

Audiocodes MediaPack 112 2 FXS Analog VoIP Gateway



Product Name: Audiocodes MediaPack 112 2 FXS Analog VoIP Gateway Manufacturer: -Model Number: MP112/2S/SIP

Please Note: This product has been discontinued. Please see AudioCodes MediaPack.

Audiocodes MediaPack 112 2 FXS Analog VoIP Gateway

The AudioCodes MP112/2S/SIP is an analogue VoIP Gateway which is a cost-effective, best-of-breed technology product. These stand-alone analogue 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 IP PBX systems. They are designed and tested to be fully interoperable with leading softswitches, SIP servers and H.323 gatekeepers. AudioCodes MP112/2S/SIP Key Features

ï¿1/2 Provides voice, fax and modem support

ï¿1/2 Offers toll quality voice compression

i¿½ Enhanced capabilities including MWI, long haul, Metering Tones Generation and, Caller ID i¿½ Allows fallback to PSTN for E911 (Emergency number PSTN breakthrough) or upon network/ power failure (FXO and/or FXS) configurations

ï¿¹/₂ Supports Standalone Survivability (SAS) for hosted communications services and centralised IP-PBX deployments

i¿1/2 Supports SIP, H.323 and MGCP standard control protocols

ï¿1/2 Proven integration with leading PBXs, IP-PBXs, and softswitches

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 carrier 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 hosted communications and centralized IP-PBX applications, the MediaPack increases the remote location availability and provides Standalone 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 SolutionsMediaPacks 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 Caller ID 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. Provide InteroperabilityMediaPacks 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 ...

AudioCodes MP112/2S/SIP IP Phone Technical Specifications MediaPack Series Features

ï¿1/2 Scales 2 to 24 analogue ports

- ï¿¹/₂ Supports PSTN/PBX analogue telephone sets or analogue trunk lines (FXS/FXO)
- ï¿1/2 Selectable, multiple LBR coders per channel
- ï¿1/2 T.38 compliant

� Rich subscriber Feature Set including; 3-Way conference with local mixing, call pickup, hunt



Audiocodes MediaPack 112 2 FXS Analog VoIP Gateway

groups, call forwarding, call hold, call transfer

ï¿1/2 Echo cancellation, Jitter Buffer, VAD and CNG

ï¿¹/₂ Complies with SIP, MGCP and H.323 (V4) control protocols

� Enhanced capabilities which include MWI, long-haul, metering tones, STUN, Security features and caller ID.

تزائ Standalone Survivability (SAS) for SIP-based hosted communications and centralised IP-PBX applications

ï¿1/2 Web Management for easy configuration and installation

ïزئ EMS for comprehensive management operations (FCAPS)

ïزائ Voice quality monitoring support via AudioCodes Session Experience Manager (SEM).

ïزائ Automatic, secure provisioning. Useful for large-scale deployments

ïزئ Internal Access List firewall for network traffic filtering

Voice, Fax, Modem

 $\ddot{\imath}_{2}$ Voice over Packet capabilities: G.168-2004 Echo Cancelation, VAD, CNG, Dynamic programmable Jitter, modem detection and auto-switch to PCM

� Voice Compression: G.711, G.723.1, G.726 ADPCM, G.727 ADPCM, G.729A/B, G.722

 $i_{\xi}1_{2}^{\prime}$ Fax over IP: T.38 compliant, Group 3 fax relay up to 14.4 kbps with automatic switching to PCM or ADPCM

ï¿1/2 3-Way Conference: 3-Way conference with local mixing

� VLAN QoS: DiffServ, TOS, 802.1 p/Q VLAN tagging, RTCP-XR

i¿1/2 IP Transport (bandwidth): RTP/RTCP per IETF RFC 3550 and 3551

Signalling

ï¿1/2 Signalling: FXS, FXO Loop-start

i¿1/2 In-band Signaling DTMF (TIA 464B): User-defined and call progress tones

ïزئ Out-of-Band Signaling: DTMF Relay (RFC 2833), DTMF via SIP INFO/NOTIFY

ï¿1/2 Control SIP: (RFC 3261), MGCP (RFC 2- 05), H.323 (V4),

Provisioning

� Protocols: BootP, DHCP, TFTP and HTTP for Automatic Installation, DHCP options 66.67 in auto update mode, Remote management using Web browser, EMS (Element Management System) / SNMP V3, 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

Additional Features

� Message Waiting Indication: Applying 100V DC online for lighting bulb in the handset, FSK, Stutter Dial Tone

i¿1/2 PSTN Fallback: Support of PSTN fallback due to Power failure, if the IP connection is down or due to customer defined IP QOS thresholds

i¿¹/₂ Stand Alone Survivability (SAS): Supports SAS of up to 25 SIP users (UA) per MediaPack i¿¹/₂ Ring voltage: Sine: 54 VRMS typical (balanced ringing only)

ï¿¹/₂ Ring Frequency: 25-100Hz



Audiocodes MediaPack 112 2 FXS Analog VoIP Gateway

ïزئ Voice Quality Monitoring: AudioCodes Session Experience Manager (SEM) ïزئ Maximum Ringer Load: REN3

تزنائد Loop Impedance (Including Phone impendance): Up to 1500 ohm for the MP-11x, Up to 1600 ohm for the MP-124

ï¿1/2 Line current up to 32 MA on 4 ports

� 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

i¿1/2 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)

ïزئ Polarity Reversal / Wink: Immediate or smooth to prevent erroneous ringing

ïزئ Metering Tones: 12/16 KHz sinusoidal bursts, Generation on FXS

i¿1/2 Distinctive Ringing: By frequency (15-100 Hz) and cadence patterns

Outdoor Protection

 i_{ℓ} ^{1/2} Over-voltage protection and surge immunity. This applies only to the MP-124 FXS telephony cables, which can be routed outdoors. In such a case, power surge protection means are required (refer to the Installation Manual for detailed instructions)

Physical

 \ddot{i}_{2} Power: 100-240 V AC/50-60 Hz or -48V DC* \ddot{i}_{2} Environmental Operational: 5 to 40o C/ 41 to 104o F, Storage: -25 to 85o C/ -13 to 185o F Humidity: 10 to 90% non-condensing \ddot{i}_{2} Dimensions: 42x172x220mm (MP-112 and MP-118) \ddot{i}_{2} Mounting: Rack mount, Tabletop, Wall mount \ddot{i}_{2} Weight: MP-1xx: 0.5 kg (1.1 lbs.) approx.

Please Enquire