

# AudioCodes™

## MP-20x Telephone Adapter

### *Frequently Asked Questions (FAQs)*





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## Notice

This document provides Frequently Asked Questions (FAQ's) for AudioCodes MP-20x Telephone Adapter.

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## Introduction

This document provides answers to frequently asked questions (FAQs) about AudioCodes MP-20x analog telephone adapter. These FAQs are grouped under the following topics:

- Web access
- Default configuration
- Session Initiation Protocol (SIP) configuration
- Voice-over-IP (VoIP) troubleshooting
- Regional settings
- Bandwidth
- Functionalities
- Software upgrade
- Telnet/advanced configuration

For further information regarding MP-20x, you can refer to the following Web URL: <http://www.audiocodes.com/products/mediapack-20x>.

If you have any questions that are not addressed in this document, please contact AudioCodes Customer Support for further assistance.

# Frequently Asked Questions

## Web Access

**Q1: How must I configure my PC in order to connect to the MP-20x Web interface?**

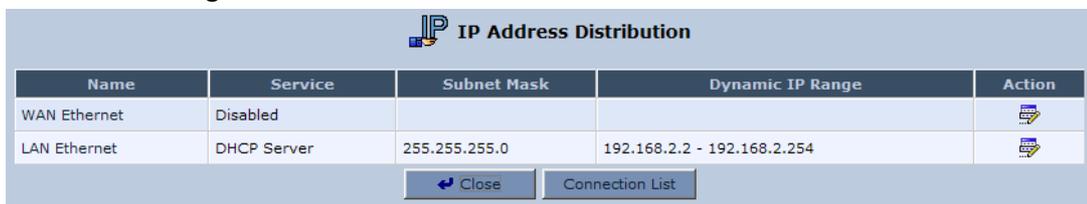
**A:** For an explanation on configuring your PC for connection to MP-20x Web interface, refer to Section “Configuring Network Parameters” in *LTRT-52702 MP-20x FXS Telephone Adapter Quick Installation Guide*.

**Q2: Why can't I access MP-20x Web interface when my PC is connected to the LAN?**

**A:** Access problems to MP-20x Web interface from the LAN may be caused by one of the following:

- Your PC is not connected to the LAN - verify that your PC is physically connected to the LAN and not to the WAN.
- The IP address (URL) that you enter in your Web browser is incorrect (the MP-20x default LAN IP address is 192.168.2.1).
- The Web interface's login name and password are incorrect (the default login user name and password is “admin”).
- By default, MP-20x allocates IP addresses to the LAN (i.e., DHCP server) with the IP address range of 192.168.2.2 to 192.168.2.254. The 'IP Address Distribution' screen allows you define the MP-20x DHCP server IP address pool, as shown below:

**Figure 1: MP-20x IP Address Pool Allocated to LAN Hosts**



Name	Service	Subnet Mask	Dynamic IP Range	Action
WAN Ethernet	Disabled			
LAN Ethernet	DHCP Server	255.255.255.0	192.168.2.2 - 192.168.2.254	

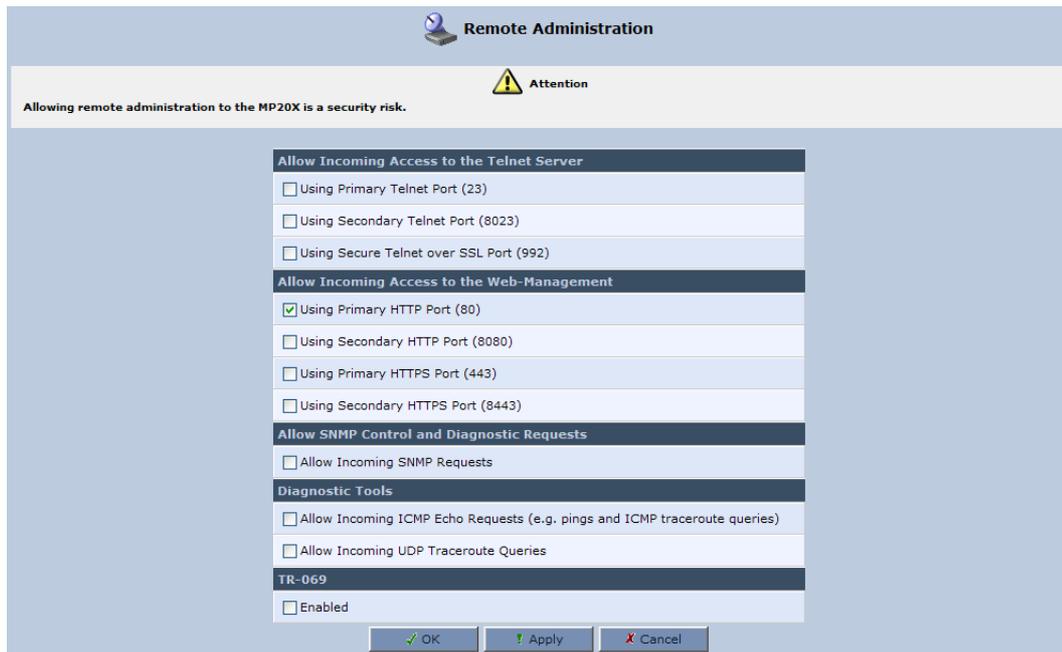
Any connected PC can behave as a DHCP client or with a static IP address within the range defined above.

- Two PCs cannot use the same IP address for 60 minutes or less, even if the first PC was disconnected. Each IP address within the distribution range is held for a maximum of 60 minutes if used before.

**Q3: Why can't I access the MP-20x Web interface while connected to the WAN?**

- A:** By default, access to the Web interface from the WAN is disabled.
- **To enable access from the WAN:**
    1. Logon to the Web interface from the LAN.
    2. Access the 'Remote Administration' screen (**Advanced** menu > **Remote Administration** icon).
    3. Under the **Allow Incoming Access to the Web-Management** group, select the Using Primary HTTP Port check box to enable remote connection, as shown below:

**Figure 2: Enabling Remote Administration (WAN) in the Remote Administration Screen**



**Q4: Why can't I ping MP-20x?**

- A:** By default, the parameter Allow Incoming ICMP Echo Requests is disabled (in the 'Remote Administration' screen - see figure Figure 2). Therefore, to successfully ping MP-20x, enable this parameter by selecting this check box.

**Q5: How do I ping/trace route from MP-20x to other devices in the network?**

- A:** To ping/trace from MP-20x to other devices in the network, refer to *LTRT-50606 MP-20x Telephone Adapter User's Manual Ver. 2.6.2*, Section 13.6, "Diagnostics".

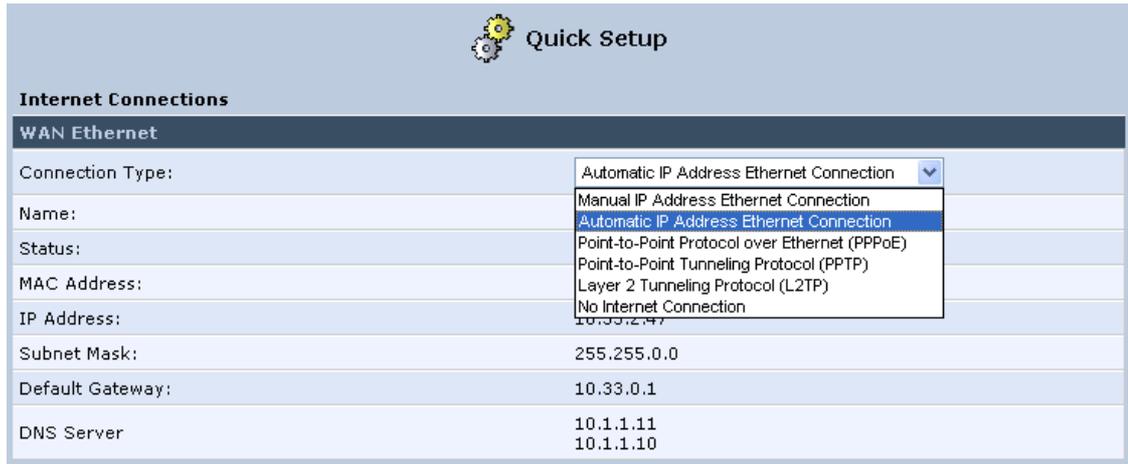
**Q6: Can I use different types of Internet connections such as L2TP, PPTP, and PPPoE?**

**A:** Yes.

➤ **To change the Internet connection type:**

1. Logon to the Web interface from the LAN.
2. Access the 'Quick Setup' screen (**Quick Setup** menu).
3. From the 'Connection Type' drop-down list, select the required Internet connection type.

**Figure 3: Selecting Internet Connection Type in the Web Interface**



The screenshot shows the 'Quick Setup' web interface. At the top, there is a gear icon and the text 'Quick Setup'. Below this is the 'Internet Connections' section, which is currently set to 'WAN Ethernet'. A dropdown menu is open for the 'Connection Type' field, showing the following options: 'Automatic IP Address Ethernet Connection' (selected), 'Manual IP Address Ethernet Connection', 'Automatic IP Address Ethernet Connection', 'Point-to-Point Protocol over Ethernet (PPPoE)', 'Point-to-Point Tunneling Protocol (PPTP)', 'Layer 2 Tunneling Protocol (L2TP)', and 'No Internet Connection'. The rest of the form fields are filled with default values.

Internet Connections	
WAN Ethernet	
Connection Type:	Automatic IP Address Ethernet Connection
Name:	
Status:	
MAC Address:	
IP Address:	10.33.0.1
Subnet Mask:	255.255.0.0
Default Gateway:	10.33.0.1
DNS Server:	10.1.1.11 10.1.1.10

## Default Configuration

**Q1: How do I restore MP-20x to factory default settings?**

**A:** You can restore MP-20x to default settings using one of the following methods:

■ **Web interface:**

- a) Logon to the Web interface.
- b) Access the 'Restore Defaults' screen (**Advanced** menu > **Restore Defaults**  icon).
- c) Click **OK**.

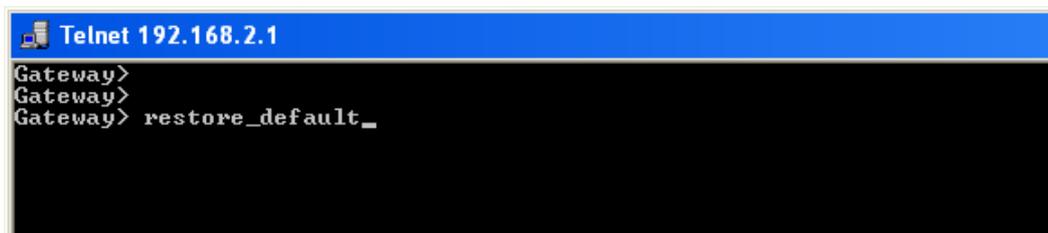
**Figure 4: Restoring Factory Defaults in the Restore Defaults Screen**



■ **Telnet:**

- a) Telnet MP-20x from the LAN.
- b) At the CLI prompt, run the following command:  
`restore_defaults`

**Figure 5: Restoring Defaults using Telnet**



■ **Hardware Reset button:** The **Reset** button is located on the rear panel of MP-20x.

- a) Disconnect MP-20x from power.
- b) Using a sharp pointed object (such as a pin), press the button and at the same time, re-connect power to MP-20x.

For additional information, refer to *LTRT-50606 MP-20x Telephone Adapter User's Manual Ver. 2.6.2*, Section 13.5, "Restoring Default Settings".

## *SIP Configurations*

**Q1: Which SIP parameters do I have to configure for MP-20x?**

**A:** For configuring SIP parameters for MP-20x, refer to the following documents:

- *LTRT-52702 MP-20x FXS Telephone Adapter Quick Installation Guide*, Section “Configuring VoIP Parameters”
- *LTRT-50606 MP-20x Telephone Adapter User’s Manual Ver. 2.6.2*, Chapter 5, “Configuring VoIP Parameters” (providing a detailed description)

## VoIP Troubleshooting

**Q1: How do I troubleshoot SIP calls?**

**A:** ➤ **To troubleshoot SIP calls:**

1. Telnet MP-20x from the LAN.
2. Run the following command:  
`voip_set_log_level 1`

**Figure 6: Defining Log Level using Telnet**

```

Telnet 192.168.2.1
Gateway>
Gateway>
Gateway>
Gateway> voip_set_log_level 1
Setting the VoIP logging level.

Returned 0
Gateway>
    
```

3. Logon to the Web interface.
4. Access the 'System Monitoring' screen (**System Monitoring** menu), and then select the **System Log** tab:

**Figure 7: Viewing System Logs in the Web Interface**

The screenshot shows the 'System Monitoring' web interface. At the top, there are tabs for 'Connections', 'Traffic', 'System Log' (which is selected), 'System', and 'Voice Over IP'. Below the tabs are buttons for 'Close', 'Clear Log', and 'Refresh'. A message below the buttons says 'Press the Refresh button to update the data.' Below this is a table with columns: Time, Event, Event-Type, and Details.

Time	Event	Event-Type	Details
Jan 1 02:11:38 2003	System Log	Message	<pre> daemon.warn ----- TX SIP MESSAGE ----- REGISTER sip:22.212.21.11 SIP/2.0 From: &lt;sip:0000000002@22.212.21.11&gt;;tag=100dfd88-2f02210a-13c4-50029-6-ec7fa41-6 To: &lt;sip:0000000002@22.212.21.11&gt; Call-ID: 100f1098-2f02210a-13c4-50029-6-7a4cc16b-6 CSeq: 1 REGISTER Via: SIP/2.0/UDP 10.33.2.47:5060;branch=z9hG4bK-6-1a56-a71c236 Max-Forwards: 70 Supported: replaces,100rel Allow: REGISTER, INVITE, ACK, BYE, REFER, NOTIFY, CANCEL, INFO, OPTIONS, PRACK, SUBSCRIBE Expires: 3600 Contact: &lt;sip:0000000002@10.33.2.47:5060&gt; User-Agent: MP202 FXS/2.6.3_build_12_test_3 Content-Length: 0  ----- END OF SIP MESSAGE -----                     </pre>
Jan 1 02:11:38 2003	System Log	Message	<pre> daemon.warn ----- TX SIP MESSAGE ----- REGISTER sip:22.212.21.11 SIP/2.0 From: &lt;sip:0000000001@22.212.21.11&gt;;tag=100dfbf0-2f02210a-13c4-50029-5-2973c9da-5 To: &lt;sip:0000000001@22.212.21.11&gt; Call-ID: 100f1098-2f02210a-13c4-50029-4-2e71707c-4 CSeq: 1 REGISTER Via: SIP/2.0/UDP 10.33.2.47:5060;branch=z9hG4bK-5-162f-74c9d24 Max-Forwards: 70 Supported: replaces,100rel Allow: REGISTER, INVITE, ACK, BYE, REFER, NOTIFY, CANCEL, INFO, OPTIONS, PRACK, SUBSCRIBE Expires: 3600 Contact: &lt;sip:0000000001@10.33.2.47:5060&gt; User-Agent: MP202 FXS/2.6.3_build_12_test_3 Content-Length: 0  ----- END OF SIP MESSAGE -----                     </pre>
Jan 1 02:11:33 2003	System Log	Message	<pre> daemon.err get_host_name() - could not find start of host string                     </pre>
Jan 1 02:11:33 2003	System Log	Message	<pre> daemon.warn estream: Can't read from fd 39 Connection reset by peer(131)                     </pre>

Alternatively, you can monitor the WAN connection with a network sniffer such as Wireshark. For additional information, refer to *LTRT-58201 MP-20x Debugging and Diagnostic Tools Application Note Ver. 2.6.1*, Sections 2.1 and 2.2, and Chapter 4 "SIP Logs".

**Q2: Why do I hear a fast busy tone after I pick up the phone (even though all SIP parameters have been configured)?**

- A:** The fast busy tone heard after the phone is off-hooked may be caused by the following:
- The SIP user is not registered. To check if your user is registered, perform the following:
    - a) Logon to the Web interface.
    - b) Access the 'System Monitoring' screen (**System Monitoring** menu), and then select the **Voice Over IP** tab.

**Figure 8: Verifying SIP User Registration in the Web Interface**



Line	Line 1	Line 2
Phone State	On Hook	On Hook
SIP registration	Not Registered	Not Registered
State	Idle	Idle
Origin	-	-
Remote Phone Number	-	-
Remote ID	-	-
Duration	-	-
Type	-	-
Encoder	-	-
Decoder	-	-
Packets Sent	-	-
Packets Received	-	-
Bytes Sent	-	-
Bytes Received	-	-
Packets Lost	-	-
Packets Loss Percentage	-	-
Jitter (ms)	-	-
Round Trip Delay (ms)	-	-

- The line's SIP Proxy user name and password is incorrect. To check this, perform the following:
  - a) Logon to the Web interface.
  - b) Access the 'Voice Over IP' screen (**Voice Over IP** menu), and then select the **Line Settings** tab.
  - c) Edit the required line.

**Figure 9: Verifying SIP Proxy User Name and Password in the Web Interface**



**Line Settings**

Line Number: 1

User ID: 0000000001

Block Caller ID

Display Name: Line 1

**SIP Proxy**

Authentication User Name: user1

Authentication Password: .....

**Advanced Line Parameters**

Line Voice Volume (-31 to +31 db): 0

Enable Supplementary Services

OK Cancel Basic <<

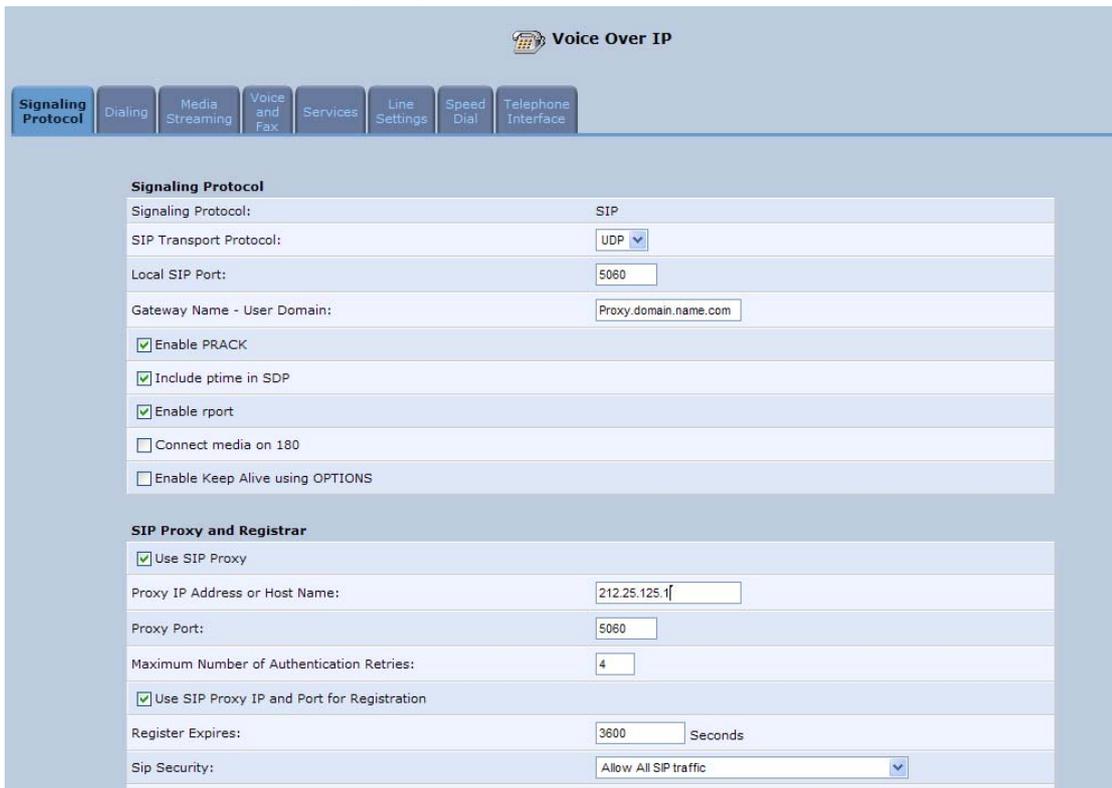
**Q3: Why do I hear a fast busy tone when making outgoing calls (even though SIP parameters have been configured)?**

**A:** The fast busy tone may be caused by the Proxy server not been configured with a domain name:

➤ **To configure the Proxy with a domain name:**

1. Logon to the Web interface.
2. Access the 'Voice Over IP' screen (**Voice Over IP** menu), and then select the **Signaling Protocol** tab.
3. Under **Signaling Protocol** group, in the Gateway Name – User Domain field, enter the Proxy's domain name.

**Figure 10: Configuring Proxy's Domain Name in the Web Interface**



The screenshot shows the 'Voice Over IP' configuration page in the AudioCodes web interface. The 'Signaling Protocol' tab is active. The 'Gateway Name - User Domain' field is set to 'Proxy.domain.name.com'. Other fields include 'Signaling Protocol' (SIP), 'SIP Transport Protocol' (UDP), 'Local SIP Port' (5060), and 'SIP Proxy and Registrar' settings.

Signaling Protocol	
Signaling Protocol:	SIP
SIP Transport Protocol:	UDP
Local SIP Port:	5060
Gateway Name - User Domain:	Proxy.domain.name.com
<input checked="" type="checkbox"/> Enable PRACK	
<input checked="" type="checkbox"/> Includeptime in SDP	
<input checked="" type="checkbox"/> Enable rport	
<input type="checkbox"/> Connect media on 180	
<input type="checkbox"/> Enable Keep Alive using OPTIONS	
SIP Proxy and Registrar	
<input checked="" type="checkbox"/> Use SIP Proxy	
Proxy IP Address or Host Name:	212.25.125.1
Proxy Port:	5060
Maximum Number of Authentication Retries:	4
<input checked="" type="checkbox"/> Use SIP Proxy IP and Port for Registration	
Register Expires:	3600 Seconds
Sip Security:	Allow All SIP traffic

**Q4: How do I troubleshoot voice echo?**

- A:**
- If you hear the echo, then the echo is caused from one of the following sources:
    - ✓ **Your own handset:** try replacing the handset with a new one.
    - ✓ **The remote party:** notify this to the far-end for it to resolve this issue.
  - If the echo is heard by the remote party, ensure that echo cancellation is activated in MP-20x:
    - a) Logon to the Web interface.
    - b) Access the 'Voice Over IP' screen (**Voice Over IP** menu), and then select the **Voice and Fax** tab.
    - c) Under the **Echo Cancellation** group, select the 'Enable Echo Cancellation' check box.

## Regional Settings

**Q1:** The ring back tone and busy tone are unfamiliar. How can I adjust my call progress tone?

- A:** To suite your call progress tones to your geographical location, perform the following:
1. Logon to the Web interface.
  2. Access the 'Regional Settings' screen (**Advanced** menu > **Regional Settings**  icon).
  3. Select your location for regional settings.

**Figure 11: Selecting Regional Location in the Web Interface**



For additional information, refer to *LTRT-50606 MP-20x Telephone Adapter User's Manual Ver. 2.6.2*, Section 13.8, "Regional Settings".

## Bandwidth

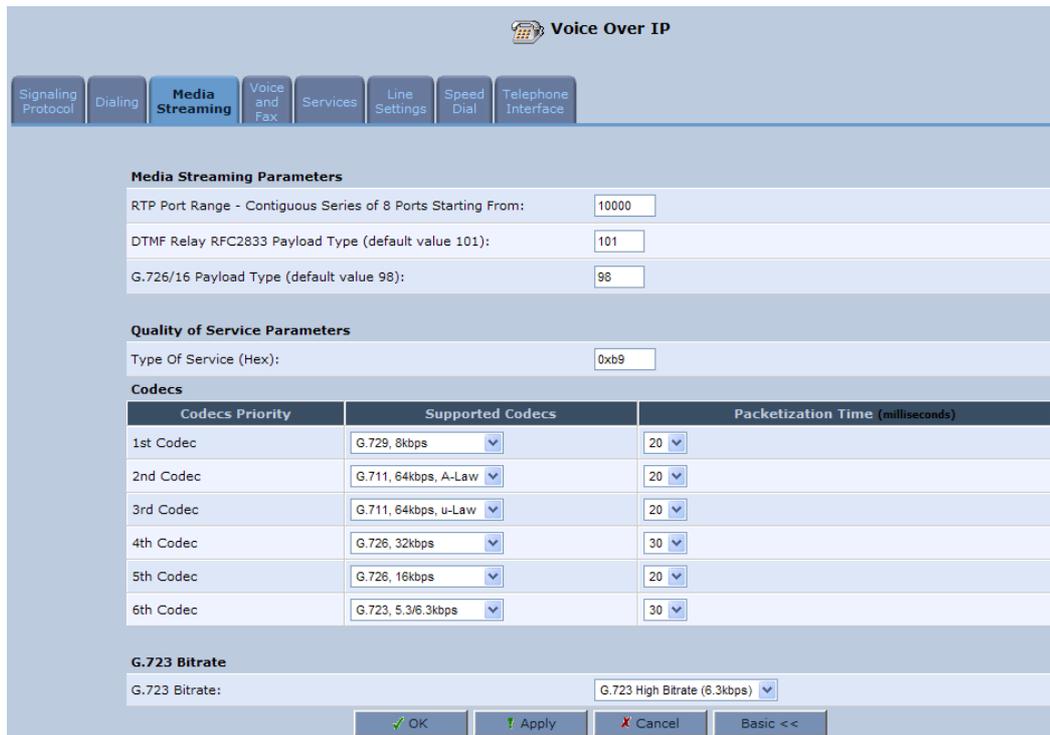
**Q1: How can I reduce bandwidth utilization for calls from MP-20x?**

**A:** It is recommended to use the coder G.729 to reduce bandwidth utilization to the minimum.

➤ **To select a lower bit-rate codec:**

1. Logon to the Web interface.
2. Access the 'Voice Over IP' screen (**Voice Over IP** menu), and then select the **Media Streaming** tab.
3. Under the **Codecs** group, select the required codecs.

**Figure 12: Selecting Codecs in the Web Interface**



**Voice Over IP**

Signaling Protocol
Dialing
**Media Streaming**
Voice and Fax
Services
Line Settings
Speed Dial
Telephone Interface

---

**Media Streaming Parameters**

RTP Port Range - Contiguous Series of 8 Ports Starting From:

DTMF Relay RFC2833 Payload Type (default value 101):

G.726/16 Payload Type (default value 98):

---

**Quality of Service Parameters**

Type Of Service (Hex):

---

**Codecs**

Codecs Priority	Supported Codecs	Packetization Time (milliseconds)
1st Codec	G.729, 8kbps	20
2nd Codec	G.711, 64kbps, A-Law	20
3rd Codec	G.711, 64kbps, u-Law	20
4th Codec	G.726, 32kbps	30
5th Codec	G.726, 16kbps	20
6th Codec	G.723, 5.3/6.3kbps	30

---

**G.723 Bitrate**

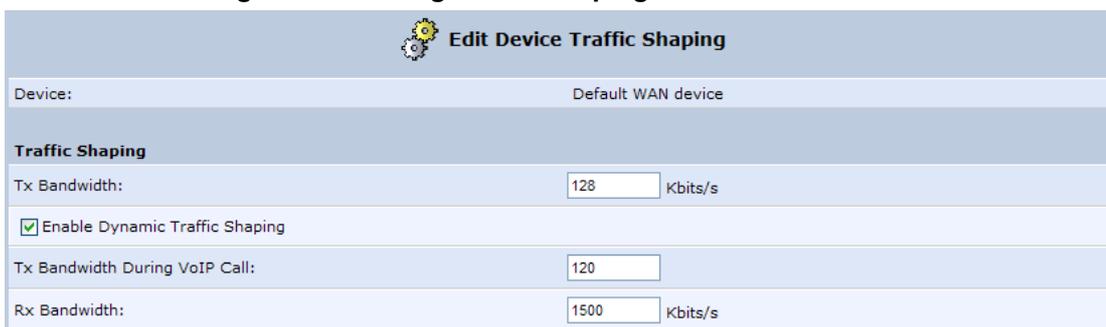
G.723 Bitrate:

**Q2: How can I improve voice quality when using Emule/BitTorrent on a PC behind MP-20x?**

**A:** You must configure MP-20x to prioritize voice packets over data packets.

1. Logon to the Web interface.
2. Open the 'Traffic Shaping' screen (**QoS** menu > **Traffic Shaping** tab).
3. In the Tx Bandwidth field, enter the upload speed.
4. In the Rx Bandwidth field, enter the download speed.
5. Select the 'Enable Dynamic Traffic Shaping' check box, and then in the Tx Bandwidth During VoIP Call field, enter the value for upload rate for data bandwidth during a voice call (original upload data rate - Codec bandwidth and header size).

**Figure 13: Editing Traffic Shaping in the Web Interface**



Edit Device Traffic Shaping	
Device:	Default WAN device
<b>Traffic Shaping</b>	
Tx Bandwidth:	128 Kbits/s
<input checked="" type="checkbox"/> Enable Dynamic Traffic Shaping	
Tx Bandwidth During VoIP Call:	120
Rx Bandwidth:	1500 Kbits/s

For additional information, refer to *LTRT-50606 MP-20x Telephone Adapter User's Manual Ver. 2.6.2*, Section 8.1 "Traffic Shaping".

**Note:** If you are still experiencing choppy or delayed audio, place your MP-20x directly behind the broadband Internet modem.

## Functionalities

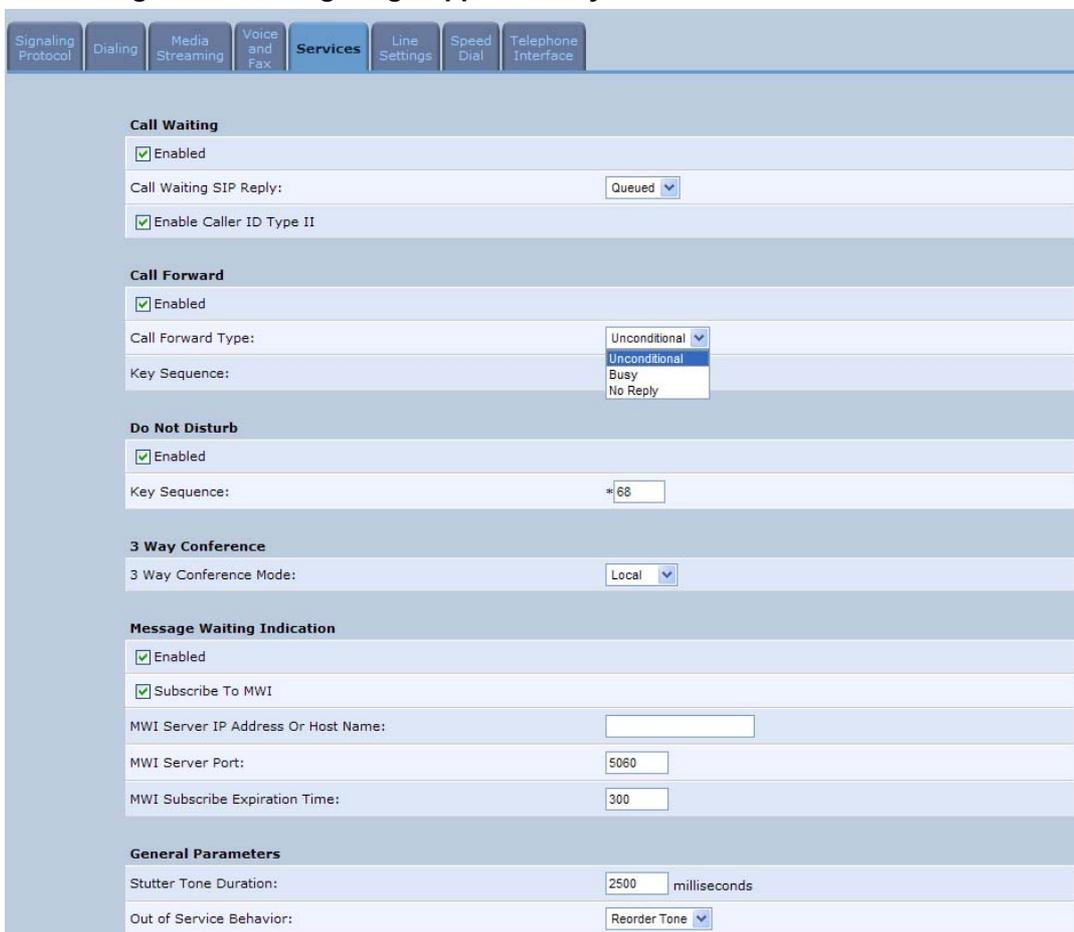
**Q1: Does MP-20x support special services such as call forwarding, conference calls, and do-not-disturb?**

**A:** Yes, MP-20x supports the following supplementary services:

- Call waiting
- Call forwarding
- Do-not-disturb
- Three-way conferencing
- Message waiting indication (MWI)
- Stutter tone

These services are configured in the Web interface's 'Services' screen (**Voice Over IP** menu > **Services** tab).

**Figure 14: Configuring Supplementary Services in the Web Interface**



The screenshot displays the 'Services' configuration page in a web interface. The page is organized into several sections, each with a title and a set of configuration options:

- Call Waiting:** Includes a checked 'Enabled' checkbox, a 'Call Waiting SIP Reply' dropdown menu set to 'Queued', and a checked 'Enable Caller ID Type II' checkbox.
- Call Forward:** Includes a checked 'Enabled' checkbox, a 'Call Forward Type' dropdown menu with options 'Unconditional', 'Unconditional', 'Busy', and 'No Reply', and a 'Key Sequence' field.
- Do Not Disturb:** Includes a checked 'Enabled' checkbox and a 'Key Sequence' field containing '\*68'.
- 3 Way Conference:** Includes a '3 Way Conference Mode' dropdown menu set to 'Local'.
- Message Waiting Indication:** Includes a checked 'Enabled' checkbox, a checked 'Subscribe To MWI' checkbox, and fields for 'MWI Server IP Address Or Host Name', 'MWI Server Port' (set to 5060), and 'MWI Subscribe Expiration Time' (set to 300).
- General Parameters:** Includes a 'Stutter Tone Duration' field set to 2500 milliseconds and an 'Out of Service Behavior' dropdown menu set to 'Reorder Tone'.

For additional information, refer to *LTRT-50606 MP-20x Telephone Adapter User's Manual Ver. 2.6.2*, Section 5.5, "Configuring Services Parameters".

**Q2: What types of fax transport methods are supported by MP-20x?**

**A:** MP-20x supports the following fax transport types:

- Transparent
- T.38
- Voice-Band Data
- Bypass

The fax transport method is configured in the Web interface’s ‘Voice and Fax’ screen (**Voice Over IP** menu > **Voice and Fax** tab).

**Figure 15: Configuring Transparent Fax Transport Method**

Fax and Modem Settings	
Fax Transport Mode:	Transparent
Modem Transport Mode:	Transparent
Fax/Modem Bypass Codec:	G.711, 64kbps, A-Law
<input checked="" type="checkbox"/> Enable CNG Detection	

**Figure 16: Configuring T.38 Fax Transport Method**

Fax and Modem Settings	
Fax Transport Mode:	T.38 Relay
Max Rate:	14.4 Kbps
Max Buffer:	1024
Max Datagram:	320
Image Data Redundancy Level:	0
T30 Control Data Redundancy Level:	0
Fax Relay Jitter Buffer Delay:	0
<input type="checkbox"/> Error Correction Mode	
Modem Transport Mode:	Transparent
Fax/Modem Bypass Codec:	G.711, 64kbps, A-Law
<input checked="" type="checkbox"/> Enable CNG Detection	

**Figure 17: Configuring Voice Band Data Fax Transport Method**

Fax and Modem Settings	
Fax Transport Mode:	Voice Band Data
Modem Transport Mode:	Transparent
Fax/Modem Bypass Codec:	G.711, 64kbps, A-Law
<input checked="" type="checkbox"/> Enable CNG Detection	

**Figure 18: Configuring Bypass Fax Transport Method**

Fax and Modem Settings	
Fax Transport Mode:	Bypass
Fax Bypass Payload Type:	102
Modem Transport Mode:	Transparent
Fax/Modem Bypass Codec:	G.711, 64kbps, A-Law
<input checked="" type="checkbox"/> Enable CNG Detection	

For additional information, refer to *LTRT-50606 MP-20x Telephone Adapter User’s Manual Ver. 2.6.2*, Section 5.4, “Configuring Voice & Fax Parameters”.

**Q3: How do I make a three-way conference call (using Flash Hook Only)?**

**A:** ➤ **To make a 3-way conference call:**

1. Make a call to A, and then press the flash hook button.
2. Make a call to B, and then press the flash hook button again.

Flash Hook Only is configured in the Web interface's 'Dialing' screen (**Voice Over IP** menu > **Dialing** tab). Under the **Key Sequence** group, configure the flash hook sequence method.

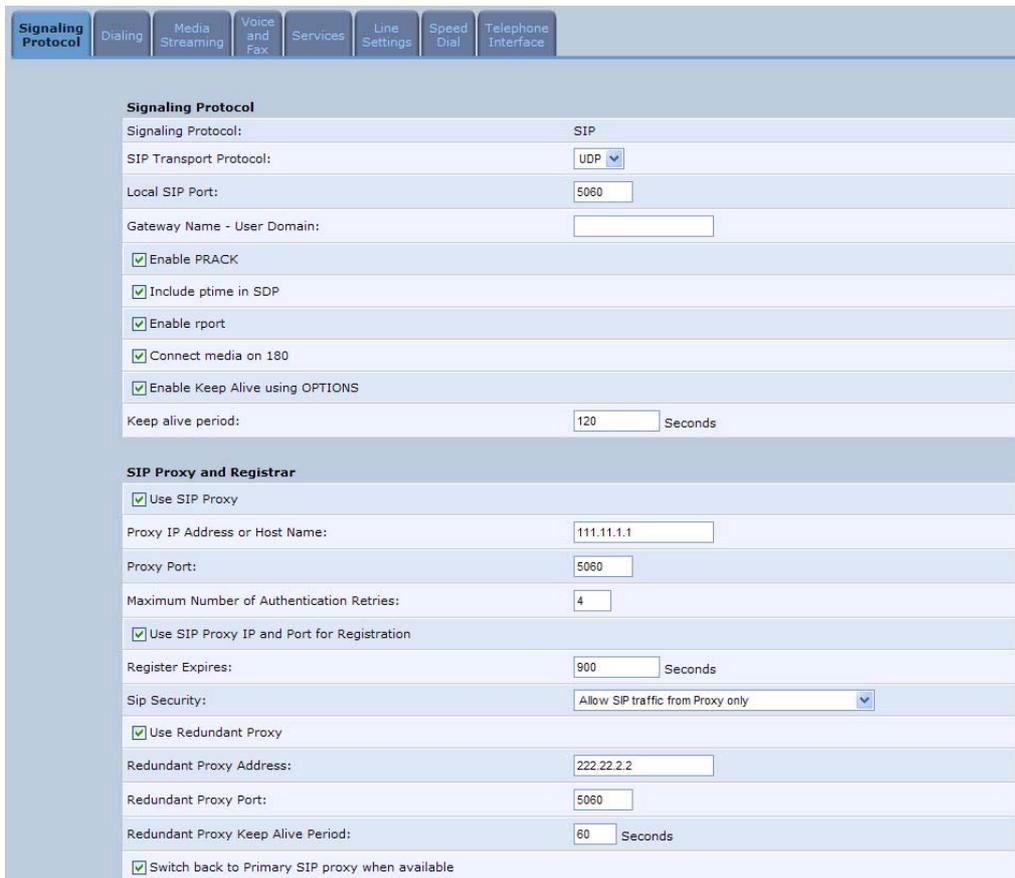
**Q4: Does MP-20x support SIP proxy redundancy?**

**A:** Yes.

➤ **To configure SIP proxy redundancy:**

1. Logon to the Web interface.
2. Open the 'Signaling Protocol' screen (**Voice Over IP** menu > **Signaling Protocol** tab).
3. Under the **SIP Proxy and Registrar** group, select the 'Use Redundant Proxy' check box, and then configure the redundant proxy parameters.

**Figure 19: Configuring SIP Proxy Redundancy in the Web Interface**



Signaling Protocol	
Signaling Protocol:	SIP
SIP Transport Protocol:	UDP
Local SIP Port:	5060
Gateway Name - User Domain:	
<input checked="" type="checkbox"/> Enable PRACK	
<input checked="" type="checkbox"/> Include ptime in SDP	
<input checked="" type="checkbox"/> Enable rport	
<input checked="" type="checkbox"/> Connect media on 180	
<input checked="" type="checkbox"/> Enable Keep Alive using OPTIONS	
Keep alive period:	120 Seconds
SIP Proxy and Registrar	
<input checked="" type="checkbox"/> Use SIP Proxy	
Proxy IP Address or Host Name:	111.11.1.1
Proxy Port:	5060
Maximum Number of Authentication Retries:	4
<input checked="" type="checkbox"/> Use SIP Proxy IP and Port for Registration	
Register Expires:	900 Seconds
Sip Security:	Allow SIP traffic from Proxy only
<input checked="" type="checkbox"/> Use Redundant Proxy	
Redundant Proxy Address:	222.22.2.2
Redundant Proxy Port:	5060
Redundant Proxy Keep Alive Period:	60 Seconds
<input checked="" type="checkbox"/> Switch back to Primary SIP proxy when available	

For additional information, refer to *LTRT-58301 MP-20x Redundant Proxy Application Note Ver. 2.6.2*.

**Q5: Can MP-20x operate behind a NAT?**

**A:** Yes. It is recommended to connect the MP-20x directly to your Cable or ADSL modem. However, if this is not possible, then:

- If the MP-20x is located behind a router that uses Application Layer Gateway (ALG), then there is no need to make any additional modifications.
- If ALG is not supported by the router, then it can operate behind the NAT using an external STUN server.

➤ **To configure NAT:**

4. Logon to the Web interface.
5. Open the 'Signaling Protocol' screen (**Voice Over IP** menu > **Signaling Protocol** tab).
6. Under the **NAT Traversal** group, select the 'Enable STUN' check box.

**Figure 20: Configuring NAT in the Web Interface**



The screenshot shows the 'NAT Traversal' configuration page. It features a 'NAT Traversal' header, a checked 'Enable STUN' checkbox, and three input fields: 'STUN Server Address' (0.0.0.0), 'STUN Server Port' (3478), and 'Subnet Mask' (0.0.0.0).

**Q6: What is the maximum number of phones that can be connected in parallel to a single MP-20x FXS port?**

**A:** Up to five phones can be connected to a single MP-20x FXS port.

**Q7: Can I set up my service so that my caller ID on outbound calls is blocked by default for all calls, without having to dial additional digits?**

**A:** Yes.

➤ **To block caller ID:**

1. Logon to the Web interface.
2. Open the 'Line Settings' screen (**Voice Over IP** menu > **Line Settings** tab).
3. Click the **Action** icon corresponding to the line for which you want to block caller ID.
4. Select the 'Block Caller ID' check box.

**Figure 21: Blocking Caller ID in the Web Interface**



The screenshot shows the 'Line Settings' configuration page. It includes a 'Line Settings' header with a phone icon, and several fields: 'Line Number' (1), 'User ID' (0000000001), a checked 'Block Caller ID' checkbox, and 'Display Name' (Line 1).

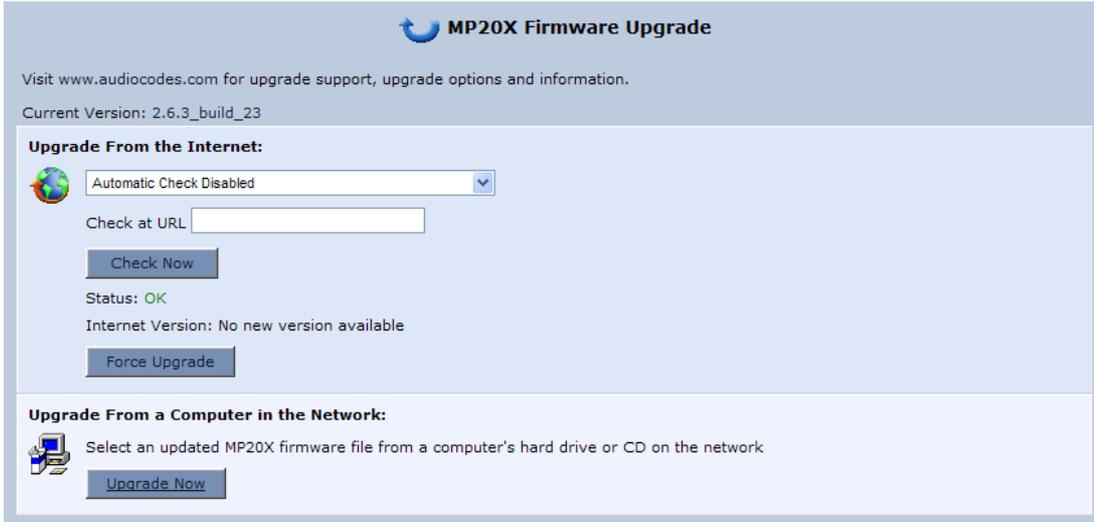
## Software Upgrade

**Q1: How do I upgrade the MP-20x software?**

**A:** ➤ **To upgrade the MP-20x software:**

1. Logon to the Web interface.
2. Open the 'MP20X Firmware Upgrade' screen (**Advanced** menu > **MP20X Firmware Upgrade**  icon).

**Figure 22: Upgrading MP-20x using the Web Interface**



 **MP20X Firmware Upgrade**

Visit [www.audiocodes.com](http://www.audiocodes.com) for upgrade support, upgrade options and information.

Current Version: 2.6.3\_build\_23

**Upgrade From the Internet:**

 Automatic Check Disabled

Check at URL

Status: OK

Internet Version: No new version available

**Upgrade From a Computer in the Network:**

 Select an updated MP20X firmware file from a computer's hard drive or CD on the network

3. Click the **Upgrade Now** button, and then browse to the folder in which the *rmt* file is located.

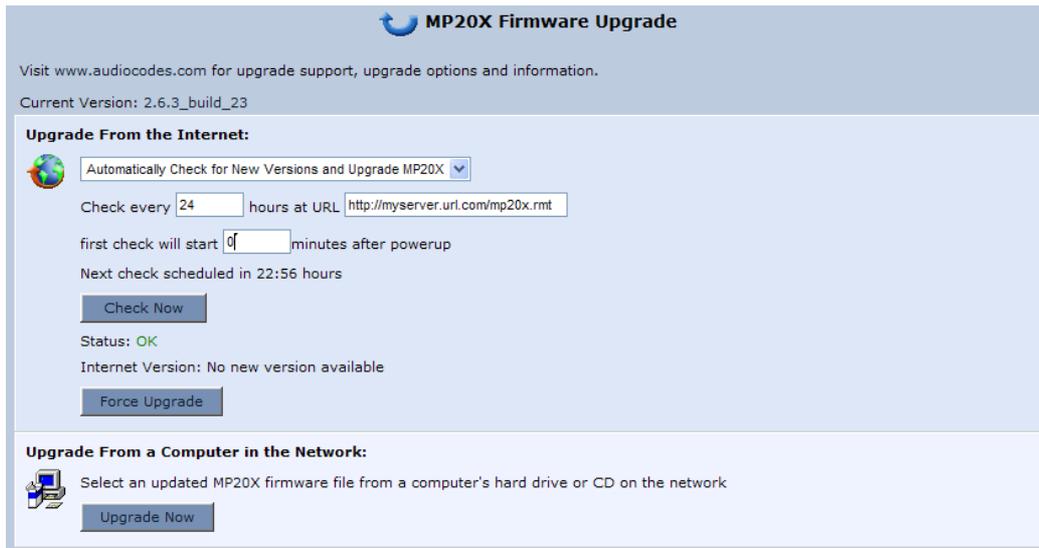
After loading the *rmt* file, MP-20x automatically reboots to factory defaults.

**Q2: How can I automatically upgrade one or more MP-20x devices?**

**A:** To automatically upgrade MP-20x, you can locate the firmware (*rmt* file) on an HTTP/HTTPS or FTP/TFTP server, and then define the URL of this server and the name of the *rmt* file, as follows:

1. Logon to the Web interface.
2. Open the 'MP20X Firmware Upgrade' screen (**Advanced** menu > **MP20X Firmware Upgrade**  icon).

**Figure 23: Automatically Upgrading MP-20x using the Web Interface**



3. From the Upgrade From the Internet drop-down list, select “Automatic Check for New Version and Upgrade MP20X”.

MP-20x checks the URL every 24 hours (user-defined) and upgrades MP-20x only if there is a later version. The firmware file is checked according to the code in the *rmt* file (and not according to the *rmt*'s file name).

For additional information, refer to *LTRT-55201 MP-20x Remote Management Application Note*, Section 2.2.1 “Firmware Upgrade”.

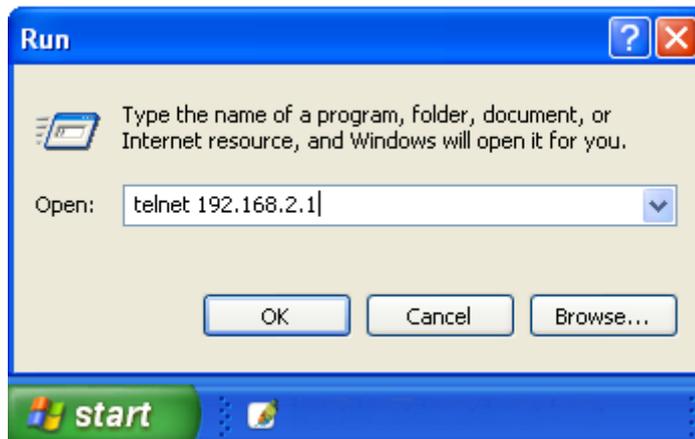
## Telnet and Advanced Configuration

**Q1: Can I manage MP-20x from the Telnet?**

**A:** Yes.

- The LAN interface is open by default. From the PC's command line, run the following command: `telnet <MP-20x LAN's IP address>`

**Figure 24: Connecting to LAN Interface using Telnet**



- The WAN interface is closed by default. To allow access from the WAN, perform the following:
  - a) Logon to the Web interface.
  - b) Access the 'Remote Administration' screen (**Advanced** menu > **Remote Administration**  icon).
  - a) Under the **Allow Incoming Access to the Telnet Server** group, select the 'Using Primary Telnet Port (23)' check box.

**Figure 25: Enabling Access to MP-20x from the WAN**



**Q2: How can I hide the configuration file in the Web?**

**A:** Run the following command:

```
rg_conf_set /rmt_config/hide_config_file_page 1
```

**Q3: Why doesn't MP-20x upgrade from a file in the HTTP server after it is retrieved?**

**A:** Open the Configuration file (located on the HTTP server), and then check the version value of the command line `/rmt_config/version`. The file on the server must have a higher version value than the current version running on MP-20x.

➤ **To view the current version in MP-20x:**

1. Telnet the MP-20x.
2. At the CLI prompt, run the following command:  
`rg_conf_print /rmt_config/version`

**Q4: When is the configuration file retrieved from the HTTP server?**

**A:** The configuration file is automatically retrieved four times after MP-20x reboots (every two minutes), and thereafter, according to the timer of the check interval value. This interval value can be viewed by performing a Telnet to MP-20x, and then running the following command:

```
rg_conf_print / rmt_config/check_interval
```

For example, if `check_interval= 43200sec`, this means  $60\text{sec} \times 60\text{min} \times 12\text{hours}$  = every 12 hours

**Q5: How do I open a TR-069 license?**

**A:** ➤ **To open a TR-069 license:**

1. Telnet the MP-20x.
2. At the CLI prompt, run the following command:  
`rg_conf_set /cwmp/tr069_licensed 1`

# AudioCodes Customer Support

## MP-20x Telephone Adapter

### *Frequently Asked Questions*



[www.audiocodes.com](http://www.audiocodes.com)