White Paper

AudioCodes VoIP Networking for Enterprises: Migrating to the New Voice Infrastructure

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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 Voice over IP (VoIP). VoIP is integrated into the enterprise network non-intrusively (or in "hybrid" mode) by acquiring VoIP-enabled network equipment, such as media gateways, without the prohibitive cost of upgrading or replacing the existing telephone or LAN equipment. In other words, only a suitable media gateway needs to be added to existing equipment, as opposed to replacing the entire IT infrastructure with VoIP-enabled routers, VoIP-enabled PBXs or pure IP-PBXs, and IP telephone terminals. Today it is necessary to keep costs down. Therefore, getting more value out of the current 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 IT managers appreciate the need for investing in converged TDM/IP enterprise solutions is reflected by the fact that in 2004, the Converged Telephony (TDM/IP) market value alone was $1.4 billion, whereas in 2009, the forecasted market value is $2.3 billion[1].

Architecture

AudioCodes VoIP Networking for Enterprise is made possible with a digital VoIP gateway (such as AudioCodes **Mediant™ 1000**) and analog VoIP gateways, (such as AudioCodes' **MediaPack™ FXS/FXO** series), which provide IP connectivity in the infrastructure of the legacy corporate (PBX) environment and also for connecting small branches with only analog terminals (e.g., telephone or fax) or key telephone systems (KTS) 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 3rd Quarter 2005 Enterprise VOIP Worldwide Forecast*

Figure 1: Present Enterprise Network

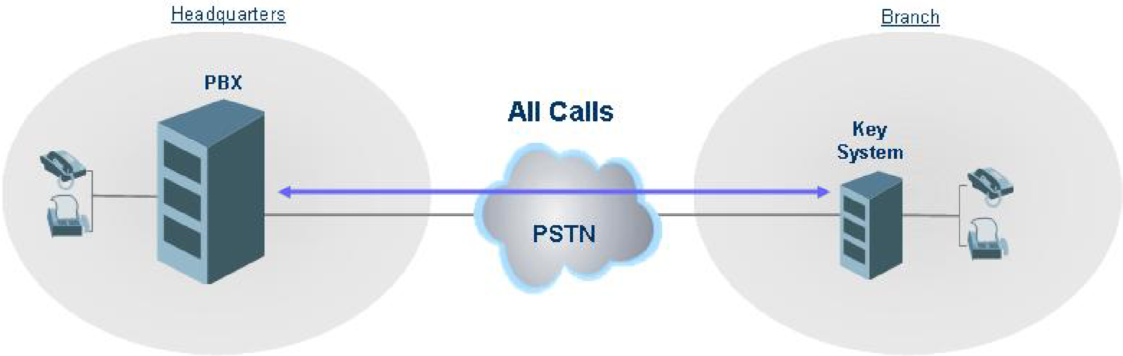
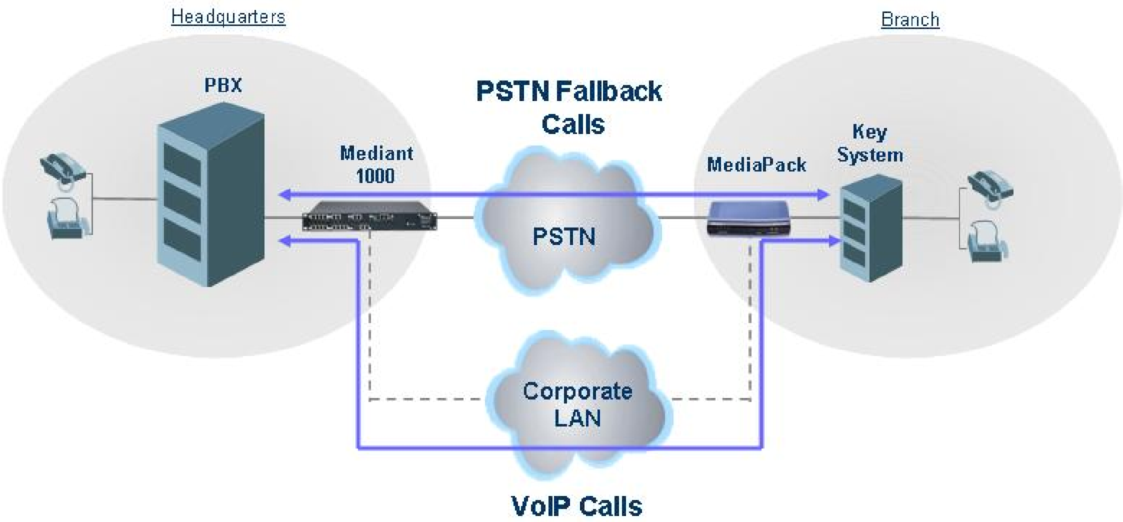


Figure 2: VoIP Based Enterprise Network



Four Major Benefits of Enterprise VoIP Networking

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

Savings

The existing multi-site data network will be used for connecting local and remote branches over the data 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 other real-time data applications. Since the calls are routed over the data network the cost model is normally 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. In addition, the cost of leased (T1/E1) telephony lines can be reduced (except for access to emergency services or fallback to the PSTN). 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, note that 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 calling the nearest branch from their residential VoIP gateway or via the local PSTN and via an FXO or trunk-based media gateway. The call will be routed via the VoIP network, dramatically improving productivity at little or no extra cost. Calls from one branch to another can be made as if both subscribers are located in the same branch. Employees will be encouraged to communicate verbally (as opposed to electronic mail) with other employees more often, since they can expect ease of dialing combined with the knowledge that these calls to their colleagues will not be billed as long-distance calls.

Increased Availability & Enhanced Features

VoIP combined with the PSTN (in hybrid mode) offers a reliable alternative to pure legacy PSTN, allowing communications even during a local disaster (e.g., “9/11”). In addition, by adding at least one call server (e.g. typically a SIP Proxy), 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. For added reliability the media gateway will also serve as a PSTN gateway, offering alternative routing via the PSTN in case of IP network failure or degradation.

Reliability (Routing and Fallback)

If connectivity is degraded or lost with the IP network, the Mediant 1000 (E1/T1 or FXO) and the MediaPack (FXO) will automatically continue to route calls via the PSTN using the Phone to IP Routing function until the IP connectivity is restored.

The Mediant 1000 and the MediaPack can function completely independently of a Softswitch (gatekeeper or proxy), by using the Phone to IP Routing function, which enables the gateway 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).

The Mediant 1000 and the MediaPack support “lifeline” switching. In case of power or network failure, they will switch calls to the PSTN/TDM interface seamlessly, thereby enabling new call setups to be routed by the PBX via the PSTN.

Mediant™ 1000

Features

The **Mediant™ 1000** is a low-density VoIP gateway, appropriate for enterprise applications. The Mediant 1000 can be used with two types of TDM signaling interfaces: CAS or PRI. The Mediant 1000 is a sensible choice for customers who require 1 to 4 trunk spans or up to 24 analog FXS or FXO line interfaces for dedicated voice applications. Specifically, the Mediant 1000 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 1000 offers scalability – from 1 to 4 E1/T1 spans and up to 24 analog lines. The Mediant 1000 offers the full range of VoIP and TDM signaling protocols. 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.

The Mediant 1000 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 voice module replacement
- Dual Ethernet interface to LAN
- Optional dual/redundant power supplies

The Mediant 1000 provides the OSN Server, an optional processor module and single or dual hard disks that can be used for a third party application. The OSN Server can be used for general applications such as:

- IP-PBX
- Gatekeeper
- SIP Proxy
- RADIUS server
- Application server

Size and Scalability

- 1, 2, or 4 E1 or T1 spans for connection to PBXs or to the PSTN network
- Up to 24 analog ports for connection to POTS and fax equipment (FXS) or to the PSTN network (FXO)
- Up to 120 simultaneous channels
- Exceptionally compact form factor: 1 Rack Unit high, installed in a 19-inch rack

Best-of-breed VoIP and DSP Technology

- High traffic handling ability
- Low latency
- Echo cancellation (G.168-2002, tail length of 32, 64, or 128 msec)
- Dynamic jitter buffer

- Robust and reliable (redundant 10/100 BaseT LAN connection, optional redundant power supply)
- Broad support for industry standard vocoders: G.711, G.723.1, G.726, and G.729A, selectable per channel.
- Silence suppression with Comfort Noise Generation
- Easy to use Web interface for setup and configuration of VoIP/PSTN parameters
- Highly scalable equipping of E1/T1 spans or analog lines 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 or mixed FXS/FXO ports)
- Supported vocoders: G.711, G.723.1, G.726, G.729A, selectable per channel.
- 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.168-2002, tail length of 32 msec)
- Dynamic Jitter Buffer
- Short and long haul
- Lifeline support
- Outdoor protection
- Message Waiting Indication (MWI) support
- Polarity reversal
- Metering pulses (12 and 16 kHz)
- Distinctive ringing

Control (Mediant™ 1000 and MediaPack™)

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 1000 supports many variants of PSTN signaling protocols, such as:

- ISDN PRI protocols, such as Euro-ISDN, 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, T1 “robbed-bit”, as well as “E911” CAMA signaling.

The Mediant 1000 and the MediaPack support a broad range of standard VoIP signaling protocols:

- ITU-H.323, version 4, VoIP signaling and control protocol
- MGCP (IETF RFC 2705)
- MEGACO (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 1000

The Mediant 1000 and MediaPack Series support the following analog signaling interfaces:

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

The Mediant 1000 and MediaPack Series supports a RADIUS interface, which can be used with third-party billing application servers.

Management

The Mediant 1000 and the MediaPack are configured using a friendly Web interface. System parameters related to the TDM and IP networks can be configured online; otherwise they are defined according to defaults provided by the INI file. The AudioCodes EMS, using on a PC-based or alternately on a UNIX-based management platform, supports configuration and management of multiple Mediant 1000 and MediaPack gateways.

Security

The Mediant 1000 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". As well, user logins can be protected by using an external RADIUS server. The Web interface can also be disabled or configured as "read only" (i.e., for monitoring only). As well, encryption of control protocols (e.g., SIPS or H.235), management protocol (e.g., IPSEC for SNMP), and voice (e.g., using SRTP for SIP) is supported.

In addition, IP access to the Mediant 1000 and MediaPack gateways is limited to a list of IP protocols, port ID's or authorized IP addresses, (e.g., other gateways or gatekeepers). NAT traversal is supported using the STUN standard. In addition, different IP addresses (Public and Private) can be used to support working with static NAT servers.

Interoperability

The Mediant and the MediaPack Series have undergone extensive testing for interoperability with various vendor equipment (e.g. Cisco, HP, Nortel, Siemens, 3Com, Alcatel, RADVISION, NetCentrex, etc.). As well, many types of analog or digital PBXs have been successfully tested and deployed with the Mediant and MediaPack series. Interoperability has also been performed extensively for the standard VoIP protocols, supported by the Mediant 1000 and the MediaPack (SIP, H.323, MGCP, MEGACO, SIGTRAN, etc.). AudioCodes invests a great deal of effort in interoperability testing with all elements of the enterprise solution in order to ensure customers that their VoIP network will start up more quickly and operate more smoothly.

Conclusion

AudioCodes VoIP Networking for Enterprise solutions provide customers a solution for enabling VoIP in their networks while protecting their present investment in IT equipment. AudioCodes' **Mediant 1000** 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.

About AudioCodes

AudioCodes Ltd. (NASDAQ: AUDC) enables the new voice infrastructure by providing innovative, reliable and cost-effective Voice over Packet technology and Voice Network products to OEMs, network equipment providers and system integrators. AudioCodes provides its customers and partners with a diverse range of flexible, comprehensive media gateway and media processing technologies, based on VoIPerfect™ – AudioCodes' underlying, best-of-breed, core media gateway architecture. The company is a market leader in voice compression technology and is a key originator of the ITU G.723.1 standard for the emerging Voice over IP market. AudioCodes' voice network products feature media gateway and media server platforms for packet-based applications in the converged, wireline, wireless, broadband access, and enhanced voice services markets. AudioCodes enabling technology products include VoIP and CTI communication boards, VoIP media gateway processors and modules, and CPE devices. AudioCodes' headquarters and R&D facilities are located in Israel with an R&D extension in the U.S. Other AudioCodes' offices are located in Europe, the Far East, and Latin America.

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