Release Notes

AudioCodes CPE & Access Gateway Products

MP-20x Telephone Adapter

Version 4.5.1

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Related Documentation

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

Document Revision Record

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50526	New features added for Version 4.4.9 Build 111.
50527	Resolved constraints added for Version 4.4.9 Build 142.
50528	Resolved constraints added for Version 4.4.9 Build 146.
50529	New features and resolved constraints added for Version 4.5.0 and Version 4.4.9 Build 156.
50530	Resolved constraints for Version 4.5.0 and new features for Version 4.4.9 Build 156 were updated.
50531	New features and resolved constraints added for Version 4.4.9 Build 163.
50532	New features and resolved constraints added for Version 4.5.1.

Documentation Feedback

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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, 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 Version 4.5.x

This section describes Version 4.5.x.

2.1 Version 4.5.1

This section describes Version 4.5.1. This version is supported on specific modules and not on existing devices.

2.1.1 New Feature

This section describes the new feature introduced in this version:

- Obtaining IPv6 NTP or DNS server using DHCPv6 with 'O' flag
- Device management over Telnet/SSH/HTTP/S in IPv6 mode
- VoIP calls over UDP/TCP/TLS in IPv6 mode
- VoIP fax over UDP/TCP/TLS in IPv6 mode
- Provisioning over HTTP/S/TFTP in IPv6 mode
- VoIP logs over IPv6
- Network diagnostic - ICMPv6

2.1.2 Known Constraints

- The following functionalities are NOT supported in IPv6 mode:
  - IPv6 Static Address Configuration
  - IPv6 Stateful Autoconfiguration (DHCPv6)
  - IPv6/IPv4 dual stack mode
  - Management over SNMP
  - Management over CWMP
  - Syslog (System, Security, CDR)
  - Quality of Service (QoS)
  - Firewall
  - MWI subscribe
  - SIP Security with IPv6 hostname
  - DNS resolving for NTP server might take up to two minutes
  - Because the device web management is unstable, it may occasionally disconnect
  - Remote Conference is not supported

2.1.3 Resolved Constraints

Table 2-1: Resolved Constraints for Version 4.5.0

Incident	Description
MP-322	Support provisioning over IPv6
MP-356	Support for TLS1.3
MP-397	AAAA query doesn't start following SRV response
MP-538	NTP domain resolving over IPv6

2.2 Version 4.5.0

This section describes Version 4.5.0. This version is supported on specific HW models

2.2.1 New Feature

This section describes the new feature introduced in this version:

- **Support SLAAC mode for IPv6**

  This version supports StateLess Address AutoConfiguration (SLAAC) mode for IPv6.

2.2.2 Known Constraints

- The following functionalities are NOT supported in IPv6 mode:
  - IPv6 static address configuration
  - IPv6 SLAAC - Router Solicitation (RS) message
  - IPv6 Stateful Autoconfiguration (DHCPv6)
  - IPv6/IPv4 dual stack mode
  - Management over SNMP
  - Management over Telnet
  - Management over CWMP
  - Syslog (system, VoIP, CDR)
  - SIP over TLS
  - Fax
  - Quality of Service (QoS)
  - Firewall
  - VoIP redundancy based on DNS SRV/NAPTR
- When changing the WAN/LAN state, a reboot is required for the change to take effect.
- SIP Proxy and Outbound SIP Proxy can be configured with IPv6 addresses only, but not with hostnames.

2.2.3 Resolved Constraints

Table 2-2: Resolved Constraints for Version 4.5.0

Incident	Description
MP-224	Improvements were made to the VoIP logs.
MP-350	The device stops answering for two minutes after receiving SIP Error 480.
MP-351	The device does not send ACK for a SIP Error 500.
MP-368	New hardware with increased RAM and flash memory was added to enable IPv6 to run on the device.
MP-390	The device stops responding to incoming calls after a certain period of idle time.
MP-391	A new TR-69 parameter called InternetGatewayDevice.Services.VoiceService.1.X_00908F_UpdateSupp, was added.
MP-392	The device responds with a "491 Server Internal Error - re-Invite inside a re-Invite" in specific call scenarios.
MP-395	When PacketSmart was removed from the web interface, it still remained in the configuration.

Incident	Description
MP-396	The device shows 4th FXS port even though Feature Key was set to 2 FXS ports.
MP-419	After making a change to the 'accept_ra' parameter, a reboot is required for the change to take effect.
MP-422	SIP lines become disabled when IPv6 mode is disabled and the proxy is configured with the IPv6 address.
MP-425	The eth0 was incorrectly getting the address of the SLAAC.
MP-428	When Local/Remote management settings were modified, a reboot was necessary for the changes to take effect.
MP-431	Tabs did not appear when IPv4 was disabled.
MP-434	Not all the IPv6 prefixes appearing on the SLAAC, are displayed on the WAN Ethernet Properties page.
MP-437	Sometimes, the IPv6 address is not sent from SLAAC.
MP-438	When disabled, IPv6 can still access the device via its link-local address.
MP-439	After doing a system restore to factory defaults on the device, the global address still incorrectly appeared on the LAN side.
MP-442	A failed CLI debugging message appeared when IPv6 was disabled.
MP-458	When configuring the Multi-Service Business Routers (MSBRs) with IPv6, after disabling IPv6, the IPv6 NTP server still appeared.
MP-473	Incorrect status for Network Bridge interface.
MP-475	On previous builds, the DNS server of IPv6 did not appear in the configuration log file.
MP-482	When IPv6 was disabled and the SIP Proxy was set to IPv6, the disconnect tone was not played.
MP-491	A button was added to the Quick Setup GUI to enable IPv4/IPv6 mode selection.
MP-493	An internal call would work only when the proxy was set to IPv4.

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3 Version 4.4.9

This section describes Version 4.4.9.

3.1 Version 4.4.9 Build 163

This section describes Version 4.4.9 Build 163.

3.1.1 New Features

This section describes the new features introduced in this version:

- **Support for IEEE 802.1X as the authenticated device:**

  The device supports two Extensible Authentication Protocol (EAP) variants:

  - MD5-Challenge (EAP-MD5):
  - EAP-TLS

  To configure these modes, refer to the IEEE 802.1X section in the latest *MP-20x Telephone Adapter User's Manual*.

3.1.2 Known Constraints

Not applicable

3.1.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved:

Table 3-1: Resolved Constraints for Version 4.4.9 Build 163

Incident	Description
MP-534	The device status is not monitored when the MP serial number is added in the One Voice Operations Center (OVOC).
MP-570	The device does not reset correctly in OVOC.

3.2 Version 4.4.9 Build 156

This section describes Version 4.4.9 Build 156.

3.2.1 New Features

This section describes the new features introduced in this version:

- **Support UPDATE method for SIP**

  This version supports the SIP UPDATE method.

- **Password values replaced with asterisks in the CLI/Configuration**

  This feature is controlled by a new parameter:

  rg_conf/hide_passwords

  By default, this parameter does not exist. Once the feature has been activated, the whole password is hidden by asterisks and the 'Configuration File' icon on the 'Advanced' web page disappears.

  - Enabling feature:

    ```
    rg_conf/hide_config=1
    ```

- Disabling feature:

```
rg_conf/hide_config=0
```

3.2.2 Known Constraints

Not applicable

3.2.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved:

Table 3-2: Resolved Constraints for Version 4.4.9 Build 156

Incident	Description
MP-384	An incorrect Network Connection status was shown in the Web interface.
MP-392	The device responds with a "491 Server Internal Error - re-Invite inside a re-Invite" in specific call scenarios.
MP-490	Unable to set the certificate with the 4098 bit RSA key.
MP-494	Incorrect default packetization time for the first voice codec.
MP-495	DTAG: Hiding passwords by displaying asterisk instead
MP-496	Merge new Faxback application [1.20.11.30].
MP-501	Logging the history of generated 'random_tod/check_tod' and actual provisioning attempt times.
MP-505	Serial number value reported in DHCP Option 125 is missing the 'VT' prefix.

3.3 Version 4.4.9 Build 146

This section describes Version 4.4.9 Build 146.

3.3.1 New Features

Not applicable

3.3.2 Known Constraints

Not applicable

3.3.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved:

Table 3-3: Resolved Constraints for Version 4.4.9 Build 146

Incident	Description
MP-351	MP-202 does not send ACK message on ‘500 Server Internal Error’.
MP-390	MP-202 does not respond to incoming calls after a specific period of idle time.
MP-391	CWMP: Adding X.parameter for 'voip/signalling/sip/update_support_enabled'.

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3.4 Version 4.4.9 Build 142

This section describes Version 4.4.9 Build 142.

3.4.1 New Features

Not applicable

3.4.2 Known Constraints

Not applicable

3.4.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved:

Table 3-4: Resolved Constraints for Version 4.4.9 Build 142

Incident	Description
MP-211	Failed EMS response when upgrading same file twice
MP-215	Drop from the conference with an internal call failed
MP-232	MP-202: Phone1 LED turns red after hang-up internal call
MP-253	MP-202 incorrect CID on analog device in specific scenarios
MP-259	[HGC]: Cancel all registered ABBD not work as expected
MP-278	Update default client certificate
MP-280	Syslog format improvement
MP-285	Support SIP Update method
MP-307	WEB: missing GW entry in static route rule
MP-317	Supporting provisioning based on random Time-Of-Day
MP-318	[HGC]: Cannot block/unblock Caller ID per call
MP-339	MP-202 response 491 then receiving INVITE inside re-INVITE
MP-341	WEB: Pressing on ‘check now’ button (configuration) cause reboot

3.5 Version 4.4.9 Build 111

This section describes Version 4.4.9 Build 111.

3.5.1 New Features

This section describes the new features introduced in this version:

- **Changing Default Cipher Suites for SIP over TLS**

You can now change the default cipher suites to be used (or removed) for SIP over TLS.

- **Changing Default Cipher Suits for Provisioning**

You can now change the default cipher suites to be used (or removed) for Provisioning.

- **Setting Provisioning Time of Day (TOD)**

You can now configure the device to perform provisioning at random or fixed times using a specified configuration. You can also configure default behavior when if no random TOD or Fixed TOD has been configured,

3.5.2 Known Constraints

This section lists known constraints:

Table 3-5: Known Constraints for Version 4.4.9 Build 111

Incident	Description
106501	Connected Line Presentation ('P-Asserted-Identity') is not updated after 'Forward on no answer'.
109783	Speed Dial: Destination SIP port cannot be changed from '5060'.
109789	Internal service: Call Forward 'Busy' mode does not work.
109790	Internal service: Call Forward 'No Reply' mode does not work.
109791	Internal service: DND prevents incoming calls to all SIP lines instead of the relevant SIP line.

3.5.3 Resolved Constraints

This section lists constraints from previous releases that have now been resolved:

Table 3-6: Resolved Constraints for Version 4.4.9 Build 111

Incident	Description
MP-1	ARP requests received on WAN, answered with LAN IP instead of a WAN IP.
MP-6	Incorrect Caller ID is displayed for forwarded incoming calls.
MP-14	Unnecessary DNS NAPTR query is sent before an INVITE is sent.
MP-15	An old configuration path for Remote Conference URI was removed.
MP-25	Software Fastpath doesn't work on bridge configuration.
MP-45	Internal calls do not work in specific scenarios.
MP-54	Payment terminal transaction failed when Transparent mode was used.
MP-62	SIP destination port remains unchanged for calls executed by speed dial.
MP-70	The device does not answer incoming SIP keep-alive OPTIONS.
MP-75	FaxBack: Signaling was routed to the network instead of the pFax application.
MP-93	Incoming DTMF (SIP INFO method) for "*" and "# " are played like the "0".
MP-98	Internal calls do not work when ATA is configured for TLS/SRTP.
MP-101	VoIP failure was encountered when receiving specific RTCP packets.
MP-106	Support for Huawei's softswitch format for MWI Notify event.
MP-116	URL value with .cfg extensions provided by DHCP Option 66 is being ignored by CPE.
MP-138	Call forward busy embedded service doesn't work on Line 2.
MP-139	Caller ID + prefix was not converted into ‘00’ in the Call Log history of the phone.
MP-167	Bad handling of REFER call flow for specific scenarios.
MP-168	Voice is not established when INVITE is missing the SDP header.

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4 Device Gateway Specifications

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

Table 4-1: 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, G.722 ▪ 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 (not applicable to MP-202) ▪ Link Layer Discovery Protocol (LLDP) support ▪ Faxback integration (Fax over HTTPS) – Optional ▪ File server ▪ Printer server ▪ DLNA Media server

Feature	Details
	▪ FXS Voice Menu
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	▪ MP-202: Power Supply 12VDC/1A ▪ MP-202R/MP-204R/MP204: 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 (not applicable to MP-202)
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

Feature	Details
	✓ ZTE K3805-Z ✓ 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-204B: 280 grams; 167 x 133 x 33 mm ▪ MP-204R: 280 grams; 167 x 133 x 33 mm

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