AudioCoded™ Enterprise VoIP Networking (EVN): Migrating to the New Voice Infrastructure

White Paper

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Introduction

This paper describes a solution whereby business customers can lower their telephone expenditures, while improving communications between employees (both from the workplace and the home) by using VoIP. Integration of VoIP into the realm of IT equipment is made possible by acquiring network equipment, which is VoIP-enabled. This is as opposed to extending the separate traditional telephony equipment, such as PBXs or key systems and the interfaces to the Public Switched Telephone Network (PSTN) or by replacing the entire IT infrastructure with VoIP-enabled PBXs and special IP telephone terminals. Today it is necessary to keep costs down. Therefore, getting more value out of the IT infrastructure investment especially makes sense. Many companies have come to the conclusion that enabling their IT infrastructure with VoIP is a good investment. The fact that manufacturers and distributors of VoIP technology should understand the need for investing in enterprise VoIP is reflected in the following conclusions by Synergy Research Group[1]:

"VoIP in the Enterprise will continue to gain momentum as a means to consolidate voice and data traffic over WAN connections and as a migration to "next generation PBXs". We believe what has initially started at the Enterprise level will begin to filter down to small businesses and commercial residences as non-traditional voice services are made available." Synergy also had this to say about worldwide enterprise VoIP equipment sales: "...In Q1 2002, Enterprise VoIP Equipment sales were $277 million .... and grew 30 percent year over year. Synergy is forecasting the Enterprise IP Telephony market to grow to $666 million in 2002, representing an increase of 14 percent over 2001 sales. Further, Synergy is forecasting the Enterprise IP Telephony market to grow to $2.2 billion by 2006..."

Architecture

AudioCoded™ **Enterprise VoIP Networking (EVN)** is made possible with a digital VoIP gateway (such as AudioCodes **Mediant™ 2000**) and analog VoIP gateways, (such as AudioCodes' **MediaPack™** series) enabling the "new voice infrastructure" in the legacy corporate (PBX) environment and also for connecting branches only with analog terminals (e.g., telephone or fax), key systems (without PBX) using line-level trunks. Traditional PBXs are found in many enterprise locations (small, medium, and large businesses) and will exist for quite some time. Eventually, when the legacy PBX is replaced by an IP-enabled PBX, the digital gateway can be used to bridge the VoIP network and the Public Switched Telephone Network (PSTN). Thus an ideal migration path from legacy telephony to the more advanced, converged voice/video/data network is available for enterprises wishing to protect their investment in PBX and key systems (and related "smart" phones, POTS phones, and faxes), while migrating to a cost-effective VoIP solution. The migration path is shown in Figures 1 to 2 as follows:

[1] *Synergy Research Group: from 1st Quarter 2002 Analysis and Forecast report*

Figure 1: First Stage

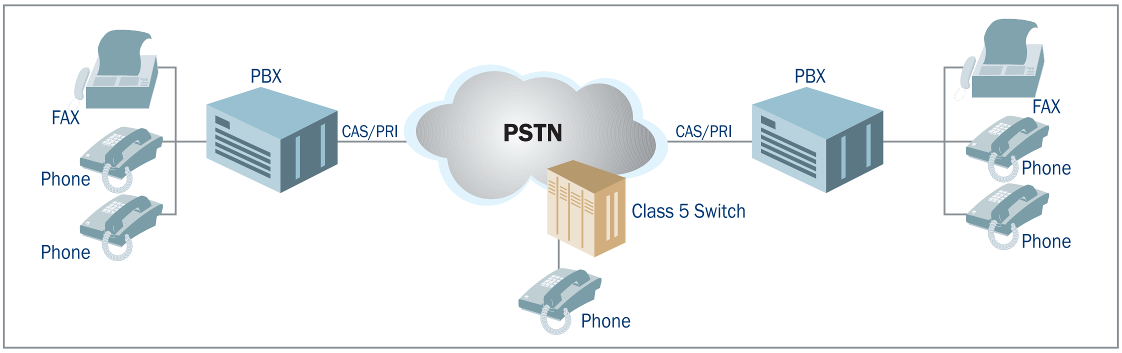
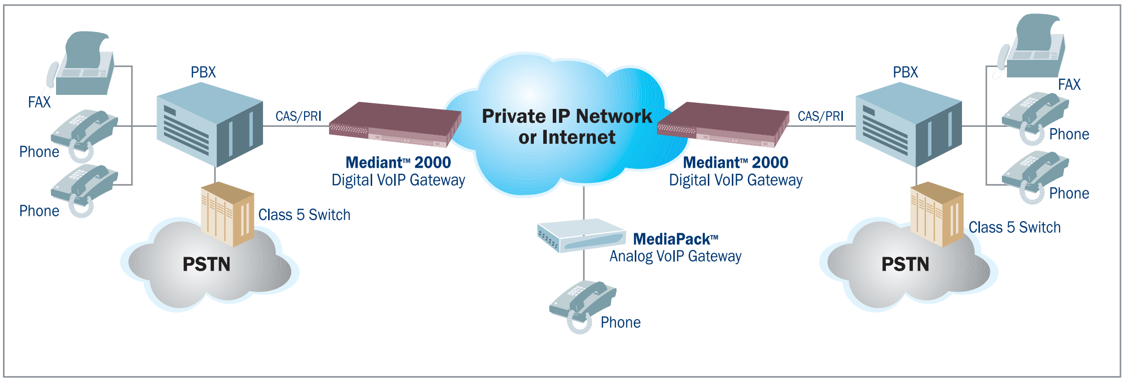


Figure 2: Second Stage



Four Major Benefits of Enterprise VoIP Networking

Businesses can enjoy four major benefits of Enterprise VoIP Networking: increased savings, productivity, availability and features.

Savings

The existing multi-site data network will be used for connecting local and remote branches. An ISP can also be used for multi-branch PBX connectivity via the Internet for enterprises without a managed IP network. The VoIP Gateway is used to access the IP network, allowing long-distance and international calls at significantly reduced rates. As in the case of the private data network, telephone calls are mere data packets sent and received over the network in the same way as electronic mail. If the calls are routed via ISP, over the Internet, the cost model is based on a flat rate for use of the data network, based on data packets, not on geographical distance. Return on Investment (ROI) can easily be calculated by checking the bill from the local exchange carrier for all calls made between branches as well as long distance calls made from home or from mobile phones to the remote branches or their local area (LATA). In addition, the cost of any leased (T1/E1) lines will be included. This will be compared with the extra spending required for VoIP-enabled gateways, beyond the LAN, routers, firewall, and various servers that are required in any case for the company's day-to-day data requirements. Since the data and voice networks are now converged, additional savings occur as a result of reduced staff workload. Voice and data will use the same distribution system of Ethernet ports, cables, and connectors, thereby reducing maintenance costs (e.g., for adding or moving lines). Another cost-benefit with the converged voice/data network is that surplus bandwidth may be exploited by telephone calls instead of data usage (e.g., after normal work-hours for calls to remote locations in different time zones). In addition, an extendable (scalable) gateway will increase the value of the initial investment in the long term, since costs will go down as savings increase over time.

Increased Productivity

Even when subscribers are not at the office, they can take advantage of the VoIP network from home or on the road by accessing the nearest branch via the PSTN and then dialing via the VoIP network, dramatically improving productivity. Calls from one branch to another can be made as if both subscribers are located in the same branch; that is, calls can be made by dialing an optional access code and the extension only (virtual private numbering plan). Employees will be encouraged to communicate directly (as opposed to electronic mail) with other employees more often, since they can expect ease of dialing combined with the knowledge that long-distance calls to their colleagues will be treated as internal calls.

Increased Availability & Enhanced Features

VoIP offers a reliable alternative to the PSTN, allowing communications even during a local disaster (e.g., "September 11"). In addition, by adding at least one gatekeeper, subscriber features, such as call forwarding, call diversion, and call pickup can be supported between the branches in a seamless fashion. For example, during certain hours, all calls from one operator to another can be forwarded from one branch to another, in order to provide around-the-clock service, without incurring expensive long-distance charges.

Mediant™ 2000

Features

The **Mediant™ 2000** is a low-density VoIP gateway, appropriate for small to medium enterprise applications. The Mediant 2000 can be used with two types of TDM signaling interfaces: CAS or PRI. The Mediant 2000 makes sense for a customer not requiring the costly infrastructure of large universal gateways or dial-access routers, which are typically not dedicated just to voice applications (but also to data applications, such as routing). Specifically, the Mediant 2000 is advantageous over alternative VoIP gateways, since it is cost effective at the low end of the (T1/E1) scale. On the other hand, the Mediant 2000 offers scalability – up to 16 E1/T1 spans. In addition, the Mediant 2000 offers the full range of VoIP and TDM signaling portfolios. Thus, the customer can protect their investment by beginning to deploy the ubiquitous H.323 standard for enterprise networks, and to migrate to a SIP-based network at a later stage, when such products become more widely available.

The Mediant 2000 also offers various carrier-class features normally found on larger, more costly gateways, while not offered by smaller media gateway alternatives, such as:

- Hot swap board replacement
- Dual Ethernet interface to LAN
- Optional dual/redundant power supplies

The Mediant 2000 provides an optional second cPCI slot that can be used for a CPU board with a third party application. This CPU board can be used for general applications such as:

- Gatekeeper
- Call Agent
- SIP Proxy
- Softswitch
- RADIUS server
- Application server

Size and Scalability

- 1, 2, 4, 8 or 16 E1 or T1 spans for connection to PBXs or to the PSTN network
- Up to 480 simultaneous channels
- 5000 subscriber lines can be supported (depending on the given subscriber/DSP port concentration ratio). Using a typical 1:10 ratio, hundreds to thousands of concurrent calls can be set up on the private voice network)
- Exceptionally compact form factor: 1 Rack Unit high, installed in a 19-inch rack

Routing and Fallback

If communication is lost with the gatekeeper, the Mediant 2000 and the MediaPacks will automatically continue to route calls using the Phone to IP Routing function.

The Mediant 2000 and the MediaPacks can function completely independent of a gatekeeper, by using the Phone to IP Routing function, which enables the Mediant to translate between (E.164 based) prefixes and the other gateways' IP addresses, while supporting user-inputted routing rules (e.g., adding and truncating prefixes). This is intended as a solution for small organizations, requiring basic telephone service without using a gatekeeper.

In case of loss or degradation of service on the IP network, the Mediant 2000 can block the PSTN/TDM interface, thereby enabling new call setups to be routed by the PBX via the PSTN until the available quality of service returns to acceptable levels[2] on the VoIP network.

Best-of-breed VoIP and DSP Technology

- High traffic handling ability
- Low latency

[2] Feature planned for future release

- Echo cancellation (G.168-2000, tail length of 32, 64, or 128 msec[3])
- Dynamic jitter buffer
- High quality voice (including inband fax up to 14.4 kbps and modem up to V.92) and fax (T.38) over IP calls
- Robust and reliable (redundant 10/100BaseT LAN connection, optional redundant power supply)
- Broad support for industry standard vocoders: G.711, G.723.1, G.726, G.727, G.728, G.729A, G.729E, and NetCoder® at 6.4 to 9.6 kbps, selectable per channel. Vocoders are optional and are subject to IPR licensing
- Silence suppression with Comfort Noise Generation
- T.38 Fax with superior performance (round trip delay up to 9 sec)
- Tone detection and generation (e.g., dial tone, ringing tone, busy tone, etc.) and announcement generation
- Easy to use Web interface for setup and configuration of VoIP/PSTN parameters
- Highly scalable equipping of E1/T1 spans allows "pay as you grow" planning

MediaPack™

Features

- Size and Scalability: 2, 4, 8, or 24 analog telephone/fax/modem (RJ-11) ports within a compact, stand-alone enclosure
- Analog line or trunk configurations (available with either FXS or FXO ports)
- Dual Ethernet ports to LAN
- Supported vocoders: G.711, G.723.1, G.726, G.729A, and Netcoder®, selectable per channel. Vocoders are optional and are subject to IPR licensing.
- Fax over IP (transparent), T.38 compliant
- Transparency for Caller ID (FSK or DTMF) and Modem (up to V.92 using PCM or ADPCM)
- Echo cancellation, G.168-2000, tail length of 25 msec
- Dynamic Jitter Buffer
- Short and long haul
- Lifeline support
- Outdoor and indoor models
- Message Waiting Indication (MWI) support
- Polarity reversal
- Metering pulses (12 and 16 kHz)
- Distinctive ringing

Typical Configuration

A typical configuration is depicted in Figure 3. The deployment architecture can include several **MediaPack™** analog VoIP gateways in small branch or departmental offices

[3] 64, 128 msec available in future; contact AudioCodes regarding availability

connected to key systems or analog phones, and the **Mediant 2000** connected to digital PBXs and to the PSTN. Central call processing and management of the Enterprise VoIP Network is typically supported by an H.323-based gatekeeper for a more advanced subscriber feature set.
The Mediant 2000 can modify the digit stream, via user-inputted dialing plans. For example, calls routed via the IP network from one corporate branch to another can be modified (prefix truncated) to appear as internally dialed calls.

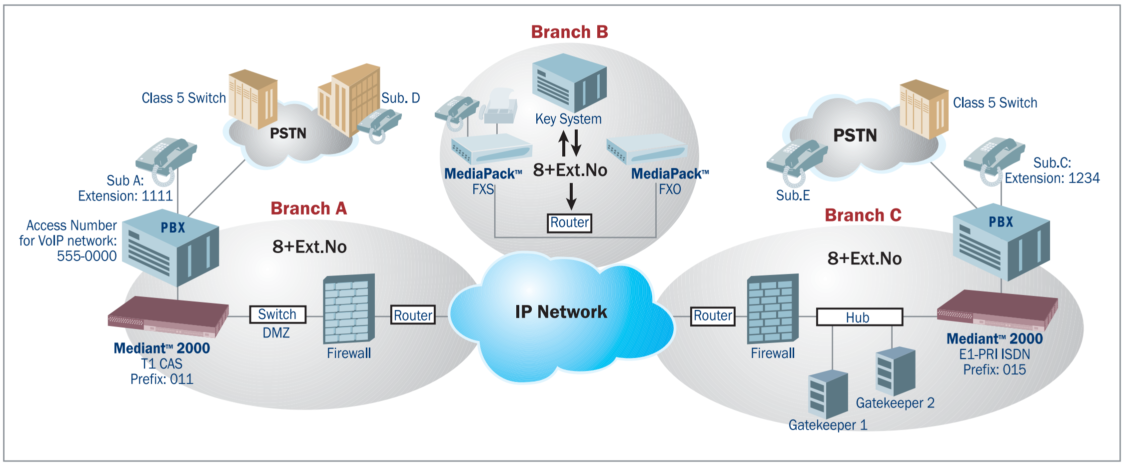
In Figure 3, an example of how to dial from one location to another by using the VoIP network is shown:
1. Local branch to remote branch dialing:
 - Subscriber A dials access digit to bypass the PSTN plus subscriber C's extension: 8 + 1234

2. Local branch to remote branch's local area dialing:
 - Subscriber A dials access digit to bypass the PSTN plus prefix of Branch C plus access digit for local dialing from Branch C: 8 + 015 + 9
 - Subscriber A receives dial tone from Branch C local exchange
 - Subscriber A dials number of subscriber E, who resides outside Branch C (e.g., in same city)

3. Subscriber dials to remote branch from home:
Subscriber D (at home) dials access number for VoIP network in Branch A plus PSTN bypass digit plus subscriber C's extension: 555-0000 + 8 + 1234

Note for points 1-3: Initial authorization is requested (via call setup request from gateway) and granted by central gatekeeper in Branch C. Call signaling is routed via gatekeeper (Routed mode) or directly (Direct mode) between gateways in local branch and remote Branch.

Figure 3: Example of Dialing Plan within EVN



Control and Management

The Mediant 2000 supports the ***H.323*** protocol, thus enabling deployment of "Voice over Packet" solutions in enterprise environments. The Mediant 2000 Gateway is connected to the PBX via CAS or ISDN-PRI, whereas the MediaPacks are connected to key systems or analog lines. Call control is normally provided by an H.323 gatekeeper. A single gatekeeper (or multiple gatekeepers for redundancy) is installed at the main site to serve a number of geographically distributed sites or one-two gatekeepers per large site may also be configured.

The Mediant 2000 supports many variants of PSTN signaling protocols, such as:

- ISDN PRI protocols, such as EuroISDN, North American NI2, Lucent™ 5ESS, Nortel™ DMS100, NFAS, and others. As well, QSIG (Q.931 Basic Calls) is supported.
- Different variants of Channel Associated Signaling (CAS) protocols for E1 and T1 spans, including MFC/R2, E&M immediate start, E&M delay dial/start, loop start, as well as T1 "robbed-bit".

The Mediant 2000 and the MediaPacks support a broad range of standard VoIP signaling protocols, both for the enterprise environment and for the Carrier environment:

- ITU-H.323, version 4, VoIP signaling and control protocol
- MGCP (Media Gateway Control Protocol) (IETF RFC 2705)
- MEGACO (***M***edia ***G***ateway ***C***ontrol) protocol (IETF RFC 3015, ITU-T H.248)
- SIP (IETF RFC 2543)
- IUA (ISDN User Application), SIGTRAN transport protocol (IETF RFC 3057) available on the Mediant 2000
- **TrunkPack®** Network Control Protocol (TPNCP) – AudioCodes' proprietary VoIP transport protocol, available on the Mediant 2000 for OEMs as an API library

The MediaPack Series supports the following PSTN signaling interfaces:

- Analog line (FXS)
- Analog (line-level) trunk (FXS)
- Analog trunk (FXO), PSTN/PBX

Many variants of both PSTN/TDM protocols and standardized VoIP protocols are supported, making AudioCodes' line of VoIP gateways one of the most extensively supported in the market.

The Mediant 2000 supports a RADIUS interface, which can be used with third-party billing application servers[4].

Management

The Mediant 2000 and the MediaPacks are configured using the friendly Web interface. System parameters, such as those related to the TDM and IP networks, can be configured online; otherwise they behave according to defaults provided by the INI file. The AudioCoded™ EMS[5], based on an external UNIX-based management platform, supports configuration and management of multiple Mediant 2000 and MediaPack gateways.

[4] Contact AudioCodes regarding specific customer requirements

[5] Regarding AudioCoded EMS for Mediant 2000, contact AudioCodes about availability

Security

The Mediant 2000 and MediaPack's Web interface is password protected and allows up to 3 concurrent users, so that access is easily managed in a secure way. Usernames and passwords are protected (by encryption) from packet "sniffers". The Web interface can also be disabled or configured as "read only" (i.e., for monitoring only).

In addition to the authentication and authorization of users that the gatekeeper normally supplies, IP access to the Mediant 2000 and MediaPack gateways is limited to a listed of authorized IP addresses, (e.g., other gateways or gatekeepers). In addition, different IP addresses (Public and Private) can be used to support working with NAT servers[6]. These features are particularly useful, when the EVN is configured without a gatekeeper (e.g., for small subscriber deployments of basic telephone service).

Interoperability

The Mediant 2000 and the MediaPack Series have undergone extensive testing for interoperability with various H.323 gatekeepers (e.g. Cisco, HP, Nortel, Siemens, 3Com, Alcatel, RADVISION, NetCentrex, etc.). As well, many types of analog VoIP gateways (CPE) and types of PBX have been successfully tested and deployed with the Mediant 2000. Interoperability has also been performed extensively for other protocols, supported by the Mediant 2000 and the MediaPack, such as with SIP proxy servers and other VoIP gateways[7]. AudioCodes invests a great deal of effort in interoperability testing with all elements of the EVN solution in order to ensure customers that their VoIP network will start up more quickly and operate more smoothly.

Conclusion

AudioCoded™ EVN solutions provide customers a full solution for enabling VoIP in their networks while protecting their present investment in IT equipment. AudioCodes' **Mediant 2000** and **MediaPack** VoIP gateways combine a flexible range of port densities, call-carrying capacities, high voice quality, and low latency in a small footprint. They are a cost effective blend of multi-featured digital and analog VoIP gateways, purpose-built for the enterprise market, thus providing a compelling choice today for enabling Enterprise VoIP Networking.

Ref. #LTRT-00425 10/02/ V.1

[6] Contact AudioCodes regarding availability

[7] For specific details regarding interoperability, testing and test configurations, contact AudioCodes