New Rock Technologies, Inc.

OM20/OM50 Installation and Configuration Guide

http://www.newrocktech.com

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Amendment Records

Guide Description

This all-in-one guide intends to help the VAR, SI and Service Providers to correctly install and manage the New Rock OM20 and OM50 IP telephony systems.

The manual guides administrators in setting OM parameters on Web interfaces. Some parameters can be set by using a telephone.

Contents

1 Installation and Configuration Process Overview

This chapter provides an overview of the installation and configuration process.

1.1 Preparation

In **Chapter 3, “Preparation,”** you will learn about the components and service requirements, codec and network bandwidth requirements, call capacity, and related topics, to ensure that the system is well planned to meet the needs of the customer. This chapter also describes basic procedures such as updating the firmware, which should be completed before you begin installing the equipment.

1.2 Connecting the Components

In this stage, you physically connect the OM20/50 IP Telephony System equipment to the LAN. **Chapter 4, “Connecting the Components”** explains how to connect the OM20/50, which provides the PBX service for the phones, and the ATA, which provides voice mail service and PSTN access. You also learn how to install the IP phones and any accessories. Furthermore, you also verify the latest upgrade firmware that you have updated during the Preparation phase.

1.3 Configuring Basic Voice Services

After you connect the components, you need to configure voice features such as ITSP service and PSTN access. **Chapter 5, “Configuring Basic Voice Services**” guides you through these steps. You also set up call routing for both outbound and inbound calls.

1.4 Configuring Advanced Features

Now you are ready to begin configuring advanced features, depending on the deployment scenarios. In **Chapter 6, “Configuring Advanced Features,”** you will learn how to configure Music On Hold, to set up the Auto Attendant with Multilevel IVR, to activate voice mail and to route calls with hunt groups and shared line appearances.

1.5 Localizing the System

For customers outside China, you need to localize the system. **Chapter 7, “Localizing the System,”** guides you through the steps.

2 Preparation

This chapter is important reading before you begin installing the equipment or configuring the system. To ensure that the installation process goes smoothly, verify that you have the required services, components, and information.

2.1 Site Survey

The site surveys consists of gathering relevant information about the customer, the existing infrastructure, the network, the telephone equipment, and the available services. This survey helps you to prepare for the installation of the OM20/50 IP Telephony System and to anticipate the design considerations. The site survey can be conducted on the customer premises or remotely over the phone and email. Various site survey templates can be used. **Chapter 10, “Installation Guide and User Training,”** contains a site survey template example that you can use to record the customer.

2.1.1 *ITSP and PSTN Connectivity*

ITSP and PSTN lines: Ensure that the lines are operative and that any features, such as caller identification, operate properly before starting the installation. Ensure that the cables are available in the location where you are installing the New Rock Devices.

2.1.2 *Local Network Connectivity*

The Local Area Network (LAN) is the communication platform used by the OM20/50 IP Telephony System for allowing communications among the telephone users and between the telephone users and the external VoIP and PSTN services. This LAN is composed of the data wiring (UTP cabling), networking equipment (switches and routers/access device) and the telecommunication (PSTN) lines.

The Local Area Network (LAN) may be already installed or it can be installed and configured at the time of installing the OM20/50 IP Telephony System Below are the general recommendations to ensure proper operation of the OM20/50 IP Telephony System.

Ethernet cabling: Ensure there is an Ethernet cabling system and that there is an outlet for each NewRock VoIP device. It is recommended that Ethernet cables are UTP CAT 5e or better.

2.1.3 *Customer Phone Requirements*

The IP phones require access to power outlets or can receive power from a Power over Ethernet (PoE) switch and are not supplied with power supplies. If you are not using the recommended PoE switch, you need to purchase a suitable power supply.

2.2 Deployment Considerations

When installing and configuring the OM20/50 IP Telephony System, it is necessary to analyze and meet some design considerations to ensure the best quality and user experience. The design considerations cover available bandwidth and quality of service.

2.2.1 *Codec and Network bandwidth*

The available connection bandwidth determines the maximum number of simultaneous calls that the system can support with the appropriate audio quality. Before installing and configuring the OM 20/50 IP Telephony System, use this information to determine the maximum number of simultaneous VoIP connections that the system can support. In general it is a best practice to use no more than 75% of the total available bandwidth for calls. This provides space for data traffic and helps ensure good voice quality.

NOTE: Some ITSP SIP trunk services limit the maximum number of simultaneous calls. Please check with your Service Provider to understand the maximum number of simultaneous calls each SIP trunk supports.

The following table provides the approximate bandwidth allocation for different codecs.

Codec	**Approximate bandwidth allocation for each side of conversation (kbit/s)**	**2 calls (kbit/s)**	**10 calls (kbit/s)**
G.729A	55	110	550
G.711U/G.711A	110	220	1100

2.2.2 Factors which will effect voice quality

Quality and reliability were the two darkest spots on VoIP's reputation for the past years. Here are the main things that affect voice quality in VoIP and what can be done to maximize quality.

Bandwidth

Your Internet connection always tops the list of factors affecting voice quality in VoIP conversations. The bandwidth you have for VoIP is the key for voice quality. For instance, if you have dial-up connection, don't expect great quality. A broadband connection will work right, as long as it is not spotty, and not shared with too many other communication applications.

Equipment

The VoIP hardware equipment you use can greatly impact on your quality. Poor quality equipment are normally the cheapest ones (but not always!). It is therefore always good to have as much information as possible on an ATA, router or IP phone before investing on it and starting to use it.

NAT Mapping

If an IP extension registered to OM (OM puts behind router) via internet or if multi-site networking is used, some form of NAT (Network Address Translation) traversal is required. NAT is a function that allows multiple devices to share the same public, routable, IP address to establish connections over the Internet. NAT is present in many broadband access devices to translate public and private IP addresses.

Some ITSPs provide NAT traversal, but some do not. If your ITSP does not provide NAT traversal, you have several options. Use NAT mapping with SIP-ALG (Application Layer Gateway) in the router.

It is necessary to configure remote-address information and configure port mapping on the Internet ingress router or used the P2P traversal feature on OM. This enables devices on external networks to traverse NAT (Network Address Translation) to get access to the OM.

Weather Conditions

The voice is terribly distorted by something called **static**, which is a small static electricity generated on broadband lines due to thunderstorms, heavy rain, strong gusts, electrical impulses etc. This static is not very much noticeable when you surf the net or download files, which is why we don't complain about it when we use the Internet for data despite it be here; but when you are listening to voice, it becomes disturbing. It is easy to get rid of static: unplug your hardware (OM20/50, ATA, router or phone) and plug it back again. The static will be brought to naught.

The effect of weather conditions on your connection is not something you can change. You can have some short-term relief in some cases, but most of the time, it is up to your service provider to do something. At times, properly grounding the device, modifying some parameter values on device and changing the cables solves the problem completely.

Location of your hardware

Interference is another factor for voice quality during voice communication. Often, VoIP equipment interfere with each other thus producing noise and other problems. For example, if OM20/50 and ATA are too close to your broadband router or any appliances, you might experience voice quality problems. This is caused by electrical feedback. Try moving them away from each other to get rid of the garbled calls, echoes, dropped calls etc.

2.2.3 *Static or Dynamic IP address for devices*

The OM20/50 becomes a DHCP client of any server on the network. The recommended setting is to use a static IP address. This configuration provides ease of installation and prevents connectivity issues that would occur if the IP address of the OM20/50 changed.

When the extension needs to register with the OM20/50 from an external network, New Rock recommends mapping an external (public) Static IP address or dynamic domain name on the OM20/50.

2.2.4 *Fax and POS machine*

OM20 supports two fax modes: T.38 and G.711 transparent transmission. It is highly recommended to check if the facsimile and POS machine can be connected to the FXS port of OM20.When fax messages are received or sent through an analog trunk, the G.711 transparent transmission mode is required. When fax messages are received or sent through an IP trunk, a T.38 or a G.711 transparent transmission mode needs to be selected according to an actual requirement and the mode supported by the IP phone operation platform. If both T.38 and G.711 transparent transmission modes are supported, T.38 is recommended because it is more stable.

2.3 Updating the Firmware

It is highly recommended that you check for recent updates before you install your equipment. Later instructions in this guide will help you to install the firmware that you download in this preparation phase. To find the latest firmware for a device, go to http://en.newrocktech.com/software/

3 Connecting the Components

3.1 Connecting the Administration Computer

3.1.1 *Connecting the Computer to the LAN*

A user could fail to log in with the default IP address if the IP address of user's computer and the default OM IP address are not at the same network segment. Set the IP address of user's computer to be identical with the same network segment of the device. For example, if the OM IP address is 192.168.2.218, set the computer's IP address to any address at the network segment of 192.168.2.XXX.

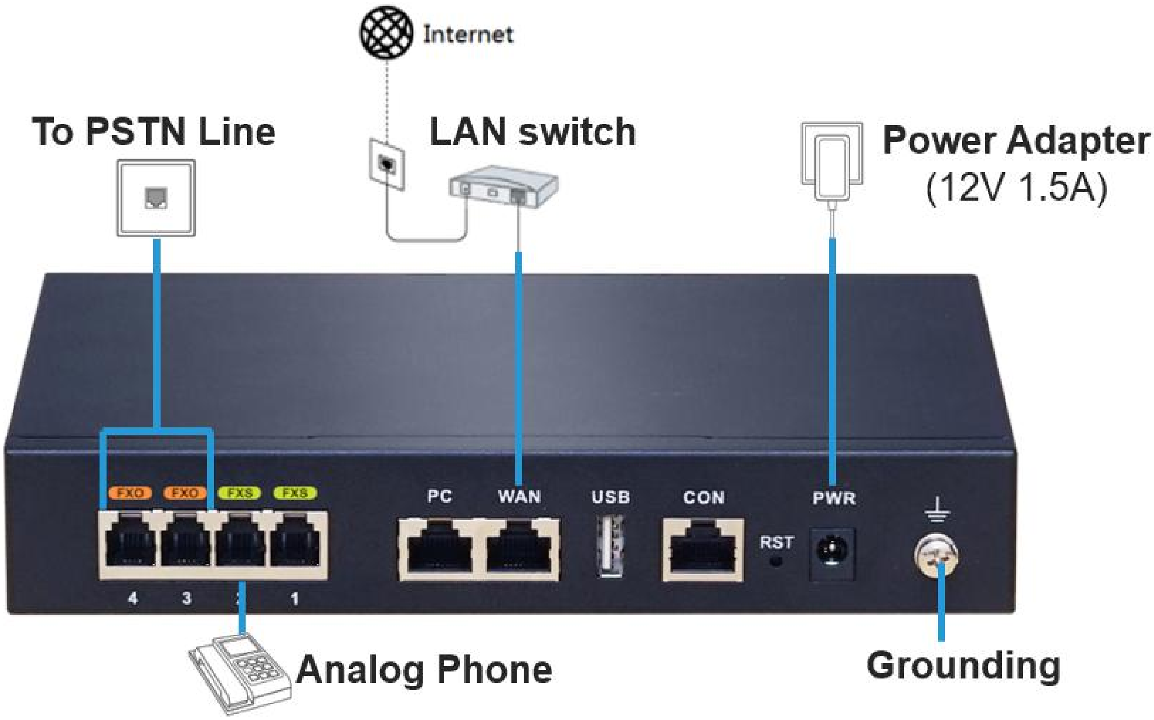
3.1.2 *Download Finder software*

Finder is a PC-based software for obtaining the IP address of New Rock products, especially FXO devices, in a LAN environment.

Link to download Finder software installation package: http://en.newrocktech.com/softwares/700.html

3.2 Installing the OM20

3.2.1 *Connecting the OM20 to the LAN*



Step 1 Connect the WAN port of the OM20 to the LAN switch.

Step 2 Connect the FXO port of the OM20 to the telephone line provided by a telecom operator or an extension line from another PBX.

Step 3 Connect the FXS port of the OM20 to an analog phone or a fax machine.

Step 4 Connect the grounding cable: connect the end with a smaller diameter to the OM20, and connect the other end to a ground bar.

Step 5 Connect the PWR port to the power adapter supplied with the OM20 and then connect the power adapter to an electrical outlet. The PWR LED turns blinking green to indicate that the device is starting and will be in steady green to indicate that the device is running normally.

3.2.2 *Finding the IP address of the OM20*

By default, the IP address of OM20 is automatically assigned by the DHCP server. The IP address can be found by using these methods:

Method 1: If OM20 has FXS port: dial "##" to obtain device IP address by an analog telephone connected to the FXS port after the equipment is powered on.

Method 2: Obtain device IP address via New Rock's Finder software as follows:

Step 1 Double click and allow to run this application on your PC.

Step 2 Click Refresh, and the MAC address and IP address of target models will be listed.

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

3.2.3 *Logon to the OM20*

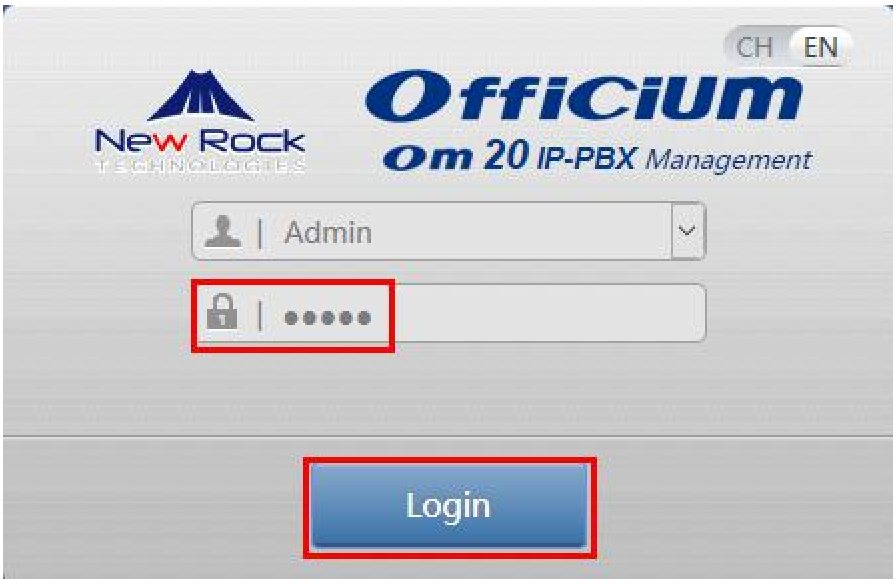
Step 1 Make sure that the PC and the device are on the same network segment.

Step 2 Enter the device IP address in the browser address bar (e.g. 192.168.2.218).

Note

- The Web GUI can be accessed using browsers such as Internet Explorer 8 to 11, Firefox, and Google Chrome.
- The device is only allowed to access using HTTPS. Since the factory default certificate is used, a prompt like "There is a problem with this website's security certificate" may occur. Click **Continue to this website** to access the login page.

Step 3 Enter the login interface for OM20 configuration by selecting your role as **admin** and entering the password, which by default is **admin**.



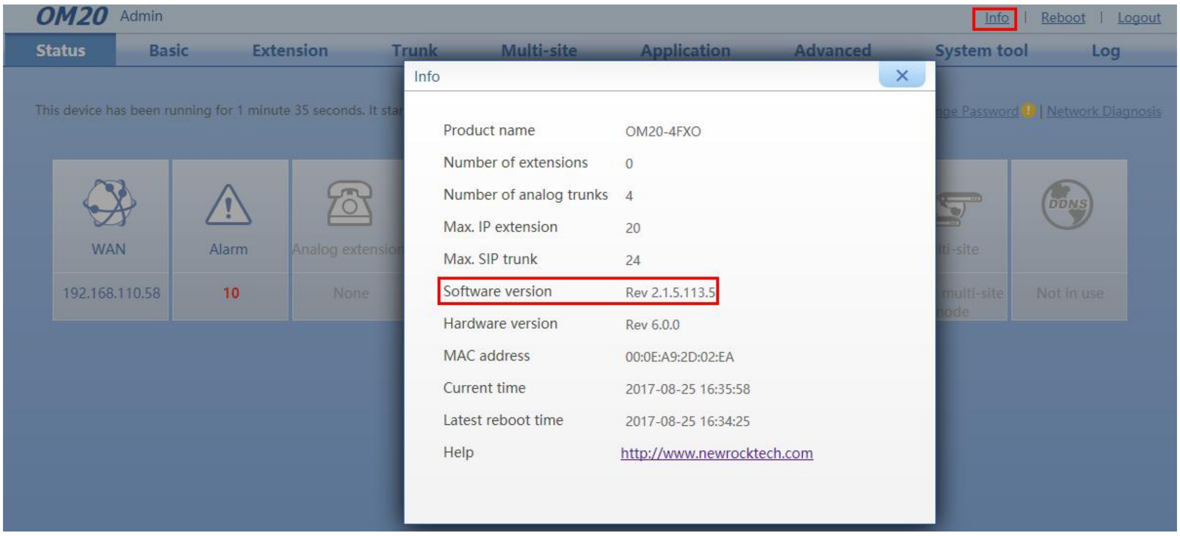
Note

Item	Description
Role	The Web utility provides two authority levels: • An administrator is allowed to make changes to any configuration. 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 corner. The device allows multiple admin 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 corner. **Note:** In the “Welcome user” mode, the operation only can browse certain pages. The

Item	Description
	pages that cannot be browsed include: **Advanced** > **Security**, **System tool** > **Change password**, **System tool** > **Software upgrade**, **System tool** > **Import data**, **System tool** > **Export data**.

3.2.4 *Verifying the Firmware Version*

Step 1 Click **Info** and you can view the software version of OM20



3.3 Installing the NRP IP Phones

Note: : The plug and play procedure described below is only applicable to New Rock NRP series IP Phones. For installation of IP phones of other brands please refer to the installation manual supplied by the phone manufacturer.

3.3.1 *Connecting the Phones to the LAN*

Step 1 Make sure that the NRP IP phone and the OM20 are on the same network segment

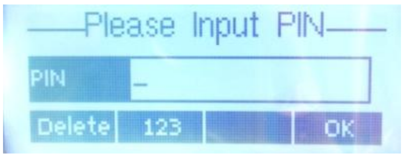
Step 1 Connect the WAN port of the IP Phone to the LAN switch.

Step 2 Connect the PC port of the IP Phone to a PC if there is one.

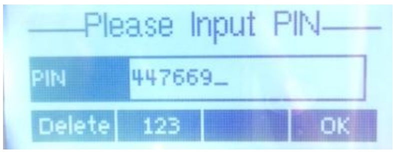
Step 3 Connect the PWR port of the IP Phone to the power adapter supplied with the IP phone and then connect the power adapter to an electrical outlet. You can view the system progress and the **INITIALIZING** prompt on the screen.

3.3.2 *Activating the Phones using PIN*

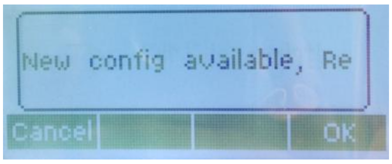
Step 1 After connection and powered on, the screen will display "Please Input PIN "



Step 2 Input PIN for the extension and click **OK**. (e.g. 447669)



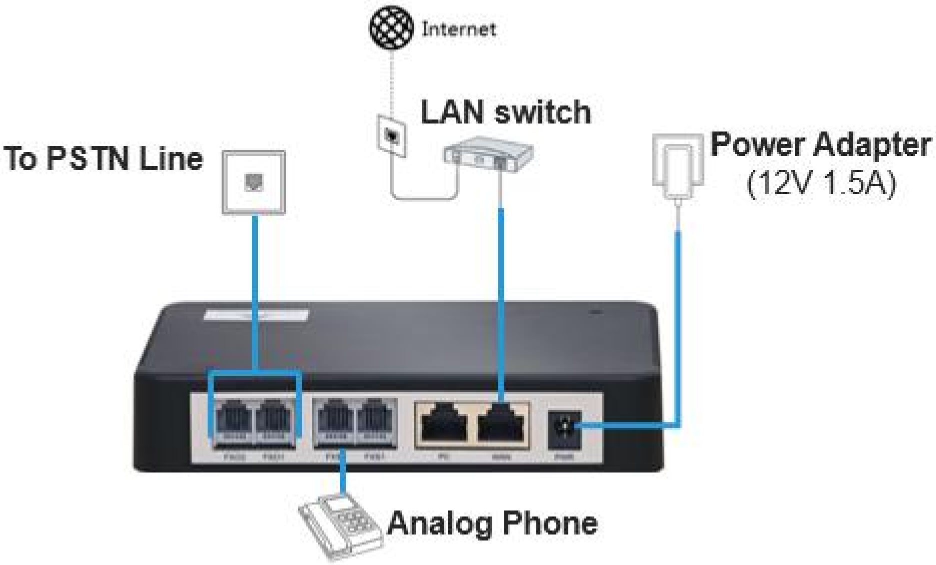
Step 3 When "**The PIN is right!"** prompts, click **OK** to reboot to make the configuration effective.



Step 4 The extension number will display at the upper left corner after a reboot

3.4 Installing the ATA

3.4.1 *Connecting the ATA to the LAN*



Step 1 Connect the WAN port of the ATA to the LAN switch.

Step 2 Connect the FXO port of the ATA to the telephone line provided by a telecom operator or an extension line from another PBX.

Step 3 Connect the FXS port of the ATA to an analog phone or a fax machine.

Step 4 Connect the PWR port of the ATA to the power adapter supplied with the ATA and then connect the power adapter to an electrical outlet. The PWR LED turns blinking green to indicate that the ATA is starting and will be in steady green to indicate that the device is running normally.

3.4.2 *Finding the IP address of the ATA*

By default, the IP address of ATA is automatically obtained by DHCP. The IP address can be obtained by using these methods:

Method 1: If ATA has FXS port: dial "##" to obtain device IP address by an analog telephone connected to the FXS port after the equipment is powered on.

Method 2: If ATA without FXS port: obtain device IP address via New Rock's Finder software.

How to obtain device IP address via Finder:

Step 1 Double click and allow to run this application on your PC.

Step 2 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.

3.4.3 *Logon to the ATA*

Step 1 Make sure that the PC and the ATA are on the same network segment.

Step 2 Enter the ATA IP address in the browser address bar (e.g. 192.168.2.218)

Note

- The Web GUI can be accessed using browsers such as Internet Explorer 8 to 11, Firefox, and Google Chrome.
- The device is only allowed to access using HTTPS. Since the factory default certificate is used, a prompt like "There is a problem with this website's security certificate" may occur. Click **Continue to this website** to access the login page.

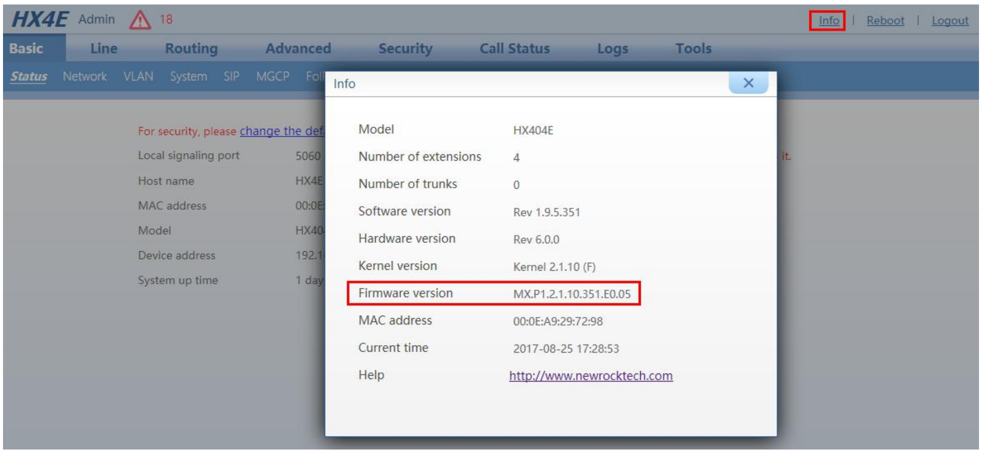
Step 3 You can enter the login interface for ATA configuration by selecting your role and entering a password on the login interface. The default administrator password is **admin**.

Note

Item	Description
Role	The Web utility provides two 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. The device 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. **Note:** In the "Welcome user" mode, the operation only can browse certain pages. The pages that cannot be browsed include: **Advanced** > **Security**, **System tool** > **Change password**, **System tool** > **Software upgrade**, **System tool** > **Import data**, **System tool** > **Export data**.

3.4.4 *Verifying the Firmware Version*

Step 1 Click **Info** and you can view the firmware version of ATA



4 Configuring the Basic Voice Services

This chapter guides you through the basic tasks that are required to get your voice system running. After you complete these procedures, users will be able to place and receive calls from the ITSP and from the PSTN.

4.1 Configuring the OM20

4.1.1 *Configuring the General Settings on the OM20*

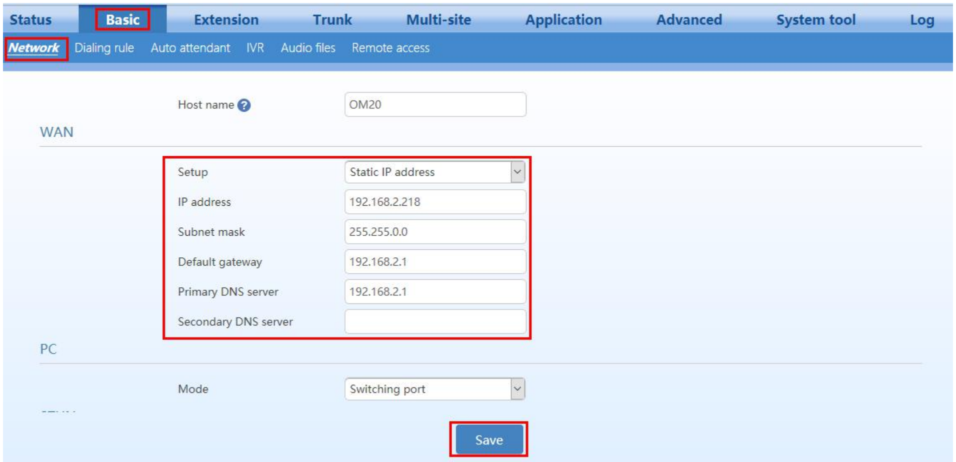
There are several settings that are recommended to ensure good performance on your voice network.

Complete this procedure before performing any other configuration tasks.

Step 1 You need to localize the system configuration by modifying some parameters according to your regional preference, including time zone, country and area code, Caller ID detection mode, call progress tones and digit map. Refer on **Chapter 7, "Localizing the System,"**

Step 2 It is recommended to assign a static IP address for your OM, especially if there are IP phones and or ATAs which are registered to the OM.

Method 1: On OM20 interface, go to **Basic > Network** page to setup Static IP address on WAN port and configure the network parameters.



Method 2: On Analog phone , dial * 90 and configure your network parameters as the following example:

192*168*2*218#255*255*0*0#192*168*2*1#0#

IP address   Subnet mask   Default gateway

Note *Configuration will only take effect after a reboot.*

4.1.1 Configuring the OM20 for ITSP Connectivity

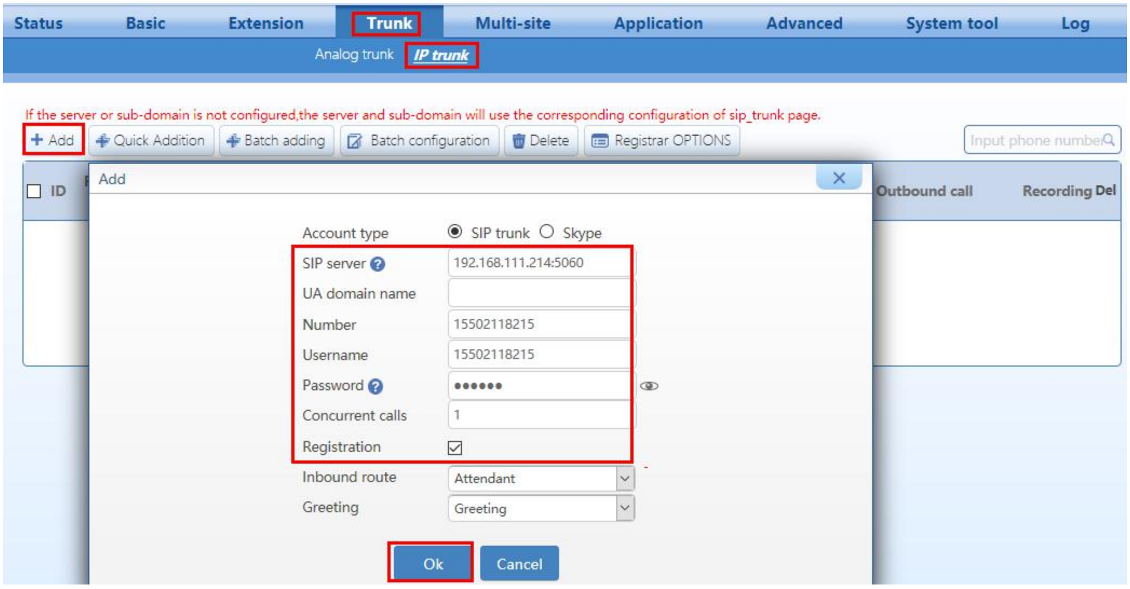
The OM supports standard SIP specifications and Skype Connect. Before setting the IP trunk, you need to obtain an account from your ITSP.

Step 1 Log in to **OM20** as **Admin** account

Step 2 Go to **Trunk** > **IP trunk.**

Step 3 Click **Add** and enter the account information for your ITSP account:

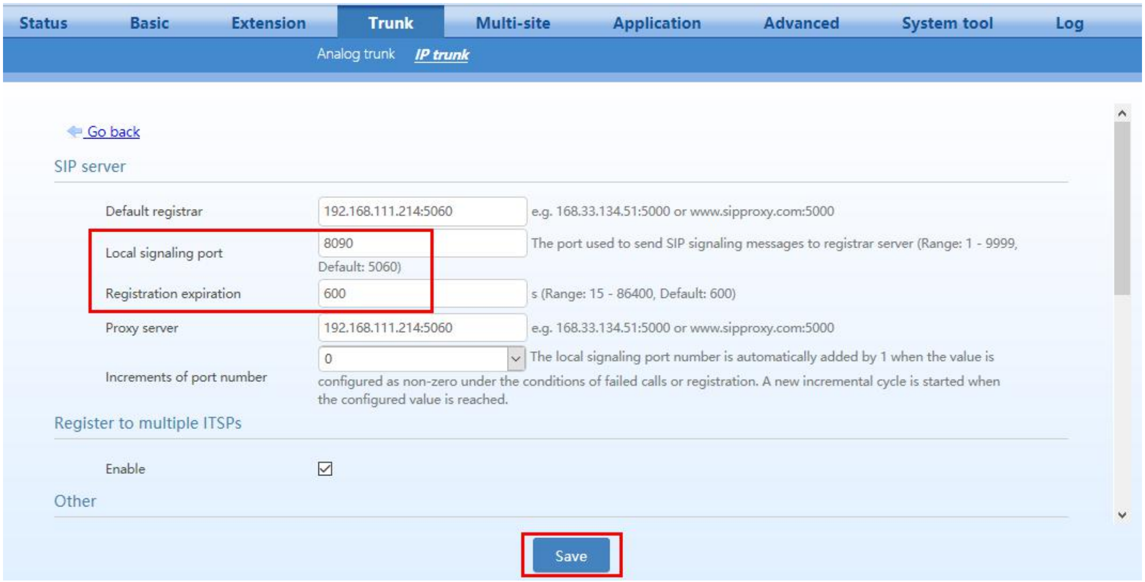
Note To register trunks to different SIP servers, go to **Trunk** > **IP trunk** > **Register OPTIONS** and enable **Permit to use multiple ITSPs** first.



Item	Description
SIP server	Enter the server IP address and port provided by the ITSP.
UA domain name	Provided by your ITSP, for example, salesdepart.abccompany.com.
Number	Provided by your ITSP.
Username	Provided by your ITSP. It is used for authentication when registering an IP trunk. If no username is entered, the trunk ID will be used for authentication.
Password	Provided by your ITSP. The password is encrypted by default. Please contact your service provider if you forgot the password. The registration password can contain up to 16 characters if encryption is enabled (The password can contain up to 30 characters if encryption is disabled).
Concurrent calls	The number of concurrent calls supported by the trunk. Note that the total number of all trunks must not exceed the maximum number for the device. The OM50 supports a maximum of 30 concurrent calls, and the OM20 supports a maximum of 24 concurrent calls.
Registration	Select this to enable registration.

Step 4 Click **OK** to return to the IP trunk setting interface, and view the registration status for the configured IP trunk.

Step 5 Click **Registrar OPTIONS**. You can modify information such as local signaling port and registration expiration. It is recommended to change the **Local signaling port** to prevent SIP attacks.



Step 6 Click **Save.** The configuration takes effect after the device is restarted.

Step 7 To verify your progress, perform the following tasks:

- After the devices reboot, click **Trunk** > **IP trunk.** Verify that the line is registered. You may need to refresh the browser screen to see the new status.
- Use an external phone, such as a cell phone, to place an inbound call to the telephone number that was assigned by your ITSP. Assuming that you have left the default settings in place, the Auto Attendant answers the call. You can then dial an extension number to verify that the call rings to the station.

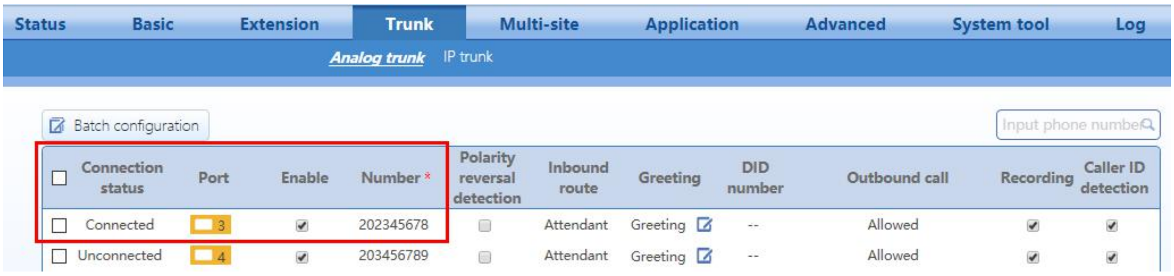
4.1.2 *Configuring the OM20 for PSTN Connectivity*

Public Switched Telephone Network (PSTN) access is an optional if you have an internet telephone service. You can connect up to four standard analog telephone lines to OM20 depending on the model you purchased and then configure it so that the users can place and receive calls through the PSTN lines.

Follow this procedure:

Step 1 Log in to **OM20** as **Admin** account

Step 2 Go to **Trunk** > **Analog trunk** to verify if the line connection line status is connected and configure the actual trunk number.



Item	Description
Connection status	Displays whether the current port is connected to an analog trunk.
Port	FXO port number on the device.
Enable	Select to enable the line. By default, the trunk line is enabled.
Number	The trunk number will be displayed as the calling number when the incoming call number is not displayed or the caller ID is disabled, so it is recommended to use an actual trunk number. The number is 2xx by default.

Step 3 To verify your progress, perform the following tasks:

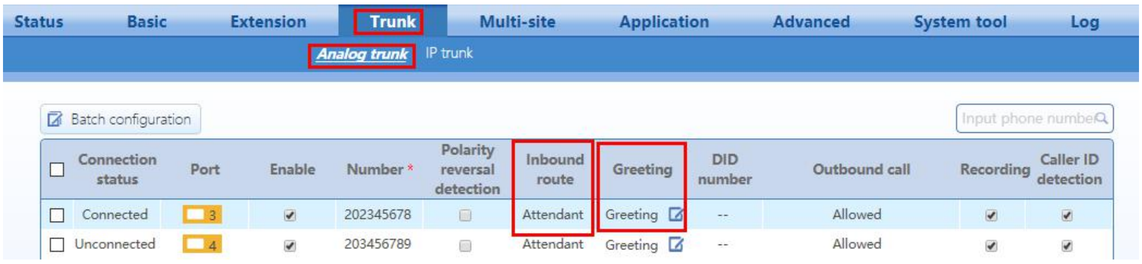
- Use an external phone, such as a cell phone, to place an inbound call to the telephone number (trunk line number). Assuming that you have left the default settings in place, the Auto Attendant answers the call. You can then dial an extension number to verify that the call rings to the station.

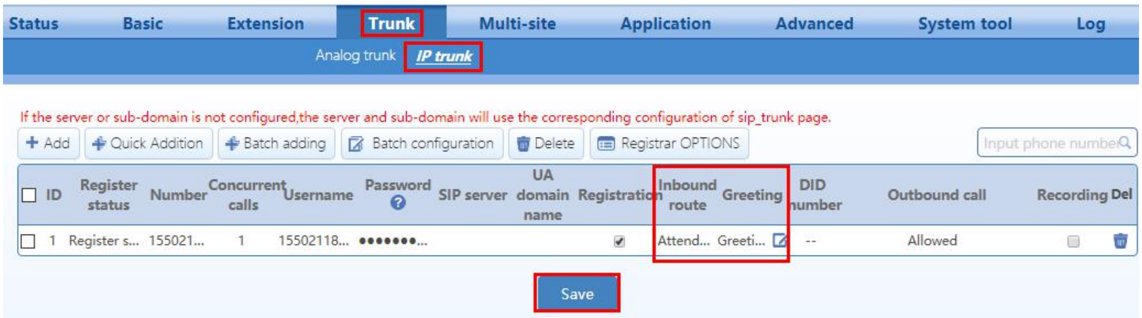
4.1.3 *Configuring the Inbound Handling – Auto Attendant*

By default, all incoming calls are directed to the auto attendant, where callers are prompted to enter the extension number of the destination phone providing immediate and professional service. You can schedule different auto attendants to play, based on the time and day of the week.

Step 1 Log in to **OM20** as **Admin** account

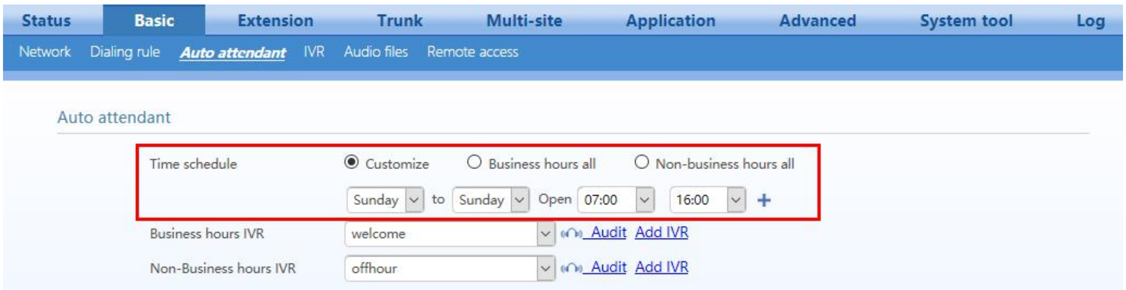
Step 2 On **Trunk > Analog trunk** or **IP trunk** page, verify if the inbound route is set to **Attendant** and set the greeting for the trunk. By default, the greetings for the auto attendant are configured on the **Basic** > **Auto attendant** page.





Step 3 Go to **Basic** > **Auto attendant** to enter **Auto attendant configuration interface.**

Step 4 Assign time schedule. The default is business hours all.



Note

Item	Description
Customize	You can set the range of business hours in a week. The hours outside of business hours are non-business hours. You can click + to divide one day into up to three business-hour segments. The device will play corresponding greetings according to the preset business hours or non-business hours.
Business hours all	The device plays business-hour greetings at any time.
Non-business hours all	The device plays non-business-hour greetings at any time.

Step 5 Click **Save** to save the configuration.

Step 6 To verify your progress, make an incoming call and check if the Auto Attendant answers the call and you are routed directly to the desired extension number.

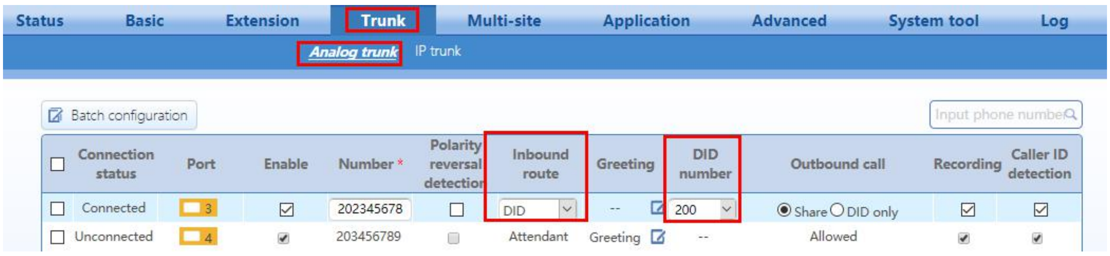
If your business requires more flexible Auto Attendant to offer callers better experience, you may refer ?????

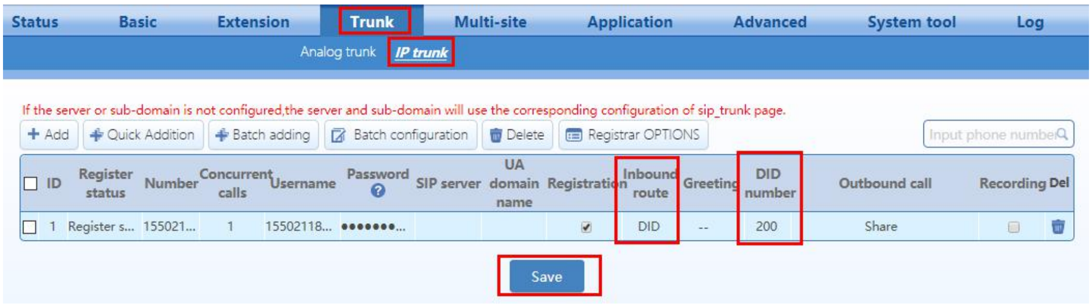
4.1.4 *Configuring the Inbound Handling – DID*

Direct Inward Dialing (DID) directs an incoming call to the bound extension or extension group without playing voice greetings or the need for auto-attendant.

Step 1 Log in to **OM20** as **Admin** account

Step 2 On **Trunk > Analog trunk** or **IP trunk** page, select DID in the **Inbound route** field and fill in the extension number in the DID number field for directing incoming call to the bound extension





Note

- Before setting a DID number for an IP trunk, you need to register the number on the service platform of the service provider.

Step 3 To verify your progress, make an inbound call to the DID number and verify that the call is directed to the specified target extension.

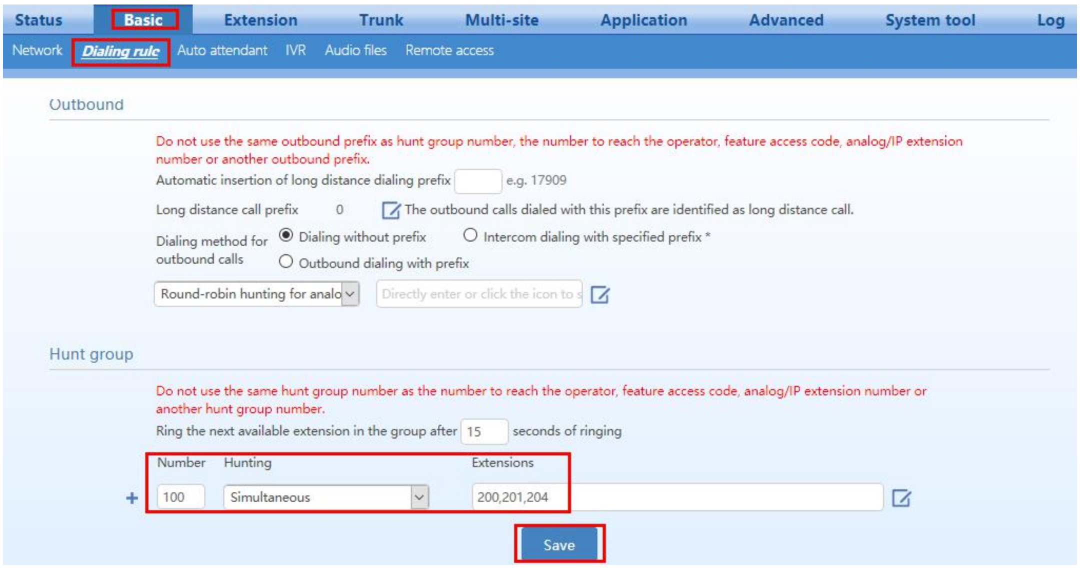
4.1.5 *Configuring the Inbound Handling – Key System*

A typical key telephone system supports sharing the same Trunk line (Analog or IP) appearance on IP phones.

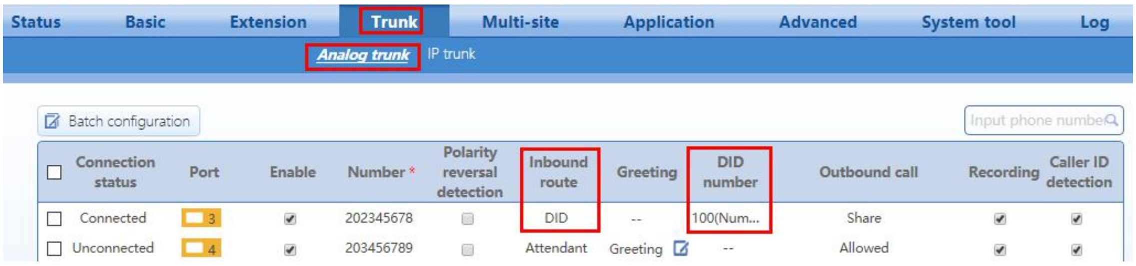
Step 1 Log in to **OM20** as **Admin** account

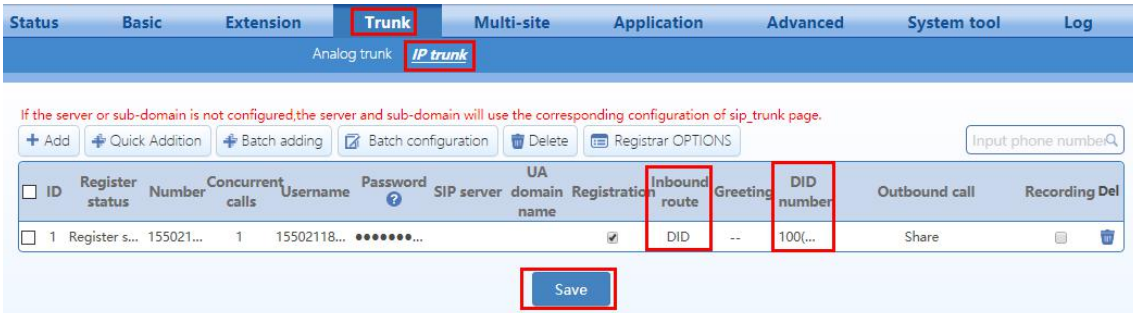
Step 2 Go to **Basic** > **Dialing rule** and set the **Hunting group.** You can allocate multiple extensions to a hunt group of extensions. When a caller dials the hunt group number, the device will ring an idle extension in the group according to the preset allocation.

Step 3 Add the **extension number** to the hunt group, then set the hunting group method into **Simultaneous** to terminate incoming calls to all available extensions, and fill in the **extension numbers** included in a hunting group.

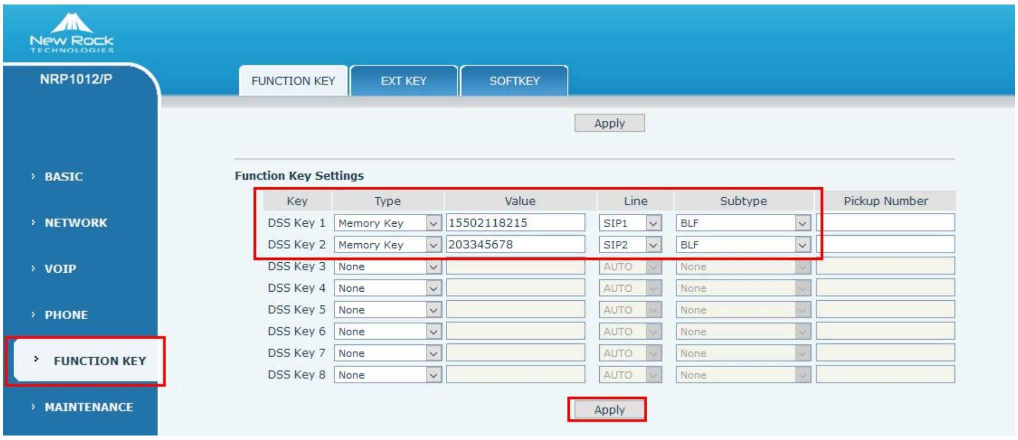


Step 4 On **Trunk > Analog trunk** or **IP trunk** page, select DID in the **Inbound route** field and fill in the hunt group number in the DID number field for directing incoming call to the bound hunt group number.





Step 5 Log in to **IP phone** as **Admin** account and configure the function keys as follows:

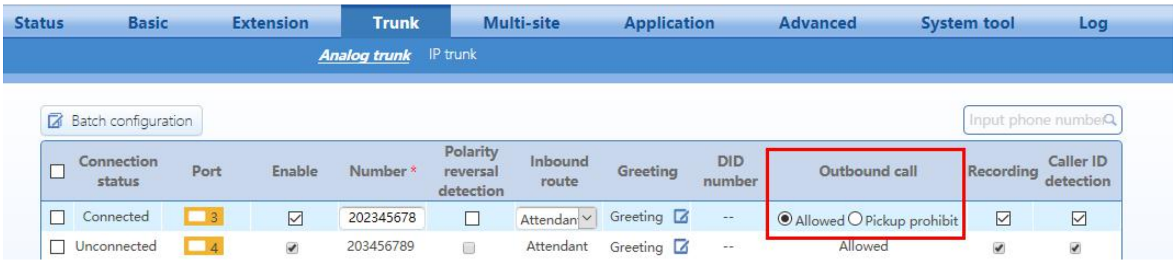


Step 6 To verify your progress, make an inbound call and verify if the call comes in Trunk line 1, every phone will ring simultaneously and the BLF status will be flashing red on IP phones. Also verify if Trunk line 1 is busy, BLF of Trunk line 1 on every phone is red.

4.1.6 *Configuring the Outbound Handling*

Step 1 Log in to **OM20** as **Admin** account

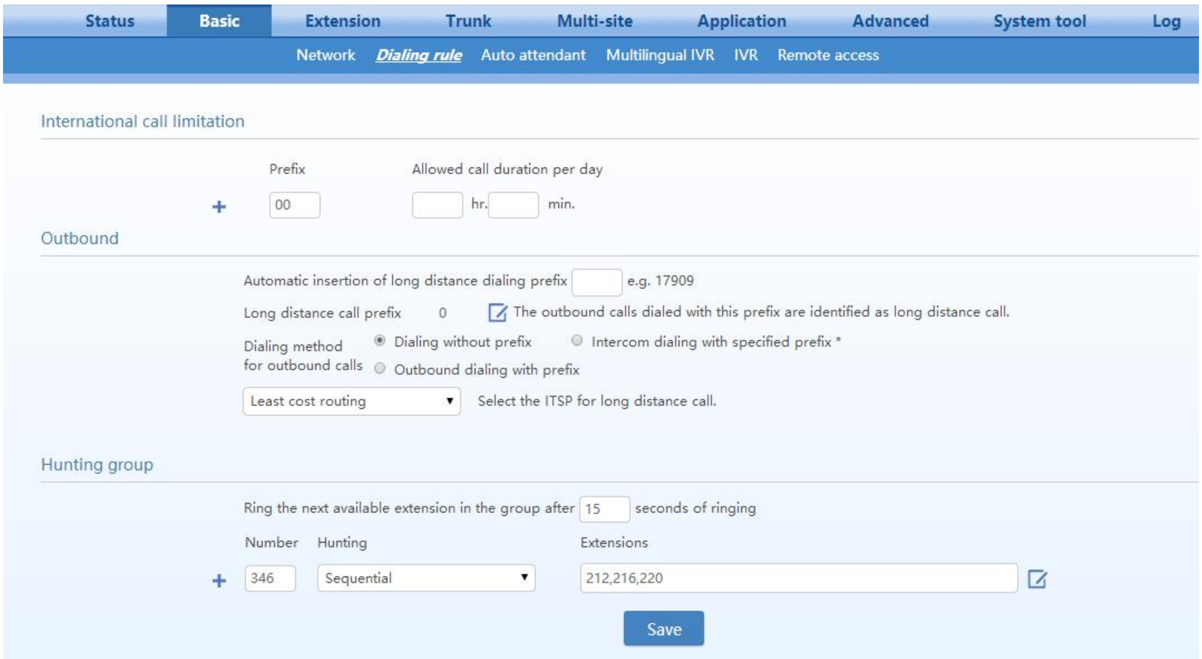
Step 2 On **Trunk > Analog trunk** or **IP trunk** page, select **Outbound call** characteristic for the trunk.



Note

Item	Description
Outbound call	When the **Inbound route** is configured as **Attendant**, there are two choices : • **Allowed**: Allowed to make outbound calls; • **Pickup prohibit**: Not allowed to make outbound calls. When the **Inbound route** is configured as **DID**, there are two choices : • **Share**: Other extensions are allowed to make outbound calls. • **DID only**: Only the extension or hunt group specified in **DID number** is allowed to make outbound calls through this trunk. And the bound extensions can only use this trunk to make outbound calls.

Step 3 Go to **Basic** > **Dialing rule** > **Outbound** to configure outbound dialing rule**.**



Note

Item	Description
International call limitation	The outbound calls dialed with the prefix configured here are identified as international call. You can limit the international call time for every day. **Note:** If the Call restriction of the extension is configured as **Prohibited**, **Internal**, **local** or **long distance**, the number with the prefix configured here cannot be dialed.
Automatic insertion of long distance dialing prefix	When the user makes a long distance call using the analog trunk, the prefix set will be automatically added.
Long distance call prefix	The outbound calls dialed with the prefix configured here are identified as long distance call. The default value is 0. **Note:** If the **Call restriction** of the extension is configured as **Prohibited**, **Internal** or **local**, no long distance outbound call will be allowed.
Dialing method for outbound calls	• **Dialing without prefix**: Directly dial internal extension or external numbers. • **Intercom dialing with specified prefix***: Dial external numbers directly while dial internal numbers by adding prefix *. • **Outbound dialing with prefix**: Dial internal extension numbers directly while dial internal numbers with prefix.
Routing	When the device is configured with multiple trunks, a corresponding outbound call hunting method can be selected as required. The device provides six routes of outbound call for users to select. • **Sequential hunting for analog trunk:** Make the outgoing call through the first available analog trunk on the analog trunk list starting from the first one. • **Round-robin hunting for analog trunk:** Make the outgoing call through the analog trunk in Round-robin order. • **Sequential hunting for IP trunk:** Make the outgoing call through the first available IP trunk on the IP trunk list starting from the first one. • **Round-robin hunting for IP trunk:** Make the outgoing call through the IP trunk

Item	Description
	in Round-robin order. • **Least cost routing**: Analog trunks are selected for local calls, and IP trunks are selected for long distance/international calls. The device determines the long distance/international calls based on the prefix. For example: If the long distance call prefix is 0 and the international call prefix is 00, an IP trunk is selected for the calls made with the numbers starting with "0" or "00". An analog trunk is selected for local calls. If the IP trunk is not activated or a network failure occurs, an analog trunk is also selected for calls made with the numbers starting with "0" or "00". **Note:** If a long distance/international call is made with all IP trunks occupied, the following announcement will be played: All circuits are busy. Please try your call again later. • **Route:** The routing table rules are used to make the call to the PSTN.
Prefix	The user can make an outbound call with a routing method identified by the prefix configured here. For example, if the prefix for **Sequential hunting for IP trunk** is 9, the device will make sequential hunting over IP trunk when dialing the number starting with 9. **Note:** • The parameter can be configured only when the **Dialing method for outbound calls** is **Outbound dialing with prefix**. • To avoid collision, the prefix must be different from extension number, hunting group number, number to reach the operator, feature access code, and other outbound call prefixes.
Secondary dial tone	After the extension user dials the prefix, the device prompts the user to dial the called number with secondary dial tone. **Note:** Applicable only when the **Dialing method for outbound calls** is **Outbound dialing with prefix**.
Trunk	Specify the corresponding trunk numbers for the outbound group of analog trunks and IP trunks. Select the trunk numbers directly or enter them manually. The trunk numbers must be separated by ",".

Step 4 To verify your progress, place an outbound call to an external cell phone or other phone with caller ID, and confirm that the specified line was used for the call.

4.2 Configuring the ATA

4.2.1 *Configuring the ATA General Settings*

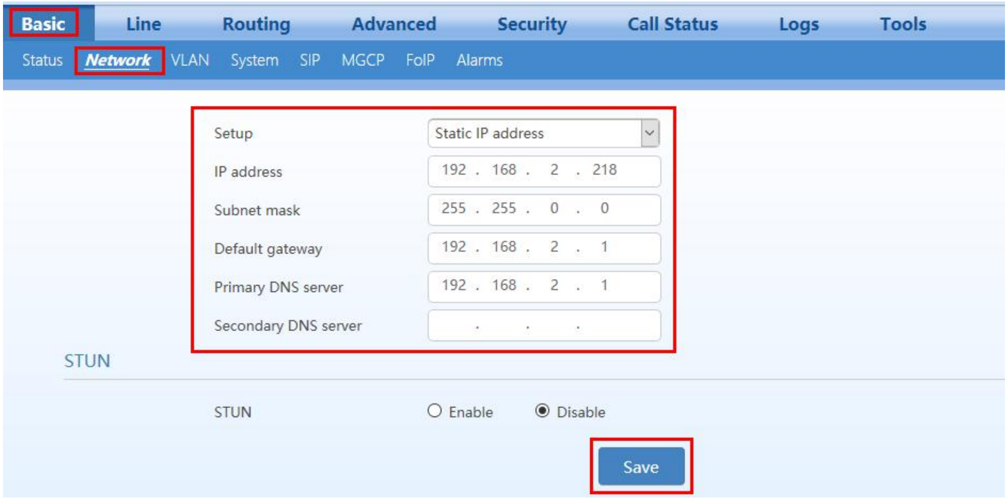
There are several settings that are recommended to ensure good performance on your voice network.

Complete this procedure before performing any other configuration tasks.

Step 1 You need to localize the system configuration by modifying some parameters according to your regional preference, including time zone, Caller ID detection mode, call progress tones and digit map.

Step 2 It is recommended to assign a static IP address for your ATA.

Method 1: On ATA interface, go to **Basic > Network** page to setup Static IP address on WAN port and configure the network parameters.



Method 2: On analog phone, dial * 90 and configure your network parameters as the following example:

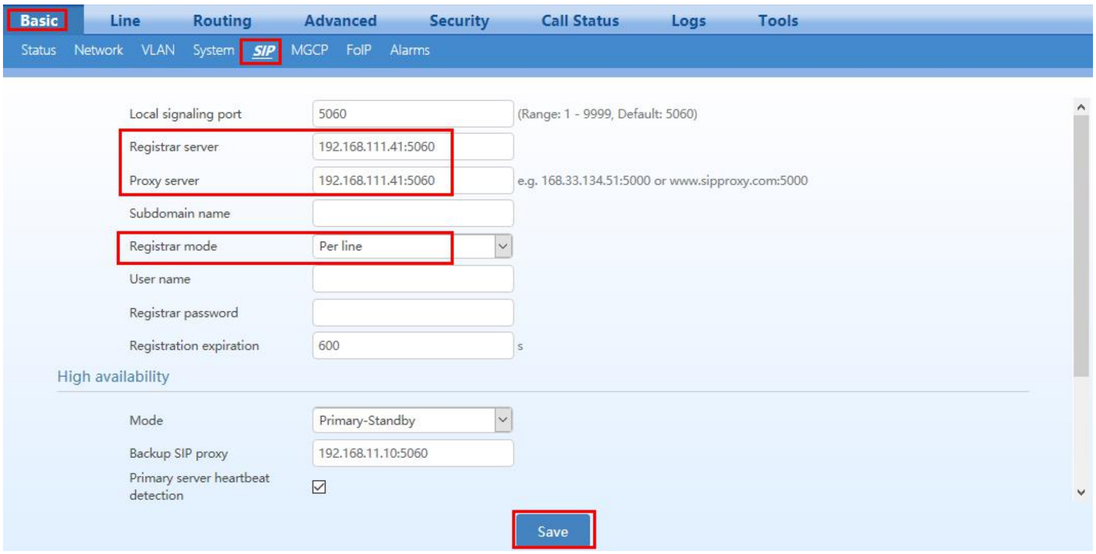
$$\underbrace{192*168*2*218\#}_{\text{IP address}}\underbrace{255*255*0*0\#}_{\text{Subnet mask}}\underbrace{192*168*2*1\#}_{\text{Default gateway}}0\#$$

Note *Configuration will only take effect after a reboot.*

4.2.2 *Configuring the ATA to Register to OM20*

Step 1 Log in to **ATA** as **Admin** account

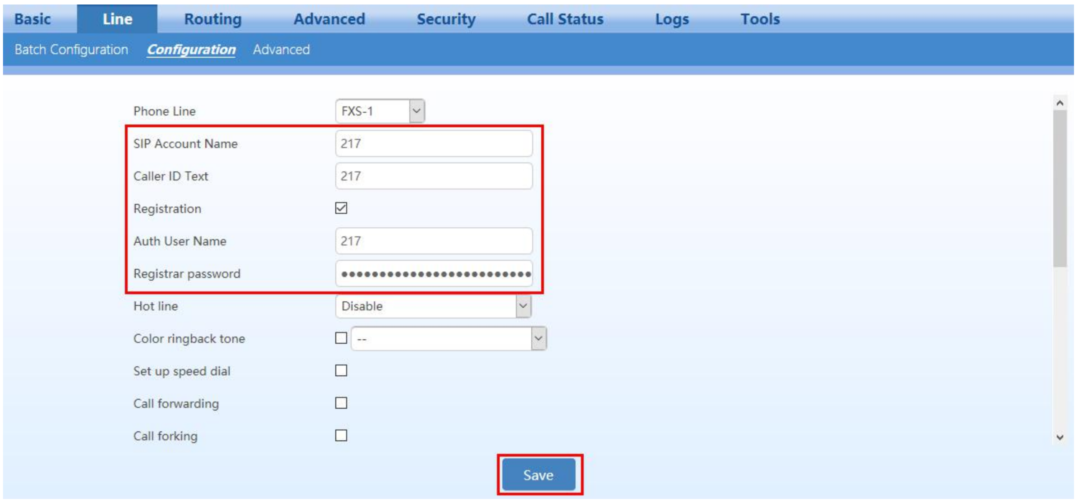
Step 2 On **Basic** > **SIP** page, choose **Registrar mode** to **Per line** and fill in both **Registrar** and **Proxy server** fields using OM IP address and then click **Save**.



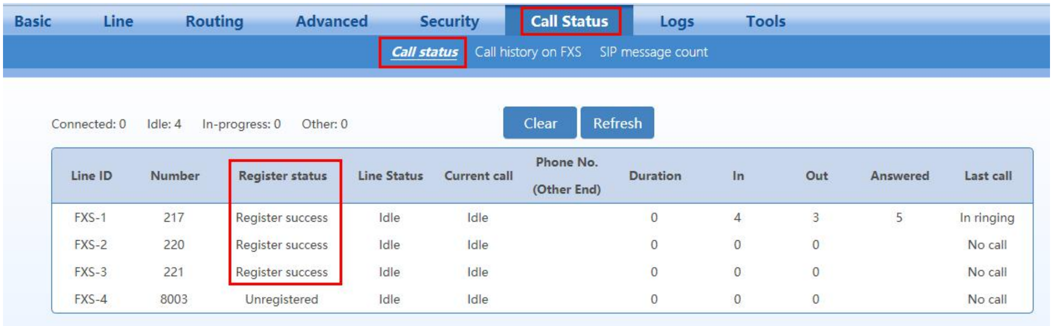
Note

Item	Description
Registrar mode	The gateway supports three registration schemes: • **Per line** (default): authenticate and register per line. • **Per gateway**: authenticate and register per gateway. • **Per line/GW auth**: Enable registration per line. Use the number configuration per line. Use the global account and password in authentication. **Per line** registration is the most preferred one.
Registrar server	Configure the address and port number of the SIP registration server. The address and port number are separated by ":". It has no default value. The register server address can be an IP address or a domain name. When a domain name is used, you must activate DNS service and configure DNS server parameters on the network-configuration page.
Proxy server	Configure the IP address and port number of the SIP proxy server. The address and port number are separated by ":". It has no default value. The proxy server address can be set to an IP address or a domain name. When a domain name is used, you must activate DNS service and configure DNS server parameters on the network-configuration page.

Step 3 On **Line > Configuration** page, fill in the **SIP Account Name**, **Call ID text, Auth User Name** and **Registrar password** fields using the OM's IP extension and password. Enable the **Registration** button and click **Save**.

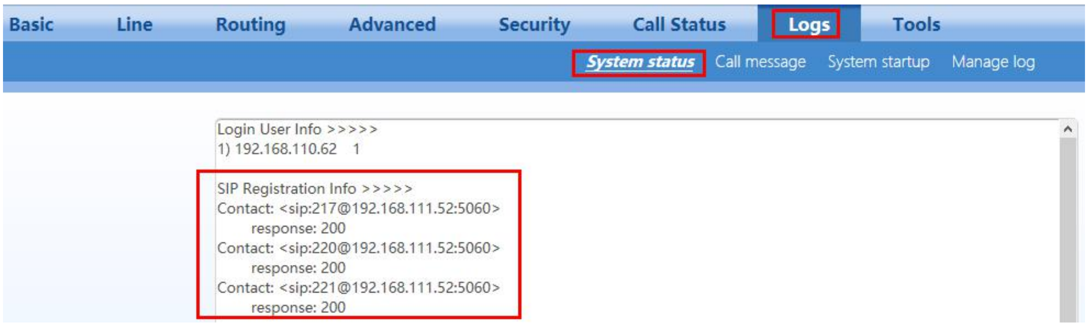


Step 4 To verify your registration status is successful, go to **Call Status > Call Status > Register status**



Line ID	Number	Register status	Line Status	Current call	Phone No. (Other End)	Duration	In	Out	Answered	Last call
FXS-1	217	Register success	Idle	Idle		0	4	3	5	In ringing
FXS-2	220	Register success	Idle	Idle		0	0	0		No call
FXS-3	221	Register success	Idle	Idle		0	0	0		No call
FXS-4	8003	Unregistered	Idle	Idle		0	0	0		No call

Step 5 (Optional) Another way to verify your registration status is successful, go to **Logs > System status > SIP Registration Info ,** latest response should be **200**



Note

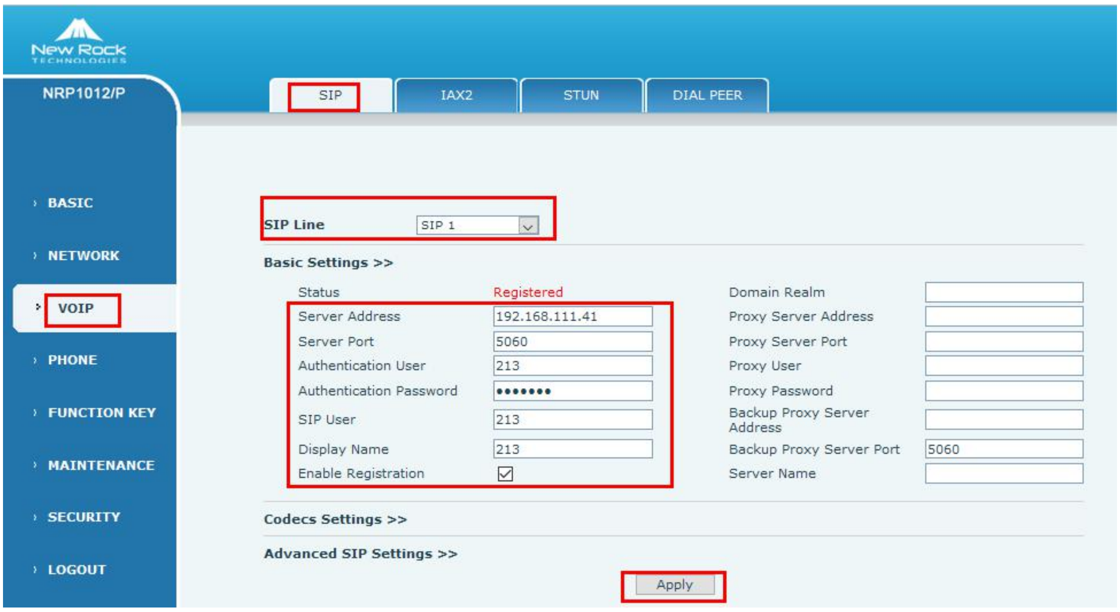
If **No response**: no response from registration server. The cause may be contributed to 1) incorrect address for the registration server; 2) IP network failure; or, 3) the registration server is not reachable.

4.3 Configuring the IP Phones

For New Rock NRP IP phone, the default configuration file is generated on the OM, and downloaded to the phones through plug and play procedure described above. Other than achieving special features, there is no need to configure the phone individually.

For a phone that is not a New Rock NRP phone, registration information must be entered. The following describes the registration information using the NRP1012 as an example.

Step 1 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**.



Note

Item	Description
Server Address	Enter the IP address or dynamic domain name of the OM. For Example : 192.168.111.41
Server port	Enter the SIP listening port of the OM. The default port is 5060. **Note:** By default, the SIP listening port of the device and the SIP trunk share a port, that is, port 5060. You can set a different registration port on the **Extension** > **IP**> **Registrar OPTIONS**.
Authentication User	Enter the number of the IP extension that is set in the OM. For example: 213.
Authentication Password	Enter the password corresponding to the number of the extension. For example: the password corresponding to the number 213 is 8240051.

Item	Description
SIP user	Enter the number of the IP extension that is set on the OM. For example: 213.
Display name	The name to be displayed on the other party's phone. The name of the extension user can be set. If it is not set, the **Authentication User** will be displayed on the other party's phone. For example: 213.

Step 2 Select **Enable registration**, and click **Apply**.

Step 3 On the web interface of the OM, go to **Extension** > **IP** to view the registration status of the IP extension.

Note

For an IP phone, it is recommended that G.729 codec standard be selected, and that the DTMF processing mode be the same as that on the device.

5 Configuring the Advance Features

5.1 Customizing a Simple Auto Attendant

5.1.1 *Customizing the Greeting*

As a simple auto attendant the OM20 plays a greeting message to the caller depending on if the call comes in at business hours or non-business hour. The system default greeting files are shown in table below, or your own greeting files can be made.

Type	File name	Content
Business hours	welcome	Thank you for calling. If you know your party's extension, please dial it now. Or, to transfer to an operator, press zero.
Non-business hours	Off-hour	Thank you for calling. Our office is closed. If you know the extension, please dial it now.

Creating greetings

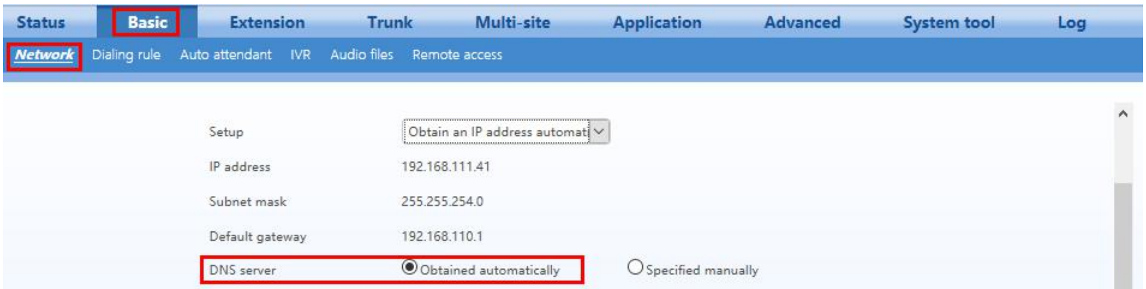
The following three methods can be selected:

- Text-to-greeting conversion
- Recording the greeting file on a phone
- Upload greetings prepared by other means

1) Text-to-greeting conversion

This is a simple way to customize the greetings with high voice quality, however limited in Chinese or English . The synthesizing service offered by New Rock Technologies, Inc. is powered by a speech-synthesis engine which is accessible on the Internet. To perform the synthesis, the device is required to be connected to the Internet. Follow this procedure:

Step 1 Go to **Basic** >**Network** page to configure DNS server.

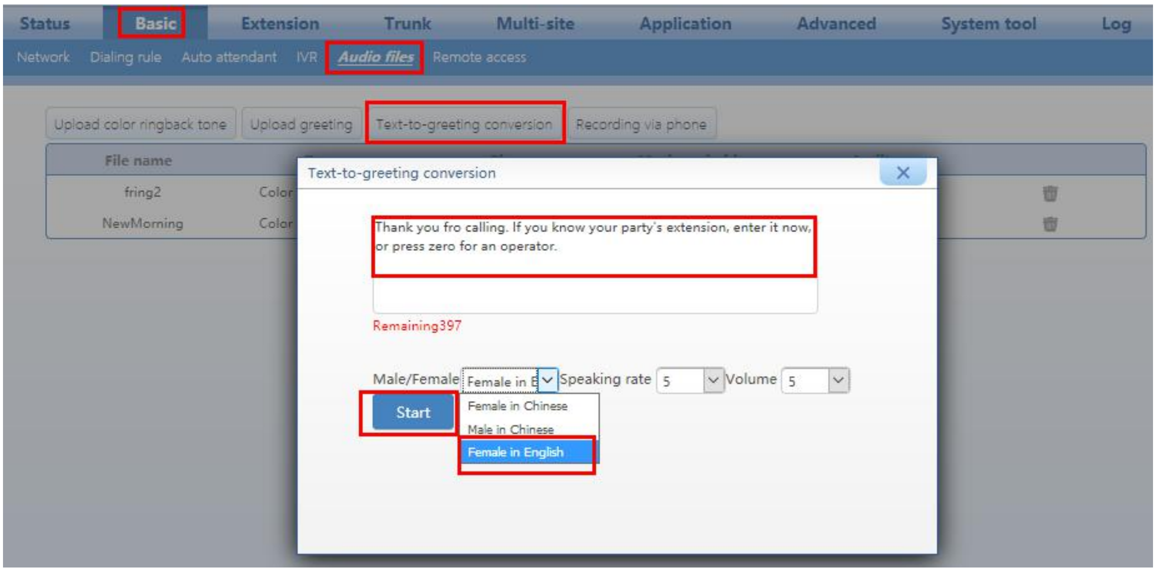


Note

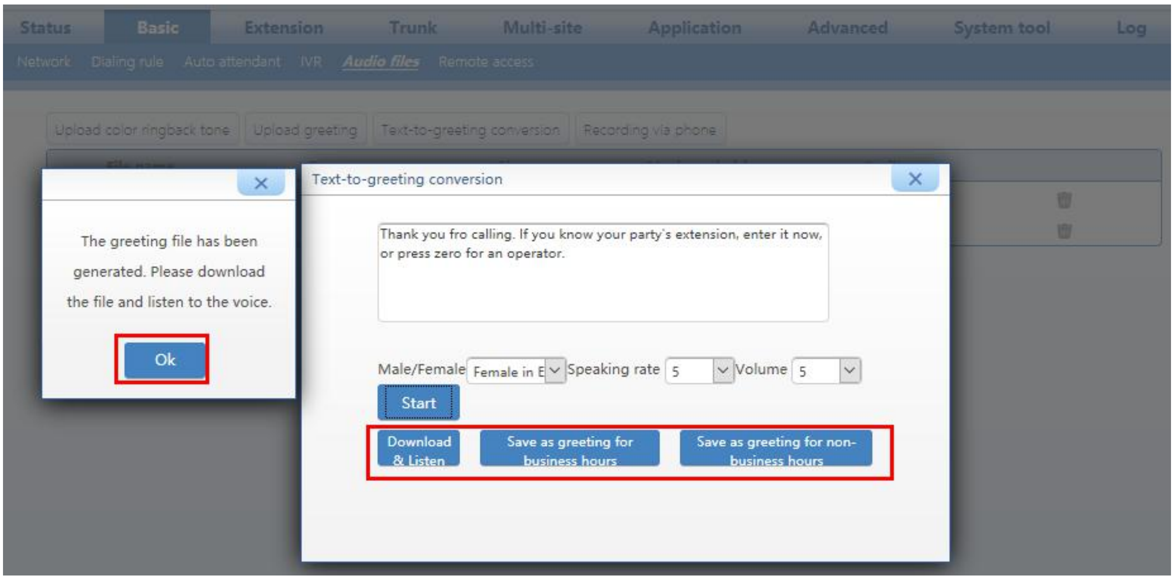
Obtained automatically can be selected only when the network connection mode is set to **DHCP** or **PPPoE**.

If **Specified manually** is selected, the network IP address of the **Primary and Secondary DNS server** must be entered

Step 2 Go to **Basic** > **Audio Files** > **Text-to-greeting conversion** page. Enter the greeting content in English and click **Start**.



Step 3 After synthesis, you can **Download & Listen** or save as a greeting for business hours or non-business hours.



Note

Please ensure that the device can access the Internet before starting the text-to-greeting conversion.

Step 4 You can also audit or delete the saved greetings for business hours or non-business hours.

2) Recording by phone

The greeting file can be recorded directly on an IP or analog phone that is connected to the device. To ensure high quality, it is recommended to make the recording in a quiet environment.

Item	Description
Recording	Pick up any phone connected to the device and press *81 to start the recording after the prompt, and hang up to finish the recording.
Listen	Press *8200 to listen to the voice recording
Save	• Press *8301 and hang up to replace the welcome file. • Press *8302 and hang up to replace the off-hours file.
Play the latest greeting file	• Press *8201 to listen to greetings; • Press *8202 to listen to off-hours greetings.

Item	Description
Recovery	Press *8300 to recover a replaced voice greeting file.

Note

Never restart your device during recording.

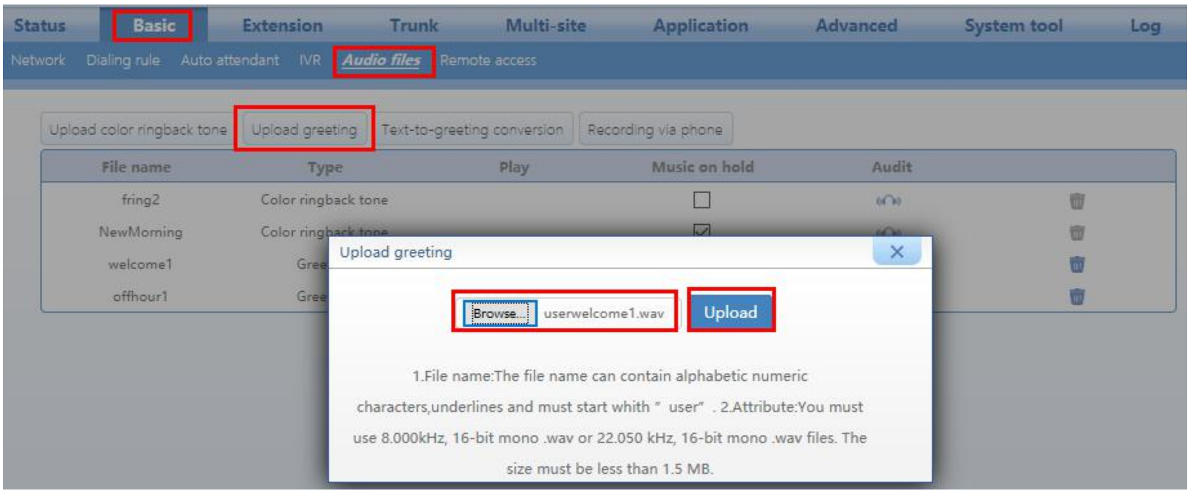
3) Upload greetings

Your customized greetings must be converted into an 8 kHz, 16-bit mono .wav file that can be used on device Telegreeting, an audio-file-conversion tool developed by New Rock. You can download the Telegreeting from http://en.newrocktech.com/show/592.html

Step 1 Go to **Basic** > **Audio files** > **Upload greeting**.

Step 2 Click **Browse** to select the audio files for uploading.

Step 3 Click **Upload**



Note

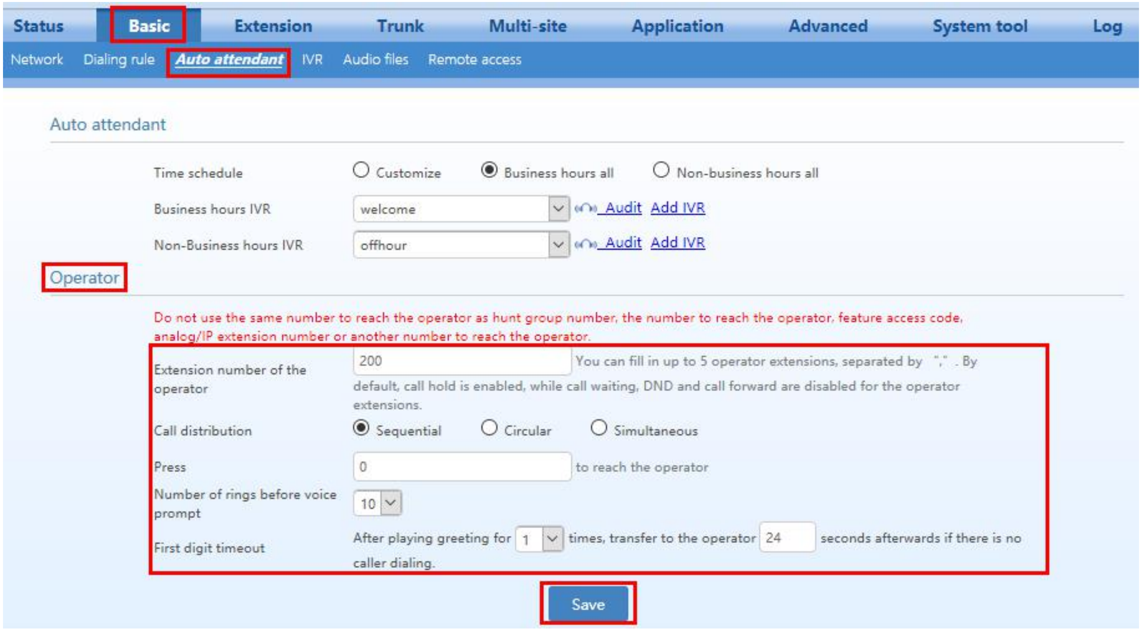
- The name of the audio file to be uploaded should begin with "user", and can contain letters, digits or underscores only. You must use a .wav file format.
- The sampling rate of the .wav file can only be 22.050 kHz or 8.000 kHz.

5.1.2 Assigning the Attendant Extensions

When the caller dials default-number 0, the call is transferred to an operator. By default, the first FXS port is reserved for the operator phone with the extension number 200. For devices without an FXS port (such as OM20-NA), no default operator is available.

But to add an operator or modify other related information, follow these procedures:

Step 1 Go to **Basic** > **Auto Attendant** to configure **Operator** settings.



Note

Item	Description
Operator	• **Extension number for the operator**: You can fill in up to five extension numbers, separated by a comma in this format: “,”. The default extension number is 200. Call waiting and DND are disabled by default. The call transfer function for the receptionist's extension will function only during non-business hours. **Note**: No default extension number for an operator exists on a non-FXS device (e.g. OM20-NA, OM50-8FXO). • **Call Distribution**: Select a call distribution scheme below when there is more than one operator: - **Sequential**: Terminate the incoming call to the first available extension on the operator list starting from the first one. - **Circular**: Terminate the incoming call to extension in Round-robin order; - **Simultaneous**: Terminate incoming calls to all available extensions on the operator list simultaneously and the first one to pick up is connected. • **Press (number) to reach the operator**: The number to reach the operator. The default value is 0. **Note:** If the default value is changed, you must modify related greetings, such as “To transfer to an operator, press zero”. • **No. of rings before voice prompt**: The device will play prompts when the number of rings reaches the value set here. The default value is 10. • **First digit timeout:** The device plays a greeting to incoming callers. The call will be transferred to the operator within the preset time after the greeting is played. After playing the greeting the first time, transfer it to the operator after 24 seconds if there is no caller dialing.

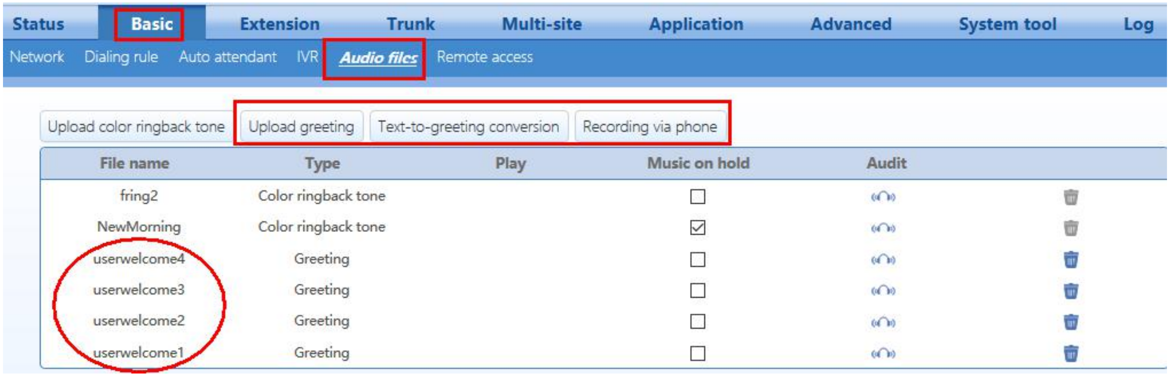
Step 2 Click **Save** to save the configuration.

5.2 Making an Auto Attendant with Multi-Level IVR

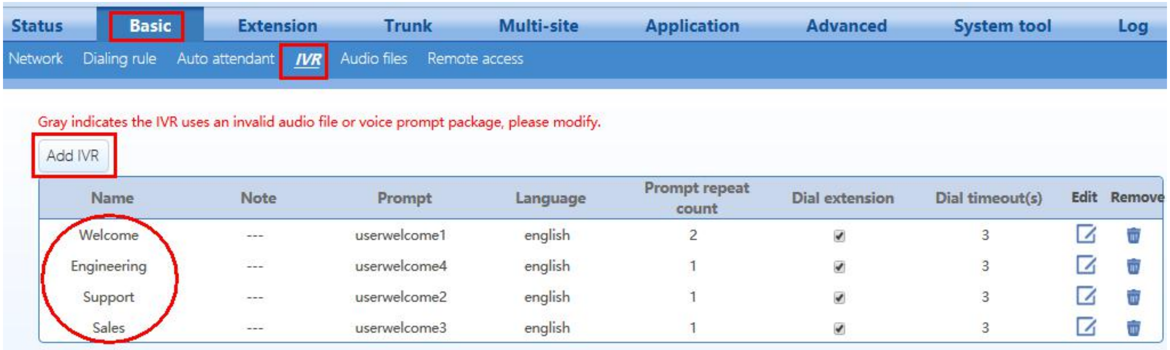
Step 1 Log in to **OM20** as **Admin** account

Step 1 Generate and upload IVR audio files on **Basic** > **Audio files** using any of the three methods: Upload greeting or text-to-greeting conversion or recording via phone.

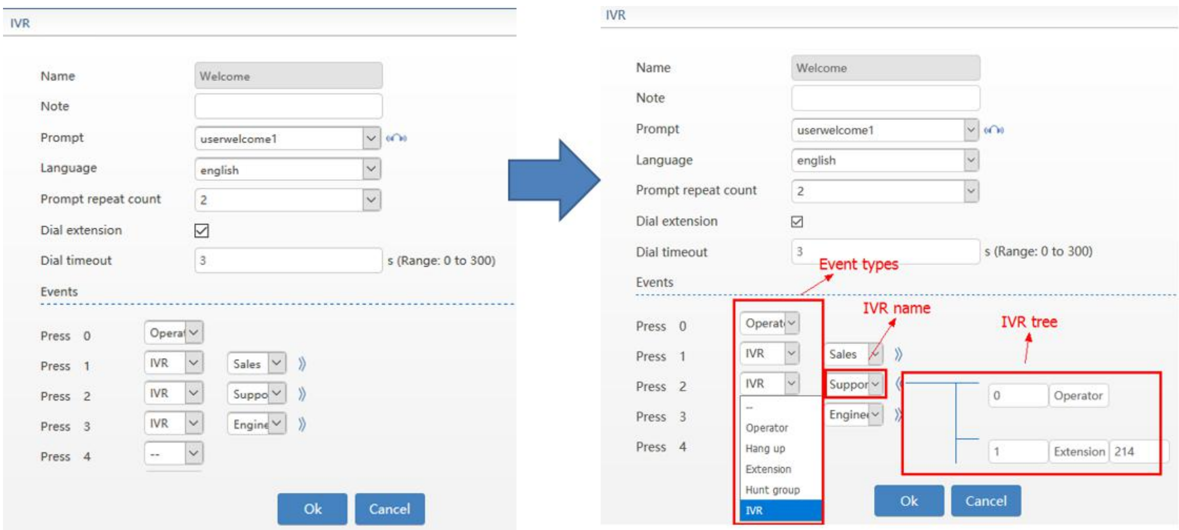
Step 2



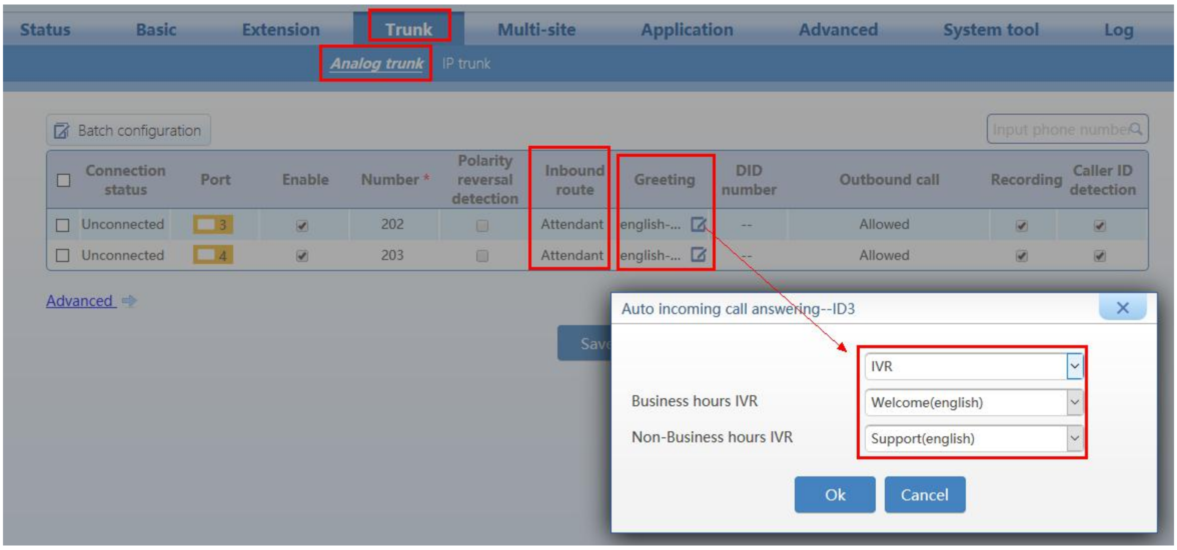
Step 3 Add and modify IVR settings on **Basic** > **IVR**



Step 4 Edit IVR template



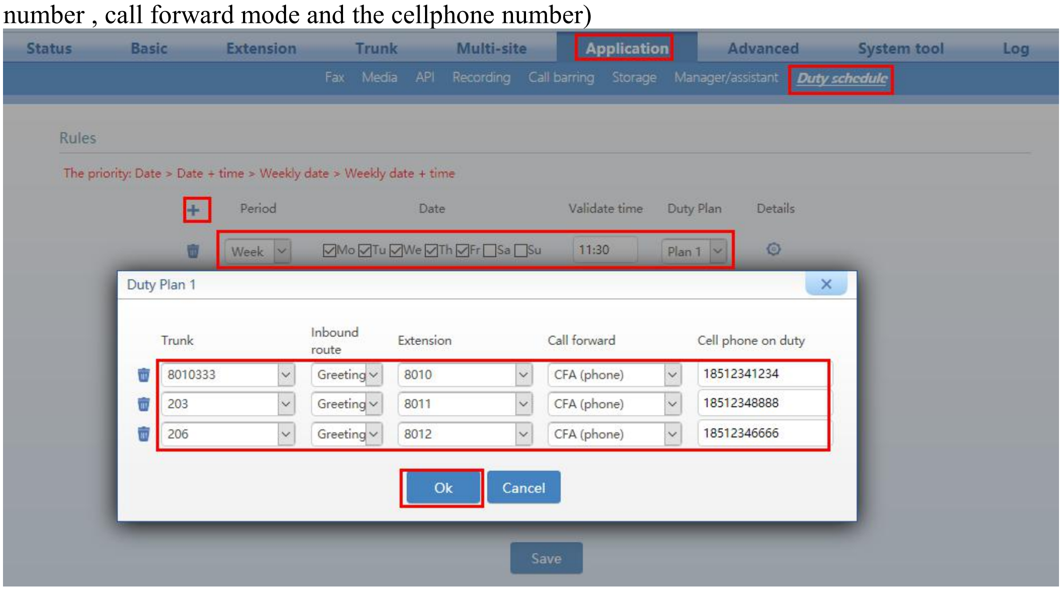
Step 5 On **Trunk > Analog trunk** or **IP trunk** page, set the inbound route to **Attendant** and set the greeting for the trunk to assign an **IVR for business hours and Non-business hours**.



5.3 Making an Auto Attendant with Scheduled Routing Rules

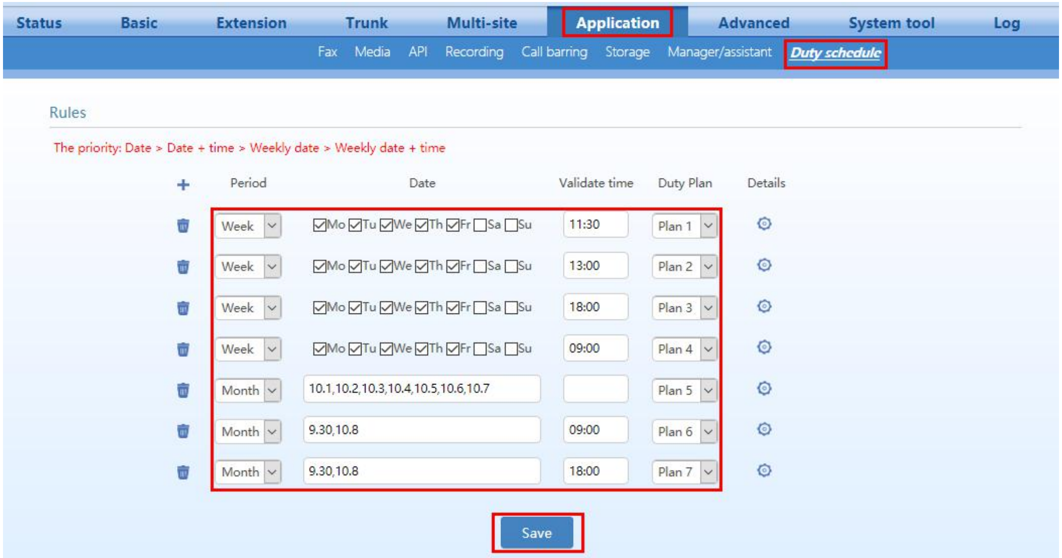
Step 1 Log in to **OM20** as **Admin** account

Step 2 On **Application > Duty Schedule**, select **week or month** for the **Period** , and select which the day of the week or fill in the specific day of the month for **Date**, input the **time** for the **Validate time,** and create the **Duty Plan** according to the given template (fill in the trunk number , inbound route, extension number , call forward mode and the cellphone number)



Step 3 Click **Save** to save the configuration.

Step 4 You can create the **Duty schedule** upto 30 plans.



Step 5 You need to reboot the device for the configuration to take effect.

5.4 Customizing the Music for Call Hold

Background music will be played for the party that is placed on hold. Two background music files are available by default: fring2 and NewMorning. You can customize and upload audio files.

CRBT audio files and background music audio files can be shared by the OM50/OM20.

After uploading an audio file, follow this procedure to set background music:

Step 1 Go to **Basic** > **IVR**, select the check box of the desired audio file below **Music on hold**, and then click **OK**.

To play the audio file, click .

To delete an audio file, click .

5.5 Selecting a Preferred Audio Codec

Step 1 Log in to **OM20** as **Admin** account

Step 2 Go to **Application** > **Media**, and set the preferred audio codec

Note

Item	Description
Codec	Support G729A/20, PCMU/20 and PCMA/20. Multiple codec types can be set separated by a comma, and in this case the device will sequentially negotiate a codec with the peer SIP device.

5.6 Making change to the Extension Numbers

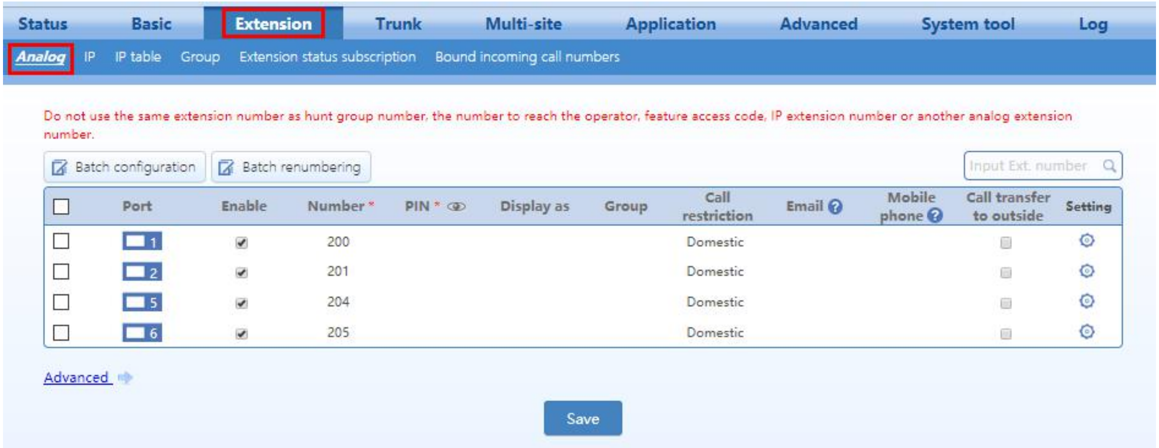
The OM supports analog and IP extensions, which are separately described below.

5.6.1 Analog extensions

Each FXS port corresponds to one analog extension. To configure an analog extension, follow this procedure:

Step 1 Log in to **OM20** as **Admin** account

Step 2 Go to **Extension** > **Analog** to configure analog extensions.



Note

Item	Description
Batch configuration	Configure extensions in batch mode. Batch mode allows the configuration of multiple parameters for multiple extensions at once.
Batch renumbering	Modify extension numbers in batch.
Port	Analog extension port (FXS port) on the device.
Enable	Select to use this line. By default, the line is enabled.
Number	Number of the analog extension. The allocated numbers start from 200 by default.
PIN	Used for verification when operating via *33 and *99 navigation. **Note:** • The caller can perform the operations according to the *33 and *99 menus on analog phone. See the *OM User Manual*. • Both **DISA** and **Authorization with PIN** use this PIN, which is also used to set automatic downloading by an IP phone.
Display as	Set the display name of the analog or IP extension. This feature is only limited to calls between extensions, and it requires that the name display feature be supported at the called terminal. If display names are configured on both the OM and the IP extension, the display name configured on the OM prevails.
Group	Select a department for the extension. The extensions within the same department can use group call pickup.
Call restriction	Each extension has an assigned privilege for making outbound calls. When a user makes a call beyond its restriction, the device rejects the call with a voice announcement. If extension A is allowed only to make internal calls, when it tries to make outgoing call, the following announcement is heard, “Sorry, you are not authorized to make outgoing call, please contact the administrator.” By default, the extension is allowed to make long distance calls. • **Internal:** Internal calls are allowed. • **Local:** Internal calls and local calls are allowed. • **Long distance:** Internal calls, local calls, and long distance calls are allowed. • **International:** Internal calls, local calls, long distance calls, and international calls are allowed. • **Prohibited:** The extension is only allowed to receive calls.
Email	Enter e-mail address to forward the call recording file or voicemail file to the user via email. For information on settings for voice mail, see 2.5.2 Voicemail.

Item	Description
Mobile phone	Instead of a PIN number a user's mobile phone number can be used for auto authentication of *33 and *99 for external access. If the **Authorization with *33 (Simplified-DISA)** function is selected, you can make outbound calls without dialing *33 for verification.
Call transfer to outside	An incoming call is allowed to be transferred to an external party. **Note 1**: Before using this function, the extension must have corresponding outbound rights. **Note 2:** During an outbound transfer, two lines are used.
Setting	Set multiple functions for the extension such as Authorization with PIN, Speed dialing, Call forking, Blocked numbers, Assistant, Block from being picked up, Call waiting, DND, Call hold, Call transfer, Call transfer to outside, and so on.
Advanced	Set advanced parameters of the analog extension.

Step 3 Click **Save** to save the configuration.

Step 4 Click **Advanced**, and set advanced properties of the extension such as Gain and Impedance. There is no need to make changes to the default values, unless there are issues when using the extension.

5.6.2 IP Extensions

The IP phone or SIP softphone registered to the OM successfully can be used as an IP extension.

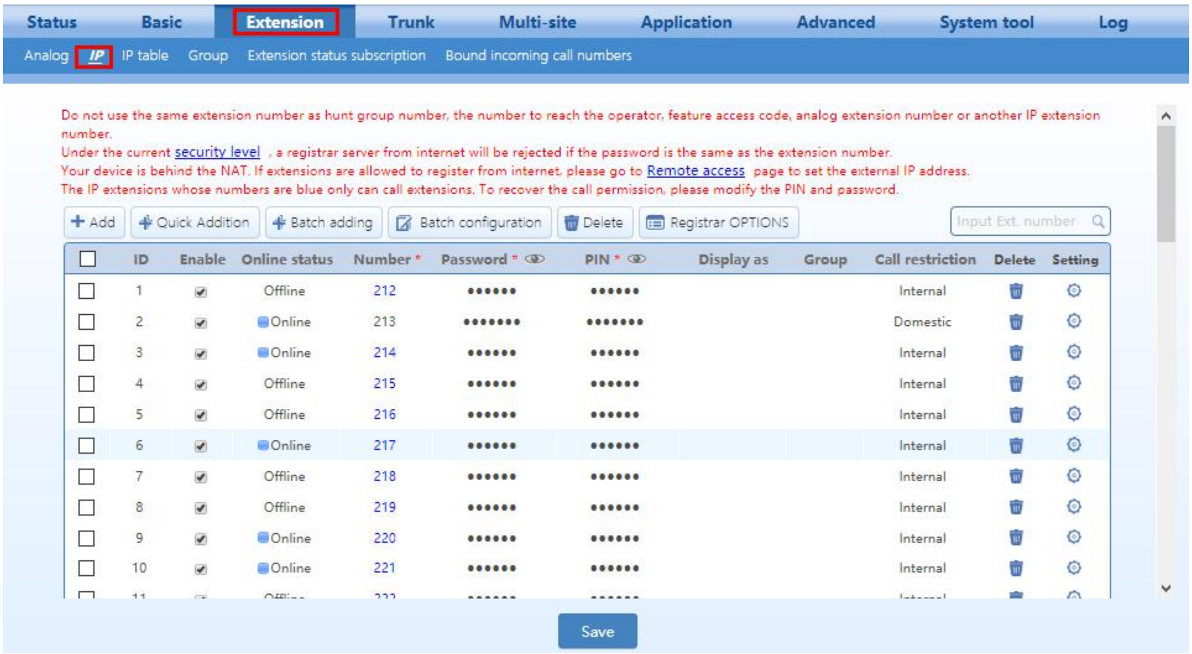
Before using an IP extension, you need to set the extension number and registration password on the OM. Follow this procedure:

Step 1 Go to **Extension** > **IP**.

Step 2 Click **Add**, and enter the IP extension number (for example, 208) and password (for example, 187986)

Note

- Do not use the same extension number as hunt group number, the number to reach the operator, feature access code, analog extension number or another IP extension number.
- The extension number and the password cannot be same.
- The IP extensions whose numbers are blue only can call extensions. To recover the call permission, please modify the PIN and password.



Note

Item	Description
Enable	Select to use this line. By default, the line is enabled.
Online status	Displays the current status of the IP extension.
Number	Phone number
Password	The password used for extension registration. The password is the same as PIN by default. After the password is changed, the PIN does not change.
PIN, **Group**, **display as**, **Call restriction**, **Email**, **Mobile phone**, **Call transfer to outside**, **Settings**	See the description on Analog extensions

Step 3 Click **Save** to save the configuration.

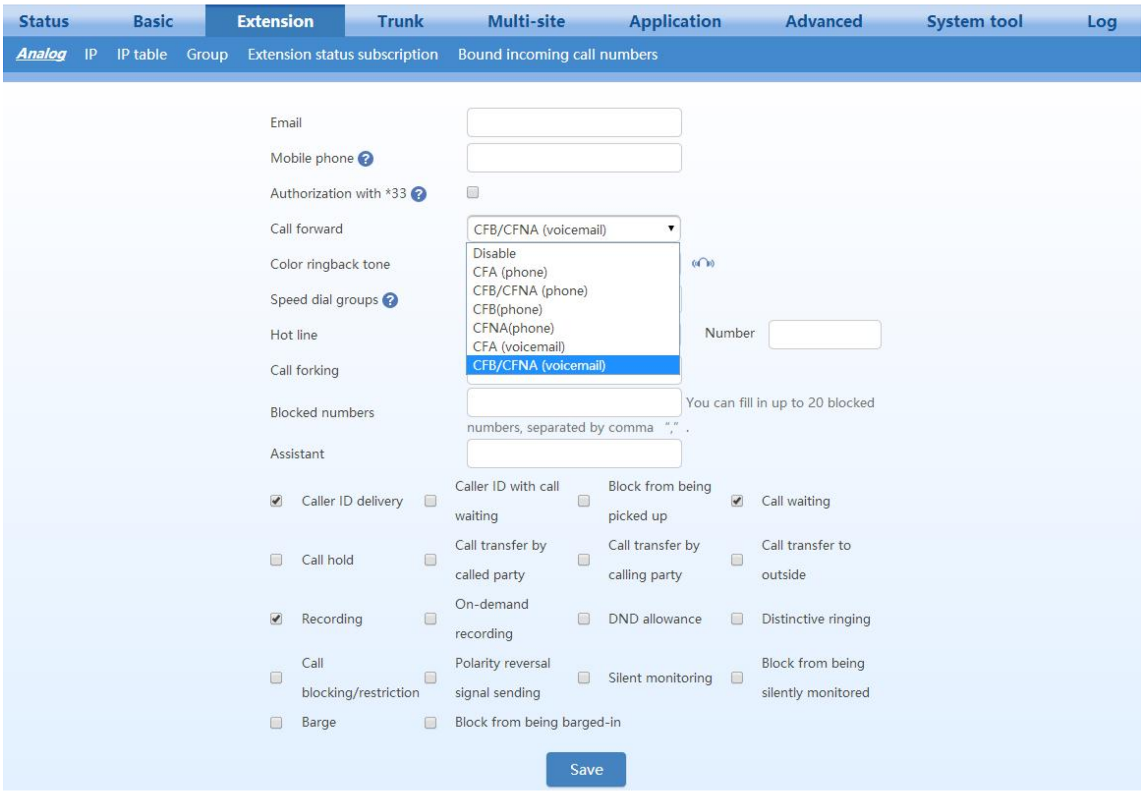
5.7 Activating the Voice Mail

When the extension user cannot receive the incoming call, the calling party can leave a message after the prompt.

Step 1 Go to **Application** > **Recording**, and enable the recording.

The storage path of recorded files is varied depending on the recording mode, for example, if **Internal USB flash drive** is selected, the voicemail messages will be stored in the internal USB flash drive.

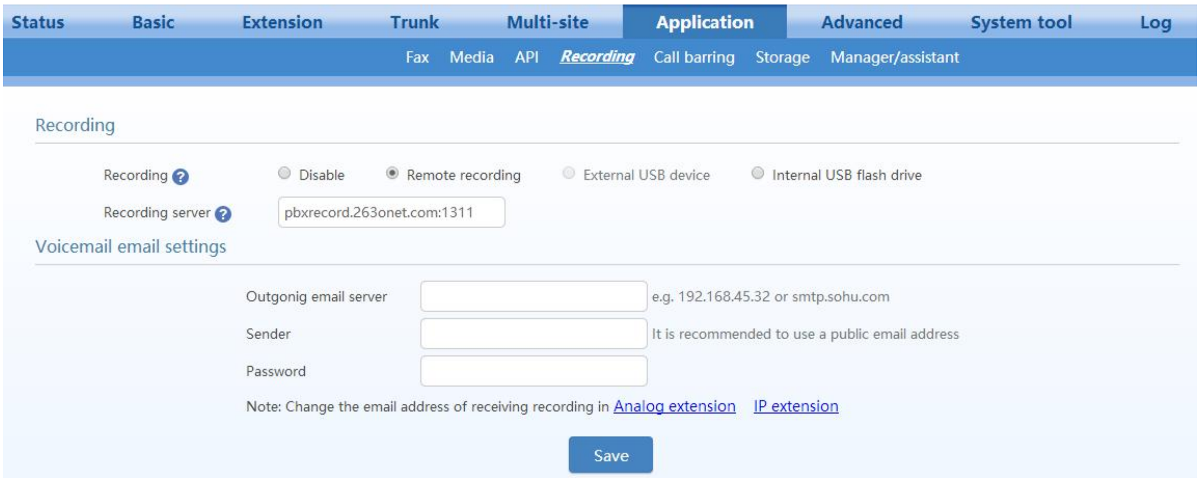
Step 2 Go to **Extension** > **Analog/IP** > **Setting**, and set the **Call Forward** to **CFA (voicemail)** or **CFB/CFNA (voicemail)**.



Step 3 Click **Save** to save the configuration.

When the recording mode is **Remote recording**, the voicemail messages can be sent to the mailbox of the extension user. To configure it, follow this procedure:

Step 1 Select **Remote recording** on the **Recording** page and configure the **Recording server** and **Outgoing email server**.



Note

Item	Description
Outgoing email server	Enter the IP address or domain name of the mail server. The device supports Sina mailbox and Sohu mailbox.
Sender	Enter the mailbox address of the sender.
Password	Enter the mailbox password of the sender.

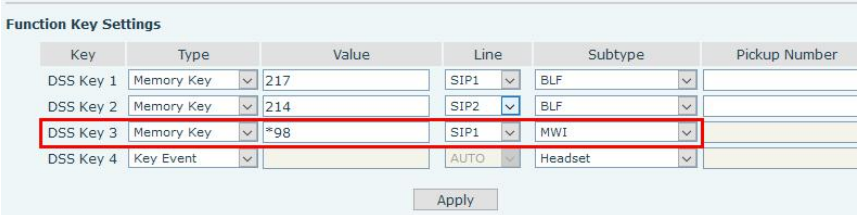
Step 2 Click **Save**.

Step 3 Go to **Extension** > **Analog/IP** > **Setting**, and configure **Email**. The mailbox will serve as the mailbox for receiving voicemail messages.

Step 4 Click **Save**.

Managing voicemail message

<table>
<tr><td>Listen to the voicemail message</td><td>

Analog phone: Press *98 and listen to the voicemail messages. You may hear:
“You have no voicemail messages”.
“You have n new/saved voicemail messages“
After the voicemail message is played, you may hear:
“To repeat the message, press one. To delete, press two. To listen to the next message, press three”.
IP phone: Configure the feature access code for listening to the voice messages by MWI function key and memory key on IP phone.


Listen from outside: you need to pre-configure to allow to use DISA

</td></tr>
<tr><td>Format</td><td>The name of a voicemail message file is in this format: vm_Called party-Calling party-Random code.pcm. For example: vm_200-6033432345-946685192.pcm. If the user presses Replay or Next when listening to a voicemail message or the voicemail message is played, the message will be identified with the file name become oldvm_200-6033432345-946685192.pcm.
Note: The filename extensions are varied depending on the codec of the voicemail message:

G.711μ: The file extension is .pcm.
G.711a: The file extension is .pcma.
G.729: The file extension is .dat.

</td></tr>
<tr><td>View the voicemail message</td><td>

Go to Application > Storage to get the access path and view the voicemail message files in the browser. Click builtin to view the files recorded through internal USB flash drive, and usb to view the files recorded through external USB device. The typical example of the storage path is: Recorder/voicemail.
If recorded files are stored in an external USB device, you can remove the USB device from the OM and connect it to your PC, then view the recorded files. The typical example of the storage path is: G:\Recorder\ voicemail.

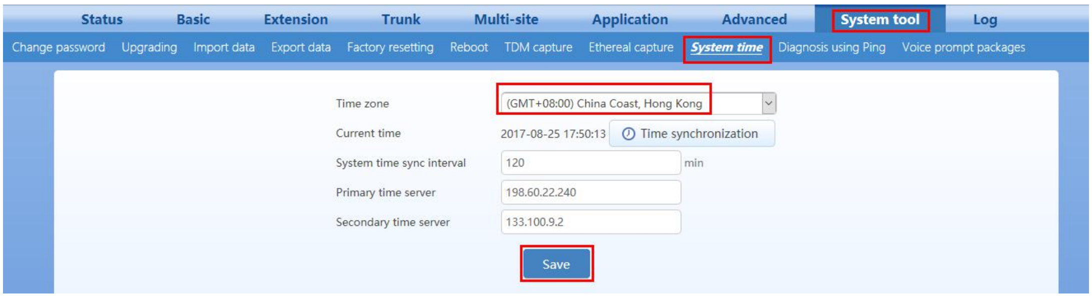
</td></tr>
</table>

6 Localizing the System

This chapter describes the configuration of parameters which you need to modify according to your regional preference, including time zone, country and area code, Caller ID detection mode, call progress tones and digit map. It is important to set up these parameters correctly before you start using the device.

6.1 Setting the System Time

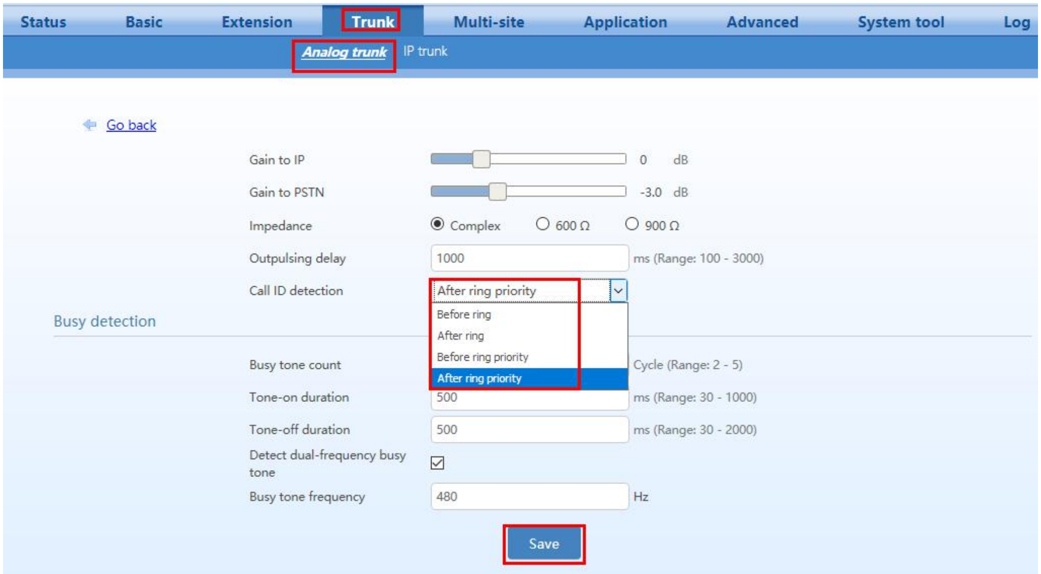
The time and time stamps are used in features and logs. The factory default time zone is GMT+08:00 hours. You can make the change at **System tool > System time**.



6.2 Selecting the Caller ID Detection Mode to PSTN

Select the Caller ID detection mode according to the features of the peer switch. The default value is **Automatic**, which means that the device detects Caller ID in either **after ringing** or **before ringing** mode automatically.

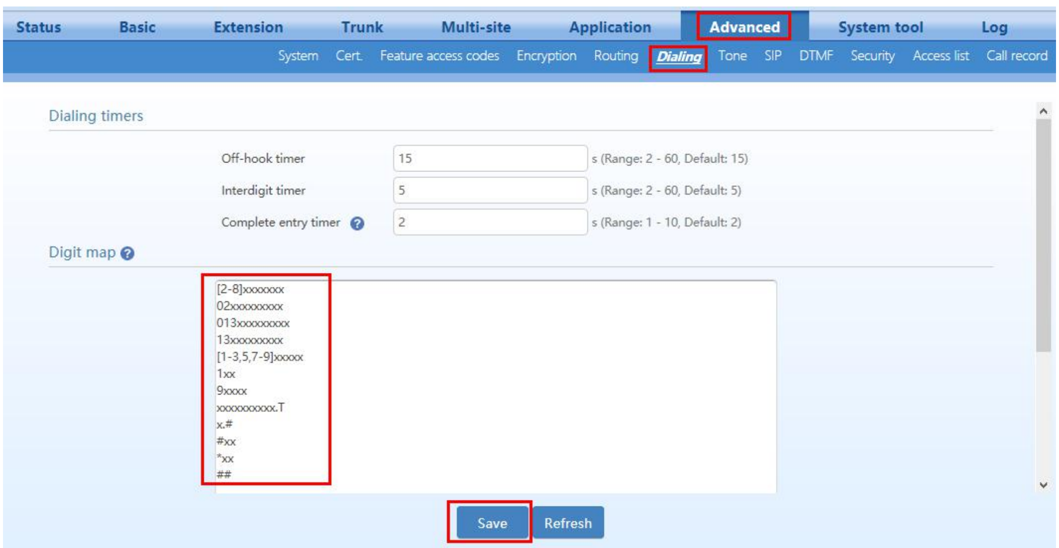
You can make the change at **Trunk > Analog trunk > Advanced > Call ID detection**



6.3 Making Changes to the Digit Map

The Digit map defines the dial plan of your device. Carefully setting up the rules in the digit map helps the device to recognize the ending of dialed numbers and thus speeds up the call process.

The default digit map only contains system function rules. If it does not fit your dial plan, please input the rules you want to the text box at **Advanced > Dialing > Digit map**.



An example of digit map associating with a dial plan is provided and explained below.

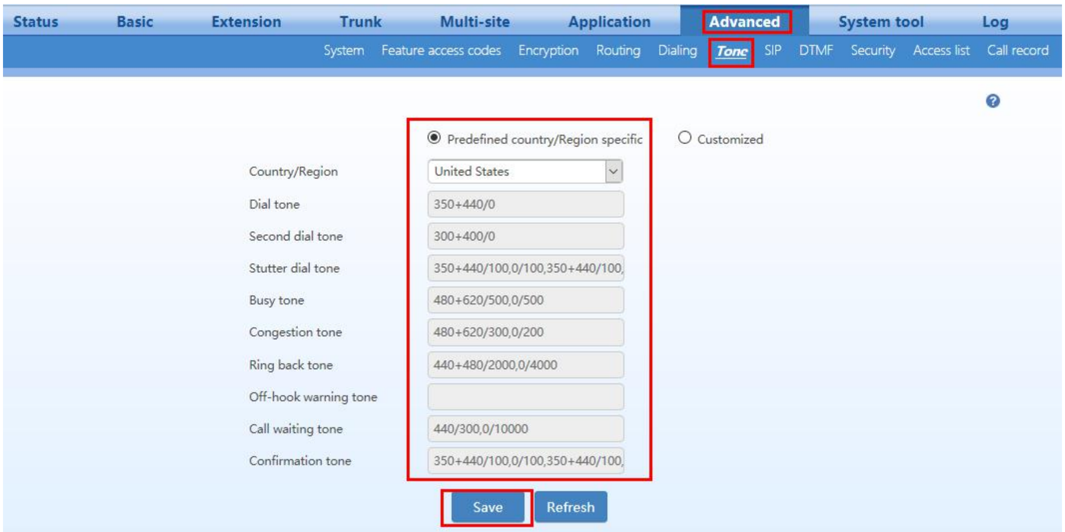
Item	Description
[2-8]xxxxxxx	The device terminates receiving digits after receiving eight digits starting with any digits between 2 and 8.
02xxxxxxxxx	The device terminates receiving digits after receiving 11 digits starting with 02.
013xxxxxxxxx	The device terminates receiving digits after receiving 12 digits starting with 013.
13xxxxxxxxx	The device terminates receiving digits after receiving 11 digits starting with 13.
[1-3,5,7-9]xxxxx	The device terminates receiving digits after receiving 6 digits starting with 1, 2, 3, 5, 7, 8 or 9.
1xx	The device terminates receiving digits after receiving three digits starting with 1.
9xxxx	The device terminates receiving digits after receiving five digits starting with 9.
xxxxxxxxxx.T	For a number with 10 digits, or less than 10 digits, the device terminates receiving digits and sends detected numbers if the duration of no dialing period exceeded the value of the **Interdigit timer** parameter. For a number with more than 10 digits, the device terminates receiving digits and sends detected numbers if the duration of no dialing period exceeded the value of the **Complete entry timer** parameter.
0-9, *, #	Matches a specific a DTMF digit.
x	Matches any single digit. For example, x can match 1 or 2 or 3...
.	Matches any string of DTMF digits. For example: “1.” can match any DTMF numbers starting with "1".
T	End of collecting DTMF digits after the timeout waiting for the next digit. For example, x.T means matching a string (a DTMF string) with any length. The ending is triggered by the timeout for waiting for the next digit.
x.#	If "#" is received after any digit is received, the device terminates receiving digits.

6.4 Selecting the impedance of FXO ports

The impedance setting of FXO port must match the expectation of your local PSTN. The factory default is **Complex**, and you can select **600 (Ohm)** or **900 (Ohm).** You can make the change at **Trunk > Analog trunk > Advanced > Impedance**.

6.5 Selecting the Call progress tone plan

The device generates the call progress tones according to the tone setting. There are tone plans of 27 countries or regions predefined in the device, and you can select one of them. Or, you can define the tone plan according to the national standard. You can make the change at **Advanced > Tone**.



7 Security

This document describes how to configure the device to protect from malicious users in the same public network. Safety and security are the primary concerns, when the device is deployed in the public network.

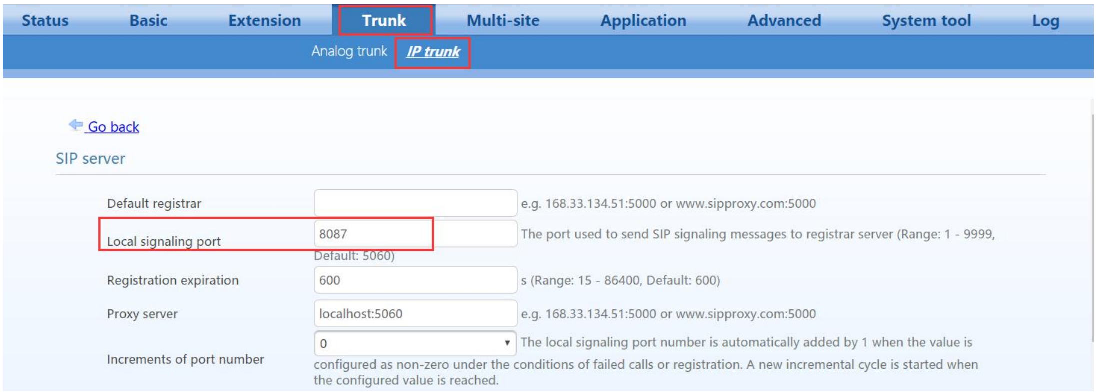
Malicious users are extremely dangerous because in most cases they are trying to steal your account information or even remotely gain access to your device and control it.

Most routers have some sort of firewall, but that's not enough to protect you from other malicious users, thus it is recommended to use the following security procedures in order to protect your device while using a public network.

7.1 Making Change to the Ports

Modify the default SIP port number (5060). You can set any number within the port range (1-9999)

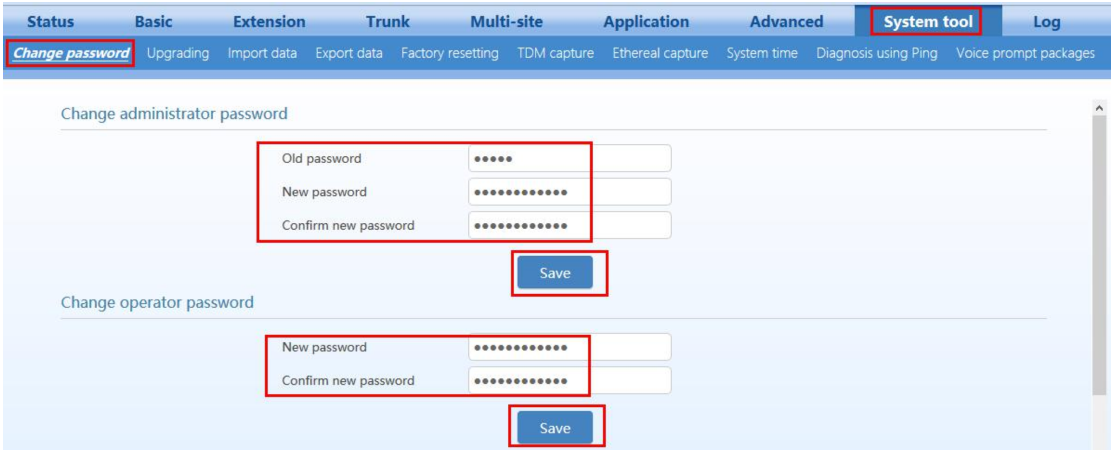
OM configuration page: go to **Trunk > IP trunk**



7.2 Making Change to the web GUI Password

Modify the default web interface password for both administrator and operator. It is recommended increasing the web password strength requirements such as the device MAC address which can be found on its printed label at the back of the device or any complex password combination.

OM configuration page: go to **System tool > Change password**



Example: MAC address is 000EA92D01B2 which can be assigned as your new administrator and operator password.

8 Multi-Sites Deployment

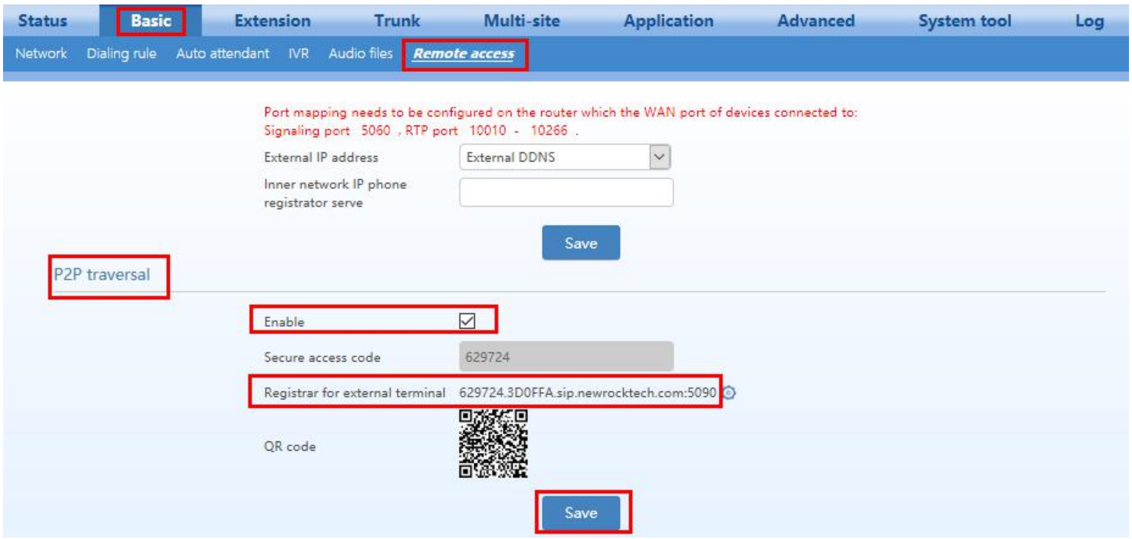
When the OM is located in the intranet (the private network), if you register with the system from an external IP extension or if multi-site networking is used, it is necessary to configure remote-address information and configure port mapping on the Internet ingress router or used the P2P traversal feature on OM. This enables devices on external networks to traverse NAT (Network Address Translation) to get access to the OM.

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.

Step 1 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.

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

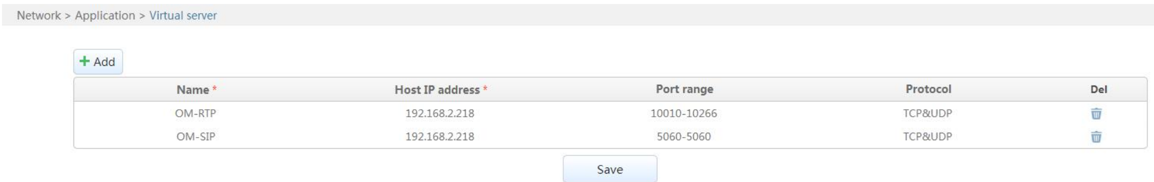
Step 1 Click **Basic** > **Remote access**, and set remote address.

Note

Item	Description
Build-in DDNS of the device	This option should be selected when the ingress router does not have a static IP address and nor DDNS support. The device will perform DDNS queries to determine its external IP address using the provided credentials. The domain name, user name, and password must be obtained from the DDNS service provider. The OM supports the following DDNS service providers: Dyndns.org, freedns.afraid.org, and www.no-ip.com.
External IP	This option should be selected when the ingress has a static IP address. In the **WAN IP address** field, enter the public IP address of the WAN port on the router.
External host name (DDNS on the router)	This option can be selected when the ingress of the external network does not have a static IP address. You need to enter the DDNS domain name of the WAN port on the ingress router.

Step 2 Click **Save**.

Step 3 Configure port mapping on the Internet ingress router. Take a New Rock WROC3000 as an example to show the ingress router configuration:



Note

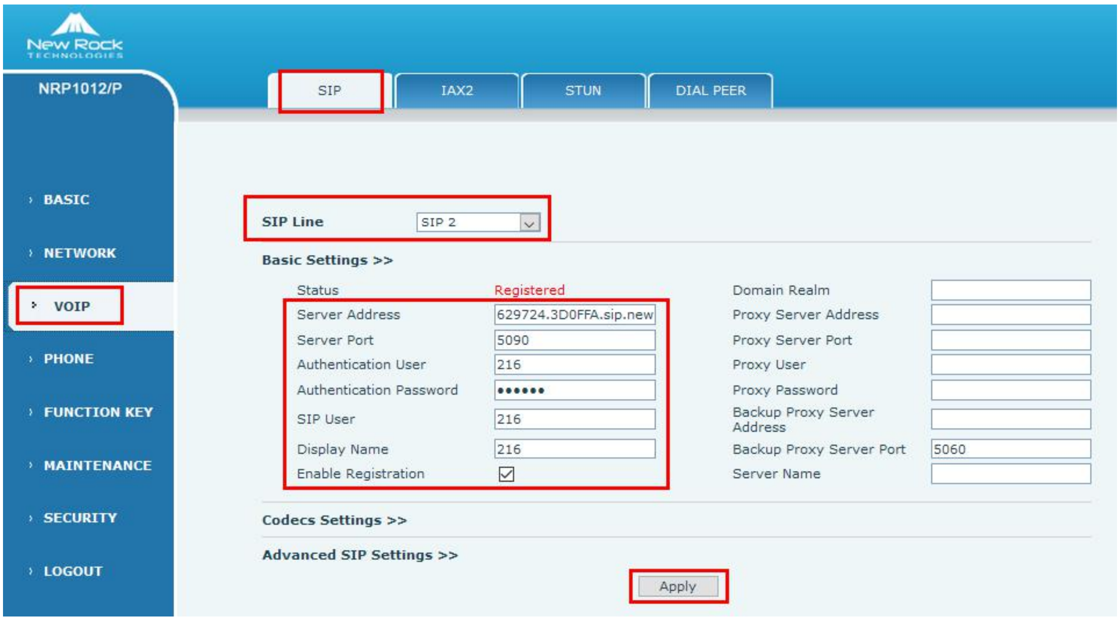
Item	Description
Host IP address	Enter the IP address of the OM. The current IP address of the device can be seen in the network part on the **Status** interface of the OM. Note: It is a must to set a static IP address which can be configured on **Basic** > **Network** page. For the OM80, it is the ETH1 IP address.
Port range	Enter the SIP signaling port and the RTP port range of the OM. You can go to **Trunk** >**IP trunk**> **Registrar OPTIONS** to view the SIP signaling port. You can view the maximum value and minimum value of the RTP port on the **Application**>**Media** interface. Note: Keep the mapping target port number the same as the port number of the OM.

Step 4 On the external IP extension, set the registrar address to the IP address or domain name configured on the **Basic** > **Remote Access** page.

8.1 Registering Remote IP Phones to the OM20

The following describes the registration information using Method 1: Remote Access with Peer-to-Peer communication across NAT and the NRP1012 as an example.

Step 1 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**.



Note

Item	Description
Server Address	Enter the External IP address or dynamic domain name of the OM. When the extension needs to register with the OM from an external network, the external access address of the device needs to be entered. Go to **Basic** > **Remote access** to view the **External IP address of the device** or the "**Registrar for External Terminal**" which is uniquely generated domain name for sip extension registration. For example: 629724.3D0FFA.sip.newrocktech.com:5090
Server port	Enter the SIP listening port of the OM. The default port is 5060. **Note:** By default, the SIP listening port of the device and the SIP trunk share a port, that is, port 5060. You can set a different registration port on the **Extension** > **IP**> **Registrar OPTIONS**.
Authentication User	Enter the number of the IP extension that is set in the OM. For example: 216.
Authentication Password	Enter the password corresponding to the number of the extension. For example: the password corresponding to the number 216 is 050026.
SIP user	Enter the number of the IP extension that is set on the OM. For example: 216.
Display name	The name to be displayed on the other party's phone. The name of the

Item	Description
	extension user can be set. If it is not set, the **Authentication User** will be displayed on the other party's phone. For example: 216.

Step 2 Select **Enable registration**, and click **Apply**.

Step 3 On the web interface of the OM, go to **Extension** > **IP** to view the registration status of the IP extension.

Note

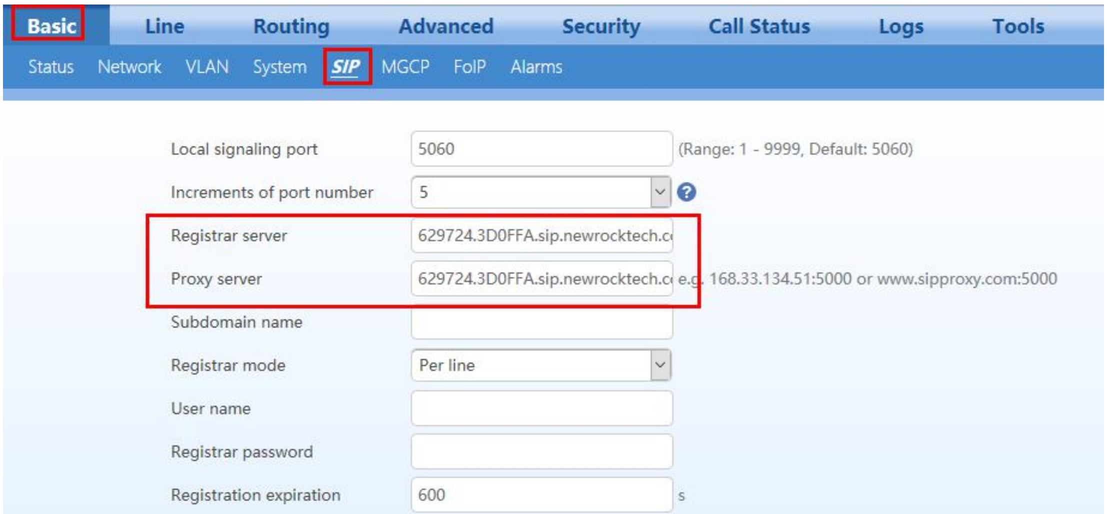
For an IP phone, it is recommended that G.729 codec standard be selected, and that the DTMF processing mode be the same as that on the device.

8.2 Registering the Remote ATA to the OM20

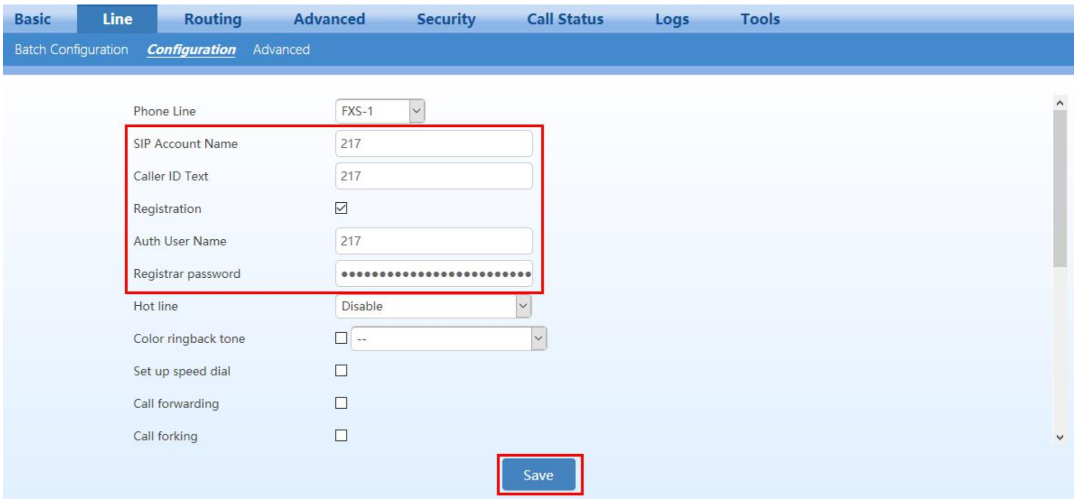
The following describes the registration information using Method 1: Remote Access with Peer-to-Peer communication across NAT and the ATA as an example.

Step 1 Log in to **ATA** as **Admin** account

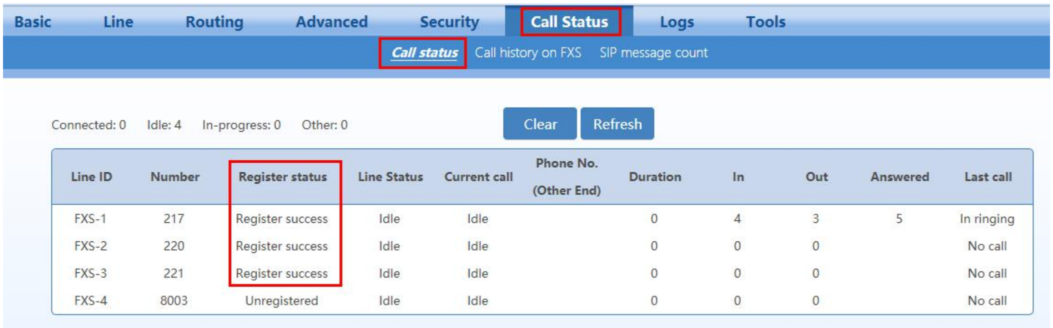
Step 2 On **Basic > SIP** page, fill in both **Registrar** and **Proxy server** fields using the **OM's** "**Registrar for External Terminal**" which is uniquely generated domain name for sip extension registration. For example: 629724.3D0FFA.sip.newrocktech.com:5090 and then click **Save**.



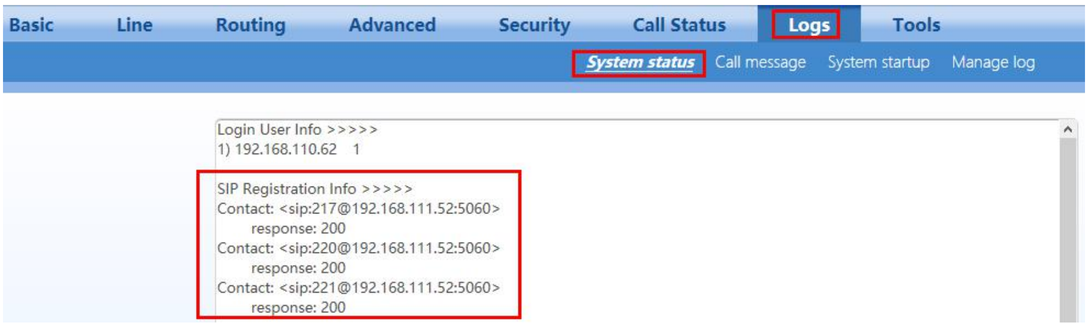
Step 3 On **Line > Configuration** page, fill in the **SIP Account Name**, **Call ID text, Auth User Name** and **Registrar password** fields using the OM's IP extension and password. Enable the **Registration** button and click **Save**.



Step 4 To verify your registration status is successful, go to **Call Status > Call Status > Register status**



Step 5 (Optional) Another way to verify your registration status is successful, go to **Logs > System status > SIP Registration Info ,** latest response should be **200**



Note

If **No response**: no response from registration server. The cause may be contributed to 1) incorrect address for the registration server; 2) IP network failure; or, 3) the registration server is not reachable.

9 Installation Guide and User Training

9.1 Installation Guide

This guideline is intended to help you to prepare and record information about the customer's network environment as well as the order and service information, before installing the OM20/50 IP Telephony System. By using this guideline, you can minimize the installation time and ensure that all setup requirements are met.

This guideline is designed to help OM20/50 IP Telephony System installation technicians and can be used as a training guide and checklist for Service Providers.

Customer Information:

Company Name:

Contact Name – Commercial:

Contact Phone Number –Commercial:

Contact Email Address – Commercial:

Alternate Contact Phone Number:

Contact Name – IT Responsible:

Contact Phone Number – IT Responsible:

Contact email address – IT Responsible:

Installation Location:

Site Survey Date :

Site Survey Method (circle one) : ON SITE / BY PHONE

Installation Schedule Date:

Service Provider Information:

Service Provider Name:

Service Provider Contact Information:

Service Order Number:

Service Activation Date:

Service Order Type:

Provisioning Information:

SIP Proxy:

User Name:

Password:

Service provider additional parameters (e.g. DID numbers):

Audio preferred codec (Circle one): G.711A G.711u G.729a G.726 G.723.1

Telephony System Survey:

Number of IP Phones to install:

Is it there an existing traditional PBX to replace (Circle one): YES / NO

If yes, please list the existing features provided by the system:

Is the customer setup requiring any of the following features? (Circle all that apply.)

Receptionist telephone

Automatic attendant

Direct Inward Dialing

Voice mail

Other (please specify):

Infrastructure Survey:

New Cable wiring required (Circle one):   YES / NO

If yes, how many and where?

AC Outlet available for each NewRock device location (Circle one.): YES / NO

If no, where are the missing locations?

PSTN Line (Circle one.): YES / NO

If yes, how many?

UPS backup (Circle one.):YES / NO

If yes, what devices are covered?

Fax Machine (Circle one.): YES / NO

If yes, is there a telephone cable available from OM20/50 to the fax machine? YES / NO

Broadband Type:

Broadband connection type (Circle one.): T1 / ADSL / xDSL / FTTH / Other

If other please specify:

IP addressing type (Circle one.) DYNAMIC / STATIC

If static, IP address:

If static, network mask:

Primary DNS:

Secondary DNS:

Bandwidth Uplink (kbps):

Bandwidth Downlink (kbps):

LAN:

Gateway LAN IP Address:

Network Mask:

DNS :

DHCP Server: YES / NO

NAT: YES / NO

QoS Switch: YES / NO

If yes, Type of QoS enforcement (Circle one.): IP TOS / VLAN ID / 802.1p

VLAN tagging (Circle one.) YES / NO

If yes, Voice VLAN ID:

If yes, Data VLAN ID:

Power over Ethernet: YES / NO

If yes, how many ports available:

Total number of ports on switch:

10 Troubleshooting

10.1 Troubleshooting Process

A troubleshooting process generally consists of the following stages:

- Collecting information
- Troubleshooting the fault
- Resolving the fault

10.1.1 Collecting Information

Fault information can be collected from:

- Feedback from the customer
- Alarms generated by the network management system
- Routine maintenance or inspection

Collecting raw information is of great importance during the initial troubleshooting stage. Raw information helps maintenance personnel narrow down the possible causes, ensuring a fast and accurate fault location. In case of faults, especially major faults, take caution that no action is performed until all necessary information is collected.

10.1.2 Troubleshooting the Fault

The cause of a fault is unique at each specific circumstance. Through analysis and comparisons, the exact cause of the fault can be identified. A fast and accurate fault diagnosis improves the troubleshooting efficiency, preventing the fault from further deterioration due to aimless troubleshooting. Locating the cause of fault is an important step in the process of technical troubleshooting, which provides guide for determining the means or measures for resolving the fault.

10.1.3 Resolving the Fault

After identifying causes of a fault, you can troubleshoot the fault accordingly.

Fault troubleshooting is a process of taking proper measures or steps (such as changing configurations or restarting the device) to rectify a fault and restore the system.

10.2 Troubleshooting Cases

10.2.1 IP terminals and ATA are unable to register successfully with the OM20/50

Fault Description

IP terminals (IP phone and softphone) and ATA cannot register to the OM20/50

Cause Analysis

- The network between the IP terminals (IP phone and softphone) and ATA, and OM20/50 fails.
- The registration server and signaling ports are improperly configured.
- The registration account name, user name and password are incorrectly set.
- There are special restrictions.

Troubleshooting and Solution

- 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, 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.

In addition, you can log in to the device and view the system status to fast locate the fault on SIP registration status:

- If it shows not enabled, then the registration server's address has not been entered.
- If it shows no response from the registration server, check whether you enter the correct address for the registrar server and the network connection between IP terminals (IP phone and softphone) and ATA , and OM20/50 fails.
- If it shows the code 403 in response, check whether the registration password is correct.
- If it shows the code 401 in response, check whether the registration account name and user name are correct.

10.2.2 Incoming call on FXO gateway is always disconnected

Fault Description

Incoming call on FXO gateway call will be automatically disconnected after 1 or 2 minutes of conversation.

Cause Analysis

- PSTN line connected to the FXO port of the device do not support "Polarity reverse signal"
- If PSTN don't provide polarity reverse signal but we enabled the feature "Polarity reversed signal detection"
- SIP server send BYE to FXO gateway to end the call.

Troubleshooting and Solution

- 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) disconnecte d
- If it has, set http://x.x.x.x/xml?method=gw.config.set&id215=no
- Check the log, if device receive ACK after 200OK message.
- Check the log to see if server send BYE to device to end the call.

10.3 Frequently Ask Questions

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 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 HX4E).
- 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 **System Tools > Factory resetting.**

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. **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. 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 150~200ms (default =75) and **Hook debouncing** to 75~100ms **(**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 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 **Advance > Line**, then increase the Ring Voltage value
- Go to **Advance > Line**, 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. 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 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

- 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

- Go to **Extension > Analog > Advanced,** increase (or decrease) the **Gain to IP** 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

10.4 Assistance in Fault Analysis

If the on-spot engineer is unable to troubleshoot the fault, New Rock can provide assistance in fault analysis. To ensure a quick and accurate troubleshooting, the on-spot engineer needs to provide the following information:

- Logs when the fault occurs (If the fault can be reproduced, logs with the level of 4 need to be captured. Follow the steps for downloading the log:
  1. Go to **Log > Log download**
  2. Select log level **4**
  3. Click **Save**
  4. Click **Clear** in **Log > Call message.**
  5. Do a test call with problem
  6. Click **Download** in **Log > Log download** to download

- Packets captured on the network when the fault occurs.
  Follow the step to get the **Ethereal Capture on OM20/50**
  1. Go to **System Tool > Ethereal Capture**
  2. Click **Start** to initiate the capture procedure.
  3. Make the test call.
  4. Click **Stop** to terminate the capture procedure. Then you will be notified for download.

- Configuration files from the OM20/50.
  Follow the step to export the **configuration file form the OM 20/50**
  1. Go to **System Tool > Configuration Maintenance**
  2. Click **Export data** to export the configuration file.

- Fault information and on-site environment, including the detailed fault information, network topology, and IP address of the devices used.

The preceding information can be sent to gs@newrocktech.com through e-mail for instant help.