AudioCodes CPE & Access Gateway Products

MP-20x Series Telephone Adapter

MP-20x Telephone Adapter
Release Notes

Version 4.4.5

Document #: LTRT-50521

AudioCodes

Table of Contents

List of Tables

Notice

Date Published: May-18-2017

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Related Documentation

Document Name
MP-20x Quick Installation Guides
MP-20x Telephone Adapter User's Manual

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1 Introduction

AudioCodes MP-20x series of analog Telephone Adapters are cost-effective, feature-rich gateways, allowing the connection of ordinary POTS analog telephones or fax machines to a Voice-over-Broadband (VoBB) service provider.

The MP-20x series is designed for the rapidly growing residential and Small Office/Home Office (SOHO) voice-over-IP (VoIP) market. The MP-20x series typically connects to an existing Broadband Internet device (Cable and ADSL modem, - depending on model), and establishes a communications path with the service provider network through 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 with capabilities such as DHCP, NAT, Firewall, PPPoE, PPTP supporting connectivity of home PC networks.

The MP-20x VoIP Gateway is an all-in-one unit featuring (depending on model) a VoIP adapter, FXS lines, FXO interfaces, Ethernet LAN interfaces (with an internal Layer-2 switch), and Ethernet WAN interface.

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.

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2 New Features

Version 4.4.5 offers the following new features and support:

- Support V.34 for T.38 Fax Relay

  Default mode has now been upgraded from T.38 Version 0, to T.38 Version 3 (Super G3 Fax = max rate 33,600 bit/s).

- Certificate Protection on the Device from Being Overwritten by the Configuration File

  - By default, the rg_conf/rmt_config/override_cert_configuration parameter doesn't exist. In this case, remote configuration update will not overwrite the certificates on the device. The same applies if the rg_conf/rmt_config/override_cert_configuration parameter is set to Disabled (0).
  - To overwrite certificates on the device, set the rg_conf/rmt_config/override_cert_configuration parameter to Enabled ('1')

- Remote Configuration Provisioning Based on MD5 Checksum Comparison

  By default, a new method for remote configuration provisioning based on MD5 checksum comparison has been added with the following steps:

  1. The device downloads the configuration file and calculates its MD5-checksum.
  2. The device compares the calculated value to the previous MD5-checksum ('rmt_config/checksum').
  3. If the value is different, the device applies the downloaded configuration and saves the new MD5-checksum value.
  4. If the value is the same, the downloaded file will be discarded.
  5. To revert back to the methods used in previous versions, (e.g., "use_if_modified_since" or "version" counter), the following parameter needs to be added to the configuration: rg_conf/rmt_config/default_prov_disabled=1

Note:

Version 4.4.5 is applicable only to MP-20x Rev. D for ATA devices for the following models:

- MP-202
- MP-202R
- MP-204R

The following software features are not supported in this version:

- SRTP for media
- SIP-over-TLS for signaling

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3 Resolved Constraints

The following constraints have now been resolved:

Table 3-1: Resolved Constraints for Version 4.4.5

Incident	Description
106290	When testing SIP over TLS (signaling) and SRTP (media), the remote side disconnects during 3-Way Conferencing and the Device Under Test (DUT) freezes.
106059	The DUT sends Name Authority Pointer (NAPTR) instead of a SRV record.
104295	VLANs on top of the Network Bridge cannot be used.
104268	No dial tone after the conference initiator (A) is dropped from a conference call.
104267	World Wide Telecom: Under TR-069, cannot disable/enable the PPTP dialer.
105892	During Packet Smart provisioning, the device tries to untar the downloaded file and gets in a loop.
105890	During Packet Smart provisioning, the periodic check is absent after an unsuccessful Packetsmart upgrade.
105783	Orange France: Incorrect presentation of boundary values for Number of Messages.
105654	VoiP Redundancy: Failback doesn't work when using TLS as signaling.
105621	A 'Basic' role user was incorrectly created from the Auto Configuration Servers (ACS).
105611	Orange France: The number of messages of the Message Waiting Indicator (MWI) is not being updated.
105418	SRTP: One-way voice occurs after toggling for a few seconds.
105287	A new pFax version (1.16.1.29) was updated.
105286	Unnecessary PacketSmart parameters were removed from the GUI.
104824	Orange France: SIP NOTIFY caused an immediate re-boot during an active call.
104822	Orange France: Caller ID - French accents missing.
104752	MP-20x: Added timestamp to serial prints.
104681	Aastra-Mitel: The transfer failed when the first call was incoming.
104487	MP-20x - Disable default QoS-Traffic Priority rules.
104175	No CALLER ID on the first call.
102892	Active calls disconnected during update/upgrade firmware/configuration.

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4 Known Constraints

Version 4.4.5 includes the following known constraints:

Table 4-1: Known Constraints for Version 4.4.5

Incident	Description
106184	Device sends an incorrect DNS SRV query when tls mode configured.
106176	DUT sends DNS SRV query instead of an A query, when outbound proxy enabled
105131	Network Bridge: It is impossible to use the PPPoE dialer on top of the Network bridge.
104890	Flash Mode: Incorrect behavior of Flash + digit type 3 mode.
103145	Voice Menu: 'Cancel' message is missing.
103041	Voice Menu: Restore to Factory Settings is missing.
102691	Local Do Not Disturb (DND) does not work.

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5 Device Gateway Specifications

The specifications for the router and VoIP functionality are listed in the table below:

Table 5-1: MP-20x Telephone Adapter Software Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP - RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ IPv4, TCP, UDP, ICMP, ARP, LLDP, TLS ▪ PPPoE (RFC 2516) ▪ PPTP (RFC 2637) ▪ DNS, Dynamic DNS ▪ WAN–to-LAN Layer-3 routing with: ✓ DHCP Client/Server (RFC 2132) ✓ NAT: RFC 3022, Application Layer Gateway (ALG) ✓ Stateful Packet Inspection Firewall ✓ QoS - Priority queues, VLAN 802.1p,Q tagging, traffic shaping ▪ STUN (RFC 3489)
Media Processing	▪ Voice Coders: G.711, G.729A/B, G.726 ▪ Echo Cancelation: G.168-2004 compliant, 64-msec tail length ▪ Silence Compression ▪ Adaptive Jitter Buffer 300 msec ▪ Fax bypass, Voice-Band Data and T.38 fax relay ▪ Automatic Gain Control
Telephony Features	▪ Call Hold and Transfer ▪ Call Waiting Type 1 and 2 ▪ Message Waiting Indication ▪ 3-Way Conferencing ▪ Speed Dial ▪ Polarity Reversal
Configuration/ Management	▪ Embedded Web Server for configuration and management ▪ TR-069 and TR-104 for remote configuration and management ▪ Remote firmware upgrade and configuration by HTTP, TFTP, FTP, and HTTPS ▪ Configuration file encryption (3DES) ▪ SIP-triggered remote firmware and configuration upgrade ▪ Command-Line Interface (CLI) over Telnet ▪ Dual image management ▪ SNMP ▪ Local Support and Troubleshooting with Web interception and voice notification ▪ BroadSoft BroadCloud certification ▪ BroadCloud PacketSmart integration (applicable only to MP-20xR) ▪ Optional 3G/4G dongle (applicable only to MP-20xR) ▪ Link Layer Discovery Protocol (LLDP) support ▪ Faxback integration (Fax over HTTPS) – Optional ▪ File server ▪ Printer server ▪ DLNA Media server ▪ FXS Voice menu

Feature	Details
Packetization	▪ RTP/RTCP Packetization (RFC 3550, RFC 3551) ▪ DTMF Relay (RFC 2833)
Security	▪ HTTPS for Web-based configuration ▪ Password protected Web pages (MD5)
Telephony Signaling	▪ In-band: ✓ DTMF: Detection and Generation, TIA464B ✓ Caller ID: Telcordia, ETSI, NTT - Type I, Telcordia Type II ✓ Call Progress Tones ▪ Out-of-band: ✓ FXS Loop-start Signaling ✓ On/Off Hook, Flash Hook
Hardware	
Power	▪ MP202B – Power Supply 12VDC/1A ▪ MP20xR – Power Supply 12VDC/2A
Interfaces	▪ WAN 10/100Base-T (RJ-45) ▪ LAN 10/100Base-T (RJ-45) ▪ RJ-11 FXS ports for telephones (POTS) ▪ Network Interface WAN/LAN 10/100 Base-T(RJ-45) ▪ Ethernet ports are 10/100/1000Base-T (applicable to MP-20xR) ▪ USB 2.0 (applicable to MP-20xR)
LED Indications	▪ FXS Phone lines (1 to 4, depending on MP-20x model) - Registered, In Use, Alert ▪ LAN activity on Ethernet Port ▪ WAN ▪ USB (applicable for MP20xR only) ▪ Status ▪ Power on
3G Backup (Optional)	▪ Support for 3G USB dongles for primary WAN backup including: ✓ Alcatel 4G LTE (1bbb:f000->1bbb:0195) ✓ Alcatel AL720 ✓ Huawei E1550 ✓ Huawei E156 ✓ Huawei E156G ✓ Huawei E160 ✓ Huawei E169 ✓ Huawei E1750 ✓ Huawei E1756 ✓ Huawei E1756 Movistar ✓ Huawei E303 ✓ Huawei E3131 ✓ Huawei E3272 ✓ Huawei E3276 ✓ Huawei E3372 ✓ Huawei E367 ✓ Huawei E392 ✓ Huawei E398 ✓ Huawei K3765 ✓ Huawei K3772 ✓ Sierra AC326U ✓ Vertex ✓ ZTE K3805-Z

Feature	Details
	✓ ZTE MF110 ✓ ZTE MF190 ✓ ZTE MF626 ✓ ZTE MF823
SLIC Characteristics	▪ Maximum Ringer Load (REN) = 5 ▪ Short Haul ▪ Ringer Voltage - up to 65Vrms ▪ Configurable Terminating Impedance
Environmental	▪ **Operating Temperature:** 0 to 40°C ▪ **Storage Temperature:** -25 to 70°C ▪ **Operating Humidity:** 10 to 90% non-condensing ▪ **Storage Humidity:** 10 to 90% non-condensing
Weight and Dimensions	▪ MP-202B: 230 grams; 167 x 133 x 33 mm ▪ MP-202R: 280 grams; 167 x 133 x 33 mm ▪ MP-204R: 280 grams; 167 x 133 x 33 mm

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