Frequently Asked Questions (FAQ)

This document provides information on the most frequently asked questions (FAQ) and the answers to those questions about New Rock products.

General

Q. Do New Rock devices provides different level of authority to log in the device?

A. Yes, all New Rock devices provides two different authority levels:

- An administrator is allowed to make changes to any configuration, such as login passwords. After login, "Welcome Admin" is displayed on the upper left corner.
- An operator is allowed to navigate configuration pages and make limited changes to configurations. After login, "Operator" is displayed on the upper left side of the interface.

Q. Do New Rock devices allows multiple users to log in?

A. Yes, all New Rock devices allows multiple users to log in, in which case the first user can modify, while others can only browse. After login, "Welcome User" is displayed on the upper left side of the interface. However, in the "Welcome user" mode, the operation only can browse certain pages.

Q. How to obtain the IP address of my device?

A. There are several ways to obtain the IP address of New Rock products:

- For FXO only devices:
  - Use Finder software
  - Use Wireshark tool

- For FXO+FXS devices or FXS only devices:
  - Connect the analog phone to any FXS port and dial ##
  - Use Finder software
  - Use Wireshark tool

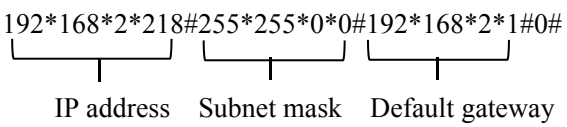
Link to download Finder software installation package:
http://en.newrocktech.com/show/594.html
How to obtain device IP address via Finder:

- Double click and allow to run this application on your PC.
- Click Refresh, and the MAC address and IP address of target models will be listed.

Note: Please connect your PC to the device via Ethernet cable if no devices are found.

Q. How to assign a static IP address for my device?

A. To assign a static IP address for your device, dial *90 and configure your network parameters as the following example on an analog phone:



Note: Configuration will only take effect after a reboot.

Q. How to obtain a DHCP IP address for my device?

A. To obtain a DHCP IP address for your device, dial *90###1# and reboot the device after you hear the voice prompt "The feature is now activated"

Note: Configuration will only take effect after a reboot.

Q. How to log in to the web GUI?

A. To log in to the web GUI, please perform the following procedures:

- Use a CAT5 cable to connect the PC port or ETH1/ETH2 port of the device to your PC.
- Obtain the IP address of the device
- Make sure that the PC and the device are on the same network segment
- Enter the device IP address in the browser address bar (e.g. default: 192.168.2.218)
- Select your role and enter the password into the device. (pls. refer to your administrator or quick installation guide for your default password)
- Enter the CAPTCHA as shown on the log in page.
- Be sure to change the password after the first log in. If the web browser for risk action, select "**allow**".

Q. How can I restore the device to factory default settings?

A. There are several ways to restore the device to factory default settings:

- Press the RST button for more than 3 seconds. (all new devices already has the RST button except HX4G).
- Connect an analog phone to the FXS port of the device and dial *911234#, then hung up and reboot the device.
- Log in as admin to the device and go to **Tools > Restore factory settings.**

Q. What if I cannot Log On to the device because I forgot the preset whitelist IP address?

A. The OM provides the embedded white-listed address of 192.168.2.100 upon factory delivery. When the Whitelist function is enabled, if you forget the whitelisted IP address previously set, the following steps can be performed for recovery.

- Connect a PC directly to the OM through a network cable.
- Press *90 to set the IP address of the OM to one that is located in the same network segment as the embedded white-listed address, such as 192.168.2.101. To do so, continuously dial *90192*168*2*101#255*255*255*0#192*168*2*1#0# after off-hook, and then hook on after hearing the successful service registration announcement.
- Restart the OM.
- Set the IP address of the PC to 192.168.2.100.
- Enter the new IP address of the OM on the Internet Explorer or the Telnet client of the PC to access the OM.

Q. What is the difference between the two outbound call modes: DID share and DID only?

A. With DID enabled, an incoming call can be directly routed to a specified extension or hunt group without passing through the auto attendant. You can select the outbound call mode as follows:

- **Share:** The trunk can be used by all extensions or hunt groups.

- **DID only**: The trunk can be used only by the specified extension or hunt group.

Q. How to make sure that I am using the correct Caller ID detection mode: Before ringing or After ringing?

A. Firstly, you need to connect the PSTN subscriber line directly to an analog phone. Then make an incoming call (using another line or mobile phone) from the PSTN to the analog phone and check if the analog phone displays Caller ID. If the Caller ID is not being shown on the phone, please contact your telco or PSTN service provider and ask them how to enable this feature. If the analog phone shows the correct Caller ID on every call, please take note if the Caller ID shows before the first ring tone or after the first ring tone. More explanation between the two modes:

- Before ringing indicates that the PSTN line will send Caller ID before the first ring signal
- After ringing indicates that the PSTN line will send Caller ID between the first ring and second ring

Q. Do New Rock products support IAX or IAX2?

A. Inter-Asterisk Exchange (IAX or IAX2) are proprietary protocols and New Rock devices do not currently support these protocols.

Q. Do New Rock IPPBX support video call?

A. Yes, we already supports video call as long as the IP phone already has this capability. There is no special configurations needed from our device.

Q. Can I press the R key on an analog extension?

A. Pressing the R key after off-hook is equivalent to hook-flash. However, because R keys on different phones may follow different design specifications, pressing the R key on an extension is not always reliable. It is recommended that you press ** for functions such as three-way calling, call transfer, and call parking.

Q. What is the maximum concurrent calls capacity for NewRock's device?

A. Each model can handle different maximum concurrent call capacity

Device model	OM20G	OM50G	OM80E	OM200G	OM500	OM8000
max concurrent calls	24	30	90	120	200	800

Device model	HX4G	MX8G	MX60E	MX120G	MX100G-S
max concurrent calls	4	8	48	96	120

Q. How VoIP gateway works?

A. Once you initiate a VoIP call, the VoIP Gateway converts voice and fax calls, in real time, between the public switched telephone network (PSTN) and an IP network. Traffic coming in from the public PSTN is fed through a VoIP gateway and converted to digital packets so traffic can be transported over a local area network (LAN) or other IP-based network. Conversely, digital IP traffic is fed back through a VoIP gateway for conversion back to analog so it can be transported out over the PSTN. In other words, a VoIP gateway acts as a bridge between an IP network and the PSTN.

Q. How to configure SIP account on VoIP gateway step-by-step?

A. The following are the recommended step-by-step procedure:

Step 1. Get the VoIP gateway IP address.

- Connect the analog phone to the gateway FXS port.
- Pick up the handset and dial *"##"* to hear the device's current IP address and port number

Note: For full FXO gateway, you can download Finder App to get the device's IP address. By default, the HX4G/ MX8A gateway can use DHCP to get IP address, while MX60E/MX120G is the static IP address: 192.168.2.240

Step 2. Make sure that the PC and the device are on the same network segment

Step 3. Log in to the gateway configuration web GUI.

- Open the browser, enter the gateway IP address that was queried, log in to the gateway configuration interface.
- HX4G/MX8G, for example, select Administrator in the user bar drop-down box then enter the device password in the password bar (the default password is a random 8 digits number found on the label of the device while the default password for MX60E is mx60 and for MX120G is mx120)
- Enter the CAPTCHA as shown on the log in page.

Step 4. Register the gateway to the SIP server.

- Select **Basic > SIP** from the configuration screen, enter SIP server address, select per line to register and save.
- Select **Line > Configuration**, select the line you need, click on the registration, enter the registered user name and password then save.

Step 5. Verify that the gateway registration was successful.

- Select **Call Status > Call status** from configuration interface, check the registration status of the line name and number, and show the registration is successful, the configuration is completed.

Note: If the registration status shown has failed, recheck the following:

① The registered server address and port is correct.
② The registration of line you chose is open.
③ The registered user name and password is correct.
④ Check if the gateway can ping the SIP server.

Q. Can I use any brand of IP phone to connect with OM device?

A. Yes, you can use any brand of IP phone as long as it supports standard SIP. Also, VoIP phones can be either soft phones that use software on a computer or smart phone or hard phones that use an IP phone, or an analog phone with an analog phone adapter (ATA), connected with Ethernet cables or WiFi directly to a network.

Q. Does NewRock sell wireless IP phone?

A. Yes, we offer and sell NRP2000/W IP Phone which is a multi-access network, easy deployment, excellent sound quality IP Phone. You can refer to this link for more details: http://en.newrocktech.com/show/602.html

Q. How to configure IP phone registration to the OM step-by-step?

A. The following are the recommended step-by-step procedure:

Step 1. Log in as **admin** to the OM device's configuration web GUI.

Step 2. Go **Extension > IP > add IP phone**

Notes: OM20G/50G has a factory built-in extension number so no need to add extension number, just directly chose the number to register. In such instances that your device has no built in extension number, you can choose **batch adding** which more convenient.

Step 3. Go **Extension > IP > add IP phone**

Step 4. Verify the local port of the OM device on **Trunk > IP trunk >Registrar OPTIONS.**

Step 5. Click on the IP phone LED screen, go to **Menu > Settings > Advanced Settings > Enter password 123 > Account > SIP 1 > SIP Basic ,** then fill in the important registration information:

1. Server address: the OM device IP address.
2. Server port.
3. SIP user name.
4. Verify user name.
5. Verify password.
6. Display name: same as SIP user name.
7. Outbound proxy: off.
8. Open registration: open.

Step 6. Click "**Save**" after configuring all the above contents. Click "Return" after seeing "**Saved**" to return

to the main interface.

Note: Another option is to open the web management interface of the IP phone, click **VOIP** > **SIP**, select the desired SIP line, and then enter the registration information in **Basic setting**.

Step 7. Verify the phone number displayed in the upper left corner if it is registered successfully

Note: If it is a wireless phone NRP2000/W, just connect to the wireless network and follow the same steps to configure it.

Q. Can VoIP phones be used at home?

A. Yes, there are two methods to follow for remote access:

Method 1: Remote Access with Peer-to-Peer communication across NAT: In conjunction with New Rock Cloud allows other entities to access an OM located behind a NAT without the need of port mapping on the access router.

- On OM, click **Basic** > **Remote access**, and make sure to enable the **P2P traversal** feature.

Note: **Registrar for External Terminal** is the uniquely generated domain name for sip extension registration; use this domain name to register your sip phone as OM extension.

- Open the Web management interface of the IP phone, click **VOIP** > **SIP**, select the desired SIP line, and then enter the registration information in **Basic setting**.

Method 2: Remote Access with NAT Traversal using DDNS or External IP address: it is necessary to configure remote-address information and configure port mapping on the Internet ingress router

- On OM, click **Basic** > **Remote access**, and set remote address
- Configure port mapping on the Internet ingress router.

  Note: Keep the mapping target port number the same as the SIP signaling port and the RTP port range of the OM.
- On the external IP extension, set the registrar address to the IP address or domain name configured on the **Basic** > **Remote Access** page.

Q. Are VoIP calls secured and encrypted?

A. Yes, it is recommended to use the following security procedures in order to protect your device while using a public network:

1. Pay attention to alerts and alarm levels on device web GUI. The administrator can see the total number of security incidents after logging in the device, click the **Basic > Alarms** to view the details. After confirmation, the entry will not be displayed.
2. Change the default Web password for the device.
   Note: The operator password cannot be the same as the administrator password.
3. Change the default SIP port for the device.
4. Check the anti-brute force mechanism including CAPTCHA for logging into Web GUI limiting the number of login attempts, and access whitelist of trusted IP addresses.
   Main manifestation: The same attack source IP address are made to the Web GUI or SSH in a short time. The IP addresses whose login attempts exceeding the specified limit will be added to the locked IP address list with the locked time which can be set by minute or hour or day.
5. Disable the Telnet and SSH services on the device.
   Note: You need to set root and operator password to access SSH.
6. Block the inbound Ping request on the device.
   Note: Ping is allowed by factory default but it is recommended to change, as the hacker could trigger an attack if he detects an address.
7. Added restrictions on special number of roads.
8. Let the operator open only those international call areas that need to be opened.
9. Check the voice security configuration
   Note: When the gateway cannot call outside, first check whether the connection is normal （as long as there is a FXO connection, the defense mechanism is effective) , then examine the configuration.
10. Set the access whitelist.
11. Change the http and https port.
12. Configure static defense.
    Notes: ①Multiple rules can be added and working from the top down.
    ②Individual IP address of IP address segments can be configured but domain names are not supported.
    ③Configuration protocol types are optional including TCP,UDP and any.
    ④The local port can be configured in the range 0~65535.
13. Configure dynamic defense.
14. Enable TLS and SRTP encryption.
    Notes: There must be a supported device or soft switch in the environment.
    It is recommended to force TLS encrypted signaling and SRTP encrypted voice to be selected and to be

checked/enabled on the line because it is disabled by default.

15. Establish a VPN connection.

Q. Does the IP phone have a default IP address?

A. No, the IP phone does not have a default IP address. You must wait for the phone to get the IP address or set the IP address for the phone.

Network and improper operation

Q. Why I cannot connect the device to my router, thus I cannot make both outbound and inbound call?

A. Verify that you have set up the network correctly. Make sure that your device is not disconnected to the router. If still having problem connecting to the router, you can perform the following procedures:

- Check for any lose or disconnected Ethernet cables
- Ensure that the MAC address filtering on router is "off"
- Check if RJ45 connectors and network ports are not broken
- Device should be configured using FIXED IP address
- If the client rebooted the router or the service provider address tenancy period has been changed then you need to reboot the device and configure the STUN server on the device and find the public IP address changed at once.

Q. Why only one of the parties receives audio (one-way communication) when a phone call is established?

A. On analog gateways, go to **Advanced > System**, NAT and check the configuration for **SDP address**

- If **SDP address is set to Local IP address**, then you need make sure that the router ALG function is **disabled** (some router enabled this function by default. e.g. H3C and TP-link).
- If still having problem after disabling the router's ALG function, then you can set **SDP address** to **NAT IP Address**

Q. Why I cannot make an outgoing call from IP trunk even though there is a dial tone?

A. Go to the **Log** > **Running log** page to check the IP trunk registration status. See the table below for details.

Table 1-1 Solutions to IP trunk registration failures

Displayed Content	Registration Status	Solution
SIP Registration Info >>>>> Contact:<sip:61208000@192.168.250.5:5060> response: **200**	Registration is successful.	Check the network configuration and wiring, and analyze call SIP signaling.
SIP Registration Info >>>>> Contact: <sip:61208000@192.168.250.5:5060>	There is no response to the registration	Contact the VoIP service provider to confirm whether the address of the IP trunk registration server is correct, and test whether the network

Displayed Content	Registration Status	Solution
No Response	request.	communications from the device to the registration platform are normal.
SIP Registration Info >>>>> Contact: <sip:61202000@192.168.250.5:5060> response:**404**	IP trunk registration number is incorrect.	Contact the VoIP service provider to confirm whether the IP trunk registration number is correct.
SIP Registration Info >>>>> Contact: <sip:61208000@192.168.250.5:5060> response: **403**	Registration password is incorrect.	Contact the VoIP service provider to confirm whether the IP trunk registration password is correct.

Q. Why I cannot login to the web administrators' interface?

A. Generally, this could be related on the network parameters of the device. You can perform the following procedures:

- Connect your phone to the FXS port on the device, pick up the phone, and press ## to listen and check whether the network parameters of the device are correct.
- Check the LAN where the device is located.
- Check the connection between the LAN and the device.
- Also you can try log in using different web browsers.

Q. What should I do if I can hear a busy tone when making an outgoing call even though my MX gateway has registered successfully to IPPBX?

A. Make sure that the network cable is connected on WAN port and not on PC port. If still has issue, you may check the following:

- Check the network configuration and wiring,
- Make a test call, then download log level 4 and analyze the SIP call process.
- You can read the error information from error.log

Q. Why do SIP registration always failed?

A. Make sure that you are using the correct username and password for SIP registration. If still registration is unsuccessful and verified the following are true:

- Log in to the device web GUI and go to **Log > System status.** Displayed "**No Response**"
- Busy tone or engaged tone is heard when making outgoing call
- A voice prompt "power off" is heard from other side (incoming call)

Then go to **Basic > SIP** and enable the **"Increments of port number"** for local signaling port so that the local SIP port will be automatically selected. You can select any values from 1-10 to register with the new signaling port value until it succeeds.

Q. Why I cannot make a new outgoing call when I dial a number immediately right after I

hang-up?

A. This issue usually happens after on-hook and then immediately off-hook to make a call (on/off-hook quickly, most customer are accustomed to use on-hook as a flash-hook method) . In order to avoid such issue, you need to check and try the following suggestions

- Recommend to change user's behavior by not using on-hook as a flash-hook method.
- Disable the **call hold** function if customer do not use the **call transfer** function.
- Go to **Line > Advanced,** adjust the **Hook flash timer min** to 150~780ms (default =75) and **Hook debouncing** to 100~1000ms **(**default=50)

Q. All of the analog phones are without dial tone after off-hook, thus I can neither make outgoing nor receive an incoming call?

A. It is usually on RJ45 &RJ11 cable connectors and power supply. You need to check and verify the following:

- Check if the RJ45 & RJ11 cable connectors is working and being plugged properly.
- If all LED indicators of the device status are ON, the issue is on power supply, thus you can replace another power adapter with the correct voltage and current ratings.
- If all LED indicators of the device status are OFF, the issue is on power supply and power board, thus you can replace the power adapter and the power board. If still not working, you need to replace the whole device.

Q. Why do the time and date displayed with the calling number on the phone is inconsistent with those in the device?

A. You need to set properly the time server. You may check the following:

- Check whether time information can be obtained from a time server on the Internet.
- If the device cannot access the time server on the Internet, select a PC in the LAN to serve as the time server. If the operating system is Windows Vista, Windows 7, or Windows server 2008, manually start the Windows time service.
- Check whether the firewall of the Windows operating system is enabled on the PC. If the firewall is enabled, perform the following steps to enable the port through which the device accesses the time server on the firewall.
  - Open the firewall window, and choose **Exceptions** > **Add Port**.
  - Add port 1 Name the port ntp-tcp, specify the port number as 123, and select the TCP mode.
  - Add port 2 Name the port ntp-udp, specify the port number as 123, and select the UDP mode.
  - Go to the **Control Panel** > **Administrative Tools** > **Services page**, and confirm that the Windows time service has been enabled.
- Call extension B from any extension A. Extension B shows the same time information as that on the device.

Q. Why do the IP phone could not get the IP address?

A. You may check the following:

- Check whether the telephone network cable is plugged in.
- Check that the DHCP server in the previous router is on or that the DHCP address pool has been allocated.

Q. How to set the static IP address for the IP phone if it cannot get IP address automatically from the previous router?

A. You may check the following:

- Press "**Menu > Settings > Advanced**", enter password 123, select "**Network>WAN Settings>Network Type**", and then select "**static IP**" through the left and right navigation keys of the phone to save the settings.
- Go back to the previous menu and select Static Settings.
- Enter your IP address, subnet mask, gateway, domain name, and save settings.
- Return to the standby interface and gently press "**Menu > Status**" to view the status information. "**Mode: Static IP**" is displayed on the screen. The second line shows the set network address, indicating that the static mode of the phone is effective.

Q. Where does the device get its IP address if it is connected to the PC network port of the IP phone?

A. The IP phone factory default is set to bridge mode. The device connected to the PC port gets the IP address from the previous router, which is in the same network segment with the IP address of the IP phone.

Q. How to switch the phone from bridge mode to routing mode so that the PC terminal of the phone does not get IP address from the superior router?

A. Enter the Web interface of the phone, click "**Network Setting > LAN**", and remove the bridge mode.

Q. How to find the IP address and Web port of the registered server on the IP phone?

A. When used with OM series IP-PBX, long press the * key of the phone, and the IP address and Web port of OM series IP-PBX will be displayed on the phone screen.

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Q. How to get a data packet capture on the IP phone?

A. Log into the Web interface of the phone, click "**Manage and set > System log > Webpage packet capture**" and then perform the steps to capture the phone packet as follows:

- Click the "**start**" button, and the browser (Google is recommended: Chrome) prompts you to save the pcap file. Please save the file.
- Reproduce the problem you encountered.
- After the problem reappears, click the "**Stop**" button on the web page to grab the data packet.
- The PCAP file saved in step (2) is the captured packet.

Q. How to set the correct time on IP phone if the time displayed on the IP phone screen is not accurate?

A. This could be due to a network outage or a problem with the time server set up in the phone.
Click "**Network Settings > SNTP**" to change the time server address or manually set the phone time.

Configurations

Q. How to configure the device in order to reduce the connection time of telephone calls?

A. You need to set properly the dialing rules to reduce the connection time of telephone calls. Digit map is used to define the dial plan of your device in order to recognize the ending of dialed numbers thus speeds up the call process. The factory default digit map only contain system function rule. If it does not fit your dial plan, you need to set the rules as follows:

- Go to **Routing > Digit map**, then remove all rules in the digit map but the last five, which allows use timeout or # as the ending of dialed numbers
- Go to **Routing > Digit map**, then redefine the digit map to fit your dial plan

Q. What should I do if incoming call on an FXO gateway is always disconnected?

A. Generally, if the PSTN line connected to the FXO port of the device do not support "Polarity reverse signal", and you enabled the feature "Polarity reversed signal detection" on the device, then the call will be automatically disconnected after 1 or 2 minutes of conversation. You need to check and configure the following:

- Verify from telco if the PSTN line provide polarity reverse signal
- Check if the feature "Polarity reversed signal detection" is enabled on the device.
- If PSTN don't provide polarity reverse signal but we enabled, it will cause the call to disconnect
- Check the log if it has following information:[07/06 11:19:51.652184]FXO-8075(76) disconnected
- If it has, set http://x.x.x.x/xml?method=gw.config.set&id215=no
- Check the **Logs > Call message**, if device receive ACK after 200OK message.
- Check the **Log > Call message** to see if server send BYE to device to end the call.

Q. How to resolve the Caller ID detection issues on FXO port?

A. You need to check and configure the following:

- Verify from telco if the PSTN subscriber line supports Caller ID feature (enabled)
- Disconnect the PSTN subscriber line from the FXO port of MX or OM device and connect this line directly to an analog phone. Make an incoming call (using another line or mobile phone) from the PSTN to the analog phone and check if the analog phone displays Caller ID. If the Caller ID is not being shown on the phone, please contact your telco or PSTN service provider and ask them how to enable this feature.
- If the analog phone shows the correct Caller ID on every call, please take note if the Caller ID shows before the first ring tone or after the first ring tone.
- Make sure that the Caller ID detection feature on the device is enabled.
  - If you are using OM devices, go to **Trunk > Analog trunk** and enabled **Caller ID Detection**
  - If you are using MX devices, go to **Trunk > Feature** and enabled **Caller ID Detection**
- Make sure to select the correct Caller ID detection mode on the web configuration of the MX or OM device.
  - If you are using OM devices, go to **Trunk > Analog trunk > Advanced** and select the appropriate **Call ID Detection** mode (Before or after ring)
  - If you are using MX devices, go to **Trunk > Advanced** and select the appropriate **Call ID Detection** mode (Before or after ring)

Q. Why I could not modify the outbound dialing rule on OM?

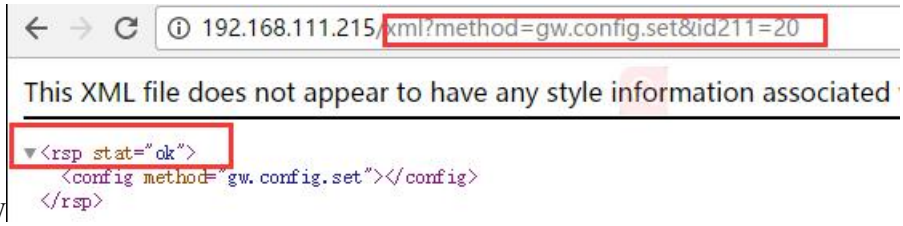
A. The configured prefix is in conflict with the extension number, hunting group number, number to reach the operator, feature access code and other call prefixes.

Q. Why the line status on the web GUI shows disconnected while the FXO port is connected to the PSTN line and when made an incoming call to the line, the caller will hear a ring back tone but extension will not ring and also cannot make an outgoing call?

A. You need to check and configure the following:

- Disconnect the PSTN subscriber line from the FXO port of MX or OM device and connect this line directly to an analog phone. Verify if the line is normal and can make incoming and outgoing call.
- If the PSTN line is normal, then the reason could be that the PSTN line voltage is too low, thus we need adjust the parameter FXO_DISC_VOLT (ID211) and set it to 20/10, default is 28. That parameter is FXO port ring detect threshold.
- Procedure in setting FXO_DISC_VOLT parameter:
  - ➢ Log in to GW with admin account
  - Set the value to 20 for disabling SIP_FG_REQ_USING_TO parameter. Input the xml command on the URL: https://x.x.x.x/xml?method=gw.config.set&id211=20

    Note: x.x.x.x refers to the IP address of your GW



Q. How to configure the device in order to have a successful POS call?

A. In order to have a successful POS call, you need to check and configure the following:

- Make sure that your POS machine is connected to the FXS port of the device.
- Make sure you have the best network quality to avoid packet loss.
- It is recommended to use T.30 mode and PCMU/20 or PCMA/20 codecs.
- Go to **Routing > Routing table** , then add a route " **FXS 143 CODEC PCMU/20/0**" (note:143 is the Bank's access account number and you are using PCMU codec with 20ms of package size with disabled echo cancellation)

Q. Why I don't hear my phone is ringing every time I received an incoming call? Other side can heard a ring back tone and when I answered the phone call is established.

A. The default Voltage and frequency is too low, thus it could not provide enough power to the analog phone to make it ring. You need to adjust and configure the following:

- Go to **Line > Advanced**, then increase the Ring Voltage value.
- Go to **Line > Advanced**, then increase Ring Frequency value. (Optional if increasing the ring voltage resolves the issue)
- Replaced your analog phone. (Optional if increasing the ring voltage resolves the issue)

Q. Why IP terminals (IP phone and softphone) can directly register to the SIP server, but the MX gateway cannot register per line to the SIP server?

A. It is recommended to check the following:

- SSH into MX gateway and run the **ping** command to check the network connection between MX gateway SIP server. If the network fails, debug the network and try again.
- Check whether the Registrar server is the address of SIP server and whether the signaling port is correct.
- Check whether registration account name, user name and registrar password are correct.
- Check whether there are special restrictions on SIP signaling. For example, some SIP server will check whether or not there is a field of specific **User-Agent** in the registration message , thus allowing only those terminal that are allowed to register.

Q. Why there is no pop-up prompt for the PIN code when the IP phone is connected to a plug-and-play device?

A. You need to check and configure the following:

- Make sure that you are using NRP IP phones and the device you are trying to connect could be any of the following OM Series:OM20G/OM50G/OM80E/OM200G/OM500/OM8000
- On the IP phone, Click "**Menu > Status**", check whether the phone has obtained WAN port IP address; If you did not get an IP address, check your network connection.

Q. Why the IP phone screen prompts for the wrong PIN code after inputting the PIN code provided by the device on the phone?

A. You need to check and configure the following:

- Check whether there is another plug and play device in the network environment. If you enter the PIN code for device A and the phone takes the configuration file from device B, then the PIN code is not match and will not work.
- Check if the IP phone was not reset to its factory settings especially when the phone has been upgraded

Q. How to set up a phone to block nuisance calls?

A. Log in to the Web interface of the phone, click "**VOIP > global Settings**" and check "**strictly match UA**" to block harassing calls.

Q. If the IP phone is not registered, how to dial each other directly?

A. This requires knowing the correct IP address of each IP phone. When dialing, enter # to dial the phone IP#, where the ". " in the IP address is replaced by" * ".

Q. How to change the password of advanced setting?

A. The factory default advanced password of the phone is 123, which can be modified by the user on the Web page:

Enter "**Management Settings > user Settings**", enter the password to be changed in the Settings menu password, click "**Settings**", prompt save successfully;

It can also be changed in the phone interface using the "**Set key > Menu password**" in advanced Settings.

Q. How to deal with the loss of login password?

A. You need to do the following:

- Power off and restart the phone, press # when INITIALIZING is displayed on the screen;
- After the phone enters post mode, press ***#168** and wait for 5 seconds before power off and restart the

phone.

Q. How to prevent the IP phone from popping out the PIN box?

A. You need to do either the following:

- **Method 1:**
  - Disconnect the network cable and restart the phone.
  - Press "**Menu > to set > advanced Settings**", enter password 123, select "**Maintain > and automatically deploy > plug and play**", then select "**close**" and save the Settings through the left and right navigation keys of the phone.
- **Method 2:**

➢ In the Web interface of the phone, click "**manage > automatically configure > Plug and Play (PnP) Settings**", and **uncheck** "**open PnP**".

Voice Quality

Q. How to remove the buzzing or humming noise during the call?

A. Usually if the grounding environment is not properly installed, it will bring more noise or AC interference (50/60 Hz noise or line frequency hum) with the device. Thus, it is recommended to check if the grounding wire is connected properly. You can perform the following procedures:

- Avoided sharing of AC power outlet with other devices (as this may generate electrical interference)
- Check if the rack is earth grounded properly
- Isolate the equipment from the rack
- Cut off the ground pin on the power plug

Q. What can I do if I can hear a sound like a DTMF tone during the call without even pressing any key on the keypad?

A. Background noise disturbance will generate the sound that can be pick up by the microphone on the handset. This issue is also common to hands-free device. In order to prevent the false detection of DTMF signal, you need to increase the DTMF actual detection threshold on the VoIP device by system configuration.

- If you are using OM devices:

  Go to **Advanced** > **DTMF**, then input 32 or 48 on **DTMF detection duration increment against talk-off** field (The range must be 16 to 240 in multiples of 16)
- If you are using MX devices:

  Go to **Basic** > **System**, then input 32 or 48 on the **DTMF detection duration increment against talk-off** field (The range must be 16 to 240 in multiples of 16)

Q. What can I do if the voice volume is too low (or high) on an extension during the call?

A. When the voice volume is too low (or high) on **an analog extension side or internal party,** you can increase (or decrease) the Gain to terminal parameter value by line configuration. Test several values higher (or lower) than default value=0 until you get the normal voice volume.

- If you are using MX devices:

  Go to **Line > Advanced,** increase (or decrease) the **Gain to terminal** value
- If you are using OM devices:

Go to **Extension > Analog > Advanced,** increase (or decrease) the **Gain to terminal** value

A. When the voice volume is too low (or high) on **the other side or external party,** you can increase (or decrease) the Gain to IP parameter value by line configuration. Test several values higher (or lower) than default value=0 until you get the normal voice volume.

- If you are using MX devices:
  Go to **Line > Advanced,** increase (or decrease) the **Gain to IP** value
- If you are using OM devices:
  Go to **Extension > Analog > Advanced,** increase (or decrease) the **Gain to IP** value

Q. What can I do to reduce the echo that I heard during the call?

A. When the a**nalog extension side or internal party** has echo**,** you can decrease the Gain to terminal parameter value by line configuration. Test several values lower than default value=0 until you get the normal voice volume.

- If you are using MX devices:
  Go to **Line > Advanced,** decrease the **Gain to terminal** value
- If you are using OM devices:
  Go to **Extension > Analog > Advanced,** decrease the **Gain to terminal** value

A. When the **other side or external party** has echo**,** you can decrease the Gain to IP and Gain to PSTN parameter value by line and trunkconfiguration. Test several values lower than default value=0 until you get the normal voice volume.

- If you are using MX devices:
  Go to **Line > Advanced,** decrease the **Gain to IP** value
  Go to **Trunk > Advanced,** decrease the **Gain to PSTN** value
- If you are using OM devices:
  Go to **Extension > Analog > Advanced,** decrease the **Gain to IP** value
  Go to **Trunk > Analog trunk > Advanced,** decrease the **Gain to PSTN** value

Q. How can I eliminate the crosstalk on an analog extension? (Conversation on another extension is heard during a call)

A. Generally, crosstalk is caused by telephone line short-circuits. Check the connection line of the FXS port and remove the line fault.

Q. How can I remove echo and noise during the call?

A. To mitigate the echo and noise related issues, make sure that Echo cancellation is enabled and FXO impedance is complex. You can perform the following procedures:

- If you are using HX4G/MX8G:
  - Go to **Trunk > Feature,** enable **Echo cancellation**
  - Go to **Trunk > Advanced,** set **Impedance as Complex**
- If still has this issue, you can try to configure the echo and noise related parameters
  - **ECHO_TAIL**, ID =300. Value range is 1~300 (Adjust every 30ms. From our testing results, **30ms** is the best value, so we can start from 30ms then increase the value if we want to

adjustit)

- Input the xml command on the URL: https://x.x.x.x/xml?method=gw.config.set&id300=30
- **ECHO_CANCEL_LEN**, ID =66. Value range is 16~64 (Adjust every 16ms. From our testing results, **64ms** is the best value, so we can start from 64ms then decrease the value if we want to adjust it.)
- Input the xml command on the URL: https://x.x.x.x/xml?method=gw.config.set&id66=64

Firmware Upgrade

Q. Where to download the firmware upgrade package?

A. They can be downloaded from New Rock's website link:

http://en.newrocktech.com/software/

Q. How to upgrade the firmware?

A. It is highly recommend to **follow the upgrade guide procedure for every new release firmware version**. Generally, using an **upgrade tool** through the Web GUI of the device (Method 1).

The device can be upgraded using the following:

Method 1: Manual Upgrade Procedure

- Place the upgrade package (xxxxxxxx.img) to the local computer.
- Upgrade xxxxxxxx.img in **WEB page > Tools> Software upgrade,** automatically restart after the upgrade is complete.
- After the upgrade, view **WEB page > Info**, if the Firmware version is the same as **Packages Info**, it is confirmed the upgrade is successful.

Method 2: Auto-Provisioning Upgrade Procedure

- Place xxxxxxxx.img to the ACS server.
- Set target version package URL in common configuration file *xxxxxxxx.cfg* as the following:

Note: ftp, tftp, http and https are all supported in FIRM_URL.

<config.ini>

[AUTOPROVISION]

//Enable firmware upgrade

FIRM_UPGRADE = Y

//Target version package URL

FIRM_URL = ftp://username:passwd@IP/xxxxxxxx. img

- Reboot the device or set it periodically to obtain configurations from ACS. Then device will automatically upgrade to the target version
- After device start up, check firmware version to confirm a successful upgrade in **WEB > Info**.

Q. What are the safety measures must I take before performing an upgrade on the device?

A. Performing an upgrade is simple yet you need to be extra careful following the procedure. Please do not close the window or remove power or disconnect the Ethernet cable from the device until the upgrade is complete. Doing so could cause an upgrade failure that would result in the device having to be sent back for repair.

Troubleshooting Device Access Issues

Scenario 1	**Unable to access device due to incorrect IP address or incorrect port number**
Device Status	Device successfully powers up with all status LEDs light normally
User Experience	1. When phone on FXS port goes off-hook, a dial tone is heard 2. Cannot load web page by browsing the default IP address of the device, "Connection Timed Out" shown on screen 3. Forgot device IP address or port number (if modified)

Solution	Operations
Obtain or set IP address through a phone attached to any FXS port	1. Pick up the handset and dial *"##"* to hear the device's current IP address and port number 2. Pick up the handset and dial "**90###1#*" to automatically receive IP address from DHCP server 3. To set new IP address for the device, pick up the handset and dial "**90"* + "*IP address followed by #"* + "*net mask followed by #"* + "*gateway IP followed by #"* + "*0#"* Use "*" for the "." within the IP address Note: If your operation is accepted, you should hear the confirmation tone or voice prompt, otherwise you should retry it. Power cycle your device for the configuration to take effect
Get IP address through Finder tool	Visit www.newrocktech.com, click **product > applications > Finder** and download the Finder tool. Unzip to run the program and follow prompts on screen
Applicable Devices	All product series (except for NRP phones)

Scenario 2	**Forgot the password, unable to log in web GUI**
Device Status	Device successfully powers up with all status LEDs light normally
User Experience	1. When phone on FXS port goes off-hook, a dial tone is heard 2. User is able to load web login page and enter a password, but received "Incorrect Password" prompt

Solutions	Operations
Perform factory reset through a phone attached to any FXS port	Pick up the handset and dial *"*911234#"*, after hearing three beeps, power cycle the device
Perform factory reset through the reset button on the device	Hold the reset button for 3 seconds or longer, release after STU LED flashes red, and wait for device to complete reboot
Perform factory reset through Finder tool	Visit www.newrocktech.com, click **product > applications > Finder** and download the Finder tool. Unzip to run the program and follow prompts on screen
Applicable Devices	Factory reset by phone only works on HX/MX series Reset button only works on MX8G,MX60E, MX120G, OM20G, OM50G, OM80E, OM200G, OM500 Factory reset through Finder only works on HX4G

Scenario 3	**Web login is denied because of too many incorrect password or PIN attempts**
Device Status	Device successfully powers up with all status LEDs light normally
User Experience	1. When phone on FXS port goes off-hook, a dial tone is heard 2. User is able to load web login page, however access has been denied (500 Error page is displayed) due to more than 6 incorrect password or PIN attempts
Solutions	Operations
1. Re-attempt login after 600 seconds 2. Re-attempt login from another PC or change PC's IP address 3. Reboot the device and try again	
Applicable Devices	All product series (except for NRP phones)

Scenario 4	**Unable to access device because the PC's IP address is not in the access whitelist (if configured)**
Device Status	Device successfully powers up with all status LEDs light normally
User Experience	1. When phone on FXS port goes off-hook, a dial tone is heard 2. Cannot load the web page even though device IP address is correct, and "Connection Timed Out" shown on screen
Solution	Operations
Access the device through internal preset whitelist address 192.168.2.100	Make a direct connection between the device and PC with an Ethernet cable 1. Use the *"90*"* commands to set the device's IP address to be one in the IP segment of 192.168.2.x, for example 192.168.2.101 2. Configure the PC's IP address to 192.168.2.100, and then access the web GUI through a browser
Applicable Devices	All product series (except for NRP phones)

Scenario 5.1	**Unable to access device after firmware upgrade**
Device Status	Device successfully powers up with all status LEDs light normally
User Experience	1. When phone on FXS port goes off-hook, a dial tone is heard 2. Cannot load web page by browsing the device's IP address, and "Connection Timed Out" shown on screen
Solution	Operations
Perform factory reset through a phone attached to any FXS port	Pick up the handset and dial *"*911234#"*, after hearing three beeps, power cycle the device
Applicable Devices	HX/MX series

Scenario 5.2	**Unable to access device after firmware upgrade**
Device Status	Device powers up abnormally, STU LED flashes red
User Experience	1. When phone on FXS port goes off-hook, no dial tone can be heard 2. Cannot load web page by browsing the device's IP address, and "Connection Timed Out" shown on screen
Solution	Operations
Perform factory reset through the reset button on the device	Hold the reset button for 3 seconds or longer, release after STU LED flashes red, and wait for device to complete reboot

	Note: If reset doesn't bring device back to normal, contact customer service for repair
Applicable Devices	OM series, MX8G, MX60E, MX120G

Scenario 6	**Unable to access device due to IP address conflict**
Device Status	1. On the desktop device, STU LED flashes red 2. On the chassis mounting device, ALM LED flashes, letter "C" is shown on LED matrix
User Experience	1. When phone on FXS port goes off-hook, a dial tone is heard 2. The web page sometimes cannot load even though device IP address is correct, and "Connection Timed Out" shown on screen
Solution	Operations
Re-obtain or set new IP address through the phone attached to any FXS port	1. Pick up the handset and dial "**90###1#*" to automatically receive IP address from DHCP server 2. To set new IP address for the device, pick up the handset and dial "**90" + "IP address followed by #" + "net mask followed by #" + "gateway IP followed by #" + "0#"* Use "*" for the "." within the IP address Note: If your operation is accepted, you should hear the confirmation tone or voice prompt, otherwise you should retry it. Power cycle your device for the configuration to take effect
Access through direct Ethernet cable connection from a PC	Direct connect the device to a computer through an Ethernet cable, login to the web GUI to reconfigure device's IP address
Applicable Devices	All product series (except NRP phones)