

AudioCodes and MailVision Application Description

Call Back Server

Powered by
AudioCodes Mediant™ 2000 VoP Media
Gateway and MailVision's B2BUA Call Back
Application

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Application Description

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Overview

MailVision has successfully designed, developed and deployed several voice products and services solutions based on the SIP protocols and open standards interoperating with other components such as voice gateways, SIP proxies, softswitches and VoIP clients. The main characteristics of the Value Plus-SIP Value Added Services- Application are: Telephony Recording over IP, VoIP Conferencing, SIP and IVR Middleware for Prepaid and Call Back application, IP Voice Mail and VoiceXML IVR.



Figure 1: Mediant™ 2000 VoP Media Gateway

About the Mediant™ 2000 VoP Gateway

AudioCodes Mediant 2000 is based on the VoIPerfect™ architecture, AudioCodes' underlying, best-of-breed core technology for all of its products, presently supporting close to 10 million VoP ports deployed worldwide. The Mediant™ 2000 VoP Gateway is the cost-effective, entry-level member in the AudioCodes family of market-ready, standards-compliant, media gateway voice network products. Intelligently packaged in a stackable 1U chassis especially designed for small 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.

System Capabilities

Users will register to the system via the Web where they will be provided with an access number.

Registered users will be offered the following capabilities:

- Unlimited multi user Call Back system – based on user DID detection.
- Unlimited multi user Call Through system – User identification by CID or Password.
- Management tool – based on customer's Intranet (BUI).
- IVR (Interactive Voice Response).
- Sequence of calls.
- User's Phone Book – Short-dial numbers to prioritized destinations
- Interface with integrative billing system (CDR).
- Simple and convenient system operation.
- E1/T1 connectivity to the PBX infrastructure or PSTN switch
- Connectivity to IP-PBX or IP network via standard SIP protocol
- National Call Back – User identification by CID detection
- SMS Call Back (optional)
- Call Through – User identification by CID detection or password
- Provisioning (Multi Tenant)
- Defines period of use for each user
- Interface with integrative billing system
- Reports

Method of Operation

The system will be comprised of a Linux Server, which will also be a Web server. Clients will be able to login to the server via the Web in order to register, obtain pin number, and to perform administrative tasks.

The AudioCodes Mediant™ 2000 VoIP Gateway will be used as the PSTN Access Gateway (E1/T1 – PBX interface).

The system architecture is based on IP, RTP and SIP for communication. The platform is modular and can scale from 15 call back sessions for the enterprise market to thousands of sessions based on the same platform.

Incoming and Outgoing Calls

A trunk (E1 or T1) from the telephone company or the PBX will be connected directly to the AudioCodes Gateway. This trunk will be used for incoming and outgoing calls to and from the system. A set of DID numbers will be allocated by the system to subscribers calling the system.

Call Back Server Administrator

A system administrator with high priority privileges will be able to login to the system. The administrator's task includes maintaining the list of users allowed to use the system as well as updating system parameters .

Hardware Requirements

	QTY	Description	Redundancy
Entry Level			
MP104/108 Or Mediant™ 2000 VoIP SIP Gateway	1	4/8 ports 30 ports	N+1
Intel Linux server	1	ProLiant DL140 Intel® Xeon™ Processor 2.40GHz/533MHz- 512KB, 512MB (Rack) 80GB AT	Dual power supply, N+1
Small ASP /Large Corporate			
Mediant™ 2000 VoIP SIP Gateway	1	24/30 –192/240 ports	N+1
Intel Linux Server	2-3	ProLiant DL360 G3 Intel® Xeon™ Processor 2.40GHz/512KB (533MHz FSB), 512MB	Dual power supply, Raid 5, N+1
Telco's/Large ASPs			
Mediant™ 2000 VoIP SIP Gateway	2-N	768-960 ports	N+1
Sun Fire V210/240 Server	3		N+1
IP BPX			
Intel Linux server	1-2	30-120 concurrent users	Dual power supply, N+1
Soft Switch			
Sun Fire V210/240 Server	3	Up to 1000 concurrent users	N+1

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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About MailVision

MailVision develops and markets enhanced services platforms for Wireless service providers, Telco's and NexGen communications service providers. The MailVision value-added service platform, "Value-Plus", is the platform of choice for operators seeking rapid scalable deployment of revenue-generating new services over existing networks. The MailVision SIP Server is a call-control software package that enables service providers to build scalable, reliable packet-voice networks. MailVision's unique IP-based architecture directly integrates into a VoIP network infrastructure forging the convergence of Internet and telephony and delivering cost-effective, revenue-producing enhanced communications services. Founded in 1997, MailVision has offices in Sunnyvale, California and an R&D center in Haifa, Israel.

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