Preliminary

AudioCodes CPE and Access Gateway Products

MediaPack™ 1288

High Density Analog VoIP Gateway

The AudioCodes **MediaPack(MP)-1288** is a cost-effective, best-of-breed **high density analog media gateway**. The MP-1288 analog VoIP gateway provides superior voice technology for connecting legacy telephones, fax machines and modems with IP-based telephony networks, as well as for integration with IP PBX systems. It is designed and tested to be fully interoperable with leading softswitches, unified communications (UC) servers and SIP proxies.

Benefits

- High density analog media gateway supporting up to 288 FXS Ports
- Ideal for large analog deployments for converting voice, fax and modem calls to IP
- Scalable solution with three capacity options: 288, 216 and 144 ports
- Cost-effective - single management interface, single IP, no need to stack and cable multiple small analog gateways
- Reduced footprint - 3U Chassis
- Designed for carrier environments with high availably power supply and Ethernet redundancy
- Rich interoperability and partnerships that extend across multiple vendor devices and protocol implementations
- Delivers high service performance and voice quality

Key Features

- Support for advanced coders such as NB-AMR and NB-OPUS
- Support for SRTP on all channels without capacity hit
- PSTN fallback (FXO) on power or network failure
- Integrated protection against surge damage on FXS ports (ITU-T K.21 - basic level compliance)
- Supports short and long haul up to 7.5 Km
- Support for emergency / elevator phones that require higher loop current and increased ring voltage
- Rich and Powerful SIP normalization and routing mechanisms for seamless interoperability
- SIP header manipulation
- Extensive fax support including T.38 version 3
- Supports Standalone Survivability (SAS) for hosted communications services and centralized IP-PBX deployments

Proven Interoperability

The MP-1288 is part of AudioCodes' comprehensive family of standalone VoIP gateways. AudioCodes has a long history of investing significant effort in complying with the leading and evolving VoIP standards. Our products have proven SIP interoperability with leading softswitch vendors. As a provider for OEMs, system integrators and network equipment providers, AudioCodes offers reduced time-to-market with field-proven products.

Reliability

The MP-1288 is designed for carrier environments including 1+1 power supplies and 1+1 Ethernet redundancy maintaining high voice quality to deliver reliable enterprise VoIP communications. Advanced call routing mechanisms, network voice quality monitoring and Standalone Survivability (SAS) capabilities (including PSTN fallback) result in minimum communications downtime.

Applications

- Enterprise campus deployments
- PSTN emulation for service providers
- Large-scale analog integration with Lync/Skype for Business or other cloud-based or hybrid PBX deployments

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Specifications

System			
Telephony Capacity	Up to 288 ports in 4 line-cards (each line-card supports 72 ports) Three available capacity options: 288, 216 and 144 ports		
Hardware Elements	• Single System Controller (SC) • 4 line cards with analog interfaces (hot-swappable*)	• 1+1 Power Supplies • Fan Module (front-to-rear air flow)	
Signaling			
Control	SIP (RFC 3261), mature & broadly deployed SIP stack		
Message Manipulation	Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)		
SIP Routing			
Routing Methods	Request URL, IP Address, FQDN, ENUM, advanced LDAP, third-party routing control through REST API		
Redundancy	Detection of proxy failures and subsequent routing to alternative proxies		
Routing Features	Least-cost routing, call forking, load balancing, emergency call detection and prioritization		
Voice Capabilities			
Voice over Packet	G.168-2004 compliant Echo Cancellation, Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation, RTP redundancy, broken connection detection		
Voice Compression	G.711, G.723.1, G.726 ADPCM, G.727 ADPCM, G.729A/B, G.722, NB-AMR, NB-OPUS		
Fax-over-IP	Bypass, T.38 and T.38v3		
3-Way Conference	3-way conference with local mixing across all line-cards		
In-band Signaling	DTMF (TIA 464B), User-defined and call progress tones		
Out-of-Band Signaling	DTMF Relay (RFC 2833), DTMF via SIP INFO/NOTIFY		
Network Protocols			
IP Transport	IPv4, IPv6 for media and control, RTP/RTCP per IETF RFC 3550		
Security			
Media	SRTP	Control	TLS/SIPS
Management	HTTPS, SSH, SNMPv3, Access List, RADIUS Web and Telnet authorization		
Voice Quality and SLA			
Standalone Survivability (SAS)	Ensures call continuity between LAN SIP clients upon connectivity failure. Support 300 registered users		
Packet Marking	802.1p/Q VLAN tagging, DiffServ, TOS		
Voice Enhancement	RTCP-XR, acoustic echo cancellation, replacing voice profile due to impairment detection		
Test Agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs		
Management			
OAM&P	Browser-based GUI, CLI, SNMP, EMS, INI configuration file, REST API		
Automatic Configuration	DHCP, TFTP and HTTP for automatic installation		
Physical Interfaces			
Telephone Interfaces	Up to 288 FXS ports		
Lifeline	Automatic lifeline to PSTN via 3 FXO interfaces per line-card		
Network Interfaces	• Dual Redundant 10/100/1000 Base-T Ethernet ports • Dual Redundant Small Form-Factor Pluggable (SFP)-based connectivity* Note: Hardware installation selectable option		
Console	RJ-45 serial interface for local management		
USB Interface	USB 2.0 for supporting external USB dongle*		
Power			
AC Input Voltage	100 - 240 V AC	DC Input Voltage*	-48 V DC
Redundant Power Supply	Optional, dual feed, redundant power supply		
Physical / Environmental			
Width	17.13 inches (435.2 mm)	Height	5.16 inches (131.2 mm)
Depth	17.75 inches (451 mm)	Weight	21 Kg (fully populated system)
Mounting	3U, 19 inches rack		
Environment			
Temperature	Operational Temp.: 0 to 40°C (41 to 104°F)	Storage Temp.: -40 to 70°C (-40 to 158°F)	Humidity: 5 to 90% non-condensing
Over-voltage protection and surge immunity	ITU-T K.21 (basic) compliant. Note: Routing of FXS telephony cables outdoors can be done only in conjunction with AudioCodes-approved primary surge protector and proper installation and grounding.		
FXS Port Specifications			
Interface Type	FXS RJ-11 connection via 50-pin CHAMP connector		
FXS Signaling Formats	In-band signaling DTMF (TIA 464B), Out-of-band pulse signalling*		
FXS Loop Impedance	Up to 1600 ohm (including phone impedance)		
Off-hook Loop Current	25 mA (maximum) on all ports 35 mA (maximum) on two ports per line-card (for emergency / elevator phones)		
Ring Voltage (Sine)	54 Vrms 80 Vrms on two ports per line-card (for emergency / elevator phones) Notes: Balanced ringing only, Enables simultaneous ringing of 80 phones (20 per line-card given EN3 load)		
Ring Frequency	25-100Hz		
Maximum Ringer Load	Ringer Equivalency Number (REN) 3		
Caller ID	Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation, ETSI Type 1, NTT, Denmark, India, Brazil, British and DTMF ETSI CID (ETS 300-659-1)		
Polarity Reversal / Wink	Immediate or smooth to prevent erroneous ringing		
Metering Tones	12/16 KHz sinusoidal bursts, Generation on FXS		
Distinctive Ringing	By frequency (15-100 Hz) and cadence patterns		
Message Waiting Indication (MWI)	DC voltage generation (TIA/EIA-464-B), V23 FSK data, Stutter dial tone		
Regulatory Compliance			
EMC	EN55022 Class A , CFR Part 15 Class A, EN55024, EN61000-3-3, EN61000-3-2, VCCI Class X1 (equal to class B)		
Safty	EN60950-1, UL60950-1		

*Roadmap feature

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 market leader focused on converged VoIP & data communications and its products are deployed globally in Broadband, Mobile, Enterprise networks and Cable. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Routers, Session Border Controllers (SBC), Residential Gateways, IP Phones, Media Servers and Value Added Applications. AudioCodes' underlying technology, VoIPerfect HDTM, relies 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 Voice communications.

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