Ascend

PRODUCT OVERVIEW

MultiVoice for the MAX

Release 1.0

ASCEND

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1. Introduction

Product Overview

In today's communications networks, real time voice information is transmitted via the Public Switched Telephone Network (PSTN). Voice is transmitted on the PSTN using circuit switched technology where each call is provided with dedicated bandwidth, usually 64 Kbps. In most public and private enterprises, computer generated information is transmitted over a separately maintained packet data network. In data networks bandwidth is usually shared between different computing endpoints.

Recently new technology and international standards have been introduced which allow enterprise customers and network service providers to introduce real time communications (voice) to traditional packet networks that utilize the Internet Protocol (IP). For example, Voice-over-IP (VoIP) technology and the ITU-T H.323 standards define a framework for the transmission of real time voice over IP-based packet networks such as the public Internet, private intranets, and Extranets. The implementations of these standards allow for products from different vendors to interoperate on the same network.

The Ascend MultiVoice™ for the MAX™ consists of a set of hardware and software components that allow Network Service Providers (NSPs) or Enterprise customers to add real-time voice transport services to their existing IP backbone network. The components of the Ascend MultiVoice solution for the MAX product family is illustrated in the following figure:

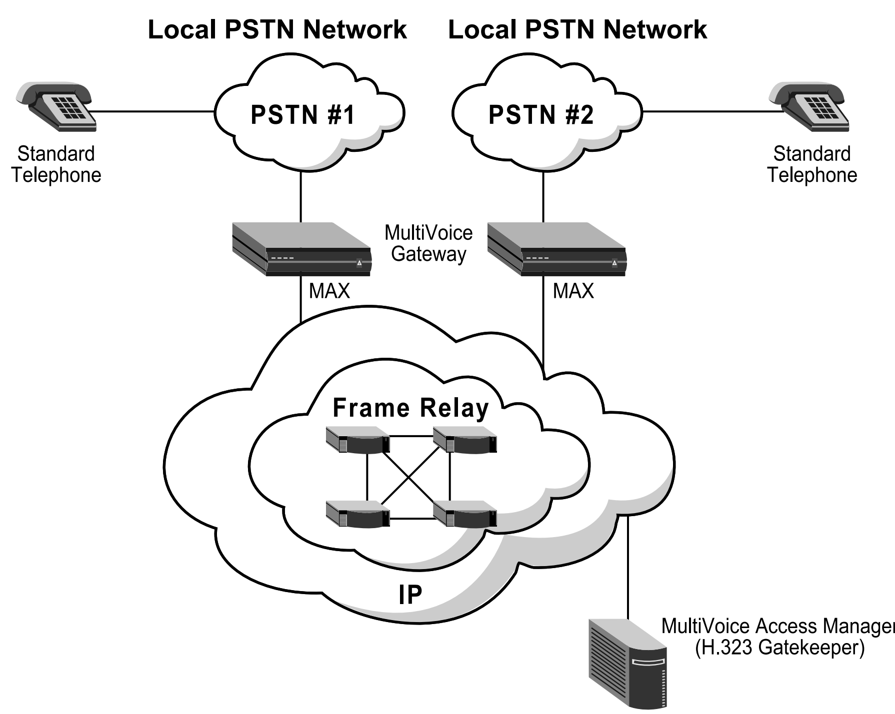

Figure 1 – Long-haul VoIP Network

The individual MultiVoice components work as a single network to support end-to-end calling between telephone endpoints on different PSTNs. The MultiVoice components for the MAX are as follows:

MultiVoice Gateway for the MAX

The MultiVoice Gateway for the MAX provides the interface between the PSTN and the IP-based packet network. It is the point at which traditional telephone calls "*hop-on*" and "*hop-off*" the packet telephone network. The MultiVoice Gateway performs the following functions:

- Terminates the native PSTN network interfaces (e.g. T1, PRI, E1, and BRI)
- Support various voice codecs which provide different levels of voice compression to reduce data throughput requirements on the packet network
- Supports DTMF tone detection/generation in order to emulate PSTN phone networks
- Supports the ITU-T H.323 protocol stack for Phone-to-Phone conversations over the IP network
- Works in conjunction with the MultiVoice Access Manager to establish and terminate VoIP calls

The MultiVoice Gateway consists of a MAX hardware platform, MultiVoice DSP slot cards for the MAX, and the MultiVoice Gateway software. For more information on the MultiVoice Gateway feature see "*Gateway Feature Description*".

MultiVoice Access Manager

The MultiVoice Access Manager[1] for the MAX provides network routing functions for connecting voice calls on the IP network. The Access Manager performs the following functions:

- Manages the H.323 *zone* for a set of MultiVoice Gateways. A "zone" is defined as a set of H.323 Gateways that are under the control of a specific Access Manager.
- Provides address translation from standard national and international telephone numbers (E.164 numbers or private dialing plan numbers) to IP address and vice-versa.
- Supports user (authentication) and gateway registration.

The MultiVoice Access Manager consists of the following software components: a Access Manager application, gateway and user database files, configuration files, web pages, and a Common Gateway Interface (CGI) adapter. For more information on the MultiVoice Access Manager feature see "MultiVoice Access Manager Feature Description".

The Release 1.0 MultiVoice platform for the MAX is primarily new software that works on the existing MAX 6000 and MAX 400X (upgrade only) WAN access switch populated with MultiVoice DSP slot cards. The R1.0 product will support phone-to-phone voice communications between two or more MultiVoice Gateways managed by a single MultiVoice Access Manager.

Target Customers

The MultiVoice R1.0 product for the MAX is ideal for existing Network Service Providers, i.e. ISPs, CLECs, and carriers, who want to offer public telephone service across managed IP networks in a cost-effective manner. It is also an excellent vehicle for enterprise customers to leverage their existing, private packet networks to support their basic voice telecommunications needs. For Network Service Providers (NSPs) the MultiVoice product for the MAX provides a high performance, high quality, and scalable Voice-over-IP (VoIP) platform which can allow NSPs to quickly and cost effectively add voice telephone services to their existing network.

For typical ISPs, the MultiVoice product means new service offerings that leverage their existing packet network and customer base. It also means that they can offer a new service that incorporates transaction-based billing that scales with network usage. Since the typical consumer-based, telephone call pricing model is a *cents-per-minute* model, ISPs

[1] The MultiVoice Access Manager is also known as a "*Gatekeeper*" in ITU-T H.323 terminology.

can now offer a service where the end-user pays for network bandwidth usage. This is a more lucrative billing model than the current monthly flat-fee, unlimited access pricing models used for Internet access services.

For Competitive Local Exchange Carriers (CLECs) and carriers, the Ascend product allows them to build a *single* backbone network that can provide both voice and data services over the same network. Carriers who already offer standard PSTN-based telephone service can use the MultiVoice product to offer customers a two-tier pricing structure. One price for circuit switched calls and another price for packet switched calls in order to compete with CLECs or ISPs that are offering packet switched telephone service.

For corporations and business, the Ascend Voice product is an excellent choice for private network deployment of VoIP services. The MultiVoice Gateway can be used to support a variety of applications within the enterprise on private voice networks and behind PBXs. For more information on these applications see the “*Applications*” sections of this document.

Ascend customers with large, managed backbone packet networks are ideal customers for the MultiVoice solution. These customers can provide voice services on a “*Port Wholesale*” basis to enterprise customers and to smaller ISPs who can then retail the service to end-users.

Complete Solution Including QoS

The MultiVoice product for the MAX is an excellent solution for VoIP deployment because it is highly scalable, integrated and based on international standards. Although the MultiVoice product can be used to support VoIP services over an unmanaged IP network such as the Internet, in order to provide end-users with reliable and high-quality voice it is recommended that the MultiVoice product be deployed on managed IP-networks. Since there is currently no Quality of Service (QoS) guarantees on unmanaged IP networks like the Internet, voice quality can suffer from delays or congestion on the network. This may result in inconsistent voice quality between calls or even within a call.

A managed public or private IP-network based on Ascend’s Core Switching products can provide QoS that can improve the overall reliability and quality of packet voice calls as compared to today’s Internet Telephony products. In this case, end-to-end QoS can be maintained resulting in higher quality packet voice calls. The QoS features of Ascend’s IP Navigator product will provide guaranteed bandwidth and delay characteristics across a backbone Frame Relay and/or ATM network. This guaranteed QoS is know as “Absolute QoS” and results in an ideal packet channel for transmitting real-time information such as voice. See the IP Navigator Product Marketing Guide for more information on the Absolute QoS feature.

The MultiVoice for the MAX platform provides a much more compact, integrated, and scalable design than other VoIP gateways from vendors such as VocalTec, NetSpeak, Vienna Systems, and Cisco. The current VocalTec, NetSpeak, and Vienna gateways are based on a standard PC architecture with add-in PSTN interface cards (T1 cards) and DSP cards. The PC-based solutions are:

Applications

A MAX-based MultiVoice network can be implemented to support many different applications in both public networks and in the private enterprise. The following applications will be discussed in this section:

- Basic Public Long Distance Service
- Local “800” Service for Customer Service Applications
- Point-to-Point PBX Trunk Extension
- PBX Trunk Intraflow over Routed IP Networks

Basic Public Long Distance Service

This application is intended for ISPs or CLECs that have an extensive domestic or international packet backbone network. It is assumed that the backbone network is a managed IP network where a QoS can be maintained for voice calls that are routed on the IP network. The following figure illustrates a Public IP Network that is being used to route both data and voice traffic:

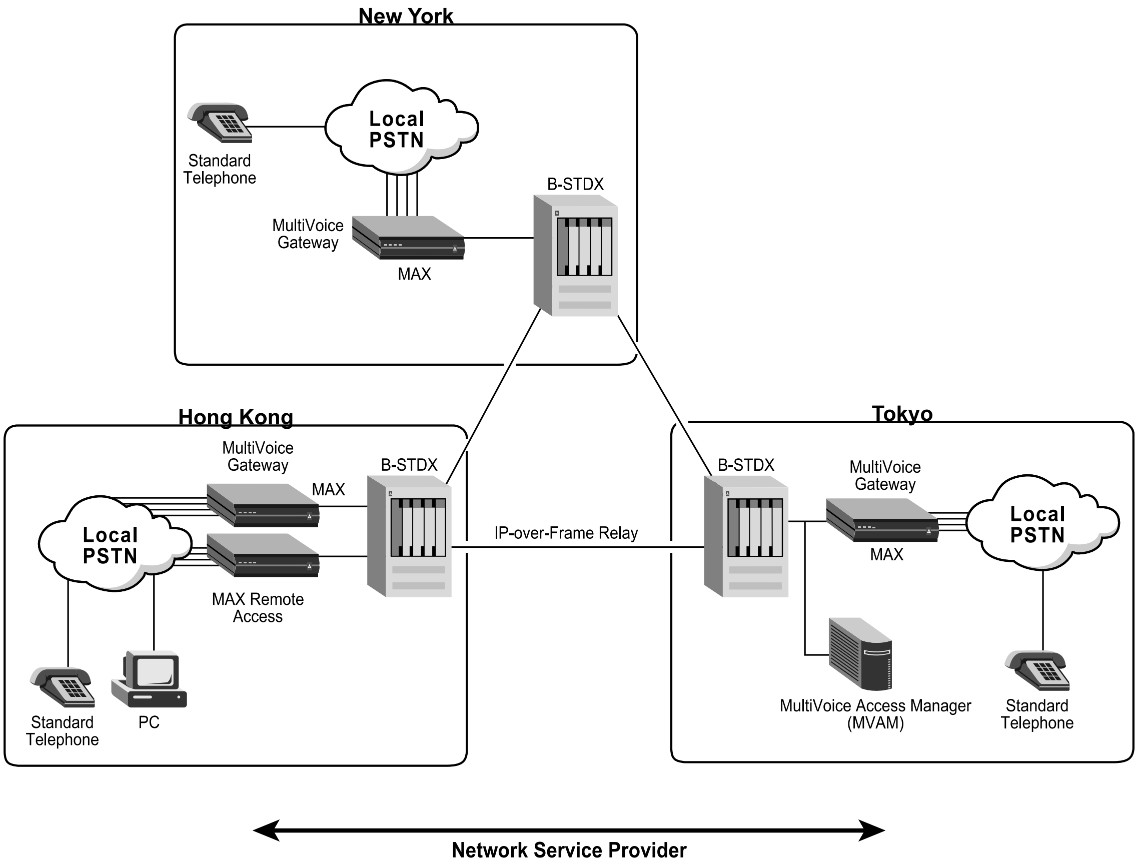

Figure 2 – Public IP Network with data and voice

In this network configuration, the Network Service Provider only builds a single network that can carry both voice and data traffic. The NSP utilizes interfaces to a local, public PSTN network to deliver the last "leg" of a voice call to an end-user. The advantages of this configuration are as follows:

- *Single Network Design* – utilizing a managed IP-backbone network using Ascend's products for the core network and the MultiVoice for the MAX platform, the NSP can build a single data network which can be used to deliver multiple services including long distance voice calling. This avoids the need to develop separate voice and data networks to offer the same services.

- *Billing that Scales with Network Usage* – since voice calls are traditionally billed on a per-minute basis, addition of a voice service on a traditional data network can provide a billing scenario that scales with the usage of network bandwidth. This provides ISPs with a billing model that is usage-per-minute based as opposed to a single flat monthly fee for unlimited access.

- *New Network Service Revenues* – for traditional NSPs that provide data only services, adding voice services via the MultiVoice platform allows them to generate new revenue from the existing network infrastructure.

Local "800" Service for Customer Service Application

The Local "800" Service application allows ISPs or other network service providers to lower the cost of operation for customer service for their own operations or as a service to other enterprise customers. Since ISPs already have Point-of-Presence (POP) locations for their existing internet services, adding MultiVoice Gateways at those POP locations allow the ISP to publish a local customer service telephone number associated with each POP. This circumvents the long distance calling costs associated with traditional 800-based customer service operation. This application is illustrated in the following figure:

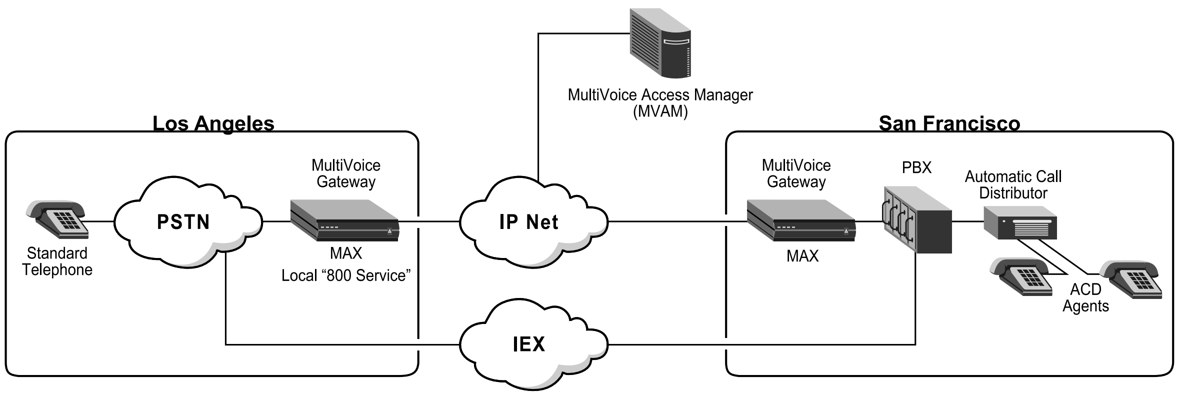

Figure 3 – Local 800 Service

In the case of traditional 800-based customer service, the caller dials a toll-free 800/888 number that is routed by the PSTN to the centrally located customer service center. The customer service center usually consists of an Automatic Call Distributor (ACD) with agents that are trained to provide customer service. Since the call is toll free the service center incurs the long distance charges for the call made by the end-user.

In the case of a Local "800" service network configuration, the end-user is provided with a local telephone number and extension that they dial to get to customer service agent. This local number is located in the same POP as the modem number used for access to basic Internet services. The PSTN call is terminated in a MultiVoice Gateway. The Gateway then extends the voice call across the ISPs packet network to another Gateway that is located at the Customer Service Center. Once the call hop-off the far-end Gateway, it is routed by the PBX/ACD to an agent who answers the call and provides customer service.

Point-to-Point PBX Trunk Extension

Enterprise customers, who want to use their existing packet data network to carry voice traffic between their PBXs, can use the MultiVoice Gateway for the MAX to interconnect different locations. In this network configuration, the MultiVoice Gateways are used in a point-to-point configuration between two PBXs at different locations. This is illustrated in the following figure:

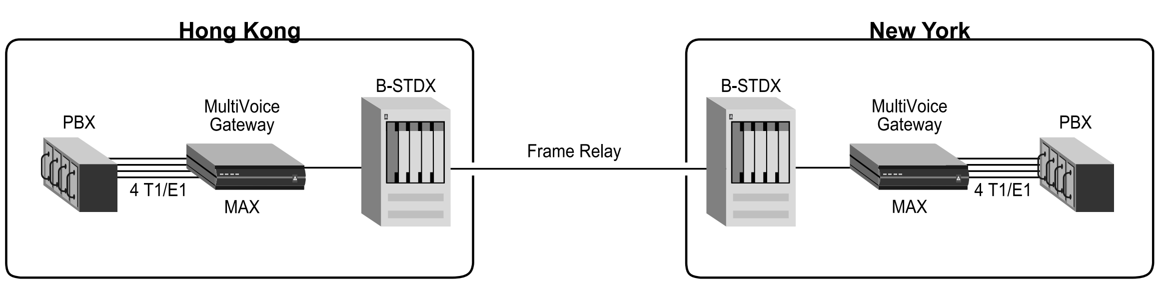

Figure 4 – MultiVoice Gateways in a point-to-point configuration

In this configuration the core B-STDX network used for data applications can also be used to carry packet voice along with data. IP Navigator within the B-STDX network can also be used to provide better QoS in order to offset delay variations and enhance voice delivery and quality. The two MultiVoice Gateways are essentially "*nailed-up*" across the packet network.

PBX Trunk Intraflow over Routed IP Networks

An enterprise customer with multiple PBX locations and a managed IP network can use the MultiVoice for the MAX platform to provide alternate routes for voice calls across their private network. This is illustrated in the following figure:

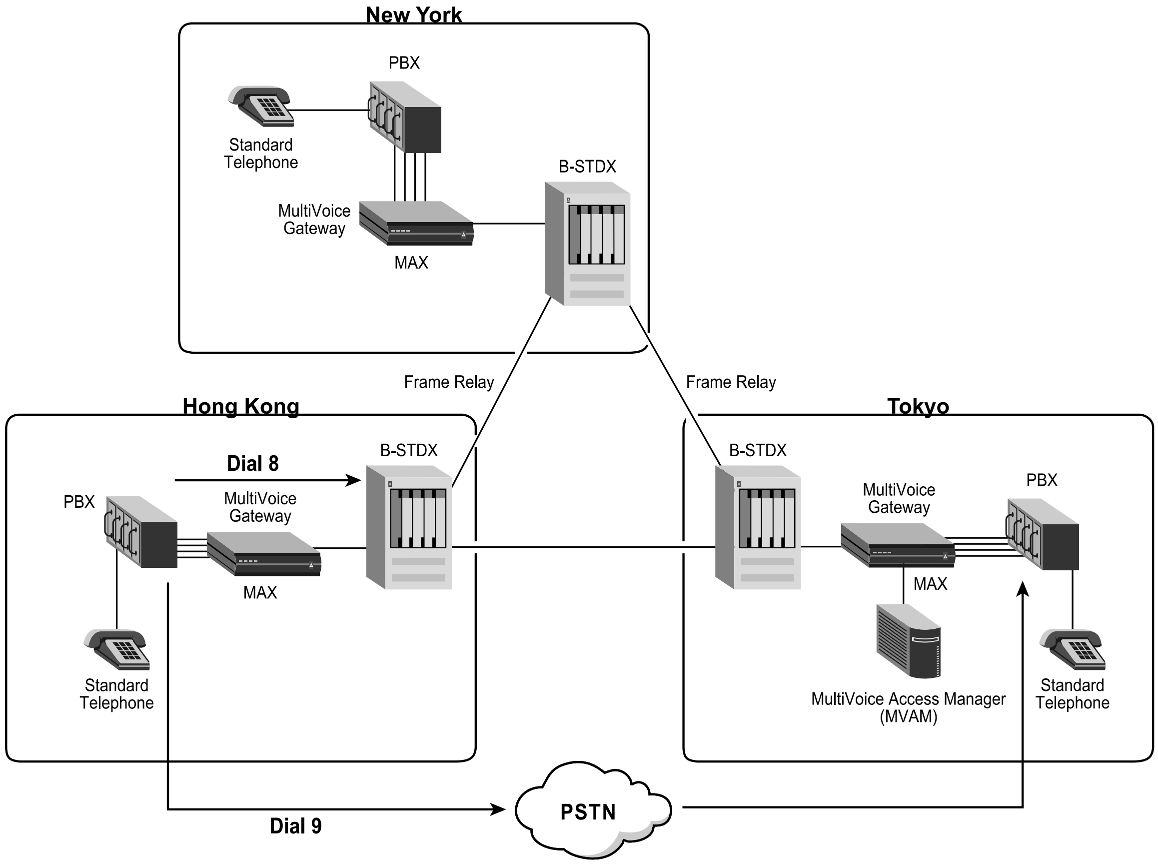

Figure 5 – Ascend offers alternate routes for voice calls.

In this configuration the caller can accomplish the following:

- Use the traditional "Dial 9" sequence to access the traditional PSTN network
- Use a "Dial 8" sequence to access the MultiVoice Gateway and establish a packet telephony call
  - The Gateway answers the call and provides dial tone using the Voice VPN Feature (see "*Gateway Feature Description*" for more information)
  - The user enters the extension of the far-end destination
  - The MultiVoice Access Manager routes the call across the IP network to the remote Gateway and rings the remote caller using the PBX

This configuration can also be used to Intraflow calls between the PSTN and the MultiVoice network. Intraflow occurs when all trunks between two PBXs are busy and new calls flow to an alternative network. The Intraflow application reduces the number of inter-PBX trunks need to handle busy-hour calling rates. The PBX can be configured to automatically route calls via the MultiVoice network when all outgoing trunks are busy as the case may be during a busy hour period.

Target Beta Sites

The initial MultiVoice Beta trials for the MAX are being targeted at large ISPs, CLECs, or carriers which currently have a managed IP-network and can "Port Wholesale" their voice services to smaller ISPs or enterprise customers on a retail basis. Enterprise customers will be also be targeted in order to test the enterprise features of the MultiVoice for the MAX platform.

2. Software Features

The MultiVoice for the MAX Release 1.0 product supports the following features:

MultiVoice Gateway R1.0 for the MAX

- Phone-to-Phone H.323 Operations
- Telephony WAN Interfaces
- Packet Network Interfaces
- Voice Codec Support
- Voice VPN Support
- Hybrid-Line Echo Cancellation Support
- DTMF Detection and Generation

MultiVoice Access Manager R1.0 for the MAX

- Microsoft Windows NT v4.0 Support
- ITU-T H.323 Compliant Gatekeeper Implementation
- Phone-to-IP Address Translation
- Web-based Administration Interface
- PIN-based User Authentication
- Voice VPN Support
- Telephone Number Aliases
- Call Accounting Records (CDR)
- Gateway and User Database Support
- Third-Party Billing System Support

Gateway Feature Description

The MultiVoice Gateway for the MAX provides the interface between the PSTN and the IP-based packet network. It is the point at which traditional, circuit switched telephone calls "*hop-on*" and "*hop-off*" the MultiVoice packet telephone network. The following sections provide a more detailed description of the major features supported by the MultiVoice Gateway Release 1.0.

Phone-to-Phone H.323 Operations

The MultiVoice Gateway provides all the H.323 terminal and protocol functions required by the ITU-T standard. It essentially provides a "proxy" H.323 terminal for every PSTN connection that is established via the Gateway. The H.323 voice terminal proxy supported on the MultiVoice Gateway consists of the following components:

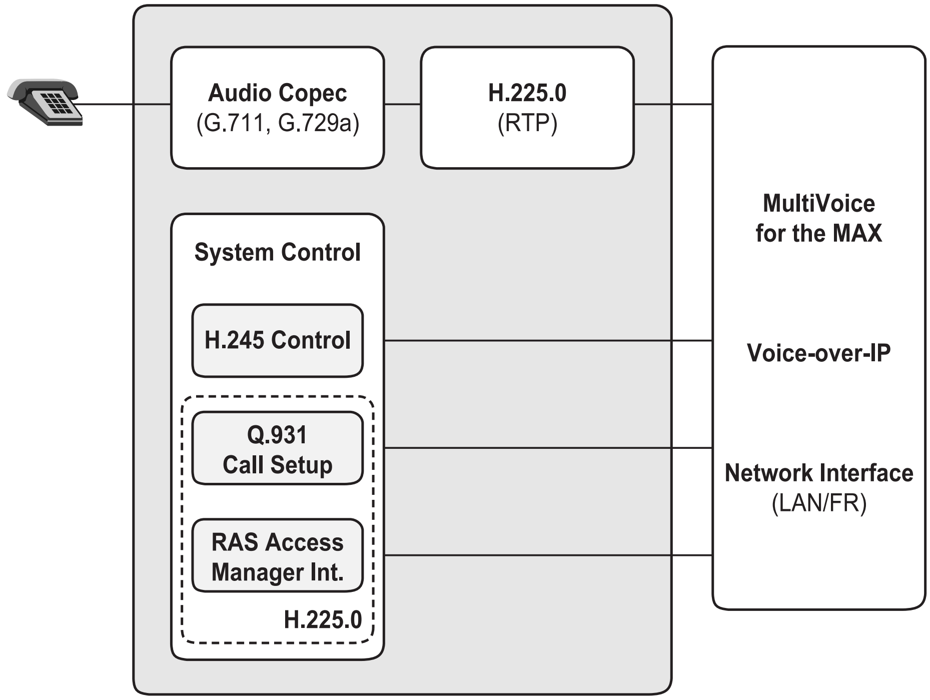

Figure 6 – MultiVoice Gateway H.323 Implementation

- *Voice Codec* – a codec defines the type of voice compression and coding that will be used within a call. The R1.0 MultiVoice Gateway for the MAX supports G.711 and G.729 (A) voice coding. See "*Voice Codec Support*" for more information.
- *H.225.0 (RTP)* – Real-Time Protocol or RTP is a protocol used to insure proper sequencing of the audio packets being transmitted across the IP network.
- *H.245 Control* – the H.245 control protocol is used to negotiate channel usage/bandwidth and capabilities. This protocol exists between the two H.323 proxy terminals created when a call is established between two Gateways in a MultiVoice network.
- *H.225.0 Q.931-derived Call Setup* – this protocol is used between the two Gateways to establish and terminate a voice call across the IP network.
- *RAS Interface* – Registration, Admission, and Status or RAS is the protocol used between the MultiVoice Gateway and the MultiVoice Access Manager to do registration, admission, and status. This protocol is also used to provide user authentication when accessing a MultiVoice network.

All these components are defined by the H.323 standard and are implemented in the MultiVoice Gateway for the MAX.

Telephony WAN Interfaces

The MultiVoice Gateway for the MAX provides several interfaces for connecting to the Public Switched Telephone Network (PSTN). All phone-to-phone VoIP calls originate/terminate on the Telephony WAN Interfaces on a MultiVoice Gateway. The R1.0 MultiVoice Gateway software will support the following Telephony WAN signaling interfaces:

- T1
- T1/PRI
- E1
- E1/PRI
- BRI

For more information on the number and configurations of Telephony WAN Interfaces supported on a single MultiVoice Gateway please see the "*Hardware Features*" section of this document.

In addition to the signaling interface, the MultiVoice Gateway will also support μ-Law to/from A-Law conversion for calls that involve both a T1 and E1-based MultiVoice Gateway. This allows for proper voice coding across certain international calls.

Packet Network Interfaces

The MultiVoice Gateway software supports both standard Ethernet LAN and Frame Relay connections to the IP packet network. For more information on these interfaces see "*Hardware Features*" section of this document.

Voice Codec Support

The audio signals that pass through a MultiVoice Gateway for the MAX will be compressed in order to transmit the speech across the IP packet network. As such, the H.323 standard has defined a set of voice codecs that can be used to compress speech for a typical phone call. The reason the standard recommends different compression algorithms is that there are different tradeoffs between speech quality, bit rate, computing power, and signal delay that are addressed by each of the recommended codecs.

The R1.0 MultiVoice Gateway for the MAX will support the following codecs:

- *G.711* – the G.711 codec transmits digitized voice at 64 Kbps and provides toll quality voice on managed IP networks with sufficient available bandwidth.
- *G.729 Annex A* – the G.729 (A) codec is a low-complexity codec that transmits digitized and compressed voice at 8 Kbps. This codec provides toll quality voice on managed IP networks.

Note: future releases of the MultiVoice Gateway for the MAX will support other ITU-T recommended codecs.

The codec to be used for a specific packet voice call is negotiated between the two Gateways using the H.245 protocol (see "*Phone-to-Phone H.323 Operation*"). This negotiation will be influenced by the specific codec configuration on the MultiVoice Gateway. The following are the administered Gateway criteria's for selecting a specific codec at the time the call is established:

- *Use G.711 only* – the G.711 codec will be used for all calls
- *Use G.729 (A) only* – the G.729 (A) codec will be used for all calls
- *Use Highest Common Bit Rate Codec* – select the codec that will provide the best voice quality
- *Use Lowest Common Bit Rate Codec* – select the codec that will provide the lowest packet bandwidth requirement

This feature allows NSPs to establish offers that provide different quality voice services at different price points.

Voice VPN Support

In conjunction with the MultiVoice Access Manager, the MultiVoice Gateway for the MAX will supports a voice Virtual Private Network mode where user authentication is bypassed. When using this feature, users trying to access a MultiVoice network via a Gateway will not be prompted for a PIN and will be allowed to directly dial the called number. See "Voice *VPN Support*" in the MultiVoice Access Manager Feature Description section of this document.

Hybrid-Line Echo Cancellation Support

For PSTN networks that require 2-4 wire hybrid echo cancellation, the R1.0 MultiVoice Gateway for the MAX will support network echo cancellation with performance better than required by ITU-T G.165. This echo canceller is designed to eliminate the echo generated in a PSTN network when a voice call is transmitted across a two-wire/four-wire boundary. In PSTN networks where echo cancellation is required, e.g. some international PSTN networks, the MultiVoice Gateway will perform this function and deliver end-to-end echo cancelled voice calls.

The reader should note that the G.165 echo canceller supported on the MultiVoice Gateway only addresses network echo and not the acoustic echo that is normally generated at the end-point (receiver/transmitter echo). This type of echo can only be addressed in the telephone end-point equipment.

DTMF Detection and Generation

The R1.0 MultiVoice Gateway for the MAX provides Dual Tone Multi-Frequency (DTMF) detection and generation in order to support telephony features required to handle voice calls. Specifically the DTMF Detection and Generation feature in the R1.0 Gateway is used for the following functions:

- Detecting the numeric PIN entered by the user for MultiVoice network authentication.
- Detecting the destination number to be dialed from the far-end Gateway, i.e. the dialed telephone number
- To provide tone-based prompting to the user including dial tone, busy tone, re-order tone, ringback tone, etc.

MultiVoice Access Manager Feature Description

The MultiVoice Access Manager provides network call routing functions for connecting voice calls over an IP network, user authentication, alias translation, and call accounting features. The following sections provide a more detailed description of the major features supported by MVAM Release 1.0.

Microsoft Windows NT v4.0 Support

The MultiVoice Access Manager Release 1.0 is a software application that requires any Intel-based Workstation/Server Platform that is compatible with Windows NT v4.0 Workstation or Server. The NT Workstation or Server must be connected to the same IP network as the MultiVoice Gateways that are to be managed by a specific Access Manager. The specific PC hardware configuration required will depend on the size and scope of the MultiVoice network. Larger networks with higher call volumes may require larger and more scalable Windows NT servers.

Future releases of the MultiVoice Access Manager will provide support for UNIX-based server platforms.

ITU-T H.323 Gatekeeper Implementation

The Ascend MultiVoice Access Manager provides an implementation of an H.323 “Gatekeeper” as specified in the ITU-T H.323 standard. The Access Manager supports the H.323 Registration, Admissions, and Status or **RAS** protocol for gateway to gatekeeper communications as required by the ITU standard. The MultiVoice Gateway and the MultiVoice Access Manager use the RAS channel for:

1. Registration of gateways within a network – used to indicate to MVAM that a specific MultiVoice Gateway is ready to handle calls, i.e. it is ready to *hop-off* calls into the far-end PSTN network
2. Call Admission which includes:
    - User authorization with Personal Identification Numbers (PINs)
    - Address translation (alias translation and telephone number to IP address translation)

Phone-to-IP Address Translation

The MultiVoice Access Manager provides support for the translation of standard E.164 telephone numbers and/or private dialing plan numbers to IP Addresses. This translation is required in order to route voice calls from the near-end Gateway to the far-end Gateway across the IP network.

Web-based Administration Interfaces

The MultiVoice Access Manager supports local and remote system management functions through a co-located, third party web server. Release 1.0 of the MultiVoice Access Manager has been tested with the following Web Servers:

- Microsoft Internet Information Server (IIS) v3.0 available from Microsoft Corporation at http://www.microsoft.com
- Xitami Web Server v1.3c from Imatix. This is a “free” web server that is available from http://www.imatix.com

Management of the MultiVoice Access Manager is accomplished through a set of management web pages that can be viewed using a standard web browser. The Access Manager management web pages have been tested with the following browsers:

- Microsoft’s Internet Explorer V3.02
- Netscape Navigator V3.01

The web-based management interface allows the network administrator to:

- Configure the MultiVoice Access Manager and get status information.
- Administer the User Database – this includes the ability to **add**, **modify**, **remove**, and **display** users that are authorized to access the MultiVoice network. Each user record in the database can include *user name*, *user alias*, *telephone number*, *telephone alias*, and *PIN*.
- Administer the Gateway Database – this includes the ability to **add**, **modify**, **remove**, and **display** Gateways that are part of the MultiVoice network. Each Gateway record can include the *name*, *telephone number*, *street*, *city/town*, *state/province*, *zip code*, and *country* associated with a specific Gateway on the MultiVoice network. The *name* field specifies the DNS name assigned to the IP address associated with a specific Gateway.
- Administer Gateway Coverage Areas – the network administrator can also configure **Gateway Inclusion Areas** that define the set of telephone numbers that are covered by a specific Gateway on the *hop-off* end of a call. The inclusion areas uniquely specify the set of telephone numbers that can be out-dialed from a specific far-end Gateway. This allows the network designers to assign different Gateways to a specific set or range of telephone numbers. For example, a Gateway administered with an inclusion area of “1732” can only terminate calls which are targeted to a country code of “1” and an area code of “732”. The inclusion area can be a partial or complete telephone number depending on the desired coverage area for a specific far-end Gateway.

PIN-based User Authentication

With PIN-based User Authentication, users that access a MultiVoice network are required to enter a PIN number to authenticate them as a user of the network. The process for user authentication is very similar to the process used for standard Calling Card calling on today’s PSTN. The MultiVoice for the MAX authentication process is as follows:

- The user dials the access telephone number of the near-end Gateway. Usually this is a local telephone number.
- The MultiVoice Gateway answers the call and prompts the user to enter their PIN and the number to be dialed at the far-end Gateway
- The Gateway collects the PIN and called number using the DTMF detection feature and forwards the collected digits to the MultiVoice Access Manager for authentication against the centralized User Database.
- The MultiVoice Access Manager authenticates the user and either accepts or rejects the user call request.
- If the user is accepted, then the VoIP call is established to the far-end Gateway and then the call is completed via the local PSTN at the far-end
- If the user is rejected, the Gateway prompts the user and informs them that their call request has been denied.

Voice VPN Support

The MultiVoice Access Manager supports a voice Virtual Private Networking (VPN) feature where no Personal Identification Number (PIN) is required to authenticate a user trying to access the MultiVoice network. When this feature is activated for a specific MultiVoice Access Manager, the PIN authentication mechanism will be disabled for all Gateways that are defined within the *zone* associated with the Access Manager. This feature is useful for Enterprise customer with private MultiVoice networks where billing and authentication is not required.

Telephone Number Aliases

The R1.0 MultiVoice Access Manager will also support the ability to define telephone number aliases. This feature allows a full endpoint telephone number to be expressed as a different alias or extension. For example, the number 1-732-555-1234 can be expressed as the alias 51234. When the caller enters this number after the authentication process, the MultiVoice Access Manager *translates* this alias to a complete and valid telephone number based on the administered alias list.

This feature is useful for enterprise or voice Virtual Private Networks for assigning shorthand extensions to users that are part of the network.

Call Accounting Records (CDR)

The MultiVoice Access Manager provides the ability to generate Call Record Files on a daily, weekly, monthly basis, or immediately as specified by the network administrator. The files contain the *start* and *stop* call record for all calls established on the MultiVoice network. This information can be used to determine call duration for billing purposes.

Gateway and User Database Support

The MultiVoice Access Manager maintains two separate databases that define the extent of the MultiVoice VoIP network and the list of users who are authorized to use the network. The R1.0 Access Manager maintains the Gateway Databases in a flat file on the Windows NT Server. The User Database can be maintained in a flat file or on a commercially available Open Database Connectivity (ODBC) compliant database engine. Future versions of the MultiVoice Access Manager will support ODBC-compliant database engines for *both* the User and Gateway Databases.

Third-Party Billing Support

The MultiVoice Access Manager supports a third-party billing interface that allows the management of the User Database to be controlled by a separate billing application. This results in a single user database that is under the control of the billing application. The administrator uses the billing application, instead of the Access Manager, to *add*, *modify*, *remove* and *display* users of the MultiVoice network. The Access Manager performs its PIN-based User Authentication function using an ODBC link to the user database maintained by the third-party billing application.

The MultiVoice Access Manager also provides third-party billing application access to the call records generated for every call that is established on the MultiVoice network. These records can then be used by the billing application to generate end-user bills for calling usage of the MultiVoice network. The Access Manager measures call duration in seconds thus allowing billing resolution down to the second.

3. Hardware Features

The R1.0 MultiVoice Gateway system software will be supported on the same MAX platform that is currently deployed in most of the largest ISP's in the world. As such, the MultiVoice Gateway for the MAX is supported on hardware that has been field tested in some of the world's largest remote access network deployments. The R1.0 MultiVoice Gateway requires a dedicated MAX platform. This implies that the MAX platform used for the MultiVoice application *must* be dedicated to this function and cannot support any other simultaneous MAX applications (e.g. remote access).

Note: This limitation applies to Release 1.0 and will be removed in a future release of the MultiVoice Gateway for the MAX platform.

The R1.0 MultiVoice platform will support the following hardware features:

- MAX 6000 Platform Support
- MAX 400X Platform Support (Upgrade Only)
- MultiVoice DSP16 Slot Card for the MAX 400X and 6000
- MultiVoice DSP12 Slot Card for the MAX 400X and 6000
- MultiVoice DSP8 Slot Card for the MAX 400X and 6000
- ISDN BRI Slot Cards
- WinTel Platform Support for the MultiVoice Access Manager

All hardware supported by the R1.0 MultiVoice Gateway and MultiVoice Access Manager software is discussed in this section of this document.

Note: All other currently available MAX hardware options not mentioned in this document are currently not supported by the R1.0 MultiVoice platform for the MAX.

MAX 6000 Platform Support

The main deployment of the MultiVoice Gateway will be supported on the MAX 6000 platform[2]. The MAX 6000 is Ascend's next-generation WAN access switch that provides Internet Service Providers, Network Service Providers/Carriers, and Corporations the increased performance and expandability needed for both current and future access services including Voice-over-IP.

The MultiVoice Gateway consists of the MAX 6000 chassis plus MultiVoice DSP slot cards for the MAX. The MAX 6000 chassis provides all the PSTN interfaces, the packet interfaces, and the slots for inserting MultiVoice DSP slot cards. The DSP slot cards are used to implement most of the MultiVoice Gateway software. For example, the voice compression is accomplished on the MultiVoice DSP slot cards (see *"MultiVoice DSP16 Slot Card*" for more information).

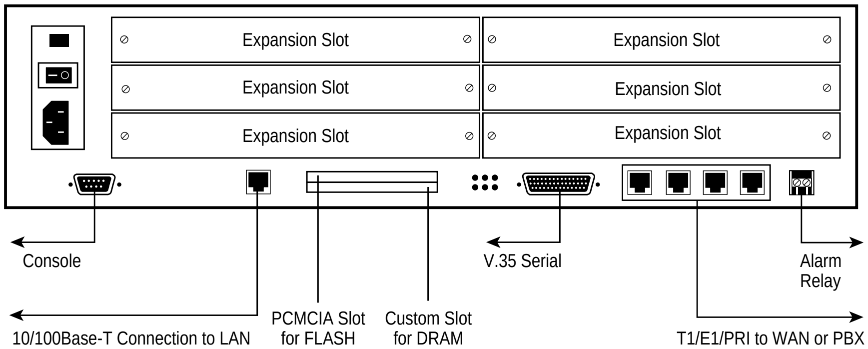

Figure 7 – MAX 6000 Back Panel

- *Console Port* – used for local administration and configuration of the MAX 6000 platform
- *V.35 Serial Interface* – used for interfacing to Frame Relay packet networks
- *10/100Base-T Connection to LAN* – autosensing Ethernet 10Base-T/Fast Ethernet 100Base-T (RJ45) connections for accessing a LAN
- *FLASH Memory Card Slots* – built in FLASH RAM or PCMCIA FLASH card used for software upgrades
- *Telephony WAN Interface Ports* – the MAX 6000 support four T1/E1 lines for connecting to the standard PSTN network.

The R1.0 MultiVoice Gateway uses the ports on the MAX 6000 chassis as follows:

- *V.35 Serial and the 10/100Base-T Connection* – these connections are used to access the IP-based packet network. This network can be located on a local Ethernet LAN or on a Frame Relay network that support the IP protocol. All packetized voice calls will be carried across these interfaces.
- *Telephony WAN Interface Ports* – these ports are used to interface to the PSTN via a T1, T1/PRI, E1, and E1/PRI. These connections can be used with a network based central office switch (e.g. 5ESS, 4ESS, DMS-100, etc.) or a premises-based PBX that supports these interfaces. All phone-to-phone voice traffic on a MultiVoice network will enter/exit the VoIP packet network via the Telephony WAN interfaces.

The R1.0 MultiVoice Gateway will be offered in several different bundled configurations on the MAX 6000 platform. These configurations are described in the following table:

MAX 400X Platform Support (Upgrade Only)

The R1.0 MultiVoice Gateway software will also be supported on existing MAX 400X platforms. Support will only be available as a software upgrade to an *existing, unpopulated* MAX 400X chassis. This option is being provided to allow Ascend customers who have upgraded a 4000 chassis to a MAX 6000 or MAX TNT™ to re-use the MAX 400X chassis to deploy Voice-over-IP services within their managed IP network. As is the case with the MAX 6000, the MAX 400X chassis with the MultiVoice Gateway software upgrade must be dedicated to the MultiVoice application only. No other concurrent MAX application will be supported on a 4000-based MultiVoice Gateway.

The R1.0 MultiVoice Gateway will be supported on the MAX 4000 (international), MAX 4002 or MAX 4004 chassis. The MAX 400x chassis has the following hardware specification:

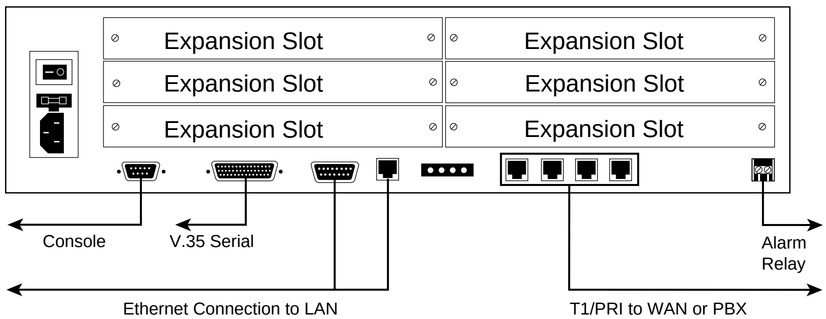

Figure 8 – MAX 400X back panel

- *Console Port* – used for local administration and configuration of the MAX 400X platform
- *V.35 Serial Interface* – used for interfacing to Frame Relay packet networks.
- *10Base-T/AUI Connection to LAN* – Ethernet 10Base-T(RJ45)/AUI connector for accessing a LAN
- *Telephony WAN Interface Ports* – the MAX 400X support four T1/E1 lines (4004) or two T1/E1 lines (4002) for connecting to the standard PSTN network.

The R1.0 MultiVoice Gateway uses the ports on the MAX 400X chassis as follows:

- *V.35 Serial and the 10Base-T Connection* – these connections are used to access the IP-based packet network. This network can be located on a local Ethernet LAN or on a Frame Relay network that support the IP protocol. All packetized voice calls will be carried across these interfaces.
- *Telephony WAN Interface Ports* – these ports are used to interface to the PSTN via a T1, T1/PRI, E1, and E1/PRI. These connections can be used with a network based central office switch (e.g. 5ESS, 4ESS, DMS-100, etc.) or a premises-based PBX that supports these interfaces. All phone-to-phone voice traffic on a MultiVoice network will enter/exit the VoIP packet network via the Telephony WAN interfaces.

Bundled configurations of the R1.0 MultiVoice Gateway will not be offered on the MAX 400X platform.

MultiVoice DSP16 Slot Card Support

For the most part, the MultiVoice Gateway software runs on special purpose processors that are integrated on the MultiVoice DSP Slot Cards. Each DSP Slot Card can be inserted into any available MAX expansion card slot. On the MAX 6000 platforms a maximum of six (6) cards can be inserted into the base chassis (see MAX back panel figures shown above).

Each MultiVoice DSP card contains a control Digital Signal Processor (DSP) and a set of 16, 12, or 8 slave DSPs. The control DSP is used for local control functions and data transfer between the Slave DSPs and the core processor. The slave DSPs are used for voice packetization and compression on a per call basis. Each slave DSP is connected to a single DS0 channel (a single voice call) on the PSTN T1/E1 interface line.

The Control DSP communicates with the Host CPU on the MAX chassis motherboard in order to interface with the IP network and provide other control functions. Once the voice is packetized and compressed it is handed over to the Host CPU to transmit across the IP packet network. This architecture is illustrated in the following figure:

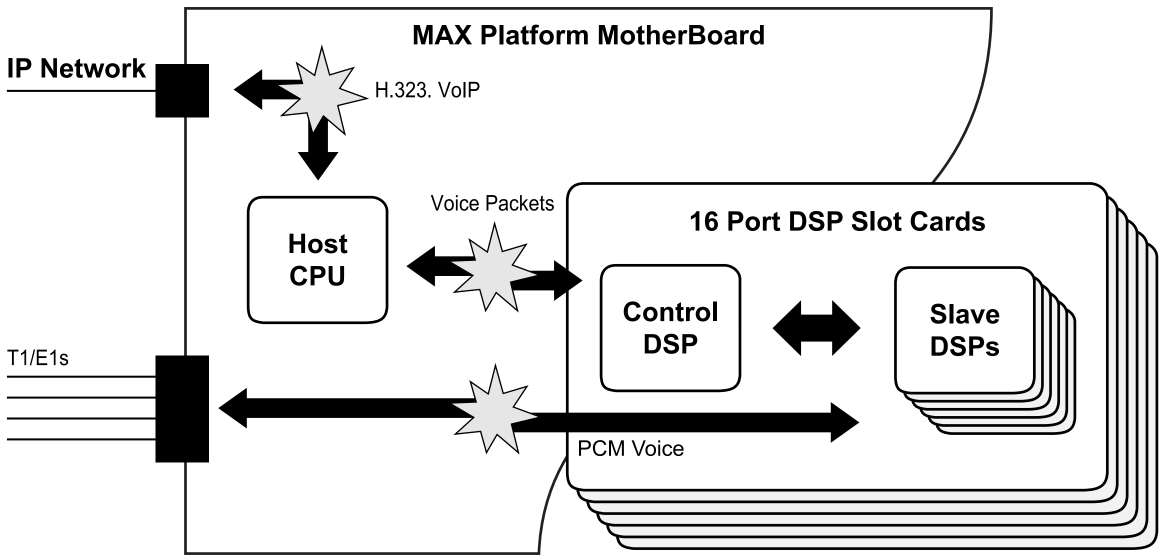

Figure 9 – MultiVoice slot card architecture

MultiVoice DSP12 Slot Card Support

The MultiVoice DSP12 Slot Cards has the same features and functions as the DSP16 card except that it contains only 12 slaves DSPs. The DSP12 card supports twelve (12) simultaneous channels of Voice-over-IP communications per DSP12 card.

MultiVoice DSP8 Slot Card Support

The MultiVoice DSP8 Slot Cards has the same features and functions as the DSP16 card except that it contains only 8 slaves DSPs. The DSP8 card supports eight (8) simultaneous channels of Voice-over-IP communications per DSP8 card.

ISDN BRI Slot Cards

The MultiVoice Gateway also supports voice calls which originate/terminate via the standard MAX ISDN BRI network interface card. This slot card provides support for up to eight (8) ISDN BRI network ports and can be used to connect VoIP calls. Support for the BRI network interface slot card is being provided for MultiVoice customers who have a MAX and wish to use BRI instead of the higher-density T1/E1 network interfaces. This solution can be used for MultiVoice VoIP trials or full-blown service deployment where the cost of T1/E1 network lines is prohibitive or not available. With respect to the MultiVoice Gateway, the ISDN BRI Slot Card serves the same functions as the T1/E1 network interface.

WinTel Platform Support

The R1.0 MultiVoice Access Manager is a software application that will generally run on any Intel PC-compatible platform that can support Microsoft Windows NT v4.0. The minimum WinTel configuration for running the MultiVoice Access Manager application is as follows:

- Pentium 100 MHz CPU
- 32 Mb of System RAM
- 2 GB of Hard Disk space
- CD-ROM Drive
- 10/100Base-T Network Interface Card
- Windows NT v4.0 Server or Workstation

Although this is a minimum configuration, the exact requirements for a specific deployment of the MultiVoice Access Manager will depend on many different factors. The following guidelines can be used for determining the proper hardware configuration of the MultiVoice Access Manager server:

- *VoIP Traffic Volume* – since each VoIP call may require user authentication and E.164/private dial plan ↔ IP address translation, MultiVoice networks with large call volumes should use high-end Windows NT Server platforms to insure acceptable network performance. For example, this may include a multi-Pentium processor server platform with fast disk arrays.
- *Network Reliability* – in large MultiVoice networks it may be desirable to run the MultiVoice Access Manager on highly reliable or fault-tolerant network servers in order to insure network availability. Any such Intel-based server that has been certified with Windows NT v4.0 can be used to run the MultiVoice Access Manager.

Since the MultiVoice Access Manager is a standard Windows NT-based application, MultiVoice networks can take advantage of any performance and reliability technology advances that are provided by the broad WinTel industry.

Worldwide and North American Headquarters
Ascend Communications, Inc.
One Ascend Plaza
1701 Harbor Bay Parkway
Alameda, CA 94502, United States
Tel: 510.769.6001
Fax: 510.747.2300
E-mail: info@ascend.com
Toll Free: 800.621.9578
FAX Server: 415.688.4343
Web Page: http://www.ascend.com

European Headquarters
Aspen House
Barley Way
Ancells Business Park
Fleet
Hampshire GU13 8UT
United Kingdom
Tel: +44 1252.360000
Fax: +44 1252.360001

Asia-Pacific Headquarters
Suite 1908
Bank of America Tower
12 Harcourt Road
Hong Kong
Tel: +852.2844.7600
Fax: +852.2810.0298

Japan Headquarters
Level 19 Shinjuku Daiichi-Seimei Bldg.
2-7-1 Nishi-Shinjuku
Shinjuku-ku, Tokyo 163-07, Japan
Tel: +81.3.5325.7397
Fax: +81.3.5325.7399
Web Site: http://www.ascend.co.jp

Latin, South America and the Caribbean Headquarters
One Ascend Plaza
1701 Harbor Bay Parkway
Alameda, CA 94502, United States
Tel: 510.769.6001
Fax: 510.747.2300

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