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Audiocodes M600 1E1/T1 VoIP Gateway

Product Name: Audiocodes M600 1E1/T1 VoIP Gateway
Manufacturer: -
Model Number: M600/1SPAN

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The Mediant 600 is AudioCodes’ cost-effective, wireline VoIP media gateway. The Mediant 600 is designed to interface between TDM & IP networks in enterprises, matching the density requirements for small locations. Incorporating AudioCodes’ innovative Voice over IP technology, the Mediant 600 enables rapid time-to-market and reliable cost-effective deployment of next-generation networks.SIP Connect CompliantThe Mediant 600 is based on VoIPerfectTM, AudioCodes underlying, best-of-breed, media gateway core technology, providing superior voice technology for smooth connectivity of legacy telephones and PBX systems to IP networks and IP-PBX systems to the PSTN. The Mediant 600 is fully interoperable with IP-PBXs, IP Centrex application servers, Softswitches, gateways, IP Phones, and Session Border Controllers. The Mediant 600 Gateway supports 1 or 2 E1/T1/J1 spans, 4 to 8 BRI ports, or 4 FXS analog ports.Seamless Interface with Legacy Enterprise NetworksThe enhanced hardware and software capabilities of the Mediant 600 provide easy installation and continuous maintenance of voice quality. If the measured voice quality falls beneath a preconfigured threshold, or if the path to a destination is disconnected, the Mediant 600 assures 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.SAS - Stand Alone Survivability for Service ContinuityCustomers who connect to centralized IP Centrex services, as well as branch offices of enterprises, using a centralized IP-PBX server, may face a survivability challenge. Stand Alone Survivability (SAS), supported in the Mediant 600, uses SIP B2BUA (Back to Back User Agent) functionality. It provides backup for SIP clients, such as IP and Soft Phones, in case connectivity with the centralized SIP server is lost.Proven InteroperabilityThe Mediant 600 is part of AudioCodes’ family of stand-alone VoIP Gateways and supports multiple VoIP control protocols which have been tested with leading vendors of Softswitches, gateways, IP phones, and Session Border Controllers. Through years of interoperability experience and extensive investment in complying with leading and evolving VoIP standards, AudioCodes offers field-proven products with short time-to-market for OEMs, System Integrators and Network Equipment Providers.Benefit From Extensive ExperienceAudioCodes, established in 1993, is one of the world’s leading providers of VoIP technology. AudioCodes’ commitment to high quality yields consistently superior voice processing products that are feature-rich and field proven. AudioCodes has deployed tens of millions of VoP ports in over 100 countries to date.
Technical Specifications
Interfaces

ï¿½ E1/T1/J1: 1, 2 or fractional (15 DS0) span spans using RJ-48c connectors
ï¿½ BRI S/T: 4 or 8 ports (8/16 calls) per gateway using RJ-45 connectors
ï¿½ Analog: 4 FXS ports using RJ-11 connectors
ï¿½ Ethernet: Dual Redundant Ethernet 10/100 Base-TX Ethernet ports via 2 RJ-45 connectors
ï¿½ RS232: RS-232 for configuration and troubleshooting

Media Processing

ï¿½ Voice Coders: G.711, G.726, G.723.1, G.729A, GSM-FR, EG.711, EG.711, iLBC
ï¿½ Independent dynamic vocoder selection per channel, VAD, CNG
ï¿½ Echo Cancellation: G.165 and G.168-2002, with 32, 64 or 128 tail length
ï¿½ QoS: 802.1p/Q VLAN tagging, DiffServ, voice quality monitoring, RTCP-XR
ï¿½ DTMF/MF Transport Packet side or PSTN side detection and generation, RFC 2833 compliant
ï¿½ DTMF relay, Call Progress tone detection and generation

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ï¿½ IP Transport, VoIP (RTP/RTCP) per IETF RFC 3550 and 3551
ï¿½ Fax and Modem: T.38 compliant (real time fax), Automatic bypass to PCM or ADPCM

Signaling

ï¿½ E1/T1 CAS: E&M, Loop Start, Feature Group-D, E911CAMA, R1.5, R2 MFC, numerous protocol and country variants
ï¿½ ISDN PRI: ETSI/EURO, ANSI NI2, DMS100, 5ESS, VN3, VN4, VN6 QSIG (Basic Call and Supplementary Services) and other variants, UIA (SIGTRAN)

Control Protocol SIP Operations & Management

ï¿½ AudioCodes Element Management System
ï¿½ Embedded HTTP Web Server, Telnet, SNMP V2/V3
ï¿½ Remote configuration and software download TFTP, HTTP, HTTPS, DHCP and BootP, RADIUS, Syslog (for events, alarms and CDRs)

Security Protocols

ï¿½ IPSEC, HTTPS, TLS (SIPS), SSL, Web access list, RADIUS login, SRTP

Hardware Specifications

ï¿½ Power Supply: Single universal 90-260 V AC power supply
ï¿½ Physical: 1U high, 19-inch wide

Dimensions

ï¿½ 306x273x44mm

Regulatory Compliance

ï¿½ Telecommunication Standards TIA/EIA-IS-968, TBR-4, TBR-13, and TBR-21
ï¿½ Safety and EMC Standards UL60950-1; FCC 47 CFR part 15 Class B
ï¿½ CE Mark (EN55022: 2006, EN55024: 1998 + A1: 2001 +A2: 2003 EN6600-3-2: 2000 + A2: 2005, EN6600-3-3: 1995 + A1: 2001 EN60950-1:2001, A11: 2004)
ï¿½ Environmental Specifications ETS 300019-2-1 Storage T1.2

Benefits

ï¿½ Employs AudioCodes VoIPerfect™ technology for outstanding voice quality
ï¿½ Offers digital E1/T1/J1 or BRI or analog (FXS) interfaces
ï¿½ Lifeline relay to PSTN in case of power failure or network degradation
ï¿½ PSTN fallback for assured connectivity
ï¿½ Stand Alone Survivability (SAS) for service continuity

Please Enquire