AudioCodes Analog VoIP Gateways (MP-114 & MP-118)

The AudioCodes MediaPack stand-alone Analog VoIP Gateways provide technology for connecting legacy telephone or fax equipment and PBX systems with IP-based telephony networks, as well as for integration with new IP-based PBX architecture.

MediaPack is designed and tested to be fully interoperable with leading soft switches, H.323 gatekeepers and SIP servers such as MESSAGEmanager.

The MediaPack series Analog VoIP Gateways are used in the voice VPN environment, centralized IVR and Quality Monitoring, calling card and pay phone markets. Ideal markets also include MTU (Multi Tenant Units), rural areas and remote analog extensions in a VoIP-based PBX or IP-PBX architecture.

FEATURES

- Supports 4 and 8 ports
- Supports PSTN/PBX analog telephone sets or analog trunk lines
- Selectable, multiple LBR coders per channel
- Internal power supply
- T.38 compliant
- Echo canceller, Jitter Buffer, VAD and CNG
- Complies with MGCP, H.323 (V4) and SIP control protocols
- Web management for easy configuration and installation
- SNMP and syslog support
- EMS for all-embracing management operations (FCAPS)

PROVIDE INTEROPERABILITY

The MediaPack Series is 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 all control protocols by the MP Series has been tested with leading softswitch vendors. As providers for OEMs, System Integrators and Network Equipment Providers, AudioCodes offers short time to market with second generation, field-proven products.

BENEFIT FROM EXTENSIVE EXPERIENCE

AudioCodes, extablished 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 field-proven, AudioCodes has deployed over 12 million VoiP ports to date.

BANDWIDTH

As with any IP-based service, the network should be provisioned to provide the necessary bandwidth which will vary depending on usage patterns. Fax over IP traffic typically uses only about 1/3 the bandwidth of VoIP traffic, and therefore will have a much smaller impact on the required network bandwidth needs.

MESSAGEmanager uses T.38 over UDP whereas other products may use a codec such as G.711. G.711 over RTP is a VoIP connection, much like streaming a fax over a VoIP line and consumes 64kbps.

MESSAGEmanager puts traffic on the IP line at the modulation rate negotiated with the end fax device on the PSTN. With a V.17 connection, this will consume IP bandwidth at a rate of 14.4 kbps.

T.38 is reliable on an IP network using UDP as it has the capability to send redundant packets to be sure the connection stays up despite jitter and latency. This is a user configurable parameter which is generally set to send 2x redundancy, which means a V.17 connection would consume 28.8 kbps of bandwidth.

If the modulation rate to the end fax device is less than 14.4 kbps,less bandwidth is required.

NETWORK DELAY/LATENCY/JITTER

Acceptable VoIP latency ranges from 250-300 milliseconds or less for each packet. A network properly configured for VoIP will support IP Fax.

- Loss of consecutive packets can cause the fax process to fail
- T.30 will fail if 3 consecutive signals are missing overcome by the T.38 driver sending redundant packets
- Jitter, variable timing between packets, is overcome by adding time stamps to the T.38 packets ensuring signals are 'played' at the right instance by the gateway

MP-114 & MP-118 SPECIFICATIONS

INTERFACES

Voice Ports

MP-114 and MP-118	4 and 8 ports

Telephone Interface

MP-114 and MP-118	Automatic cut through of a single analog line

VOICE, FAX, MODEM

Capabilities	Buffer, modem detection and auto-switch to PCM
Voice Compression	G.711, G.723.1, G.726, G.729A
Fax over IP	T.38 compliant Group 3 fax rely up to 14.4kbps with automatic switching to PCM or ADPCM
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging
IP Transport	RTP/RTCP per IETF RFC 3550 and 3551 PPPoE, ThruPacket™ (aggregated RTP streams of several channels for saving network bandwith)

SIGNALING

Signaling

MP-114 and MP-118	FXO Loop-start
In-band Signaling	DTMF (TIA 464-B) User defined and call pregress tones
Control	MGCP (RFC 2-05), H.232 (V4), SIP (RFC 3261)
Provisioning	BootP, DHCP, TFTP and HTTP for automatic installation Remote management using Web Browser Element Management System Syslog support RS-232 for basic configuration 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 V AC/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

MP-114 and MP-118	42x172x220mm
Mounting	Rack mount, Table top, Wall mount

ADDITIONAL FEATURES

Message Waiting Indication	Applying 100V DC on line for lighting bulb in handset
High Availability	**PSTN Fallback:** Support of PSTN fallback due to Power failure of the IP connection is down or customer defined if IP QOS thresholds Supported in all ports of and in first port of MP-114 and MP-118 using special Lifeline cable **Stand Alone Survivability (SA):** Supports SAS of up to 30 SIP users (UA) per MediaPack
Short and Long Haul	REN3: Up to 9Km (24 AWG line)
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	DC voltage generation (TIA/EIA-464-B), V23 FSK data, Stutter Dial Tone
Outdoor Protection	Over-voltage protection and surge immunity