

ANALOG VOIP GATEWAYS (MP-11X, MP-124-D)

MediaPack™ Series AudioCodes CPE & Access Gateway Products

The MediaPack™ Series Analog VoIP Gateways are cost-effective, best-of-breed technology products. These stand-alone analog VoIP Gateways provide superior voice technology for connecting legacy telephones, fax machines and PBX systems with IP-based telephony networks, as well as for integration with new IP-based PBX systems. These products are designed and tested to be fully interoperable with leading Softswitches, SIP servers and H.323 gatekeepers.

MediaPacks are well suited for commercial VoIP deployment because of their mature and field-proven voice and fax technology. Their rich feature set allows integration with a wide range of Carriers and Enterprise network applications. MediaPack gateways are used by Carriers and Service Providers in Access networks for connecting Multi-Tenant Units (MTU), IP Centrex subscribers, payphones and rural users over various wireless and satellite links. Enterprises use MediaPack gateways to connect their legacy PBX systems over an IP infrastructure.

In addition, in IP Centrex and central IP-PBX applications, the MediaPack increases the remote location availability and provides Stand Alone Survivability (SAS) when there is no IP connection between branch locations and the central SIP servers, SIP Proxy or central IP-PBX.

DELIVER FEATURE-RICH SOLUTIONS

MediaPacks are third generation products that have been designed to meet real market needs. In addition to superior voice technology, the products provide advanced telephony features such as long-haul, metering tones generation, country dependent MWI and CID for true integration with the existing telephony infrastructure.

A variety of management and provisioning tools, such as AudioCodes' EMS, embedded web server, Telnet and SNMP enable fast deployment and management of large and complex networks. MediaPacks are based on VoIPerfect™ architecture, AudioCodes underlying, best-of-breed, core media gateway technology for all of its products.

PROVIDE INTEROPERABILITY

MediaPacks are part of AudioCodes' complete family of stand-alone VoIP Gateways for OEM system integration. Throughout the years, AudioCodes has invested significant effort in complying with the leading and evolving VoIP standards. Support of multiple VoIP control protocols has been tested with leading Softswitch vendors. As a provider for OEMs, System Integrators and Network Equipment Providers, AudioCodes offers short time-to-market with field-proven products.



- Provides voice, fax & modem support
- Offers toll quality voice compression
- Enhanced capabilities including MWI, long haul, Metering Tones Generation, CID & out door protection**
- Allows Fallback to PSTN for E911 (Emergency number PSTN breakthrough) or upon network/power failure (FXO and/or FXS) configurations
- Supports Stand Alone Survivability (SAS) for IP Centrex and Central IP-PBX
- Supports SIP, H.323 and MGCP standard control protocols
- Proven integration with leading PBXs, IP-PBXs, Softswitches and Gatekeepers



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MediaPack™ Series

BENEFIT FROM EXTENSIVE EXPERIENCE

AudioCodes, established in 1993, is one of the world's leading providers of VoIP technology. AudioCodes' commitment to innovation yields consistently high-quality voice processing products that are feature-rich and fieldproven. AudioCodes has deployed tens of millions of VoP ports in over 100 countries to date.

MEDIAPACK SERIES (11X) FEATURES

- Spans a range of 2 to 24 analog ports
- Supports PSTN/PBX analog telephone sets or analog trunk lines (FXS/FXO)
- Selectable, multiple LBR coders per channel
- T.38 compliant
- Echo cancelation, Jitter Buffer, VAD and CNG
- Rich subscriber Feature Set including; 3-Way conference with local mixing, call pickup, hunt groups, call forwarding, call hold, call transfer
- Complies with MGCP, H.323 (V4) and SIP control protocols
- Enhanced capabilities which including MWI, long-haul, Metering Tones, STUN, Security features, Generation, CID & outdoor protection**
- Stand Alone Survivability (SAS) for SIP based IP Centrex and Central IP-PBX applications
- Web Management for easy configuration and installation
- EMS for comprehensive management operations (FCAPS)
- Automatic provisioning via TFTP/HTTP
- Internal Access List firewall for network traffic filtering

APPLICATIONS

- Survivable IP Centrex for hosted services and central IP-PBX IP Centrex for hosted services
- Multi Tenant Units
- POTs and fax extensions for IP-PBX
- Voice VPN
- VoIP-enabled PBX
- Unified messaging and recording
- IP-PBX FXS Analog extensions and FXO

SPECIFICATIONS

	MP-112	MP-114 and MP-118	MP-124D
Interfaces			
Voice Ports	2 ports	4 and 8 ports	24 ports
Telephone Interfaces	FXS, RJ11	FXS, FXO or mixed FXS/FXO, RJ11	FXS, 50-pin Telco Connector
Life Line	Automatic cut through of a single analog line		
Network Interfaces	10/100 BASE-TX, RJ45		
Indicators	Channel status and activity LEDs		
Voice, Fax, Modem			
Voice over Packet Capabilities	G.168-2002 compliant Echo Cancellation, VAD, CNG, Dynamic programmable Jitter Buffer, modem detection and auto switch to PCM		
Voice Compression	G.711, G.723.1, G.726, G.729A, EG.711, G.722		
Fax over IP	T.38 compliant Group 3 fax relay up to 14.4 kbps with automatic switching to PCM or ADPCM		
3-Way Conference	3-Way conference with local mixing		
QoS	Diffserve, TOS, 802.1 P/Q tagging		
IP Transport	RTP/RTCP per IETF RFC 3550 and 3551 PPPoE, ThruPacket™ (aggregated RTP streams of several channels for saving network bandwidth)		
Signaling			
Signaling	FXS Loop-start	FXS, FXO Loop-start	FXS Loop-start
In-band Signaling	<ul style="list-style-type: none"> ■ DTMF (TIA 464B) ■ User-defined and call progress tones 		
Control	MGCP (RFC 2705), H.323 (V4), SIP (RFC 3261)		
Provisioning	<ul style="list-style-type: none"> ■ BootP, DHCP, TFTP & HTTP for automatic installation ■ DHCP options 66.67 in auto update mode ■ Remote management using Web browser ■ EMS (Element Management System) / SNMP V3 ■ Syslog support ■ RS-232 for basic configuration (via CLI) ■ Voice Menu using touch tone phone for basic configuration 		
Security			
Media	SRTP		
Control	H.235, IPSEC, TLS/SIPS		
Management	HTTPS, Access List, IPSEC		
Physical			
Power	100-240 VAC/50-60 Hz or 48V DC*		
Environmental	Operational: 5 to 40° C - 41 to 104° F Storage: -25 to 85° C - 13 to 185° F Humidity: 10 to 90% non-condensing		
Dimensions (Hx W x D)	42 x 172 x 220 mm	42 x 172 x 220 mm	44 x 445 x 269 mm
Mounting	Rack mount, Table top, Wall mount		
Additional Features			
Message Waiting Indication	Applying 100V DC online for lightingbulb in handset, FSK, Stutter Dial Tone		
High Availability	PSTN Fallback Support of PSTN fallback due to Power failure, if the IP Connection is down or due to customer defined IP QoS thresholds Stand Alone Survivability (SAS), Supports SAS of up to 25 SIP users (UA) per MediaPack		
Short and Long Haul	REN 3, up to 9 km (24 AWG)		
Lifeline	Supported in all ports of Mixed FXS/FXO and in first port of MP-114/FXS and MP-118/FXS using special Lifeline cable		
Caller ID	Bellcore GR-30-CORE Typ 1 using Bell 202 FSK modulation, ETSI Typ1. NTT. British, Sweden, Denmark, Brazil 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		
Outdoor Protection*	Over-voltage protection and surge immunity		
Homologation			
EMC	EN55022 Class B, CFR Part 15 Class B, EN55024, EN61000-3-3, EN61000-3-2, VCCI Class X1 (equal to class B)		
Safety	EN60950-1 Safety of information technology equipment UL60950-1 Including compliance to section 6 (over-voltage protection)		
Telecom	TBR-21, TIA-968		