



MX Series VoIP Gateway

(HX4G/MX8G/MX60E/MX120G)

By New Rock Global Services Team



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Main Functions

- Connect analog telephone, PBX, FAX and POS machine to the IP core network, or PSTN
- Up to **500** rules for routing and number transformation(keep, remove, replace and add prefix/suffix to number).
- Compatible with unified communication solutions, such as CallManager, OCS and Asterisk;
- Support call progress tones for various countries and regions;
- G.711, G.729A, and G.723.1, GSM, iLBC, G.168 echo cancellation, RFC2833 and SIP INFO for DTMF transmission and fax relay (T.30/T.38)
- Password management, ciphertext data storage, IP table for external network access control, and system log backup.
- Web based GUI to configure parameters, upgrade firmware, import/export configuration files, monitor status, visit history, call traces
- Remote management with auto-provisioning, SNMP, TR069
- Supports protocols including SIP, RTP, TFTP, FTP, HTTP, STUN, and SNMP
- Flexible configuration of Phone/Line interfaces;
- Support polarity inverse detection and busy tone detection;
- Registration and authentication of SIP terminals

Highlighted Features



Terminals: Gateway

Highlighted Features

Service



Outstanding Points

Closed source system :

Write our own source code and platform to be more stable and security.

Capacity :

New Rock Gateway support from 2 up to 96 ports while other brand maximum 48 ports

Flexibility:

New Rock gateway are very flexible with different configuration of FXO and FXS combination, while the other band mainly support only FXS.

High Availability:

All Newrock gateways supports many ways to keep working under disaster such as active standby and load balance.

Stability :

we sold millions of gateway products to different customers from SMB users to big enterprises even governments , banks and Armies , we gain our good reputation with high quality sound and stable performance.

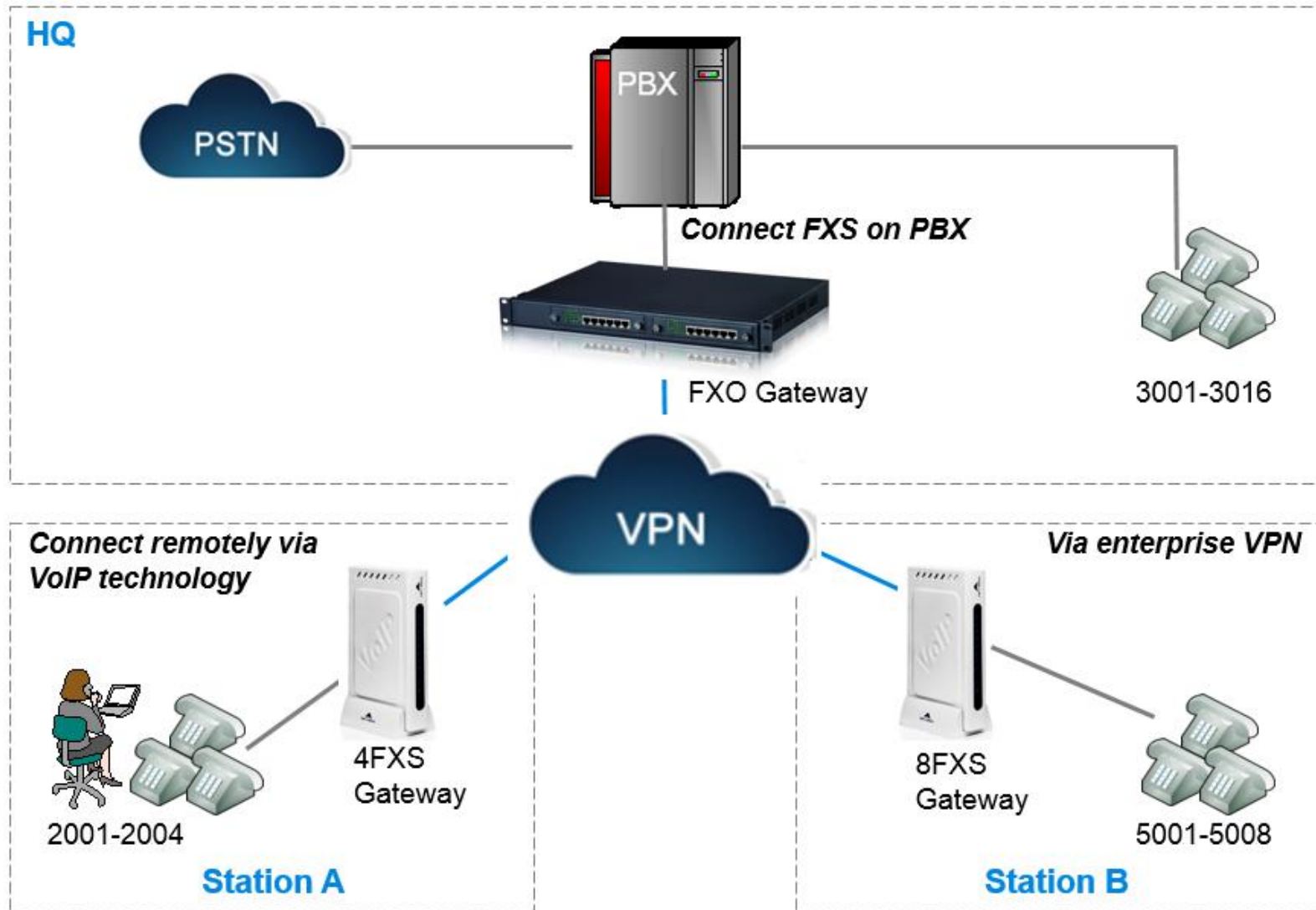
Compatibility :

Among our users , there are different platforms such as Cisco , Microsoft Lync , Broadsoft , Asterisk , Avaya , Tadiran Ours is always the best choice.

Real time recording :

Sending voice recordings in real time to an external storage server running with a recording agent.

Application Scenario



HX4G Front Panel



Item	Description
	Power Indicator
WAN	WAN interface indicator
PC	PC interface indicator
FXO/FXS	FXS /FXO port indicator

Indicators on HX4G Front Panel

	Status	Description
PWR	Blinking green	The device is starting.
	Steady green	The device is running.
	Off	The device is powered off or a power supply fault occurred
STU	Steady red	The WAN interface failed to acquire the IP address. Possibly the WAN interface is not connected to a network cable, the WAN interface address failed to be acquired by DHCP, the IP addresses are conflicted, and the PPPoE dialing failed.
	Blinking red	The device is starting or the KUPDATE is upgrading.
	Steady green	Registration is successful.
	Blinking alternatively between red and green	Registration failed.
	Blinking green	Calling.
	Off	Registration has not started.
WAN	Steady green	A WAN connection is established without any service flow.
	Blinking green	A WAN connection is established with service flow.
	Off	WAN interface is disconnected.
PC	Steady green	A link is connected without any service flow.
	Blinking green	A service flow is being transmitted.
	Off	A link is not connected.
FXS/FXO	Steady green	Off-hook or call established
	Blinking green	Ringing on incoming call
	Off	The port is in idle status

HX4G Back Panel



Item	Description
PWR	Power interface, 12 VDC input
PC and WAN	The PC port is used to connect a computer. The WAN port is used to connect the uplink network. Both are 10/100 Mbps Ethernet ports (RJ45). They share one IP address, which, by default, is obtained through DHCP. If no IP address is obtained, 192.168.2.218 is used by default, and you can change it on Basic > Network page.
FXO/FXS	FXS port or FXO port

MX8G Front Panel

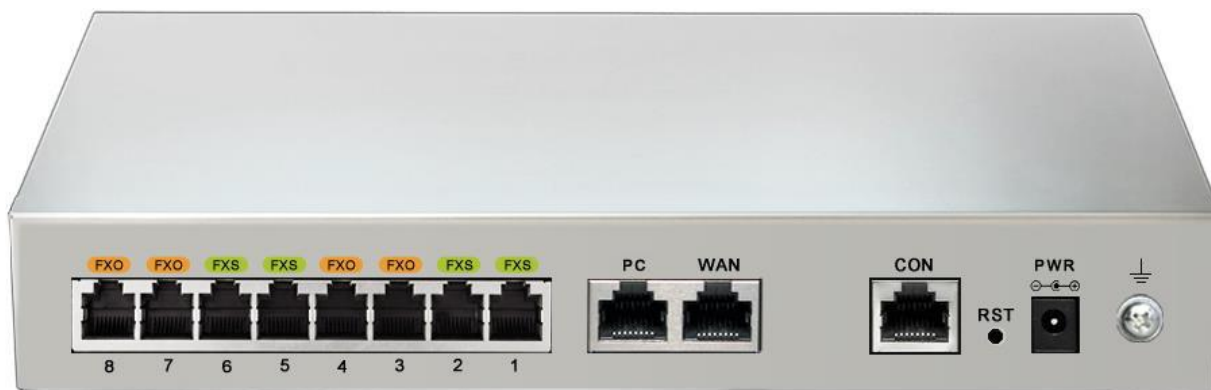


Item	Description
PWR	Power indicator , Light-on indicates that it has been powered.
STU	Status indicator
WAN	WAN interface indicator
PC	PC interface indicator
VOICE	FXS/FXO port indicator

Indicators on MX8G Front Panel

	Status	Description
PWR	Green Flashing	The device is starting.
	Steady Green	The device is running.
	Off	Power off
STU	Steady red	The WAN interface failed to acquire the IP address.
	Red flashing	The device is starting or the KUPDATE is upgrading.
	Steady green	Registration is successful.
	Flashing alternatively between red and green	Registration failed.
	Green flashing	Calling.
	Off	Registration has not started.
WAN	Steady Green	A WAN connection is established without any service flow
	Green Flashing	A WAN connection is established with service flow
	Off	WAN interface is disconnected
Voice	Steady yellow	Calling in or out via an analog trunk (FXO).
	Yellow flashing	Ringin of calling in for an analog trunk.
	Steady green	Off-hook or call established (FXS)
	Green Flashing	Ringin on incoming call
	Green flashing	The port is in idle status

MX8G Back Panel



Item	Description
PWR	Power interface, 12 VDC input
CON	The console port is used for local management and testing. PCs can be connected to device by linking the RS232 port to CON port.
PC/WAN	The PC port is used to connect a computer. The WAN port is used to connect the uplink network. Both ports share one IP address, which, by default, is obtained through DHCP.
FXO/FXS	FXS port or FXO port
RST	To restore the MX8G to factory default, press the RST for more than 3 seconds and release it when the STU light starts blinking in red. This setting will be valid after rebooting the device.

MX60E Front Panel



Item	Description
RJ45 sockets	Each RJ45socket can correspond with four pairs of analog lines. Note: numbers of interface slots vary from different configuration. Each interface slot can correspond with four RJ45 sockets or can contain one 16-port interface card.
LED Matrix	Each matrix of 4x4 LED corresponds to the line status , line type and system operation status on one interface card

MX60E Back Panel



Item	Description
AC power socket	Power supply: 100 to 240V; 50/60Hz; 1A
Cooling fan	12VDC; 92mA; 5800RPM
ETH	ETH1 and ETH2 share the same IP address for allowing access to the external network. Dual-LAN redundancy is supported.
CON	A configuration interface
USB	USB interface, reserved for future use
PWR/STU/ALM	Indicators for power supply, system status and alarm
Ground pole	Use the grounding cable to connect MX60E and the grounding terminal of the rack.

Indicators on MX60E Back Panel

	Status	Description
PWR	Green	Power on
	Off	Power off
STU	Green	Normal operation
	Off	System locked and inactive
ALM	Red	Alarms existed and all alarm information confirmed
	Red Constant	System in the process of power up and not in the normal operation mode
	Red Flash	New alarms occurred but not confirmed
	Off	No alarms

MX120G Front Panel



Item	Description
RJ45 sockets	Each RJ45socket can correspond with four pairs of analog lines. Note: numbers of interface slots vary from different configuration. Each interface slot can correspond with six RJ45 sockets or can contain one 24-port interface card.
LED Matrix	Each matrix of 6x4 LED corresponds to the line status , line type and system operation status on one interface card

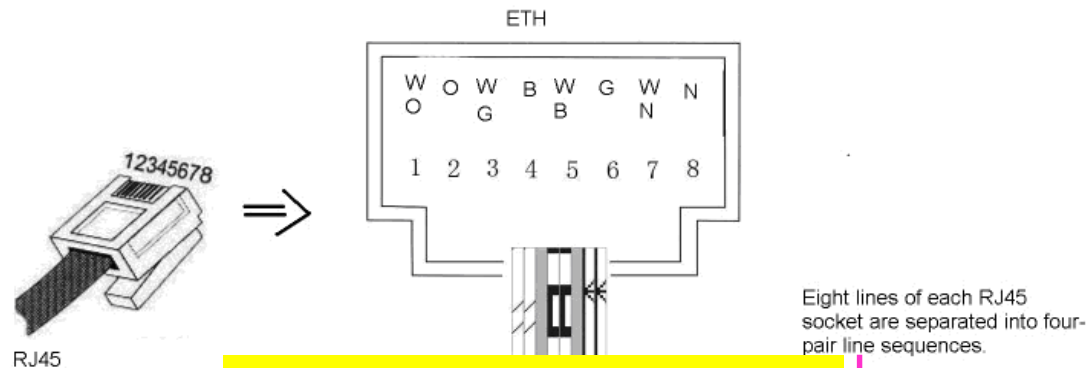
MX120G Back Panel



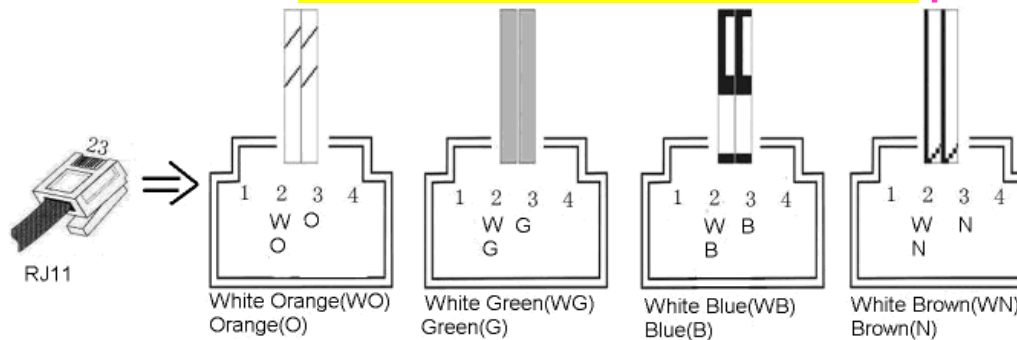
Item	Description
AC power socket	Power supply: 100 to 240V; 50/60Hz; 1A
Cooling fan	12VDC; 92mA; 5800RPM
ETH	ETH1 and ETH2 share the same IP address for allowing access to the external network. Dual-LAN redundancy is supported.
CON	A configuration interface
USB	USB interface, reserved for future use
PWR/STU/ALM	Indicators for power supply, system status and alarm
Ground pole	Use the grounding cable to connect MX60E and the grounding terminal of the rack.

Pin Specs for MX60E/MX120G RJ45 Socket

RJ45 Pin #	1	2	3	4	5	6	7	8
Analog line pair	1 st Pair		2 nd Pair	3 rd Pair		2 nd Pair	4 th Pair	
	TIP1	RING1	TIP2	TIP3	RING3	RING2	TIP4	RING4
Reference color	White Orange	Orange	White Green	Blue	White Blue	Green	White Brown	Brown



MX60E Subscriber Line Connection



Corresponding Relation Between MX60E/MX120G RJ45 Socket and Line Number

RJ45 Socket # (From Left to Right)	1	2	3	4	5	6
Line Number (From Top to Bottom) of MX60E interface card	1 ~ 4	5 ~ 8	9 ~ 12	13 ~ 16	N/A	N/A
Line Number (From Top to Bottom) of MX120G interface card	1 ~ 4	5 ~ 8	9 ~ 12	13 ~ 16	17 ~ 20	21 ~ 24

LED Indicators on MX60E/MX120G Front Panel

Indicator	Status	Description
Line Status (normal use)	On	Line is in a call or in use
	Off	Line is idle
Line Type (Both WAN and LAN ports must be disconnected)	Steady On	Line is equipped as analog subscriber line type
	Flashing	Line is equipped as analog foreign exchange line type
	Off	Line is not equipped or not ready for use
System Operation Status (Startup, diagnosis or operation)	Flashing letter C	The IP address of gateway conflicts with that of other equipment in LAN. Please settle this problem before the gateway can be operated normally.
	Flashing letter D	Internal failures have been encountered during gateway start up procedure. Please contact your local distributor for further diagnosis.
	Flashing letter P	The gateway is in progress of system software upgrade. Please guarantee stable power supply and do not conduct other operations during this period.
	Flashing letter T	The application software of gateway has been exited. If it can not be restored by rebooting the system, please contact your local distributor for further diagnosis.

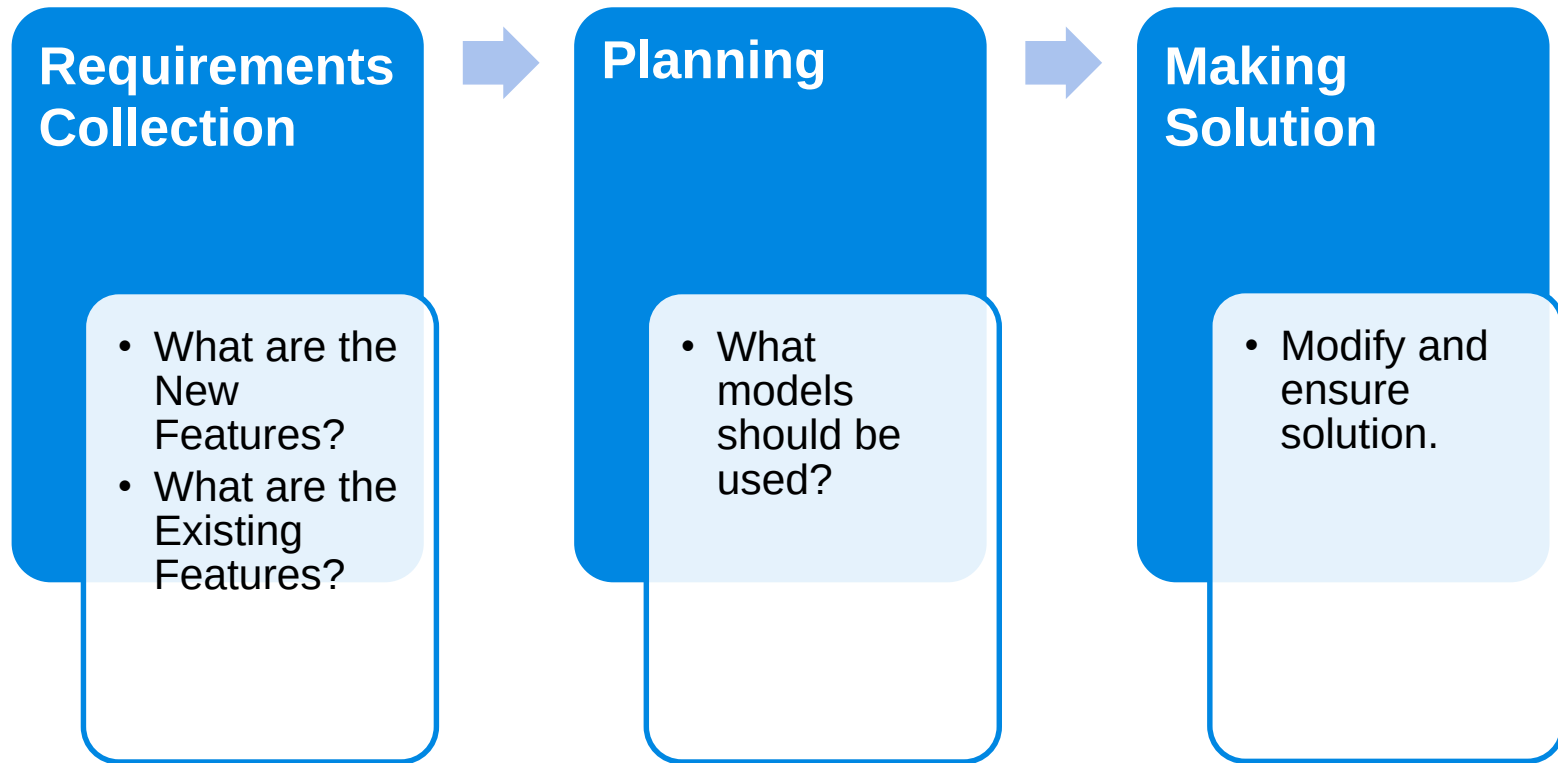
Hardware Specifications

Item	MX8G	MX60E	MX120G
System Memory	32MB	64MB	256MB
System Flash	8MB	16MB	32MB
Processor	Motorola Power PC852	ATMEL AT91SAM9G20	FREESCALE MPC8247
DSP	TI C5509	TI C5509	TI C5509
Power	12VDC, 3A	100-240V, 50/60Hz,1A	100-240V, 50/60Hz,1A
Power Consumption	15 W (Max)	75 W (Max)	125 W (Max)
Size	195x118x34 mm	440x260x44 mm	440X300X85 mm
Net Weight	0.65 kg	3 kg	6.32kg
Operating Environment	0 to 40°C, Non-Condensing Humidity 10% to 90%		

Other Specifications

Security	Encryption Type / Method	Key Length
Auto provision	Advanced Encryption Standard (AES)	256 bit
Password	Advanced Encryption Standard (AES)	192 bit

Preparing for Installation



Lightning Protection

Wiring Outdoor

Use PT2400 or
PT4800
Distribution
Panel with
lighting
protection



Install PT2400/ PT4800 Distribution Panel

Connect the RJ45 ports on front of GW to RJ45 ports on the left of PT2400 or PT4800 by Straight-through Cat-5 cable and output through RJ11 ports on the right side of PT2400 or PT4800.



Good Installation



Grounding Rules

Select the reliable grounding system.

The resistance between gateway and grounding system should be less than 3Ω .

Power System should be installed a reliable grounding system.

Making RJ45 to RJ11 Cable

What models need to use it?

- MX60E
- MX120G

The usage

- If you don't use PT2400 or PT4800 distribution panel, you need to make RJ45 to RJ11 cable to separate lines from RJ45.

Contents

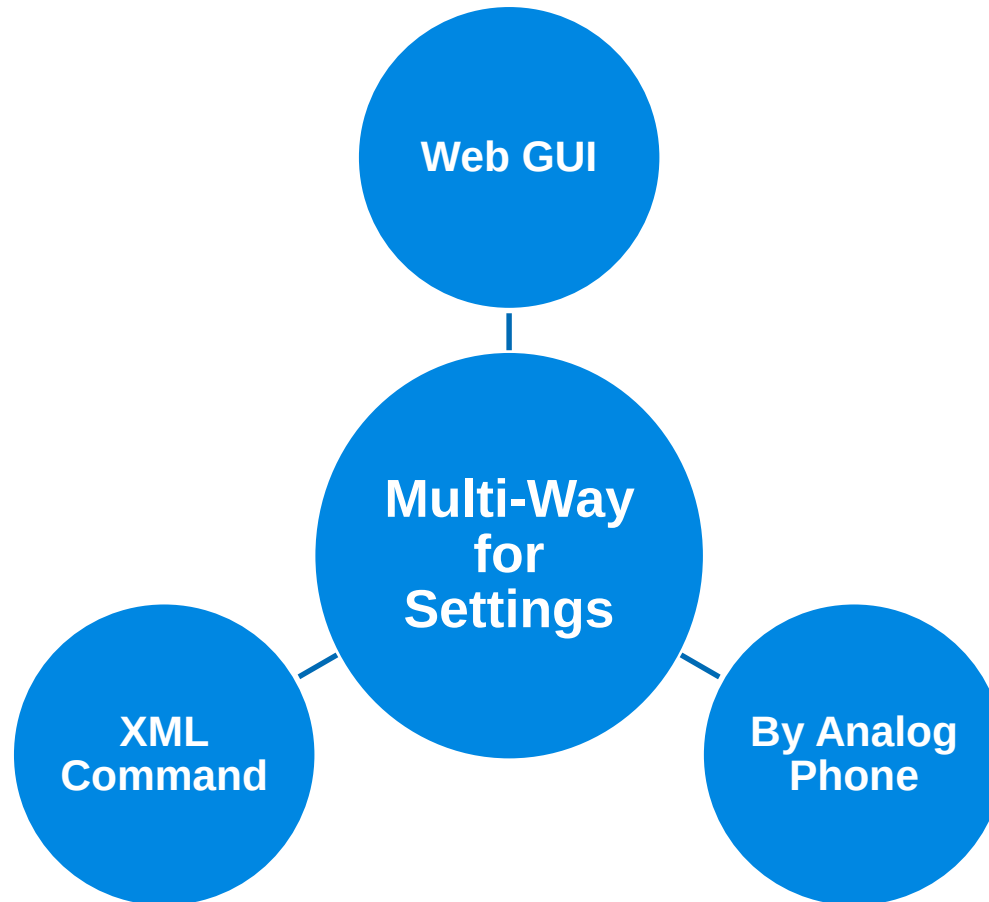
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Configuration Interfaces



Setting by Analog Phone

Using the function keys on your analog phone to set up some features of gateway with FXS ports.



Common settings :

- **Enable DHCP:** pickup the phone, dial ***90###1#**, you will hear “Beep beep” confirmation tone, then reboot gateway.
- **Restore factory settings:** pickup the phone, dial ***921234#**, you will hear “Beep beep” confirmation tone, then reboot gateway.
Note: only restore the network settings in gateway .
- **Setting IP address:** pickup the phone, dial ***90 + IP address + # + Subnet mask + # gateway IP + # + 0 + #**, using “*” to replace dot in IP address and subnet mask, you will hear “Beep beep” confirmation tone, then reboot gateway.
Example: setting IP address:192.168.3.5, subnet mask:255.255.255.0, default gateway:192.168.3.1, you should dial:
***90 192*168*3*5#255*255*0*0#192*168*3*1#0#**

XML Command



- Log in Web GUI by administrator.
- Input and Execute XML Command.
- Check if it is success through infos on the page.

XML Command Syntax:

`http://x.x.x.x/xml?method=gw.config.set&idnnn=value`

x.x.x.x is IP address of GW, ***nnn*** is parameter id. In the figure, it means the parameter which id is 215 is set to yes.

There will be some infos on the page after you finish the command.

stat="ok" means the setting has taken effect; ***stat="restart"*** means you need to restart the GW.

stat="fail" means the setting is refused, there will be a note in ***cause="Not login yet"***, have not log in.

Feature Setting by XML Command

HSB Always Proxy

- <http://x.x.x.x/xml?method=gw.config.set&id120=yes>
- Note: Firmware must be Version 323 to later.

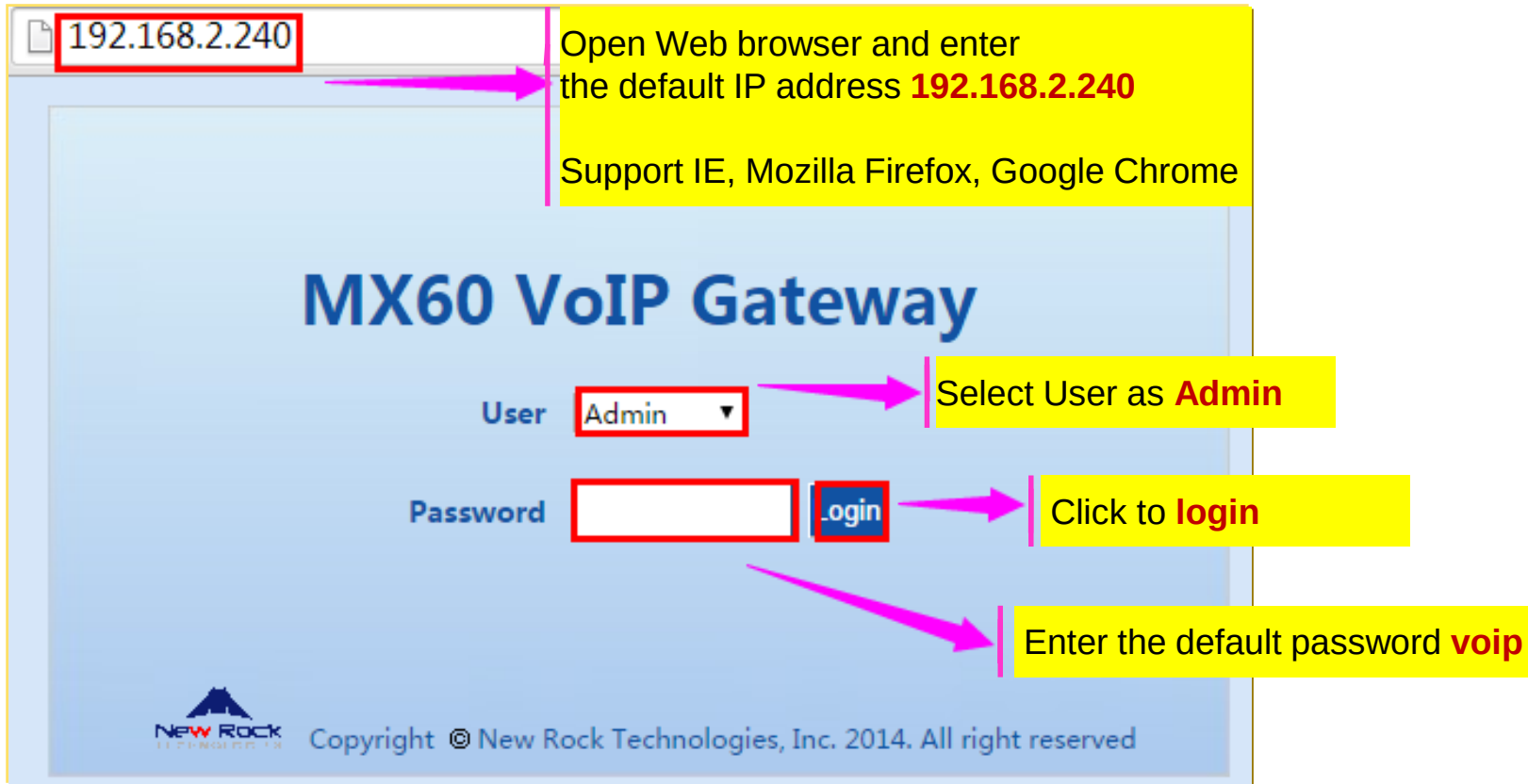
Enable display name detection on TG

- <http://x.x.x.x/xml?method=gw.config.set&id199=1>

Enable high-voltage MWI feature on MX60E

- <http://x.x.x.x/xml?method=gw.config.set&id196=4>
- Note: Firmware must be Version 323 or later, and line driver should be upgraded to Version 227 or later.

Log in to Web Interface



The screenshot shows a web browser window with the address bar containing **192.168.2.240**. The page title is **MX60 VoIP Gateway**. The login form includes a **User** dropdown menu set to **Admin**, a **Password** text input field, and a **login** button. Annotations with arrows point to these elements: the IP address, the **Admin** user selection, the password field, and the **login** button.

Open Web browser and enter the default IP address **192.168.2.240**

Support IE, Mozilla Firefox, Google Chrome

Select User as **Admin**

Click to **login**

Enter the default password **voip**

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- The system will confirm timeout if users do not conduct any operation within 10 minutes after login. Users are required to log in again for continuing operations.
- Upon completion of configuration, click the **Logout** button to return to the login page, so as not to affect the login permission of other users.

Basic Network Configuration

Basic	Routing	Line	Advanced	Status	Logs	Tools	Info
Status Network System SIP MGCP FoIP Logout							
Host name		MX60-VoIP-AG		Contain letter, number and "-" but must start with letter			
eth1							
MAC address		DA:FD:D1:18:AF:C6					
IP address assignment		Fixed		Select Fixed to obtain Static IP address			
IP address		192.168.2.240					
Netmask		255.255.0.0		Fill in IP address/Netmask/ Gateway IP address			
Gateway IP address		192.168.2.1					
DNS							
Enable		☒		Check Enable to activate DNS service			
Primary server		192.168.2.1		e.g. 202.96.209.6			
Secondary server		202.96.209.133		e.g. 202.96.209.133			
SNTP							
Primary server		198.60.22.240					
Secondary server		133.100.9.2					
Time zone		(GMT+08:00) Beijing					
		Submit		Click to Submit			

Basic SIP Configuration

Welcome admin Login time: 1970-01-01 08:06:39

Basic

Routing

Line

Advanced

Status

Logs

Tools

Info

[Status](#) | [Network](#) | [System](#) | [SIP](#) | [MGCP](#) | [FoIP](#) | [Logout](#)

Signaling port	5060	(range:1-9999,default:5060)
Auto SIP port selection	Off ▼	1-10:Local SIP port will auto select, based 5060 increasing the value
Registrar server	168.33.134.50:5060	
Proxy server	168.33.134.50:5060	e.g. 168.33.134.50:5060
User agent domain name		e.g. www.gatewaysip.com
Registration mode	Per line ▼	
User name		
Password		You may obtain it from service provider
Registration period	600	s(range:15-86400, default:600)

Fill in **Registrar and Proxy server**

[Primary-Standby configuration](#)

Submit

Click to **Submit**

Basic System Configuration

Basic	Routing	Line	Advanced	Status	Logs	Tools	Info
Status Network System SIP MGCP FoIP Logout							
First digit timer	12	s(range:2-60, default:12)					
Inter-digit timer	12	s(range:2-60, default:12)					
Critical digit timer	3	s(range:1-10, default:5)					
Codec	PCMU/20,G723/30,PCMA/20,iLBC/30,G729A/20 G729A/20, G723/30, PCMA/20, iLBC/30, GSM/20						
Hook-flash handle	Internal						
DTMF method	RFC 2833						
2833 payload type	101	(range:96-127, default:101). This value should be as the value in server					
DTMF on-time	100	ms(range:80-150, default:100). This is the on-time of sending					
DTMF digit							
DTMF off-time	100	ms(range:80-150, default:100). This is the off-time of sending					
DTMF digit							
DTMF detection threshold	48	ms(range:32-96, default:48). This is the detection threshold for receiving DTMF digit					
DTMF detection adjust	16	ms					
Submit							

Fill in **Codec**

Choose **DTMF Method** you want

Fill in **DTMF on/off-time**

Click to **Submit**

Advance Routing Table Configuration

The screenshot shows a web-based configuration interface for a routing table. At the top, there is a navigation bar with tabs: Basic, Routing (selected), Line, Advanced, Status, Logs, Tools, and Info. Below the navigation bar, there is a breadcrumb trail: [Digit map](#) | [Routing table](#) | [IP table](#) | [Logout](#). The main content area is a large white text box with a red border. A yellow callout box with the text "Input **Routing rules** in the text box" has a pink arrow pointing to the text box. At the bottom of the interface, there are two buttons: "Submit" and "Help". A yellow callout box with the text "Click to **Submit**" has a pink arrow pointing to the "Submit" button. Another yellow callout box with the text "Click **Help** for tips on how to write routing rules" has a pink arrow pointing to the "Help" button.

Basic Routing Line Advanced Status Logs Tools Info X

[Digit map](#) | [Routing table](#) | [IP table](#) | [Logout](#)

Input **Routing rules** in the text box

Click to **Submit** Submit Help Click **Help** for tips on how to write routing rules

FXS phone number configuration

Basic

Routing

Line

Advanced

Status

Logs

Tools

Info

[FXS phone number](#) | [Feature](#) | [Feature batch](#) | [Logout](#)

Click to **Submit**

Submit

FXS 1st line No.		Batch
ID1	300	
ID2	301	
ID3	302	
ID4	303	
ID5	304	
ID6	305	
ID7	306	
ID8	307	
ID9	308	
ID10	309	
ID11	310	
ID12	311	
ID13	312	

Fill in **Telephone number**

FXS phone feature configuration

Basic	Routing	Line	Advanced	Status	Logs	Tools	Info
FXS phone number Feature Feature batch Logout							
Line ID			Select FXS port Line ID				
Phone number			Max 24 characters				
Registration			Enable Registration				
Password			Enter Password for registration of this line				
Hot line							
Hot line number			Max 20 digits				
CRBT			Color ring back tone				
Speed dials							
Call forwarding							
Forking							
Release control by caller			Also see " Caller release " in page " Advanced > Line "				
Loop open disconnect			Also see " Loop open interval " in page " Advanced > Line "				
RFC6913							
☐ Call waiting			☒ Call hold				
☒ Caller ID display			☒ Caller transfer				
☐ Outgoing call b			☐ Caller ID restriction				
☐ 3-way			☐ DND(Do Not Disturb)				
☐ Maintenance			☐ Polarity reversed signa				
☐ Subscribe MWI(Also see " MWI subscription " in page " Advanced > SIP ")							
☐ DDI(Direct Dialing in)							
Submit			Click to Submit				

Advance Security Configuration-1

Basic	Routing	Line	Advanced	Status	Logs	Tools	Info
System Security White list Media stream SIP Line RADIUS Encryption Greeting Tones Feature codes Logout							
Telnet service							
Telnet		☒ On ☐ Off		Click On to enable Telnet server			
Password		sensitive		Input Telnet password and			
Confirm password				Confirm password			
Web service							
Port		80		Input Web server port			
				Submit		Click to Submit	

Advance Security Configuration-2

Tips	Security Measures
1	To ensure the device security, changing the web GUI and Telnet passwords periodically is recommended.
2	Disabling Telnet services is highly recommended for MX8G/MX60E/MX120G so that there will be no Telnet-based access to the device is permitted.
3	The default web server port is 80. It is also recommended to change the port number of the Web server . The value is a 2-5 digits number.
4	Administrators can add the authorized IP addresses to IP table . This table is designed to ensure the safe use of gateways as it will only process the information from authorized IP addresses.
5	Administrators can add up to maximum of 20 IP Addresses to the Whitelist on the MX8G/MX60E/MX120G. Only the addresses on the list are allowed to access the web and telnet services.
6	Configure the UDP signaling port for transmitting and receiving SIP messages (default value is 5060). If the MX8G/MX60E/MX120G is connected directly to the Internet, it's recommended to change the default signaling port value to prevent hacker attacks.

Advance Media Stream Configuration

Basic	Routing	Line	Advanced	Status
System Security White list Media stream SIP Line Trunk				
Voice				
Min. RTP port	10010	3000-6553		
Max. RTP port	10250	3020-6553		
iLBC payload type	97	97-127, default 97		
G.723.1 rate	6300(bit/s) ▾			
TOS bits	0x0C	Normally 0		
Min. jitter buffer	3	0-30(frame) set the value with caution		
Max. jitter buffer	50	10-250(frame), default 50		
RTP drop SID	☐			
Enable VAD	☒			
RTP destination address	☒ From SDP global connection ☐ From SDP media connection			
Submit				

Input Minimum RTP port
Note: The minimum value of UDP ports for RTP transmission and receiving must be greater than or equal to 3000. Recommended value to be equal or greater than 10000. Each phone call will occupy RTP and RTCP ports.

Input Maximum RTP port
Note: It's advisable to be greater than or equal to "2 × number of lines + min. RTP port".

Click to **Submit**

Call Status

Welcome admin Login time: 1970-01-06 06:09:36

Basic Routing Line Advanced **Status** Logs Tools Info

Call status | Call history on FXS | SIP message count Logout

Connected:0 Idle:16 In-progress:0 Other:0

Click Clear

Line ID	Phone No.(This End)	Registration	Line	Call	Phone No. (Other End)	Duration	In	Out	Answered	Operation
FXS-1	8000	Not registered	On-hook	Idle			0	0		-
FXS-2	8001	Not registered	On-hook	Idle			0	0		-
FXS-3	8002	Not registered	On-hook	Idle			0	7		-
FXS-4	8003	Not registered	On-hook	Idle			0	0		-
FXS-5	8004	Not registered	On-hook	Idle			0	0		-
FXS-6	8005	Not registered	On-hook	Idle			0	0		-
FXS-7							0	0		-
FXS-8							0	0		-
FXS-9							0	0		-
FXS-10	8009	Not registered	On-hook	Idle			0	0		-
FXS-11	8010	Not registered	On-hook	Idle			0	0		-
FXS-12	8011	Not registered	On-hook	Idle			0	0		-
FXS-13	8012	Not registered	On-hook	Idle			0	0		-
FXS-14	8013	Not registered	On-hook	Idle			0	0		-

Parameters of call state will be displayed

Log Management

Choose **Log level** (Default is 4. The higher the level goes, the more details the log file will be)
Log level should be set to be 4 or lower when gateway is used in normal operation,
avoiding influencing the system performance.

The screenshot shows a web interface with tabs: Basic, Routing, Line, Advanced, Status, Logs, Tools, and Info. The 'Logs' tab is active. Below the tabs are links: System status, Call message, System startup, Manage log, and Logout. The 'Manage log' section contains a form with the following fields:

- Download log (text input)
- Log level (dropdown menu, currently set to 4)
- Download (button)
- Advanced (Syslog) (checkbox)
- System log server (text input, value: e.g. 137.61.68.25)
- Local log port (text input, value: 4)
- Log server (text input, value: e.g. 137.61.68.26)
- Submit (button)

Annotations with pink arrows point to the 'Log level' dropdown and the 'Submit' button. A yellow box next to the 'Download' button contains the text: 'Click **Download** to download logs'. Another yellow box next to the 'Submit' button contains the text: 'Click to **Submit**'.

Log Level	Description	Log Level	Description
1	System Information	5	Call Information
2		6	RTP Event
3	DSP Instructions	7	T38 Fax Information
4	DSP Event	8	Registration Information

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Maintenance Tools

The screenshot shows a web interface for 'Maintenance Tools' with a navigation bar at the top containing 'Basic', 'Routing', 'Line', 'Advanced', and 'Info' (with a close icon). The 'Advanced' tab is selected. On the right side of the navigation bar, there is a 'Logout' link. The main content area is divided into a left sidebar and a right main panel. The sidebar contains a list of buttons: 'Change password', 'Export data', 'Import data', 'Upgrade', 'Restore factory settings', 'Restart', 'Reboot', 'TDM capture', and 'Ethereal capture'. The main panel displays the 'Administrator password' form, which includes fields for 'New password' and 'Confirm new password', and a 'Submit' button. Several yellow callout boxes with red text and pink arrows point to specific features: 'For configuration export and import data (only voice configuration)' points to the 'Export data' and 'Import data' buttons; 'For changing Administrator password' points to the password form; 'For software upgrading' points to the 'Upgrade' button; 'For software restart and device reboot' points to the 'Restart' and 'Reboot' buttons; 'For capturing voice stream from Phone or Line interface' points to the 'TDM capture' button; and 'For capturing IP voice data' points to the 'Ethereal capture' button.

Basic Routing Line Advanced Info X

Logout

For configuration export and import data (only voice configuration)

Change password

Export data

Import data

Upgrade

Restore factory settings

Restart

Reboot

TDM capture

Ethereal capture

Administrator password

New password

Confirm new password

Submit

For changing Administrator password

For software upgrading

For software restart and device reboot

For capturing voice stream from Phone or Line interface

For capturing IP voice data

Device Information

Basic	Routing	Line	Advanced	Status	Logs	Tools	Info
Logout							
		Software version	Rev 1.9.82.333	Indicates the Software version			
		Hardware version	Rev 1.0.1 MX60-16S	Indicates the Hardware version			
		Kernel version	Kernel 2.0.0 (F)	Indicates the Kernel version			
		DSP version	Rev 1.8.203	Indicates the DSP version			
		More info	http://www.newrocktech.com.cn				

Contents

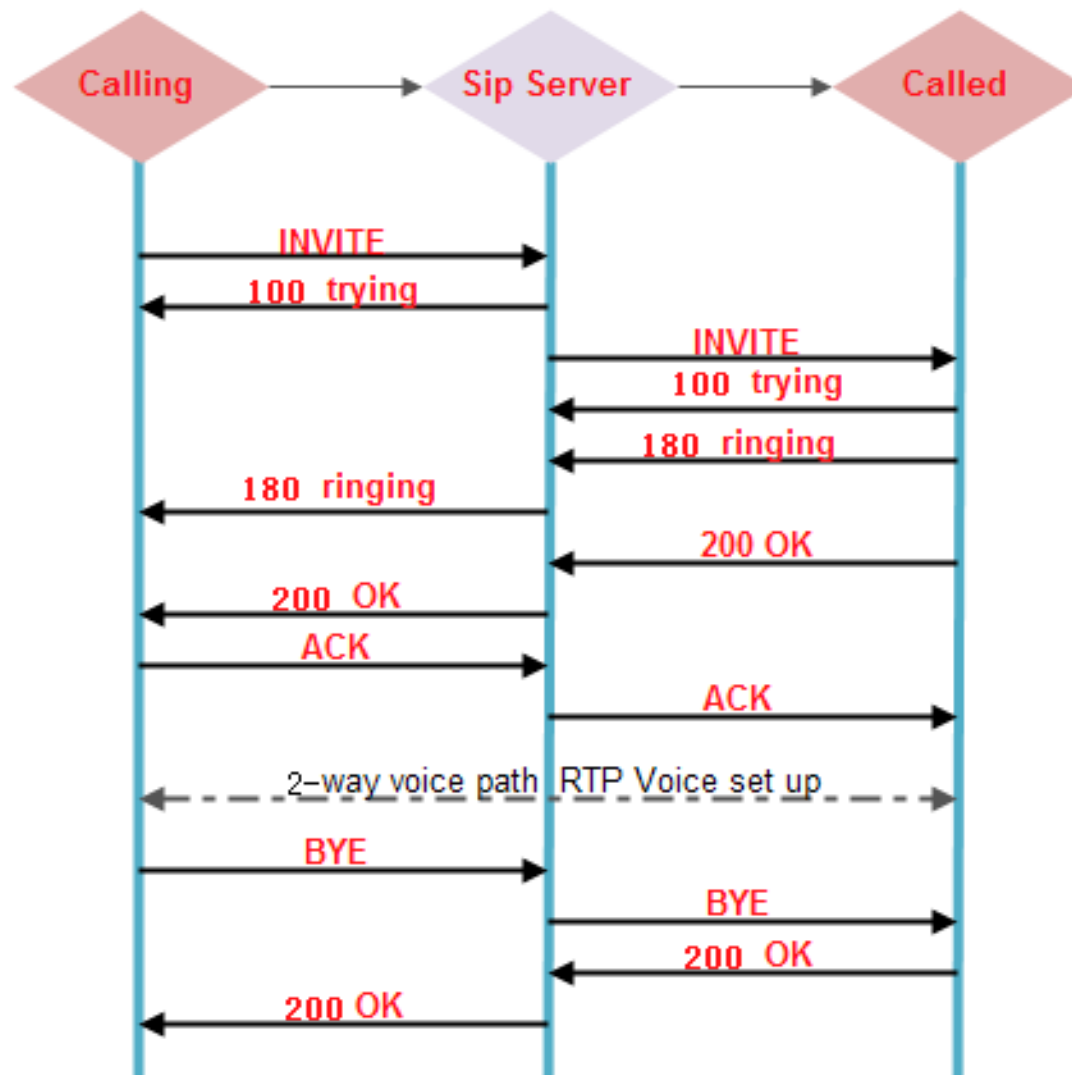
1 Product Introduction

2 Basic Configurations

3 Maintenance Operations

4 Troubleshooting Techniques

SIP Basic Call Process (RFC3261)



SIP Protocol Analysis on Log Level 4 (For invite)

[12/10 13:59:04.577820] **LINE-143(2) offhook** || The extension 143 off-hook prepare to make a call
[12/10 13:59:07.795907] LINE-143(2) **outgoing** 1418191147866553213-0@192.168.2.218 || Outgoing call
[12/10 13:59:07.853078] SIP:**send[192.168.2.218:5060]** len[844] || 192.168.2.218 for the destination IP address
INVITE sip:234@192.168.2.218 SIP/2.0 || 234 for the invite request number
Via: SIP/2.0/UDP 192.168.2.218:5060;rport;branch=z9hG4bK--1062731046141819114788
To: <sip:234@192.168.2.218>;local=c007 || 234 for the called party number
From: <sip:143@192.168.2.218>;tag=141819114787-1 || 143 for the calling party number
Call-ID: 1418191147866553213-0@192.168.2.218 || The call ID header field acts as a unique identifier to group together a series of message
CSeq: 100080 INVITE || The CSeq header field serves as a way to identify or order transactions
Max-Forwards: 70
Expires: 60000
Contact: <sip:143@192.168.250.224:5060>
Allow: INVITE,ACK,CANCEL,BYE,OPTIONS,INFO,REFER,NOTIFY,PRACK,UPDATE
User-Agent: NewRockTech-MX8-v1.9.82.333/UA 2.0 || The user agent represents an end system. The User-Agent header field contains information about the User Agent Client originating the request.
Content-Type: application/sdp
Content-Length: 321
v=0
o=- 1418191147 820678 IN IP4 192.168.250.224 s=-
c=IN IP4 192.168.2.218 t=0 0
m=audio 10016 **RTP/AVP 18 0 4 8 97 101** || Can use the following codec: 18 for G729; 0 for PCMU or G711U; 4 for G723; 8 for PCMA or G711A; 97 for iLBC; 101 for DTMF method
a=ptime:20
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:97 iLBC/8000
a=fmtp:97 mode=30
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

SIP Protocol Analysis on Log Level 4 (For 100 trying)

[12/10 13:59:08.135866]SIP:recv[192.168.2.218:5060] len[332] || received destination response

SIP/2.0 **100 Trying** || This response indicates that the INVITE has been received and that the proxy is working on behalf of the source to route the INVITE to the destination.

Via: SIP/2.0/UDP 192.168.2.218:5060;rport;received=192.168.250.224;branch=z9hG4bK--1062731046141819114788

To: <sip:234@192.168.2.218>;tag=14181911471418190163-1;local=c007

From: <sip:143@192.168.2.218>;tag=141819114787-1

Call-ID: 1418191147866553213-0@192.168.2.218

CSeq: 100080 INVITE

Content-Length: 0

SIP Protocol Analysis on Log Level 4 (For 180 Ringing)

[12/10 113:59:09.652253]SIP:recv[192.168.2.218:5060] en[372] || Received destination information

SIP/2.0 **180 Ringing** ||This response may be used to initiate local ringback

Via: SIP/2.0/UDP 192.168.2.218:5060;rport;received=192.168.250.224;branch=z9hG4bK--1062731046141819114788

To: <sip:234@192.168.2.218>;tag=14181911471418190163-1;local=c007

From: <sip:143@192.168.2.218>;tag=141819114787-1

Call-ID: 1418191147866553213-0@192.168.2.218

CSeq: 100080 INVITE

Contact: <sip:234@192.168.2.218:5060>

Content-Length: 0

SIP Protocol Analysis on Log Level 4 (For 200 OK)

[12/10 13:59:11.329202]SIP:recv[192.168.2.218:5060] len[614]

SIP/2.0 **200 OK** || Received this response which indicates the request has succeeded

Via: SIP/2.0/UDP 192.168.2.218:5060;rport;received=192.168.250.224;branch=z9hG4bK--1062731046141819114788

To: <sip:234@192.168.2.218>;tag=14181911471418190163-1;local=c007

From: <sip:143@192.168.2.218>;tag=141819114787-1

Call-ID: 1418191147866553213-0@192.168.2.218

CSeq: 100080 INVITE

Contact: <sip:234@192.168.2.218:5060>

Content-Type: application/sdp

Content-Length: 214

v=0

o=- 1418191151 202024 IN IP4 192.168.2.218

s=-

c=IN IP4 192.168.2.218

t=0 0

m=audio 10024 RTP/AVP 18 101

a=ptime:20

a=rtpmap:18 G729/8000 || This indicates that the other end has decided to use codec G729

a=sendrecv

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

SIP Protocol Analysis on Log Level 4 (For ACK)

[12/10 13:59:11.402431]SIP:send[192.168.2.218:5060] len[336] || After receiving 200 OK response the source will send a message back to destination to let it know that the response has been received.

ACK sip:234@192.168.2.218:5060 SIP/2.0 || This indicates an acknowledgement message to confirm the reception of the final response [200(OK)]. An ACK is only sent in response to a response to an INVITE request.

Via: SIP/2.0/UDP 192.168.2.218:5060;branch=z9hG4bK--
1062731046141819115191

To: <sip:234@192.168.2.218>;tag=14181911471418190163-1;local=c007

From: <sip:143@192.168.2.218>;tag=141819114787-1

Call-ID: 1418191147866553213-0@192.168.2.218

CSeq: 100080 ACK

Max-Forwards: 70

Content-Length: 0

SIP Protocol Analysis on Log Level 4 (For BYE)

[12/10 13:59:24.097669]**LINE-143(2) onhook** || **The calling party on-hook**

[12/10 13:59:24.180263]SIP:**send[192.168.2.218:5060]** len[317] || **It will send bye information to destination system**

BYE sip:143@192.168.2.218:5060 SIP/2.0

Via: SIP/2.0/UDP 192.168.250.224:5060;branch=z9hG4bK--1062731046141819116492

To: <sip:143@127.0.0.1>;tag=14181911481418190169-1

From: <sip:234@127.0.0.1>;tag=141819114989-1

Call-ID: 14181911481341771996-0@Officium10

CSeq: 100101 BYE

Max-Forwards: 70

Content-Length: 0

SIP Protocol Analysis on Log Level 4 (200OK for BYE)

[12/10 13:59:24.322854]SIP:recv[192.168.2.218:5060] len[314]
SIP/2.0 **200 OK** || This response confirms receipt of the BYE
Via: SIP/2.0/UDP 192.168.250.224:5060;branch=z9hG4bK--
1062731046141819116492
To: <sip:143@127.0.0.1>;tag=14181911481418190169-1
From: <sip:234@127.0.0.1>;tag=141819114989-1
Call-ID: 14181911481341771996-0@Officium10
CSeq: 100101 BYE || This message indicates the BYE request method
Contact: <sip:143@192.168.2.218:5060>
Content-Length: 0

User Action on Caller ID Display Issue

CID Display Issue	User Action
Can detect Caller ID, but the Caller ID is not complete. Cause: PSTN line signal is too strong or weak.	1. Log in MX8G/MX60E/MX120G 2. Then input the xml command [to adjust parameter: FXO_CID_GAIN (ID341)] on the URL: http://x.x.x.x/xml?method=gw.config.set&id341=9 or http://x.x.x.x/xml?method=gw.config.set&id341=11 Note: FXO_CID_GAIN (ID341) default value is 10; If signal too strong adjust to 9, if too weak adjust to 11.
Caller ID display “P”	1. Contact PSTN operators to solve the issue since “P” indicates that PSTN operators hide the Caller ID
Call from FXO incoming can not display Caller ID (Caller ID time-out)	1. Go to Advanced >Trunk > CID Detection Mode 2. Set the mode to Before Ringing A 3. Check if PSTN supports this function 4. If PSTN supported but still display “time-out” , then perform TDM capture on FXO port: Go to Tools > TDM Capture 3. Select the line ID (FXO port) to which you want to perform the capture. 4. Click Start to initiate the capture procedure. 5. Make the test call. 6. Click Stop to terminate the capture procedure. Then you will be notified for download.

User Action on 60s call issue

60s call issue	User Action
Each outgoing call will be cut off after 60 seconds. Cause: Enable polarity reversed signal detection for FXO port, but the PSTN line does not support this function.	1. Disable polarity reversed signal detection on MX8G/MX60E/MX120G . Go to Line > Trunk 2. Can be verified on the log, that the call cannot received ACK message.

Analysis on Log Level 4:

|| When received destination information 127.0.0.1:7890 200 OK

01/01 08:19:39.142260]SIP:recv[127.0.0.1:7890] len[591]

SIP/2.0 **200 OK**

Via:SIP/2.0/UDP192.168.186.177:7890;received=127.0.0.1;branch=z9hG4bK—
13131671681175226

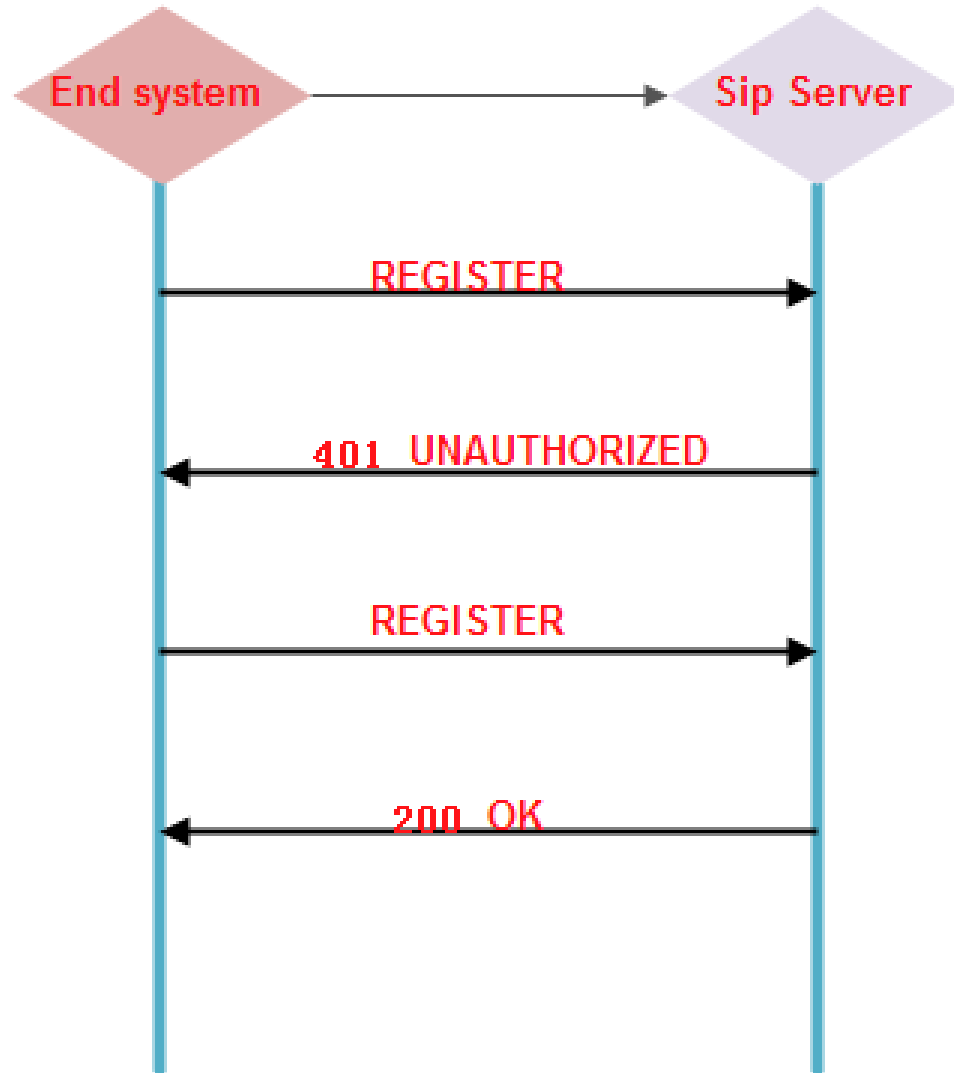
|| Then send 59.124.80.61:7890 to ACK

[01/01 08:19:39.189721]SIP:send[59.124.80.61:7890] len[327]

ACK sip:7003@59.124.80.61:7890 SIP/2.0

|| But the 127.0.0.1:7890 can not received the ACK message, so the call will be cut off after 60s.

SIP Basic Register Process (RFC3261)



Register Message Analysis on Log Level 8

[12/11 10:08:36.419279]SIP:**send[192.168.8.89:8888]** len[435] || **Send register message to sip server**

REGISTER sip:192.168.8.89:8888 SIP/2.0 || **This header field is Register**

Via: SIP/2.0/UDP 192.168.250.224:5060;rport;branch=z9hG4bK--
106266755214182637161418263620

To: <sip:232@192.168.8.89:8888>

From: <sip:232@192.168.8.89:8888>;tag=14182637161418263619-1

Call-ID: 1418263716835237579-0@192.168.250.224

CSeq: 1 REGISTER

Max-Forwards: 70

Expires: 600 || **The registration will expires after 10 minutes**

Contact: <sip:232@192.168.250.224:5060>

User-Agent: NewRockTech-MX8-v1.9.82.333/UA 2.0

Content-Length: 0

Register Message Analysis on Log Level 8

[12/11 10:08:36.469332]SIP:recv[192.168.8.89:8888] len[427] || Rec
SIP/2.0 **401 Unauthorized** || SIP server needs end user to re-send register
message to him again and must bring an authenticated message. The request requires
user authentication
Via: SIP/2.0/UDP 192.168.250.224:5060;rport=5060;branch=z9hG4bK--
106266755214182637161418263620
To: <sip:232@192.168.8.89:8888>;tag=336d3577
From: <sip:232@192.168.8.89:8888>;tag=14182637161418263619-1
Call-ID: 1418263716835237579-0@192.168.250.224
CSeq: 1 REGISTER
WWW-Authenticate: Digest
algorithm=MD5,nonce="613EC12B4B10C01C67307DB6681073BB57175E3A",realm="ne
wrocktech" || This is the message content information for sending second
register request
Content-Length: 0

Register Message Analysis on Log Level 8

[12/11 10:08:36.565061]SIP:**send[192.168.8.89:8888]** len[628]

REGISTER sip:192.168.8.89:8888 SIP/2.0 || **Thus, send second register request to SIP server and also brought authorization message**

Via: SIP/2.0/UDP 192.168.250.224:5060;rport;branch=z9hG4bK--

106266755214182637161418263621

To: <sip:232@192.168.8.89:8888>

From: <sip:232@192.168.8.89:8888>;tag=14182637161418263619-1

Call-ID: 1418263716835237579-0@192.168.250.224

CSeq: 2 REGISTER

Max-Forwards: 70

Expires: 600

Contact: <sip:232@192.168.250.224:5060>

Authorization: Digest

algorithm=MD5,nonce="613EC12B4B10C01C67307DB6681073BB57175E3A",realm="newrocktech",response="6889a89d7fe41af72a23eaaec949c45f",uri="sip:192.168.8.89:8888",user name="232"

User-Agent: NewRockTech-MX8-v1.9.82.333/UA 2.0

Content-Length: 0

Register Message Analysis on Log Level 8

[12/11 10:08:36.665067]SIP:recv[192.168.8.89:8888] len[427]
SIP/2.0 200 OK || Received the SIP server 200 OK indicates the register is successful
Via: SIP/2.0/UDP 192.168.250.224:5060;rport=5060;branch=z9hG4bK--
106266755214182637891418263636
To: <sip:232@192.168.8.89:8888>;tag=aee05745
From: <sip:232@192.168.8.89:8888>;tag=14182637891418263634-1
Date: Thu, 11 Dec 2014 02:09:56 GMT
Call-ID: 1418263716835237579-0@192.168.250.224
CSeq: 2 REGISTER
Server: OM80(Rev 2.1.5.85)
Contact: <sip:232@192.168.250.224:5060>;expires=600
Content-Length: 0

User Action on SIP Registration Issue

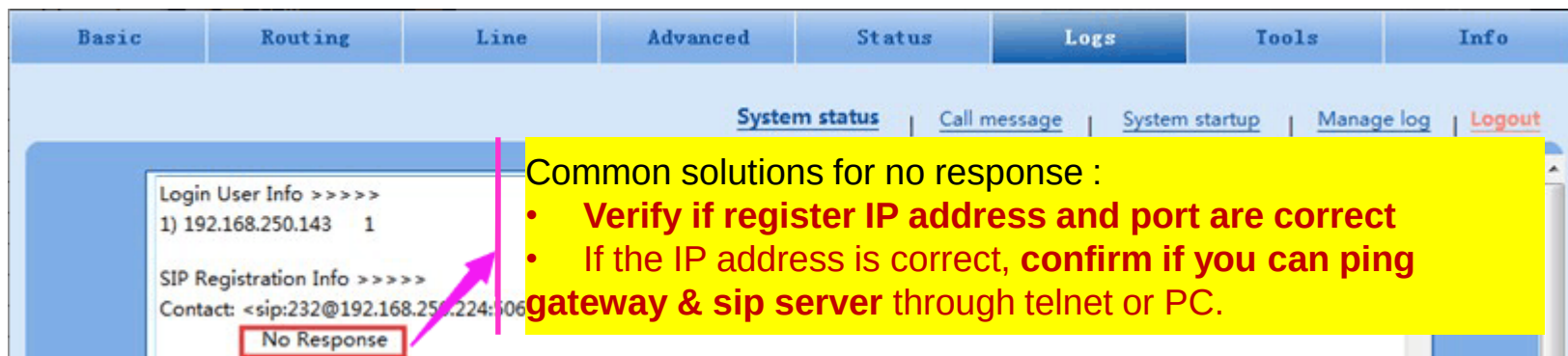
Click Logs > System status



The screenshot shows the 'System status' page with a tabbed interface. The 'Logs' tab is selected, and the 'System status' sub-tab is active. The 'Login User Info' section shows '1) 192.168.250.143 1'. The 'SIP Registration Info' section shows 'Contact: <sip:232@192.168.250.224:5060>' and a red box highlights 'response: 403'. A yellow callout box contains the following text:

Common solutions for response 403 or 404:

- **Verify if registration account is correct**
- **Verify if registration account password is correct**



The screenshot shows the 'System status' page with a tabbed interface. The 'Logs' tab is selected, and the 'System status' sub-tab is active. The 'Login User Info' section shows '1) 192.168.250.143 1'. The 'SIP Registration Info' section shows 'Contact: <sip:232@192.168.250.224:5060>' and a red box highlights 'No Response'. A yellow callout box contains the following text:

Common solutions for no response :

- **Verify if register IP address and port are correct**
- **If the IP address is correct, confirm if you can ping gateway & sip server through telnet or PC.**

Sound Quality Problems

Noise

- Check the grounding is good
- Check the wiring is good
- Check the installation environments for any potential sources of interference
- Check whether the circuit gain is set too high

Echo

- Check the wiring is good
- Check if the echo cancellation is enabled
- Check whether the circuit gain is set too high
- Check the line matches the impedance setting

DTMF Sending & Detection Problem

Send DTMF to softswitch

- Soft switch does not correctly receive DTMF signals from gateway
- **setting: *Basic***
> *System*
DTMF transmission mode is set same with soft switch

Detect DTMF from PSTN(1)

- PSTN call is in progress and gateway does not send DTMF to the softswitch
- Check whether FXO received DTMF signals and the DTMF signals are distorted.

Detect DTMF from PSTN(2)

- FXO port signals is non-standard DTMF signals (such as signal is too short)
- **Setting: *Basic***
> *System*,
adjusted DTMF detection threshold according to the actual received signal

FXO send DTMF to PSTN

- PBX, CO needs DTMF lengths and spacing of FXO gateway adjustment issue
- **Setting: *Basic***
> *System* ,
adjust gateway DTMF signal sending interval and hold time

FXO Not Detect Caller ID

Step 1

- Check whether the PSTN is enabled caller ID.
- Connecting the PSTN to an analog phone, make a incoming call to determine whether the PSTN do provide caller ID.

Step 2

- ***Line > Trunk*** check whether caller ID detection feature of the FXO port is turned on.

Step 3

- After the first 2 steps, if it still does not detect the caller ID, may try to enter ***Advanced > Trunk***, change caller ID and ring detection mode, then watch whether is able to detect Caller ID.

Step 4

- If the three steps mentioned above are invalid, can enter ***Tools > TDM Capture*** to capture PSTN signal to analysis.

FXO Port Busy Tone Detection

Analysis of busy tone characteristic

- Busy tone standards are different in different countries, if busy tone detection parameters set wrong, will cause PSTN lines could not be released
- By capturing the PSTN signal to see the busy tone characteristic, such as a busy tone frequency and on/off time.

Adjust

- After getting the PSTN busy tone parameters, enter ***Advance > Trunk***, Busy tone detection parameters (on/off) is set according to the PSTN actually signal, if there is a dual frequency of busy tone, can input with either one frequency.

How to Use Data Capture

Tools > Ethereal Capture, click ***Start***, make a test call, after end the call, click ***Stop***, the system will automatically package and compress a captured packet files, then download the file to analysis.

Ethereal capture

Start

Stop

How to Capture PSTN Signal

Enter **Tools > TDM Capture**, Select a FXO port by typing line # into Input box, click **Start**, then make a test incoming call to this FXO port and talk something, after finish the call, click **Stop**, the system will automatically download a capture file with extension as .PCM.

TDM capture	
Line ID	8000  ID: FXS-1,Number: 8000
	Start Stop

Tips

1. Capture caller ID signal: make a incoming call from PSTN to FXO port, and wait about 10 seconds and click stop button, don't need to pickup the phone in this case.
2. Capturing busy tone signal: make a incoming call from PSTN to FXO port, you need to connect, then PSTN side on hook, wait a while for PSTN playing busy tone, then click stop button to end the test.

Communication without Limits



Thank you!