UAS Basics

Universal Audio Server (UAS)

The Nortel Networks Universal Audio Server (UAS) is a server platform that can play voice announcements, collect Dual Tone Multi-Frequency (DTMF) digits, and can support speech recognition, text-to speech synthesis, speaker verification, audio conferences, facsimile, and other media features. The UAS uses a conventional Microsoft Windows-based computing server architecture, industry standard internal busses, standard external physical interfaces, and standard protocols for communication with the network. The UAS is available as a -48 V dc powered, rack mount chassis suitable for deployment in a central office, and as a 100-220 V ac powered configuration suitable for computer room installations.

As an application server supporting audio services, the UAS provides an interface for caller interactive features that require the collection of user input and prompt playback. In this capacity, the UAS supports the following functions:

- plays announcements stored as G.711 encoded mulaw and alaw
- plays a set of announcements to the caller which, for the UAS-IP bearer (not ATM bearer), can be interruptible by Dual Tone MultiFrequency (DTMF) digit entry
- plays an announcement and collects DTMF digits, for the UAS-IP bearer (not ATM bearer)
- plays a particular announcement and collects DTMF digits, potentially looking for a specific DTMF digit response using a specified DTMF digit pattern (specific digits, maximum number of digits, or specific digits that can interrupt the announcement), for the UAS-IP bearer (not ATM bearer)
- plays an announcement that is stored in the runtime database

Hardware and software details

The UAS runs Microsoft Windows 2K Server on a conventional Intel-based computing server platform. The UAS system architecture provides industry standard physical interfaces, standard internal buses, and standard protocols for communication with the network. Standard external physical interfaces supported by the UAS include 10/100 BaseT Ethernet (IP), and SONET OC-3c (ATM – AAL1 and ATM – AAL2).

The UAS system architecture uses an industry-standard compact Peripheral Component Interconnect (cPCI) bus for data transfer and H.110 voice bus for time division multiplexed (TDM) voice transfer between circuit packs. The PCI based cards provide voice capabilities, interfaces, and other special functions for the cPCI bus on each server.

The UAS is highly scalable. Each shelf (chassis) can contain up to two separate UAS nodes, and additional shelves can be added as capacity demands. Redundant hardware is provisioned on an N + 1 basis, thus ensuring that the engineered service capacity will not be degraded due to a single server outage. If the busy hour call volume does not exceed engineered capacity, the UAS blocks an average of one call per 10,000 due to server or service circuit outages.

Hardware redundancy is provided where it is required to ensure high reliability. For example, redundant power supplies and fans are provided since these hardware elements have a lower MTBF. Dual network interfaces prevent a failed router or bridge from taking a UAS out of service.

The following is a summary of the capabilities, configuration, and remote operation of the UAS:

- basic capability components:
  - Motorola CPX8216T NEBS compatible chassis (also known as the "SAM16" chassis)
  - dc (rectifier required where the source is ac)
  - Windows2K Server operating system with Service Pack 2 for access to robust personality card drivers
  - cPCI data bus
  - H.110 voice bus
- configuration:
  - Dual Ethernet Network Interface Card
  - redundant power and fans
  - scalable by adding UAS to a system with no maximum limit
  - can locate the UAS where required

  - N + 1 chassis redundancy
  - MGCP or H.248 control for each IPConnect SoftSwitch implementation
- remote operation:
  - all UAS nodes are connected through a Keyboard Video Mouse (KVM) switch
  - Rose electronics, UPB-8U, 8 port peripheral Switchbox is the supported KVM switch

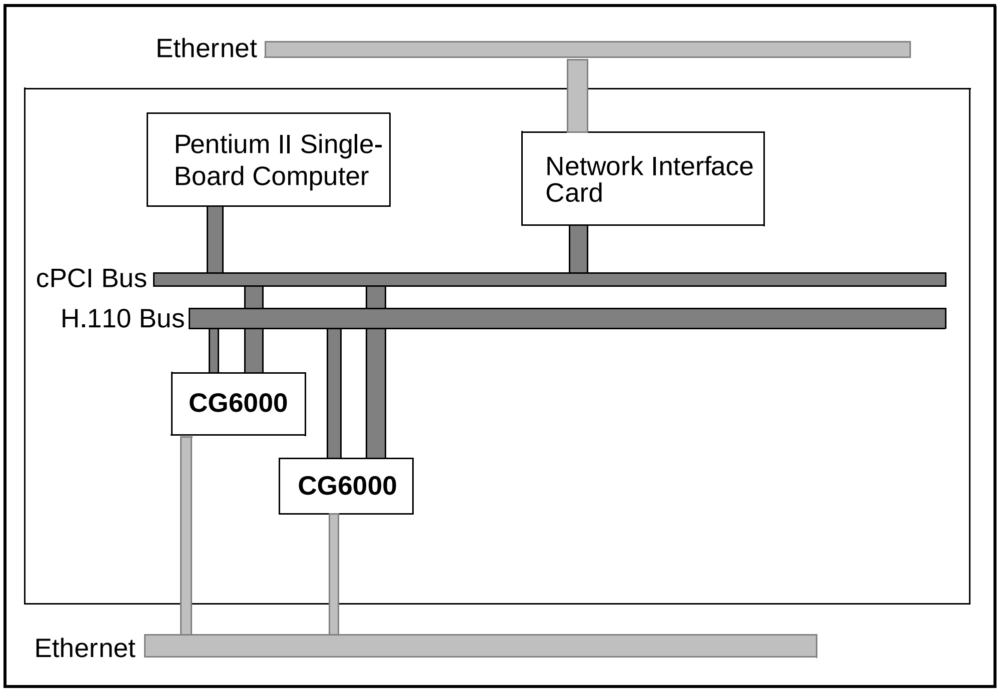
UAS VoIP Configuration

In a VoIP-based configuration, the UAS is provisioned with the following basic components:

- compact peripheral component interconnect (cPCI) Natural Microsystems CG6000 cards, which contain 16 dual-core DSPs each, providing 32000 MIPs of processing power for each card. The CG6000 card provides the network interfaces, the translation of packet audio from the IP network to Time Division Multiplex (TDM) audio, vocoding, interactive voice response (IVR) capabilities, conferencing capabilities, and BCT capabilities. The UAS can be configured with a maximum of six CG6000 cPCI cards.
- H.110 bus cable, which provides the physical, electrical, and timing requirements for a bus that is capable of carrying 4096 simplex or 2048 duplex voice conversations at a rate of 8000 8-bit samples per second per data stream.

The following figure illustrates the UAS VoIP hardware architecture.

UAS VoIP hardware architecture



The CPX 8216T chassis packfill, when the UAS is configured in a VoIP network, is shown in the following illustration.

VoIP UAS CPX8216T chassis packfill

	Chassis Front	Chassis Rear
1	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
2	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
3	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
4	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
5	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
6	(see Note 2)	(see Note 2)
7	CPV5370 Processor 700 MHz (see Note 1)	CPV5370 Transition Module (see Note 1)
8	Hot Swap Controller	
9	CPV5370 Processor 700 MHz (see Note 1)	CPV5370 Transition Module (see Note 1)
10	Hot Swap Controller	
11	(see Note 2)	(see Note 2)
12	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
13	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
14	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
15	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)
16	NMS CG6000C (main board)	NMS CG6000C (rear I/O card)

Note 1: The system processor may be either a CPV5350 500MHz processor or a CPV5370 700MHz processor.

Note 2: For systems configured with the CPV5350 processor, these slots contain the CPV8540 SCSI Controller (front and rear modules).

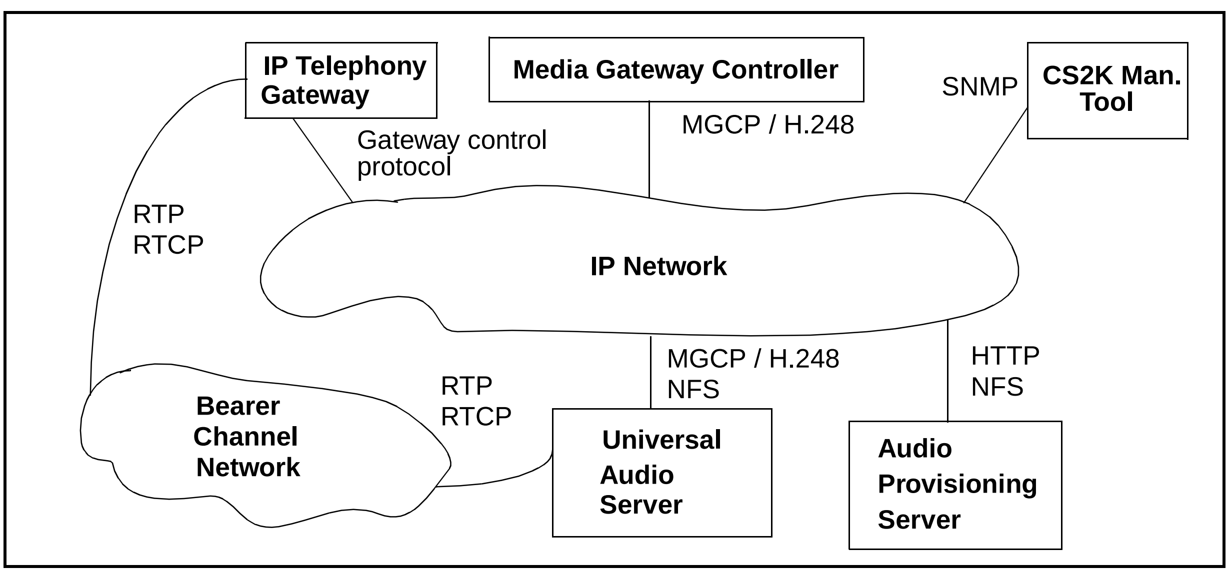
In systems configured with the CPV5370 processor, these slots can be either empty or configured with CG6000c front and rear cards.

The following control protocols are supported by the UAS when it is configured in a VoIP network:

- media gateway control protocol (MGCP) and H.248 protocol, which direct a telephony media gateway from a call agent running on a media gateway controller. The UAS is a specialized media gateway. The MGCP Messaging Interface function translates MGCP messages sent from the call agent to the UAS and builds MGCP messages to be sent from the UAS to the call agent.
- simple network management protocol (SNMP), to implement fault management, configuration management, and performance management
- real time protocol/real time control protocol (RTP/RTCP), for transmission of audio on the bearer channel network
- network file system protocol (NFS), which supports the transfer of audio files stored on the APS to the UAS

The use of these protocols when a UAS is configured in a VoIP network is shown in the following illustration.

Universal Audio Server in a VoIP network



In a VoIP-based UAS, the number of available ports per CG6000 card depends on the capabilities, including interactive voice response (IVR), conferencing, and BCT, that the card supports. If the card supports only IVR, the number of available ports is 80. If the card supports both IVR and conferencing, the number of available ports is 66. If the card supports BCT, the number of available ports is 90. The total number of available ports in a fully-configured VoIP UAS with conferencing disabled is 480 (80 x 6 cards). The total number of available ports in a VoIP UAS with conferencing enabled is 396 (66 x 6 cards).

UAS VoATM Configuration

In an ATM-based configuration, the UAS is provisioned with the following components:

- Natural Microsystems PA200 cards, which provide announcement capability to the ATM-AAL1 or AAL2 network. In support of AAL2 connections, up to 247 channel IDs (CID) can be assigned to each provisioned virtual channel (VC) for simultaneous communication between common endpoints. The range of CIDs is from 9 through 255, with CIDs assigned on a per-VC basis.

  The ATM interface is provided through a rear transition module of the PA200 card set, by a single mode fiber (SMF) using SC connectors. The interface speed is 155 Mb per second (OC-3c / STM-1).

  The card is designed to support 1000 simultaneous full duplex DS0s. AAL1 VCs are configured to be switched virtual channels (SVC); AAL2 VCs support both SVCs and permanent virtual channels (PVC).

  One PA200 card set (front and rear modules) can be configured in each domain.

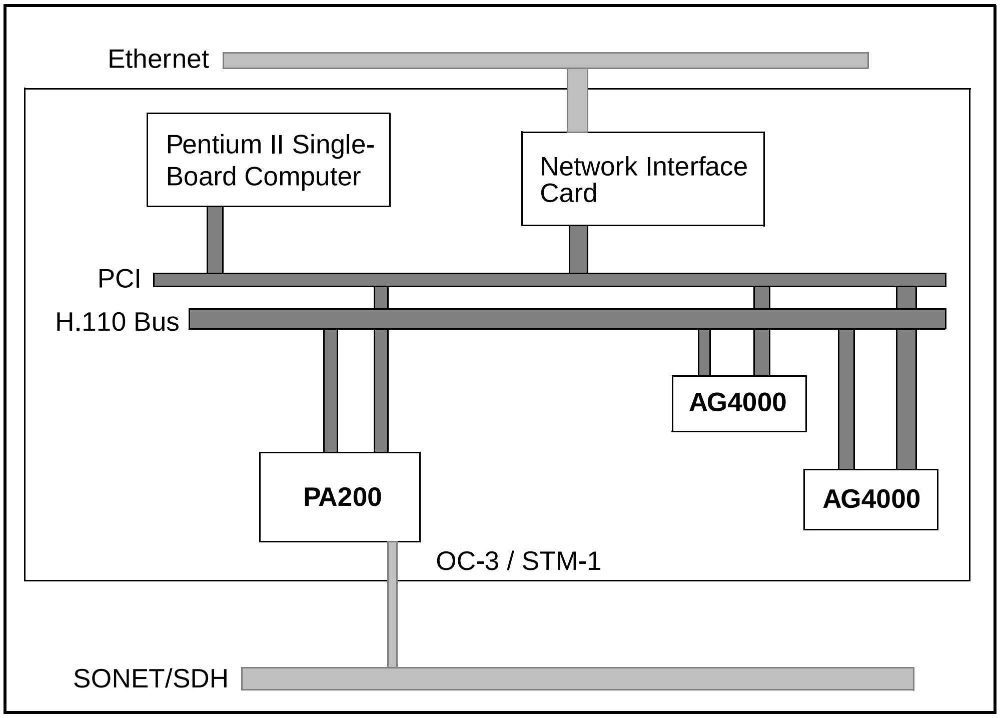
  Note 1: Support for the NMS S007/BX4000C ATM card has been discontinued starting with release UAS07.

  Note 2: While the PA200 provides support for two ATM port interfaces, only one of these ATM port interfaces can be used to support AAL1 SVC-based applications.

- AG4000 cPCI cards, which contain 16 single-core Digital Signal Processors (DSP) each, providing 1600 MIPs of processing power for each card. The AG4000c card provides IVR services and vocoding functionality. The card also provides a switching matrix capable of transmitting and receiving audio from the H.110 bus. AG4000-T1 cards provide announcements. International UAS hardware configurations require AG4000-E1 cards. The UAS can be configured with a maximum of five AG4000/40 cPCI cards.

The following illustration shows the UAS VoATM hardware architecture.

UAS VoATM hardware architecture



The CPX 8216T chassis packfill, when the UAS is configured in a VoATM network, is shown in the following illustration.

VoATM UAS CPX8216T chassis packfill

	Chassis Front	Chassis Rear
1	NMS AG4000 (main board)	
2	NMS AG4000 (main board)	
3	NMS AG4000 (main board)	(AG4000 rear I/O card - see Note)
4	NMS AG4000 (main board)	
5	NMS AG4000 (main board)	
6	NMS PA200 (main board)	NMS R200 (rear I/O card)
7	CPV5370 Processor 700 MHz	CPV5370 Transition Module
8	Hot Swap Controller	
9	CPV5370 Processor 700 MHz	CPV5370 Transition Module
10	Hot Swap Controller	
11	NMS PA200 (main board)	NMS R200 (rear IO card)
12	NMS AG4000 (main board)	
13	NMS AG4000 (main board)	
14	NMS AG4000 (main board)	(AG4000 rear I/O card - see Note)
15	NMS AG4000 (main board)	
16	NMS AG4000 (main board)	

Note: An AG4000 transition module (also called "rear I/O card") is installed only for the following reasons:

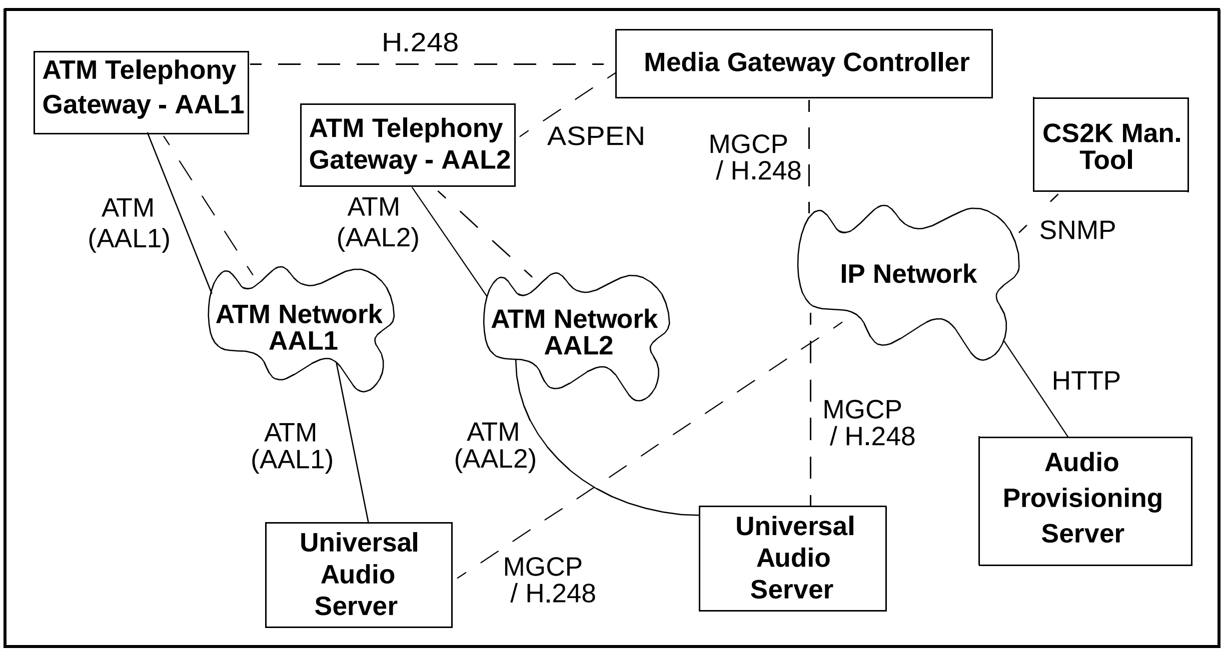
- when a trunk interface for synchronizing clocking of the internal H.110 bus to the PSTN is required
- when test trunk support is required

The UAS VoATM network uses the following control protocols:

- asynchronous transfer mode adaptation layer 2 (ATM-AAL2), which enables the ATM telephony gateway to connect with the UAS
- ASPEN protocol, which enables the gateway controller to send signaling messages to the ATM telephony gateway (PVG)
- media gateway control protocol (MGCP) and H.248 protocol, which direct a telephony media gateway from a call agent running on a media gateway controller. The UAS is a specialized media gateway. The MGCP Messaging Interface function translates MGCP messages sent from the call agent to the UAS and builds MGCP messages to be sent from the UAS to the call agent.
- simple network management protocol (SNMP), to implement fault management, configuration management, and performance management

The use of these protocols when a UAS is configured in a VoATM network is shown in the following illustration.

UAS in a VoATM network



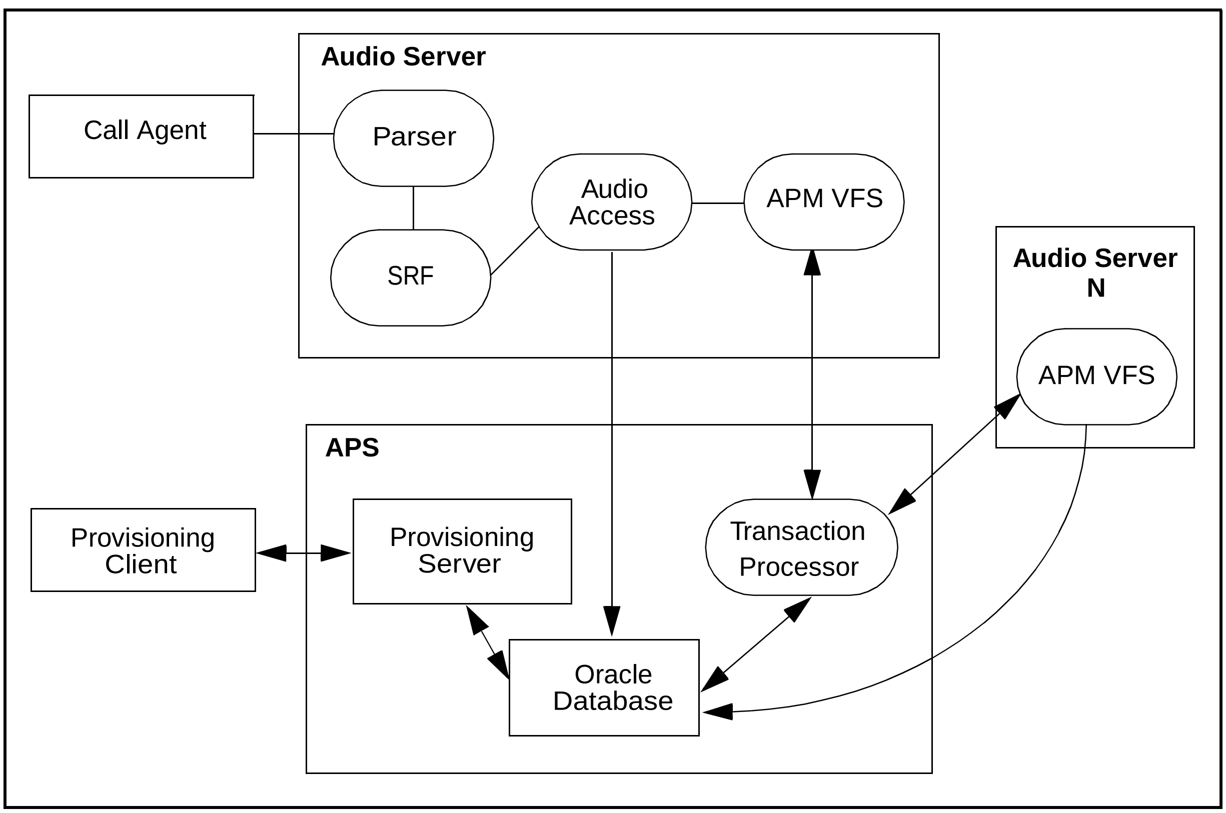
In an ATM-based configuration, up to 128 ports are available per ATM bearer card. Thus, a UAS can support a maximum of 512 ports in an ATM-based configuration.

Audio Access

One of the primary roles of the UAS is to play audio as directed by a Call Agent. The Call Agent can specify that a single piece of audio be played, or it can specify a more complex audio assemblage be played, such as the current date in Spanish. The "audio access" feature provides a simple and centralized point of access to the UAS audio repository.

The basic relationship between the parts of the audio access feature is shown in the following illustration. In its basic operation, the Call Agent sends messages to the Audio Server with instruction to play an audio segment or to provide conferencing resources. The Parser in the Audio Server interprets the messages sent by the Call Agent and, through small, dedicated programs known as "SRFs (Service Resource Functions)," calls on other programs within the audio server to perform the work requested by the Call Agent.

Audio access



If audio is to be played, the SRFs make calls on the audio access feature to retrieve audio from the audio repository, APM VFS. If audio is to be recorded, the APM VFS forwards the new audio to the APS Transaction Processor. The Transaction Processor adds the audio to the Oracle database on the APS platform and then forwards the new audio to other audio server nodes in the system that need to use it.

Audio is added to the system through the Provisioning Client. The new audio is sent from the Provisioning Client to the Provisioning Server, which, in turn, forwards the audio to the Oracle database in the APS, known as the "Interactive Provisioning System (IPS)" database. After the audio has been added to the database, the Transaction Processor can forward it on to other audio server nodes in the system, where the new audio is added to the nodes' APM VFS. Audio is normally automatically distributed by the Transaction Process to other audio server nodes in the system on a scheduled basis, typically every hour.

Audio Provisioning Server (APS)
The G.711 encoded mulaw and alaw announcements that the UAS plays are provisioned using the Audio Provisioning Server (APS). The APS is built on the Interactive Provisioning System (IPS) platform, which provides a centralized location and Web-based Generic User Interface (GUI) (through Internet Explorer 5.0 or higher, or through Netscape Navigator 4.5 or higher) for uploading announcement files from a client machine to the APS, where the files can be prepared for use by a gateway.

APS hardware The APS server is configured on a server that also hosts the CS 2000 Management Tools. For information about the server on which the APS resides, see the CS 2000 Management Tools document, NN10020-111, "CS 2000 Management Tools Basics" in your Succession Solution document suite.

APS software The following list shows the layered software architecture elements of the APS:

- Java Applet on client workstation
- Remote database proxy on client workstation
- HTTP/ftp communication over Inter/Intra-net between APS and Client(s)
- Web server (Database Servlets)
- IPS base (Audits, Permissions)
- SSPFS (Installs, Job Scheduler, System Admin)
- DMP (MBrow, Apache, JDBC)
- Database (Oracle 8.1.7)
- Operating System (Solaris 2.8. JRE 1.4.1)

The *applets* provide the client interface to the database. The applets gather information from the APS GUI that is sent to the servlet in the form of a request. The applets communicate with the database servlets through the network using the hypertext transport protocol (HTTP).

For the audio management GUI, the *servlets* act as a request and response handler between the applets and the database server application. The servlets use the remote method invocation (RMI) capabilities provided by the Java class libraries to communicate with the database server application. For the administrative GUI, the servlets communicate directly with the IPS database using the Java Database Connectivity (JDBC) interface.

The *database server application* communicates with the IPS database using the JDBC interface. The database server application translates servlet requests into database operations. It provides a response back to the servlet, which includes a success or failure of the database's ability to process the request.

The *Interactive Provisioning System (IPS)* relational database stores the actual physical audio segments and the properties or the relationships that have been defined for physical segments, packages, sequences, sets and variables. It interacts with the UAS runtime database hourly to provide update audio management data.

Note: Trademark and copyright acknowledgement information for products listed above can be found in the Customer Letter for this Succession Solution.

Conference service

The audio server conference component is a specialized function that manages creation and modification of conferences on the audio server and manages addition and removal of conferees from these conferences.

For systems connecting to an IP network, conferencing service is provided through the Natural MicroSystems CG6000C card, in four possible configurations:

- Conference Spanning capability enabled, and IVR capability disabled. In this configuration, 128 ports are provided through one conference resource. The large conference resource can maintain either 64 simultaneous conferences consisting of two conferees each, a single conference consisting of 128 conferees, or any combination of conferees that adds up to 128.
- Conference Spanning capability disabled, and IVR capability disabled. In this configuration, 128 ports are provided through four

conference resources, each of which can accommodate 32 conferees.

- Conference Spanning capability enabled, and IVR capability enabled. In this configuration, 96 ports are provided through one conference resource, per card.
- Conference Spanning capability disabled, and IVR capability enabled. In this configuration, each CG6000 card provides 3 conference resources. Each conference resource provides 32 ports.

The benefit of using the Conference Spanning capability is that one resource can accommodate a larger than normal pool of potential conferees - up to 128 as opposed to 32.

Although up to 128 ports can be supported by each CG6000C, there is an effective limit of 120 conferencing ports, in a conferencing-only configuration, and 96 conferencing ports, in a configuration that supports both conferencing service and IVR service. The actual conference capacity of the system is also determined by the number of CG6000C cards provisioned and the amount of system play and monitor port usage.