CPE & Access Gateway Products

MP-20x series

Telephone Adapters with Integrated Router

MP-202 Telephone Adapter Release Notes

Version 2.4.0

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NUERA

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- US: 1-800-966-8372 opt 3
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Abbreviations and Conventions

Each abbreviation, unless widely used, is spelled out in full when first used, and only industry-standard terms are used throughout this manual. 0x before a number denotes hexadecimal notation. DSP (Digital Signaling Processor) and VoPP (Voice over Packet Processor) may be used interchangeably.

1 MP-202 Telephone Adapter

The MP-202 is a 2-line SIP gateway allowing residential and SOHO subscribers to connect ordinary POTS telephones or fax machines, and is interoperable with leading Softswitches and SIP Application Servers for enabling legacy phone services such as caller ID, call waiting, and call forwarding. In addition, the MP-202 includes an internal router with DHCP, NAT and PPPoE capabilities enabling subscribers to connect their home PC or LAN hub/switch to it.

Utilizing Nuera 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 save valuable bandwidth. The “Voice over Data” prioritization algorithm prevents degradation in voice quality even during large data transfers.

The MP-20x Series is designed for full interoperability with leading Softswitches and SIP Servers for deployment in various network environments. Throughout the years, Nuera has invested significant effort in complying with the leading and evolving VoIP standards. Support of the Session Initiation Protocol (SIP), which is commonly found in Voice over Broadband (VoB) networks, assures seamless integration and rapid deployment.

1.1 What’s New in Version 2.4.0

- The following new features are supported in version 2.4.0:
    - MWI (Message Waiting Indication). If a user has an unheard voice mail message, a stutter dial tone is heard when the user picks up the phone. In addition, the MP-202 generates an FSK signal to the phone to indicate that a message is waiting. If the telephone connected to the MP-202 supports this feature, an MWI ‘envelope icon’ is displayed.
    - PRACK (Provisional Response Ack) messages. The PRACK request plays the same role as ACK, but for provisional responses. PRACK also allows the media negotiation to be performed at an earlier stage, before the call is answered.
    - User Agent field in a REGISTER message contains the product version (VI43857).
    - User Permissions. Whenever a user with user name ‘user’ is added via Advanced>Users, the user is only able to access the Quick Setup screen. (VI44244).
    - Added support of SIP OPTIONS messages. The SIP method OPTIONS allows a UA to query another UA or a proxy server as to its capabilities. This allows a client to discover information about the supported methods, content types, extensions, codecs, etc., without ‘ringing’ the other party.
    - Added TR-069 WAN Management Protocol. TR-069 is a protocol that enables remote server management of an MP-202. The protocol is useful, for example, for remotely and securely controlling the MP-202. TR-069 defines several Remote Procedure Call (RPC) methods, as well as a large number of parameters which can be set or read.

- Call Forward (CFW). Permits a user to redirect incoming calls addressed to him/her to another number. The user's ability to originate calls is unaffected by Call Forward. Three types of Call Forwarding exist:

    1. CF Unconditional (CFU). After CFU is activated, incoming calls are forwarded independently of the status of the endpoint.
    2. CF Busy (CFB). After CFB is activated, incoming calls are forwarded only if the endpoint is busy, i.e., if all lines are active.
    3. CF No Reply (CFNR). After CFNR is activated, incoming calls are forwarded only if the endpoint does not answer before a pre-configured timeout.

    A known constraint of CFW is that activation is not saved after reboot.

- Added support for G.723 5.3 kpbs (LBR) encoding and 60 msec frames for both G.723 High and Low Bit Rate.

■ In addition, the following changes were implemented:

- The user can choose a waiting call SIP reply message - 180 Ringing or 182 Queued. (VI45119)
- Upgrade to RADVISION MTF2.1.0.0.
- Dial tone only after SIP proxy registration succeeds (VI44246).
- Restore default is not automatically performed when upgrading from version 2.2.X to 2.4.0.

■ Bug Fixes: The following bugs were fixed in version 2.4.0:

- VI46177: Failed to get the local address of type TLS to set in the via header.
- VI45550: Performing a call transfer and pressing the Flash button while the phone is still ringing causes the phone to ring ceaselessly.
- VI46963: VLANs in Bridge mode isn't functioning.
- VI47052: CID bug: When receiving a call; the time which is displayed is GMT and not local time.
- VI47346: The MP-202 sends the field 'User-Agent' with a typo. It sends it as 'UserAgent'; without the '-'.
- VI47399: The value of fax Transparent mode was incorrect.
- VI48199: Modem calls malfunction.
- VI48270: Some Proxy servers rejected a SIP REGISTER Message that contains transport=UDP in the request line.
- VI48690: Echo Canceler performance improvements were made; false IBS detection was solved.
- VI48762: RTP wasn't stopped when the MP-202 was held.
- VI43410: Some statistical information in RTCP packets is not calculated correctly.

1.1.1 Known Limitations

Following are known limitations:

- The default device software version does not support IPSec. Contact Nuera for availability of IPSec support as a separate software option.
- Fax can be sent between the two local lines only if the chosen fax transport mode is Transparent (in G.711).
- The Web is not refreshed automatically during the firmware upgrade process. [VI43754]
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call. [VI43562]
- A silence period of about 3 seconds is created after pressing the ‘Flash’ key during a conversation (normally, the user presses ‘Flash’+’1’, ‘Flash’+’2’ or ‘Flash’+’3’). This limitation does not occur when in ‘Flash only’ key sequence mode. [VI43424]
- When pressing ‘Flash’+’1’ or ‘Flash’+’2’ (for call hold and transfer), the DTMF is sometimes heard at the remote side. [VI42919]
- QOS traffic shaping: Enabling ‘TCP Serialization’ may cause problems viewing real-time video streams on a PC which is connected to the device.
- It may take up to 2 minutes for the PPTP or L2TP tunneling protocols to reconnect to the WAN after the device is rebooted. [VI44056]
- 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 a 2833 DTMF. [VI47366]
- SIP packets longer than 1500 bytes may be handled incorrectly by the MP-202. [VI47051]
- In some scenarios, performance degradation occurs after a long L2TP session [VI44016].

Table 1-1 shows the supported features:

Table 1-1: MP-202 Software Specifications

Feature	Details
VoIP Signaling Protocols	SIP - RFC 3261, 2327 (SDP)
Data Protocols	IPv4, TCP, UDP, ICMP, ARP PPPoE – RFC2516 L2TP – RFC 2661 PPTP – RFC 2637 DNS, Dynamic DNS WAN to LAN Layer3 routing with: ▪ DHCP Client/Server – RFC 2132 ▪ NAT – RFC 3022, Application Layer Gateway (ALG) ▪ Firewall ▪ QoS - Priority queues, VLAN 802.1p,Q tagging, traffic shaping or Layer 2 switching (not supported in this version)
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 Remote firmware-upgrade and configuration by HTTP Telnet
Packetization	RTP/RTCP Packetization - RFC 3550, 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 ▪ On/Off Hook, Flash Hook

2 Previous Versions

2.1 Version 2.2.2

2.1.1 What's New

- Key Sequence now features two options. In screen VoIP > Dialing > Advanced, a new section called 'Keys Sequence' provides the two options:
    - The first option (default) is 'Flash + digits sequence', where a sequence of Flash + 1 holds a call or toggles between two existing calls. Flash + 2 makes a call transfer. Flash + 3 establishes a 3-way conference.
    - The second option is 'Flash only' and uses only the Flash button. Pressing Flash holds an existing call; a dial tone is heard and the user is able to initiate a second call. Once the second call is established, on-hooking transfers the first (held) call to the second call. Alternatively, the user can also hold an existing call by pressing the Flash button, getting a dial tone, initiating a second call and establishing a 3-way conference by again pressing the Flash button after the second call was initiated. In the case of call waiting, pressing the Flash button puts the active call on hold and answers the waiting call; pressing Flash again toggles between these two calls.
- Remote Configuration File Update - In the firmware upgrade screen, a new section called 'Configuration File Update' was added, enabling users to configure the URL details of the server from which the configuration file is downloaded and to enable/disable the automatic configuration file update process.
- When upgrading from version 2.2 to version 2.2.2, the device's configuration file is preserved. Users are not required to reconfigure it.
- Increase in the maximum length of a dialed number to 19 digits
- A Watchdog software mechanism was added to prevent L2TP connection losses due to low resources.
- The Fax Bypass payload type number can be configured from the Web management.
- A new tab 'Services' was added to the VoIP screens. The user can select (via this tab in the Web management) one of two SIP response methods: Ringing (180) or Queued (182). This is used when dialing to the MP-202 during an ongoing call.
- Bug Fixes: The following issues are fixed in version 2.2.2:
    - VI 46444: Caller ID Type II was functioning incorrectly when the regional settings were configured to 'ISRAEL'.
    - VI 44260: RTP session didn't open when receiving SIP 183 Session Progress Message with SDP message. Call Progress Tones were not relayed correctly over the network.
    - VI 44497: There was a deviation on the DC-BIAS that caused an Echo Canceler problem on some devices
    - Fixed the range validation of DTMF Relay RFC 2833 Payload Type and G.726/16 Payload Type. The new validity range is 96-127.
    - VI 46979: Bridge mode was functioning incorrectly in earlier versions.

2.1.2 Known Limitations

Following are known limitations:

- When upgrading from version 1.4.0 (and lower), the device's factory settings are automatically restored. Users must reconfigure their settings after running the new version.
- The default device software version does not support IPSec. Contact Nuera for availability of IPSec support as a separate software option.
- Fax can be sent between the two local lines only if the chosen fax transport mode is Transparent (in G.711).
- Calls between two local lines can't be placed when the LAN connection and the WAN connection are both disconnected.
- In Fax Bypass and VBD modes, if the user dials from a regular phone to a fax machine the channel locks. [VI43944]
- The Web is not refreshed automatically during the firmware upgrade process. [VI43754]
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call. [VI43562]
- A silence period of about 3 seconds is created after pressing the 'Flash' key during a conversation (normally, the user presses 'Flash'+'1', 'Flash'+'2' or 'Flash'+'3'). This limitation does not occur when in 'Flash only' key sequence mode. [VI43424]
- Some statistical information in RTCP packets is not calculated correctly. [VI43410]
- When pressing 'Flash'+'1' or 'Flash'+'2' (for call hold and transfer), the DTMF is sometimes heard at the remote side. [VI42919]
- QOS traffic shaping: Enabling 'TCP Serialization' may cause problems viewing real-time video streams on a PC which is connected to the device.
- It may take up to 2 minutes for the PPTP or L2TP tunneling protocols to reconnect to the WAN after the device is rebooted. [VI44056]
- Caller ID Type II audio indication is sometimes heard by both the calling and the called parties. [VI47366]
- The SIP header field syntax 'UserAgent' is incorrect. It should be built with the hyphen. It should be 'User-Agent' and not 'UserAgent'. [VI47346]
- When receiving a call, the Caller ID time displayed on the phone is always GMT regardless of the MP-202 configured system time. [VI 47052]
- SIP packets longer than 1500 bytes may be handled incorrectly by the MP-202. [VI47051]

2.2 Version 2.2

2.2.1 New Features

- PPTP and L2TP Internet connection - in addition to existing support for DHCP and PPPoE Internet connections, version 2.2.0 adds support for PPTP and L2TP, common protocols in Internet over cable.
- Outbound SIP proxy - users can now configure the address of an outbound proxy. All SIP messages are sent directly to the outbound proxy as the first hop. The outbound proxy forwards the SIP messages to their actual destination.
- Call Waiting Caller ID - in addition to existing support for Type I (on-hook) Caller ID, version 2.2.0 adds support for Type II (off-hook) Caller ID (the Bellcore standard).
- Support added for the pound key (#) to indicate the end of a dialed string.
- Multiple languages - the MP-202's Web interface now supports multiple languages (English, French, Russian, Spanish, Korean, Chinese, Japanese, Hebrew, German and Italian).
- SIP registration status LED indication - a successful registration ('online' state) is indicated by a LED flashing slowly (a very short flash every 4 seconds). This indication is per phone line.
- VoIP line status web page - a new web page displays information on the two lines - the hook state, registration status, and information about each active call.
- Additional options for setting the voice codec frame duration were added.
- Support for blind call transfer was added, in addition to the existing support for attended transfer.
- Automatic Gain Control - when enabled, the MP-202 automatically adjusts the voice gain (towards the local phone or towards the remote side), according to the signal strength.
- Support for Silence Compression and Comfort Noise Generation was added.
- Support was added for digit map and dialing plan. The digit map defines the format of the dialed string. The dialing plan translates the dialed strings to SIP destination addresses.
- Support for 3-way conferencing was added.
- A new QoS mechanism enables the service provider to define traffic priorities using internal priority queues, to tag different traffic types in Layer 2 (802.1p) or Layer 3 (DSCP) and to define traffic shaping / bandwidth limitation rules. The QoS mechanism enables users to make VoIP calls behind cable or DSL modems with limited uplink bandwidth, while running bandwidth-consuming applications on their PC.
- Support for VBD (Voice Band Data) fax transport mode. In this mode, when a fax tone is detected, a re-INVITE message is sent to switch the call to G.711.
- Alternating ringing - to reduce power consumption, when two incoming calls are received simultaneously (one for each phone line), an alternating ringing is generated.
- To improve interoperability, support was added for additional coder strings in the SDP message (for example, "G711U", "G.711U", "g711u", "g.711u", "pcmu" in addition to "PCMU").
- To avoid conflicting WAN and LAN subnet masks when connecting the MP-202 behind some 3rd party routers, the default LAN IP address was changed from 192.168.1.1 to 192.168.2.1. The IP address distribution range was changed accordingly.

- Support for out-of-band DTMF relay via SIP INFO messages was added.
- The maximum length of a dialed number increased from 15 to 18 digits.
- Toggling between two active calls is performed by pressing 'Flash' + '1'; in previous versions, 'Flash' + '1' and 'Flash' were used alternately.
- Advanced VoIP configuration - new configuration parameters added:
  - Gateway Name - User Domain (this string appears in INVITE messages)
  - SIP registrar address and port
  - Outbound proxy address and port
  - SIP T1, T2, T4 and INVITE timers
  - Dial-tone timeout
  - Voice volume for line 1 and line 2
  - Automatic Gain Control parameters
  - Jitter Buffer minimal delay and optimization factor
  - Silence compression and comfort noise parameters
  - Echo Cancelation (enable / disable)
  - Fax transport parameters (mode, bypass coder, CNG detection, max rate, error correction mode)
  - Configuration of RTP payload types for RFC 2833 and G.726-16 packets.

- Bug Fixes: The following issues are fixed in version 2.2.0:
  - There was no timeout when dialing a speed-dial number.
  - The system crashed when disconnecting the WAN cable during an active call.
  - Caller ID blocking did not function.
  - Several call hold, transfer, and call waiting related bugs.
  - There was an invalid digit length when using RFC 2833 DTMF relay.
  - When the remote side was configured to 10 msec frames and the local to 20 msec, voice was distorted.
  - There was no Web indication on a disconnected WAN cable.
  - When configuring the remote SIP UDP port, SIP registration was sent to the configured port but SIP INVITEs were still sent to the default UDP port - 5060.
  - The system crashed when the local side was configured to Fax Bypass and the remote side to T.38.
  - When dialing to a phone off-hooked but not in an active call (as well as dialing to an extension's own phone number), a call waiting ringback tone was heard instead of a busy tone.
  - There was no SIP indication for coder negotiation failure.

2.2.2 Known Limitations

Following are the known limitations:

- The default MP-202 software version does not support IPSec. Contact Nuera for availability of IPSec support as a separate software option.
- When upgrading to version 2.2.0, the MP-202's factory settings are automatically restored. Users must reconfigure their settings after running the new version.
- Fax can be sent between the 2 lines only if the chosen fax transport mode is Transparent (in G.711).
- Calls can't be placed when the LAN connection and the WAN connection are both disconnected.
- In some revisions of the MP-202 hardware (FXSS module version 0112124-02), the level of the background noise is higher than in other hardware versions.
- In fax bypass and VBD modes, if the user dials from a regular phone to a fax machine, the channel will become locked. [VI43944]
- The Web is not refreshed automatically during the firmware upgrade process. [VI43754]
- After the dial tone timeout has expired (and a fast-busy tone is played), the user can still make an outgoing call. [VI43562]
- A silence period of about 3 seconds is created after pressing the 'flash' key during a conversation (normally the user will press 'flash'+'1', 'flash'+'2' or 'flash'+'3'). [VI43424]
- Some statistical information in the RTCP packets is not calculated correctly. [VI43410]
- When pressing 'flash'+'1' or 'flash'+'2' (for call hold and transfer), the DTMF is sometimes heard at the remote side. [VI42919]
- When using fax bypass or VBD, the selected bypass coder must be included in the list of coders enabled for voice. [VI42918]
- It occasionally takes 2 minutes for the PPTP or L2TP tunneling protocols to reconnect to the WAN after the device is rebooted. [VI 44056]

2.3 Version 1.4.0

2.3.1 New Features

- Regional settings – Version 1.4.0 supports telephony parameters for multiple countries. Users can now choose their location from a list of supported countries and the MP-202 will automatically configure the correct set of Call Progress Tones, FXS impedance values, and Caller ID type. In addition, the Service Provider can modify the parameters of a specific country or add support for a new country. The following countries are currently supported: China, France, Germany, Israel, Netherlands, UK, and USA.
- Configuration of the proxy/registrar domain name (FQDN) - previous MP-202 versions allowed users to configure the IP address of the proxy/registrar. Version 1.4.0 allows users to configure either the IP address or the domain name.
- Separate authentication values per phone port - the SIP authentication values (user and password) can now be configured independently per phone port. This allows the user to have two independent VoIP lines.
- Configuration of the remote SIP UDP port - it is now possible to configure the UDP port of the SIP registrar (the default value is 5060).
- Fax support - Version 1.4.0 supports fax bypass and T.38 fax relay.
- Call waiting - Version 1.4.0 adds support for call waiting. When an incoming call arrives during an active call, a short tone is heard and the user is able to switch between the current call and the new call (using Flash + ’1’).
- Configuration of G.729 packetization time - Users can now configure the packetization time of G.729 (to 10, 20 or 30 msec). Previously, only 10 msec was supported.
- The following issues were fixed in version 1.4.0:
  - An off-hooked phone loses its dial tone when another phone dials to it.
  - Several interoperability problems, including call transfer with some SIP proxies.

2.3.2 Known Limitations

- When upgrading to version 1.4.0, the MP-202’s factory settings are automatically restored. Users must reconfigure their settings after running the new version.
- When configuring the remote SIP UDP port, SIP registration is sent to the configured port, but SIP INVITEs are still sent to the default UDP port - 5060.
- During fax bypass, no SIP reINVITE is issued. The coder automatically changes to G.711 with a payload type of 102.
- When dialing to a phone that is currently off-hook but not in an active call (including dialing to an extension’s own phone number), a call waiting ringback tone is heard instead of a busy tone.
- Invalid RFC 2833 DTMF relay - The duration of the DTMF digits relayed over RTP per RFC 2833 is incorrect.
- 3-way conferencing is not supported in this version.
- There is no SIP indication for coder negotiation failure.

2.4 Version 1.2.0

2.4.1 New Features

- Additional codecs: G.726, iLBC and AMR are now supported, in addition to the existing support for G.711, G.729 and G.723. Note that iLBC and AMR are enabled as a separate MP-202 image, on request. Also note that when upgrading from version 1.0.1, in order to use the added codecs users must first restore the default configuration (Advanced > Restore Defaults).
- Users can select codec priorities: In previous versions, the priority (order) of the selected codecs was fixed. Version 1.2.0 enables users to configure the order of the selected codecs.
- Pushbutton to restore factory settings: A new mechanism enables users to restore default factory settings. This can be useful to users who have changed the MP-202's configuration so that they're prevented from logging into the device's web management. To restore factory settings, press (with a paper clip) the pushbutton (located at the base of the MP-202) for 5 seconds during power-up.
  Note: This feature requires Bootloader version 2.3.0.2 or higher.
- LED indications: The LEDs labelded 'Phone 1' and 'Phone 2' now reflect the status of the phones connected to the MP-202.

Phone 1 / Phone 2 LED Status	Indication
ON	During power-up and when the phone is off-hook.
OFF	Normal / the phone is on-hook
Flashing	Phone is ringing

- Remote configuration: Version 1.2.0 enables the MP-202 to be configured automatically by downloading a configuration file from a remote server. The automatic configuration can be activated via CLI (Command Line Interface) as well.
- Support for symbol '.' in the authentication user name.
- The following issues were fixed in version 1.2.0:
  - One-way voice on the first call after power-up.
  - Improved FXS configuration - solved problem of a 'click' that was heard several seconds after the beginning of the call.

2.4.2 Known Limitations

- Invalid RFC 2833 DTMF relay - The duration of the DTMF digits relayed over RTP per RFC 2833 is incorrect.
- An off-hooked phone loses its dial tone when another phone dials to it.
- 3-way conferencing is not supported in this version.
- Call waiting is not supported in this version.
- T.38 fax relay is not supported in this version.
- In some scenarios (for example, after calling the Vonage SIP proxy), the MP-202 stops generating the dial tone.
- There is no SIP indication for coder negotiation failure.

2.5 Version 1.0.1

2.5.1 New Features

- Version 1.0.1 is a maintenance version that fixes issues found in version 1.0.0 beta. The following major issues were fixed in version 1.0.1:
  - One-way voice - Occassionally, voice towards the FXS phone is muted.
  - Stability – the MP-202 crashes after 48 hours due to a memory leak.

2.5.2 Known Limitations

- Invalid RFC 2833 DTMF relay - The duration of the DTMF digits relayed over RTP per RFC 2833 is incorrect.
- An off-hooked phone loses its dial tone when another phone dials to it.
- 3-way conferencing is not supported in this version.
- Call waiting is not supported in this version.
- T.38 fax relay is not supported in this version.
- A single click is heard several seconds after the beginning of a call.
- In some scenarios (for example, after calling the Vonage SIP proxy), the MP-202 stops generating the dial tone.
- There is no SIP indication for coder negotiation failure.

2.6 Version 1.0.0

2.6.1 New Features

- Version 1.0.0 was the first software release of the MP-202; it should be regarded as a beta version for evaluation.

2.6.2 Known Limitations

- Invalid RFC 2833 DTMF relay - The duration of the DTMF digits relayed over RTP per RFC 2833 is incorrect.
- An off-hooked phone loses its dial tone when another phone dials to it.
- 3-way conferencing is not supported in this version.
- Call waiting is not supported in this version.
- T.38 fax relay is not supported in this version.
- One-way voice - Occassionally, voice towards the FXS phone is muted.
- A single click is heard several seconds after the beginning of a call.
- In some scenarios (for example, after calling the Vonage SIP proxy), the MP-202 stops generating the dial tone.
- There is no SIP indication for coder negotiation failure.

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