

SS7 Support for WebSphere Voice Response



SS7 User's Guide

Version 6.1

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Note

Before using this information and the product it supports, read the general information under “Notices” on page 119.

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This edition applies to Version 6, Release 1 of IBMSS7 Support for WebSphereVoice Response (program numbers 5799-GZL, 5799-GZQ), and to all subsequent releases and modifications until otherwise indicated in new editions. Make sure you are using the correct edition for the level of the product.

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About this information

This book provides an overview of how you can use the IBM® SS7 Support for WebSphere® Voice Response product with IBM WebSphere Voice Response for AIX® to connect to telephone switches using the SS7 protocol. This book aims to help you install and configure the product, and provide system administration support for the product once it is running.

This book describes SS7 Support for WebSphere Voice Response Version 6.1. If you are a newcomer to using the SS7 protocol with WebSphere Voice Response, start with Chapter 1, “Introducing SS7 Support for WebSphere Voice Response,” on page 1, which provides an overview and describes some typical installations.

In this book, IBM SS7 Support for WebSphere Voice Response is generally referred to as SS7 Support for WebSphere Voice Response, and Distributed7 is abbreviated to D7.

This edition applies to Version 6, Release 1 of IBM®SS7 Support for WebSphereVoice Response (program numbers 5799-GZL, 5799-GZQ), and to all subsequent releases and modifications until otherwise indicated in new editions. Make sure you are using the correct edition for the level of the product.

Who should use this information

This book is for the person who installs and configures SS7 Support for WebSphere Voice Response in your business. It assumes some knowledge of telephony (in particular SS7), WebSphere Voice Response for AIX, the AIX operating system, and pSeries® computer hardware.

Typographic conventions

This book uses the following typographic conventions:

boldface

Identifies an **item** that is in a WebSphere Voice Response window. The item might be a keyword, an action, a field label, or a pushbutton. Whenever one of the steps in a procedure includes a word in boldface, look in the window for an item that is labeled with that word.

boldface italics

Are used for emphasis. *Take extra care* wherever you see bold italics.

italics Identify one of the following:

- New *terms* that describe WebSphere Voice Response components or concepts. A term that is printed in italics is usually followed by its definition.
- *Parameters* for which you supply the actual names or values.
- References to other *books*.

monospace

Identifies one of the following:

- Text that you type in an AIX window. Because AIX is case sensitive, ensure that you type the uppercase and lowercase characters exactly as shown.
- Names of files and directories (path names).

Accessibility

WebSphere Voice Response for AIX is a voice application enabler. The applications that are developed to run on WebSphere Voice Response provide telephone access to business data and services. In this way, WebSphere Voice Response provides accessibility for people who cannot access the data and services by using regular Web pages or traditional graphic interfaces. These telephone user interfaces are fully accessible to people who are blind or have low vision and, if speech recognition is used, to people with mobility impairments or limited hand use. Speech recognition capability can be provided by products such as IBM WebSphere Voice Server. In addition, support for users of Telephony Devices for the Deaf (TDD) is provided as part of the WebSphere Voice Response product.

With WebSphere Voice Response you can perform many application development and system administration tasks with a text editor or line commands—these are accessible if you use a screen reader product to interface with them. Also, the default settings of the WebSphere Voice Response graphical user interface can be changed to produce large fonts and high contrast colors. Details of how to use these accessibility features can be found in the *WebSphere Voice Response for AIX: User Interface Guide* book. Alternatively, application development can be done with Java or VoiceXML development tools that are supplied by IBM and third parties.

You can also use a screen-reader product to access the WebSphere Voice Response publications in HTML format (for details of their availability see “List of WebSphere Voice Response and associated documentation” on page 157).

Notes on terminology

- A glossary of commonly-used terms is at the end of this book.
- The full product name of *WebSphere Voice Response for AIX with DirectTalk Technology* is generally abbreviated in this book to *WebSphere Voice Response*.
- The term *pSeries*[®] is generically used in this book to refer both to PCI-based RS/6000[®] computers and to appropriate models of the System p5[®] and pSeries ranges. (Consult your IBM representative for details of models that are supported for use with WebSphere Voice Response.) RS/6000 computers with an MCA bus are not supported.
- The *IBM Quad Digital Trunk Telephony PCI Adapter* is generally referred to in this book by its abbreviation DTTA. This adapter is a replacement for the IBM ARTIC960RxD Quad Digital Trunk PCI Adapter, which is generally referred to by the abbreviation DTXA. The DTXA is not supported with WebSphere Voice Response Version 6.1.
- References made to the *VoiceXML 2.1* specification are intended to include VoiceXML 2.0 unless otherwise specified.

Where to find more information

The information provided in the WebSphere Voice Response library will help you complete WebSphere Voice Response tasks more quickly. A complete list of the available publications and where you can obtain them is shown in “List of WebSphere Voice Response and associated documentation” on page 157.

Useful Web sites

The following Web sites are useful sources of information about WebSphere Voice Response and related products:

WebSphere Voice Response

http://www.ibm.com/software/pervasive/voice_response_aix/

IBM WebSphere developerWorks resources (including WebSphere Voice products)

<http://www.ibm.com/developerworks/websphere/zones/voice/>

VoiceXML Version 2.0 and 2.1 specifications

<http://www.w3.org/TR/voicexml21/>

<http://www.w3.org/TR/voicexml20/>

CCXML Version 1.0 specification

<http://www.w3.org/TR/2011/PR-ccxml-20110510/>

Genesys

For more information on Genesys products go to the Genesys Web site at <http://www.genesyslab.com>

Making comments on this book

If you especially like or dislike anything about this book, feel free to send us your comments.

You can comment on what you regard as specific errors or omissions, and on the accuracy, organization, subject matter, or completeness of this book. Please limit your comments to the information that is in this book and to the way in which the information is presented. Speak to your IBM representative if you have suggestions about the product itself.

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Chapter 1. Introducing SS7 Support for WebSphere Voice Response

This chapter introduces Signaling System 7 (SS7), and how SS7 works with WebSphere Voice Response for AIX. This chapter also includes four examples of how you might configure WebSphere Voice Response when using SS7, from the simplest to the maximum supported configuration.

What is SS7?

SS7 is the ubiquitous communications protocol that is used within and between telephony networks of the world for system management, call setup and tear-down, and other network functions. Although there are slight variations in the fine detail of SS7 implementations used around the world, all telephone companies use the same set of protocols known generically as SS7.

Because the signaling is on a separate network to voice traffic, SS7 allows the use of features such as automatic callback when a called number is busy. Messages about the states of the voice lines are sent and received on the signaling network so voice lines are not tied up until both parties are known to be available to speak.

SS7 is used for many other functions within a telephone network. Essentially, when the computer in one telephone exchange or *switch* needs to talk to computers elsewhere in the network (either to other switches or to standalone machines such as those used for billing and advanced services), SS7 messages are always used.

SS7 links and network components

SS7 uses discrete messages to exchange information, the messages being passed between switches and other end points using packet switched data links. There are two types of connections in a telephony network:

- Packet-switched links used for SS7 messages (but not voice data).
- Circuit-switched links used for voice data (but not SS7 messages).

The SS7 protocol stack

SS7 is structured in a multi-layered stack which corresponds closely to the layers of the standard OSI model, although some SS7 components span a number of layers, as illustrated in here.

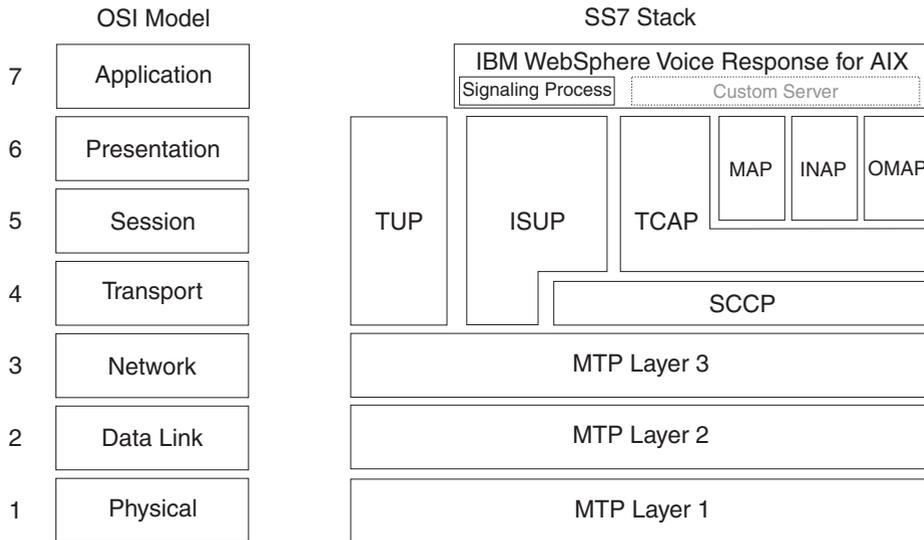


Figure 1. SS7 protocol stack

The SS7 component parts are:

Layer 1 (Physical): MTP-1 (Message Transfer Part-1)

MTP-1 defines the physical means by which SS7 messages are transferred from one node to another. For E1 or T1 networks, the physical layer is usually a timeslot of an E1 or T1 frame respectively.

The physical layer specifies only how a sequence of bits is conveyed from one SS7 node to another. It says nothing about the actual meaning of the bits or how they are grouped together to form a message.

Layer 2 (Data Link): MTP-2

MTP-2 defines how an MTP-1 bit transfer mechanism is used to reliably pass variable length messages from one SS7 node to another. MTP-2 uses a variant of the High level Data Link Control (HDLC) used in most modern data transfer protocols. This uses a delimiter to define the start and end of a data frame, prevents flags occurring in a

frame (bit-stuffing) and protection for the entire frame (CRC at the end). It also defines how CRC errors are handled (by error response and retransmission).

MTP-2 says nothing about the actual content of a message. It simply defines a mechanism by which a message of any length can be sent 100% reliably between SS7 nodes and can be used by higher layers of the SS7 protocol.

MTP-2 knows nothing beyond the single point-to-point link it operates on.

Layer 3 (Network Layer): MTP-3

MTP-3 builds on top of the lower-level MTP layers to allow the creation of a network of telephony network nodes interconnected by SS7 links. Each node is assigned a unique address in the network (known as a Signaling Point Code or SPC). Messages can be sent at the MTP-3 level in one node to a topologically distant node (that is with one or more intermediate SS7 nodes) simply by specifying the Destination Point Code (DPC). MTP-3 entities on the SPC node, the DPC node, and all intermediate nodes coordinate the transfer of a higher-layer message through the network.

MTP-3 can use multiple parallel routes from SPC to DPC through the network to take account of link loading and availability (there should always be more than one way to get from any SPC to any DPC).

Upper Layers: TUP (Telephone User Part)

The Telephone User Part (TUP) is used to set up a telephone call between two SS7 nodes. It defines a set of messages and a protocol using these messages that allows a telephone call to be set up and torn down.

TUP messages flow only immediately before a call is established and then immediately before it is terminated. No TUP messages are exchanged when a call is in progress.

TUP was one of the first SS7 protocols and designed to support simple analog phones (with little function over and above call setup and tear-down).

Upper Layers: ISUP (Integrated Services User Part)

The ISUP performs the same function as the TUP (that is, it handles the setup and tear-down of telephone calls) but it is much more sophisticated providing function available with primary rate ISDN. This includes calling and called number notification (or suppression),

the ability to control billing (charging) rates, advanced telephony functions such as transfer, and control over whether the voice channel is used for voice, fax, or data.

As with TUP, ISUP messages flow only during the setup and tear-down phases of a call.

Upper layers: SCCP (Signaling Connection Control Part)

The SCCP runs above the MTP layers and provides a set of facilities similar to those provided by the UDP and TCP layers of TCP/IP. Specifically, SCCP provides five classes of service such as connectionless (like UDP) and connection-oriented (like TCP) with options of error recovery and flow control. It also provides what is known in SS7 as Global Title Translation.

Upper layers: TCAP (Transaction Capabilities Application Part)

The TCAP is designed to implement functions in the SS7 network which are unrelated to the origination and termination of actual telephone calls. TCAP provides a means by which information can be transferred from an application at a switch location to another application in another network entity.

One example of TCAP usage is number translation and database transactions and lookup. Another example of the use of TCAP is the Message Waiting Indicator (MWI) on some telephones which indicates that a voice message is waiting for the subscriber. An SS7-connected voice mail system sends a TCAP message to the network to set the MWI flag in a subscriber's database.

Note that TCAP can be used by itself (on top of SCCP and the MTP layers), or it can be used as a transport layer for higher-level layers such as MAP and INAP (described in following sections).

Upper layers: MAP (Mobile Application Part)

Mobile Application Part (MAP) is the most complex SS7 component and is used in GSM mobile telephone systems to pass information between the components of the network.

Upper layers: INAP (Intelligent Network Application Part)

The Intelligent Network Application Part (INAP) is used to implement services within a network, which involve accesses to an SCP and might also involve the use of an Intelligent Peripheral (IP). INAP messages are sent between network entities using TCAP transactions.

Upper layers: OMAP (Operations and Administration Application Part)

The OMAP is typically used by a network administration facility to control an entire network from a central point. Facilities provided in

OMAP include administration of system databases, maintenance access and performance monitoring.

SS7 Support for WebSphere Voice Response, which is discussed in the next section, supports the MTP layers and the ISUP. If you need other layers to be supported contact your IBM representative.

Using SS7 with IBM WebSphere Voice Response

WebSphere Voice Response for AIX is a high performance interactive voice response (IVR) system which is capable of handling a wide range of voice response applications including voice messaging.

To interact with telephone callers, WebSphere Voice Response needs to be connected to the telephone network or a switch (telephone exchange). For SS7 Support for WebSphere Voice Response this connection must be digital and can be either E1 (30 channels multiplexed at 2.048 Mb per second) or T1 (24 channels multiplexed at 1.536 Mb per second).

Two types of information must flow between the network or switch and WebSphere Voice Response: PCM (Pulse Code Modulated) voice data, and signaling.

There are many options for signaling but the range of signaling types breaks down into two categories: Channel Associated Signaling (CAS) or Common Channel Signaling (CCS). There are two types of Common Channel Signaling protocols in general use: ISDN and SS7. We discuss only SS7 in this book.

The basic principle of CCS is that a data channel (usually 64 kb/s) is allocated to one of the timeslots on a trunk. This data channel is used to send messages between WebSphere Voice Response and the switch or network to control the flow of calls. Such messages for all channels are sent down the same signaling channel, hence the name, *common channel* signaling.

WebSphere Voice Response can receive incoming telephone calls and make outgoing telephone calls. As such, the only required signaling is to handle the setup and tear-down of telephone calls. This is provided for attachment outside telephone networks (customer premises equipment (CPE) environments) by the built-in CAS and CCS-ISDN signaling protocol handlers of WebSphere Voice Response.

SS7 Support for WebSphere Voice Response provides the equivalent capability for WebSphere Voice Response applications in a network. SS7 Support for WebSphere Voice Response is implemented as follows:

- An SS7 hardware adapter (available from SS8 Networks Inc) is plugged into the PCI bus of the pSeries computer. A separate E1 or T1 trunk carrying the

signaling traffic for one or more voice trunks is connected to the SS7 adapter. The adapter handles the MTP-1 layer of the SS7 stack and contains firmware supporting the MTP-2 layer.

- An SS7 software stack that runs on top of the SS7 adapter, and provides standards-compliant support for the MTP-1 and MTP-2 layers of the SS7 protocol stack. Although the stack is able to support most components of the stack above MTP-3, only ISUP is used for the WebSphere Voice Response basic call signaling application.
- The server component of SS7 Support for WebSphere Voice Response provides the support which allows the SS7 Server machine to connect to one or more WebSphere Voice Response SS7 Clients and provides the system configuration utilities. Configuration is discussed in Chapter 4, “Configuration using SS7itty,” on page 41.
- The client component of SS7 Support for WebSphere Voice Response contains a WebSphere Voice Response *signaling process* (a special WebSphere Voice Response custom server) and the SS7 Support for WebSphere Voice Response software provides the interface between the top of the ISUP layer of the SS7 protocol stack and the WebSphere Voice Response *Signaling Library*. This allows WebSphere Voice Response applications to receive incoming calls and to make outgoing calls without knowing the type of signaling being used so they don't need to change between CAS, ISDN, and SS7.

Although the above describes a logical view of the structure of WebSphere Voice Response's SS7 support, the actual implementation is more complicated in that the SS7 stack and signaling processes can run on different pSeries computers connected by a fast TCP/IP LAN. This allows for a distributed system where a single SS7 controller machine running the SS7 adapter and stack (called the SS7 Server in this book) can control up to six voice processor machines each running WebSphere Voice Response and the SS7 signaling process (called the SS7 Client in this book).

The SS7 Server can be duplicated in a load-sharing configuration to provide high reliability.

Example configurations

This section shows four diagrams giving examples of different configurations that you can set up with SS7 Support for WebSphere Voice Response. The first example represents the simplest single-machine non-redundant configuration, and the fourth example shows a fully-redundant eight machine (maximum) configuration. Two intermediate configurations are also shown.

The example configurations are as follows:

- “A minimum configuration using one pSeries computer” on page 7

- “A configuration using one SS7 Server and two SS7 Clients” on page 8
- “A configuration using two SS7 Servers and two SS7 Clients with some redundancy” on page 9
- “A maximum configuration with full redundancy” on page 10

A minimum configuration using one pSeries computer

Figure 2 shows the minimum configuration you can have using SS7 with WebSphere Voice Response. This configuration uses one pSeries computer which contains the SS7 adapter, the SS7 Server, and the SS7 Client running on WebSphere Voice Response.

From the network the signaling component of a telephone call goes through the SS7 switch and into the pSeries computer through the SS7 adapter, then to the SS7 Server, then to WebSphere Voice Response through the SS7 Client.

This is the simplest configuration you can use, and supports a maximum of 480 E1 or 384 T1 channels.

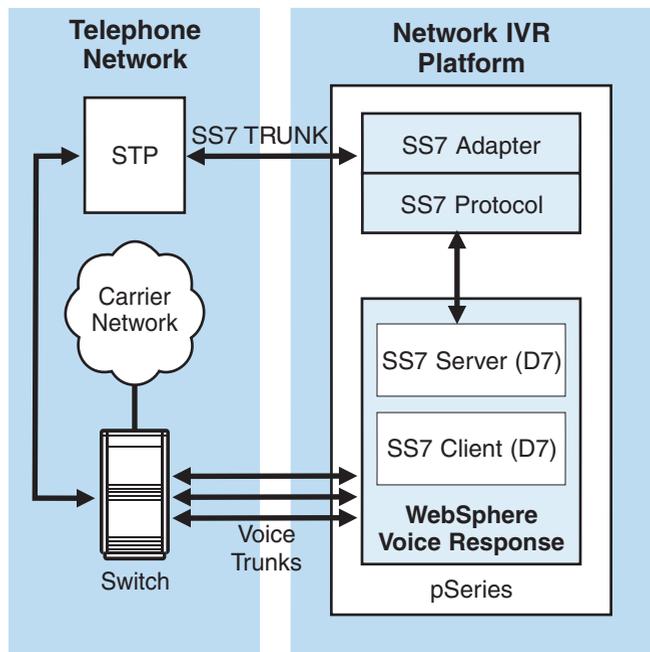


Figure 2. Typical layout for SS7 Support for WebSphere Voice Response signaling components (minimum configuration)

A configuration using one SS7 Server and two SS7 Clients

Figure 3 shows a configuration using one SS7 Server and two SS7 Clients. This configuration lets you use more channels and means that the load on your system can be spread.

Using this configuration, you can use up to 960 E1 or 768 T1 channels.

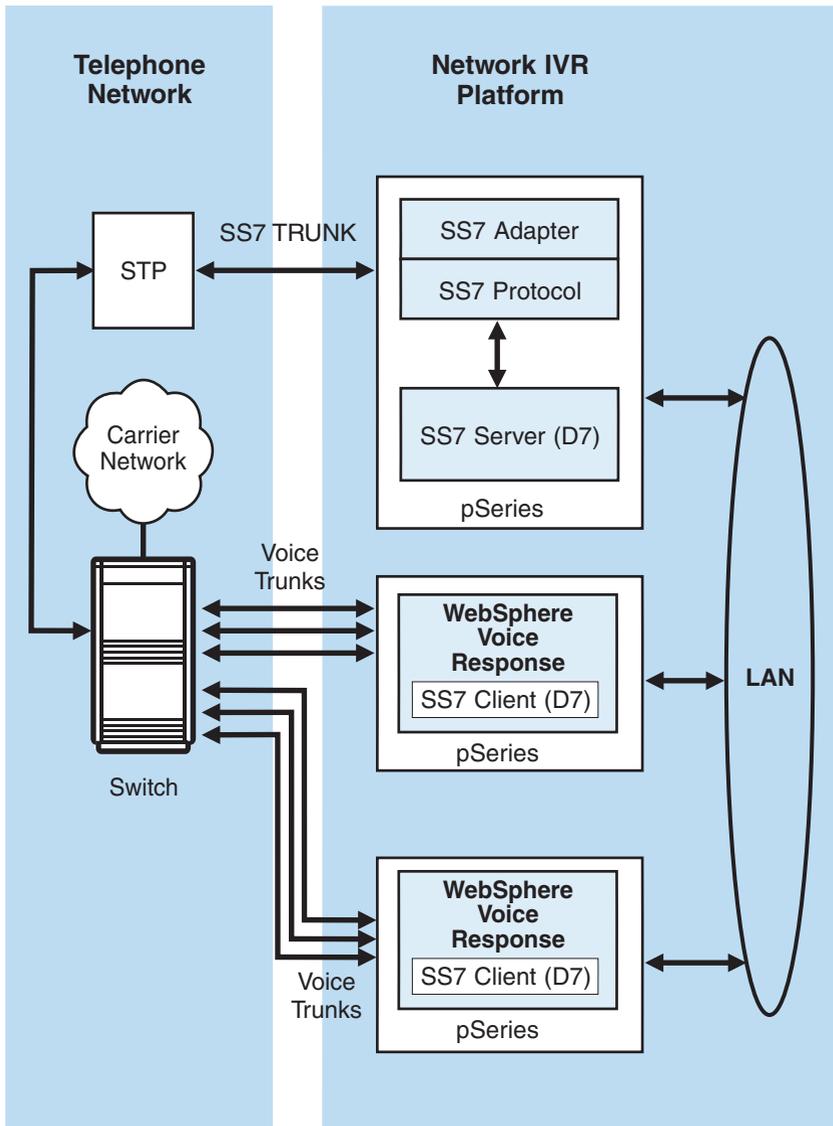


Figure 3. Configuration using one SS7 Server and two SS7 Clients

A configuration using two SS7 Servers and two SS7 Clients with some redundancy

Figure 4 shows a configuration using two SS7 Servers and two SS7 Clients. This configuration shows the two SS7 Servers each with links to the same STP.

Both SS7 Servers are active, sharing the SS7 traffic in a load-sharing configuration. The SS7 links are also both active and sharing the SS7 traffic. The SS7 Servers and the SS7 Clients are linked by a single LAN.

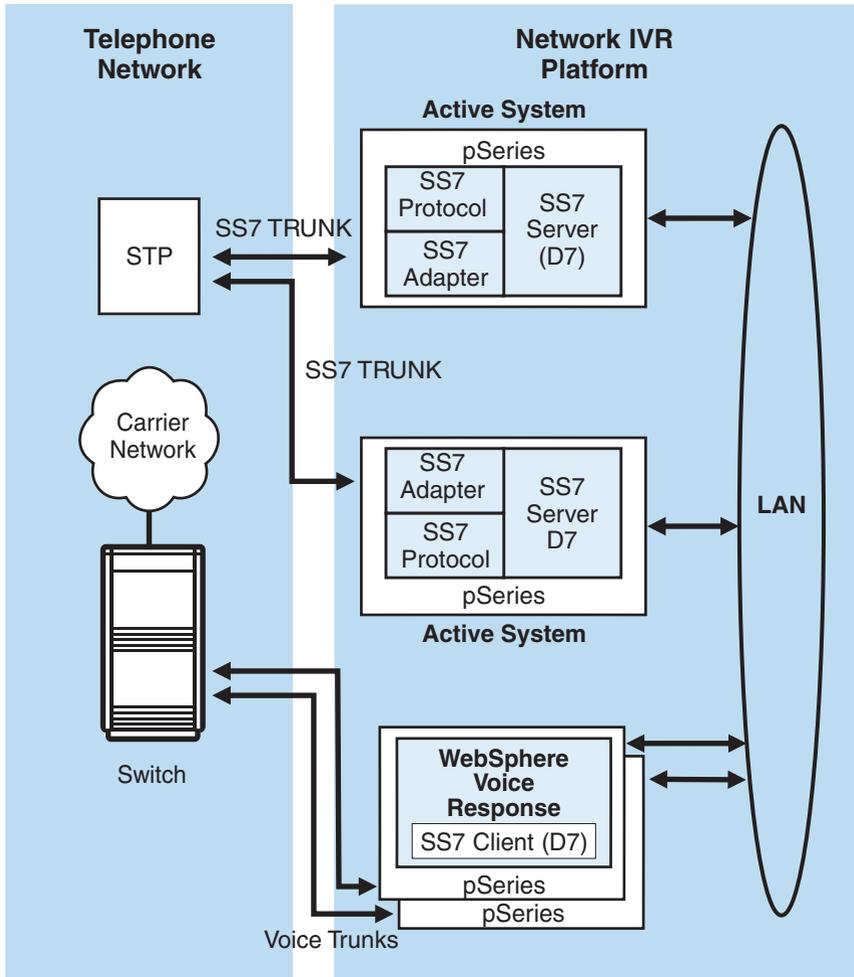


Figure 4. Configuration using two SS7 Servers and two SS7 Clients

Using this configuration, you can use up to 960 E1 or 786 T1 channels.

A maximum configuration with full redundancy

When SS7 Support for WebSphere Voice Response is configured for full redundancy you can expect it to have a high degree of resilience against all single point of failure situations, thereby ensuring high availability. Full redundancy provides the following:

SS7 adapter redundancy

This is provided in a SS7 configuration by multiple adapters in multiple SS7 Servers which are always active and processing SS7 Traffic. A failure in an adapter will result in the remaining adapters taking the additional SS7 traffic.

SS7 Server redundancy

A failure in a SS7 Server will result in the other SS7 Server taking the additional SS7 traffic.

WebSphere Voice Response SS7 Client redundancy

This can be provided by configuring sufficient SS7 Clients to have more voice bearers that are required. In this way, if an SS7 Client fails, the remaining SS7 Clients have spare capacity to take the additional call traffic.

LAN connection redundancy

This is provided by connecting your SS7 components on a dual LAN. Should the primary LAN fail, the LAN traffic is processed through the second LAN.

Note: WebSphere Voice Response database redundancy is not shown in the diagram, for information on this see *WebSphere Voice Response for AIX: Configuring the System*.

In this configuration, there is a "cross-over" format, with each SS7 Server having SS7 Links to two STPs, and similarly each STP has two SS7 Links to each SS7 Server. The two pSeries® computers are SS7 Servers and both are in active mode. When a problem is detected on a machine, the second machine takes over the traffic load.

The SS7 Servers are connected on a dual LAN to six pSeries computers that are running WebSphere Voice Response, with the SS7 Client giving support for redundancy in the case of LAN failure. This configuration supports a maximum of 2880 E1 or 2304 T1 channels, which can be distributed over the six SS7 Clients.

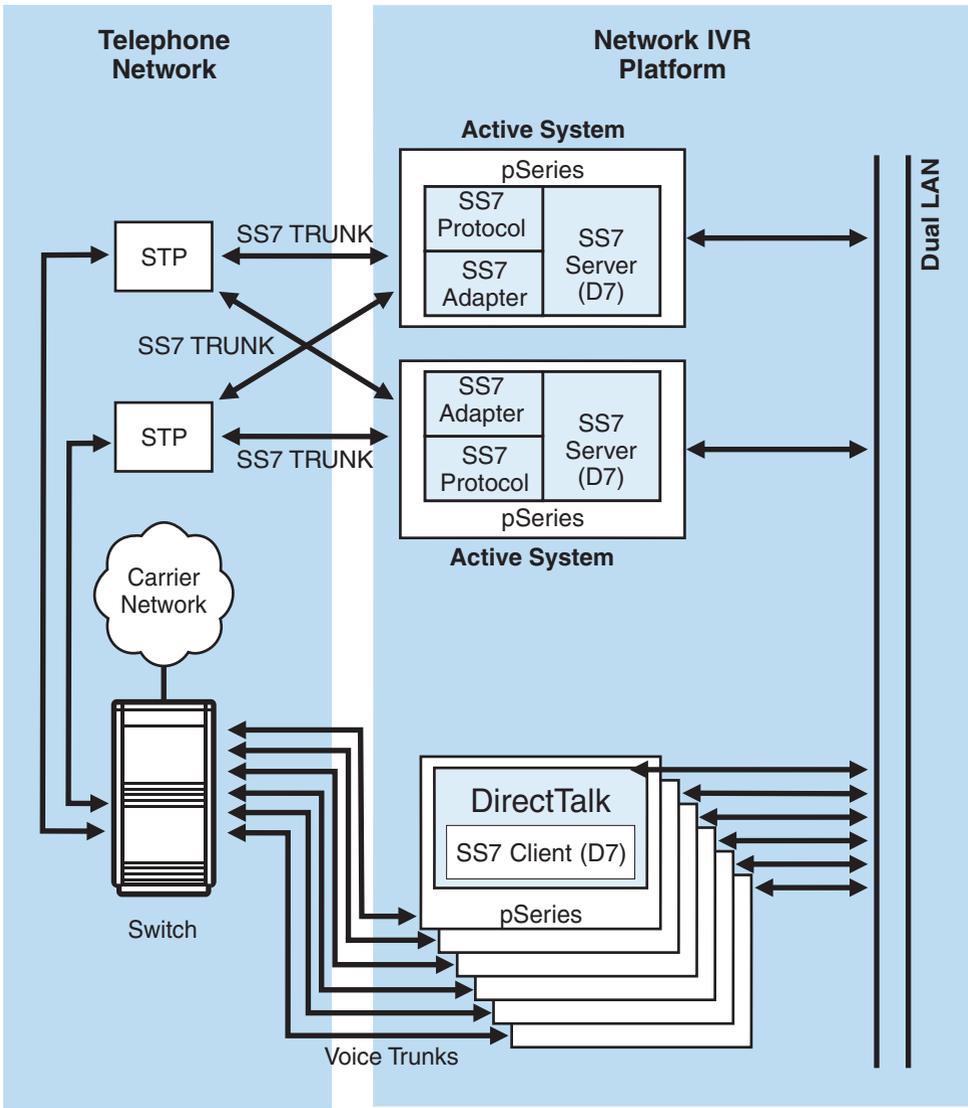


Figure 5. Maximum configuration with full redundancy

Chapter 2. Planning

This section of the book looks at how to prepare for the installation of SS7 Support for WebSphere Voice Response. You should also read the relevant section of the *WebSphere Voice Response for AIX: General Information and Planning* book.

Software

SS7 Support for WebSphere Voice Response is delivered in two parts:

- D7 Enablement, which contains software for the SS7 Server and WebSphere Voice Response client, is available as part of the WebSphere Voice Response base code. The SS7 Server software provides utilities and tools to support the configuration of the SS7 Server. The SS7 Client software provides the interface between Distributed7 and WebSphere Voice Response.
- A second CD, delivered as an IBM PRPQ (7J0465), which contains the Distributed7 software for both SS7 Server and WebSphere Voice Response client machines

Hardware

There are specific hardware requirements when running SS7 Support for WebSphere Voice Response.

SS7 adapter (mandatory)

This is a PCI adapter which must be installed in the same machine as your SS7 Server. This adapter handles the SS7 signaling from your switch and routes the signals over your network to your SS7 Server and your SS7 Client or Clients. You need at least one SS7 adapter for each SS7 Server.

The following SS7 adapter is supported. It is available from NewNet Communications Technologies.

- HAX50PCGEN T1 / E1 - quad port card

For more information see Chapter 3, “Installing SS7 Support for WebSphere Voice Response,” on page 21.

Supported pSeries computers

For use as an SS7 Server, all Power Systems in which the DTTA adapter has been previously supported and which is capable of running AIX 6.1. Refer to the WVR adapter guidance:

<http://www-01.ibm.com/support/doview.wss?rs=761&uid=swg21253839>

In addition, the following Power6 Power Systems:

- Power Systems p6 520, 550
- Power Systems p6 570 using 7311-D20 drawer for attachment of one or more SS7 adapters to the system

pSeries support for an SS7 Client is the same as for base WebSphere Voice Response installations. For detailed information refer to the *WebSphere Voice Response for AIX: General Information and Planning* book.

Configuration considerations

The number of SS7 Clients you choose to have is primarily determined by the number of calls you aim to support. SS7 Support for WebSphere Voice Response supports up to a maximum of 96 trunks, distributed over six WebSphere Voice Response clients. With SS7, WebSphere Voice Response supports both T1 and E1 telephony, which provide 24 or 30 channels (calls) per trunk respectively.

Therefore the total number of channels supported is:

Telephony	Circuits supported
T1	2304 (96 x 24)
E1	2880 (96 x 30)

WebSphere Voice Response currently supports a maximum of 480 E1 or 384 T1 channels per pSeries computer (that is, 120 channels per DTTA, up to a maximum of four DTTAs per pSeries computer). To achieve the maximum number of channels supported by SS7 you need six WebSphere Voice Response clients, each with four DTTAs.

The number of SS7 Servers you choose to have is determined by the level of redundancy you require. Redundancy requires two SS7 Servers, each with a SS7 adapter, working in parallel in a load-sharing format. If one of the SS7 Servers has a problem, the remaining one will handle the call capacity for both. If you choose to have a non-redundant single SS7 Server configuration and your SS7 Server fails, all calls to your WebSphere Voice Response application are lost and no new calls can be accepted until the situation is recovered.

Licensing the Distributed7 package

The Distributed7 package is licensed using a key file, which is included in the price, but is not shipped with the package. Although you can install the software without license key file `/usr/ss8/d7/access/etc/license.dat`, it will not run. To obtain the license key file, contact your IBM representative.

The license key file is specific to the machine in which you install your software, so you need to provide the machine id when you request the key file from your IBM representative. To find out the machine id, log onto the machine and use the following command:

```
uname -m
```

Each machine within a SS7 cluster needs a license file. When requesting licenses for a cluster, *do so as a complete group*, as each license needs to know the size of the cluster. If an additional WebSphere Voice Response client is added to a SS7 cluster, a new group of licenses is required, and the existing licenses need to be updated to reflect the increase in cluster size.

Note:

1. If you install additional SS7 adapters in a system unit, you may need a replacement license key file to handle the increased number of Links.
2. If you move Distributed7 to a different computer, you need to request a new license key file for that computer.

Call capacity

The number of SS7 Links (connections) that are required is calculated on the number of SS7 Servers and the expected maximum call capacity. The smallest unit of SS7 communication is a Link. This is a single timeslot within an E1 or T1 trunk, and is capable of transferring data at 64 kbps. Based on a typical SS7 message structure and 40% utilization, a single Link can support up to 30 calls per second. For example, if 105 calls/second is required, 4 Links are required to handle the capacity.

The number of SS7 Servers will already be defined. If the redundancy method of two SS7 Servers is used, the calculated number of Links can be divided evenly between them. If the number of Links is an odd number, increase the Link count by one. As a 40% utilization is used in the calculation, this configuration can still function correctly with one SS7 Server failing.

Preparing for installation

By this stage, you should know how many SS7 Servers and SS7 Clients you are going to use in your SS7 system. Below is a list of questions you can ask yourself to check that you have covered everything you are likely to need:

- How many SS7 Servers am I using in my configuration?
- How many SS7 adapters am I using?
- Do I have a license key file for each machine, and do they match the machine numbers?
- How many WebSphere Voice Response clients am I using in my configuration?
- How many trunks and channels are there on each WebSphere Voice Response machine?
- Am I using single or dual LAN?
- What is the OPC?
- What is the DPC?
- What CICs /trunks are to be assigned to which WebSphere Voice Response client?
- Has my switch been configured for T1 SS7 or E1 SS7?
- Does ISUP support my switch and network?
- How much redundancy am I using, and where in the system?
- How many SS7 Links are required?

If you can answer all the above questions, go to the next section, and focus on identifying and collecting the relevant information. This will reduce the likelihood of problems when performing the installation and configuration.

Collecting System Parameters

This section describes what SS7 system information is required and in what format it is needed. Appendix A has a number of useful tables to assist collection of this information while reading this section. By this stage of planning you should have a basic plan for what components are required, SS7 Servers, WebSphere Voice Response client, STP and so on.

SS7 interconnection information

Identifying the machines

The first stage is to identify the machines within the SS7 Cluster. Complete the "AIX machines" table in Appendix A, "Sample forms for recording information," on page 101 as follows:

Machine name

The value that is returned by the machine when executing the AIX command `hostname -s`.

Primary IP

The name given to the Primary IP address of this machine. Typically this would be the same as Machine Name. Note that this name cannot exceed 15 characters in length.

Secondary IP

This is optional and is the name given to the Secondary IP address of the machine. This is required if running with a Dual LAN configuration.

Machine type

Either SS7 Server or WebSphere Voice Response client

Serial number

The value returned by the machine when executing the AIX command `uname -m`. This information is required when requesting licences for these machines.

Physical Connections

The SS7 Server is connected to the Networks STP via E1 or T1 connections. These connections (known as Links) are made to the SS7 adapter in the SS7 Server and a similar port on the STP.

1. Decide how many connections are required; for example in Figure 2 on page 7, for a single connection between one SS7 Server and the network, there would be one connection. However in a configuration with full redundancy, as in Figure 5 on page 11, four connections are required.
2. Decide how many Links (time slots) are required in each connection. Typically, for a balanced system the same number of time slots would be used in each physical connection. The number of timeslots required is governed by the expected maximum call capacity. This is described in "Call capacity" on page 15.
3. Now complete the "Signaling Links" table in Appendix A, "Sample forms for recording information," on page 101. At this stage only complete the following columns:

Machine name

The primary host name of the SS7 Server to which the connection will be made.

SS7 adapter

The adapter type installed in the Machine.

Instance

The Instance number is required if there is more than one of the same type of adapter in the same machine. The first adapter is known as instance zero.

Port The port index for which the connection will be made to on the given adapter. The first port is known as port 1 and is the lowest connection seen when the adapter is in the vertical direction. If the adapter shows labelling as Port A, read that as Port 1 and Port B as Port 2.

Timeslot

The reference within the connection indicating which timeslot will be carrying the SS7 signaling. These values are 0-23 for T1 and 1-31 for E1. Note that if more than one timeslot is to be used, complete a row entry in the table for each timeslot.

Link Name

An optional field to assist in describing the connection.

Link Set/DPC & SLC

This is explained in the next section.

Grouping Connection and Addressing SS7

This section describes how to collect the defined Links (or connections) before allocating them into logical groups of Link Sets.

1. Decide how many Link Sets are going to be required. There should be one Link Set per STP (Signalling Transfer Point) and typically there will be one or two STPs, depending on the level of redundancy being used.
2. Identify the SS7 STPs with Point Codes; these values define the unit address of each STP within the SS7 network (similar to a LAN IP address). These values can take the form of x-y-z or a hex value—either is permitted when you enter these values into the Configurator. The SS7 cluster can have only one Point Code, even though there may be two SS7 Servers in the cluster (both SS7 Servers have the same Point Code for redundancy purposes).
3. Complete the Originating Point Code table in Appendix A, with the value issued by the Network. Each Link Set has a single Destination Point Code (DPC) which is the Point Code of the switch or STP. It can have the same format as that for the OPC. Review the Signaling Links table, group the Links into Link Sets (if there is more than one) and enter the DPC value for each entry. If there is only one Link Set, the DPC will be the same for all Links.
4. Identify each Link with a Signaling Link Code. These will have been issued by the Network Operator and each Link within a Link Set has a unique SLC. Add these to the Signaling Links table.

Additional Routing

This section describes routing messages beyond the Link Sets. Typically all the physical SS7 connections are to one of more Signaling Transfer Points (STPs), but most call control messages need to be routed beyond this point to a switch, using the following procedure:

1. Decide how many Route Sets are required. Typically this is the number of switches. If the network does not require additional routing, either because it is directly connected to the Switch, or because the STP already knows how to route the message, then a Route Set is required for each Link Set.
2. For each Route Set, identify the destination point code (DPC) and the trunks associated with that switch.
3. For each Route Set, identify which of the Link Sets can route the messages. Up to four Link Sets can be specified.

Connecting the Voice Bearers

The final stage is to identify how the Voice Bearers are connected between the various WebSphere Voice Response systems and the switches. Each Route Set has a defined set of trunks or CICs connected to the switches. Similarly, there will be one or more WebSphere Voice Response systems, also with a number of trunks. Complete the "Trunks" table in Appendix A as described below, creating a new entry for each combination of Route Set name and WebSphere Voice Response name. Multiple trunks from the same Route Set or WebSphere Voice Response system can be listed in one entry. Notes that CIC ranges must be unique across all Route Sets. If they are not, ask your network operator re-configure them.

SS7 Trunk

A reference of trunks on the network side. If you have the references in a range of CICs they should be converted as follows:

- For T1 connections, first take the lowest CIC in the range, divide by 24 and add one to the answer (for example, CIC 0 would be trunk 1). Then take the highest CIC in the range, add one to it and divide by 24 (for example CIC 383 would be trunk 16). Note that the lowest CIC must be multiple of 24 and similarly the last CIC after adding one must also be a multiple of 24. Trunks cannot be subdivided.
- For E1 connections, the same process is followed as for T1, except the value 24 is replaced with the value 32. Some CIC ranges may start with the first voice circuit and not the framing timeslot, in which case subtract one from the lowest CIC before dividing by 32. As an example a CIC range of 0-511 would translate to Trunks 1-16.

Route Set Name

The Route Set name associated with the SS7 Trunks.

WVR Trunk

This is the allocated trunks from the WebSphere Voice Response client to the SS7 cluster. Typically this would be all the trunks, for example 1-8 (WebSphere Voice Response with two adapters) or 1-16 (WebSphere Voice Response with four adapters).

WVR Name

This is the name given to the WebSphere Voice Response client in the "AIX Machines" table. Note that the number of trunks listed in an SS7 Trunk must be the same as the number of WebSphere Voice Response trunks. That is, if there are 16 trunks from a WebSphere Voice Response system, they must be connected to 16 trunks on a switch.

Chapter 3. Installing SS7 Support for WebSphere Voice Response

This section describes the installation of the Distributed7 (D7) package and its associated D7 Enablement package on different types of SS7 configuration.. To simplify the installation in an SS7 cluster, the order of machine-types on which you install should be as follows:

1. Single system image (SSI) server
2. SS7 Server
3. WebSphere Voice Response client

Pre-installation checking

A critical element in ensuring that the D7 package runs correctly is for the **dtuser** userid (or whichever userid is used) to be the same on all machines within the SS7 cluster.

Before the D7 and the D7 Enablement packages are installed, ensure that all WebSphere Voice Response systems have had the WebSphere Voice Response base components installed, and that the WebSphere Voice Response user IDs (typically **dtuser**) are matching. This can be verified and changed by using the following command:

```
smitty user
```

For more detailed information about setting up WebSphere Voice Response configurations, refer to the *WebSphere Voice Response for AIX: Installation* book.

When the D7 and the D7 Enablement packages are installed on the SS7 Server, a userid of dtuser is created (if not already present). Verify that the userid also matches the userids on the WebSphere Voice Response Clients. If dtuser is not the preferred userid, delete it and create a new user with the desired name and matching userid. It is important that you do this before the use of the SS7_MAINT tools, as a number of files are created and copied with the userid.

Locating the required files

The D7 and the D7 Enablement packages are installed from the following locations:

- D7 is supplied on the PRPQ CD entitled "IBM SS7 Support for WebSphere Voice Response".

When installing from this CD, using the command `smitty`, select the **d7.isup** fileset, ensuring that **AUTOMATICALLY install requisite software?** is set to *yes*.

- D7 Enablement is provided on CD as part of the base WebSphere Voice Response package. The fileset name is `dirTalk.SS7_D7.D7_Enablement`. Instructions on how to install this package are provided in *WebSphere Voice Response for AIX: Installation*.

Installation on an SSI server

This section tells you how to install SS7 Support for WebSphere Voice Response on a WebSphere Voice Response SSI server. The installation of these packages onto this machine must conform to WebSphere Voice Response standards on loading custom servers onto SSI machines (see the *WebSphere Voice Response for AIX: General Information and Planning* for more information).

Having the D7 Enablement package installed on this machine assists when using the `SS7itty` Configurator, as the resulting files are made available on the relevant WebSphere Voice Response client machines without the need for copying. Note that only the D7 Enablement package will be installed, as the SSI Server has no active D7 code and so no D7 license file is required.

There are five stages to installing and configuring the SSI Server:

1. Pre-installation
2. Installing the D7 Enablement package.
3. Configuring the SS7 Cluster
4. Configuring the AIX machine for SSI Server usage
5. Installation and configuration complete

Pre-installation

Before installing this software package read the `README.TXT` file on the CD-ROM to see if there any special instructions. Ensure this SSI Server has been fully loaded with the WebSphere Voice Response package and configured for SSI Server usage.

Installing the D7 Enablement package

Use the following procedure to install the D7 Enablement package on to your SSI Server:

1. Log on as root to the machine on which you are going to install this package.
2. Use `smitty` to install the following fileset:

- dirTalk.SS7_D7.D7_Enablement (D7 Enablement, from the base WebSphere Voice Response package)

The D7 package fileset is not required.

Configuring the SS7 Cluster

Refer to Chapter 4, "Configuration using SS7itty," on page 41 for details. The configurator performs the configuration of the SS7 Cluster.

Configuring the AIX machine for SSI Server usage

To configure the machine to function as an SSI Server for SS7 operations:

1. Log on as dtuser. If you are already logged on, log off and then log on again, as the login profile will have changed.
2. Start WebSphere Voice Response by entering the command vaeinit. Wait for the startup process to complete.
3. Type the command SS7_MAINT. The Top Level Menu screen is displayed.
4. Type S to select the **SS7 Server/WebSphere Voice Response Client** choice. The Server/Client Configuration screen is displayed.
5. Type I to select the **Configure machine as SSI Server (No D7 present)** choice.

If error messages relating to the importing of the SS7_D7 custom server are displayed, these can be ignored as they refer only to the fact that the custom server is already loaded.

Once completed, the SSI Server is configured for SS7 operation.

Installation and configuration complete

All the necessary actions to install and configure the SSI Server for SS7 operations are now complete. To install and configure other types of system in this SS7 cluster, see the subsequent sections. When the installation of the complete SS7 cluster has been proven, refer to the section "Put into Production" to ensure that all operations have been setup in automatic mode.

Installation of SS7 Adapters

The SS7 adapters can be installed in the same PCI slots that are supported by DTTA cards. For more information see *WebSphere Voice Response for AIX: General Information and Planning*, and refer to the SS8 "Distributed7 Installation and Maintenance" book that is provided on the SS7 Support for WebSphere Voice Response CD.

Installation on an SS7 Server

This section tells you how to install the SS7 Support for WebSphere Voice Response onto an AIX SS7 Server. There are five stages to installing and configuring the SS7 Server:

1. Pre-installation
2. Installing the D7 and the D7 Enablement packages
3. Configuring the SS7 Cluster
4. Configuring the AIX machine for SS7 Server usage
5. Installation and configuration complete

Pre-installation

Before installing these software packages read the README.TXT file on the CD-ROM to see if there are any special instructions.

Installing the D7 and the D7 Enablement packages

Use the following procedure to install the D7 and the D7 Enablement packages onto your SS7 Server.

1. Logon to the computer with a userid of root.
2. Using smitty, install the following file sets:
 - dirTalk.SS7_D7_Enablement (D7 Enablement, from the base WebSphere Voice Response package)
 - D7.isup (D7, from the SS7 Support for WebSphere Voice Response CD). Refer to the readme file located in /usr/lpp/dirTalk/readme/SS7README, as this stipulates which level of D7 is required. If more than one level is required, install the packages in chronological order, then refer to “D7 software release” on page 57 to select the relevant package, before proceeding.
3. Now copy the relevant Distributed7 license for this machine (see “Licensing the Distributed7 package” on page 15) into /usr/ss8/d7/access/etc/license.dat .

The D7 package is installed into the directory /usr/ss8/d7. The D7 Enablement package is installed into /usr/lpp/dirTalk, and creates a dtuser userid if one does not already exist. This userid has not yet been assigned a password and therefore can only be accessed via the root userid.

Configuring the SS7 cluster

If this is the first SS7 Server, refer to Chapter 4, “Configuration using SS7itty,” on page 41, and then continue to the next section. Otherwise use the following instructions:

1. Logon as dtuser.
 2. Change directory to `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg`. Using `mkdir`, create a directory of the same name as this machine (the name returned if you enter the `hostname -s` command).
- Note:** If the `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg` directory does not exist, create it by running the `ss7itty` command, followed by `exit`.
3. Using `ftp`, connect to the original SS7 Server on which `SS7itty` was run.
 4. Change both the local and remote directories to `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/AnyMachine`, by using the `ftp` commands `cd` and `lcd`.
 5. Using the `ftp` command `mget *`, copy all the files that are in this directory.
 6. Change both local and remote directories to `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/hostname`
 7. Using the `ftp` command `mget *`, copy all the files that are in this directory
 8. Exit from `ftp`

Configuring the AIX machine for SS7 Server Usage

This section describes how to configure the machine for SS7 Server operation.

1. Logon as dtuser.
2. Enter the command `SS7_MAINT` and select the following options:
 - S** SS7 Server/WebSphere Voice Response client
 - S** Configure machine as an SS7 Server.

Once completed the AIX machine is configured for SS7 Server operation. The status line shown just before the prompt should show:

```
<name> is configured for D7 Server usage
```

 - R** Return to top level menu
 - D** D7 Operation (Start/Stop)
 - M** Start the D7 package with Alarm Message to screen. The following text appears:

```

Starting D7 package
Application Process Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
D7 Status - [init] : D7 core starting
Select an option :
Master Event Log Daemon - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Signalling Point Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Network Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
+++++
+
+          <<< IMPORTANT NOTE >>>          +
+
+ Operations of the Application Process Manager +
+ are now suspended indefinitely !!!          +
+
+ To continue, use either MMI/MML or AccessMOB +
+ interface to set the product operation mode +
+ (i.e., stand-alone vs. distributed) and      +
+ introduce any remote host machines involved. +
+
+ It is only after you perform these tasks,    +
+ Application Process Manager will resume its  +
+ normal operations & proceed with its routine +
+ system software initialization/start-up ...   +
+
+++++

```

Ignore the message in the "Important Note" and press Enter to recover the command prompt.

- R** Return to top level menu
- C** MML configuration loader
- H** Load Host MML configuration for this machine. The following text appears:

```

*** Script loading operation - completed
Loading the script has been successful
The log output can be reviewed in /usr/lpp/dirTalk/sw/ss7/log/mml-host.log

```

After about 30-60 seconds the following text appears:

```
Alarm Daemon - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Distributed Shared Memory Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Distributed Kernel Memory Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
User Part Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
ISDN User Part - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
```

If this is the first of two SS7 Servers, you should now suspend installation on this server, and repeat on the second server the instructions that you have completed up to this point. Once you have completed that activity on the second server, return to this server and complete the rest of the instructions in this section (note that there is no further action required on the second server).

- S Load SS7 MML configuration. This loads the SS7-specific information, such as card configuration, Link, Link Sets and circuit configuration. The following text appears:

```
Waiting for MTP to be ready ...
Installing MTP components ...
NOTICE: spmd: pci3xapq[0] configured.
Waiting for ISUP to be ready ...
NOTICE: isup0: configuration started
NOTICE: isup0: database sync in progress ...
NOTICE: isup0: ready
Installing ISUP components ...
*** Script loading operation - completed
Loading the script has been successful
Review the log output in /usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/log/mml.log
```

Note: The action of loading this configuration automatically enables other functions, such as activating Links etc. Therefore the text may be interspersed with D7 error messages. This is normal if the remote end of the SS7 links has not yet been configured.

Installation and configuration complete

All the necessary actions to install and configure the SS7 Server have now been completed. Continue the installation and configuration for the other machines in this SS7 cluster. If error messages are still being generated once all the machines have been configured, refer to Chapter 7, “Problem determination,” on page 79 for assistance. Once the complete installation of the SS7 cluster has been validated, refer to “Putting into production” on page 38 to ensure that all operations have been set to automatic mode.

Installation on a WebSphere Voice Response client

This section tells you how to install SS7 Support for WebSphere Voice Response onto a standard WebSphere Voice Response with or without SSI support. There are five stages to installing and configuring the WebSphere Voice Response client:

1. Pre-installation
2. Installing the D7 and the D7 Enablement packages
3. Configuring the SS7 Cluster
4. Configuring for WebSphere Voice Response client usage
5. Installation and configuration complete.

Pre-installation

Before installing these software packages, read the README.TXT file on the CD-ROM to see if there are any special instructions. Ensure that the WebSphere Voice Response base packages have already been installed on this client machine, and that any SSI configuration (if relevant) has been completed.

You should configure the SS7 Server first and, if relevant, install the D7 Enablement package on the SSI Server *before* starting this client installation.

Installing the D7 and the D7 Enablement packages

Use the following procedure to install the D7 and the D7 Enablement packages on your AIX machine:

1. Logon as root.
2. Using smitty, install the following file sets:
 - dirTalk.SS7_D7_Enablement (D7 Enablement, from the base WebSphere Voice Response package)
 - D7.isup (D7, from the SS7 Support for WebSphere Voice Response CD). Refer to the readme file located in /usr/lpp/dirTalk/readme/SS7README, as this stipulates which level of D7 is required. If more than one level is required, install the packages in chronological order, then refer to “D7 software release” on page 57 to select the relevant package, before proceeding.
3. Now copy the relevant Distributed7 license for this machine (see “Licensing the Distributed7 package” on page 15) into /usr/ss8/d7/access/etc/license.dat .

The D7 package is installed into the /usr/ss8/d7 directory, and the D7 Enablement package is installed into /usr/lpp/dirTalk.

Configuring the SS7 cluster

This section describes how to locate the necessary configuration for D7. It is assumed that the guidance provided in “Pre-installation” on page 28 has been followed and one of the following two scenarios will apply:

1. SS7 Cluster with SSI support.

The D7 Enablement package is already loaded onto the SSI machine and the SS7itty configuration has been performed. No additional work is required as the SSI will make all relevant files available to the WebSphere Voice Response machines.

2. SS7 Cluster without SSI support

The D7 and the D7 Enablement packages are already loaded onto one or more SS7 Servers and the SS7itty configuration is complete. In this scenario, follow the instructions below to copy the relevant configuration files for this machine:

- a. Logon as dtuser.
- b. Change directory to `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg`, then use the `mkdir` command to create a directory with the same name as this machine (the name returned if you enter the `hostname -s` command).

Note: If the `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg` directory does not exist, create it by running the `ss7itty` command, followed by `exit`.

- c. Using `ftp`, connect to the original SS7 Server on which the `SS7itty` command was entered.
- d. Using the `ftp` commands `cd` and `lcd`, change both the local and remote directories to `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/AnyMachine`.
- e. Using the `ftp` command `mget *`, copy all the files in this directory.
- f. Change both local and remote directories to `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/hostname`.
- g. Using the `ftp` command `mget *`, copy all files in this directory.
- h. Exit from `ftp`.

Configuring the AIX machine for WebSphere Voice Response client usage

This section describes how to configure the machine for WebSphere Voice Response client operation:

1. Login using the `dtuser` userid. If you are already logged in, logoff and then login again, because the login profiles will have changed.

2. Start WebSphere Voice Response by entering the command `vaeinit`. Wait for the startup process to complete.
3. Returning to the original command prompt window, enter the command `SS7_MAINT`, selecting the following options:

S SS7 Server/WebSphere Voice Response client

W Configure machine as a WebSphere Voice Response client.

Once completed, the AIX machine is configured for WebSphere Voice Response client operation. Two custom servers (`SS7_D7` and `D7WVRErrorReport`) will have been installed into WebSphere Voice Response and can be viewed in the Custom Server Manager GUI.

R Return to top level menu

D D7 Operation (Start/Stop)

M Start the D7 package with Alarm Message to screen. The following text appears:

```
Starting D7 package
Application Process Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
D7 Status - [init] : D7 core starting
Select an option :
Master Event Log Daemon - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Signalling Point Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Network Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
+++++
+
+               <<< IMPORTANT NOTE >>>               +
+
+ Operations of the Application Process Manager         +
+ are now suspended indefinitely !!!                   +
+
+ To continue, use either MMI/MML or AccessMOB       +
+ interface to set the product operation mode         +
+ (i.e., stand-alone vs. distributed) and             +
+ introduce any remote host machines involved.       +
+
+ It is only after you perform these tasks,          +
+ Application Process Manager will resume its         +
+ normal operations & proceed with its routine       +
+ system software initialization/start-up ...         +
+
+
+++++
```

Ignore the message in the "Important Note" and press Enter to recover the command prompt.

R Return to top level menu

C MML configuration loader

H Load Host MML configuration for this machine. The following text appears:

```
*** Script loading operation - completed
Loading the script has been successful
The log output can be reviewed in /usr/lpp/dirTalk/sw/ss7/log/mml-host.log
```

After about 30-60 seconds the following text appears:

```
Alarm Daemon - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Distributed Shared Memory Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Distributed Kernel Memory Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
```

Installation and configuration complete

All the necessary actions to install and configure the WebSphere Voice Response client are now complete. Now continue with any remaining installations and configurations for other machines in this SS7 cluster. If error messages are still being generated once all machines have been configured, refer to Chapter 7, “Problem determination,” on page 79 for assistance. Once the complete installation of the SS7 cluster has been validated, refer to “Putting into production” on page 38 to ensure that all operations have been set to automatic mode.

Installation of a combined SS7 Server and WebSphere Voice Response client

This section tells you how to install SS7 Support for WebSphere Voice Response on an AIX machine for both SS7 Server and WebSphere Voice Response client configuration. There are five stages to installing and configuring the SS7 Server:

1. Pre-installation
2. Installing the D7 and the D7 Enablement packages
3. Configuring the WebSphere Voice Response client and SS7 Cluster.
4. Configuring the AIX machine for WebSphere Voice Response client and SS7 Server usage.
5. Installation and configuration complete

Pre-installation

Before installing these software package read the README.TXT file on the CD-ROM to see if there any special instructions.

Installing the D7 and the D7 Enablement packages

Use the following procedure to install the D7 and the D7 Enablement packages on your AIX machine:

1. Logon with a user id of root.
2. Using smitty, install the following file sets:
 - dirTalk.SS7_D7_Enablement (D7 Enablement, from the base WebSphere Voice Response package)
 - D7.isup (D7, from the SS7 Support for WebSphere Voice Response CD). Refer to the readme file located in /usr/lpp/dirTalk/readme/SS7README, as this stipulates which level of D7 is required. If more than one level is required, install the packages in chronological order, then refer to "D7 software release" on page 57 to select the relevant package, before proceeding.
3. Now copy the relevant Distributed7 license for this machine (see "Licensing the Distributed7 package" on page 15) into /usr/ss8/d7/access/etc/license.dat .

The D7 package is installed into the /usr/ss8/d7 directory, and the D7 Enablement package is installed into /usr/lpp/dirTalk, creating a dtuser user id. This user id has not yet been assigned a password, and so can only be accessed via the root user id..

Configuring for combined WebSphere Voice Response client and SS7 Cluster

If this is a first machine, refer to Chapter 4, "Configuration using SS7itty," on page 41. Otherwise, if this is a second or subsequent machine for the same cluster, follow the instructions below:

1. Logon as dtuser.
2. Change directory to /usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg, then use the mkdir command to create a directory with the same name as this machine (the name returned if you enter the hostname -s command).

Note: If the /usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg directory does not exist, create it by running the ss7itty command, followed by exit.

3. Using ftp, connect the original SS7 Server on which the SS7itty command was entered.

4. Using the ftp commands `cd` and `lcd`, change both the local and remote directories to `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/AnyMachine`.
5. Using the ftp command `mget *`, copy all files in this directory.
6. Change both local and remote directories to `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/hostname`.
7. Using the ftp command `mget *`, copy all files in this directory.
8. Exit from ftp.

Configuring the AIX machine for WebSphere Voice Response client and SS7 Server usage

This section describes how to configure the machine for WebSphere Voice Response client and SS7 Server operation.

1. Logon as dtuser.
2. Enter the command `SS7_MAINT`, and select the following options:
 - S** SS7 Server/WebSphere Voice Response client
 - S** Configure machine as a combined SS7 Server and WebSphere Voice Response client.

Once completed the AIX machine is configured for SS7 Server and WebSphere Voice Response client operation. The status line shown just before the prompt should show:

`<name>` is configured for combined D7 WebSphere Voice Response and SS7 Server usage

- R** Return to top level menu
- D** D7 Operation (Start/Stop)
- M** Start the D7 package with Alarm Message to screen. The following text appears:


```

Alarm Daemon - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Distributed Shared Memory Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Distributed Kernel Memory Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
User Part Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
ISDN User Part - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved

```

If this is the first of two SS7 Servers, you should now suspend installation on this server, and repeat on the second server the instructions that you have completed up to this point. Once you have completed that activity on the second server, return to this server and complete the rest of the instructions in this section (note that no further action is required on the second server).

- S Load SS7 MML configuration. This loads the SS7-specific information, such as card configuration, Link, LinkSets and circuit configuration. The following text appears:

```

Waiting for MTP to be ready ...
Installing MTP components ...
NOTICE: spmd: pci3xapq[0] configured.
Waiting for ISUP to be ready ...
NOTICE: isup0: configuration started
NOTICE: isup0: database sync in progress ...
NOTICE: isup0: ready
Installing ISUP components ...
*** Script loading operation - completed
Loading the script has been successful
Review the log output in /usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/log/mml.log

```

Note: The action of loading this configuration automatically enables other functions, such as activating Links etc. Therefore the text may be interspersed with D7 error messages. This is normal if the remote end of the SS7 link has not yet been configured.

- S SS7 Server/WebSphere Voice Response client
- S Configure machine as combined SS7 Server and WebSphere Voice Response client.

Once completed, the AIX machine is configured for SS7 Server and WebSphere Voice Response client operation. The status line shown just before the prompt should show:

<name> is configured for combined D7 WebSphere Voice Response and SS7 Server usage

- R Return to top level menu
- D D7 Operation (Start/Stop)
- M Start the D7 package with Alarm Message to screen. The following text appears:

```

Starting D7 package
Application Process Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
D7 Status - [init] : D7 core starting
Select an option :
Master Event Log Daemon - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Signalling Point Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Network Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
+++++
+
+           <<< IMPORTANT NOTE >>>           +
+
+ Operations of the Application Process Manager +
+ are now suspended indefinitely !!!           +
+
+ To continue, use either MMI/MML or AccessMOB +
+ interface to set the product operation mode  +
+ (i.e., stand-alone vs. distributed) and      +
+ introduce any remote host machines involved. +
+
+ It is only after you perform these tasks,    +
+ Application Process Manager will resume its  +
+ normal operations & proceed with its routine +
+ system software initialization/start-up ...  +
+
+
+++++

```

Ignore the message in the "Important Note" and press Enter to recover the command prompt.

- R Return to top level menu
- C MML configuration loader
- H Load Host MML configuration for this machine. The following text appears:

```

*** Script loading operation - completed
Loading the script has been successful
The log output can be reviewed in /usr/lpp/dirTalk/sw/ss7/log/mml-host.log

```

After about 30-60 seconds the following text appears:

```

Alarm Daemon - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Distributed Shared Memory Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
Distributed Kernel Memory Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
User Part Manager - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved
ISDN User Part - Version v1.3.1
Copyright (c) 1991-2004, SS8 Networks, Inc.
All Rights Reserved

```

If this is the first of two SS7 Servers, you should now suspend installation on this server, and repeat on the second server the instructions that you have completed up to this point. Once you have completed that activity on the second server, return to the first server and complete the rest of the instructions in this section (note that no further action is required on the second server).

S

Load SS7 MML configuration. This loads the SS7-specific information, such as card configuration, Link, LinkSets and circuit configuration. The following text appears:

```

Waiting for MTP to be ready ...
Installing MTP components ...
NOTICE: spmd: pci3xapq[0] configured.
Waiting for ISUP to be ready ...
NOTICE: isup0: configuration started
NOTICE: isup0: database sync in progress ...
NOTICE: isup0: ready
Installing ISUP components ...
*** Script loading operation - completed
Loading the script has been successful
Review the log output in /usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/log/mml.log

```

Note: The action of loading this configuration automatically enables other functions, such as activating Links. Therefore the text may be interspersed with D7 error messages. This is normal if the remote end of the SS7 link has not yet been configured.

Installation and configuration Complete

All the necessary actions to install and configure the WebSphere Voice Response client and the SS7 Server are now completed. Now continue with the installation and configuration for any remaining machines in this SS7 cluster. If error messages are still being generated after all the machines are configured, refer to Chapter 7, “Problem determination,” on page 79 for

assistance. Once the complete installation of the SS7 cluster has been validated, refer to “Putting into production” to ensure that all operations have been set to automatic mode.

Putting into production

Once the D7 and D7 Enablement packages have been installed, successfully configured, and have passed testing, there are a number of actions required to ensure that the packages will auto-restart if the machine is rebooted:

Custom servers

To ensure that the custom servers are started automatically when WebSphere Voice Response is started, use the Custom Server Manager GUI (which is accessed from the top level GUI) to check that the IPL Status for both the custom servers (SS7_D7 and D7WVRErrorReport) are set to AUTOEXEC. This should be done for all WebSphere Voice Response clients.

D7 package

To ensure that D7 automatically restarts after a system reboot, run the SS7_MAINT command when logged in as dtuser, and select the menu option **D** to enable operation of D7, followed by **E** to enable D7 to start after system boot. This action should be performed on all machines in the SS7 cluster.

D7 Alarm Logging

The default operation of the D7WVRErrorReport custom server is to collect D7 alarms that are generated by the machine on which it is running. For those SS7 Servers where WebSphere Voice Response is not installed, the custom server is not available to collect these errors. To ensure that D7 Alarms are collected and stored into the WebSphere Voice Response system, one or more WebSphere Voice Response systems can be selected to collect D7 Alarms from the SS7 Servers.

This operation involves instructing the D7WVRErrorReport custom server to register itself globally in the SS7 Cluster, and thereby be able to collect D7 alarms from other machines. Note that a custom server that is globally registered in this way does not collect D7 Alarms from other WebSphere Voice Response machines if they are running D7WVRErrorReport themselves, as a local custom server takes precedence.

To enable a D7WVRErrorReport custom server for global registration, follow the menu path below from the top level menu in the WebSphere Voice Response GUI.

1. From the top-level WebSphere Voice Response menu, click **Applications**.

2. From the drop-down menu, select **Custom Servers**.
3. From the custom server list, double-click **D7WVRErrorReport**.
4. Click **File**
5. Click **Properties**

The Properties panel is displayed, showing six separate windows. In the top left window, labelled **main () args**, ensure that the characters **-r** and **-q** appear on separate lines, as shown in Figure 6.

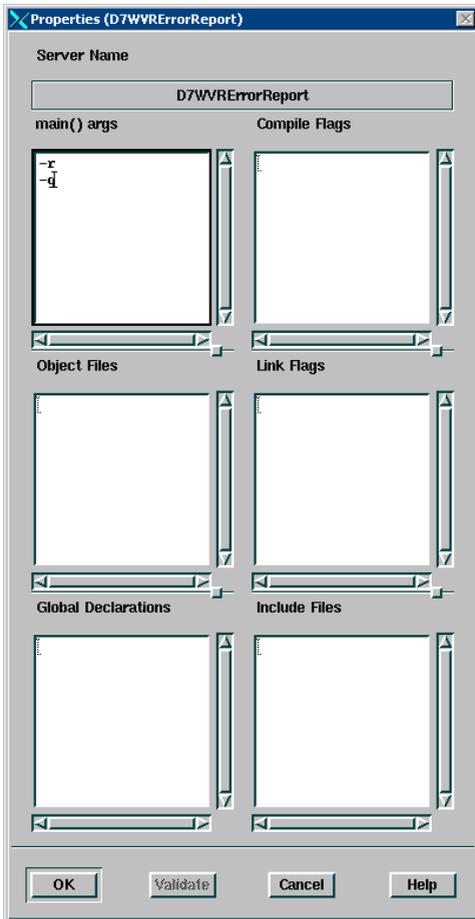


Figure 6. The Properties panel for the D7WVRErrorReport custom server

Chapter 4. Configuration using SS7itty

The section covers the use of the SS7itty configurator, which is based on the architecture of the AIX smitty facility. It takes the requirements of the SS7 Cluster and constructs the relevant configuration files.

Preparation

The first stage is to collect the relevant information together as described in Chapter 2, “Planning,” on page 13.

The SS7itty only needs to be run on one machine in the SS7 Cluster as it creates files for all the machines. If a WebSphere Voice Response SSI server is present in the cluster, use this to run SS7itty, as the configuration is then automatically distributed to the other WebSphere Voice Response machines (although manual copying to the SS7 Servers is required).

If no SSI is present, select one of the SS7 Servers.

Running SS7itty

To run SS7itty, first log on to the selected machine as dtuser, and enter the command SS7itty.

The follow screen appears:

```

                                     SS7 Configuration
Move cursor to desired item and press Enter.
Configure Environmental Conditions
Configure AIX Systems
Configure SS7 Server
Configure WVR Client
Configure SS7 Links
Configure SS7 Route Sets
Configure SS7 Trunks to WVR Trunks
Generate SS7 Configuration Files
Export SS7 Configuration
Import SS7 Configuration
F1=Help          F2=Refresh      F3=Cancel      F8=Image
F9=Shell        F10=Exit       Enter=Do
```

The natural progression is to go to each configuration option in turn and then perform the Generate (after pressing Enter). The Export/Import function, which is described later, enables backing-up of data and recovery of previous configurations. The key operations are the same as for the AIX smitty command.

On most submenus, there are four options:

Add Creates a new entry

Change/Show

Modifies or displays the details of an existing entry

Delete

Deletes an entry

Show Shows all the current entries

As an example, the submenu for Configure AIX Systems is shown below:

```

                                Configure AIX Systems
Move cursor to desired item and press Enter.
  Add an AIX Computer
  Change / Show Characteristics of an AIX Computer
  Delete an AIX Computer
  Show Summary of AIX Computers
F1=Help          F2=Refresh          F3=Cancel          F8=Image
F9=Shell         F10=Exit           Enter=Do
```

The following sections provide more detail about each of the choices available from the SS7itty main menu:

Configure Environmental Conditions

The following screen is displayed:

```

                                Change / Show Characteristics of an SS7 Environment
Type or select values in entry fields.
Press Enter AFTER making all desired changes.
* SS7 Variant                                [Entry Fields]
* SS7 Configuration Pack                    [ANSI/T1 GENERIC (2000)]
                                           [Standard]
F1=Help          F2=Refresh          F3=Cancel          F4=List
F5=Reset         F6=Command          F7=Edit           F8=Image
F9=Shell         F10=Exit           Enter=Do
```

The SS7 Environment defines the operational conditions for the SS7 Cluster. There are two configurable options, which can be selected with the F4 key:

SS7 Variant

Defines the SS7 variant of the connecting network

SS7 Configuration Pack

Handles the special conditions that are outside the SS7 variant. From the options that are displayed, unless you have been directed otherwise, select **Standard**

Configure AIX Systems

The Configure AIX Systems submenu is used to define all the machines within the SS7 Cluster. When the **Add** option is selected, the following screen is displayed:

```

                                Add an AIX Computer
Type or select values in entry fields.
Press Enter AFTER making all desired changes.
                                [Entry Fields]
* Computer Name                               []
* First TCPIP Address (15 char LAN name)      []
  Second TCPIP Address (15 char LAN name)     []
F1=Help           F2=Refresh           F3=Cancel           F4=List
F5=Reset          F6=Command           F7=Edit             F8=Image
F9=Shell         F10=Exit              Enter=Do

```

Review the information collected in the table entitled "AIX machines" in Appendix A, "Sample forms for recording information," on page 101, and enter the data as follows:

Computer Name

Name of the machine

First TCPIP Address

The primary IP address

Second TCPIP Address

The secondary IP address (this is optional)

For any additional AIX computer entries, you can either modify the previous entry or start a new one from the submenu.

On completion, review the information that is displayed when you select the **Show Summary** option

Configure SS7 Server

This submenu is used to further define which AIX Computers are to be SS7 Servers.

The following screen appears when you select the Add option:

Add an SS7 Server

Type or select values in entry fields.
Press Enter AFTER making all desired changes.

	[Entry Fields]	
* SS7 Server (AIX Computer name)	<input type="text"/>	+
* SS7 Original Point Code (x-x-x or 0x???)	<input type="text"/>	

F1=Help	F2=Refresh	F3=Cancel	F4=List
F5=Reset	F6=Command	F7=Edit	F8=Image
F9=Shell	F10=Exit	Enter=Do	

After reviewing the information collected in the Table entitled AIX MACHINES in Appendix A, "Sample forms for recording information," on page 101, for each entry of Machine Type SS7 Server name, enter the following data.

- SS7 Server - select the Computer Name from the list that appears when you press F4. If the name is not present, return to the Configure AIX System menu, and review the machine names.
- SS7 Original Point – enter the OPC as defined in the Originating Point Code table.

On completion, review the information that is displayed when you select the **Show Summary** option

Configure WebSphere Voice Response client

This submenu is used to further define which AIX Computers are to be WebSphere Voice Response clients.

The following screen appears when you select the Add operation:

Add a WebSphere Voice Response Client

Type or select values in entry fields.
Press Enter AFTER making all desired changes.

	[Entry Fields]	
* WebSphere Voice Response Client (AIX Computer name)	[]	+
* Telephone Trunks Attach to Computer (1 .. 16)	[]	
* Voice Bearer card type	[DTTA]	+

F1=Help	F2=Refresh	F3=Cancel	F4=List
F5=Reset	F6=Command	F7=Edit	F8=Image
F9=Shell	F10=Exit	Enter=Do	

Review the information that has been collected in the AIX MACHINES table, and for each entry of machine type WebSphere Voice Response Client, enter the following data:

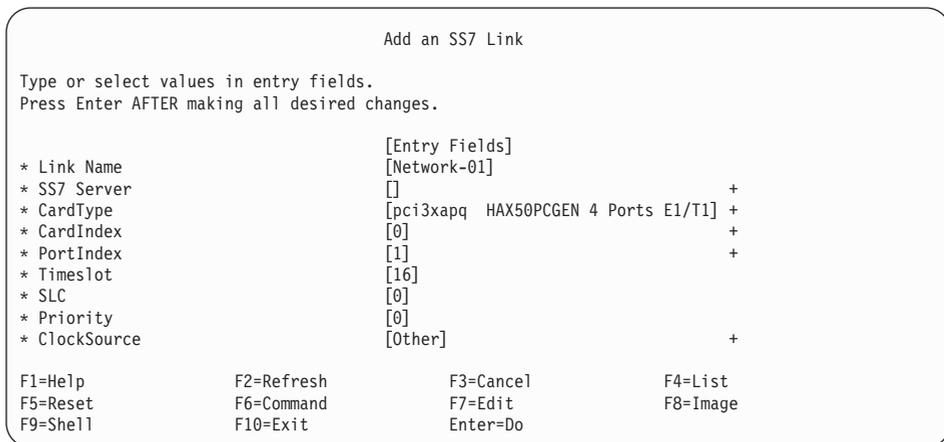
- **WebSphere Voice Response Client.** Select the Computer Name list that is generated by pressing F4. If the name is not present, return to the Configure AIX System sub menu and review the machine names.
- **Telephone Trunks.** This defines the trunks within the WebSphere Voice Response system that are to be allocated to SS7. For example, if all 16 trunks are to be allocated, enter 1-16. For a fragmented selection, for example where trunks 9 through 14 are used for ISDN, enter either 1-8,15,16 or 1-8,15-16.
- **Voice Bearer card type.** Currently there is only one possible option, DTTA.

On completion, review the information that is displayed when you select the **Show Summary** option.

Configure SS7 Links

This submenu defines the Links, or connections to the SS7 network from the SS7 Servers. To assist you, use the information collected in the Signaling links table in Appendix A, "Sample forms for recording information," on page 101.

The following screen appears when you select the Add operation:



Enter the following data for each entry in the Signaling links table:

Link Name

This is the name given to this Link or connection. SS7itty creates a unique name for you, which you can override. Note that to get a unique name created when creating subsequent Links, you must reenter the submenu.

SS7 Server

Use the list generated when you press F4 to select a predefined SS7 Server. If the machine name is not present, return to the Configure SS7 Server menu and review information entered.

CardType

Use the information entered into the table under SS7 Adaptor to make a selective choice.

Card Index

Use the Instance value to define the card index. If only one card is present, set this to zero.

PortIndex

This is the value entered in the table as Port. The valid range of values will be dependent on the CardType that is selected.

Timeslot

This is the value that appears as Timeslot in the table. Valid values are 0-23 for T1 and 1-31 for E1.

SLC

This is the value that is listed under SLC. This will have been defined for the Network operator.

Priority

This value needs to be unique for each connection. Enter a value starting from 0.

Clocksource

The adapter needs to take a clock source from one of the connections. For the first connection on an adapter, select **Line**, and then select **Other** for any subsequent entries that are using the same adapter. In the rare case when the Network is not supplying a stable clock, replace **Line** with **Local**.

On completion, review the information that is displayed when you select the **Show Summary** option.

Configure SS7 Link sets

This submenu defines the grouping of the Links that have the same Destination Point Code. Use the information collected in the Signaling links table from Appendix A to create Link Sets. In most cases there will be only one Link Set.

The following screen appears when you select the Add operation:

Add an SS7 LinkSet

Type or select values in entry fields.
Press Enter AFTER making all desired changes.

* SS7 Link Set Name	Entry Fields]		
* SS7 Link 1	NetworkB]	<input type="checkbox"/>	+
SS7 Link 2		<input type="checkbox"/>	+
SS7 Link 3		<input type="checkbox"/>	+
SS7 Link 4		<input type="checkbox"/>	+
SS7 Link 5		<input type="checkbox"/>	+
SS7 Link 6		<input type="checkbox"/>	+
SS7 Link 7		<input type="checkbox"/>	+
SS7 Link 8		<input type="checkbox"/>	+
* SS7 Destination Point Code[x-x-x or hex]		<input type="checkbox"/>	+

F1=Help	F2=Refresh	F3=Cancel	F4=List
F5=Reset	F6=Command	F7=Edit	F8=Image
F9=Shell	F10=Exit	Enter=Do	

Complete one of these screens for each different DPC entry in the Signaling links table, as follows:

- **Link Set Name.** The name given to this Link Set. SS7itty create a unique name for you, which you can override. Note that when you create subsequent Links, to ensure a unique name you must reenter the submenu.
- **SS7 Link.** Use the list that is generated by pressing F4 to select predefined Links.

- **Destination Point Code.** Enter the value of the DPC from the table.

On completion, review the information that is displayed when you select the **Show Summary** option.

Configure SS7 Route Sets

This submenu defines the routing to the switches via the Link Sets. Use the information collected in the Route Set table from appendix A.

The following screen appears when you select the Add operation:

Add an SS7 Route

Type or select values in entry fields.
Press Enter AFTER making all desired changes.

	[Entry Fields]
* SS7 Route Set Name	[]
* Switch Destination Point Code[x-x-x or hex]	<input type="checkbox"/>	
* SS7 LinkSet 1	<input type="checkbox"/>	+
SS7 LinkSet 2	<input type="checkbox"/>	+
SS7 LinkSet 3	<input type="checkbox"/>	+
SS7 LinkSet 4	<input type="checkbox"/>	+
* SS7 Bearer Trunks associated with this Route Set	<input type="checkbox"/>	

Complete one of these screens for each Route Set in the Route Set table, as follows:

- **Route Set Name.** The name given to this Route Set. SS7itty creates a unique name for this, which you can override. Typically, this would be the name of the switch. Note that when you create subsequent Route Sets, you must reenter the submenu to ensure that you assign a unique Route Set Name.
- **Switch Destination Point Code.** Enter the value of the DPC from the table.
- **LinkSets.** To select a predefined Link Set, use the list that is generated by pressing F4.
- **Bearer Trunks.** This is a list of trunks that are being supplied by the Network under this DPC.

On completion, select the **Show Summary** option, and review the information that is displayed.

Configure SS7 trunks to WebSphere Voice Response trunks

This submenu defines the Voice Bearer interconnections between the network and the WebSphere Voice Response systems. Use the information collected in the Trunk table to create entries.

The following screen appears when you select the Add operation:

Add an SS7 Trunk

Type or select values in entry fields.
Press Enter AFTER making all desired changes.

	[Entry Fields]	
* SS7 Trunk (Route Set name)	□	+
* SS7_Trunks	□	
* WebSphere Voice Response Client	□	+
* WebSphere Voice Response Trunks	□	

F1=Help	F2=Refresh	F3=Cancel	F4=List
F5=Reset	F6=Command	F7=Edit	F8=Image
F9=Shell	F10=Exit	Enter=Do	

Complete one of these screens for each entry in the TRUNK table, as follows:

- **SS7 Trunk (Route Set name)**. Use the F4=List option to select a predefined SS7 Trunk or Route Set name.
- **SS7 Trunks** Select a range of network trunks to be used in this entry of the interconnect.
- **WebSphere Voice Response Client** . Use the F4=List option to select a predefined WebSphere Voice Response client.
- **WebSphere Voice Response Trunks** . Select a range of WebSphere Voice Response trunks to be used in this entry of the interconnect.

On completion, review the information that is displayed when you select the **Show Summary** option.

Generate SS7 Configuration Files

When all the necessary information has been entered into SS7itty, you can generate the configuration. To perform this task, select the Generate SS7 Configuration Files option from the top level menu.

For a successful generation, the screen that is displayed should look similar to the one below:

```

                                COMMAND STATUS
Command: OK                      stdout: yes          stderr: no

Before command completion, additional instructions may appear below.

Stage 1: Load environment information.
Stage 2: Check configuration association.
Stage 3: Check Trunk usage.
Stage 4: Check SS7 adapter card usage.

***** Checking data has been successful.

Stage 5: Creating WebSphere Voice Response Client files.
Stage 6: Creating Distributed7 files.
Stage 6a: Machine Mercury
Stage 6b: Machine Venue
Stage 6c: Machine Mars

F1=Help          F2=Refresh      F3=Cancel      F6=Command
F8=Image        F9=Shell       F10=Exit      /=Find
n=Find Next

```

The first four stages of the process check the form of the entered data. The last two stages perform the actual generation of the configuration files. If problems are found during the checking stage, a screen is displayed similar to the one below:

```

                                COMMAND STATUS
Command: failed                 stdout: yes          stderr: no
Before command completion, additional instructions may appear below.
Stage 1: Load environment information.
Stage 2: Check configuration association.
Stage 3: Check Trunk usage.
----- SS7 Trunks to WebSphere Voice Response Trunks/SS7 Trunks -----
Error: Trunk 17 is present in the following RouteSets/Clients
      RouteSet/Client: NetworkA/Venus
      RouteSet/Client: NetworkA/Mars
Remedy: Return to the "SS7 Trunks to WebSphere Voice Response Trunks" change screen, select one of
        The of two RouteSets/Clients and review the SS7 Trunks selected.
-----
The configuration generation has been aborted.
F1=Help          F2=Refresh      F3=Cancel      F6=Command
F8=Image        F9=Shell       F10=Exit      /=Find
n=Find Next

```

In this example, Network Trunk 17 has been defined for connection with two WebSphere Voice Response systems. Generation has terminated at stage 3, reporting this problem, together with directions about how to possibly rectify the situation.

Importing or exporting an SS7 configuration

The Export option on the top level menu allows the current information to be written to a file.

These files are stored in `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/archive` and have a file extension of `.arc`.

Information can similarly be imported from previously exported files using the Import option and the F4=List option.

Pack Configuration

For detailed about information on pack configuration refer to the *WebSphere Voice Response for AIX: Configuring the System* book.

1. Login to WebSphere Voice Response.
2. From the WebSphere Voice Response Welcome window, select **Configuration** → **Pack Configuration** → **Change**.
3. Make the following changes to each of the packs indicated:
 - a. Set Trunk Parameters to CCS-SP.
 - b. Set Line Signaling to SS7-ISUP. For optimum SS7 performance, put all the channels on each pack in a single group.
 - c. Check the **Save** check-box for each pack that has been modified.
 - d. Click **File** → **Save**.
 - e. Confirm the save.
 - f. Click **File** → **Close** to return to the Welcome window.

Note: You need to restart WebSphere Voice Response for the new pack configuration to take effect.

Chapter 5. Managing and Monitoring SS7 Support for WebSphere Voice Response

This part of the book describes how to manage Distributed7 after you have installed it, and how to monitor activity.

SS7_MAINT – the general maintenance utility

The SS7_MAINT is a general purpose facility to run other commands that are associated with SS7 Support for WebSphere Voice Response. This command has been designed to simplify the SS7 maintenance by having the most frequently-used commands accessible from a central point. The basic operation of the menu structure is a single letter selection from a menu choice displayed on the screen.

The following options are available on all levels:

- R** Return to previous level
- Q** Quit the SS7_MAINT utility (this is not published on all menus but is always available for use)
- ?** Redisplay the menu option from the current level

<Enter>

Redisplay the current prompt. Some menu levels redisplay an active status, so using <Enter> without selecting an option can be useful in monitoring a changing status.

The top-level menu for SS7_MAINT is shown below:

```
WebSphere Voice Response
  Signaling System #7
  Maintenance Program
  Top Level Menu
-----
E - Enablement Operation (Start/Stop).
D - D7 Operation (Start/Stop)
C - MML configuration loader
H - Housekeeping
M - D7 Maintenance programs
S - SS7 Server/WVR Client
I - iFix Loading
F - Fault Finding
V - D7 software release
? - Redisplay these options.
Q - Quit this program
Select an option :
```

The functionality of each of the options is described in the following sections:

Enablement Operation

This menu handles the starting and stopping of the `SS7_D7` signaling custom server. These same operations can be performed from the WebSphere Voice Response Custom Server Manager GUI. This menu has an active prompt that indicates the current execution state of the `SS7_D7` custom server.

The menu has two options to start and to stop the custom server. When one of these options is selected, the prompt returns immediately without confirming the action. However, the status can be monitored by periodically pressing Enter to update the status.

D7 Operation

This menu handles the starting and stopping of the Distributed7 package, and has an active prompt indicating the current state of the package. The menu options are:

- S** Start the D7 package in silence mode, without any alarm message being copied to screen. This is the default method of starting D7 unless you have just completed installation or are fault-finding.
- M** Start the D7 package with alarm message being copied to screen. This method is useful when first installing D7 or fault-finding.
- T** Terminate the D7 package. This initiates the sequence to close down the D7 package.
- E** This enables automatic start of the D7 package following a system startup.
- D** This disables automatic start of the D7 package following a system startup. The D7 package must be started by hand (using `SS7_MAINT`).

Note: The options S, M, and T initiate the command sequence, and the return of the prompt does not imply the sequence has completed. The current state of the D7 package can be monitored by periodically pressing Enter to update the active prompt.

MML Configuration Loader

This handles the options around MML (the D7 configuration database), and should be used when initially configuring the SS7 cluster or updating. Possible options are:

- H** Install the Host MML configuration for this machine from the `mml-host` file generated by `SS7itty`.

- S** Load the SS7 configuration for the whole SS7 Cluster from the mml-ss7 file generated by SS7itty.
 - F** Remove the MML database from the current machine, ready to receive a new copy of the Host MML and SS7 MML.
- Note:**
1. The option cannot be performed when the D7 package is running.
 2. The root password will be requested in order to gain access to the root-owned database files.
- B** Backup the currentMML configuration by creating the file /tmp/D7-dbBackup.
 - Z** Restore a previously backed-up MML configuration. It overwrites the current configuration in the MML database with the data stored in file /tmp/D7-dbBackup.

HouseKeeping

This menu option allows the monitoring and control of the Log/Event and Trace files generated by the SS7/D7 Enablement custom server. On initial entry to this submenu the current Log/Error and Trace files are listed. The options are as follows:

- A** Delete all the listed files.
- E** Deletes only the error files listed.
- V** Delete only the event files listed.
- T** Deletes the SS7 trace file.
- D** Rescreen the file information list on entry.

D7 Maintenance programs

This submenu gives access to a number of system health-monitoring programs that are used to verify the D7 package. Possible options are:

- C** Show the SS7 card that is present in the current machine. It runs the D7 command getcfg and shows its output. getcfg shows the state and presence of the SS7 adapters in the current machine.
- P** Show the current D7 processes that are running. It runs the D7 command apm_ps, which identifies which D7 processes are running on the current machine.
- O** Show the status of the OSI components. It initiates the D7 command AccessMonitor, which shows all the currently active machines within

the SS7 cluster, and what options are currently running. Note AccessMonitor is a GUI, for further information refer to “SS8 commands” on page 64.

- L** Show the status of the SS7 Links by initiating the D7 command AccessStatus, which shows the status of the Link in the current SS7 cluster, and the currently message capacity. For more information on this command refer to “SS8 commands” on page 64.
- D** Show the D7 interconnect status by running the D7 command ebs_showlink. The output shows the connected machine on the SS7 cluster and the available SS7 adapters. For more information on this command refer to “SS8 commands” on page 64.
- M** Run the ss7view –circuit command, showing the current status of the associated circuits from this machine. Note that this option is available only on WebSphere Voice Response client machines. For more information on this command refer to “ss7view – D7 Enablement monitoring tool” on page 58.

SS7 Server/WebSphere Voice Response client

This menu handles the options for configuring a machine for either server or client operation. Possible options are as follows:

- S** Configure the current machine for SS7 Server operation
- W** Configure the current machine for WebSphere Voice Response client operation. Also use this option for configuring this package on a SSI machine.
- I** Configure a WebSphere Voice Response client connected to a single system image.
- C** Configure a combination of a WebSphere Voice Response client and an SS7 Server in the same machine.

iFix Loading

This option is used to install a temporary software fix for the D7 package. Should a software correction be issued, copy the file to /tmp/iFix and select this option. The iFix file is in TAR format and contains a script to run the necessary file-copying of the supplied corrections.

Fault Finding

This menu is designed to assist when using Chapter 7, “Problem determination,” on page 79 from this book. During the identification of the

root cause of problem, you may be asked to run certain options and act on the output. Options for this menu are as follows:

- 1 D7 Network Check. Gives the output from the D7 command `ebs_showlink`
- 2 D7 Process Check. Gives the output from the D7 command `ebs_ps`
- 3 D7 Card Check. Gives a status report of the SS7 adapters.

ss7problem

the options S, L, and D each run the `ss7problem` command on level 1, 2 and 3 respectively. Each of these options generates data that can be sent to IBM to assist in problem determination.

D7 software release

This menu provides control over which release Distributed 7 package is to be run. On selection of this option a screen similar to below will be shown:

```
-----  
                                WebSphere Voice Response  
                                Signaling System #7  
                                Maintenance Program  
                                D7 Release Menu  
-----  
Current active release: 1.3.1.7  
Available releases:    1.3.1.0  
                      1.3.1.7  
    C - Change active release  
    ? - Redisplay these options.  
    R - Return to top level menu  
Select an option :
```

The currently-selected release is shown, followed by the available releases. The readme in `/usr/lpp/dirTalk/readme/SS7.README` defines the release that you should select. To change the selected release, use option C, and then select the required release from the subsequent menu. As this changes elements within the Distributed7 package, you must enter the root password before you can proceed.

Note:

1. Before changing a release, ensure that Distributed7 is not running; if it is, it will be terminated.
2. A question regarding the copy MML database might appear at this stage; if so, you should normally respond with "yes", as this is a new or upgrade installation.

ss7view – D7 Enablement monitoring tool

The `ss7view` command is a multi purpose tool for monitoring the operation of the SS7_D7 Custom Server. It has the following functions:

1. Monitoring the service state and call connection states of all circuits
2. Monitoring the internal event log
3. Monitoring the SS7 and WebSphere Voice Response message flows
4. Control event capture through error message triggers
5. Monitoring call statistical information
6. Translating release cause code
7. Providing information about the signaling state table

The following sections describe how to use the `ss7view` command for each of the functions in the above list.

ss7view – help options

If you run `ss7view` without any argument, or with the `-help` argument, the following screen is displayed, listing all the possible operations and how to view help information for each of them :

```
Prompt> ss7view -help
ss7view :: SS7_D7 monitoring tools
-----
For help on ....
SS7-D7 Event ..... Type ss7view -help event
SS7 ISUP Messages ..... Type ss7view -help message
Monitors circuits ..... Type ss7view -help circuit
SS7 Stats ..... Type ss7view -help stats
Event triggering ..... Type ss7view -help trigger
Trace level monitoring ... Type ss7view -help trace
Cause Code explanation ... Type ss7view -help cause
Information ..... Type ss7view -help info
```

ss7view – monitor circuits

This monitor circuit option shows the current service state and call state of all circuits on the current WebSphere Voice Response client. To use it, enter `ss7view -circuit [<trunks>]` The `[<trunks>]` is optional, and defines how many trunks to display at one time. The default is 16. A typical screen might look like this:

```

SS7 Circuit Status on b80h []
  0  2  4  6  8 10 12 14 16 18 20
-----
 1 *
 2
 3
 4
 5
 6
 7
 8
 9
10
11
12
13
14
15
16
-----
Selected Circuit ... 0
Call state           = answ
Near End Service state = Ubk
Far End Service state = Ubk
Press Q to quit

```

The central area of the screen shows a real-time updating interpretation of the current state of each circuit. This is ordered in rows of trunks and columns of circuits within trunks. Note that the number of the column is 0-23 for T1 and 0-31 for E1. The E1 display will show the letters F and S in the zero and sixteenth column. This is to indicate the frame and signaling timeslot.

Within the table a single character is used to denote the current state. For an SS7 cluster with more trunks than that are being displayed, others can be shown by either using the up and down cursors keys or the Page Up and Page Down keys to scroll through the display.

Although all the circuits of the SS7 cluster are displayed, only those pertaining to this client will show their call state. Trunks for other WebSphere Voice Response clients show only their redundancy state relative to this client.

The lower part of the screen shows details of an individual circuit. The CIC value of the circuit being viewed is shown in the first line. This can be selected using the cursor keys to move the reversed video character to the desired circuit. The other three lines show the internal states of Call, Near End Service and Far End Service. Refer to “ss7view –info ” on page 64 for a legend to interpret the characters and phrases that are displayed on this screen.

ss7view –event

The SS7_D7 custom server is based on a state table design (this is different to a WebSphere Voice Response state table). Each time a message is received, the

SS7_D7 state table engine is run to generate the required reaction and also log events. These events are logged in shared memory which can in turn be read by `ss7view`.

The simplest form of `ss7view -event` shows all logged events since `SS7_D7` started running, or when it wrapped. Other options are as follows:

- t** Tail event buffer. Once the current events have been displayed, `ss7view` continues to monitor the event buffer and display new events as they are detected. Use `Ctrl + C` to exit from this mode
- c <cic> or -c <cic>-<cic>**
Filters the event display so that only events relating to the `<cic>` or the `<cic>-<cic>` range are displayed.
- s** Use with the `-c` option to re-enable displaying of non `cic`-based events.
- R <filename>**
Copies the event buffer in raw mode into a named file for later processing. This option is used in the `ss7view -trigger` option (see “`ss7view -trigger`” on page 61).
- f <filename> or -f <filename>,<state table>**
instead of using the current event buffer, this option processes data stored in `<filename>` that was previously created with the `-R` option. The additional `<state table>` is for machines that are not currently using the same state table as the originating machine.

ss7view -message

The `ss7view -message` option uses the event information to interpret the message flows through `SS7_D7` custom server, WebSphere Voice Response and `SS7` Stack. Below is an example output from `ss7view` for a single inbound call on `CIC 0`.

```

13:29:38.264:0000: IAM/ISUP_SETUP[1234] ----->>
13:29:38.265:0000:                               SL_CALL_SETUP_IND ----->>
13:29:38.265:0000: <<----- ACM/ISUP_OGC_SELECTED
13:29:38.265:0000: <<----- ACM/ISUP_ALERT
13:29:38.268:0000:                               <<----- SL_CALL_ANSWER_REQ
13:29:38.268:0000:                               SL_CALL_ANSWER_CNF [SL_REPLY_SUCCESS] ----->>
13:29:38.269:0000: <<----- ANM/ISUP_SETUP
.....
13:29:41.846:0000:                               <<----- SL_CALL_TERMINATE_REQ
13:29:41.846:0000: <<----- REL/ISUP_RELEASE[Cause:CSV_NORMALCALLCLR]
13:29:41.861:0000: RLC/ISUP_RELEASE ----->>
13:29:41.862:0000:                               SL_CALL_TERMINATE_CNF [SL_REPLY_SUCCESS] ----->>

```

The information show is:

hh:mm:ss:mmm

The time stamp of the event down to milliseconds

cccc CIC number of circuits

Message flows

From left to right, these are:

- Network/SS7 Stack
- SS7_D7 custom server
- WebSphere Voice Response

The arrows show the direction that the message travels. In the above example, the first line shows IAM message being sent from the SS7 Stack to the SS7_D7. The next shows that SS7_D7 has converted this message into a SL_CALL_SETUP_ID message, which is then passed onto WebSphere Voice Response, and so on.

The following arguments are available:

-t After displaying the current message, monitor the event buffer for new messages, and display when detected. Use Ctrl + C to exit from this mode.

-<cic> filters events to show only messages for the given CIC.

ss7view -trigger

This option allows `ss7view` to define to SS7_D7 which errors, when detected, should cause the execution of a predefined command. This is generally used to capture event-buffer information prior to an error being detected. The collected information can then be subsequently examined to discover the reason for the error.

The command that is run is defined in the `Logevent.cfg` (see “Logging and event file” on page 109 for further information). The `ss7view -trigger` option can define and redefine which errors trigger a dump of the event buffer. The format of the command is `ss7view -trigger <trigger options>`, with possible options as follows:

d<event>

Deletes event <event>

i<event>=<times>

Inserts event <event> to trigger a maximum of <times> times.

r<event>

Refreshes event <event>. The maximum <times> is reset.

u<event>=<times>

Updates event <event> as that the maximum time is <times>

s Shows the setting of all current active triggers.

For example `ss7view -trigger i30000=2d30002s` performs the following:

Insert event 30000 to trigger twice

Delete event 30002

Show the resulting setting

Note: only errors within the range 30000 and 31999 that are generated by SS7_D7 can be used as triggers.

ss7view -trace

The `ss7view -trace` option controls a secondary tracing facility that involves writing text messages to a file. This form of tracing can be used to identify specific problem areas in very low call volumes and is not recommended for general use. An example of such a use would be to identify message and parameter problems in SS7 when used on a new or different variant of an SS7 network. The command takes the format `ss7view -trace <option>`, where the options are list below. Each option is used with either a + or - preceding them. The + and - mean enable and disable the mode respectively.

- A** Action class
- i** Information class
- z** Maintenance
- S** Subsystem (AIX operations) class
- m** WebSphere Voice Response and ISUP messages
- t** Dump of table information
- c** clear – disable all modes previous enabled
- d** display the currently enabled modes

For example, `ss7view -trace +iA-td :`

Enables information

Enables actions

Disables tables

Displays current settings.

Trace information is appended to `/usr/lpp/dirTalk/db/current_dir/oam_log/SS7/SS7-Trace`.

ss7view –stat

The SS7_D7 custom server can collect some basic statistical information about call rates and process waiting times. The enablement of this feature and its sampling rates are defined in the LogEvent.cfg file (see “Logging and event file” on page 109 for more information). To run the command enter `ss7view –stat`. This command continuously updates the screen with new information as each time period is complete. To exit this program use Control + C.

The columns are defined as follows:-

Time Range

The timestamp for the end of the period just sampled.

Busy Ratio

A rough calculation of the ratio of time spent waiting for messages against the time processing messages. A typical value of 14% indicates that in one second, approximately 1/7 of that second was spent processing messages. This is a useful indicator to show when the SS7_D7 custom server is not getting sufficient processing time.

Network IAM

A count of inbound call setup messages.

Network REL

A count of release messages from the network.

Network ABRT

A count of inbound calls that have been aborted (calls that did not reach the answer state).

WebSphere Voice Response IAM

A count of outbound call setup messages.

WebSphere Voice Response REL

A count of release messages generated by WebSphere Voice Response.

WebSphere Voice Response ABRT

A count of outbound calls that have been aborted.

BHCA A rough projection of the Busy Hour call attempts. It is a calculation of the number of call setups over the current sample period and the previous five sample periods as if consecutive over an hour.

BHCC A rough projection of the Busy Hour call completion. This is calculated in the same way as BHCA, but is based on the release messages.

ss7view -info

Information that relates to the Signaling State Table can be displayed through the `ss7view -info` option.

Enter `ss7view -info -p` to display the current protocol and variant information of the loaded Signaling State Table.

Enter `ss7view -info -c` to display a legend of the possible circuit states as shown in `ss7view -circuit`, with brief and full descriptions.

ss7view -cause <name/value>

This option interprets a Release cause code into a full description. The option takes a single argument of either a numerical value or a mnemonic label. For example `ss7view -cause 16` or `ss7view -cause CSV_NORMALCALLCLR`.

All mnemonics start with the letters CSV and may appear in other `ss7view` options.

SS8 commands

This section describes the following SS8 commands:

- `AccessMonitor`
- `AccessStatus`
- `ebs_showlink`

AccessMonitor

`AccessMonitor` is a D7 GUI utility that monitors the operational states of the various components within the D7 SS7 cluster. It can be run either from `SS7_MAINT` (option M, then O) or from the command line by entering `AccessMonitor 0 &`.

The example screen below shows a three machine configuration. `b80c` and `b80e` are both WebSphere Voice Response clients and have the D7 base processes running – denoted in green. Machine `h80a` has been configured for SS7 Server operation and has the MTP-L3 and ISUP layer processes running. The SS7 Server contains a configured SS7 adapter with MTP-L2 running.

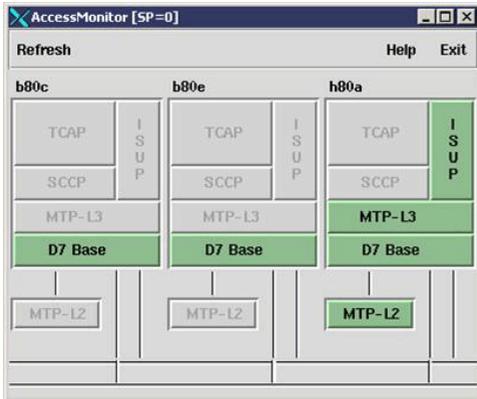


Figure 7. The AccessMonitor screen

If a process stops running or has problems, the corresponding panel on the screen changes to red. If a machine stops running D7, all of its boxes change to grey.

Only one copy of the command can be run per machine. If a second is started, it silently terminates. More information can be found about this by referring to the Distributed7 User manual included on the Distributed7 disk.

AccessStatus

AccessStatus is a D7 GUI utility to monitor the operational states of the various Links within the D7 SS7 cluster. It can be run either from SS7_MAINT (option M then L) or from the command line by entering `AccessStatus 0 & .`

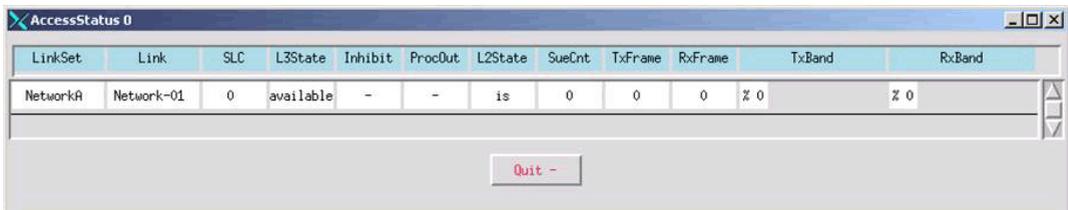


Figure 8. The AccessStatus screen

The example display above shows a cluster with a single Link called Network-01, with an SLC of 0, belonging to the LinkSet NetworkA. The L2 and L3 states shown are "is" (in service) and available, indicating a functioning link. The TxFrame/TxBand and RxFrame/RxBand indicate that there is no SS7 traffic at this time. The TxBand and RxBand give a graphical representation of the current loading.

Only one copy of the command can be run on each machine. If a second is started it will silently terminate. More information can be found about this by referring to the Distributed7 User manual that is included on the Distributed7 disc.

ebs_showlink

ebs_showlink is a D7 text-based utility that shows the interconnection status between the various machines in the SS7 cluster.

LINK	TYPE	HOST	RMTHOST	STAT	TIME	HBEAT
8	tcp/ip	b80c	h80a	L A	Tue May 25 11:04:31	ok
9	tcp/ip	b80c	b80e	L A	Tue May 25 09:23:32	ok
0	pci3xapq	h80a	-	L A	Tue May 25 11:04:26	-

The text above shows a three machine SS7 cluster when viewed from machine b80c. The first two lines denote the LAN connections to machines h80a and b80e respectively. The final line denotes the presence of an SS7 adapter on the machine h80a (SS7 Server). If U is present under the STAT column, this indicates that the machine was present but has been disconnected.

More information can be found about this utility by referring to the Distributed7 User manual included on the Distributed7 disc.

D7WVRErrorReport

D7WVRErrorReport is a second custom server of the D7 Enablement package. Its sole purpose is to translate Alarm messages from the D7 package into WebSphere Voice Response Alarm messages.

Translated messages can be collected from the local D7 package, or can be received globally from any D7 package on the SS7 cluster. A number of D7 alarm messages have their own translation, with parameters translated into more identifiable values (for example machine names) and descriptions to highlight the condition and potential remedies.

Startup parameters

The startup parameters listed below can be applied to the custom server from the top level WebSphere Voice Response GUI, after making the following selections: **Application** —> **Application** —> **SS7_D7** —> **Custom servers** —> **D7WVRErrorReport** —> **File** —> **Properties**.

Remember to put only one parameter on each line.

Global Reporting

Use the `-g` parameter (together with `-r`) to register this D7 host as a global collector of D7 alarms from other D7 hosts (SS7 Servers). See “D7 Alarm Logging” on page 38.

Repetition Suppression

Use the `-R <number>` facility to reduce the number of repetitions of the same Alarm message being passed to WebSphere Voice Response. For example by entering `-R 3` following the first occurrence of an alarm causes the subsequent three identical alarms to be suppressed. Use the `-r` parameter to disable this feature.

Reporting Level

Use the `-d <1...3>` parameter to define the reporting level. For an Alarm message to be passed to WebSphere Voice Response it must have a reporting level of greater than or equal to the parameter supplied. For example `-d 1` is the default and passes all alarm messages.

Compile Translation File

Use `-C` to create a binary translation file based on the D7 package error files. This option is used once on installation, to create a translation file from D7 to WebSphere Voice Response, and is not required during normal operation.

Configuration Files

D7WVRErrorReport has two configuration files:

D7_ErrorList

The binary translation file created by the Compile Translation file option above.

AlarmTranslation.cfg

This contains the mapping of D7 error codes to WebSphere Voice Response alarm codes and associated reporting level. For more information on this configuration refer to Appendix C, “Configuration files,” on page 107.

Collecting event and trace information

Event information

As the SS7_D7 custom server processes messages between the WebSphere Voice Response system and the SS7 stack, events are logged into shared memory. This is a continuous activity and minimises the effect on the performance of the function. This information can be read to screen and recorded to files using the `ss7view` (see “`ss7view -event`” on page 59 for details).

The events can be interpreted into message flows between the various components with the `ss7view -message` option (see “`ss7view -message`” on page 60 for details). The `ss7view -event` also has an option `-R` for copying the shared memory direct to disk for offline processing. The primary purpose of this event facility is to capture events prior to an error being report.

Reviewing the event enables diagnostics to be performed. The `ss7view -trigger` (see “`ss7view -trigger`” on page 61) enables automatic capture of event information direct to disk when selected errors message are detected. The default operation of this function is defined in the logging and event file configuration (see “Logging and event file” on page 109 for details).

Trace information

Trace information is a method of writing diagnostic information direct to disk file. By default this function is disabled as it affects the performance of the custom server when it is under load. The purpose of this trace facility is to capture detailed information about messages and their parameters. This is useful when first connecting to a new network or variant.

The `ss7view -trace` option controls whether this facility is enabled and what elements are traced (see “`ss7view -trace` ” on page 62 for details) .

Chapter 6. Information elements passed to WebSphere Voice Response

This section describes the WebSphere Voice Response tags that are supported in the SS7 product.

Tag presentation and omissions

For all mandatory SS7 Information Elements that do not have tags supplied, or part-completed tags, these fields are filled by the system defaults. Refer to Appendix C, "Configuration files," on page 107 for more information. The following table shows a list of valid tags, describes the relationship between call activities, and indicates when tags can be used. If any unsupported tags are used, they are silently ignored.

Tag Name	Inbound call setup (SV542)	Outbound call setup (SV541)	Inbound call termination	Outbound call termination
USR2USR	✓	✓	✓	✓
CLDN	✓	✓	×	×
CLGP	✓	✓	×	×
ORIGIN	✓	✓	×	×
REDIRN	✓	✓	×	×
REDINFO	✓	✓	×	×
CHARGEN	✓	✓	×	×
UTI	✓	✓	×	×
USI	✓	✓	×	×
CAUSE	×	×	✓	✓
NEWCLDN	×	×	×	✓
PROTOCOL	✓	×	✓	×
GENERICADDR	✓	✓	✓	✓
ISUPPARM	✓	✓	✓	✓
SERVICEACTIVATION	×	×	✓	✓
JINFO	✓	✓	×	×
PINFO	✓	✓	×	×

Country and network variants

The values specified in tags, and their attributes below, are a small subset of the potential available values. Depending on the country or network variant these values may change or different ones be preferred. In general this package passes the values unchanged to and from the network, so if any special values must be passed, this is possible without resorting to software changes.

When changing values, both in tags and configuration files, it is important to understand the effect and consequences of the change. For example, it may not always be the best choice to turn off the Charging flag when answering all calls

Tags types and descriptions

This section provides details of each tag and its attributes.

User-to-user information [USR2USR]

This is a sequence of free-format numbers that are not defined within the SS7 ITU/ANSI system. The maximum number in the sequence is 131, with each number having the range of one byte unassigned (0 – 255). The number can be specified either as decimal or hexadecimal—the latter must be preceded by 0x. Each number is separated by a comma, for example: 1, 3, 5, 7, 0x2, 0x4, 0x6 .

Called Number [CLDN]

This tag specifies the called number and associated attributes. The maximum number of digits in the number is 24 (ITU/ANSI), and only digits from 0 – 9 are supported. To use this tag, leave the Phone Number and Format fields blank in the MakeCall state table Action. The optional attributes are:

CLDN.NUMBER_TYPE

A numerical value defining the Nature of Address of the call, with the following possible values:

- 1 Subscriber
- 3 National
- 4 International
- 5 Network Specific

CLDN.NUMBER_PLAN

a numerical value defining the Numbering Plan:

- 1 ISDN

Calling Number [CLGP]

This tag specifies the calling number. The maximum number of digits in the number is 24 (ITU/ANSI), and only digits from 0 – 9 are supported. The optional attributes are:

CLGP.NUMBER_TYPE

A numerical value defining the Nature of Address of the call, with the following possible values:

- 1 Subscriber
- 3 National
- 4 International
- 5 Network Specific

CLGP.NUMBER_PLAN

a numerical value defining the Numbering Plan:

- 1 ISDN

CLGP.SCREEN

Defines the screening indicator for the calling number, with the following possible values:

- 0 User provided, not verified
- 1 User provided, verified and passed
- 2 User provided, verified and failed
- 3 Network provided

CLGP.PRESENT

Defines the address presentation indicator, with possible values as follows:

- 0 Presentation allowed
- 1 Presentation restricted
- 2 Address not available

Original Number [ORIGIN]

This tag specifies the original number. It has the same specification and optional attributes as Called Number, but has a tag name of ORIGIN.

Redirecting Number [REDIRN]

This tag specifies the redirecting number. It has the same specification and optional attributes as Calling Number, but has a tag name of REDIRN.

Redirecting Information [REDINFO]

This tag specifies the Redirecting Information associated with a call. There is no tag information to be processed but the following attributes are processed:

REDINFO.IND

The redirecting method, with the following possible values:

- 0 No redirection
- 1 Call rerouted
- 2 Call rerouted, all redirection information presentation restricted
- 3 Call diverted
- 4 Call diverted, all redirection information presentation restricted
- 5 Call rerouted, redirection number presentation restricted
- 6 Call diversion, redirection number presentation restricted

REDINFO.ORIG

Indicates the original diverting reason, with the following possible values:

- 0 Unknown or not available
- 1 User busy
- 2 No reply
- 3 Unconditional

REDINFO.COUNT

Indicates the number of redirections the call has undergone. The value has a range of 1 – 5.

REDINFO.REASON

Reason for redirecting, with the following possible values:

- 0 Unknown or not available
- 1 User busy
- 2 No reply
- 3 Unconditional
- 4 Deflection during alerting
- 5 Deflection immediate response
- 6 Mobile subscriber not reachable.

User Teleservices Information [UTI] (ITU Only)

This tag and its attributes define the User Teleservices Information IE. There are three elements in the IE. The first will be present in the tag UTI and the other two in attributes UTI.HLC_1 and UTI.HLC_2 respectively. These tags are passed to WebSphere Voice Response as decimal, however an application can present these numbers as decimal or hexadecimal (a two digit number preceded by 0x).

User Services Information [USI]

This tag and its attributes define the User Services Information IE. There is no value to the tag, but up to five attributes can be defined. Depending on the attribute, one or a series of numbers can be specified. The number in this tag is passed as decimal; however, the application can present these numbers as decimal or hexadecimal. For series of numbers, each number is separated by a comma. The attribute labels are as follows:

USI.OCTET_1

one value

USI.OCTET_2

up to three values

USI.OCTET_3

up to four values

USI.OCTET_4

one value

USI.OCTET_5

one value

Generic Address [GENERICADDR]

This tag specifies the Generic Address number and associated attributes. The optional attributes are:

GENERICADDR.ADDR_TYPE

Various values, refer to relevant documentation for ANSI – T1.113 3.20A or ITU – Q.763 326

GENERICADDR.NUMBER_TYPE

A numerical value defining the Nature of Address of the call, with the following possible values:

- 1 Subscriber
- 3 National
- 4 International

5 Network Specific

GENERICADDR.NUMBER_PLAN

a numerical value defining the Numbering Plan:

1 ISDN

GENERICADDR.SCREEN

this attribute defines the Screening indicator for the calling number:

0 User provided, not verified

1 User provided, verified and passed

2 User provided, verified and failed

3 Network provided.

GENERICADDR.PRESENT

This attribute defines the address presentation indicator:

0 Presentation allowed

1 Presentation restricted

2 Address not available

Service Activation Parameter [SERVICEACTIVATION]

This tag is available only during application call termination, and contains the optional IE that is used to pass a switch-specific request, for example, transfer this call on termination. The tag value is a numeric list (up to a maximum of 128 comma separated decimal or hexadecimal numbers) the decode of which is specific to the local switch implementation of supplementary services (if any). There are no attributes defined for this tag.

Jurisdiction Information Parameter [JINFO]

For an SS7 T1 inbound call application, the Jurisdiction Information is presented as the value of Tag JINFO in system variable SV542.

For an outbound SS7 T1 call application, the Jurisdiction Information is set in SV541 through the value of Tag JINFO.

The JINFO tag has a value of *xxxxxx* (digit string of up to six unpacked IA5 address digits) and has no tag attributes. The assumed format of this optional IAM parameter received from the SS7 network is:

```
byte  nameOfParam = 0xC4;    /* JURISDICTION_INFO_PARM */
byte  lenOfParam;    /* (in bytes) */
byte  DigitInfo[3];    /* Packed 2 digits per byte */
```

The DigitInfo array (packed) is mapped to the JINFO Tag value (unpacked).

Party Information Parameter [PINFO]

For an SS7 T1 inbound call application, the Party Information is presented as the two attributes of Tag PINFO (PINFO.CALLING_NAME and PINFO.REDIRECTING_NAME) in system variable SV542. The value of Tag PINFO is set to zero.

For an outbound SS7 T1 call application, the Party Information is set in SV541 through PINFO Tag string attributes PINFO.CALLING_NAME and PINFO.REDIRECTING_NAME.

The PINFO Tag value is unused but should be set by the application to zero. The format of this optional IAM parameter sent to the SS7 network is:

```
byte nameOfParam = 0xFC;
    /* PTY_INFO_PARM */
byte lenOfParam;
    /* Overall length of parameter (in bytes) */
byte NameInfoType = 0xFE;
    /* Name Information (sub-parameter code) */
byte NameInfoLength;
    /* Length of sub-parameter (in bytes) */
byte CallingNameType = 0x01;
    /* Calling Party Name element indicator */
byte CallingNameLength;
    /* Length of Calling string (in bytes) */
byte CallingName[15];
    /* IA5 character string 1 - 15 bytes */
byte RedirectingNameType = 0x03;
    /* Redirecting Party Name element indicator */
byte RedirectingNameLength;
    /* Length of Redirecting string (in bytes) */
byte RedirectingName[15];
    /* IA5 character string 1 - 15 bytes */
```

The character arrays CallingName and RedirectingName are mapped to the PINFO Tag attributes PINFO.CALLING_NAME and PINFO.REDIRECTING_NAME respectively.

Release Cause Codes [CAUSE]

This tag is available only during call termination, and contains the IE that defines the reason for termination. The value of the tag CAUSE defines the termination reason as a number between 1 and 127. Refer to “ss7view – D7 Enablement monitoring tool” on page 58 to interpret these values. The attributes are:

CAUSE.LOCATION

This attribute defines the location field:

0 User

- 1 Private network serving the local user
- 2 Public network serving the local user
- 3 Transit network
- 4 Public network serving the remote user
- 5 Private network serving the remote user
- 7 International network
- 10 Network beyond internet working point

CAUSE.CODING_STD

This attribute defines the coding standard field, and reflects the meaning of the Cause field:

- 0 ITU-T standardized coding
- 1 ISO/IEC standard
- 2 National standard
- 3 Standard specific to identified location

CAUSE.DIAGS

A series of numbers, separated by commas, in the range of 0 – 255, to a maximum of 28 numbers

Protocol Type [PROTOCOL]

The PROTOCOL tag defines the signaling protocol—in this case it will always be SS7, with the following attributes:

PROTOCOL.VARIANT

This has the same value as selected in the Environment screen of SS7itty

PROTOCOL.ISUP

This is the ISUP variant.

Note: This tag is generated by D7 Enablement only and has no effect when passed in as a tag from WebSphere Voice Response.

Switching SS7 Default values [ISUPPARM]

The configuration file ISUPParm.cfg contains the SS7 default values. However these values can be cloned and modified into different "named" sets. The tag ISUPPARM allows these different parameter sets to be used with the tag value naming the cloned set. This facility is useful if :

- Values need to be changed on a call by call basis but the tags for the value are not available.

- A number of tags are being used on a regular basis to justify creating their own set.

For details in creating these cloned sets refer to Appendix C, “Configuration files,” on page 107.

Special case tags

The following tag is a special case and its availability and operation may be dependant on the local switch or network.

Release with new Called Party Number[NEWCLDN]

The NEWCLDN tag is specifically formed to decode a replacement Called Party Number into the Diagnostic field of a Release message. The format of the Tag and attributes is the same as Called Number above. Since this tag affects the Cause and Diagnostics fields of the Release message it must following a CAUSE tag.

System variables

For backwards compatibility the following system variables are supported in SS7. Note that these variables are available only to the State table applications, and tag values have precedence over System Variables.

The table below shows the list of system variables and the equivalent tag information.

Note:

1. For SV187 and SV188, these are the default assignments and can be reassigned via the service.cfg configuration file (see “Service configurable options” on page 112 for details).
2. WebSphere Voice Response truncates to 20 digits numbers that are stored in these system variables (if a tag supports more than 20 digits).

System variable	Associated tags
SV185 : Called Number	CLDN
SV186 : Calling Number	CLGP
SV187 : User 1	ORIGIN
SV188 : User 2	REDIRN

Chapter 7. Problem determination

This section aims to identify any problems in SS7 Support for WebSphere Voice Response, and suggests possible fixes to them.

Stage 1: Is the SS7 Server working?

This test investigates whether the D7 package is present on this machine and checks its current state of operation, by using the following instructions:

1. Log onto the SS7 Server machine as dtuser
2. Enter the SS7_MAINT command
3. From the top level menu, enter F (Fault Finding)
4. On the submenu, enter 2 (D7 Process Check)

A screen similar to that below is displayed:

```

*****
*** D7 Process checking ***
*****
  PID  STAT  MODE  HOST          MUX      OBJECT
 9852  ok    LX | A  b80ss7b     spm/0    daemon [name=apmd]
 7358  ok    LX | A  b80ss7b     spm/2    daemon [name=mlugd]
 8716  ok    LX | A  b80ss7b     spm/3    daemon [name=spmd]
11428  ok    LX | A  b80ss7b     spm/4    daemon [name=netd]
27248  ok    LX | A  b80ss7b     spm/5    daemon [name=alarmd]
27248  ok    L | A  b80ss7b     spm/6    nmdobj [name=AlarmExt]
23362  ok    LX | A  b80ss7b     spm/7    daemon [name=dsmd]
23622  ok    LX | A  b80ss7b     spm/8    daemon [name=dkmd]
24032  ok    LX | A+ b80ss7b  upm/0/0  daemon [name=upmd0]
27488  ok    LX | A  b80ss7b     upm/0/3  ss7obj [sp=0 up=5]
27154  ok    N | A  b80h        spm/11   nmdobj [name=b80h]
 4182  ok    L | A  b80ss7b     spm/16   nmdobj [name=eps_ps]
*****
*** SUCCESS: D7 system configured for SS7 Server

```

Based on the message at the bottom of your screen, go to the corresponding section as defined in this table:

Message	Action
SUCCESS : D7 system configured for SS7 Server	Go to stage 1c
SUCCESS : D7 system configured for WVR operation	Go to stage 1a
FAILED : D7 system is not running	Go to stage 1b
FAILED : MTP-3(upmd) is not running	Go to stage 1d
FAILED : ISUP(isupd) is not running.	Go to stage 1d

Stage 1a: Machine is not a Server

This machine appears to be configured for WebSphere Voice Response client use. Try the following steps:

1. Use option D from the SS7_MAINT top-level menu to stop and then restart the D7 package.
2. Then repeat the machine configuration; enter S followed by S again.

Stage 1b: D7 package is not running

As it suggests, it appears that the D7 package is not running. Try using option D in SS7_MAINT to start the D7 package. Using the M option may offer useful information as to why it does not start.

Stage 1c: D7 system configured for SS7 Server

This is the expected outcome, so continue to Stage 2. If the machine has been configured as a combined server/client, this is also the correct outcome.

Stage 1d: MTP-3(upmd) / ISUP(isupd) is not running

The machine is configured for SS7 Server operation, however one or more of the higher layers of the SS7 Stack do not appear to be working. Try the following steps:

1. Use option D of SS7_MAINT to stop and then restart the D7 package.
2. Use SS7_MAINT to stop and then restart the D7 package with the M option, and monitor any error messages being generated by the D7 package. Cross-reference these problems with those listed in "Obscure problems" on page 96.

Stage 2: Which other machines are present?

After completing Stage 1, you should have an SS7 Server machine with a D7 package running. The next stage is to investigate which other machines are present in the SS7 Cluster.

If the SS7 cluster is a single machine then skip this stage and go to Stage 3

Otherwise use the following steps to investigate which other machines are present, and their state of health:

1. Log onto the SS7 Server machine as dtuser.
2. Enter the SS7_MAINT command.
3. From the top level menu, enter F (Fault Finding).
4. From the submenu enter 1 (D7 Network Check).

A screen is displayed as below:

```

*****
*** D7 Network checking ***
*****
LINK  TYPE      HOST          RMTHOST  STAT  TIME                HBEAT
  8  tcp/ip    Mercury      Venus    L|A   Mon May 17 13:49:24  ok
  9  tcp/ip    Mercury      Mars     U     Tue May 18 10:52:16  -
  0  pci3xapq  Venus        -        L|A   Mon May 17 13:49:17  -
Select an option :

```

The output above is from a machine named Mercury, and indicates the following:

- The first line shows a good connection between machines Mercury and Venus.
- The second line shows a failed connection between machines Mercury and Mars, but that it was once a valid connection.
- The third line shows an SS7 Adapter has been activated on machine Venus.

By matching the data displayed on your screen with the symptoms listed below, go to the corresponding section listed in the Action column:

Symptom	Action
A machine is missing from the list	Go to stage 2a
A listed machine under RMTHOST has a U under the STAT column	Go to stage 2b
An SS7 adapter card is missing from the list, but the machine holding the adapter is present in the list	Go to stage 2c
If all expected machines and SS7 adapters are present	Go to stage 2d

Stage 2a : A machine is missing from the cluster

If the machine is configured as an SS7 Server (or Combined) then repeat Stage 1 for that machine. Otherwise proceed to stage 3, assuming it is configured as a WebSphere Voice Response client.

Stage 2b: The named machine in RMTHOST has been part of the SS7 cluster but has since failed

Go to Stage 4

Stage 2c: A machine containing an SS7 adapter card is present in the SS7 cluster, but is not activated

Go to Stage 6

Stage 2d: The machines within the SS7 cluster appear to be in order

Go to Stage 5

Stage 3: Is D7 working on a WebSphere Voice Response Client?

This test attempts to identify why the D7 package is not working on a machine that is configured for WebSphere Voice Response Client operation. Perform the following steps:

1. Log onto the SS7 Server machine as dtuser
2. Enter the SS7_MAINT command
3. From the top level menu, enter F (Fault Finding)
4. On the submenu, enter 2 (D7 Process Check)

A screen similar to the one below is displayed:

```
*****
*** D7 Process checking ***
*****
  PID  STAT  MODE  HOST          MUX      OBJECT
124806 ok  LX|A  b80e          spm/0    daemon [name=apmd]
124234 ok  LX|A  b80e          spm/2    daemon [name=mlogd]
43552  ok  LX|A  b80e          spm/3    daemon [name=spmd]
36588  ok  LX|A  b80e          spm/4    daemon [name=netd]
147110 ok  LX|A  b80e          spm/5    daemon [name=alarmd]
147110 ok  L|A   b80e          spm/6    nmdobj  [name=AlarmExt]
108324 ok  LX|A  b80e          spm/7    daemon [name=dsmd]
153932 ok  LX|A  b80e          spm/8    daemon [name=dkmd]
43802  ok  L|A   b80e          spm/10   nmdobj  [name=ebs_ps]
*****
*** SUCCESS : D7 system configured for WVR operation
*****
```

Based on the message at the bottom of your screen, go to the corresponding section as defined in this table:

Message	Action
SUCCESS : D7 system configured for SS7 Server .	Go to stage 3a
SUCCESS : D7 system configured for WVR operation	Go to stage 3b
FAILED : D7 system is not running	Go to stage 3c
FAILED : MTP-3(upmd) is not running	Go to stage 3a
FAILED : ISUP(isupd) is not running.	Go to stage 3a

Stage 3a: The machine is not a WebSphere Voice Response client

This machine appears to be configured for SS7 Server use. Try the following steps:

1. Use option D of SS7_MAINT to stop and then restart the D7 package
2. Repeat the machine configuration with SS7_MAINT; use option S followed by option W.

Stage 3b: The machine is correctly running as a WebSphere Voice Response client.

If this machine was not present in a Stage 2 test, proceed to Stage 5

Stage 3c: D7 package is not running

It appears that the D7 package is not running. Try using option D of SS7_MAINT to start the D7 package. Using the M option may offer useful information as to why it does not start.

Stage 4: Machine has been present in the SS7 cluster but is not now?

Try the following step on the machine that has disappeared from the SS7 cluster.

1. Log onto the SS7 Server machine as dtuser
2. Enter the SS7_MAINT command
3. From the top level menu, enter F (Fault Finding)
4. On the submenu, enter 2 (D7 Process Check)

A screen similar to the one below is displayed:

```
*****
*** D7 Process checking ***
*****
  PID  STAT  MODE  HOST          MUX          OBJECT
124806 ok  LX|A  b80e          spm/0         daemon [name=apmd]
124234 ok  LX|A  b80e          spm/2         daemon [name=mlogd]
43552  ok  LX|A  b80e          spm/3         daemon [name=spmd]
36588  ok  LX|A  b80e          spm/4         daemon [name=netd]
147110 ok  LX|A  b80e          spm/5         daemon [name=alarmd]
147110 ok  L|A  b80e          spm/6         nmdobj [name=A1armExt]
108324 ok  LX|A  b80e          spm/7         daemon [name=dsmd]
153932 ok  LX|A  b80e          spm/8         daemon [name=dkmd]
43802  ok  L|A  b80e          spm/10        nmdobj [name=ebs_ps]
*****
*** SUCCESS : D7 system configured for WVR operation
*****
```

Based on the message at the bottom of your screen, go to the corresponding section as defined in this table:

Message	Action
SUCCESS : D7 system configured for SS7 Server .	Go to stage 4a
SUCCESS : D7 system configured for WVR operation	Go to stage 4b
FAILED : D7 system is not running	Go to stage 4c
FAILED : MTP-3(upmd) is not running	Go to stage 4d
FAILED : ISUP(isupd) is not running.	Go to stage 4d

Stage 4a: The machine has been configured for SS7 Server

The report suggests that the machine has been configured for SS7 Server (or Combined). If this is correct, proceed to Stage 5. Otherwise repeat stage 3 with this machine.

Stage 4b: The machine has been configured as a WebSphere Voice Response client

The report suggests that the machine has been configured for WebSphere Voice Response client. If this is correct, proceed to Stage 5. Otherwise repeat stage 1 with this machine.

Stage 4c: D7 package is not running

It appears that the D7 package is not running. Try using option D of SS7_MAINT to start the D7 package. Using the M option may offer useful information as to why it does not start.

Stage 4d: There is a problem starting this machine as an SS7 Server

There appears to a problem in starting this machine as an SS7 Server or combined client/server. If this is the correct configuration, then return to Stage 1. Otherwise return to Stage 3.

Stage 5: Two machines are not communicating

This stage investigates why two machines are not communicating with each other. Perform the following tests on both machines.

1. Log onto the SS7 Server machine as dtuser
2. Enter the SS7_MAINT command
3. From the top level menu, enter F (Fault Finding)

4. From the submenu, enter 1 (D7 Network Check)

An example of the output is shown in “Stage 2: Which other machines are present?” on page 80.

By comparing the data displayed on your screen with the symptoms listed below, go to the corresponding section listed in the Action column:

Symptom	Action
Both entries have a U in the STAT column in both outputs	Go to stage 5a
One output shows L A and the other shows U in the STAT column	Go to stage 5b
Both entries show L A in the STAT column	Go to stage 5c
One or both of the machines is missing from the other's output	Go to stage 5d

Stage 5a: There is no communication between the machines.

Try the following steps

1. Verify the LAN connection between the two machines and its quality.
2. Review the SS7itty configuration to ensure that the specified Primary LAN address for each machine is correct, that is, they match the values returned when you enter `hostname -s`.
3. Restart the D7 package on both machines.

Stage 5b: There has been a connection between the machines at one time.

Try the following actions:

1. Verify the quality of the LAN connections and that communication between the two machines is possible.
2. Restart the D7 package on both machines and then repeat this stage.

Stage 5c: There is a good connection between the two machines

In this case the problem has been resolved. It may be that the problem is intermittent and this stage should be repeated.

Stage 5d: There is a configuration problem in defining the machines within the SS7 Cluster

Try the following steps:

1. Review the SS7itty configuration to ensure that all machines have been declared, and have correct Primary LAN addresses.
2. Restart the D7 packages for the two machines.
3. Reapply the Host MML configuration – use options C and H from top level menu of SS7_MAINT. Review the log file for errors, then restart the D7 package on the two machines.
4. Refer to the subsection on flush configuration in Chapter 8, “Migration,” on page 99.

Stage 6: SS7 Adapter checks

This stage investigates the presence of SS7 Adapters. Perform the following tests:

1. Log onto the SS7 Server machine as dtuser
2. Enter the SS7_MAINT command
3. From the top level menu, enter F (Fault Finding)
4. On the submenu, enter 1 (D7 Network Check)

An example of the output is shown in “Stage 2: Which other machines are present?” on page 80.

By comparing the data displayed on your screen with the symptoms listed below, go to the corresponding section listed in the Action column:

Symptom	Action
The expected number of adapters are listed	Go to “Stage 7: SS7 Links.” on page 87
A machine that has an SS7 adapter is not listed	Go to “Stage 1: Is the SS7 Server working?” on page 79
A machine is listed but no adapter is shown	Go to stage 6a

Stage 6a: Locate a missing adapter

Perform the following steps:

1. On the machine that has a missing adapter, logon as dtuser
2. Enter the SS7_MAINT command
3. From the top level menu, enter F (Fault Finding)
4. On the submenu, enter 3 (D7 Card Check)

If the adapter is located, the output is similar to that shown below :

Driver	Board Type	Slot	Instance	State
pci3xapq	N/A	08	0	Availble

If the adapter is not located, output similar to that shown below is displayed:

Driver	Board Type	Slot	Instance	State
pci3xapq	N/A	08	0	Define

*** FAILED : Status is Define				
*** Card : pci3xpq				
*** Inst : 0				

If the expected SS7 adapter is missing from the output, or if the output indicates a failure case (showing Define in the State field), try the following:

1. Check that the adapter is located in an active PCI slot.
2. Verify that the adapter is functioning.
3. Try a different slot on a different PCI bus, for example try a slot on the far left or far right.

If there have been a number of adapters being plugged in or removed in the past, the ODM might be confused. Try the following:

1. Remove all the adapters and restart the machine.
2. Log in as root.
3. Run `cfgmgr`

Repeat stage 6a; if cards are still being shown as missing, seek help in correcting the ODM database.

Stage 7: SS7 Links.

Reaching this stage indicates that the D7 package is functioning correctly on all machines in the SS7 cluster, and that this machine has the correct configuration.

This stage checks if all the SS7 links are functional. Perform the following steps:

1. Logon to one of the SS7 Servers as `dtuser`
2. Enter the `SS7_MAINT` command
3. From the top level menu, enter **M** (D7 Maintenance Programs)
4. On the submenu, enter **L** (Link Status)

The link monitoring program (AccessStatus) is started, and a screen similar to that below appears:



Figure 9. The link monitoring program screen

There should be one entry displayed for each Link that was configured in SS7itty. If one is missing, return to SS7itty and review the Link configuration. There might also be additional information present in the log file `/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/log/mml.log`. Once you have corrected the link information, perform the following steps:

1. Reapply the SS7 configure in SS7_MAINT, by selecting options C, followed by S alongside the name of the machine on which the Link is configured to be present. Review the error log, but ignore any errors that suggest duplication.
2. Otherwise perform the reconfiguration as outlined in Chapter 8, "Migration," on page 99.

Based on the status of the links in the link monitoring display, select from the following:

Status	Action
If an entry is not showing "is" under the column L2State	Go to stage 8a
If an entry does not show "available" under L3State, but does show "is" under L2State	Go to stage 8b
If all entries show "available" and "is"	Go to "Stage 9: Is the Enablement package working?" on page 89

Stage 8a: The physical connection for this Link is faulty

Try the following procedure:

1. Check that the correct port on the adapter has been cabled, and that the cable is connected to a valid port at the remote end. If a T1/E1 analyser is present, this will help to prove it.
2. Examine the alarm files in `/usr/opt/D7/access/RUN/alarmlog`. Entries that are similar to the following would point to a problem with the physical connection:

```

Tue May 11 11:11:18 2004
ALARM $870101 HOST: Venus SP: 0 LVL: Minor
MTP-L2: T2 expired [lset=1 link=0]
Tue May 11 11:11:18 2004
ALARM $87010a HOST: Venus SP: 0 LVL: Minor
MTP-L2: Alignment not possible [lset=1 link=0]

```

3. Verify that the correct timeslot has been selected when using SS7itty.

Stage 8b: Incorrect Point Code value

This suggests that the cabling is functioning correctly, but what is connected at the remote end does not match the expected Point Code. Verify that the DPC value configured in SS7itty under Link Sets matches the Point Code of the far end.

Stage 9: Is the Enablement package working?

Repeat this stage for each WebSphere Voice Response client in the SS7 cluster:

1. Log onto the WebSphere Voice Response client as dtuser.
2. From the top level of the WebSphere Voice Response GUI select **Operation** —> **Custom Server Manager**. From the list, check that the following custom servers are present:
 - SS7_D7
 - D7WVRErrorReport

Status	Action
If either is missing	Go to stage 9a
If both are present but either is not running	Go to stage 9b
If neither of the above conditions apply	Go to stage 9c

Stage 9a: The custom servers have not been imported.

Try the following solution:

1. Logon as dtuser.
2. Enter the SS7_MAINT command.
3. From the top level menu, enter **S** (SS7 Server/WebSphere Voice Response Client).
4. From the submenu, enter **W** (Configure this Machine). This performs the WebSphere Voice Response client configuration process, including the import of the custom server at step 5. Watch for any errors being reported.

Stage 9b: A custom server is not running

While monitoring the WebSphere Voice Response system monitor screen, start the custom server(s). A white information alarm will report that the SS7_D7 has started. Select from the following outcomes:

Outcome	Action
No further alarms are reported	Go to "Stage 10" on page 91.
Alarm 30038 is reported	Go to stage 9c
Alarm 30001 is reported	Go to stage 9d
A number of alarms 30008 are reported	Go to Stage 9e

Stage 9c: One or more configuration files cannot be located

Try the following to correct this situation:

1. Logon as dtuser.
2. Enter the SS7_MAINT command.
3. From the top level menu, enter **S** (SS7 Server/WebSphere Voice Response Client).
4. From the submenu, enter **W** (Configure this Machine). This performs the WebSphere Voice Response client configuration, including copying the configuration file at step 7. Watch for any errors being reported.

Stage 9d

The 30001 alarm indicates that the SS7_D7 custom server has been rejected when attempting to bind and activate the ISUP interface. Try the following:

1. If the SS7 Server has been redefined recently, check that this WebSphere Voice Response client has received an updated copy of its configuration files. Refer to "Installation on a WebSphere Voice Response client" on page 28.
2. If subtext suggests that the a GRPID does not exist, check that the WebSphere Voice Response pack configuration matches the values that were configured in SS7itty.

Stage 9e

Clear the alarms from the system monitor and if they do not reoccur, go to Stage 10. Otherwise, this might suggest that there is a compatibility problem between the SS7 variants, in which case seek further advice.

Stage 10

At this stage the following have been tested and proved to be working:

- All D7 packages are installed and functioning correctly
- All SS7 links are activated and working
- All custom servers relating to D7 Enablement are installed and working

The circuit is now brought into service and verified. Perform the following on all WebSphere Voice Response clients:

1. Logon to the system as dtuser
2. Enter SS7_MAINT
3. Select the option M from the top level menu, and then M again. The following screen is displayed:

```
SS7 Circuit Status on b80e []
 0  2  4  6  8 10 12 14 16 18 20 22
-----
17
18| B B B B B B B B B B B B B B B B B B B B B B B B
19| B B B B B B B B B B B B B B B B B B B B B B B B
20| B B B B B B B B B B B B B B B B B B B B B B B B
21| B B B B B B B B B B B B B B B B B B B B B B B B
22| B B B B B B B B B B B B B B B B B B B B B B B B
23| B B B B B B B B B B B B B B B B B B B B B B B B
24| B B B B B B B B B B B B B B B B B B B B B B B B
25| B B B B B B B B B B B B B B B B B B B B B B B B
26| B B B B B B B B B B B B B B B B B B B B B B B B
27| B B B B B B B B B B B B B B B B B B B B B B B B
28| B B B B B B B B B B B B B B B B B B B B B B B B
29| B B B B B B B B B B B B B B B B B B B B B B B B
30| B B B B B B B B B B B B B B B B B B B B B B B B
31| B B B B B B B B B B B B B B B B B B B B B B B B
32| B B B B B B B B B B B B B B B B B B B B B B B B

Selected Circuit ... 384
Call state           = IDLE
Near End Service state = Ubk
Far End Service state = Ubk
Press Q to quit
```

4. Now bring the WebSphere Voice Response circuit into service, either from the System Monitor screen, or by using the wvrtrunk command from the system prompt.

The example above shows the first trunk is in service (not blocked by either end), while the remainder are blocked locally, as their trunks have not been brought into service. After completing the steps above, wait for a short period while the changes resulting from the last step take effect. Then select from the following:

Status	Action
No Bs (Block) are present on the display	Go to "Stage 11: Inbound Traffic" on page 92

Status	Action
Bs are present and the lower part of the screen indicates that there are near-end blocks.	Verify that all trunks have been brought into service.
Bs are present and the lower part of the screen indicates that there are far-end blocks .	Verify with the network operator and have the circuit brought into service.
Characters I or W are present.	Go to stage 10a.

Stage 10a: Possible message synchronization problem

. Try the following actions:

1. Using the Custom Server Manager GUI, stop and restart the SS7_D7 custom server. Then repeat stage 10.
2. Verify with the network operator and have the circuit brought into service.

Stage 11: Inbound Traffic

This suggests that all the circuits are now working and ready for call traffic, but you need to test that this is the case, as described in this section.

To test, make an inbound call to a valid number on the WebSphere Voice Response Client, then select from the following outcomes:

Outcome	Action
After dialling, silence is heard	Go to stage 11a
After dialling, a busy tone is heard, or a similar verbal report	Go to stage 11c
You successfully connect to WebSphere Voice Response, and receive the selected service	Go to "Stage 12: Testing outbound calls" on page 94

Stage 11a: Verify that the call is coming into a WebSphere Voice Response client

To simplify this task, select one WebSphere Voice Response client, which should be configured to receive inbound call for the dialled service. Take the remaining WebSphere Voice Response systems out of service. Now perform the following steps:

1. Log into the selected system as dtuser.
2. Enter `ss7view -message`.
3. Make an inbound call to the selected service. If the inbound call is successfully answered, the following output is displayed:

```

13:29:38.264:0000:   IAM/ISUP_SETUP[1234] ----->>
13:29:38.265:0000:                               SL_CALL_SETUP_IND ----->>
13:29:38.265:0000:   <<----- ACM/ISUP_OGC_SELECTED
13:29:38.265:0000:   <<----- ACM/ISUP_ALERT
13:29:38.268:0000:                               <<----- SL_CALL_ANSWER_REQ
13:29:38.268:0000:                               SL_CALL_ANSWER_CNF [SL_REPLY_SUCCESS] ----->>
13:29:38.269:0000:   <<----- ANM/ISUP_SETUP
.....
13:29:41.846:0000:                               <<----- SL_CALL_TERMINATE_REQ
13:29:41.846:0000:   <<----- REL/ISUP_RELEASE[Cause:CSV_NORMALCALLCLR]
13:29:41.861:0000:   RLC/ISUP_RELEASE ----->>
13:29:41.862:0000:                               SL_CALL_TERMINATE_CNF [SL_REPLY_SUCCESS] ----->>

```

Now select from the following:

Status	Action
There was no output	Go to stage 11b
The IAM/ISUP_SETUP was shown but no SL_CALL_SETUP_IND	Go to stage 11c
The SL_CALL_SETUP_IND was shown but no SL_CALL_ANSWER_REQ	Go to stage 11d
The sequence as shown in the screen above (down to the line ANM/ISUP_SETUP) was displayed	Go to stage 11e

Stage 11b: An inbound call is not being received by the SS7 cluster

Verify with the network operator that the configuration is correct and has been enabled.

Stage 11c: SS7_D7 has rejected the call request, possibility with invalid parameters.

This condition should be supported by alarms appearing in the WebSphere Voice Response System Monitor. If further help is required collect event and trace data (the test will need to be repeated). For more information see "Collecting event and trace information" on page 67. Pass this information to IBM support for assistance.

Stage 11d: Possible problem in the call-answering mechanism of WebSphere Voice Response

Refer to the relevant documentation for the application language used, to verify that it has been correctly assigned.

Stage 11e: The inbound call is connected and the application is running.

Try the following steps:

1. Ensure that the application is delivering voice information.
2. Ensure the voice bearer cables for connections to the associated DTTA cards are present, and in the correct order (voice could be transmitted down the wrong cable).

Stage 12: Testing outbound calls

Arrange for an application to generate an outbound call, and monitor the SS7 message as before by entering `ss7view -message -t`. Perform the outbound call.

If it works, the display will look similar to below:

```
13:49:03.918:0383:                <<----- SL_CALL_SETUP_REQ
13:49:03.924:0383:                <<----- IAM/ISUP_SETUP[624185]
13:49:03.947:0383:                ACM/ISUP_PROCEED ----->>
.....
13:49:05.879:0383:                ANM/ISUP_SETUP ----->>
13:49:05.879:0383:                SL_CALL_SETUP_CNF [SL_REPLY_SUCCESS] ----->>
.....
13:49:10.110:0383:                <<----- SL_CALL_TERMINATE_REQ
13:49:10.110:0383:                <<----- REL/ISUP_RELEASE[Cause:CSV_NORMALCALLCLR]
13:49:10.127:0383:                RLC/ISUP_RELEASE ----->>
13:49:10.127:0383:                SL_CALL_TERMINATE_CNF [SL_REPLY_SUCCESS] ----->>
```

If not, select from the following:

Status	Action
No message are displayed	Go to stage 12a
The first line containing SL_CALL_SETUP_REQ is present but the second is not, or is not the same as containing IAM/ISUP_SETUP	Go to stage 12b
The message format matches, but the destination was not contacted (did not ring)	Verify that the outbound called number is correct for the device you are trying to contact.

Stage 12a: Check the returned value from the application's Makecall instruction

You might find the following problems:

- The MakeCall request is rejected due to bad numbers or parameters. Check all the parameters. Try to simplify the tags or parameters.

- No Outbound lines available. Check that the relevant circuits (both local and remote) are in service. The remote end can be checked using the `ss7view -circuit` command.

Stage 12b: The network is rejecting the call

Look at the output from `ss7view -message` for a REL/ISUP RELEASE going from the network to WebSphere Voice Response (left to right). It contains a cause code in square brackets—this is the reason the call is being rejected. Use the `ss7view -cause <name>` to translate the cause code. For example `ss7view -cause CSV_NORMALCALLCLR` returns "Normal call, clearance".

Redundancy – WebSphere Voice Response client failure.

When a WebSphere Voice Response client fails or is taken offline, the associated SS7 Server that is managing its circuits passes responsibility and control to another WebSphere Voice Response client. As a WebSphere Voice Response client that is taking over the circuits of another client cannot perform call control on these circuits (the voice bearers are not physically connected to it), it can only take the circuits out of service in a controlled manner, and thereby ensure no further calls are taken by these circuits.

When the original WebSphere Voice Response client comes back online, it takes back the reassigned circuits from the other WebSphere Voice Response clients. The WebSphere Voice Response System Monitor reports alarm messages when it is given control of another client's circuits, and again when it releases these circuits.

The `ss7view -circuit` command shows this changeover. It replaces with A the `-` character for the CICs of other WebSphere Voice Response systems, when this client is requested to take over.

The effects of call overload

Whenever a new application is planned for your configuration, you should carefully consider the call volume or call rate. The planning section earlier in this document describes how to calculate the Link requirements. However, if these are not reviewed when new applications are added to the WebSphere Voice Response system, you might find that the original calculation does not meet the new requirements.

The effect of overloading Links can vary from application to application but possible problems might include:

- Calls do not connect, or are slow to connect even when there are many free circuits.

- The HMCG messages from the D7 package occur frequently.
- Quiescing circuits does not seem to work.
- Large numbers of unrelated error messages are reported.

Obscure problems

This section describes other miscellaneous problems.

HMCG Alarm Message

The HMCG Alarm messages are generated by the D7 package (SS7 Stack) when there are too many messages buffered for transmission. These messages indicate that new messages will be discarded until the "low-water mark" in the message buffer has been reached.

This message typically occurs when circuits are taken in and out of service, possibly when they are under heavy call load. If large numbers of these messages are occurring, revisit Chapter 2, "Planning," on page 13. However if the problem occurs in small bursts around circuits being taken in and out of service, it is possible to change the "high-water mark" to a higher value.

LAN Configuration problems

If you suspect that the SS7 cluster is having LAN problems, check whether SS7 Machines are reporting other machines appearing, and then disappearing from the LAN.

Most LAN configurations are set for auto-negotiation, which typically configures at 100 MB per second, full duplex. However some machine types (for example H80), when set for auto-negotiation, select half duplex which causes problems on the LAN. You can check this by entering the AIX command `entstat -d <LAN device>`, where <LAN device> is the interface being used, for example `en0`.

Look towards the bottom of the data displayed for the lines:

```
Media Speed Selected: Auto negotiation
Media Speed Running: 100 Mbps Full Duplex
```

If the "Speed Running" is not set to Full Duplex, contact your AIX expert to correct the status.

Unexpected primitive

Occasionally, the D7 package may report unexpected primitives. These are warnings and are not necessarily errors. A primitive is likely to be

"unexpected" because it is out of sequence. This typically occurs when both parties release at the same moment in time.

However if this occurs in more than 1 in 100 calls it may suggest there is a problem in the ISUP variant being used. In this case you should contact your IBM representative for more information.

Machine locks up in dual LAN configuration

If one or more machines in an SS7 Cluster no longer respond to LAN-based commands or other operations on the stream package, it may be caused by D7 having become confused. In this case, you should review the dual LAN configuration. An imbalance in any machine in the configuration may cause confusion within D7. Typical failures include:

- Incorrect routing
- Mixed routing
- Inability to resolve LAN addresses

Chapter 8. Migration

Upgrading from an existing Distributed7 or D7 Enablement package to a new version is straightforward. Before starting, always ensure that you:

- Read any README file in the upgrade software package.
- Have put a copy of the Distributed7 licence in a safe place.

SS7 Server migration

Remove the existing packages (D7 and D7 Enablement). The D7 Enablement configuration directories remain on your machine and are not overwritten, so the SS7_MAINT operation is not required, other than to restart the packages. Install the upgrade packages as outlined in Chapter 3, “Installing SS7 Support for WebSphere Voice Response,” on page 21 earlier in this document.

WebSphere Voice Response client migration

Remove the existing packages (D7 and D7 Enablement). If you are updating the WebSphere Voice Response base product, you should make a backup of the following directories:

```
/usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg/*  
/usr/lpp/dirTalk/db/current_dir/ca/D7WVRErrorReport/*
```

After completing the upgrade of the WebSphere Voice Response base product, check that the above directories are still present. If they are not, recover them from the backup. Install the upgrade packages as outlined in Chapter 3, “Installing SS7 Support for WebSphere Voice Response,” on page 21.

Adding additional machines to the SS7 Cluster

This section describes the operation of adding additional machines to your SS7 cluster. If there are any changes or removals from the configuration, read “Changes or Removals from SS7 Cluster” on page 100first.

- Review Chapter 2, “Planning,” on page 13 to ensure that there are sufficient Links to carry the new traffic to your machines.
- Review your license needs with your IBM representative. New licenses may need to be installed on existing machines as well as on the new machine.
- Update the entries in SS7itty to include the new machine, plus any additional configurations it may have.

Updating existing machines

Take the following actions:

- (All Machines) If necessary update the license files on the existing machines and stop and restart the D7 package.
- (All Machines) Copy across the configuration files that were updated by SS7itty, as outlined in the installation section.
- (All Machines) Use SS7_MAINT to reinstall the host configuration file. Note that there will be error messages recorded in the log, but these are errors that are caused by duplication of the existing machine. These duplication errors can be ignored.
- (SS7 Server) If there are two servers, ensure that they are both active, and then re-apply the SS7 configuration to one of them. The reloading generates a number of errors referring to duplication, which can be ignored.
- (New machines) Perform an installation on new machines as outlined in Chapter 3, "Installing SS7 Support for WebSphere Voice Response," on page 21.

After the above actions are completed, verify that the system is working.

Changes or Removals from SS7 Cluster

If part of an upgrade requires the moving or removal of machines, Links, or changes of Point Code, then the SS7 Server needs to have the databases flushed. This requires the whole SS7 Cluster to be closed down.

- (All Machines) Stop the D7 packages running on all machines.
- (All Machines) Update any license files as necessary.
- (All Machines) Copy across the configuration files that were updated by SS7itty as outlined in Chapter 3, "Installing SS7 Support for WebSphere Voice Response," on page 21.
- (SS7 Server) Use SS7_MAINT to flush the database. From the top-level menu, use option C and then option F.
- (WebSphere Voice Response Client) If a machine is being removed or renamed (including change of LAN address) perform the same flush operation on the WebSphere Voice Response clients.
- (SS7 Server) Start the D7 package, and apply the new configuration as described in Chapter 3, "Installing SS7 Support for WebSphere Voice Response," on page 21.
- (WebSphere Voice Response Client) If the client databases were flushed, repeat the operation described above.

Once the above actions are complete, you should verify that the system is working.

Appendix A. Sample forms for recording information

AIX Machines

	Machine name	Primary IP	Secondary IP	Machine type	Serial number
1					
2					
3					
4					
5					
6					
7					
8					
9					
10					
11					
12					

SS7 Adapters

	SS7 adapter type	Machine name
1		
2		
3		
4		
5		
6		
7		
8		

Originating Point Code

Originating Point Code	
------------------------	--

Signaling Links

	Link name	Machine name	DPC and SLC	SS7 adapter and instance	Port and timeslot
1					
2					
3					
4					
5					
6					
7					
8					

Route Sets

	Route Set name	DPC	Bearer Trunks	First Link Set	Second Link Set	Third Link Set	Fourth Link Set
1							
2							
3							
4							
5							
6							
7							
8							

Trunks

	SS7 trunk	Route Set name	WebSphere Voice Response Trunk	WebSphere Voice Response Name
1				
2				
3				
4				
5				
6				
7				

	SS7 trunk	Route Set name	WebSphere Voice Response Trunk	WebSphere Voice Response Name
8				

Appendix B. Uninstalling SS7 Support for WebSphere Voice Response

This section covers the removal of the D7 and D7 Enablement packages. Use the following procedure:

1. Logon as dtuser and use the WebSphere Voice Response GUI to stop and remove the SS7_D7 and D7WVRErrorReport custom servers.
2. Shutdown WebSphere Voice Response by running DT_shutdown from a dtuser command prompt.
3. Use either SS7_MAINT or the D7 command ebs_stop to terminate the D7 package.
4. Change the userid to root and run smitty remove
5. Select the following packages and remove them:
 - d7.*
 - dirTalk.SS7_D7.D7_Enablement (D7 Enablement)
6. If the configuration files are no longer required, delete the files by entering:

```
rm -rf /usr/lpp/dirTalk/db/current_dir/ca/SS7_D7_cfg
rm -rf /usr/lpp/dirTalk/db/current_dir/ca/D7WVRErrorReport_cfg
```

Appendix C. Configuration files

This appendix covers the extensive area of configuration files for the D7 Enablement package and SS7_D7 custom Server. All configuration files are located in subdirectories of the /home/dirTalk/current_dir/ca/SS7_D7_cfg directory. Configuration files are read only once at startup, so any changes to these files requires a restart of the SS7_D7 custom server, to ensure that they are read.

Each configuration file can be located in one of two different directories. The SS7_D7 custom server first looks in a directory with the same name as its given machine name (as returned by the hostname -s command). However, if they are not present, it expects the file to be in directory AnyMachine.

Master configuration file

This file defines the list of other configuration files and the order in which they are processed. There are no active configuration options, but calls are made to the following configuration files:

Filename: SS7-Master.cfg
Designated Editor: Fixed at manufacture (read only)

D7Trunks.cfg

This is created by the SS7itty program.

LogEvent.cfg

This configuration, which relates to event logging, is controlled by ss7view.

Service.cfg

Contains a configuration which might be configured by IBM.

User.cfg

Contains a configuration which might be configured by the end user.

ISUPParms.cfg

Contains the configuration with the default for various IE elements.

Trunk association file

This configuration file is generated by the SS7itty and contains the CIC (trunks) to WebSphere Voice Response (trunks) associations.

The syntax is as follows:

Trunk <WebSphere Voice Response Trunk> at <CIC Trunk>

Filename: D7Trunks.cfg
Designated Editor: SS7itty

where

<WebSphere Voice Response Trunk>

is a one base trunk number that relates to the WebSphere Voice Response trunks.

<CIC Trunk>

is a zero base CIC number which is the first CIC of the associated D7 trunk.

This option defines the association between WebSphere Voice Response trunks and SS7 CICs. The CIC must be multiple of the trunk size, that is 0, 32, 64 for E1 and 0, 24, 48 for T1.

LoopBackSupport = Enable/Disable

This option indicates whether the voice hardware can support the Traffic loop back required when a Local COT test is being performed. Currently only DTTA are supported.

PointCode = Higher/Lower

This option indicates whether this SS7 Cluster has a higher or lower point code relative to the connecting Switch or STP. When set to Auto, this option is used by the Outbound CIC Allocation feature .

StateTableFile = "<FileName>"

This option defines the Signaling State Table to be loaded. This in turn defines the SS7 Trunks size and variant.

ISUPHost=[*"<Host name>"[:<STAYALIVE TRUNK>],]

This option defines the preference list of host machine names that are expected to be running the ISUP demon. The parameter is a quoted list of host machine names with the first name being the preferred choice. On startup, the custom server will contact each given host machine in turn until it finds an ISUP demon that accepts the connection. An error message is generated for each failed connection that gives the host machine.

<STAYALIVETRUNK> is an optional reference to define a non-existent ISUP trunk. This is used to ensure that active connections are present on all SS7 Servers, in case of an SS7 Server failover.

If the ISUPHost option is not present, the default host machine is local.

Logging and event file

This configuration file holds the options for controlling the event handler, trace handling and alert suppression.

Filename: LogEvent.cfg
Designated Editor: ss7view (program)

Trace logging

The trace logging options define what information is written to the SS7-Trace file. In normal operation, the trace-to-text file is disabled, but can be enabled with the following options:

Trace = [<trace options>]

where the <trace options> are:

Info Useful information relating to the passage of the program, such as activities in the internal state table.

Action Typically logs information relating to entry to other functions.

Message Logs detailed information about messages sent and received.

Subsystem Details AIX system requests, for example creating shared memory.

Screen Indicates that the log information should be sent to standard out and not to the trace file. This is only useful if SS7SigLib is being run from the command line.

Table Details the outbound call allocation tables.

Function Enables function and line reporting in the D7 Enablement package. This option does not affect the trace logging — SS7 Trace is not created or updated.

An example of the trace options would be:

```
Trace = Info Subsystem Function
```

Error options

```
Error=[<error options>]
```

where the error options are:

Screen

Indicates that the error information is to be sent to standard out as well as to the error file and WebSphere Voice Response. This is useful only if SS7SigLib is being run from the command line.

Event options

The following lines are used to control the event-capturing facility:

```
EventTrigger = <list>
```

```
EventTrigCmd = <valid shell command>
```

```
EventSize = <size>
```

EventTrigger=<list>

This defines which error events cause the current contents of the Event buffer to be written to disk file. Each item in the list consist of the following :

```
<WVError>:<times>
```

where WVError is the error number as passed to WebSphere Voice Response, in the range 30000-30199. The special error of ANY can be used to refer to any error events within the valid range. The <times> is the number of occurrences after which no further captures are generated. The <times> and the colon (:) can be omitted, in which case event files will continue to be issued.

An example would be EventTrigger = ANY: 3, 30000:4, 30020:2. This matches any error three times, error 30000 four times, and finally error 30020 twice. Note that if the first error detected by the system is 30020, this is handled under the ANY error as it is first in the list.

When an error trigger is detected, the EventTrigCmd is processed. This calls ss7view to copy the shared memory into disk file.

EventTrigCmd = <valid shell command>

This describes the command you should run when an event trigger has been detected. Typically this would call the `ss7view` command to copy the event shared memory to disk file. However, other commands can be substituted.

Note that the command is run in the background relative to `SS7_D7` custom server, so does not delay the real-time execution. However, there is no monitoring of the program and over-triggering or badly chosen commands could result in the system running out of CPU and processing slots.

The command can utilize two substitutions to describe the event with the use of the `%d` parameter. The first `%d` is replaced with the Error number that triggered this capture and the second is replaced with the issue number.

An example would be:

```
EventTrigCmd = "ss7view -event R Events/Event-%d-%d"
```

This example shows the `ss7view` being run to copy the event-shared memory in raw mode into the file `Events/Event-%d-%d`. The first `%d` will be substituted with the error number that triggered this execution and the second `%d` is an incrementing issue number for this error.

EventSize = <size>

The default value for size is 300000. This defines the size of the shared memory block used to capture the events. The value refers to the number of events captured before wrapping around

AlertSupress=[*(<AlertNumber>,| <Alertnumber>-<Alertnumber>,)]

This option defines a number of alerts which are to be suppressed when an attempt is made to send to the WebSphere Voice Response reporting mechanism. The condition is still recorded in the `SS7_D7` custom server error file.

The list of Alerts to suppress can be given as a list of single numbers or as ranges separated by commas. This suppress option works only for Alert in the range 30000-30499. An example would be: `AlertSupress=30021, 30030-30035`

Statistical Information

`Stats = Off/Interval <interval> Buffer <buffer>`

This controls the statistical collection option. `<interval>` defines how many seconds between collecting samples and `<buffer>` defines how many samples to remember.

Service configurable options

This configuration file holds a series of options under the following groups:

Filename: Service.cfg
Designated Editor: SS7 knowledgeable user

Outbound Channel Assignment

The Allocation option defines how outbound calls are allocated circuits. There are three modes of operation: Auto, Manual and Raw:

Allocation = Auto

This is the default mode and means that SS7_D7 custom server follows the ITU Q.764 2.9.1.3. or ANSI described method. SS7_D7 examines the signaling point code and then allocates either Odd circuits from 1 or Even circuits from the last circuit. In general, the connecting switch allocates circuits in the opposite sense, and thus minimizes glare conditions.

Allocation = Manual <manual definition> [then <manual definition>]
<manual definition> = [<Odd/Even>]<Up/Down> by circuit <cic/trunk>

where

- Odd/Even is optional and only Odd or Even circuits are allocated.
- Up/Down — in the case of Up circuits are allocated from 1 and in the case of Down from the last circuits.
- cic/trunk, the cic allocation by cic number or trunk by trunk number, then cic numbers.

By using the option THEN a second sequence can be defined

Allocation Raw = <list of comma separated cics>

This is a simple list of CIC and the order that they are to be allocated in. The Allocation Raw line can occur any number of times to build up a large list of CICs to allocate.

The SS7_D7 custom server follows the channel groups issued by WebSphere Voice Response, and if an outbound call requests a circuit from a sub-range of Channel groups, the selected allocation method above is followed, but within the Channel Group selected.

Parameter compatibility checking

These options define how SS7_D7 reacts to invalid SS7 parameters and the parameter compatibility messages.

SS7Checking=<Check Item>:<Check option> *

Check Items are:

ParmValue

Relates to problems regarding the parameter values; typically these are values listed as Spare or reserved in the ITU/ANSI spec.

ParmUnknown

Relates to parameters received which have no software routine to process them.

MsgUnknown

Relates to messages received which have no software routine to process them.

Check options are:

Ignore Does nothing.

Report

Creates error message and passes to WebSphere Voice Response.

Confusion

Same as Report, but also returns a Confusion message.

Release

Same as Report but also returns a Release message.

ParmComp

Same as Report, and follows the Parameter Comp. rules.

The ANSI default option is to set all three Check Items to ParmComp, to follow the ITU specification. However, in some situations it might be worthwhile setting some or all of these options to Report or Ignore.

SS7Checking = ParmRejected:<parameter comma separated list>

This option allows a list of SS7 parameters to be defined which will be applied according to the rules above. If a problem parameter is detected but not on this list it is ignored. This option is useful if checking is required, but if one or more parameters are creating unnecessary errors and need to be removed from the checking loop.

The default is to process all parameters and allow bad values to be reported and follow the Parameter Comp. rules. For example:

SS7Checking = ParmValue:Report, MsgUnknown:Ignore
SS7Checking = ParmRejected:UUIIN

This will report only those problems with values in the UUIIN parameters.

Tag handling

These options defines which Tag fields are loaded when sending indication messages to WebSphere Voice Response. Currently there are two Tag groups:

Tag Setup <Tag comma separated list>

Tag Release <Tag comma separated list>

Tag Setup refers to the setup indicator message that is sent to WebSphere Voice Response when an inbound call is detected. The associated list of tags is loaded in the Application tag field. Similarly the Tag Release refers to a network release and Disconnect indicator.

The purpose of these options is to control the number of tags that are processed and passed to WebSphere Voice Response, as in the future the number of Tag options could become considerable.

StayAlive

StayAlive = Disable/Enable [<WaitPeriod>]

The StayAlive option controls the restart of the software in the event of failure. Disable indicates that should the software fail, it is not to be restarted, whereas enable indicates that the software is to be restarted after a defined period. The is optional and defines the number of seconds to wait before attempting a restart.

Miscellaneous

UpperCIC=<value>[Default:4096]

This defines the highest—supported CIC.

Distributed7 parameters

SignalingPointCode=<0..7>

This defines the Signaling Point Code to pass to D7 when initializing the ISUP and MTP interfaces. The default value is zero.

UserServicePoint=<0..10>

This defines the UserServicePoint parameter for D7 when the MTP interface is initialized. The default value is disabled and thereby the MTP interface is not initialized.

SV187 Call User 1 / SV188 Call User 2 assignments

This defines which numbers are assigned to the Call User system variables when processing an inbound call setup:

CallUser1 = [Origin, Redirect, Generic] Default = Origin

CallUser2 = [Origin, Redirect, Generic] Default = Redirect

ISUP default IE parameters

This configuration file holds the default mandatory and optional values for IEs of SS7 messages that are sent by WebSphere Voice Response. These values can be overridden by state table tags, if supported.

Filename: ISUPParms.cfg
Designated Editor: SS7 knowledgeable user

The supported IEs and parameters are not documented here due to the complexity of the value, but are self-documented within ISUPParm.cfg. These values should be reviewed and, if necessary changed with the aid of SS7 protocol documents.

Creating a Clone

The original ISUPParm.cfg has only one set of default values, but these values can be cloned to assist in specific instances. To create a clone, append the following line to the ISUPParm.cfg file:

```
ISUPbank="<name>"
```

where <name> is a short descriptive name. Copy any previous declared IEs from earlier in the file to a point below this instruction and modify the parameters as necessary.

To select this clone, use the WebSphere Voice Response tag ISUPPARM, with the same name when performing a setup or release (without the double quotation marks used in the .cfg file). Additional clones can be created by repeating the above; note that clones are always based on the default, and are not compounded.

AlarmTranslation.cfg

This configuration file defines the translation between D7 Error messages and WebSphere Voice Response alert messages. Each line defines the relationship between the translations.

Location: /usr/lpp/dirTalk/db/current_dir/ca/D7WVRErrorReport
Machine type: WebSphere Voice Response client only
Uses process: D7WVRErrorReport custom server.

D7 Error Code

a six digit hexadecimal value as defined in the D7 documentation

WVR Alert message

the WebSphere Voice Response alert code (between 32200 and 32299). This value must be predefined for any meaningful information to be displayed. If the value -1 is used, this is interpreted as "suppress error message".

Reporting Level

this is a value between 1 and 3, that defines the message reporting level.

User file

Following installation, this configuration file does not contain any information, but when necessary the user can add entries when directed.

Filename: User.cfg
Designated editor: User


```

15:29:48.596:0000:                                <<----- SL_CHANNEL_QUIESCE_REQ
15:29:48.597:0000:                                <<----- BLO/ISUP_BLOCKING
15:29:48.613:0000:    BLA/ISUP_BLOCKING ----->>
15:29:48.613:0000:                                SL_CHANNEL_QUIESCE_CNF ----->>

```

Outbound Call with suspend

```

15:38:46.938:0383:                                <<----- SL_CALL_SETUP_REQ
15:38:46.938:0383:                                <<----- IAM/ISUP_SETUP[624185]
15:38:46.960:0383:    ACM/ISUP_PROCEED ----->>
.....
15:38:48.047:0383:    ANM/ISUP_SETUP ----->>
15:38:48.047:0383:                                SL_CALL_SETUP_CNF [SL_REPLY_SUCCESS] ----->>
.....
15:38:50.078:0383:    SUS/ISUP_SUSPEND ----->>
.....
15:38:52.269:0383:                                <<----- SL_CALL_TERMINATE_REQ
.....
15:39:20.070:0383:    RES/ISUP_RESUME ----->>
15:39:20.070:0383:                                <<----- REL/ISUP_RELEASE[Cause:CSV_NORMALCALLCLR]
15:39:20.086:0383:    RLC/ISUP_RELEASE ----->>
15:39:20.087:0383:                                SL_CALL_TERMINATE_CNF [SL_REPLY_SUCCESS] ----->>

```

Quiescing a circuit with active call

```

15:53:06.953:0381:                                <<----- SL_CALL_SETUP_REQ
15:53:06.953:0381:                                <<----- IAM/ISUP_SETUP[624185]
15:53:06.976:0381:    ACM/ISUP_PROCEED ----->>
15:53:07.041:0381:    ANM/ISUP_SETUP ----->>
15:53:07.041:0381:                                SL_CALL_SETUP_CNF [SL_REPLY_SUCCESS] ----->>
.....
15:53:11.835:0381:                                <<----- SL_CHANNEL_QUIESCE_REQ
15:53:11.835:0381:                                <<----- BLO/ISUP_BLOCKING
15:53:11.851:0381:    BLA/ISUP_BLOCKING ----->>
.....
15:53:37.130:0381:                                <<----- SL_CALL_TERMINATE_REQ
15:53:37.130:0381:                                <<----- REL/ISUP_RELEASE[Cause:CSV_NORMALCALLCLR]
15:53:37.146:0381:    RLC/ISUP_RELEASE ----->>
15:53:37.146:0381:                                SL_CALL_TERMINATE_CNF [SL_REPLY_SUCCESS] ----->>
15:53:37.146:0381:                                SL_CHANNEL_QUIESCE_CNF ----->>

```

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For country-specific notes on the use of WebSphere Voice Response, refer to the README file located in the directory /usr/lpp/dirTalk/homologation. The file name is in the format README_homologation.xxxx, where xxxx is the country/region identifier.

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Glossary

The following terms and abbreviations are defined as they are used in the context of WebSphere Voice Response. If you do not find the term or abbreviation you are looking for, see *IBM Dictionary of Computing, McGraw-Hill, 1994* or the *AIX: Topic Index and Glossary, SC23–2513*.

Special Characters

μ -law The companding algorithm that is used primarily in North America and Japan when converting from analog to digital speech data. (Compand is a contraction of compress and expand.) Contrast with *A-law*.

Numerics

2 B-channel transfer feature

See *Integrated Services Digital Network (ISDN) two B-channel transfer*.

3270 host application

An application on the IBM System/370™ System/390®, or AS/400® that interacts with terminals that support the 3270 data stream.

3270 script language

See *script language*.

3270 server

A function of WebSphere Voice Response that provides a software interface between WebSphere Voice Response and IBM System/370, System/390, or AS/400 architecture business applications that interact with terminals that support the 3270 data stream. Contrast with *custom server*.

5ESS (1) A Lucent Technologies switch.

(2) The ISDN protocol that is used on the 5ESS switch. It provides 23 B-channels and a D-channel over a T1 trunk.

6312 Digital Trunk Telephony Adapter (DTTA)

See *Digital Trunk Telephony Adapter*.

6313 Digital Trunk Telephony Adapter (DTTA) with Blind Swap Cassette (BSC)

See *Digital Trunk Telephony Adapter with Blind Swap Cassette*.

A

A-law The companding algorithm that is used in Europe, Latin America, and other countries when converting from analog to digital speech data. (Compand is a contraction of compress and expand.) Contrast with *μ -law*.

access protocol

A protocol that is used between an external subscriber and a switch in a telephone network.

ACD See *automatic call distributor*.

ACL See *application connectivity link*.

action See *state table action*.

Action Palette

An area that contains folders and icons that can be selected to create state table actions.

Address Resolution Protocol (ARP)

In HACMP, the Internet communication protocol that dynamically maps Internet addresses to physical (hardware) addresses on local area networks. Limited to networks that support hardware broadcast.

The `usr/sbin/cluster/etc/clinfo.rc` script, which is invoked by the `clinfo` daemon whenever a network or node event occurs, updates the system ARP cache. This ensures that the IP addresses of all cluster nodes are updated after an IP address takeover. The script can be further customized to handle site-specific needs.

administrator profile

Data that describes a WebSphere Voice Response user. Information that is in an administrator profile includes ID, password, language preference, and access privileges.

ADSI See *analog display services interface*.

ADSI telephone

A "smart" telephone that can interpret and return ADSI data.

advanced intelligent network (AIN)

A telephone network that expands the idea of the *intelligent network (IN)* to provide special services more efficiently; for example, by giving users the ability to program many of the services themselves.

AIN See *advanced intelligent network*.

alarm Any condition that WebSphere Voice Response thinks worthy of documenting with an *error message*. Strictly, the term **alarm** should include only red (immediate attention) and yellow (problem

condition), but it is also used to refer to green (a red or yellow message has been cleared) and white (information) conditions. Contrast with *alert*.

alert A message that is sent to a central monitoring station, as the result of an alarm. Contrast with *alarm*.

alternate mark inversion (AMI)

A T1 line coding scheme in which binary 1 bits are represented by alternate positive and negative pulses and binary 0 bits by spaces (no pulse). The purpose is to make the average dc level on the line equal to zero.

AMI See *alternate mark inversion*.

analog

Data in the form of continuously variable signals, such as voice or light signals.

analog display services interface (ADSI)

A Bellcore signaling protocol that is used with existing voice networks. ADSI supports analog transmission of voice and text-based information between a host or switch, voice mail system, service bureau, or similar, and a subscriber's ADSI-compatible screen telephone. A single voice-grade telephony channel is shared between voice and data, using a technique by which the channel is taken over for the transmission of modem-encoded data.

ANI See *automatic number identification*.

annotation

In speech recognition, an alphanumeric string that is used to mark a grammar when it is defined. When the grammar is used in an

application, both the word and the alphanumeric string are returned to the application.

announcement-only greeting

In voice mail, a *greeting* that does not give the *caller* a chance to leave a *voice message*.

application

A (usually) customer-written program or set of programs that might consist of one or more state tables or custom servers that are running on WebSphere Voice Response, with associated voice segments. See *voice application*.

application connectivity link (ACL)

A service that transmits out-of-band information between WebSphere Voice Response and the Siemens Hicom 300 switch.

application profile

Data that describes initial actions that are to be performed when the telephone is answered. Information in an application profile indicates to the channel process which state table to load.

application server interface (ASI)

The principal software component of WebSphere Voice Response that manages the real-time channel processing.

application server platform (ASP)

A platform that is used for Web and voice applications for e-business.

ASI See *application server interface*.

ASP See *application server platform*.

audio name

The audible name that relates to a specific application profile ID and mailbox.

auto-attendant

Automated attendant. A voice application that answers incoming calls and asks callers which number or other service they would like.

automatic call distributor (ACD)

A telephone system feature that automatically queues and processes inbound calls according to predefined rules. For example, a call might be routed to the agent whose line has been idle longest.

automatic number identification (ANI)

A service available in the U.S. that provides the telephone number of the calling party. It is generated by the caller's originating central office switch, sent to a telephone network carrier if required, then sent directly either to a switch or to a voice processing system.

autostubbing

A state table icon view utility that automatically converts lines into *stubs* when they cross a specified number of columns.

B

B8ZS Bipolar with 8-zero substitution. A T1 line code that is required for 64Kb channels such as ISDN.

B-channel

See *bearer channel*. See also *Integrated Services Digital Network (ISDN)* .

background music

Any audio data that is to be played on a *music channel*.

barge-in

The capability that allows a prompt to be interrupted by an utterance

that is then passed to a speech recognizer. See also *cut-through channel*.

baseforms

The set of phonetic pronunciations that are associated with a grammar. In WebSphere Voice Server, the IBM dictionary of pronunciations is used.

basic rate interface (BRI)

The means of ISDN access that is normally used by private subscribers. It provides two B-channels of 64 Kb per second and one D-channel of 16 Kb per second for signaling. This is often known as 2B+D. Contrast with *primary rate interface (PRI)*.

beans Java beans with which you can build voice applications to use the services of WebSphere Voice Response on any platform.

bearer channel

In an ISDN interface, a duplex channel for transmitting data or digital voice between the terminal and the network. The B-channel operates at 64 Kb per second.

bearer service

The type of service that defines how an ISDN connection will be used. Typical bearer services are speech telephony, 64 Kb per second data, and high-quality speech.

blind transfer

A type of call transfer in which the call is routed to another extension and the original call is ended. No check is made to determine whether the transferred call is answered or if the number is busy. Contrast with *screened transfer*.

bnf Abbreviation for Backus-Naur Form,

which is used to describe the syntax of a given language and its notation. In speech recognition, a special adaptation of grammar representation that is specified by *Speech Recognition Control Language (SRCL)* (pronounced "circle").

bos Base Operating System.

bps bits per second.

BRI See *basic rate interface*.

bridge See *DVT bridge*.

British Approvals Board for Telecommunications

The British standards organization that is responsible for approval of equipment that is to be attached to the PSTN.

C

cadence

The modulated and rhythmic recurrence of an audio signal. For example, a series of beeps or a series of rings.

call Telephone call. Often used to mean a single run-time instance of a *voice application*.

call center

A central point at which all inbound calls are handled by a group of individuals in a controlled sequential way. Call centers are usually a front end to a business such as airline ticketing or mail order.

Call Control eXtensible Markup Language (CCXML)

Language designed to provide telephony call control support for VoiceXML or other dialog systems.

Refer to the CCXML forum web site at <http://www.w3.org/TR/ccxml>

call forwarding

The process of sending incoming calls to a different number.

called party

Any person, device, or system that receives a telephone call. Contrast with *caller*.

caller

(1) Any person, device, or system that makes a telephone call. (2) Often used to refer to any user of a voice application, although WebSphere Voice Response might have made an outbound call and the user is really the *called party*. (3) In voice mail, any person who makes a telephone call to a *subscriber*. Contrast with *user*.

calling line identification presentation (CLIP)

An ISDN supplementary service that advises the called party of the caller's number; for example, by displaying it on a telephone display panel.

CallPath

Software that provides basic computer-telephony integration (CTI) enablement and comprehensive CTI functionality. This includes access to, and management of, inbound and outbound telecommunications.

call session

The sequence of events that occurs from the time a call is started to the time all activities related to answering and processing the call are completed.

call transfer

A series of actions that directs a call

to another telephone number. See also *dual-line call transfer*.

CAS See *channel associated signaling*.

cascading resources

Resources that can be taken over by more than one node. A takeover priority is assigned to each configured cluster resource group in a per-node way. In the event of a takeover, the node with the highest priority gets the resource group. If that node is unavailable, the node with the next-highest priority gets the resource group, and so on.

CAS tone

Customer Premise Equipment Alerting Signal tone. In ADSI, this tone is sent to the ADSI telephone to switch the phone to data mode.

CBX See *computerized branch exchange*.

CCH See *Comité de Coordination de l'Harmonisation*.

CCITT

See *Comité Consultatif International Télégraphique et Téléphonique*.

CCS See *common channel signaling (CCS)*.

central office (CO)

A telephone switching system that resides in the telephone service provider's network. Different types of central office switches exist, depending upon the role of the switch in the telephone network. Commonly, a central office switch connects customer lines to other customer lines or trunks, and is the point at which local subscriber lines end for switching to other lines or trunks.

central registry

A component of the Licence Use

Management network topology. A server's database that logs requests for licenses, upgrades for licenses, and journals all license activity in a tamper-proof auditable file.

CEPT See *Conférence Européenne des Administrations des Postes et Télécommunications*.

CGI See *Common Gateway Interface*.

channel

One of the 24 channels that are on a T1 trunk, or one of the 30 channels that are on an E1 trunk. See also *speech recognition session*, *music channel*.

channel-associated signaling (CAS)

A method of communicating telephony supervisory or line signaling (on-hook and off-hook) and address signaling on T1 and E1 digital links. The signaling information for each traffic (voice) channel is transmitted in a signaling channel that is permanently associated with the traffic channel. On T1 links, supervisory signaling is sent in the traffic channel by using *robbed-bit signaling (RBS)*. On E1 links, a separate channel is used to send signaling. Address signaling can be transmitted either in the signaling channel (out-of-band) or in the traffic channel (in-band). Contrast with *common channel signaling (CCS)*.

channel bank

A device that converts an analog line signal to a digital trunk signal.

channel number

The identifying number that is assigned to a licensed channel on the T1 or E1 trunk that connects

WebSphere Voice Response to the switch, channel bank, or channel service unit.

channel process (CHP)

The AIX process that runs the logic of the state table; each active caller session has one active channel process.

channel service unit (CSU)

A device that is used to connect a digital phone line to a multiplexer, a channel bank, or directly to another device that generates a digital signal. A CSU performs specific line-conditioning and equalization functions, and responds to loopback commands that are sent from the CO.

CHP See *channel process*.

CIC See *circuit identification code*.

CICS See *customer information control system*.

circuit identification code (CIC)

A 12-bit number that identifies a trunk and channel on which a call is carried.

clear message

A message that is displayed by WebSphere Voice Response to tell the operator that a red or yellow error message has been cleared.

client node

In a single system image (SSI), a WebSphere Voice Response system that handles interactions with callers. A client node must have a telephony connection. It does not store application or voice data; it gets data from the server node of the SSI.

CLIP See *calling line identification presentation*.

cluster

Loosely-coupled collection of independent systems (nodes) that are organized into a network to share resources and to communicate with each other. HACMP defines relationships among cooperating systems where peer cluster nodes provide the services that a cluster node offers if that node cannot do so.

cluster configuration

User definition of all cluster components. Component information is stored in the *Object Data Manager*. Components include cluster name and ID, and information about member nodes, adapters, and network modules.

CO See *central office*.

codec Refers to adapters that compress and decompress video files. The letters "codec" represent "compression/decompression"; in the past, they represented "coder/decoder."

Comité de Coordination de l'Harmonization

The CEPT committee responsible for standards.

Comitato Elettrotecnico Italiano

The Italian standards organization responsible for signaling protocols.

Comité Consultatif International Télégraphique et Téléphonique (CCITT)

This organization has been renamed and is now known as the International Telecommunications Union - Telecommunication Standardization Sector (ITU-T).

common channel signaling (CCS)

A method of communicating telephony information and line signaling events (for example, call setup and call clearing) on a dedicated signaling channel. The signaling channel is either a predefined channel on an E1 or T1 digital link, or a completely separate link between the switch and WebSphere Voice Response. For data integrity and reliability, the information is usually communicated using a data link protocol. The telephone information and line signaling events are sent as data packets. SS7 and ISDN are common-channel signaling protocols. Contrast with *channel associated signaling*.

Common Gateway Interface (CGI)

An interface to programs that provide services on the world wide Web.

compiled grammar file

A *grammar* in binary format that was built by the WebSphere Voice Server grammar development tools.

compound license

In License Use Management, a type of license that allows a system administrator to generate license passwords for a given number of licenses. A compound license can generate either nodelocked or non-nodelocked licenses, but not both

computer-telephony integration (CTI)

The use of a general-purpose computer to issue commands to a telephone switch to transfer calls and provide other services. Typically, CTI is used in call centers.

computerized branch exchange (CBX)

A computer-driven, digital communications controller that provides telephone communication between internal stations and external networks.

Conférence Européenne des Administrations des Postes et Télécommunications (CEPT)

European Conference of Postal and Telecommunications Administrations.

configuration file

See *parameter file*.

configuration parameter

A *variable* that controls the behavior of the system or the behavior of all applications that are running on the system. See *parameter file*, *system parameter*.

container window

A window that lists the names of all existing objects of the same type.

context

A set of one or more grammars that is enabled and used during a recognition action. The grammars are specified by a FILELIST file. Parameters that influence the recognition, such as the maximum initial silence period and the ending silence period, are also defined by the context. More than one context can be enabled for a recognition.

context name

The name given to a context in a context profile that is used for WebSphere Voice Server.

context profile

Describes to the WebSphere Voice Server process which contexts should be loaded into an engine. A

WebSphere Voice Response for Windows application specifies which context profiles to load into the engine it has reserved.

context type

Indicates to the recognition engine how to interpret the grammar file. Possible types are: VOCAB_FILE, GRAMMAR_FILE, TEXT, MNR_FILE, MNR, PERSONAL_FILE, PERSONAL_WDS, BASEFORM_FILE.

continuous speech recognition

Recognition of words that are spoken in a continuous stream. Unlike isolated or discrete word recognition, users do not have to pause between words.

conversation

See *speech recognition session*.

CPE See *customer premises equipment*.

CSU See *channel service unit*.

CTI See *computer-telephony integration*.

customer information control system (CICS)

A licensed program that enables transactions that are entered at remote workstations to be processed concurrently by user-written application programs. It includes facilities for building, using, and maintaining databases.

custom server

A C language or C++ language program that provides data manipulation and local or remote data stream, database, or other services that are additional to those that the state table interface provides. Custom servers provide an interface between WebSphere

Voice Response and business applications, functions, or other processes to give callers access to business information and voice processing functions such as speech recognition.

customer premises equipment (CPE)

Telephony equipment that is on the premises of a business or domestic customer of the telephone company. An example is a private branch exchange (PBX).

cut-through channel

A channel of voice data that has been passed through echo-cancellation algorithms. The channel provides echo-canceled voice data that can then be used by the engine in a recognition attempt. This is similar to *barge-in*.

D

daemon

In the AIX operating system, a program that runs unattended to perform a standard service.

database server node

In a single system image (SSI), a WebSphere Voice Response system that contains the WebSphere Voice Response DB2® database. This is usually the same node as the voice server node.

DBIM The internal database manager of WebSphere Voice Response.

DBS The database server of WebSphere Voice Response.

DCBU See *D-channel backup*.

D-channel

See *delta channel*.

D-channel backup (DCBU)

An ISDN NFAS configuration where two of the T1 facilities have a D-channel, one of which is used for signaling, and the other as a backup if the other fails. See also *non-facility associated signaling*.

DDI See *direct inward dialing*.

DDS See *production system*.

delay start

A procedure that is used with some channel-associated signaling protocols to indicate when a switch or PABX is ready to accept address signaling. After seizure, the switch sends off-hook until it is ready to accept address signaling, at which time it sends on-hook. Contrast with *immediate start* and *wink start*.

delta channel

In an ISDN interface, the D-channel or delta channel carries the signaling between the terminal and the network. In a basic rate interface, the D-channel operates at 16 Kb per second. In a primary rate interface, the D-channel operates at 64 Kb per second.

destination point code (DPC)

A code that identifies the signaling point to which an MTP signal unit is to be sent. Unique in a particular network.

development system

A WebSphere Voice Response system that is not used to respond to, or make, "live" calls; it is used only to develop and test applications. Contrast with *production system*.

dial To start a telephone call. In telecommunication, this action is

performed to make a connection between a terminal and a telecommunication device over a switched line.

dial by name

To press the keys that are related to subscribers' names instead of to their telephone numbers or extensions.

dialed number identification service (DNIS)

A number that is supplied by the public telephone network to identify a logical called party. For example, two toll-free numbers might both be translated to a single real number. The DNIS information distinguishes which of the two toll-free numbers was dialed.

dialog box

A secondary window that presents information or requests data for a selected action.

dial tone

An audible signal (call progress tone) that indicates that a device such as a PABX or central office switch is ready to accept address information (DTMF or dial pulses).

DID See *direct inward dialing*.

digital signal processing (DSP)

A set of algorithms and procedures that processes electronic signals after their conversion to digital format. Because of the specific mathematical models that are required to perform this processing, specialized processors are generally used.

Digital Subscriber signaling System Number 1 (DSS1)

A signaling protocol that is used

between ISDN subscriber equipment and the network. It is carried on the ISDN D-channel. ITU-T recommendations Q.920 to Q.940 describe this protocol.

Digital Trunk Ethernet Adapter (DTEA)

A Radysis adapter card that provides the audio streaming (RTP) interface between the WebSphere Voice Response internal H.100 bus and Ethernet for a maximum of 120 channels using uncompressed (G.711) voice, and compressed G.723.2 and G.729A compressed voice.

Digital Trunk No Adapter (DTNA)

A device driver that supports uncompressed (G.711) voice RTP streaming.

Digital Trunk Telephony Adapter (DTTA)

The IBM Quad Digital Trunk Telephony PCI Adapter. In WebSphere Voice Response, this adapter is known as a DTTA. It allows you to connect directly to the telephony network from a pSeries computer without the need for an external pack.

Digital Trunk Telephony Adapter (DTTA) with Blind Swap Cassette (BSC)

The IBM Quad Digital Trunk Telephony PCI Adapter. In WebSphere Voice Response, this adapter is known as a DTTA. It allows you to connect directly to the telephony network from a pSeries computer without the need for an external pack. This DTTA includes a short Blind Swap Cassette (BSC) which is required for installing the DTTA in machines that use the BSC (for example, the pSeries 650–6M2).

diphone

A transitional phase from one sound to the next that is used as a building block for speech synthesis. Typically, between one thousand and two thousand diphones exist in any national language.

direct dial in (DDI)

See *direct inward dialing*.

direct inward dialing (DID)

A service that allows outside parties to call directly to an extension of a PABX. Known in Europe as direct dial in (DDI).

direct speech recognition

Identification of words from spoken input that are read directly from the telephony channel. Contrast with *indirect speech recognition*.

DirectTalk bean

One of the beans that is provided with WebSphere Voice Response. It provides access from a voice application to simple call control functions: waiting for a call, making an outgoing call, handing a call over to another application, and returning a call when finished.

discrete word recognition

Identification of spoken words that are separated by periods of silence, or input one at a time. Contrast with *continuous speech recognition*.

disconnect

To hang up or terminate a call.

Distributed Voice Technologies (DVT)

A component of WebSphere Voice Response that provides an interface to allow you to integrate your own voice technology (such as a speech recognizer) with your WebSphere Voice Response system.

distribution list

In voice mail, a list of subscribers to whom the same message can be sent.

DMS100

(1) A Northern Telecom switch. (2) The custom ISDN protocol that is run on the DMS100 switch, providing 23 B-channels and a D-channel over a T1 trunk.

DNIS See *dialed number identification service*.

double-trunking

See *trombone*.

down The condition in which a device is unusable as a result of an internal fault or of an external condition, such as loss of power.

downstream physical unit (DSPU)

Any remote physical unit (data link, storage, or input/output device) that is attached to a single network host system.

DPC See *destination point code*.

drop-in grammar

A set of precompiled grammar rules that can be used by an application-specific grammar to improve the recognition performance.

DSP See *digital signal processing*.

DSPU See *downstream physical unit*.

DSS1 See *Digital Subscriber signaling System Number 1*.

DTMF

See *dual-tone multifrequency*.

DTEA See *Digital Trunk Ethernet Adapter*.

DTNA

See *Digital Trunk No Adapter*.

DTTA See *Digital Trunk Telephony Adapter*.

dtuser The name of the AIX account that is set up during the installation process for the use of all users of WebSphere Voice Response.

dual-line call transfer

A call transfer method in which the primary and secondary lines remain bridged until a call is completed. (Also known as tromboning; see *trombone*).

dual-tone multifrequency (DTMF)

The signals are sent when one of the telephone keys is pressed. Each signal is composed of two different tones.

DVT See *Distributed Voice Technologies*.

DVT bridge

The interface between a voice technology component (such as a speech recognizer) and the DVT server. A bridge must exist for each technology that you want to integrate with DVT.

DVT_Client2

A WebSphere Voice Response custom server that passes commands and data to DVT_Server.

DVT interface

A WebSphere Voice Response programming interface that is used by a DVT bridge. It enables integration of voice applications with *Distributed Voice Technologies* to provide functions such as speech recognition.

DVT_Server

A component of DVT that allocates and manages system resources in response to requests from DVT_Client2.

DVT service

The combination of a voice application, a DVT bridge, and a voice technology that allows a caller to interact with your business.

dynamic vocabulary

A vocabulary that is defined while an application is running.

E

E&M A channel-associated signaling protocol in which signaling is done using two leads: an M-lead that transmits battery or ground and an E-lead that receives open or ground.

E1 A digital trunking facility standard that is used in Europe and elsewhere. It can transmit and receive 30 digitized voice or data channels. Two additional channels are used for synchronization, framing, and signaling. The transmission rate is 2048 Kb per second. Contrast with *T1*.

echo cancelation

A filter algorithm that compares a copy of the voice data that is being sent to a caller, with the voice data being that is received from the caller. Any echo of the sent data is removed before the received data is sent on, for example, to a speech recognizer.

edge See *result*.

EDL See *exchange data link*.

emulation

The imitation of all or part of one computer system by another, so that the imitating system accepts the same data, runs the same programs, and gets the same results as the imitated computer system does.

endpoint

In *Voice over Internet Protocol*, a place where calls are originated and ended.

engine

A speech recognition process that accepts voice data as input and returns the text of what was said as output. It is the process that performs the recognition.

engine type

Each engine must be configured with a specific type. The type is a textual tag that is associated with a specific engine and does not change the operation or functionality of the engine.

error message

Any message that is displayed by WebSphere Voice Response in the System Monitor as an *alarm* and optionally written to the WebSphere Voice Response error log, or to the AIX error log (as an *alert*). Strictly, the term **error message** should include only red (immediate attention) and yellow (problem situation) messages, but it is also used to refer to green (a red or yellow message has been cleared) and white (informational) messages.

Ethernet

A 10/100 network connection between the VoIP gateway and the Speech Server that supports VoIP.

ETS European Telecommunications Standard or European Telecommunication Specification.

ETSI European Telecommunications Standards Institute.

Euro-ISDN

The common European ISDN

standard, agreed in 1993, that provides a basic range of services and supplementary services using 30 B-channels plus a D-channel over an E1 trunk.

exchange data link

A serial connection that carries messaging information between WebSphere Voice Response and the Lucent Technologies 1AESS, Northern Telecom DMS100, Ericsson MD110 switch, or Siemens Hicom 300.

exit A point in a supplied application from which control can be passed to another custom-written application. On completion, the custom-written application passes control back to the supplied application.

F**fade in**

To gradually increase the volume of sounds, such as background music.

fade out

To gradually decrease the volume of sounds, such as background music.

failover

A transparent operation that, in the event of a system failure, switches responsibility for managing resources to a redundant or standby system. Also known as fallover.

FDM See *Feature Download Management*.

Feature Download Management (FDM)

An ADSI protocol that enables several alternative key and screen overlays to be stored in an ADSI telephone, and to be selected by predetermined events at the telephone.

Federal Communication Commission

(FCC) The standard body in the United States that is responsible for communication.

field An identifiable area in a window that is used to enter or display data.

FILELIST

A WebSphere Voice Server Telephony runtime file that defines which files to load into a WebSphere Voice Server engine. It contains a list in the form:

```
context type grammar filename
      ...           ...
```

Recursion is not permitted; that is, no contexts of type FILELIST can be specified in a FILELIST. When a FILELIST is loaded, all the grammars that are specified in it are loaded into the engine. From then on, the grammars that are loaded when the FILELIST is specified are regarded as a single context.

Foreign Exchange Subscriber (FXS)

A signaling protocol that links a user's location to a remote exchange that would not normally be serving that user, to provide, for example, calls to outside the local area at the local rate.

frame A group of data bits that is surrounded by a beginning sequence and an ending sequence.

fsg Abbreviation for finite state grammar. In WebSphere Voice Server, the extension of a file that contains grammar specifications in compiled, binary form. It is generated from a .bnf file and is called a .fsg file.

function

In ADSI, an ADSI instruction or group of instructions.

FXS See *Foreign Exchange Subscriber*.

G

gatekeeper

A component of a *Voice over Internet Protocol* that provides services such as admission to the network and address translation.

gateway

A component of *Voice over Internet Protocol* that provides a bridge between VoIP and circuit-switched environments.

G.711 Specification for uncompressed voice for PSTN and *Voice over Internet Protocol* access.

G.723.1

Compressed audio codecs that are used on *Voice over Internet Protocol* connection for voice.

G.729A

Compressed audio codecs that are used on *Voice over Internet Protocol* connection for voice.

glare A condition that occurs when both ends of a telephone line or trunk are seized at the same time.

grammar

A structured collection of words and phrases that are bound together by rules. A grammar defines the set of all words, phrases, and sentences that might be spoken by a caller and are recognized by the engine. A grammar differs from a *vocabulary* in that it provides rules that govern the sequence in which words and phrases can be joined together.

greeting

In voice mail, the recording that is heard by a caller on reaching *subscriber's* mailbox. See also *announcement-only greeting*. Contrast with *voice message*.

greeting header

In voice mail, a recording that is made by a subscriber and played to *callers* either before or instead of a *personal greeting*.

Groupe Special Mobile (GSM)

A CEPT/CCH standard for mobile telephony.

H**HACMP (High-Availability Cluster Multi-Processing) for AIX**

Licensed Program Product (LPP) that provides custom software that recognizes changes in a cluster and coordinates the use of AIX features to create a highly-available environment for critical data and applications.

HACMP/ES

Licensed Program Product (LPP) that provides Enhanced Scalability to the HACMP for AIX LPP. An HACMP/ES cluster can include up to 32 nodes.

hang up

To end a call. See also *disconnect*.

HDB3 High-density bipolar of order 3. An E1 line coding method in which each block of four successive zeros is replaced by 000V or B00V, so that the number of B pulses between consecutive V pulses is odd. Therefore, successive V pulses are of alternate polarity so that no dc component is introduced. Note: B represents an inserted pulse that

observes the *alternate mark inversion (AMI)* rule and V represents an AMI violation. HDB3 is similar to B8ZS that is used with T1.

HDLC See *high-level data link control*.

high-level data link control

An X.25 protocol.

homologation

The process of getting a telephony product approved and certified by a country's telecommunications authority.

hook flash

A signal that is sent to a switch to request a switch feature (such as call transfer).

host application

An application residing on the host computer.

hunt group

A set of telephone lines from which a non-busy line is found to handle, for example, an incoming call.

I**immediate start**

A procedure that is used with some channel-associated signaling protocols, when the address signaling is sent within 65 milliseconds of going off-hook. Contrast with *delay start* and *wink start*.

IN See *intelligent network*.

in-band

In the telephony voice channel, signals are said to be carried in-band. Contrast with *out-of-band*.

indirect speech recognition

Identification of words from spoken

input that are read from a file.
Contrast with *direct speech recognition*.

initialize

To prepare a system, device, or program for operation; for example, to initialize a diskette.

input parameter

Data that is received by a program such as a prompt, 3270 script, custom server, or state table from the program that called it. Contrast with *local variable* and *system variable*.

integrated messaging

A messaging system in which more than one copy of a single message is stored, the copies being kept synchronized by the applications that are used to access them. Contrast with *unified messaging*.

Integrated Services Digital Network (ISDN)

A digital end-to-end telecommunication network that supports multiple services including, but not limited to, voice and data.

Integrated Services Digital Network (ISDN) call transfer

In WebSphere Voice Response, an application that allows you to transfer calls on Nortel DMS-100 switches using *Integrated Services Digital Network (ISDN) two B-channel transfer*, and on Nortel DMS-100 and DMS-250 switches using Nortel's proprietary Release Link Trunk (RLT) call transfer protocol.

Integrated Services Digital Network (ISDN) two B-channel transfer

A call transfer feature that is defined by Bellcore GR-2865-CORE

specification, and used on Nortel and Lucent switches.

Integrated Services Digital Network user part (ISUP)

Part of the SS7 protocol that supports telephony signaling applications. The ISDN user part is defined to carry signaling information that relates to digital telephones, terminals, and PABXs in customer premises.

intelligent network (IN)

A telephone network that includes programmable software that is not resident on the switch. It allows the service provider to provide special services, such as special call-handling, that are not dependent on the capabilities of the switch. See also *advanced intelligent network*.

intelligent peripheral (IP)

A voice processing system (such as WebSphere Voice Response) that provides enhanced services such as voice response, speech recognition, text-to-speech, voice messaging, and database access in an *advanced intelligent network*.

interactive voice response (IVR)

A computer application that communicates information and interacts with the caller via the telephone voice channel.

International Telecommunications Union – Telecommunication Standardization Sector (ITU-T)

The name of the organization that was previously known as the CCITT.

IP See *intelligent peripheral*.

ISDN See *Integrated Services Digital Network (ISDN)* .

ISDN two B-channel transfer
See *Integrated Services Digital Network (ISDN) two B-channel transfer*.

ISDN-UP
See *Integrated Services Digital Network user part*.

ISUP See *Integrated Services Digital Network user part*.

ITU-T See *International Telecommunications Union – Telecommunication Standardization Sector*.

IVR See *interactive voice response*.

J

Java Bean
A reusable Java component. See *beans*.

jump out
See *call transfer*.

K

key (1) One of the pushbuttons on the telephone handset; sometimes referred to as a *DTMF* key. (2) A component of the keyboard that is attached to the computer system.

key pad
The part of the telephone that contains the pushbutton keys.

key pad mapping
The process of assigning special alphanumeric characters to the keys that are on a telephone key pad, so that the telephone can be used as a computer-terminal keyboard.

L

LAN See *local area network*.

language model

For speech recognition, a set of acoustic shapes (in binary format) for a given set of words, in which word-to-word differences are maximized, but speaker-to-speaker differences are minimized. See also *vocabulary*.

LAPD See *link access protocol for the D-channel*.

licensed program product (LPP)

A separately-priced program and its associated materials that bear an IBM copyright and are offered under the terms and conditions of a licensing agreement.

license server

A machine on a network that holds licenses and distributes them on request to other machines on the network.

line error

An error on the telephone line that causes the signal to be impaired.

link access protocol for the D-channel

An HDLC protocol used in ISDN that ensures a reliable connection between the network and the user. Often used as another name for Q.921.

local area network (LAN)

A network in which computers are connected to one another in a limited geographical area. WebSphere Voice Response communication with WebSphere Voice Server speech recognition, text-to-speech, and single system image (SSI) requires a LAN that is dedicated to that purpose (unless

both are installed on the same system). A *token-ring network* is a type of LAN.

local variable

A user-defined temporary variable that can be accessed only by the program (state table, prompt, or 3270 script) for which it is defined. Contrast with *input parameter*, *system variable*.

M

macro See *system prompt*.

MAP See *mobile application part*.

MB See *megabyte*.

megabyte

(1) For processor storage and real and virtual memory, 1 048 576 bytes. (2) For disk storage capacity and transmission rates, 1 000 000 bytes.

Message Center

See *Unified Messaging*

message delivery preference

The subscriber's choice of whether voice mail is stored as voice mail only, as e-mail only, or as both voice mail and e-mail.

message delivery type

The format in which a voice message is delivered.

message signal unit (MSU)

An MTP packet that contains data.

message transfer part (MTP)

Part of the SS7 protocol that is normally used to provide a connectionless service that is roughly similar to levels one through three of the OSI reference model.

message waiting indicator (MWI)

A visible or audible indication (such as a light or a stutter tone) that a voice message is waiting to be retrieved.

MFR1 An in-band address signaling system that uses six tone frequencies, two at a time. MFR1 is used principally in North America and is described in ITU-T recommendations Q.310 through Q.332.

MIME See *multipurpose Internet mail extensions*.

mobile application part (MAP)

Optional layer 7 application for SS7 that runs on top of TCAP for use with mobile network applications.

MP See *multiprocessor*.

MSU See *message signal unit*.

MTP See *message transfer part*.

mu(μ)-law

The companding algorithm that is used primarily in North America and Japan when converting from analog to digital speech data. (Compand is a contraction of compress and expand.) Contrast with *A-law*.

multiprocessor (MP)

A computer that includes two or more processing units that can access a common main storage.

multipurpose Internet mail extensions (MIME)

A protocol that is used on Internet for extending e-mail capability and merging it with other forms of communication, such as voice mail and fax.

mumble

Non speech noise that a user interjects while speaking.

music channel

A channel on which sounds can be broadcast to one or more telephony (voice) channels.

music title

The name by which WebSphere Voice Response knows a tune.

MWI See *message waiting indicator*.

N**National ISDN**

A common ISDN standard that was developed for use in the U.S.

NAU See *network addressable unit*.

N-Best

The ability to return more than one speech recognition result. Typically, an array of results is available in the application in sequence of descending probability.

NCP See *network control program*.

NET Norme Européenne de Télécommunication.

Net 5 The test specification for conformance to the Euro-ISDN standard for primary rate access to ISDN.

network addressable unit (NAU)

Any network component that can be addressed separately by other members of the network.

network control program (NCP)

Used for requests and responses that are exchanged between physical units in a network for data flow control.

Network File System (NFS)

A protocol, developed by Sun Microsystems, Incorporated, that allows any host in a network to gain access to another host or netgroup and their file directories. In a single system image (SSI), NFS is used to attach the WebSphere Voice Response DB2 database.

network termination

See *NT mode*.

NFAS See *non-facility associated signaling*.

NFS See *Network File System*.

node In a single system image (SSI), one of the WebSphere Voice Response systems that are in the cluster.

non-facility associated signaling (NFAS)

An ISDN configuration where several T1 facilities can be controlled by a single D-channel, instead of the normal T1 configuration where each T1 facility has 23 B-channels and a D-channel (23B+D). With NFAS, all 24 timeslots of the non signaling trunks are available for voice, whereas only 23 channels can be used on the trunk that carries signaling traffic (23B+D+n24B).

NT mode

Attachment to the ISDN network is asymmetric. The network side of the connection operates in network termination, or NT, mode. User equipment operates in terminal equipment, or TE, mode.

O

ODM See *Object Data Manager*.

Object Data Manager (ODM)

A data manager intended for the storage of system data. The ODM is

used for many system management functions. Information that is used in many commands and SMIT functions is stored and maintained in the ODM as objects with associated characteristics.

off-hook

A telephone line state, usually induced by lifting a receiver, in which the line is ready to make a call.

offline

Not attached or known to the existing system configuration, and therefore not in active operation.

on-hook

A telephone line state, usually induced by hanging up a receiver, in which the line is ready to receive a call.

online In active operation.

OPC See *originating point code*.

Open Systems Interconnection (OSI)

(1.) The interconnection of open systems as specified in particular ISO standards. (2.) The use of standardized procedures to enable the interconnection of data processing systems.

Open Systems Interconnection (OSI) architecture

Network architecture that observes the particular set of ISO standards that relate to Open Systems Interconnection.

Open Systems Interconnection (OSI) Reference Model

A conceptual model composed of seven layers, each specifying particular network functions. Developed by the International Organization for Standardization

(ISO) in 1984, it is considered to be the primary architectural model for intercomputer communications

originating point code (OPC)

A code that identifies the signaling Point that originated an MTP signal unit. Unique in a particular network.

OSI See *Open Systems Interconnection*.

outgoing mail

In voice mail, messages that are sent by a subscriber to another subscriber on the same system, and have not yet been listened to by the addressee.

out-of-band

In the telephony signaling channel, as opposed to the voice channel. Signals are said to be carried out-of-band. Contrast with *in-band*.

P

PABX See *private automatic branch exchange*.

pack Each DTTA contains the equivalent of four packs. The pack is a digital trunk processor built into the digital trunk adapter, so there is no need for external hardware. See also *TPACK*.

parameter file

An ASCII file that sets *configuration parameters*.

password

A unique string of characters that is known to a computer system and to a user. The user must specify the character string to gain access to the system and to the information that is stored in it.

PBX See *private branch exchange*.

PCI See *peripheral component interconnect*.

PCM See *Pulse Code Modulation*.

PCM fault condition

A fault, such as power supply failure, or loss of incoming signal, in T1 or E1 equipment. (ITU-T G.732 and G.733.)

peripheral component interconnect (PCI)

A computer busing architecture that defines electrical and physical standards for electronic interconnection.

personal greeting

In voice mail, a *greeting* that is recorded by a *subscriber*. Contrast with *system greeting*.

phone recognition

Communicating with a computer using voice via a telephone, over a telephone line. The computer application recognizes what was said and takes suitable action.

port In time-slot management, one end of a 64 Kbps unidirectional stream that can be attached to the TDM bus.

port set

In time-slot management, a collection of ports that can be connected using a single CA_TDM_Connect() API call to a complementary collection of ports.

PRA Primary rate access (PRA). Used as another name for primary rate interface (PRI).

PRI See *primary rate interface*.

primary rate access (PRA)

See *primary rate interface*.

primary rate interface (PRI)

The means of ISDN access that is

normally used by large sites. It provides 30 (E1) or 23 (T1) B-channels of 64 Kb per second and one D-channel for signaling. This is often known as 30B+D or 23B+D. Contrast with *basic rate interface*.

primary rate ISDN (PRI)

See *primary rate interface*.

primitive

A message that is sent from one process to another.

private automatic branch exchange (PABX)

An automatic private switching system that services an organization and is usually located on a customer's premises. Often used as another name for *private branch exchange (PBX)* .

private branch exchange (PBX)

A switch inside a private business that concentrates the number of inside lines into a smaller number of outside lines (trunks). Many PBXs also provide advanced voice and data communication features. Often used as another name for *private automatic branch exchange* .

process a call

To answer the telephone and perform the correct tasks.

Process Manager

In WebSphere Voice Server, the process that manages the interaction of all telephony system processes; for example, starting and stopping text-to-speech or speech recognition sessions.

production system

A WebSphere Voice Response system that responds to or makes "live" calls. A production system can also be used to develop new

applications. Contrast with *development system*.

program temporary fix (PTF)

An update to IBM software.

program data

Application-specific data that can be associated with a call transfer from CallPath to WebSphere Voice Response, or in the opposite direction. This is equivalent to CallPath program data, but WebSphere Voice Response imposes the restriction that the data must be a printable ASCII character string, with a maximum length of 512 bytes.

prompt

(1) A message that requests input or provides information. Prompts are seen on the computer display screen and heard over the telephone. (2) In WebSphere Voice Response, a program that uses logic to determine dynamically the voice segments that are to be played as a voice prompt.

prompt directory

A list of all the prompts that are used in a particular voice application. Used by the state table to play the requested voice prompts.

pronunciation

The possible phonetic representations of a word. A word can have multiple pronunciations; for example, "the" has at least two pronunciations, "thee" and "thuh".

pronunciation dictionary

A file that contains the phonetic representation of all of the words, phrases, and sentences for an application grammar.

pronunciation pool

A WebSphere Voice Server resource that contains the set of all pronunciations.

protocol

A set of semantic and syntactic rules that determines the behavior of functional units when they get communication. Examples of WebSphere Voice Response protocols are FXS, RE, and R2.

PSTN An ITU-T abbreviation for public switched telephone network.

PTF See *program temporary fix*.

Pulse Code Modulation (PCM)

Variation of a digital signal to represent information.

pushbutton

(1) A key that is on a telephone key pad. (2) A component in a window that allows the user to start a specific action.

pushbutton telephone

A type of telephone that has pushbuttons. It might or might not send tone signals. If it does, each number and symbol on the key pad has its own specific tone.

Q

Q.921 The ITU-T (formerly CCITT) recommendation that defines the link layer of the DSS1 protocol. Q.921 defines an HDLC protocol that ensures a reliable connection between the network and the user. Often used as another name for LAPD.

Q.931 The ITU-T recommendation that defines the network layer of the DSS1 protocol. This layer carries the

ISDN messages that control the making and clearing of calls.

quiesce

To shut down a channel, a trunk line, or the whole system after allowing normal completion of any active operations. The shutdown is performed channel-by-channel.

Channels that are in an idle state are shut down immediately.

Channels that are processing calls are shut down at call completion.

R

RAI See *remote alarm indication*.

RBS See *robbed-bit signaling*.

RE See *remote extension*.

Recognition Engine server

In WebSphere Voice Server, the software that performs the speech recognition and sends the results to the client. This consists of one 'Tsm router' and at least one 'tsmp' and one 'engine'.

reduced instruction set computer (RISC)

A computer that uses a small, simplified set of frequently-used instructions to improve processing speed.

referral number

The phone number to which calls are routed, when call forwarding is active.

rejection

The identification of an utterance as one that is not allowed by a grammar.

release link trunk (RLT)

A custom specification from Nortel for ISDN call transfer.

remote alarm indication (RAI)

A remote alarm (also referred to as a yellow alarm) indicates that the far-end of a T1 connection has lost frame synchronization. The Send RAI system parameter can be set to prevent WebSphere Voice Response from sending RAI.

remote extension (RE)

An E1 signaling protocol that is similar to FXS loop start.

resource element

A component of an Intelligent Network. The resource element contains specialized resources such as speech recognizers or text-to-speech converters.

response

In speech recognition, the character string that is returned by the recognizer, through DVT_Client, to the state table. The string represents the result of a recognition attempt. This is the word or words that the recognizer considers to be the best match with the speech input.

result An indicator of the success or failure of a state table action. It is returned by WebSphere Voice Response to the state table. Also known as an edge.

result state

The state that follows each of the possible results of an action.

return code

A code that indicates the status of an application action when it completes.

RISC See *reduced instruction set computer*.

RLT See *release link trunk*.

robbed-bit signaling (RBS)

The T1 channel -associated signaling scheme that uses the least significant bit (bit 8) of each information channel byte for signaling every sixth frame. This is known as 7-5/6-bit coding rather than 8-bit coding. The signaling bit in each channel is associated only with the channel in which it is contained.

S

SAP See *service access point*.

SAS A T1 signaling protocol that is similar to FXS.

SCbus See *Signal Computing bus*.

SCCP See *signaling connection control part*.

SCP See *service control point*.

screened transfer

A type of call transfer in which the transfer of the held party to the third party is completed only if the third party answers the call. Contrast with *blind transfer*.

script The logical flow of actions for a 3270 server program.

script language

A high-level, application-specific scripting language, which consists of statements that are used to develop 3270 scripts. These scripts are part of the interface between a state table and a 3270-based host business application.

SCSA See *Signal Computing System Architecture*.

SDC See *Server Display Control*.

SDLC See *Synchronous Data Link Control*.

segment ID number

One or more numbers that are used to identify a voice or prompt segment.

Server Display Control (SDC)

An ADSI control mode in which the ADSI telephone is controlled through a dialog with a voice response system.

server node

In a single system image (SSI), a WebSphere Voice Response system that contains either the WebSphere Voice Response DB2 database, or the voice data, or both.

service access point (SAP)

An OSI term for the port through which a service user (layer N+1) accesses the services of a service provider (layer N).

service control point (SCP)

A component of the intelligent network that provides transactional services, such as translation of toll-free numbers to subscriber numbers.

service information octet (SIO)

A field that is in an MTP message signal unit. It identifies a higher layer user of MTP, and whether the message relates to a national or international network.

service node

An element of an Intelligent Network. The service node contains the service logic that controls an intelligent network application and resources.

service provider

Any company that provides services for a fee to its customers, such as telecommunication companies,

application service providers, enterprise IT, and Internet service providers.

service provider equipment (SPE)

The switching equipment that is owned by the telephone company.

session

See *speech recognition session*.

Session Initiation Protocol

A signaling protocol used for internet conferencing, telephony, presence, events notification and instant messaging.

short message service center (SMSC)

A component of the mobile telephony network, specified by the GSM group of standards, that provides for exchange of alphanumeric messages of less than 160 bytes. Messages can be exchanged between different types of system such as mobile telephone, alphanumeric pager, terminal, e-mail, telex, or DTMF telephone.

SIF See *signaling information field*.

Signal Computing System Architecture (SCSA)

An architecture that was defined by Dialogic to support interoperability of software and hardware components that are developed by different vendors in the computer telephony industry.

Signal Computing bus (SCbus)

A time division multiplexed (TDM) hardware bus that was originated by Dialogic to interconnect different vendors' computer telephony adapters. Specified as part of *Signal Computing System Architecture (SCSA)*.

signaling

The exchange of control information between functional parts of the system in a telecommunications network.

signaling connection control part (SCCP)

A layer 3 protocol that observes OSI.

signaling information field (SIF)

The user data portion of an MTP message signal unit.

signaling link code (SLC)

A code that identifies a particular signaling link that connects the destination and originating signaling points. This is used in MTP signaling network management messages to indicate the signaling link to which the message relates.

signaling link selection (SLS)

A field that is used to distribute MTP signal units across multiple signaling links.

signaling mode

The type of signaling protocol, either channel-associated signaling, or common-channel signaling.

signaling point

A node in a signaling network that either originates and receives signaling messages, or transfers signaling messages from one signaling link to another, or both.

signaling process

A WebSphere Voice Response component that controls signaling for an exchange data link or common-channel signaling protocol. Some signaling processes are

supplied with WebSphere Voice Response, and others can be custom-written.

signaling System Number 7 (SS7)

The international high-speed signaling backbone used for the public-switched telephone network.

silence

A short pause between utterances.

simple mail transfer protocol (SMTP)

An Ethernet protocol that is related to TCP/IP.

simple network management protocol (SNMP)

In the Internet suite of protocols, a network management protocol that is used to monitor routers and attached networks. SNMP is an application layer protocol. Information on devices managed is defined and stored in the application's Management Information Base (MIB). SNMP provides a means of monitoring WebSphere Voice Response resources remotely.

Simplified Message Desk Interface (SMDI)

A Northern Telecom service that transmits out-of-band information between WebSphere Voice Response and particular switches.

Simplified Message Service Interface (SMSI)

A Lucent Technologies service that transmits out-of-band information between WebSphere Voice Response and particular switches.

single system image (SSI)

A cluster of WebSphere Voice Response systems that are connected together using a *local area*

network. Each system (known as a node) in the cluster is configured as either a client or a server. A single system image typically consists of one server node and multiple client nodes. The client nodes retrieve applications and voice data from the server. A second server can be configured for redundancy.

sink A port that takes voice data from the TDM bus. Contrast with *source*.

SIO See *service information octet*.

SIP See *Session Initiation Protocol*.

SLC See *signaling link code*.

SLS See *signaling link selection*.

SMDI See *Simplified Message Desk Interface*.

SMIT See *System Management Interface Tool*.

SMP See *symmetric multiprocessor*.

SMSC See *short message service center*.

SMSI See *Simplified Message Service Interface*.

SMTP See *simple mail transfer protocol*.

SNA Systems Network Architecture.

SNMP See *simple network management protocol*.

source A port that puts voice data on to the TDM bus. Contrast with *sink*.

SPACK A logical component that consists of a base card, which connects to the digital trunk adapter in the pSeries computer, and a trunk interface card (TIC), which manages the trunk connection to the switch. Contrast with *VPACK* and *TPACK*.

SPE See *service provider equipment*.

speaker-dependent speech recognition

Identification of spoken words that is related to knowledge of the speech characteristics of one speaker. Contrast with *speaker-independent speech recognition*.

speaker-independent speech recognition

Identification of spoken words that is related to collected knowledge of the speech characteristics of a population of speakers. Contrast with *speaker-dependent speech recognition*.

special character

A character that is not alphabetic, numeric, or blank. For example, a comma (,) or an asterisk (*).

speech recognition

The process of identifying spoken words. See *discrete word recognition*, *continuous speech recognition*, *speaker-dependent speech recognition*, *speaker-independent speech recognition*.

Speech Recognition Control Language (SRCL)

In WebSphere Voice Server, a structured syntax and notation that defines speech grammars, annotations, repetitions, words, phrases, and associated rules.

speech recognition session

In WebSphere Voice Server, a sequence of recognition commands that allocate a recognition engine, and return a unique identifier to identify the engine.

speech synthesis

The creation of an approximation to human speech by a computer that concatenates basic speech parts together. See also *text-to-speech*.

SRCL See *Speech Recognition Control Language (SRCL)*.

SS7 See *signaling System Number 7*.

SSI See *single system image*.

SSI-compliant custom server

A custom server that runs correctly in a single system image. The custom server observes all the guidelines for the operation of custom servers in an SSI environment.

SSI-tolerant custom server

A custom server that runs in a single system image, but with only some restrictions.

standalone system

A WebSphere Voice Response system that is not part of a single system image (SSI). A standalone system is not connected to other WebSphere Voice Response systems, so it contains its own application and voice data.

state One step in the logical sequence of actions that makes a WebSphere Voice Response voice application.

state table

A list of all the actions that are used in a particular voice application. A component of WebSphere Voice Response.

state table action

One instruction in a set of instructions that is in a WebSphere Voice Response state table that controls how WebSphere Voice Response processes various operations such as playing voice prompts or recording voice messages. See also *state*.

stub A line in a state table that is only partially displayed.

subscriber

In voice mail, any person who owns a mailbox.

subscriber class

A named set of variables that defines a specific level of service available to telephone subscribers, such as maximum number of messages per mailbox and maximum number of members per mailbox distribution list.

subvocabulary

A vocabulary that is called by another vocabulary.

supplementary service

In Euro-ISDN, a service outside the minimum service offering that each signatory is obliged to provide. For example, *calling line identification presentation (CLIP)* and *call session*.

switch A generic term that describes a telecommunications system that provides connections between telephone lines and trunks.

symmetric multiprocessor (SMP)

A system in which functionally-identical multiple processors are used in parallel, providing simple and efficient load-balancing.

Synchronous Data Link Control (SDLC)

A discipline for managing synchronous, code-transparent, serial-by-bit information transfer over a link connection. Transmission exchanges can be duplex or half-duplex over switched or nonswitched links.

system administrator

The person who controls and

manages the WebSphere Voice Response system by adding users, assigning account numbers, and changing authorizations.

system greeting

In voice mail, a default *greeting* that is heard by callers to the mailboxes of subscribers who have not recorded a personal greeting or who have selected the system greeting. Contrast with *personal greeting*.

System Management Interface Tool (SMIT)

A set of utilities that can be used for various purposes, such as loading WebSphere Voice Response software, installing the exchange data link, and configuring SNA.

Systems Network Architecture (SNA)

An architecture that describes the logical structure, formats, protocols, and operational sequences for transmitting information units through the networks and also the operational sequences for controlling the configuration and operation of networks.

system parameter

A *variable* that controls some of the behavior of WebSphere Voice Response or applications that are running under WebSphere Voice Response. System parameters are set through System Configuration or Pack Configuration options on the Configuration menu. Some system parameter values are assigned to *system variables* when an application is initialized. Contrast with *input parameter*, *local variable*, *system variable*.

system prompt

The symbol that appears at the

command line of an operating system, indicating that the operating system is ready for the user to enter a command.

system variable

A permanent global variable that is defined by WebSphere Voice Response for use by state tables. Many system variables are loaded with values when the state table is initialized. Some values are taken from system parameters. Contrast with *input parameter*, *local variable*, *system parameter*.

T

T1 A digital trunking facility standard that is used in the United States and elsewhere. It can transmit and receive 24 digitized voice or data channels. Signaling can be imbedded in the voice channel transmission when robbed-bit signaling is used. The transmission rate is 1544 kilobits per second. Contrast with *E1*.

T1/D3 A framing format that is used in T1 transmission.

T1/D4 A framing format that is used in T1 transmission.

tag A text string that is attached to any instance of a word in a *grammar*. A tag can be used (1) to distinguish two occurrences of the same word in a grammar or (2) to identify more than one word in a grammar as having the same meaning.

Tag Image File Format-Fax (TIFF-F)

A graphic file format that is used to store and exchange scanned fax images.

TCAP See *transaction capabilities application part*.

TCP/IP See *Transmission Control Protocol/Internet Protocol*.

TDD See *Telecommunications Device for the Deaf*.

TDM See *time-division multiplex bus*.

technology

A program, external to WebSphere Voice Response, that provides processing for functions such as text-to-speech or speech recognition.

Telecommunications Device for the Deaf (TDD)

A telephony device that has a QWERTY keyboard and a small display and, optionally, a printer.

telephone input field

A field type that contains information that is entered by a caller who is using pushbutton signals. See also *field*.

terminal

(1) A point in a system or communication network at which data can enter or leave. (2) In data communication, a device, usually equipped with a keyboard and display device, that can send and receive information.

termination character

A character that defines the end of a telephone data entry.

text-to-speech (TTS)

The process by which ASCII text data is converted into synthesized speech. See also *speech synthesis*.

TIC See *trunk interface card*.

time-division multiplex bus (TDM)

A method of transmitting many

channels of data over a smaller number of physical connections by multiplexing the data into timeslots, and demultiplexing at the receiving end. In this document, one such channel can be considered to be a half-duplex unidirectional stream of 64 Kb per second.

TIFF-F

See *Tag Image File Format-Fax*

timeslot

The smallest switchable data unit on a data bus. It consists of eight consecutive bits of data. One timeslot is similar to a data path with a bandwidth of 64 Kb per second.

token A particular message or bit pattern that indicates permission or temporary control to transmit.

token-ring network

A local area network that connects devices in a ring topology and allows unidirectional data transmission between devices by a token-passing procedure. A device must receive a token before it can transmit data.

tone An audible signal that is sent across a telephone network. Single (one-frequency) tones, tritones (three sequential tones at different frequencies), dual tones (two simultaneous tones at different frequencies), and dual sequential tones exist. Each has a different meaning.

TPACK

A digital trunk processor that is implemented using DSP technology on the digital trunk adapter without the need for external hardware. One

DTTA digital trunk adapter provides up to four TPACKs on a PCI card.

transaction

A specific, related set of tasks in an application that retrieve information from a file or database. For example, a request for the account balance or the available credit limit.

transaction capabilities application part (TCAP)

Part of the SS7 protocol that provides transactions in the signaling network. A typical use of TCAP is to verify a card number, for the credit card calling service.

transaction messaging

The ability to associate an item of data, such as a transaction identifier, with a voice message. The voice message can later be retrieved by referencing the data value.

transfer

See *call transfer*.

Transmission Control Protocol/Internet Protocol (TCP/IP)

A communication subsystem that is used to create local area and wide area networks.

trombone

A connected voice path that enters an IVR from a switch on one circuit, then returns to the same switch on a parallel circuit. Two IVR ports and two circuits are consumed, but in some circumstances this might be the only way to make a connection between two callers if the attached switch does not support a Call Transfer function. Also known as double-trunking.

trunk A telephone connection between

two central offices or switching devices. In WebSphere Voice Response, a trunk refers to 24 or 30 channels that are carried on the same T1 or E1 digital interface.

trunk interface card (TIC)

The component of the pack that manages the trunk connection to the switch.

Tsm Router

In WebSphere Voice Server, a process that controls which engine processes are in use at any time. Requests for an engine by a WebSphere Voice Server Client are accepted or rejected depending on whether an engine that meets the Tsm Client's requirements is available.

tsmp In WebSphere Voice Server, a process that is running on the Recognition engine server machine that passes messages between an engine and a Tsm Client. One tsmp exists for every engine.

TTS See *text-to-speech*.

tune A piece of music or other audio data that is intended to be played as background music.

U

underrun

To run out of audio data to play, causing voice or music to be audibly broken up or cut off.

unified messaging

A messaging system in which a single copy of a message is stored and accessed by multiple applications (for example, voice mail and e-mail). Contrast with *integrated messaging*.

Unified Messaging

An IBM product that uses WebSphere Voice Response's voice processing capabilities to provide a wide range of voice mail, fax, and e-mail functions. Previously known as Message Center.

user Someone who uses WebSphere Voice Response as a system administrator, application developer, or similar. Contrast with *caller*.

utterance

A spoken word, phrase, or sentence that can be preceded and followed by silence.

V

variable

A system or user-defined element that contains data values that are used by WebSphere Voice Response voice applications. See *input parameter*, *local variable*, *system parameter*, *system variable*.

VMS See *Voice Message Service*.

vocabulary

A list of words with which WebSphere Voice Response matches input that is spoken by a caller. See also *language model*.

voice application

A WebSphere Voice Response application that answers or makes calls, plays recorded voice segments to callers, and responds to the caller's input.

voice directory

A list of *voice segments* that is identified by a group ID. Voice directories can be referenced by prompts and state tables. Contrast with *voice table*.

voice mail

The capability to record, play back, distribute, and route voice messages.

voice mailbox

The notional hard disk space where the incoming messages for a voice mail subscriber are stored.

voice message

In voice mail, a recording that is made by a caller for later retrieval by a *subscriber*.

Voice Message Service (VMS)

An Ericsson service that transmits information between WebSphere Voice Response and particular switches.

voice messaging

The capability to record, play back, distribute, route, and manage voice recordings of telephone calls through the use of a processor, without the intervention of agents other than the callers and those who receive messages.

voice model

A file that contains parameters that describe the sounds of the language that are to be recognized on behalf of an application. In WebSphere Voice Server, this is a *bnf* file. See also *grammar*.

Voice over Internet Protocol (VoIP)

The sending of telephony voice over Internet Protocol (IP) data connections instead of over existing dedicated voice networks, switching and transmission equipment. See also *gatekeeper* and *gateway*.

voice port library

A library that manages a socket connection from the client to the

voice technology. The library uses entry points that are provided by DVT.

Voice Protocol for Internet Messaging (VPIM)

The standard for digital exchange of voice messages between different voice mail systems, as defined in Internet Request For Comments (RFC) 1911.

voice response unit (VRU)

A telephony device that uses prerecorded voice responses to provide information in response to DTMF or voice input from a telephone caller.

voice segment

The spoken words or sounds that make recorded voice prompts. Each segment in an application is identified by a group ID and a segment ID and usually includes text.

voice server node

In a single system image (SSI), a server node that contains the voice data. This is usually the same node as the database server node.

voice table

A grouping of *voice segments* that is used for organizational purposes. Voice tables can be referenced by prompts, but not by state tables. Contrast with *voice directory*.

voice technology

See *technology*.

VoiceXML

VoiceXtensible Markup Language. An XML-based markup language for creating distributed voice applications. Refer to the VoiceXML forum web site at www.voicexml.org

VoIP See *Voice over Internet Protocol*.

VPACK

A component consisting of a base card, which connects to the digital trunk adapter in the pSeries computer, and a trunk interface card (TIC), which manages the trunk connection to the switch. The single digital trunk processor contains one VPACK, and the multiple digital trunk processor contains slots for up to five VPACKs. Contrast with *SPACK* and *TPACK*.

VPIM See *Voice Protocol for Internet Messaging*.

VRU See *voice response unit*.

W

World Wide Web Consortium (W3C)

An organization that develops interoperable technologies (specifications, guidelines, software, and tools) to lead the Web to its full potential. W3C is a forum for information, commerce, communication, and collective understanding. Refer to the web site at <http://www.w3.org>

WebSphere Voice Response

A voice processing system, that combines telephone and data communications networks to use, directly from a telephone, information that is stored in databases.

wink start

A procedure that is used with some channel-associated signaling protocols to indicate when a switch or PABX is ready to accept address signaling. After seizure, the switch sends a short off-hook signal (wink) when it is ready to accept address

information. Contrast with *delay start* and *immediate start*.

word spotting

In speech recognition, the ability to recognize a single word in a stream of words.

wrap

In ADSI, the concatenation of two columns of display data to form a single column.

Y

yellow alarm

See *remote alarm indication*.

Z

zero code suppression (ZCS)

A coding method that is used with alternate mark inversion to prevent sending eight successive zeros. If eight successive zeros occur, the second-least significant bit (bit 7, with the bits labeled 1 through 8 from the most significant to the least significant) is changed from a 0 to a 1. AMI with ZCS does not support clear channel operation.

List of WebSphere Voice Response and associated documentation

Here is a list of the documentation for WebSphere Voice Response for AIX and associated products. PDF and HTML versions of the documentation are available from the IBM Publications Center at <http://www.ibm.com/shop/publications/order>. Hardcopy books, where available, can be ordered through your IBM representative or at this Web site.

WebSphere Voice Response for AIX documentation can also be found by going to the IBM Pervasive software Web site at <http://www.ibm.com/software/pervasive>, selecting the **WebSphere Voice products** link, and then selecting the **library** link from the WebSphere Voice Response page.

PDF and HTML versions of the WebSphere Voice Response for AIX publications are available on the CD-ROM supplied with the product. In addition, WebSphere Voice Response for AIX, WebSphere Voice Response for Windows, Unified Messaging, and other WebSphere Voice publications are available together in PDF and HTML formats on a separately-orderable CD-ROM (order number SK2T-1787).

Note: To read PDF versions of books you need to have the Adobe Acrobat Reader (it can also be installed as a plug-in to a Web browser). It is available from Adobe Systems at <http://www.adobe.com>.

WebSphere Voice Response software

- *WebSphere Voice Response for AIX: General Information and Planning*, GC34-7084
- *WebSphere Voice Response for AIX: Installation*, GC34-7095
- *WebSphere Voice Response for AIX: User Interface Guide*, SC34-7091
- *WebSphere Voice Response for AIX: Configuring the System*, SC34-7078
- *WebSphere Voice Response for AIX: Managing and Monitoring the System*, SC34-7085
- *WebSphere Voice Response for AIX: Designing and Managing State Table Applications*, SC34-7081
- *WebSphere Voice Response for AIX: Application Development using State Tables*, SC34-7076
- *WebSphere Voice Response for AIX: Developing Java applications*, GC34-7082
- *WebSphere Voice Response for AIX: Deploying and Managing VoiceXML and Java Applications*, GC34-7080

- *WebSphere Voice Response for AIX: Custom Servers*, SC34-7079
- *WebSphere Voice Response for AIX: 3270 Servers*, SC34-7075
- *WebSphere Voice Response for AIX: Problem Determination*, GC34-7087
- *WebSphere Voice Response for AIX: Fax using Brooktrout* , GC34-7083
- *WebSphere Voice Response for AIX: Cisco ICM Interface User's Guide*, SC34-7077
- *WebSphere Voice Response for AIX: MRCP for State Tables*, SC34-7086
- *WebSphere Voice Response for AIX: Programming for the ADSI Feature*, SC34-7088
- *WebSphere Voice Response for AIX: Programming for the Signaling Interface*, SC34-7089
- *WebSphere Voice Response for AIX: Voice over IP using Session Initiation Protocol*, GC34-7093
- *WebSphere Voice Response for AIX: Using the CCXML Browser*, SC34-7092
- *WebSphere Voice Response for AIX: VoiceXML Programmer's Guide*, SC34-7117

IBM hardware for use with WebSphere Voice Response

- *IBM Quad Digital Trunk Telephony PCI Adapter (DTTA): Installation and User's Guide*, part number 00P3119 (DTTA card)

WebSphere Voice Response related products

WebSphere Voice Server

The documentation for Version 5.1 of WebSphere Voice Server is provided in the form of an HTML-based information center, and can be found at:

<http://publib.boulder.ibm.com/pvc/wvs/51/en/infocenter/index.html>

Unified Messaging for WebSphere Voice Response

- *Unified Messaging: General Information and Planning*, GC34-6398
- *Unified Messaging: Subscriber's Guide (Types 0, 1, 2, 3, 4 and 9)*, SC34-6403
- *Unified Messaging: Subscriber's Guide (Types 5, 6, 7 and 8)*, SC34-6400
- *Unified Messaging: Administrator's Guide*, SC34-6399
- *Unified Messaging: Voice Interface*, GC34-6401
- *Unified Messaging: Web Services Voicemail API*, SC34-6975

Unified Messaging publications can be found by going to the IBM Pervasive software Web site at <http://www.ibm.com/software/pervasive>, selecting the products link, and then selecting the library link from the Unified Messaging page.

AIX and the IBM pSeries computer

For information on AIX Version 6.1, refer to the AIX V6.1 infocenter

For information on System p5 and BladeCenter computers, refer to the IBM Power hardware infocenter

HACMP

- *HACMP for AIX: HACMP 5.4 Concepts and Facilities*, SC23-4864-09
- *HACMP for AIX: HACMP 5.4 Planning Guide*, SC23-4861-09
- *HACMP for AIX: HACMP 5.4 Installation Guide*, SC23-5209-00
- *HACMP for AIX: HACMP 5.4 Administration Guide*, SC23-4862-09
- *HACMP for AIX: HACMP 5.4 Smart Assist for DB2*, SC23-5179-03
- *HACMP for AIX: HACMP 5.4 Troubleshooting*, SC23-5177-03
- *HACMP for AIX: Enhanced Scalability Installation and Administration Guide, Volume 1*, SC23-4284
- *HACMP for AIX: Enhanced Scalability Installation and Administration Guide, Volume 2*, SC23-4306

For more information on HACMP, refer to the HACMP Library and the AIX V6.1 infocenter.

SS7

- *SS7 Support for WebSphere Voice Response: SS7 User's Guide*, GC34-7090

IBM SS7 Support for WebSphere Voice Response observes the applicable parts of the following specifications for ISUP:

- *CCITT Blue book (1988) Q.701 - Q.707*
- *ITU-T (formerly CCITT) Recommendations Q.700 - Q.716, Volume VI Fascicle VI.7*
- *CCITT Blue book (1988) Q.711 - Q.714*
- *ITU-T White book (1993) Q.711 - Q.714*
- *CCITT Blue book (1988) Q.721 - Q.724*
- *ITU-T (formerly CCITT) Recommendations Q.721 - Q.725, Volume VI Fascicle VI.8*
- *ITU-T White book (1992) Q.730 group*
- *CCITT Blue book (1988) Q.761 - Q.764*
- *ITU-T White book (1992) Q.761 - Q.764*
- *CCITT Blue book (1988) Q.771 - Q.775*
- *ITU-T (formerly CCITT) Recommendations Q.771 - Q.775, Q.791, Volume VI Fascicle VI.9*

ADC

- *ADC NewNet AccessMANAGER™: Installation and Maintenance Manual*

- *ADC NewNet AccessMANAGER™: User Manual*

Integrated Services Digital Network

WebSphere Voice Response ISDN support observes the applicable parts of the following standards for User Side protocol:

Custom ISDN Standards:

- *Northern Telecom DMS/250 Primary Rate Interface NIS A211-4 Release 8, July 1995. (IEC05 level)*
- *Northern Telecom DMS/100 Primary Rate Interface NIS A211-1 Release 7.05, May 1998. (NA007 & RLT)*
- *AT&T 5ESS Switch. ISDN Primary Rate Interface Specification. 5E7 and 5E8 Software Release AT&T 235-900-332. Issue 2.00 December 1991*
- *AT&T 5ESS Switch. ISDN Primary Rate Interface Specification. 5E9 Software Release AT&T 235-900-342. Issue 1.00 November 1993 (National ISDN only)*
- *Lucent 5ESS-2000 Switch ISDN Primary Rate Interface, Interface Specification, 5E9(2) and Later Software Releases, 235-900-342. Issue 5.00 January 1997 (National ISDN only)*
- *AT&T ISDN Primary Rate Specification TR41449 July 1989*
- *AT&T ISDN Primary Rate Specification TR41459 August 1996*

Euro-ISDN

The following documents refer to the specifications required for observing ISDN:

- *TBR4-ISDN; Attachment Requirements For Terminal Equipment To Connect To An ISDN Using ISDN Primary Rate Access, Edition 1, Nov. 95, English*
- *CTR 4 - European Communities Commission Decision 94/796/EC published in the Official Journal of the European Communities L 329, 20 December 94 (ISDN PRA)*

National ISDN

National ISDN is described in the following publications:

- *National ISDN, SR-NWT-002006, Issue 1, August 1991, published by Bellcore*
- *National ISDN-1, SR-NWT-001937, Issue 1, February 1991, published by Bellcore*
- *National ISDN-2, SR-NWT-002120, Issue 1, May 1992, published by Bellcore*

INS Net Service 1500

INS Net Service is described in the following publications:

- *Interface for the INS Net Service Volume 1 (Outline)*, 7th Edition, published by Nippon Telegraph and Telephone Corporation
- *Interface for the INS Net Service Volume 2 (Layer 1 & 2 Specifications)*, 4th Edition, published by Nippon Telegraph and Telephone Corporation
- *Interface for the INS Net Service Volume 3 (Layer 3 Circuit Switching)*, 5th Edition, published by Nippon Telegraph and Telephone Corporation

Bellcore Specifications for ADSI Telephones

The following Bellcore specification documents contain technical details of the requirements for ADSI telephones, and the interface to voice response systems such as WebSphere Voice Response:

- SR-INS-002461: *Customer Premises Equipment Compatibility Considerations for the Analog Display Services Interface*
- TR-NWT-001273: *Generic Requirements for an SPCS to Customer Premises Equipment Data Interface for Analog Display Services*

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