eSIP Evolution Series

Administrator Guide

Contents

About This Guide

Thanks for choosing ESI's eSIP Evolution Series VoIP system. This guide is intended for Installing Technicians and Administrators who need to prepare for, configure, and operate their eSIP Evolution Series server. In this guide, we describe every detail on the functionality and configuration of the server. We begin by assuming that you are interested in the eSIP Evolution Series VoIP system and are familiar with networking and other IT disciplines.

Products Covered

This guide explains how to configure the following products:

- ESI 20x
- ESI 50x
- ESI 200x
- ESI 500x

Related Documents

The following related documents are available on ESI's website: www.esiacademy.com.

Document	Description
ESI eSIP Evolution Series Datasheet	Datasheet for the ESI eSIP Evolution Series.
ESI eSIP Evolution Series Hardware Installation Manual	Installation Manual for all models of the ESI eSIP Evolution Series.
ESI eSIP Evolution Series Extension User Guide	Users should refer to the manual for instructions on how to login to the user portal, how to configure their accounts, listen to call recordings, check voicemail messages, etc.
ESI eSIP Evolution Series LDAP Server Guide	Administrator's guide for configuring the internal LDAP Server.
ESI eSIP Evolution Series Auto Provisioning Guide	Administrator's guide for using the internal Auto Provisioning feature.
ESI eSIP Evolution Series VPN Server Guide	Administrator's guide for configuring the internal VPN Server.
ESI eSIP Evolution Series QueueMetrics Integration App Guide	Administrator's guide for integrating with QueueMetrics Live
ESI eSIP Evolution Series First Login and Installation Wizard Guide	Installing Technician/Administrator's walk-through of the basic, initial system configuration.

Safety when working with electricity

- Do not use a 3rd party power adaptor.
- Do not power on the device during the installation.
- Do not work on the device, connect or disconnect cables when lightning strikes.

eSIP Evolution Series Overview

This chapter provides the following sections:

- Introduction
- Feature Highlights
- Expansion Board
- Hardware Overview

Introduction

Designed with the small and medium sized enterprises in mind, supporting up to 500 users and built using the very latest technology, the ESI eSIP Evolution Series delivers exceptional cost savings, productivity and efficiency improvements, delivering power, performance, quality and peace of mind.

The all new eSIP Evolution Series is engineered for the communications needs of today and tomorrow, and with the unique modular design, future proofs your investment choice.

Feature Highlights

Appreciate the Easy-to-use Solution

- Intuitive graphical user interface brings point-and-click configuration.
- Convenient Phone Provisioning feature saves you tremendous time.
- Everything can be managed from anywhere with Internet access.

Your Choice of Technologies and Features

- Embedded VoIP capability and analog phone connections.
- External trunking options include SIP, PSTN, ISDN, T1/PRI, and Cellular.
- Concurrent calls and maximum users are expandable with modules.
- Integrated Utility features that are already there when you need them.

Telephone System without Risk

- Meanwell power supply featuring MTBF>560Kh.
- High-quality Freescale CPU processor and industry leading TI DSP voice processor.
- Connectors from TE Connectivity with a gold plating layer as thick as 15 μ.
- Lightning protection on analog ports complying with ITU-T K.20/45/59021 8/20 μs and GR-1089 standard.

Expect Security and Reliability

- TLS, SRTP, and HTTPS standards for better security.
- Defend against malicious attacks with built-in Firewall and intrusion detection.
- Monitor system status and behavior and be notified when abnormalities occur.

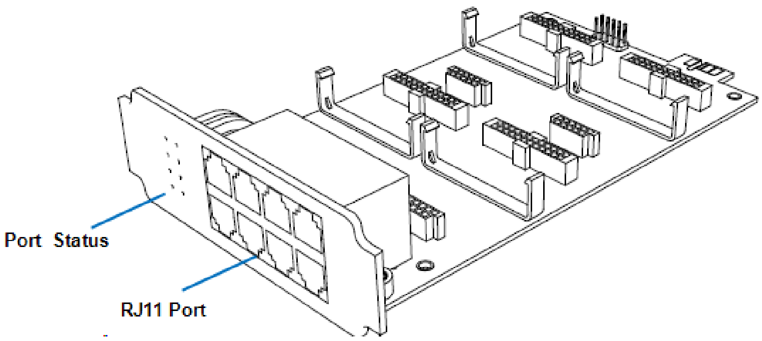
Expansion Boards

The eSIP 200x and 500x are expandable.

- 200x supports up to 2 EXP04/EXPT1PRI Expansion Spans; supports 1 EXP100 Module.
- 500x supports up to 3 EXP04/EXPT1PRI Expansion Spans; supports up to 2 EXP100 Modules.

Expansion Board - EXP04

EXP04 board supports up to 4 modules (8 ports).

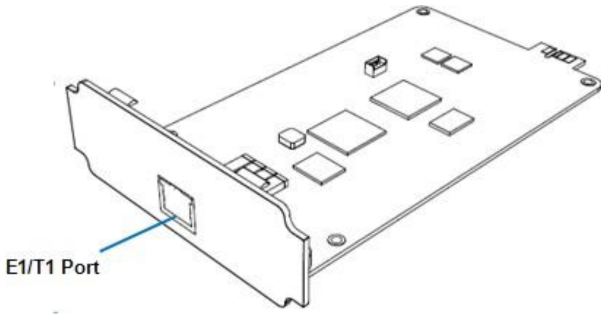


Optional Modules

- 2FXO Module
- 2FXS Module
- FXOFXS Module
- 3G Module
- 4G/LTE Module

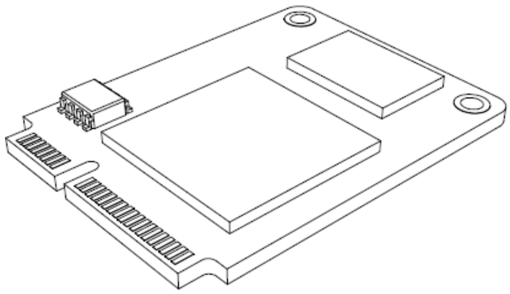
Expansion Board - EXPT1PRI

EXPT1PRI board supports 1 T1/PRI port.



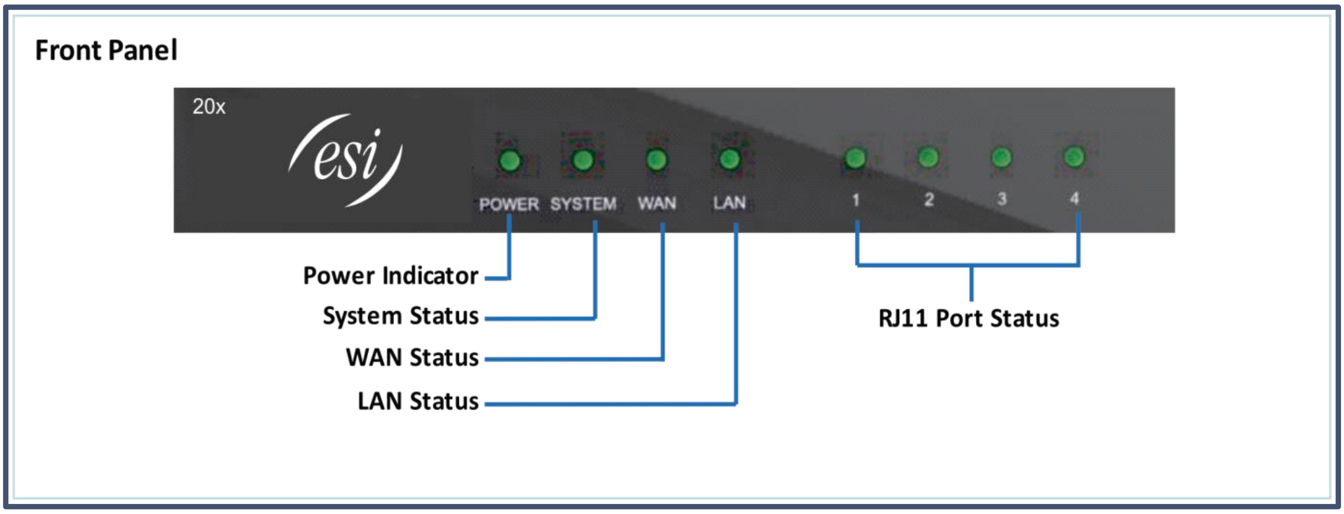
EXP100 Module

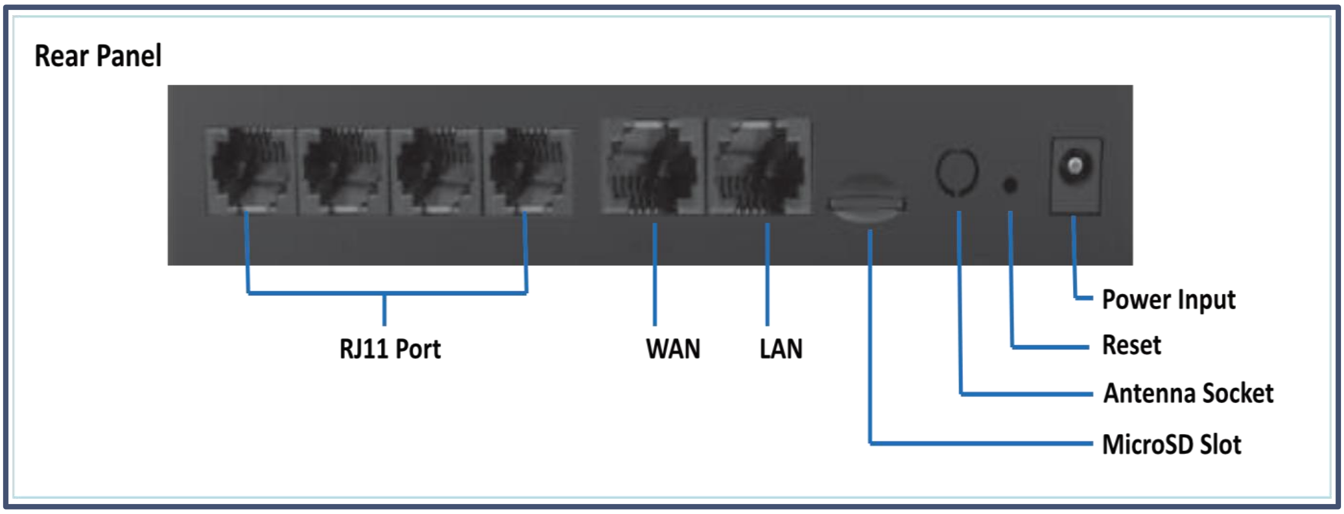
EXP100 is a DSP module, used to expand the capability of the 200x and 500x. With each additional EXP100 module, the max extensions increases by 100 and max concurrent calls increases by 30.



Hardware Overview

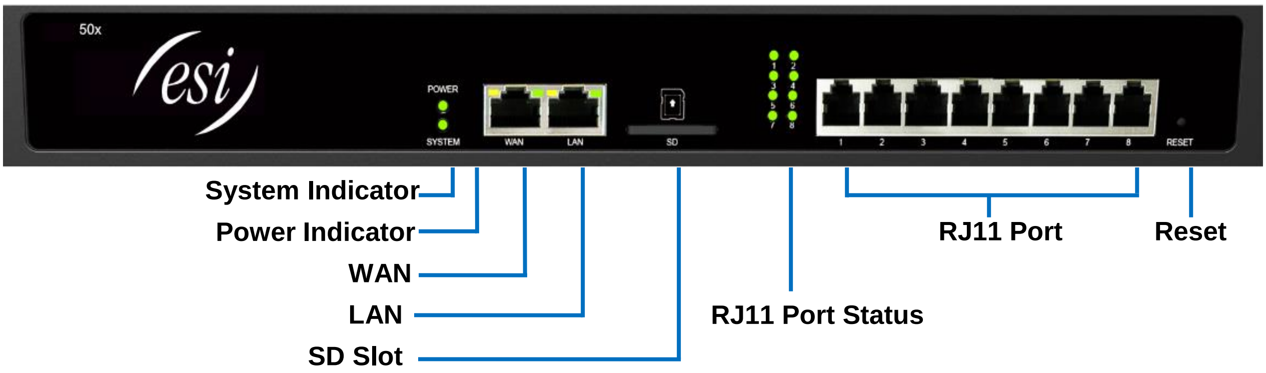
eSIP 20x



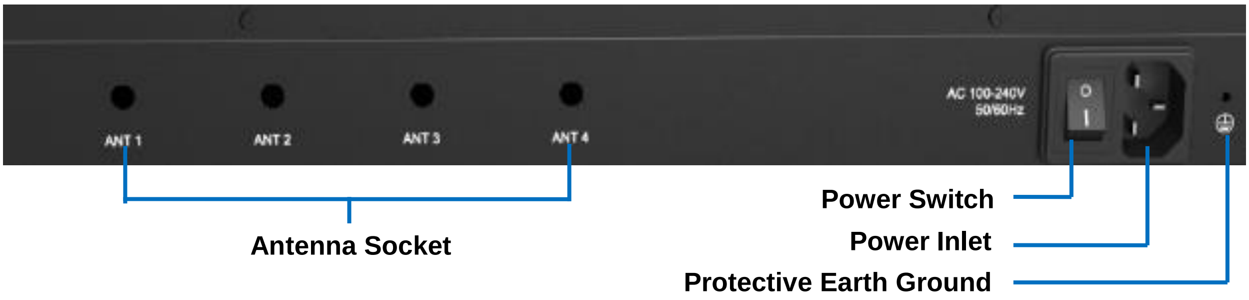


eSIP 50x

Front Panel



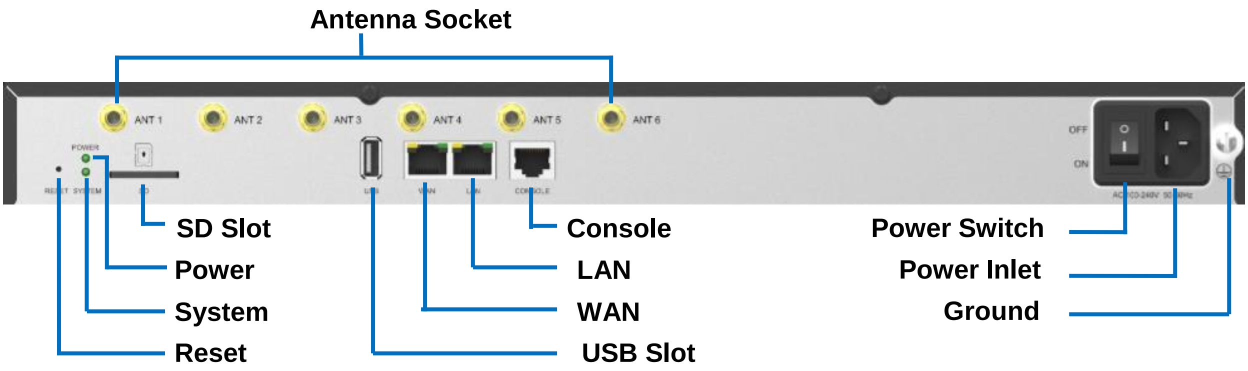
Rear Panel



eSIP 200x

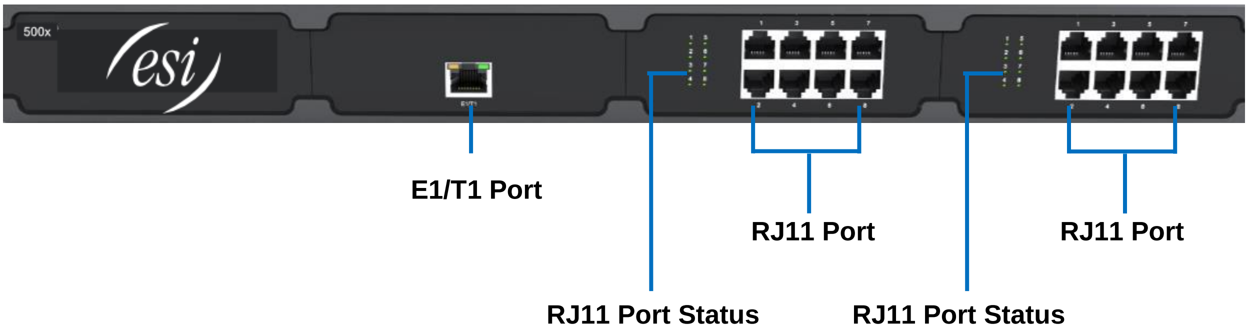
Front Panel (1*EXPT1PRI + 1*EXP04)

Rear Panel

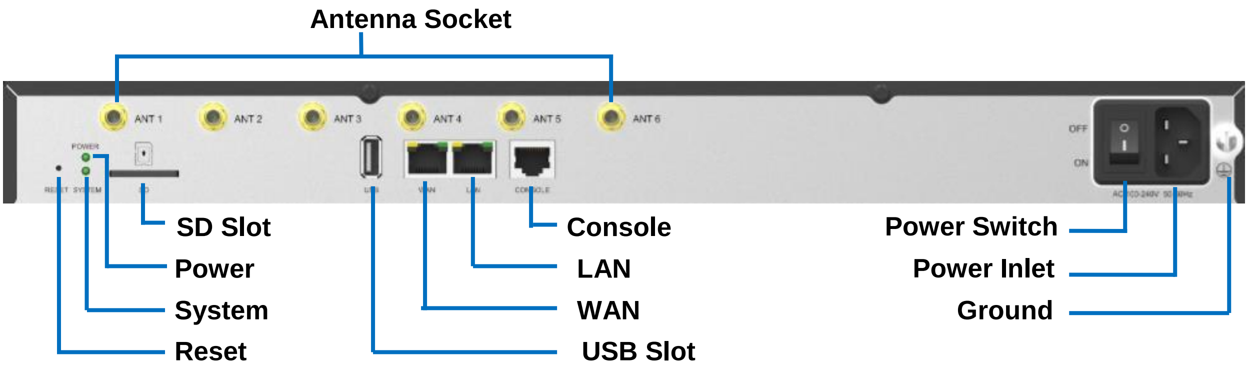


eSIP 500x

Front Panel (1*EXPT1PRI +



Rear Panel



LED Indicators and Ports

LED Indicators

LED	Indication	Status	Description
POWER	Power status	On	The power is switched on
		Off	The power is switched off
System	System status	Blinking	The system is running properly
		Static/Off	The system goes wrong
WAN	WAN status	Static Green light	Linked normally, 10/100 Mbps.
		Static Orange light	Linked normally, 1000 Mbps.
		Blinking	In communication.
		Off	Off-line.
LAN	LAN status	Static Green light	Linked normally, 10/100 Mbps.
		Static Orange light	Linked normally, 1000 Mbps.
		Blinking	In communication.
		Off	Off-line.
RJ11 Port Status	FXS	Green light	• Static: The port is idle. • Blinking: There is an ongoing call on the port.
	Cellular/4G	Red light	• Static: the trunk is idle. • Blinking slowly: there is no SIM card inserted. • Blinking rapidly: the trunk is in use.
	FXO	Red light	• Blinking slowly: no PSTN line is connected to the port. • Static: the PSTN line is idle. • Blinking rapidly: the PSTN line is busy.

Port Description

Ports	Description
RJ11 Port	**FXO port** (red light): for the connection of PSTN lines or FXS ports of traditional servers. **FXS port** (green light): for the connection of analog phones. **Note:** the sequence number of the ports corresponds to that of the Indicator lights in the front panel. (I.e. the LED lights in the front indicate the connection status of the corresponding ports at the front panel.)
ANT	Connect to 3G/4G Antenna.
T1/PRI	Connect to T1 or PRI.
Console	Connect to the RS-232 Cable to debug to system.
MicroSD/TF Slot	Insert MicroSD/TF card.
SD Slot	Insert SD card.
USB Slot	Connect to USB external disk.
Ethernet Port	The eSIP 20x provides two 10/100M adaptive RJ45 Ethernet ports, 50x/200x/500x supports two 10/100/1000M Ethernet ports. There are 3 Ethernet modes for the system. The default mode is “Bridge”. • **Bridge:** LAN port interface will be used for uplink connection. WAN port interface will be used as bridge for PC connection. • **Dual:** both ports can be used for uplink connection. • **LAN Only:** LAN port interface will be used for uplink connection. WAN port interface will be disabled.
Reset Button	Press and hold for 10 seconds to restore the factory defaults
Power Inlet	Connect the supplied power supply to the port.
Power Switch	Toggle this switch to switch on/off the device.

Getting Started

This chapter explains how to log in to the eSIP Evolution Series Web GUI, use the taskbar and widgets, and open applications with the Main Menu.

- Accessing Web GUI
- Web Configuration Desktop
- Make Your First Call

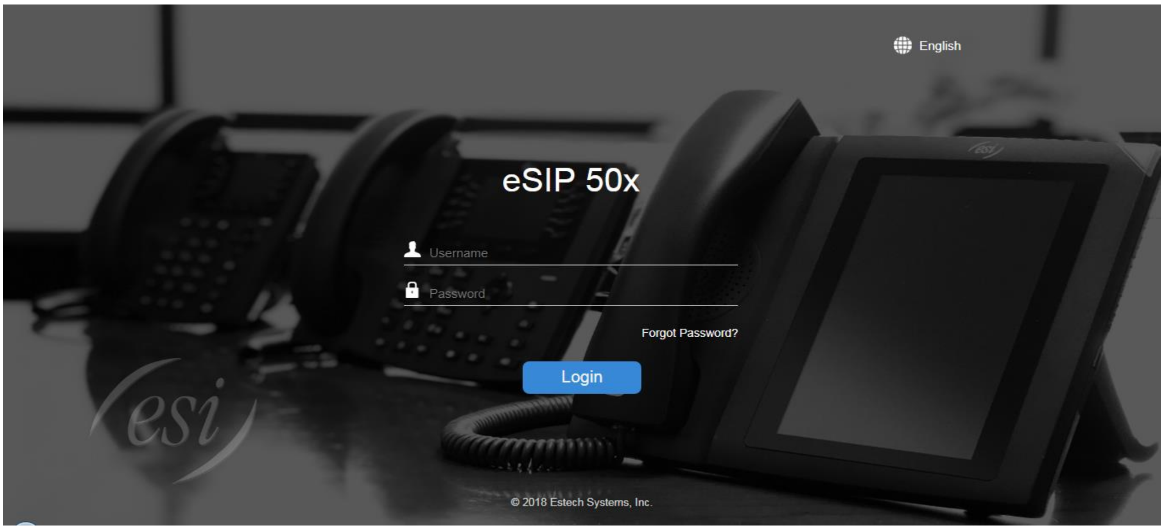
Accessing Web GUI

ESI's eSIP Evolution Series provides a web-based configuration interface for the installer, administrator, and extension users. The Installer/Administrator can manage the device by logging in to the Web interface. See the factory defaults below:

- IP address: http://192.168.5.150:59088
- User Name: admin
- Default Password: esi789*#

To log in to the eSIP Evolution Series

1. Make sure your computer is connected to the same network as the server.
   a. **Note**: An easy way to accomplish this is statically assign the Ethernet adapter on your PC to 192.168.5.151.
2. Open a web browser on your PC, enter the IP address, press **Enter** on your keyboard.
3. Enter your user name and password, click **Login**.



eSIP Evolution Series Web Configuration Panel Login Page

Note: To ensure your connection to the eSIP Evolution Series Web GUI runs smoothly, please use the following browsers:

- **Chrome**
- **Firefox**
- **Internet Explorer** 11.0 or later

Web Configuration Desktop

When you first log in to the ESI eSIP Evolution Series Web GUI, you will be walked through some initial configuration steps. For more details, please refer to the ***eSIP Evolution Series First Login and Installation Wizard Guide***. Once you have completed the Installation Wizard, you will see the Desktop below. From here, you can manage settings, applications, and view system resource information.

Desktop

The desktop is where your application windows are displayed.

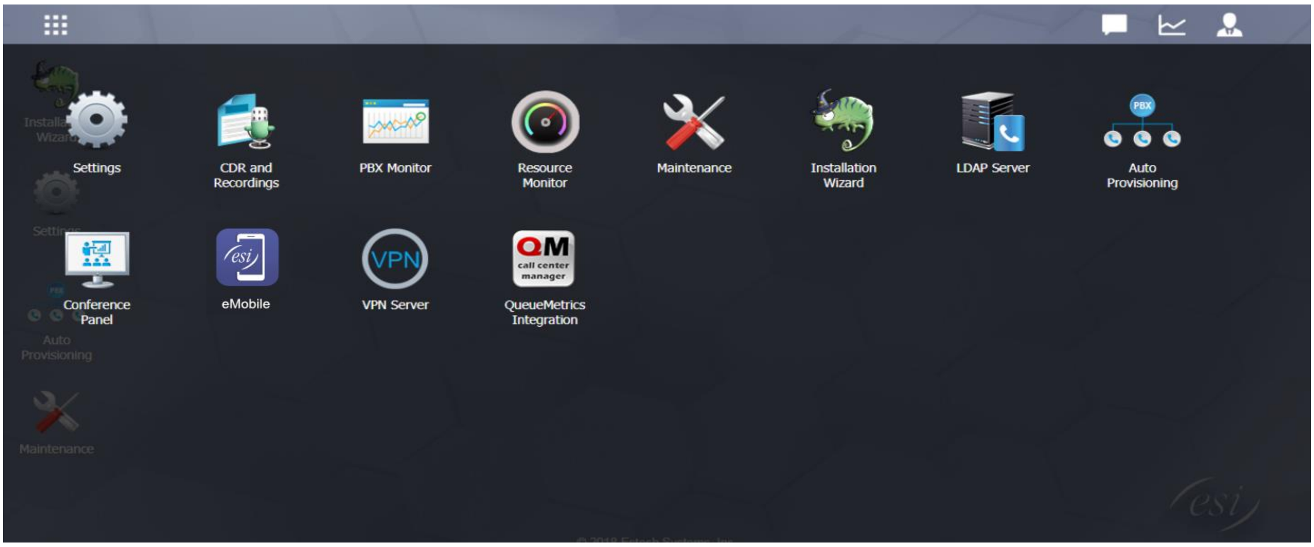
Taskbar

The taskbar at the top of the desktop includes the following items:

1. **Main Menu:** view and open applications installed on your eSIP Evolution Series system. Right-click an application icon and you can add the application to the desktop for easy access.
2. **Open Application**
3. Click the icon of an application to show or hide its window on the desktop.
4. Right-click the icon and choose from the shortcut menu to manage the application window (**Maximize**, **Minimize**, **Restore**, **Close**).
5. **Notifications:** displays notifications, like errors, status updates, and app installation notifications.
6. **Resource Monitor:** click the icon to check the system information, network status and storage usage.
7. **Options**: logout, change Web language or modify personal account options.

Main Menu

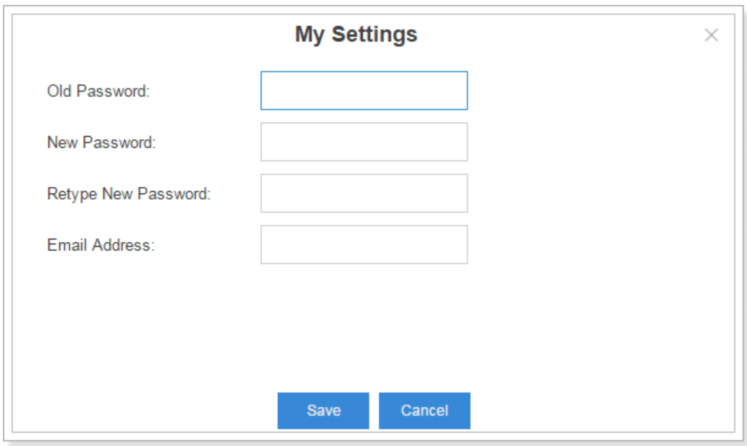
Click the **Main Menu** at the top-left of the desktop, you can find all of the installed applications on your eSIP Evolution Series system.



Options

Click the options icon  to logout, change Web language or modify your account settings.

- **Language**
  Select Language to change the web language.
- **My Settings**
  Click My Settings to modify your account settings. Here you can change the login password and configure your email address on the account.
- **Logout**
  Click Logout to log out the Web GUI.



Save and Apply Changes

Click **Save** button after your configurations on the eSIP Evolution Series system, do not forget to click **Apply** button on the upper right of the desktop to submit all the changes. If the change requires a reboot to take effect, the system will prompt you with a pop-up window.

Make Your First Call

Connect your phone and the eSIP Evolution Series device to the same network. Then register an extension to the IP phone and make your first call through the eSIP Evolution Series system.

1. Log in your eSIP Evolution Series Web GUI, go to **Settings > PBX > Extensions**.
2. Click **Add** to create a new extension, set the type as “SIP”. You will need the Registration Name and Registration Password to register the extension later.
3. Register the extension on your phone with the Registration Name and Registration Password, the SIP server address is your eSIP Evolution Series IP address.
4. When the extensions is registered to eSIP Evolution Series, you can dial *2 to access your voicemail box. The default password to enter the voicemail box is your extension number.
5. Once entering the voicemail box, you are connected to the eSIP Evolution Series system!

Note: Hopefully, you've followed the ***eSIP Evolution Series First Login and Installation Wizard Guide*** and you've made your first call already. If you haven't, please refer to that guide.

System Settings

This chapter explains system settings on eSIP Evolution Series. Go to **Settings > System** to check the system settings.

- Network
- Security
- User Permission
- Date & Time
- Email
- Storage

Network

After successfully logging in to the eSIP Evolution Series Web GUI for the first time with the factory IP address, users could go to **Settings > System > Network** to configure the network for eSIP Evolution Series.

The eSIP Evolution Series supports 3 Ethernet modes: Single, Dual and Bridge.

Basic Settings

Please check the basic network settings below.

Basic Settings	
Hostname	Set the hostname for the system.
Mode	Select the Ethernet mode. The default mode is Single. • Single: only LAN port will be used for uplink, WAN port is disabled. • Bridge: LAN port interface will be used for uplink connection. WAN port interface will be used as bridge for PC connection. • Dual: the two Ethernet interfaces will use different IP addresses. Assign two IP addresses in this mode.
Default Interface	In Dual mode, you need to choose the default interface.
LAN/WAN Settings (DHCP Mode)	
If you choose this mode, the system will act as DHCP client to get an available IP address from your local network.	
LAN/WAN Settings (Static IP Address)	
IP Address	Enter the IP address (xxx.xxx.xxx.xxx).
Subnet Mask	Enter the subnet mask (xxx.xxx.xxx.xxx). For example, 255.255.255.0
Gateway	Enter the gateway address (xxx.xxx.xxx.xxx).
Preferred DNS Server	Enter the IP address of the preferred DNS server (xxx.xxx.xxx.xxx).
Alternate DNS Server	Enter the IP address of the alternative DNS server (xxx.xxx.xxx.xxx).
LAN/WAN Settings (PPPoE)	
Username	Enter the PPPoE username.
Password	Enter the PPPoE password.
VLAN	
Enable VLAN	Check this option to enable VLAN.
VLAN ID	Enter the VLAN ID.
VLAN Priority	Set the VLAN priority. The default is 0.

OpenVPN

The eSIP Evolution Series supports OpenVPN. The system provides detailed VPN configurations on the Web GUI and you can also upload the VPN configuration package to the system to make it work.

Before using OpenVPN feature, please Enable OpenVPN first, then choose the Type to configure OpenVPN:

- Manual Configuration
- Upload OpenVPN Package

Check the VPN configurations parameters below.

OpenVPN Configuration	
Server Address	Enter the server address of OpenVPN.
Server Port	Enter the server port of OpenVPN. The default is **1194**.
Protocol	Select the protocol type. The server and client must use the same protocol.
Device Mode	Select the network device. The client and server must use the same setting. • TUN: a TUN device is a virtual point-to-point IP link. • TAP: a TAP device is a virtual Ethernet adapter.
Username	Specify the username.
Password	Specify the password.
Encryption	Select the encryption method. The server and client must use the same setting.
Compression	Enable or disable compression for data stream. The server and client must use the same setting.
Proxy Server	Specify the proxy server.
Proxy Port	Specify the proxy port.
CA Cert	Upload a CA certificate.
Cert	Upload a Client certificate.
Key	Upload a Client key.
TLS Authentication	Enable or disable TLS authentication. If enabled, please upload a TA key via **Settings > System> Security>Certificate**.

DDNS Settings
Dynamic DNS or DDNS is a method of updating, in real time, a Domain Name System (DNS) to point to a changing IP address on the Internet. This is used to provide a persistent domain name for a resource that may change location on the network. DDNS is usually configured on a router. If the router cannot support DDNS, we can set up DDNS on the eSIP Evolution Series.

ESI's eSIP Evolution Series supports the following DDNS servers:

- dyndns.org
- freedns.afraid.org
- www.no-ip.com
- www.zoneedit.com
- www.oray.com
- 3322.org

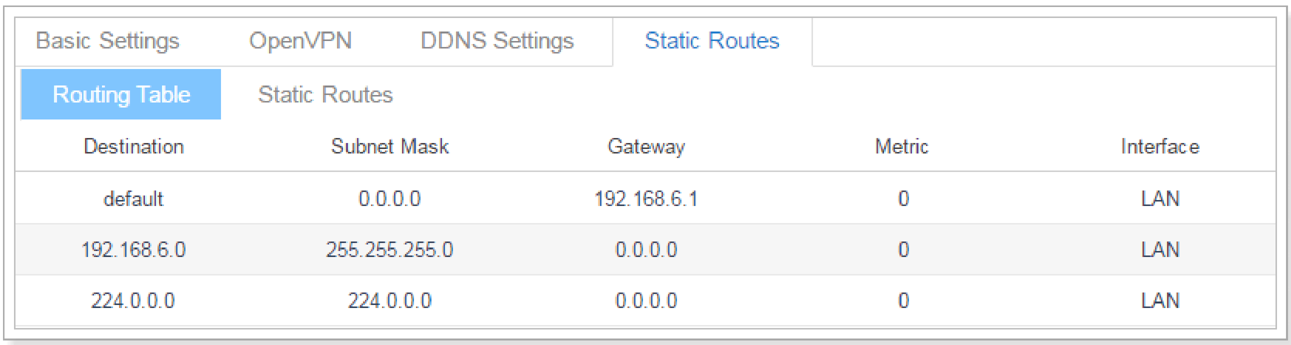
See the DDNS configuration parameters below

DDNS	
DDNS Status	This shows the current DDNS status of the device.
Enable DDNS	Check this box to enable DDNS.
DDNS Server	Choose a DDNS provider from the list.
Username	Enter the username of your DDNS account.
Password	Enter the password of you DDNS account.
Hash	Enter your string of Hash as provided by freedns.afraid.org.
Domain	Enter the domain name.

Static Route

In computer networking, a routing table is a data table stored in a router or a networked device that lists the routes to particular network destinations, and in some cases, metrics (distances) associated with those routes. Static routes are entries made in a routing table by non-automatic means and which are fixed rather than being the result of some network topology “discovery” procedure.
Static route on the system is used to configure the route of the connection/packets to a particular network destination, usually a specific gateway.

Routing Table
All the static routes are displayed on the Routing Table.



Destination	Subnet Mask	Gateway	Metric	Interface
default	0.0.0.0	192.168.6.1	0	LAN
192.168.6.0	255.255.255.0	0.0.0.0	0	LAN
224.0.0.0	224.0.0.0	0.0.0.0	0	LAN

Static Routes
Click Static Routes tab, you can add static routes here. Click
Add to add a static route.

- Click to edit the static route.
- Click to delete the static route.

Check the Static route settings below.

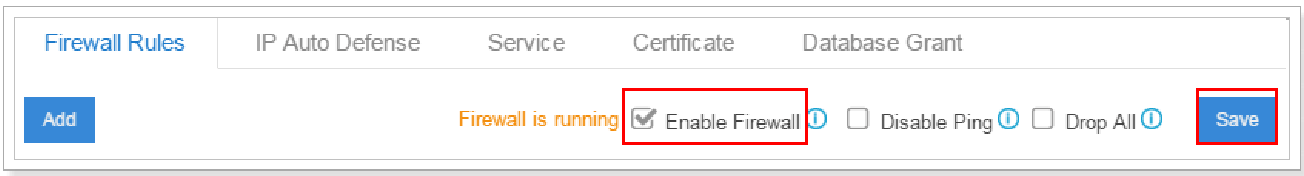
Static Route	
Destination	Enter the destination IP address or IP subnet for the eSIP Evolution Series to reach using the static route. **Example:** • IP address: *192.168.6.120* • IP subnet: *192.168.6.0*
Subnet Mask	Enter the subnet mask for the destination address. **Example:** *255.255.255.255*
Gateway	Enter the gateway address. The eSIP Evolution Series system will reach the destination address via this gateway. **Example:** *192.168.6.1*
Metric	The cost of a route is calculated using what are called routing metric. Routing metrics are assigned to routes by routing protocols to provide measurable values that can be used to judge how useful a route will be.
Interface	Select the network interface. The system will reach the destination address using the static route through the selected network interface.

Security

VoIP attacks, although not an everyday occurrence should always be a consideration. When using VoIP, system security is undoubtedly one of the issues we care about most. With appropriate configuration, and some basic safety habits, we can improve the security of the telephone system. Moreover, the powerful built-in firewall function in the eSIP Evolution Series is adequate to enable the system to run safely and stably.
We strongly recommend that you configure the firewall and other security options to prevent attacks and fraud that may cause system failure, functionality loss, or fraudulent use.

Note: ESI-specific firewall rules are configured by default.

Firewall Rules
Administrators should add rules to accept or reject traffic through the system. Go to **Settings** > **System** > **Security** > **Firewall Rules** to configure firewall for the system. Before adding firewall rules, please check the option **Enable Firewall**, then click **Save** to enable the firewall.



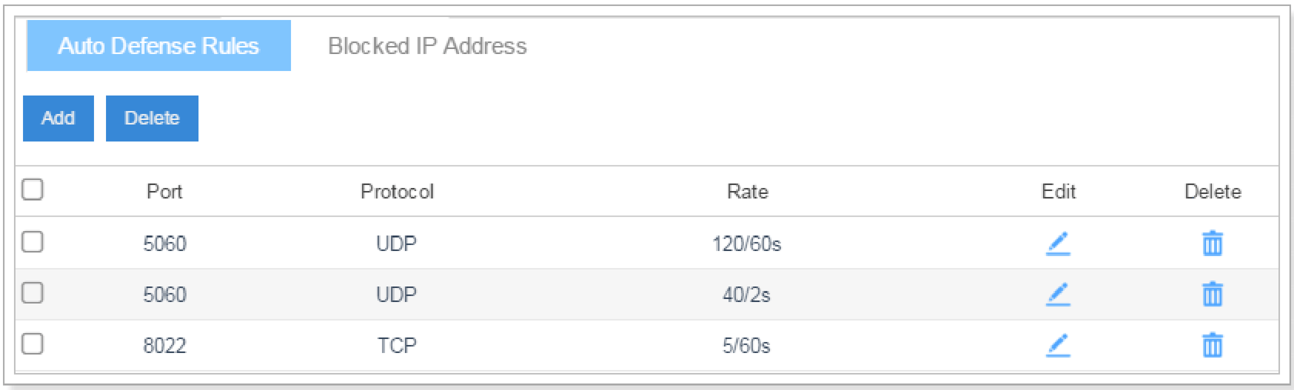
- Click Add to add a new rule.
- Click to edit the rule.
- Click to delete the rule.

Check the firewall configuration parameters below.

Firewall	
Enable Firewall	Enable Firewall to protect the system from malicious attack. Click Save icon to apply the changes.
Disable Ping	Enable this item; net ping from remote hosts will be dropped. Click Save icon to apply the changes.
Drop All	When you enable Drop All feature, the system will drop all packets and connections from other hosts if there are no other rules defined. To avoid locking the device, at least one TCP Accept common rule must be created for port used for SSH access and port used for HTTP access.
Firewall Rules	
Name	Specify a name to identify the firewall rule.
Description	Description for this firewall rule.
Action	Select the action for the firewall rule: • Accept • Ignore • Reject
Protocol	Select the protocol applied for the rule: • UDP • TCP • BOTH
Source IP address/ Subnet mask	The IP address for this rule. **Example:** 192.168.5.100/255.255.255.255 means this rule is for 192.168.5.100. 192.168.5.100/255.255.255.0 is for IP from 192.168.5.0 to 192.168.5.100.
Port	Set the port for the firewall rule. The end port must be equal to or greater than start port.

IP Auto Defense

Administrators should create auto defense rules, and the system will prevent massive connection attempts or brute force attacks. The offending IP addresses would then be listed in the **Blocked IP Address** table. There are 3 default auto defense rules; we recommend you keep the rules there.



Please see the auto defense rule configuration parameters below.

IP Auto Defense Rule	
Port	Auto defense port, for example, 59022.
Protocol	Select auto defense protocol: • UDP • TCP
The Number of IP Packets	The number of IP Packets permitted within a specific time interval.
Time Interval	The time interval to receive IP Packets. For example, Number of IP Packets set to 90 and Time Interval set to 60 mean 90 IP packets are allowed within a 60 second window.

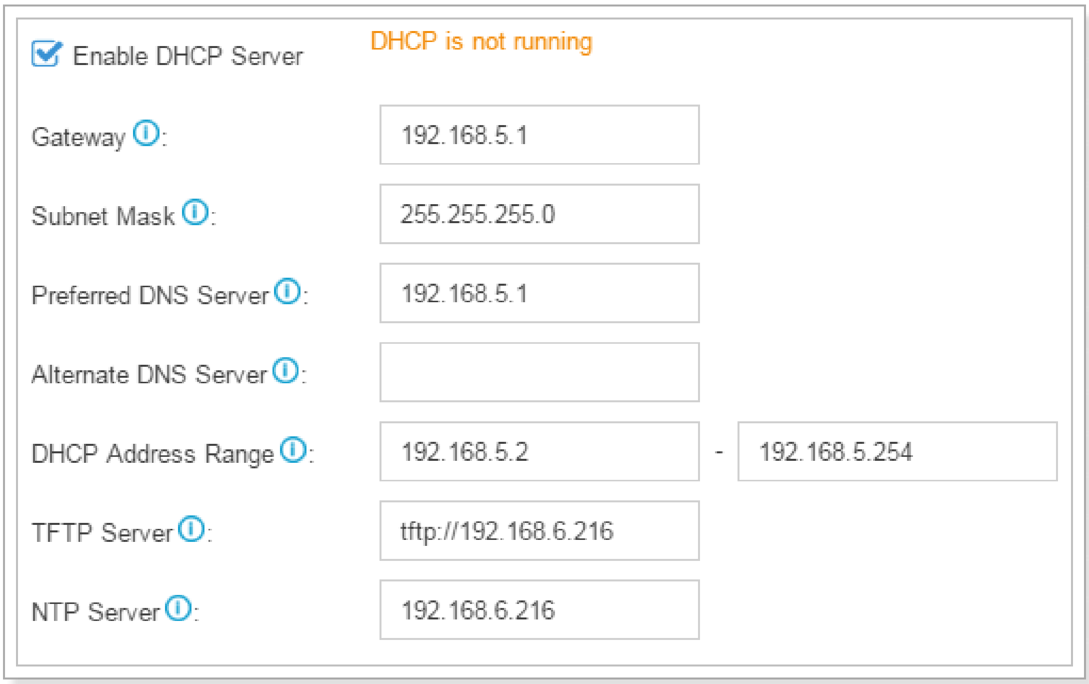
Service

The service page displays all the service status and port on eSIP Evolution Series.

Protocol or Service	Description
HTTPS	The default access protocol is HTTPS and the port is 59088.
Redirect from port 80	If the option is enabled, when you access eSIP Evolution Series using HTTP with port 80, it will be redirected to HTTPS with port 59088.
Certificate	If you have uploaded HTTPS certificates to eSIP Evolution Series, select it from the drop-down menu.
HTTP	The default port for HTTP is 80.
SSH	SSH port is used to access the eSIP Evolution Series underlying configurations to debug the system. The default port is 59022. We recommend you disable SSH port if you do not need it.
FTP	With FTP service, you can connect to the server via web browser. Thedefault port is 59021.
TFTP	To upload files to eSIP Evolution Series through TFTP, you need to enable this option.
IAX	The default port is 4569.
SIP UDP	The default port is 5060.
SIP TCP	The default port is 5060.
SIP TLS	The default port is 5061.

DHCP

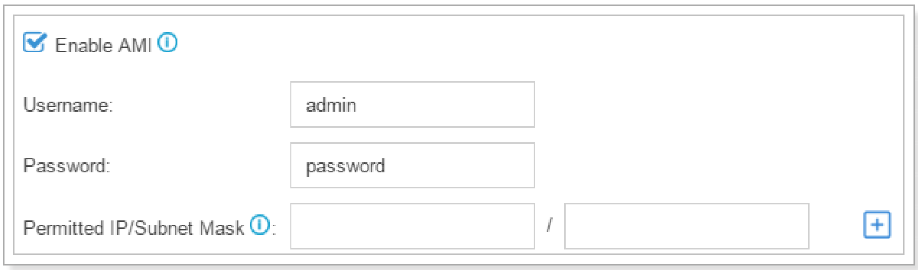
Check the box **Enable DHCP Server**, and the eSIP Evolution Series will act as a DHCP server. This feature is used when you do phone provisioning through DHCP mode.



- **Gateway**: enter the gateway IP address.
- **Subnet Mask**: enter the subnet mask.
- **Preferred DNS Server**: enter the preferred DNS server.
- **Alternate DNS Server**: enter the alternate DNS server.
- **DHCP Address Range**: this sets the IP address that the DHCP server can assign to network devices. Start IP address is on the left and end IP on the right.
- **TFTP Server**: this option is for Phone Provisioning feature, so 3rd Party SIP phones can get configuration file from this address. For some SIP phones, remember to specify the protocol, for example, tftp://192.168.5.150.
- **NTP Server**: the server can be a NTP server. By default, it is the server’s IP address.

AMI

The Asterisk Manager Interface (AMI) is a system monitoring and management interface provided by Asterisk. The 3rd party software can work with the eSIP Evolution Series using the AMI interface. The default port is 5038.

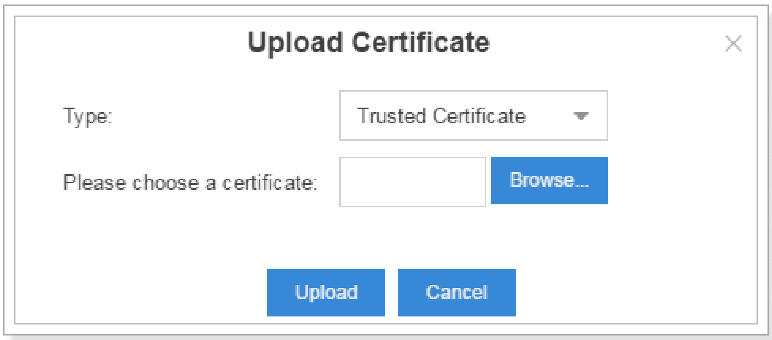


- **Username**: specify a name for the AMI user.
- **Password**: specify a password for the user to connect to AMI.
- **Permitted IP/Subnet mask**: configure permitted IP address and subnet mask that would be allowed to authenticate as the AMI user. If you do not set this option, all IPs will be denied.

Certificate

The eSIP Evolution Series supports TLS and HTTPS protocols. Before using these two protocols, you need to upload the relevant certificates to the system.

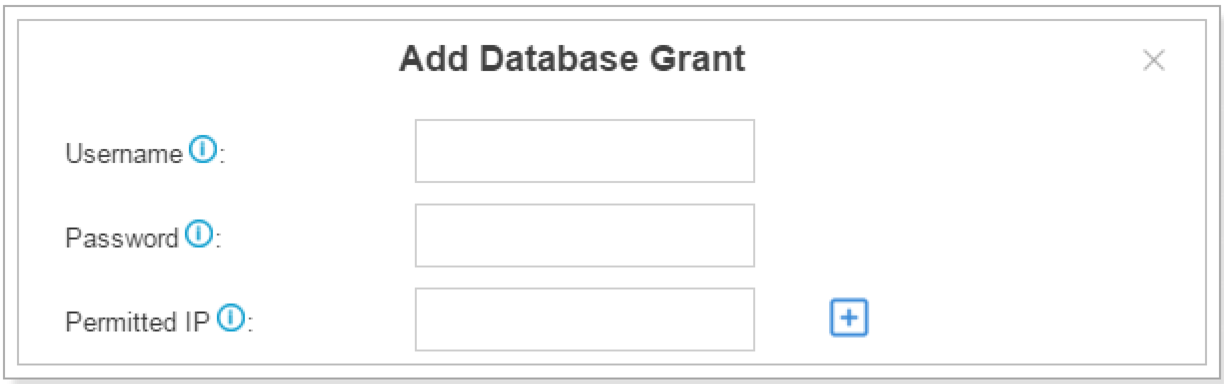
Click Upload to upload a certificate.



- **Trusted Certificate**: This certificate is a CA certificate. When selecting “TLS Verify Client” as “Yes”, you should upload a CA. The relevant TLS client (i.e. SIP phone) should also have this certificate.
- **PBX Certificate**:
  This certificate is the server-side certificate. Whether selecting “TLS Verify Client” as ”Yes” or “No”, you should upload this certificate to the eSIP Evolution Series. If a TLS client (i.e. SIP phone) enables “TLS Verify server”, you should also upload the relevant CA certificate on the SIP phone.

Database Grant

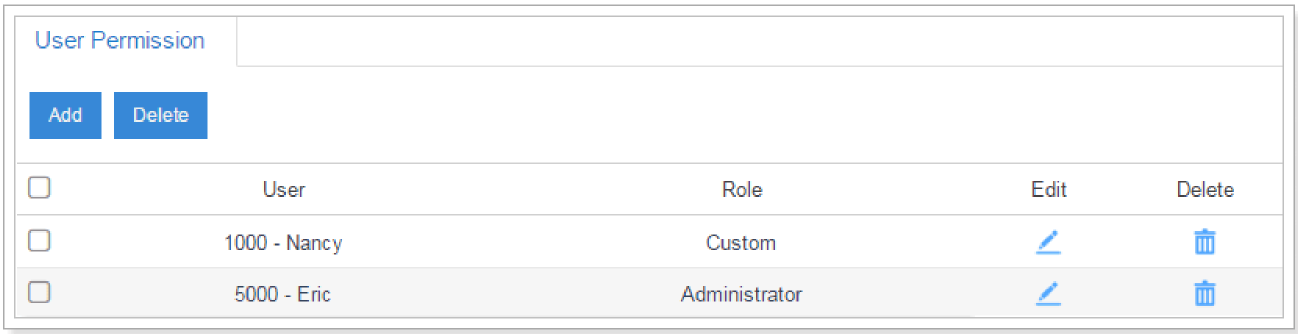
ESI's eSIP Evolution Series uses MySQL database. 3rd party software can access MySQL via the Internet. Before that can happen, you need to grant the authority to the database user. Go to Database Grant page, click Add to add a database user, and specify the username and password.



- **Username**: configure the username which can be used by a third party to access the database of THE SEVER.
- **Password**: configure the password which can be used by a third party to access the database of THE SEVER.
- **Permitted IP**: enter the permitted IP address.

User Permission

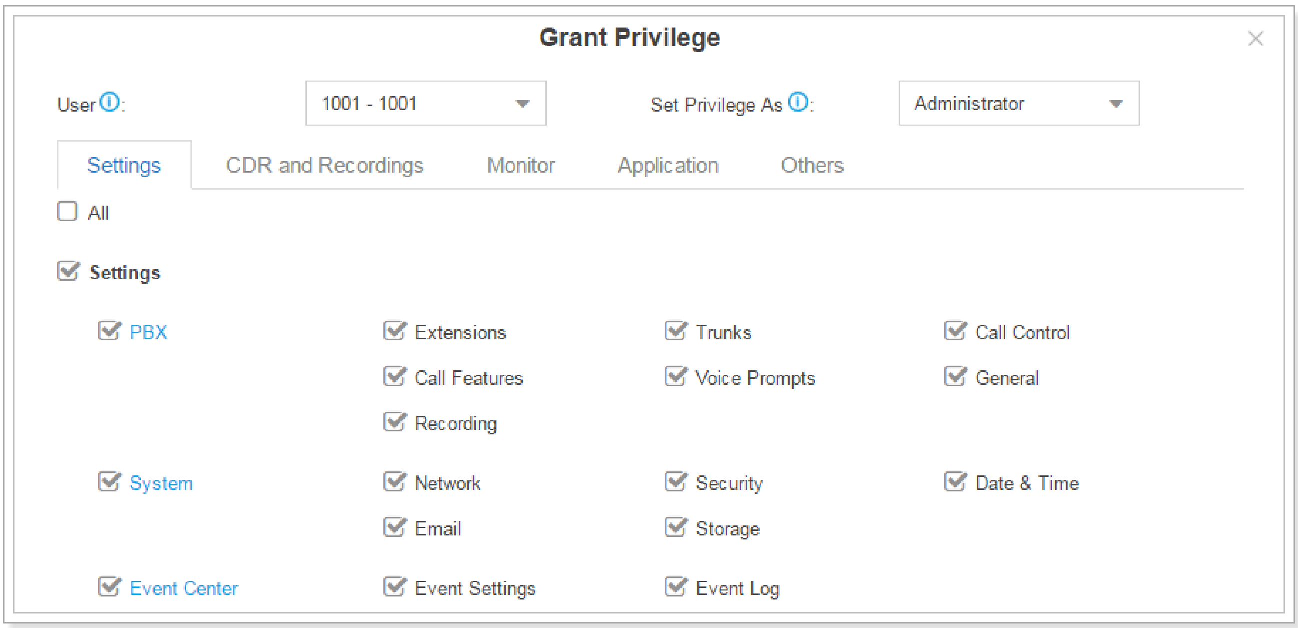
The system has one default administrator account, which has the highest privileges. Here the administrator is referred to as Super Admin. The system will automatically create user accounts when new extensions are created. By default, the extension users can log in the system and check their own settings and CDR. The Super Admin can grant more privileges for extension users. All the created users will be displayed on the User Permission page.



- **Super Admin** has the highest privilege. The super administrator can access all pages on eSIP Evolution Series Web GUI and make all the configurations on the system.
  - Username: admin
  - Default Password: esi789*#
- **Administrator** is created by the Super Admin. The administrator has all the privileges but cannot create new users for login.
- **Custom User** is created by the Super Admin. The Super Admin sets the privileges for those users according to different situations.

Add New User Permission

Log in to the eSIP Evolution Series Web GUI with the Super Admin account, go to **Settings > System > User Permission**. Click Add to add a new User Permission. The following window pops up. Choose the user and privilege type, then check the options to enable the privileges for the user.



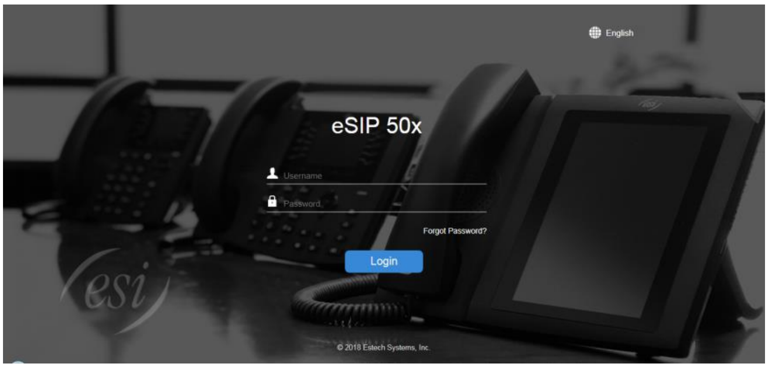
Once created, the Super Admin can edit the users by clicking or delete the users by clicking .

User Portal

The extension user can log in to the eSIP Evolution Series Web GUI with the extension username and password. The extension user account is created automatically when an extension is created on the system.

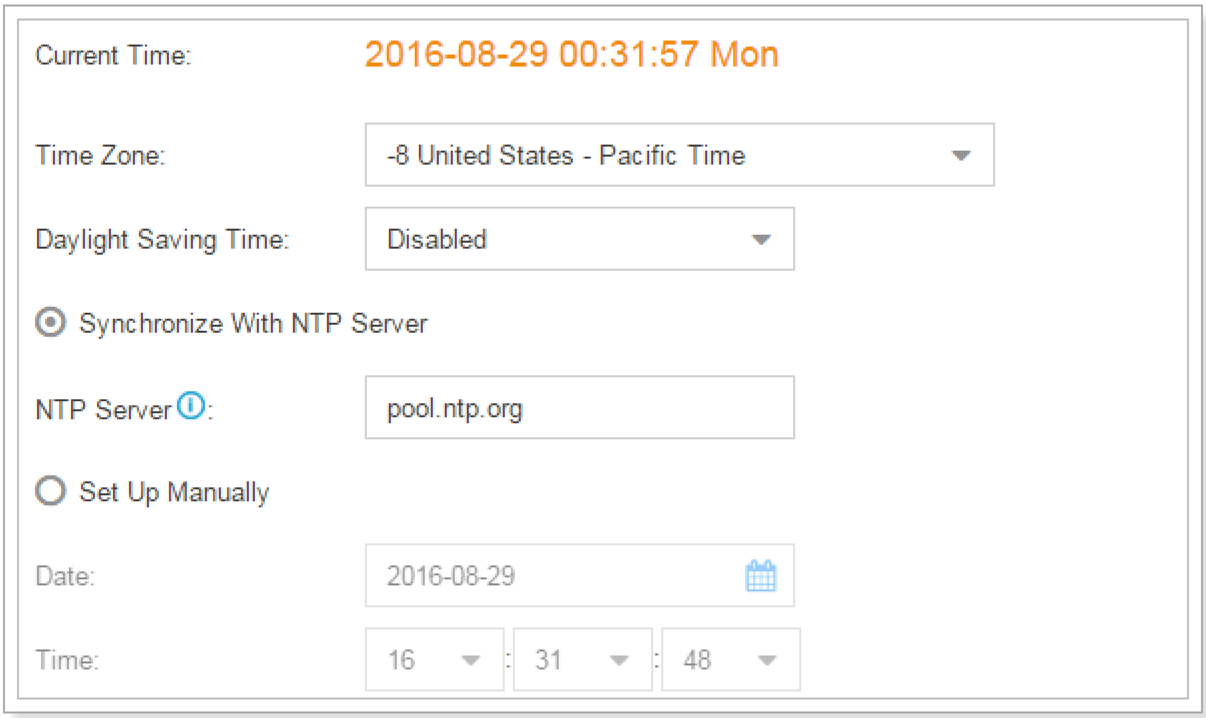
- **Username**: extension number (i.e. 1000)
- **Default password**: “Pass” plus extension number (i.e.Pass1000)

Below is an example of the login page using extension number 1000.



Date & Time

Go to **Settings > System > Date & Time** to check the current time on the system. Here you can adjust the time of the system (including time zone) to your local time.



- **Time Zone**: Select your current time zone.
- **Daylight Saving Time**: The option is disabled by default. Enable it when necessary.
- **Synchronize With NTP Server**: if you choose this mode, the system will adjust its internal clock to a central network server. Please note, the eSIP Evolution Series should be able to access the Internet if you choose this mode.
- **NTP Server**: Enter a NTP server.
- **Set Up Manually**: if you choose this mode, you need to set the time manually.
    - Date: choose the date.
    - Time: choose the time.

Email

Set the system's email server to send voicemail to email, alert event emails, fax to email, email to SMS and SMS to email. Go to **Settings > System > Email** to configure the system email.
Check the email settings parameters below.

Option	Description
Email Address	Enter the email address.
Password	Enter the password.
Outgoing Mail Server (SMTP)	Enter SMTP server and port. **Example**: *smtp.sina.com:25*
Incoming Mail Server (POP3)	Enter the POP3 server and port. **Example**: *pop.sina.com:110*
Enable TLS	Use TLS to send secure message to server. If the email sending server needs to authenticate the sender, you need to select the check box. **Note:** if you use Gmail or Exchange, you need to enable this option.

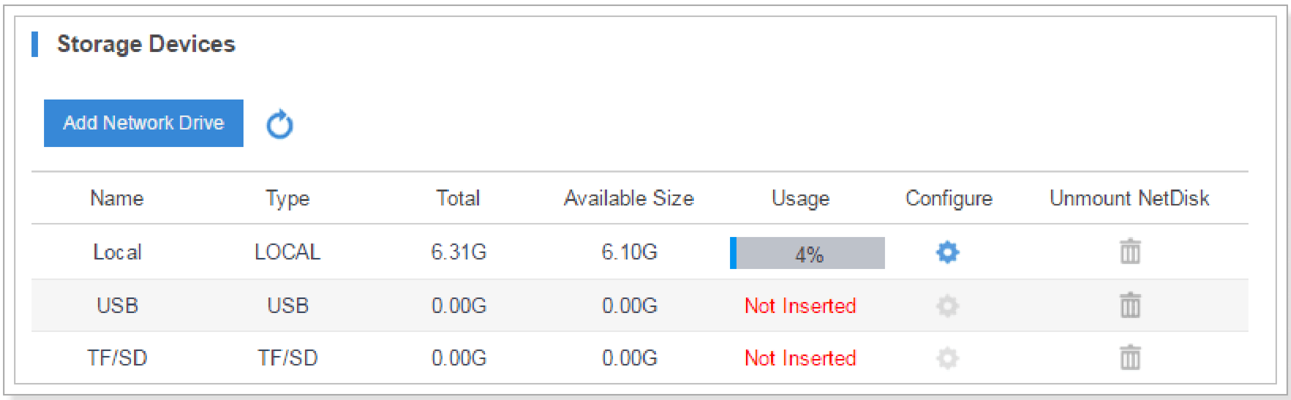
After finishing the configuration, click Test to test the email. In the prompt, fill in an email address to send a test email to verify the Email settings.

Storage

ESI's eSIP Evolution Series provides local storage (Flash) and supports external storage (MicroSD/SD card). Users can choose where to store the voicemails, CDR, recordings, and logs.

Storage Devices

Go to **Settings > System > Storage** to configure the storage. All the local storage and external storage status shows on the page.



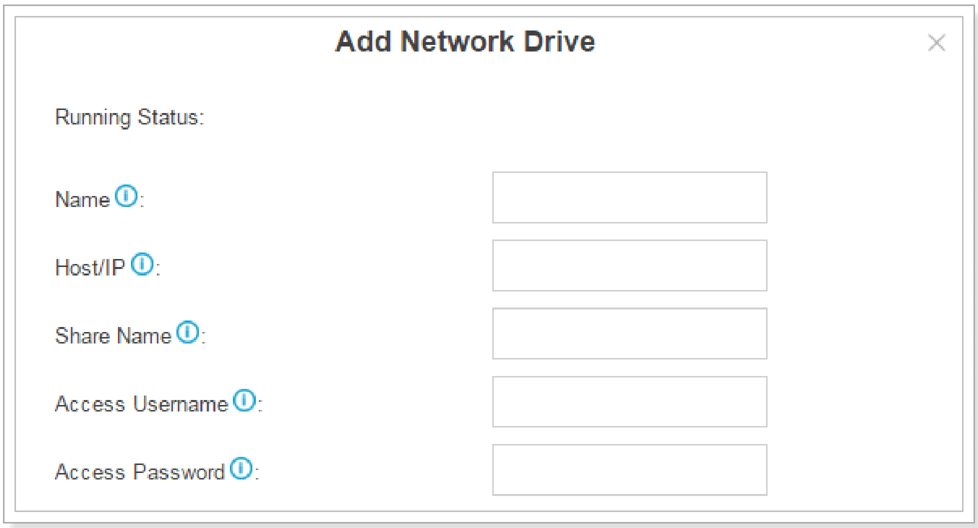
To format an external storage:

1. Click .
2. Click Format on the pop-up window to start formatting.

To add Network Drive:

The Network Drive feature is used to extend storage space. Before a network drive can be properly configured, an SMB share folder accessible from the eSIP system must be set up on a Windows based machine. Once that has been set up, please follow the following instructions to configure a network drive:

1. Choose a window-based computer/server that is always in service.
2. Create a folder.
3. Share this folder to Everyone.
4. Click Add Network Drive and input the Net-Disk information in the eSIP Evolution Series:

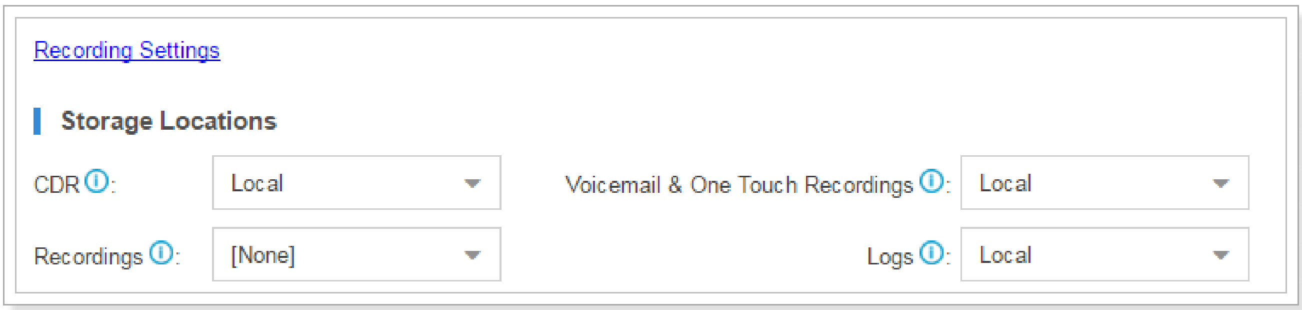


- **Name**: give this network drive a name to help you identify it.
- **Host/IP**: set the IP address where the recordings will be stored.
- **Share Name**: the shared folder name where the recordings will be stored.
- **Access User Name**: the User name used to log in the Network share. Leave this blank if it is not required. In general, you use the administrator account on PC as a user name here.
- **Access Password**: the password used to log into the network share. Leave this blank if it is not required.

5. If the configuration is correct, you can see the NETDISK status shown as below.

Storage Locations

When the storage devices are configured and ready to use, you can select where to store CDR, Recordings, Voicemail, one-touch recordings, logs.



Auto Cleanup

ESI's eSIP Evolution Series supports auto clean for CDR, logs, voicemails, one-touch recordings and recordings.

CDR Auto Cleanup	
Max Number of CDR	Set the maximum number of CDR that should be retained. The default is 500000. The oldest CDR will be deleted when the threshold is reached.
CDR Preservation Duration	Set the maximum number of days that CDR should be retained. The default is left blank.
Voicemail and One Touch Recording Auto Cleanup	
Max Number of Files	Set the maximum number of voicemail and one touch recording files that should be retained. The default is 50 for each user. The old CDR will be deleted when the threshold is reached.
Files Preservation Duration	Set the maximum number of minutes that voicemails and one touch recordings should be retained. The default is left blank.
Recordings Auto Cleanup	
Max Usage of Device	Set the maximum storage percentage the device is allowed to store. The default is 80%. The recordings will be deleted when the threshold is reached.
Recordings Preservation Duration	Set the maximum number of days that recording files should be retained. The default is left blank.
Logs Auto Cleanup	
Logs Preservation Duration	Set the maximum number of days that logs should be retained. “Logs Preservation Duration”. The default is 7. This setting is for system logs.
Max Number of Logs	Set the maximum number of logs that should be retained. The default is unlimited. The old logs will be deleted when the threshold is reached. This setting is for operation logs.

Extensions

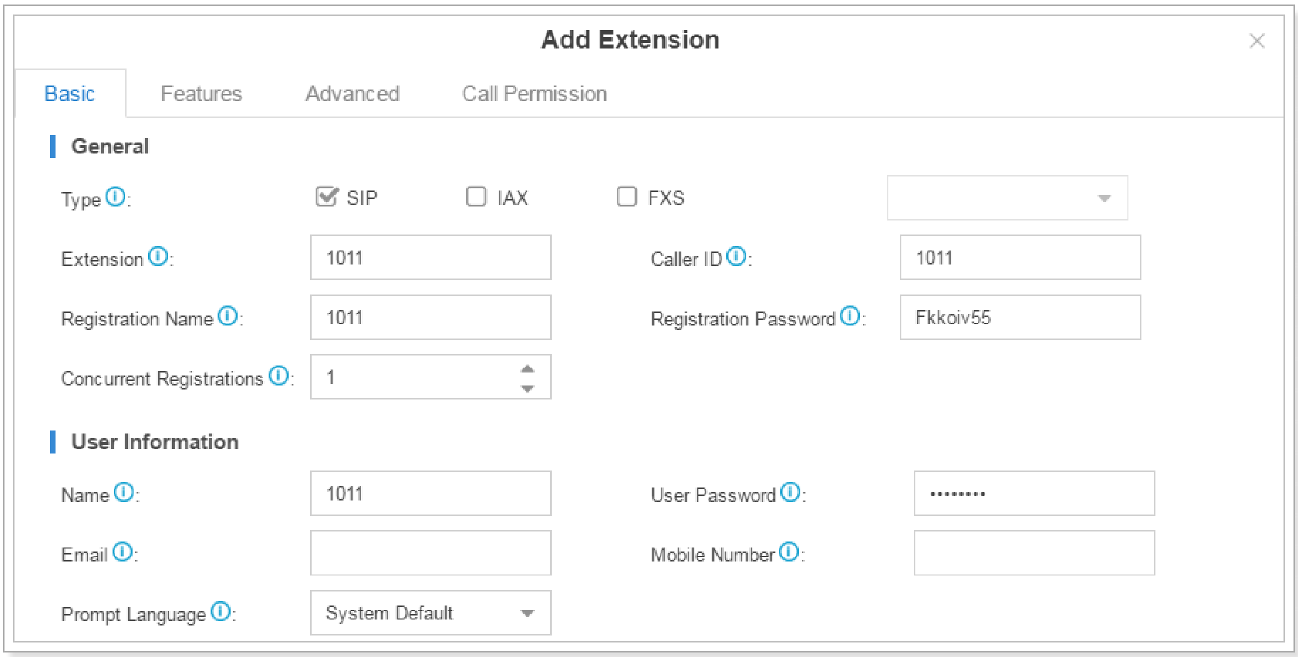
This chapter explains how to create and configure extensions on the eSIP Evolution Series. ESI's eSIP Evolution Series supports SIP, IAX and FXS extensions. An extension can be set to the 3 types and be registered to different devices. Go to **PBX > Extensions** page to configure the extensions.

- Add New Extension
- Add Bulk Extensions
- Search and Edit Extensions
- Import and Export Extensions
- Extension Group

Add New Extension

Click 

to add a new extension; you will see the pop-up window appear as below.



Extension settings are divided to 4 categories:

- Basic
- Features
- Advanced
- Call Permission

Click on the tab to view or edit the relevant settings. See the configuration parameters below.

Note: different settings will appear for different types of extensions.

Basic Settings

General	
Type	Check the box to set the extension type. You can set the extension to multiple types. • SIP • IAX • FXS: 2FXS or FXOFXS module should be installed on the device if you want to create FXS extensions.
Extension	The extension number that will be associated with this particular user or phone.
Caller ID	The Caller ID string that appears on outbound calls for this extension.
Registration Name	For extension registration validation.
Registration Password	The password for the user to register the SIP or IAX account. For example, 12t3f6.
Concurrent Registrations	The eSIP Evolution Series server supports SIP forking. **SIP forking** refers to the process of “forking” a single SIP call to multiple SIP endpoints. The value of Concurrent Registrations limits how many SIP endpoints the extension can be registered.
User Information	
User Password	The password for this extension user to log in the system. For example, 12t3f6.
Email	Email address of this extension user. The email will be used to recover their password, receive forwarding voicemails, receive fax as an attachment, and receive event notifications.
Mobile Number	Mobile Number of this user. The number can receive forwarded calls and event notifications.
Prompt Language	The language of voice prompts. The default is the same with system language.

Features

Voicemail	
Enable Voicemail	Check this box to enable voicemail for this extension.
Send Voicemail to Email	Check this box to send voicemail to the user's email address. **Note:** to use this feature, "Email Settings" under "System" need to be configured correctly.
Voicemail Access PIN	Voicemail password used to access Voicemail system. This password can contain only numbers.
Call Forwarding	
Always	Always redirect the call to the designated destination. • Voicemail: redirect the caller to leave a voice message. • Extension: redirect the caller to another extension. • Users' Mobile Number: redirect the caller to the mobile number filled in User Information. • Custom Number: fill in the number manually and redirect the caller to this number.
No Answer	Redirect the call to the designated destination when it is not answered.
When Busy	Redirect the call when the extension is busy.
Mobility Extension	
Enable Mobility Extension	If you enabled, when the User's Mobile Number dials into the system, the phone will have the same user permissions as a desktop extension. The mobile number will be able to reach other extensions, dial out with the trunk, and play voicemail. **Note:** This is not the same as using the mobile app as a registered device for the user.
Mobility Extension	It is the same with the User's Mobile Number. A prefix matching the outbound route also needs to be filled in.
Ring Simultaneously	When the extension has an incoming call, it rings the mobile number simultaneously.
Monitor Settings	
Allow Being Monitored	Check this option to allow this user to be monitored.

Monitor Mode	Decide how you will monitor another extension's current call. • None: you will not be allowed to monitor others. • Extensive: all of the below modes will be available to use. • Listen: you can only listen to the call, but can't talk (default feature code: *90). • Whisper: you can talk to the extension you're monitoring without being heard by the other party (default feature code: *91). • Barge-in: you can talk to both parties (default feature code: *92).
Other Settings	
Ring Timeout	Customize the ring timeout in seconds. The phone will stop ringing and follow the No Answer destination once this threshold is met.
Max Call Duration	Select the maximum call duration in seconds for every call of this extension. If you wish to customize, enter the value in the text box directly. This option is valid only for outbound calls. If you choose "Follow System", it would be equal to the "Max Call Duration" value in the "General" page.
Call Waiting	Check this option if the extension should have Call Waiting capability. If this option is checked, the "When busy" call forwarding options will not be available. The call waiting setting of the IP phone has higher priority than the server's call waiting setting.
DND	Do Not Disturb. When DND is enabled for an extension, the extension will not be available.

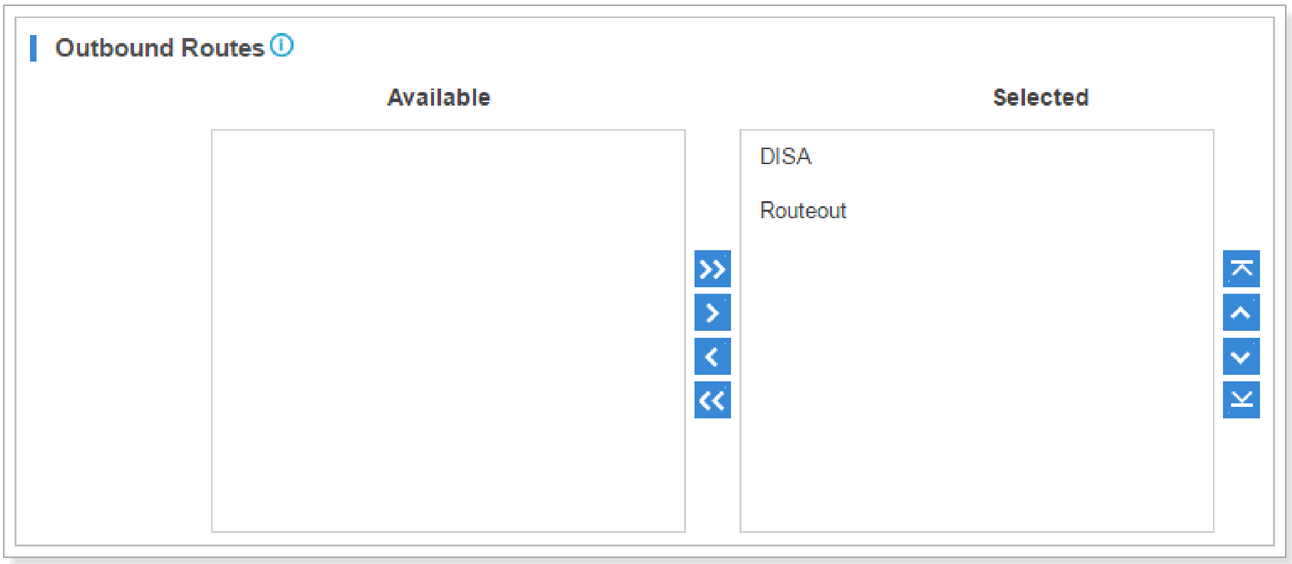
Advanced Settings

VoIP Settings	
NAT	This setting should be used when the system is using a public IP address, communicating with devices hidden behind a NAT device (such as a broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP and/or RTP ports.
Qualify	Check the box to send SIP OPTIONS regularly to the device to check if the device is still online.
Enable SRTP	Enable SRTP for voice encryption.
Register Remotely	Check the box to allow registration of a remote extension.
Transport	Select the allowed transport.
DTMF Mode	Set the default mode for sending DTMF tones. • RFC4733: DTMF will be carried in the RTP stream in different RTP packets than the audio signal • Info: DTMF will be carried in the SIP Info messages • Inband: DTMF will be carried in the audio signal • Auto: will use RFC4733 or Info automatically. RFC4733 is the default mode.
IP Restriction	
Enable IP Restriction	This option is used for IP access control. Check this option to enhance the system's security. Once enabled, only the IP address or IP subnet matching the settings will be able to register as this extension number.
Permitted IP/Subnet mask	Define the IP address or IP subnet which is allowed to register to the server. The input format should be IP address/Subnet mask. **Example:** • 192.168.5.100/255.255.255.255 means only the device whose IP address is 192.168.5.100 is allowed to register this extension number; • 192.168.5.0/255.255.255.0 means only the device whose IP section is 192.168.5.XXX is allowed to register this extension number.
Analog Settings	
Min Flash Detection	Set the minimum amount of time, in milliseconds, that a hook flash must remain depressed in order for the system to consider it as a valid flash event. The default is 300 ms.

Max Flash Detection	Set the maximum amount of time, in milliseconds, that a hook flash must remain depressed in order for the system to consider it as a valid flash event. The default is 1000 ms.
Echo Cancellation	Enable or disable echo cancellation on the FXS port.
Rx Volume	The volume of the voice sent from the analog phone to the FXS port of the server Set the value from 5% to 100% or choose Custom to define the RX gain below.
Rx Gain	The gain of the voice sent from the analog phone to the FXS port of the server. (Unit: db). The valid range is -30db to 6.0db.
Tx Volume	The gain of the voice sent from the FXS port of the server to the analog phone. (Unit: db) The valid range is -30db to 6.0db.

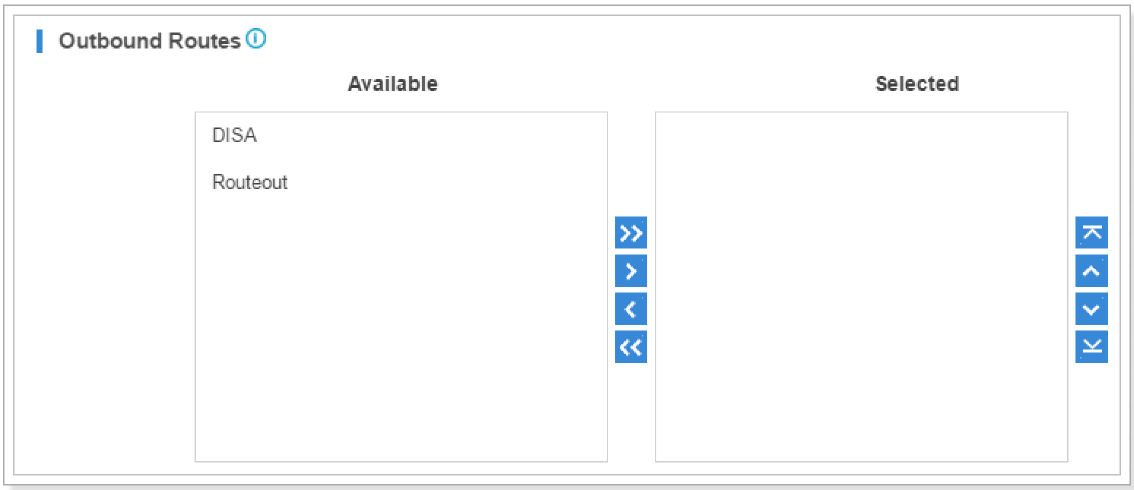
Call Permission

Choose the outbound routes the user is allowed to use.



Add Bulk Extensions

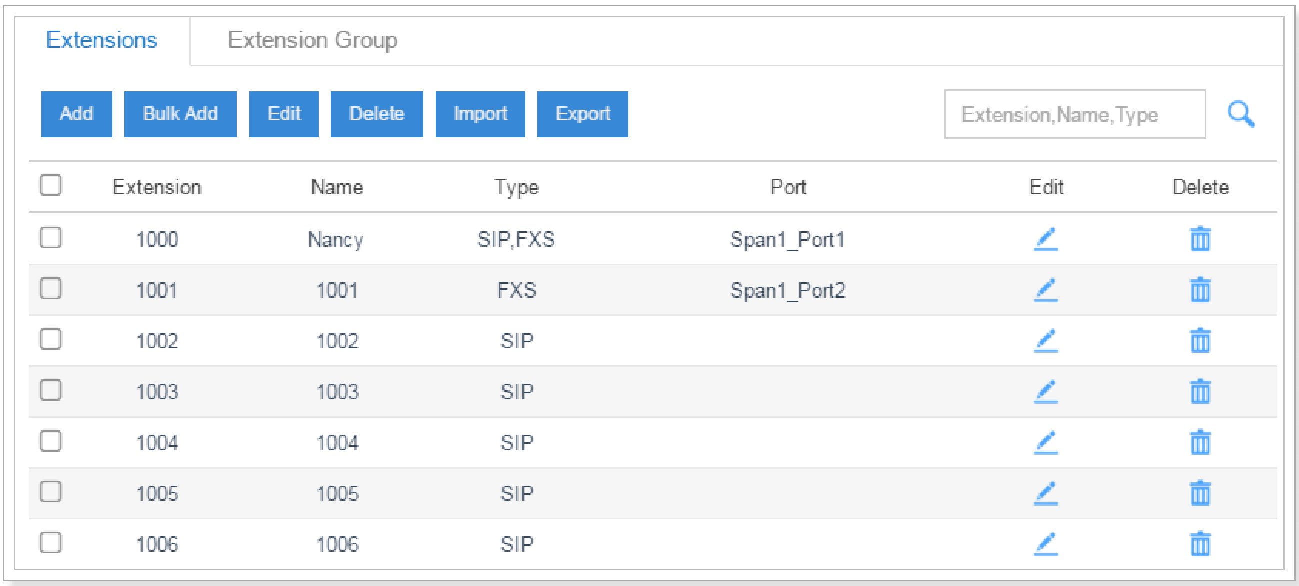
You can bulk add SIP/IAX extensions on the system, which helps you add a large amount of extensions quickly. Click **Bulk Add** to add extensions in bulk.



General	
Type	Choose the type for the extensions: • SIP • IAX
Start Extension	Set the starting extension number of the batch of extensions to be added.
Create Number	The number of extensions to be created.
Registration Password	Decide which type of registration password will be used. There are 3 options. • Random: generate a random password for each extension. • Fixed: use the text filled in as the password for all extensions. • Prefix + extension number: fill in a prefix and the password will be the text plus the extension's number.
User Password	Decide which type of user password will be used. There are 3 options. • Fixed: use the text filled in as the password for all extensions. • Prefix + extension number: fill in a prefix and the password will be the text plus the extension's number.
Concurrent Registrations	Set the max concurrent registrations for SIP extensions.
Prompt Language	Set the language of voice prompt for extensions.

Search and Edit Extensions

All the extensions are listed on the extension page. Each extension has a check box for you to edit or delete in bulk. Also, you can edit or delete per extension by clicking or .



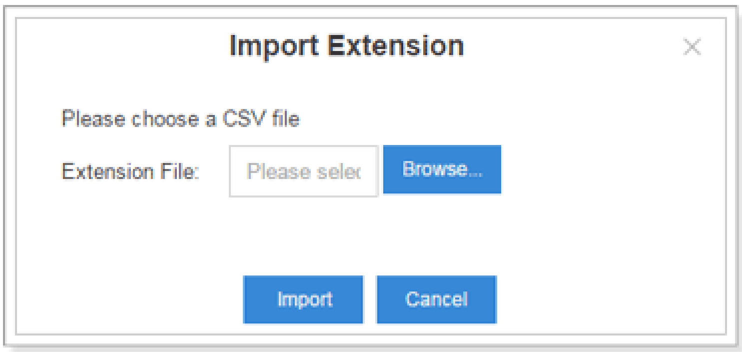
- **Search Extension**
  You can search extensions by entering the extension number, name or type.
- **Edit an Extension**
  Click to edit the desired extension.
- **Delete an Extension**
  Click to delete the desired extension.
- **Bulk Edit Extensions**
  Select the check box for the extensions, click Edit to edit the extensions.
- **Bulk Delete Extensions**
  Select the check box for the extensions, click Delete to delete the extensions.

Importing and Exporting Extensions

Administrators can import and export extension configurations, which helps you manage extensions easily.

To Import Extensions

1. Click Import, you will see a dialog window shown as below.



2. Click **Browse** and select the file to start uploading. The file must be a csv file. See the sample file below. You can export an extension file from the server and use it as a template to start with.

	A	B	C	D	E	F	G	H	I
1	type	username	registerpassword	fullname	callerid	registerr	vmsecret	hasvoicem	enablevm
2	SIP,FXS	1000	Password1000	Nancy	1000	1000	1000	yes	no
3	FXS	1001		1001	1001		1001	yes	no
4	SIP	1002	ejWH3Yqx	1002	1002	1002	1002	yes	no
5	SIP	1003	2JIikoPH	1003	1003	1003	1003	yes	no
6	SIP	1004	dA8A2yuS	1004	1004	1004	1004	yes	no
7	SIP	1005	zK54FQ1E	1005	1005	1005	1005	yes	no
8	SIP	1006	vTech1006	1006	1006	1006	1006	yes	no
9	SIP	1007	vTech1007	1007	1007	1007	1007	yes	no
10	SIP	1008	Pincode1008	1008	1008	1008	1008	yes	no
11	SIP	1009	Pincode1009	1009	1009	1009	1009	yes	no
12	SIP	1010	Pincode1010	1010	1010	1010	1010	yes	no
13	SIP	5000	Inwwgd21	Eric	5000	5000	1011	yes	no

The sample csv file will result in the following extensions in the server.

Add | Bulk Add | Edit | Delete | Import | Export | Extension,Name,Type

	Extension	Name	Type	Port	Edit	Delete
☐	1000	Nancy	SIP,FXS	Span1_Port1		
☐	1001	1001	FXS	Span1_Port2		
☐	1002	1002	SIP			
☐	1003	1003	SIP			
☐	1004	1004	SIP			
☐	1005	1005	SIP			
☐	1006	1006	SIP			
☐	1007	1007	SIP			
☐	1008	1008	SIP			
☐	1009	1009	SIP			

To Export Extensions

1. Select the check box of the extensions, click Export, the selected extensions would be exported to your local PC.

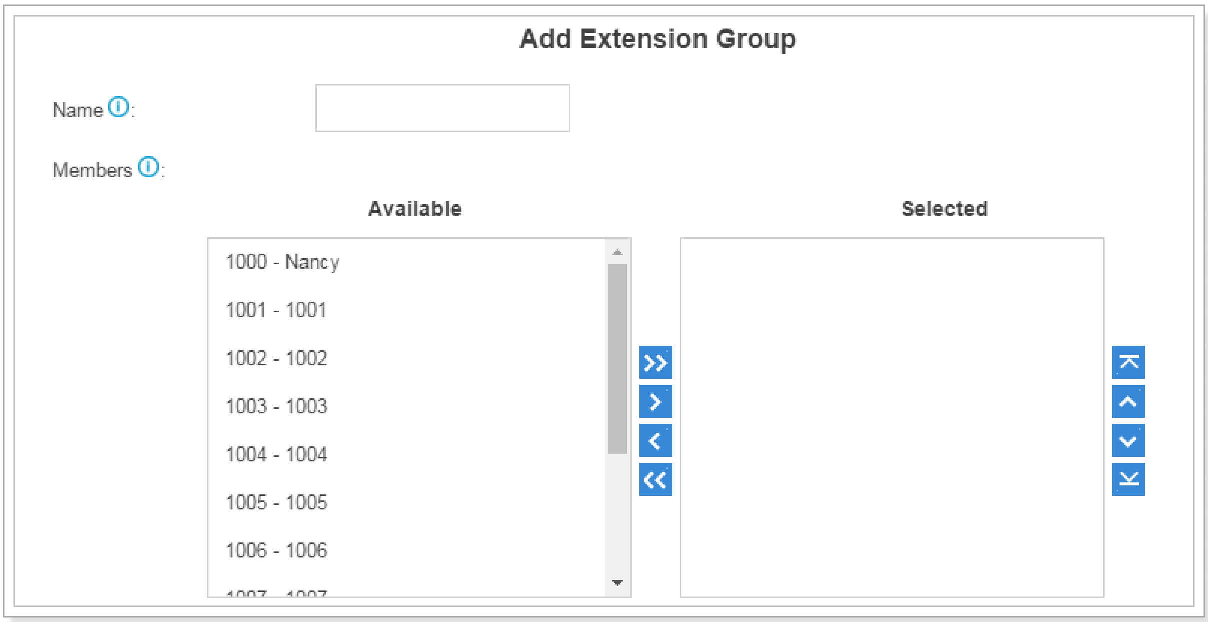
Add | Bulk Add | Edit | Delete | Import | Export | Extension,Name,Type

☑	Extension	Name	Type	Port	Edit	Delete
☑	1000	Nancy	SIP,FXS	Span1_Port1		
☑	1001	1001	FXS	Span1_Port2		
☑	1002	1002	SIP			
☑	1003	1003	SIP			
☑	1004	1004	SIP			
☑	1005	1005	SIP			
☑	1006	1006	SIP			
☑	1007	1007	SIP			
☑	1008	1008	SIP			
☑	1009	1009	SIP			

Extension Group

The Extension Group feature allows you to assign and categorize extensions in different groups, which helps you to better manage the configurations in the system. For example, you can create Support and Sales groups, when configuring Outbound Route, you can select an extension group instead of each extension. This feature simplifies the configuration process.

Click  to create an extension group.



Trunks

The ESI eSIP Evolution Series supports FXO trunks, 3G/4G trunks, SIP trunks, and T1/PRI trunks. In this chapter, we give a simplified guide of setting up trunks.

- FXO Trunk
- 3G/4G Trunk
- SIP Trunk
- T1/PRI Trunk

FXO Trunk

FXO trunks are also known as PSTN trunks or analog trunks. The public switched telephone network (PSTN) is the network of the world's public circuit-switched telephone networks.

To support FXO trunks on the system, you need to insert 2FXO or FXOFXS module to the server. Go to **Settings > PBX > Trunks** to edit the FXO trunk. Before configuring a FXO trunk, please make sure that the analog line is connected to the eSIP Evolution Series' FXO port.

Click to edit the FXO trunk. Please check the FXO trunk configuration parameters below.

Basic Settings

General	
Trunk Name	Give this trunk a name to help you identify this trunk.
Rx Volume	Set the receiving volume of FXO port or choose Custom to define the RX gain below.
RxGain	The RX Gain for the receiving channel of FXO Port. The valid range is -30db to 12db.
Tx Volume	Set the transmitting volume of FXO port or choose Custom to define the TX gain below.
TxGain	The TX Gain for the transmitting channel of FXO Port. The valid range is -30db to 12db.
Enable SLA	If enabled, this trunk will not be available in routes or other channels.
Allow Barge	Whether to allow other SLA stations to join a call by pressing the SLA key.
Hold Access	Specify hold permission for the station. • **Open:** other stations that share the same line could retrieve the call. • **Private:** the call can be retrieved only by the station that previously put the call on hold, not by others sharing the same line.

Advanced Settings

Hang-up detection settings help the system to detect if a call is hung up. If you find the PSTN call could not be disconnected, these settings need to be changed.

Hang-up Detection	Description
Hang up Detection Method	Detect if a call is hung up with one of the following methods: • Busy Tone: listen for a busy tone to detect if the line got hung up. • Polarity Reversal: the call will be considered as “hang up” on a polarity reversal.
Busy Count	Specify how many busy tones to wait for before hanging up. The default is 4. If you wish to customize, enter the value in the text box directly. Setting this too high might cause failure of busy detection.
Busy Pattern	Select the cadence of your busy signal. The default is None. If you wish to customize, enter the value in the text box directly. The input format should be "Sound,Silence". E.g. "500,500" means 500ms on, 500ms off. • If you choose None, the system will accept any regular sound- silence pattern that repeats Busy Count times as a busy signal. • If you specify Busy Pattern, the system will further check the length of the tone and silence, which will further reduce the chance of a false positive disconnection.
Busy Interval	The busy detection interval. The default is 1. If you wish to customize, enter the value in the text box directly.

Frequency Detection	Decide whether to enable detecting the busy signal frequency or not.
Busy Frequency	If Frequency Detection is enabled, you must specify the local frequency. The default is 480,620. If you wish to customize, enter the value in the text box directly. Unit: Hz.

Answer Detection Type

Answer Detection will help the system to accurately bill your calls.

- None:
- Polarity: choose this option if the FXO trunk could send polarity reversal signal after a call is established.

Caller ID Settings

Caller ID Settings will help the system to detect Caller ID. If an incoming PSTN call does not display Caller ID, you need to confirm with your service provider if the line has enabled Caller ID feature. If this line does support Caller ID, configure these settings to solve this problem.

Option	Description
Caller ID Detection	Whether to enable Caller ID detection.
Caller ID Start	Define the start of a Caller ID signal. The options are: • After Ring: detect Caller ID after first ring; • Before Ring: detect Caller ID before first ring; • After Polarity: detect Caller ID after polarity reversal; the default is After Ring.
Caller ID Signaling	This option defines the type of caller ID signaling to use. • Bell202 • ETSI-V23 • V23-Japan • DTMF

Other Settings

Option	Description
Ring Detect Timeout	FXO (FXS devices) must have a timeout to determine if there was a hang up before the line is answered. This value can be used to configure how long it takes before the system considers a non- ringing line with hang-up activity. The default is 5000. If you wish to customize, enter the value in the text box directly. The valid range is 1000-8000.
Echo Cancellation	Whether to enable echo cancellation for this trunk.
Enable DNIS	Dialed Number Identification Service is a telephone service that enables a company to identify which telephone number was dialed. Users could configure DNIS to allow the IP phones to display which trunk is passing the call.
DNIS Name	A name for this DNIS, when a call reaches the selected trunk, the name will be displayed on the ringing phone.

3G/4G Trunk

ESI's eSIP Evolution Series supports 3G/4G trunks. To support the trunks, you need to install either a 3G or 4G/LTE module on the eSIP Evolution Series and insert the SIM card on the module.

Click  to edit the trunk. Please check the 3G/4G trunk configuration parameters below.

Option	Description
Trunk Name	Give this trunk a name to help you identify this trunk.
PIN Code	Enter the SIM card PIN code if the card has one. **Note:** If you fail to enter your correct PIN code 3 times in succession, the SIM card will be permanently locked, which means you would need a new card.
Rx Volume	Set the receiving volume of the cellular port or choose Custom to define the RX gain below.
RX Gain (db)	The RX Gain for the receiving channel of the cellular port. The valid range is - 20db to 20db.
Tx Volume	Set the transmitting volume of cellular port or choose Custom to define the TX gain below.
TX Gain (db)	Set the transmitting volume of cellular port or choose Custom to define the TX gain below.
Echo Cancellation	Whether to enable echo cancellation for the trunk.
Enable DNIS	Dialed Number Identification Service is a telephone service that enables a company to identify which telephone number was dialed. Users could configure DNIS to allow the IP phones to display which trunk is passing the call.
DNIS Name	A name for this DNIS, when a call reaches the selected trunk, the name will be displayed on the ringing phone.

VoIP Trunk

ESI's eSIP Evolution Series supports SIP and IAX protocols and provides 2 types of VoIP trunks:

- **Register Trunk:** registration based VoIP trunk. A Register Trunk requires the eSIP Evolution Series to register with the provider using an authentication name and password.
- **Peer Trunk:** IP based VoIP trunk. A Peer VoIP trunk does not require the eSIP Evolution Series to register with the provider. The IP address of the eSIP Evolution Series needs to be configured with the provider, so that it knows where calls to your number should be routed.

Go to **Settings > PBX > Trunks** to add a VoIP trunk.

Note: Choosing a different trunk protocol will have different settings.

Basic Settings

SIP Register Trunk	
Protocol	Set the trunk protocol "SIP".
Trunk Type	Choose the trunk type "Register Trunk".
Name	Give this trunk a name to help you identify this trunk.
Transport	Set the transport method used by the trunk. If Hostname/IP Address is the server's Hostname and the port is 0 or blank, NAPTR and SRV lookup will be executed to search for transport, port and server. If Hostname/IP Address is a legal IP address or a designated port, then UDP will be used.
Hostname/IP	Service provider's hostname or IP address. The default SIP port is 5060.
Domain	VoIP provider's server domain name. If the provider has no domain name fill in the IP address instead.
User Name	The username used to register to the trunk from the VoIP provider.
Password	The password to register to the trunk from the VoIP provider.
From User	All outgoing calls from the SIP trunk will use the From User (in this case the account name for SIP Registration) in From Header of the SIP Invite package. Keep this field blank if not needed.
Authentication Name	Used for SIP authentication. In most cases, it is the same with the username.
Enable Outbound Proxy	A proxy that receives requests from a client. Even though it may not be the server resolved by the Request-URI.
Outbound Proxy Server	Configure the address of outbound proxy server. The address can be domain name or IP address.
Enable SLA	If enabled, this trunk will not be available in routes or other channels.
Allow Barge	Whether to allow other SLA stations to join a call by pressing the SLA key.
Hold Access	Specify hold permission for the station. • **Open:** other stations that share the same line could retrieve the call. • **Private:** the call can be retrieved only by the station that previously put the call on hold, not by others sharing the same line.
SIP Peer Trunk	
Protocol	Set the trunk protocol as "SIP".
Trunk Type	Choose the trunk type "Peer Trunk".
Name	Give this trunk a name to help you identify this trunk.
Transport	Set the transport method used by the trunk. If Hostname/IP Address is the server's Hostname and the port is 0 or blank, NAPTR and SRV lookup will be executed to search for transport, port and server. If Hostname/IP Address is a legal IP address or a designated port, then UDP will be used.
Hostname/IP	Service provider's hostname or IP address. The default SIP port is 5060.
Domain	VoIP provider's server domain name. If the provider has no domain name fill in the IP address instead.

Enable SLA	If enabled, this trunk will not be available in routes or other channels.
Allow Barge	Whether to allow other SLA stations to join a call by pressing the SLA key.
Hold Access	Specify hold permission for the station. • **Open:** other stations that share the same line could retrieve the call. • **Private:** the call can be retrieved only by the station that previously put the call on hold, not by others sharing the same line.
IAX Register Trunk	
Protocol	Set the trunk protocol "IAX".
Trunk Type	Choose the trunk type "Register Trunk".
Name	Give this trunk a name to help you identify this trunk.
Hostname/IP	Service provider's hostname or IP address. The default IAX port is 4569.
User Name	The username used to register to the trunk from the VoIP provider.
Password	The password to register to the trunk from the VoIP provider.
IAX Peer Trunk	
Protocol	Set the trunk protocol "IAX".
Trunk Type	Choose the trunk type "Peer Trunk".
Name	Give this trunk a name to help you identify this trunk.
Hostname/IP	Service provider's hostname or IP address. The default IAX port is 4569.
Domain	VoIP provider's server domain name. If the provider has no domain name fill in the IP address instead.

Codec

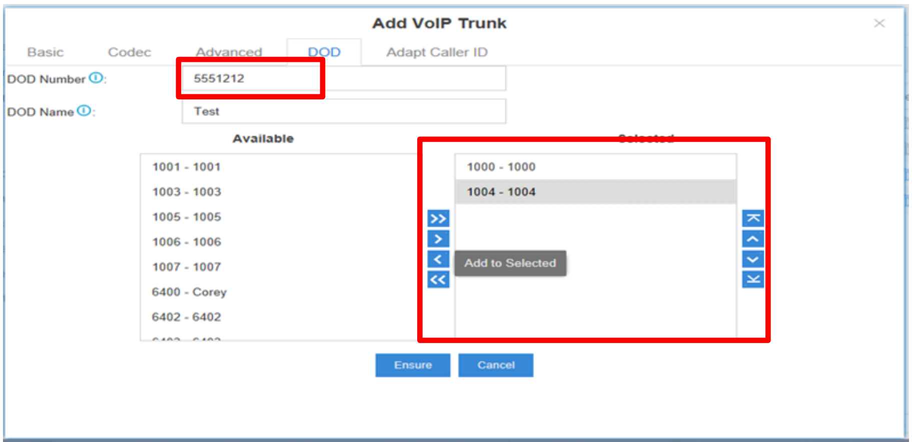
Select the supported codecs for the VoIP trunk. The eSIP Evolution Series supports the codecs: a-law, u-law, GSM, iLBC, SPEEX, G722, G726, ADPCM, G729A, H261, H263, H263P, H264, MPEG4 and iLBC.

Advanced

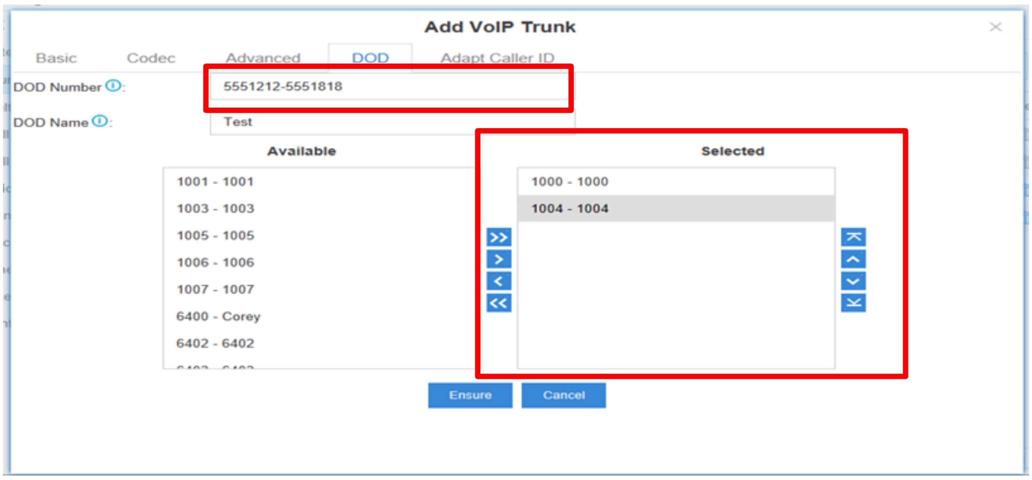
VoIP Settings	
Qualify	Enable this to send SIP OPTIONS packet to SIP device to check if the device is up.
Enable SRTP	This option enables or disables SRTP (encrypted RTP) for the trunk.
T.38 Support	Whether to enable T.38 fax for the trunk.
DTMF Mode	Set the default mode for sending DTMF tones. • RFC4733: DTMF will be carried in the RTP stream in different RTP packets than the audio signal • Info: DTMF will be carried in the SIP Info messages • Inband: DTMF will be carried in the audio signal • Auto: will attempt to detect if the device supports RFC4733 DTMF. If so, it will choose RFC4733; if not, it will choose Inband. RFC4733 is the default mode.
Other Settings	
Realm	Realm is a string to be displayed to users so they know which username and password to use. If you don't know what to fill in, contact your service provider for further instructions.
Send Privacy ID	Check this check box to send privacy ID.
Enable DNIS	Dialed Number Identification Service is a telephone service that enables a company to identify which telephone number was dialed. Users could configure DNIS to allow the IP phones to display which trunk is passing the call.
DID Number	This number is used to identify which line of the trunk is passing the call.
DNIS Name	A name for this DNIS, when a call reaches the selected trunk, the name will be displayed on the ringing phone.

DOD

DOD (Direct Outward Dialing) is the caller ID displayed when dialing out. Before configuring this, please make sure the provider supports this feature.



- **Global DOD**
  Configure Global direct outward dialing number. DOD (Direct Outward Dialing) is the caller ID displayed when dialing out. Before configuring this, please make sure the carrier supports this feature.
- **Add One DOD with Multiple Extensions**
  Enter one DOD number and select multiple extensions.
- **Bind Consecutive DOD Numbers to Multiple Extensions**
  Enter the DOD number range and select the extensions.



T1/PRI Trunk

The eSIP 200x supports expanding up to 2 digital trunk ports, while the 500x supports expanding up to 3 digital trunk ports.

Go to **Settings > PBX > Trunks** to edit the digital trunk.

Note: Choosing a different trunk signaling will have different settings.

Basic Settings

PRI Signaling	
Trunk Name	Give this trunk a name to help you identify this trunk.
Interface Type	Specify the interface type according to the trunk specification.
Framing	Choose the frame format for this trunk. The options are: ESF, D4
Line Code	Choose the line code for this trunk. The options are: B8ZS, AMI
Codec	Choose the codec for this trunk.
Echo Cancellation	This option enables or disables echo cancellation. The default is checked.
D Channel	Set the channel used to carry control information and signaling information. Enter a channel number from 1 to 24.
Switch Type	Configure the switch type according to the direction provided by your service provider.
Signaling Role	Specify whether this interface will act like the user or the network. The default is User.
Overlap Dial	Define whether the system can dial this switch using overlap digits or not. If you need Direct Dial-in, then enable this option. The default is Disable.
MFC/R2 Signaling	
Trunk Name	Give this trunk a name to help you identify this trunk.
Framing	Choose the frame format for this trunk. The options are: ESF, D4
Line Code	Choose the line code for this trunk. The options are: B8ZS, AMI
Echo Cancellation	This option enables or disables echo cancellation. The default is checked.
Variant	Set the MFC/R2 variant.
Category	Set the category of calling party.
MAX DNIS	Select max amount of DNIS to ask for If you wish to customize, enter the value in the text box directly.
MAX ANI	Max amount of ANI to ask for if you wish to customize, enter the value in the text box directly.
SS7 Signaling	
Trunk Name	Give this trunk a name to help you identify this trunk.
Framing	Choose the frame format for this trunk. The options are: ESF, D4
Line Code	Choose the line code for this trunk. The options are: B8ZS, AMI
Codec	Choose the codec for this trunk.
Echo Cancellation	This option enables or disables echo cancellation. The default is checked.
D Channel	Set the channel used to carry control information and signaling information. Enter a channel number from 1 to 24.
Variant	Specify the SS7 Signaling variant. The options are: • ITU: 14 bits • ANSI: 24 bits
Link set	Define SS7 link set numbers.
Network Indicator	Specify the network indicator according to the network environment.
SLC	Specify the Signaling Link Code.
OPC	Specify the Originating Point Code. This is generally assigned by your carrier.
DPC	Specify the Destination Point Code. This is generally assigned by your carrier.
E&M Signaling	

Trunk Name	Give this trunk a name to help you identify this trunk.
Interface Type	Specify the interface type according to the trunk specification.
Framing	Choose the frame format for this trunk. The options are: ESF, D4
Line Code	Choose the line code for this trunk. The options are: B8ZS, AMI
Codec	Choose the codec for this trunk.
Echo Cancellation	This option enables or disables echo cancellation. The default is checked.

Advanced

PRI Signaling	
Facility-based ISDN Supplementary Services	Decide whether to enable transmission of facility-based ISDN supplementary services (such as caller name from CPE over facility) or not. The default is checked.
Reset Interval	This sets the time in seconds between restart of unused B channels. Set the interval to Never if you don't like the channel to restart. The default is Never.
PRI Indication	Tells how the server should indicate busy and congestion to the switch/user. The options are: • In-band: server plays indication tones without answering; not available on all PRI/BRI subscription lines; • Out-of-Band: server disconnects with busy/congestion information code so the switch will play the indication tones to the caller. The default is Out-of-Band.
Enable DNIS	Dialed Number Identification Service is a telephone service that enables a company to identify which telephone number was dialed. Users could configure DNIS to allow the IP phones to display which trunk is passing the call.
DID Number	This number is used to identify which line of the trunk is passing the call.
DNIS Name	A name for this DNIS, when a call reaches the selected trunk, the name will be displayed on the ringing phone.
Dial Plan	
Calling Party Numbering Plan	Select the Calling Party Numbering Plan.
Calling Party Numbering Type	Select the Calling Party Numbering Type.
Called Party Numbering Plan	Select the Called Party Numbering Plan.
Called Party Numbering Type	Select the Called Party Numbering Type.
Presentation Indicator	The PI provides instructions on whether or not the provided calling line identity is allowed to be presented, or indicate that the number is not available.
Screen Indicator	The SI provides information on the source and the quality of the provided information.
ISDN Dialplan	ISDN/telephony numbering plan (Recommendation E.164)
International Prefix	Dialplan: '(Remote Dialplan: ISDN +) Remote Number Type: international'.
National Prefix	Dialplan: '(Remote Dialplan:ISDN +)Remote Number Type:national'.
Local Prefix	Dialplan: '(Remote Dialplan:ISDN +)Remote Number Type:subscriber'.
Private Prefix	Dialplan: 'Remote Dialplan:private + Remote Number Type: subscriber'.
Unknown Prefix	Dialplan: '(Remote Dialplan:ISDN +)Remote Number Type:unknown'.
MFC/R2 Signaling	
Enable DNIS	Dialed Number Identification Service is a telephone service that enables a company to identify which telephone number was dialed. Users could configure DNIS to allow the IP phones to display which trunk is passing the call.
DID Number	This number is used to identify which line of the trunk is passing the call.
DNIS Name	A name for this DNIS, when a call reaches the selected trunk, the name

	will be displayed on the ringing phone.
Forced Release	This option enables or disables forced release of channel. The default is unchecked.
Immediate Accept	Most variants of MFC/R2 offer a way to go directly to the call accepted state, by passing the use of group B and II tones. This option enables or disables the use of that feature for incoming calls. The default is unchecked.
Double Answer	Block collect calls with double answer. This will cause that every answer signal is changed by answer -> clear back -> answer. The default is unchecked.
Charge Calls	Whether or not report to the other end "accept call with charge".
Allow Collect Calls	Specify whether to accept collect calls or not.
MF Back Timeout MF	MFC/R2 value in milliseconds for the MF timeout. The default is None.
Metering Pulse Timeout	MFC/R2 value in milliseconds for the metering pulse timeout. Enter -1 to use the default value.
DTMF Detection Timeout	Specify the DTMF Detection timeout in milliseconds. The default is 5000 ms.
Incoming DTMF Mode	Specify the incoming DTMF mode.
First Number of Get	Choose which number to get first.
Outgoing DTMF Mode	Specify the outgoing DTMF mode.
SS7 Signaling	
Enable DNIS	Dialed Number Identification Service is a telephone service that enables a company to identify which telephone number was dialed. Users could configure DNIS to allow the IP phones to display which trunk is passing the call.
DID Number	This number is used to identify which line of the trunk is passing the call.
DNIS Name	A name for this DNIS, when a call reaches the selected trunk, the name will be displayed on the ringing phone.
Start CIC No.	Specify the Circuit Identification Code number of the first B channel of E1 line (SS7). **Note:** the suggested value is the multiples of 32 plus 1, for example: 1, 33, 65...
Calling Party Number Type	Calling Party Numbering Type
Called Party Number Type	Called Party Number Type

DOD

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out. Before configuring this, please make sure the provider supports this feature.

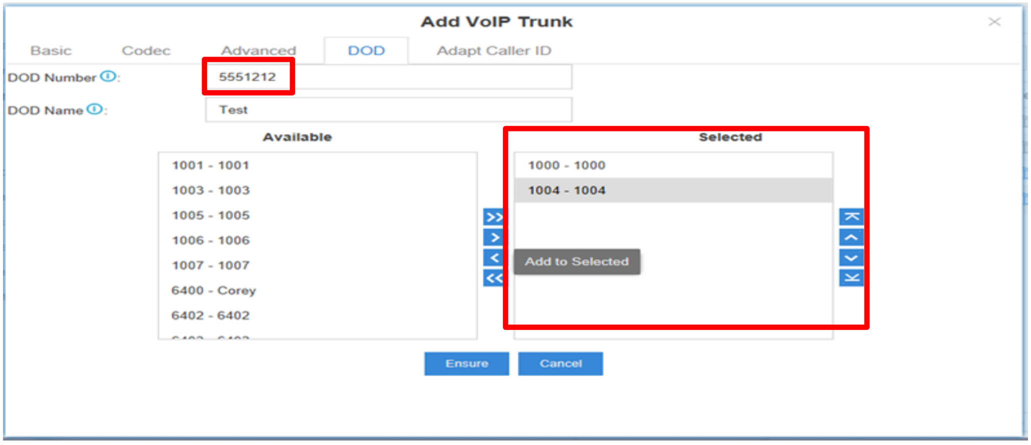
- **Global DOD**

  Configure Global direct outward dialing number. DOD (Direct Outward Dialing) is the caller ID displayed when dialing out. Before configuring this, please make sure the carrier supports this feature.

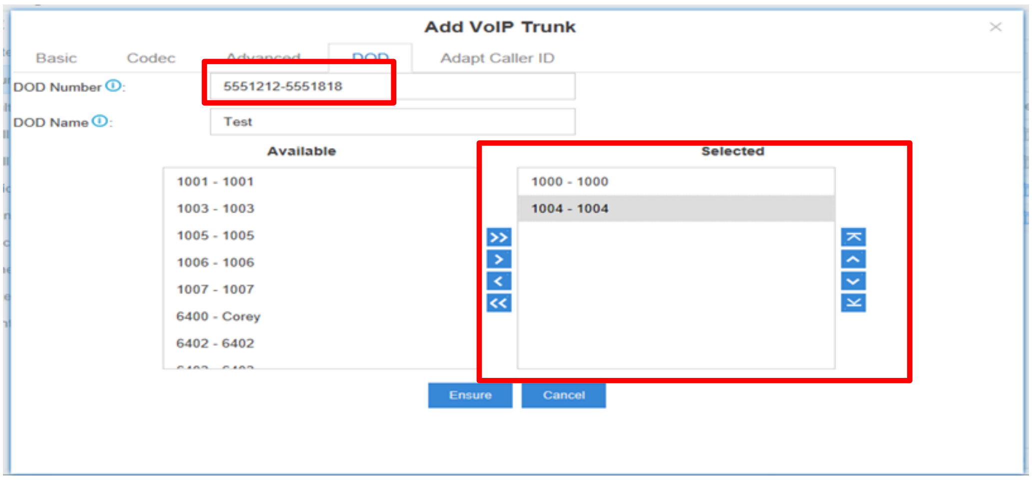
(continued on next page)

- **Add One DOD with Multiple Extensions**

Enter one DOD number and select multiple extensions.



- **Bind Consecutive DOD Numbers to Multiple Extensions**
  Enter the DOD number range and select the extensions.



Call Control

This chapter shows you how to control outgoing calls and incoming calls.

- Inbound Routes
- Outbound Routes
- Auto CLIP Routes
- Time Conditions

Inbound Routes

When a call comes into the eSIP Evolution Series from the outside, the eSIP Evolution Series needs to know where to direct it. It can be directed to an extension, a ring group, a queue or an Auto Attendant (IVR) etc.

Go to **Settings** > **PBX** > **Call Control** > **Inbound Routes** to edit inbound routes. Please check the inbound route configuration parameters below.

1. Name

   Give this inbound route a brief name to help you identify it.
2. DID Pattern

   Match the DID Pattern in this field to pass incoming call through. Leave this blank to match calls with any or no DID info. You can use a pattern match to map a range of numbers. Only Peer to Peer Trunks, PRI trunks, and SIP Trunks need to configure this option.
   In patterns, the following characters have special meanings:

Patterns	
X	Refers to any digit between 0 and 9
Z	Refers to any digit between 1 and 9
N	Refers to any digit between 2 and 9
[###]	Refers to any digit in the brackets, example [123] is 1 or 2 or 3. Note that multiple numbers can be separated by commas and ranges of numbers can be specified with a dash ([1.3.6-8]) would match the numbers 1,3,6,7 and 8.
. (dot)	Wildcard. Match any number of anything.
!	Used to initiate call processing as soon as it can be determined that no other matches are possible.

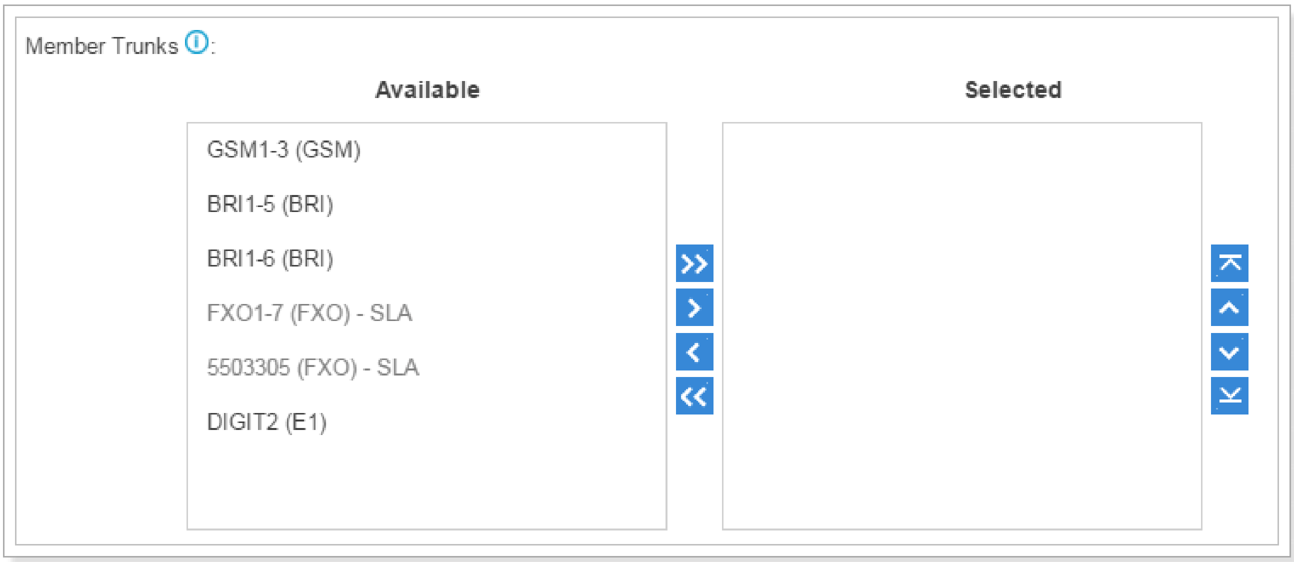
   If you want to route consecutive DID numbers to a range of consecutive extensions directly through SIP, SIP Peer to Peer, IAX Peer to Peer trunk, you need to enter the DID number range (separate the first number and the last number by “-”), choose the Destination as Extension Range, and fill in the relevant extension numbers (separated by “-”).

3. Caller ID Pattern

   Define the Caller ID Number that is allowed to call in through this inbound route. Leave this field blank to match any or no CID info. You can also use patterns match to map a range of numbers. Press Enter to input multiple patterns.

4. Member Trunks

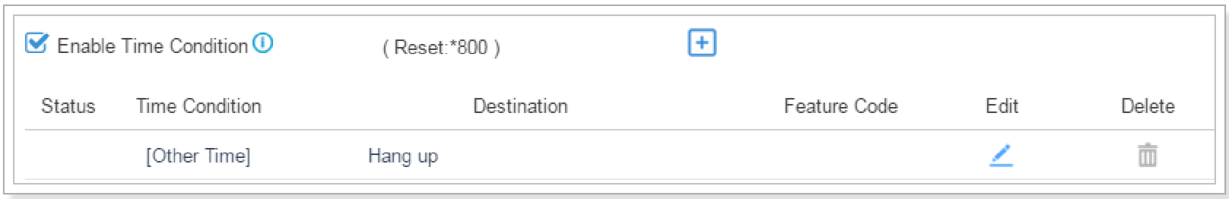
   Select which trunks will be used in this route. To make a trunk a member of this route, please move it to the “Selected” box.



5. Enable Time Condition

   Decide if you want to route incoming calls based on Time Condition.

   - If disabled, all calls will be routed to the Destination.
   - If enabled, you could route calls to different destinations at different time. Calls that do not match the time periods will be routed to “Other Time” destination. The system will assign each Time Condition with a feature code, so you could use this code to force change the destination of a Time Condition and restore to its original destination.



6. Distinctive Ring Tone

   The system supports mapping to custom ring tone files. For example, if you configure the distinctive ringing for custom ring tone to "Family", the ring tone will be played if the phone receives the incoming call.

7. Enable Fax Detection

   Decide if you want to enable Fax Detection.

   - If disabled, the system will not detect fax tone nor will it send fax tone.
   - If enabled, the system will send the fax to Fax Destination if a fax tone is detected.

Fax Destination

Sets the destination where to send the fax to. You can set it to:

- Extension: Send the fax to the designated extension. If it is an FXS extension, the fax will be sent to the FXS fort (fax machine).
- Fax to Email: Send the fax as an email attachment to the designated email address, which could be associated to an extension or a custom one.

  Note: Please make sure the sender email address is correctly configured in “System > Email”.

Outbound Routes

An outbound route works like a traffic cop giving directions to road users to use a predefined route to reach a predefined destination. Outbound routes are used to specify what numbers are allowed to go out a particular route. When a call is placed, the actual number dialed by the user is compared with the dial patterns in each route (from highest to lowest priority) until a match is found. If no match is found, the call fails. If the number dialed matches a pattern in more than one route, only the rules with the highest priority in the route are used.

Notes:

- The eSIP Evolution Series compares the dialed number with the pattern that you have defined in the first Outbound Route listed. If it matches, it will initiate the call using the selected trunks. If it does not, it will compare the number with the pattern you have defined in the second Outbound Route listed, and so on. The outbound route which is in a higher position will be matched first.
- Adjust the outbound route sequence by clicking these buttons .

Go to **Settings** > **PBX** > **Call Control** > **Outbound Routes** to edit outbound routes. Please check the outbound route configuration parameters below.

1. Name

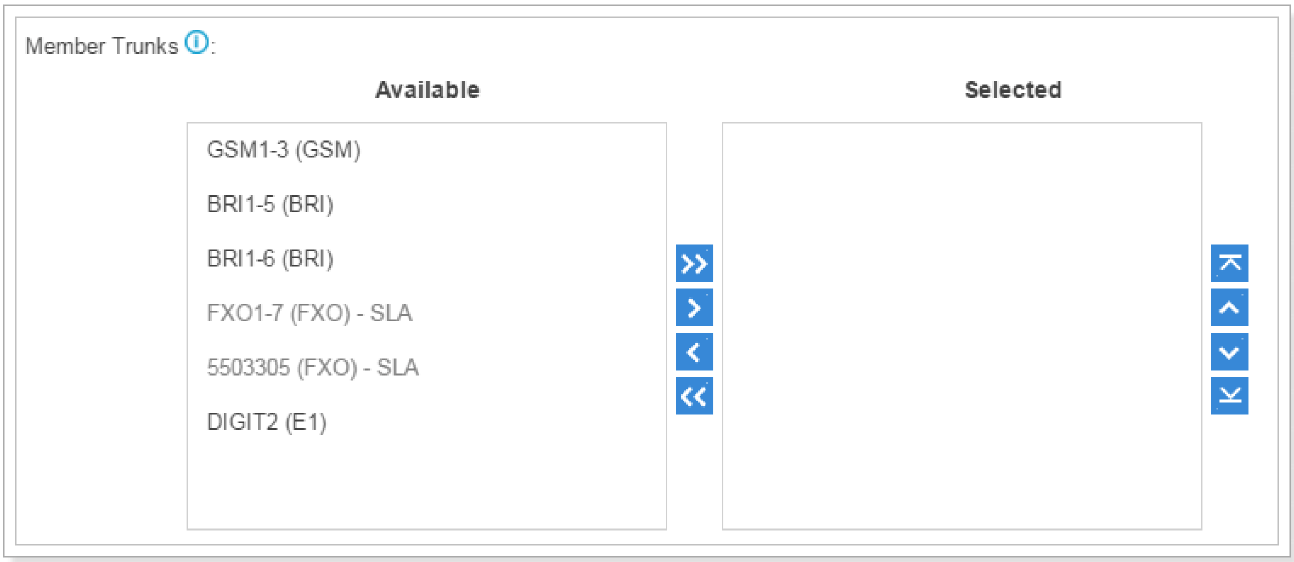
   Give this outbound route a brief name to help you identify it.
2. Dial Patterns

   Outbound calls that match this dial pattern will use this outbound route.

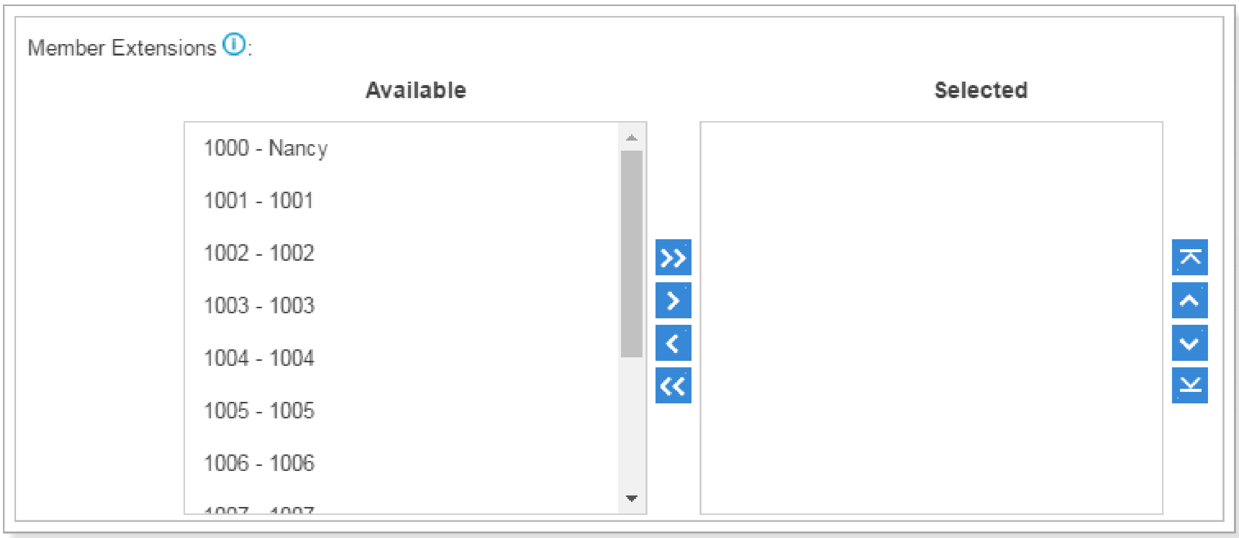
Patterns	
X	Refers to any digit between 0 and 9
Z	Refers to any digit between 1 and 9
N	Refers to any digit between 2 and 9
[###]	Refers to any digit in the brackets, example [123] is 1 or 2 or 3. Note that multiple numbers can be separated by commas and ranges of numbers can be specified with a dash ([1,3,6-8]) would match the numbers 1,3,6,7 and 8.
. (dot)	Wildcard. Match any number of anything.
!	Used to initiate call processing as soon as it can be determined that no other matches are possible.
Strip	
Administrators can specify the number of digits that will be stripped from the front of the phone number before the call is placed. For example, if users must press 0 before dialing a phone number, one digit should be stripped from the dial string before the call is placed.	
Prepend	
Digits to prepend to a successful match. If the dialed number matches the patterns, then this will be prepended before sending to the trunks. For example if a trunk requires 10-digit dialing, but users are more comfortable with 7-digit dialing, this field could be used to prepend a 3-digit area code to all 7-digit phone numbers before the calls are placed. When using analog trunks, a “w” character may also be prepended to provide a slight delay before dialing.	

3. Member Trunks

Select which trunks will be used in this route.



4. Member Extensions
   Select extensions that will be permitted to use this outbound route.

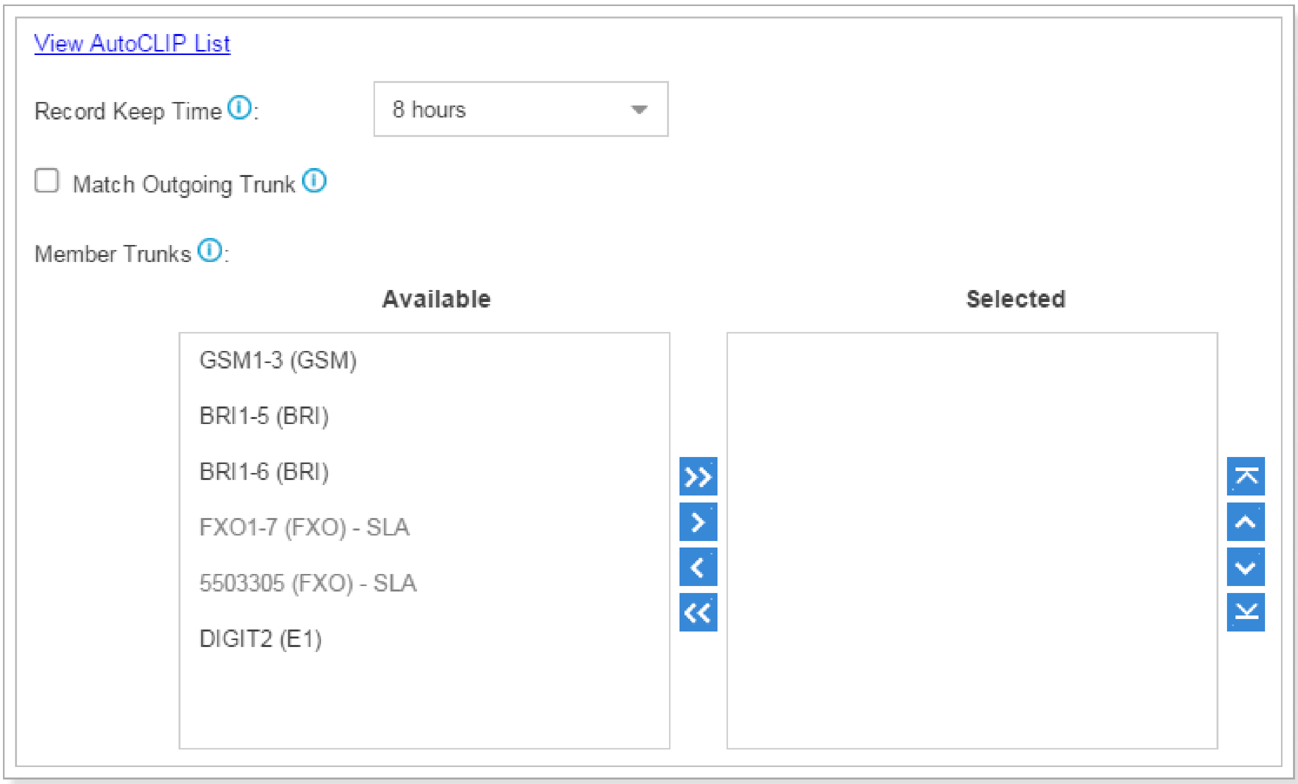


5. Password
   You can prompt users for a password before allowing calls to progress. The options are:
   - None
   - PIN List: select a list of PIN
   - Password: enter a single password which will be needed when dialing through this outbound route
6. Rrmemory Hunt
   Round robin with memory, remembers which trunk was used last time, and then use the next available trunk to call out.
7. Time Condition
   This defines the time conditions during which, use of this outbound route will be allowed.

Auto CLIP Routes

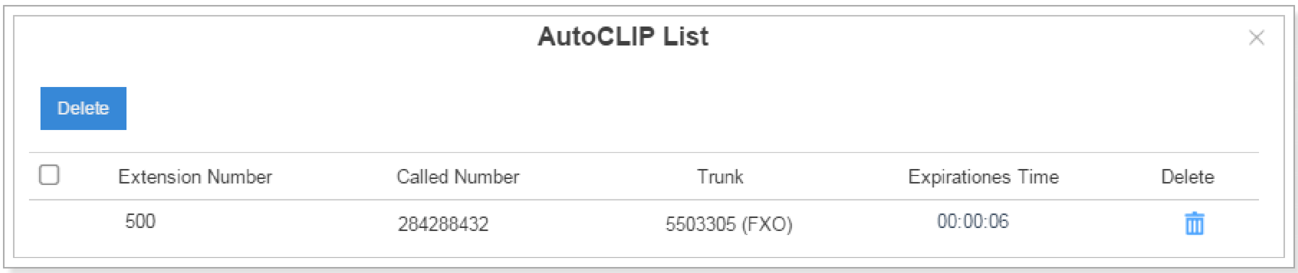
The system automatically stores information about outgoing calls to the AutoCLIP routing table. When a person calls back the call is routed directly to the original number.

Go to **Settings** > **PBX** > **Call Control** > **Auto CLIP Routes** to configure Auto CLIP:



- **Record Keep Time:** set the time duration for which records should be kept in the AutoCLIP List. Default is 8 hours.
- **Match Outgoing Trunk:** if enabled, only the incoming call that came to the server through the same trunk which made the call will be match against the AutoCLIP List.
- **Member Trunks:** choose the trunks, AutoCLIP Route will apply to the selected trunks.

Click View AutoCLIP List to view the records. In the AutoCLIP List you can see the record of the unconnected call.



As the above figure shows, when the Caller (284288432) has a missed call and returns the call, he will be directly forwarded to extension 500 as shown in the AutoCLIP List.

SLA

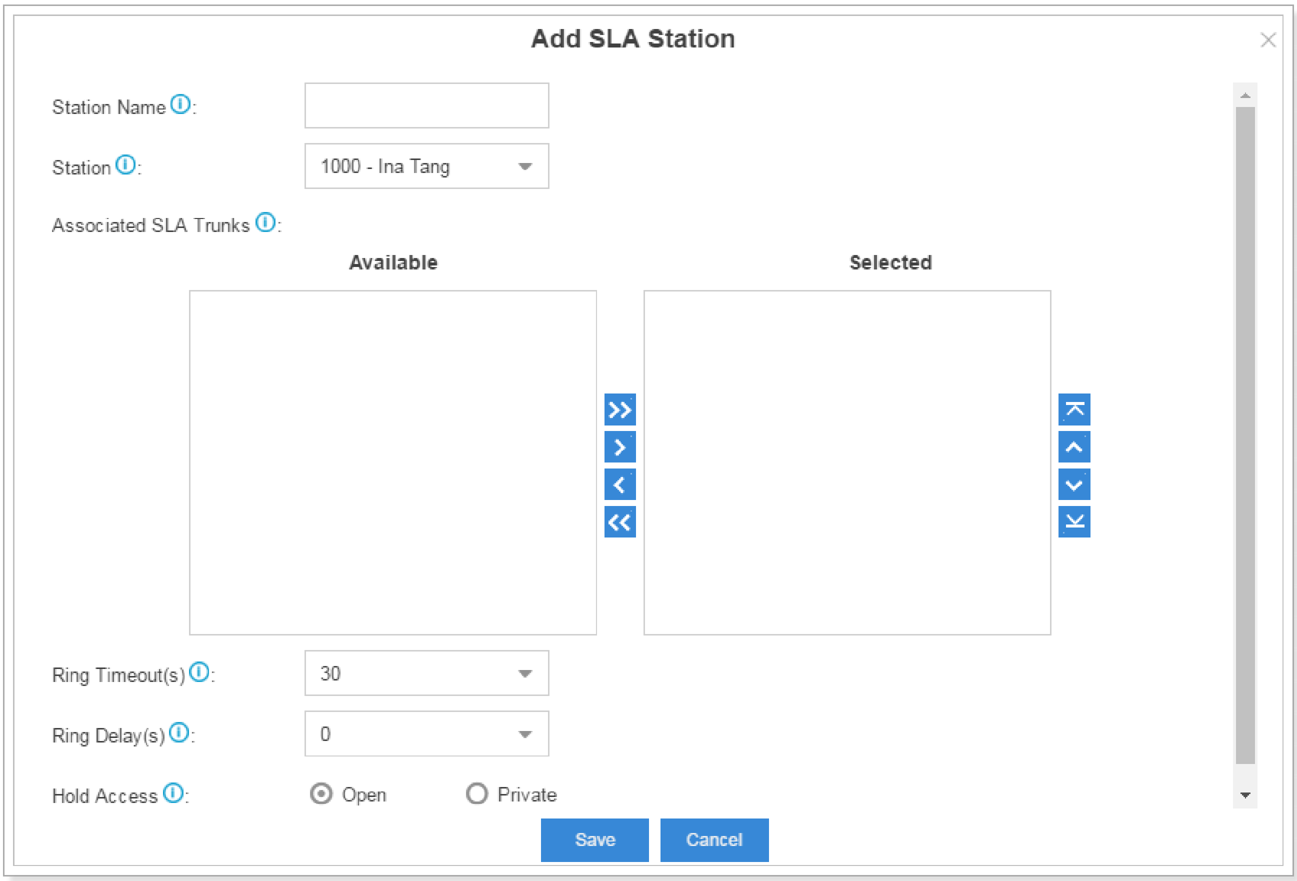
Shared Line Appearance (SLA) feature helps users share SIP trunks and FXO trunks. It also helps to monitor the status of the shared line. SLA feature works with BLF keys on IP phones.

- When an incoming call is received, all the SLA stations are informed of it and may join it if the shared line allows to barge in.
- When an outgoing call is made by one SLA station, all members shared with the same line are informed about the call, and will be blocked from this line until the line goes back to idle or the call is put on hold.

To use SLA, you need do the following:

- Enable SLA feature on a FXO trunk or VoIP trunk.
- Create SLA Stations.
- Configure BLF keys for the shared line on the stations' IP phones. The BLF key value is "**extension number_trunkname**".

Go to **Settings** > **PBX** > **Call Control** > **SLA**, click Add to create SLA stations.



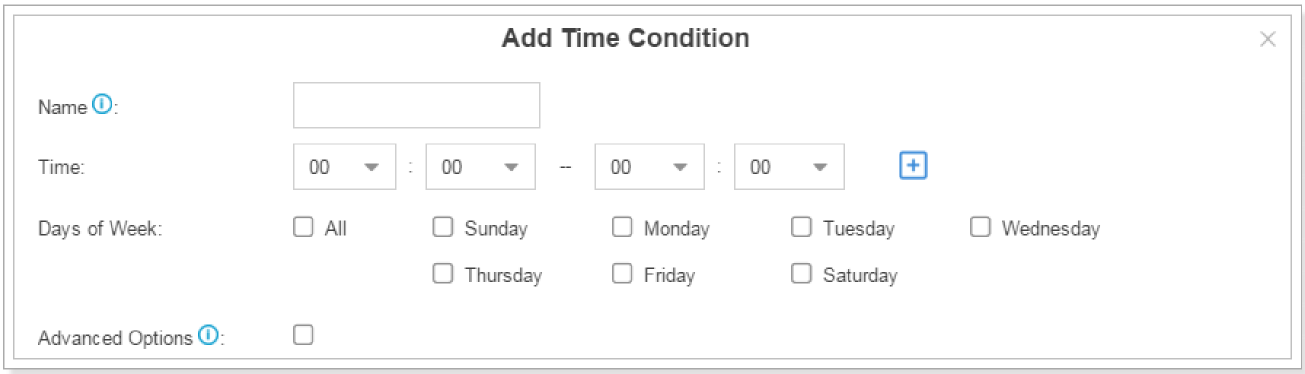
- **Station Name:** set a name for the SLA name.
- **Station:** choose a SIP extension to monitor and use the SLA trunks.
- **Associated SLA Trunks:** choose the SLA trunks.
- **Ring Timeout:** set the ring timeout in seconds, phone will stop ringing after the time defined.
- **Ring Delay:** set the delay time in seconds. Phone will delay ringing after the time defined. Ring Delay time cannot be longer than "Ring Timeout".
- **Hold Access:** specify hold permission for the station.
    - **Open:** any station can place this trunk on hold and any other station is allowed to take it back off of hold.
    - **Private:** only the station that placed the trunk on hold is allowed to take it back off of hold.

Time Conditions

On the Time Condition page, you can create time groups. A time group is a list of times against which incoming or outgoing calls are checked. The rules specify a time range, by the time, day of the week, day of the month, and month of the year. Time conditions can be assigned to an inbound route, which control the destination of a call based on the time. Time conditions can also be assigned to an outbound route in order to limit the use of that route.

Add Time Condition

Go to **Settings** > **PBX** > **Call Control** > **Time Conditions**, click Add Time Condition to add time condition.

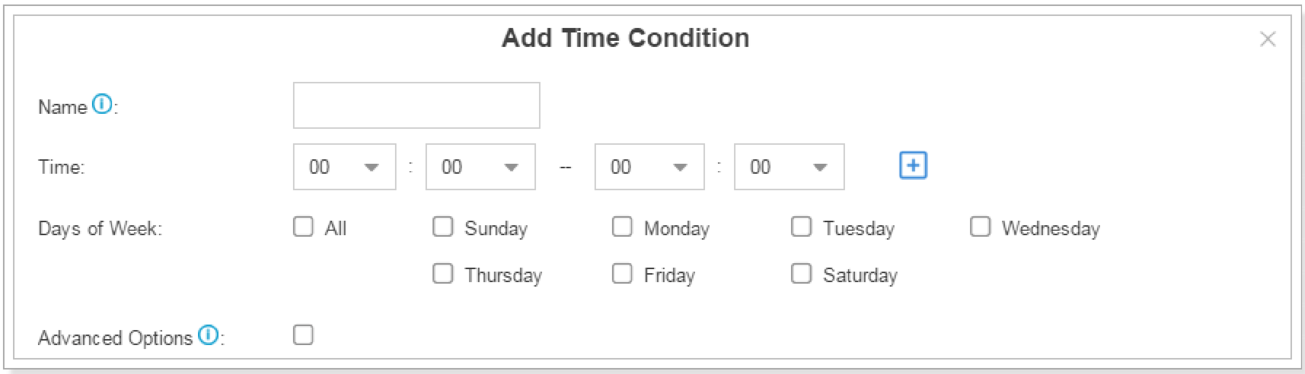


- **Name**: give this Time Condition a brief name to help you identify it.
- **Time**: this is where you will define a time range. You can define multiple ranges in the same time group by clicking .
- **Days of Week**: select a week day, month day, and/or month range in which you want this time range to apply.
- **Advanced Options**: this option is disabled by default. If it is enabled, you need to set the month and the day of the month. If it is disabled, it means that the time range defined above will apply to every day of the month, every month of the year.

Add a Holiday

After you have defined your office time conditions, you may need to create a holiday time groups. For example, you want to create a Holiday for New Year's Day.

Click Holiday tab and click Add Holiday to add a holiday.



Assigning Time Conditions to Inbound Routes

The created Time Conditions will become available for selection in the Inbound Routes.

Assigning Time Conditions to Outbound Routes

You can also assign Time Conditions to outbound routes, which may help you to control the route that can be used. For example, you can limit the users to make outbound calls when your office is closed.

Call Features

This chapter explains various call features on ESI's eSIP Evolution Series.

- IVR
- Ring Group
- Queue
- Conference
- Pickup Group
- Speed Dial
- Callback
- DISA
- Blacklist/Whitelist
- Pin List
- Paging/Intercom
- SMS

IVR

Like most organizations, where possible, we would like to route incoming calls to an Auto Attendant. You can create one or more IVR (Auto Attendant) on the eSIP Evolution Series to achieve it. When calls are routed to an IVR, the eSIP Evolution Series will play a recording, prompting the caller to select from the available options. For example: “Welcome to XX, press 1 for Sales and press 2 for Technical Support”.
Go to **Settings** > **PBX** > **Call Features** > **IVR** to configure IVR.

- Click Add to add a new IVR.
- Click Delete to delete the selected IVR.
- Click to edit one IVR.
- Click to delete one IVR.

Please check the IVR configuration parameters below.

Basic Settings	
Number	ESI's eSIP Evolution Series treats an IVR as an extension; you can dial this extension number to reach the IVR from internal extensions.
Name	Give this IVR a brief name to help you identify it.
Prompt	The prompt that will be played when the caller reaches this IVR.
Prompt Repeat Count	The number of times that the selected IVR prompt will be played.
Response Timeout	The number of seconds to wait for a digit input after prompt.
Digit Timeout	How long (in seconds) to wait for the caller to enter an option on their phone keypad before we consider it timed out and it follows the Timeout Destination as defined below.
Dial Extension	If this option is enabled, the callers can enter a user's extension number when entering the IVR to go direct to the users.
Dial Outbound Routes	Allow the caller to dial through outbound routes.
Keypress Events	

Key Press Event 0 1 2 3 4 5 6 7 8 9 # * Timeout Invalid	Select the destination for each key pressing: digits 0-9, “#”, “*”, Timeout and Invalid. When the callers press the corresponding key, the call will be routed to: • Extension • Voicemail • Ring Group • IVR • Conference • Queue • Fax to Email • Dial by Name • Hang up

Ring Group

A ring group helps you to ring a group of extensions in a variety of simple ring strategies. For example, you could define all the technical support guys' extensions in a ring group and ring the support guys one by one.

Go to **Settings** > **PBX** > **Call Features** > **Ring Group** to configure ring groups.

- Click Add to add a new ring group.
- Click Delete to delete the selected ring groups.
- Click  to edit one ring group.
- Click  to delete one ring group.

Please check the ring group configuration parameters below.

Option	Description
Number	The extension number dialed to reach this ring group.
Name	Give this ring group a brief name to help you identify it.
Ring Strategy	Select an appropriate ring strategy for this ring group. • Ring All Simultaneously: ring all the available extensions simultaneously. • Ring Sequentially: ring each extension in the group one at a time.
Seconds to ring each member	Set the number of seconds to ring a single extension before moving to the next one.
Members	Choose the member of this ring group
Failover Destination If No Answer	Choose the failover destination.

Queue

Queues are designed to receive calls in a call center. A queue is like a virtual waiting room, in which callers wait in line to talk with the available agent. Once the caller calls in to the eSIP Evolution Series and reaches the queue, they will hear hold music and prompts, while the queue sends out the call to the logged-in and available agents. A number of configuration options on the queue help you to control how the incoming calls are routed to the agents and what callers hear and do while waiting in the line.

Go to **Settings** > **PBX** > **Call Features** > **Queue** to configure queue.

- Click Add to add a new queue.
- Click Delete to delete the selected queues.
- Click to edit one queue.
- Click to delete one queue.

Please see the queue configuration parameters below.

1. Basic Settings

Basic Settings	
Number	Use this number to dial into the queue, or transfer callers to this number to put them into the queue.
Name	Give this queue a brief name to help you identify it.
Password	You can require agents to enter a password before they can login to this queue.
Ring Strategy	This option sets the Ringing Strategy for this Queue. The options are: • Ring All: ring all available agents simultaneously until one answer. • Least Recent: ring the agent which was least recently called. • Fewest Calls: ring the agent with the fewest completed calls. • Random: ring a random agent. • Rrmemory: Round Robin with Memory, remembers where it left off in the last ring pass. • Linear: rings agents in the order specified in the configuration file.
Failover Destination	Set the failover destination.
Static Agents	This selection shows all users. Selecting a user here makes them a dynamic agent of the current queue. The dynamic agent is allowed to log in and log out the queue at any time. • Dial "Queue number" + "*" to log in the queue. • Dial "Queue number" + "**" to log out the queue.
Agent Timeout	The number of seconds an agent's phone can ring before we consider it a timeout. If you wish to customize, enter the value in the text box directly.
Agent Announcement	Announcement played to the Agent prior to bridging in the caller.
Wrap-up Time	How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call .If you wish to customize, enter the value in the text box directly. Input 0 for no delay.
Ring In Use	If set to “no”, unchecked, the queue will avoid sending calls to members whose device are known to be “in use”.
Retry	The number of seconds to wait before trying all the phones again. If you wish to customize, enter the value in the text box directly.

2. Caller Experience Settings

Caller Settings	
Music On Hold	Select the "Music on Hold" playlist for this Queue.
Caller Max Wait Time	Select the maximum number of seconds a caller can wait in a queue before being pulled out. If you wish to customize, enter the value in the text box directly. Input 0 for unlimited.
Leave When Empty	If enabled, callers already on hold will be forced out of a queue when no agents available.
Join Empty	If enabled, callers can join a queue that has no agents.
Join Announcement	Announcement played to callers once prior to joining the queue.
Caller Position Announcements	
Announce Position	Announce position of caller in the queue.
Announce Hold Time	Enabling this option causes the server to announce the hold time to the caller periodically based on the frequency timer. Either yes or no; hold time will be announced after one minute.
Frequency	How often to announce queue position and estimated hold time.
Periodic Announcements	
Prompt	Select a prompt file to play periodically.
Frequency	How often to play the periodic announcements.
Events	
Once the events settings are configured, the callers are able to press the key to enter the destination you set. Usually, a prompt should be set on **Periodic Announcements** to guide the callers to press the key.	

Conference

Conference Calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings. Conference agents can dial * to access the settings options and the admin can kick the last user out and can lock the conference room.

Go to **Settings** > **PBX** > **Call Features** > **Conference** to configure conferences.

- Click Add to add a new conference.
- Click Delete to delete the selected conferences.
- Click to edit one conference.
- Click to delete one conference.

Please see the conference configuration parameters below.

Options	Description
Number	Use this number to dial into the conference room.
Name	Give the conference a brief name to help you identify it.
Participant Password	You can require callers to enter a password before they can enter this conference. This setting is optional.
Wait For Moderator	If set, the participant couldn't hear each other until the moderator joins in the conference.
Sound Prompt	Set the sound prompt used for the login and logout of conference members.
Allow Participant To Invite	If enabled, all paticipants could press "#" to invite other users to this conference. Please note that during the invitation, the inviter will be forced out of the conference room until invitee accept or reject the call.
Moderator Password	The participant who has not been assigned a moderator role can enter the password to take on the role.
Member Moderators	Specify the moderator for this conference. The moderator does not need to enter password when joining the conference.

Join a Conference Room
Users on the eSIP Evolution Series can dial the conference extension to join the conference room. If a password is set for the conference, users would be prompted to enter a Participant Password.

How do I join the conference room if I am calling from outside (i.e. calling from my mobile phone)?
In this case, an inbound route for conferences should be set on the eSIP Evolution Series. A trunk should be selected in the inbound route and destination should be set to a conference room. When the outside users dial in the trunk number, the call will be routed to the conference room.

Manage the Conference
During the conference call, the users can manage the conference by pressing * key on their phones to access voice menu for conference room. Please see the options for the voice menu below.

Conference Administrator Menu	
1	Mute/ un-mute yourself.
2	Lock /unlock the conference.
3	Eject the last user.
4	Decrease the conference volume.
6	Increase the conference volume.
7	Decrease your volume.
8	Exit the IVR menu.

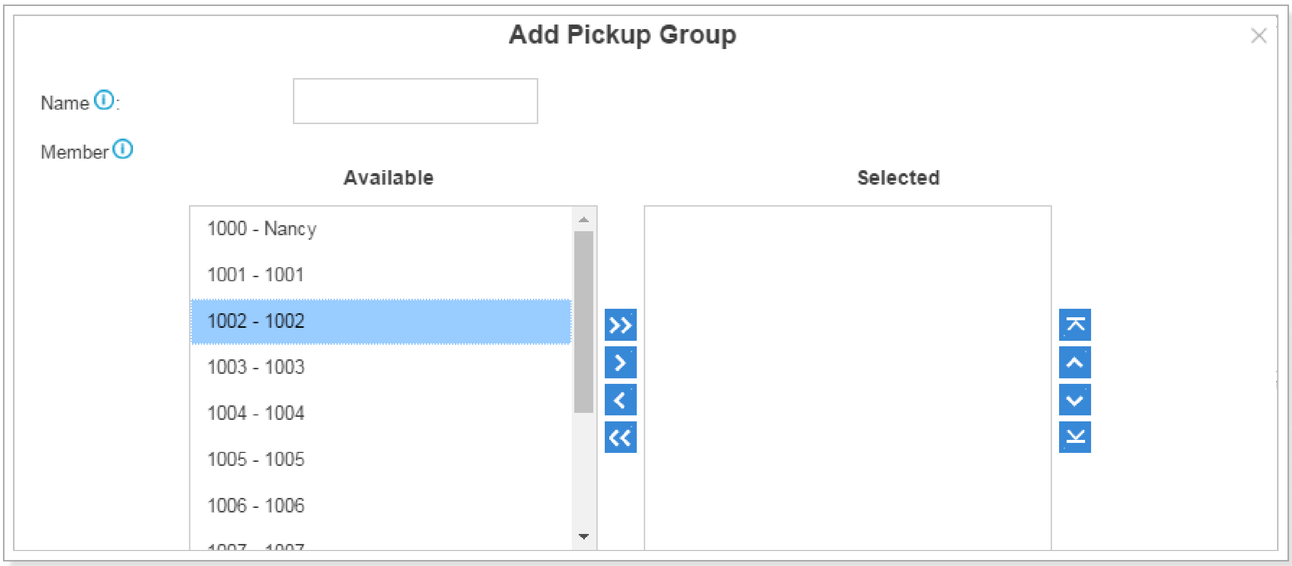
Conference Users Menu	
1	Mute/ un-mute yourself.
4	Decrease the conference volume.
6	Increase the conference volume.
7	Decrease your volume.
8	Exit the IVR menu.
9	Increase your volume.

9	Increase your volume.			

Pickup Group

Call pickup allows one to answer someone else’s call. You can add pickup groups. The default call pickup for Group Call Pickup is *4. It allows you to pick up a call from a ringing phone which is in the same group as you. Go to **Settings** > **PBX** > **Call Features** > **Pickup Group** to add pickup group.

- Click **Add** to add a new pickup group.
- Click **Delete** to delete the selected pickup groups.
- Click to edit one pickup group.
- Click to delete one pickup group.



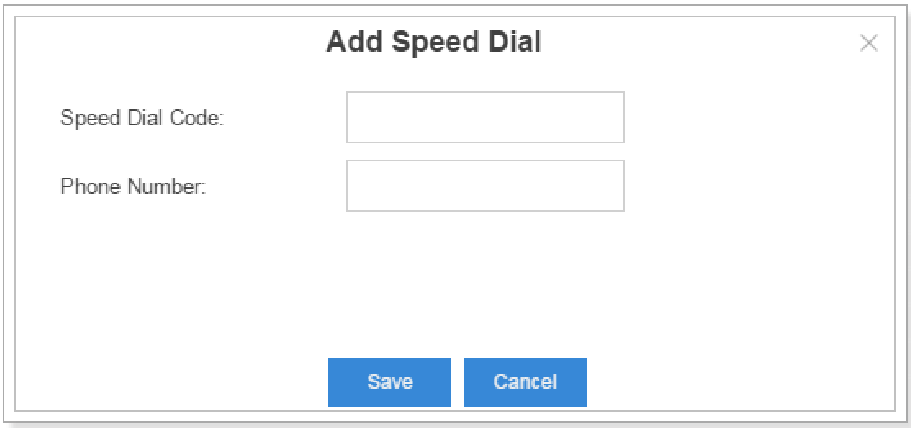
Speed Dial

Sometimes you may just need to call someone quickly without having to look up his/her phone number. You can by simply define a shortcut number. The Speed Dial feature is available on ESI's eSIP Evolution Series, allowing you to place a call by pressing a reduced number of keys.

1. Add Speed Dial

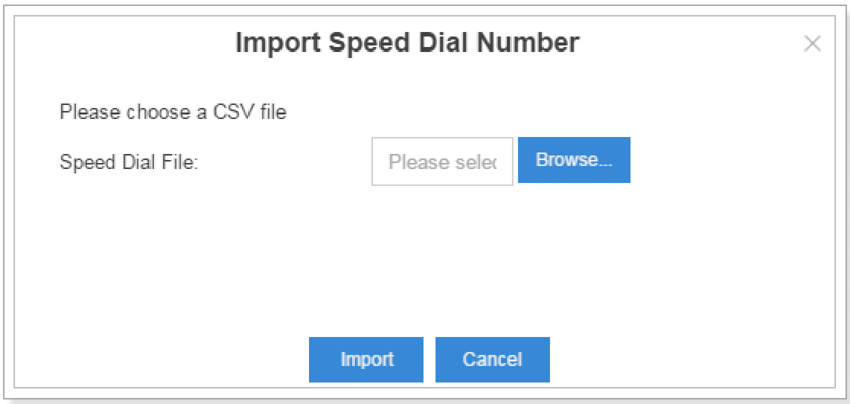
   Click **Add** to add a speed dial.

- **Speed Dial Code**: enter the speed dial code.
- **Phone Number**: enter the number you want to call.



2. Import Speed Dial

   Click **Import**, you will see a dialog window shown as below.



Click **Browse** and select the file to start uploading. The file must be a .csv file. See the sample file below. You can export a speed dial file from eSIP Evolution Series and use it as a sample to start with.

	A	B	C	D	E	F
1	Speed Dial Code	Phone Number				
2	1	3545454				
3	2	4645745656				
4	3	456576666				
5	4	67585666				
6	5	24645555				
7						

The sample csv file will result in the following speed dial in ESI's eSIP Evolution Series.

☐	Speed Dial Code	Phone Number	Edit	Delete
☐	1	3545454		
☐	2	4645745656		
☐	3	456576666		
☐	4	67585666		
☐	5	24645555		

3. Export Speed Dial

   Select the check box for the speed dial, click Export, and the selected speed dial will be exported to your local PC.

Add Delete Import Export   Speed Dial Prefix: *99   Save

☐	Speed Dial Code	Phone Number	Edit	Delete
☑	1	218737823882		
☑	2	1237823147831		
☑	3	7834273928838833		
☐	4	2347187744444		

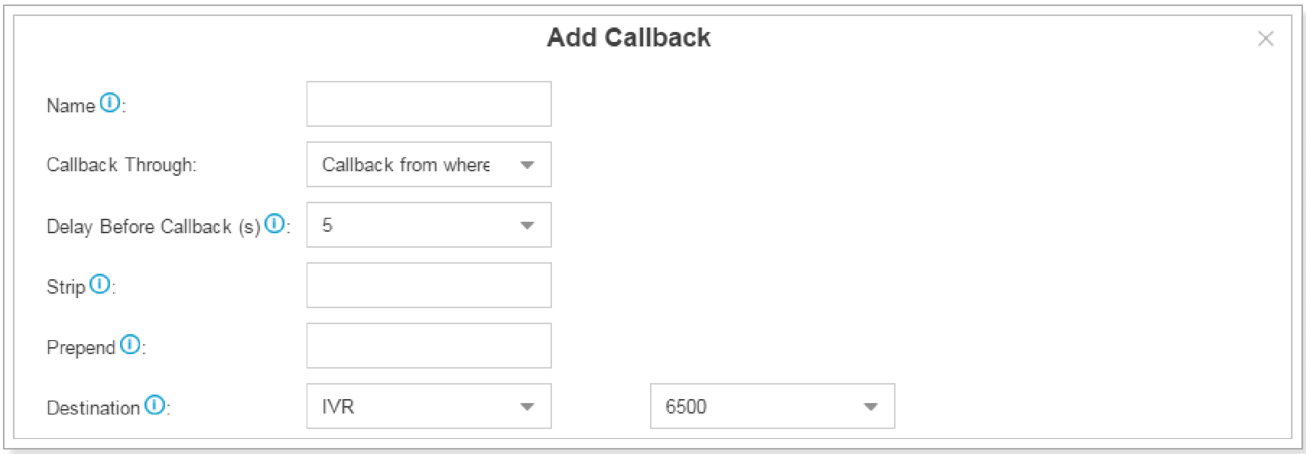
Callback

Callback feature allows callers to hang up and get called back by ESI's eSIP Evolution Series. The Callback feature could reduce the cost for the users who work out of the office using their own mobile phones.

Go to **Settings** > **PBX** > **Call Features** > **Callback** to configure Callback.

- Click Add to add a new callback.
- Click Delete to delete the selected callbacks.
- Click  to edit one callback.
- Click  to delete one callback.

To use the callback feature, you need to select callback as destination on the inbound route. Please check the callback configuration parameters below.
Note: you don't need to configure "Strip" and "Prepend" options if the trunk supports call back with the caller ID directly.

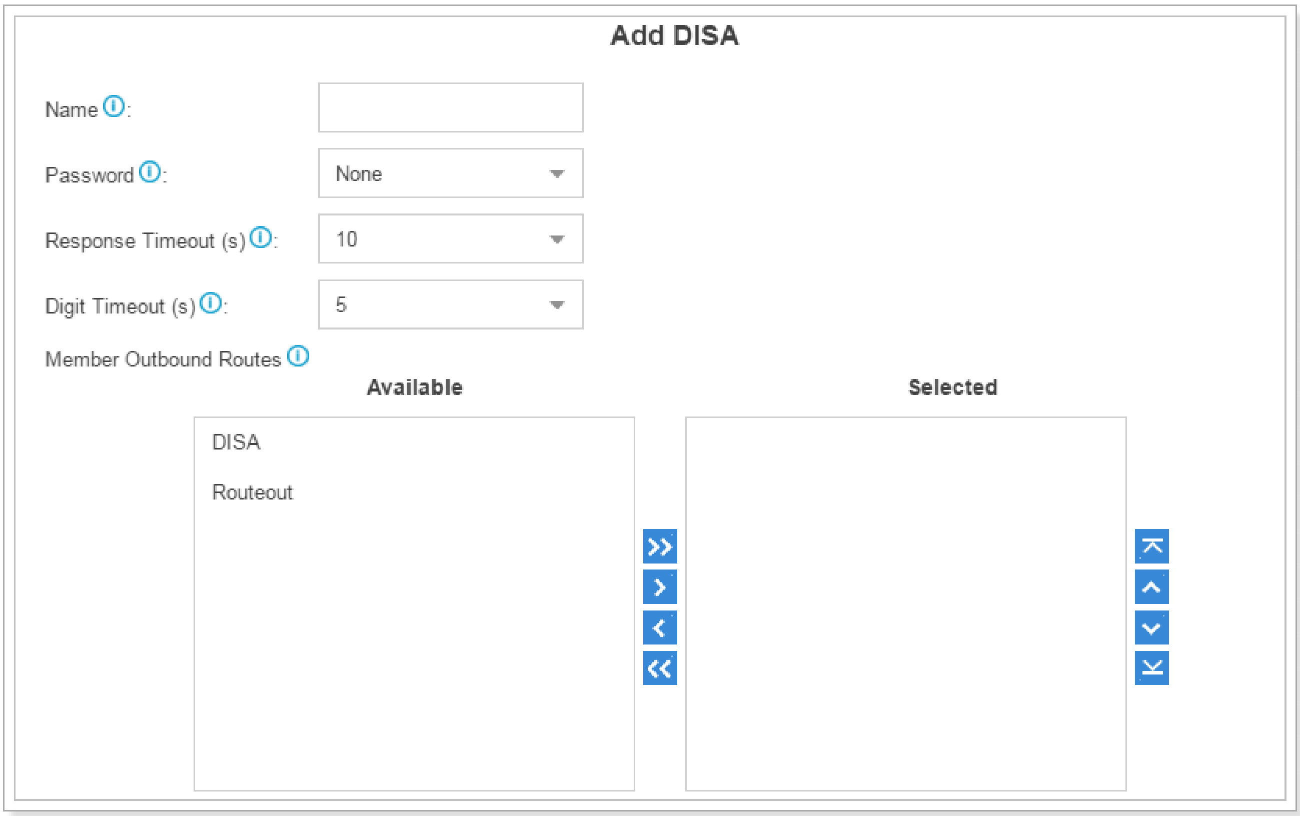


Option	Description
Name	Give this Callback a brief name to help you identify it.
Callback Through	Choose a trunk; the call will be called back through the selected trunk.
Delay Before Callback	Set the number of seconds before calling back a caller.
Strip	Defines how many digits will be stripped from the call in number before the callback is placed.
Prepend	Defines digits added before a callback number before the callback is placed.
Destination	The destination which the callback will direct the caller to.

DISA

DISA (Direct Inward System Access) allows someone calling in from outside of the eSIP Evolution Series to obtain an “internal” system dial tone and make calls as if they were using one of the internal extensions.
To use DISA, a user calls a DISA number, which invokes the DISA application. The DISA application in turn requires the user to enter a PIN number, followed by the pound key (#). If the PIN number is correct, the user will hear dial tone on which a call may be placed.

Please check the callback configuration parameters below.



Option	Description
Name	Give this DISA a brief name to help you identify it.
Password	The password for this DISA.
Response Timeout	The maximum amount of time the system will wait before hanging up the call if the user has dialed an incomplete or invalid number. The default value is 10s.
Digit Timeout	The maximum amount of time permitted between each digit when the user is dialing an extension number. The default value is: 5s.
Member Outbound Routes	Defines the outbound routes that can be accessed from this DISA.

Blacklist/Whitelist

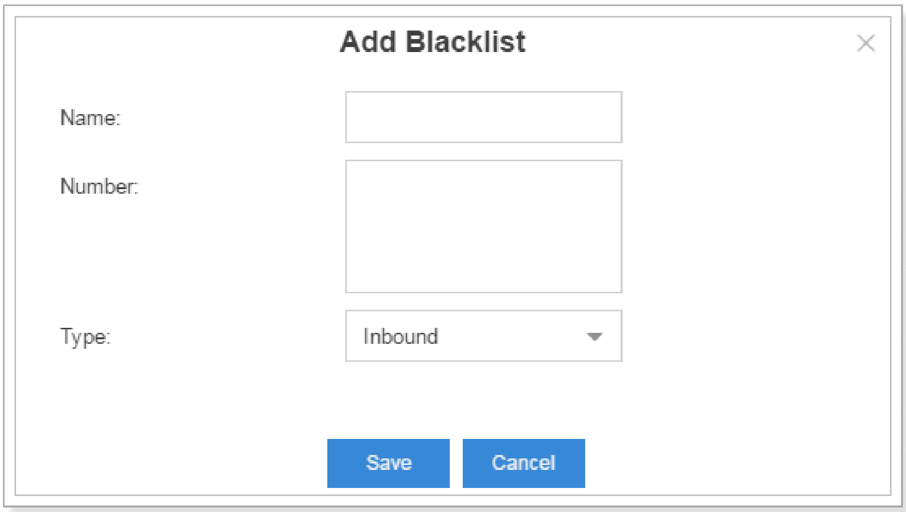
Blacklist is used to block an incoming/outgoing call. If the number of incoming or outgoing call is listed in the number blacklist, the caller will hear the following prompt: “The number you have dialed is not in service. Please check the number and try again”. The system will then disconnect the call.
Whitelist is used to allow incoming/outgoing numbers.

The system supports to block or allow 3 types of numbers:

- **Inbound**: the number would be disallowed or allowed to call into the system.
- **Outbound**: users are disallowed or allowed to call the number out from the system.
- **Both**: both inbound and outbound calls are disallowed or allowed.

1. Add Blacklist/Whitelist

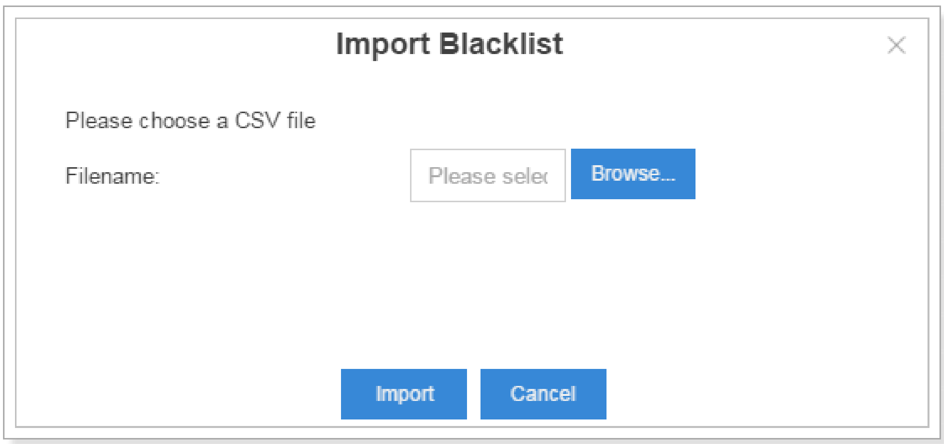
   Select Blacklist or Whitelist tag, click Add to add a number to Blacklist or Whitelist.



- **Name**: give a name for the blacklist/whitelist.
- **Number**: enter the numbers, one number per row.
- **Type**: choose the type.

2. Import Blacklist/Whitelist

   Click Import, you will see a dialog window shown as below.



Click Browse and select the file to start uploading. The file must be a .csv file. Open the file with notepad, check the sample below. You can export a blacklist/whitelist file from the eSIP Evolution Series and use it as a sample to start with.

```
Name,Number,Type
international,18288383,73829911,outbound
ads,28192828,83829920,88287373,inbound
blacklist,18283883,89388383,both
```

The sample csv file will result in the following speed dial in the eSIP Evolution Series.

☐	Name	Number	Type	Edit	Delete
☐	international	18288383,73829911	Outbound		
☐	ads	28192828,83829920,8828...	Inbound		
☐	blacklist	18283883,89388383	Both		

3. Export Blacklist/Whitelist

Select the check box of the blacklist/whitelist, click Export, the selected blacklist/whitelist will be exported to your local PC.

Pin List

PIN List is used to manage lists of PINs (numerical passwords) that can be used to access restricted features such as outbound routes. The PIN can also be presented in the CDR record.

Go to **Settings** > **PBX** > **Call Features** > **Pin List** and click Add to add Pin list.

Linking a PIN List to Outbound Routes/DISA

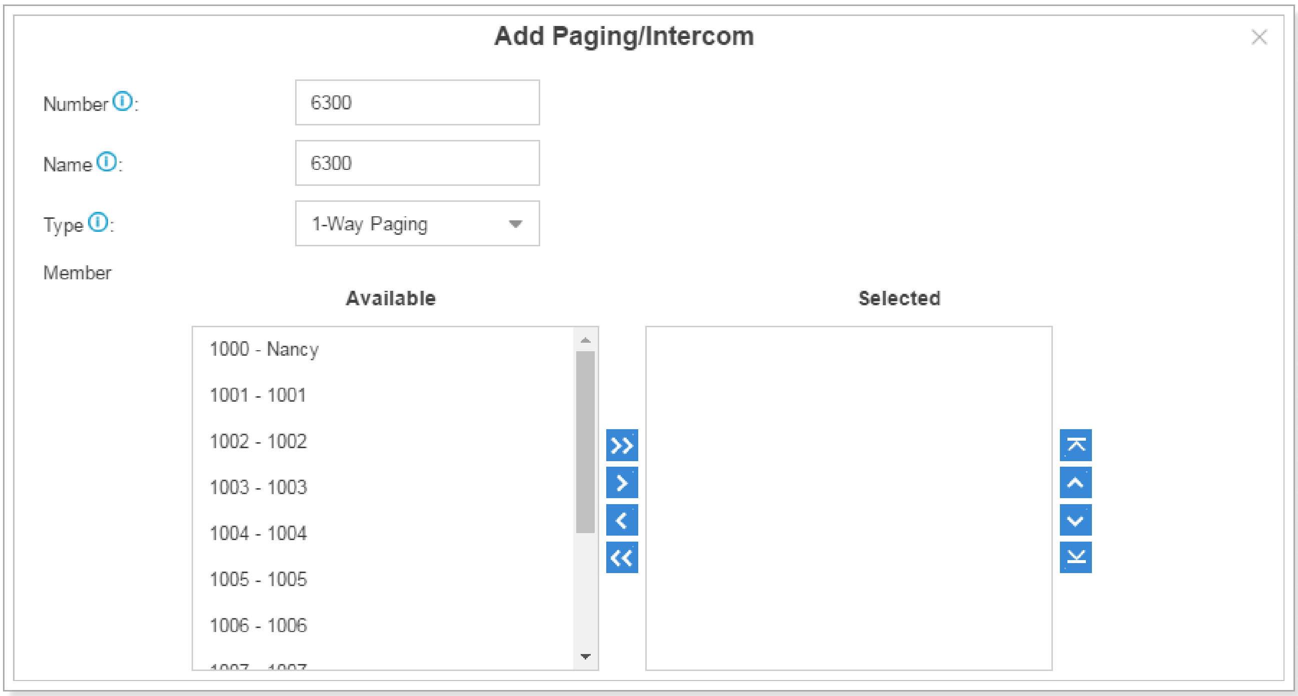
After creating PIN lists, you can link the PIN lists to Outbound Routes or DISA. On outbound route/DISA edit page, you can select the PIN list from the **Password** drop-down menu.

Paging/Intercom

Intercom is a feature that allows you to make an announcement to one extension via a phone speaker. The called party does not need to pick up the handset. It can be achieved by pressing the feature code on your phone and it is a two-way audio call.
The default Intercom feature code is *5. To make an announcement to a specific extension, you need to dial *5+ extension number on your phone. For example, make an announcement to extension 500, you need to dial *5500, then the extension 500 will be automatically picked up.

Paging is used to make an announcement over the speakerphone to a phone or group of phones. Targeted phones will not ring, but instead answer immediately into speakerphone mode. Paging is typically one way for announcements only, but you can set the paging group as a duplex mode to allow all users in the group to talk and be heard by all.

Go to **Settings > PBX > Call Features > Paging/Intercom**, click Add to add a paging group.



- **Number**: the extension number dialed to reach this Paging Group.
- **Name**: give this Paging Group a brief name to help you identify it.
- **Type**: select the mode of paging group.
  - 1-Way Paging: typically one way for announcement only.
  - 2-Way Paging: make paging duplex, allowing all users in the group to talk and be heard by all.
- **Member**: select the members of the group.

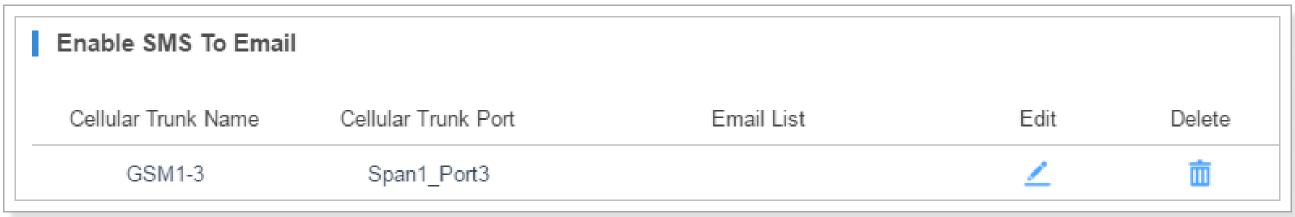
SMS

ESI's eSIP Evolution Series supports **SMS to Email** and **Email to SMS** features. To use these two features, you must do the following:

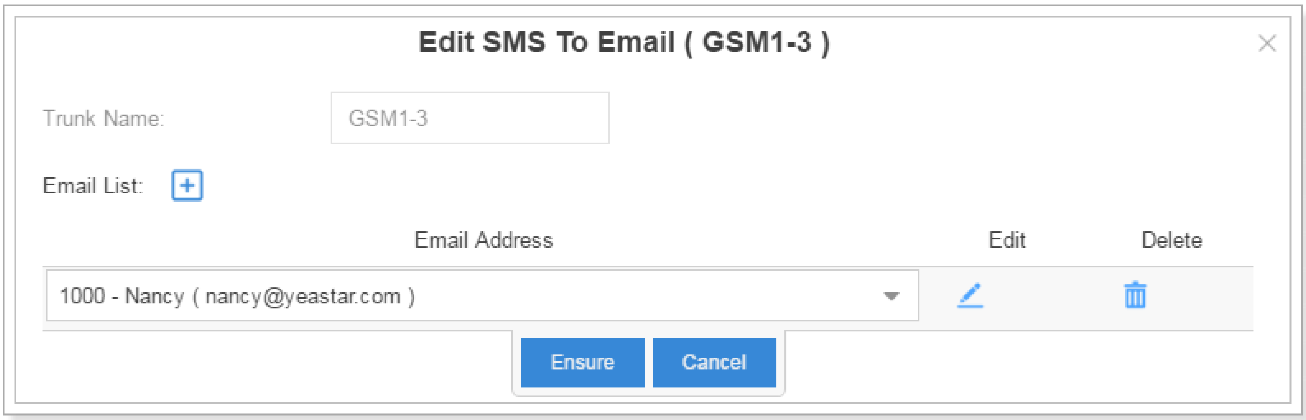
- Install **3G/4G** module on the device.
- Insert **SIM card** on the 3G/4G module.
- Check the trunk status and make sure that the 3G/4G trunk is ready to be used.
- Set an email address for the system (Settings > System > Email).

SMS to Email

SMS to Email is a feature that allows users' email to receive the SMS of a cellular network. The SMS sent to the cellular ports will be received first by the application of the eSIP system and then forwarded to the pre-configured email address (the email set in Settings > System > Email). Thus, users can receive the SMS through email.



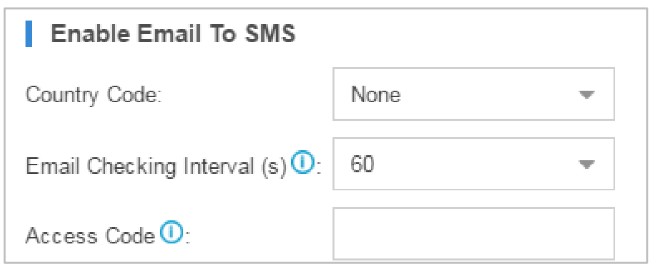
Choose a cellular trunk and click , you will see the dialog appear as below. Click to add email address then click Ensure.



When you send a SMS from your mobile to the cellular trunk's number, the SMS message will be delivered to the email addresses.

Email to SMS

Email to SMS is a feature that allows users to send SMS to mobile phone number via email. When users would like to send a SMS, they just need to send an email to the ESI system's email address, with the destination mobile phone number as the email subject. The system will then receive the email and forward the email to the cellular port, so that the email can be sent out through SMS to expected destinations.



Sending Email to SMS, the Email subject format is as below:
port:[port];num:[number];code:[code];

Note: for the 200x and 500x, you need to specify the cellular port and on which expansion board. For example, "port:2_1", means Expansion board 2 port 1 is cellular port.

1. Send Email to SMS without Access Code through default cellular Port
   Email Subject: num:[number];

2. Send Email to SMS without Access Code through a specific cellular Port
   Email Subject: port:[port];num:[number];

3. Send Email to SMS with Access Code through Default cellular Port
   Email Subject: port:[port];num:[number];code:[code];

4. Send Email to SMS with Access Code through a specific cellular Port
   Email Subject: port:[port];num:[number];code:[code];

Voice Prompts

In this chapter, we introduce how to manage voice prompts on ESI's eSIP Evolution Series, including the following sections:

- Prompt Preference
- System Prompt
- Music on Hold
- Custom Prompts

Prompt Preference

Select prompt files for the relevant options on this page.

Option	Description
Music On Hold	The music to play when a call is being held.
Play Call Forwarding Prompt	If enabled, system will play a prompt before transferring the call. Otherwise, the call will be transferred directly without any prompt. It is enabled by default.
Music On Hold for Call Forwarding	This decides what to play when the caller is put on hold during call forwarding. The options are: • Music, which will be the same with the one selected in Music on Hold. • Ringing Tone The default is to play Music.
Invalid Phone Number Prompt	The prompt to play when the dialed phone number is invalid.
Busy Line Prompt	The prompt to play when the dialed phone number is busy.
Dial Failure Prompt	The prompt to play when a dial failed due to conjunction and lack of available trunks.

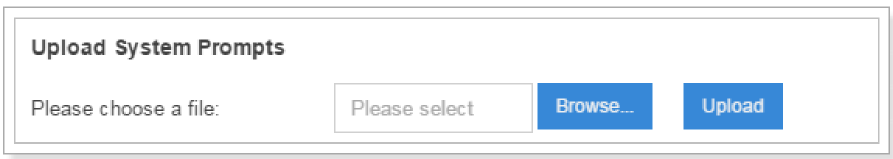
System Prompt

ESI's eSIP Evolution Series ships with ESI’s English prompt, as well as Spanish and French, loaded by default. The system supports multiple languages, though, so you could update the system prompt from a local PC, as needed.

Go to **Settings** > **PBX** > **Voice Prompt** > **System Prompt** to update the system prompt.

Upload System Prompts

Click Browse to select the system prompt file from local computer, then click Upload to start uploading.



Music on Hold

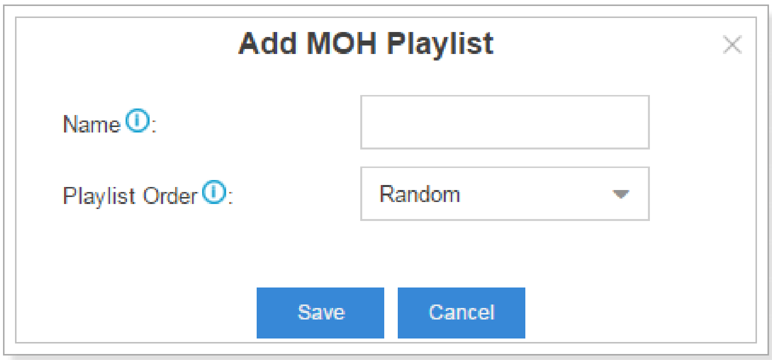
Music on hold (MOH) is the business practice of playing recorded music to fill the silence that would be heard by callers who have been placed on hold. Administrators can configure the Music on Hold Folder and upload music files to the system. The "default" Music on Hold Playlists include a variety of music files for users to use.

Go to **Settings** > **PBX** > **Voice Prompts** > **Music on Hold**.

1. Create New Playlist

   Click Create New Playlist to create a new playlist.

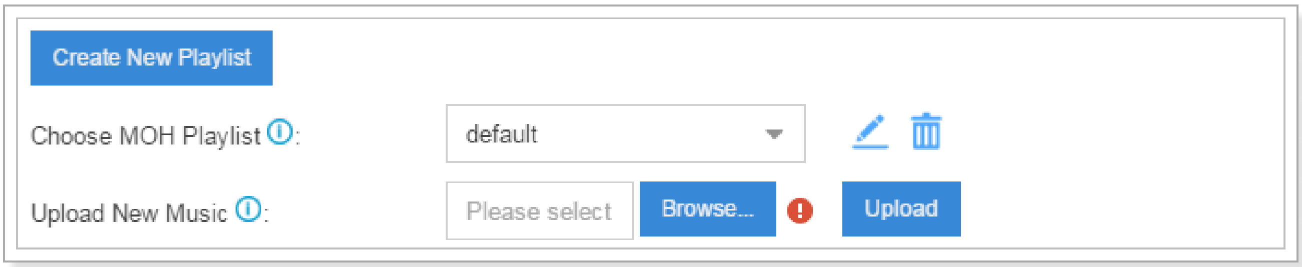
   - **Name:** give this playlist a name to help you identify it.
   - **Playlist Order:** select the playing order of the playlist.



2. Upload New Music

   Choose a MOH Playlist from the drop-down menu.

   Click Browse to select music file from your local computer, click Upload to start uploading.

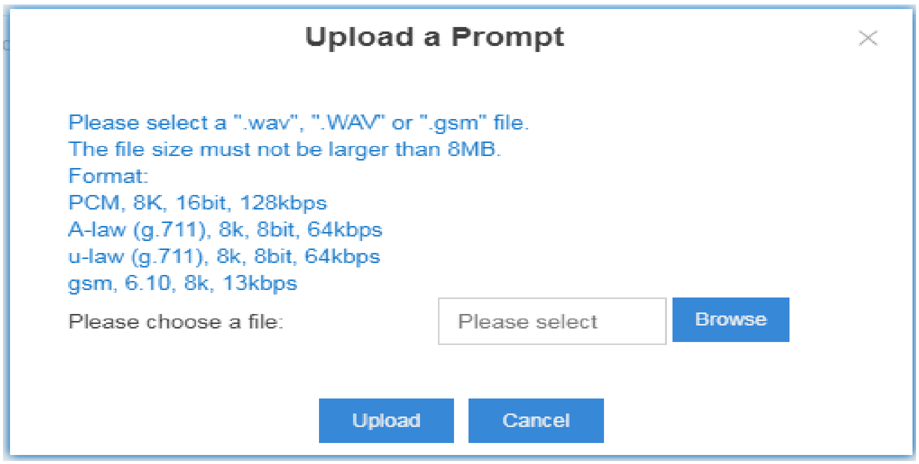


Custom Prompt

The default voice prompts and announcements in the system are suitable for almost every situation. However, you may want to use your own voice prompt to make it more meaningful and suitable for your case. In this case, you need to upload a custom prompt to the system or record a new prompt and apply it to the place you want to change.
Go to **Settings > PBX > Voice Prompts > Custom Prompts** to record and upload custom prompts.

1. Upload Custom Prompt

   Click Upload, the following dialog window appears. Click Browse... to choose a music file from your computer. Click Upload to start uploading.

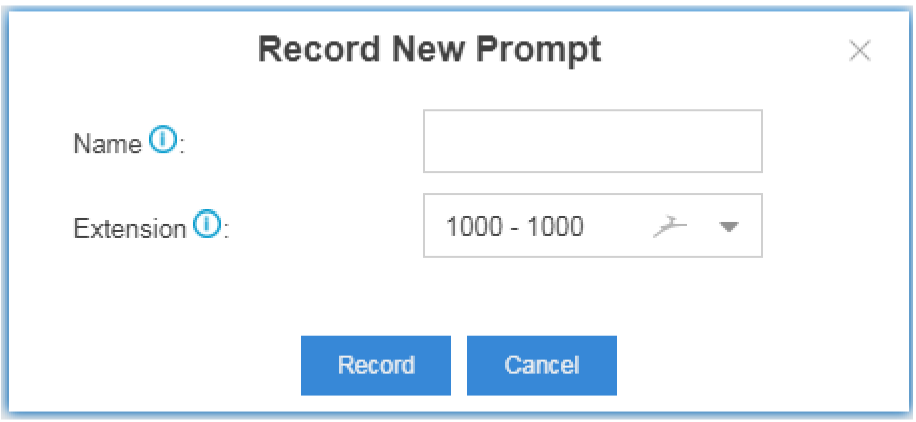


Note: Please select a ".wav", ".WAV", or ".gsm" file. The file size must not be larger than 8MB.
Supported Formats include:

- PCM, 8k, 16bit, 128kbps
- A-law, (g.711), 8k, 8bit, 64kbps
- u-law, (g.711), 8k, 8bit, 64kbps
- gsm, 6.10, 8k, 13kbps

2. Record Custom Prompt

   Click Record New, the following dialog window shows. Specify the name and choose an extension to make the record.



Click Record, the selected extension will ring, pick up the call to start recording.

General

This chapter explains general settings in the system, which can be applied globally to the eSIP Evolution Series.

- Preference
- Feature Code
- Voicemail
- SIP
- IAX

Preference

Option	Description
Max Call Duration	Select the absolute maximum number of seconds permitted for a call. If you wish to customize, enter the value in the text box directly. Input "0" disables the timeout.
Attended Transfer Caller ID	The Caller ID that will be displayed on the recipient's phone. For example: 500 calls 501 and then 501 transfers the call to 502. If set to "Transferor", the Caller ID displayed on 502's phone will be 501. If set to "Transferee", 500 will be displayed on 502's phone. If set to "Auto", when 501 is connected with 502, the Caller ID will be 501. When 501 hangs up after the transfer is done, and 500 and 502 are connected, it will display 500. When set to "Auto", you also need to check "Send Remote Party ID" under "Advanced" tab of "SIP" settings.
Flash Event	Specify which event will be triggered by pressing the hook flash: • 3 - Way Calling • Transfer
Virtual Ring Back Tone	Once enabled, when the caller calls out with cellular trunks, the caller will hear the virtual ring back tone generated by the system before the callee answers the call.
Distinctive Caller ID	When the incoming call is routed from Ring Group, Queue or IVR, the Caller ID would display where it comes from.
Match Route Permission When Seizing A Line	If checked, when users seize a line to place an outbound call, the dial will succeed only when the route permission is matched.
FXO Mode	Select a mode to set the On Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage, Adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is "FCC" for USA.
Tone Region	Select your country or nearest neighboring country to enable the default dial tone, busy tone, and ring tone for your region.
DTMF Duration (ms)	This set the duration (in milliseconds) of a DTMF tone on the FXO Trunk. The default is 120 ms.
DTMF Gap (ms)	This set the interval (in milliseconds) between each DTMF tone on the FXO Trunk. The default is 120 ms.
Extension Preferences	
User Extensions	Specify the user extension range. The default range is 1000-5999.
Ring Group Extensions	Specify the Ring Group extension range. The default range is 6200-6299.
Paging Group Extensions	Specify the Paging Group extension range. The default range is 6300-6399.
Conference Extensions	Specify the Conference extension range. The default range is 6400-6499.
IVR Extensions	Specify the IVR extension range. The default range is 6500-6599.
Queue Extensions	Specify the Queue extension range. The default range is 6600-6699.

Feature Code

Feature Codes are used to enable and disable certain features available in the system. The eSIP Evolution Series users can dial feature codes on their phones to use a particular feature. The default feature codes can be checked and changed via **Settings > PBX > General > Feature Code**.

Feature Code	
Feature Code Digits Timeout	The timeout to input next digit (in milliseconds). The default is 4000.
Recording	
One Touch Record	The feature code that is used to start or stop call recording. The default feature code is *1.
Voicemail	
Check Voicemail	The feature code that is used to check voicemail. The system will prompt you for password. The default feature code is *2.
Voicemail for Extension	You can leave a voicemail to other extensions by dialing feature code on their phone or forward an incoming call to an extension’s voicemail directly. The default feature code is **. For example, dial “**501” to leave a message for Ext. 501.
Voicemail Main Menu	The feature code that is used to access voicemail main menu. The default feature code is *02.
Transfer	
Blind Transfer	Dial this feature code and an extension number to blind transfer the call. The default feature code is *03.
Attended Transfer	Dial this feature code and an extension number to transfer the call. Hang up after contacting the destination. The default feature code is *3.
Attended Transfer Timeout	The timeout to transfer a call, in seconds. The default is 15 seconds.
Call Pickup	
Call Pickup	This feature code allows you to answer another ringing phone that is in the same pickup group. The default feature code is *4.
Enable Forward When Busy	Dial this feature code to forward calls to voicemail or a designated number when busy. For example: dial *72 to forward calls to voicemail when busy, and dial *72500 to forward all calls to number 500 when busy (this number does not include prefix, if you are required to dial with prefix, you need to configure it in Call Forwarding in Edit Extension window). The default feature code is *72.
Disable Forward When Busy	Dial this feature code to disable when busy call forwarding. The default feature code is *072.
Enable Forward No Answer	Dial this feature code to forward calls to voicemail or a designated number when no answer. For example: dial *73 to forward calls to voicemail when no answer, and dial *73500 to forward all calls to number 500 when no answer (this number does not include prefix, if you are required to dial with prefix, you need to configure it in Call Forwarding in Edit Extension window). The default feature code is *73.
Disable Forward No Answer	Dial this feature code to disable no answer call forwarding. The default feature code is *073.
Call Monitor	
Listen	Dial this feature code and the monitored extension number to initiate Listen monitoring. In this mode, the monitor can only listen to the call but can't talk. The default feature code is *90. Note: to monitor an extension, you need to configure the Monitor Settings for this extension first.
Whisper	Dial this feature code and the monitored extension number to initiate

	Whisper monitoring. In this mode, the monitor can listen to and talk with the monitored extension without being heard by the other party. The default feature code is *91. Note: to monitor an extension, you need to configure the Monitor Settings for this extension first.
Barge-in	Dial this feature code and the monitored extension number to initiate Barge-in monitoring. In this mode, the monitor can listen to and talk with both parties. The default feature code is *92. Note: to monitor an extension, you need to configure the Monitor Settings for this extension first.
DND	
Enable Do Not Disturb	Dial this feature code to put the extension in Do Not Disturb state. The default feature code is *74.
Disable Do Not Disturb	Dial this feature code to take the extension out of Do Not Disturb state. The default feature doe is *074.

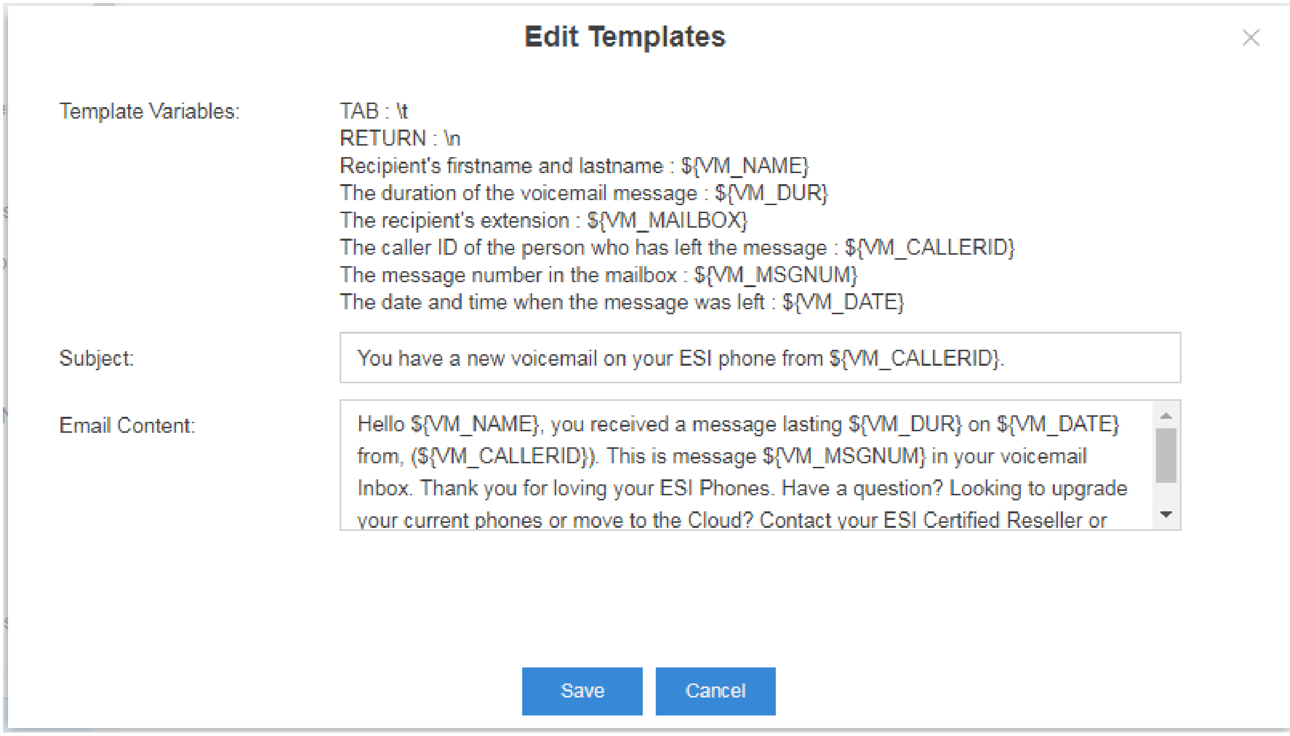
Voicemail

The configurations of voicemail can be globally set up and managed on the Voicemail page. Go to **Settings > PBX > General > Voicemail**, you can configure the Message Options, Greeting Options and Playback Options.

Message Options	
Max Messages per Folder	This sets the maximum number of messages that can be stored in a single folder of voicemail.
Max Message Time	This sets the maximum length of a single voicemail message (in seconds).
Min Message Time	This sets the minimum length of a single voicemail message (in seconds). Messages below this threshold will be automatically deleted.
Ask Caller to Dial 5	If this option is enabled, the caller will be prompted to press 5 before leaving a message.
Operator Breakout from Voicemail	If this option is set, the caller can jump out of the voicemail and go to the pre-configured destination by dialing 0.
Destination	This sets the breakout destination.
Greeting Options	
Busy Prompt	Greeting played when the extension is busy.
Unavailable Prompt	Greeting played when the extension is unavailable.
Voicemail Prompt	Select the greeting that will be played before the caller leave a message. If [Default] is selected, the default voicemail prompt will be played. If [None] is selected, no voice prompt will be played.
Playback Options	
Announce Message Caller ID	If this option is enabled, the caller ID of the party that left the message will be announced before the voicemail message begins playing.
Announce Message Duration	If this option is enabled, the duration of the message in minutes will be announced before the voicemail message begins playing.
Announce Message Arrival Time	If this option is enabled, the arrival time of the message will be played back before the voicemail message begins playing.
Allow Users to Review Messages	Allow the callers to review their recorded messages before sending them to the voicemail box.

Voicemail to Email Template

You can customize the Voicemail Email contents by clicking Voicemail To Email Template Settings.



SIP

Go to **Settings > PBX > General > SIP** to configure SIP settings. It is wise to leave the default setting as provided on this page. However, for a few fields, you need to change them to suit your situation.

General

Port Type and Description	
UDP Port	UDP Port used for SIP registrations. The default is 5060.
TCP Port	TCP Port used for SIP registrations. The default is 5060.
RTP Port	RTP Port for transmitting data. The From-port should start from 10000. From-port and To-port should have a difference value between 100 and 10000. The default is 10000-12000.
Local SIP Port	A random port in the port range will be used when sending packets to SIP server. The default range is 5062-5082.
Registration And Subscription Timers	
Max Registration and Subscription Timers	Maximum duration (in seconds) of incoming registrations and subscriptions. The default is 3600 seconds.
Min Registration and Subscription Timers	Minimum duration (in seconds) of incoming registration and subscriptions. The default is 60 seconds for Registration and 90 seconds for Subscription.
Qualify Frequency	How often to send SIP OPTIONS packet to SIP device to check if the device is up. The default is 30 per second.
Outbound SIP Registrations	
Registration Attempts	The number of registration attempts before giving up (0 for no limit).
Default Incoming/Outgoing Registration Time	Default duration (in seconds) of incoming/outgoing registration. The default is 1800 seconds. **Note:** the actual duration needs to be 10 seconds less than the value you filled in.

NAT

If your server is operating in a network connected to the internet through a single router, your server is likely using NAT. The NAT device has to be instructed to forward the right inbound packets (from internet) to the server. Usually you must configure NAT settings when you want to register a remote extension to the server or when you need to connect to the server via SIP trunk.

ESI's eSIP Evolution Series supports 3 methods to configure NAT: STUN, External IP Address and External Host. You can select one method to configure NAT or disable NAT.

1. STUN

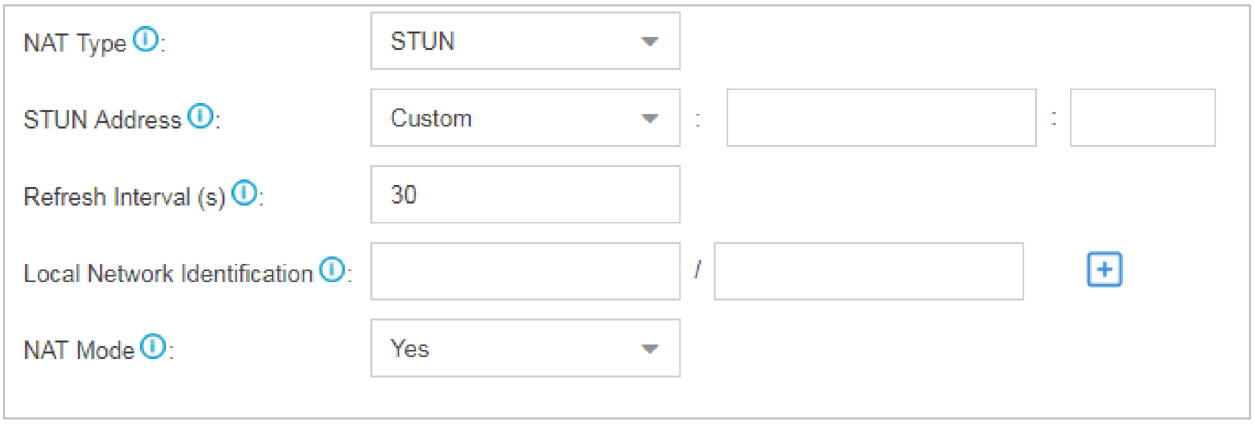
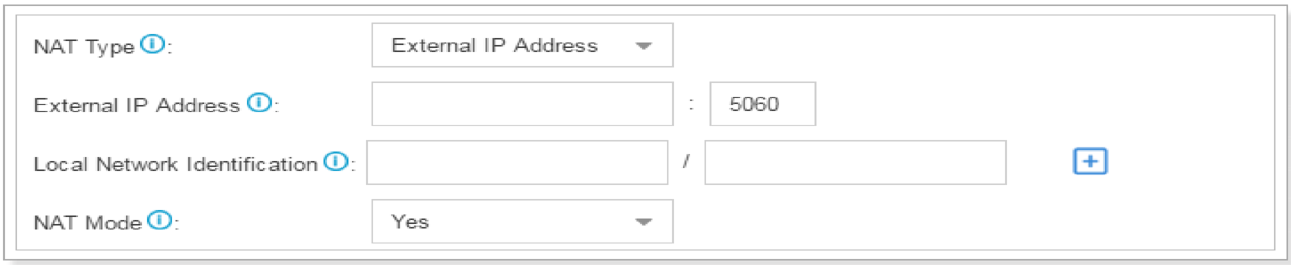


Figure 9-2 STUN

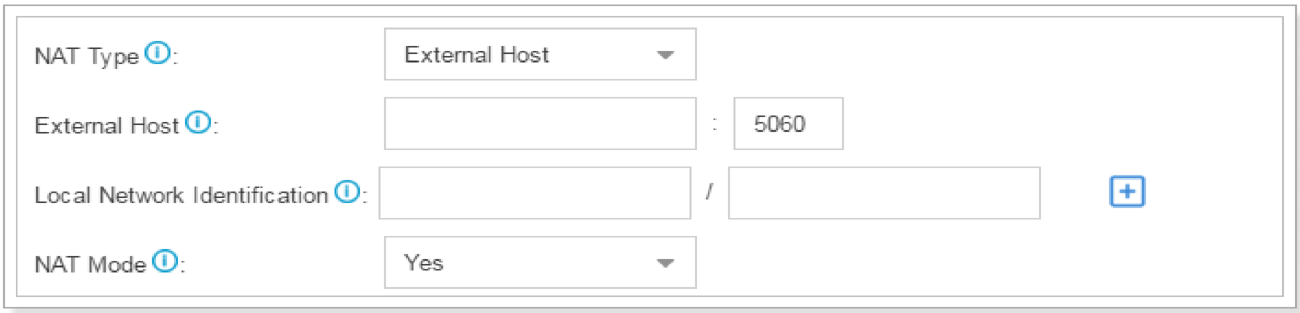
Option	Description
STUN Address	Choose a STUN address in the drop-down list or customize with a STUN address and STUN port.
Refresh Interval	If an external host has been supplied, you may specify how often the system will perform a DNS query on this host. This value is specified in seconds.
Local Network Identification	Used to identify the local network using a network number/subnet mask pair when the system is behind a NAT or firewall. Some examples are as follows: “192.168.0.0/255.255.0.0”, “10.0.0.0/255.0.0.0”, and “172.16.0.0/12”.
NAT Mode	Global NAT configuration for the system. The options are as follows: • Yes: use NAT and ignore the address information in the SIP/SDP headers and reply to the sender's IP address/port. • No: use NAT mode only according to RFC3581. • Never: never attempt NAT mode or RFC3581 support. • Route: use NAT but do not include remote port in headers.

2. External IP Address



Option	Description
External IP Address	The IP address that will be associated with outbound SIP messages if the system is in a NAT environment.
Local Network Identification	Used to identify the local network using a network number/subnet mask pair when the system is behind a NAT or firewall. Some examples are as follows: “192.168.0.0/255.255.0.0”, “10.0.0.0/255.0.0.0”, and “172.16.0.0/12”.
NAT Mode	Global NAT configuration for the system. The options are as follows: • Yes: use NAT and ignore the address information in the SIP/SDP headers and reply to the sender's IP address/port. • No: use NAT mode only according to RFC3581. • Never: never attempt NAT mode or RFC3581 support. • Route: use NAT but do not include rport in headers.

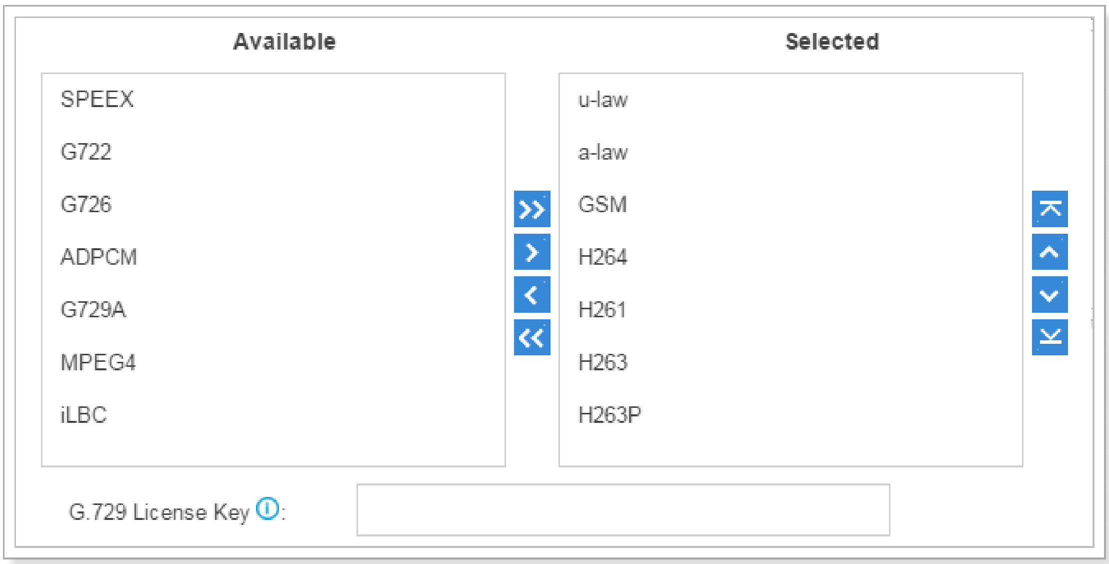
3. External Host



Option	Description
External Host	Alternatively you can specify an external host, and the system will perform DNS queries periodically. This setting is only required when your external IP address is not static. It is recommended that a static public IP address be used with this system. Please contact your ISP for more information.
External Refresh Interval	If an external host has been supplied, you may specify how often the system will perform a DNS query on this host. This value is specified in seconds.
Local Network Identification	Used to identify the local network using a network number/subnet mask pair when the system is behind a NAT or firewall. Some examples are as follows: “192.168.0.0/255.255.0.0”, “10.0.0.0/255.0.0.0”, and “172.16.0.0/12”.
NAT Mode	Global NAT configuration for the system. The options are as follows: • Yes: use NAT and ignore the address information in the SIP/SDP headers and reply to the sender's IP address/port. • No: use NAT mode only according to RFC3581. • Never: never attempt NAT mode or RFC3581 support. • Route: use NAT but do not include rport in headers.

Codec

A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet. The eSIP Evolution Series supports G711 a-law, u-law, GSM, H261, H263, H263P, H264, SPEEX, G722, G726, G729, ADPCM, MPEG4, H263P and iLBC.



TLS

ESI's eSIP Evolution Series supports TLS protocol, to use TLS, you need enable TLS via **Settings > PBX > General > SIP > TLS**. Check the TLS configuration parameters below.

Option	Description
Enable TLS	Check the check box to enable TLS.
TLS Port	TLS Port used for SIP registrations. The default is 5061.
Certificate	Choose the TLS certificates.
TLS Verify Server	If checkbox is unchecked, don't verify the server's certificate when acting as a client. If you don't have the server's CA certificate you can set this, and it will connect without requiring TLS CA file. The default is unchecked.
TLS Verify Client	If checkbox is unchecked, don't verify the server's certificate when acting as a client. If you don't have the server's CA certificate you can set this, and it will connect without requiring TLS CA file. The default is unchecked.
TLS Ignore Common Name	If checkbox is checked, verify certificate when acting as server. The default is unchecked.
TLS Client Method	Specify protocol for outbound client connections. The default is tslv2.

Session Timer

A periodic refreshing of a SIP session that allows both the user agent and proxy to determine if the SIP session is still active.

Option	Description
Session-timers	Choose the session timers mode on the system: • No: do not include "timer" value in any field • Supported: include "timer" value in Supported header • Require: include "timer" value in Require header • Forced: include "timer" value in both "Supported" and "Required" header. The default is Supported.
Session Expires	The max refresh interval in seconds.
Min-SE (s)	The min refresh interval in seconds, it must not be less than 90.

QOS

QoS (Quality of Service) is a major concern in VoIP implementations. The concern is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due to interference from other, lower priority traffic. When the network capacity is insufficient, QoS could provide priority to users by setting the value.

T.38

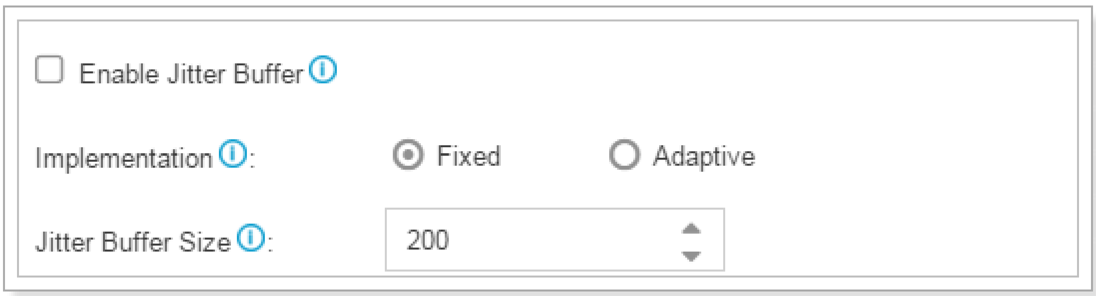
- **No T.38 Attributes In Re-invite SDP**
  If checked, SDP in re-invite packet will not add T.38 attributes.
- **Error Correction**
  This sets the Error Correction Mode (ECM) for the Fax.
- **T.38 Max BitRate**
  Set theT38 Max Bit Rate.

Advanced

Option	Description
Allow RTP Re-invite	By default, the system will route media steams from SIP endpoints through itself. Enabling this option causes the system to attempt to negotiate the endpoints to route packets to each other directly, bypassing the system. It is not always possible for the system to negotiate endpoint-to-endpoint media routing.
Get Caller ID From	Determines from which header field the system will pull the Caller ID header.
User Agent	This allows you to change the User-Agent field.
Get DID From	Determines from which header field system will pull DID. If Remote-Party-ID is selected but the line does not support this, DID will be pulled from Invite header.
Send Remote Party ID	Whether to send the Remote-Party-ID in SIP header or not. The Default is unchecked.
Send P Asserted Identify	Whether to send the P-Asserted-Identify in SIP header or not. The Default is unchecked.
100rel	Check the option to enable 100rel.
Send Diversion ID	Whether to send the Diversion ID in SIP header or not. The Default is unchecked.
Allow Guest	If enabled, it will allow the unauthorized INVITE coming into the server and the calls can be made. The default is unchecked.

Jitter Buffer

Jitter is the variation in the time between packets arriving on a VoIP system. These variations can be caused by network congestion, timing drift or route changes. Jitter buffers are used to counter delay or latency, dropped packets, and jitter. They temporarily store arriving packets to minimize jitter and discard packets that arrive too late.



Configuring the Jitter Buffer settings on the eSIP Evolution Series server can help to improve the call quality through VoIP. Jitter buffers must be correctly configured to be effective.

- **Enable Jitter Buffer**: Check to enable this feature.
- **Implementation:** Choose the implementation of jitter buffer.
  - **Fixed:** The length of jitter buffer will always be the size defined by “Jitter Buffer Size”. The default is 200 ms.
  - **Adaptive:** The length of jitter buffer will vary in size within the range of min size and max size based on current network condition. The default is from 100 ms to 200 ms.
- **Jitter Buffer Size:** Set a fixed jitter buffer size.
- **Adaptive Adjustment Size:** The size of each adaptive adjustment of jitter buffer.
- **Max Jitter Buffer Size:** The maximum jitter buffer size.

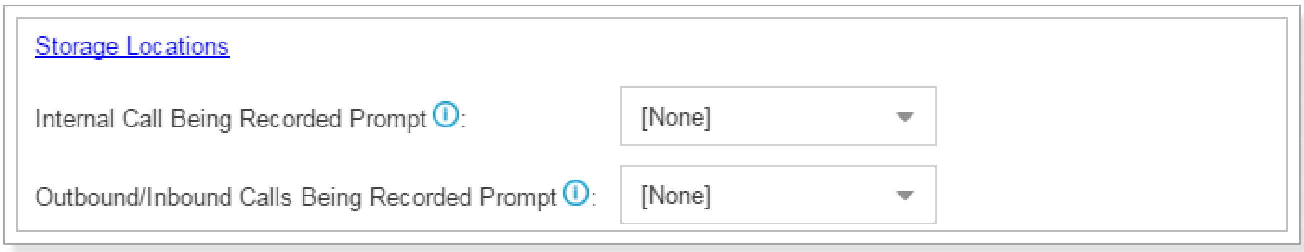
IAX

Option	Description
UDP Port	UDP port used for IAX2 registrations. The default is 4569.
Bandwidth	Control which codecs to be used based on bandwidth consumption.
Maximum Registration/ Subscription Time	Maximum duration (in seconds) of an IAX registration. The default is 1200 seconds.
Minimum Registration/ Subscription Time	Minimum duration (in seconds) of an IAX registration. The default is 60 seconds.
Codec	Choose the codec.

Recording

This chapter explains how to configure auto recording on ESI's eSIP Evolution Series.

The eSIP Evolution Series supports auto recording for an established call. Go to **Settings > PBX > Recording** to configure auto recording settings.



General Preferences	
Storage Location	Click the option to link the Storage settings. In the storage settings, you can configure where to store recording files.
Internal Call Being Recorded Prompt	If the internal call has enabled call recording, this prompt will notify the called party that the call will be recorded.
Outbound/Inbound Call Being Recorded Prompt	If the external call (outbound/inbound/callback) has enabled call recording, this prompt will notify the called party that the call will be recorded.
Record Trunks	When a call reaches the selected trunk, it will be recorded.
Record Extensions	The selected extensions will be recorded.
Record Conferences	The selected conferences will be recorded.

Event Center

The eSIP Evolution Series can monitor system events and logs and send notifications to the specified contacts.

Event Settings

The system events are divided into three categories:

Operation

- Modify Administrator Password
- User Login Success
- User Login Failed
- User Lockout
- API Authentication Lockout
- Extension User Password Changed
- eMobile Login Failed
- eMobile Client has been Locked

Telephony

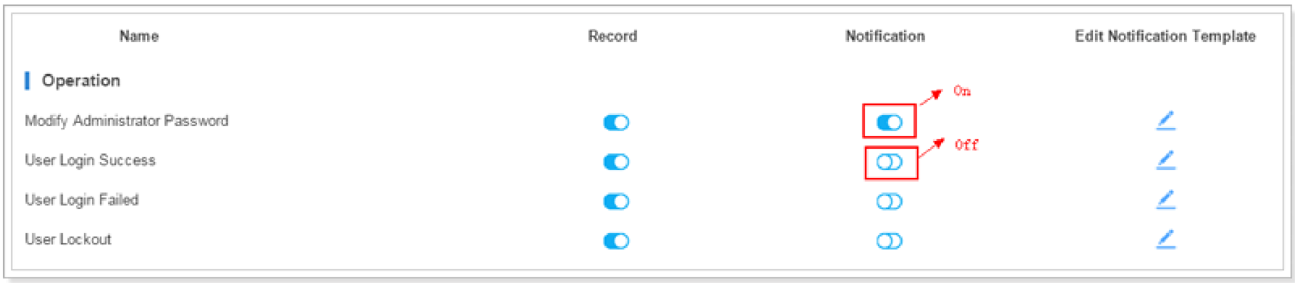
- VoIP Peer Trunk Registration Failed
- VoIP Register Trunk Registration Failed
- Outgoing Call Failed
- Concurrent Calls Overload
- GSM Registration Failure
- Emergency Call
- Extension Outbound Calls Prohibited
- VoIP Peer Trunk Re-Registered
- VoIP Register Trunk Re-Registered

System

- CPU Overload
- Memory Overload
- Disk Failure
- Storage Full
- Network Failure
- Network Attacked
- System Reboot
- System Upgrade
- System Restore
- SMS To Email Failed
- Hot Standby Failover Action
- Abnormal EXP100 Module
- Network Drive Lost Connection
- Auto Cleanup Reminder
- Cellular Network Connected
- About To Reach Data Allowance

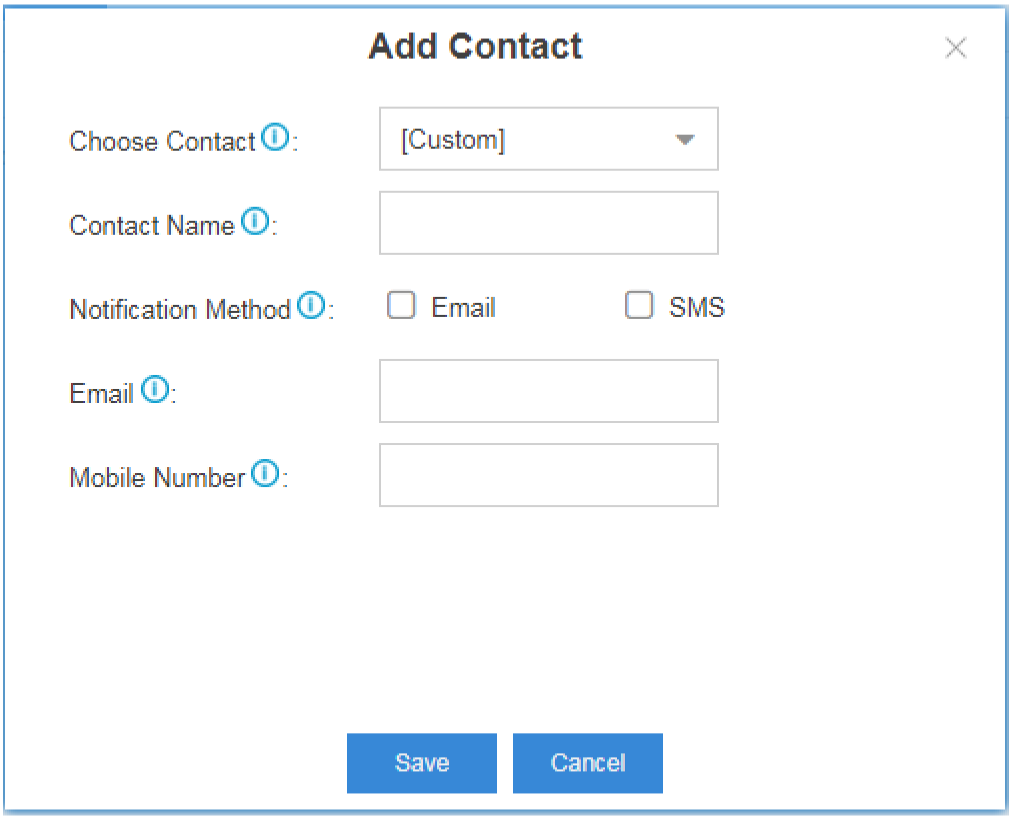
Turn on **Record** to decide whether to record the event.
Turn on **Notification** to decide whether to send notification.

- Click to edit the notification template.



Notification Contacts

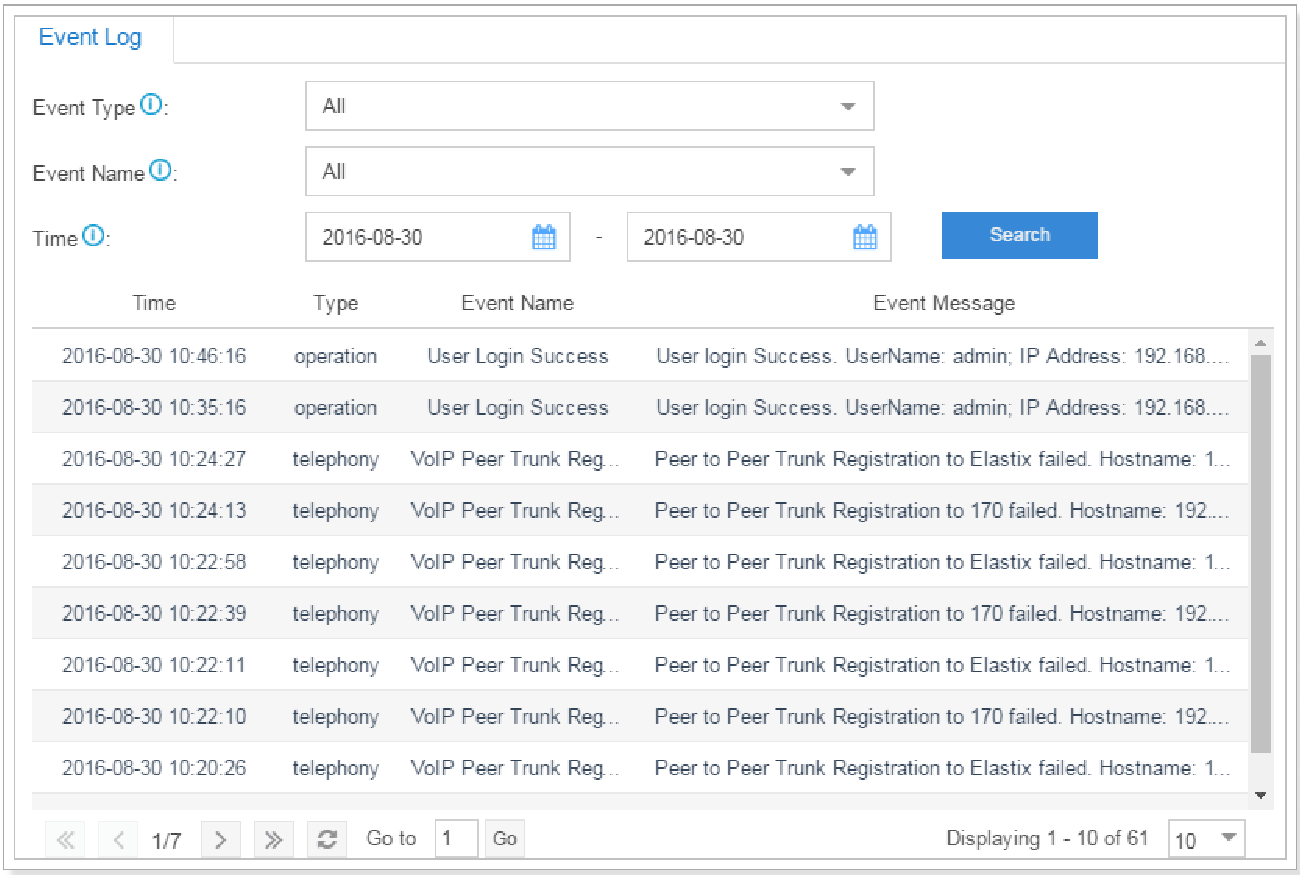
The administrator can add contacts here to define where to send the notifications. The system supports the ability to send Email notification, Call notification, and SMS notification. Click Add to add a contact.



Option	Description
Choose Contact	Choose a contact from the drop-down menu. The selected contact will receive alert emails, SMS messages or calls.
Notification Method	Select how to notify the contact when the event occurs. • Email • SMS
Email	When events occur, send notification emails to this address. If the Notification Method is Email, this field must be entered.
Mobile Number	When events occur, call or send SMS to this mobile number. If the Notification Method is Phone Call or SMS, this field must be entered.

Event Log

Go to **Settings > Event Center > Event Log** to check the event log.
You can filter the event logs by selecting an event type, event name, and specifying a certain time period.
Click Search, the matching results will be displayed.



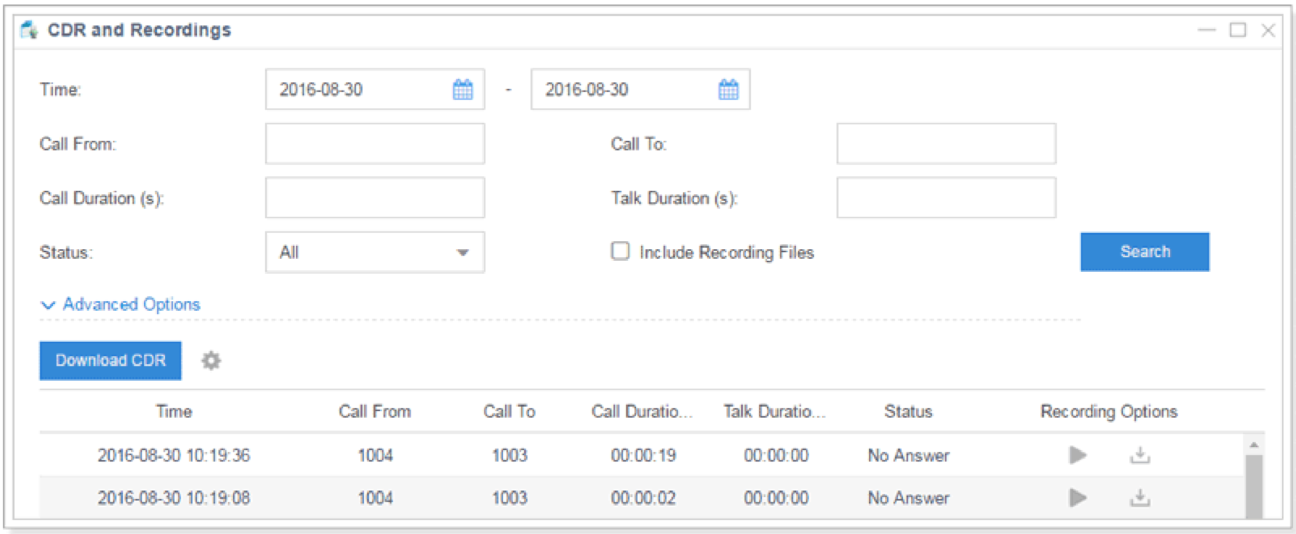
CDR and Recording

In the CDR and Recording center, you can check all the call logs and recordings on the system. You can run reports against the logs and filter based on the following:

- Time
- Call From
- Call To
- Call Duration
- Talk Duration
- Status
- Trunk
- Communication Type
- PIN Code

You can perform the following operations on the filtered call report:

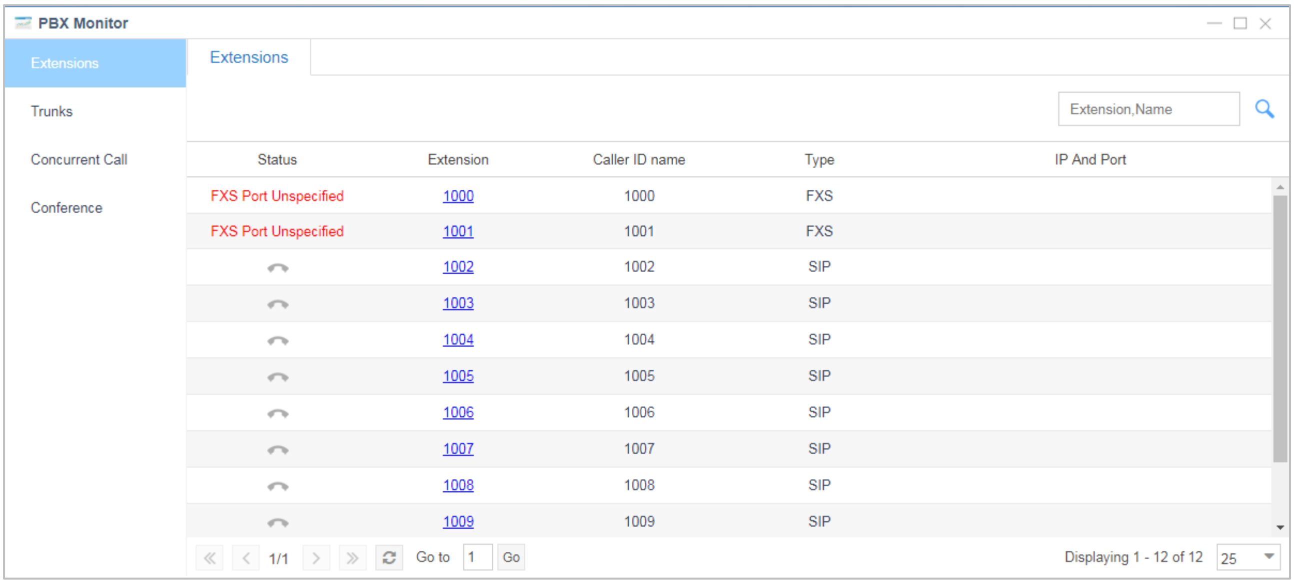
- **Download Searched Result**
  Click Download CDR to download the searched records.
- **Edit List Options**
  Click to choose which options will be displayed on the logs page.
- **Play Recording File**
  Click to play the recording file.
- **Download Recording File**
  Click to Download the recording file.



PBX Monitor

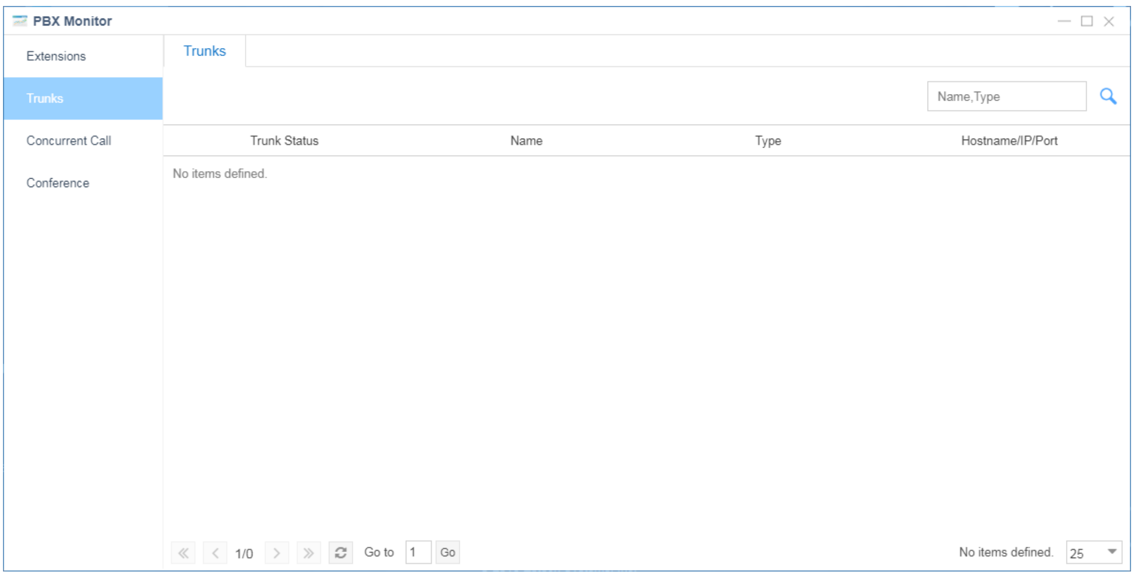
The server monitors the status of Extensions, Trunks and Concurrent Calls. Go to **PBX Monitor** to check the real time status.

Extension Status



Status	Description
	The extension is idle.
	The extension is ringing.
	The extension is unavailable.
	The extension is busy.
	The extension is on hold.
	Malfunction in FXS interface; please examine the relevant interface and module.

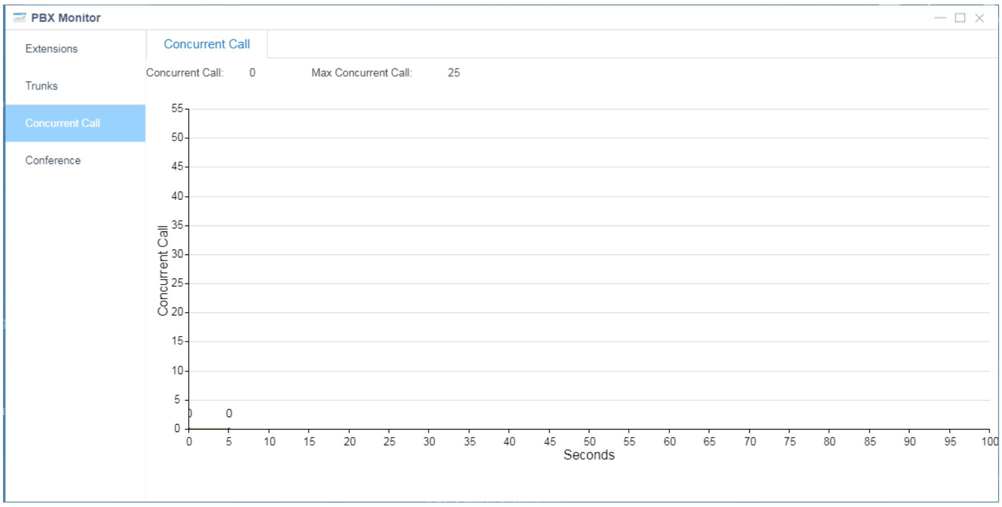
Trunk Status



FXO Trunk Status	
	The trunk is idle.
	The trunk is busy in use.
	No PSTN line plugged in FXO interface.
	Malfunction in FXO interface; please examine the relevant interface and module.
Cellular Trunk Status	
	The trunk is idle, the icon shows the signal strength.
	The trunk is busy.
	The module is powered off.
	No SIM card inserted.
	No signal.
	PIN/PUK Error.
	GSM network registration failed.
	Malfunction in module; please examine the relevant module.
T1/PRI Trunk Status	
	The trunk is idle.
	1. Broken module/interface. 2. Incorrect physical layer configuration. 3. Service provider did not activate the trunk.
	1. Incorrect protocol layer configuration. 2. Service provider did not activate the trunk.
	1. Malfunction in interface/module; please examine the relevant interface/module. 2. No trunk plugged in. 3. Service provider did not activate the trunk.
VoIP Trunk Status	
	1. Registered 2. Unmonitored
	Registering.
	1. Unreachable 2. Registration failed, caused by: ● wrong password ● wrong authentication name ● wrong username ● transport type inconsistent

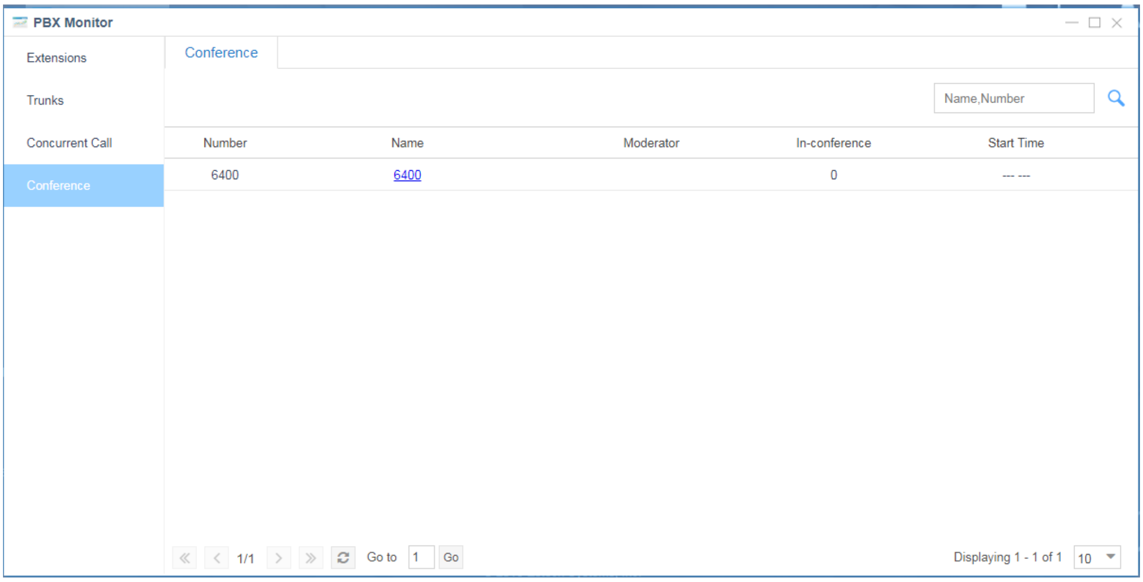
Concurrent Call

Monitor the concurrent calls on the system.



Conference

You can check the conference moderator, how many members in the conference, when the conference starts.



Resource Monitor

Resource Monitor allows you to monitor the CPU usage, memory usage, disk utilization and network flow.

Information

On this page, you can check the system information, including Product, SN, Hardware version, Software version etc.

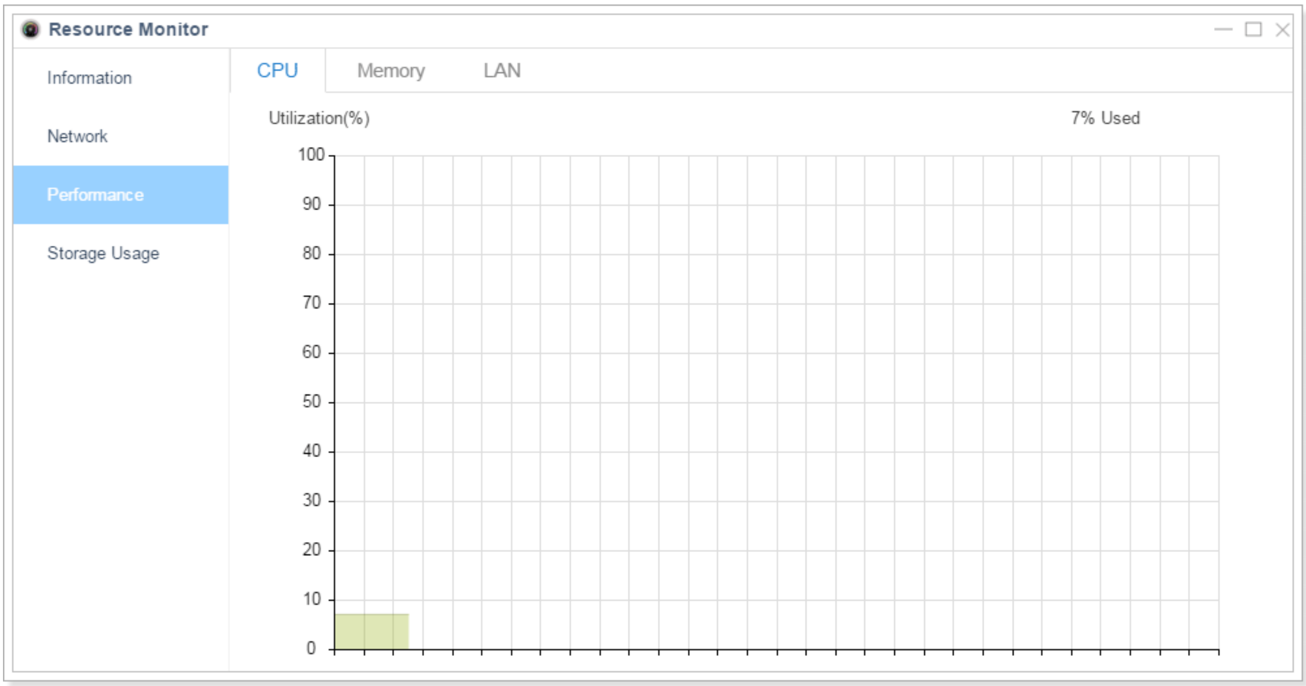
Network

Click on **Network** tab to view the system's network status.

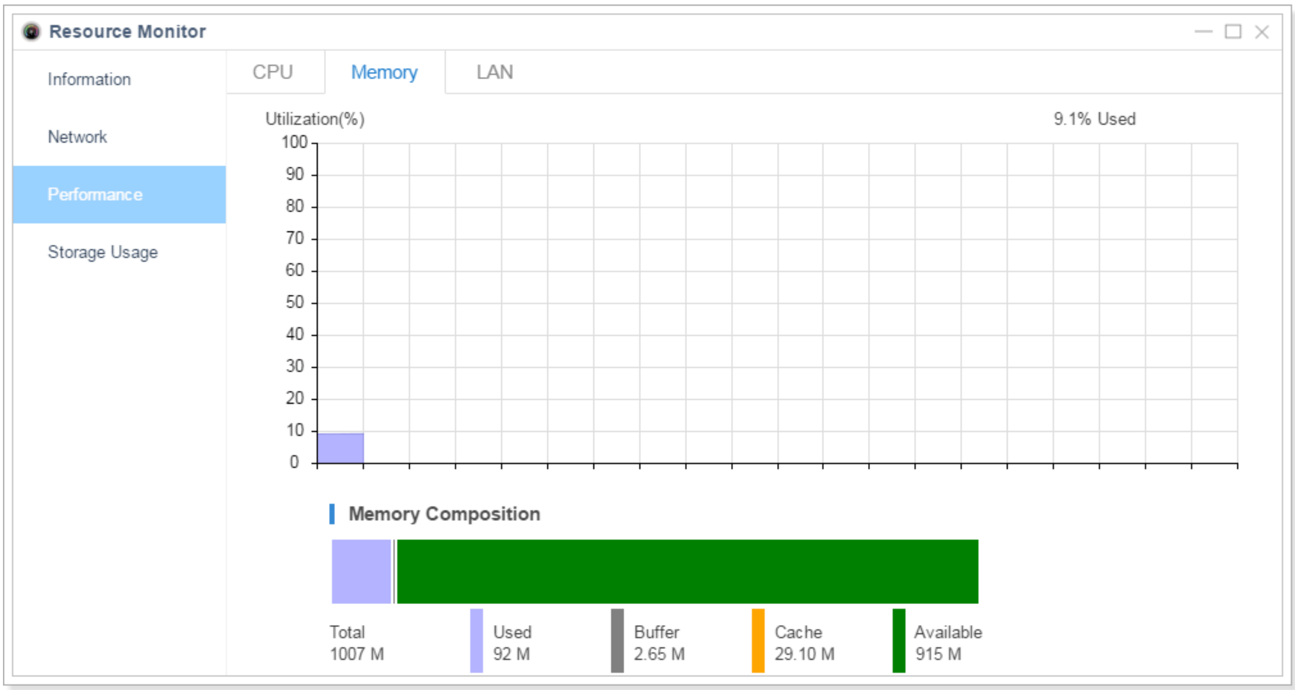
Performance

Click on **Performance** tab to view the resource utilization data. The information of the chart will be shown upon mouse-over.

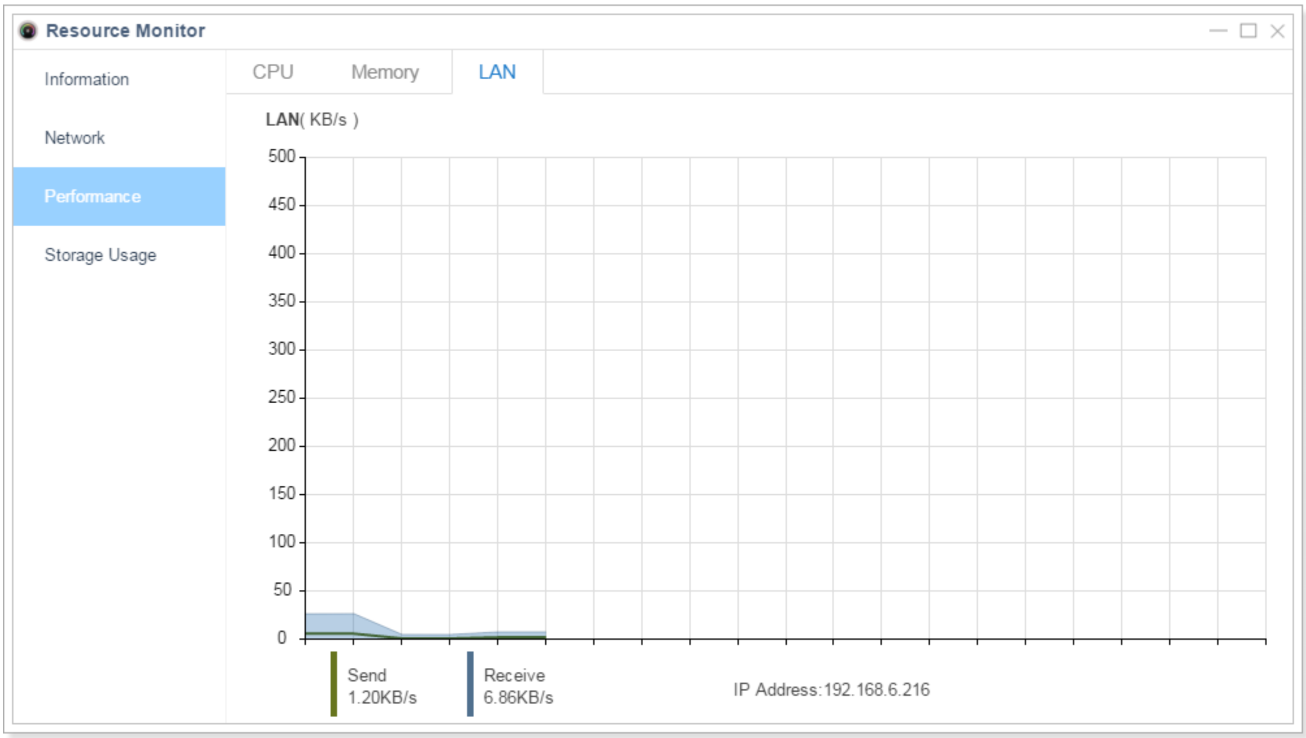
CPU



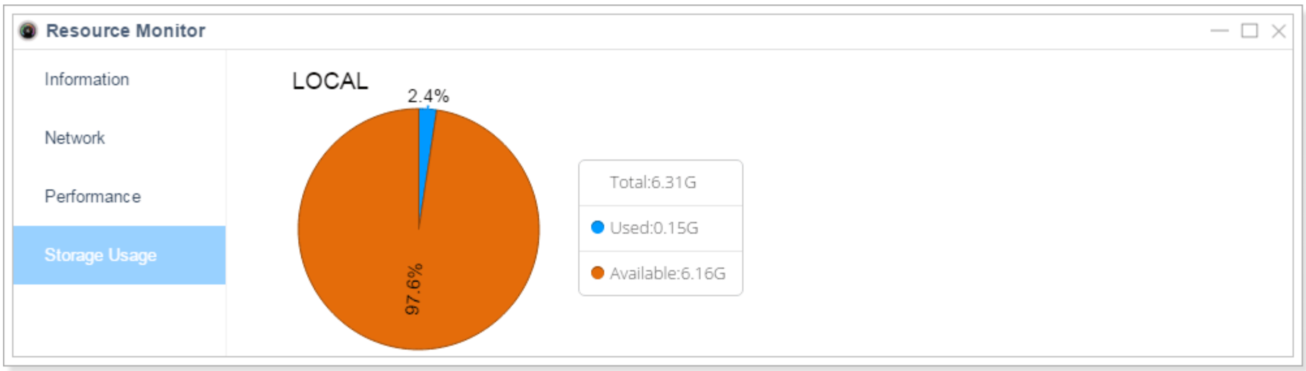
Memory



LAN



Storage Usage



Maintenance

This chapter describes system maintenance settings including the followings:

- Upgrade
- Backup and Restore
- Reboot
- Reset
- System Log
- Operation Log
- Trouble Shooting

Upgrade

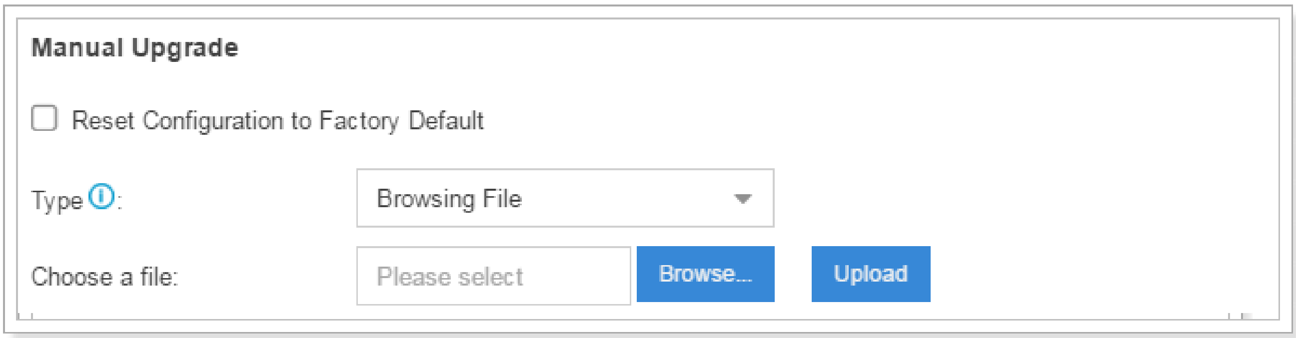
ESI's eSIP Evolution Series requires current software assurance to obtain the latest system firmware version. That firmware can then be manually upgraded. The system supports browsing to the firmware file from the local PC as well as pulling from HTTP or TFTP servers.
Go to **Maintenance > Upgrade** to do upgrade.

Notes:

- If “Reset configuration to Factory Defaults” is enabled, the system will reset to factory default settings.
- When updating the firmware, please don’t turn off the power or the system may become unresponsive.
- If you are trying to upgrade through HTTP, please make sure that the system is able to reach the Internet, or it will not be able to retrieve the firmware file, causing the upgrade to fail.

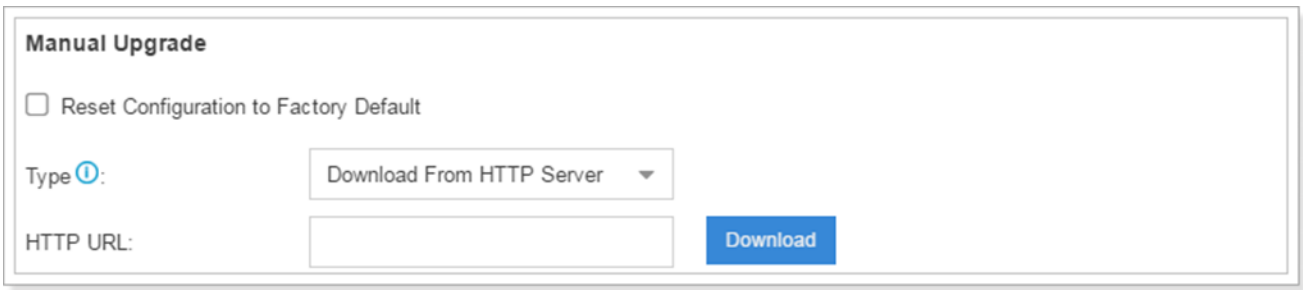
Browsing to the File from a Local PC to Upgrade

1. Choose **Type** "Browsing File".
2. Click Browse, select the firmware file from your local PC. Note that the file should be a BIN file.
3. Click Upload to start uploading.



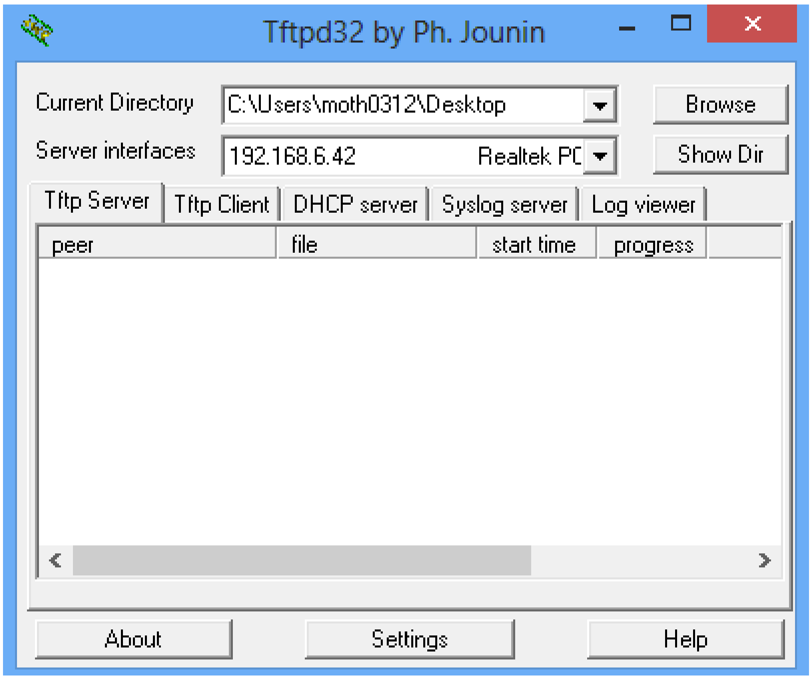
Upgrade through HTTP

1. On the Firmware Upgrade page, choose “Download From HTTP Server”.
2. Enter the HTTP URL.
   Note: the HTTP URL should be a BIN file download link.
3. Click Download to start downloading the file from ESI HTTP server.

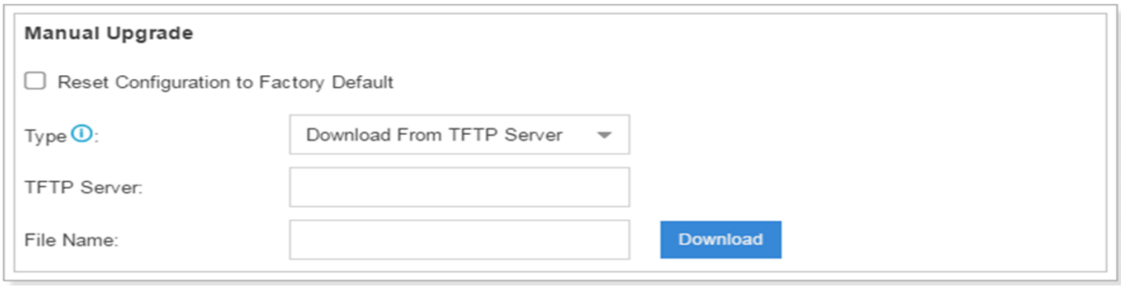


Upgrade through TFTP

1. Download firmware file from ESI to your local PC.
2. Create a tftp server, here take Tftpd32 for example.
3. Configure tftp server. Click Browse button to select the firmware file.



4. Go to **Maintenance > Upgrade** in the eSIP Web UI, select **Type** as "Download From TFTP Server".
5. Fill in the **TFTP Server IP**, the IP should be the local PC's IP address.
6. Fill in the name of firmware update. It should be a BIN file name.
7. Click **Download** to download the file and start to upgrade.



Backup and Restore

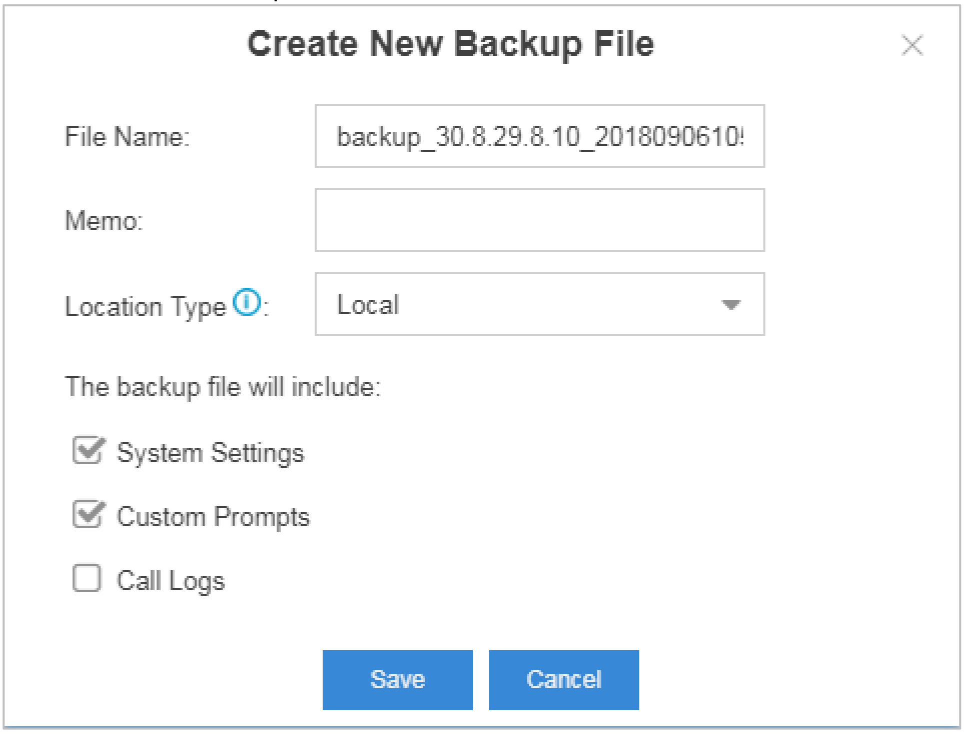
The eSIP Evolution Series provides a Backup and Restore feature, which allows you to create a complete backup of the system's configurations to a file.

Notes:

- When you have updated the firmware version, it's not recommended to restore an old backup package.
- Backup from an earlier version cannot be restored on the system of a later version.

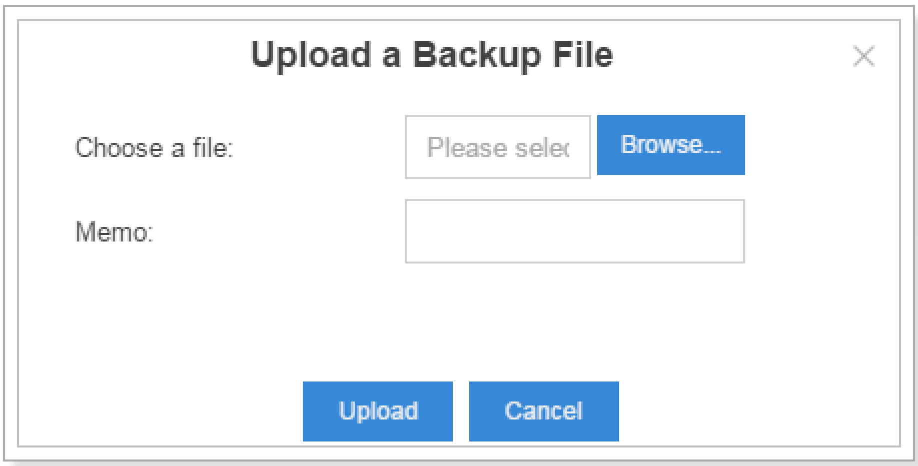
Create a New Backup

Click Backup to create a new backup.



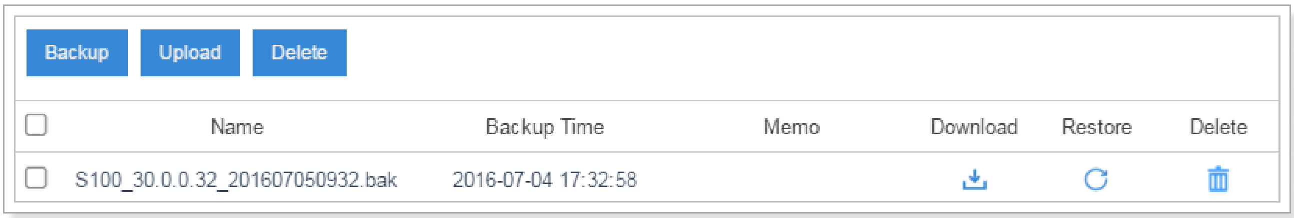
Upload a Backup

Click Upload to upload a backup.



Restore

To restore the configuration data, select a backup and click . Reboot the system to take effect.
Note: The current configurations will be OVERWRITTEN with the backup data.



Reset and Reboot

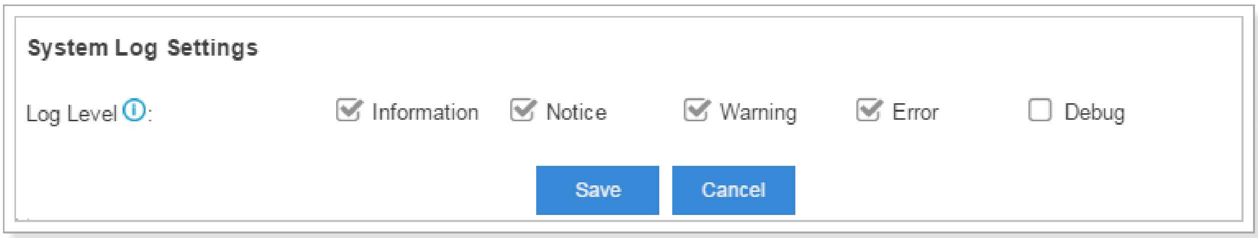
Admins can reset and reboot the system via **Maintenance > Reset and Reboot**.

- Click Reboot to reboot the system
- Click Reset to reset the system to factory configurations.

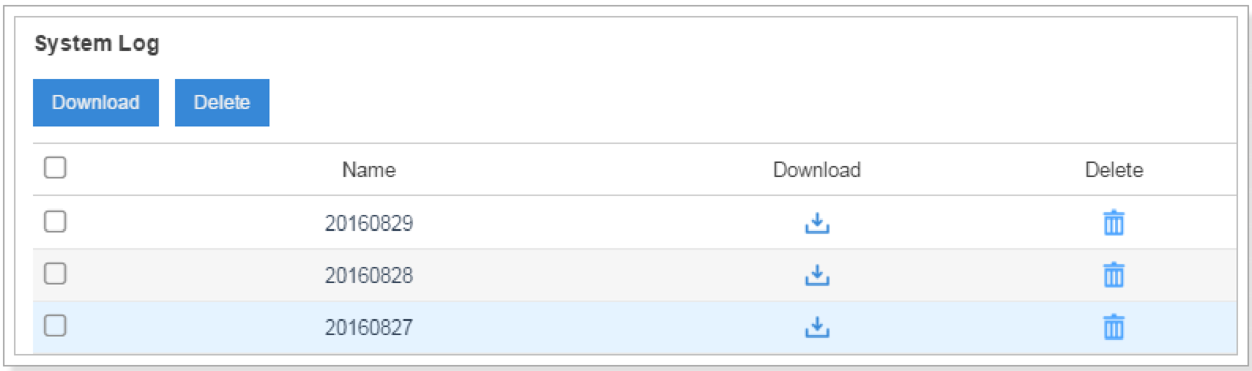
System Log

Admins can check system logs under **Maintenance > System Log**.
The system logs will be generated every day, automatically, and a log file will be listed in the System Log.

1. System Log Settings
   You can set the debug level by checking/unchecking the options "Info", "Notice", "Warning", "Error" and "Debug", click Save and Apply to save the changes.



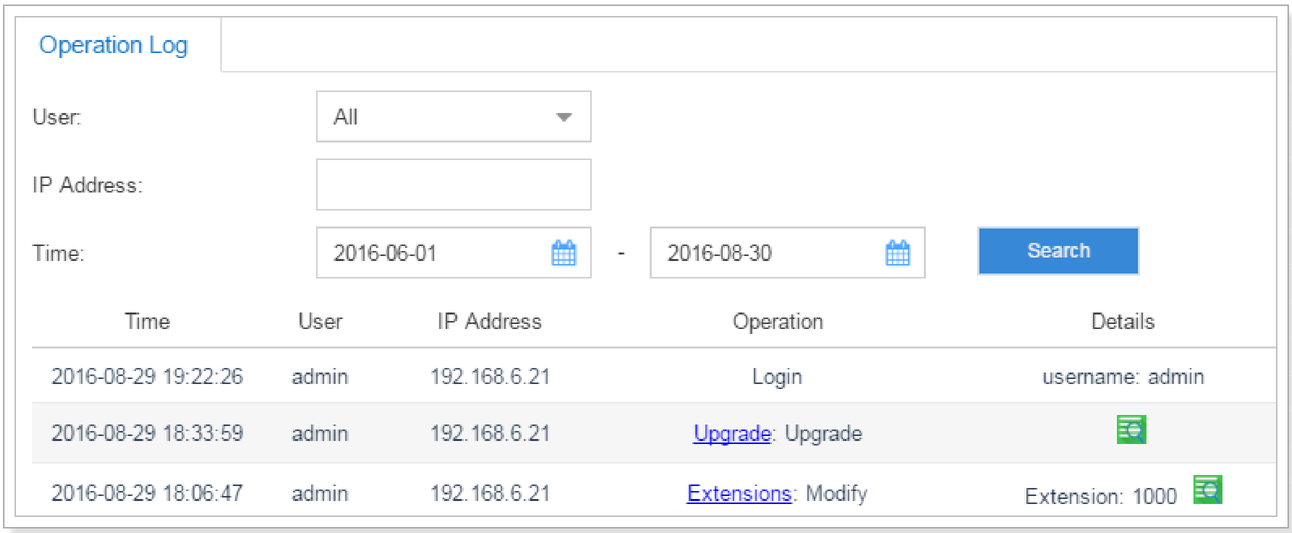
2. System Log
   - Click to download the file to your local PC.
   - Click to delete the log file.



Operation Log

Go to **Maintenance > Operation Log** to check the operation log.

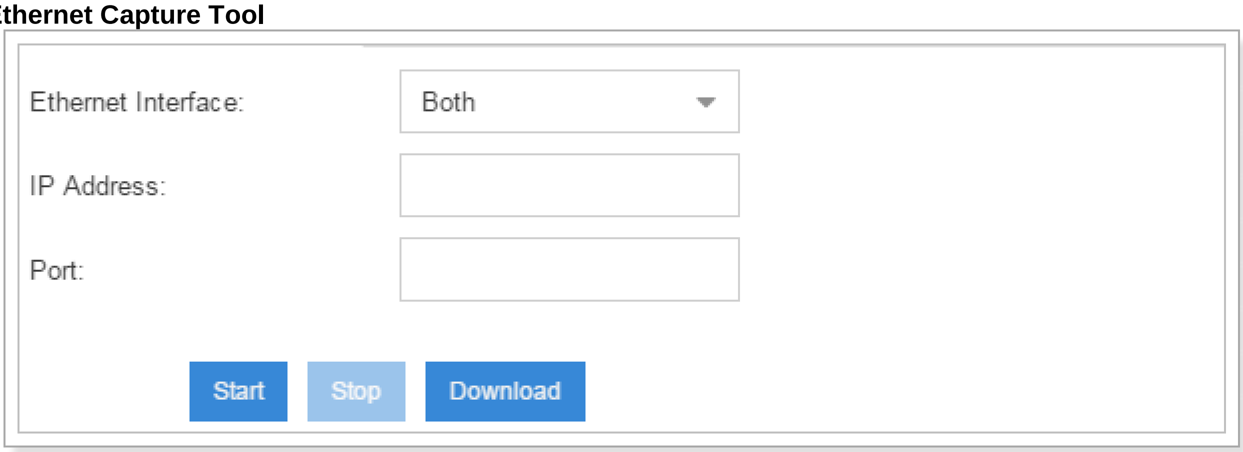
You can filter the logs by user, IP address, and specifying a certain time period. Click Search, the matching results will be displayed.



Troubleshooting

The eSIP Evolution Series provides multiple tools within the Web GUI for you to perform troubleshooting. Go to **Maintenance > Troubleshooting** to access the tools.

Ethernet Capture Tool



1. Fill in the target IP address and port.
2. Click **Start** to start capturing logs.
3. Click **Stop** to stop capturing.
4. Click **Download** to download the file to your local PC and analyze it.

The output result is in .tar format. Decompress/unzip the file and open the .pcap file using Wireshark software.

Port Monitor Tool

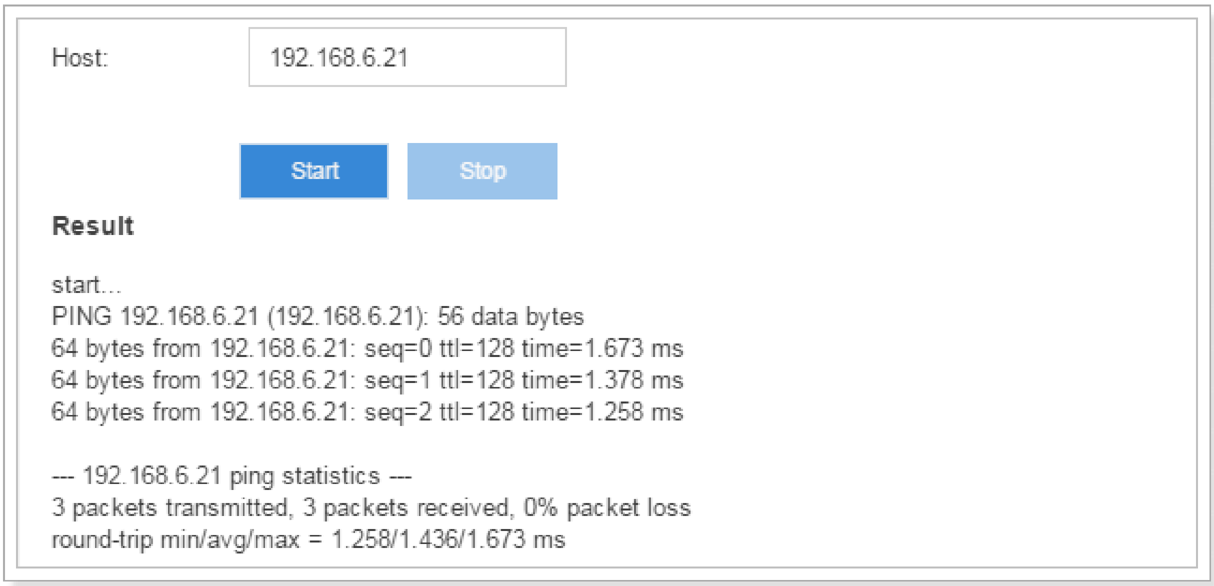
This feature is used to monitor PSTN trunks on the system.
Users could choose a PSTN trunk, then start to monitor the trunk.

1. Choose a trunk from the drop-down menu.
2. Click **Start** to start capturing logs.
3. Click **Stop** to stop capturing.
4. Click **Download** to download the file to your local PC and analysis it.
   The output result is in .tar format. Decompress the file and open the .raw files using Audition, or similar, software.

IP Ping

1. Enter the target IP address or hostname.
2. Click **Start** to start capturing logs.

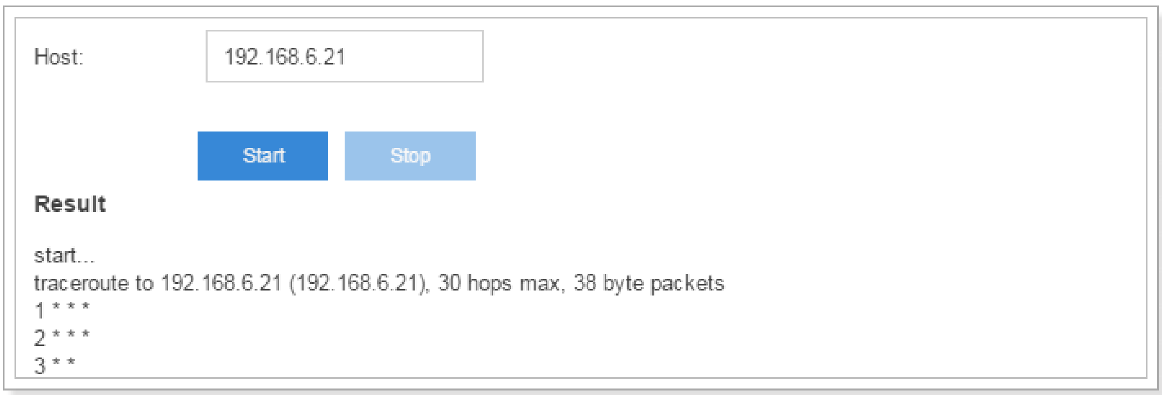
The output result will display in the window as below.



Traceroute

Enter the target IP address or hostname.

1. Click **Start** to start capturing logs.
   The output result will display in the window as below.



Applications

This chapter describes the additional applications pre-loaded on ESI's eSIP Evolution Series:

- LDAP Server
- Auto Provisioning
- Conference Panel
- VPN Server
- QueueMetrics Integration
- eMobile

LDAP Server

LDAP stands for *Lightweight Directory Access Protocol*, which is a client-server protocol for accessing a directory service. ESI's eSIP Evolution Series has an integrated LDAP server, to provide centralized phonebook management. With the LDAP phone book, you can quickly launch calls without wasting time finding a contact's number and subsequently entering it on your phone.

Note: Refer to the ***ESI eSIP Evolution Series LDAP Server Guide*** for detailed information.

Auto Provisioning

The Auto Provision Application makes quick work of provisioning phones that are on the local network by instructing the phones to retrieve its configuration file from the eSIP Evolution Series server.

Note: Refer to the ***ESI eSIP Evolution Series Auto Provisioning Guide*** for detailed information.

Conference Panel

Conference Panel is a visual control panel for your conference calls. You can batch invite people with the dial-out feature in the panel or use your telephone. You can also save all the attendees' contact information to the "Contact Group", so you can reuse it next time.

Note: Refer to the ***ESI eSIP Evolution Series Conference Panel Guide*** for detailed information.

VPN Server

A Virtual Private Network (VPN) allows you to traverse networks privately and securely as if you were on a private network. The VPN server application on ESI's eSIP Evolution Series will help you configure the server as a VPN server. You can setup multiple VPN clients to access the eSIP Evolution Series VPN server safely and securely.

Note: Refer to the ***ESI eSIP Evolution Series VPN Server Guide*** for detailed information.

QueueMetrics Integration

QueueMetrics Live Integration provides the interface to connect the eSIP Evolution Series and QueueMetrics Live. QueueMetrics Live is a cloud-based call center suite for Asterisk telephony systems. The integration helps you to easily generate the report of the queue daily, weekly and monthly. What's more, QueueMetrics Live is also a call center control platform, including features like hot-desking, agent log in and log out, real-time queue monitoring, and call "spying".

Note: Refer to the ***ESI eSIP Evolution Series QueueMetrics Live Integration Guide*** for detailed information.

eMobile

ESI eMobile enables users of the ESI eSIP Evolution Series to use their personal smart phones to communicate when they leave their offices, while providing mobile functionality that mirrors what they experience with their desk phones.

Note: Refer to the ***ESI eSIP Evolution Series eMobile User Guides*** for detailed information.