mode is "Agent ID," set **Nortel-M1 Agent ID Mode** to **Yes**. Otherwise, set this parameter to **No** (this indicates that the login mode is "Position ID").

   To determine which login mode is used, see page 29.

4 If applicable, use the **Agent ID - Port** *x* settings to specify the agent ID for each port. If the Multiple Queue Assignment feature is used, agent IDs must be followed by ##.

The following settings are also related to ACD queues, but don't need to be adjusted now:

- **Agent Logged Out** contains the character string that appears on the phone display when an agent is logged out.
- **Required Action When Port Idle** is accessible from level 1 or 2. It can be used to log out idle ports so that the application can be stopped in an orderly fashion. It also enables the systematic logging in of ports to restart the application or ensure continuous service. Possible values are:
  - **Logout**, for idle ports to be logged out within the next 100 seconds, and busy ports to be logged out as soon as ongoing calls end. When all the ports are logged out, you can stop the telephony application (from the **Admin Tools** window) without interrupting calls.
  - **Login**, to enable the systematic logging in of ports that the PBX may have logged out, or for the system to log in ports that were logged out to stop the system (ports will be logged in within the next 100 seconds).

Avaya

This section explains how to integrate SpeechAttendant in a digital Avaya ACD environment with expert agent selection (EAS) enabled[1]. It assumes that you are familiar with programming an Avaya Definity PBX. Refer to the Avaya documentation for more information.

Introduction

Integrating SpeechAttendant in an Avaya ACD environment involves the following steps:

In the PBX

1. Creating a hunt group (see below)
2. Programming agents (see page 34)
3. Programming stations (see page 34)
4. Programming a call vector (see page 35)
5. Programming a vector directory number (see page 35)

In the SpeechAttendant configuration panel

6. Configuring the settings related to automatic call distribution (see page 36)

Creating a hunt group

You must create a hunt group for SpeechAttendant. To do so:

1. In the **ACD**, **Queue**, and **Vector** fields, type `y`.
2. In the **Group type** field, type `ead-mia`.
3. In the **Skill** field, type `y`.
4. Set **Timed ACW interval** to `1` second.

1. SpeechAttendant is compatible with 2-wire Avaya line cards. Integration with 4-wire line cards has not been tested.

5 Set **Redirect on no answer** to `2` rings, and enter the appropriate destination in the **Redirect to VDN** field (for example, the operator extension).

Programming agents

You must program an agent for each port dedicated to SpeechAttendant. To do so:

1 Specify the agent name and login ID (to keep things simple, enter the same number for both).

2 In the **TN** field, type `1` as the tenant number.

3 In the **COR** field, type `1` as the class of restriction.

4 Enter the agent password twice (to keep things simple, use the same number as the agent's name and login ID).

5 In the **Auto Answer** field, type `none`.

6 In the **SN** field, type the hunt group number.

7 In the **SL** field, type `1` as the skill level.

Programming stations

Each port assigned to SpeechAttendant must be programmed as Model 8434D stations, with default settings. You must then program each station buttons as indicated below:

Button	Assignment	Additional data
6	`auto-in`	N/A (Ignore **Grp**)
7	`aux-work`	N/A (Ignore **RC** and **Grp**)
8	`autodial`	Enter the ACD logout feature access code followed by #.
9	`autodial`	Enter the ACD login feature access code.

Programming a call vector

Program a vector as indicated below:

Line	Command	Additional data
01	`wait-time`	`0` **seconds hearing** `ringback`
02	`queue-to`	**Skill** *`SN`* **pri** `m` Where *SN* should be replaced with the actual hunt group number.
03	`stop`	

If desired, you can configure a vector for each SpeechAttendant entry point. This allows you for more flexibility in call routing on no answer and best service routing (BSR). For example, you could program different call destinations on no answer condition.

Programming a vector directory number (VDN)

You must now program a VDN to point to the previously programmed call vector. If you programmed several call vectors for several entry points, you must also program several VDNs.

To allow for proper DNIS capture, the VDN name must include the VDN number as a prefix (e.g., `24500SpeechAttendant`).

In the **Measured** field, type **internal**. This allows you to keep track of calls to the VDN using the basic call management system.

Setting ACD parameters in the configuration panel

Once you have programmed the ports, you must configure ACD settings in the configuration panel (about other configuration settings, see Chapter 7, page 73).

1 Access level 2 of the SpeechAttendant configuration panel.

2 Set **ACD Enabled** to **Yes**.

3 Set the **Agent ID - Port** *x* settings for each port by entering the login ID and password of the corresponding agent (without spaces).

The following settings also pertain to ACD queues, but don't need to be adjusted now:

- **Agent Logged Out** contains the character string that appears on the phone display when an agent is logged out.
- **Required Action When Port Idle** is accessible from level 1 or 2. It can be used to log out idle ports so that the application can be stopped in an orderly fashion. It also enables the systematic logging in of ports to restart the application or ensure continuous service. Possible values are:
  - **Logout**, for idle ports to be logged out within the next 100 seconds, and busy ports to be logged out as soon as ongoing calls end. When all the ports are logged out, you can stop the telephony application (from the **Admin Tools** window) without interrupting calls.
  - **Login**, to enable the systematic logging in of ports that the PBX may have logged out, or for the system to log in ports that were logged out to stop the system (ports will be logged in within the next 100 seconds).

The **Nortel-M1 Agent ID Mode** parameter is not relevant for Avaya ACD.

Voice files for call centers

Some of the default voice files in the factory profiles are not suitable for call center environments.

However, you can replace them with other voice files supplied by ScanSoft. To do so:

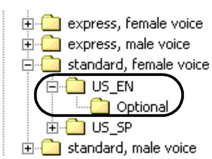
1 Locate the voice profile for your application in `C:\SpeechAttendant\data\voice\system\factory\` *`voice_profile`*`\`*`language`*.

Where *`voice_profile`* corresponds to voice profile used (for example, `standard, female voice`) and *`language`*, to the system language (for example, `US_EN`).

2 In the **Optional** folder, copy the call center vox files[1], then paste them in the same folder.

3 Rename each copy by deleting `CALL_CENTER` and `Copy of` (for example, so that `Copy of PARDONME_2_1_CALL CENTER.vox` becomes `PARDONME_2_1.vox`).

4 Copy the renamed files in the parent folder, for example, **US_EN**.



5 Repeat this procedure for all express and standard voice profiles used in the application, in all system languages.

1. `LOUDER2_1_CALL_CENTER.vox`, `LOUDER2_2_CALL_CENTER.vox`, `PARDONME2_1_CALL_CENTER.vox`, and `PARDONME2_2_CALL_CENTER.vox`.

5

Business Communications Manager (BCM)

SpeechAttendant can be integrated to Nortel Networks BCM. This chapter contains all the necessary information to do so.

Introduction

Integration to Nortel BCM requires a Dialogic voice card, analog lines to BCM, and LAN connectivity to BCM. The whole process involves the following steps:

1 Gathering the information needed and connecting SpeechAttendant to its dedicated analog lines (see below)

2 Downloading and installing Unified Manager Client on the BCM server (see page 43)

3 Checking the configuration of the directory numbers dedicated to SpeechAttendant (see page 46)

4 Programming a hunt group for SpeechAttendant (see page 49)

5 Configuring LAN CTE (see page 51)

6 Downloading and installing LAN CTE Client on the SpeechAttendant server (see page 53)

PBX, voice card and transfer mode

SpeechAttendant must be equipped with a Dialogic voice card such as D/41 JCT-LS. When activating the system's license (see page 11), verify that the PBX and telephony card selected in the **Config Setup** windows are correct (respectively BCM_TAPI and D/41JCT-LS), and set **Default Transfer Mode** to **Unsupervised**.

Analog lines

To enable BCM connection, you must provide one analog line per SpeechAttendant port.

The analog lines may come either from an internal ATA line card or from Nortel ATA-2 adapters. The internal ATA line card is installed in the BCM chassis and provides 8 analog lines that can be connected to the Dialogic card. The ATA-2 adapters are small modem-like boxes with 2 RJ-11 connectors. One of them connects to a digital line card in the BCM and the other one provides an analog line that can be connected to the Dialogic card.

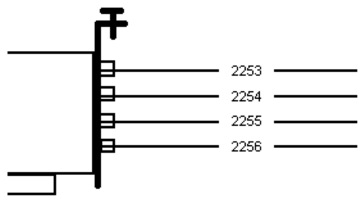
Information needed

The table provided on page 55 will help you gather this information.

Directory numbers

Obtain the directory number of each analog line dedicated to SpeechAttendant. Test each line by connecting them to an analog phone and calling their directory number. Label the lines with their directory number.

Connect the lines to the SpeechAttendant from the lowest to the highest directory number, starting from the top of the Dialogic card.



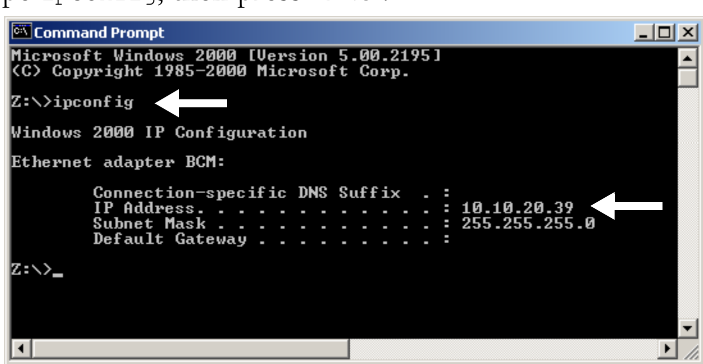
IP addresses

You need to get the IP address of the SpeechAttendant and BCM servers. You must also enable SpeechAttendant to ping BCM.

To get the IP address of the SpeechAttendant server:

1 Open the **Command Prompt** window (click **Start** > **Programs** > **Accessories** > **Command Prompt**).

2 Type `ipconfig`, then press **Enter**.

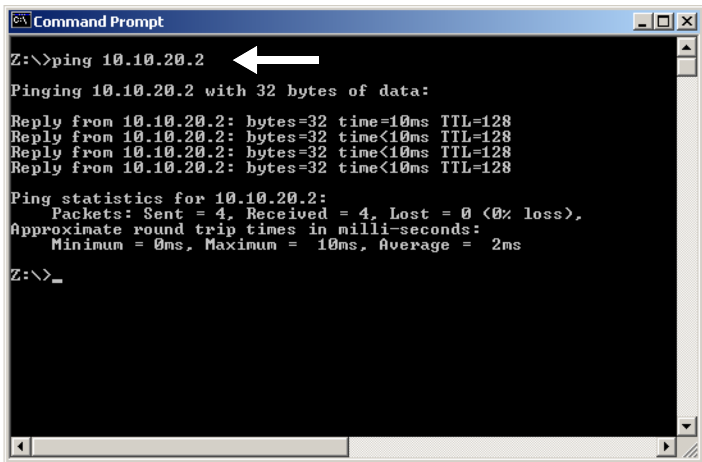


To get the IP address of the BCM server:

◆ Ask the person who installed BCM.

To enable ping:

1. Open the **Command Prompt** window.
2. Type `ping` followed by the IP address of the BCM server, then press **Enter**.



BCM user ID and password

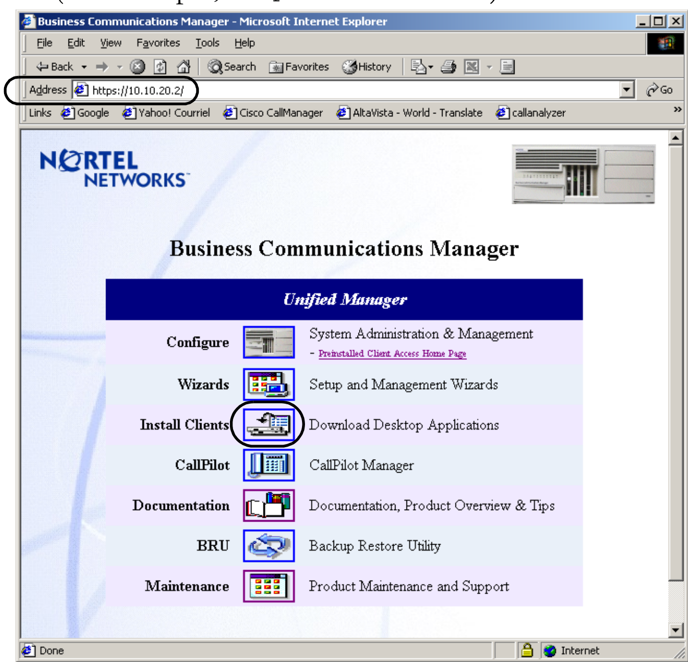
To be able to customize BCM configuration, you need a user ID and password. Get this information from the person who installed the BCM server.

Downloading and installing Unified Manager Client

Unified Manager Client accelerates access to Unified Manager. It must be installed on the computer used to manage BCM.

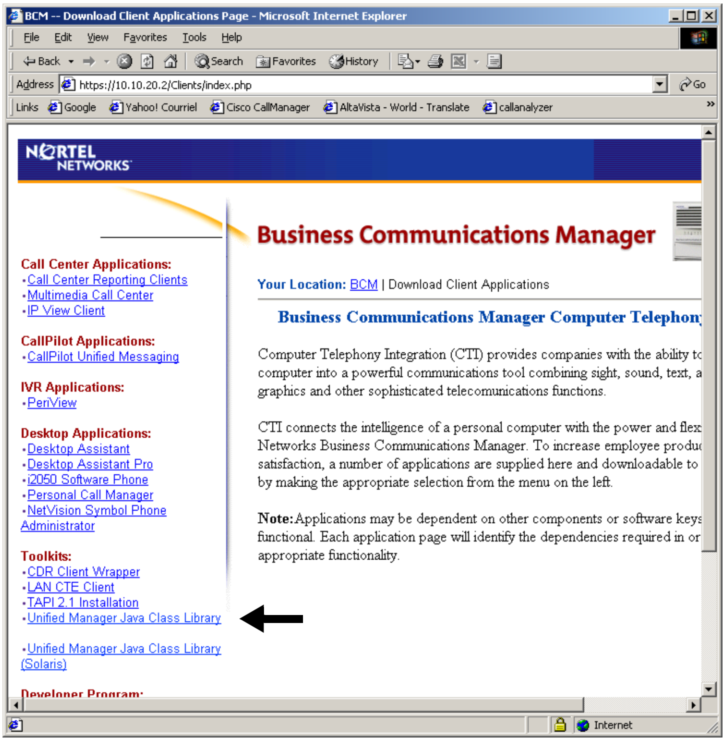
To install Unified Manager Client:

1 Access the BCM Web site using the IP address of the BCM server (for example, `https://10.10.20.2`).

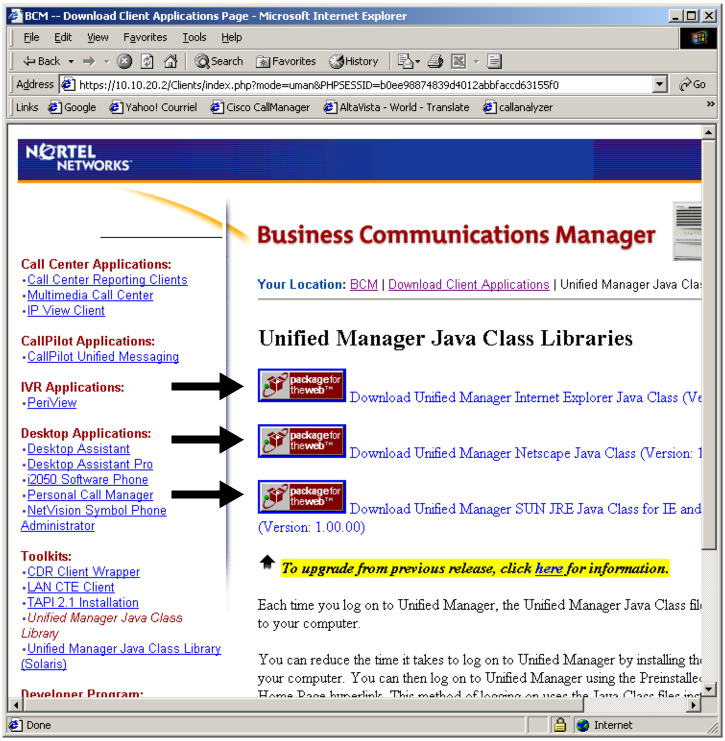


2 Click **Install Clients**.

3 Click **Unified Manager Java Class Library.**



4 Click the **Package for the Web** corresponding to your browser.



5 In the **File Download** window, select **Save this program to disk**, then click **OK**.

6 Specify the desired location for the Unified Manager Client file, then click **Save**.

7 Using Windows Explorer, locate the downloaded file, then double-click it to begin the installation.

8 When the installation is complete, click **Finish**.

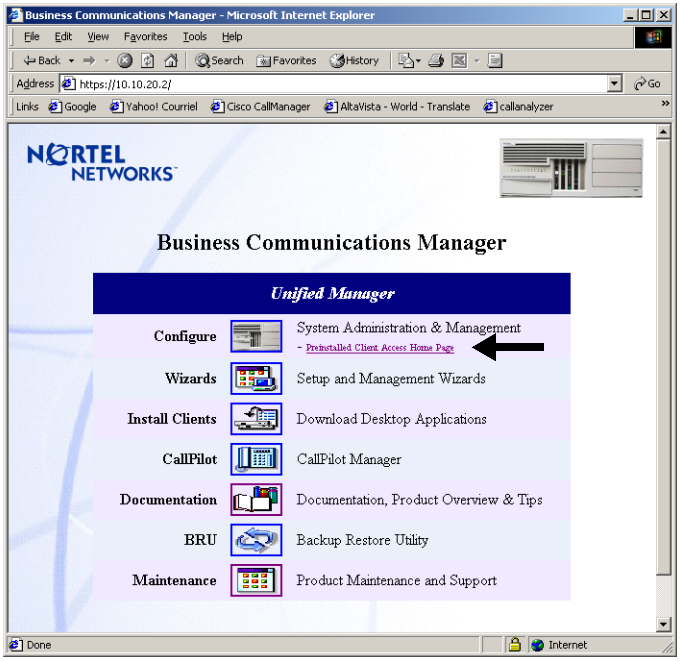
Validating the directory numbers

You must check the configuration of the directory numbers that will handled by SpeechAttendant.

To validate the directory numbers:

1 Access the BCM Web site.

2 Click **Preinstalled Client Access Home Page**.

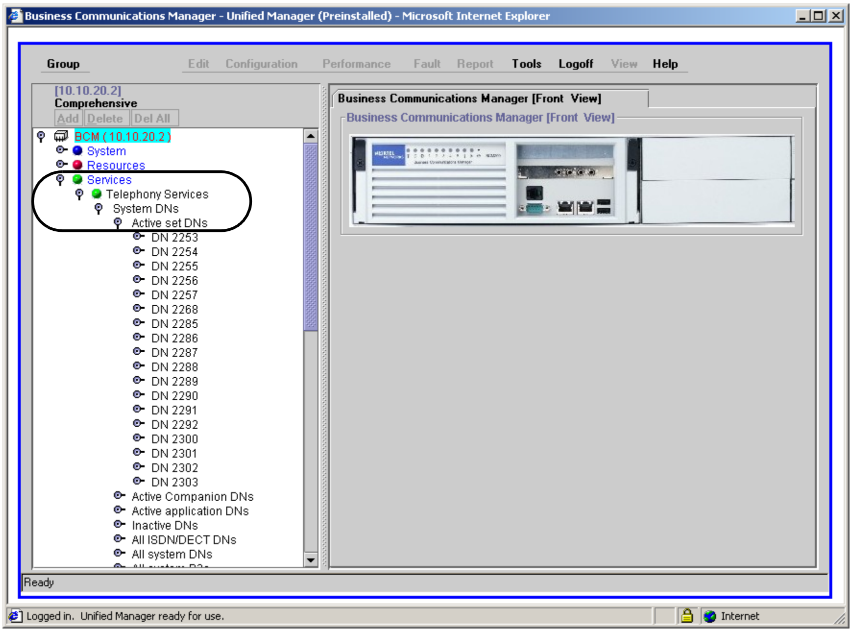


3 Click **Configure**.

4 Enter the user ID and password, then click **Login**.

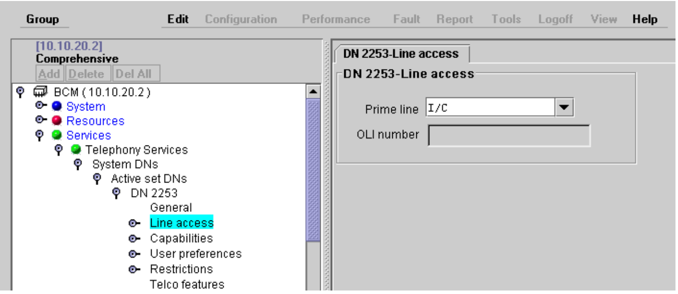
You can get this information from the person who installed BCM.

5 Select **Services** > **Telephony Services** > **System DNs** > **Active Set DNs**.

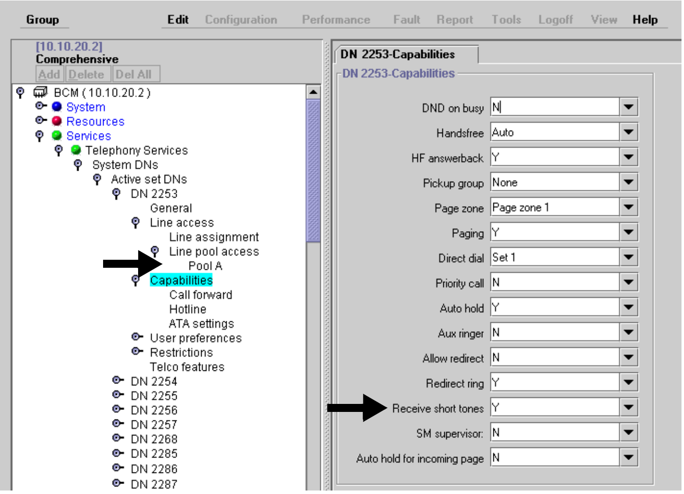


6 For each DN dedicated to SpeechAttendant, verify that:

- **Prime line** is set to **I/C**.



- **Receive short tones** is set to **Y**.
- The pool selected allows external transfers (if SpeechAttendant must be able to transfer calls to external numbers[1]).



1. The lines connected to SpeechAttendant must belong to the same pool.

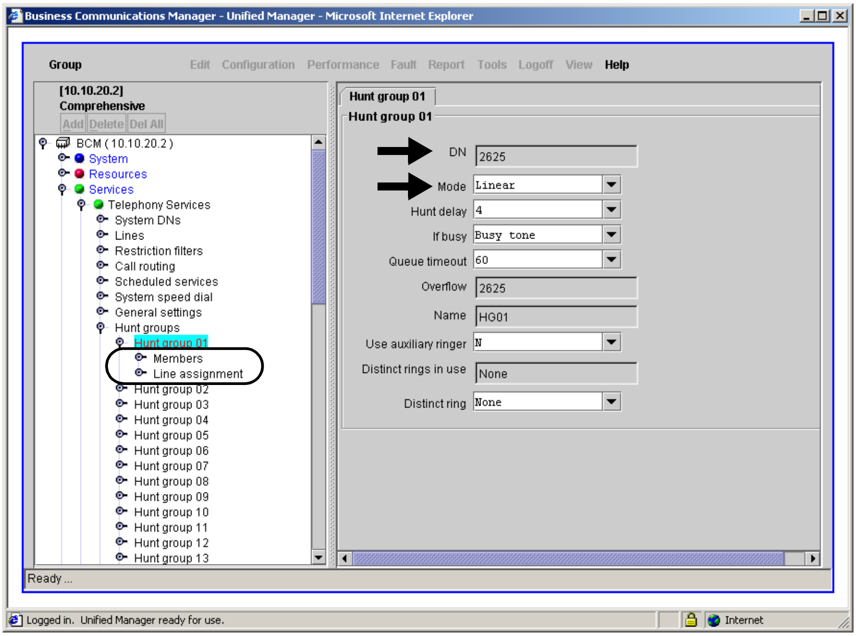
Defining a hunt group for SpeechAttendant

You must define a linear hunt group with the directory numbers and lines dedicated to SpeechAttendant.

To configure the hunt group for SpeechAttendant:

1 Select **Services** > **Telephony Services** > **Hunt groups**.

2 Click an available hunt group.



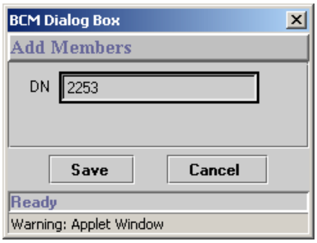
The **DN** field contains the directory number that internal callers will use to access SpeechAttendant. As a reference, you should enter this number in the table on page 55.

3 Verify that the **Mode** field is set to **Linear**.

4 Click **Members**.

5 For each directory number dedicated to SpeechAttendant:

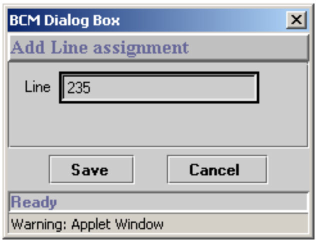
- Click **Add**.
- Enter the directory number, then click **Save**.



6 Click **Line assignment**.

7 For each line that will be handled by SpeechAttendant:

- Click **Add**.
- Enter the line number, then click **Save**.

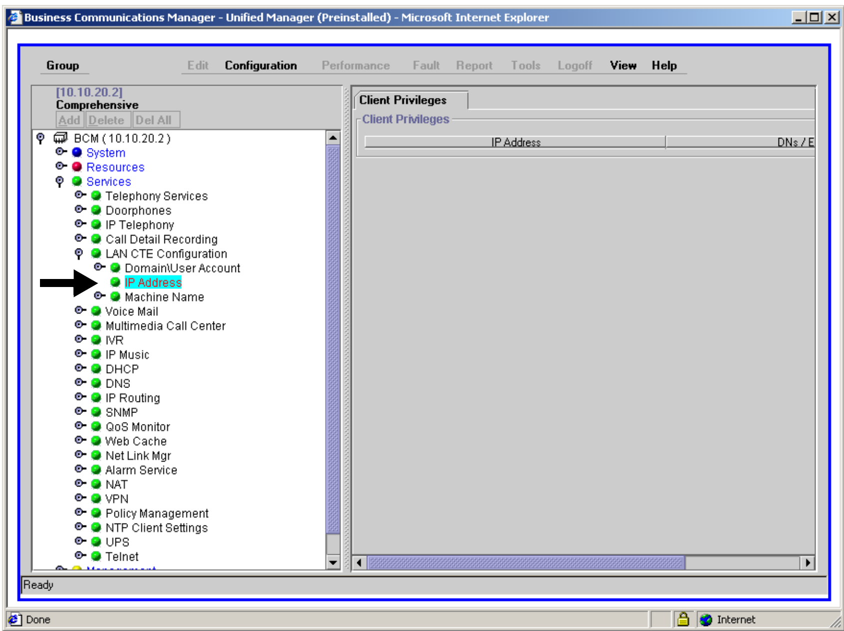


Configuring LAN CTE

In BCM environments, call control is achieved through a LAN connection using LAN CTE. You must therefore specify which directory numbers are dedicated to SpeechAttendant in LAN CTE.

To do so:

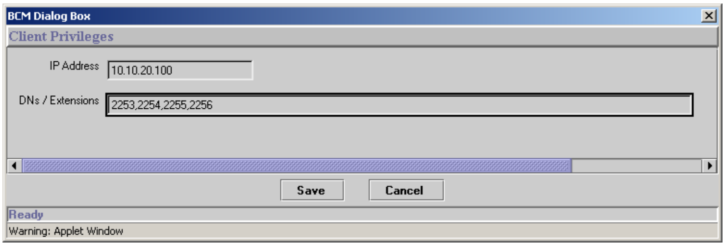
1 Select **Services** > **LAN CTE Configuration** > **IP Address**.



2 In the menu bar, select **Configuration** > **Add Entry to Privileges List**.

3 In the **IP Address** field enter the IP address of the SpeechAttendant server.

4 In the **DNs / Extensions** field enter the directory numbers of the lines connected to SpeechAttendant.



The directory numbers must be separated by a comma with no interleaving spaces.

Downloading and installing LAN CTE Client

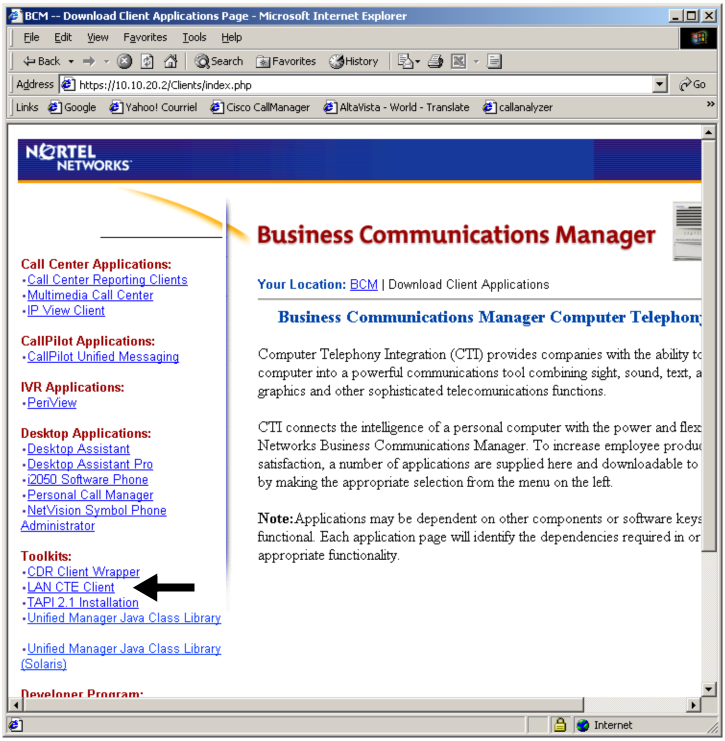
LAN CTE Client must be installed on the SpeechAttendant server.

To do so:

1 Access the BCM Web site from the LocusDialogSTS server.

2 Click **Install Clients**.

3 Click **LAN CTE Client**.



4. Click **Package for the Web**.
5. In the **File Download** window, select **Save this program to disk**, then click **OK**.
6. Specify the desired location for the LAN CTE Client file, then click **Save**.
7. Using Windows Explorer, locate the downloaded file, then double-click it.
8. Accept the terms in the license agreement, then click **Next**.
9. Click **Next** again, then click **Yes**.
10. Click **Yes** again for the Telephony Service Provider to be installed on your system.
11. Enter the IP address of the BCM server, then click **Yes**.
12. Click **Yes** again to confirm the IP address of the BCM server.
13. Select **Yes, I want to restart my computer now**, then click **Finish**.
14. Click **Next**.
15. Click **Yes**.
16. Select **Yes, I want to restart my computer now**, then click **Finish**.

You can now proceed with the implementation of SpeechAttendant.

Records

Fill-out this form and refers to it when integrating SpeechAttendant to BCM.

DN1	
DN2	
DN3	
DN4	
DN5	
DN6	
DN7	
DN8	
SpeechAttendant IP address	
BCM IP address	
BCM user ID	
BCM password	
Hunt group DN†	

† The number that will be used by internal callers to reach SpeechAttendant.

6

Multi-server application

A SpeechAttendant application can consist of two or more computers connected so that they operate together.

Server clustering ensures that calls will be answered without interruption in the event of a system failure. Clustering also allows for call sharing between servers.

This chapter contains all the information pertaining to the implementation of a multi-server environment.

Introduction

The computers to be connected in a cluster must be equipped with:

- Windows 2000 Server operating system, service pack 4
- SQL Server 2000 Standard Edition, service pack 3

If MSDE is installed on those computers, you must un-install it and install SQL Server instead.

The global procedure for configuring a server cluster involves:

1. Connecting the master and slave servers to the network, with write access rights to each other[1]
2. Programming the PBX for a multi-server environment (see page 59)
3. Activating the license on each server (see page 11)
4. Removing the SpeechAttendant database from the slave server (see page 60)
5. Configuring replication (see page 61)
6. Identifying servers and defining port assignment (see page 71)

You can then proceed with the application design and initial load of the directory, as explained in the *Administration Guide*. This can be done on any server: all changes made to the application are automatically reflected on the other server within seconds.

It is then recommended to test PBX integration and SpeechAttendant functions as explained in Chapter 9, page 115.

1. It is also possible to connect the slave server to the master using a network crossover cable.

Programming the PBX

In multi-server applications, the PBX ports can be programmed as ACD agents[1] or a hunt group. The ACD method is recommended, even when the system is not actually installed in a call center.

If you choose the ACD queue method:

1. Program all the ports on both servers as ACD agents on a single queue (see Chapter 4, page 27).
2. Determine how calls should be distributed (for example, in certain types of PBXs, incoming calls could be presented to the longest idle agent—port, in this case).
3. If the PBX uses agent ID mode, enter the ID of each agent in the configuration panel of either the master or the slave server, depending on the case[2] (see page 32).
4. If the PBX uses position ID mode, you must change the **Nortel-M1 Agent ID Mode** setting in the configuration panel of both the master and the slave servers (see page 32).

If you choose the hunt group method:

- Connect the PBX ports to each server so that they are interlaced.

  For example, connect PBX port 1 to port 1 on the master server, PBX port 2 to port 1 on the slave server, PBX port 3 to port 2 on the slave server, and so on.

  The hunting method can be either sequential or rotary. See page 15 for details.

1. Only with Nortel Meridian 1 and Avaya Definity.
2. For example, in a 64-port application, the ID numbers of the first 32 agents must be specified in the master server's configuration panel, and those of the last 32 agents in the slave server.

Removing the database from the slave server

Once you have installed the companion programs, OpenSpeech Recognizer, Speechify TTS (if included in the license) and SpeechAttendant core software on the slave server, you must remove the SpeechAttendant database from this server.

To do so:

1. Make sure that you are on the **slave** server.
2. Stop the telephony application from the **Admin Tools** window.
3. Open the **Command Prompt** window (click **Start** > **Programs** > **Accessories** > **Command Prompt**).
4. Type `osql -Usa -P`*`database_password`*, then press Enter.

   Where *`database_password`* must be replaced with the actual password for accessing the SQL Server databases. If you don’t know this password, contact the Customer Service Center.
5. At **1>** command prompt, type `drop database liaison`, then press Enter.
6. At **2>** command prompt, type `go`, then press Enter.
7. At **1>** command prompt, type `drop database sum_sts_db`, then press Enter.
8. At **2>** command prompt, type `go`, then press Enter.
9. At **1>** command prompt, type `quit`, then press Enter.
10. Close the **Command Prompt** window.

Configuring replication

The two main steps of this procedure consist in configuring the publisher (that is, the master server) and configuring the subscriber (the slave server).

Configuring the publisher

This involves the following tasks:

1 Publishing liaison database (see below)

2 Publishing sum_sts_db database (see page 64)

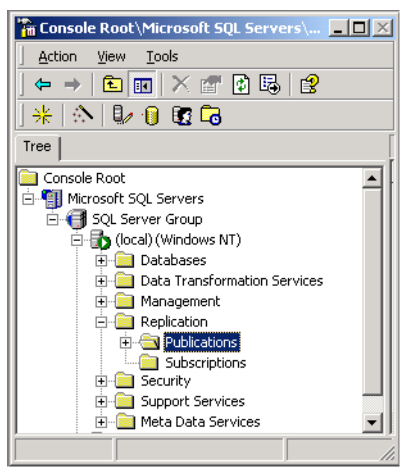
3 Publishing files from the master server (see page 66)

Publishing liaison database

1 Make sure that you are on the **master** server.

2 Click **Start** > **Programs** > **Microsoft SQL Server** > **Enterprise Manager**.

3 In **Console Root\Microsoft SQL Servers\SQL Server Group\(local) (WindowsNT)\Replication**, right-click **Publications**, then select **New Publication**.



The Create Publication Wizard appears.

4 Check **Show advanced options in this wizard**, then click **Next**.

5 When prompted to select a distributor, select **Make** *'name_of_master'* **its own Distributor**, then click **Next**.

Where *name_of_master* stands for the actual name of the master server.

6 Select **Yes, configure the SQL Server Agent service to start automatically**, then click **Next**.

7 Without modifying the proposed snapshot folder (`\\Computer_Name\C$\Program File\Microsoft SQL Server\MSSQL\ReplData`), click **Next**.

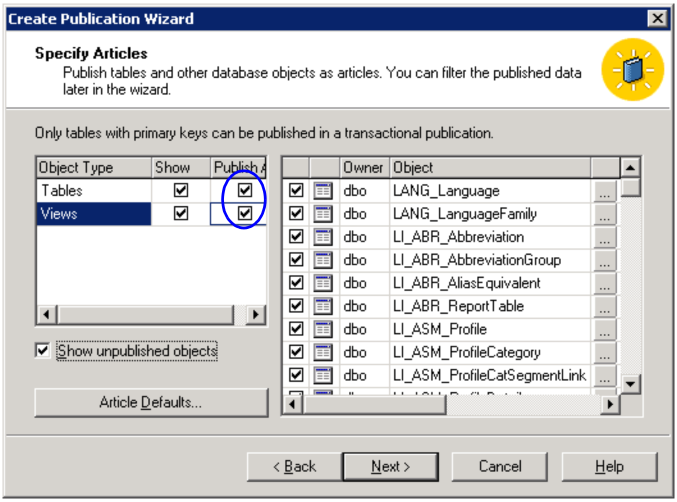
8 Select **liaison** as "the database that contains the data or objects you want to publish," then click **Next**.

9 Select **Transactional publication**, then click **Next**.

10 Check **Queued updating**, then click **Next**.

11 Check **Servers running SQL Server 2000**, then click **Next**.

12 As indicated below, check the boxes for tables and views to be published, then click **Next**.



13 In the **Article Issues** window, click **Next**.

14 Enter `master_liaison` as the publication name, then click **Next**.

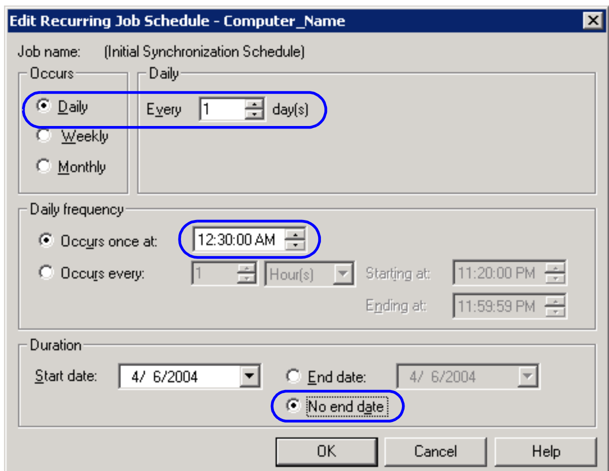
15 Select **Yes, I will define data filters, enable anonymous subscriptions, or customize other properties**, then click **Next**.

16 In the **Filter Data** window, click **Next**, leaving the checkboxes empty.

17 Select **Yes, allow anonymous subscriptions**, then click **Next**.

18 Click **Change** to “set snapshot agent schedule”.

19 For a snapshot (a backup) of the liaison database to be taken every night, at 12:30, set up the schedule as indicated below:



This doesn’t mean that the servers will be synchronized only once a day: with the current clustering method, servers are constantly synchronized.

20 Click **OK**.

21 Back to **Set Snapshot Agent Schedule**, click **Next**.

22 Click **Finish**.

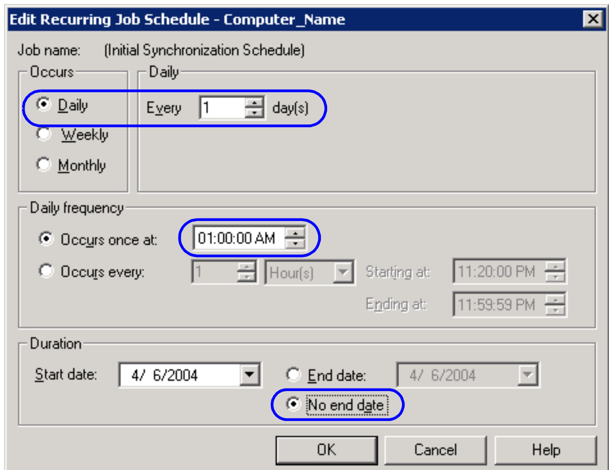
The creation process begins.

23 When informed that publication has been “successfully created,” click **Close**.

Publishing sum_sts_db database

1. In **Console Root\Microsoft SQL Servers\SQL Server Group\(local) (WindowsNT)\Replication**, right-click **Publications**, then select **New Publication**.
2. When the Create Publication Wizard appears, check **Show advanced options in this wizard**, then click **Next**.
3. Select **sum_sts_db** as “the database that contains the data or objects you want to publish,” then click **Next**.
4. Select **Transactional publication**, then click **Next**.
5. Check **Queued updating**, then click **Next**.
6. Check **Servers running SQL Server 2000**, then click **Next**.
7. Check the boxes for tables and views to be published, then click **Next**.
8. In the **Article Issues** window, click **Next**.
9. Enter `master_sum_sts_db` as the publication name, then click **Next**.
10. Select **Yes, I will define data filters, enable anonymous subscriptions, or customize other properties**, then click **Next**.
11. In the **Filter Data** window, click **Next**, leaving the checkboxes empty.
12. Select **Yes, allow anonymous subscriptions**, then click **Next**.
13. Click **Change** to “set snapshot agent schedule”.

14 Schedule the snapshot of the sum_sts_db database to occur once a night, half an hour after the liaison database snapshot, as indicated below:



15 Click **OK**.

16 Back to **Set Snapshot Agent Schedule**, click **Next**.

17 Click **Finish**.

The creation process begins.

18 When informed that publication has been "successfully created," click **Close**, then exit the SQL server enterprise manager.

Publishing files from the master server

Once you have published the liaison and sum_sts_db databases, you must set the directories used for replication on the master server.

1 Make sure that you are on the **master** server.

2 In **C:****SpeechAttendant**, create a **Published** folder.

3 Right-click **Published**, then select **Sharing**.

4 At the **Sharing** tab of the **Published Properties** window, select **Share this folder**, then click **Permissions**.

5 Allow read access to everyone (only read access), then click **OK**.

6 Access level 3 of the SpeechAttendant configuration panel and set the following parameters:

Parameter	Value to enter
Replication/Target_Directory	`$HOME$/Published`
Replication/Source_subscriber_directory	`\\`*`name_of_slave`*`\Published`†
Activate the File Publisher	$SERVER_NAMES@FR_PUBLISHER$
Activate the File Subscriber	$SERVER_NAMES@FR_SUBSCRIBER$
Run-Time : Secondary Database Server	*`name_of_slave`*†

† Where *`name_of_slave`* must be replaced with the actual name of the slave server.

7 Click **Apply**, then exit the configuration panel.

8 In **C:\SpeechAttendant\bin**, double-click **FR_Subscriber Config.exe**.

9 When prompted, enter the user name (locus), domain, and password, then click **OK**.

10 Restart the computer.

Configure the subscriber

This involves the following tasks:

1. Connecting to the slave server from the master (see below)
2. Subscribing to liaison database (see page 68)
3. Subscribing to sum_sts_db database (see page 69)
4. Publishing files from the slave server (see page 70)

Connecting to the slave server from the master

1. Make sure that you are on the **master** server.
2. Click **Start** > **Programs** > **Microsoft SQL Server** > **Enterprise Manager**.
3. In **Console Root\Microsoft SQL Servers**, right-click **SQL Server Group**, then select **New SQL Server Registration**.

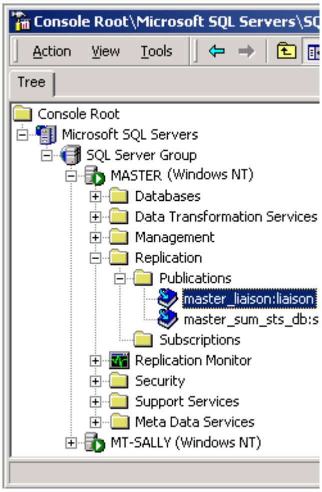
   The Register SQL Server Wizard appears.
4. Click **Next**.
5. In the list of available servers, select the **slave** server, click **Add**, then click **Next**.
6. Select **The SQL Server login information that was assigned to me by the system administrator** as the authentication mode, then click **Next**.
7. Select **Login automatically using my SQL Server account information**, enter the login name (`sa`) and corresponding password, then click **Next**.
8. Select **Add the SQL Server(s) to an existing SQL Server group**, then click **Next**.
9. Click **Finish**.
10. When informed that server registration is completed, click **Close**.

Subscribing to liaison database

1 In **Console Root\Microsoft SQL Servers\SQL Server Group***name_of_master* **(WindowsNT)\Replication\Publications**, right-click **master_liaison : liaison**, then select **Push New Subscription**.

   Where *name_of_master* stands for the actual name of the master server.

   The Push Subscription Wizard appears.



2 Check **Show advanced options in this wizard**, then click **Next**.

3 When prompted to choose subscribers, select the **slave** server, then click **Next**.

4 When prompted to choose the destination database, enter `liaison`, then click **Next**.

5 Select **Run the agent at the Distributor**, then click **Next**.

6 When prompted to set the distribution agent schedule, select **Continuously**, then click **Next**.

7 Select **Yes, initialize the schema and data**, then click **Next**.

8 Select **Queued updating**, then click **Next**.

9 Click **Next** again to start required service, then click **Finish**.

   The process begins.

10 When informed that the subscriptions were successfully created, click **OK**.

Subscribing to sum_sts_db database

1. In **Console Root\Microsoft SQL Servers\SQL Server Group***name_of_master* **(WindowsNT)\Replication\Publications**, right-click **master_sum_sts_db : sum_sts_db**, then select **Push New Subscription**.

   The Push Subscription Wizard appears.

2. When prompted to choose subscribers, select the **slave** server, then click **Next**.
3. When prompted to choose the destination database, enter `sum_sts_db`, then click **Next**.
4. Select **Run the agent at the Distributor**, then click **Next**.
5. When prompted to set the distribution agent schedule, select **Continuously**, then click **Next**.
6. Select **Yes, initialize the schema and data**, then click **Next**.
7. Select **Queued updating**, then click **Next**.
8. Click **Next** again to start required service, then click **Finish**.

   The process begins.

9. When informed that the subscriptions were successfully created, click **OK**.

Publishing files from the slave server

You have already set the directories used for replication on the master server. To enable data sharing, you must now do the same thing on the slave server.

1. Make sure that you are on the **slave** server.
2. On **C:\SpeechAttendant**, create a **Published** folder.
3. Right-click **Published**, then select **Sharing**.
4. At the **Sharing** tab of the **Published Properties** window, select **Share this folder**, then click **Permissions**.
5. Allow read access to everyone (only read access), then click **OK**.
6. Access level 3 of the configuration panel and set the following parameters:

Parameter	Value to enter
Replication/Target_Directory	`$HOME$/Published`
Replication/Source_subscriber_directory	`\\`*`name_of_master`*`\Published`†
Activate the File Publisher	$SERVER_NAMES@FR_PUBLISHER$
Activate the File Subscriber	$SERVER_NAMES@FR_SUBSCRIBER$
Run-Time : Secondary Database Server	*`name_of_master`*†

† Where *`name_of_master`* must be replaced with the actual name of the master server.

7. Click **Apply**, then exit the configuration panel.
8. In **C:\SpeechAttendant\bin**, double-click **FR_Subscriber Config.exe**.
9. When prompted, enter the user name (locus), domain, and password, then click **OK**.
10. Restart the computer.

Identifying servers and defining port assignment

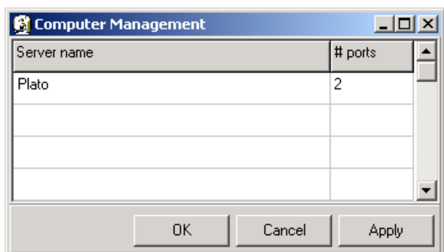
This procedure only needs to be done on the master server: information provided on this server will be automatically reflected on the slave server.

To identify servers and set port assignment:

1 Access the menu editor from the **master** server.

2 In the toolbar, click **Ports and entry points**.

3 Click **Computer management**.



4 Replace **(local)** with the name of the master server and enter the actual number of ports for this server.

5 On the following line, enter the name of the slave server and number of ports for this server.

6 Click **OK**, then exit the **Ports and Entry Points** window.

About configuration settings in a server cluster

In a multi-server environment, most of the configuration settings only need to be adjusted on the master server. As any other change made to the application, new setting values are automatically reflected on the slave server.

However, you must adjust settings pertaining to ACD queues on both servers (unless you prefer to create a hunt group). For details on how to program an ACD queue in a multi-server environment, see page 59.

There are other settings that are not synchronized and that can be adjusted differently on each server. They are:

Prefixes for External Numbers (see page 76)
Flash Length (see page 79)
Pause Length (see page 79)
Ring On Time (see page 79)
Ring Off Time (see page 79)
SWI Server Trace Level (see page 80)
CLID Available (see page 82)
Volume (see page 87)
Default Account Code (see page 88)
Max. Number of Unauthorized Requests (see page 88)
Show Menu Editor IDs (see page 98)

7

SpeechAttendant Configuration

This chapter explains how to configure SpeechAttendant to suit the organization's needs. Configuring SpeechAttendant involves:

- Customizing the settings in the configuration panel
- Configuring Windows parameters
- Connecting to a printer
- Changing default passwords

Subsequently, you can define schedules, create the menu structure, perform the initial load of the directory and define entry points (refer to the Administration Guide for details).

Configuration panel

There are two levels in which you can access the configuration panel:

- Level 2, which provides access to all settings, including the PBX model and voice board
- Level 1, which provides access only to the level 1 settings

All the configuration settings are described starting on page 75.

You already configured some of these settings during license validation. Other settings have default values that vary depending on the type of PBX; you many need to modify some of them.

For details on configuration settings in a multi-server application, see page 72.

To modify configuration settings:

1 Turn on the SpeechAttendant computer and log on as `locus`.

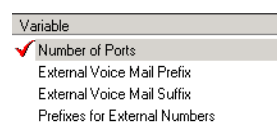
2 In the **Admin Tools** window, click **Configuration Panel**.

3 When prompted, select **Level 2** and enter the corresponding password.

4 In the left pane of the window, select a setting.

5 In the right pane, modify its value.

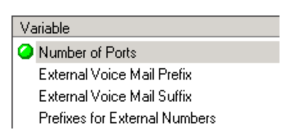
In the left pane of the window, a red check mark appears next to the setting name to highlight the change.



6 Customize all other necessary settings.

7 Click **Apply**.

Check marks turn into green bullets.



8 When your are done, close the **Configuration** window.

9 Follow the system instructions.

Once you have customized the configuration settings, you are ready to proceed with the creation of the menu structure and the initial directory upload (refer to the *Administration Guide*).

Setting descriptions

The settings are presented in the order in which they appear in the configuration panel, with the exception of settings pertaining to call accounting, which are grouped together. The level (1 or 2) of each setting is always specified.

For settings pertaining to ACD queues, see Chapter 4, page 27.

PBX and voice board

The **PBX** and **Voice board** fields appear at the top of the configuration panel. They contain the information that you specified during license validation.

You can change the values of these fields, but only in level 2 and only if:

- You are switching from an analog to a digital integration (this change has an impact on both field values).
- The PBX with which SpeechAttendant must be integrated has changed.
- A new type of voice board is installed following an increase in the number of system ports.

Number of Ports

This level 2 setting corresponds to the number of active ports in the system (that is, ports able to receive calls). The number of active ports can be less than the number listed in the license or that are physically installed.

External Voice Mail Prefix

This level 2 setting corresponds to the string that SpeechAttendant must dial before a voice mailbox number. It must contain the usual transfer string (see page 78), followed by the guest access mailbox number or, in a Meridian environment, the express messaging access number.

External Voice Mail Suffix

This level 2 setting corresponds to the string that SpeechAttendant must dial after the voice mailbox number. It can be used to enable other options.

Prefixes for External Numbers

If SpeechAttendant will be used to transfer calls to external numbers, this level 2 setting corresponds to the access code (or codes) to place external calls (9 is the common access code). If there is more than one prefix, type the digits one after the other, without spaces or commas.

Number of Digits in Area Code

This level 2 setting corresponds to the number of digits in area codes.

Number of Digits in External Phone Numbers

This level 2 setting corresponds to the number of digits in external local phone numbers.

Default Transfer Mode

This level 2 setting allows you to determine one of the following as the default transfer mode:

- **Supervised** (this mode is recommended, see below)
- **Unsupervised** (see page 78)

About Supervised Mode

One advantage of the supervised mode is that it gives callers the choice of asking for someone else when the desired party's line is busy.

In supervised mode, when the called party is either an internal extension, a voice mail or fax/modem number, SpeechAttendant waits for a ring signal before releasing the call to the PBX. If it detects a busy signal, it will ask if the caller wants to speak to another person or tell the caller to call back later (depending on the value of the System Behavior on Busy setting; see page 98). If SpeechAttendant detects neither a ring nor a busy signal after five seconds, it will release the port automatically as in the unsupervised mode.

In supervised mode, the called party is an external phone number on the public switched telephone network (PSTN), SpeechAttendant waits up to five seconds for a ring signal before releasing the port. If it detects a busy signal, it will ask if the caller wants to speak to another person or tell the caller to call back later. If, after five seconds, SpeechAttendant detects neither a ring nor a busy signal, it will release the port automatically as in the unsupervised mode.

About Unsupervised Mode

In unsupervised mode, SpeechAttendant performs a *blind transfer*, that is, it releases the port after sending the digits to the PBX without waiting to hear a ring or busy signal.

This mode is used when ring or busy detection is problematic. For example, on Norstar systems where the ring tone on the calling line is not synchronized with the one on the called station, the user might pick up before the call is released.

This mode can be used for extensions equipped with a voice mail system. Since these extensions are usually programmed to forward calls to the called party's voice mail on a busy signal, SpeechAttendant does not need to offer callers another choice.

It can also be used for devices that have no dial tone, such as pagers and voice mailboxes, or for overseas and other destination numbers whose ring or busy signals differ from those in North America.

Because of a connection limitation in certain telephone systems, there could be a delay between the time the called party answers and when the connection with the caller is made. If the called party actually answers the phone, this delay is not apparent. However, the delay might clip the beginning of the called person's voice mail message. Users with voice mail should record a personal greeting rather than just their name.

Transfer String

This level 2 setting corresponds to the string required to initiate a transfer.

Flashback String

This level 2 setting corresponds to the string required to initiate a flashback.

Flash Length

In an analog environment, this level 2 setting corresponds to the desired duration of a hookflash, in hundredths of a second (1/100 s).

Pause Length

This level 2 setting determines how long, in hundredths of a second (1/100 s), SpeechAttendant will pause when interpreting commas between transfer strings.

Ring On Time

SpeechAttendant answers a call when it detects a ring tone of a certain length followed by a silent interval of a certain length (ring off time, see below).

The Ring On Time setting (level 2) determines the ring tone duration in tenths of a second (1/10 s) of the calls SpeechAttendant must answer. This value should only be changed when SpeechAttendant must detect short ring tones or to match other ring/no ring patterns defined in PBX configuration.

Short ring on times may cause erroneous ring tone detection.

Ring Off Time

SpeechAttendant answers a call after it detects a ring tone of a certain length (see ring on time, above), followed by a silent interval of the length specified here. This level 2 setting determines the length in tenths of a second (1/10 s) of the silent interval.

If the ring off time specified is longer than as delivered by the PBX, SpeechAttendant will not answer the call.

SpeechAttendant Trace Level

This level 2 setting determines the level of detail recorded in the log files for the application. **Prod** (production) is recommended for normal operations. **Dev** (development) or **Beta** (beta monitoring) generate greater log file detail, but can cause memory problems. Change this setting only under the direction of the Customer Service Center.

SWI Server Trace Level

This level 2 setting determines the level of detail recorded in the log files for the OpenSpeech recognition engine. **Prod** (production) is recommended for normal operations. **Dev** (development) or **Beta** (beta monitoring) generate greater log file detail, but can cause memory problems. Change this setting only under the direction of the Customer Service Center.

Barge-In Mode

This level 2 setting determines if callers will be allowed to interrupt greetings and prompts.

The value selected in the configuration panel applies to the entire application, but the barge-in function can be set differently in certain menus or entry points. For details, refer to the *Administration Guide*.

Call Analyzer Call Filter

This level 2 setting should not be modified. For more information, contact the Customer Service Center.

Call Analyzer Recorded Channels

This level 2 setting determines from which ports call data should be gathered (notably for the call analyzer; refer to the *Administration Guide* for details). Change this setting only under the direction of the Customer Service Center.

Transfer Connect Prefix

If your company already uses AT&T's Transfer Connect service, you can activate it in a SpeechAttendant entry point. This is useful for handling calls to toll free numbers. If that is the case, you must configure this level 2 setting so that it corresponds to the string that SpeechAttendant must dial before external phone numbers to access the Transfer Connect service.

System Languages

This level 2 setting determines which language or language combination the system can recognize (according to the license). The possible languages are:

- US English (US_EN)
- Canadian French (CA_FR)
- US Spanish (US_SP)

US English can be combined with one of the other languages to create a bilingual system. Using the appropriate speed-dial key (by default, 9; see page 89), callers can then choose the language in which the system addresses them.

To specify default languages for menus and entry points, consult the *Administration Guide*.

CLID Available

This level 2 setting only concerns analog installations[1]. It allows you to specify whether the telephone system is capable of transmitting CLID. If that is the case, choose **On**; otherwise, choose **Off**. For details about dynamic call routing, see below.

SpeechAttendant uses CLID to identify callers when the application includes secured features and menus or to allow dynamic call routing and call accounting. For details, refer to the *Administration Guide*.

DNIS and CLID Masks

To enable dynamic call routing and caller identification, SpeechAttendant must extract the DNIS and CLID from a character string transmitted by the PBX with each call received[2]. Since the syntax of this character string varies from one system to another, masks are needed to identify the DNIS and CLID within the character string.

Default DNIS and CLID masks are defined in level 2 of the configuration panel for all supported digital PBXs. These masks correspond to the different types of calls: external, internal, or calls where a specific number was dialed.

In analog systems allowing dynamic call routing, these masks are not visible. If you have a digital PBX and corresponding DNIS and CLID masks are not displayed, call the Customer Service Center.

1. Digital PBXs are all capable of transmitting CLID as well as DNIS, which enables dynamic call routing.

2. Analog systems cannot transmit DNIS; some cannot even transmit CLID (see the previous procedure).

For some of the PBXs, there are no masks selected by default, but you can select masks from the drop-down list. In some cases, masks are selected by default, but you must verify that they are the correct ones. To do so, you need to understand their syntax.

Character	Description
C	An alphabetical character, capitalized or not, with or without accents. If preceded by a number, this indicates a sequence of that number of alphabetical characters (**4C** means a sequence of four alphabetical characters). **C=** followed by a space indicates a space.
N	A number from 0 to 9. If preceded by a number, this indicates a sequence of that number of digits (**4N** means a sequence of four digits).
A	Same as **C**, plus any number from 0 to 9.
T	A character belonging to a telephone number, such as a hyphen, a parenthesis or a number from 0 to 9.
S	A character not included in **C** or **A**, such as a comma, a period or hyphen (**S=,** would therefore mean a comma, **S=.** would mean a period, and so on).
Z	Any character or symbol.
*	Any number of characters belonging to the category that follows it (***C** would therefore mean any number of alphabetical characters and/or spaces).
@	A range of characters comprised between the preceding and following values (**4N=1000@1999** would therefore mean a number between 1000 and 1999).
a, b, c (where *a, b, c* can be replaced by any value)	Any of the indicated values in the group (**3C= sep, oct, nov** would therefore signify one of the abbreviations from the group)**.**

All these elements appear in curly brackets, except for the DNIS and CLID, which appear between *greater than* and *less than* signs.

An actual DNIS mask might look like this:

Setting	Current value
1st DNIS Mask	{26Z}+{*T}+{C= }+<*T>+{C= }+{S="}+{*C}+{S="}

The above example represents a 26 character string, a telephone number (the CLID), a space, another telephone number (the DNIS), a space, a quotation mark, a string of alphabetical characters, and a quotation mark.

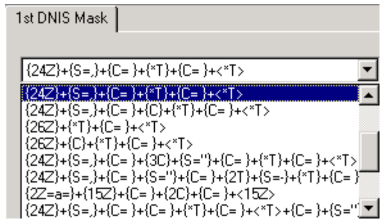
The corresponding CLID mask might look like this:

Setting	Current value
1st CLID Mask	{26Z}+<*T>

To identify the correct masks:

1. Have two telephones, with display screens and connected to the PBX, and one cell phone available close to the SpeechAttendant computer.
2. Using the two display screen telephones, make an internal call.
3. As the call is transferred, carefully observe the display screen of the answering telephone. Note the order of the DNIS and CLID information as it is displayed (this information may be displayed very quickly and it may be necessary to call again).
4. In the configuration panel, verify that the default values for the **1st DNIS mask** and **1st CLID mask** settings match the string that appears in the telephone display.

5 If the default values don't match, select the appropriate ones in the drop-down list.



If you don't find a mask corresponding to the information supplied by the PBX, contact the Customer Service Center.

6 Using a cell phone to make an external call, repeat steps 3 to 5 to specify the second DNIS/CLID masks.

7 Click **Apply** in the **Configuration** window.

At this point in the installation, only the masks for internal and external calls can be defined.

SMTP Server Name

This level 2 setting is not used in this version of SpeechAttendant.

Access to Secured Features

This level 2 setting allows you to specify the security level for accessing secured features. The possible levels are:

- **1**, for PIN only
- **2**, for voice biometrics only
- **3**, for PIN and voice biometrics

For details, refer to the *Administration Guide*.

Beep During ASR Processing

This level 2 setting determines whether SpeechAttendant will play beep tones while it processes requests. This lets callers know that they are not expected to say anything at that moment and avoids dead air in applications with large directories. It is not activated by default.

Beep Interval

This level 2 setting determines the duration, in seconds, of the interval between the beeps that play when the system is processing a request (see above). The possible values are **1** through **6**.

Volume

This level 2 setting is used to adjust the sound level of the Dialogic cards, and consequently, the volume of SpeechAttendant greetings and prompts.

0 or **-8** are the default values, depending on the card. If the volume of the prompts is too high, choose a lower setting.

Refusal and Request Together

This level 2 setting determines whether SpeechAttendant can handle one-step corrections (when callers reject a suggestion and repeat their request in the same utterance). For example:

SpeechAttendant:	Do you wish to talk to... Mary Lipton?
Caller:	No, Larry Benton!
SpeechAttendant:	Thank you. Transferring your call to... Larry Benton.

This option is not activated by default. In applications with large directories, one-step corrections may hinder voice recognition.

Call Accounting Settings

SpeechAttendant can be integrated with the Avotus Professional call accounting system (in Meridian environments). If that is the case, you must configure the following settings.

Setting	Level	Description
Account Code Prefix	2	The value of this setting must correspond to the SPRE,—or special prefix number— programmed in the PBX, followed by 5. For example, if the SPRE is 11, the account code prefix would be 115.
Number of Digits in Account Codes	2	This setting determines the number of digits account codes can contain.
Authorization Code Prefix	2	The value of this setting must correspond to the SPRE—or special prefix number—programmed in the PBX, followed by 6. For example, if the SPRE is 11, the authorization code prefix would be 116.
Default Account Code	1	Certain directory entries must be associated with a specific account code, while others can be associated with a generic one. This setting corresponds to the account code used for entries that do not have a specific account code.
Max. Number of Unauthorized Requests	1	This setting determines how many times employees can attempt to make a call they are not authorized to make. When this number is reached, the system will hang up.

When SpeechAttendant is integrated with a call accounting system and an employee uses it to make a call, the system first establishes the employee’s identity (with CLID) to determine what type of call the employee can make and which account code to use.

The employee can then make a request. If the system recognizes the request, it transmits a sequence to the PBX containing the following information, in this order: prefix for authorization codes, employee’s authorization code, prefix for account codes, employee’s account code and requested phone number.

Percentage of Ports Available for PAM

This level 1 setting determines how many ports can be used simultaneously by employees to record their names, create their voiceprints, change their passwords or redirect their calls via the personal administration mode (PAM).

The value of this setting should be a percentage. If 0 is entered, no ports are available. For two ports to be available in a four-port system, enter 50, three ports, enter 75, and so on.

For details about PAM features, refer to the *Administration Guide*.

Operator Extension Number

The value of this level 1 setting should correspond to a system-wide operator extension. This number can be overridden in menu entries (refer to the *Administration Guide*).

A call is automatically transferred to the operator when all attempts to understand the caller's request have failed. Callers can also be transferred to the operator on request or by pressing an assigned speed-dial key.

Speed-Dial Keys

To quickly access particular services (for example, technical support), callers can use speed-dial keys rather than dialing extensions or phone numbers.

The following speed-dial keys have already been programmed:

Preset Speed-Dial Keys

Telephone keys	Purpose
0	Activates transfer to the operator (see page 89)
9	Changes the language in a bilingual system (see page 81)
*99	Activates the personal administration mode (refer to the *Administration Guide*)
*	Activates the maintenance mode (refer to the *Administration Guide*)
123	Provides access to the name recorder (refer to the *Administration Guide*)

You can program other speed-dial keys according to your organization's needs.

To program a speed-dial key:

1 Select the corresponding setting in the left pane of the configuration panel.

2 In the **Speed-Dial Key** field, specify the telephone pad key you want to assign to the selected setting.

> The number of the speed-dial key does not have to be the same as the telephone key. For example, default **Speed-Dial Key 1** is associated with telephone key **0**.

3 In the **Speed-Dial Key Extension** field, enter the extension or external phone number you want to assign to the speed-dial key, or select one of the following commands in the list:

- **Operator**, to transfer calls to the operator.
- **Goto Maintenance**, to activate the maintenance mode, allowing callers to choose between the name recorder or system administration functions.
- **Toggle Language**, to switch to the other language of a bilingual system.
- **Goto Name Recorder**, to activate the name recorder directly.
- **Goto System Admin**, to activate the system administration functions of the maintenance mode directly.
- **Goto Personal Admin**, to activate the personal administration mode.

External phone numbers must contain any required prefix, pause (represented by a comma) or area code. In bilingual systems, an extension or phone number must be specified for each language, in the appropriate field.

4 In the **Speed-Dial Key Timer** field, specify how much time (in tenths of a second) the system must wait before dialing the extension number.

   If any extensions begin with the number of the assigned key, or if the speed-dial key is associated with a combination of digits[1], a delay of 20 tenths of a second is required to let the caller press another key.

   If the speed-dial key is associated with a single digit, and no extensions begin with this digit, set the timer to 0 to have calls transferred as soon as the key is pressed.

5 In the **Speed-Dial Prefix** field, enter any string the system must dial before the extension number, instead of the usual transfer string.

6 In the **Speed-Dial Suffix** field, enter any string the system must dial after the extension number.

To deactivate a preset speed-dial key:

1 Select it in the left pane of the **Configuration** window.

2 Delete the contents of the **Speed-Dial Key** field.

1. A combination of digits can be associated to a command such as Toggle Language or Goto NameRecorder, but not to an external phone number.

Homophone Resolution Method

This level 1 setting offers three different methods for dealing with homophones (names or aliases that have the same pronunciation):

- With the **Operator** method, each time a homophone is requested, SpeechAttendant plays a message saying that there is more than one person with that name, then transfers the call to the operator (for details on the operator extension, see page 89).
- With the **Name** method, the recording associated to each homophone entry must contain a distinctive element in addition to the person's name; for example, "John Smith, sales manager" and "John Smith, webmaster." When a homophone is requested, the system suggests one of the homophones using the recording (for example, "Do you wish to speak to... *John Smith, sales manager*?"). If the caller rejects the first suggestion, SpeechAttendant suggests the second homophone. For details on name recordings, refer to the *Administration Guide*.
- With the **Department** method, the department to which each homophone employee belongs should be specified in their respective entries (refer to the *Administration Guide*). When a homophone is requested, SpeechAttendant suggests one of the homophones using the requested name plus that person's department; for example, "Do you wish to speak to... Jane Wyatt, *Customer Service Center*?" If the caller rejects the first suggestion, SpeechAttendant suggests the second homophone (for example, "Then, do yo wish to speak to... Jane Wyatt, *Marketing Department*?").

  This method only works for employees with like-sounding names who work in different departments.

  If this method is selected but no department is specified for a requested homophone, SpeechAttendant transfers calls to the operator.

Homophone Resolution Threshold

This level 1 setting determines the maximum number of names forming a group of homophones that SpeechAttendant can attempt to handle using either the **Name** or **Department** homophone resolution method (see page 92). If the number of homophones corresponding to a caller's request exceeds the number specified here, the call is transferred to the operator.

This setting does not concern the number of homophone groups listed in the directory, but the number of names in each group.

For example, if the value is **2**, SpeechAttendant attempts to handle homophones (based on either departments or recorded names) if there are only two names involved. If there are three or more, SpeechAttendant transfers the call to the operator.

Confirmation Before Homophone Listing

This level 1 setting determines whether SpeechAttendant must confirm that the name recognized is correct before suggesting homophones (if the homophone resolution threshold has not been reached and homophone resolution method is Name or Department). This will also apply when the caller didn't specify the first name of the person he wants to reach and the requested last name corresponds to several people.

By default, the system ask for confirmation each time that the voice recognition result is uncertain. This allows to avoid the following scenario:

SpeechAttendant:	Welcome to XYZ. Which person or...
Caller:	Bob Corey!
SpeechAttendant:	Do you want to talk to John Murray from Purchasing?
Caller:	No.
SpeechAttendant:	Then, do you want to talk to John Murray from Registrar's Office?
Caller:	No!
SpeechAttendant:	Which person or service would you like to reach?
	And so on.

Instead, you will get the following call flow:

SpeechAttendant:	Welcome to XYZ. Which person or...
Caller:	Bob Corey!
SpeechAttendant:	Did you say John Murray?
Caller:	No.
SpeechAttendant:	Then, do you want to talk to Bob Corey?
Caller:	Yes!
SpeechAttendant:	Thank you! Transferring your call.

Consecutive Rejections

When SpeechAttendant is totally unable to understand an utterance, it rejects it and asks the caller to repeat it. Rejection can be caused by background noise, coughing, inaudible speech, line noise, or prolonged silence.

The **Consecutive Rejections** setting (level 1) allows you to determine how many consecutive rejected utterances the system will allow before either transferring the call to the operator or hanging up (based on the value of the **Transfer to Operator on Silence** setting, described on page 96).

Invalid Password Counter

The value of this level 1 setting indicates the number of times someone has tried to access the maintenance mode with an invalid password in one day. When this counter reaches **4**, access is denied for 24 hours, unless you restore it.

To restore access after it has been disabled:

- Set the counter to **0**.

Maintenance Mode Password

This level 1 setting contains the maintenance mode password, which provides access to the name recorder and system administration functions (refer to the *Administration Guide*). The maintenance mode password can contain up to 12 digits, and it must not contain a star or pound sign.

You can modify the maintenance mode password from the configuration panel or directly in the maintenance mode using a touch-tone phone (see page 103). See also “Name Recorder Password” on page 96.

Maximum Number of Digits in Extensions

The value of this level 1 setting indicates what is the maximum number of digits internal extensions can contain according to the numbering plan. It must be set high enough to accommodate the longest extension number in the system. The default setting is **3** or **4**, depending on the PBX type.

Some extensions may be shorter than the maximum. When dialing short extensions, callers can press the # key to indicate that they have finished.

Announce Destination Number

This level 1 setting determines whether or not SpeechAttendant announces the destination phone number when transferring a call. For example: “Transferring your call to... Wilma Patterson, *extension 3321*.”

Possible values for this setting are:

- **No**, for SpeechAttendant not to announce destination numbers.
- **Extension**, for SpeechAttendant to announce only local extensions (excluding voice mailbox numbers).
- **External**, for SpeechAttendant to announce only external numbers (such as off-site employees’ office numbers, home and cell phone numbers).
- **Voice mail**, for SpeechAttendant to announce only voice mailbox numbers.
- **All**, for SpeechAttendant to announce extensions, external numbers and voice mailbox numbers, depending on the case.

Ports are in use for longer periods of time when this value is selected.

Transfer to Operator on Silence

This level 1 setting determines the system behavior when no utterance is received.

To have silent calls transferred to the operator:

- Select **Yes**.

This allows you to create prompts such as: "If you know the name of the employee you want to reach, say it now. Otherwise, please hold the line and your call will be transferred to the operator." This also gives a chance to intimidated callers to speak to someone.

For SpeechAttendant to hang up on silent calls:

- Select **No**.

Name Recorder Password

This setting contains the password providing direct access to the name recorder. It can contain up to twelve digits, and it must not contain a star or pound sign.

The name recorder is also accessible from the maintenance and personal administration modes (refer to the *Administration Guide* for details).

Emergency Barge-in Mode

This level 1 setting determines whether the barge-in function is enabled or not during the emergency greeting.

For the emergency greeting to be played completely before any interaction is possible:

- Select **Off**.

To allow callers to make their requests before the emergency message is finished:

- Select **On**.

GUI Language

This setting determines the language of the graphical tools allowing system administration. Two languages are available: English and French.

To change the GUI language:

- Select **Fr** for French, and **En** for English.

Open windows are not automatically refreshed.

Some of the windows, error messages and options that are used in SpeechAttendant are generated by the Windows 2000 operating system and are therefore not affected by this setting: their language is determined by the language of the operating system.

LDAP Server Timeout

This level 1 setting determines how many seconds SpeechAttendant should wait before abandoning its attempt to connect to the LDAP server in order to perform a directory update by interface (refer to the *Administration Guide*).

Single Access to Name Recorder

This level 1 setting determines whether the name recorder should be disabled for entries whose names have been recorded. By default, it is not possible to rerecord a name unless the name recorder access is reset (for details, refer to the *Administration Guide*).

To allow multiple uses of the name recorder:

- Select **No**.

To limit name recorder use to one attempt:

- Select **Yes**.

Default PIN

This level 1 setting allows you to specify a starting PIN that will be assigned to all employees, until they modify it in the PAM[1]. The default PIN should consist of at least four digits and should not contain a star or pound sign.

Reminder for Name Recorder

This level 1 setting determines whether employees who have not recorded their names should be systematically given the opportunity to do so each time they access SpeechAttendant from their default phone number.

To disable the reminder:

◆ Select **Off**.

Show Menu Editor IDs

This level 1 setting determines if menu editor IDs must systematically appear along with entry names in the directory.

By default, only entry names appear in the phone directory or in the menu editor. Displaying the menu editor IDs can be useful if your application contains several entries with the same name.

For menu editor IDs to appear along with their corresponding entry names:

◆ Set the variable to **Yes**.

System Behavior on Busy

This level 1 setting only applies to supervised transfers (see page 77). It determines what the system does when it has recognized the caller's request, but cannot perform the transfer because the line is busy. By default, it gives the caller the option of asking for someone else. It can also hang up after having told the caller to call back later.

To prompt callers to call back later:

◆ Select **Hang up**.

To allow callers to ask for another person or service:

◆ Select **Ask for someone else**.

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1. It is also possible to assign a specific PIN in an individual's entry. Refer to the *Administration Guide* for details.

Verify Caller Identity by Default

This level 1 setting determines whether the system should validate (using voice biometrics or a PIN) the identity of callers trying to access secured features or menus from any phone number contained in the directory.

If the calling number is not contained in the directory or if the PBX is not capable of transmitting the CLID, SpeechAttendant will systematically verify the identity of anyone trying to access a secured feature or menu.

If most of the phones associated with your application are private (that is, each phone is used by a specific employee), it may not be necessary for the system to validate caller identity. If that is the case, the value of this setting should be **False**.

The **Verify Caller Identity by Default** setting determines the default value of the **Private phone** attribute in transfer entries. You can always change this attribute manually for any phone number (refer to the *Administration Guide* for details).

Configuring Windows parameters

Verify that the two following parameters are set in the operating system:

- Date and time
- Keyboard language

Do not change the display settings. This can significantly reduce the display quality for people accessing the system remotely (especially for troubleshooting).

Connecting to a printer

SpeechAttendant can be connected to a printer, which can be useful for printing reports or using other diagnostic features related to the system's performance. Refer to the Windows documentation or follow the installation procedures supplied by the printer's manufacturer for details.

Modifying passwords

For security reasons, you should change all the default passwords:

- Locus account
- Windows Administrator
- Level 1 of the configuration panel
- Level 2 of the configuration panel
- Maintenance mode
- Name recorder
- Default PIN
- Admin (in a multi-administrator environment)

Note the new passwords, keep them in a safe place, and above all, remember to notify the Customer Service Center.

The default passwords don't appear in this guide. If you don't know them, contact the Customer Service Center at 1 866 434-2564 or (514) 390-3922.

Locus and Windows administrator

There are two user accounts on a SpeechAttendant computer:

- **Locus**, which will be used by the system manager to run SpeechAttendant after it has been installed.
- **Windows Administrator**, which is part of the operating system and has the highest authority.

To change the Locus or Windows Administrator account password:

1. Log on to SpeechAttendant as:
   - `locus`, if changing the Locus account password
   - `Administrator`, if changing the Windows Administrator password
2. Press **Ctrl+Alt+Delete**.
3. Click **Change Password**.
4. Type the current password.
5. Type the new password.
6. Type the new password again in the **Confirm Password** field.
7. Press **Enter**.

In the case of a multi-server environment, if you change the Locus account password on the master server, you must also change it on the slave server.

Level 1 and level 2

Configuration settings can be defined by the system administrator (level 1) or telephony distributor (level 2). Level 1 settings deal mainly with the day-to-day operation and maintenance of the system, while level 2 settings are set at installation time according to the customer's needs (refer to the signature pages of the *Site Preparation Document*).

To change the level 1 or 2 password:

1. In the **Admin Tools** window, click **Configuration**.
2. When prompted, select **Level 2** and enter the corresponding password.
3. From the **Tools** menu, choose:
   - **Change Password** > **Level 2**

   or

   - **Change Password** > **Level 1**

   The **Change Password** window appears.
4. Type the current password.
5. Type the new password.
6. Type the new password again in the **Confirm password** field.
7. Click **Update**.

PIN, maintenance mode and name recorder

These three passwords are registered and can be modified in the configuration panel. See page 97, page 94, and page 96.

You can also modify the maintenance mode password from any touch-tone phone.

To modify the maintenance mode password by phone:

1. Call SpeechAttendant from a touch-tone telephone.
2. Say "Maintenance mode," then press **2**.

   Another way to access the maintenance mode is to press the assigned speed-dial key (by default, the * key). For details, see page 89.

3. When prompted, enter the maintenance mode password.
4. Press **5**.
5. When prompted, enter the current password on the telephone keypad.
6. When prompted, enter a new password.
7. When prompted, enter the new password again.
8. Exit the maintenance mode by pressing the * key or follow the system instructions to perform another task.

The new password is now active. The change is reflected automatically in the configuration panel.

Admin

In a multi-administrator environment, **Admin** is the name of the principal administrator. The **Admin** password provides the administrator with access to the phone directory and the prompt recorder. Refer to the *Administration Guide* for instructions on how to change this password.

8

pcAnywhere

SpeechAttendant comes equipped with a modem and pcAnywhere host software. This enables ScanSoft's Customer Service Center to optimize the directory, ensure that the system is functioning properly, and, when the need arises, perform remote troubleshooting.

The customer can also use pcAnywhere to administer SpeechAttendant from a remote workstation. For example, if the computer is installed in a server room, the system administrator may find it useful to manage the system from another workstation.

The computer that will be used to manage SpeechAttendant must be equipped with pcAnywhere and configured to allow a remote connection.

Introduction

pcAnywhere allows remote management of the SpeechAttendant application via either:

- A LAN connection
- Dial-Up Networking through a Point-to-Point Protocol (PPP) connection

To be able to establish a pcAnywhere connection, you must:

1. Verify the configuration of the remote computer.
2. Create a DUN file on the remote computer (if you want to use a PPP connection).
3. Configure pcAnywhere on the remote computer.

Verifying the remote system

To run the host software, the remote computer must:

- Be a Pentium-class computer running Windows 2000, Windows XP, Windows NT or Windows 95/98
- Be equipped with an analog 33.6 Kbps modem (or faster)
- Be connected to an analog telephone line
- Be equipped with pcAnywhere remote access software

pcAnywhere remote access software is not included in the SpeechAttendant package; only the host software is provided.

Creating a DUN file

To be able to manage SpeechAttendant remotely, you must first create a DUN file on the remote computer. The process differs slightly depending on the operating system installed on the remote computer. Refer to the appropriate procedure.

To create a DUN file using Windows 2000:

1. On the desktop of the remote computer, right-click **My Network Places** and choose **Properties**.
2. In the **Network and Dial-up Connections** window, double-click **Make New Connection**.
3. In the first **Network Connection Wizard** window, click **Next**.
4. Select **Dial-up to private network**, then click **Next**.
5. Enter the remote computer's phone number, then click **Next**.
6. Select **For all users**, then click **Next**.
7. Enter connection's name, then click **Finish**.

You can now configure pcAnywhere (see page 112).

To create a DUN file using Windows XP:

1. On the desktop of the remote computer, right-click **My Network Places** and choose **Properties**.
2. In the **Network Connections** window, double-click **New Connection Wizard**.
3. In the first **New Connection Wizard** window, click **Next**.
4. Select **Connect to the network at my workplace**, then click **Next**.
5. Select **Dial-up connection**, then click **Next**.
6. Enter the connection's name, then click **Next**.
7. Enter remote computer's phone number, then click **Next**.
8. Select **For anyone's use**, then click **Finish**.

You can now configure pcAnywhere (see page 112).

To create a DUN file using Windows NT:

1 On the remote computer, click **Start**, then choose **Programs** > **Accessories** > **Dial-Up Networking**.

2 In the **Dial-Up Networking** window, click **New**.

3 In the **New Phonebook Entry Wizard** window, enter the name of the new phone book entry, then click **Next**.

4 In the **Server** window, click **Next**.

5 In the **Phone Number** window, specify the required information.

6 In the **New Phonebook Entry Wizard** window, click **Finish**.

7 In the **Dial-Up Networking** window, click **More**, then select **Edit entry and modem properties**.

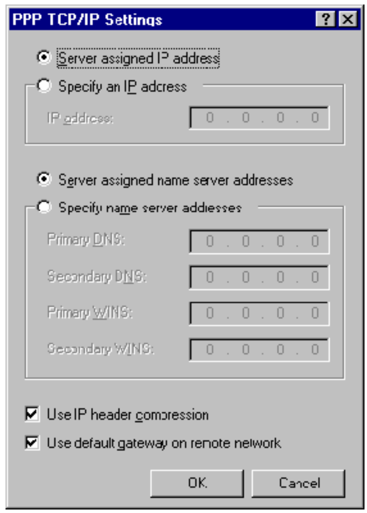
8 In the **Edit Phonebook Entry** window, click the **Server** tab.

9 In the **Dial-up server type** field, select **PPP: Windows NT, Windows 95 Plus, Internet**.

10 Under **Network protocols**, check **TCP/IP**, then click **TCP/IP Settings**.

11 Select the required parameters, then click **OK**.

Parameters should be the same as those indicated below. If the remote computer is connected to a network, contact the network administrator to get the correct values.



12 Under the **Server** tab of the **Edit Phonebook Entry** window, check **Enable software compression** and **Enable PPP LCP extensions**, then click **OK**.

13 In the **Dial-Up Networking** window, click **Close**.

You can now configure pcAnywhere (see page 112).

To create a DUN file using Windows 95/98:

1 On the desktop of the remote computer, double-click **My Computer**.

2 Double-click **Dial-Up Networking**.

3 Double-click **Make New Connection**.

4 Type a name to identify the connection, select a modem from the drop-down list, then click **Next**.

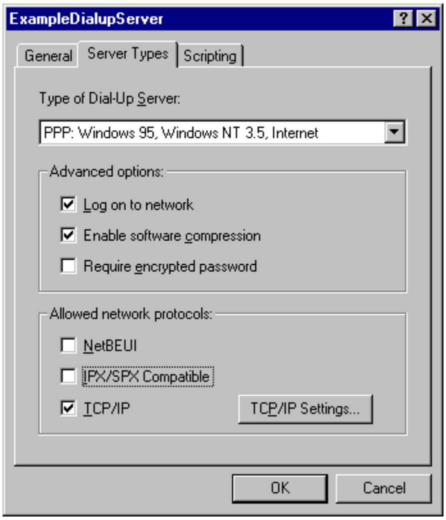
5 Type the area code and telephone number of SpeechAttendant, select the country code (USA and Canada use the same code), then click **Next**.

6 Click **Finish**.

The **Dial-Up Networking** window appears, with an icon for the connection you have created.

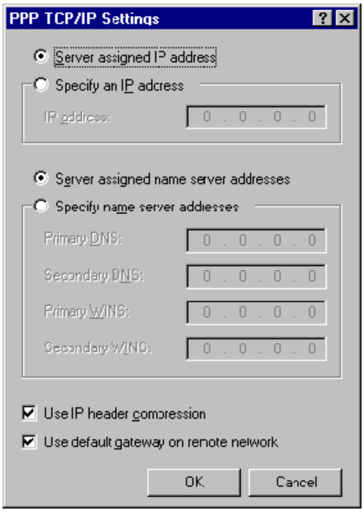
7 Right-click the new connection icon, then choose **Properties**.

8 Click the **Server Types** tab, then select required parameters as indicated below.



9 Click **TCP/IP Settings**, select the required parameters, then click **OK**.

Parameters should be the same as those indicated below. If the remote computer is connected to a network, contact the network administrator to get the correct values, as indicated below.



10 In the **Edit Phonebook Entry** window, click **OK**.

You can now configure pcAnywhere (see page 112).

Configuring pcAnywhere

pcAnywhere is already configured on SpeechAttendant. However, it must be configured on the remote computer before you can log on to the SpeechAttendant computer.

To configure the pcAnywhere remote software:

1 On the remote computer, click **Start**, then choose **Programs** > **Symantec pcAnywhere**.

2 In the **pcAnywhere** window, click **Remote Control**, then double-click **Add Remote Control Item**.

3 In the first **Remote Control Wizard** window, enter a name to identify the remote connection configuration, then click **Next**.

4 When prompted to specify a connection device, select **TCP/IP**, then click **Next**.

5 When prompted to enter the "Host PC's name," enter the IP address of SpeechAttendant, then click **Next**.

SpeechAttendant configuration requires an IP address rather than a host name.

6 If you wish to establish the remote connection immediately, check **Automatically begin remote**.

7 Click **Finish**.

pcAnywhere configuration is complete. An icon with the remote control connection's name appears in the **pcAnywhere** window.

Logging on to SpeechAttendant

Once pcAnywhere has been configured on the remote computer, you can use it to manage SpeechAttendant remotely.

To log on to the SpeechAttendant computer:

- pcAnywhere must be “waiting” on the SpeechAttendant computer.
- You must know the following information about the SpeechAttendant computer:
  - The modem phone number and/or the IP address provided by your distributor
  - The user name and password

Refer to pcAnywhere documentation for specific instructions.

9

Testing

Once the phone directory is loaded, you must test:

- The integration of SpeechAttendant with the PBX/KSU
- The SpeechAttendant functions

PBX integration

To validate the integration of SpeechAttendant with the PBX/KSU, you must test the following:

- Noise level
- Barge-in activation
- Ring tone detection
- Busy tone detection
- Hangup detection
- DTMF transfers
- Call forward capabilities

The purpose, procedure, and desired results of each test is explained below, along with solutions to potential problems.

Noise level

Purpose	Assure that the quality of the telephone lines is good. Excessively noisy lines will interfere with voice recognition and, in extreme cases, may trigger barge-in. This test is not required if you previously tested the PBX/KSU ports (as described on page 18).
Procedure	**1** From a relatively quiet environment, call all the SpeechAttendant ports, one after the other. **2** Remain silent when the system plays the greeting and prompt.
Result	The greeting and prompt should play without interruption. During the silences you should hear no unusual noise on the line.
Troubleshooting	◆ If the greeting or prompt is interrupted, barge-in has been triggered. Make sure that this is not due to noise in the room. If you are unsure, repeat the test a few times. Test the other ports to see if the problem is confined to one port. ◆ If the greeting or prompt is interrupted on some but not all the ports, the sound quality is uneven. Ask a telephony expert to verify the quality of the lines and make the necessary repairs. ◆ If you have done the above and perceive no improvement in the system performance, call the Customer Service Center.

Barge-in activation

Purpose	Verify that the barge-in function is triggered by normal speech at normal volume.
Procedure	**1** From an internal telephone located in a relatively quiet environment, call all SpeechAttendant ports, one after the other. **2** When the system answers, interrupt the greeting or prompt by saying the name of a person or service. Speak normally, at a natural speed. **3** Repeat from an external line.
Result	As soon as you make your request, the greeting or prompt should stop and the system should transfer your call to the corresponding extension.
Troubleshooting	If the greeting or prompt does not stop, try speaking a bit louder. In a noisy environment, the caller needs to speak louder to trigger the barge-in; in a quiet environment, barge-in should be triggered by normal speech. If this is not the case, call the Customer Service Center.

Ring tone detection

Purpose	Verify that SpeechAttendant detects ring tones properly.
Procedure	**1** Secure the cooperation of a volunteer to answer the phone as soon as it starts ringing. **2** Call SpeechAttendant and ask for a directory entry whose extension is physically close to the SpeechAttendant computer.
Result	You should hear your assistant say “Hello” immediately.
Troubleshooting	◆ If the system does not detect the ring properly, make sure the right PBX is selected in the configuration panel. ◆ If you do not hear your assistant saying “Hello,” and are connected a few seconds later, repeat the test. This time, your assistant should wait at least half a ring before picking up the phone. You should immediately hear the person answering. ◆ If the connection is made after more than one ring, the ring tone on the calling line is not synchronized with the one on the called station. To verify this, use an analog phone connected to the analog port that SpeechAttendant uses. Call an extension located close by. Listen carefully on the phone you are calling from and also for the actual ringing sound of the target phone to determine if the rings occur at the same time. ◆ On certain PBXs, like the Norstar, the telephone receiving the call may ring before the caller hears a ring in the receiver. In this case, the caller will not hear if the person picks up on the first ring and says “Hello.” The parties are connected when SpeechAttendant detects the ring a couple of seconds later. This results in the person who answers repeating “Hello.”

Ring tone detection (*cont'd*)

Troubleshooting	To resolve this issue, you must select **Unsupervised** as the default transfer mode in the configuration panel. SpeechAttendant will not wait for ring detection but will simply dial the extension and release the call. Busy signal detection, however, will not be possible: If a call is transferred to a busy extension, the caller will either hear the busy signal or the call will be dropped, depending on the PBX port possibilities. If the **Unsupervised** mode is selected and the problem persists, call the Customer Service Center. Some parameters can be adjusted to improve ring detection.

Busy tone detection

Purpose	Verify that SpeechAttendant advises callers when requested lines are busy (if the **Supervised** transfer mode is selected).
Procedure	**1** Take a telephone that is not programmed with **Forward on busy capability** off hook (a directory entry must be associated with this telephone number). **2** Using another telephone, call SpeechAttendant and ask for the entry corresponding with the off hook telephone.
Result	You should be told that the line is busy and offered to speak to someone else.
Troubleshooting	If SpeechAttendant does not detect the busy signal: ◆ Ensure that the default transfer mode in the configuration panel is **Supervised** (see page 77). ◆ Ensure that the correct PBX is selected in the configuration panel. ◆ Ensure that the line generates a busy signal by calling it directly with another telephone. This test is not required if you previously tested the PBX/KSU line (as described on page 18). If the line actually generates a busy signal which SpeechAttendant does not detect, call the Customer Service Center.

Hangup detection

Purpose	Assure that SpeechAttendant detects that callers have hung up and quickly releases the port to make it available for the next call.
Procedure	**1** Start the SpeechAttendant monitor and select the port you want to check. **2** Call SpeechAttendant and hang up as soon as it answers. **3** See how the status of the port changes. **4** Repeat for each SpeechAttendant port.
Result	On the SpeechAttendant monitor, the port should detect the "idle" state within 5 to 15 seconds.
Troubleshooting	Otherwise: ◆ Ensure that the correct PBX is selected in the configuration panel. ◆ Ensure that the PBX port is assigned the right class of service (LTDA, on some PBXs) and dial tone attribute. ◆ Ensure that the PBX port generates a dial tone: **1** Depending on the type of integration, connect an analog or digital telephone to the PBX port used by SpeechAttendant. **2** Call this port. **3** Answer the ringing phone, then hang up the phone from which you called. You should hear a dial tone on the telephone that received the call. SpeechAttendant can detect loop current drops (LC_DROP). The most efficient way to detect a hangup is if the PBX port can generate this signal when a user hangs up. If the system still does not detect the hangup, call the Customer Service Center.

DTMF transfers

Purpose	Assure that SpeechAttendant processes DTMF.
Procedure	◆ Call SpeechAttendant, then key in the number of any directory entry.
Result	SpeechAttendant should transfer the call.
Troubleshooting	Otherwise: ◆ In the configuration panel, verify that the **Maximum Number of Digits in Extensions** setting is correct (see page 95), to avoid SpeechAttendant ignoring extra digits. ◆ In Norstar systems, to generate DTMF during a single call, the Long Tone function must be activated. To do so, press **Feature button**, then dial **808**. ◆ Verify that the requested entry is associated with the right phone number and that **DTMF access** is checked. ◆ If the problem persists, call the Customer Service Center.

Call forward capabilities

Purpose	Assure that calls transferred by SpeechAttendant are re-directed to the appropriate destination.
Procedure	**1** Call SpeechAttendant from an internal telephone and ask for a directory entry whose phone is located nearby. **2** Let the phone ring until the call is forwarded to the voice mailbox. **3** Take the phone off-hook and call again (if the phone has two lines, make sure both are busy). **4** Program the target telephone to forward calls directly to the voice mailbox and call again. **5** Program the target telephone to forward calls to another destination and call again. **6** Repeat steps 1 to 5 by calling SpeechAttendant from an external line.
Result	All calls should end up in the voice mailbox or programmed destination. The results should be the same when calling from an internal telephone and from an external line.
Troubleshooting	On some PBXs, forwarding calls can result in the telephone not ringing at all. When this occurs, the beginning of the personal greeting may not play. To resolve this issue, select **Unsupervised** as the default transfer mode.

SpeechAttendant functions

To make sure that the SpeechAttendant application is functional, you must test the following:

- Transfer to the operator
- Transfer to directory entries
- Transfer to external and voice mail numbers (if applicable)
- Homophone resolution (if applicable)
- Maintenance mode functions
- Personal administration mode (PAM)
- Dynamic call routing (if applicable)
- Transfer to personal contacts (if applicable)
- Voice biometrics and private phones (if applicable and not previously tested and personal contacts)
- Language toggling (if applicable)
- Call forward
- Remote connection

The purpose, procedure, and desired results of each test is explained below, along with solutions to potential problems.

Transfer to the operator

Purpose	Assure that the operator extension is correctly programmed and can be accessed on voice command as well as using a speed-dial key.
Procedure	**1** From a relatively quiet environment, call SpeechAttendant and say "Operator." **2** Call again and press the operator speed-dial key (by default, the **0**).
Result	Your call should be routed to the operator in both cases.
Troubleshooting	Otherwise: ◆ Verify that the operator extension is correctly programmed in the configuration panel (see page 89). ◆ Verify that the operator speed-dial key is correctly programmed (see page 89).

Transfer to directory entries

Purpose	Assure that speech recognition works well and that calls can be transferred to any type of entry.
Procedure	**1** If the phone directory doesn't already contain menu, transfer and audiotex entries, create some (refer to the *Administration Guide*). Remember to activate your changes for the entries to be accessible on the phone. **2** From a relatively quiet environment, call SpeechAttendant and say the name of a transfer entry whose default phone number is an extension number. **3** Try again with a few other transfer entries. **4** Call again and ask for a sub-menu. **5** Call again and ask for an audiotex entry.
Result	Your calls should be routed to the correct destination, directly or after one or two confirmations. If this is the case, you can delete any temporary modification you made to the phone directory.
Troubleshooting	Otherwise, verify that all entry names are correctly spelled and that each entry is associated with correct phone numbers. If the problem persists, call the Customer Service Center.

Transfer to external and voice mail numbers

Purpose	Assure that SpeechAttendant can route calls to external and voice mail numbers (if this is required).
Procedure	**1** If the phone directory doesn't already contain entries associated with external (cell phone, home or office) and voice mail numbers as their default numbers, create some. Remember to activate your changes for the entries to be accessible on the phone. **2** From a relatively quiet environment, call SpeechAttendant and say the name of a transfer entry whose default phone number is external. **3** Call again and ask for a transfer entry associated with a voice mail number.
Result	Your calls should be routed to the correct destination, directly or after one or two confirmations. When asking for the entry associated with the voice mail number, you should hear a personal greeting. If this is the case, you can delete any temporary modification you made to the phone directory.

Transfer to external and voice mail numbers (***cont'd***)

Troubleshooting	Otherwise: ◆ Verify that all entry names are correctly spelled and that each entry is associated with an adequate phone number and phone type. Remember that external phone numbers must include any required prefix such as access and area codes. ◆ For external transfers, ensure that the prefixes for external numbers are correctly set (see page 76) and that the PBX ports are programmed to allow external calls. Test this by calling the external number using an analog telephone connected to the PBX port. ◆ For voice mail transfers, verify that the voice mail prefix and suffix are correctly set (see page 76). ◆ If the problem persists, call the Customer Service Center.

Homophone resolution

Purpose	Assure that SpeechAttendant can handle homophones.
Procedure	**1** If the phone directory doesn't already contain homophone entries, create two transfer entries with the same name. Each entry must be associated with a different department and phone number. Remember to activate your changes for the entries to be accessible on the phone. **2** From a relatively quiet environment, call SpeechAttendant and ask for the homophone name.
Result	The system should suggest one of the homophones using the requested name plus that person's department. If you accept this suggestion, your call should be transferred to this person. Otherwise, the system should suggest the second homophone. If this is the case, you can delete any temporary modification you made to the phone directory.
Troubleshooting	Otherwise: ◆ Verify that all entry names are correctly spelled and that each entry is associated with the right phone number and department. ◆ Using the Directory Assistant (accessible from the **Menu Editor** toolbar), validate that the homophones you created are detected as such (refer to the *Administration Guide*). ◆ Verify that the homophone resolution method is set to **Department** and that homophone resolution threshold is correct (see pages 92 and 93). ◆ If the problem persists, call the Customer Service Center.

Maintenance mode functions

Purpose	Assure that the maintenance mode is accessible either on voice command or using a speed-dial key, and that its functions are working properly.
Procedure	**1** In the phone directory, create a transfer entry for yourself. Make sure name recorder access is allowed. **2** From a phone close to the SpeechAttendant computer, call the system and say "Maintenance Mode." **3** Press **1**, enter the name recorder password, then follow the instructions to record your name. **4** Call again, and press the maintenance mode speed-dial key (by default, the ***** key). **5** Press **2**, then enter the maintenance mode password. **6** Referring to the *Administration Guide*, access the prompt recorder and create an emergency greeting. **7** If you have programmed a speed-dial key for accessing the name recorder or system administration functions directly, call again and press these keys.
Result	You should be able to access the maintenance mode and activate all of its functions.
Troubleshooting	Otherwise: ◆ Verify that the speed-dial keys and passwords are correctly programmed (see pages 89, 94 and 96). ◆ If access to the maintenance is blocked, reset the invalid password counter (see page 94).

Personal administration mode (PAM)

Purpose	Assure that the personal administration mode is accessible and that its functions are working properly.
Procedure	**1** In the phone directory, create a transfer entry for yourself, making sure you are allowed to use the name recorder and voice biometrics (if applicable). **2** From a quiet environment, call SpeechAttendant, then press ***99** to access the PAM. **3** Referring to the *Administration Guide*, record your name, create your voiceprint, redirect your calls and change your PIN (your default PIN is the one specified in the configuration panel, unless you entered a specific PIN in your directory entry).
Result	You should be able to record your name, create your voice print (if applicable) and change your PIN. You should also be given the option of redirecting your calls, but this will be tested later (see page 128).

Personal administration mode (PAM) (*cont'd*)

Troubleshooting	Otherwise: ◆ Verify that the PAM speed-dial key is correctly programmed (see page 89). ◆ Verify that **Name recorder** and **Voice biometrics** are checked in your entry. ◆ If you weren't able to change your PIN, go back to your directory entry and assign yourself a new PIN, then try again. ◆ If you experienced difficulty creating your voiceprint, try again from a more quiet environment. ◆ If the problem persists, call the Customer Service Center.

Dynamic call routing

Purpose	If the system allows dynamic call routing, assure that it recognizes DNIS and CLID and answers calls accordingly.
Procedure	**1** Referring to the *Administration Guide*, create three sub-menus named: ◆ "Testing internal CLID" ◆ "Testing external CLID" ◆ "Testing DNIS" (if applicable) It is not necessary to create any greetings or prompts, or modify any settings. Once you are done, don't forget to activate your changes. **2** Create an entry point named "Internal CLID," selecting "Testing internal CLID" as the home and main menus. In the **CLID** column of the **DNIS/CLID** tab, enter the extension number of the phone close to the SpeechAttendant computer. **3** Create a second entry point named "External CLID," selecting "Testing external CLID" as the home and main menus; in the **CLID** column of the **DNIS/CLID** tab, enter your cell phone number. **4** Create another entry point named "DNIS," selecting "Testing DNIS" as the home and main menus; in the **DNIS** column of the **DNIS/CLID** tab, enter the phantom number programmed as explained on page 15. **5** Using the internal phone that you programmed in the CLID internal entry point, call SpeechAttendant. You should hear "Testing internal CLID. Which person or service do you want to reach?" **6** From your cell phone, call SpeechAttendant. You should hear "Testing external CLID. Which person or service do you want to reach?" **7** From any other phone, dial the phantom number programmed in step 4. You should hear "Testing DNIS. Which person or service do you want to reach?"

Dynamic call routing (*cont'd*)

Result	Each of your calls should be automatically routed to the appropriate menu, which is confirmed by the greeting. If this is the case, you can delete any temporary modification you made to the phone directory.

Troubleshooting	Otherwise:
	◆ Check that your entry points are correctly programmed.
	◆ Make sure the phantom number is correctly programmed (see page 15).
	◆ Verify that DNIS and CLID are correctly deciphered by the system:
	  ◆ From **Admin Tools** window, start the monitor.
	  ◆ Repeat steps 5 to 7 of the above procedure, making sure the correct DNIS and CLID are displayed.
	  ◆ If this is not the case, modify the DNIS and CLID masks (see page 82).
	◆ If the problem persists, call the Customer Service Center.

Transfer to personal contacts

Purpose	If the SpeechContacts option is included in the license, this test assures that it is functional.
Procedure	**1** Ensure the cooperation of an employee who has a personal directory. **2** Have this person create a voiceprint in the PAM. **3** Referring to the *Administration Guide*, create this person's personal directory. **4** Ask your assistant to access the personal directory management Web site and integrate contacts into the personal directory (instructions can be accessed by clicking **Help**). **5** Ask your assistant to call SpeechAttendant and ask for one of the contacts that has just been integrated into the personal directory (to avoid disturbing people, your assistant should hang up as soon as the transfer is announced).
Result	Your assistant should be transferred to the requested contact.
Troubleshooting	◆ If your assistant cannot create a voiceprint, refer to the troubleshooting section of the personal administration mode test (see page 124). ◆ If your assistant cannot access the personal directory management Web site, verify that the network connection is correctly programmed (see page 10). ◆ If your assistant's Outlook contacts are not displayed, refer to the online help. ◆ If your assistant cannot reach contacts, make sure their names are properly spelled. ◆ If the problem persists, call the Customer Service Center.

Voice biometrics and private phones

Purpose	Assure that the voice biometrics module is working properly. **NOTE** This test is only required if the SpeechAuthenticator option is included in the license and if you haven't previously tested voice biometrics with personal contacts (see page 126).
Procedure	**1** Referring to the *Administration Guide*, create a menu protected with voice biometrics, adding your name to the list of people authorized to access it. Remember to activate your changes in the directory for the newly created menu to be accessible on the phone. **2** Call SpeechAttendant and ask for the protected menu. **3** Call again, but when prompted to identify yourself, say another name. **4** Ask someone to try accessing the menu using your identity. **5** In your transfer entry, designate your phone number as private. **6** From this extension, call SpeechAttendant and ask for the protected menu.
Result	You should be the only person granted to access the menu, and only when using your real name. When calling from your private phone, the system should not ask you to identify yourself.
Troubleshooting	Otherwise, try again from a more quiet environment. If the problem persists, call the Customer Service Center.

Language toggling

Purpose	In a bilingual system, assure that callers can select another language.
Procedure	**1** Call SpeechAttendant. **2** When the system answers: ◆ Press **9** (or the actual speed-dial key), if the language selection method is **Optional**. ◆ Say the desired language, if the language selection method is **Mandatory**.
Result	SpeechAttendant should repeat the greeting in the second language.
Troubleshooting	Otherwise, verify that the speed-dial key is correctly programed (see page 89).

Call forward

Purpose	Assure that the call forward function (also known as *Call ReDirect*) is working properly.
Procedure	**1** In your directory entry, enter your cell phone number in addition to the extension close to the computer. Remember to activate your changes. **2** Call SpeechAttendant and say "Call forward." **3** Follow the instructions to redirect your calls to your cell phone number. **4** Call SpeechAttendant again and ask for yourself.
Result	Your call should be transferred to your cell phone. If this is the case, delete your directory entry (if it was temporary) or call the system again to redirect your calls to your default phone number.
Troubleshooting	Otherwise: ◆ Verify that the second phone number specified in your directory entry is correct. ◆ Call again from a more quiet environment.

Remote connection

Purpose	Assure that SpeechAttendant can be accessed via pcAnywhere through dial-up connection.
Procedure	**1** From the remote computer, connect to the SpeechAttendant computer (for details, refer to pcAnywhere documentation). The desktop of the SpeechAttendant computer should appear. **2** From the **Admin Tools** window, access the configuration panel, monitor or phone directory and menu editor.
Result	You should be able to administer SpeechAttendant from the remote computer.
Troubleshooting	Otherwise, verify that the remote computer is correctly configured for pcAnywhere connection (see Chapter 8, page 105). If the problem persists, call the Customer Service Center.

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