



Interactive Response Overview

December 2008

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Contents

Chapter 1: Avaya IR	5
Chapter 2: Functionality	7
Call progress functionality	7
Circuit-based call progress	7
Packet-based (VoIP) call progress	10
Call handling example	13
Data exchange operations	16
Database access	16
Avaya Interaction Center Integration	16
Vesp_dip and the VOX server	17
Speech functionality	18
Speech detection	18
Greeting and prompting callers	18
Supported audio format	18
Caller speech recognition	18
Supported speech software	19
Supported transfers	19
Configurable ASR and TTS resources	20
TTS dictionary support	20
Fax functionality	21
Voice response functionality	21
How voice response works	21
Sample voice response transaction	22
Chapter 3: Features	25
Alarm administration	25
ASAI feature	25
Backup and restore	27
Symantec NetBackup support	28
ASG feature	29
Barge-in	29
Call data reports	29
Call functions	30
CTI data integration process	33
Converting DIP Into Service	34
Data collection	34
Dial Pulse Recognition	34
Dialout Alarming	35
Disk mirroring	35
Digital protocol - Loop Start	37
Digital protocol - PRI	38
Digital protocol - R2MFC	39
Digital protocol - Wink Start feature	40
DTMF type ahead	40
Echo cancellation	41
Enhanced Basic Speech	41
Fault Management Facility	43
Fax	43

Contents

Host interface.....	44
IC Integration.....	45
Call handling guidelines between the Avaya IR and IC systems.....	46
JDBC database interface.....	47
Logging.....	48
Message administration.....	48
MRCP overview.....	49
MRCP v1.....	50
MRCP v2.....	53
MRCP implementation considerations for IBM MRCP.....	56
MRCP load balancing.....	56
MRCP backup servers.....	57
Network Management (SNMP).....	57
Proactive Contact integration.....	58
ROI and TCO reports.....	58
Service creation environment.....	59
Service Management Facility.....	61
TDD Modem.....	63
Tone masking.....	64
Two B-Channel Transfer.....	64
Universal Call ID Expansion.....	64
Voice over IP.....	66
Voice over IP feature capabilities.....	67
VoiceXML.....	69
VXML log commands.....	71
VXML 2.1 support.....	71
VXI channel modes.....	73
vrop_tng.....	73
Web administration.....	74
WebLicensingManager.....	74
Web services.....	74
Avaya IR Web Services related commands.....	75
Avaya Web Services authorization and authentication.....	75
Chapter 4: Security.....	77
IR security.....	77
System access security.....	77
Using a firewall.....	77
Physically isolating the LAN.....	78
Disabling a network service.....	78
Disabling the telnet feature.....	79
Running the DisableServices utility.....	79
Encrypting CTI Password.....	80
Restricting administration permissions.....	80
Logging in as root.....	81
IR command permissions by login type.....	82
Chapter 5: Related documents.....	85
Chapter 6: Training.....	89
Index.....	91

Chapter 1: Avaya IR

The Avaya IR system is an interactive voice response system for automatic telephone transactions, which can be used in the following ways:

- In a stand-alone self service configuration, either directly connected to the Public Switched Telephone Network (PSTN), or connected to the PSTN behind an on premise Avaya or non-Avaya switch.
- With a contact center to provide seamless integration from self service to assisted service.
- With an outbound dialer to provide automated alerts and notifications.
- With TAS (proprietary), VoiceXML (open standard), and custom Independent Software Vendor applications.
- With TDM (circuit switched) and VoIP (H.323) based switches.

Using synthesized or prerecorded speech, the system can:

- Respond to (answer) an incoming call.
- Request specific information from the caller.
- Provide information or services to the caller based on data from the caller.

Depending on how the system is configured, callers can respond to information requested by the system in the following ways:

- Speech
- Touch Tone (DTMF) input
- Dial Pulse input
- Using a Telecommunications Device for the Deaf (TDD)

Chapter 2: Functionality

Call progress functionality

Circuit-based call progress

On circuit-based connections, call progress is managed through digital signaling, interaction with voice response applications and IRAPI commands and instructions. Call handling involves T1/E1 lines, the LAN, and telephony boards.

In circuit-based operations, calls may proceed:

- From a switch, such as a DEFINITY or MultiVantage system, to the IR system.
- Directly from the public switched telephone network (PSTN) to the IR system.

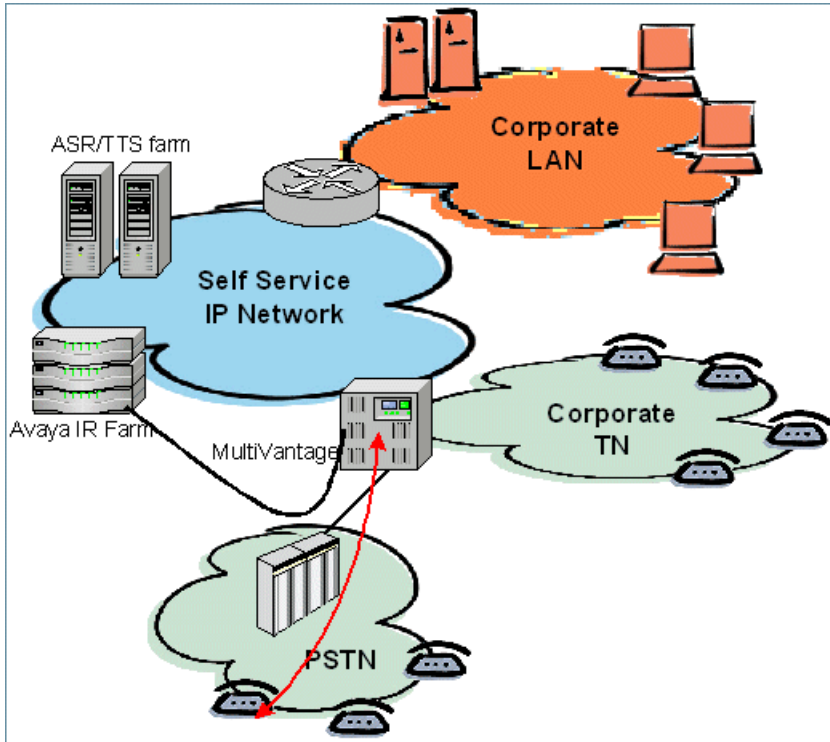
Circuit-based operations with a switch

When a switch is used, direct digital trunk connections are set up between the switch and the Avaya IR system. A variety of digital protocols are available.

Because an Avaya IR system is on the LAN, it can communicate with:

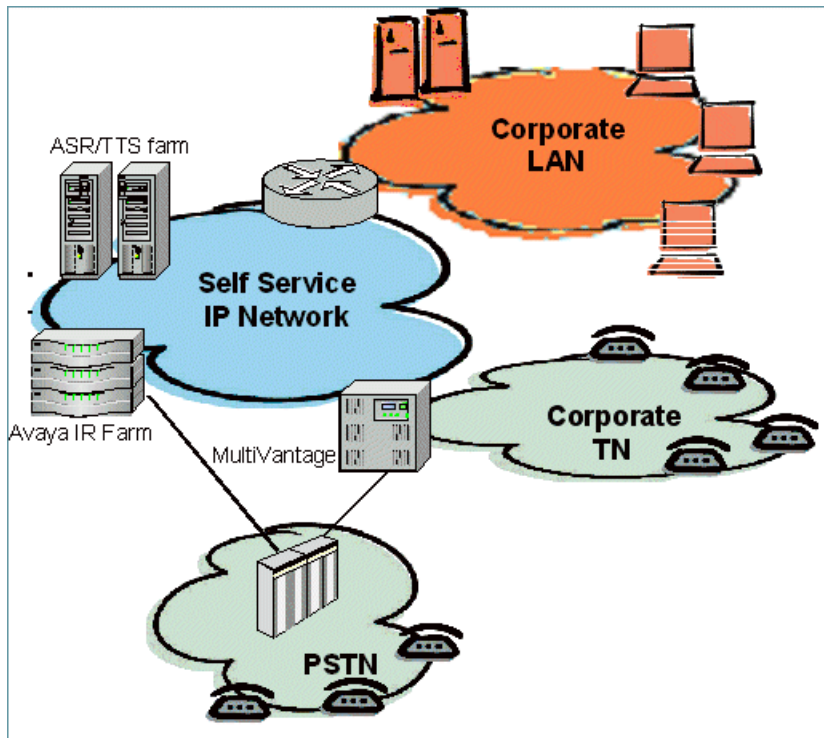
- Remote databases containing information requested by callers.
- Speech servers that provide features such as Text-to-Speech (TTS).

Recorded speech files may reside on the hard drive of the Avaya IR system, or on a speech server. The Avaya IR system retrieves information from databases and converts it into spoken information for callers. Calls may then go back to the switch for distribution. Interactions are controlled by voice response applications. Optional features, such as ASAI and CTI, enhance communications with switches and telephony systems. The following figure depicts a circuit-based configuration with a switch. Multiple Avaya IR systems are in use, and are referred to as a "farm."



Circuit-based operations without a switch

A circuit-based system can be configured so that calls from the PSTN come directly to the Avaya IR system. When configured this way, the Avaya IR system provides information requested by callers and handles call routing. The only difference is where the call comes into the enterprise. The functionality is the same. The following figure depicts this type of configuration.



Circuit-based call answer

The following steps are typically involved in the answering of a call:

1. If a switch is being used:
 - a. The incoming call is detected by the switch.
 - b. The switch connects the line for the incoming call to the Avaya IR system.
2. The Avaya IR system detects the incoming call as a signal on the digital telephone line.
3. The Avaya IR system checks the instructions in the application assigned to the telephone line to find out what it should do with an incoming call. It finds an answer instruction.
4. One of the processes running on the operating system executes the IRAPI answer instruction. The execution involves sending a command to the digital telephony board, which tells it to send an off hook signal through the telephone line.

Related topics:

[Circuit-based call termination](#) on page 10

[Circuit-based call transfer](#) on page 10

Circuit-based call transfer

Transferring a call involves changing to whom or to what the caller is connected.

The application that is assigned to the channel associated with the telephone line carrying the call initiates a transfer instruction. The instruction may be a response to an event (such as a caller asking to talk to a live operator), or may simply be the next in a set of instructions (for example, transferring to an agent following the collection of an account number from the caller).

The application instruction results in a process being run on the system executing an IRAPI command. Depending on the command (and whether the features required to support the command are installed and configured), the system either performs the transfer itself, or tells the switch to transfer the call. If the system performs the transfer, the transfer is done by the telephony boards.

Related topics:

[Circuit-based call answer](#) on page 9

[Circuit-based call termination](#) on page 10

Circuit-based call termination

Ending (or terminating) a call is done by either a caller going on hook or by the Avaya IR system.

The application that is assigned to the channel handling the call initiates a termination instruction. The instruction is typically in response to an event (such as a caller selecting a menu option for ending the call), or the last in a set of instructions (for example, ending the call following the speaking of information to the caller).

The application instruction results in a process being run on the system executing an IRAPI command. Depending on the command (and whether the features required to support the command are installed and configured), the system terminates the call through the telephony board or tells the switch to end the call.

Related topics:

[Circuit-based call answer](#) on page 9

[Circuit-based call transfer](#) on page 10

Packet-based (VoIP) call progress

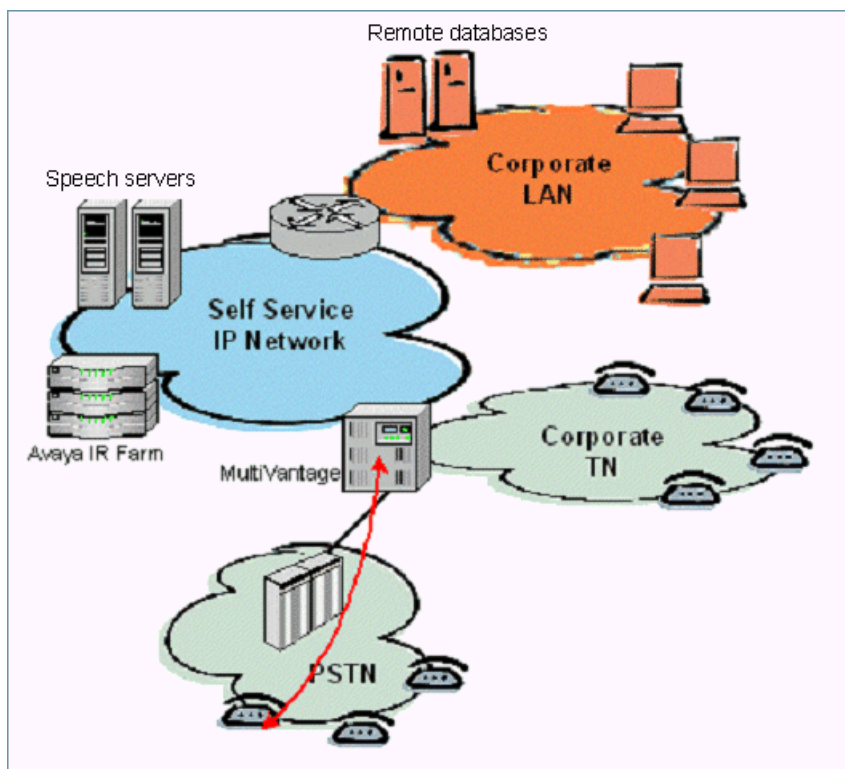
On packet-based configurations, call progress is managed through interaction with voice response applications and communication of messages and events between the DEFINITY or MultiVantage system and the Avaya IR VoIP subsystem. Call progress involves the LAN, with

no use of T1/E1 lines or telephony boards. The DEFINITY or MultiVantage system acts as the Gateway to the VoIP subsystem.

With the Voice over IP (VoIP) feature, voice data is transported in small packets using the same Internet Protocol (IP) as corporate LANs and the Internet. In a VoIP configuration, a DEFINITY or MultiVantage system is always present.

The VoIP subsystem may be placed on the corporate LAN (the same network that is used for e-mail and document transfer). However, a dedicated network is generally used to connect the switch to the Avaya IR system. Using a dedicated network guarantees voice quality without using complex bandwidth management technologies.

In the VoIP configuration, the switch is the gateway between the public and corporate telephony network and the Avaya IR system or systems. The following figure depicts a VoIP configuration.



Packet-based call answer

The following steps are typically involved in answering a call in a packet-based configuration:

1. The DEFINITY or MultiVantage system (switch) detects a new call.
2. The switch selects a VoIP port to an Avaya IR system based on the dialed digits.
3. The switch forwards the call to the Avaya IR VoIP subsystem.
4. The VoIP subsystem detects a new call and:

- a. Identifies the correct Avaya IR VoIP channel to use for the call.
 - b. Sends a message to the process that owns the channel, informing the process that a new call has arrived on the channel.
5. The process that owns the channel instructs the VoIP subsystem to answer the call.
By default, the Application Dispatch (AD) process owns all free channels and starts the other processes required to handle calls.
6. The VoIP subsystem answers the call.
Once the call is answered, there is a media path between the caller and the Avaya IR system.
7. Acting as a Gateway, the switch sends and receives media to and from the Avaya IR system in packets over the LAN.
8. Standard IR system call processing continues until the call ends.

Packet-based call transfer

The following steps are typically involved in transferring a call in a packet-based configuration:

1. To transfer calls on VoIP channels, the VoIP subsystem:
 - a. Places the primary call on hold.
 - b. Places a secondary call to the DEFINITY or MultiVantage system, which acts as the Gateway.
 - c. Forwards DNIS, ANI and UUI information, if it is available.
 - d. Sends an instruction to the DEFINITY or MultiVantage system to establish the transferred call.
2. The DEFINITY or MultiVantage system completes the transfer and clears the secondary call.
3. The VoIP subsystem clears the primary call.

Packet-based call termination

The following steps are typically involved in answering a call in a packet-based configuration:

1. When a caller hangs up, the DEFINITY or MultiVantage system sends a message to the VoIP subsystem.
2. The VoIP subsystem notifies the voice response application to disconnect the call.

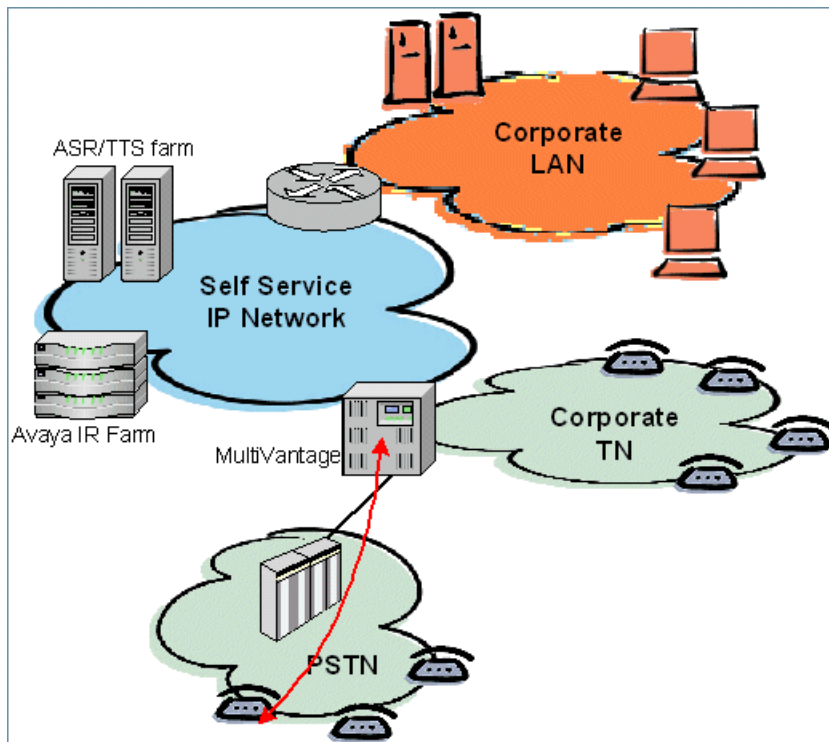
Additionally, voice response applications can terminate calls on VoIP channels based on conditions identified in the application, such as lack of response from the caller. In that case, a message is sent to the DEFINITY or MultiVantage system to terminate the call.

Call handling example

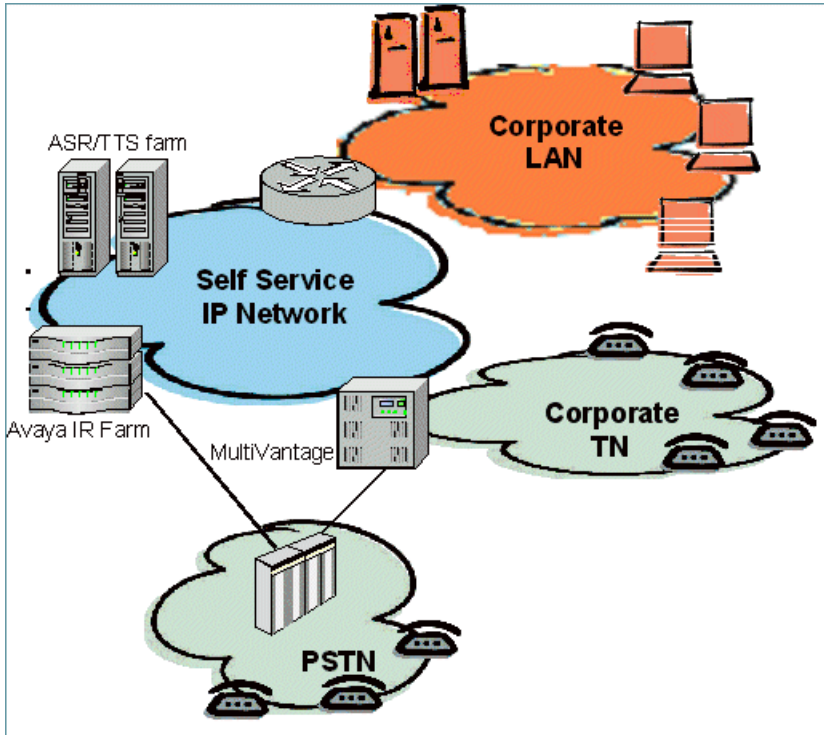
This example shows how the Avaya IR system interacts with other systems to handle calls. Operations for various Avaya IR system configurations are depicted.

1. Call comes into the switch.

A caller from the public switched telephony network (PSTN) dials a telephone number. The PSTN routes the call to the switch associated with the number and establishes a circuit-based connection between the caller and the switch. The following figure shows a call coming into the switch.

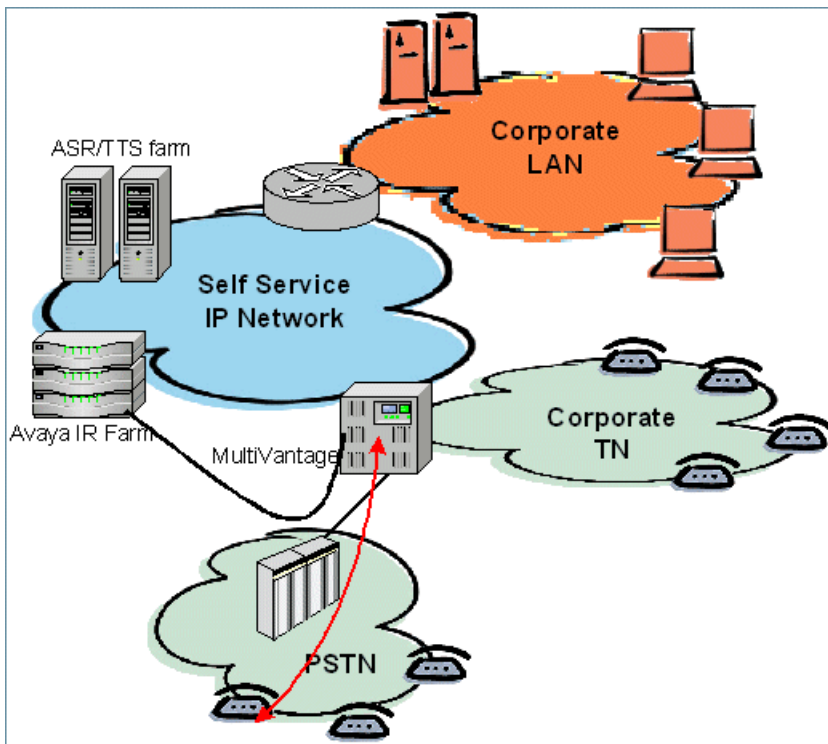


Another possibility is for calls to go directly to Avaya IR system or farm. The following figure shows the configuration without a switch.

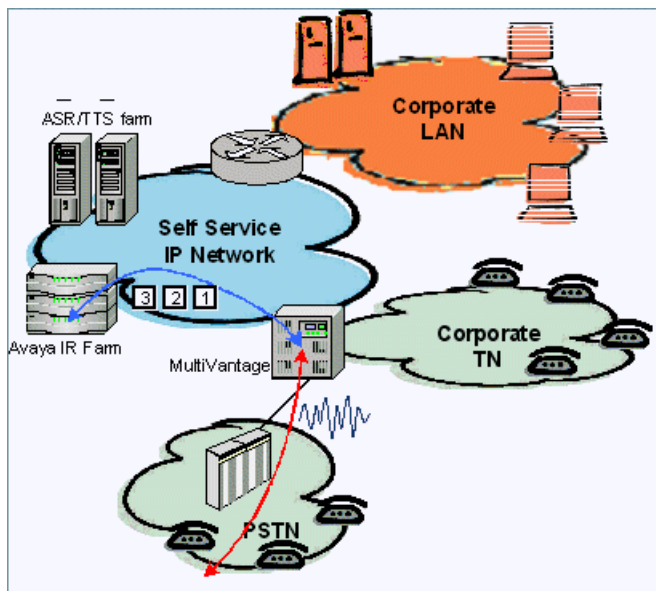


2. Call goes from switch to an Avaya IR system.

When a switch receives a call from the PSTN, it establishes a call to the Avaya IR system farm. In a circuit-based configuration, connection is through T1/E1. The figure that follows shows a call being sent to a Avaya IR system farm through a direct connection.



If the VoIP feature is in use, the voice data for the call is broken up into packets and sent over the LAN to the Avaya IR system. The figure below shows an Avaya IR system and the speech servers on a separate LAN segment.



3. Avaya IR system interacts with caller.

When a call reaches the Avaya IR system, the voice response application associated with the called number interacts with the caller by playing a prompt. The caller responds by entering DTMF digits or by speaking. Interpretation of speech requires the Speech Recognition feature.

4. Speech is interpreted, if necessary.

If the voice response application uses speech recognition instead of DTMF digits to collect data, the Avaya IR system establishes a connection to a server where a speech recognition application resides. The recorded voice segments are forwarded to the server for processing.

5. Data is extracted and returned to the Avaya IR system.

When the information provided by the caller is in a form that can be interpreted, it is passed to the appropriate database. An Oracle database may reside on the hard drive of the Avaya IR system. In most cases, however, the database resides on the LAN, which is used to complete the data transfer.

6. The Avaya IR system speaks information to the caller.

The Avaya IR system interprets information from the database into spoken form and the application plays it for the caller. With TAS voice response applications, Enhanced Basic Speech can be used to convert information from the database into spoken form. With VXML voice response applications, the Text-to-Speech feature is highly recommended for converting information.

7. Self-service ends.

If the caller hangs up, the PSTN clears the call to the system that received it--the switch or the Avaya IR system. If the calls clear first at the switch, it will, in turn, clear the call from the switch. If the call is transferred, the transfer process may be handled by the switch or by the Avaya IR system.

Data exchange operations

Data exchange involves the following.

Receiving data from a caller: When a caller inputs touch tones, the content of the response is placed into a system variable. It can also be placed into custom variables. When a caller speaks, speech recognition software converts the voice into data. This data is analyzed by the speech recognition servers and the relevant parts are extracted. This data is also placed into variables.

Putting data in a database: Applications on the IR system can define database tables and place content from them into database table fields.

Retrieving data from a database: Applications on the IR system can retrieve data from database tables.

Database access

An ORACLE database may reside directly on the Avaya IR system. ORACLE databases, as well as other types of JDBC-compliant databases, may also reside on servers outside the system, with access via the LAN. When remote servers are used, an Avaya IR system can interact simultaneously with up to five databases using various database applications. The JDBC feature provides database connectivity and activity on the Avaya IR server and to remote servers.

Related topics:

[JDBC database interface](#) on page 47

Avaya Interaction Center Integration

Avaya Interaction Center (IC) is a comprehensive multimedia contact center solution for customer relationship management (CRM). It routes, manages, records, and reports on customer interactions across communication channels including Web, e-mail, and telephony.

Avaya Computer Telephony for IC supports the Avaya IR system, enabling the Avaya IR system to communicate with both the telephone caller and the Avaya IC computer system.

With IC Integration, you can develop IVR-D applications that perform actions such as:

- Retrieve values stored in a database through Avaya Computer Telephony
- Set values in a database through Avaya Computer Telephony

- Generate Avaya Computer Telephony alarms
- Invoke a subset of the transactions defined by the Avaya IC Workflow Designer for a specific site

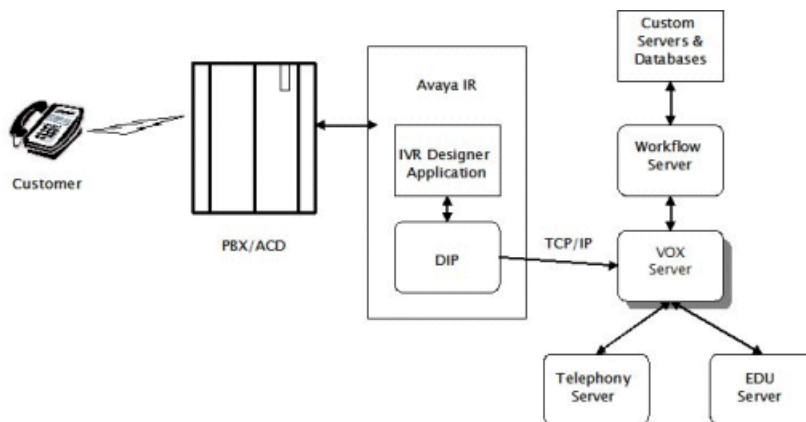
While the application is running, it has synchronous access to Avaya Computer Telephony for IC services. For example, an application can retrieve information from a database in response to a caller's request, and then read that information to the customer.

An application can also allow callers to hear messages recorded in their native languages. If the customer's language preference is identified in the database, you can issue the Avaya Computer Telephony external function `getvox` to find out which language the Avaya IR system should speak to the caller. Then, if you have recorded messages in several languages, you could call selected vocabulary items from the appropriate foreign language vocabulary file or play text-to-speech prompts in the appropriate language.

Vesp_dip and the VOX server

The `Vesp_dip` installed on the Avaya IR system provides the means to access Avaya Computer Telephony services such as data retrieval and call transfers. It enables the Avaya IR system to interact transparently with Avaya Computer Telephony for IC across the network.

The following figure depicts the data flow in an IC integration.



Avaya IR applications invoke `Vesp_dip` functions to communicate with Avaya Computer Telephony. Requests for Avaya Computer Telephony services pass through a TCP/IP connection into the VOX Server, which interprets the DIP commands. The VOX Server passes the application requests to the appropriate Avaya Computer Telephony components and returns the responses in a form that the Avaya IR system can understand. The commands issued to the VOX Server are described in detail in the *Avaya Interaction Center VOX Server Programmer's Guide*.

The functions contained in the DIP are described in the `IVR DesignerHelp`.

Speech functionality

Speech detection

The telephony card has on-board speech detection. Processing is handled through a proxy interface to the Natural Language Speech Recognition (NLSR) Proxy. G.711 speech encoding is supported.

Greeting and prompting callers

Applications on the Avaya IR system determine how callers are greeted and prompted for information or choices. Greetings and prompts may be played using recorded messages or Text-to-Speech (TTS), which converts text phrases to spoken phrases.

TTS allows you to include speech in an application using ASCII text as input, which is converted to synthesized speech. TTS can also convert text retrieved from a database or host. TTS is an alternative to using prerecorded phrases for voice response.

The TTS technology can distinguish between different classes of text, such as zip codes and telephone numbers, and pronounces the text string in the appropriate spoken format. When constructing speech, parameters such as pitch and duration are adjusted to make the outcome sound more natural. The ASCII text is preprocessed to expand abbreviations. For example, "Dr." is expanded to "doctor" or "drive," depending on the context.

TTS application development is supported by IVR-D. Speech processing is done using one or more auxiliary computers connected to the Avaya IR system through a proxy interface.

Supported audio format

Avaya IR supports the following .wav audio file type for speech and recording:

- Audio format: G.711 mu-law
- Sample rate: 8 kHz
- Channels: Mono

Caller speech recognition

In an interactive voice response (IVR) system or application, speech recognition, sometimes called automatic speech recognition or advanced speech recognition (ASR), is the ability of

an IVR system to recognize spoken responses from a caller and either convert the responses to text or use the results to initiate some system action. On the Avaya IR system, speech recognition is provided by the Speech Proxy package (AVsproxy).

Supported speech software

Product	Interface	Proxy	Type
Quantum ASR (Nuance Recognizer 9.0) / NSS 5.0	mrCP-v1-d04	AVmrcpasr	mrCP-scansoft
Quantum TTS (Nuance RealSpeak 4.5) / NSS 5.0	mrCP-v1-d04	AVmrcptts	mrCP-scansoft
Quantum ASR (Nuance Recognizer 9.0) / NSS 5.0	mrCP-v2-d15	AVmrcpv2asr	mrCP-v2
Quantum TTS (Nuance RealSpeak 4.5) / NSS 5.0	mrCP-v2-d15	AVmrcpv2asr	mrCP-v2
Scansoft OpenSpeech Recognizer 3.0.9/SWMS 3.1.9	mrCP-v1-d04	AVmrcpasr	mrCP-scansoft
Scansoft SpeechSecure	SOAP	SOAP	-
Scansoft RealSpeak 4.0.12	mrCP-v1-d04	AVmrcptts	mrCP-scansoft
Nuance 8.5	Native APIs	NPC 3.7	-
Nuance Verifier	Native APIs	NPC 3.7	-
Nuance Vocalizer 4.0.6	Native APIs	NUANvtpxy-v1.1	-
IBM WVS 5.1.3.1	mrCP-v1-d04	AVmrcpasr/ AVmrcptts	mrCP-ibm

Supported transfers

IR supports different transfer types on different protocols.

Protocol	Blind transfer	Bridge transfer	Consultation transfer
T1/E1 Loop	Supported	Supported	Supported
T1/E1 Pri	Not supported	Supported	Supported with QSIG, E1 ETSI, and T1 4ESS/5ESS.
T1 Wink	Not supported	Supported	Not supported
E1 R2MFC	Not supported	Supported	Not supported

Protocol	Blind transfer	Bridge transfer	Consultation transfer
VoIP	Supported	Supported	Supported only with patch SA 8874 for Communication Manager 3.1 and later.

Configurable ASR and TTS resources

You can configure how speech resources are allocated in a call.

You are licensed for a specific number of speech resources. The ASR and TTS resource allocation keys allow you to more effectively manage your speech resources. You must use the resource allocation keys in the `mrcp.cfg` and `pal.cfg` files to configure ASR and TTS resource allocation for VXML applications using MRCP interfaces. You cannot use the `mrcp.cfg` file and `pal.cfg` file to configure resource allocation for non-MRCP or proprietary interfaces.

TTS dictionary support

TTS dictionary support is available for MRCP proxies. TTS dictionaries are supported by loading the dictionary onto the TTS server.

Methods for loading dictionaries

You can load a TTS dictionary by:

- Adding vendor specific parameters to the `mrcp.cfg` file
- Adding a lexicon tag in the SSML for the prompt in a VXML application

Note:

You can also load a TTS dictionary on a Scansoft RealSpeak TTS server. You must use the `default_dictionaries` element tag, and the `dictionary_default_path` element tag in the RealSpeak TTS servers `config\ttserver.xml` file.

Considerations for TTS dictionary loading methods

You must consider the following before you load your TTS dictionaries:

- Dictionaries loaded through the lexicon tag are limited to one session, but this allows for dynamic manipulation of the dictionary during the call.
- Continuous use of dictionaries loaded through the lexicon tag can impact system performance.
- Dictionaries loaded with vendor specific parameters in the `mrcp.cfg` file persist through multiple sessions, but the dictionary cannot be manipulated dynamically.
- Dictionaries loaded with vendor specific parameters must be enabled in each session.
- Dictionaries loaded with vendor specific parameters allow you to load and enable a single dictionary for all applications on all channels.

Fax functionality

When configured as a circuit-based system, the Avaya IR system supports sending and receiving faxes. On a packet-based Avaya IR system, fax services are not supported, and the switch must be used to send and receive faxes.

Fax capabilities on the Avaya IR system are supported by NMS NaturalFax software and make use of the NMS digital telephony cards.

The Fax feature can be invoked:

- From applications written with fax external functions and assigned to Avaya IR channels
- Using command line user tools

The Fax Actions package includes the fax external functions and tools.

The NGFAX software from NMS includes external functions that can be called from applications on the Avaya IR system to:

- Create cover pages
- Queue and send faxes
- Receive faxes

Voice response functionality

How voice response works

The Avaya IR system allows either full or partial automation of telephone transactions that would otherwise be performed by an operator or attendant. These automated transactions are known as applications. Each application is designed and developed to meet a specific customer's need. An application script is a set of instructions written for the system that informs it how to carry out the automated transaction. Scripts define the flows of calls and determine what callers hear and how callers respond to the system.

When an incoming call is connected to the system:

1. The system prompts the caller with synthesized or prerecorded speech
2. The caller responds by entering touch-tones or by speaking into the telephone.

The dialog between the system and the caller is determined by the particular application and its corresponding script.

The application script can be simple or complex, depending on the purpose of the call. For example, a simple script may accept a caller's request for information, perform a quick search of a local database, and respond to the caller with that information. A more complex script can accept a caller's request for information, prompt the caller to provide additional touch-tone or spoken format information, and access a remote host computer database to retrieve information related to the caller. The script forwards the call to an operator or attendant who uses the information previously acquired to respond and interact with the caller.

Related topics:

[Sample voice response transaction](#) on page 22

Sample voice response transaction

This sample voice response transaction describes a simple, non-automated transaction between an operator and caller and then describes how the system can automate that same transaction.

In this sample, a bank has several operators whose duties include providing callers with certain information, such as account balances and current interest rates for different types of accounts. The operators also answer a variety of questions. Some of the information, such as interest rates, is located on a sheet of paper in front of the operator. Other information, such as account balances, must be obtained from the bank's computer. When necessary, these operators can also transfer callers to specialized customer service representatives for further information. Finally, when the caller requests information on automobile loans, the operator must transfer the call to a loan officer who has the means to further investigate and service the inquiry.

Think in terms of the application script replacing the operator when you automate a transaction using the system. The transaction steps remain the same, but the caller interacts with the system instead of an operator. The system follows the same basic steps as the non-automated transaction. It uses the application script for instructions about the setting, what to say during a transaction, and possible options the caller may wish to investigate.

In an automated call, you can see the following types of interactions between the caller and the system:

1. The system greets the caller.
2. The system prompts the caller and receives a request for information. In the sample call, this includes interest rate and account balance information.
3. The system takes the following action on the caller request:
 - a. If necessary, the system prompts the caller for further information (type of rate, type of account, ID number, and so forth). In this sample call, the caller is asked to enter the last four digits of his or her social security number.
 - b. The system looks up the information from the bank database.
 - c. The system reports the information to the caller.
 - d. The system repeats steps 2 and 3.

4. The caller requests information that can only be provided by a non-automated operation.
5. The system transfers the caller to a loan office representative.

Sample conversation

The following table represents a typical conversation, or transaction, between a caller and an operator at a bank.

Sample Non-automated Transaction	
Operator:	Thank you for calling River Bank. How may I help you?
Caller:	What is the current interest rate on your automobile loans?
Operator:	(Referring to a chart of interest rates) The interest rate for our auto loans is 7.9%. May I help you with anything else?
Caller:	Yes. I'd like to check my savings account balance.
Operator:	What is your account number?
Caller:	My account number is 0653202782.
Operator:	To verify that this is your account, what are the last four digits of your social security number?
Caller:	9087
Operator:	One moment, please." (Accessing account balance using a computer terminal) Your savings account balance is \$2,010.27. May I help you with anything else?
Caller:	Yes. I'd like to speak to someone about an auto loan.
Operator:	I'll transfer you to one of our loan office representatives. One moment, please. (Transferring caller to loan officer, who must access the computer to gain the customer's credit history)

Sample transaction process

To understand how calls are automated, break down the non-automated call transactions into its more basic steps. In the non-automated sample call above, you can see the following types of interactions between the caller and the operator:

1. The operator greets the caller.
2. The operator prompts the caller and receives a request for information. In the sample call, this includes interest rate and account balance information.
3. The operator takes the following actions at the caller's request:
 - a. If necessary, the operator prompts the caller for further information (type of rate, type of account, ID number, and so forth).
 - b. The operator looks up the information.

- c. The operator reports the information.
- d. The operator repeats steps 2 and 3.
4. The caller requests information that can only be provided by someone other than the operator.
5. The operator transfers the caller.

Virtually all transactions consist of the basic steps listed above. The caller's "request for information" shown in step 2 above may be as simple as the need to hear a checking account balance or the latest stock market value. More complex requests might include placing a sales order or requesting information on a particular product via a fax.

In the sample banking transaction, when the caller asks for an interest rate, the operator simply looks at a chart and reads the information to the caller. However, when the caller wants to know account balance information, the operator must ask for additional information (the caller's account number and social security number), and use a computer terminal to enter the caller information and read the balance displayed on the screen.

Types of interactions in the sample transaction

When you automate a transaction, think in terms of the application script replacing the operator when you automate a transaction using the system. The transaction steps remain the same, but the caller interacts with the system instead of an operator. The system follows the same basic steps as the non-automated transaction. It uses the application script for instructions about the setting, what to say during a transaction, and possible options the caller may wish to investigate.

In an automated call, you can see the following types of interactions between the caller and the system:

1. The system greets the caller.
2. The system prompts the caller and receives a request for information. In the sample call, this includes interest rate and account balance information.
3. The system takes the following action on the caller request:
 - a. If necessary, the system prompts the caller for further information (type of rate, type of account, ID number, and so forth). In this sample call, the caller is asked to enter the last four digits of his or her social security number.
 - b. The system looks up the information from the bank database.
 - c. The system reports the information to the caller.
 - d. The system repeats steps 2 and 3.
4. The caller requests information that can only be provided by a non-automated operation.
5. The system transfers the caller to a loan office representative.

Related topics:

[How voice response works](#) on page 21

Chapter 3: Features

Alarm administration

The Avaya IR system routinely tracks, records, and reports the occurrence of events so that service personnel can troubleshoot problems. Most of this reporting is done through the writing of messages to destinations (typically log files) on the system. Some events are associated with known problems, and are reported as alarms to service personnel through the system's external modem (see Dialout Alarming).

Alarm administration is a base feature that allows you to administer system alarm functions and to manage events and thresholds from a local console or a remote location. Users can view, add, change, and delete events, event thresholds, and actions that occur when event thresholds are reached. Administrators have the same privileges as users, as well as the ability to add, change, and delete user privileges. It is recommended that only Avaya service personnel administer alarms and messages.

When the system generates an alarm, the alarm is considered active. If you want the system to dial out the next time it generates an alarm, you must first retire the alarm.

Requirements

- The alarm administration feature requires an analog modem or Point-to-Point Protocol (PPP) that is supported over serial connections.
- The Base system package (AVir) is required to support Alarm Administration.

ASAI feature

Adjunct/Switch Application Interface (ASAI) is a digital signaling interface that provides a LAN or VoIP interface between a MultiVantage switch and adjuncts. The optional ASAI feature:

- Routes calls to agents based on information from a database.
- Delivers account information or caller profiles to the agent terminal at the same time as the call.
- Provides the ability to adjust application parameters.

With ASAI, the voice system can monitor and route calls on the switch. This interface operates over an Ethernet TCP/IP link connected to MAPD in CVLAN mode. When the ASAI interface is used in conjunction with digital line side T1 or E1 loop start interfaces or VoIP, the voice system can monitor and control incoming calls. It also supports access to ANI and DNIS and supports ASAI transfer, which is faster and more reliable than a flash transfer.

The ASAI feature includes the following capabilities.

Capability	Description
Universal Call ID (UCID)	UCID provides a unique identifier (8-byte binary or 20-character ASCII) for every call in a MultiVantage call center customer environment for uniform data-tracking for all call-related data in a call center, regardless of the system. MultiVantage uses the ASAI interface to pass the UCID to adjuncts.
ANI Information Indicator (ANI-II)	ANI-II provides a number that indicates the class of service of the customer who is calling, such as residential, coin, or wireless.
User-to-User Information element (UUI)	With UUI, the customer can specify additional information to be passed in external function arguments, which can contain up to 96 bytes of information.
CallVisor	CallVisor libraries are supported over a TCP/IP stack on an Ethernet LAN.

The full CallVisor CVLAN client of ASAI interface software is also provided with the ASAI feature package to facilitate building ASAI applications in C code. Avaya Professional Services provides development expertise in ASAI and the system, and other independent software vendors (ISVs) can develop custom applications using the ASAI API, thereby providing the optimum solution when you require full ASAI integration with the application.

Requirements

- The ASAI feature requires a MultiVantage switch.
- The ASAI package (AVasai) is required to support the ASAI feature.

Setup considerations

An ASAI link between the MultiVantage switch and the Avaya IR system delivers control and supervisory messages about each loop start T1 channel. The link must be implemented as an Ethernet connection. One ASAI link is supported per IR system.

- In a typical Avaya IR system and MultiVantage configuration, the public network provides a PRI connection to the MultiVantage switch for an application to receive calling number information. To establish an ASAI link between a MultiVantage switch and an Avaya IR system, connect one of the integrated LAN connections on the Avaya IR system to a MAPD circuit card in the MultiVantage switch.
- To make connections from the Avaya IR system to the switch, provision ASAI with loop start T1, loop start E1, or VoIP to support digital connections between the Avaya IR system and the line side of the switch. With this type of connection, various switch features that are not compatible with an ordinary T1 trunk connected between the Avaya IR system and switch can be used. These features include call transfer and call

progress tone (CPT) detection, either in conjunction with CCA or where an E1/T1 interface circuit card is used for communications.

Backup and restore

The backup and restore feature is part of basic system functionality that allows you to:

- Recover a single failed disk
- Recover a single Solaris file
- Backup all system files
- Incrementally backup specified changed files
- Back up and restore over a tape drive and over a LAN connection
- Verify backup while the system is operational
- Schedule backups to run automatically during off hours
- Delete scheduled backups

The maximum single file size that you can be backup using Interactive Response backup utility is up to 4 GB. The backup utility uses Solaris command `flarcreate` to create an archive. This command internally uses "cpio" utility which is the default copy method and supports individual file archives up to 4 GB. This limits you to backup files of size up to 4 GB each only.

About full system backups

A full system backup includes file structures, configured users and passwords, recorded speech, flat files, and application data. If a complete system failure occurs, you can use the backup data to reestablish a base functional system configuration.

Avaya strongly recommends that you create a full system backup:

- After the system is first installed, configured, and tested
- Weekly as part of regular system maintenance
- After installing or removing software packages

About partial system backups

You can use partial backups for disaster recovery or mistake recovery. Partial backups can include:

- A list of specified files
- All files in a specified directory
- All files that have changed in a specified time frame within the last 90 days

Avaya recommends that you perform partial system backups daily to record system changes.

About restoring the system

You can restore system data from any full or partial backup. You can only restore system data to a fully operational system.

About e-mail notifications

The e-mail notification service for backups or restores requires the Sendmail service. Sendmail can be disabled as a security hardening measure which will cause e-mail notifications to fail. If you want to have e-mail notifications for backups or restores, Avaya recommends that Sendmail be configured to only send outgoing mail. You should not use Sendmail as an incoming mail server.

Sendmail security vulnerabilities can be hardened in the `/etc/mail/sendmail.cf` file by setting the `DaemonPortOptions` attribute to a value of `Addr=127.0.0.1`.

If you do not use the e-mail notification service, you must disable the Sendmail startup script and the Sendmail SMTP and submission ports.

Refer to the Sendmail documentation provided by Sun for more information on configuring the Sendmail service.

Requirements

- The Base system package (AVir) is required to support backup and restore.
- The Backup and Restore package (AVbackrst) is required to support the Backup and restore feature.

Symantec NetBackup support

Avaya IR is certified with Symantec NetBackup 6.5 MP3. The scope of the certification has the following limitations:

- There should be no conflicts between IR and NetBackup Solaris 10 Agent on the network port usage.
- IR should continue to function properly when NetBackup Solaris 10 Agent is installed. The installation of the Agent should not degrade the performance of IR.

IR is not certified to support the following:

- NetBackup BMR restore capabilities
- NetBackup Oracle Agent software

Avaya recommends that you stop the voice system when you are using Symantec Netbackup to perform a backup.

Note:

Avaya cannot be responsible for any NetBackup related issues. NetBackup related issues should be referred directly to Symantec.

ASG feature

Access Security Gateway (ASG) is a base feature that allows you to ensure that only authorized persons have access to the system.

ASG uses an authentication protocol to confirm the validity of users. Successful authentication is accomplished when ASG-enabled devices communicate with compatible keys. The authentication transaction consists of a challenge (essentially a random number), which must be followed by the receipt of a response entered by the user. At the core of this scheme is a secret key, which is information possessed by the system and an authorized user. Interception of the challenge or the response will not compromise the security of the system, as the relevance of the authentication token used to perform the challenge and response is limited to the current challenge and response exchange.

Requirements

The Base system package (AVir) is required to support the ASG feature.

Barge-in

The barge-in feature is also sometimes referred to as allow interrupt or recognize during prompt. This feature allows callers to interrupt a voice prompt using speech or a touch-tone input on the telephone keypad. Speech recognition accepts either speech or touch tone input in response to a prompt. Barge-in operates for speech much like the talk-off option does for touch tone input. Barge-in is supported using echo cancellation supplied by the telephony cards

The VoiceXML interpreter supports selective barge-in, which enables a response to predefined keywords, called hotwords. A hotword can be voice or touch-tone input. When a user interrupts a voice prompt using a hotword, the current voice prompt stops playing and the user is directed to the next prompt of the voice application. Selective barge-in enables a response to predefined hotwords in the speech proxy of a speech recognition server.

Call data reports

The Call Data Handler (CDH) is a software process that accumulates data associated with calls, ports, resources, and application events.

The following call data reports are available:

Call Classification Data Summary: Provides information about each telephone number dialed by the Avaya IR system and the outcome for call transfers for the specified time interval.

Call Data Detail: Provides detailed information about incoming calls including call duration and the service (application) and channel that handled the call.

Call Data Summary: Provides an hourly summary of incoming calls by service rather than on a per-call basis as in the Call Data Detail report.

Call Traffic: Provides information about call traffic and call occupancy including the number of incoming calls and the percentage of time that channels were occupied for the specified time period and the average holding time.

Message Log: Provides information about Avaya IR system alarms and events including severity level, date and time the alarm or event occurred, source of the message, and card and channel numbers.

The Avaya IR system uses database DIPs to communicate with local and remote databases. These DIPs need to be configured with information regarding where and how to access the databases. By default, call information is stored in Call Data Handler (CDH) text files on the Avaya IR system. You can also administer the system to store CDH records in a local Oracle database. If the Avaya IR system does not have a local Oracle database, the CDH text files can be uploaded to a remote database. The upload can be scheduled to run at a convenient (low traffic) time using a cron job. You can administer both File CDH and Database CDH configurations but only one configuration can be active.

Requirements

- The JDBC Integration package (AVjdbcint) is required to store call data in a local or remote database.
- The Base system package (AVir) is required.

Call functions

Call functions include nodes and script instructions that allow you to build voice applications.

The following call functions are supported by IR, and are included as part of the base software. For each call function, the associated node and script instruction are listed. Voice@Work and IVR Designer use nodes as the basic building blocks to create a voice application. Script instructions are part of the TAS instruction set.

Call function	Capability
Announce	Speaks a message to a caller. Up to 15 phrases, values, or lines of text (from Text-to-Speech) may be played in succession in a single Announcement node.
Answer	Answers a call (takes a line off-hook). This can be accomplished through the tic('a') script instruction or the Answer Call node.
Blind Transfer	Transfers a call and then drops control without monitoring the outcome of the transfer. Transfers are accomplished

Call function	Capability
	through tic script instructions (with several different options) or the Transfer Call node with the Blind Transfer option. Only intra-board transfer is supported, and ANI, UUI, or both can be provided when transferring a call using T1 or E1 interfaces.
Call Bridge	Places an outbound call to a third party and maintains the connection while the caller interacts with the third party. When the third party hangs up, the application continues with the next instruction or step. The call bridge feature is used most often when call transfer is not available on the PBX or from the central office. Only intra-board bridging is supported on Avaya IR. Call bridging is accomplished through the hbridge script instruction or the Call_Bridge external function. A Call Bridge can be set up using the Transfer Call node with the Bridge action. Only intra-board bridging is supported, and ANI, UUI, or both can be provided when bridging a call using T1 or E1 interfaces.
Converse Vector Step	Maintains control of a call while callers have access to announcements and transaction processing as they wait in queue. The converse vector step supports the DEFINITY ECS call vectoring (routing) feature where the switch retains control of vector processing in the system environment. Converse vector step is accomplished through the Conv_data external function.
Disconnect	Disconnects or hangs up the call. Disconnecting the call does not stop the execution of the application script. An application terminates execution when it reaches a Quit instruction. Disconnect is accomplished through the tic('h') script instruction or the Disconnect node.
Intelligent Transfer	Transfers a call and stays on the line to monitor the outcome of the transfer.

Call function	Capability
	<p>Intelligent transfers are accomplished through the tic script instructions (with several different options) or the Transfer Call node with the Intelligent Transfer option. Only intra-board transfer is supported, and ANI, UUI, or both can be provided when transferring a call using T1 or E1 interfaces.</p>
Originate	<p>Makes an outgoing call. Calls can be originated with the Make_Call external function.</p>
Full CCA	<p>Classifies the disposition of originated and transferred calls. Some of the dispositions include busy, answered, ring no-answer, and reorder. Full CCA provides more complete interpretation of network progress tones than intelligent CCA.</p> <p>Full CCA also provides voice detection and answering machine detection. Full CCA can classify a remote greeting as either human or machine.</p> <p>CCA can be activated when a call is dialed out during a switch-hook-flash transfer, a call bridge (internal transfer), or a make call (call origination). An error is generated if a script attempts to use CCA and the maximum number of CCA instances are running.</p> <p>Full CCA on trunk protocols such as PRI, WINK, and R2MFC needs sound. If the call is on mute or has no sound, the call is disconnected.</p>
Intelligent CCA	<p>Classifies the disposition of originated and transferred calls. Some of the dispositions include busy, answered, ring no-answer, and reorder.</p> <p>CCA can be activated when a call is dialed out during a switch-hook-flash transfer, a call bridge (internal transfer), or a make call (call origination). An error is generated if a script attempts to use CCA and the maximum number of CCA instances are running.</p>

Requirements

The Base system package (AVir) is required to support the call functions feature.

CTI data integration process

The computer telephony integration data integration process (CTI DIP) is an optional feature that provides an alternative to the manipulation of a switch via a direct ASA1 connection to the Avaya IR system. Applications that make function calls to the CTI DIP on the Avaya IR system can use an Avaya Computer Telephony (CT) server to control ports on a switch.

To administer the CTI DIP for IR, you must configure the CTI DIP on the Avaya IR system and set up and administer a Siebel Client and a CVCT server for CTI.

One of the main benefits of CTI is redundancy. Up to two extra Avaya CT servers can be connected to an Avaya IR system as backups to the server handling calls. If connection to the primary Avaya CT server is lost, then connection is made to a second server. If this connection is lost, then the third server is used.

CTI also lets IVR Designer applications communicate with an Avaya CT server that controls a MultiVantage switch.

Applications can make function calls to the associated DIP to:

- Put calls on hold.
- Retrieve calls from hold.
- Disconnect calls.
- Transfer two calls together.
- Report information about a call (port extension, call ID, ANI, Called Number) to the application.
- Report the state of each call to the application.
- Provide answer notification to a voice script.
- Dial a call.
- Get and report private data (UUI and UCID information) to the application.

An application on the Avaya IR system uses the CTI DIP to interact with a JTAPI client, which sends and receives JTAPI messages over a LAN to a network interface card (NIC) on the Avaya CT server.

The Avaya CT server uses a telephony server (TSRV) to route messages to the switch with the aid of a special driver and another NIC. The MAPD circuit card on the MultiVantage switch handles the messages.

Requirements

The CTI DIP feature is supported by only the MultiVantage switch.

The CTI DIP package (AVctidip) is required to support the CTI DIP feature.

The following software supports the CTI DIP feature:

- CentreVu CT Release 9.1 Version 1 or later
- Private Data Version 6

Converting DIP Into Service

To convert a DIP into a service:

-
1. Create an XML file for the required DIP.
 2. Make the DIP dependant on the relevant Avaya IR service.
 3. Copy the file to the `$IR_HOME/vs/data/services/manifest/` directory.
 4. Create a shell script to start the service.
 5. Execute the service as daemon (use `&`).
 6. Copy the shell script to the `$IR_HOME/vs/data/services/scripts` directory.
 7. Import the service by executing the `svccfg import IR_HOME/vs/data/services/*.xml`.
-

Data collection

The Data Collection feature is a base feature that provides support for collecting speech samples for application debugging and for building model applications for Natural Language Speech Recognition (NLSR).

Use data collection by implementing the following proxy external functions in TAS applications:

- Get data collection files for analysis (getdcPROXY)
- Remove data collection files when they are no longer needed (rmdcPROXY)

For VoiceXML applications, data collection will be turned on for all calls when DataCollection is set to yes. There is no control over data collection for individual calls when using VoiceXML.

For more information about using external functions in TAS applications, see the *IVR-D for TAS Online Help*.

Requirements

The Base system package (AVir) is required to support the data collection feature.

Dial Pulse Recognition

The Dial Pulse Recognition (DPR) feature enables the system to collect and recognize dial pulse input from callers. Dial pulse input can be generated by callers using rotary-dial

telephones, or by callers using push-button telephones that send dial pulses. The DPR feature can collect and recognize the digits 0 through 9 on T1 or E1 digital interfaces and supports barge-in. The DPR feature cannot be used with Voice over IP calls.

DPR supports barge-in. False recognition can occur if the caller is in a noisy environment or on a noisy telephone line. For example, an electrical disturbance that causes a "pop" sound on the telephone line can be recognized as a dial pulse digit. A telephone with a speaker can aggravate this condition because the speaker amplifies room noise. False recognition is reduced, however, once training is completed. Training is the ability of the system to learn the characteristics of each caller's connection.

Dial pulse processing is done using a speech-proxy interface that resides on the Avaya IR system.

Requirements

- At least one digital telephony card must be installed and operational.
- The Dial Pulse Recognition package (AVdpr) is required to support this feature.
- Speech Proxy Base Software package (AVsproxy)
- Speech Proxy SR - Speech Recognition package (AVsrproxy)

Dialout Alarming

Dialout alarming is a base feature that provides the ability for the system to automatically notify maintenance organizations when predetermined conditions exist. The system can dial out to at least two predetermined (user administrable) telephone numbers and provide an alarm status. It can also call the Initialization and Administration System (INADS) contact number (not administrable).

You can call into the Avaya IR system to query platform conditions. Connection to a remote session is maintained when the system initiates a dialout alarm. Users can turn off dialout alarming during a remote session. Dialout alarm messages will queue, and the user will be notified that they are being queued. Dialout alarming is re-enabled when the remote session ends. Queued dialout alarm messages are released at that time. Users can clear dialout alarm messages from the queue.

Requirements

- The dialout alarming feature requires an analog modem or Point-to-Point Protocol (PPP) that is supported over serial connections.
- The Base system package (AVir) is required to support dialout alarming.

Disk mirroring

Disk mirroring helps to ensure data security and availability by providing a completely redundant set of data. A redundant set of data greatly reduces the risk of data loss due to a

hard disk drive or system failure. If one hard disk drive is disabled or removed, the Avaya IR system can boot automatically from the other hard disk.

Software disk mirroring

Software disk mirroring is an optional feature that is only available on the Sun Fire 280R, Sun Fire V240, or equivalent hardware platforms because it requires two hard disk drives. The disk mirroring feature is provided by the Solstice DiskSuite 4.2.1 functionality of the Solaris operating system. Only full disk mirroring is supported.

Before enabling Software disk mirroring and synchronizing the hard disks, Unix File System (UFS) logging should be enabled. See *Enabling Unix File System logging* for more information.

The following table shows the mirrors, submirrors, directories, and partition sizes that are set up by default on complete (hardware and software) systems purchased from Avaya. For software-only solutions, these partitions are not set up by default. If you want to use Disk Mirroring on a system in which you have acquired the hardware on your own, you must first verify that your system is partitioned as shown in the following table.

Mirror	Primary submirror	Secondary submirror	Directory	Partition size in MB for 35-49 GB drives	Partition size in MB for 50+ GB drives
d0	d01 (c1t0d0s0)	d02 (c1t0d0s0)	/	10240	20480
d1	d11 (c1t0d0s1)	d12 (c1t1d0s1)	swap	2048	2048
d3	d31 (c1t0d0s3)	d32 (c1t1d0s3)	/voice1	5120	10240
d4	d41 (c1t0d0s4)	d42 (c1t1d004)		14	14
d5	d51 (c1t0d0s5)	d52 (c1t1d0s5)	/unused1	5120	10240
d6	d61 (c1t0d0s6)	d62 (c1t1d0s6)	/unused2	5120	10240
d7	d71 (c1t0d0s7)	d72 (c1t1d0s7)	/export	free	free

After disk mirroring is set up, the partitioning of the Solaris file system should not be changed. The status of the disk mirroring configuration is checked daily, and the system attempts to fix any problems that are found. If a problem cannot be fixed, the system raises a CGEN043 alarm and sends a message to the `$IR_HOME/vs/mirror/elog` log file.

If the system reboots after a hard disk drive has failed or has been removed and the drive has not been properly detached (using the `mirror_admin` command), it will come up in single-user mode. In single-user mode, the system only allows a single login from the console. To

restore the system to multi-user mode, log in as root, enter `mirror_admin detach` at the command prompt, and reboot the system.

Requirements:

- Two hard disk drives that have the same size and geography are required for disk mirroring.
- The disk mirroring feature is installed using the Disk Mirroring software package (AVdm).
- If you bought the software-only solution, and you are installing it on a Sun Fire 280R, Sun Fire V240, or Sun Fire V245 system, the Disk Mirroring software is copied to the `$IR_HOME/exports/optional_features` directory during installation. However, it is not automatically installed, and you must enable it manually.

Hardware disk mirroring

Sun Netra T2000 and Sun V245 Dual platforms support internal hardware disk mirroring. For more information about Sun Netra T2000 internal hardware disk mirroring, see http://docs.sun.com/source/819-7990-10/ontario-volume_man.html.

However, software disk mirroring through the AVdm package in Avaya IR is not supported on the Sun Netra T2000 and Sun V245 Dual platforms.

Note:

For Raid hardware control, you must configure Raid before installing Solaris. Alternatively, if you install Solaris and configure Raid, the system clears the disk and you must install the system again. This statement is true for both Sun Netra T2000 and Sun V245 Dual platforms.

Digital protocol - Loop Start

The Loop Start digital protocol is a standard loop start Foreign eXchange Station (FXS) protocol. An outgoing call is originated by closing the loop between the Tip and Ring leads toward the distant switch and waiting for a signal on the loop leads before sending the digits of the called number. On incoming calls, a loop closure from the distant switch is sufficient to cause the call to route to a predetermined destination.

This protocol is widely supported by customer-based switches, central office switches, and channel banks under names like line side digital, FXS, or off premise station (OPS). The Loop Start protocol works over digital cards configured as T1 or E1. Switches known to support either or both of the loop start and ground start FXS protocols include Avaya DEFINITY, Nortel Meridian, and Rockwell Spectrum.

T1 Loop Start provides 24 voice channels per trunk. E1 Loop Start provides 30 voice channels per trunk between a customer-based switch and the Avaya IR system.

The following limitations apply when you use a line-side digital interface:

- When a switch is excessively loaded and a timed delay is used prior to dialing, a call can be lost if the switch is not properly engineered and administered.
- Dial pulse is not supported on either T1 or E1 channels; however, dialing of DTMF tones is supported.

Requirements

- When you use the Loop Start digital protocol with ports on an Avaya IR system, you must have WebLM licenses for each port that uses the protocol.
- At least one digital telephony card must be installed and operational.
- The Loop Start digital protocol feature requires the NMS package (AVnms).
- When Loop Start is used to provide an ASAI link between the Avaya IR system and a switch, a separate path must be provided for communications between the two systems. The path must be provided by a MAPD connected to a local area network.

Digital protocol - PRI

PRI is an ISDN configuration that functions like an enhanced T1 circuit. The Avaya IR system supports and can communicate with any switches that support the following digital signaling protocols:

- AT&T PRI
- National ISDN
- NTT (INS-1500)
- ETSI
- Nortel PRI
- QSIG

Information elements such as ANI, DNIS, redirecting number, and service type are available for incoming PRI calls. Outbound calls can provide information elements like outbound ANI and service type.

QSIG is an ISDN-based signaling protocol for signaling between nodes of a Private Integrated Services Network (PISN). The QSIG protocol is based on the ISDN Q.931 standard. QSIG protocol supports the following features:

- Obtain ANI and DNIS from incoming QSIG calls.
- Transfer a QSIG call using the Transfer by Join feature (as defined by the QSIG ECMA 178, June 1992 standard). This feature allows an incoming QSIG call to the Avaya IR to be transferred to another termination point outside of the Avaya IR. After the incoming call has been established, a second call is placed out to the new termination point. The two calls are then joined together in the Avaya IR system to complete the transfer.
- Participate in a QSIG Path Replacement operation (as defined by the QSIG ECMA 176, June 1992 standard). The Avaya IR system does not initiate the Path Replacement request, but can pass messages between the PBXs involved in the Path Replacement operation. The QSIG Path Replacement feature optimizes the voice-

connections between multiple QSIG switching sites by reducing the number of hops (trunk facilities) it takes to route a call between sites.

- Use the Explicit Call Transfer (ECT) feature to transfer a call that uses the ETSI signaling protocol. With ECT, a user who has two calls, one active and one on hold, can connect the two calls together and then release the connections to both parties. ECT works only with Intelligent transfer. Blind transfer is not supported.

The Avaya IR system supports non-facility associated signaling (NFAS) across all spans on a single digital telephony card.

The Avaya IR system supports only one primary D-channel for each PRI interface. Avaya IR also supports configuring a backup D-channel. You can optionally configure a backup D-channel.

Requirements

- When you use the Primary Rate Interface (PRI) digital protocol with ports on an Avaya IR system, you must have WebLM licenses for each port that uses the protocol.
- At least one digital telephony card must be installed and operational.
- The PRI digital protocol feature requires the NMS package (AVnms).

Digital protocol - R2MFC

Multi-Frequency Compelled R2 (R2MFC) is an ITU-T Q400 series standard signaling protocol that operates on E1 connections and is used internationally between switches.

Avaya IR supports the digital version of line signals and inter-register signals.

Switch-to-switch R2 signaling supports the following MFC call types:

Simple calls (Incoming and outgoing calls): This is the signaling used between an Avaya IR system and a Central Office for local calls and toll-free call.

Calls with Caller ID: This is used between central offices for tracking the call to report the number to the switch being called and to apply billing charges when necessary.

Collect calls: For Brazil, incoming collect calls that are not agent assisted can be accepted or blocked for E1 trunks that are administered to use the R2MFC protocol. Incoming collect calls can be blocked only for E1 trunks that use the R2MFC protocol. The Avaya IR system can be administered to block collect calls by choosing a collect call blocking method. The collect call blocking method can be set to none (collect calls are accepted), Group II Signal 8, or double answer. Although not part of the R2MFC protocol, the double answer option for blocking collect calls is administered using the E1 R2MFC web administration screens.

Countries supporting R2MFC

R2MFC is supported in the following countries:

- Argentina
- Brazil

- Chile
- China
- Colombia (Telecom and ComSel)
- India
- Indonesia
- Korea
- Malaysia
- Mexico
- Philippines
- Singapore
- Thailand

Requirements

- At least one E1 digital telephony card must be installed and operational.
- When you use the R2MFC signaling protocol with ports on an Avaya IR system, you must have WebLM licenses for each port that uses the protocol.
- The R2MFC digital protocol feature requires the NMS package (Avnms).

Digital protocol - Wink Start feature

T1 Wink Start is a trunk arrangement where either side can initiate actions. After a connection is established with a distant system for an outgoing call, the system waits for a momentary signal (wink) before sending the digits of the called number. Similarly, on an incoming call, the system sends the wink signal when ready to receive digits.

Requirements

- When you use the Wink Start digital protocol with ports on an Avaya IR system, you must have WebLM licenses for each port that uses the protocol.
- At least one digital telephony card must be installed and operational.
- The Wink Start digital protocol feature requires the NMS package (AVnms).

DTMF type ahead

DTMF type ahead is supported by the Avaya VXML browser.

DTMF type ahead allows a user to enter DTMF values for multiple input fields without having to wait for individual prompts.

For example, if a caller knows the IR will prompt for a language, then a credit card type, and then an activity type, the caller can enter the input fields in quick succession instead of giving input separately for each prompt.

DTMF type ahead only works with:

- Fixed length fields
- Bargeable prompts

Echo cancellation

Echo cancellation is a base feature that involves the removal of the echo that occurs when a caller speaks or dials during a prompt. This facilitates barge-in and talk-off, and improves the accuracy of speech recognition.

Echo cancellation is built into the digital telephony boards.

Requirements

The NMS package (AVnms) is required to support the echo cancellation feature.

Application development considerations

Echo cancellation is controlled by IRAPI calls, including the following:

- irStartEcho starts echo cancellation.
- irStopEcho stops echo cancellation.
- irCheckEcho checks whether echo cancellation is on or off.

Enhanced Basic Speech

Enhanced basic speech is an optional feature intended for TAS applications.

The enhanced basic speech vocabulary package includes simple words that are frequently used to provide information to callers during transactions. Enhanced basic speech has been standardized to include the phrases needed to speak numeric, date, and monetary quantities.

The range of numeric, date, and monetary quantities that can be spoken is the same as in previous releases.

A maximum of one language can be used per application. Multiple languages can be installed on a system, and multiple applications can be running simultaneously, but only one language can be active per each application.

Enhanced basic speech is available in the following languages and voices:

- AE: Australian English (female voice)
- BP: Brazilian Portuguese (female voice)
- CF: Canadian French (female voice)
- CT: Cantonese Chinese (female voice)
- CS: Castilian Spanish (female voice)
- CZ: Czech (female voice)
- DT: Dutch (female voice)
- FR: French (female voice)
- GR: German (female voice)
- GK: Greek (female voice)
- HB: Hebrew (female voice)
- HD: Hindi (female voice)
- HU: Hungarian (female)
- IN: Indonesian (female voice)
- IT: Italian (female voice)
- JP: Japanese (female voice)
- KO: Korean (female voice)
- LS: Latin-American Spanish (female voice)
- MA: Malaysian (female voice)
- MD: Mandarin Chinese (female voice)
- PL: Polish (female voice)
- SQ: Slovak (female voice)
- TH: Thai (female voice)
- UE: United Kingdom (UK) English (female voice)

- US: US English (female voice)
- USM: United States English (male voice)

Requirements

The enhanced basic speech feature requires that the Enhanced Basic Speech package with at least one Avaya IR enhanced basic speech software language package be installed and operational.

Fault Management Facility

The Solaris Fault Management Facility is designed to be integrated into the Service Management Facility to provide a self-healing capability to Solaris 10 systems. The `fmd` daemon is responsible for monitoring several aspects of system health. The `fmadm config` command shows the current configuration for `fmd`. The Fault Manager logs can be viewed with `fmdump -v` and `fmdump -e -v.fmadm faulty` lists any devices flagged as faulty. `fmstat` shows statistics gathered by `fmd`.

With Solaris 10, Sun has implemented a daemon, `fmd`, to track and react to fault management. In addition to sending traditional `syslog` messages, the system sends binary telemetry events to `fmd` for correlation and analysis.

This feature is used to report error messages in a well-defined and explicit format. Each error code is uniquely specified by a Universal Unique Identifier. Resources are uniquely identified by a Fault Managed Resource Identifier (FMRI), a string that acts as the formal name for a particular resource for which Solaris can perform automated fault management activities.

For more information about the Fault Management Facility, see the relevant Sun documentation.

Fax

The optional Fax feature allows you to send and receive faxes. IRAPI commands are used to control the process. Fax is not supported over a VoIP connection.

NMS NaturalFax is a C function library component of Natural Access that provides the following fax functionality:

- Enhanced Group 3 fax features, including the following:
 - 1D, 2D and MMR encoding
 - Error Correction Mode (ECM)
 - Low, high, and superhigh resolutions
 - Standard page width formats (A3, A4, and B4)
- The T.37 format; specified as 1D encoding, low resolution, and A4 page width.

- The ability to convert documents into different fax formats online or offline.
- Text-to fax on the fly is a capability provided as part of NMS NaturalFax that converts text (ASCII) files into fax (TIFF) files. This is typically used to generate cover sheets.

The following are restrictions on the use of NMS NaturalFax:

- The document queue only supports TIFF-F or TIFF-S format file; it does not support text files or a text buffer.
- Conversion from an ASCII text file to a TIFF file is done on the host process.
- The file in the document queue must not exist on reception.
- There is no detailed information for transmission status, such as bytes sent or received.

For more information, see *NaturalFax Developer's Reference Manual*.

Requirements

- The Fax feature requires the Next Generation FAX package (AVfax) and the Fax Actions package (AVnmsfax).
- You must define the NFXTEXTFONT and NFXHEADERFONT environment variables to run this utility.

Host interface

The optional Host interface feature includes the following interface software that provides access to IBM mainframe applications:

- The Cleo TN3270 host interface software provides a 3270 Mod 2-5 terminal emulator and programmable interface to look like an IBM 3270 terminal.
- The TN3270 software uses a TCP/IP connection to a TNSERVER system that physically connects to the IBM mainframe.

The following capabilities of the host interface software are available with an Avaya IR system and Cleo software.

IVR Designer applications interface with host programs

You can use IVR Designer to create an application that interfaces with a complicated host computer application. The application developer logs in to the host computer and captures screen images. The developer then identifies the screens and fields on those screens that are needed during the transaction. An external function can also be created for Script Builder interface with custom data interface processes (DIPs) that require data communications protocols other than TN3270, SNA 3270, or the High Level Language Application Programming Interface (HLLAPI).

3270 Terminal Emulation

With this capability, a device or program can imitate another device or program. The 3270 terminal emulation software temporarily transforms itself into a look-alike of an IBM 3270

terminal. In addition to providing full 3270 functionality, the 3270 terminal emulator supports the transfer of files to and from UNIX.

IND\$FILE File Transfer

With the file transfer capability, you can transfer text or binary files between a mainframe using the IBM host program IND\$FILE and your Avaya IR system. Once installed, file transfer can be initiated interactively through the Terminal Emulator (TE) program or directly from the Solaris command line either by entering the FTS commands or by running a shell script containing the commands.

Note:

The host configuration for file transfer must use the write structured fields (WSF), also known as DFT mode, in the logmode table entry. This entry must be verified in both upgrades and new systems because it is a change from previous product releases.

FTS can work with multiple IBM mainframe operating environments or processing subsystems. These host systems and their IBM IND\$FILE program product numbers include:

- Time Sharing Option (TSO), #5665-311
- Conversational Monitor System (CMS), #5663-281
- Customer Information Control System (CICS), #5789-DQH

HLLAPI

HLLAPI is an application programming interface that can be used to write custom applications that can communicate with the host via an API. See the Cleo TN3270 or SNA 3270 HLLAPI Programmer's Guide for more information.

Host Data Interface Process (DIP)

(agdip3270)

Requirements

The host interface is a combination of software and optional hardware designed to support the transmission of information over a network. This network usually includes remote host computers and databases. With the host interface software package, applications running on the Avaya IR system can send and receive screens from applications running on the host mainframe.

IC Integration

IC Integration is an optional feature that supports the integration of the Avaya IR system with computer telephony services provided by Interaction Center. Avaya Computer Telephony for IC supports the Avaya IR system, enabling the Avaya IR system to communicate with both the telephone caller and the computer system.

With IC Integration, you can develop IVR Designer applications that perform actions such as:

- Retrieve values stored in a database through Avaya Computer Telephony
- Set values in a database through Avaya Computer Telephony

- Raise Avaya Computer Telephony alarms
- Invoke a subset of the transactions defined by the Interaction Center Workflow Designer for your specific site

While the application is running, it has synchronous access to Avaya Computer Telephony for IC services. For example, an application can be directed to retrieve information from a database in response to a caller's request, and then read that information aloud to the customer by playing prerecorded sound bites or messages.

An application on the Avaya IR system uses the Vesp_dip to access Avaya Computer Telephony services such as data retrieval and telephone call transfers. It enables the Avaya IR system to interact transparently with Avaya Computer Telephony for IC across the network.

The CTI DIP feature uses a CentreVu Computer Telephony (CVCT) server to control ports on a PBX or to interact with Siebel eBusiness applications that are located on a client connected to the server.

The following software is supported by the IC Integration feature:

- Interaction Center 6.1
- Interaction Center 6.1.3

Requirements

- This feature requires the Interaction Center Vesp_dip package software (AVvespdip).
- For the Interaction Center VOX Server to recognize the Avaya IR system, it must be appropriately configured. The configuration parameters for the VOX Server are described in the *Avaya Interaction Center VOX Server Programmer's Guide*. These configuration parameters are set using the IC Manager administrative tool, as described in the *IC Administration Volume 1: Servers & Domains*.

Call handling guidelines between the Avaya IR and IC systems

Use the following guidelines to improve call handling between the Avaya IR and Interaction Center systems.

- For systems that use the NMS package:
 - Add a very short (between 2 and 5 seconds in length) announcement to the very beginning of the application.
 - Invoke the newcall function.
- For systems that do not use the NMS package, as soon as a call is accepted by the application through the Answer Call node, the application must immediately invoke the newcall function. If it does not, the call is not handled properly and subsequent calls could be affected.
- When a call ends, regardless of the reason, the application must immediately invoke the gone function of the DIP. If not, both that call and all subsequent calls on that line may be lost or otherwise improperly handled by the system. Do not handle new calls on

that line until the gone function returns a response. Do not invoke the gone function more than one time within a call. Sending a second gone request within the same call causes a high priority alarm on the IC system.

JDBC database interface

Java Database Connectivity (JDBC) is a base feature that provides database connectivity and activity on Avaya IR and remote servers. The JDBC interface process executes basic SQL commands. It also creates and populates call data into local database tables or Solaris flat files.

How to access the JDBC DIP

You can access the JDBC Database Interface Process (DIP) from a TAS application through the dbase instruction. The name of the database DIP to use is specified as the first parameter to the dbase instruction. The parameter value could be DBDIP1, DBDIP2, DBDIP3, etc. depending on the DIP that is configured on the platform.

Using the JDBC DIP, applications can perform the following operations on a database:

- SELECT
- NEXT
- INSERT
- DELETE
- UPDATE

Applications can also execute stored procedures created on a database and retrieve the resultset/values returned by the procedure. For more information about how to write TAS applications that use these features, see *Avaya IVR Designer Help for TAS Applications*.

You can also access the JDBC DIP from VoiceXML applications through a servlet interface. VXML applications can post a request to JDBCServlet provided by the AVjdbcint package. Through this interface, applications can perform the following operations on a database:

- SELECT
- NEXT
- INSERT
- DELETE
- UPDATE

Applications can also execute stored procedures created on a database and retrieve the resultset/values returned by the procedure.

Database connectivity

The JDBC interface supports connectivity to the following remote databases:

- Oracle standard edition 9i and 10g
- Oracle enterprise edition 9i and 10g

- Sybase 12.x
- Informix 9.21.UC4 and 9.4
- IBM DB2 V8.1 and V8.2
- MS SQL Server 2000, 2005, and MSDE
- PostgreSQL V8.2

 **Note:**

Oracle 10g is the only database that is certified to run locally on Avaya IR systems. All other databases are not supported. The system does not support stored procedures on Informix or MS SQL Server 7.0.

Requirements

You must install the following software to use the JDBC database interface feature:

- The JDBC integration package (AVjdbcint) is required to support the JDBC Interface feature
- JDBC type4 drivers

Related topics:

[Database access](#) on page 16

Logging

Logging is a base feature that uses the Prism Logger to save system events in a log file. The system uses messages to alert you to problems, potential problems, or a change in the status of the system, including a record of system status when thresholds are reached. Messages are collected in the Message Log Report.

Requirements

- The alarm administration feature requires an analog modem or Point-to-Point Protocol (PPP) that is supported over serial connections.
- The Base system package (AVir) is required to support the Logging feature.

Message administration

System messages provide system status. You can use the Message administration feature to:

- Define the destinations to which the system messages are to be sent, such as a message log.
- The conditions under which the messages are sent to the specified locations (including by priority or the number of messages generated in a specified time period).

- Create a report that extracts specified information from the message log.
- Assign a priority to a message that reflects the severity of the associated event.

Sometimes an abnormal frequency of events signals that there is a problem worthy of attention. You can have a specific message (called a threshold message ID) sent to a destination indicating when a specific frequency (threshold frequency) has been detected for a particular event. The threshold frequency is administrable, and is defined as the number of messages (the message threshold) generated over a period of time.

Requirements

The Base system package (AVir) is required to support the message administration feature.

MRCP overview

The Media Resource Control Protocol (MRCP) is an Internet Engineering Task Force (IETF) specification. It is a communication protocol that describes a standard interface to media processing resources providing capabilities such as:

- Automatic Speech Recognition (ASR)
- Text-to-Speech (TTS)

An MRCP resource server or an MRCP server provides access to one or more of these speech resources.

Design

MRCP works similar to HTTP and many other Internet protocols, where each message contains the following design:

- *First line* indicates the message type and information such as response codes.
- *Header* contains a number of lines, each of these lines must be in <header>:<data> format.
- *Body* contains message details and the length specified by the header.

Like HTTP, MRCP contains requests and responses that acknowledge receipt of a request or provide other information about its processing. For example, when an MRCP client sends some audio data for processing, the server responds by providing a suitable port to send the data. This happens because MRCP does not support audio data specifically, and this request has to be handled by some other protocol, such as Real-time Transport Protocol (RTP).

Versions

Interactive voice response applications typically rely on the recognition of caller speech (through ASR) and the use of synthesized voice in system responses (through TTS). The

earlier releases of Avaya IR required a proprietary integration for each vendor's ASR and TTS system.

MRCP contains two versions, they are:

MRCP v1

- Uses Real Time Streaming Protocol (RTSP) to embed MRCP messages and signal requests for session set-up and tear-down.
- Uses RTP to carry audio data (speech delivered to an ASR engine for recognition and synthesized speech delivered from a TTS engine).
- Enables you to have ASR and TTS services independent of each other.
- Enables you to have ASR and TTS servers from different vendors that do not interoperate.
- Allows you to sell ASR and TTS licenses separately.
- Allows proxies to fit into existing design without affecting major architectural changes.
- Is compliant with the MRCP v1 specification (RFC 4463).

MRCP v2

- Uses Session Initiation Protocol (SIP) to signal requests for MRCP session set-up and tear-down.
- Uses RTP to carry audio data (speech delivered to an ASR engine for recognition and synthesized speech delivered from a TTS engine).
- Uses Transmission Control Protocol (TCP), a dedicated connection-oriented control channel to route actual MRCP protocol messages.
- Enables you to configure port allocation, failover, and load balancing independently.
- Allows distinct ASR and TTS sessions that are independent of each other.
- Allows both the MRCP versions to co-exist on different resource types.
- Bridges ASR and TTS sessions to allow the endpointer, bargain, and prompt-end information shared between two proxies.
- Is compliant with the MRCP v2 specification draft-15. For more information about this draft, see <http://tools.ietf.org/html/draft-ietf-speechsc-mrcpv2-15>.

Related topics:

[MRCP v1](#) on page 50

[MRCP v2](#) on page 53

MRCP v1

You can use the MRCP v1 feature to connect to the speech engines other than those supported by the original speech proxy.

Avaya IR applications use recognition of caller speech (Advanced Speech Recognition or ASR) and synthesized voice in system responses (Text-to-Speech or TTS). In the earlier releases

of Avaya IR, the software required a proprietary integration of each vendor's ASR and TTS system.

The MRCP v1 ASR and TTS proxy servers support:

- Scansoft
- Nuance
- IBM WebSphere Voice Server (WVS) for Multiplatforms V5.1.

You can install the ASR and TTS vendor engines on a remote server (a non Avaya IR system), media servers that are used to implement the MRCP v1 protocol, or completely different servers.

You can use the following media servers to implement the MRCP v1 open standard protocol:

- Scansoft SpeechWorks Media Server (Release 3.1.9 or later)
- Nuance MRCP Server (Release 1.0.0 SP 9 or later)
- IBM WVS for Multiplatforms V5.1 (Release 5.1.3 or later)
- Nuance Speech Server 5.0

The MRCP v1 feature includes the following optional software packages:

- MRCP Advanced SpeechRecognition package (AVmrcpasr). For more information, see MRCP ASR package
- MRCP Text-to-Speech package (AVmrcptts). For more information, see MRCP TTS package.

Requirements

- The MRCP v1 ASR and TTS packages require the Speech Proxy package (AVsproxy).
- The MRCP v1 ASR package requires the Speech Proxy SR-Speech Recognition package (AVsrproxy).
- The MRCP v1 TTS package requires the Proxy Text-to-Speech package (AVttsprxy).

Related topics:

[MRCP v1 protocols and connections](#) on page 51

[MRCP v1 requests and responses](#) on page 52

[MRCP v1 protocols and connections](#) on page 51

[MRCP implementation considerations for IBM MRCP](#) on page 56

[MRCP v1 requests and responses](#) on page 52

[VoiceXML](#) on page 69

[MRCP v2](#) on page 53

MRCP v1 protocols and connections

Protocols

Avaya IR communicates with the MRCP v1 servers using the following protocols:

- Signaling requests for call set-up and tear-down between servers use Real-time Streaming Protocol (RTSP).
- Audio data (speech delivered to an ASR engine for recognition and synthesized speech delivered from a TTS engine) is carried over a Real-time Transport Protocol (RTP).

 **Note:**

- RTSP and RTP use different ports and port ranges.
- The Avaya IR MRCP v1 implementation does not use Real-time Control Protocol (RTCP), a control protocol that is normally used with RTP.

Avaya IR neither sends nor receives RTCP packets.

Connections

The Avaya IR MRCP v1 connection sends the following RTSP methods and receives their associated responses:

- SETUP
- ANNOUNCE
- TEARDOWN
- DESCRIBE

The software receives the following RTSP methods and responds accordingly:

- ANNOUNCE
- TEARDOWN

All other received methods are ignored.

Avaya IR sends the RTSP DESCRIBE message with a sequence number of 0 and incremented further, primarily to verify its connection with the MRCP v1 server and periodically to confirm the server's ongoing connection. The responses to the DESCRIBE message with a sequence number of 0 confirm that the MRCP server is still connected and active.

Related topics:

[MRCP v1](#) on page 50

[MRCP implementation considerations for IBM MRCP](#) on page 56

[MRCP v1 requests and responses](#) on page 52

MRCP v1 requests and responses

ASR requests and responses

Avaya IR sends the following MRCP v1 requests for ASR resources and receives their associated responses:

- SET-PARAMS
- DEFINE-GRAMMAR

- RECOGNIZE
- RECOGNITION-START-TIMERS
- STOP

The software receives the following MRCP v1 events and responds accordingly:

- START-OF-SPEECH
- RECOGNITION-COMPLETE

All other events are ignored.

TTS requests and responses

Avaya IR sends the following MRCP v1 requests for TTS resources and receives their associated responses:

- SET-PARAMS
- SPEAK
- STOP

The software receives the SPEAK-COMPLETE event and responds accordingly.

All other events are ignored.

Related topics:

[MRCP v1](#) on page 50

[MRCP v1 protocols and connections](#) on page 51

[MRCP implementation considerations for IBM MRCP](#) on page 56

MRCP v2

You can use the MRCP v2 feature to connect to the speech engines other than those supported by the original speech proxy.

Avaya IR applications use recognition of caller speech (Advanced Speech Recognition or ASR) and synthesized voice in system responses (Text-to-Speech or TTS). In the earlier Avaya IR releases, the software required a proprietary integration of each vendor's ASR and TTS system.



Note:

The MRCP v2 ASR and TTS proxy servers only support Nuance.

You can install the ASR and TTS vendor engines on a remote server (a non Avaya IR system), media servers that are used to implement the MRCP protocol, or completely different servers.

You can use media servers that implement MRCP v2 open standard protocol. For example, Nuance Speech Server 5.0.

The MRCP v2 feature includes the following optional software packages:

- MRCP v2 Advanced Speech Recognition package (AVmrcpv2asr). For more information, see MRCP v2 ASR package.
- MRCP v2 Text-to-Speech package (AVmrcpv2tts). For more information, see MRCP v2 TTS package.

Avaya IR does not support the following MRCP v2 functionalities:

- Secure RTP (SRTP)
- Speaker Identification (SI) and Speaker Verification (SV)

Requirements

- The MRCP v2 ASR and TTS packages require the Speech Proxy package (AVsproxy).
- The MRCP v2 ASR package requires the Speech Proxy SR-Speech Recognition package (AVsrproxy).
- The MRCP v2 TTS package requires the Proxy Text-to-Speech package (AVttsprxy).

Related topics:

[MRCP v2 protocols and connections](#) on page 54

[MRCP v2 requests and responses](#) on page 55

[VoiceXML](#) on page 69

[MRCP v1](#) on page 50

MRCP v2 protocols and connections

Protocols

Avaya IR communicates with the MRCP v2 servers using the following protocols:

- Signaling requests for call set-up and tear-down between servers use Session Initiation Protocol (SIP). You can secure it further by using TLS.
- Audio data (speech delivered to an ASR engine for recognition and synthesized speech delivered from a TTS engine) is carried over a Real-time Transport Protocol (RTP).

Note:

- SIP and RTP use different ports and port ranges.
- The Avaya IR MRCP v2 implementation does not use Real-time Control Protocol (RTCP), a control protocol that is normally used with RTP.

Avaya IR neither sends nor receives RTCP packets.

Connections

The Avaya IR speech proxy connection uses the Transport Control Protocol (TCP), a dedicated connection oriented channel to send and receive MRCP v2 messages.

The Avaya IR MRCP v2 connection sends the following SIP methods and receives their associated responses:

- OPTIONS
- INVITE
- ACK
- OK
- BYE

Avaya IR sends the SIP OPTIONS message primarily to verify its connection with the MRCP v2 server and periodically to confirm the server's ongoing connection.

MRCP v2 requests and responses

ASR requests and responses

Avaya IR sends the following MRCP v2 requests for ASR resources and receives their associated responses:

- DEFINE-GRAMMAR
- SET-PARAMS
- RECOGNIZE
- START-INPUT-TIMERS
- STOP

The software receives the following MRCP v2 events and responds accordingly:

- START-OF-INPUT
- RECOGNITION-COMPLETE

All other events are ignored.

TTS requests and responses

Avaya IR sends the following MRCP v2 requests for TTS resources and receives their associated responses:

- SPEAK
- STOP
- SET-PARAMS

The software receives the SPEAK-COMPLETE event and responds accordingly.

All other events are ignored.

MRCP implementation considerations for IBM MRCP

For ASR or TTS, you can configure an Avaya IR system to support an MRCP compliant engine (ASR or TTS) or the original set of vendor engines as described in the following cases:

- An Avaya IR system with the MRCP ASR package can only support ASR engines that are MRCP compliant.
- An Avaya IR system with the MRCP TTS package can only support TTS adjuncts that are MRCP compliant.
- An Avaya IR system configured to support MRCP-compliant ASR engines can support the original set of TTS engines.
- An Avaya IR system configured to support MRCP-compliant TTS engines can support the original set of ASR engines.

When multiple Avaya IR systems share the same LAN, some can be configured to support MRCP-compliant engines and others can support non-MRCP engines. Avaya IR with MRCP also supports the ASR and TTS languages supported by IBM WVS V5.1.

As a result, you can configure an Avaya IR system in the following ways:

- A single Avaya IR system to a single MRCP server.
- A single Avaya IR system to multiple MRCP servers.
- Multiple Avaya IR systems to a single MRCP server.
- Multiple Avaya IR systems to multiple MRCP servers.

 **Note:**

For IBM WVS server, you must set the `PerPortConnection` value to `true` in the `mrcp.cfg` file.

Related topics:

[MRCP v1](#) on page 50

[MRCP v1 protocols and connections](#) on page 51

[MRCP v1 requests and responses](#) on page 52

MRCP load balancing

Several methods of load balancing are supported for MRCP servers.

Balancing your MRCP server load can improve system performance. Load balancing is supported by TAS and VXML applications using a MRCP proxy.

The supported methods of load balancing are:

- RoundRobin
- nSessions
- nServices

MRCP backup servers

You can associate every MRCP server with a backup MRCP server. If the primary MRCP server loses TCP/IP connectivity, the backup MRCP server automatically starts servicing calls. The backup MRCP server will continue servicing calls until the primary MRCP server can resume normal operations.

MRCP transactions

The backup server will start servicing new MRCP transactions if there is a loss of TCP/IP connectivity to the primary MRCP server. A MRCP transaction is lost if the primary MRCP server fails during the transaction.

Considerations for MRCP backup servers

Consider the following points for your backup servers:

- Each primary server can have a single backup server, or several primary servers can have a backup server.
- Primary servers and backup servers must be configured in a similar manner from the **Web Administration** menu.
- The ports on the backup server should be equal to the total number of ports for all the servers it will backup.
- Load balancing does not involve backup servers, until they become active.

Network Management (SNMP)

The optional Network Management feature allows customers to monitor and administer their IR systems remotely, using a network management system. It provides client-side service for Simple Network Management Protocol (SNMP) communications with network management servers using such service monitoring software as HP OpenView or Tivoli.

When designated alarms or severe errors occur on an IR system, this feature can be used to send alarm traps to the network management server (NMS) using SNMP. The network management server then typically uses a graphical display to indicate that the IR server is in trouble.

This feature also allows you to use the NMS to remotely get and set administration data (or a subset of it) on the IR server.

Requirements

- In addition to the Avaya IR hardware platform, this feature requires you to have a network management server and the appropriate LAN or other connections. Avaya does not provide any of these other hardware elements.
- the Network Management package (AVsnmp) is required to support the Network Management (SNMP).
- This feature requires the use of an SNMP network management software tool, such as HP OpenView or Tivoli, which are the only two SNMP software tools tested by Avaya for use with the Networking Management (SNMP) feature.

Proactive Contact integration

The Proactive Contact Integration feature is an optional feature that integrates the functionality of the Avaya IR system and Proactive Contact. The PDS Integration feature provides a subset of PDS commands for the Avaya IR system, which allows the Avaya IR to act as agents attached to Proactive Contact. These commands can be used as external functions in IVR Designer applications to add outbound functionality to the Avaya IR system. Use Dialer external functions in IVR Designer to develop applications that interact with Proactive Contact.

The Avaya IR then processes the call as it would any other inbound call. The flexibility of the Avaya IR allows the IVR Designer application to provide a high level of complexity to the subsequent caller interaction. When the call is complete, certain information related to the call is passed back to Proactive Contact so the record in the calling list is updated.

Requirements

- The Proactive Contact Integration feature requires the PDS Integration package software (AVdialer).
- For Proactive Contact to recognize the Avaya IR system, it must be installed and appropriately configured.

ROI and TCO reports

IR has standard reports you can use to track the return on investment (ROI) and the total cost of ownership (TCO) of every call. TCO is the cost spent on operations with your investments.

ROI is the calculation of benefits returned on your investments. The TCO and ROI report calculations are based on the events grouped with a call service.

ROI report calculations

ROI is calculated on per call basis. You assign a specific ROI value to each event in a call service. The ROI of a call is the number of occurring events multiplied with the ROI value of the event .

TCO report calculations

You assign a specific TCO value to each event in a call service. The TCO value must be based on operation cost, not on the initial purchase cost of equipment and software. The TCO of a call is the number of occurring events multiplied by the TCO value which is then added to the total call cost. The total call cost is the number of call units multiplied by the call cost.

ROI and TCO report considerations

- The ROI and TCO reports feature has been designed primarily for TAS applications, but it can be used with a VoiceXML application also, if the application is designed carefully by using the object tag.
- ROI and TCO reports are not supported with an Informix database.

Service creation environment

The service creation environment feature is a base feature that provides tools for developing automated transactions, known as applications.

IVR Designer is the icon-based, object-oriented application development tool provided with the Avaya IR system. This PC-based tool can be used by individuals without programming experience to create and modify voice applications.

With the IVR Designer service creation environment, you can create TAS and VoiceXML applications (for VoiceXML applications, only when using IVR Designer 5.3 or earlier) on a Windows-based PC, with a graphical user interface (GUI) that you can customize. IVR Designer features standard Windows operations, with right-click menus and enhanced cut, copy, and paste capabilities.

With IVR Designer, you develop applications by specifying the details of interaction between the Avaya IR system and its callers. IVR Designer includes a set of tools that you use to design, edit, test, simulate, generate, transfer, and install the applications. In addition, you use these tools to diagnose and debug your applications.

The standard system software also supports applications developed using the tools described in the following table:

Tool	Description
Script Builder	Applications created with the Script Builder tool on CONVERSANT systems can be ported to the Avaya IR system or converted to IVR Designer applications.

Tool	Description
	<p>You can convert existing Script Builder applications to IVR Designer automatically, without having to recreate them in IVR Designer. The Script Builder Conversion Tool is activated by an option on the File Menu within IVR Designer. When you select this option, the Script Builder Conversion Tool finds the desired application on the target Avaya IR system, downloads it to the IVR Designer PC, and converts it to IVR Designer format. You can then modify it as desired before reinstalling it on the Avaya IR system. It is likely that converted applications will require some manual editing to maintain the original functionality.</p>
Transaction Assembly Script (TAS)	<p>TAS is an assembly-type instruction language that can be used by programmers who are experienced in assembly language development to develop voice applications. A sequence of instruction calls run within the generic transaction state machine (TSM) software that manages the low-level interactions required to operate the system. At any time, TAS can be used to assemble, load, change, or replace a script without affecting the other scripts running on TSM or other IRAPI programs running the system.</p>
Voice eXtensible Markup Language (VoiceXML)	<p>VoiceXML applications run on the Avaya IR system. VoiceXML applications can be created using IVR Designer 5.3 or earlier.</p>
Intuity Response Application Programming Interface (IRAPI)	<p>IRAPI is a standard C language programming interface. Applications developed with IRAPI communicate with and execute script applications on the Avaya IR system.</p>

Hardware requirements

The following hardware is required for installation and operation of IVR Designer:

- Pentium personal computer, or equivalent, with the following configuration:

Minimum and Recommended PC Requirements				
Operating System	Processor		RAM	
	Minimum	Recommended	Minimum	Recommended
Windows 2000, NT 4.0, XP	133 MHz	900 MHz	64 MB	128 MB

- VGA monitor (or better)
- Microsoft-compatible mouse
- 65 MB of free hard disk space. The required amount of available disk space depends on the number and complexity of the applications you plan to develop on your personal computer and whether or not you plan to record speech. The entire IVR Designer software package requires 65 MB of hard drive space. You need at least 10 MB of additional free space for the applications you develop and considerably more disk space if you plan to record speech. For help on determining the hard drive requirements for your applications, contact Avaya.
- CD-ROM drive, any speed
- Sound Blaster compatible card. With the sound card, you can listen to speech that you record with IVR Designer. You can use this speech during simulation, as well as on the system. You can run IVR Designer without a sound card, but you will not be able to record or listen to speech on your PC.
- Microphone. Although any microphone is sufficient for recording speech with IVR Designer, the quality of the microphone you choose affects the fidelity of speech.
- Speaker. Although any speaker is sufficient for listening to recorded speech with IVR Designer, the quality of the speaker you choose affects the fidelity of the recorded speech you hear.

Software requirements

Your personal computer must have the following software installed:

- IVR Designer Version 5.0 or later
- Microsoft Windows 2000, XP, or NT 4.0

The following software must be installed on the Avaya IR system:

- Service Creation Integration package
- Transaction State Machine package for TAS applications
- VoiceXML Interpreter package for VoiceXML applications

Service Management Facility

Service Management Facility (SMF) is a Solaris Operating System feature that defines a programming model providing persistently running applications known as services. Services

can represent a number of software facilities, such as a set of running processes, a set of system configuration parameters, or a synthetic set of running services. A Solaris service is started only after the administrator marks it as enabled and all its dependencies are satisfied.

In the older versions of Avaya IR (prior to Avaya IR 4.0), at the time of installation all the packages copy their CS* files in `/etc/conf/init.d/` directory. These CS* files are having entries of processes on each line which are supposed to go in `/etc/inittab` file. The `/etc/inittab` file controls process dispatching by `init`. The processes most typically dispatched by `init` are daemons. The `mkitab` script read these CS* files and rebuild the `/etc/inittab` file resolving the order and dependency between various processes. Once the `/etc/inittab` file rebuilds with the Avaya IR related processes written into it. It is the job of the operating system to start these processes. These processes are highly dependent on run-levels. Many of the Avaya IR processes run at user defined run level 4. Many scripts also use this user-defined run level 4 to check weather the voice system of Avaya IR is running or not.

Migration to SMF architecture has following advantages for Avaya IR:

- SMF ensures that the processes which are spawned through services are started concurrently which bring up the voice system faster than older versions of Avaya IR.
- Service monitoring and integration with the Fault Management Facility. For more information, see [Fault Management Facility](#) on page 43.
- Resolves the order of starting services of Avaya IR.
- Dependency of scripts on run-levels is removed.

Following tables describes the SMF features:

Feature	Description
Dependency order	Sometimes, services depend on each other for proper operation. A robust system understands the dependencies of each service. If an underlying service fails, you must correct it before it affects the other dependant services.
Delegation of tasks to non-root users	Enables you to configure a service to run within a limited set of privileges in place of an all powerful root user. If you compromise and constrain the power of a service to a more limited user, you can minimize the damage that can be inflicted by the intruder.
Parallel starting of services	Speeds up the boot process by starting multiple services simultaneously. It allows idle CPU time resulting from a service that is temporarily blocked to be relinquished for use by other services

Feature	Description
	that can start independently of the blocked service.
Automatic service restart after failure	Works in conjunction with the Solaris Fault Manager, allowing software recovery in the event of hardware faults (CPU, Memory), administration errors, such as accidental kills and software core dumps.

You can define the SMF features to contain special states that allow finer control, monitoring, and diagnosing software failures, than having a Solaris administrator or dedicated modules to kill and restart the service.

 **Note:**

Avaya IR 4.0, Customized Data Interface Process (DIP) must be converted into service. To convert a DIP into a service, see [Converting DIP Into Service](#) on page 34.

For more information about SMF, see relevant *Sun documentation*.

TDD Modem

The TDD (Telecommunications Device for the Deaf) modem is an optional feature that can be used in IVR applications to allow communications with the hearing-impaired, in compliance with Section 508 of the federal Rehabilitation Act of 1973.

The TDD modem acts much like a speech recognizer. It uses the speech proxy server to communicate with the Avaya IR server. When the IR system receives a TDD tone, that tone is passed to the TDD modem software. The TDD software then uses a grammar-like algorithm to determine the corresponding DTMF tone, which it then returns to the IR system.

Requirements

- At least one digital telephony card must be installed and operational.
- The TDD Model feature requires the tdd package.
- An IVR application developed using IVR-D.
- An NLSR grammar (word list) created using IVR-D.

Tone masking

Tone masking feature enables you to secure sensitive information that you provide using DTMF inputs, such as PIN codes, credit card numbers, and bank account numbers. It is a base feature, applicable only for DTMF inputs.

In IVR Designer for TAS, you can use the external function to turn on the tone masking feature. The parameters of this function are similar to those of the Prompt and Collect node. When you enable tone masking and enter the required DTMF input, the application plays a file corresponding to the digit to mask your input.

The application saves the files that are played out in the `$IR_HOME/vs/bin/ag/eascripts/prompts` location. You can overwrite these files by tones of your choice.

 **Note:**

When using the tone masking feature with bargein enabled, do not set the Discard Previous Touchtones property to True in the Prompt and Collect node.

Requirements

- The IR Base System package (AVir)
- You must generate the application using IVR Designer for TAS or the <object> tag in Dialog Designer

Two B-Channel Transfer

With the optional Two B-Channel Transfer (TBCT) feature, a call coming into the Avaya IR system on one B (bearer) channel can be connected, within the switch, to a call that has been made by the Avaya IR system on another B channel. Both incoming and outgoing calls on the Avaya IR system are released.

Requirements

- The TBCT package (AVtbct) is required to support the TBCT feature
- PRI is required for TBCT.

Universal Call ID Expansion

Universal Call ID (UCID) is a base feature that assigns a unique identifier (8-byte binary or 20-character ASCII) for every call in a MultiVantage call center customer environment. With UCID, uniform data-tracking is possible for all call-related data in a call center, regardless of the

system. Also available is the user-to-user information (UUI) element that supports the specification of additional information to be passed in external function arguments.

Requirements

The UCID feature requires the following software:

- The Universal Call ID package (AVucid) is required to support the UCID feature.
- NMS package
- PRI is required for the UCID feature.

UUI data transfer

Due to a limitation in the NMS software used in Avaya IR, the software can only receive certain types of UUI data for incoming calls. The third octet of the UUI field is the UUI protocol discriminator. The Avaya IR system can only receive UUI data in which the protocol discriminator is one of the following:

- 0x0 – User-specified protocol
- 0x1 – OSI
- 0x2 – X.244
- 0x3 – Reserved for system management convergence function
- 0x4 – IA5 (ASCII)
- 0x7 – V.120
- 0x8 – Q.931

UUI elements with protocol discriminators other than the ones listed above will not be passed through the system to the application handling the call.

For IR applications designed to populate the **UUI** field, the first byte of data used as input from the application will specify the protocol discriminator (which is the third octet in the field). The first and second octets for the **UUI** field, used to specify the information element (IE) and length respectively, are not populated by the application input.

For applications designed to receive and read the **UUI** field, the first byte of data that is read from the field is the protocol discriminator (the third octet in the field). The first and second octets are not read.

Encoding user-defined UUI

Avaya IR encodes the 0x7e message for UUI based on ECMA 269 standards. It ensures that the received message is unchanged.

When you enable the UUI encoding, the **Protocol Discriminator** field in the UU packet is set to 0x00 (user defined protocol). By default, the **Protocol Discriminator** field is set to 0x04 (ASCII).

Encoded UU packet contents

The following table provides an example of an UU packet encoded using the ECMA 269 standards:

Size	Contents
------	----------

1 byte	0x7E (Information Element type)
1 byte	(Length of the element including data)
1 byte	0x00 (Protocol Discriminator)
1 byte	0x45 (Format Specifier)
1 byte	0x80 (Data type)
1 byte	(Length of the data - n)
1 byte	(Data byte 1)
1 byte	(Data byte 2)
1 byte	(Data byte 3)
1 byte	(Data byte 4)
1 byte	(Data byte 5)
1 byte	(Data byte n)
1 byte	(XOR Checksum of all the bytes starting from the Format Specifier – 4th byte)

The checksum value in the last byte of the encoded UU packet confirms that the received information is unchanged.

You must ensure the following when encoding a 0x7e message:

- Add **Inline Code** node in the IVR-D application.
- Define `setIRAPIparam(IRP_UUI_TYPE_im.0` function call in the **Inline Code** node.
- Specify `data to be transferred` in ASCII format in the **UUI** field in the transfer node.

During data transfer, Avaya IR encodes and sends the value specified in the transfer node.

Voice over IP

The licensed VoIP feature enables the Avaya IR system to serve as an IVR adjunct that connects to a DEFINITY or Communications Manager switch over a packet-based network. With VoIP, transmission to the switch is achieved without digital interfaces (T1/E1). Instead, all transmissions occur over the packet network using the network interface card (NIC) on the Avaya IR system. VoIP and T1 or E1 digital interfaces can be used on the same IR system at the same time.

With packet-based call handling:

- Each call can use variable bandwidth, as required.
- Network capacity that is not used for calls can be used for other applications, such as file-sharing, email, and Web browsing.
- There is no need to maintain separate networks for voice (telephony) and data (office LAN).
- Although VoIP works using standard hardware required for LAN communication, configuration settings are applied to a virtual VoIP card. The card concept is familiar and allows you to group VoIP channels for the purposes of management and control, similar to grouping channels for a digital telephony card. VoIP connectivity is supported using the Communications Manager H.323 Terminal (Station) protocol.

Capacities

The maximum number of channels on a VoIP subsystem, when used in conjunction with E1/T1 ports, is 240. VoIP supports 240 concurrent calls running touchtone TAS applications on a single machine. The minimum number of concurrent ASR and TTS calls, or ASR only calls, or TTS only calls supported is 96.

Requirements

- You must install the Voice over IP package.
- You must have a WebLM license for each port (channel) that uses this feature.
- You must connect the Avaya IR to a local area network (LAN) that also supports a DEFINITY MultiVantage 1.3 with R11 or later system software.
- Calls handled by VoIP must be received from a DEFINITY or Avaya Communication Manager switch. VoIP calls cannot come directly from a public network switch to the Avaya IR. The DEFINITY or Avaya Communication Manager switch has the following hardware requirements:
 - Prowler card (TN22302AP)
 - C-LAN card (TN799DP)

Related topics:

[Voice over IP feature capabilities](#) on page 67

Voice over IP feature capabilities

The VoIP subsystem provides the following capabilities:

- Incoming and outgoing calls. The Avaya IR system can receive and make calls via the VoIP subsystem. The destination of outgoing calls and the origin of incoming calls can be anywhere in the public telephony network.
- Each licensed channel can be used by an incoming or an outgoing call at any given time.
- G.711 mu-law and A-law voice encoding to and from the network.
- Voice energy detection on per channel basis. This feature is activated on demand, usually by Avaya IR applications that need to detect voice energy from the remote end. It may be described as either on-interface or in-band, meaning that the network

connection between the DEFINITY or MultiVantage system and the Avaya IR system is used.

- DTMF barge-in. Tones are passed to the IR system through the connection between the DEFINITY or MultiVantage system and the IR system (in-band). Signals may also be used and are passed off-interface or out-of-band.
- Voice barge-in. This capability is under the control of an external speech recognition system. It is described as off-interface, or out-of-band, meaning that the required packets are passed to another system.
- Call transfer. Avaya IR applications can request fully-supervised or unsupervised transfer of calls using the ASAI or CTI feature.
- IR uses a unique product ID when registering with the Communications Manager R2.1.
- To improve security, passwords can be assigned to VoIP communication channels. For more information about assigning passwords, see [Assigning telephone numbers and passwords to channels](#).
- Optional reporting to a VoIP Monitoring Manager. The VoIP Monitoring Manager is a call quality monitoring application for calls that use packet-forwarding technology. The manager is an application part of the Avaya VisAbility Management Suite.
I think our standard is to not include any trademarks.
- The VoIP subsystem is fully transparent to existing IVR applications, which can:
 - Play announcements or prompts
 - Record voice messages
 - Receive or send DTMF digits
 - Interact with speech recognizers and Text-to-Speech converters
 - Interact with enterprise servers (databases)
- Blind transfers using the <object> tag in VoiceXML applications
has an xref to the Blind transfer topic (25840).
- Bridge transfers using the <transfer bridge="true"> tag in VoiceXML applications
- Up to 10 DEFINITY, MultiVantage, or Avaya Communication Manager switches can be supported by a single IR.
- Support for ANI and DNIS protocols.
- Support for converse data return. An IVR-D external function or a Script Builder external action that supports the call vectoring (routing) feature by enabling the switch to retain control of vector processing in the system environment. It supports the converse vector command to establish a two-way routing mechanism between the switch and the system to facilitate data passing and return.
- On-system bridging of calls. Historically, the Avaya IR system did support on-system bridging of channels in order to provide for call transfer when the Avaya IR system worked as an adjunct to a switching element that did not support third-party call control.
- Voice encryption.

- C-LAN failover capability. This feature requires you to have 2 C-LAN (TN799DP or later) cards installed on the Avaya Communication Manager. The C-LAN card which controls voice communication at startup is considered to be the primary C-LAN card. When the primary C-LAN card fails, all the voice calls are automatically diverted to the secondary C-LAN card. For the voice calls currently in progress, voice communication can continue to be exchanged but DTMF inputs cannot be processed.
- Quality of Service (QoS) support using the Differentiated Services (Diffserv) parameter on the Avaya Communication Manager. This feature is turned on by default, if you have the Voice over IP (AVvoip) package installed and operational on the Avaya IR system. You can use this feature to optimize audio communication between the Avaya IR system and the Avaya Communications Manager. Using Diffserv, a QoS protocol that prioritizes IP voice and data traffic, voice quality is preserved, even when network traffic is heavy. This is achieved using DiffServ code points (DSCP), located in 6 of the 8 bits of the Type of Service (TOS) field.
- Call Classification Analysis (CCA). Classifies the disposition of originated and transferred calls. Some of the dispositions include busy, answered, ring no-answer, and reorder.

The VoIP subsystem does not support:

- Voice encoding other than the G.711 mu-law and A-law.
- Incoming or outgoing FAX calls.
- Echo cancellation. There is no need to support echo cancellation on the VoIP subsystem since this function is performed by the MultiVantage or DEFINITY switch.
- Gateway function as defined in the ITU-T H.323 Recommendation. An Avaya IR system cannot be used as an interface between a circuit and a packet network.
- Call signaling encryption.

Related topics:

[Voice over IP](#) on page 66

VoiceXML

VoiceXML (Voice Extensible Markup Language) is an optional feature that provides another language with which to write IVR applications for the Avaya IR system. VoiceXML is a standardized scripting language that is similar to HTML but VoiceXML uses a voice browser.

The VoiceXML interpreter provides the functionality for parsing and processing the VoiceXML documents. The voice browser enables VoiceXML applications to perform the following IVR functions:

- Prompting users with text-to-speech (TTS) and prerecorded digitized audio
- Recognizing speech and dual tone multi-frequency (DTMF) input

- Recording user speech
- Providing basic telephony control, such as transfer and disconnect

The AVvoicxml2-0 package enforces compliance with the VoiceXML 2.1 specification, dated June 13, 2005, and should be used with VoiceXML applications that have been written to comply with that version of the specification. There are a few exceptions, which are listed in unsupported features of the VoiceXML 2.1 specification for the AVvoicxml2-0 package. To create and deploy VoiceXML applications for the AVvoicxml2-0 package, Avaya recommends that you use Dialog Designer 4.0.

VoiceXML applications can be made more secure by using Secure Sockets Layer (SSL) technology. SSL is a protocol designed to allow the secure transmission of data over the Internet. It works by using a private key to encrypt data transmitted using the SSL connection. For companies desiring to use VoiceXML applications to request and send sensitive data, such as user account numbers or credit card numbers, SSL allows them to do this securely and safely. The supported protocols are file, http, and https.

The AVvoicxml2-0 package is backward compatible with existing IR VoiceXML applications.

Like TAS applications, VoiceXML applications can be use speech recognition and TTS. For more information, see Natural Language Speech Recognition (NLSR) and the MRCP feature. The Proxy Text-to-Speech (PTTS) feature is highly recommended to synthesize speech from the information retrieved from a database. (The Enhanced Basic Speech feature can be used for dynamic speech in TAS applications, but not in VXML applications.) For more information on PTTS, see Proxy Text-to-Speech (PTTS).

You will have to make certain modifications to your existing VoiceXML applications that do not comply with VoiceXML 2.1 specifications. For more information see New VoiceXML capabilities for the AVvoicxml2-0 package and New VoiceXML capabilities for the AVvoicxml2-0 package that may affect customer applications.

Requirements

The VoiceXML feature requires the AVvoicxml2-0 package.

Related topics:

[Natural Language Speech Recognition](#)

[MRCP v1](#) on page 50

[MRCP v2](#) on page 53

[Proxy Text-to-Speech](#)

[VoiceXML capabilities for the AVvoicxml2-0 package](#)

VXML log commands

There are two commands you can use to view VoiceXML browser (VXI) log messages. The commands are `trace`, and `logCat`.

trace

The `trace` command is your primary tool for viewing VXI log messages. Log messages are generated only when the `trace` command is run. The `trace` command allows you to view the entire call trace in a sequential manner.

logCat


You use the `logCat` command to view the following log messages:

- Fatal
- Error
- User level logs with the `<log>` tag
- Performance logs

VXML 2.1 support

IR supports standard VXML 2.1 tags and attributes.

Tag	Attributes
<code><data></code>	Supported <code><data></code> attributes: <ul style="list-style-type: none"> • <i>src</i> • <i>name</i> • <i>srcexpr</i> • <i>method</i> • <i>namelist</i> • <i>enctype</i> • <i>fetchaudio</i> • <i>fetchhint</i> • <i>fetchtimeout</i> • <i>maxage</i> • <i>maxstale</i>

Tag	Attributes
	<p> Note: The attributes <i>fetchaudio</i>, <i>fetchhint</i>, <i>fetchtimeout</i>, <i>maxstale</i>, and <i>maxage</i> are not used for external grammars.</p>
<disconnect>	Supported <disconnect> attributes: <i>namelist</i>
<grammar>	Supported <grammar> attributes: <i>srcexpr</i>
<foreach>	Supported <foreach> attributes: <ul style="list-style-type: none"> • <i>array</i> • <i>item</i>
<mark>	Supported <mark> attributes: <ul style="list-style-type: none"> • <i>name</i> • <i>nameexpr</i> • <i>markname</i> • <i>marktime</i>
recordutterance property	Supported <property> attributes: <ul style="list-style-type: none"> • <i>recording</i> • <i>recordingsize</i> • <i>recordingduration</i>
<script>	Supported <script> attributes: <i>srcexpr</i>
<transfer>	Supported <transfer> attributes: <i>type</i>

VXI channel modes

IR allows you to choose between two different VXI channel modes for processing calls. VXI can process calls in a normal channel mode or a logical channel mode.

Normal channel mode

In normal channel mode, VXI associates a call process with a physical channel. Calls to the physical channel have to wait until the previous call process completes.

Logical channel mode

In logical channel mode, VXI associates a call process with an event queue. Calls to the physical channel can be answered as soon as the previous caller disconnects.

Channel mode considerations

Consider the following points when choosing your VXI channel mode:

- Normal channel mode can result in callers experiencing rings with no answer if there is a large amount of after call processing.
- Logical channel mode could slow down your IR system because of the additional processing performed.
- Logical channel mode is the default VXI channel mode.

vrop_tng

Voice response output process (vrop_tng) manages speech database and downloads speech data to VRU. You can use vrop_tng to reduce the CPU usage under load. This is achieved by reducing the number of channel handlers from 25 to 4. It reduced their contention over a single mutex and resulted in maximum gains.

The default number of channel handlers is 4. However, you can modify the default number by providing the -h argument to the process.

For example, to change the number of channel handlers to 6, modify the command line in `$IR_HOME/vs/data/services/scripts/vrop_tng_start.sh` as //
`CV05:4:respawn:`getIRHome`/vs/bin/vrs/vrop_tng -h 6 </dev/console >`getIRHome`/voice1/vrop_tng.err 2>&1:`

The maximum number of channel handlers can be 25 and the minimum is 1. For any other values, the system reverts to the default number, 4.

If there are any delays in playing the pre-recorded or TTS prompts, you must check the action handling time of vrop_tng performance in the traces. If the average of the action handling time increases significantly, complemented by appreciable delays, you can increase the number of channel handlers. However, this process affects the CPU time and cycles.

Web administration

Web administration is a base feature that provides a web browser interface to administer the system.

Requirements

The Web Administration package (AVwebadm) is required to support Web administration.

The Web administration feature may be used with the following supported browsers:

- Netscape 6.2
- Netscape 7.x
- Internet Explorer 5.x
- Internet Explorer 6.x
- Internet Explorer 7.x
- Mozilla Firefox 2.x
- Mozilla Firefox 3.x

WebLicensingManager

WebLicensingManager (WebLM) is a base feature that provides the licensing functionality required to administer the Avaya IR system.

Requirments

- WebLM requires the LicenseManager package (AVlm) to be installed.
- If you want to install the WebLM server on the same platform as the Avaya IR system, you would need to install the WebLicensingManager package (AVweblm) on the Avaya IR platform.

Web services

Avaya IR provides a web service based interface to perform Operation, Administration, and Maintenance (OAM) activities.

Avaya IR uses Apache Axis 1.4 for Web Services Server applications and RPC SOAP messaging technique. Axis creates the necessary WSDL interface to be used to talk with the server to run IR commands from a remote client.

OAM activities in Avaya IR are performed through the Apache Tomcat server, which provides secure http connection. Users can call the Web Service from a client application written in any

programming language that supports SOAP posted/received over secure http services or https. The OAM commands can be executed from the client by accessing the web service hosted on an IR system. The client endpoint interfaces can be built independently of the server application. Each web service request is authenticated with proper username and password to perform any OAM activity.

Related topics:

[Avaya IR Web Services related commands](#) on page 75

[Avaya Web Services authorization and authentication](#) on page 75

[WSDL for IR web services](#)

[Guidelines for creating client programs for web services](#)

Avaya IR Web Services related commands

Operations supported by Avaya IR Web Services

<code>start_vs</code>	Starts the voice system.
<code>stop_vs</code>	Stops the voice system.
<code>vs_status</code>	Inquires on the voice system status.

Commands for deploying and undeploying Avaya IR Web Services

<code>irWebService deploy</code>	Deploys the web service.
<code>irWebService undeploy</code>	Undeploys the web service.
<code>irWebService list</code>	Checks whether web service is active.

Related topics:

[Client stub creation Client stubs are used by client applications to invoke a web service.](#)

[Web services](#) on page 74

Avaya Web Services authorization and authentication

Avaya Web Service security framework uses the WSSE User name Token algorithm for authorization and authentication of the client. IR will accept only those web service requests from client which are in compliance with OASIS web services security specification WSSE username token. The SOAP header of every web service request must be strictly in the form given below:

```
<wsse:Security>
<wsse:UsernameToken>
<wsse:Username>IR_Admin</wsse:Username>
<wsse:Password>IloveIR</wsse:Password>
</wsse:UsernameToken>
</wsse:Security>
```

The username and password must be sent in every web service request. There is no need to send the password in encrypted form because the transport layer on which the web service

request is sent is TLS protected. Only users with access to the IR Web Administration can access the web service.

Related topics:

[Client stub creation](#) Client stubs are used by client applications to invoke a web service.

[Web services](#) on page 74

Chapter 4: Security

IR security

Securing the IR system involves a number of tasks, including securing your network, restricting access to the system, restricting administrative passwords, and securing the CTI password by encryption.

In addition, there are many security features and other security measures that you must consider when implementing an IR system. The white paper titled *Avaya Interactive Response Security* includes information to help guide you in making security policy decisions concerning your Avaya IR system. You are strongly encouraged to read and implement the recommendations in the white paper.

System access security

You should restrict system access to authorized personnel. There are two types of access:

- Physical access
- Operational and administrative access

Physical access

To restrict physical access to the system, the system should be located in a secured location that can only be reached by authorized personnel.

Operational and administrative access

To restrict operational and administrative access to the system, the Avaya IR system uses both the built-in capabilities of the Solaris operating system and the ASG Security feature.

For information about securing the operating system and securing other parts of the system using Solaris, see the Managing System Security Topics in *Solaris System Administration Guide, Volume 2*.

Using a firewall

Avaya strongly recommends using a firewall product to protect your internal LAN, including any Avaya IR servers, from unauthorized access. Firewalls sit between your LAN and the

Internet and control access to designated ports. Most firewalls can be configured to allow specified remote IP addresses to connect to designated ports.

For information about establishing a firewall on your network, see the following resources:

- The Network Security information in *Solaris System Administration Guide, Volume 2*.
- Customer documentation for your network equipment.

If using a firewall is not an option, you can to configure your Avaya IR system on a physically isolated LAN. For more information see [Physically isolating the LAN](#).

Related topics:

[Physically isolating the LAN](#) on page 78

Physically isolating the LAN

As an alternative to using a firewall, you can protect your Avaya IR system by configuring it on a LAN that has no physical connection to the Internet. Sometimes referred to as an island LAN, this type of network environment has its own LAN switch and contains only those network elements with which the Avaya IR system needs to interface, such as speech servers, database servers, a backup server, and a MultiVantage system (for VoIP connectivity and ASAI feature functionality). Because this LAN has no physical connection to the internet, there is no need to use a firewall to protect the system from unauthorized access.

If you cannot isolate the LAN, Avaya recommends using a firewall to protect your system.

Related topics:

[Using a firewall](#) on page 77

Disabling a network service

Disabling a network service that is required or used by the Avaya IR software can adversely affect system performance or behavior. Avaya recommends that you disable only those network services that you are sure are not required by the system. For a list of which services should be enabled or disabled and why, see *Avaya Interactive Response Security*.

-
1. Open the `/etc/inetd.conf` or `/etc/services` file, depending on the network service you want to disable.
A list of the network services in each file is provided in *Avaya™ Interactive Response Security*.
 2. Add a comment symbol (`#`) to the beginning of the line referencing the network service that you want to disable.

3. Save and close the file.
 4. Reboot the system.
-

Disabling the telnet feature

The telnet service provides you with a mechanism to connect to a host machine from a remote system. The telnet service is not needed for IR operation. It is recommended that you disable the telnet feature on a software-only IR solution.

Companies should not disable the telnet service unless they have another means of accessing the system, such as using the system console or Secure Shell.

Running the DisableServices utility

Use the disableServices utility to disable unneeded network services and to harden system security.

This utility is intended for customers who purchase the Avaya IR software-only solution. However, it can be used to set any IR system to the default and preferred settings for network services. Running this utility will ensure that your system conforms to the same standards as the Avaya IR complete system solution at time of purchase

-
1. Log in as root if you are not already logged in.
 2. At the system prompt, enter `$IR_HOME/vs/bin/util/disableServices` and press `Enter`.

The utility sets all network services to the default and preferred settings, and generates a list of all the network services that it disabled in the file `$IR_HOME/$IR_HOME/voice1/disabledServices.log`.

Encrypting CTI Password

Avaya IR security is enhanced by encrypting CTI passwords. It prevents the telephony server passwords from being stored as plain text in the `$IR_HOME/vs/data/ctidip.rc` directory.

To ensure security:

- The `cticonfig` command does not display the password on the console, while adding the `AES/MAPD` server details.
- Avaya IR saves the password in the `ctidip.rc` configuration file in the encrypted form.
- Avaya IR encrypts the CTI password in the `.XML` file after running `save_conf` during upgrade.

Avaya IR uses:

- The Advanced Encryption Standards (AES) algorithm to conform to the Federal Information Processing Standards (FIPS) for encrypting passwords.
- The Java Secure Key Store (JSKS) to store the secure key generated using the Java program. The secure key encrypts the password while storing it in a configuration file and decrypts the password after reading it from the configuration file.

 **Note:**

When you upgrade from an earlier Avaya IR release to Avaya IR 4.0, the system automatically encrypts the password by upgrading the required configuration details.

Restricting administration permissions

Administration of Avaya software can be restricted by assigning permissions to users, as described in the following table:

Permission	Description
Root	The user can perform any task on the system.
Administration	The user has full control of the voice system and features through Web Administration and the command line.

Permission	Description
Operations	The user has access to configuration management, reports, administration, and system monitor capabilities, but does not have control of the voice system. Note that the system monitor is only administrable through the command line.

Logging in as root

For security purposes, the IR system is set up by default to restrict access of the root account to users logging in from the console. Users are not allowed to log in as root remotely (using telnet, for example). It is possible, however, to log in remotely as another user and then switch to root using the `su` command. For more information about root and the `su` command, see *Solaris System Administration Guide, Volume 1*.

-
1. Log in as root using the console or by logging in as another user then entering `su root` and the root password.
 2. Enter `cd /etc/default`.
 3. Enter `vi login`.
 4. Using the vi editing commands, find the line `CONSOLE=/dev/console`
 5. Comment the line by inserting `#` at the beginning of the line.
 6. Save and close the file.
-

IR command permissions by login type

Permission to execute some IR commands is controlled by IR login type.

IR Root login

The Root login has permission to run all IR commands.

IR Admin login

IR Admin can run all commands except:

- avinstall
- avuninstall
- upgrade
- assign_permissions
- unassign_permissions
- mirror_admin
- pkgadd
- pkgrm

IR Operator

IR Operator can run only these commands:

ass	assign	backuph
browseSortLogCat	callCost	capturePerfRec
ccarpt	cddrpt	cdsrpt
dalarm	del	delete
disp	display	display_permissions
docexplain	dspActAlarms	dspRetAlarms
exp	explain	fax
faxq	faxrpt	licenseConfig
licenseQuery	logMsg	mode
msgadm	remove	renumber
res	restore	retireAlarms
roitco	roitcorpt	shmview

IR command permissions by login type

snmpConfig	speech	sproxyadm
startNmsCards	start_vs	stop_vs
tts	vs_status	vxmlConfig
vxmlSetLogs		

Chapter 5: Related documents

These documents provide information on products related to the Avaya IR system. For the most current versions of these documents, contact the company responsible for publishing the information or visit their Web sites.

NMS Communications

To obtain these document, visit the NMS Web site at <http://www.nmscommunications.com>.

- *AG4000 Installation and Developer's Manual*: Installation and configuration of the NMS AG4000 telephony boards
- *AG4040 Installation and Developer's Manual* : Installation and configuration of the NMS AG4040 telephony board
- *CG6060 Series Installation and Developer's Manual*: Installation and configuration of the NMS CG 6060 Series telephony board
- *CG6565 Series Installation and Developer's Manual*: Installation and configuration of the NMS CG 6060 Series telephony board
- *NaturalFax Developer's Reference Manual*: NMS NaturalFax feature
- *NMS SNMP Reference Manual*: NMS implementation of SNMP

Sun Microsystems

To obtain these document, visit the Sun Web site at <http://www.sun.com>.

- *Sun Fire V215 and V245 Servers Installation Guide* : Installation of the platform.
- *Sun Fire V215 and V245 Servers Administration Guide*: Use of the platform
- *Sun Fire V210 and V240 Servers Installation Guide*: Installation of the platform
- *Sun Fire V210 and V240 Servers Administration Guide*: Use of the platform
- *Sun Fire V210 and V240 Servers Parts Replacement Manual*: Parts replacement
- *Sun Fire V210 and V240 Servers Product Notes*: Additional information about the platform
- *Sun Blade 150 Service Manual*: Manual that comes with the platform
- *Sun Blade 150 Getting Started*: Installation of the platform
- *Sun Fire 280R Server Service Manual*: Manual that comes with the platform
- *Sun Fire 280R Server Owner's Guide*: Installation and use of the platform
- *SunVTS 4.6 User's Guide*: How to use SunVTS

Related documents

- *System Administration Guide: Basic Administration*: Managing and monitoring basic network services for Solaris 10
- *System Administration Guide: Network Services*: Managing and monitoring advanced network services for Solaris 10

Scansoft

To obtain these document, visit the Scansoft Web site at <http://www.scansoft.com>.

- *RealSpeak TTS 3.5.1 Programmer's Guide*: Installation and configuration of the server software
- *SpeechWorks OSR 3.0 Installation Guide*: Installation and configuration of the server software
- *SpeechWorks Speechify 3.0 User's Guide*: Installation and configuration of the server software

- *SpeechWorks Speechify 2.1.5 User's Guide*: Installation and configuration of the server software
- *SpeechWorks Media Server Getting Started Guide*: Installation configuration of the media server

Nuance

To obtain these document, visit the Nuance site at <http://www.nuance.com>.

- *Speech Recognition System 8.5 Installation Guide*: Installation and configuration of the server software
- *Vocalizer 4.0 Release Notes*: Installation of the server software
- *Vocalizer 3.0 Developer's Guide*: Installation and configuration of the server

IBM

To obtain these document, visit the IBM site at <http://www.ibm.com>.

IBM WebSphere Voice Server (WVS) for Multiplatforms V5.1: Documentation for IBM WVS V5.1

US Robotics

To obtain these document, visit the US Robotics site at <http://www.usr.com/support/s-main-menu.asp>.

Sportster Manual: User's manual for the Sportster 33.6K external modem

Apache Software Foundation

To obtain these document, visit the Apache Software Foundation site at <http://www.apache.org>.

Tomcat Servlet Engine documentation: Complete documentation for installing, administering, and using the Tomcat Servlet Engine

Cleo Communications

To obtain these document, visit the Cleo Communications site at <http://www.cleo.com>.

Cleo TN3270E : Specifications for the host interface feature

Gold Systems

To obtain these document, visit the Gold Systems site at <http://www.goldsys.com>.

Vonetix documentation: Vonetix Web site that contains various documents available from Gold Systems

University of Cambridge

To obtain these document, visit the University of Cambridge site at <http://www.uk.research.att.com>.

Virtual Network Computing (VNC) documentation: Complete documentation on VNC, a remote display system that uses a thin client and works on the Internet

Avaya

Avaya documentation can be obtained from the Avaya Support site at: <http://www.avaya.com/support>

Avaya CentreVu:

Related documents

Avaya CallVisor:

CallVisor PC LAN over MAPD Installation, Administration, and Maintenance, 555-230-113

Avaya PDS:

- *PDS User's Guide*
- *Product Release Notes for Avaya PDS 12.0*
- *Product Release Notes for Service Pack 2 of Avaya PDS 12.0*

Avaya Interaction Center:

- *Avaya Interaction Center VOX Server Programmer's Guide*
- *IC Administration Volume 1: Servers & Domains*

Avaya Dialog Designer:

- *Avaya Dialog Designer Getting Started Guide*
- *Avaya Dialog Designer Developer's Guide*

Chapter 6: Training

For information about product training, go to the Avaya Web site at www.avaya.com and click **Training**.

Index

A

Alarm administration	
feature overview	25
ASAI	
feature overview	25
ASG	
feature overview	29
ASR resources	20
audio formats, supported	18
authentication, Web Services	75
authorization, Web Services	75
Avaya Interaction Center	
integration	16
Avaya IR web services	74

B

Backing up system data	
overview	27
backup servers, MRCP	57
Barge-in feature	29

C

call answer	
circuit-based	9
packet-based	11
Call data reports	
feature overview	29
Call functions feature	30
Call handling example	13
call termination	
circuit-based	10
packet-based	12
call transfer	
circuit-based	10
packet-based	12
caller speech, recognizing	18
circuit-based	
call answer	9
call termination	10
call transfer	10
operations	7
command permissions	82
commands for deploying and undeploying Web Services	
75	
Converting DIP Into Service	34
CTI DIP	
feature overview	33

CTI password encryption	80
-------------------------------	--------------------

D

Data collection feature	34
Data exchange operations	16
Dial Pulse Recognition, overview	34
Dialout Alarming feature overview	35
digital interfaces	
Two B Channel Transfer feature	64
Digital protocols	
Loop Start feature overview	37
PRI feature overview	38
R2MFC feature overview	39
Wink Start	
feature overview	40
disableServices utility, running	79
disabling	
network service	78
telnet	79
disk mirroring	
feature overview	36
DTMF type ahead	40

E

Echo cancellation	
feature overview	41
enhanced basic speech	
feature overview	41

F

Fault Management Facility	43
fax	
feature overview	43

G

Greeting and prompting callers	18
--------------------------------------	--------------------

H

Host interface feature	44
------------------------------	--------------------

<hr/>	
I	
IC integration	
feature description	45
Vesp_dip and the VOX server	17
Interactive Response	
overview	5
Introduction, MRCP	49
<hr/>	
L	
legal notices	2
load balancing	
methods	56
Logging	
feature overview	48
logging in as root	81
logical channel mode	73
Loop Start	
feature overview	37
<hr/>	
M	
Message administration	
feature overview	48
MRCP	
backup servers	57
implementation considerations	56
load balancing	56
MRCP Overview	49
MRCP v1	
feature description	50
protocols and connections	51
requests and responses	52
MRCP v2	
feature description	53
protocols and connections	54
requests and responses	55
<hr/>	
N	
Network Management feature	57
NMS NaturalFax	43
normal channel mode	73
notices, legal	2
nServices	56
nSessions	56
<hr/>	
O	
operations supported by Web Services	75
Overview, MRCP	49
<hr/>	
P	
packet-based	
(VoIP) operations	11
call answer	11
call termination	12
call transfer	12
PDSintegration	58
physically isolating the LAN	78
PRI	
feature overview	38
<hr/>	
R	
R2MFC	
feature overview	39
Recognizing caller speech	18
related documents	85
report	
ROI	59
TCO	59
report calculations	
ROI	59
TCO	59
Restoring system data	
overview	27
restricting administration permissions	80
ROI report	59
round-robin	56
Running	
disableServices utility	79
<hr/>	
S	
security	77–81
disabling the telnet feature	79
overview	77
physically isolating the LAN	78
restricting administration permissions	80
using a firewall	77
Service creation environment feature	59
Service Management Facility	61
SNMP	
Network Management feature overview	57
speech	
enhanced basic speech feature	41
recognizing caller speech	18
speech detection feature	
detecting speech	
feature overview	18
speech software	19
Supported audio format	18
supported transfers	19
Symantec NetBackup support	
overview	28
system access security	77

T

TCO report	59
TDD Modem	
feature overview	63
third-party documents	85
Tone masking	64
training	89
TTS	
dictionary support	20
resources	20
Two B Channel Transfer	
feature overview	64

U

Universal Call ID Expansion	
feature overview	64
using	
firewalls	77

V

Vesp_dip and the VOX server functionality	17
voice response	
how it works	21
sample transaction	22

VoiceXML	
feature overview	69
VoIP	
capabilities	67
feature description	66
vrop_tng	73
VXI	
logical channel mode	73
normal channel mode	73
VXML	
attributes	71
elements	71
logCat	71
trace	71

W

Web administration	
feature overview	74
web services	74
Web Services	
authentication	75
authorization	75
commands for deploying and undeploying Web	
Services	75
operations supported by Web Services	75
Wink Start	
feature overview	40

