AudioCodes CPE & Access Gateway Products

Mediant™ 1000 VoIP Media Gateway

- Employs AudioCodes VoIPerfect™ technology for outstanding voice quality
- Scalable “pay-as-you-grow” modular architecture
- Rich offering of digital (E1/T1/J1), analog (FXS/FXO), and BRI interfaces
- Cost-efficient for low density gateways
- Life-line fallback to PSTN in case of power failure or network degradation
- PSTN fallback for assured connectivity
- Internal OSN Server for hosting 3rd party application
- An ideal match as a platform for IP-PBX
- Media processing and conferencing option
- Stand Alone Survivability (SAS) for service continuity

AudioCodes

The **Mediant™ 1000** is AudioCodes’ cost-effective, converged wireline VoIP media gateway. Intelligently packaged in a stackable 1U chassis, it is designed to interface between TDM & IP networks in enterprises or small-scale carrier locations. Incorporating AudioCodes’ innovative Voice over Packet technology, the Mediant 1000 enables rapid time-to-market and reliable cost-effective deployment of next-generation networks.

The Mediant 1000 is based on VoIPerfect™, AudioCodes underlying, best-of-breed, media gateway core technology for all of its products. The Mediant 1000 provides superior voice-technology for connecting legacy telephone and PBX systems to IP networks, as well as seamless connection of the IP-PBX to the PSTN. In addition to operating as a pure media gateway, the Mediant 1000 can also host partner applications and serve as an IP-PBX platform. The Mediant 1000 is fully interoperable with multiple vendor gateways, softswitches, gatekeepers, proxy servers, IP phones, Session Border Controllers and firewalls.

Scale Up As Your Business Grows

The Mediant 1000 matches the density requirements for small locations while meeting enterprises and service providers’ demands for scalability. The compact Mediant 1000 Modular Gateway is extremely scalable and supports multiples of 1, 2, or 4 E1/T1/J1 spans, 4 to 20 BRI ports or 1 to 24 analog ports in various FXO/FXS configurations. The Mediant 1000 also supports mixed digital/analog with media processing capabilities such as conferencing, play/record configurations.

The Mediant 1000 can support a variety of telephony interfaces. The digital module can be configured as regular E1/T1/J1 interfaces, with up to 1 or 2 paired spans acting as life-line interfaces for switching to the PSTN in case of power failure or network problems. The analog module is available as regular FXS or FXO interfaces, where 1 FXS line can be used as a life-line interface for switching to the PSTN.

Interface Modules:

- Digital (E1/T1/J1) – connecting the PSTN or PBX to the IP-network
- Analog FXS – connecting analog phones and fax machines to the IP-network
- Analog FXO – connecting analog lines from the Central Office (CO) or PBX to the IP network
- BRI – connecting to PBXs or the PSTN

SAS - Stand Alone Survivability for Service Continuity

Customers who connect to centralized IP-Centrex services, as well as branch offices of enterprises who use a centralized IP-PBX server may face a survivability challenge. Stand Alone Survivability (SAS), supported in the Mediant 1000 is based on the SIP B2BUA (Back to Back User Agent) functionality, and enables the backup of SIP clients such as SIP IP and Soft Phones in the case of a connectivity failure with the centralized SIP server.

Seamless Interface with Legacy Enterprise Networks

The Mediant 1000 has enhanced hardware and software capabilities to ease its installation and to help maintain voice quality. If the measured voice quality falls beneath a pre-configured value, or the path to the destination is disconnected, the Mediant 1000 can assure voice connectivity by falling back to the PSTN. In the event of network problems, calls can be routed back to the PSTN without requiring routing modifications in the PBX. Further reliability is provided by dual Ethernet ports and optional dual AC power supply.

3rd Party Application Platform

The Mediant 1000 extends the flexibility of the Media Gateway family with additional deployment options. The open platform on the Mediant 1000 offers partners the option to host their own applications (e.g., IP-PBX, call center, conferencing and messaging applications) using the OSN (Open Solutions Network) Server platform, including a powerful processor and hard disks to provide a complete solution within the Mediant 1000 chassis along with rich SIP gateway features.

Mediant™ 1000

Specifications

Interfaces	
Modularity and Capacity	Voice interface: Equipped with 6 Slots that can host voice modules Up to a maximum of 24 analog ports or 4 digital spans
Digital Modules	1, 2 or 4 E1/T1/J1 spans using RJ-48c connectors per module Up to 4 digital modules (maximum 4 spans per gateway) Optional 1+1 or 2+2 fallback spans
Analog FXO and FXS Modules	4 ports using RJ-11 connectors per module; Up to 6 modules per gateway, Ground Start and Loop Start
BRI Module	4 BRI ports (8 calls) per module, up to 5 modules per gateway with S/T interfaces Supports Euro ISDN, NI2, 5ESS or QSIG
Media Processing Module	Hosting media processing features: conferencing, play/record over HTTP or NFS
I/O	MOH (Music On Hold), NB (Night Bell)
Ethernet	Dual Redundant 10/100 Base-TX Ethernet ports via 2 RJ-45 connectors
RS-232	Debugging and configuration
Media Processing	
Voice Coders	G.711, G.726, G.723.1, G.729A, GSM-FR, iLBC, EG.711 Independent dynamic vocoder selection per channel
Echo Cancellation	G.165 and G.168-2002, with 32, 64 or 128 tail length
Quality Enhancement	Dynamic programmable jitter buffer, VAD, CNG, 802.1p/Q VLAN tagging, DiffServ, voice quality monitoring, G.729B, RTCPXR
DTMF/MF Transport	Packet side or PSTN side detection and generation, RFC 2833 compliant DTMF relay Call Progress tones detection and generation
IP Transport	VoIP (RTP/RTCP) per IETF RFC 3550 and 3551
Fax and Modem Transport	T.38 compliant (real time fax), Automatic bypass to PCM or ADPCM
OSN Server Platform	
Single Chassis Integration	Embedded, Partner Application Platform for third party services
CPU	Intel™ Celeron™ 600 Mhz
Memory	One SODIMM slot 512M or 1G RAM
Storage	Single/Dual hard disk drives
Interfaces	10/100 Base-TX, USB, RS-232, NB relay, MOH
Signaling	
Digital –PSTN Protocols	CAS: MF-R1: T1 CAS (E&M, Loop, Start, Feature Group-D, E911CAMA), E1 CAS (R2 MFC), R1.5 numerous protocol and country variants ISDN PRI: ETSI/EURO ISDN, ANSI NI2 and other variants (DMS100, 5ESS) QSIG (Basic and supplementary), IUA (SIGTRAN), VN3, VN4, VN6
Analog Signaling	FXS; Caller ID; polarity reversal; metering tones, distinctive ringing, visual message waiting indication, Loop Start, Ground Start
Control & Management	
Control Protocols	SIP, MSCML, H.323 (MEGACO – for digital trunks)[1]
Operations & Management	AudioCodes Element Management System Embedded HTTP Web Server, Telnet, SNMP V2, V3 Remote configuration and software download via TFTP, HTTP, HTTPS, DHCP and BootP, RADIUS, Syslog (for events, alarms and CDRs)
Security	
	IPSEC, HTTPS, TLS (SIPS), SSL, Web access list. RADIUS login, SRTP[2]
Hardware Specifications	
Power Supply	Single universal 90-260 V AC, redundant power supply
Physical	1U high, 19-inch wide
Regulatory Compliance	
Telecommunication Standards	TIA/EIA-IS-968, TBR-4, TBR-13, and TBR-21
Safety and EMC Standards	UL60950-1; FCC 47 CFR part 15 Class B CE Mark (EN55022 Class B, EN60950-1, EN55024, EN300 386, EN61000-3-2/3-3)
Environmental Specifications	ETS 300019-2-1 Storage T1.2, ETS 300019-2-2 Transportation T2.3 ETS 300019-2-3 Operating T3.2

1 Some PSTN variants may not be supported with all control protocols
2 May reduce density

Applications

- PBX Networking
- IP Centrex/Hosted IP-PBX
- Partner Applications (e.g., IP-PBX, Call Center, Conferencing Messaging)
- Remote Office Applications

About AudioCodes

AudioCodes Ltd. (NASDAQ: AUDC) provides innovative, reliable and cost-effective Voice over IP (VoIP) technology, Voice Network Products, and Value Added Applications to Service Providers, Enterprises, OEMs, Network Equipment Providers and System Integrators worldwide. AudioCodes provides a diverse range of flexible, comprehensive media gateway, and media processing enabling technologies based on VoIPerfect™ – AudioCodes' underlying, best-of-breed, core media architecture. The company is a market leader in VoIP equipment, focused on VoIP Media Gateway, Media Server, Session Border Controllers (SBC), Security Gateways and Value Added Application network products. AudioCodes has deployed tens of millions of media gateway and media server channels globally over the past ten years and is a key player in the emerging best-of-breed, IMS based, VoIP market. The Company is a VoIP technology leader focused on quality and interoperability, with a proven track record in product and network interoperability with industry leaders in the Service Provider and Enterprise space. AudioCodes Voice Network Products feature media gateway and media server platforms for packet-based applications in the converged, wireline, wireless, broadband access, cable, enhanced voice services, video, and Enterprise IP Telephony markets. AudioCodes' headquarters are located in Israel with R&D in the U.S. Other AudioCodes' offices are located in Europe, India, the Far East, and Latin America.

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Your Gateway to VoIP