Application Description

AudioCodes' Enterprise VoIP Applications Description

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AudioCodes

Enterprise VoIP

Applications Description

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1. General

This paper describes how VoIP applications can be easily implemented by AudioCodes' MP-1xx Analog Media Gateway product line together with the Mediant™ 2000, AudioCodes' digital gateway.

Note that all the presented applications in this paper are exclusive of the VoIP PBX networking solution which is a major enterprise market application. The PBX networking solution is referred to by AudioCodes as EVN (Enterprise Voice Network)* and utilizes AudioCodes' analog and digital media gateways, the AudioCoded EMS, as well as appropriate third party Call Agents, Gatekeepers, Proxy Servers respectively, all of which have undergone Interop testing with AudioCodes' media gateways. (A current list of interoperable vendor equipment is available from AudioCodes upon request.)

* Please refer to AudioCodes EVN Solution Description entitled "AudioCodes Enterprise VoIP Networking (EVN) - Migrating to the New Voice Infrastructure", available on AudioCodes' Website: http://www.audiocodes.com/Main_ID55_1.html.

2. Support for Remote Extension

AudioCodes MP-1xx enables distributed enterprises to provide their employees working from small offices or small branches with direct connections via the IP data network to the corporate headquarters' PBX. In addition to cost-savings related to the installation and PBX maintenance on each remote location, the MP-1xx also enables unified phone services to all enterprise employees. **Figure 1** as follows presents a typical installation of the remote extension solution. Each FXO port is mapped to a specific FXS (extension) port.

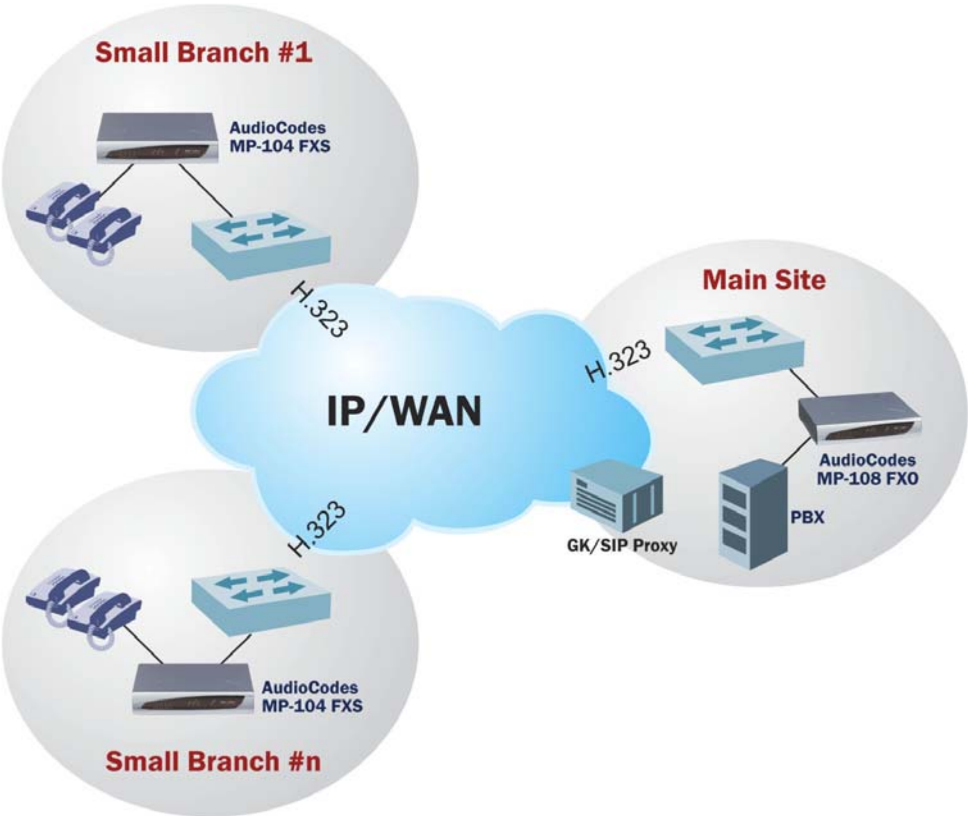

Figure 1 – Support for Remote Extension

Main Benefits:

- Enable enterprise customers to lower their investment costs related to PBX equipment and phone services at their remote branches
- Enhance the phone services to remote employees by providing them direct connectivity to the Headquarters' PBX.
- Improve the level of service (e.g., subscriber features) to enterprise customers
- Provide customers with a simple entry-level application and the confidence to gradually migrate to the new VoIP technology
- Implementation can start with a small investment to reduce the risks
- Simple to install and configure:
    - One-to-one routing
    - Internal Routing
    - Define POTS services to remote user (in PBX)

3. DID/DDI (Direct Inward Dialing) Support for KEY Systems

DID is a PSTN service enabling routing of PSTN inbound calls directly to the PBX subscriber extension. Without DID services, inbound calls are routed to the PBX subscriber extension by a human operator or by a simple IVR application called an auto-attendant. With the DID service the PBX can utilize special PSTN trunks (ISDN BRI or PRI, or CAS trunks), which are expensive and not normally supported by small PBXs, KEY systems, or telephone sets that are connected directly to the PSTN.

Figure 2 below presents a cost-effective VoIP-based DID solution for KEY systems:

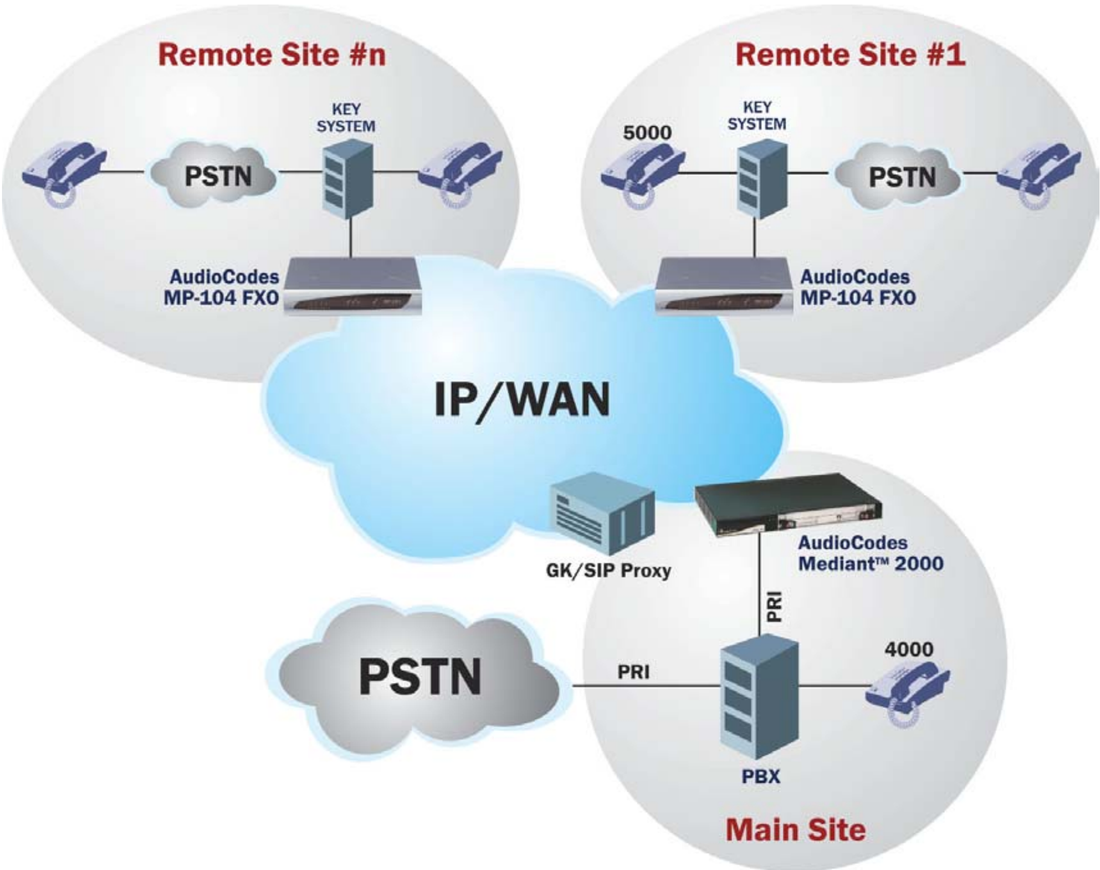

Figure 2 – DID Support for KEY System

The architecture presented above can easily be implemented by any distributed enterprise, which has a PBX in the main site and many KEY systems in their remote offices. A Mediant™ 2000, installed at the main site, is connected to the site's PBX or directly to the PSTN via PRI trunks, which support the required DID service. Each extension at the main site, as well as at the remote sites, is provided with a specific PSTN number (i.e., 03-5394000 for extension 4000 at the main site and 03-5395000 for extension 5000 at site 1). The KEY systems at the remote site are connected via an extension interface to AudioCodes' MP-104/8-FXO. AudioCodes' MP-104/8 is connected to the enterprise data network to which the Mediant™ 2000 is also connected at the main site. When a PSTN user dials, for example, 03-5395000, the call is routed by the PSTN to the Mediant™ 2000 in the enterprise main site.

Based on the third party control and management servers/GK/SIP proxies (or optional internal routing table), the call is routed over the enterprise data network to the AudioCodes' MP-104/8 in site 1, which is connected to the KEY system via one of its analog extensions (FXO). Once the MP-104/8 receives the call, it goes off-hook, discards the first 5 digits and dials 5000, the call being routed directly to the required extension as with normal (PSTN-based) DID service.

Main Benefits:

- Enables normally unavailable DID service for KEY system users without additional lines or trunk connections to the PSTN
- Optional internal routing table
- Single area dialing plan enabled country-wide or world-wide.
- The large Install Base of KEY systems that presently do not benefit from DID service presents a sales opportunity for AudioCodes' customers.
- Can be sold to distributed (multi-branch) enterprises
- Can be sold to operators who can provide this service to customers who own KEY systems

4. Centralized Cellular Gateway

The access fee that enterprises are required to pay to wireline operators, whenever they call cellular subscribers, is a common reason to install a cellular gateway next to the enterprise PBX. The cellular gateway allows them to bypass wireline operators and route calls directly to the cellular operator, thus incurring charges for “air time” only. The cellular gateway is connected to the PBX by FXO interfaces and is seen by the cellular operator as a standard cellular phone. Today distributed enterprises are required to install cellular gateways on each of their locations so that the overall solution is expensive. Furthermore, in many cases, calls to cellular subscribers are routed via wireline operators because all the cellular gateway ports on-site are engaged with previous calls. VoIP technology enables routing of calls over the enterprise data network from site to site. This enables a relatively large cellular gateway in one central location to serve several sites, thus reducing the number of required cellular gateways. This also enables routing of calls via a remote cellular gateway if all the ports of the local cellular gateway are busy.

Figure 3 below presents a typical implementation of a central cellular gateway in a distributed enterprise environment. At the remote sites, the FXO gateways support inbound calls from the cellular gateway to the PBX, while the FXS gateways support outbound calls from the PBX to the cellular gateway. AudioCodes’ proven integration with other cellular gateway vendors, as well as Gatekeeper vendors, enables AudioCodes to offer enterprises completely compatible solutions. (A current list of interoperable cellular gateway vendors is available from AudioCodes upon request).

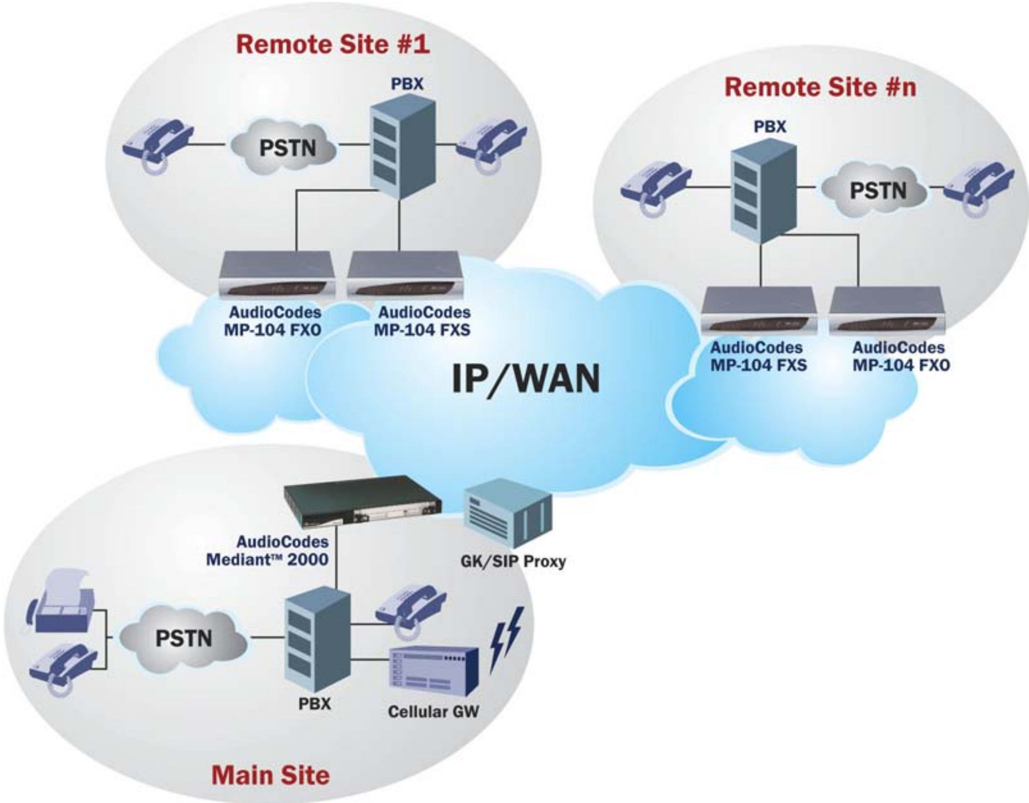

Figure 3 – Centralized Cellular Gateway

Main Benefits:

- Reduce the number of required cellular gateways
- Reduce access costs related to calls to cellular subscribers via the wireline network
- Easily established short Return of Investment (ROI)
- AudioCodes provides end-to-end integrated working solutions
- Attractive to distributed enterprises.
- Attractive to “call terminators” (IP telephony “POP”s); AudioCodes’ FXO and digital gateway can be connected directly to the cellular gateway, saving the money call terminators would be required to pay for a PBX in their POPs

5. Centralized Enterprise Voice Application (Unified Messaging/ IVR)

Today, many enterprises are operating as global organizations serving their customers all over the world with offices and branches at different cities, states and countries. The main challenge of these enterprises is to build their operations and design their enterprise infrastructure in a way that will preserve the distributed nature of the organization, also allowing it to be managed as one operation. These global organizations will find attractive any solution that will enable their distributed operation be run as one, well-managed operation.

In most enterprise implementations, a dedicated voice application server is installed on each site. The site server is connected only to the site's PBX and serves only that site's subscribers. Retrieving messages from remote location requires paying for long distance or international calls. Transferring calling customers from site to site also requires initiating long distance or international expensive PSTN calls.

In order to resolve the above limitations, enterprises are moving toward a centralized solution, using IP-based voice communications. VoIP technology enables enterprises to install the voice application server over the (data) network to serve all the enterprise's distributed sites that have some sort of connection to the enterprise IP network. Supporting all enterprise branches with just one system ensures that all the above issues are solved.

The MP-1XX and Mediant™ 2000 are used to connect the PBXs at the remote site to the IP network. In the central site, the media gateways are used to connect the centralized legacy voice server to an IP network establishing a virtual voice connection from each of the remote sites to the central location where centralized voice server is located. (The architecture is outlined in **Figure 4 below):**

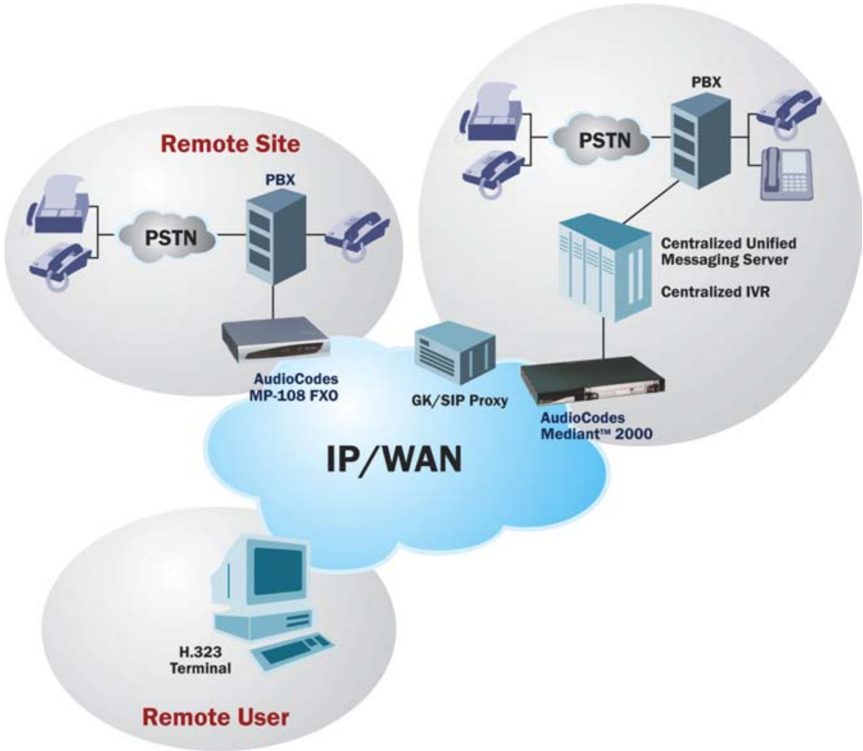

Figure 4 – Centralized Enterprise Voice Application

Main Benefits:

- One centralized system supporting all enterprise offices and branches
- User can access services via legacy or IP-enabled servers
- Reduced management and configuration costs
- Smooth migration for legacy voice server vendors who want to upgrade their solution to support distributed enterprise needs

6. Support for Analog Terminals and Analog PSTN lines in iPBX Environment

AudioCodes' MP-104/108/124 Analog Media Gateways provides iPBX customers with a cost-effective solution that connects the iPBX systems with local and remote analog telephone sets, modems and fax machines, as well as local analog PSTN lines over the customer installed telephony wiring infrastructure. The MP-104/108/124 covers a range of configurations from 4 to 24 FXS ports and 4 or 8 FXO ports. AudioCodes' Analog Media Gateway product line supports a variety of telephony-related features that allow for a smooth replacement of the legacy PBX with the new distributed iPBX architecture. ***Figure 5*** as follows describes a typical installation of AudioCodes' MP-1xx product line in an iPBX-based, distributed environment.

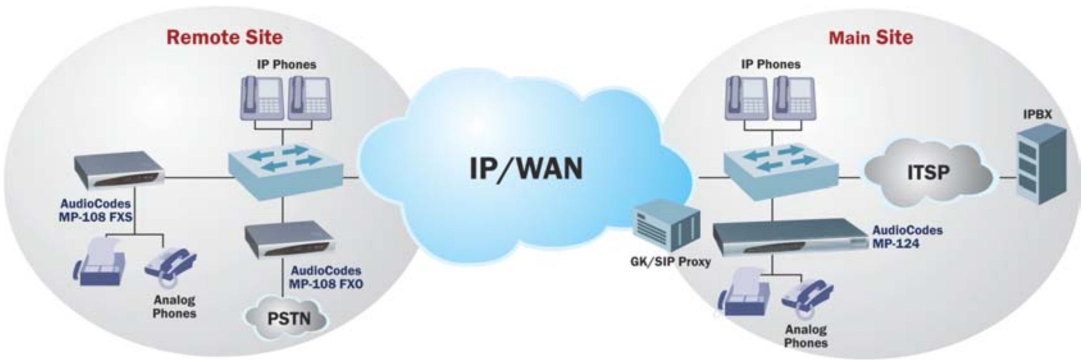

Figure 5 – Support for Analog Devices in iPBX Environment

Main Benefits:

- Support for legacy voice, fax and modem services
- Enhanced call services as Hold and Transfer provided by iPBX (e.g., via SIP or H.323 VoIP signaling)
- Message waiting indication
- Polarity reversal, e.g., to support disconnect and answer supervision features
- CLI generation and detection
- Supports T.38 Fax Relay
- Supports all standard VoIP control protocols (MGCP, MEGACO, SIP and H.323)
- FXO enables local analog connection to PSTN to meet 911 regulations
- Interoperability with major iPBX vendors (a current list of interoperable vendor equipment is available from AudioCodes upon request).

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 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.

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