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
- Lifeline 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

SAS
Stand Alone Survivability
Continuous VoIP Service

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 Solution 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.727, G.723.1, G.729, GSM FR, MS GSM, iLBC, EG.711, EVRC, QCELP, AMR, GSM EFR, G.722 Independent dynamic vocoder selection per channel		
Echo Cancelation	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 - Embedded, Partner Application Platform for third party services			
OSN Types	**OSN1**	**OSN2**	**OSN3[3] (ON AMC Chassis)**
CPU	Intel™ Celeron™ 600 Mhz	Intel Pentium M 1.4 GHz	Intel Core2Due
Memory	One SODIMM slot 512M or 1G RAM	1 or 2 GRAM	Two SODIMM slots 2-4 G RAM, ECC support
Storage	Single/Dual hard disk drives	Single SATA HDD	Single or Dual SATA HDD
Interfaces	10/100 Base-TX, USB, RS-232, NB relay, MOH	10/100 Base-TX, USB, RS-232, VGA	1000 Base-TX, USB, RS-232
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), Auto Update		
Security			
	IPSEC, HTTPS, TLS (SIPS), SSL, Web access list, RADIUS login and SRTP[2]		
Hardware Specifications			
Power Supply	100-240V, 50-60Hz, 1.5A Max, Single (default) or redundant (optional) power supply configurations		
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

3 OSN3 can be used on the Mediant 1000B Chassis only (with AMC support)

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. (NasdaqGS: AUDC) designs, develops and sells advanced Voice over IP (VoIP) and converged VoIP and Data networking products and applications to Service Providers and Enterprises. AudioCodes is a VoIP technology leader focused on VoIP communications, applications and networking elements, and its products are deployed globally in Broadband, Mobile, Cable, and Enterprise networks. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Gateways, Residential Gateways, IP Phones, Media Servers, Session Border Controllers (SBC), Security Gateways and Value Added Applications. AudioCodes underlying technology, VoIPerfectHD™, relies primarily on AudioCodes leadership in DSP, voice coding and voice processing technologies. AudioCodes High Definition (HD) VoIP technologies and products provide enhanced intelligibility, and a better end user communication experience in emerging Voice networks.

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