AudioCodes 300HD Series of High Definition IP Phones

MP-20x Series

MediaPack™ Series Telephone Adapters with Integrated Router

MP-20x Telephone Adapter Release Notes

Version 3.0.3

Document #: LTRT-50518

AudioCodes

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Related Documentation

Document Name
MP-20x Broadband VoIP Gateway & Wireless Router Quick Guide
MP-20x Telephone Adapter User's Manual

Reader's Notes

1 Introduction

Version 3.0.3 is a unified firmware version for MP-20x Revisions A, B and C. This includes four separate 3.0.3 software images, one for each of the following MP-20x models:

- MP-202A
- MP-20xB
- MP-202C-A
- MP-202C-R and MP-202C-W

MP-202A features 2 FXS lines, a single Ethernet WAN, and a single Ethernet WAN.

MP-20xB features 1 to 4 FXS lines and an optional FXO line, a single Ethernet WAN, and a single Ethernet WAN.

MP-202C features two FXS lines, an Ethernet WAN interface, four Ethernet LAN interfaces (with an internal Layer-2 switch), and an 802.11b/g Wireless LAN (WiFi). MP-202C is an all-in-one unit, combining a VoIP telephone adaptor, a wireless router, and an Ethernet switch.

The MP-20x can be connected to an external modem for connectivity to broadband networks such as ADSL, Cable, or WiMAX (only for MP-202C models).

The MP-20x is interoperable with leading softswitches and SIP Application Servers, offering legacy phone services such as caller ID, call waiting, and call forwarding. In addition, the MP-20x includes an internal router with DHCP, NAT, and L2TP/PPTP/PPPoE capabilities, enabling subscribers to connect their home PC or LAN hub/switch to the Residential Gateway (RGW).

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 state-of-the-art 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 both telephony ports to free valuable bandwidth resources. The "Voice over Data" prioritization algorithm prevents degradation in voice quality even during large data transfers.

As mentioned above, the MP-20x series is designed for full interoperability with leading softswitches and SIP servers for deployment in various network environments. Throughout the years, AudioCodes has invested significant effort in establishing, and complying with, the leading and evolving VoIP standards. Support of SIP, which is commonly found in Voice-over-Broadband (VoB) networks, assures seamless integration and rapid deployment.

The MP-20x software specifications are summarized in the following table:

Table 1-1: MP-20x Software Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP - RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ IPv4, TCP, UDP, ICMP, ARP,TLS (SIP Over TLS) [1] ▪ PPPoE (RFC 2516) ▪ L2TP (RFC 2661) ▪ PPTP (RFC 2637) ▪ DNS, Dynamic DNS [2] ▪ 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 [3], traffic shaping ▪ STUN (RFC 3489)
Media Processing	▪ Voice Coders: G.711, G.723.1, G.729A/B, G.726 [4] ▪ 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 ▪ Message Waiting Indication ▪ Call Forward ▪ 3-Way Conferencing [5]
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 [6]

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1 Not supported by MP-202A

2 Not supported by MP-202A

3 Not supported by MP-202C & MP-202A

4 MP-20x Rev B models only.

5 MP-202C does not support two concurrent three-way conference calls.

6 Not supported by MP-202C-A & MP-202A

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
Wireless LAN[7]	▪ Wireless LAN - 802.11b/g Wireless Access Point ▪ Wireless Security: ✓ RADIUS Server (802.1x/WPA Client Authentication) ✓ WPA ✓ WPA2 ✓ WPA/WEP Mixed Mode ✓ TKIP Encryption ✓ MAC Filtering

[7] MP-202C-W model only.

Reader's Notes

2 Version 3.0.3

The following describes the new features, resolved constraints and known constraints of Version 3.0.3.

2.1 New Features

Version 3.0.3 offers the following new features and support:

- Version 3.0.3 supports MP-202A, MP-202C series, and MP-20xB series, providing the following available configurations:

Model	FXS	FXO	WAN	LAN	Wi-Fi
MP-202A	2	-	1	1	-
MP-201B	1	-	1	1	-
MP-202B	2	-	1	1	-
MP-204B	4	-	1	1	-
MP-202C-A	2	-	1	1	-
MP-202C-R	2	-	1	4	-
MP-202C-W	2	-	1	4	802.11b/g

- This is the first 3.0.x version that supports MP-202A.
- TR-069 enhancements:
- Traffic Shaping configuration
  - Traffic Priority configuration
  - Multi-Telnet commands via RPC commands
  - Configurable parameters:
    - Outbound Proxy Enable
    - *RegistrationUseSIPProxyIPandPort* and *RegistrarUse*
    - Jitter Minimum Delay and Optimization Factor
    - Caller ID Type2 Enable
    - Services: Call forward keys, MWI subscribe, and 3-way conf enable
- LST (Local Support and Troubleshooting) Phase 1
- Replace # with %23 to activate special services (when *IsSpecialDigits* set to 1)
- MP-202C-A and MP-202C-R can be used in L2 Switch mode (configurable)
- CLI commands in the MP-20x configuration file
- Configuration file download through Telnet

2.2 Resolved Constraints

The following constraints have now been resolved:

- Unable to dial to a phone number that begins with '#'.
- In certain configurations (with rmt_config or rmt_upd enabled), a memory leak and a CPU load =100%, occurs within several weeks of MP-202 operation.
- Endless loop DHCP discover - MP-20x sends the BootP (DHCP) discover, it receives the offer, but a few seconds later it sends another discover message (may cause Endless loop of DHCP discover messages).
- Java links - Using specific Java URLs reveals admin pages for users.
- wget - Unit using wget to download the conf file endlessly after a 404 Error.
- Wrong call history hours – The time of a missed call (in call waiting state only) is wrong.
- The time in the system log is still incorrect.
- Daylight Saving Time timers do not work correctly.
- Traffic shaping – MP-20x does not allow making traffic shaping rules larger than 9765 on WAN Ethernet or L2TP
- TR-069 - After a terminating force upgrade action, this action remains at status: 'Active'/'Pending' task.
- TR-069 - Identify MP-20x with a MAC address (instead of Serial number) in the ACS.
- MP-202 sends an INVITE to the network with URI 999888777999.
- SIP Security - Allow setting more than one IP address in the SIP Security field.
- SIP 'INVITE' messages with 'referred-by' field are rejected by the MP-202.
- Register Failed Expires mechanism improvement (See “What’s New in Version 3.0.1" - “Improved SIP registration process” for more details of the feature).
- PPPoE fails to terminate. It then takes time to re-connect.
- MP-202 response with SIP 404 message upon incoming INVITE with UserID that includes '-‘.
- MP-202C-W - Can't share folder when connecting streamer and PC to MP-202.
- MP-202C - Certain phones do not ring (RevC SLIC update was needed).
- MP-202C – T.38 stability issues
- MP-202C – Switch To Voice from Fax doesn’t work well.
- Fix Australian ring cadences.
- Australia A-Tick Homologations Support.

2.3 Known Constraints

Version 3.0.3 includes the following known constraints:

- When upgrading from software Version 2.6.4 to 3.0.0, 3.0.1 or 3.0.3, MP-20x automatically restores to factory defaults.
- MP-202A Constraints – Due to memory limitations, the following features were removed comparing to Version 2.6.4:
  - SNMP
  - VLANs
  - Dynamic DNS
  - SIP over TLS
  - SIP logs (A separate version with logs will be provided for debugging when needed).
- MP-20xB Constraints – Due to memory limitations, the following feature was removed comparing to Version 2.6.4:
  - SIP logs (A separate version with logs will be provided for debugging when needed).
- MP-202C Constraints:
  - VLANs are not supported.
  - The SIP application logging mechanism is not enabled in the MP-202C firmware.
  - LAN-WAN bridging is not supported. For L2 switching between all Ethernet ports, please contact AudioCodes.
  - Traffic Shaping should be activated only on the “Default WAN”.
  - Bandwidth limitation in the Rx direction is not functional. Under normal residential use, limiting the Rx traffic is not required and does not improve the QoS.
  - It is not possible to perform MAC cloning to a MAC address that belongs to a PC that is connected to one of the LAN ports.
  - When there is a voice message, the MWI (“envelope”) icon is not displayed on the phone’s LCD; only a stutter tone is heard.
  - Reliable Modem Transfer is supported only in transparent mode.
  - When the three-way conferencing feature is enabled (using the Web management - **Voice Over IP** > **Services**) and two lines are enabled, the following limitations apply:
    - When both phone lines are active (off hook or ringing), a user can’t place a call on hold. Doing such an action causes a busy tone to be played.
    - When there are two active calls on one line (three-way conference call, or first call active and second call on hold), the other line will be unavailable. If the user picks up the phone, a busy tone is heard. An incoming call to the second line will be rejected.
    - When both lines are in a call, a call waiting to either line will be rejected.
    - There is no confirmation tone after a successful call transfer in flash + digits mode.
  - MP-202C Rev. A model: Establishing the first outgoing call may take 1 to 2 seconds longer than other calls.
- Remote Configuration File URL parameter length is limited to 96 characters.
- During a voice call session, the user can send a fax to the remote party only once. If the user attempts to send another fax during this call session, fax transmission fails.
- When the Fax Mode is set to “Bypass”, MP-20x fails to switch back to voice after the fax has been transmitted.
- Occasionally, after software or hardware reboot, TCP registration does not occur.
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call.

- QoS traffic shaping: enabling 'TCP Serialization' may cause problems for viewing real-time video streams on a PC that is connected to the MP-20x.
- Caller ID Type II audio indication is sometimes heard by both the calling and the called parties. The remote side does not hear the FSK; only an RFC 2833 DTMF tone.
- To ensure that the OPTIONS Keep-Alive feature is fully functional, the remote side must answer with any message, including the SIP 501 "Not implemented" SIP message.
- Enabling or disabling the Periodic Checking of Configuration File feature requires a device reboot.
- A SIP NOTIFY message with a "check-sync" event causes a device restart even if there are ongoing conversation calls.
- MP-202C-W: Wi-Fi password fields in the Web interface are displayed as clear text.
- In a scenario in which an authentication is required for Call Hold, Music on Hold may not be sent from the media server to the remote machine.

3 Previous Releases

3.1 Version 3.0.1

3.1.1 What's New in Version 3.0.1

Version 3.0.1 offers the following new features and support:

- Version 3.0.1 supports both the MP-202C series and MP-20xB series, providing the following available configurations:

Model	FXS	FXO	WAN	LAN	Wi-Fi
MP201-B	1	-	1	1	-
MP202-B	2	-	1	1	-
MP204-B	4	-	1	1	-
MP202C-A	2	-	1	1	-
MP202C-R	2	-	1	4	-
MP202C-W	2	-	1	4	802.11b/g

- When upgrading from 2.6.4 to 3.0.1, MP-20x retains the current value of the **config_url** parameter, allowing the unit to download the new configuration file.
- Restoring default configuration by dialing a specific digit sequence. The default sequence is "*factory" (or *3228679 in digits). By default, this option is disabled.
- Read-only Web access permissions for users. A read-only user can view the current configuration in the embedded Web server, but cannot modify it.
- Australia and Russia regional settings support.
- Improved SIP registration process. A new timer has been added for periodic registration attempt in case of a registration failure (e.g. due to a network problem). This is provided by the new parameter, 'Register Failed Expires' under the SIP Proxy and Registrar group (**Voice Over IP** menu > **Signaling Protocol** tab).
- Caller ID is now displayed during Call Waiting Reminder Ring (CWRR).
- Flash-hook duration detection threshold is now configurable. This is provided by the new parameters, 'Flash Min' and 'Flash Max' (**Voice Over IP** menu > **Dialing** tab). The default detection range is 100 to 1000 msec.
- To improve the general performance of MP-202C, the second VoIP call is now automatically configured to use G.711. This feature enables the service provider to select G.729 as the preferred codec and still reserve enough resources when there are two simultaneous calls.
- Basic "User" Web page now includes a link to the Port Forwarding page.
- HTTPS firmware and configuration download mechanism now supports download with or without a certificate (configurable).
- Transfer mode of the fax CED tone is now configurable (**Voice Over IP** menu > **Voice and Fax** tab).
- Login username is now not case-sensitive. However, the user password is still case-sensitive.
- Support provided for the **rmt_config** CLI command.

3.1.2 Resolved Constraints in Version 3.0.1

The following constraints have been resolved in this release:

- Regional settings configuration for Argentina has been updated.
- Various limitations with SIP over TLS.
- Redundant Proxy mechanism is now supported with SIP over TCP.
- Redundant Proxy mechanism is now supported for a non-default SIP port.
- Bug fix in the Redundant Proxy mechanism when running in “symmetric” mode.
- Bug fix for MP-202C when initiating a three-way conference from line 1, off-hooking and then on-hooking line 2 (resulted in short ringing).
- Polarity reversal now functions properly in MP-202C.
- Manual configuration of Busy Tone and Reorder Tone is now supported.
- Bug fix: increasing number of Auth header fields in the SIP Call Messages caused error 513 – message too large.
- Payload Type RFC 2833 can now be configured to 97 without colliding with iLBC.
- Dynamic payload type for VBD is now supported.
- Occasional Wi-Fi disconnection in MP-202C-W has now been resolved.
- Quick Setup Web page now supports configuring WPA security.
- Noise during Do Not Disturb (DND) mode has now been resolved.
- Pressing flash immediately after enabling call forward (*72) resulted in call-forward deactivation bug has now been resolved.
- Displayed Caller ID time information is now accurate even when in daylight saving time.
- DHCP Options 66/67 are now enabled by default (to allow out-of-the-box provisioning).
- DHCP Options 66/67 now support the ini file format.
- Unique naming of configuration files are now supported when saving using the Web interface.
- Bug fix for MP202C-A: when dialing to a busy line, one channel was not released properly.
- Default QoS profile changed to ”Triple Play User”, as a result Rx bandwidth is no longer limited.
- 'anonymous' CID in the number line is now supported.
- Bug fix in transferring of flash events via SIP.
- When performing a “blind transfer”, the ringing line did not receive the line description. This has now been resolved.
- Bug fix concerned with off-hooking the phone during DND (Do Not Disturb).
- Speed-dial for a local line is now fully supported.
- MP-20x Revision B: Call was disconnected when ringing into load higher than 5 REN.
- Bug fix for fax transmission failure if G.711 was not in the selected codec list.
- Bug fix for echo cancellation mechanism of MP-202C.
- Bug fix for SIP INVITE message with 'referred-by' field were rejected by the MP-20x.
- TR-069 now supports using HTTPS transport.
- Many unnecessary DHCP request were sent when operating with Windows 2003 DHCP server. This has now been resolved.
- Downloading a firmware file using FTP without providing user and password, caused the recovery image to reboot. This has now been resolved.
- Bug fixed for firmware upgrade using TFTP from the recovery image.
- Web Welcome page is now displayed.
- Web access issues when using Mozilla Firefox and Google chrome has now been resolved.

3.1.3 Known Limitations in Version 3.0.1

Version 3.0.1includes the following known limitations:

- When upgrading from software version 2.6.4 to 3.0.0 or 3.0.1, MP-20x automatically restores to factory defaults.
- MP-202C Limitations:
  - VLANs are not supported.
  - The SIP application logging mechanism is not enabled in the MP-202C firmware.
  - LAN-WAN bridging is not supported. For L2 switching between all Ethernet ports, please contact AudioCodes.
  - Traffic Shaping should be activated only on the “Default WAN”.
  - Bandwidth limitation in the Rx direction is not functional. Under normal residential use, limiting the Rx traffic is not required and does not improve the QoS.
  - It is not possible to perform MAC cloning to a MAC address that belongs to a PC that is connected to one of the LAN ports.
  - When there is a voice message, the MWI (“envelope”) icon is not displayed on the phone’s LCD; only a stutter tone is heard.
  - Reliable Modem transfer is supported only in Transparent mode.
  - When the three-way conferencing feature is enabled (using the Web management - **Voice Over IP** > **Services**) and two lines are enabled, the following limitations apply:
    - When both phone lines are active (off hook or ringing), a user can’t place a call on hold. Doing such an action causes a busy tone to be played.
    - When there are two active calls on one line (three-way conference call, or first call active and second call on hold), the other line will be unavailable. If the user picks up the phone, a busy tone is heard. An incoming call to the second line will be rejected.
    - When both lines are in a call, a call waiting to either line will be rejected.
    - There is no confirmation tone after a successful call transfer in flash + digits mode.
  - MP-202C Rev A model: Establishing the first outgoing call may take 1 to 2 seconds longer than other calls.
- Remote Configuration File URL parameter length is limited to 96 characters.
- Timestamp in system log is incorrect.
- During a voice call session, the user can send a fax to the remote party only once. If the user attempts to send another fax during this call session, fax transmission fails.
- When the Fax Mode is set to “Bypass”, MP-20x fails to switch back to voice after the fax has been transmitted.
- Occasionally, after software or hardware reboot, TCP registration does not occur.
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call.
- QoS traffic shaping: enabling ‘TCP Serialization’ may cause problems for viewing real-time video streams on a PC that is connected to the MP-20x.
- Caller ID Type II audio indication is sometimes heard by both the calling and the called parties. The remote side does not hear the FSK; only an RFC 2833 DTMF tone.
- To ensure that the OPTIONS Keep-Alive feature is fully functional, the remote side must answer with any message, including the SIP 501 “Not implemented” SIP message.
- Enabling or disabling the Periodic Checking of Configuration File feature requires a device reboot.

- A SIP NOTIFY message with a “check-sync” event causes a device restart even if there are ongoing conversation calls.
- MP-202C-W: Wi-Fi password fields in the Web interface are displayed as clear text.

In a scenario in which an authentication is required for Call Hold, Music on Hold may not be sent from the media server to the remote machine.

3.2 Version 3.0.0

3.2.1 What's New in Version 3.0.0

Version 3.0.0 offers the following new features and support:

- Supports both the MP-202C series and MP-20xB series, providing the following available configurations:

Model	FXS	FXO	WAN	LAN	WiFi
MP201-B	1	-	1	1	-
MP202-B	2	-	1	1	-
MP203-B	2	1	1	1	-
MP204-B	4	-	1	1	-
MP202C-A	2	-	1	1	-
MP202C-R	2	-	1	4	-
MP202C-W	2	-	1	4	802.11b/g

- Automatic detection of the Internet connection type – DHCP, PPPoE, or L2TP. This feature allows the service provider to use a single MP-20x pre-configuration (factory settings) for all connection types. When first connected to the network, MP-20x automatically detects the Internet connection type and uses the relevant set of pre-configured parameters. In addition, this feature allows users who change their connection type (e.g., from Cable/L2TP to ADSL/PPPoE) to continue using their existing MP-20x – by restoring the default settings and allowing MP-20x to detect the new connection type.
- Support for DHCP options 66 (TFTP server name) and 67 (Bootfile name).
- Support for DHCP option 61 (Client identifier) as part of the DHCP request.
- Automatic dialing per line is supported.
- Support for a new keep-alive mechanism, which sends dummy UDP packets instead of a SIP OPTION message.
- New Dial Plan behavior (same as used for the digits map).
- TR-069 and TR-104 enhancements.

The MP-20x software specifications are summarized in the following table:

Table 3-1: MP-20x Software Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP - RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ IPv4, TCP, UDP, ICMP, ARP,TLS (SIP Over TLS) ▪ PPPoE (RFC 2516) ▪ L2TP (RFC 2661) ▪ 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 or Layer-2 switching (currently not supported) ▪ STUN (RFC 3489)
Media Processing	▪ Voice Coders: G.711, G.723.1, G.729A/B, G.726[8] ▪ 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 ▪ Message Waiting Indication ▪ Call Forward ▪ 3-Way Conferencing[9]
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[10]
Packetization	▪ RTP/RTCP Packetization (RFC 3550, RFC 3551) ▪ DTMF Relay (RFC 2833)
Security	▪ HTTPS for Web-based configuration ▪ Password protected Web pages (MD5)

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[8] MP-20x Rev B models only.

[9] MP-202C does not support two concurrent three-way conference calls.

[10] Not supported by MP-202C-A.

Feature	Details
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
Wireless LAN[11]	▪ Wireless LAN - 802.11b/g Wireless Access Point ▪ Wireless Security: ✓ RADIUS Server (802.1x/WPA Client Authentication) ✓ WPA ✓ WPA2 ✓ WPA/WEP Mixed Mode ✓ TKIP Encryption ✓ MAC Filtering

3.2.2 Resolved Constraints in Version 3.0.0

The following constraints were resolved in Version 3.0.0:

- Improved TR-069 and TR-104 management.
- Remote configuration file update: if a file does not exist in the HTTP server, the WGET function is not re-called every few seconds.
- MP-20x can now receive a partial configuration file from the ACS.
- Hebrew Web management graphical user interface (GUI) errors have been corrected.
- The redundant proxy's port number can now be configured to a different number from the default port.
- MP-202C - QoS Traffic Shaping only during a VoIP session is now supported.
- Changing again to the same regional settings does not cause an unnecessary reboot.
- MP-202C now supports three-way conferencing.
- MP-202C SIP Security feature is now supported.
- MP-202C WAN and LAN status are updated when the cable is unplugged.
- Sending duplicated register requests by MP-20x has now been resolved.
- The MP-20x now receives MWI status periodically (even if it wasn’t changed), but it is updated only when there is a real change of the status.
- When MP-20x performs automatic dialing, a Ringback tone is now heard (instead of a Reorder or Howler tone).
- Non-digit characters in the calling number are now supported.
- The user can now hear the remote side between the first Call Waiting tone and the second Call Waiting tone.
- Maximum Hook Flash timeout is now configurable (and the default value is 1 second).

11 MP-202C-W model only.

3.2.3 Known Limitations in Version 3.0.0

Version 3.0.0 includes the following known limitations:

- **When upgrading from software version 2.6.4 to 3.0.0, MP-20x automatically restores to factory defaults.**
- Rev C models: When using L2TP, MP-20x is unable to connect two calls using the G.729 coder.
- Rev C models: Modem is supported only in Transparent mode .
- During a voice call session, the user can send a fax to the remote party only once. If the user attempts to send another fax during this call session, fax transmission fails.
- The Redundant Proxy operates only with the default SIP port 5060.
- The Redundant Proxy does not function when the SIP transport protocol is set to TCP.
- When the Fax Mode is set to “Bypass”, MP-20-x fails to switch back to voice after the fax has been transmitted.
- Occasionally, after software or hardware reboot, TCP registration does not occur.
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call.
- QoS traffic shaping: enabling ‘TCP Serialization’ may cause problems for viewing real-time video streams on a PC that is connected to the MP-20x.
- Caller ID Type II audio indication is sometimes heard by both the calling and the called parties. The remote side does not hear the FSK; only an RFC 2833 DTMF tone.
- To ensure that the OPTIONS Keep-Alive feature is fully functional, the remote side must send any message, including the “501 Not implemented” SIP message.
- Enabling or disabling the Periodic Checking of Configuration File feature requires a device reboot.
- A SIP NOTIFY message with a “check-sync” event causes a device restart even if there are ongoing conversation calls.

3.3 Version 2.9.2 (MP-202C)

3.3.1 What's New in Version 2.9.2

- Version 2.9.2 supports the following MP-202C models:

Model	FXS	WAN	LAN	WiFi
MP202C-A	2	1	1	-
MP202C-R	2	1	4	-
MP202C-W	2	1	4	802.11b/g

Note: Version 2.9.2 supports only MP-202 Revision C models.

The MP-202C software specifications are summarized in the following table:

Table 3-2: MP-202C Software Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP - RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ IPv4, TCP, UDP, ICMP, ARP,TLS (SIP Over TLS) ▪ PPPoE (RFC 2516) ▪ L2TP (RFC 2661) ▪ 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 or Layer 2 switching (currently not supported) ▪ STUN (RFC 3489)
Media Processing	▪ Voice Coders: G.711, G.723.1, G.729A/B ▪ 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

Feature	Details
Telephony Features	▪ Call Hold and Transfer ▪ Call Waiting ▪ Message Waiting Indication ▪ Call Forward ▪ 3 way Conference
Configuration/ Management	▪ Embedded Web Server for configuration and management ▪ Remote firmware upgrade and configuration by HTTP,TFTP,FTP and HTTPS ▪ Command-Line Interface (CLI) over Telnet
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
Wireless LAN	▪ Wireless LAN - 802.11b/g Wireless Access Point ▪ Wireless Security: ✓ RADIUS Server (802.1x/WPA Client Authentication) ✓ WPA ✓ WPA2 ✓ WPA/WEP Mixed Mode ✓ TKIP Encryption ✓ MAC Filtering

3.3.2 Resolved Constraints in Version 2.9.2

The following constraints were resolved in Version 2.9.2:

- Key sequence mode “Flash + digit” is now supported.
- Configuration File Encryption (3DES) is now supported.
- The ini-file format of the configuration file is now supported.
- Dual-image management to ensure recovery in case of power outage during the upgrade process is now supported.
- Universal Plug and Play (UPnP) feature is now supported.
- Fixed bug of Caller ID Type II when using G.729 with packetization time 30 msec.

3.3.3 Known Limitations in Version 2.9.2

Version 2.9.2 includes the following known limitations:

- TR-069 and TR-104 for remote configuration and management are currently only partially supported.
- Three-Way Conferencing is supported only when a single line is enabled.
- SIP Security feature is currently not supported.
- Modem is supported in Transparent mode only.

- WAN and LAN statuses are not updated when the cable is unplugged.
- Enabling QoS Traffic Shaping only during a VoIP session is not supported. Currently, traffic shaping limits the bandwidth at all times, even if the user is not making a VoIP call.
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call.
- QoS traffic shaping: Enabling 'TCP Serialization' may cause problems viewing real-time video streams on a PC that is connected to the device.
- Caller ID Type II audio indication is sometimes heard by both the calling and the called parties. The remote side doesn't hear the FSK; only an RFC 2833 DTMF tone.
- To ensure that the OPTIONS Keep-Alive feature is fully functional, the remote side must send any message, including the "501 Not implemented" SIP message.
- Enabling or disabling of the Periodic Checking of Configuration File feature requires a device reboot.

A SIP NOTIFY message with a "check-sync" event causes a device restart even if there are ongoing conversation calls.

3.4 Version 2.9.0 (MP-202C)

3.4.1 What's New in Version 2.9.0

- Version 2.9.0 is the first Version for MP-202 Revision C.
- Version 2.9.0 supports the following MP-202C models:

Model	FXS	WAN	LAN	WiFi
MP202C-R	2	1	4	-
MP202C-W	2	1	4	802.11b/g

Notes:

- Version 2.9.0 support only MP-202 Revision C models.
- MP-202C-A (with two FXS ports, one WAN and one LAN) will be supported in the next release.

The MP-202C software specifications are summarized in the following table:

Table 3-3: MP-202C Software Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP - RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ IPv4, TCP, UDP, ICMP, ARP,TLS (SIP Over TLS) ▪ PPPoE (RFC 2516) ▪ L2TP (RFC 2661) ▪ 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 or Layer 2 switching (currently not supported) ▪ STUN (RFC 3489)
Media Processing	▪ Voice Coders: G.711, G.723.1, G.729A/B ▪ 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 ▪ Message Waiting Indication ▪ Call Forward

Feature	Details
Configuration/ Management	▪ Embedded Web Server for configuration and management ▪ Remote firmware upgrade and configuration by HTTP,TFTP,FTP and HTTPS ▪ Command-Line Interface (CLI) over Telnet
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
Wireless LAN	▪ Wireless LAN - 802.11b/g Wireless Access Point ▪ Wireless Security: ✓ RADIUS Server (802.1x/WPA Client Authentication) ✓ WPA ✓ WPA2 ✓ WPA/WEP Mixed Mode ✓ TKIP Encryption ✓ MAC Filtering

3.4.2 Known Limitations in Version 2.9.0

Version 2.9.0 includes the following known limitations:

- Key sequence mode “Flash + digit” is currently not supported.
- Three-Way Conferencing is currently not supported.
- TR-069 and TR-104 for remote configuration and management are currently not supported.
- Configuration File Encryption (3DES) is currently not supported.
- Dual image management is currently not supported.
- SIP Security feature is currently not supported.
- Universal Plug and Play (UPnP) feature is currently not supported.
- WAN and LAN status are not updated when the cable is unplugged.
- Caller ID Type II is not supported when using G.729 with packetization time 30 msec.
- The ini-file format of the configuration file is not supported.
- Enabling QoS Traffic Shaping only during a VoIP session is not supported. Currently, traffic shaping limits the bandwidth at all times, even if the user is not making a VoIP call.
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call.
- QoS traffic shaping: Enabling ‘TCP Serialization’ may cause problems viewing real-time video streams on a PC that is connected to the device.
- Caller ID Type II audio indication is sometimes heard by both the calling and the called parties. The remote side doesn't hear the FSK; only an RFC 2833 DTMF tone.
- To ensure that the OPTIONS Keep-Alive feature is fully functional, the remote side must send any message, including the “501 Not implemented” SIP message.

- Enabling or disabling the Periodic Checking of Configuration File feature requires a device reboot.
- A SIP NOTIFY message with a “check-sync” event causes a device restart even if there are ongoing conversation calls.

3.5 Version 2.6.4 (MP-20x-B)

3.5.1 What's New in Version 2.6.4

- Support for dynamic traffic shaping (traffic shaping only during call): Traffic shaping is critical in residential VoIP gateways because of the bottleneck created in the ADSL or Cable modem, mainly in the upload direction. In previous versions, traffic shaping limited the bandwidth at all times, even if the user was not making a VoIP call. Consequently, the service provider had to configure the QoS traffic shaping transmit (Tx) bandwidth according to the user's specific upload bandwidth. Configuring a lower value resulted in a lower upload bandwidth (not just during VoIP calls).

  This version introduces a new mechanism that enables the service provider to configure two upload traffic shaping bandwidth values:

  - "Tx Bandwidth"
  - "Tx Bandwidth during Call"

  MP-20x normally uses the "Tx Bandwidth" value. When the user makes a VoIP call (i.e. any phone/s connected to MP-20x is ringing or off-hook), MP-20x switches to use the "Tx Bandwidth during Call" value.

> **Note:** To enable traffic shaping only during a call, the parameter "Tx Bandwidth" can be left at its default value (i.e. 100 Mbps).

- Support for Fax Fallback from T.38 to Transparent. MP-20x first attempts to use T.38. If it receives a SIP 488 ("Not implemented") response or a SIP Re-INVITE with G.711 audio packets from the remote side, it switches to Transparent mode and then sends a Re-INVITE with G.711 (only if not already sent by the other side).

  MP-20x should be configured as follows:

  - Fax Transport Mode = T.38
  - Modem Transport Mode = Transparent
  - Enable CNG Detection = Enable
  - Error Correction Mode = Disable
  - update_fax_to_transparent_enable = 1 (configured only from the CFG file - cannot be configured from the Web interface)
  - All other fax parameters should be set to their default values
- A new option has been added that allows the user to disable Caller ID Type II.
- AudioCodes proprietary SNMP MIB has been added for VoIP configuration and status information and for basic network configuration.

The MP-20x software specifications are summarized in the following table:

Table 3-4: MP-20x Software Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP - RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ IPv4, TCP, UDP, ICMP, ARP,TLS (SIP Over TLS) ▪ PPPoE (RFC 2516) ▪ L2TP (RFC 2661) ▪ 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 or Layer 2 switching (currently not supported) ▪ STUN (RFC 3489)
Media Processing	▪ Voice Coders: G.711, G.723.1, G.729A/B, G.726 Optional - iLBC, AMR (separate software image) ▪ 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 ▪ 3-Way Conferencing ▪ Message Waiting Indication ▪ Call Forward
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
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

3.5.2 Resolved Constraints in Version 2.6.4

The following bugs were resolved in Version 2.6.4:

- The Dial Plan size has now been increased to 1024.
- The “SwitchToVoice” Re-INVITE includes three G.729 lines; now, only a single line occurs.
- Simultaneous sending and receiving of faxes (T.38) from two lines fails; now, the line does not fail.
- A new option for the out-of-service tone has been added (“Regular Dial tone”).
- The 'Mute RTP' option (upon hold) is now configurable.
- Registration is lost after a while due to memory leak; now, registration is retained.

3.5.3 Known Limitations in Version 2.6.4

Version 2.6.4 includes the following known limitations:

- The MP-20x model MP-203 Rev B is not supported. The most updated released version for this model is Version 2.6.3.
- The Web interface is not automatically refreshed during the firmware upgrade process.
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call.
- A silence period of about three seconds occurs after pressing the ‘Flash’ key during a conversation (typically, the user presses ‘Flash’ + ’1’, ‘Flash’ + ’2’, or ‘Flash’ + ’3’). This limitation does not occur when in ”Flash only” key sequence mode.
- QoS traffic shaping: Enabling ‘TCP Serialization’ may cause problems viewing real-time video streams on a PC that is connected to MP-20x.
- Caller ID Type II audio indication is sometimes heard by both the calling and called parties. The remote side doesn't hear the FSK; only an RFC 2833 DTMF tone.
- To ensure that the OPTIONS Keep-Alive feature is fully functional, the remote side must send any message, including the “501 Not implemented” SIP message.
- Enabling or disabling the Periodic Checking of Configuration File feature requires a device reset.
- A SIP NOTIFY message with a “check-sync” event with a message body causes a device reset even if there are ongoing conversation calls.

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