New Rock Technologies, Inc.

OM20/OM50 Installation and Configuration Guide

http://www.newrocktech.com

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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.

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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.

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

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

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 <u>http://en.newrocktech.com/software/</u>

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).



- 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.



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Note	
Item	Description
	The Web utility provides two authority levels:
Role	• 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

status	Basic	Exte	ension	Trunk	Multi-site	Application	Advanced	System tor	nol log
rutus	Dusic	LAU		Info	Midlid Side	Application	X		, Log
his device ha		g for 1 minu	te 35 seconds. It s	tar Prod	uct name	OM20-4FXO		nge Password	
			1	Num	ber of extensions	0			
0	3		8	Num	ber of analog trunks	4			DONS
Y	7			Max	IP extension	20			
1AW	N	Alarm		Max	SIP trunk	24		lti-site	
192.168.	110.58	10		Soft	vare version	Rev 2.1.5.113.5		multi-site	
				Harc	ware version	Rev 6.0.0		pode	
				MAC	address	00:0E:A9:2D:02:EA			
				Curr	ent time	2017-08-25 16:35:58			
				Late	st reboot time	2017-08-25 16:34:25			
				Help		http://www.newrocktec	h.com		

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)

Ple	ase It	nput	PIN
PIN	44766	9_	
Delete	123		ОК

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)

4	
1	N.
N	oto

- 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**.

-		CH EN
New F	ROCK HX4E Vol	P Gateway
	Admin	~
	<u>∎</u> •••	
	jada 💦 🤌	Refresh
	Login	

Note

Item	Description
	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.
Role	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

HX4E Admin	18					Info <u>Reboot</u> <u>Lo</u>
Basic Line	e Routing	Advanced	Security Ca	Ill Status Logs	Tools	
		GCP Foll In	fo		×	
	For security, please <u>chan</u>	ge the def	Model	HX404E		
	Local signaling port	5060	Number of extensions	4		it.
	Host name	HX4E	Number of trunks	0		
	MAC address	00:0E	Software version	Rev 1.9.5.351		
	Model	HX40	Hardware version	Rev 6.0.0		
	Device address	192.1	Kernel version	Kernel 2.1.10 (F)		
	System up time	1 day	Firmware version	MX.P1.2.1.10.351.E0.05		
			MAC address	00:0E:A9:29:72:98		
			Current time	2017-08-25 17:28:53		
			Help	http://www.newrocktech.com	1	

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.

Status	Basic	Extension	Trur	k Multi-site	Application	Advanced	System tool	Log
Network	Dialing rule	Auto attendant IVR	Audio files	Remote access				
WAN	I	Host name 🕢		OM20				
		Setup		Static IP address	~			
		IP address		192.168.2.218				
		Subnet mask		255.255.0.0				
		Default gateway		192.168.2.1				
		Primary DNS server		192.168.2.1				
		Secondary DNS ser	ver					
PC								
		Mode		Switching port				
					Save			

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



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.

Status	Basic	Extension	Trunk	Multi-site	Ар	plication	Advanced	System tool	Log
		Analog	g trunk IP trun	<u>k</u>					
If the serve	er or sub-domain is r Duick Addition	not configured, the serve	r and sub-domair Batch configur	ation at the correspondence of the correspon	ponding c	onfiguration of s histrar OPTIONS	ip_trunk page.	Input	phone number
	A		g setter ter ingen				~		,
	Add							Outbound call	Recording Del
		Account	t type	SIP trunk OSk	kype				
		SIP serv	er 🕜 🛛 🔤	92.168.111.214:506	0				
		UA dom	nain name						
		Number	r [1	5502118215					
		Usernar	ne 1	5502118215				_	
		Passwor	rd 🕜			٩			
		Concurr	rent calls						
		Registra	ition 🔽]					
		Inbound	d route	Attendant	~				
		Greeting	9	Greeting	\sim				
			6						
			Ok	Cancel					

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.

Status	Bas	sic	Exten	sion	Trunk	Mul	ti-site	Appl	ication	A	dvance	d Systen	1 tool	Log	J
				Analog	g trunk <u>IP</u>	<u>trunk</u>									
11.4				1.1				P	6						
+ Add	rver or sub-d	Addition	A Batc	h adding	Batch con	figuration	e the corre	sponding con	rar OPTIC	ONS	c page.		Input ph	one numb	berQ
	Register status	Number	Concurre calls	ent Username	Password	SIP server	UA domain name	Registratio	Inbound route	Greeting	DID number	Outbound o	all F	Recording	g Del
1	Register s	155021	1	15502118	•••••	192.168	**		Attend	Greeti 🗹		Allowed			•

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.

-	Colback						
	GO Dack						
SIP s	erver						
	Default registrar	192.168.111.214:5060	e.g. 168	3.33.134.51:5000 or www.s	ipproxy.com:5000		
	Local signaling part	8090	The po	rt used to send SIP signali	ng messages to registra	ar server (Range: 1 - 9999,	
	Local signaling port	Default: 5060)					
	Registration expiration	600	s (Rang	e: 15 - 86400, Default: 60	D)		
	Proxy server	192.168.111.214:5060	e.g. 168	3.33.134.51:5000 or www.s	ipproxy.com:5000		
		0	✓ The loc	al signaling port number i	s automatically added b	by 1 when the value is	
	Increments of port number	configured as non-zero und	ler the condition	ns of failed calls or registr	ation. A new incrementa	al cycle is started when	
Regi	ster to multiple ITSPs	the conligued value is reac	neu.				
	Enable						
0.1							

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.

atus	Basic	E	ctension	Trunk	Mul	ti-site	Applicati	on	Advanced	System tool	Log
			<u>A</u>	nalog trunk	IP trunk						
🕼 Ba	atch configuratio	on								Input pho	ne numbeiQ
	Connection status	Port	Enable	Number *	Polarity reversal detection	Inbound route	Greeting	DID number	Outbound ca	I Recording	Caller ID detection
	Connected	3		202345678		Attendant	Greeting 🗹		Allowed	V	
U	Inconnected	4		203456789		Attendant	Greeting 🔽		Allowed		

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.

atus	Basic	E	ctension	Trunk	Mul	ti-site	Applicat	tion	Advanced	System tool	Log
			A	alog trunk	IP trunk						
🛛 Bat	tch configurati	on								Input pho	ne numbel©
	Connection status	Port	Enable	Number *	Polarity reversal detection	Inbound route	Greeting	DID number	Outbound ca	II Recording	Caller ID detection
_ c	Connected	3		202345678		Attendant	Greeting 🔽	3	Allowed		V
Un Un	nconnected	4		203456789		Attendant	Greeting 🔽	3	Allowed		
				alog trunk 👖	P trunk						
he server	er or sub-doma	in is not co lition	<mark>nfigured,the se</mark> Batch adding	erver and sub-d	lomain will use	e the correspo Delete	onding configu	ration of sip_ OPTIONS	trunk page.	Input p	hone num
	Register status Nur	mber Conc	urrent Userna IIs	me 🕜	d SIP server	UA domain Re name	egistration ro	ound oute Greet	DID Out	utbound call	Recordin
1 Reg	gister s 1550)21 1	1550211	3 ••••••.			☑ Atte	end Greeti	. 🗹	Allowed	
						Save					

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

Status	Basic	Extension	Trunk	Multi-site	Application	Advanced	System tool	Log
Network Dial	ing rule <u>A</u>	uto attendant IVR	Audio files Rei	mote access				
Auto at	tendant							
	Time so	hedule	Oustomize	O Business hours al	O Non-business	hours all		
			Sunday 🗸 to	Sunday v Open 0	7:00 🗸 16:00 🗸) +		
	Busine	ss hours IVR	welcome	× •∩•_	Audit Add IVR	1.1		
	Non-B	usiness hours IVR	offhour	~ (n)	Audit Add IVR			
Note Item		Descrip	otion					
		You can hours an	n set the ra re non-bus	nge of busine iness hours.	ess hours in a	week. The ho	ours outside of	fbusiness
Customi	ze	You can device v non-bus	n click 🛨 will play co siness hour	to divide one orresponding s.	e day into up t greetings acc	to three busin ording to the	ess-hour segn preset busine	ients. The ss hours o
Business all	s hou	Irs The dev	vice plays l	ousiness-hour	greetings at	any time.		
Non-bus hours all	iness	The dev	vice plays i	non-business-	hour greeting	s at any time		

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

- 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

atus	Basic	E	ctension	Trunk	Mu	ulti-site	Applicat	tion	Advanced	System tool	Log
			A	nalog trunk	IP trunk						
🖬 Bati	ch configurati	on								Input phor	ne numberQ
	onnection status	Port	Enable	Number *	Polarity reversal detection	Inbound route	Greeting	DID number	Outbound call	Recording	Caller ID detection
	onnected	3		202345678		DID	🛛	200 🗸	Share O DID or	ily 🗹	
🗌 Un	iconnected	4		203456789		Attendant	Greeting 🔽		Allowed		

Status	Basic	Exten	sion	Trunk	Multi-site	Арр	lication	,	Advanced	System tool	Log
				g trunk <u>IP tru</u>	<u>ink</u>						
If the serve	er or sub-domain is n	ot configu	red,the serve	r and sub-doma	in will use the corre	sponding co	nfiguration	of sip_tru	nk page.	Innut	nhono numboo
TAU		+ Datci		a bater cornig			suar of ne	5/45		Input	priorie numbera
	Register status Number	Concurre calls	Username	Password SI	P server domain name	Registratio	Inbound n route	Greeting	DID number	Outbound call	Recording Del
🗌 1 Re	gister s 155021	1	15502118	•••••			DID	19223	200	Share	0 🗑
					S	ave	4. ····· 20				



- 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.

us	Basic Extension	Trunk	Multi-site	Application	Advanced	System tool	
ork <u>Dialin</u>	Auto attendant IVR Au	dio files Remote					
Outbound	6						
	Do not use the same outbound number or another outbound n	prefix as hunt grou refix	p number, the num	ber to reach the operator	r, feature access code, a	nalog/IP extension	
	Automatic insertion of long dist	ance dialing prefix	e.g. 17909				
	Long distance call prefix 0	The out	ound calls dialed w	ith this prefix are identifi	ed as long distance call		
	Dialing method for Dialing outbound calls Outbou	without prefix Ind dialing with pre	O Intercom dialin	g with specified prefix *			
	Round-robin hunting for analo	Directly enter	er or click the icon to				
Hunt grou	p						
	Do not use the same hunt group another hunt group number.	p number as the nu	mber to reach the c	operator, feature access c	ode, analog/IP extensio	in number or	
	Ring the next available extensio	n in the group afte	r 15 seconds c	f ringing			
	Number Hunting		Extensions				
						-	
	+ 100 Simultaneous	~	200,201,204				

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.

tus	Basic	1	xtension	Trunk	Mu	lti-site	Applicat	tion	Advanced	Sys	tem tool	Log
			4	alog trunk	IP trunk							
🕼 Batc	h configurati	on									Input pho	ne numbe
	nnection status	Port	Enable	Number *	Polarity reversal detection	Inbound route	Greeting	DID number	Outbo	ound call	Recording	Caller I detectio
Co	onnected	3		202345678	8	DID		100(Num	Sh	iare		
Unc	onnected	4	4	203456789		Attendant	Greeting 🔽		Allo	owed	¥	
tus	Basic	E	xtension	Trunk	Mult	ti-site	Applicati	on a	Advanced	Syste	em tool	Log
tus	Basic	E	xtension Ana	Trunk alog trunk	Mult ² trunk	ti-site	Applicati	on a	Advanced	Syste	em tool	Log
tus he server	Basic or sub-doma	E in is not c	xtension Ana onfigured,the se	Trunk alog trunk rver and sub-do	Mult <u>P trunk</u> omain will use	ti-site	Application	on a	Advanced	Syste	em tool	Log
tus he server · Add	Basic or sub-doma ₽ Quick Add	E in is not c ition	xtension Ana onfigured,the se	Trunk alog trunk rver and sub-do Batch cor	Mult 2 trunk omain will use nfiguration	ti-site the correspon	Application Applic	tion of sip_tru	Advanced	Syste	em tool	Log ne numb
tus the server Add	Basic or sub-doma Quick Add egister tatus	in is not c ition 4 nber Con	xtension And onfigured,the se r Batch adding current Usernar alls	Trunk alog trunk rver and sub-do Batch cor ne Password @	Mult P trunk omain will use nfiguration	ti-site the correspon Delete UA domain Rey name	Application Adding configura Registrar O gistration rou	tion of sip_tru PTIONS	Advanced nk page. DID number	Syste	em tool	Log ne numbe

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

New Rock									
NRP1012/P	FUNCTION KEY	Y EXT KE	EY	SOFTKEY					
					Apply				
BASIC	Function Key Se	ttings						_	
	Key	Туре		Value	Line		Subtype		Pickup Number
and the second				15500110015		_	Factor 1		
> NETWORK	DSS Key 1	Memory Key	\sim	15502118215	SIP1	\sim	BLF	\sim	
NETWORK	DSS Key 1 DSS Key 2	Memory Key Memory Key	~	203345678	SIP1 SIP2	~	BLF	~	
 NETWORK VOIP 	DSS Key 1 DSS Key 2 DSS Key 3	Memory Key Memory Key None	~ ~	203345678	SIP1 · SIP2 ·	~	BLF None		
 NETWORK VOIP 	DSS Key 1 DSS Key 2 DSS Key 3 DSS Key 4	Memory Key Memory Key None None	> > > >	203345678	SIP1 · SIP2 · AUTO AUTO	× ×	BLF BLF None		
 NETWORK - VOIP PHONE 	DSS Key 1 DSS Key 2 DSS Key 3 DSS Key 4 DSS Key 5	Memory Key Memory Key None None None		203345678	SIP1 · SIP2 · AUTO AUTO AUTO	× × ×	BLF BLF None None None		
 NETWORK VOIP PHONE 	DSS Key 1 DSS Key 2 DSS Key 3 DSS Key 4 DSS Key 5 DSS Key 6	Memory Key Memory Key None None None None	> > > > >	203345678	AUTO AUTO AUTO AUTO	× × ×	BLF BLF None None None None		
VOIP PHONE EINCTION KEY	DSS Key 1 DSS Key 2 DSS Key 3 DSS Key 4 DSS Key 5 DSS Key 6 DSS Key 7	Memory Key Memory Key None None None None		203345678	AUTO AUTO AUTO AUTO AUTO AUTO		BLF BLF None None None None None		

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.

atus	Basic	Ex	tension	Trunk	Mu	lti-site	Applicati	on	Advanced Sys	tem tool	Log
			A	nalog trunk	IP trunk						
		_								_	
Bat	tch configurati	ion								Input phor	ne numbeiQ
	onnection status	Port	Enable	Number *	Polarity reversal detection	Inbound route	Greeting	DID number	Outbound call	Recording	Caller ID detection
C	Connected	3		202345678		Attendan 🖌	Greeting 🔽		Allowed O Pickup prohib	oit 🗹	
Un Un	connected	4		203456789		Attendant	Greeting 🔽	440	Allowed		1

Basic	Exter	sion	Trunk	Multi	i-site	Арр	lication	A	dvanced	System to	ool Log
		Analog	trunk <u>IP t</u>	runk							
or sub-domain is	not config	ured the serve	r and sub-dor	main will use t	the correst	ondina co	figuration o	f sip trur	k page.		
Quick Addition	💠 Bato	th adding	Batch conf	iguration	🗑 Delete	📼 Regis	trar OPTION	IS	in pager		nput phone number
gister Number atus	Concurre	ent Username	Password	SIP server	UA domain R name	Registratio	Inbound n route	ireeting	DID number	Outbound call	Recording De
ster s 15502118	1	1550211821	ZCGaDlbeV					7	200~		
	or sub-domain is Quick Addition gister atus	basic Excer or sub-domain is not config Quick Addition + Batc gister atus Concurre calls	Analog ar sub-domain is not configured,the serve Quick Addition + Batch adding [] gister atus Concurrent calls	Analog trunk IP the ar sub-domain is not configured,the server and sub-dou a Quick Addition + Batch adding Batch configured gister Number Concurrent calls Password	Analog trunk IP trunk	Analog trunk <u>IP trunk</u> Analog trunk <u>IP trunk</u> ar sub-domain is not configured,the server and sub-domain will use the corresp Quick Addition + Batch adding Batch configuration Delete gister Number Concurrent trunk Strees Password SIP server domain F name	Analog trunk <u>IP trunk</u> Analog trunk <u>IP trunk</u> ar sub-domain is not configured,the server and sub-domain will use the corresponding cor Quick Addition + Batch adding Batch configuration = Delete = Regis gister Number Concurrent calls Password SIP server domain Registratio name	Analog trunk <u>IP trunk</u> ar sub-domain is not configured,the server and sub-domain will use the corresponding configuration of Quick Addition + Batch adding Batch configuration = Delete Begistrar OPTION gister Number Concurrent calls Password SIP server domain Registration route of name	Analog trunk IP trunk IP trunk Analog trunk IP trunk IP trunk Analog trunk IP trunk IP t	Analog trunk IP trunk IP trunk Analog trunk IP trunk IP trunk Analog trunk IP trunk IP trunk IP trunk Analog trunk IP trunk IP trunk	Analog trunk <u>IP trunk</u> ar sub-domain is not configured,the server and sub-domain will use the corresponding configuration of sip_trunk page. Quick Addition + Batch adding Batch configuration Delete Registrar OPTIONS gister Number Concurrent atus Number Concurrent Calls Password SIP server domain Registration route DID name Outbound call

Note

Item	Description
	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.
Outbound call	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.

Status	Basic	Extension	Trunk	Multi-site	Application	Advanced	System tool	Log
		Network <u>Dia</u>	u <i>ling rule</i> Auto a	attendant Multiling	gual IVR IVR R	emote access		
International call	limitation							
		Prefix	Allowed call dura	tion per dav				
	+	00	hr.	min.				
Outbound								
	Auto	omatic insertion of lo	ong distance dialing	prefix e.g. 1	7909			
	Long	g distance call prefix	: 0 🔽 T	he outbound calls dia	led with this prefix a	are identified as long dis	tance call.	
	Diali	ing method	Dialing without prefi	x O Intercom d	lialing with specified	d prefix *		
	Lea	ast cost routing	Vutbound dialing w	th pretix he ITSP for long dista	ance call.			
		5						
Hunting group								
	Ring	g the next available e	extension in the gro	up after 15 seco	onds of ringing			
	Nun	mber Hunting		Extensions				
	+ 346	6 Sequential		212,216,220)			
				_				
				Sa	ave			

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 first available IP trunk on the IP trunk list starting from the first one.

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.
	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.
Ducfiy	Note:
Frenx	 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.

asic Line	Routing Adv	anced Security	Call Status	Logs	Tools
atus Network	VLAN System SIP MGCP				
	Setup	Static IP address	~		
	IP address	192 . 168 . 2	218		
	Subnet mask	255 . 255 . 0 .	0		
	Default gateway	192 . 168 . 2 .	1		
	Primary DNS server	192 . 168 . 2 .	1		
	Secondary DNS server				
STUN					
	STUN	O Enable	le		
		1			
			Save		

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



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.

Basic Line Routing Status Network VLAN System SIP	Advanced Secu MGCP FolP Alarms	urity	Call Status	Logs	Tools	
Local signaling port	5060		(Range: 1 - 9999, Defa	ult: 5060)		
Registrar server	192.168.111.41:5060					
Proxy server	192.168.111.41:5060		e.g. 168.33.134.51:500	0 or www.sippr	oxy.com:5000	
Subdomain name						
Registrar mode	Per line	~)			
User name						
Registrar password						
Registration expiration	600		s			
High availability						
Mode	Primary-Standby	~)			
Backup SIP proxy	192.168.11.10:5060					
Primary server heartbeat detection						
			Save			



Jane 2000 1						
Item	Description					
Registrar	The gateway supports three registration schemes:					
mode	• 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.

Basic	Line	Routing	Adva	nced	Security	Call Status	Logs	Tools		
Batch Configur	ration <u>C</u>	onfiguration								
	Pho	ne Line		FXS-1	~					^
	SIP	Account Name		217						
	Calle	er ID Text		217						
	Regi	istration								
	Auth	n User Name		217						
	Regi	istrar password		•••••	•••••	••••				
	Hot	line		Disable		~				
	Colo	or ringback tone				~				
	Set	up speed dial								
	Call	forwarding								
	Call	forking								•
					[Save				

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

Basic	Line	Rout	ting Advan	ced Se	ecurity	Call Status	Logs	Tools			
				Call st	atus Call hist	ory on FXS SIF	message count				
					_						
Con	nnected: 0	Idle: 4 In-	progress: 0 Other: ()		Clear Rel	fresh				
1	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
3	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

Basic	Line	Routing	Advanced	Security	Call Status	Logs	Tools		
				2	y stem status Call n	nessage Syst	tem startup	Manage log	
		Login User Info 1) 192.168.110	o >>>>>).62 1						^
		SIP Registratic Contact: <sip: response: Contact: <sip: response: Contact: <sip: contact: <sip:< td=""><td>on Info >>>>> 217@192.168.111.52:50 : 200 220@192.168.111.52:50 : 200 221@192.168.111.52:50 : 200</td><td>)60>)60>)60></td><td></td><td></td><td></td><td></td><td></td></sip:<></sip: </sip: </sip: 	on Info >>>>> 217@192.168.111.52:50 : 200 220@192.168.111.52:50 : 200 221@192.168.111.52:50 : 200)60>)60>)60>					

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.

	SIP IAX2	STUN	DIAL PEER	
SIP I	Line SIP 1	~		
Basi	c Settings >>			
	Status	Registered	Domain Realm	
E F	Server Address	192.168.111.41	Proxy Server Address	
	Server Port	5060	Proxy Server Port	
	Authentication User	213	Proxy User	
	Authentication Password	•••••	Proxy Password	
KEY	SIP User	213	Backup Proxy Server	
	Display Name	213	Backup Proxy Server Port	5060
			,,	1

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.

ltem	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.



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.

Туре	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.

Status	Basic	Extensio	n Trui	nk Multi-site	Application	Advanced	System tool	Log
Network	Dialing rule	Auto attendant I	VR Audio files	Remote access				
								0
		Setup		Obtain an IP address autom	at 🗸			
		IP address		192.168.111.41				
		Subnet mask		255.255.254.0				
		Default gatew	ay	192.168.110.1				
		DNS server		Obtained automatically	O Specified manu	ually		
								_

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.

Sta	tus Basic work Dialing rule Auto atte	Extension endant IVR <u>Auc</u>	Trunk <i>lio files</i> Remote	Multi-site	Application	Advanced	System tool	Log
	Upload color ringback tone	Upload greeting	Text-to-greeting	conversion Recor	rding via phone		_	
	File name	Text-to-	greeting conversi	on		×		
	fring2 NewMorning	Color	Thank you fro ca or press zero for	lling. If you know you an operator.	ır party's extension, enter	it now,	8	
			Remaining397					
		Ľ	Male/Female Fe	Imale in E Speakin Imale in Chinese ale in Chinese Imale in English	ig rate 5 Volum	ne 5 V		

Note

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

Status	Basic	Extension	Trunk	Multi-site	Application	Advanced	System tool	Log
Network B								
Upload		one Upload greet	ing Text-to-greet		arding via phone			
	Elle anne	X Tex	t-to-greeting conve	rsion		×		
Th gen the f	e greeting file ha erated. Please do ile and listen to t	as been ownload the voice.	Thank you fro or press zero f	calling. If you know yo for an operator.	ur party's extension, ente	r it now,	W	
	Ok		Male/Female Start	Female in E Speak	ng rate 5 Volu	me 5		
			Download & Listen	Save as greeting business hour	for Save as gre s busine	eting for non- ess 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.

us Basic	Extension	Trunk	Multi-site	Application	Advanced	System tool	Lo
ork Dialing rule A	Auto attendant IVR <u>A</u>	<i>idio files</i> Rem	iote access				
Upload color ringbac	k tone Upload greeting	Text-to-greet	ing conversion Re	cording via phone			
File name	Туре		Play	Music on hold	Audit		
fring2	Color ringback	tone			0 ∩ 0	Ū	
NewMorning	Color ringback	tone			000	0	
welcome1	Greeting		*8231		0 ∩ 0		
offhour1	Greeting		*8241		000	-	

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.

ltem	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.

ltem	Description						
Recovery	Press *8300 to recover a replaced voice greeting file.						
Note							
Never restart your	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

Upload color ringback tone	Upload greeting	Text-to-greeting conversion	Recording via phone			
File name	Туре	Play	Music on hold	Audit		
fring2	Color ringback to	one		000		
NewMorning	Color ringback to	ne		100		
welcome1	Gree	ad greeting		×		
offhour1	Gree	Browse	welcome1.wav Upload		Ű	
		1.File name:The file nar	ne can contain alphabetic num	eric		
	cl	naracters, underlines and must	start whith " user" . 2.Attribut	te:You must		
	us	e 8.000kHz, 16-bit mono .wav	or 22.050 kHz, 16-bit mono .w	av files. The		
		size must	be less than 1.5 MB.			

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.

us	Basic Extension	Trunk	Multi-site	Application	Advanced	System tool	
rk	Dialing rule <u>Auto attendant</u> IVR	Audio files Rem	ote access				
Juto	o attendant						
	Time schedule	O Customize	Business hours	s all O Non-busine	ss hours all		
	Business hours IVR	welcome	 ••••••••••••••••••••••••••••••••••••	Audit Add IVR			
	Non-Business hours IVR	offhour	~ •••	Audit Add IVR			
ne							
ope	erator						
ope	Do not use the same numbe	r to reach the operat	tor as hunt group num	ber, the number to reach	the operator, feature as	ccess code,	
spe	Do not use the same numbe analog/IP extension number	r to reach the operat or another number	tor as hunt group num to reach the operator.	ber, the number to reach	the operator, feature as	ccess code,	
ope	Do not use the same numbe analog/IP extension number Extension number of the	r to reach the operat or another number 200	tor as hunt group num to reach the operator. Yo	uber, the number to reach	the operator, feature ac	ccess code, d by "," . By	
ope	Do not use the same numbe analog/IP extension number Extension number of the operator	r to reach the operat or another number 200 default, call hold extensions	tor as hunt group num to reach the operator. Yo is enabled, while call	u can fill in up to 5 operat waiting, DND and call forv	the operator, feature ac tor extensions, separate ward are disabled for the	d by "," . By e operator	
Jpc	Do not use the same numbe analog/IP extension number Extension number of the operator Call distribution	to reach the operation another number for another number for a contract of the second	tor as hunt group hum to reach the operator. Yo is enabled, while call	uber, the number to reach u can fill in up to 5 operat waiting, DND and call forv Simultaneous	the operator, feature and tor extensions, separate ward are disabled for the	ccess code, d by "," . By e operator	
Jpc	Do not use the same numbe analog/IP extension number Extension number of the operator Call distribution Press	er to reach the operation or another number 200 default, call hold extensions. Sequential 0	tor as hunt group num to reach the operator. Yo is enabled, while call ' O Circular to	uber, the number to reach u can fill in up to 5 operat waiting, DND and call forv Simultaneous reach the operator	the operator, feature and tor extensions, separate ward are disabled for the	ccess code, d by "," . By e operator	
, pe	Do not use the same numbe analog/IP extension number Extension number of the operator Call distribution Press Number of rings before voic	er to reach the operation or another number of 200 default, call hold extensions. © Sequential 0 re 10 v	tor as hunt group num to reach the operator. Yo is enabled, while call O Circular to	uber, the number to reach u can fill in up to 5 operat waiting, DND and call forv Simultaneous reach the operator	the operator, feature ac tor extensions, separate ward are disabled for the	ccess code, d by "," . By e operator	
Spe	Do not use the same numbe analog/IP extension number Extension number of the operator Call distribution Press Number of rings before voic prompt	r to reach the operation of another number of another number of 200 default, call hold extensions.	tor as hunt group num to reach the operator. Yo is enabled, while call Circular to	uber, the number to reach u can fill in up to 5 operat waiting, DND and call forv Simultaneous reach the operator	the operator, feature ac tor extensions, separate ward are disabled for the	ccess code, d by "," . By e operator	
ope	Do not use the same numbe analog/IP extension number Extension number of the operator Call distribution Press Number of rings before voic prompt First digit timeout	er to reach the operation of another number of another number of 200 default, call hold extensions.	tor as hunt group num to reach the operator. Yo is enabled, while call O Circular to to	uber, the number to reach u can fill in up to 5 operat waiting, DND and call for Simultaneous reach the operator es, transfer to the operato	the operator, feature ac tor extensions, separate ward are disabled for the r 24 seconds after	ccess code, d by "," . By e operator erwards if there is no	

Note

ltem	Description
	• 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.
Operator	- Circular : Terminate the incoming call to extension in Round-robin order;
- F	- 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

Sta	tus Basic	Extension	Trunk Multi-	site Application	n Advanced	System tool	Log
	vork Dialing rule Auto atte	endant IVR <u>Aud</u>	lio files Remote access				
	Upload color ringback tone	Upload greeting	Text-to-greeting conversio	n Recording via phone			
	File name	Туре	Play	Music on ho	ld Audit		
	fring2	Color ringback to	ne		(C))	Ū	
	NewMorning	Color ringback to	ne	\checkmark	(())	•	
	userwelcome4	Greeting			(C))		
	userwelcome3	Greeting			(())	1	
	userwelcome2	Greeting			(i))		
	userwelcome1	Greeting			(())	1	

Step 3 Add and modify IVR settings on Basic > IVR

Status	Basic	Extension	Trunk	Multi-site	Application	Advanced	System tool		Log
	Dialing rule	Auto attendant <u>IVI</u>	Audio files Remot	e access					
Gray	indicates the IV	R uses an invalid audio	file or voice prompt pac	kage, please modify.					
Add	IVR								
	Name	Note	Prompt	Language	Prompt repeat count	Dial extension	Dial timeout(s)	Edit	Remove
1	Welcome		userwelcome1	english	2		3		ŵ
1	Engineering		userwelcome4	english	1		3		ŵ
	Support		userwelcome2	english	1		3		Ū
	Sales		userwelcome3	english	1	2	3		



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.

tus	Basic	E	xtension	Trunk	Mul	ti-site	Applicati	ion	Advanced	System tool	Log
	Batch configurat	ion		alog trunk							
	Connection status	Port	Enable	Number *	Polarity reversal detection	Inbound route	Greeting	DID number	Outbound call	Recording	Caller ID detection
	Unconnected	3	e	202		Attendant	english 🔀		Allowed		
	Unconnected	4	Ø	203		Attendant	english 🗹		Allowed	2	
<u>Adva</u>	ncea. 🗇					Savo	Auto incomin	ig call answ	eringID3		×
							Business ho	urs IVR	Welcome(e	nglish)	~
							Non-Busine	ss hours <mark>IV</mark> I	Support(en	glish)	~
									Ok Cance	el	

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)

Status	Bas	ic	Extension	Trun	k M	ulti-site	Applicat	ion	Advanced	System tool	Log
									inager/assistant	/ schedule	
Rules											
The prior	ity: Dat	e > Date +	time > Weekly	date > Weekly	date + time						
		+	Period		Date		Validate time	Duty	Plan Details		
		-	March No.		Calify Calify Calify	-	11-30	Dise	1 0		
		U I	vveek ×				11.50	Pidfi			
	Duty	Plan 1							2		
		Trunk		Inbound	Extension		Call forward		Cell phone on duty		
		8010333	~	Greeting ~	8010	~	CFA (phone)	~	18512341234		
	Ŵ	203	~	Greeting ~	8011	~	CFA (phone)	~	18512348888		
	Ŵ	206	~	Greeting ~	8012	~	CFA (phone)	~	18512346666		
			and the second se		-	-	_			·	
					Ok	Cance	L .				
		_	_	_		-	_	-		_	
						Si	ave				
						Si	ave				

- Step 3 Click Save to save the configuration.
- Step 4 You can create the Duty schedule upto 30 plans.

Status	Basic	Extension	Trunk	Multi-site	Applicati	on A	dvanced	System tool	Log
			Fax Media AF	Pl Recording Cal	I barring Storag	e Manager/a	assistant Duty	<u>schedule</u>	
Rules									
The prio	rity: Date > Date ·	+ time > Weekly	date > Weekly date + 1	time					
	+	Period	Da	ite	Validate time	Duty Plan	Details		
	ŧ	Week 🗸	Mo Mu We L]Th ØFr □Sa □Su	11:30	Plan 1 🖌	ø		
	Û	Week 🗸	Mo Mu Me	Th 🖉 Fr 🗌 Sa 🗌 Su	13:00	Plan 2 🗸	ø		
	Û	Week 🗸	Mo VI u VWe V]Th ØFr Sa Su	18:00	Plan 3 🗸	0		
	Ū	Week 🗸	Mo Mu Me	Th 🖉 Fr 🗌 Sa 🗌 Su	09:00	Plan 4 🗸	ø		
	T	Month ~	10.1,10.2,10.3,10.4,1	0.5,10.6,10.7		Plan 5 🗸	ø		
	ŧ	Month ~	9.30,10.8		09:00	Plan 6 🗸	0		
		Month	9.30,10.8		18:00	Plan 7 🗸	0		
				Sa	ve				
				Sa	ve				

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 (1).

To delete an audio file, click $\overline{\mathbf{m}}$.

Status	Basic	Extension	Trunk I	Multi-site	Application	Advanced	System tool	Log
Upload	color ringback tone	Upload greeting	Text-to-greeting con	version Rec	ording via phone			
	File name	Туре	Pl	ay	Music on hold	d Audit		
us	erfring1687size	CRBT				(())		
	user_offhour7	Greeting				(())	Ŵ	
	fring1687	CRBT				(())		
	user_offhour3	Greeting				(())	Ŵ	
	user_offhour5	Greeting				(())		
	fring2	CRBT				((∩ 0)	1	
	NewMorning	CRBT				(())		

5.5 Selecting a Preferred Audio Codec

Step 1	Log in	to OM20 a	as Admin	account
~~~ ~				

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

Status	Basic	Extension	Trunk	Mu	lti-site	Applicatio	on	Advanced	System tool	Log				
			Fax	<u>Media</u> Al	Pl Recordir	ig Call barring	Storage	Manager/assist	ant					
	Codeo		G.711U/2	20, G.711A/20	0, G.729A/2	.729A/20,G.711U/	/20,G. <mark>711</mark> A	/20						
	RTP p	ort min.	10010		(1	(Range: 3000 - 65535)								
	RTP p	ort max.	10266		(1	(Range: 3020 - 65535)								
	TOS/I	DSCP	0x0C		0									
	Min. j	itter buffer	2			frame (Range: 0 - 30, Default: 3). Higher value results in long delay.								
	Max.]	itter buffer	50		fi	rame (Range: 10 - 2	250, Defaul	t: 50)						
	RTP d	rop SID												
	Obtai	n Media Address From	SDP GI	obal Address		SDP Media	Address							
					Sau									
					Sav	5								
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.

99⊈ ⊮	IP table Gro	up Extensio	on status subscrip	otion Bound	incoming call nur	mbers					
lo not u: umber.	se the same exten	sion number	as hunt group n	umber, the num	ber to reach the	operator, fea	ture access code,	, IP extension nu	imber or anoth	er analog extens	ion
🖌 Bato	h configuration	🔀 Batch r	enumbering							Input Ext. nu	mber (
	Port	Enable	Number *	PIN * @	Display as	Group	Call restriction	Email 🕜	Mobile phone 🕜	Call transfer to outside	Setting
	1		200				Domestic				0
	2		201				Domestic				0
	5		204				Domestic				0
	6		205				Domestic				O



ltem	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	<ul> <li>Used for verification when operating via *33 and *99 navigation.</li> <li>Note:</li> <li>The caller can perform the operations according to the *33 and *99 menus on analog phone. See the <u>OM User Manual</u>.</li> <li>Both <b>DISA</b> and <b>Authorization with PIN</b> use this PIN, which is also used to set automatic downloading by an IP phone.</li> </ul>
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	<ul> <li>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.</li> <li>Internal: Internal calls are allowed.</li> <li>Long distance: Internal calls, local calls, and long distance calls are allowed.</li> <li>International: Internal calls, local calls, long distance calls, and international calls are allowed.</li> <li>Prohibited: The extension is only allowed to receive calls.</li> </ul>
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.

ltem	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 <u>Authorization with *33 (Simplified-DISA)</u> 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. <b>Note 1</b> : Before using this function, the extension must have corresponding outbound rights. <b>Note 2:</b> 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.

ك		Group	Extension statu	s saoscriptic	shi usana meaning	contributions.					
o not use imber. ider the iur devic e IP evte	e the s currer e is be	ame extens It <u>security</u> I hind the N	ion number as hu level , a registrar AT. If extensions a mbers are blue on	nt group nu server from re allowed t	mber, the number to r internet will be rejecte to register from interne viensions. To recover t	each the operator of if the password at, please go to <u>Re</u> the call permission	feature access code, is the same as the ex- mote access page t place modify the P	analog exte tension nun o set the ex IN and nass	ension number or a nber. :ternal IP address. :word	nother IP	extension
+ Add	40	Quick Additi	ion 👍 Batch ac	Iding	Batch configuration	Delete	Registrar OPTIONS		Inpu	it Ext. nu	mber Q
	ID	Enable	Online status	Number	* Password * @	PIN * @	Display as	Group	Call restriction	Delete	Setting
	1		Offline	212					Internal	T	0
	2		Online	213		•••••			Domestic	Ŵ	0
	3		Online	214					Internal	1	0
	4		Offline	215					Internal	ŵ	0
	5		Offline	216					Internal	Ŵ	0
	6		Online	217					Internal		0
	7		Offline	218					Internal	-	0
	8		Offline	219					Internal	亩	0
	9		Online	220					Internal	亩	0
	10		Online	221					Internal	窗	0
										-	-



ltem	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).

Status	Basic	ic Extension Trunk			Multi-site		Applica	ation		Advanced	System tool	Log	
<u>Analog</u> IF	P IP table	Group	Extensi	ion status subscript	ion	Bound incoming	call	numbers					
			Ema	ail									
			Mol	bile phone 🕜									
			Aut	horization with *33	2								
			Call	forward		CEB/CENA (voicemail)							
			Cold	or ringback tone		Disable		(a))					
			Spe	ed dial groups 🙆		CFA (phone) CFB/CFNA (pho	ne)						
			Hot	line		CFB(phone) CFNA(phone)				nber			
			Call	forling		CFA (voicemail) CFB/CFNA (voic	emai	D					
			Call	TOTKING					Vau	e fill	in up to 20 blocked		
			Bloc	cked numbers		numbers, separa	ted b	y comma ","		ri tui	in up to 20 blocked		
			Assi	istant									
				Caller ID delivery		Caller ID with call waiting		Block from b picked up	eing		Call waiting		
				Call hold		Call transfer by called party		Call transfer calling party	by		Call transfer to outside		
			🕑 Recording 📃		On-demand recording		DND allowar	nce		Distinctive ringing			
				Call blocking/restriction		Polarity reversal signal sending		Silent monito	oring		Block from being silently monitored		
				Barge		Block from being b	arge	d-in					
							Court						
							Save						

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:

Status	Basic	Extension	Trunk	М	ulti-site	Applicatio	m	Advanced	System tool	Log
			Fax	Media A	PI <u>Recordin</u>	g Call barring	Storage	Manager/assis	tant	
Recordin	ng									
Voicema	Recording ? Recording server ? ail email settings	Disable	Remo     Remo     Sonet.com:	ote recording	Extern	al USB device	Intern	al USB flash drive		
		Outgonig amail con	uar (			0 0 102 169 4	15 22 or cm	ta sobu com		
		Outgonig email serv	er _			e.g. 192.166.4	13.32 OF SIT	up.sonu.com	1	
		Sender				It is recomme	It is recommended to use a public email address			
		Password								
		Note: Change the er	mail addres	s of receiving	recording in <u>A</u>	nalog extension	IP exten	sion		
					Save	•				
Note										

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

ltem	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

Listen to the voicemail message	<ul> <li>Analog phone: Press '98 and listen to the voiceman messages. You may hear:</li> <li>"You have no voicemail messages".</li> <li>"You have n new/saved voicemail messages".</li> <li>After the voicemail message is played, you may hear:</li> <li>"To repeat the message, press one. To delete, press two. To listen to the next message, press three".</li> <li>IP phone: Configure the feature access code for listening to the voice messages by MWI function key and memory key on IP phone.</li> </ul>									
	Function Key Settings									
	Key Type Value Line Subtype Pickup Number									
	DSS Key 2 Memory Key V 21/ SIP1 V BLP V									
	DSS Key 2 Memory Key v *98 SIP1 v MWI v									
	DSS Key 4 Key Event 🗸 AUTO V Headset V									
	Listen from outside: you need to pre-configure to allow to use DISA									
Format	<ul> <li>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 <b>Replay</b> or <b>Next</b> 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.</li> <li><b>Note:</b> The filename extensions are varied depending on the codec of the voicemail message:</li> <li>G.711µ: The file extension is .pcm.</li> <li>G.729: The file extension is .dat.</li> </ul>									
View the voicemail message	<ul> <li>G.729: The file extension is .dat.</li> <li>Go to Application &gt; 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.</li> <li>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</li> </ul>									

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**.

	Statu	is B	asic	Extension	Trunk	Mu	ilti-site	Application	Advanced	System too	Log	
Change	password	Upgrading	Import data	Export data	Factory resetting	Reboot	TDM capture	Ethereal capture	e <b>System time</b> Diagno	osis using Ping Vo	ice prompt packages	
						-						
					Time zone		(GMT+08:00) C	hina Coast, Hong	Kong			
				Current time		2017-08-25 17:5	0:13 🕖 Time	synchronization				
				System time sync interval		120		min				
				Primary time server		198.60.22.240						
				Secondary time server		133.100.9.2						
							1 and the second	1				
							Save					

## 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

Status	Basic	Extension	Multi-site	Application	Advanced	System tool	Log
		Analog trunk					
4	Go back						
		Gain to IP		0 dB			
		Gain to PSTN		-3.0 dB			
		Impedance	● Complex ○ 6	500 Ω Ο 900 Ω			
		Outpulsing delay	1000	ms (Range:	100 - 3000)		
		Call ID detection	After ring priority	<b>~</b>			
Busy	detection		Before ring				
		Busy tone count	Before ring priority	Cycle (Rang	je: 2 - 5)		
		Tone-on duration	After ring priority 500	ms (Range:	30 - 1000)		
		Tone-off duration	500	ms (Range:	30 - 2000)		
		Detect dual-frequency busy tone					
		Busy tone frequency	480	Hz			
			Course				
			Save				

## 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**.

atus Basic	Extension	Trun	ik Multi-site	A A	oplication	1	Advar	nced		System t	ool	Log
	System (	Cert.	Feature access codes	Encryption	Routing	<u>Dialing</u>	Tone	SIP	DTMF	Security	Access list	Call reco
Dialing timers												
	Off-hook timer		15		s (Range:	2 - 60, Defa	ault: 15)					
	Interdigit timer		5		s (Range:	2 - 60, Defa	ault: 5)					
	Complete entry timer	0	2		s (Range:	1 - 10, Defa	ault: 2)					
Digit map 🕜												
	[2-8]x000000x 02x000000x 013x000000x 13x0000000x [1-3,5,7-9]x000x 1xx 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x000x 9x											

Item	Description
[2-8]xxxxxx	The device terminates receiving digits after receiving eight digits starting with any digits between 2 and 8.
02xxxxxxxx	The device terminates receiving digits after receiving 11 digits starting with 02.
013xxxxxxxx	The device terminates receiving digits after receiving 12 digits starting with 013.
13xxxxxxxx	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.
xxxxxxxxX.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 <b>Interdigit timer</b> 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 <b>Complete entry timer</b> 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".
Т	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.

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

## 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**.

Status	Basic	Extension Trunk	Multi-site	Application	Advanced	System tool	Log
		Analog trunk	IP trunk				
			~				
4	Go back						
		Gain to IP		0 dB			
		Gain to PSTN		-3.0 dB			
		Impedance	Ocomplex	Ο 600 Ω Ο 900 Ω	]		
		Outpulsing delay	1000	ms (Range:	100 - 3000)		
		Call ID detection	After ring priorit	у 🗸			

## 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**.

Status	Basic	Extension	Trunk	Multi-site	App	lication		Advance	d	System t	lool	Log
			System F	eature access codes	Encryption	Routing	Dialing	<u>Tone</u> S		IF Security	Access list	Call record
												0
				Predefined	country/Regior	n specific	0	Customized				
		Country/Re	gion	United States		~						
		Dial tone		350+440/0								
		Second dial	tone	300+400/0								
		Stutter dial	tone	350+440/100	0/100,350+440	/100,						
		Busy tone		480+620/500	0/500							
		Congestion	tone	480+620/300,	0/200							
		Ring back to	one	440+480/200	0,0/4000							
		Off-hook w	arning tone									
		Call waiting	tone	440/300,0/10	000							
		Confirmatic	on tone	350+440/100	0/100,350+440	/100,						
				Save	Refresh							



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** 

Status	Basic E	xtension	Trunk	Multi	-site	Application	Advanced	System tool	Log
			Analog trunk	<u>IP trunk</u>					
<u>•</u> <u>G</u>	o back								
SIP se	rver								
	Default registrar				e.g. 168.3	3.134.51:5000 or www.:	sipproxy.com:5000		
	Local signaling port		8087		The port u	used to send SIP signal	ng messages to registra	r server (Range: 1 - 9999,	
		Ľ	Default: 5060)						
	Registration expiration	n	600		s (Range:	15 - 86400, Default: 60	0)		
	Proxy server		localhost:5060		e.g. 168.3	3.134.51:5000 or www.s	ipproxy.com:5000		
		(	0		The local :	signaling port number	is automatically added b	y 1 when the value is	
	Increments of port nu	mber c	configured as not the configured states of the	on-zero under the value is reached.	e conditions	of failed calls or registr	ation. A new incrementa	l cycle is started when	

## 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

Status	Basic	Extension	Trur	nk Mult	i-site	Application	n Advance	ed System t	ool Log
Change password	Upgrading	Import data	Export data		TDM capture	e Ethereal cap		Diagnosis using Ping	Voice prompt packages
Change a	dministrator	password							^
		Old pa	assword		•				
		New p	assword	••••					
		Confir	m new passw	ord	•••••				
					Save				
Change o	perator pass	word							
		New p	assword	••••	•••••				
		Confir	m new passw	ord					
					Save				

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





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.

Status	Basic	Extension	Trunk	Multi-site	Application	Advanced	System tool	Le
Network	Dialing rule	Auto attendant IVR A	udio files <u>Rem</u>	ote access				
		Port mappir Signaling pr	ng needs to be con ort 5060 , RTP po	figured on the router w rt 10010 - <mark>1</mark> 0266 .	hich the WAN port of de	vices connected to:		
		External IP a	iddress	External DDNS	$\sim$			
		Inner netwo	rk IP phone					
		registrator s	erve					
				Sav	e			
P2	P traversal	1						
1 2	u u u u u u u							
		Enable						
		Secure acce	ss code	629724				
		Registrar fo	r external terminal	629724.3D0FFA.sip.ne	wrocktech.com:5090 🗿			
				07660				
		QR code		集合结				
				i kate				
				C				
				Sav	e			
					-			

Step 1 Click Basic > Remote access, and make sure to enable the P2P traversal feature.

**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.

Status Ba	sic Extension	Trunk	Multi-site	Application	Advanced	System tool	Log
Network Dialing rule	e Auto attendant IVR	Audio files <u>Re</u>	mote access				
	Port mappi Signaling p	ing needs to be con ort 5060 , RTP pr	nfigured on the router wh ort 10010 - 10266 .	ich the WAN port of devi	es connected to:		^
	External IP	address	OM-based DDNS	~			
	Service pro	vider	www.oray.com	~			
	Account						
	Password			œ			
	Domain na	me					
	Renewal in	terval		s			
			Save				_
Note							

ltem	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 <b>WAN IP address</b> 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:

Network > Applicati	on > Virtual server				
+ Add					
	Name *	Host IP address *	Port range	Protocol	Del
	OM-RTP	192.168.2.218	10010-10266	TCP&UDP	窗
	OM-SIP	192.168.2.218	5060-5060	TCP&UDP	窗
			Save		



ltem	Description
Heat ID address	Enter the IP address of the OM. The current IP address of the device can be seen in the network part on the <b>Status</b> interface of the OM.
nost ir address	Note: It is a must to set a static IP address which can be configured on <b>Basic</b> > <b>Network</b> page. For the OM80, it is the ETH1 IP address.
	Enter the SIP signaling port and the RTP port range of the OM.
	You can go to <b>Trunk &gt;IP trunk&gt; Registrar OPTIONS</b> to view the SIP signaling port.
Port range	You can view the maximum value and minimum value of the RTP port on the <b>Application&gt;Media</b> 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.

PW ROCK				
RP1012/P	SIP IAX2	STUN C	DIAL PEER	
ASIC				
IETWORK	SIP Line SIP 2	×		
ETWORK	Basic Settings >>			
VOTR	Status	Registered	Domain Realm	
VOIP	Server Address	629724.3D0FFA.sip.new	Proxy Server Address	
	Server Port	5090	Proxy Server Port	
HONE	Authentication User	216	Proxy User	
	Authentication Password	•••••	Proxy Password	
UNCTION KEY	SIP User	216	Backup Proxy Server Address	
TANK CONTRACTOR	Display Name	216	Backup Proxy Server Port	5060
AINTENANCE	Enable Registration		Server Name	
ECURITY	Codecs Settings >>			
	Advanced STP Settings >>			



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 <b>Basic</b> > <b>Remote access</b> to view the <b>External IP address of the device</b> or the " <b>Registrar for External Terminal</b> " 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. <b>Note:</b> 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 <b>Extension</b> > <b>IP</b> > <b>Registrar OPTIONS</b> .
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 <b>Authentication User</b> 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.



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.

Basic	Li	ne	Routin	g	Advanced	Security	Call Status	Logs	Tools
Status	Network	VLAN		<u>SIP</u>	MGCP FolP	Alarms			
		Local sig	naling port		5060		(Range: 1 - 9999, Defa	ault: 5060)	
		Incremen	nts of port i	number	5		0		
	Г	Registrar	server		629724.3D0	FFA.sip.newrocktech.	0		
		Proxy ser	ver		629724.3D0	FFA.sip.newrocktech.	e.g. 168.33.134.51:500	0 or www.sippr	oxy.com:5000
		Subdom	ain name						
		Registrar	mode		Per line	~	•		
		User nan	ne						
		Registrar	password						
		Registrat	ion expirat	ion	600		s		

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.

Basic	Line	Routing	Advanced	Security	Call Status	Logs	Tools	
Batch Confi	iguration	Configuration A						
		Phone Line	FXS-1	~				^
		SIP Account Name	217					
		Caller ID Text	217					
		Registration						
		Auth User Name	217					
		Registrar password	******					
	-	Hot line	Disable		~			
	,	Color ringback tone			~			
		Set up speed dial						
	,	Call forwarding						
		Call forking						<b>.</b>
					Save			

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

Basic	Line	Rout	ing Advand	ced Se	curity	Call Status	Logs	Tools			
				<u>Call sta</u>	atus Call histo	ory on FXS SIF	^o message count				
Cor	nnected: 0	Idle: 4 In-	progress: 0 Other: 0	1		Clear Ref	resh				
	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

Basic	Line	Routing	Advanced	Security	Call Status	Log	s Tools		
					<b>System status</b> Ca	l message	System startup	Manage log	
		Login User Info 1) 192.168.110	o >>>> .62 1						^
	SIP Registratio Contact: <sip:2 response: Contact: <sip:2 response: Contact: <sip:2< td=""><td>n Info &gt;&gt;&gt;&gt;&gt; 217@192.168.111.52:5 200 220@192.168.111.52:5 200 221@192.168.111.52:5</td><td>060&gt; 060&gt; 060&gt;</td><td></td><td></td><td></td><td></td><td></td></sip:2<></sip:2 </sip:2 	n Info >>>>> 217@192.168.111.52:5 200 220@192.168.111.52:5 200 221@192.168.111.52:5	060> 060> 060>						



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.

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):

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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 <u>http://x.x.x.x/xml?method=gw.config.set&id215=no</u>
- Check the log, if device receive ACK after 2000K 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*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: <u>https://x.x.x.x/xml?method=gw.config.set&id211=20</u>

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 <u>gs@newrocktech.com</u> through e-mail for instant help.