MP-20x Series

AudioCodes MediaPack™ Series Analog Telephone Adapters with Integrated Router

- **Residential and SOHO telephony VoIP Gateway for Voice over Broadband (VoBB) networks**
- **Up to 4 FXS ports connecting to POTS phones or fax machines**
- **Provides toll quality voice compression**
- **Robust fax handling with support for T.38**
- **Integrated router including Firewall, NAT and smart traffic prioritization**
- **Supports a rich set of subscriber calling features**
- **Easy integration into existing VoIP networks**
- **Mass deployment provisioning with TR-069 support and remote provisioning**
- **Broad interoperability list of Softswitches and SIP servers**

The **MP-20x series** of Analog Telephone Adapters are cost-effective, advanced products, which allow the connection of ordinary analog telephones or fax machines to a Voice over Broadband (VoBB) service.

As a member of the MediaPack™ series of gateways, the MP-20x series of Telephone Adapters are well-suited for commercial VoIP deployments. Field-proven voice and fax technology and a feature-rich design make these Telephone Adapters an excellent choice for Voice over Broadband (VoBB) service providers and operators.

DESIGNED FOR RESIDENTIAL AND SOHO IP TELEPHONY
The MP-20x series is designed for the rapidly growing residential and Small Office/Home Office (SOHO) VoIP market. The MP-20x series typically connects to an existing Broadband Internet device (cable, DSL modem or fixed wireless), and establishes a communications path with the service provider network via its IP Uplink connection. Supporting a rich set of subscriber calling features such as caller ID, call forwarding, and call waiting, the MP-20x series maintains a uniform user experience when migrating to VoIP services. In addition, the MP-20x series serves as a router, supporting connectivity of home PC Networks.

BENEFIT FROM EXTENSIVE EXPERIENCE
Utilizing AudioCodes' VoIPerfect™ core architecture, and gaining from its accumulated experience in providing IP telephony solutions, the MP-20x series combines superior voice quality and cutting-edge features for end users, such as T.38 Fax Relay and G.168-2004 compliant Echo Cancellation. Low bit rate vocoders (voice coders) can be used simultaneously on all the telephony ports to save valuable bandwidth.

INTEROPERABILITY
The MP-20x series is designed for interoperability with leading Softswitches and SIP servers for deployment in various network environments.

MP-20x KEY FEATURES
- Up to 4 FXS ports for connecting POTS phones or fax machines
  in case of power failure
- WAN and LAN Ethernet ports for internal routing
- Bridge mode or full router functionality with DHCP, NAT, Firewall and configurable routing logic
- Premium voice quality
  - Prioritization of voice over data traffic ensuring voice quality
  - Based on AudioCodes' robust, 3rd generation VoIPerfect™ technology
  - Dual core architecture ensures voice quality is maintained even during high CPU load

AudioCodes

MP-20x Series

SPECIFICATIONS

Telephony Features	
Supplementary Features	Call Waiting, Call Hold, Call Transfer, Call Forward Local and Network 3-Way conferencing, Hot Line
Analog Signaling	Caller ID, MWI, Polarity Reversal, Distinctive Ringing DTMF Relay - RFC 2833, DTMF via SIP INFO Hook Flash - Local Processing or Relay Configurable Call Progress tones

Media Processing	
Voice Coders	G.711, G.723.1, G.729A/B, G.726, iLBC*, GSM FR*
Fax	T.38, network delay up to 10 sec, VBD
Echo Cancellation	G.168-2004 compliant, 64 msec tail length
Silence Suppression	VAD, CNG
Adaptive Jitter Buffer	300 msec
Packetization	RTP/RTCP Packetization – RFC 3550, 3551

Protocol Support	
VoIP Signaling Support	SIP – RFC 3261 SIP over TLS MGCP – RFC 2705*
Data Protocols and Features	IPv4, TCP, UDP, ICMP, ARP, RTP, RTCP PPPoE – RFC 2516, PPTP, L2TP DNS, Dynamic DNS, Stun IEEE 802.1p/q (QoS & VLAN Tagging) ToS/Diffserv Layer 2 switching or WAN to LAN Layer 3 routing with: - DHCP Client/Server – RFC 2132 - RIP V1/V2 and static routing - NAT – RFC3022, Application Layer Gateway (ALG) - Voice over Data prioritization - QOS traffic shaping - SPI Firewall - URL filtering - Advance Access Control

Provisioning and Management	
Provisioning	Web Server for Configuration and Management Software and Configuration upgrade from HTTP/HTTPS/TFTP/FTP Configuration file encryption (3DES) TR-069, TR-104
Management	Integrated SNMP agent CLI over Telnet Syslog support for debug and diagnostics
LED Indications	Power, Status LED, Ethernet activity, Phone activity LEDs indicate upgrade in process VoIP Softswitch Registration Status

Physical Specifications	
Telephone Interfaces	Up to 4 RJ-11 FXS port options
Network Interface Power	WAN, LAN I/F (Uplink): RJ-45 - 10/100 Base-T x2 +12V DC, 0.6A Power adaptor: 110V AC/220V AC to 12V DC, 1A
Dimensions	167x133x33 mm
Environmental	Operation: 5⁰C - 40⁰C, Storage: -25⁰C - 85⁰C

Ordering options	
MP-201B/1FXS/SIP	MediaPack 201 VoIP Telephone Adapter including single FXS line, 1 WAN port, 1 LAN port
MP-202B/2FXS/SIP	MediaPack 202 VoIP Telephone Adapter including 2 FXS lines, 1 WAN port, 1 LAN port
MP-204B/4FXS/SIP	MediaPack 204 VoIP Telephone Adapter including 4 FXS lines, 1 WAN port, 1 LAN port

* Please contact AudioCodes for availability

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 Gateways, Session Border Controllers (SBC), Residential Gateways, IP Phones, Media Servers and Value Added Applications. AudioCodes' underlying technology, VoIPerfectHD™, 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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