Application Description

AudioCodes Calling Card Feature

Powered by AudioCodes
Mediant™ 2000 and MediaPack™ Gateways

Version 2
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AudioCodes

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Benefits for Internet Telephony Service Providers

- Reduce costs and deployment time by choosing VoIP technology over switched circuit telephony (TDM)
- Use standard E1/T1/J1 and loop-start FXS/FXO PSTN interfaces to access international carriers, as well as IP interfaces and RADIUS (Remote Authentication Dial-In User Service) protocol, the industry standard for remote authentication, authorization and accounting to interface with an external accounting server
- Provide same level of service and convenience to users as with TDM-based solutions
- Achieve voice quality comparable to toll quality
- Acquire a reliable VoIP network infrastructure, which is cost-effective due to bandwidth savings and offers better economy of scales than TDM-based solutions
- Lower the cost (CAPEX) and deployment time that a TDM-based calling-card service would require
- Increase call revenues from additional callers who will be attracted by the convenience of the additional service
- Interoperate with other VoIP service providers' networks and other vendors' VoIP equipment

About the Mediant™ 2000 and MediaPack™ VoIP Gateways

The **Mediant™ 2000** VoIP Gateway is the cost-effective, entry-level member in the AudioCodes family of market-ready, standards-compliant, digital media gateway voice network products. Supporting 1 to 16 E1/T1 spans and intelligently packaged in a stackable 1U chassis especially designed for small, medium or remote locations, the Mediant 2000 is the right-sized solution for various market needs, such as Enterprise Networking, Contact Centers, Toll Bypass, VoIP Trunking and IP-Centrex. The Mediant™ 2000 VoIP Gateway can be deployed in IP core network solutions for many applications including PBX Access Trunking Gateways, IP Centrex Gateways and Broadband Access Gateways.

AudioCodes series of **MediaPack™** Analog VoIP Gateways are cost-effective, stand-alone gateways, providing superior voice technology for connecting legacy telephone or fax equipment and PBX systems with IP-based telephony networks, as well as for integration with new IP-based PBX architecture. Featuring 2, 4, 8 and 24 ports with FXS or FXO interfaces, the MediaPack™ provides a comprehensive offering. Field-proven voice and fax technology and feature-rich design makes the MP-series an excellent solution for various emerging VoIP applications, such as the voice VPN environment, centralized IVR and Quality Monitoring, calling card and pay phone markets. Ideal markets also include MTU (Multi Tenant Units), rural areas and remote analog extensions in a VoIP-based PBX or IP-PBX architecture.

AudioCodes **Mediant 2000™** and **MediaPack™** are based on the VoIPerfect™ architecture, AudioCodes' underlying, best-of-breed core media gateway technology for all of its products. They are part of AudioCodes enabling technology and voice network products, of which more than 10 million Voice over Packet (VoP) ports have been deployed in over 75 countries around the world.

Figure 1: Mediant™ 2000 VoIP Gateway

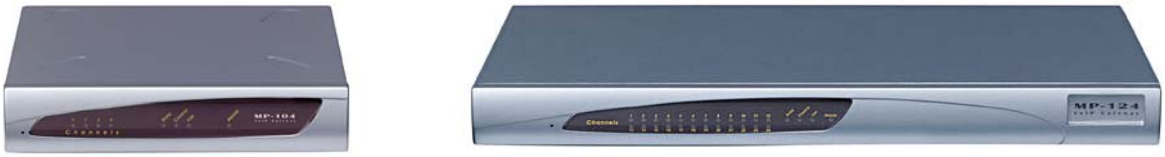

Figure 2: MediaPack™ VoIP Gateways

Overview

One of the many attractive uses of IP telephony is to utilize the IP network for calling card applications, such as pre-paid and post-paid calling cards. The Mediant 2000 (when equipped with the optional IVR - Interactive Voice Response - feature) enables Internet Telephony Service Providers (ITSPs) to provide VoIP telephone service to users who have purchased calling cards in advance or have otherwise subscribed to a calling card service. The Mediant 2000 provides an integrated calling card solution, while providing a seamless and reliable telephone experience using VoIP.

The market for calling cards is growing exponentially worldwide. Calling cards are often cheaper than collect calls and operator-assisted calls made through long distance providers and local phone companies, not to mention mobile phone companies. In addition, calling cards provide easy access to telephones or payphones, for users away from home. Judging from the great number of people who do not own a telephone, for example, in countries with a poor telephony infrastructure, combined with the growing number of users who do not use credit cards (or cannot due to credit limitations), the calling card is fast becoming the chosen method for many. In addition, a growing number of businesses are issuing calling cards for their employees in order to better manage their employees' telephone expenses and to benefit from group discounts offered by the chosen service provider.

The IVR application generally exists at the edge of the VoIP network (i.e., in the PSTN gateway). Therefore, only calls that have been authorized are set up, thus reducing the load on centralized resources (such as the gatekeeper) due to unauthorized callers or wrong numbers. ITSPs, using AudioCodes Mediant 2000 and MediaPack media gateways for calling card services, can further increase their profit margins and reduce subsequent costs, by taking advantage of excess capacity in the IP network. This is especially true for operators already taking advantage of VoIP for telephony applications, such as toll bypass. More call minutes can be added to the network by attracting new subscribers, while offering existing subscribers added services.

Network Architecture

The figure below illustrates a standard architecture for IVR-based Calling Card applications. The figure depicts in general terms an incoming PSTN→IP call (IVR in the IP direction is also supported) being conveyed to the IP network.

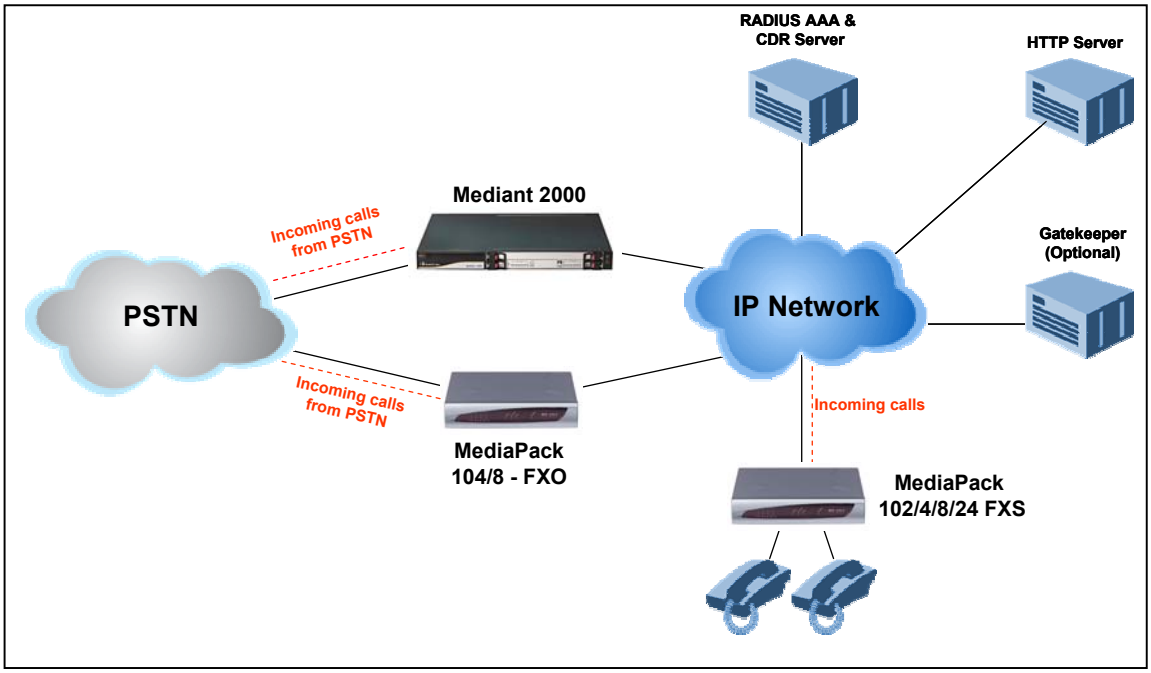

Figure 3: Typical Calling Card IVR Application

The above figure is comprised of the following physically distinct components:

Mediant 2000 - AudioCodes' digital Media Gateway that interfaces the PSTN and the IP network. It includes the VoiceXML (VXML) Interpreter which generates events in response to user actions (DTMF input) and system events (e.g. timer expiration). These events are acted upon by the VXML interpreter itself, as specified by the VXML document. Voice prompts (announcements) are stored on gateway (flash).

MediaPack – AudioCodes' Analog Media Gateway that serves both as an interfaces between the PSTN and the IP network and an interface for legacy phones. The MediaPack provides an excellent solution for PSTN interfacing where low density interconnection is required and where digital interfaces are unavailable.

HTTP Server – (Document Server) sends out the VXML script (IVR scenario) in response to the Mediant 2000 (the VXML Interpreter) request.

AAA server – A centralized Authentication, Authorization and Accounting server for remote access users that communicate via the RADIUS protocol. Call Detail Records (CDR) and subscriber database, e.g. for authentication, are stored here.

Gatekeeper (GK) – Standard call management server for H.323 networks performs essential control, administrative, and call management functions. A gatekeeper is not required when AudioCodes' internal routing feature is used (supported on Mediant 2000 and MediaPack products).

Pre-paid Call Scenario (example)

1. An incoming PSTN call with a published access number (typically a toll free number) reaches the Mediant 2000 or the MediaPack.

2. The Mediant 2000 (or the MediaPack) accepts the call (sends Ring Back Tone and/or Alert message).*

3. The Mediant 2000 searches its internal digit manipulation table for the specific prefix (of the published access number). When it is detected, if it corresponds with the predefined call scenario parameter, the Mediant 2000 determines that the incoming call is a "pre-paid" IVR call.

4. The Mediant 2000 accesses the "pre-paid" script (if the VXML script file is not presently resident, then it downloads it from the outbound HTTP Server and stores it in memory). The Mediant 2000 sends Answer/Connect message.*

5. The Mediant 2000 starts by playing an initial voice message. This message is composed of an opening greeting and an interactive menu, typically asking the caller to press '1' to make a call, '2' for access to help menu, '3' to request operator service, etc.

6. After pressing the digit '1', the caller is immediately prompted to enter his/her account number (usually the card number) and his password or PIN (Personal Identification Number). The Mediant 2000 collects the input DTMF digits and sends an Authentication message to an outbound AAA server.

7. Only after the call is authenticated successfully, the caller is asked to enter the destination number he/she wishes to reach. The Gateway collects the input DTMF digits and sends an Authorization message to the AAA server.

8. The AAA server determines if the caller is authorized to proceed with the call and specifies the maximal duration of the call; the Mediant 2000 routes the call to the IP network and starts an internal timer.

9. A short time (one minute) before the account's credit is exhausted, based on the authorized duration of the call, an announcement or tone is played; finally, a minute later, the call is disconnected and the final voice prompt is played.

10. After the conclusion of the call, the Mediant 2000 sends an accounting message to the AAA server containing the call details (CDR) and prompts the caller either to proceed with another call or to disconnect.

* When the call is connected by the MediaPack, the MediaPack forwards the call to the Mediant 2000 for IVR processing.

Mediant 2000 IVR-based Calling Card Features

- Supports up to 480 concurrent calls (scalable from 1 to 16 trunk spans), running on the same or different scripts
- PSTN and IP access – Supports calls from the PSTN to IP network and IP network to the PSTN
- Distributed IVR architecture – IVR functionality can be located at the edge of the network (distributed functionality) or in a central location for various applications, such as calling card service
- Connect with the PSTN over standard carrier (PRI or CAS) T1, E1, or J1 interfaces
- Connect with the PSTN or analog telephones over standard loop-start FXO or FXS interfaces, respectively, using MediaPack analog gateways
- Interoperate with standard RADIUS-based (Remote Authentication Dial-In User Server/Service) AAA servers to perform Authentication, Authorization, Accounting and to send and store billing records (or CDRs)
- Post-paid applications (H.323-billing-model: credit)
- Pre-paid application (H.323-billing-model: debit). When credit is exhausted the call is disconnected (a short Voice prompt is played prior to disconnection)
- On-board, voice prompts (in flash memory) for all VXML scripts
- Multiple prefix access (for defining multiple services with different branding or multiple operator access)
- Multiple VXML scripts stored on external HTTP server (up to 10 different scripts)
- Downloads the VXML scripts once and stores them in the RAM; scripts can be changed “on the fly”
- Successive calls without having to re-enter user account information, saving the caller additional access charges and without having to hang up, redial, and re-authenticate
- Barge-in dialing so that the caller can enter menu options during playing of voice prompts (to shorten menu access time)
- Flexible (user-defined) call scenario scripts and voice prompts

For up-to-date and more detailed information about AudioCodes VoIP Gateways including marketing brochures and related press releases, visit AudioCodes’ Website at http://www.audiocodes.com/Main_ID138_1.html.

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.

AudioCodes 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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