– N412 –

Administrator Manual

Sales Tel: +86-592-5503309
E-mail: sales@yeastar.com
Support Tel: +86-592-5503301
E-mail: support@yeastar.com
Web: http://www.yeastar.com

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WEEE Warning

In accordance with the requirements of council directive 2002/96/EC on Waste of Electrical and Electronic Equipment (WEEE), ensure that at end-of-life you separate this product from other waste and scrap and deliver to the WEEE collection system in your country for recycling.

Contents

N412 Overview

This chapter provides the following sections:

- Introduction
- Feature Highlights
- N412 Front Panel
- N412 Rear Panel

Introduction

Yeastar N412 is a flexible and modular PBX that provides productivity-enhancing communication platform for small business. Yeastar N412 can handle up to 4 CO/BRI lines, up to 12 analog extensions, 8 SIP extensions, and 4 SIP trunks. With Yeastar N412, small business can get business-class features with a compact and powerful analog and VoIP capable system.

Feature Highlights

- **Hybrid System**
  Pre-configured with 8 FXS ports and customizable with 4 module slots.
- **Modular Technology**
  Easily add 4 extra analog extensions and 4 external CO or BRI lines.
- **User-friendly Configuration**
  Manage the system via user-friendly Web interface without complicate operations.
- **Embedded Recording Capability**
  Record calls to monitor the conversation for various purposes required by your business.
- **Advanced Call Handling**
  Flexible call routing, effective call queuing and distribution handle incoming calls automatically.
- **Cloud Service**
  Detect the new firmware from cloud server and upgrade automatically.

Learn more about N412 here: http://www.yeastar.com/Products/Smart-PBX-N412

N412 Front Panel

Figure 1-1 N412 Front Panel

Table 1-1 N412 Front Panel - LED Description

LED	LED Status	Description
POWER	On	The power is switched on.
	Off	The power is switched off.
RUN	Blinking	The system is running properly.
	Not Blinking/Off	The system goes wrong.
LAN	Blinking	Stable LAN port connection.
	Off	No LAN port connection.
WAN	Blinking	Stable WAN port connection.
	Off	No WAN port connection.
FXS Port Status	Solid Green	The port is idle.
	Blinking Green	There is an ongoing call on the port.
Line Status	CO Red light	• Blinking slowly: no CO line is connected to the port. • Static: the CO line is idle. • Blinking rapidly: the CO line is busy.
	BRI Orange light	• Blinking: the BRI line is disconnected. • Static: the BRI line is connected or in use.

N412 Rear Panel

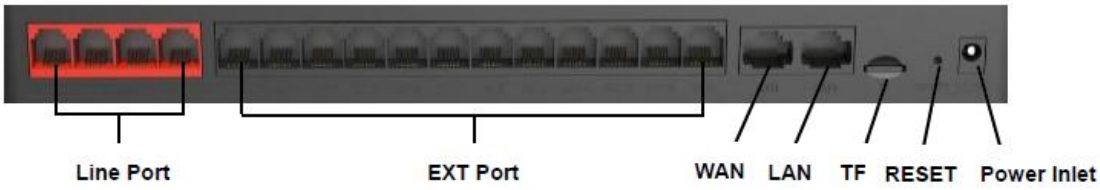


Figure 1-2 N412 Rear Panel

Table 1-2 N412 Rear Panels–Port Description

Port	Description
Line Port	For the connection of PSTN lines or ISDN BRI lines.
EXT Port	For connection of analog phones/fax machines.
TF Card Slot	Insert the TF card and restore the recording files.
LAN Port	10/100 Base-TX, connect one end of an RJ-45 Ethernet cable into the LAN port.
WAN Port	10/100 Base-TX, connect one end of an RJ-45 Ethernet cable into the WAN port, for the connection to internet.
Reset Button	Press and hold until all the LED turns off.
Power Inlet	For connection of power supply.

Application Overview

With N412, in addition to use the functions as traditional PBX, you could expand the communication flexibly with 4 SIP trunks, 8 SIP extensions. You will enjoy the N412 with its easy management that you had never experienced on a traditional PBX.

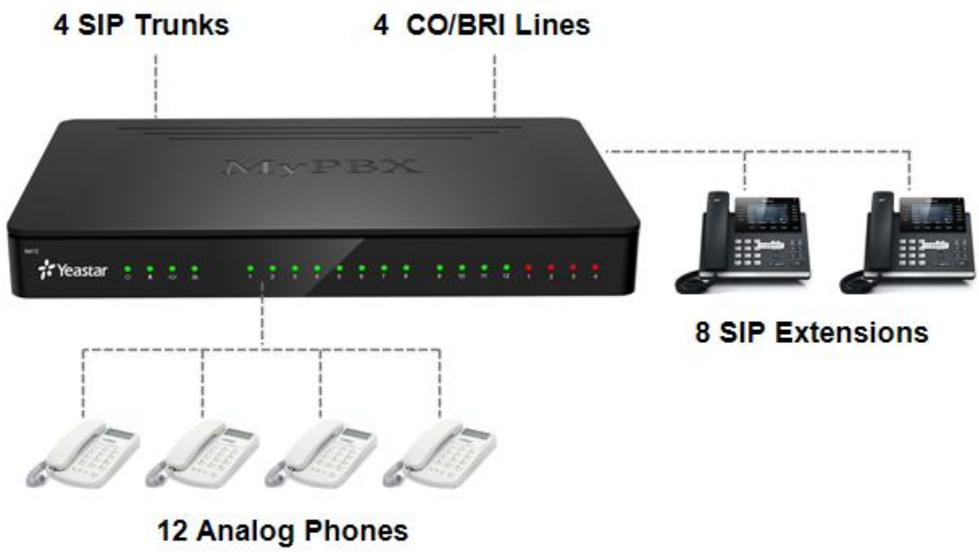


Figure 2-1 N412 Application Overview

Getting Started

In this chapter, we guide you through the basic steps to start with a new N412:

- Accessing Web GUI
- Web Configuration Panel
- User Management

Accessing Web GUI

N412 provides a web-based configuration interface for administrator and account user. The user can manage the device by logging in the Web interface. Check the factory defaults below:

IP address: http://192.168.5.150

User Name: admin

Default Password: password

1. Start the browser on PC. In the address bar, enter the IP address, click “Enter” button and then you can see the Web Configuration Panel login page.
2. Enter the Admin User Name and Password to log in.

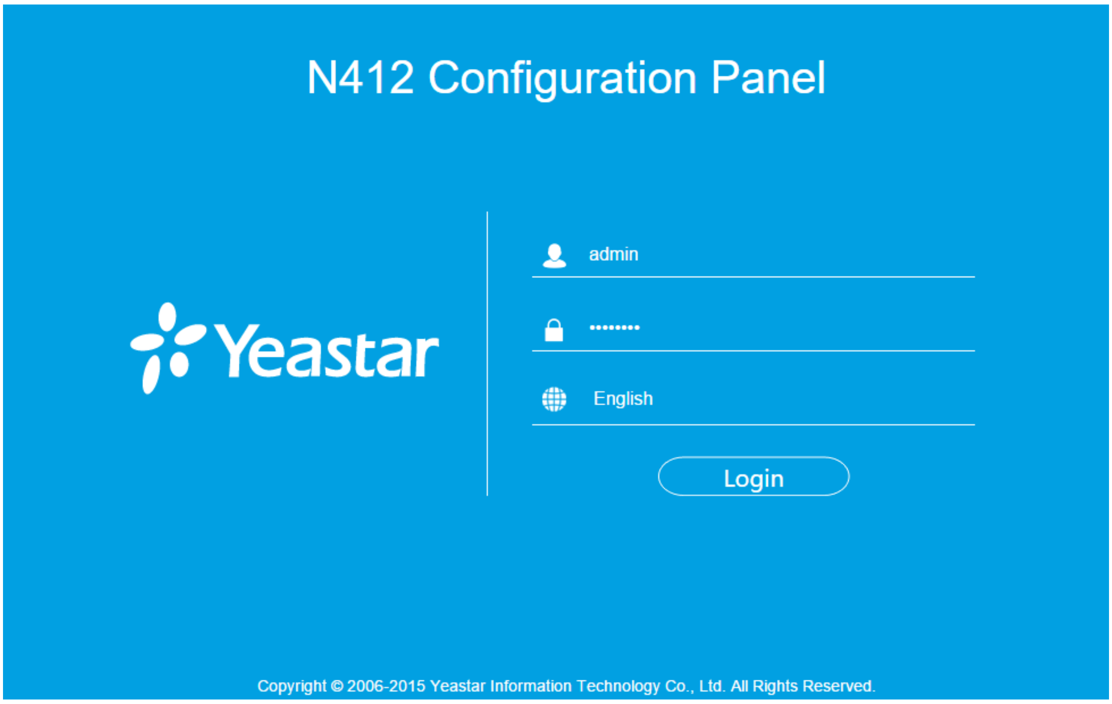


Figure 3-1 N412 Web Configuration Panel Login Page

Note:

It is highly recommended that you change the default password on first login.

Web Configuration Panel

There are 4 main sections on the Web Configuration Panel for users to check the N412's status and configure it.

- **Status:** check System Status, Extension Status, Trunk Status, Network Status and CDR.
- **System:** configure Network Settings, Security related Settings, System Date and Time, Password, Backup and Restore, Storage Management, Recording Settings etc.
- **PBX:** configure extensions, trunks, Call Routing, Call Features, Audio Settings, Voicemail Settings, SIP Settings etc.
- **Logout:** log out N412.

Note:
After saving the changes, remember to click the "Apply changes" button on the upper right of the Web GUI to make the changes take effect.

User Management

N412 supports two user types with different privileges.

User Privileges

- **Administrator** has the highest privilege. The administrator can access all pages on N412 Web and make all the configurations on the system.
  Username: admin
  Default Password: password

- **Extension User** has the privilege to check voicemails, one-touch recordings, auto recordings and CDR. The user can also configure settings and wake-up call for his own extension.
  Username: Extension number (i.e.601)
  Default Password: pass+ Extension number (i.e. pass601)

Enable Extension User

To log in N412 Web GUI using Extension User, you need to enable **User Web Interface** option for the extension.
Login N412, go to **PBX→ Extensions and Trunks→ Extensions**, choose an extension and click edit, check the **User Web Interface** options on **Account** tab.

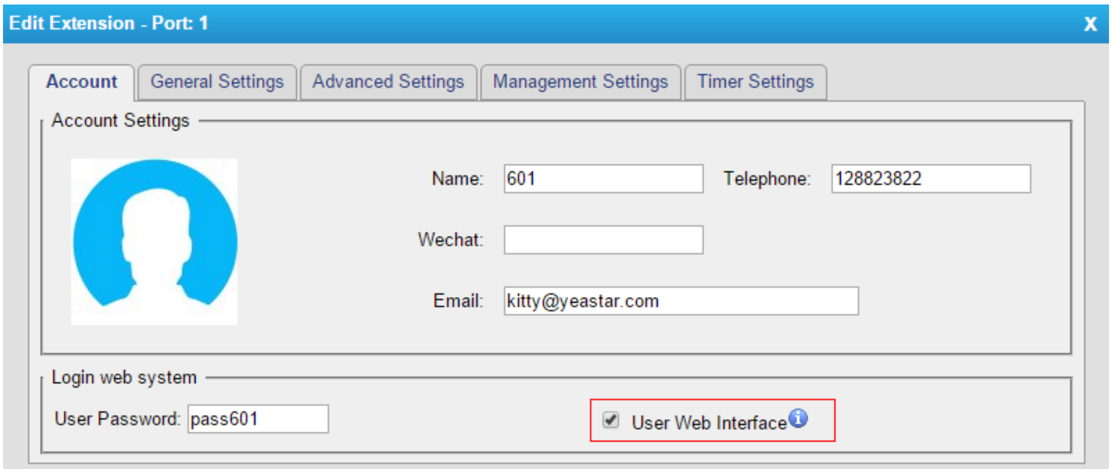


Figure 3-2 Enable Extension User

Set the privileges of CDR check and Auto Recording check on **Management Settings** tab.

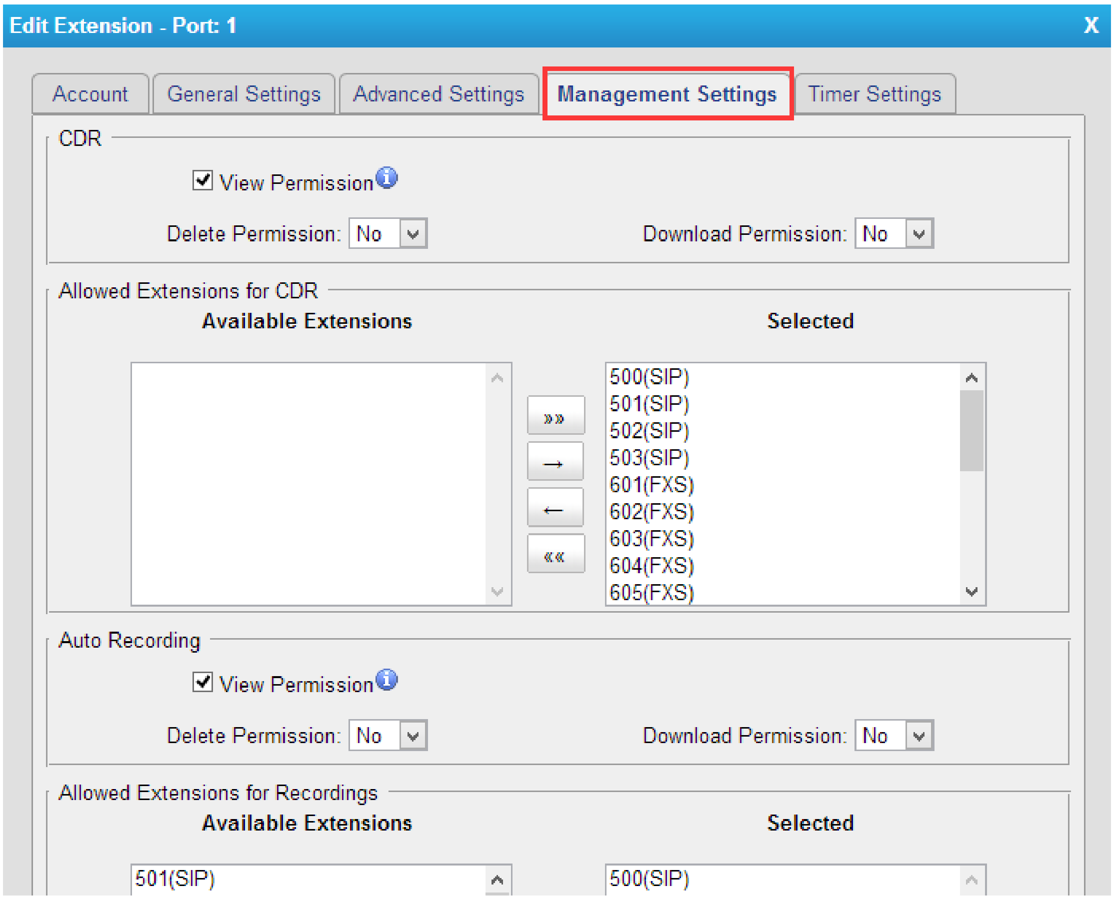


Figure 3-3 Management Settings

System Settings

This chapter explains system settings on N412. Click the main menu 

on the top of the Web GUI to check the system settings.

- Network Settings
- Security Center
- Date and Time
- Password Settings

Network Settings

LAN Settings

After successfully logging in the N412 Web GUI for the first time with the factory IP address, users could go **System→ Network Preferences →LAN Settings** to configure the network for N412.

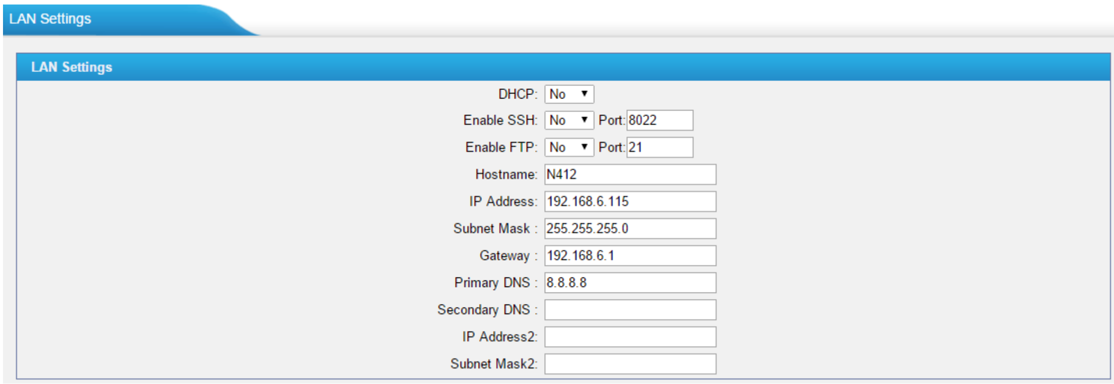


Figure 4-1 LAN Settings

Table 4-1 LAN Settings

Items	Description
DHCP	If this option is set as yes, N412 will act as DHCP client to get an available IP address from your local network. We don't recommend enabling this, as without the right IP address you cannot access N412.
Enable SSH	By using SSH, you can log in to N412 and run commands. It's disabled by default. We don't recommend enabling it if not needed. Default Port: 8022.

Enable FTP	Users could log in N412 via FTP if this option is enabled. Users could access FTP resource on N412 via Windows explorer or Web browser. FTP default user: **root**, password: **ys123456** Default Port: 21.
Hostname	Set the host name for N412.
IP Address	Set the IP Address for N412.
Subnet Mask	Set the subnet mask for N412.
Gateway	Set the gateway for N412.
Primary DNS	Set the primary DNS for N412.
Secondary DNS	Set the secondary DNS for N412.
IP Address2	Set the second IP Address for N412.
Subnet Mask2	Set the second subnet mask for N412.

WAN Settings

WAN port is disabled by default. Users should log in N412 to enable WAN. It supports “DHCP Server”, “PPPoE/dynamic DNS”, and “Static IP” for IP address assignment.

Note: N412 does not act as a router to route the internet packages from WAN port to LAN port.

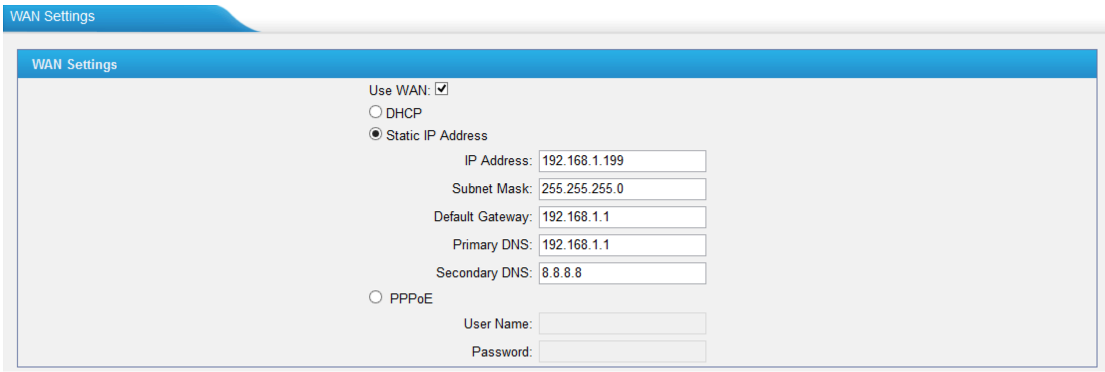


Figure 4-2 WAN Settings

Table 4-2 WAN Settings

Items	Description
DHCP	If your ISP says that you are connecting through DHCP or a dynamic IP address, perform these steps: Step1: Select “DHCP” as the WAN Connection Type. Step2: Save the changes. Step3: Reboot the device. Step4: Check the WAN Status (Status → Network status).

Static IP Address	If your ISP says that you are connecting through a static or fixed IP address, perform these steps: Step1: Select “Static IP Address” as the WAN Connection Type. Step2: Enter the IP Address. Step3: Enter the Subnet Mask. Step4: Enter the Gateway Address. Step5: Enter the Primary DNS and Secondary DNS. Step6: Save the changes. Step7: Reboot the device. Step8: Check the WAN Status (Status → Network status).
PPPOE	If your DSL provider says that you are connecting through PPPoE or if you normally enter a user name and password to access the Internet, perform these steps: Step1: Select “PPPoE” as the WAN Connection Type. Step2: Enter the User Name. Step3: Enter the Password. Step4:Save the changes. Step5: Reboot the device. Step6: Check the WAN Status (Status → Network status).

Security Center

Users are strongly recommended to configure firewall and other security options on N412 to prevent the attack fraud and the system failure or calls loss.

Security Center

All the security settings including Firewall, Service, Port Settings in N412 are displayed in Security Center. Users could rapidly check and configure the relevant security settings here.

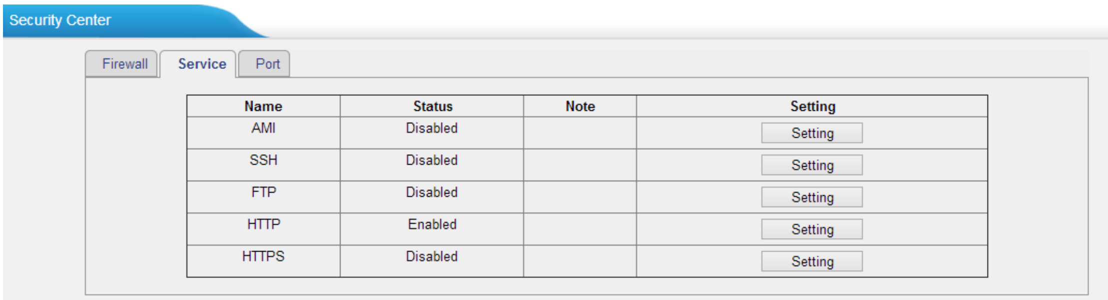
Firewall

In the “Firewall” tab, users could check firewall configuration and alert settings. By clicking the relevant button, you can enter the configuration page directly.

Figure 4-3 Security Center—Firewall

Service

In “Service” tab, you can check AMI/SSH status. For AMI/SSH, you can enter the according page by clicking the button in “Setting” column.

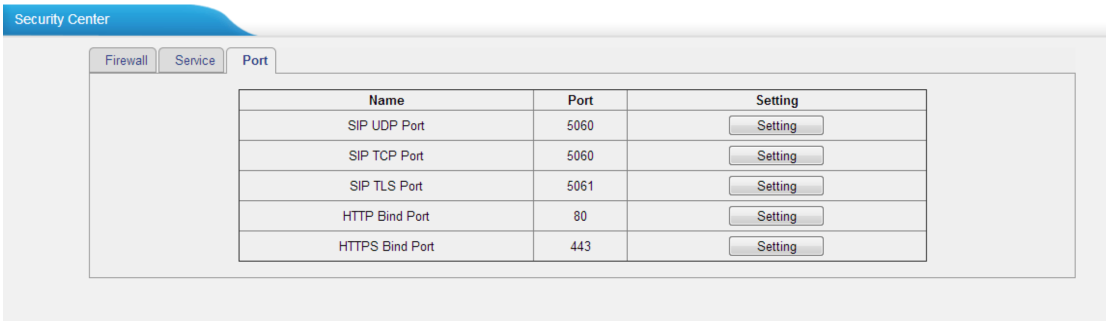


Name	Status	Note	Setting
AMI	Disabled		Setting
SSH	Disabled		Setting
FTP	Disabled		Setting
HTTP	Enabled		Setting
HTTPS	Disabled		Setting

Figure 4-4 Security Center—Service

Port

In “Port” tab, you can check SIP port and HTTP port. You can also enter the relevant page by clicking the button in “Setting” column.



Name	Port	Setting
SIP UDP Port	5060	Setting
SIP TCP Port	5060	Setting
SIP TLS Port	5061	Setting
HTTP Bind Port	80	Setting
HTTPS Bind Port	443	Setting

Figure 4-5 Security Center—Port

Firewall Rules

Firewalls are used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.

Figure 4-6 Firewall Settings

1) General Settings

Table 4-3 Description of Firewall General Settings

Items	Description
Enable Firewall	Enable the firewall to protect the device.
Disable Ping	Enable this item to drop net ping from remote hosts.
Drop All	When you enable “Drop All” feature, the system will drop all packets or connection from other hosts if there are no other rules defined. To avoid locking the devices, at least one “TCP” accept common rule must be created for port used for SSH access, port used for HTTP access and port sued for CGI access.

2) Common Rules

There is no default rule; you can create one as required.

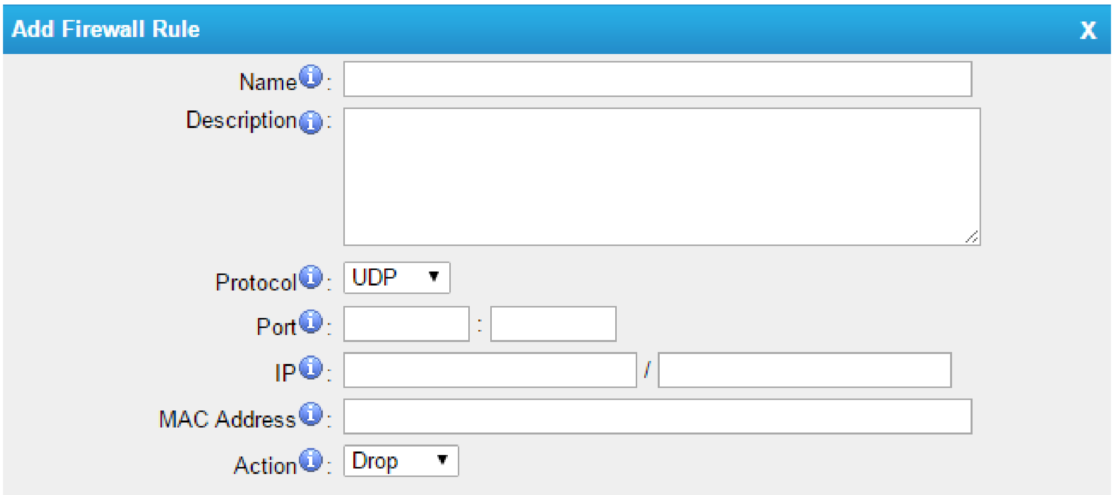


Figure 4-7 Common Rules

Table 4-4 Description of Common Rules

Items	Description
Name	A name for this rule, e.g. “HTTP”.
Description	Simple description for this rule. E.g. accept the specific host to access the Web interface for configuration.
Protocol	The protocols for this rule.
Port	Initial port should be on the left and end port should be on the right. The end port must be equal to or greater than start port.
IP	The IP address for this rule. The format of IP address is: IP/mask E.g. 192.168.5.100/255.255.255.255 for IP 192.168.5.100 E.g. 192.168.5.0/255.255.255.0 for IP from 192.168.5.0to 192.168.5.255.
MAC Address	The format of MAC Address is XX:XX:XX:XX:XX:XX, X means 0~9 or A~F in hex, the A~F are not case sensitive.
Action	Accept: Accept the access from remote hosts. Drop: Drop the access from remote hosts. Ignore: Ignore the access.

Note: the MAC address will be changed when it’s a remote device, so it will not be working to filter using MAC for remote devices.

3) Auto Defense

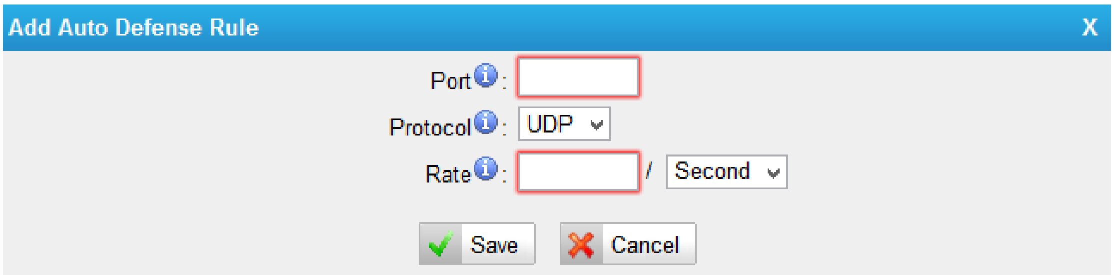


Figure 4-8 Auto Defense

Table 4-5 Description of Auto Defense

Items	Description
Port	The port you want to auto defense, for example, 8022.
Protocol	Select the protocol. You can select UDP or TCP.
Rate	The maximum packets or connections can be handled per unit time. For example, if you configure it as below: Port: 8022 Protocol: TCP Rate: 10/min Then, it means maximum 10 TCP connections can be handled in 1 minute. The 11th connection will be dropped.

IP Blacklist

You can set some packets accept speed rules here. When an IP address, which hasn't been accepted in common rules, sends packets faster than the allowed speed, it will be set as a black IP address and be blocked automatically.

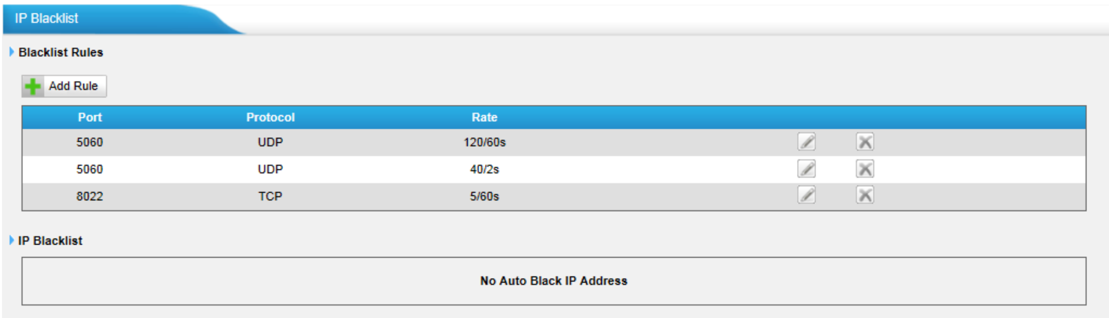


Figure 4-9 IP Blacklist Settings Page

1) Blacklist rules

We can add the rules for IP blacklist rate as demanded.

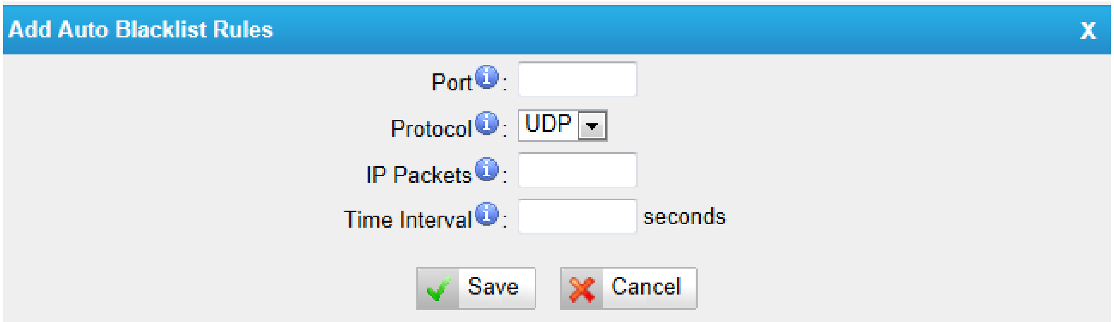


Figure 4-10 Add Blacklist Rule

Table 4-6 Description of Auto Blacklist Rules

Items	Description
Port	Auto defense port
Protocol	Auto defense protocol. TCP or UDP.
IP Packets	Allowed IP packets number in the specific time interval.
Time interval	The time interval to receive IP packets. For example, IP packets 90, time interval 60 means 90 IP packets are allowed in 60 seconds.

2) IP blacklist

The blocked IP address will display here, you can edit or delete it as you wish.

AMI Settings

The Asterisk Manager Interface (AMI) is a system monitoring and management interface provided by Asterisk. It allows live monitoring of events that occur in the system, as well enabling you to request that Asterisk perform some action. The actions that are available are wide-ranging and include things such as returning status information and originating new calls. Many interesting applications have been developed on top of Asterisk that take advantage of the AMI as their primary interface to Asterisk.

There are two main types of messages on the Asterisk Manager Interface: manager events and manager actions.

The 3rd party software can work with N412 using AMI interface. It is disabled by default. If necessary, you can enable it.

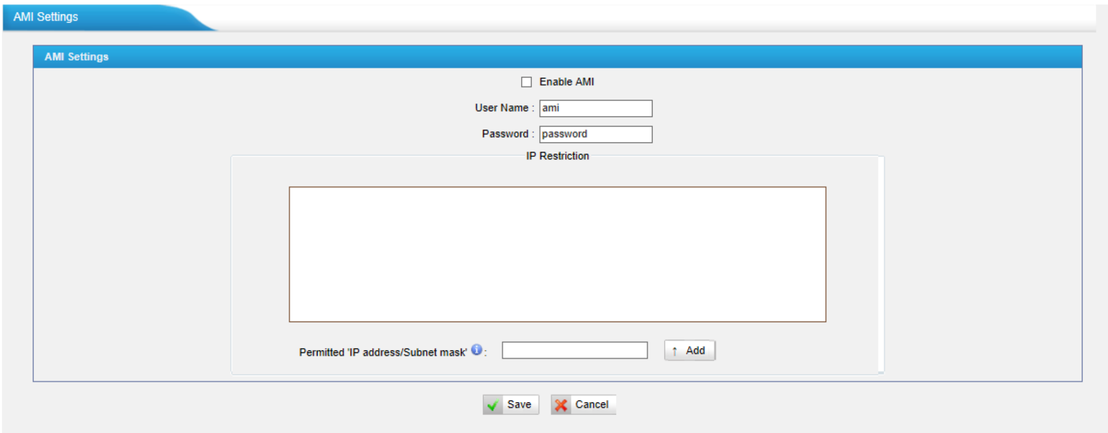


Figure 4-11 AMI Settings

- **Username & password**
  After enabling AMI, you can use this username and password to log in N412 AMI.
- **IP Restriction**
  You can set which IP is allowed to log in N412 AMI interface.

Certificates

N412 supports TLS protocol, you can register TLS extension and TLS SIP trunk on N412. To use TLS protocol, you need to upload TLS certificates to N412 first.

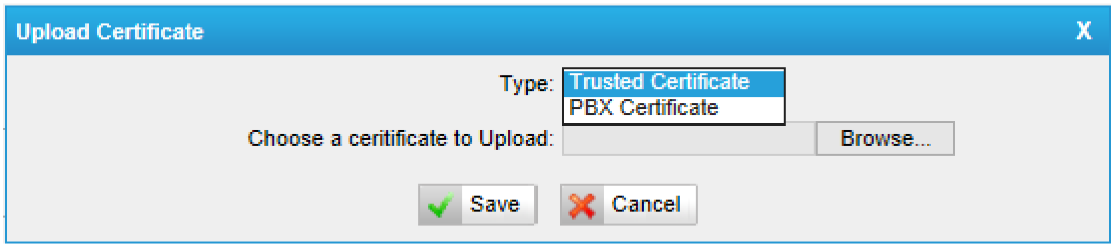


Figure 4-12 Certificates

- **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. IP phone) should also have this certificate.

- **PBX Certificate**
  This certificate is server certificate. No matter selecting “TLS Verify Client” as “Yes” or “NO”, you should upload this certificate to N412. If TLS client (i.e. IP phone) enables “TLS Verify server”, you should also upload the relevant CA certificate on IP phone.

Alert Settings

After enabling this feature, phone notification or email notification will be sent to users if the system has been attacked via IP or Web.

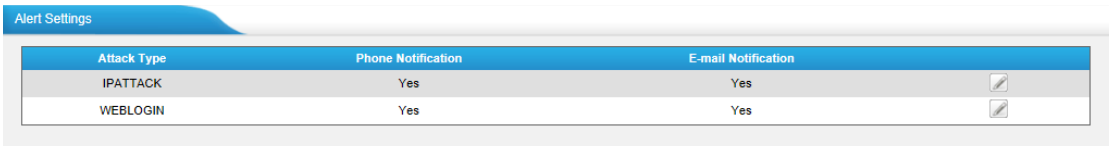


Figure 4-13 Alert Settings

- **IPATTACK**
  When the system is attacked by IP address, the firewall will add the IP to auto IP Blacklist and notify the user if it match the protection rule.
- **WEBLOGIN**
  Web Login Alert Notification: enter the incorrect password consecutively for five times will be considered as an attack, the system will limit the IP login within 10 minutes and notify the user.

1) Phone Notification Settings

Table 4-7 Description of Phone Notification Settings

Items	Description
Number	The numbers could be set for alert notification; users can setup multiple extension and outbound phone numbers. Please separate them by “;”. Example: “500;9911”, if the extension has configured Follow Me Settings, the call would go to the forwarded number directly.
Attempts	The attempts to dial a phone number when there is no answer.
Interval	The interval between each attempt to dial the phone number. Must be greater than 3 seconds, the default value is 10 seconds.
Prompt	Users will hear the prompt while receiving the phone notification.

2) Email Notification Settings

Please ensure that all voicemail settings are properly configured on the **PBX→ Basic Settings →Voicemail Settings** page before using this feature.

Table 4-8 Description of Email Notification Settings

Items	Description
Recipient's Name	The recipients for the alert notification, and multiple email addresses are allowed, please separate them by ";". Example:jerry@yeastar.com;jason@yeastar.com, 456@sina.com .
Subject	The subject of the alert email.
Email Content	Text content supports predefined variables. Variable names and corresponding instructions are as follows: $(HOSTNAME) Host name $(LOCALIP) Local IP address $(SOURCEIP) Attack source IP address $(DATETIME) Occurred $(USERNAME) User name (WEBLOGIN effective) $(DESTMAC) Attacks destination MAC (IPATTACK effective) $(DESTPORT) Attacks destination Port number (IPATTACK effective) $(PROTOCOL) Protocol type (IPATTACK effective) $(INTERFACE) Network interface name (IPATTACK effective)

Password Settings

It is highly recommended to change the system's password after first login. Go to **System→ System Preferences →Password Settings** to change the password.

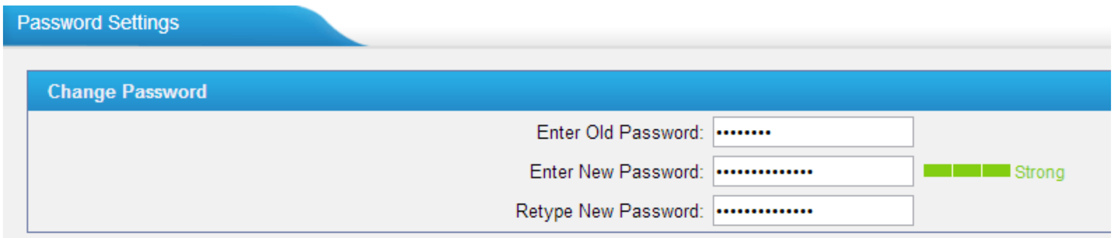


Figure 4-14 Change Password

1. Enter the old password first.
2. Enter a new password and retype the new password to confirm. The password complexity will be detected, which will help users to set a strong password and make N412 safer. A strong password is comprised of letters, numbers and characters.
3. Save the changes, the user will be automatically logged out.
4. Log in N412 using the new password.

Date and Time

Please adjust the time of N412 (including the time zone) consistent with your local time. Go to **System→ System Preferences →Date and Time** to configure the system date and time.

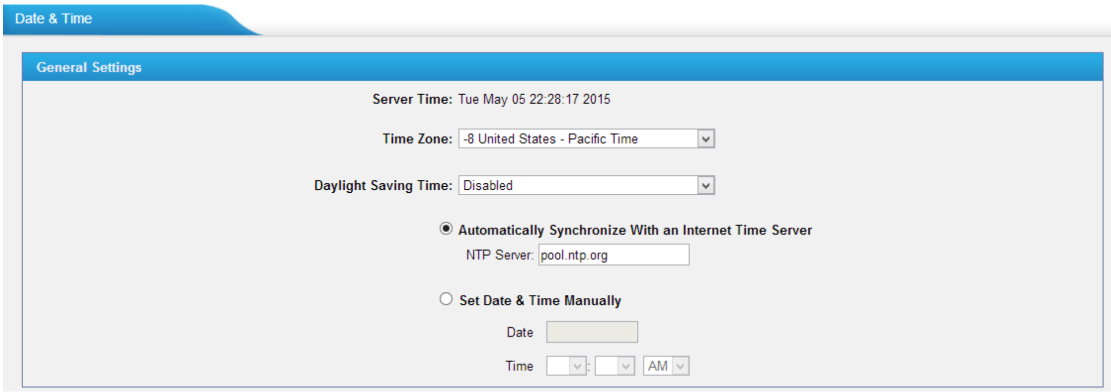


Figure 4-15 Configuring Date & Time

- **Time Zone**
  Select your current and correct time zone on N412.
- **Daylight Saving Time**
  The option is disabled by default. Enable it when necessary.
- **Automatically Synchronize with an Internet Time Server**
  N412 will adjust its internal clock to a central network server. Please note theN412 should be able to access to the Internet if you choose this method.
- **Set Date & Time Manually**
  Enter the time using the numbers on your keyboard.

Note:

You have to reboot the system to make the changes take effect.

Extensions

This chapter explains how to create and configure extensions on N412. It supports SIP extensions and FXS extensions, go to **PBX** → **Extensions and Trunks** → **Extensions** page to configure the extensions.

- FXS Extensions
- VoIP Extensions

FXS Extensions

N412 supports up to 12 FXS extensions. Users could click to edit each FXS extension.

FXS Extension Configuration

The extension settings are divided into Account, General Settings, Advanced Settings, Management Settings and Timer Settings.

1) Account

On this page, users could fill in the user information, including Name, Telephone number, Wechat ID and Email address.

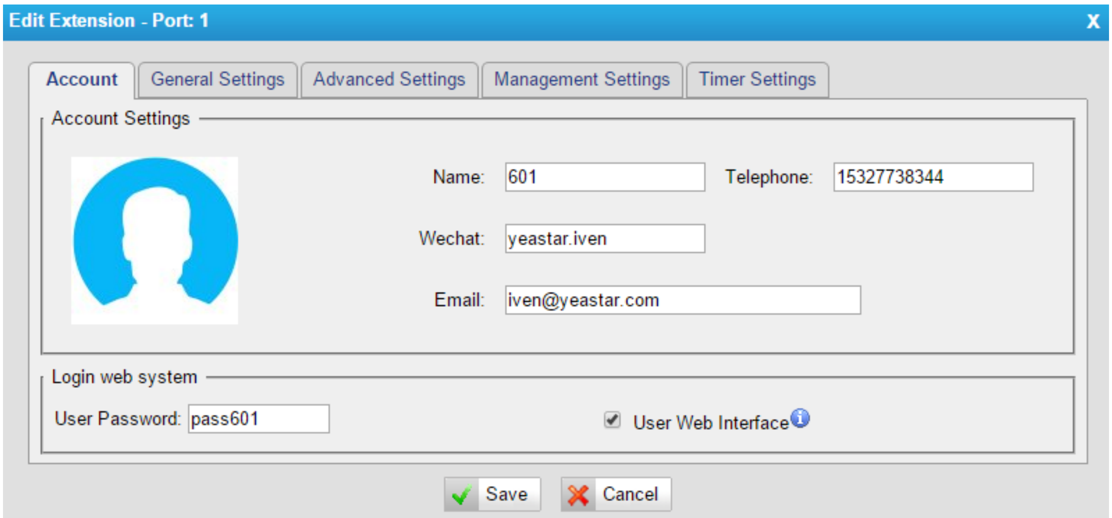


Figure 5-1 Account Information

The administrator can give a privilege to the extension user to log in N412 web interface.

Table 5-1 FXS Extension Account Settings

Items	Description
User Password	The password to log in the user Web interface.

User Web Interface	Check this option to allow the user to login to the N412 User Web interface, which can be used to check voicemail and extension recordings.

2) General Settings

Table 5-2 FXS Extension Configuration- General Settings

Items	Description
Extension	The numbered extension, which will be associated with this particular User/Phone.
Caller ID	The Caller ID will be used when this user calls another internal extension.
Voicemail	• Enable Voicemail Enable voicemail for the user. • Voicemail Access PIN The voicemail password (digits only) for the user to access the voicemail box.
Mail Setting	• Enable Send Voicemail Once enabled, the voicemail will be sent to a configured email address.
Hotline	• Enable Hotline: whether to use hotline. • Hotline Number: set a hotline number. • Delay Dial: define how long to make Hotline call after you pick up the call.
Flash	Sets the minimum/maximum time the phone is on hook before being detected as a hook flash.
Pickup Group	If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code (the default is *4). Note: *4 is the default setting, it can be changed under **Feature Codes→ General → Call Pickup**.
Max Call Duration	Setup the max cull duration for every call of this extension, but it's only valid for outbound calls. Enter "0" or leave this blank empty, the value would be equal to the max call duration configured in the "Basic Settings > General Preferences" page. Note: this setting will not be valid for internal calls.

3) Advanced Settings

Table 5-3 FXS Extension Configuration- Advanced Settings

Items	Description
Call Waiting	Check this option if the extension should have Call Waiting

	capability. If this option is checked, the “When busy” follow me options will not be available. The call waiting function of IP phone has higher priority than N412’s call waiting function.
DND	Don’t Disturb. When DND is enabled for an extension, the extension will not be available.
Busy Camp-on	If a dialed extension or a desired line is busy, with this feature, when the extension or line becomes idle, your telephone will ring automatically, so you can pick up to speak with the extension or seize the line and dial an external number.
Ring Out	Check this option if you want to customize the ring time. Ring tone will stop over the time defined.
Follow me	Call forwarding for an extension can be configured here. The administrator can configure Follow Me option for this extension. If you want to transfer the call to an outbound number, please follow the dial pattern of outbound route filled in the outbound number. For example: transferring to your mobile phone number 123456789, the dial pattern of outbound route is “9.”, you should fill in 9123456789 here.
Volume Settings	• Rxgain The Volume sent to FXS extension. • Txgain The Volume sent out by the FXS extension.
Caller ID Type	Normally, you choose the “default” option except for using N412 in Japan, in which case you should choose “Japan”.
Spy Settings	There are 4 spy modes available: • General spy You have the permission to use the following 3 modes. • Normal spy You can only hear the call, but can't talk • Whisper spy You can hear the call, and can talk with the monitored extension • Barge spy You can hear the call and talk with them both **Example:** If 500 want to monitor extension 501, we need to enable the “allow being spied” for 501, and choose the spy mode for extension 500. Then pick up 500 and dial “feature codes + 501” to start monitoring when 501 is in a call. If 500 choose “normal spy”, it should dial “*90501” to start monitoring. If 500 choose “whisper spy”, it should dial “*91501” to start

	monitoring. If 500 choose “barge spy”, it should dial “*92501” to start monitoring. If 500 choose “general spy”, it can dial “*90501”, “*91501” or “*92501” to start monitoring.

4) Management Settings

Once you enable “User Web Interface” for the extension, you need to also configure the Management settings to set the access permissions.

By default, extension users could check voicemail, one-touch Recordings, and configure settings of their own extensions when logging in User Web Interface.

If the user wants to manage the CDR and Auto Recordings, you have to set the access permissions here.

CDR

- View Permission: the permission to view CDR.
- Delete Permission: the permission to delete CDR.
- Download Permission: the permission to download CDR.
- Allowed Extension for CDR: choose which extensions’ CDR is allowed to be checked/deleted/downloaded by the user.

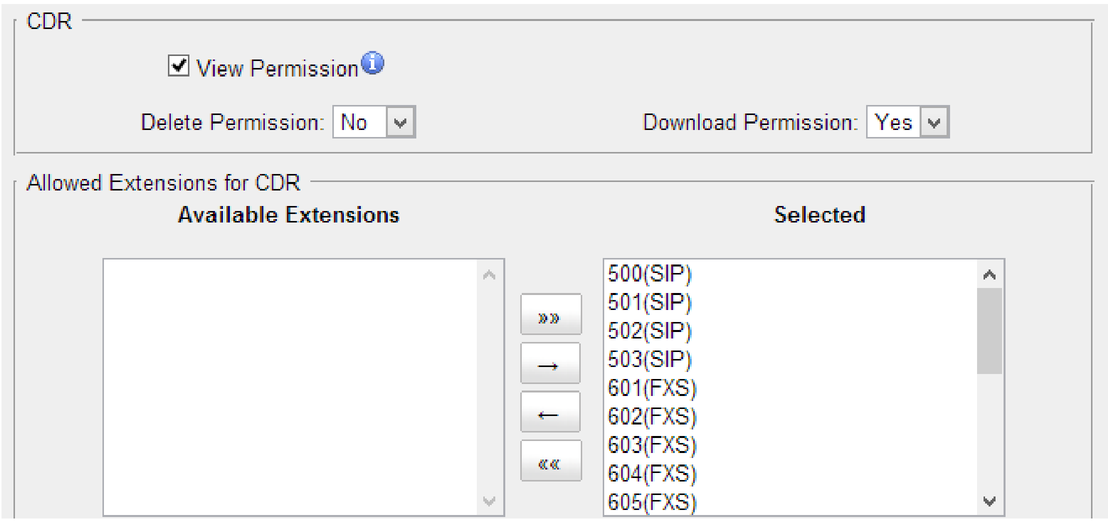


Figure 5-2 CDR Permissions for FXS Extensions

Auto Recordings

- View Permission: the permission to check auto recordings.
- Delete Permission: the permission to delete recording files.
- Download Permission: the permission to download auto recording files.
- Allowed Extension for Recordings: choose which extensions’ auto recording files are allowed to be checked/deleted/downloaded by the user.

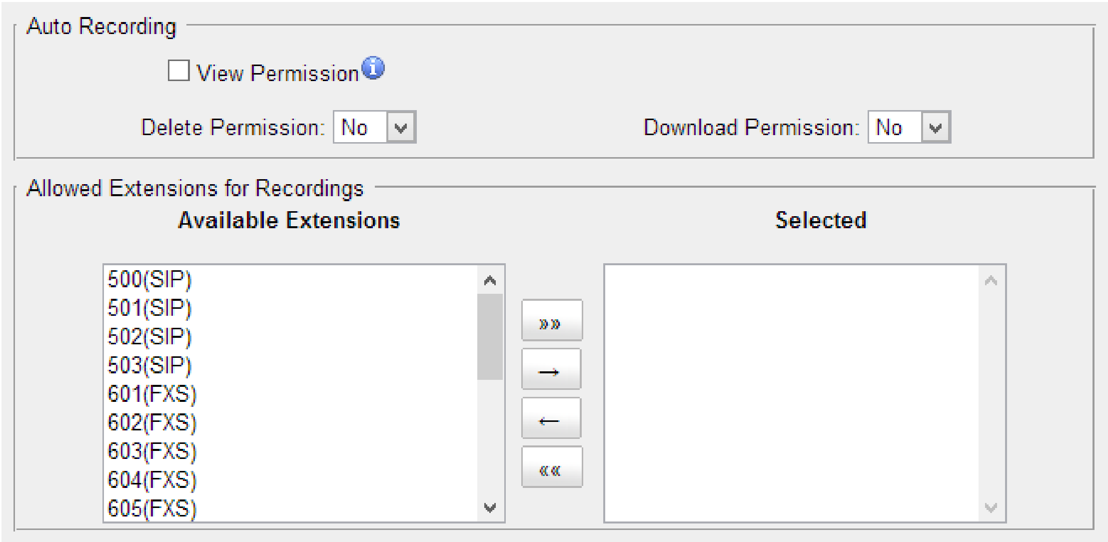


Figure 5-3 Auto Recordings Permissions for FXS Extensions

5) **Timer Settings**

Want the phone to wake you? Click Timer Settings Section, set your wake-up time and other options, and give the alarm a name (like "Good morning").

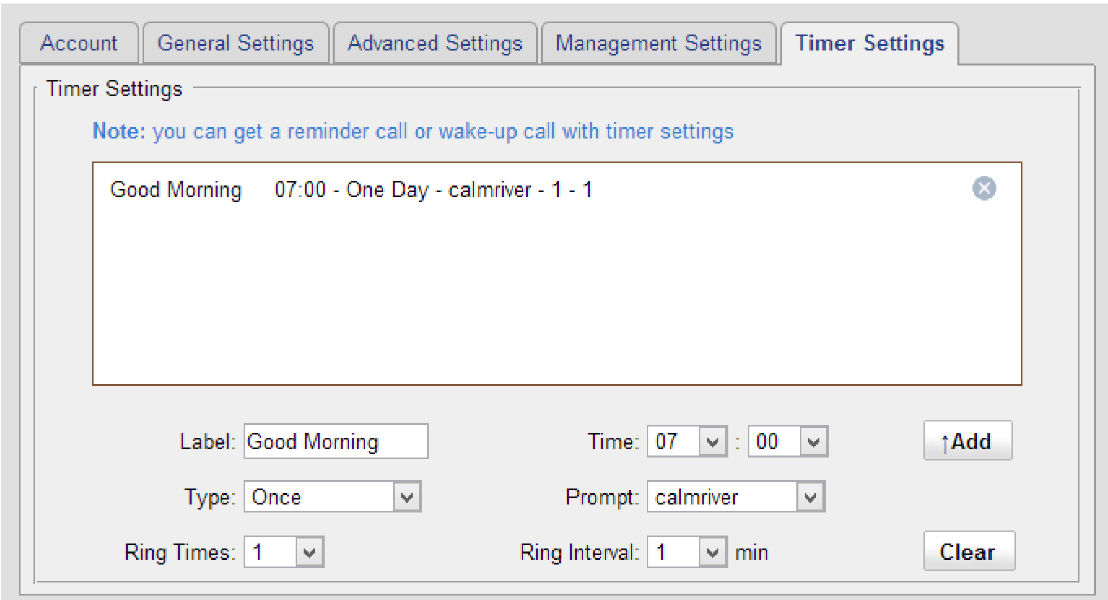


Figure 5-4 Timer Settings for FXS Extensions

Batch Edit FXS Extensions

Users could batch edit the selected FXS extensions' number, timer settings and other settings.

- **Modify the Number of the Selected Extensions**

  Click Modify Number of the Selected Extensions to modify the selected extensions. Define the extension number starting from a number.

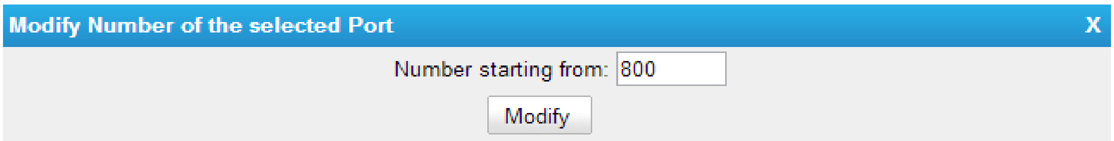


Figure 5-5 Modify Number of the Selected Ports

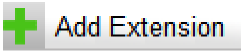
- **Edit the Selected Extensions**

  Click Edit the Selected Extensions, you can edit the selected extensions' General Settings, Voicemail, Volume, Hotline, Follow me and Group Settings.

- **Edit the Selected Timers**

  Click Edit the Selected Timers to set alarms for the selected extensions.

VoIP Extensions

Users could extend VoIP extension by clicking 

to add on VoIP extension. N412 supports up to 8 VoIP extensions.

VoIP Extension Configuration

The extension settings are divided into Account, General Settings, Advanced Settings, Management Settings and Timer Settings.

1) Account

On this page, users could fill in the user information, including Name, Telephone number, Wechat ID and Email address.

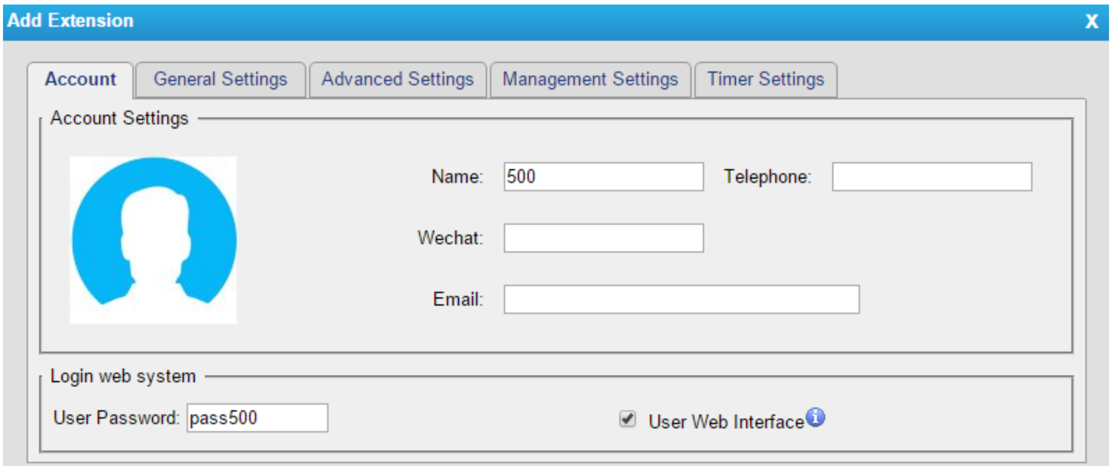


Figure 5-6 Account Information

You can also give a privilege to the extension user to log in N412 web interface.

Table 5-4 SIP Extension Account Settings

Items	Description
User Password	The password to log in the user web interface.
User Web Interface	Check this option to allow the user to login to the N412 User Web interface, which can be used to check voicemail and extension recordings.

2) General

Table 5-5 VoIP Extension Configuration- General

Items	Description
Extension	The numbered extension, which will be associated with this particular User/Phone.

Password	The password for this extension.
Caller ID	The Caller ID will be used when this user calls another internal extension.
Enable Voicemail	Enable voicemail for the user.
Voicemail Access PIN	The voicemail password (digits only) for the user to access the voicemail box.
Enable Send Voicemail	Once enabled, the voicemail will be sent to a configured email address.
Email Address	Email address used to receive the voicemail or Fax. Note: please ensure that the section “SMTP Settings For Voicemail” (in the “Voicemail Settings”) have been properly configured before using this feature.
Pickup Group	If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code (the default is *4). Note: *4 is the default setting, it can be changed under **Feature Codes→ General → Call Pickup**.
Max Call Duration	Set up the max cull duration for every call of this extension, but it’s only valid for outbound calls. Enter “0” or leave this blank empty, the value would be equal to the max call duration configured in the Option Settings page. Note: this setting will not be valid for internal calls.
NAT	This setting should be used when the system is using a public IP address to communicate 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	Send check alive packets to IP phones.
Enable SRTP	Enable extension for SRTP (RTP Encryption).
Transport	This will be the transport method used by the extension. The default is UDP. • UDP • TCP • TLS
DTMF Mode	Choose the DTMF mode: • RFC4733 • Info • Inband • Auto.
Remote Register	Allow to register remote extensions. If you enable “Remote Register”, the extension password must include uppercase letters, lowercase letters, and digits.

3) Advanced Settings

Table 5-6 VoIP Extension Configuration—Advanced Settings

Items	Description
Call Waiting	Check this option if the extension should have Call Waiting capability. If this option is checked, the "When busy" follow me options will not be available. The call waiting function of IP phone has higher priority than N412's call waiting function.
DND	Don't Disturb. When DND is enabled for an extension, the extension will not be available.
Enable Busy Camp-on	If a dialed extension or a desired line is busy, with this feature, when the extension or line becomes idle, your telephone will ring automatically, so you can pick up to speak with the extension or seize the line and dial an external number.
Ring Out	Check this option if you want to customize the ring time. Ring tone will stop over the time defined.
Follow me	Call forwarding for an extension can be configured here. The administrator can configure Follow Me option for this extension. If you want to transfer the call to an outbound number, please follow the dial pattern of outbound route filled in the outbound number. For example: transferring to your mobile phone number 123456789, the dial pattern of outbound route is "9.", you should fill in 9123456789 here.
IP Restriction	• **Enable IP Restriction** Check this option to enhance the VoIP security for N412. If this option is enabled, only the permitted IP/Subnet mask will be able to register this extension number. In this way, the VoIP security will be enhanced. • **Permitted "IP address/Subnet mask"** The input format should be "IP address" + "/" + "Subnet mask". E.g."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. E.g."192.168.5.0/255.255.255.0" means only the device whose IP address is 192.168.5.XXX is allowed to register this extension number.
Spy Settings	There are 4 spy modes available: • General spy You have the permission to use the following 3 modes. • Normal spy You can only hear the call, but can't talk. • Whisper spy

	You can hear the call, and can talk with the monitored extension. • Barge spy You can hear the call and talk with them both. Example: If 500 want to monitor extension 501, we need to enable the “allow being spied” for 501, and choose the spy mode for extension 500. Then pick up 500 and dial “feature codes + 501” to start monitoring when 501 is in a call. If 500 choose “normal spy”, it should dial “*90501” to start monitoring. If 500 choose “whisper spy”, it should dial “*91501” to start monitoring. If 500 choose “barge spy”, it should dial “*92501” to start monitoring. If 500 choose “general spy”, it can dial “*90501”, “*91501” or “*92501” to start monitoring.

4) Management Settings

Once you enable “User Web Interface” for the extension, you need to also configure the Management settings to set the access permissions.

By default, extension users could check voicemail, one-touch Recordings, and configure settings of their own extensions when logging in User Web Interface.

If the user wants to manage the CDR and Auto Recordings, you have to set the access permissions here.

CDR

- View Permission: the permission to check CDR.
- Delete Permission: the permission to delete CDR.
- Download Permission: the permission to download CDR.
- Allowed Extension for CDR: choose which extensions’ CDR is allowed to be checked/deleted/downloaded by the user.

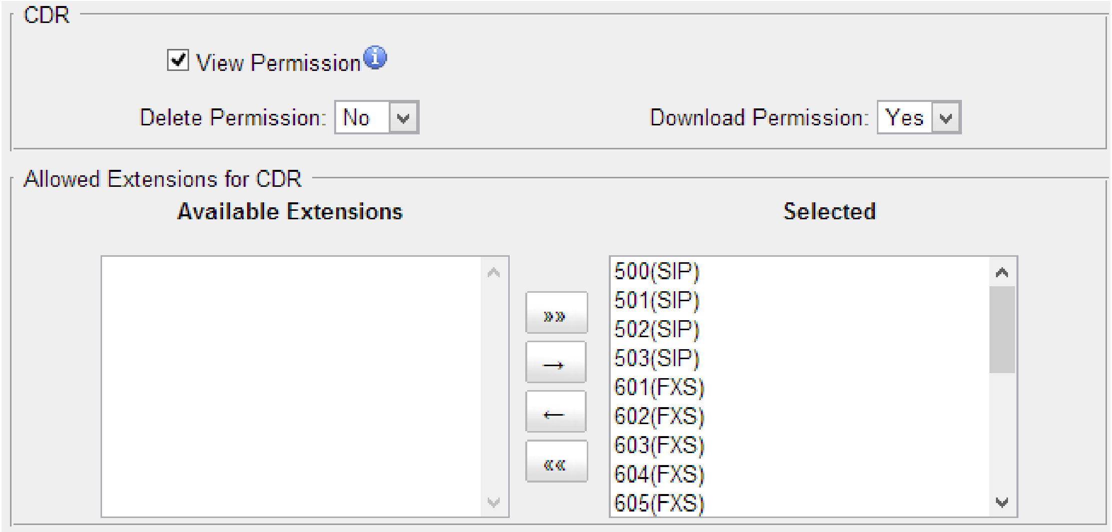


Figure 5-7 CDR Permissions

Auto Recordings

- View Permission: the permission to check auto recordings.
- Delete Permission: the permission to delete recording files.
- Download Permission: the permission to download auto recording files.
- Allowed Extension for Recordings: choose which extensions' auto recording files are allowed to be checked/deleted/downloaded by the user.

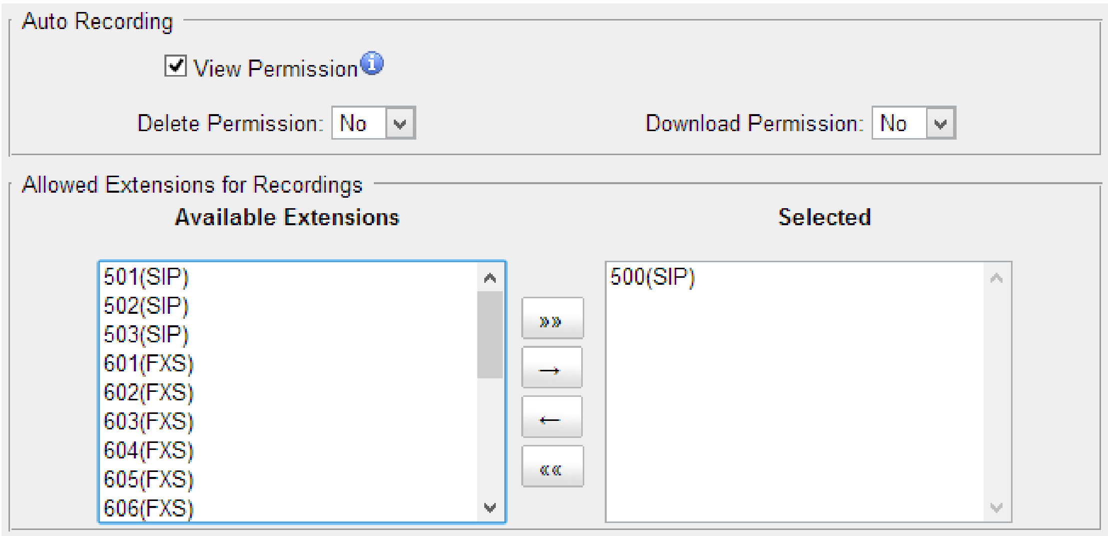


Figure 5-8 Auto Recording Permissions

5) Timer Settings

Want the phone to wake you? Click Timer Settings Section, set your wake-up time and other options, and give the alarm a name (like "Good morning").

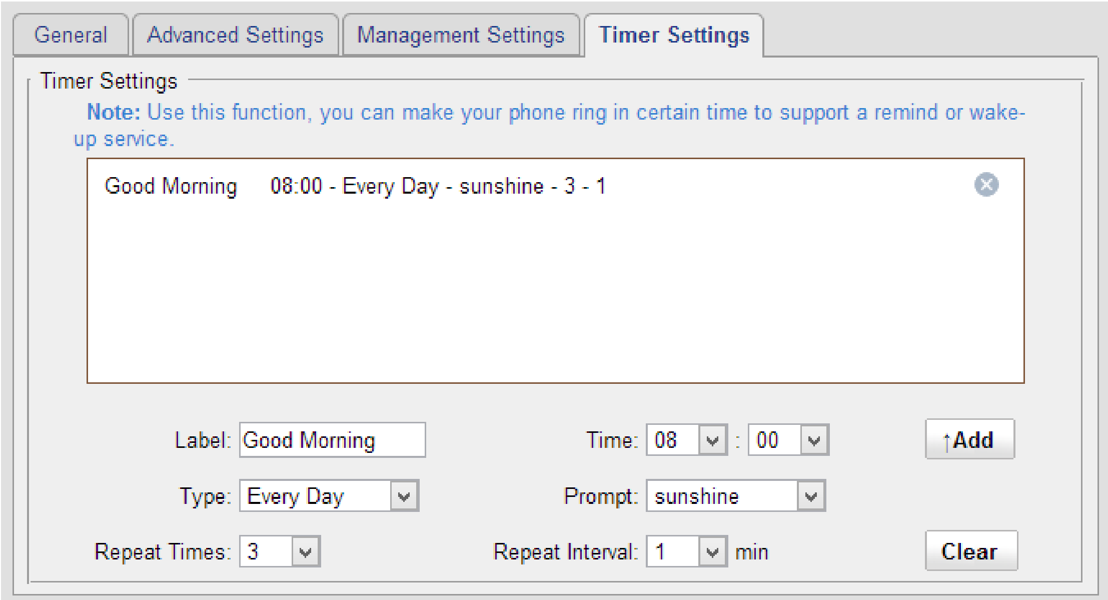


Figure 5-9 Timer Settings

Trunks

External calls can be made through CO lines, BRI trunks or via VoIP trunks on N412. In this chapter, we give a simplified guide to the N412 users in setting up trunks. We describe CO, BRI lines configurations and how to configure N412 to work with VoIP Providers.

- CO Lines
- BRI Trunks
- VoIP Trunks

CO Lines

CO lines also known as PSTN trunks. The public switched telephone network (PSTN) is the network of the world's public circuit-switched telephone networks.

Go to **PBX → Extensions and Trunks→ Trunks → CO lines** to edit the CO lines. Before configuring a CO line, please make sure that the CO line is connected to N412 CO port.

Click to edit the CO line.

CO Line Configuration

Please check the CO line configuration parameters below.

1) General Settings

Table 6-1 CO Line-General Settings

General Settings	
Trunk Name	A unique label used to identify this trunk.
Volume Setting	Set the volume for this trunk. The default is 40%.

2) Hangup Detection

Hangup 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 configured.

Table 6-2 CO Line-Hangup Detection

Hang up Detection	
Hangup Type	Choose the Hangup type. • Default • Busy Tone • Polarity

Busy Detection	Busy Detection is used to detect far end hang-up or for detecting a busy signal. Select “Yes” to turn this feature on.
Busy Count	If Busy Detection is enabled, it is also possible to specify how many busy tones to wait for before disconnecting the call. The default is 4, but better results can be achieved if set to 6 or even 8. Remember, the higher the number, the more time will be required to release a channel. A higher setting lowers the probability that you will encounter random hang-ups.
Busy Interval	The busy detection interval.
Busy Pattern	If Busy Detection is enabled, it is also possible to specify the cadence of your busy signal. In many Countries, it is 500msec on, 500msec off. Without Busy Pattern specified, N412 will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then N412 will further check the length of the tone and silence, which will further reduce the chance of a false positive disconnection.
Frequency Detection	Used for Frequency Detection (Enable detecting the busy signal frequency or not).
Busy Frequency	If the Frequency Detection is enabled, you must specify the local frequency.
Hangup Polarity Detection	The call will be considered as “hang up” on a polarity reversal.
Silence Timeout	Define the ring out value (in seconds) for this trunk.

3) Answer Detection

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

If the CO line could send polarity reversal signal after a call is established, you could choose "Polarity Detection" in this field. If not, you could choose "Ring Detection" and configure the rest of the settings accordingly.

Table 6-3 CO Line-Answer Detection

Answer Detection	
Answer Detection	Select which type to detect the call as answered. • Default: N412 will start to charge once you grab the CO line to call out, whether the call is answered or not. • Polarity Detection: if the CO line supports polarity, you can choose "Polarity detection". When the callee answers the call, the provider will send a polarity signal, and then N412 starts to bill. • Ring Detection: if you choose this option, N412 will charge the call according to CO line ring back tone

	detection. When the "ring duration" or the "ring interval duration" detected on N412 is larger than the standard parameters or custom parameters, the call is detected as ANSWERED. *Standard parameters: when you configure the "Tone Zone Settings" under **PBX → Basic Settings → General Preferences** you can get the country's standard tone parameters.
Custom Ring Tone	Enable or disable Custom Ring Tone. If the custom ring tone is enabled, you need to configure the following settings according to the ring back signal.
Max Ring Duration	Max duration of the ring tone.
Max Ring Interval Duration	Max pause between the two ring tones.
Min Ring Detection	Enable Min Ring Detection, which is useful for complex situations, like when jitter or noise occurs on the PSTN line. Generally it is disabled.
Min Ring Duration	Min duration of the received tone.
Min Ring Interval Duration	Min pause between the two received tones.

4) **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.

Table 6-4 CO Line-Caller ID Settings

Caller ID Settings	
Caller ID Detection	Enable/Disable the Caller ID detection.
Caller ID Start	This option allows you to define the start of a Caller ID signal. • Ring: start when a ring is received (Caller ID Signaling: Bell_USA, DTMF). • Polarity: start when a polarity reversal is started (Caller ID Signaling: V23_UK, V23_JP, DTMF). • Before Ring: start before a ring is received (Caller ID Signaling: DTMF).
Caller ID Signaling	This option defines the type of Caller ID signaling to use. It can be set to one of the following: • Bell: bell202 as used in the United States • v23_UK: suitable in the UK • v23_Japan: suitable in Japan

	• v23-Japan pure: suitable in Japan • DTMF: suitable in Denmark, Sweden, and Holland

5) **DNIS Settings**

DNIS (Dialed Number Identification Service) is a telephone service that identifies for the receiver of a call the number that the caller dialed.

Table 6-5 CO Line-DNIS Settings

DNIS Settings	
Enable DNIS	Tick to enable DNIS for this trunk.
DNIS Name	Define the DINS name.

6) **Other Settings**

Table 6-6 CO Line-Other Settings

Other Settings	
Ring Detect Timeout	FXO (FXS signaled) devices must have a timeout to determine if there was a hangup before the line was answered. Rang from 1000 to 8000. Default: 8000.

BRI Trunks

Basic Rate Interface (BRI, 2B+D, 2B1D) is an Integrated Services Digital Network (ISDN) configuration intended primarily for use in subscriber lines similar to those that have long been used for plain old telephone service. The BRI configuration provides 2 bearer channels (B channels) at 64 kbit/s each and 1 data channel (D channel) at 16 kbit/s. The B channels are used for voice or user data, and the D channel is used for any combination of data, control/signalling, and X.25 packet networking.

To extend BRI trunks on N412, please install B2 module on N412, and connect the BRI port to the BRI provider with a RJ45-RJ11 cable.

Select a BRI trunk and click to edit it.

1) General Settings

Table 6-7 BRI trunk-General Settings

General	
Trunk Name	A unique label used to identify this trunk when listed in outbound rules, incoming rules, etc. E.g. “BriTrunk1”.
Signaling	Set the Signaling method. • **BRI-PTP**: peer-to-peer signaling. • **BRI-PTMP**: peer-to-multiple-peer signaling.
Switch Side	• **User**: N412 connects to the BRI provider as user side. • **Network**: N412 connects to the BRI provider as network side.
Switch Type	Set switch type. • **national**: national ISDN type2 (common in the US). • **ni1**: national ISDN type 1. • **dms100**: Nortel DMS100. • **4ess**: AT&T 4ESS. • **5ess**: Lucent 5ESS. • **euroisdn**: EuroISDN. • **qsig**: D-channel signaling protocol at Q reference point for PBX networking.

2) Advanced Options

Table 6-8 BRI trunk-Advanced Options

Advanced Options	
Overlap Dial	Define whether N412 can dial this switch using overlap digits or not. If you need Direct Dial-in (DDI; in German "Durchwahl") you should change this to yes, then N412 will wait after the last digit it receives.
Reset Interval	Set the time in seconds between restart of unused channels. Some PBXs don't like channel restarts. So set the interval to a

	very long interval e.g. 100000000 or "never" to disable entirely. If you are in Israel, the following is important: As Bezeq in Israel doesn't like the B-Channel resets happening on the lines, it is best to set the reset interval to "never" when installing a box in Israel. Our past experience also shows that this parameter may also cause issues on local switches in the UK and China.
PRI Indication	Tells how Device should indicate Busy() and Congestion() to the switch/user. Accepted values are: • **inband**: Device plays indication tones without answering; not available on all PRI/BRI subscription lines . • **outofband**: Device disconnects with busy/congestion information code so the switch will play the indication tones to the caller. Busy() will now do same as setting PRI_CAUSE=17 and Hangup().
Enable Facility	To enable transmission of facility-based ISDN supplementary services (such as caller name from CPE over facility).
Nsf	Used with AT&T PRIs. If outbound calls are being rejected due to "Mandatory information element missing" and the missing IE is 0x20, then you need this setting.
Echo Cancellation	Disable or enable echo cancellation; it is recommended not to turn this off.
Hide Caller ID	If you want others to see your CID, please disable this option.
Codec	Set codec: • **alaw** • **ulaw**

3) DNIS Settings

Table 6-9 BRI Trunk-DNIS Settings

DNIS Settings	
Enable DNIS	Tick to enable DNIS for this trunk.
DNIS Name	Define the DINS name.
DID Number	Set the DID Number for this trunk.

4) Caller ID Prefix

Table 6-10 BRI trunk-Hangup Detection

Caller ID Prefix	
ISDN Dialplan	These settings are set to make the caller ID prefix work according to information sent from the BRI provider. ISDN/ telephony numbering plan Recommendation E.164.
International Prefix	When there are international calls coming in via this BRI trunk, the International Prefix you have set here will be added before the CID. So you can know this is an international call before you answer it.

National Prefix	When there are national calls coming in via this BRI trunk, the National Prefix you have set here will be added before the CID. So you can know this is a national call before you answer it.
Local Prefix	When there are Local calls coming in via this BRI trunk, the Local Prefix you have set here will be added before the CID. So you can know this is a local call before you answer it.
Private Prefix	When there are Private calls coming in via this BRI trunk, the Private Prefix you have set here will be added before the CID. So you can know this is a Private call before you answer it.
Unknown Prefix	When there are calls with unknown number coming via this BRI trunk, the Unknown Prefix you set here will be shown as the caller ID.

5) Dialplan

Table 6-11 BRI trunk-Dialplan

Dialplan	
Remote Dialplan	Calling number type.
Remote Number Type	Calling number identification.
Location Dialplan	Called number type.
Location Number Type	Called number identification.

6) DOD Settings

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

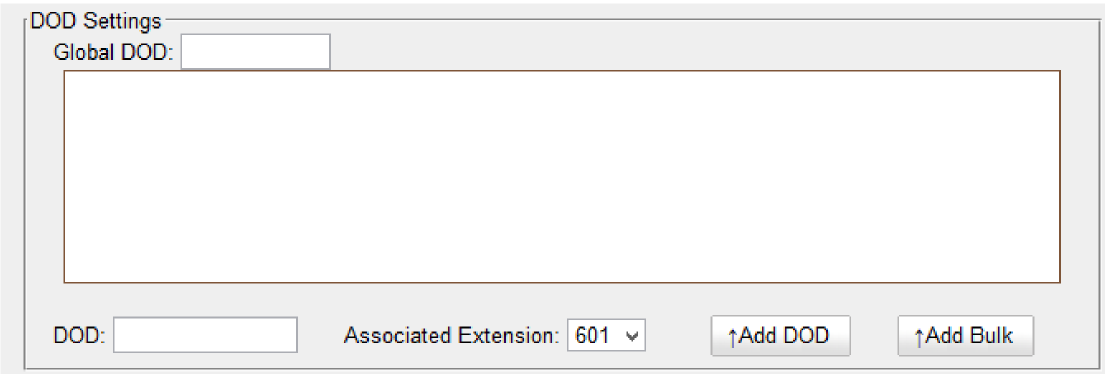


Figure 6-1 BRI trunk-DOD Settings

Table 6-12 BRI trunk-DOD Settings

DOD Settings	
Global DOD	Global direct outward dialing number.
DOD	Direct Outward Dialing Number.
Associated Extension	The extension make call out via BRI Trunk will display the associated DOD.

- **Add a DOD**

  Fill in DOD number and choose associated extension, then click ↑Add DOD to add one DOD number.

- **Add Bulk DOD**

  Click ↑Add Bulk to add DOD numbers in bulk.

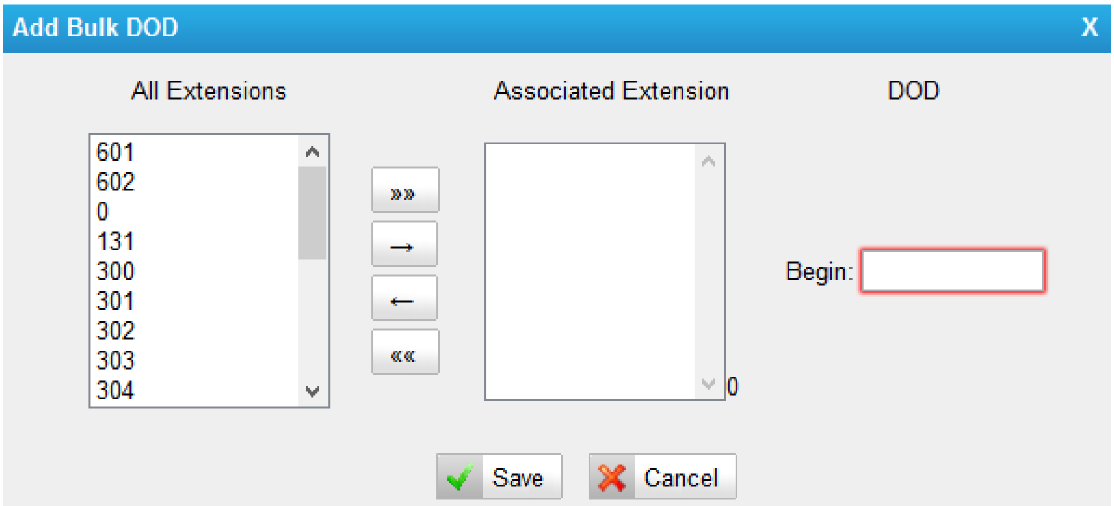


Figure 6-2 Add Bulk DOD

Add bulk DOD for bulk extensions in ascending sequence with the “Begin DOD” you fill in. For example, if the Associated Extensions are 100, 101, 102, 103, 104, 105 with “Begin DOD” as 5500100, the corresponding DOD will be 5500100, 5500101, 5500102, 5500103, 5500104, and 5500105.

VoIP Trunks

N412 provides 2 types of VoIP trunks:

- **VoIP trunk:** registration based VoIP trunk. A VoIP trunk requires N412 to register with the provider using an authentication name and password.
- **Service Provider:** IP based VoIP trunk. A Service Provider VoIP trunk does not require N412 to register with the provider. The IP address of N412 needs to be configured with the provider, so that it knows where calls to your number should be routed.

Go to **PBX→ Extensions and Trunks → VoIP Trunks** to edit the VoIP trunks.

VoIP Trunk

1) General Settings

Table 6-13 VoIP Trunk-General Settings

General Settings	
Enable/Disable	Enable or disable this VoIP trunk.
Provider Name	A unique label to help you identify this trunk.
Hostname/IP	Service provider's hostname or IP address. Default port: 5060. Don't change this part if it is not required.
Domain	VoIP provider's server domain name. If no domain name for the provider. Fill in the IP address instead.
User Name	The user name to register to the trunk from the VoIP provider.
Authorization Name	Used for SIP authentication.
Password	The password to register to the trunk from the VoIP provider.
From User	All outgoing calls from this 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.
Online Number	Define the online number that is expected by "Skype Connect" and some other SIP service providers. Leave this field blank if not needed.
Maximum Channels	Control the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Set as 0 to specify no maximum.
Caller ID	Specify the caller ID to use when making outbound calls over this trunk. The caller ID set in the "Extension" page will override

	the caller ID set in the "VOIP trunk" page. Please note that not all the service providers support this feature. Contact your service provider for more information.
Realm	Realm is a string to be displayed to users so they know which username and password to use.
Authenticating Incoming	• Yes: when an incoming call reaches N412 and sends INVITE packet to N412, N412 responds 401, but the Realm info in 401 Response does not match the Realm set on VoIP trunk, the provider will refuse to authenticate. • No: N412 will not reply a 401 Response to the provider to authenticate the incoming call.
Enable Outbound Proxy Server	A proxy that receives requests from a client. Even though it may not be the server resolved by the Request-URI.
Codecs	Define the codec for this sip trunk and its priority
Transport	This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider.
Enable SRTP	Define if SRTP is enabled for this trunk.
Qualify	Send check alive packets to the sip provider.
DTMF Mode	Set default mode for sending DTMF of this trunk. Default setting: rfc4733.

2) DNIS Settings

Table 6-14 VoIP Trunk-DNIS Settings

DNIS Settings	
Enable DNIS	Tick to enable DNIS for this trunk.
DNIS Name	Define the DINS name.
DID Number	Set the DID Number for this trunk.

3) DOD Settings

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

- **Associated Extension**
  The extension making call out via SIP Trunk will display the associated DOD.
- **Add DOD**
  Add DOD for one associated extension.
- **Add Bulk DOD**

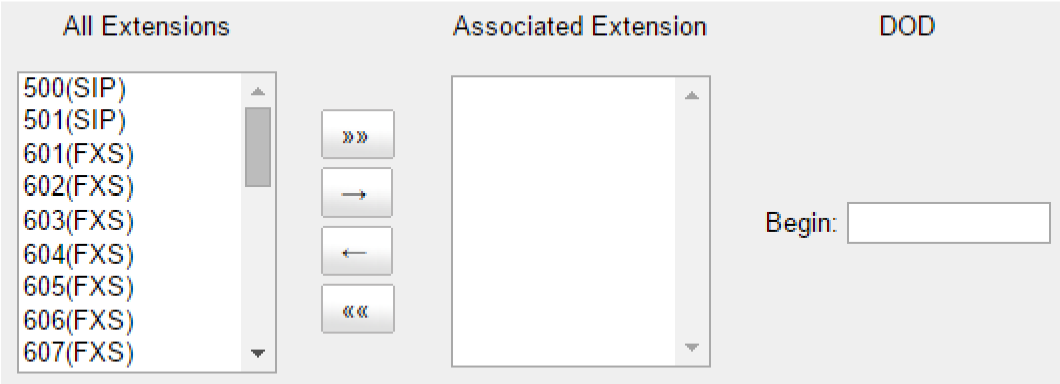


Figure 6-3 VoIP Trunk DOD Settings

Add bulk DOD for bulk extensions in ascending sequence with the “Begin DOD” you fill in. For example, if the Associated Extensions are 100, 101, 102, 103, 104, 105 with “Begin DOD” as 5500100, the corresponding DOD will be 5500100, 5500101, 5500102, 5500103, 5500104, and 5500105.

Service Provider

1) General Settings

Table 6-15 Service Provider Trunk-General Settings

General Settings	
Provider Name	A unique label to help you identify this trunk.
Hostname/IP	Service provider’s hostname or IP address. Default port: 5060.Don’t change this part if it is not required.
Maximum Channels	Control the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Set as 0 to specify no maximum.
Transport	This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider.
Qualify	Send check alive packets to the sip provider.
DTMF Mode	Set default mode for sending DTMF of this trunk. Default setting: rfc2833.

2) DOD Settings

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

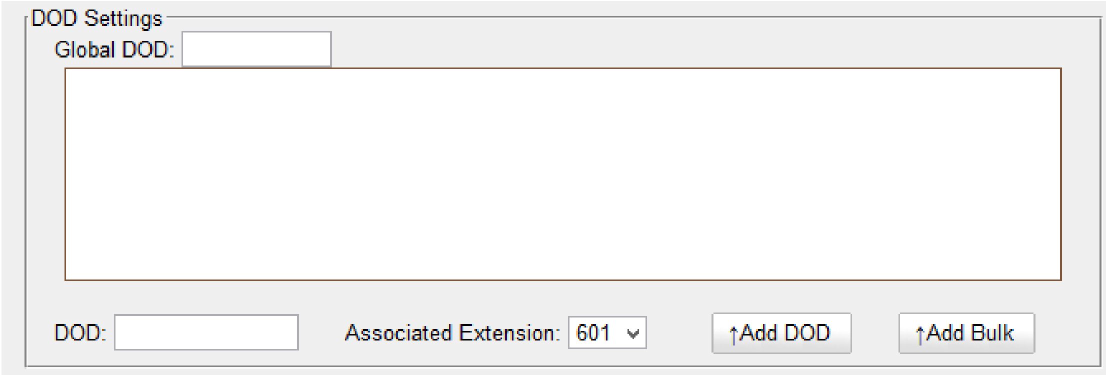


Figure 6-4 DOD Settings

Table 6-16 Service Provider Trunk-DOD Settings

DOD Settings	
Global DOD	Global direct outward dialing number.
DOD	Direct Outward Dialing Number.
Associated Extension	The extension make call out via BRI Trunk will display the associated DOD.

- **Add a DOD**

  Fill in DOD number and choose associated extension, then click ↑Add DOD to add one DOD number.

- **Add Bulk DOD**

  Click ↑Add Bulk to add DOD numbers in bulk.

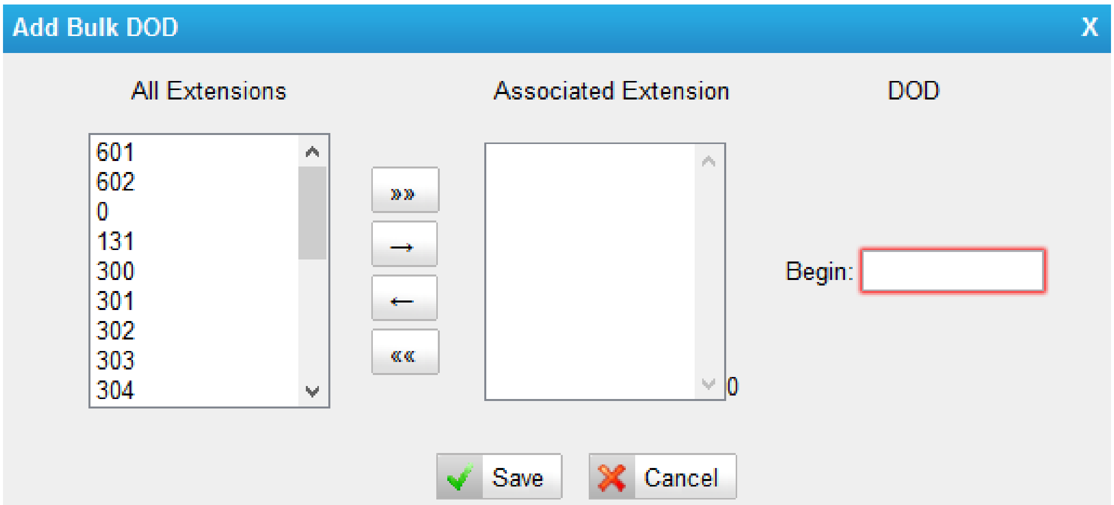


Figure 6-5 Add Bulk DOD

Add bulk DOD for bulk extensions in ascending sequence with the “Begin DOD” you fill in. For example, if the Associated Extensions are 100, 101, 102, 103, 104, 105 with “Begin DOD” as 5500100, the corresponding DOD will be 5500100, 5500101, 5500102, 5500103, 5500104, and 5500105.

Call Control

This chapter shows you how to control outbound calls and incoming calls with outbound routes and inbound routes.

- Outgoing Rules
- Incoming Rules
- PIN Settings
- Blacklist

Outgoing Rules

Outbound Route

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.

Note:

- N412 compares the number with the pattern that you have defined in your route 1. If 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 with route 2 and so on. The outbound route which is in a higher position will be matched firstly.
- Adjust the outbound route sequence by clicking these buttons .

Go to **PBX→ Outbound Call Control → Outgoing Rules** to edit outbound routes. Please check the outbound route configuration parameters below.

1) General Settings

Table 7-1 Outbound Route-General Settings

Options	Description
Route Name	Used to identify the route. The name is usually descriptive, i.e. "local" or "international".
Password	OPTIONAL. Select a PIN list from PIN Settings to set password for the

	outbound route. A route can prompt users for a password before allowing calls to process. Leave this field blank if you don't want to restrict this outbound route.
T.38 Support	Enable or disable T.38 FAX on this outbound route. Only for SIP Trunk.
Rrmemory Hunt	Round Robin with memory. If it is enabled, N412 will remember which trunk was used last time, and then use the next available trunk to call out.
Office Hours	This is an option to limit when the outbound route is available to use. Usually we can select an office hours that is same as your working hours, and the outbound route would be unavailable after work.

2) Dial Patterns

A dial pattern is a unique set of digits that will select this route and send the call to the designated trunks. Multiple Dial Patterns can be added on one outbound route by clicking 

button.

Table 7-2 Outbound Route-Dial Patterns

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	
Allow the users to 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 Extensions

Move the extensions could call through this outbound route to "Selected" Box.

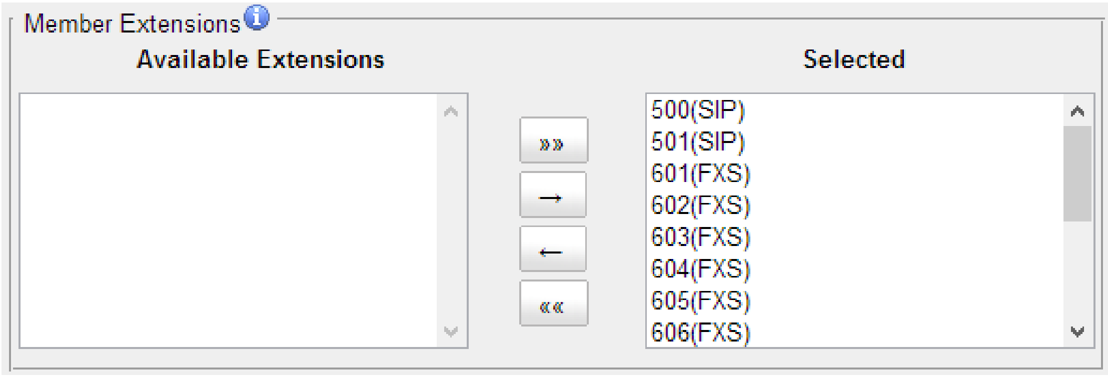


Figure 7-1 Outbound Route-Member Extensions

4) Member Trunks

Move the trunks that would be used on this outbound route to "Selected" Box.

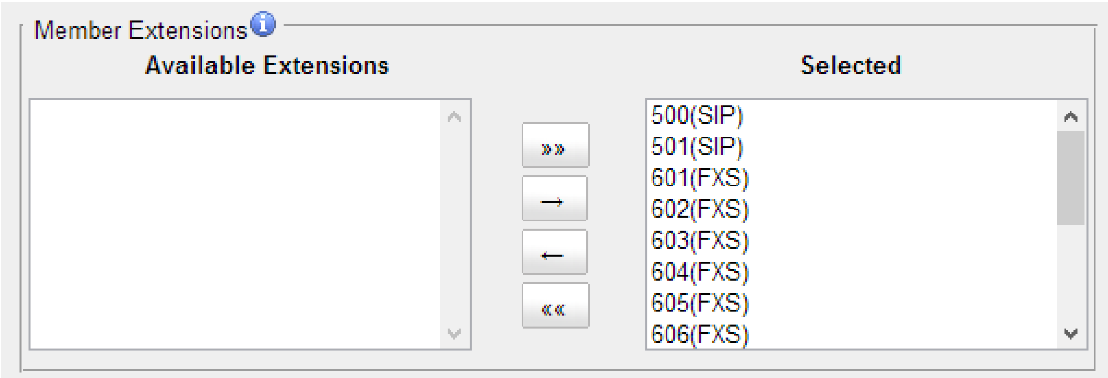


Figure 7-2 Outbound Route-Member Trunks

Seize a Line

On a traditional PBX, users have to seize a CO Line before dialing outside number. To adapt to these users' habits, we retain this feature. Users could seize an available CO Line by dialing a pre-configured number (default: 9), then get a dial tone and dial the external number to call out.

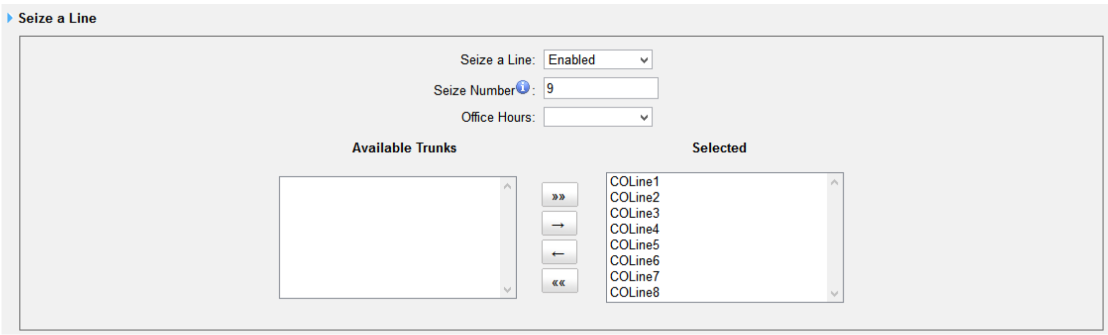


Figure 7-3 Seize a Line

Incoming Rules

When a call comes into N412 from the outside, N412 needs to know where to direct it. It can be directed to an extension, a ring group, a queue or a digital Receptionist (IVR) etc.

Go to **PBX→ Inbound Call Control → Incoming Rules** to edit incoming rules. Please check the inbound route configuration parameters below.

1) General Settings

Table 7-3 Inbound Route-General Settings

Options	Description
Route Name	Used to identify the route.
DID Number	Routing calls based on the trunk on which the call is coming in. In the DID field, you will define the expected “DID Number” if your trunk passes DID on incoming calls. Leave this blank to match calls with any or no DID info. The DID number entered must match the format of the provider sending the DID. You can also use a pattern match to match a range of numbers.
Caller ID Number	Routing calls based on the caller ID number of the person that is calling. Define the caller ID number to be matched on incoming calls. Leave this field blank to match any or no CID info.

2) Trunk Members

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

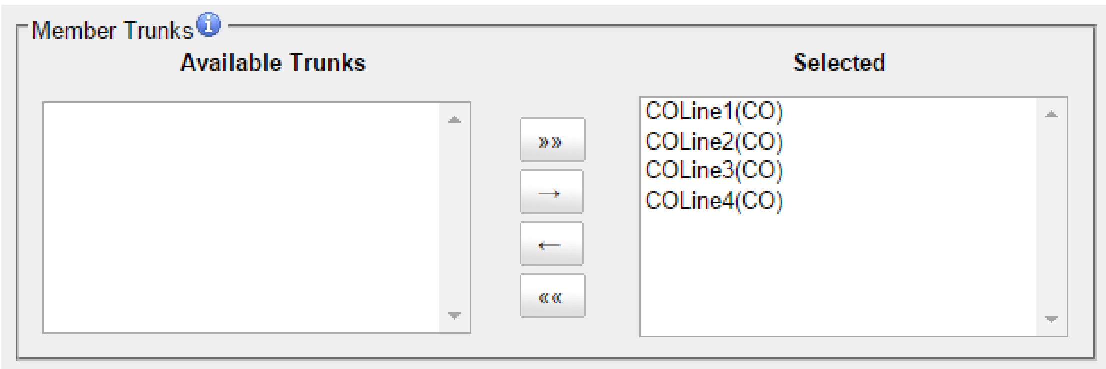


Figure 7-4 Inbound Route-Member Trunks

3) Business Days

Define where the calls will be routed during Business Days.

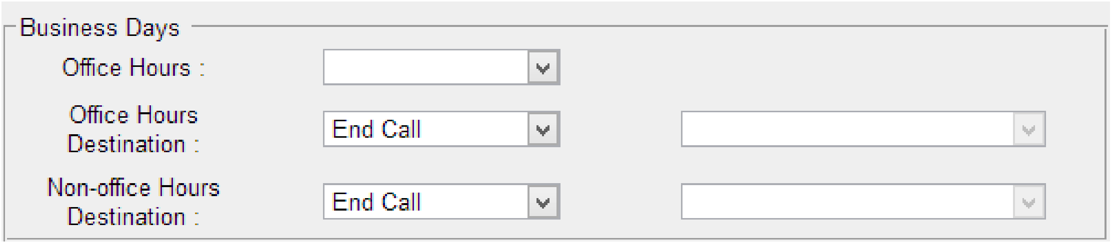


Figure 7-5 Inbound Route-Business Days

- Office Hours
  Select one defined business days office hours.
- Office Hours Destination
  Configure where to route the incoming calls during office hours.
- Non-office Hours Destination
  Configure where to route the incoming calls during non-office hours.

Table 7-4 Inbound Route-Business Days

Destination	Description
End Call	Route the incoming calls to end calls, the system will auto hang up the call.
Extension	Route the incoming calls to a specific extension.
Voicemail	Route the incoming calls to an extension's voicemail.
IVR	Route the incoming calls to a specific IVR.
Ring Group	Route the incoming calls to a specific Ring Group.
Conference Room	Used to initiate call processing as soon as it can be determined that no other matches are possible.
Queues	Route the incoming calls to a specific Queue.
Faxes	Route the incoming faxes to a specific extension's mail address. Note: this function only supports T.38 faxes.

4) During Holidays

Define where the calls will be routed during Holidays.

Figure 7-6 Inbound Route-During Holidays

- **Holiday**
  Select which defined Holiday to use. When a time is defined in both Business Days and Holidays, it will be treated as Holidays.
- **Destination**
  Configure where to route the incoming calls during holidays.

5) Fax Detection

Enable or disable the "Fax Detection" functionality on this route.

Figure 7-7 Inbound Route-Fax Detection

- **No Detect**

  No attempts are made to auto-determine the call type. All calls are sent to the defined destination.
- **Custom Email**

  Customize an E-mail address to receive the faxes. You should first configure the “Voicemail Settings->SMTP Settings for Voicemail” correctly before you use this option.
- **Faxes**

  Send faxes to an extension. If choosing a FXS extension here, the fax will be sent to the FXS port selected, you should connect a fax machine to this FXS port. If choosing a VoIP extension, the fax will be sent to the extension’s voicemail as an attachment.

Note:

If you want to receive faxes with custom Email address, the SMTP Settings should be configured successfully in advance. If you want to receive faxes with E-mail address configured in VoIP extension voicemail, you should first make sure the tested email to your email address works fine.

PIN Settings

Go to **PBX→ Advanced Settings→ PIN Settings** to create a PIN list. The PIN lists can be selected to access restricted features. The PIN can also be added to the CDR record’s “Account Code” field. PIN list can be applied to Outbound Route.

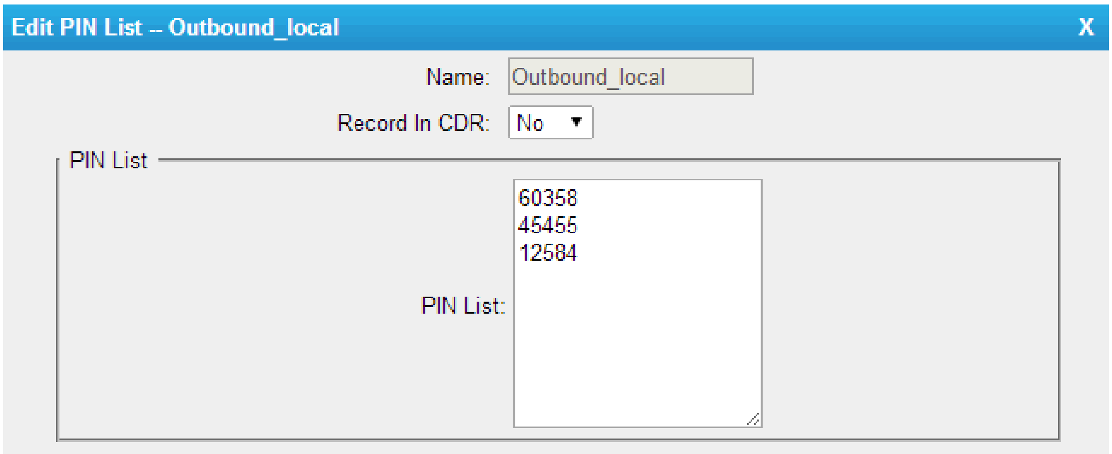


Figure 7-8 PIN Settings

Blacklist

Blacklist is used to block an incoming/outgoing call. If the number of incoming/outgoing call is registered 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.

Go to **PBX→ Advanced Settings → Blacklist** to add numbers to the blacklist.

You can choose to block the number for inbound, outbound or both.

- If the type is "inbound", then this number can't be called.
- If the type is "outbound", then the extensions in N412 can't call this number.

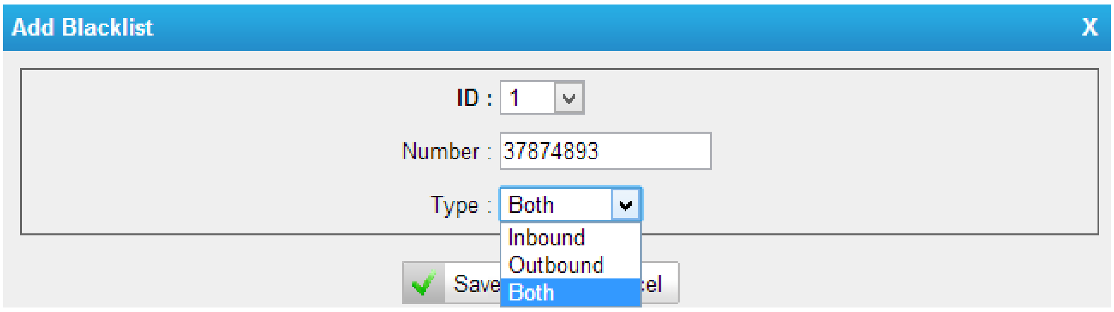


Figure 7-9 Number Blacklist

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 N412 to achieve it. When calls are routed to an IVR, N412 will play a recording prompting them what options the callers can enter such as “Welcome to XX, press 1 for Sales and press 2 for Technical Support”.

Configure IVR

Go to **PBX** → **Inbound Call Control** → **IVR** to configure IVR.

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

Please check the IVR configuration parameters below.

Table 8-1 IVR Configuration Parameters-General Settings

General Settings	
Number	N412 treats IVR as an extension; you can dial this extension number to reach the IVR from internal extensions.
Name	Set a name for the IVR.
Prompt	Choose which recording to be played to the caller when they reach the IVR. You can choose the default prompt on N412 or choose a **Custom Prompt** which is uploaded or created on N412.
Repeat Count	The number of times that the selected IVR prompt will be played.
Key Timeout	How long (in seconds) we 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.
Enable Direct Dial	Tick this option to enable Direct Dial. If Direct Dial is enabled, the callers can enter a user's extension number when entering the IVR to go direct to the users.

Key Press 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 Room • Queues • Faxes • Dial by Name • Hangup

Ring Group

A ring group helps you to ring a group of extensions in a variety of 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.

Configure Ring Group

Go to **PBX→ Inbound Call Control→ Ring Group** to configure ring group.

- Click Add Ring Group to add a new Ring Group.
- Click Delete the Selected Ring Group 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.

1) General Settings

Table 9-1 Ring Group Configuration Parameters-General Settings

General Settings	
Ring Group Name	Used to identify the ring group.
Ring Group Number	This option defines the numbered extension that can be dialed to reach this group.
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	Specify how long (in second) to ring each extension or all the extensions. • If the strategy is “Ring All Simultaneously”, it means the number of seconds to ring this group before routing the call according to the “Destination if No Answer” settings. • If the strategy is “Ring Sequentially”, it means the number of seconds to ring a single extension before moving on to the next one.

2) Ring Group Member

Specify the extensions to be part of this ring group. Move the desired ring group members to the "Selected" Box.

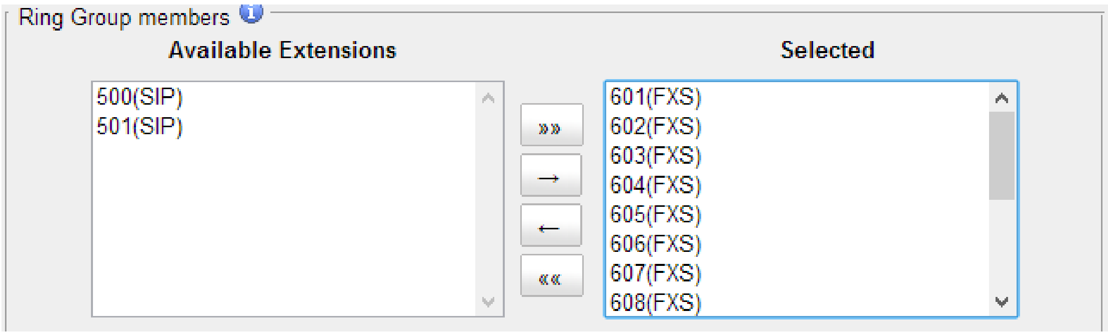


Figure 9-1 Ring Group Member

3) Destination if No Answer

When all members on this group fail to answer the call, system will handle the call according to the selected destination.

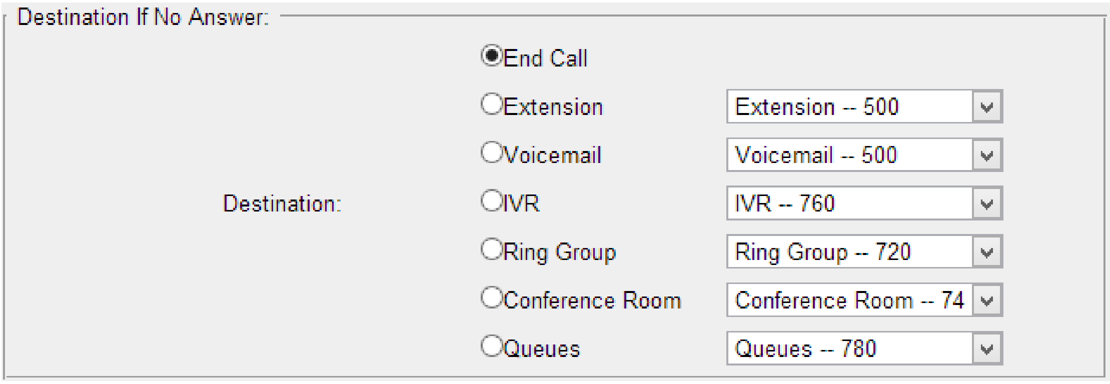


Figure 9-2 Destination if No Answer

Queue

Queues are designed to receiving 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 called in N412 and reached the queue, he/she 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.

Configure a Queue

Go to **PBX→ Inbound Call Control→ Queues** to configure queues.

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

Please check the Queue configuration parameters below.

1) General Settings

Table 10-1 Queue Configuration Parameters-General Settings

General Settings	
Name	A name for the Queue. The name is used for identification purpose throughout the user interface.
Number	Use this number to dial into the queue, or transfer callers to this number to put them into the queue.
Queue Password	You can require agents to enter a password before they can login to this queue.
Queue Agent Timeout	The number of seconds an agent's phone can ring before we consider it a timeout.
Queue Max Wait Time	The maximum number of seconds a caller can wait in a queue before being pulled out. (0 for unlimited).
Queue Ring Strategy	Multiple strategies are available for the queue. • **Ring All:** ring all available agents simultaneously until one answers. • **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.

2) 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.

The agents dial "Queue number" + "*" to log in or "Queue number" + "**" to log out the queue. For example, if the queue number is "681", then the dynamic agent can dial "681*" to log in or "681**" to log out.

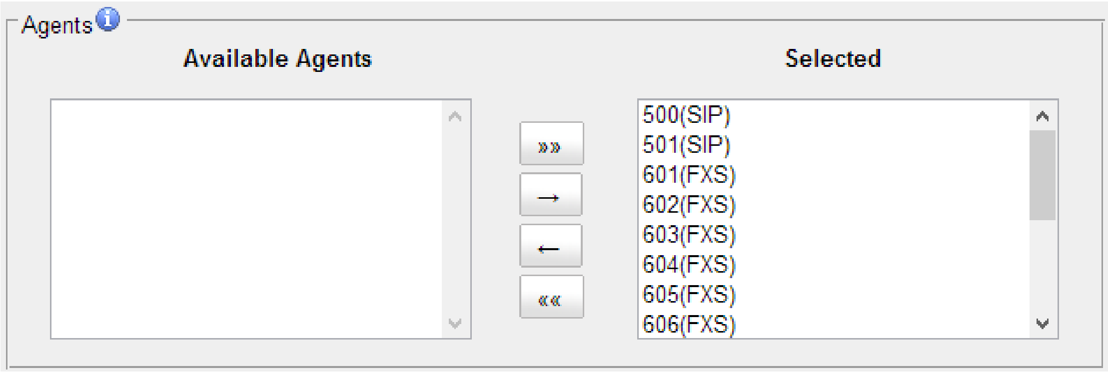


Figure 10-1 Agents

3) Caller Position Announcement

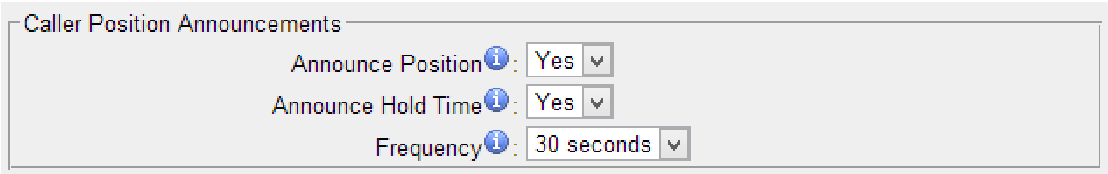


Figure 10-2 Caller Position Announcement

Table 10-2 Caller Position Announcement

Caller Position Announcement	
Announce Position	Whether to announce position of call in the queue or not.
Announce Hold Time	Enabling this option causes N412 to announce the hold time to the caller periodically based on the frequency timer. The hold time will not be announced if the time is less than 1 minute no matter this option is set to yes or no.
Frequency	How often to announce the queue position and estimated hold time.

4) Periodic Announcements

The periodic announcement is played periodically when the caller is waiting on the line.

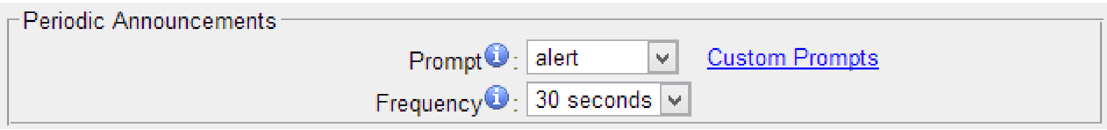


Figure 10-3 Periodic Announcements

5) **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.

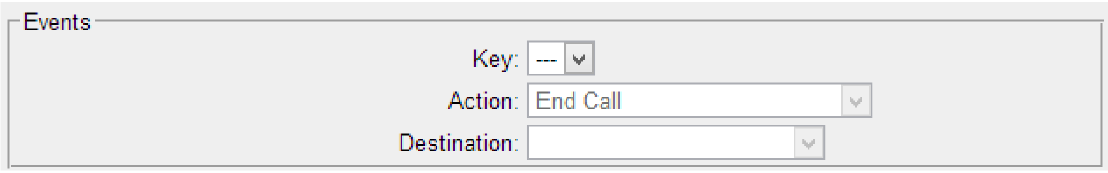


Figure 10-4 Events

6) **Failover Destination**

Define the failover action. A failover occurs after the user reach the Queue max wait time.

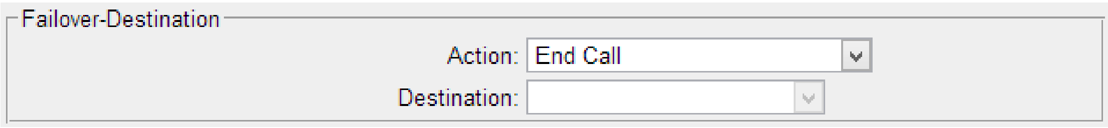


Figure 10-5 Failover Destination

7) **Others**

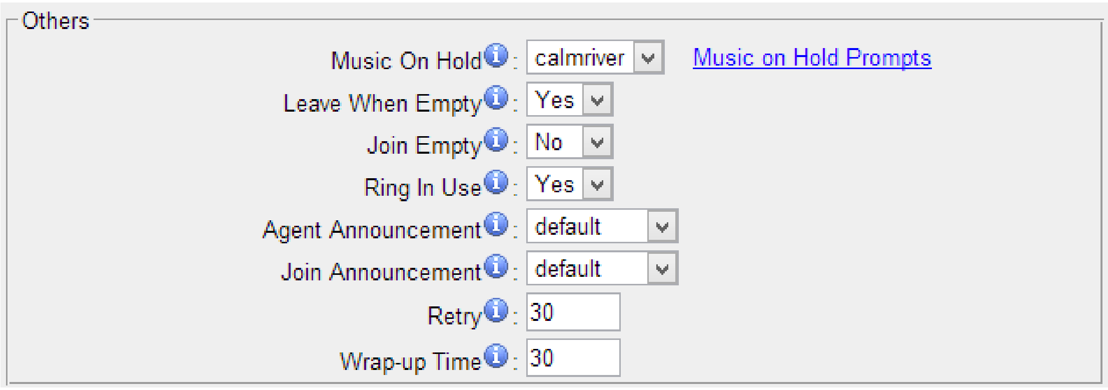


Figure 10-6 Queue Others Settings

Table 10-5 Queue Others Settings

Others	
Music on Hold	Select the "Music on Hold" Prompt for this queue.
Leave When Empty	This option controls whether callers already on hold are forced out of the queue that has no agents. • Yes: callers are forced out of a queue when no agents are logged in.

	• No: callers will remain in the queue without any agent.
Join Empty	This option controls whether callers can join a call queue that has no agents. • Yes: callers can join a call queue with no agents or only unavailable agents • No: callers cannot join a queue with no agents The default setting is "No".
Ring in Use	If set to "No", the queue will avoid sending calls to members whose devices are known to be "in use".
Agent Announcement	Announcement played to the agent prior to bringing in the caller.
Join Announcement	Announcement played to callers once prior to joining the queue.
Retry	The amount of seconds the queue waits after calling all available agents before calling them again.
Wrap-up Time	The amount of seconds the queue waits for passing another queue call to an agent who has completed a call (0 for no delay).

Conference

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

- Configure a Conference Room
- Join a Conference Room
- Manage the Conference

Configure a Conference Room

Go to **PBX→ Inbound Call Control→ Conferences** to configure conferences.

- Click Add Conference Room to add a new Conference Room.
- Click Delete the Selected Conference Rooms to delete the selected conference rooms.
- Click to edit one Conference Room.
- Click to delete one Conference Room.

Please check the Conference configuration parameters below.

Table 11-1 Conference Configuration Parameters

Options	Description
Extension	Use this number to dial into the conference room.
Admin	Admin can kick a user out and can lock the conference room.
PIN#	You can require callers to enter a password before they can enter this conference. This setting is optional.

Join a Conference Room

Users on N412 could dial the conference extension to join the conference room. If a password is set for the conference, users would be prompted to enter a PIN.

How to join a N412 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 N412. 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 could manage the conference by pressing * key on their phones to access IVR menu for conference room.
Please check the options for conference IVR below.

Table 11-2 Conference IVR Menu

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

Managing Voice on N412

In this chapter, we introduce how to manage voice on N412, including the following sections:

- System Prompt
- Custom Prompt
- Music on Hold

System Prompt

N412 ships with a US English prompt set by default. The system supports multiple languages. Users could update the system prompt in different ways. Go to **PBX → Audio Settings →System Prompts Settings** to update the system prompt.

HTTP/Auto Mode (Recommended)

Please make sure your N412 can access the internet before you update system prompt with this method.
Users could choose the desired prompts and click download to update directly without reboot.

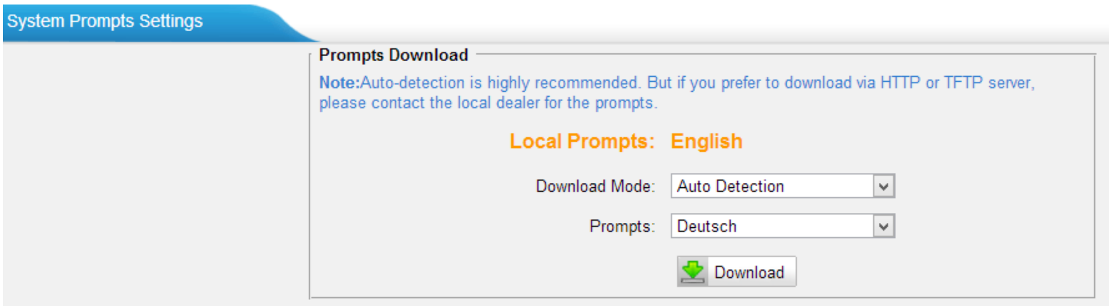


Figure 12-1 Update System Prompts- Auto Detection

Another way is choose Download Mode as "HTTP" and fill in the URL to download system prompt and update it.

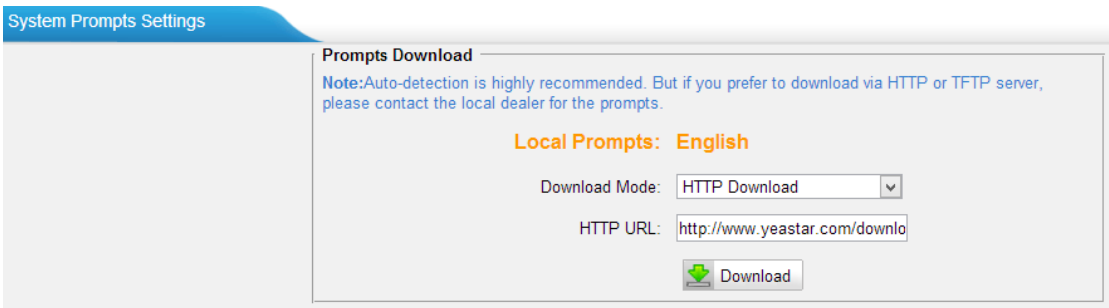


Figure 12-2 Update System Prompts- HTTP Download

Download link of system prompt is as below:

- America
- Arabic
- Australia
- British
- Chinese
- Danish
- Deutschland
- Dutch
- Finnish
- French
- French Canada
- Greek
- Hungarian
- Italian
- Korean
- Norwegian
- Persian
- Polish
- Portuguese
- Portuguese Brazil
- Russian
- Spanish
- Spanish Latin
- Spanish Mexico
- Swedish
- Thai
- Turkish

TFTP Method

If N412 cannot access the internet, please update the system prompts via TFTP.

Step1. Download the system prompt to your local PC.

Step2. Enable TFTP Server (For example, tftpd on Windows)

1) Install tftpd32 software on computer.
   Download link: http://tftpd32.jounin.net/tftpd32_download.html
2) Configure tftpd32
   For the option "Current Directory", click "Browse" button, choose the system prompt file of N412, such as D:\fr.tar.gz.

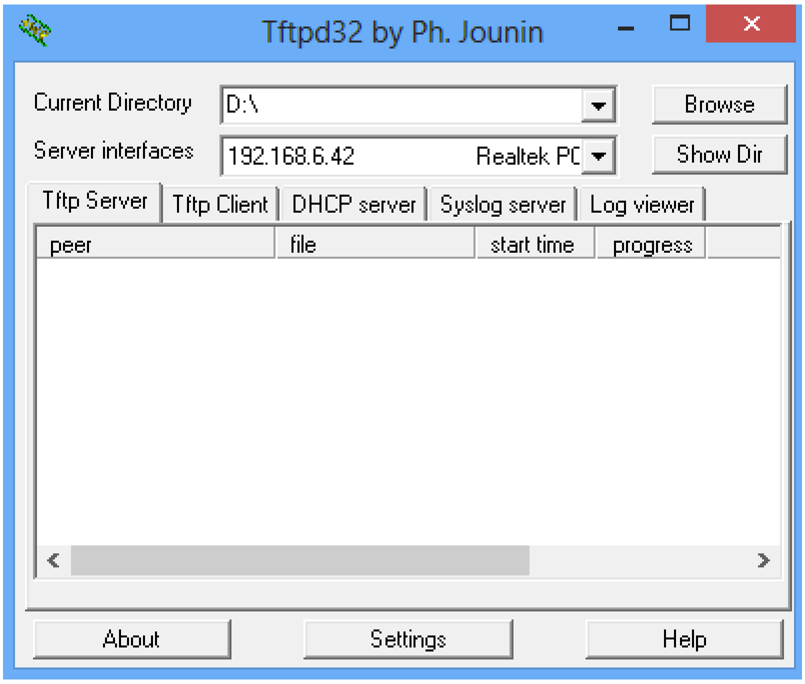


Figure 12-3 Configure Tftpd32

Step3. Update via TFTP

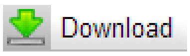
1) TFTP Server: fill in IP address of tftpd32 server, such as 192.168.6.42.
2) File Name: enter the name of voice prompt tar file name, such as "fr.tar.gz".
3) Click  to download the system prompt and update.

Figure 12-4 Update System Prompts- TFTP Download

Custom Prompt

The default voice prompts and announcements in N412 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 N412 and apply it to the place you want to change.

Custom Prompt can be found via the path **PBX→ Audio Settings → Custom Prompts**. On the Custom Prompt page, you can record a new prompt or upload your audio file to N412.

Record New Prompt

The administrator can record custom prompts by doing the following:

1) Click the button Record New Prompt.
2) Input the desired file name on the popup window and choose an extension to call for recording (such as 601).
3) Click Record. The selected extension will ring and you can pick up the phone to start recording.

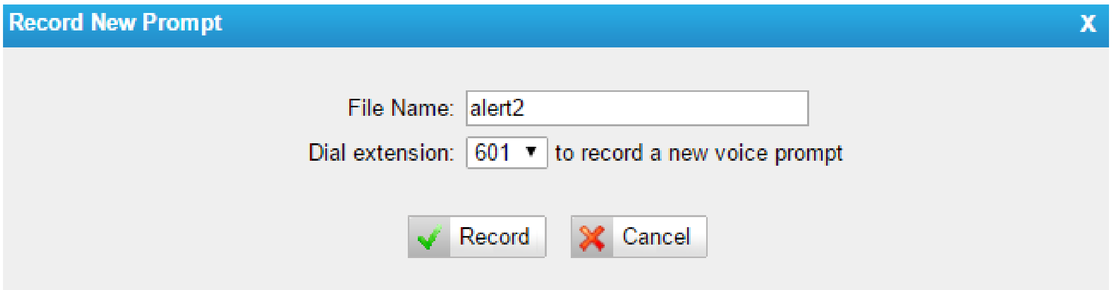


Figure 12-5 Record New Prompt

Upload Custom Prompt

1) Click the button Upload a Prompt.
2) Click Choose File to choose the desired prompt.

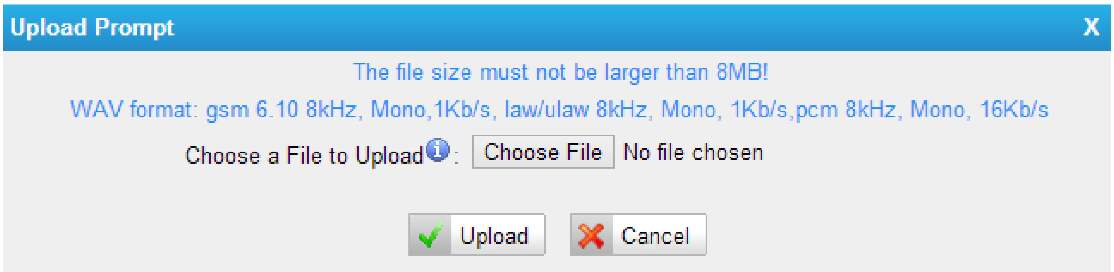


Figure 12-6 Upload Custom Prompt

3) Click Upload to upload the selected prompt.

Note:

The file size must not be larger than 8 MB, and the file must be WAV format:

- ✓ GSM 6.10 8 kHz, Mono, 1 Kb/s
- ✓ Alaw/Ulaw 8 kHz, Mono, 1 Kb/s
- ✓ PCM 8 kHz, Mono, 16 Kb/s

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. There are 3 default MOH files built in N412, you can also upload the one you want to N412.

Upload a Music on Hold Prompt

Upload a custom prompt via **PBX→ Audio Settings → Music on Hold Prompts**.

1) Click the button Upload Music on Hold Prompt.

2) Click Choose File to choose the desired prompt.

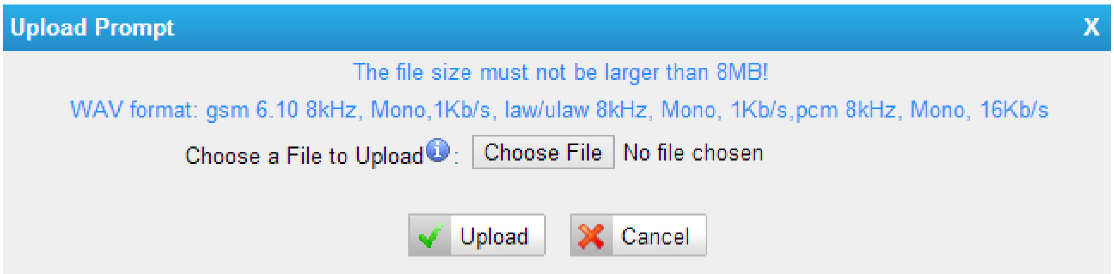


Figure 12-7 Upload Music on Hold File

3) Click Upload to upload the selected prompt.

Note:

The file size must not be larger than 8 MB, and the file must be WAV format:

✓ GSM 6.10 8 kHz, Mono, 1 Kb/s

✓ Alaw/Ulaw 8 kHz, Mono, 1 Kb/s

✓ PCM 8 kHz, Mono, 16 Kb/s

Play a Music on Hold Prompt

Choose a Music on Hold file via **PBX→ Audio Settings →Music on Hold Prompts** and click ▶ to play the prompt. Choose one extension to play the prompt. Once clicked the button Play, the selected extension will ring. Pick up the phone and listen to the music.

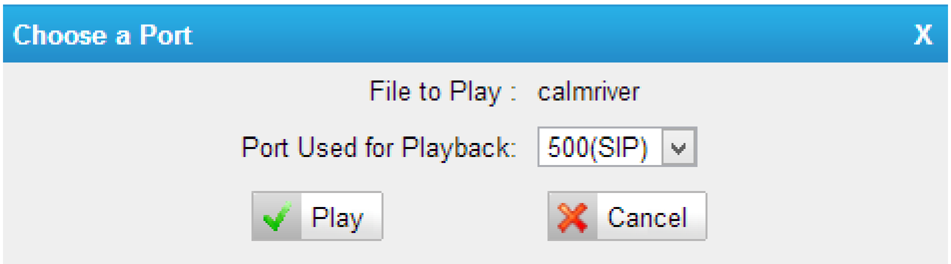


Figure 12-8 Play a Music on Hold Prompt

Voicemail

In this chapter, we introduce how to manage voicemail system on N412, including the following sections:

- Voicemail Settings
- Voicemail to Email
- How to Check Voicemail?
- How to Change Voicemail Greetings?

Voicemail Settings

Users could configure voicemail settings, including general voicemail settings and SMTP settings (which is used for “Voicemail to Email”) via **PBX**→ **Basic Settings**→ **Voicemail Settings**.

1) General Settings

Table 13-1 Voicemail- General Settings

General Voicemail Settings	
Max Message per Folder	Set the maximum number of messages that can be stored in a single voicemail box.
Max Message Time	Set the maximum length of a single voicemail message.
Min Message Time	Set the minimum length of a single voicemail message. Messages below this threshold will be automatically deleted.
Ask Caller to Dial 5	If this option is set, the caller will be prompted to press 5 before leaving a message.
Delete Voicemail	After notification, the voicemail is deleted from the server.
Operator Breakout from Voicemail	If this option is set, the caller can jump out of the voicemail and go to the destination you set by dialing “0”.
Destination	The caller will go to the destination by dialing “0”.

2) SMTP Settings

Please ensure the SMTP settings are configured correctly to make **Voicemail to Email** work properly.

After finishing the configuration, you can click on the Test SMTP Settings button to check whether the setup is OK.

- If the test is successful, you can use the email safely.

- If the test failed, please check if the above information is input correctly or if the network is OK.

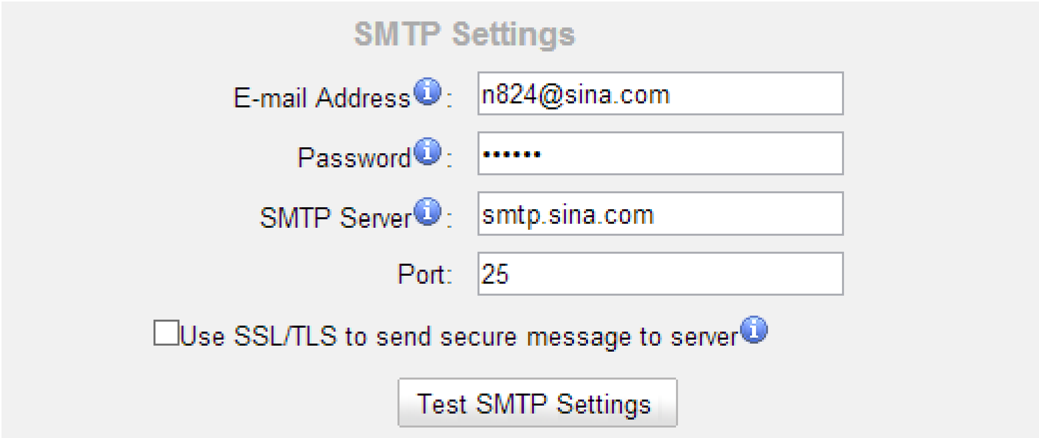


Figure 13-1 Voicemail-SMTP Settings

- **E-mail Address**
  The E-mail Address that N412 will use to send voicemail.
- **Password**
  The password for the email address used above.
- **SMTP Server**
  The IP address or hostname of an SMTP server that the N412 will connect in order to send voicemail messages via email.
- **Port**
  SMTP Port: the default value is 25.
- **Use SSL/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:
  SSL/TLS must be selected if you use Gmail or Exchange Server.

Voicemail to Email

Voicemail is enabled for each extension on N412 by default. If there is no answer for an extension, the call will be forwarded to the extension's voicemail. Email notification of voicemails are supported on N412, simply enable this feature on the desired extension edit page. Enter your email address in the Email Address field, the received voicemails will be sent to your email.

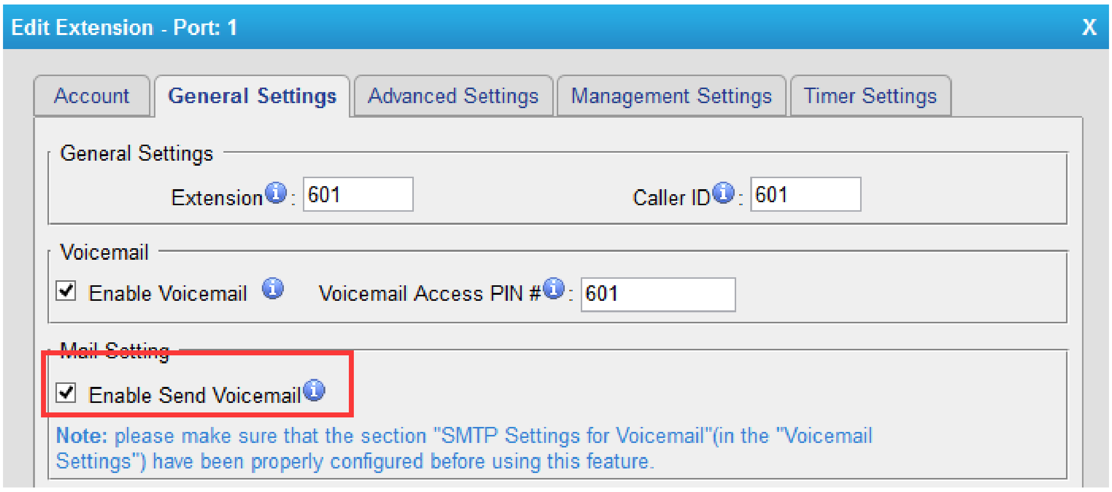


Figure 13-2 Enable Voicemail to Email

Note:

Please ensure that the section of "SMTP Settings for Voicemail" (in the "Voicemail Settings") has been properly configured before using this feature.

How to Check Voicemail?

There are multiple ways to check voicemail on N412. You can check the voicemail by pressing voicemail feature code on your phone or log in N412 by Extension account to check voicemails. In addition, you can check voicemail via Email if Voicemail to Email is enabled.

1) **Check Voicemail by Phones**

The default feature code to check a specific extension's voicemail is *2.

Dial *2 on your phone, and enter the voicemail PIN code to access your voicemail. The default voicemail PIN number is the same as your extension number. The password can be changed on the extension edit page.

You can also check other extension's voicemail on your own handset by using feature code *02. Dial *02 on your phone to enter the voicemail main menu. Entering the desired extension number and followed by the extension's voicemail PIN, you will be able to check the extension's voicemail.

2) **Check Voicemail on Web**

Another way to check voicemail is logging in N412 by Extension User Account.

Before logging in N412 Web using the extension User account, you should enable “User Web Interface” for the extension.

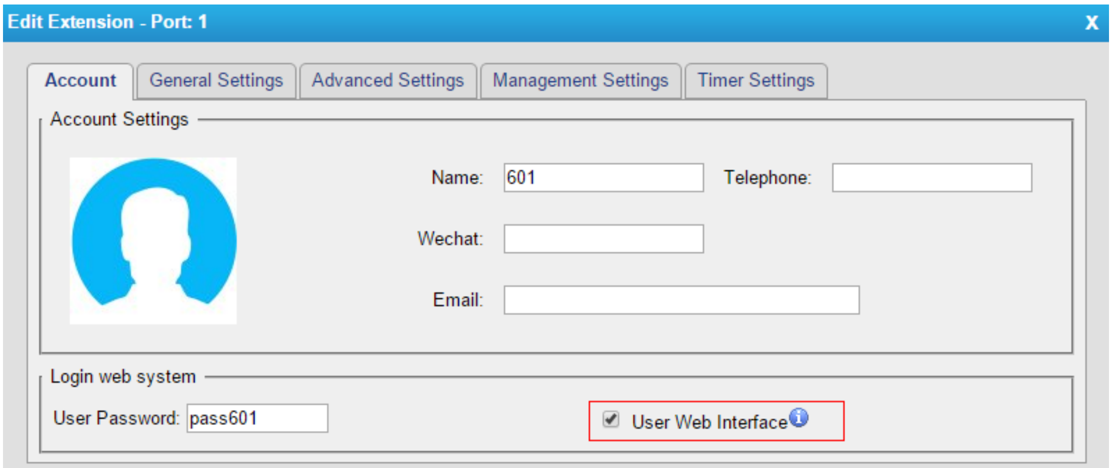


Figure 13-3 Enable User Interface

User Name: Extension Number
Password: User Password

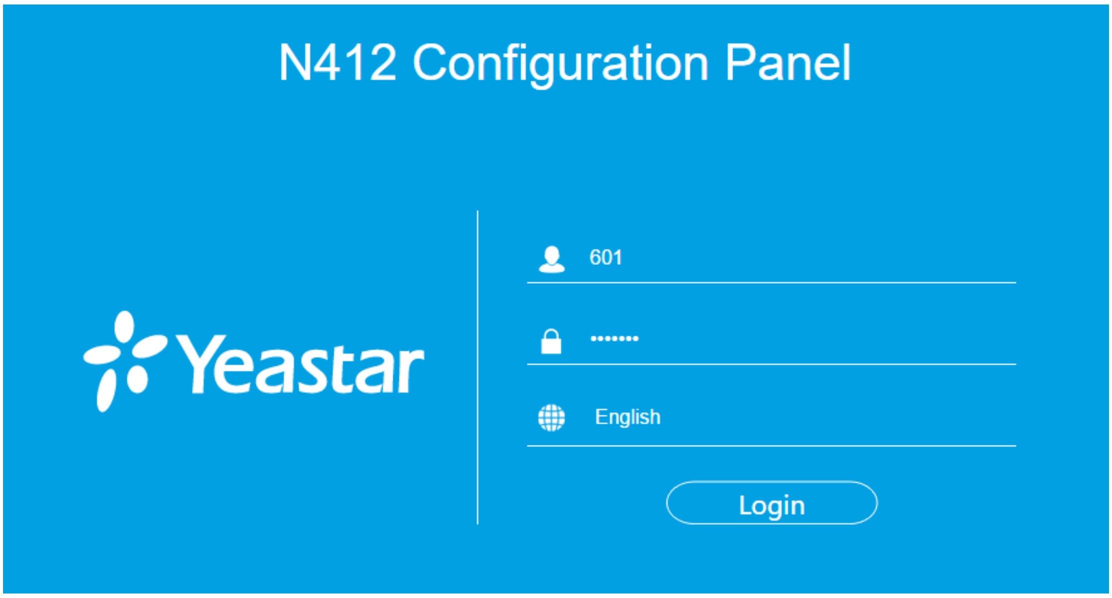


Figure 13-4 Login N412 via Extension Account

After login, you can check voicemail on "Voicemail" page.

3) **Check Voicemail via Email**
If you have enabled "Voicemail to Email" feature, you can check voicemail on your email.

How to Change Voicemail Greetings?

The default Voicemail greeting on N412 is fine but it is rather bland and quite boring. You can customize your own voicemail greetings.

1. Dial *2 to enter voicemail on your handset.

2. Enter the access password.
3. Press 0 for Mailbox Options. You will then be given the choice what type of message you want to record.
4. Press 1 to record your Unavailable Message.
5. Press 2 to record your Busy Message.
6. Press 3 to record your name.
7. Press 4 to record your Temporary Message.
8. Choose the message that you want to record, press # to finish the record.
9. Press 1 to accept your message.
10. Press 2 to listen to your message.
11. Press 3 to re-record your message if you don't like the previous message.

Business Calling Features

This chapter shows various call features on N412:

- Feature Code
- Call Transfer
- Call Pickup
- Intercom
- Spy
- Call Parking
- Speed Dialing

Feature Code

Feature Codes are used to enable and disable certain features available in N412. N412 local users can dial feature codes on their phones to use a particular feature.
The default feature codes can be checked and changed on **PBX→ Basic Settings→ Feature Codes** page.

1) General

Table14-1 Feature Code-General

General	
One Touch Record	Default Code: *1. A user may initiate or stop call recording by dialing the code during a call.
Check Extension Voicemail	Default Code: *2. Users could check their own voicemails by this code.
Voicemail for Extension	Default Code: #. Users can leave a voicemail to other extensions by dialing # on their phone or the incoming call could be forwarded to an extension's voicemail directly. (# is the default setting). For example, extension 500 want to leave a message for extension 501, users can use 500 dial "#501" to enter the voicemail of 501.
Voicemail Main Menu	Default Code: *02.
Attended Transfer	Default Code: *3. Attended Transfer Timeout: The timeout value of transferring a call.
Blind Transfer	Default Code: *03.
Call Pickup	Default Code: *4.

Extension Pickup	Default Code: *04. Users may pick up a specific extension's incoming call by dialing *04+extension number on their phone.
Intercom	Default Code: *5.
Normal Spy	Default Code: *90. In this mode, you can only listen to the extension being spied.
Whisper Spy	Default Code: *91. In this mode you can listen/whisper to the extension being spied.
Barge Spy	Default Code: *92. In this mode, you can barge in both extensions involved in the call.
Input Digit Timeout	Default: 4000ms. The timeout to input the next digit.

2) Call Parking Preferences

Table 14-2 Call Parking Preferences

Call Parking Preferences	
Call Parking	Default Code: *6.
Extension range used to park calls	Default: 690-699. User may park an incoming call on a designated extension at first and then pick up the call again on any other extensions.
Number of seconds a call can be parked for	Default: 60s. Define the time (in seconds) that a call can be parked before it is recalled to the station that parked it.

3) Call Forwarding Preferences

Table 14-3 Call Forwarding Preferences

Call Forwarding Preferences	
Reset to Defaults	Default Code: *70. The call forwarding settings will be configured as follows: • Always forward: Disabled • Busy forward to Voicemail: Enabled • No answer forward to Voicemail: Enabled • Do not disturb: Disabled
Enable Do Not Disturb	Default Code: *75.
Disable Do Not Disturb	Default Code: *075.

Call Transfer

There are 2 types of call transfers available on N412: Blind Transfer and Attended Transfer. Users can achieve call transfer by pressing the feature code during the call.

Blind Transfer

Default feature code: *03
1. Dial "*03" during the call;
2. Dial the called number after hearing a prompt "transfer";
3. The call will be transferred after the number is dialed.

Attended Transfer

Default feature code: *3
1. Dial "*3" during the call;
2. Dial the called number after hearing a prompt "transfer";
3. Talk to the transfer recipient;
4. The call will be transferred after hanging up.

On **PBX→ Basic Settings→ General Preferences** page, you can set the **Attended Transfer Caller ID**. The default display is the Caller ID of the initiator.
For example, if extension 500 makes a call to extension 501. After 501 picks up the call, user501 makes an attended transfer to extension 502.

- If selecting “Transferer”, 502 will display the Caller ID as 501;
- If selecting “Transferee”, 502 will display the Caller ID as 500.

Call Pickup

Call Pickup is a feature that allows one to answer someone else's call. The feature is accessed by pressing call pickup feature code on N412. If a colleague's phone set is ringing, one can answer that call by picking up one's own set and then using the call pick-up feature, instead of walking to the colleague’s desk.

Group Call Pickup

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.
Pickup group can be set on extension edit page. Extensions that are in the same group can pick up each other's call by feature code *4.

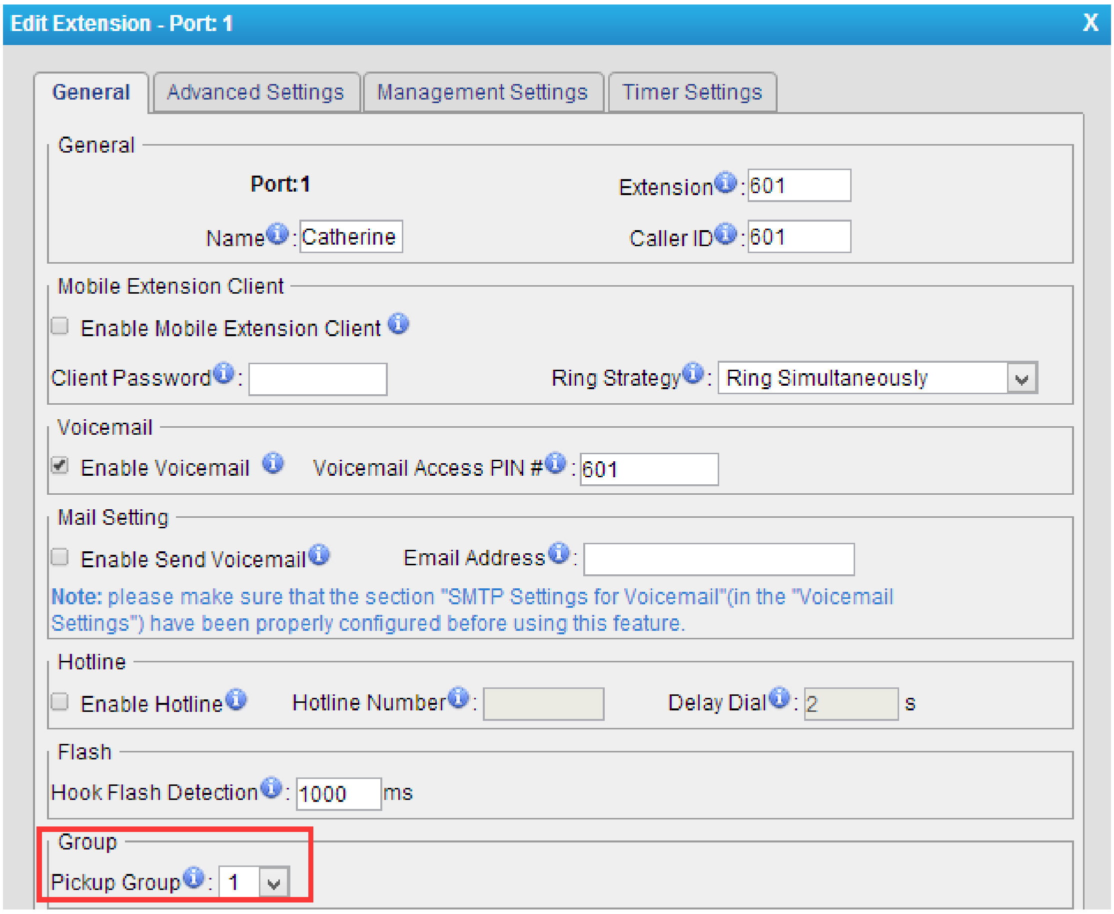


Figure 14-1 Group Extensions

Direct Call Pick

The default Direct Call Pickup (Extension Pick up) feature code is *04. It allows you to pick up a call that is made to a specific extension. If you know whose phone is ringing and what is the extension number is, you can pick up the call by pressing *04+ extension number.

For example, if a call reaches the Sales Department Manager's phone (extension number 888), but he is in a meeting, you can pick up the call by pressing *04888 on your own phone to answer the call.

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.

Spy

N412 allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode.

Spy Modes

- **General spy:** you have the permission to use the following 3 modes.
- **Normal spy:** you can only hear the call, but can't talk. Feature code: *90.
- **Whisper spy:** you can hear the call, and can talk with the monitored extension. Feature code: *91.
- **Barge spy:** you can hear the call and talk with them both. Feature code: *92.

Steps to Use Spy Feature

Example: Use Extension100 to monitor the calls of Extension 101**.**

1. Enable "Allow Being Spied" in extension 101. In this case, extension 101 is allowed to be spied by other extensions.
2. Choose the "Spy Modes" for extension 100. In this case, extension 100 has the right to use the feature code to monitor extension 101.
3. If 100 choose “normal spy”, it should dial “*90301” to start monitoring;
   If 100 choose “whisper spy”, it should dial “*91301” to start monitoring;
   If 100 choose “barge spy”, it should dial “*92301” to start monitor;
   If 100 choose “general spy”', it can dial “*90301”, “*91301” or “*92301” to start monitor.

Call Parking

Call Parking is a feature that allows the user to put a call on hold at one phone and continue the conversation from any other phone. Call parking is activated by feature code. For example, extension 8010 is in a call, but the person needs to go to another place to find the answer for a question. He can dial Call Parking feature code on the phone, and system will prompt that the call is parked at an extension, i.e. 690.Then this person can hang up the call and leave. When he finds the information, he can pick up any phone nearby and dial 690 to resume the conversation.

Uses of Call Parking

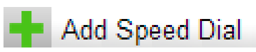
Call parking is often useful in buildings with many offices or with more than one floor, and with most of the areas having access to one or more telephone sets.

- If the desired called party is not the person who picked up the call, and the desired called party is at an unknown location, the person who picked up the call may park the call and then use the public address system to page the desired called party to pick up the call.
- During a conversation, a person may need to go to another office for some reason (for example, to retrieve an important file); parking the call allows this person to continue the conversation after arriving at the other office.

Speed Dialing

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. Speed Dial feature is available on N412 that allowing you to place a call by pressing a reduced number of keys.

Add a Speed Dial

1. Go to **PBX→ Outbound Call Control→ Speed Dial Settings**, you can see the default Speed Dial Prefix is *99. Please avoid conflict with other feature codes if you want to change the prefix.
2. Click 

to add one Speed Dial.
3. Fill in the Source Number and Destination Number.
   Number for the number you want to call.
   Speed Dial Code for speed dialing number.

Note:
Do not forget to add the outbound dial prefix if you would like to dial the speed dial number through trunk.

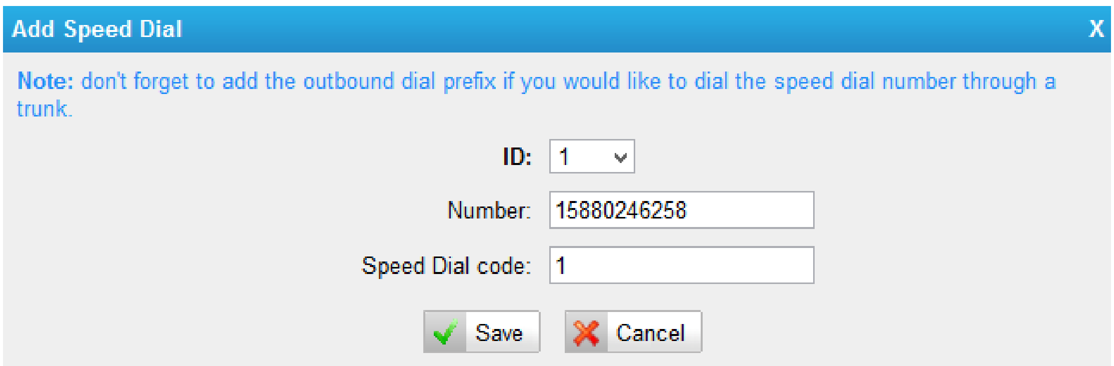


Figure 14-2 Speed Dial

To make a speed dial, e.g. you want to call 15880246258, simply dial *991. The *99 tells N412 that you want to use the Speed Dial and the 1 is the Speed Dial Code for destination number 15880246258. (Check the Speed Dial Setting for 15880246258 on the screen above.)

Auto Recording

This chapter explains how to configure auto recording on N412 and how to manage the recording files.

- Auto Recording Settings
- TF Card Management
- Store Recordings to Network Disk
- Share Recordings

Auto Recording Settings

N412 supports auto recording for an established call. Go to **System→ Auto Recording Settings→ Recording Settings** to configure auto recording settings.

Note: before enabling call recording, please make sure that the TF card or the Net Disk performs well.

Table 15-1 Auto Recording Configurations

General Preferences	
Enable Call Recording	Enable or Disable Auto Recording feature. You can choose: ✓ Record Inbound calls ✓ Record Outbound calls ✓ Record Internal calls
Storage Location	After enabling Call recording, you can choose where to store the recordings, in the Network Disk or in the TF card. • Status: shows if the TF card or Network Disk is successfully mounted to N412.
Record Inbound Prompt	If Call Recording is enabled, the caller will hear the pre-configured prompt when the inbound calls go through trunks. • Written to the Recording file: If enabled, the prompt will be recorded.
Record Outbound Prompt	If Call Recording is enabled, the callee will hear the pre-configured prompt when the outbound calls go through trunks. • Written to the Recording file: If enabled, the prompt will be recorded.
Apply To	
Record Trunks	When ticked, all calls through the selected trunks will be

	recorded.
Record Extensions	When ticked, all calls made by the selected extensions will be recorded.
Record Conferences	When ticked, all conversations through the selected conferences will be recorded.

TF Card Management

Insert a TF Card to N412, then manage and check the TF check status under **System→ Storage Management→ TF Card Management**.
You can format or clean up the TF card on this page.

Note:

- You should power off your machine when you put in or pull out your TF card.
- If you want to clear up the TF card, please check if you need to back up the files in it.

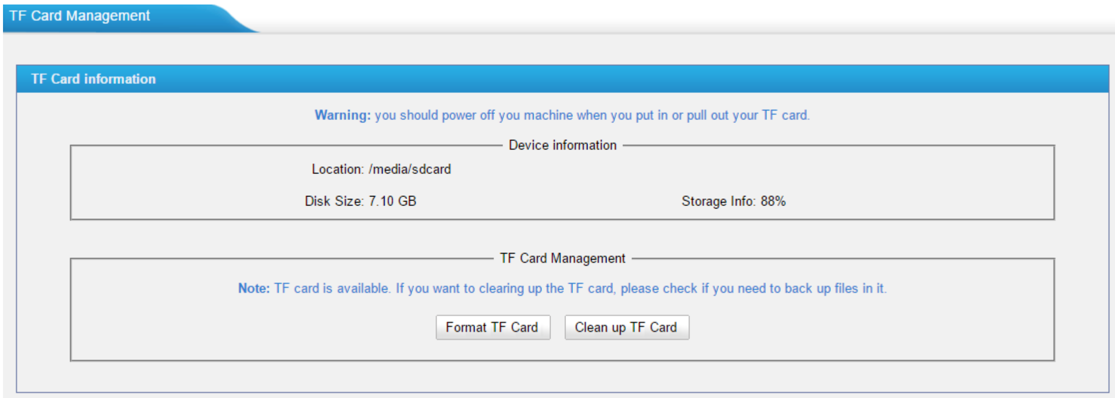


Figure 15-1 TF Card Management

Store Recordings to Network Disk

The Network Disk feature is used to extend storage space. If “Storage Location” is set to Network Disk, the call recording files created will be moved to the Network Disk. Configure the Network Disk under **System→ Storage Management→ Network Disk Settings**.

Note:

The shared folder must be based on Windows Operation System. And if it's windows Vista/2008/7, please add “Everyone” into the shared account list. After that you should ensure that the permission of “Everyone” is checked.

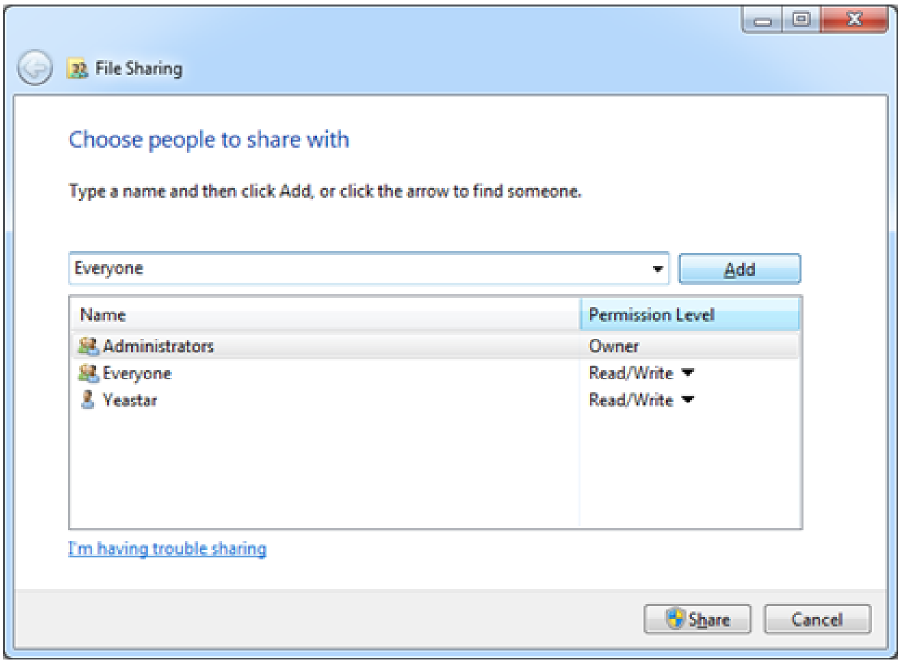


Figure 15-2 Add Everyone

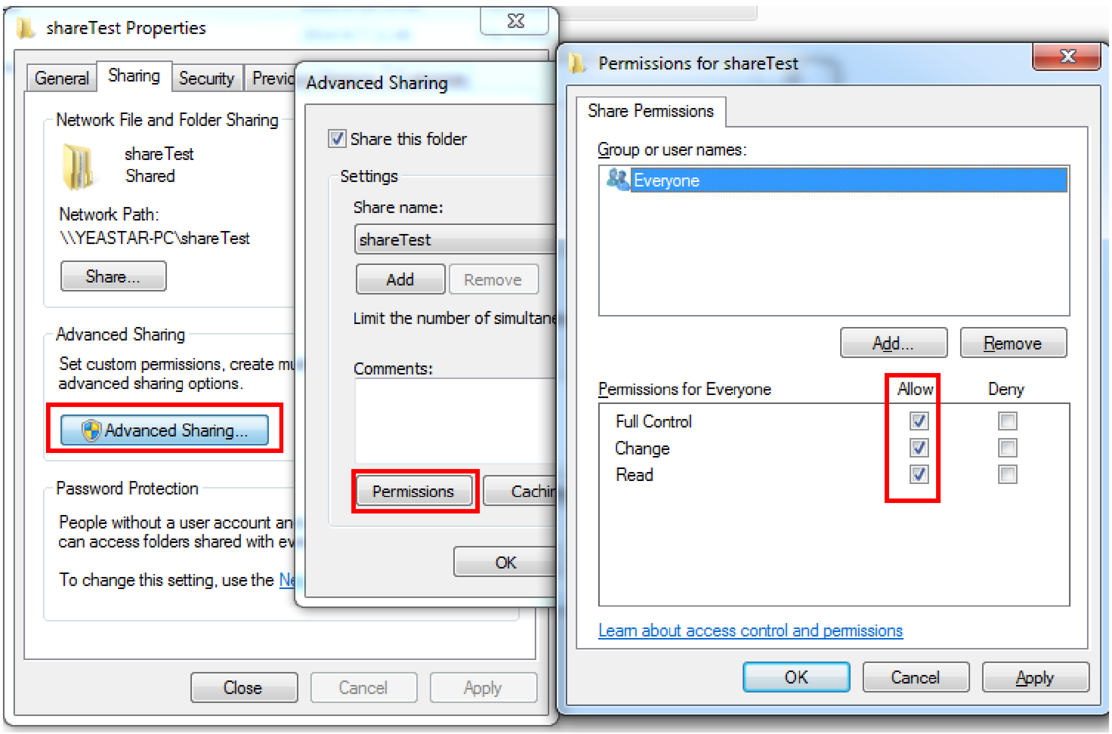


Figure 15-3 Share the File

Before network disk can be properly configured, an SMB share folder accessible from N412 must be set up on a Windows based machine. Once that has been set up, please follow the steps below.

1. Choose a window-based computer that is always in service
2. Create a folder
3. Share this folder

4. Input the Net-Disk information in “Connect to Network Disk” tab.

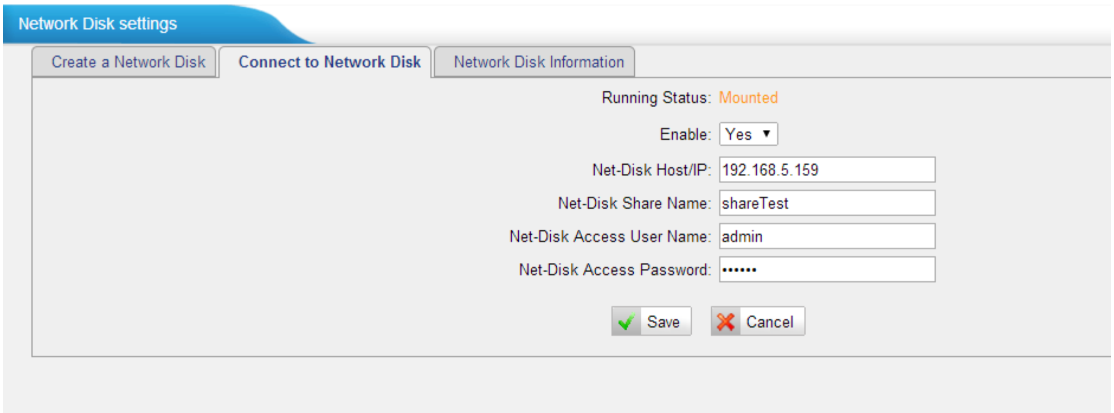


Figure 15-4 Connect to Network Disk

Table 15-2 Network Disk Settings

Network Disk Settings	
Running Status	Shows if the Net-Disk is successfully mounted to N412.
Enable	• Yes: Enable Network Disk • No: Disable Network Disk
Net-Disk Host/IP	Set the IP address where the recordings will be stored.
Net-Disk Share Name	The shared folder name where the recordings will be stored.
Net-Disk 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.
Net-Disk Access Password	The password used to log into the network share. Leave this blank if it is not required.

If the configuration is correct, you will see the Network Disk information.

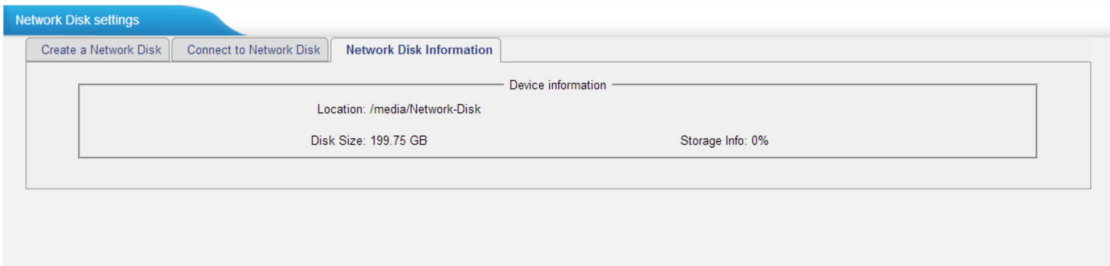


Figure 15-5 Network Disk Status

Share Recordings

Users could share recordings on the network. This setting is only applied to the Call Recording folder in the TF Card. Go to **System→ Storage Management→ Network Disk Settings** to configure the sharing settings.

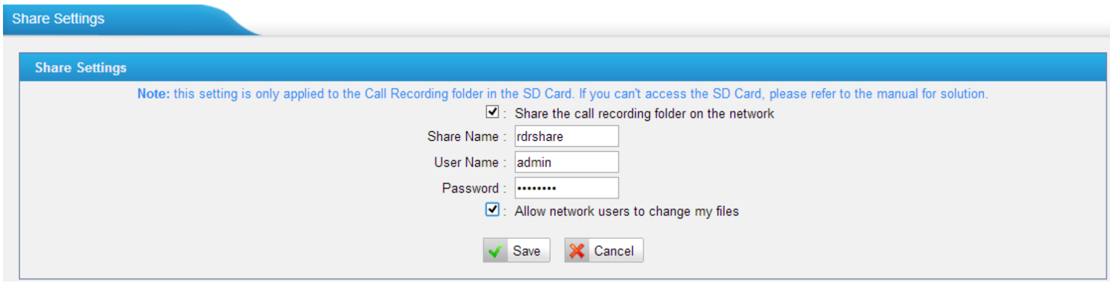


Figure 15-6 Share Recordings

Table 15-3 Sharing Settings

Sharing Settings	
Share Name	The name of the folder showed on the network.
User Name	The user name to access the files.
Password	The password for the user name.

Check Shared Recordings

Open a file folder on your PC, and type N412 IP address (\\IP Address) to check the shared recordings.

Example: \\192.168.5.149

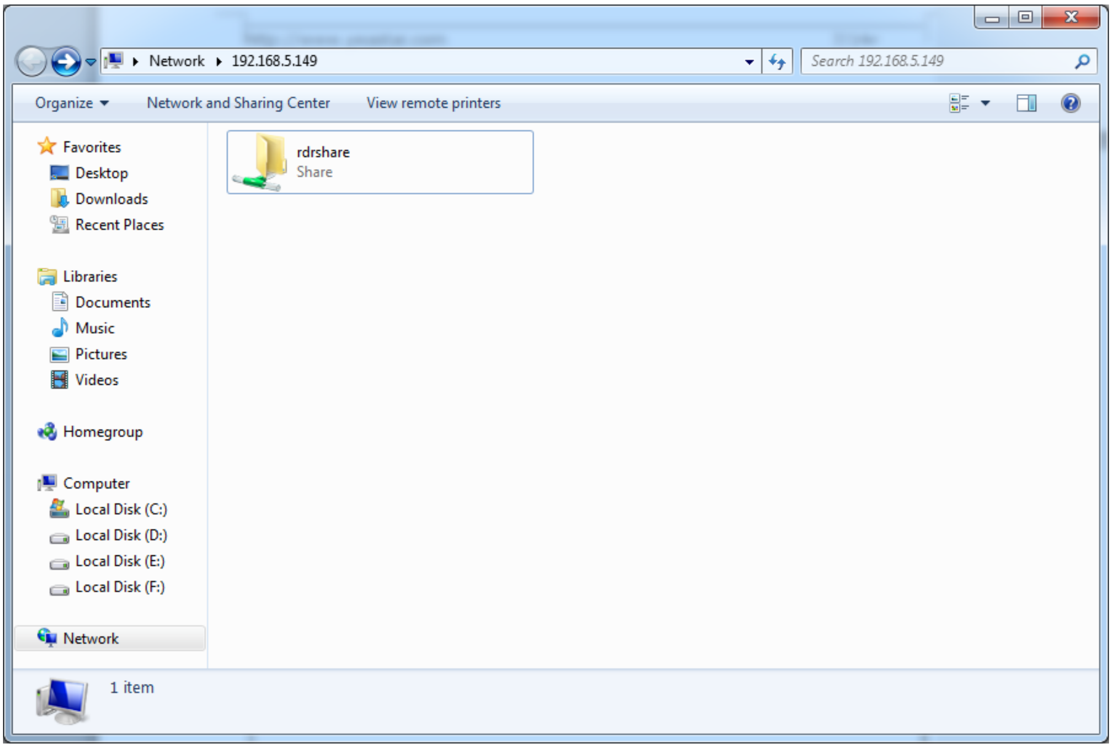


Figure 15-7 Check Shared Recordings

How to check the Network User?

1. Start-> Run ->cmd
2. Execute the command net use * /del (there is a space behind *)
3. Visit the device again with the new password

For WIN 7/Win8 Users

Please modify the registry before checking the shared file.

1. Start->Run->regedit
2. Modify the value of

HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Control\Lsa\
LmCompatibilityLevel to 1

If LmCompatibilityLevel doesn't exit, create a Dword value LmCompatibilityLevel= 1.

PBX Basic Settings

This chapter explains PBX basic settings, which can be applied globally to N412. The basic settings can be configured under **PBX→ Basic Settings**.

- General Preferences
- Business Hours

General Preferences

1) General Settings

Table 16-1 General Preferences-General

General Settings	
Max Call Duration	The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout. The default value is 6000s.
Music On Hold	Used to set hold music for the system.
Tone Region	Select country to set the default tones (dial tone, busy tone, ring tone and etc.) to be sent from FXS port. The default setting is United States/North America.
Dsp Fax	Enable Dsp to optimize Fax reception.
FXO Mode	Select country 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.
Attended Transfer Caller ID	When transferring an incoming call using the attended transfer feature code or the transfer key of IP phone, the Caller ID of transferee or transferer displayed on the screen of the callee. The default display is the Caller ID of the initiator.
Follow Me Prompt	If “Enable Follow Me Prompt” choosing yes, there will be prompt before transferring the call. Otherwise, the call will be transferred directly without any prompt. Default: Yes.
Music on hold for Follow Me	Configure whether to play a prompt “please hold while I try to locate the person you are calling” when transfer a call by follow me settings.
Invalid Phone Number Prompt	Configure the prompt when the dialed phone number is invalid.
Busy Line Prompt	Configure the prompt when the dialed phone number is busy.
Dial Failure Prompt	Configure the prompt when dial failed due to conjunction

	no-available channel.
Enable Last Caller Routing	Whether to enable the feature Last Caller Routing. When an extension is making an outbound call, the system will automatically record the information, and when the dialed number make an inbound call using the same line, this number will directly reach the corresponding internal extension.
Keep Time	How long you want to keep the Last Caller Routing records.
Internal Ring Type	Select the Ring tone type for internal calls.
Inbound Ring Type	Select the ring tone type for inbound calls.

2) Web Server

N412 supports web server responds to HTTP and HTTPS. By default, users could access the Web GUI via HTTP (default port: 80). You can also access web via HTTPS if HTTPS is enabled.

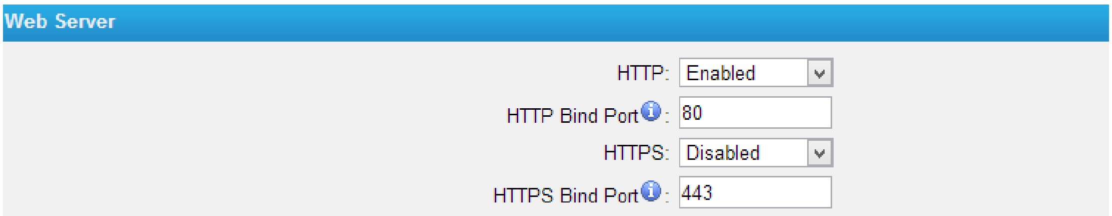


Figure 16-1 Web Sever

3) Extension Preferences

You can change extension preferences on this Section. There are 5 types of extension range, including User Extensions, Ring Group Extensions, Conference Extensions, IVR Extensions, and Queue Extensions. Assign a specific range for each type will help to distinguish and manage those different extensions.

You could change the default range or redefine it to meet your requirements. The extension number should have at least 2 digits and at most 7 digits.

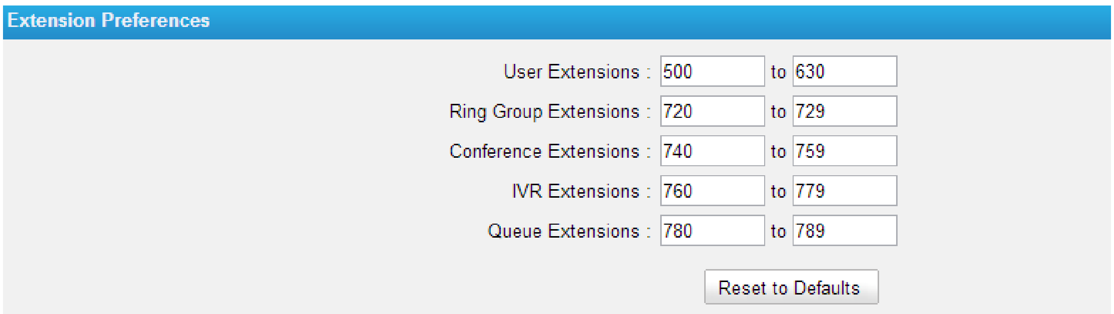


Figure 16-2 Extension Preferences

Business Hours

On Business Hours page, you can create a list of times (Office Hours, Other Office

Hours and Holiday) in which incoming or outgoing calls are checked. The rules specify a time range, by the hour and/or date. Business Hours typically are associated with time conditions, which match destinations for calls based on the time. Outbound routes can also be assigned an Office Hours, making that route only available during times defined in an Office Hours.
Go to **PBX→ Basic Settings→ Business Hours** to find Business Hours settings.

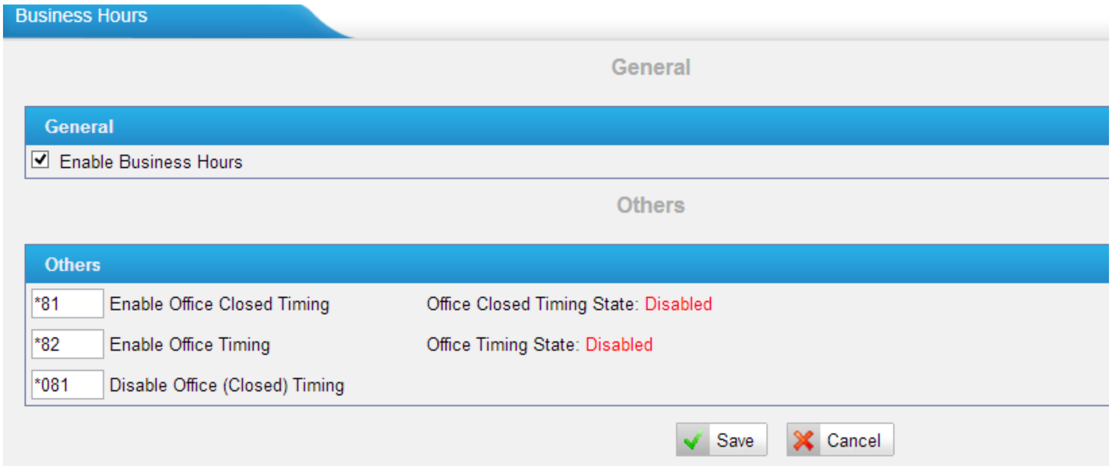


Figure 16-3 Business Hours

- **Enable Business Hours**
- **Enable Office Closed Timing**
  By dialing *81 (*81 is the default code) on an extension will force the office time closed for the device whatever the general setting is.
- **Enable Office Timing**
  By dialing *82 (*82 is the default code) on an extension will force the office time to take effect for the device whatever the general setting is.
- **Disable Office closed timing**
  By dialing *081 (*081 is the default code) on an extension will disable the Office Closed Timing.

Business Days

Generally, we add office hours according to your working time and set different destinations for Office Hours and Non-office Hours.
Define the Office Hours or Other Office Hours name and set the time as the following picture shows.

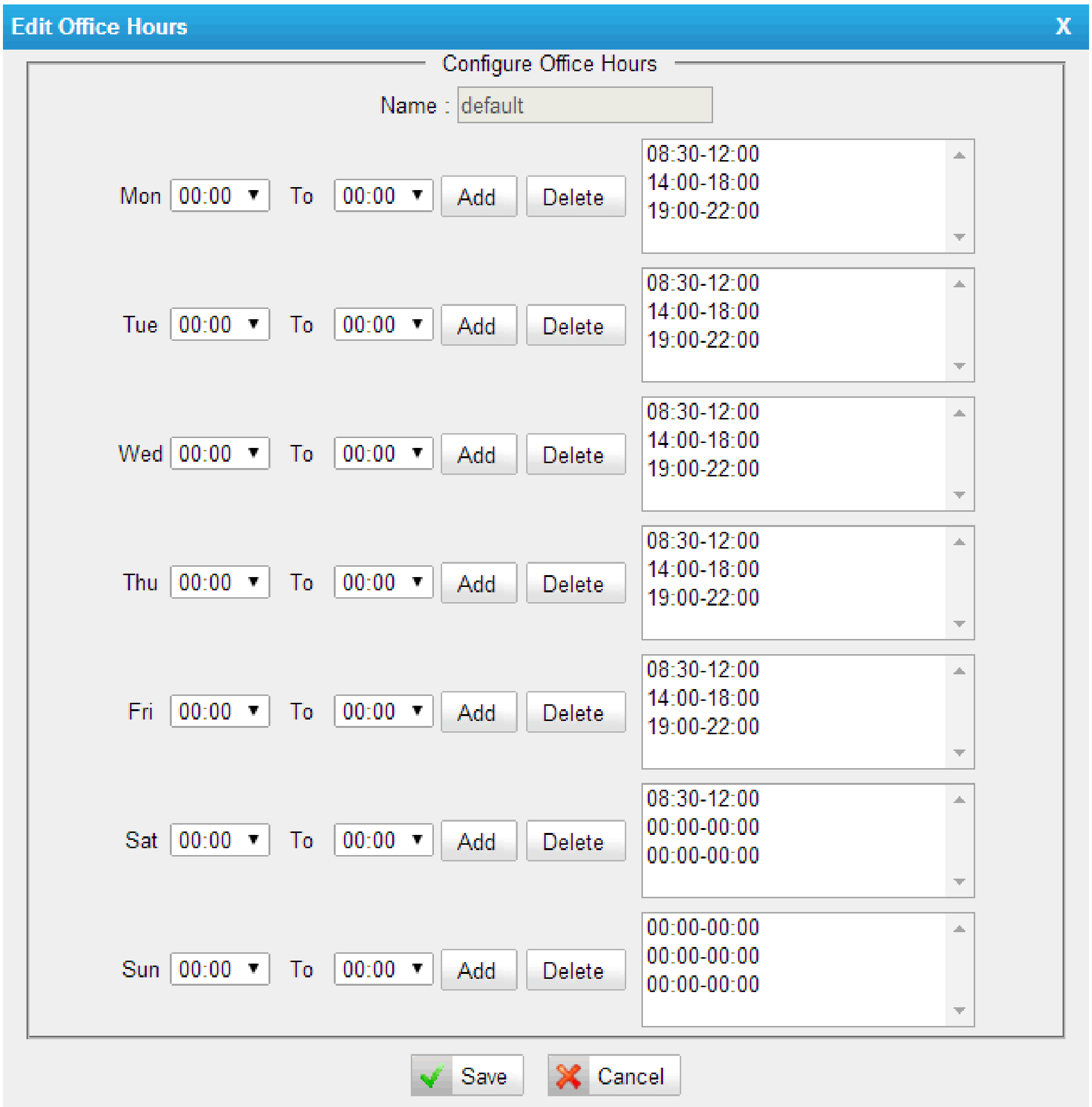


Figure 16-4 Business Days

Holidays

You can set up the holidays here.

If a time period is configured as both Holidays and office hours, it will be treated as Holidays.

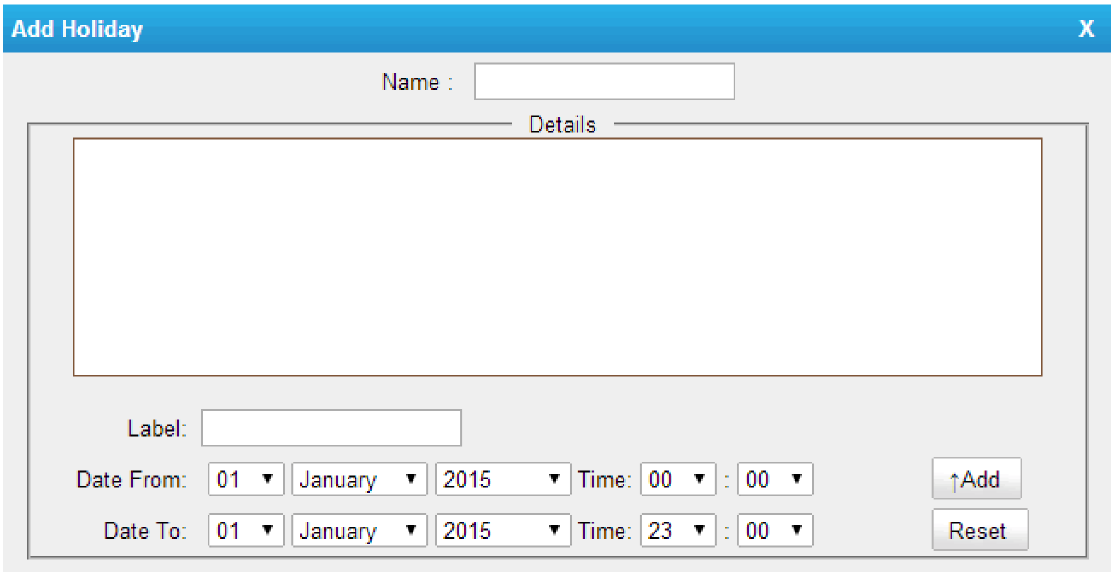


Figure 16-5 Holidays

SIP Settings

SIP settings can be configured on **PBX→ Advanced Settings →SIP Settings** page. 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
- NAT
- Codecs

General

Table 17-1 SIP Settings-General

General Settings	
UDP Port	Port used for SIP registrations. The default is 5060.
TCP Port	Port used for SIP registrations. The default is 5060.
TLS Port	Port used for SIP registrations. The default is 5060.
TLS Verify Server	When using N412 as a TLS client, whether or not to verify server's certificate. It is "No" by default.
TLS Verify Client	When using N412 as a TLS server, whether or not to verify client's certificate. It is "No" by default.
TLS Client Method	When using N412 as TLS client, specify the protocol for outbound TLS connections. You can select it as tlsv1, sslv2 or sslv3.
RTP Port Start/End	Set the RTP Port range.
DTMF Mode	Set default mode for sending DTMF. Default setting: rfc4733.
Min Registration/ Subscription Time	Minimum duration (in seconds) of a SIP registration. The default is 60 seconds.

NAT

Configuration of this section is only required when you use remote extensions.

Table 17-2 SIP Settings- NAT

NAT Settings	
Enable STUN	Whether to enable STUN.
STUN Address	STUN IP address.
STUN Port	STUN port.

External IP Address	The IP address that will be associated with outbound SIP messages if the system is in a NAT environment.
External Host	Alternatively you can specify an external host, and the system will perform DNS queries periodically. This setting is only required when your public 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 of this are as follows: “192.168.0.0/255.255.0.0”: all RFC 1918 addresses are local networks; “10.0.0.0/255.0.0.0”: also RFC1918; “172.16.0.0/12”: another RFC1918 with CIDR notation; “169.254.0.0/255.255.0.0”: zero conf local network. Please refer to RFC1918 for more information.
NAT Mode	Global NAT configuration for the system; the options for this setting are as follows: Yes = Use NAT. Ignore 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.
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.

Codecs

A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet. N412 supports G711 a-law, u-law, GSM, SPEEX, iLBC, G722, G726, G729A, G729B and G729AB,

Note:
If you would like to use G.729, please enter your license. Our device have embedded the G729, you can test it directly without purchasing license. But for copyright protection, we suggest you to buy it after testing it successfully. After you buy the

license from DIGIUM, you should enter G729 license at the "G729 License Key".

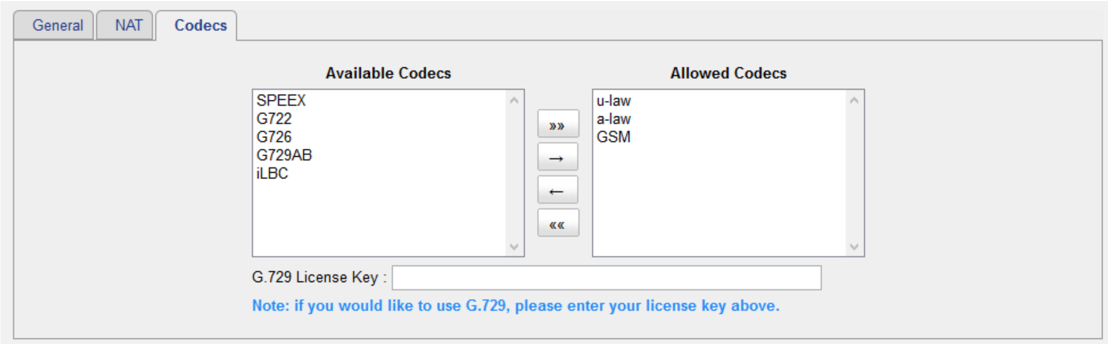


Figure 17-1 SIP Settings-Codecs

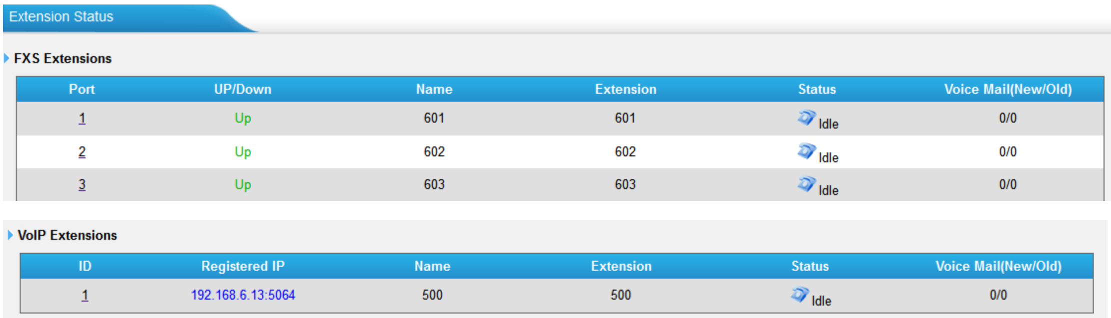
Status and Call Reports

Users could check the system status on **Status→ System Status**, where Extension Status, Trunk Status, Network Status and System Info can be checked. CDR and Call Recordings can be checked under **Status→ Reports.**

- Extension Status
- Trunk Status
- Network Status
- System Info
- Call Logs
- Record Logs

Extension Status

Users could view all the extension status on this page.


Extension Status

FXS Extensions

Port	UP/Down	Name	Extension	Status	Voice Mail(New/Old)
1	Up	601	601	Idle	0/0
2	Up	602	602	Idle	0/0
3	Up	603	603	Idle	0/0

VoIP Extensions

ID	Registered IP	Name	Extension	Status	Voice Mail(New/Old)
1	192.168.6.13:5064	500	500	Idle	0/0

Figure 18-1 Extension Status

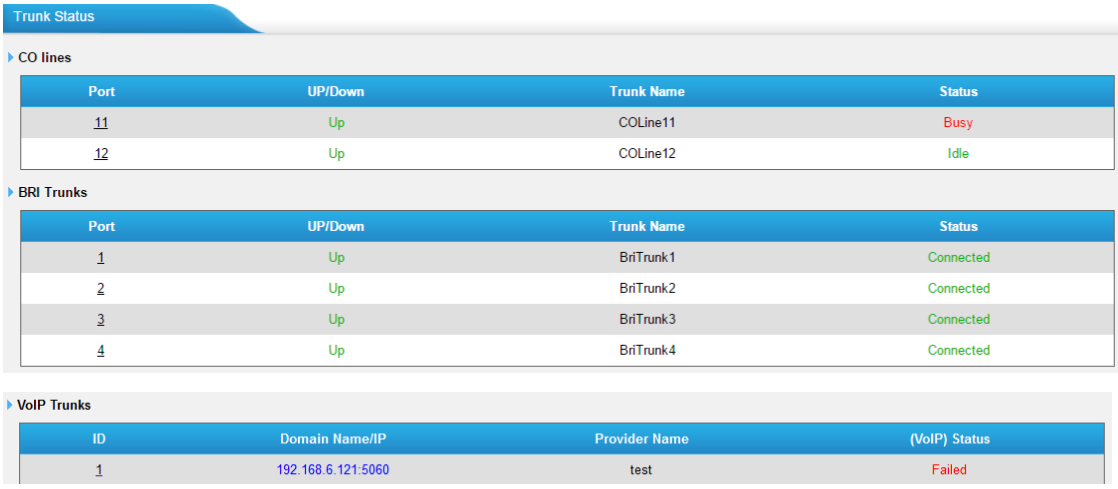
Table 18-1 Extension Status

Up/Down	Status for FXS Extensions: • Up: the FXS module works well. • Down: the FXS module is broken.
Registered IP	Status for SIP Extensions: • Unregistered: The SIP extension is not registered. • [IP]:[Port]: The SIP is successfully registered with the IP. Example: 192.168.6.142:50113 Description: The extension is registered on IP 192.168.6.142.
Name	Display the extension name.
Extension	Display the extension number.
Status	Monitor the extension’s call status in real time. •  Extension is unavailable

	• Extension is idle • Extension is ringing • Extension is busy • Extension is on hold
Voicemail	Display message status of the extension. Format: New/Old Example: 1/3 Description: There are 1 new voice message, and 3 old messages.

Trunk Status

Users could check all the CO trunks status and VoIP trunk status if VoIP trunk is created.



Trunk Status

CO lines

Port	UP/Down	Trunk Name	Status
11	Up	COLine11	Busy
12	Up	COLine12	Idle

BRI Trunks

Port	UP/Down	Trunk Name	Status
1	Up	BriTrunk1	Connected
2	Up	BriTrunk2	Connected
3	Up	BriTrunk3	Connected
4	Up	BriTrunk4	Connected

VoIP Trunks

ID	Domain Name/IP	Provider Name	(VoIP) Status
1	192.168.6.121:5060	test	Failed

Figure 18-2 Trunk Status

Table 18-2 VoIP Trunk Status

Domain Name/IP	Display the domain name/IP for the VoIP trunk. • [IP]:[Port]: display the IP address of the VoIP service provider and the SIP registry port. Example: 110.92.83.4:5060 Description: the extension is registered on IP 192.168.6.142 and the SIP port is 5060.
Provider Name	Display the trunk name.
(VoIP) Status	VoIP trunk status: • Registered: successful registration, trunk is ready for use. • Unreachable: cannot reach the VoIP service provider.

	• Failed: trunk registry failed.

Table 18-3 CO Line Status

Up/Down	Status for CO lines: • Up: the CO line works well. • Down: the CO line is broken.
Trunk Name	Display the trunk name.
(FXO) Status	PSTN trunk status: • Idle: the port is idle. • Busy: the port is in use. • Disconnected: there is no line connected to the port.

Table 18-4 BRI Trunk Status

Status	BRI trunk status: • Idle: the port is idle. • Disconnected: there is no line connected to the port.
Trunk Name	Display the trunk name.
Type	Display the trunk type: BRI.
Port	Display the relevant physical port of the trunk.

Network Status

Users could check the network status under **Status→ System Status→ Network Status**.

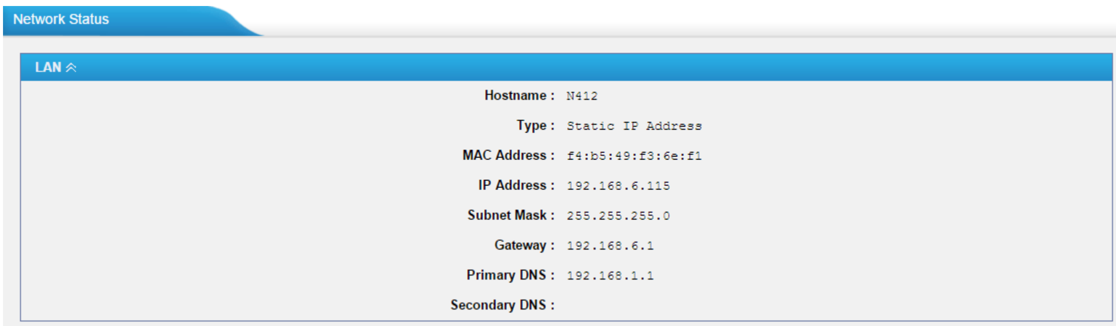


Figure 18-3 Network Status

System Info

The system info: product type, hardware version, firmware version, disk usage and memory usage can be viewed under **Status→ System Status→ System Info**.

Figure 18-4 System Info

Call Logs

The call Log captures all call details, including call time, caller number, callee number, call type, call duration, etc. An administrator can search and filter call data by filter the call logs by call date, caller/callee, trunk, duration, billing duration, status, communication type.

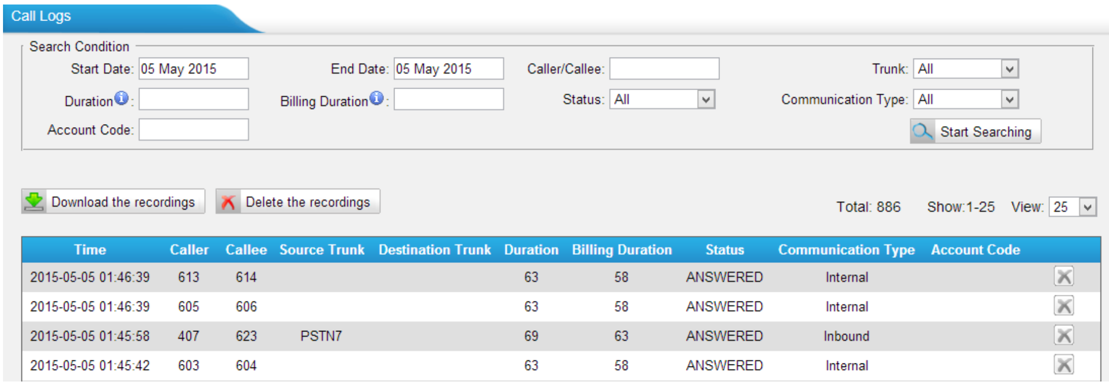


Figure 18-5 Call Log List

- **Search**
  The administrator can search and filter call data by specifying the call date, caller/callee, trunk, duration, billing duration, status, communication type.
- **Delete**
  Click to delete the chosen record.
- **Download Searched Results**
  Click Download the recordings to export the filtered records to a .csv file.
- **Delete Searched Results**
  Click Delete the recordings to delete the filtered records.

Record Logs

Under **Status→System Status→Reports→Record Logs**, users could check all the auto recording logs. Record logs are composed of Call time, caller, callee, trunk, Duration and Communication Type.

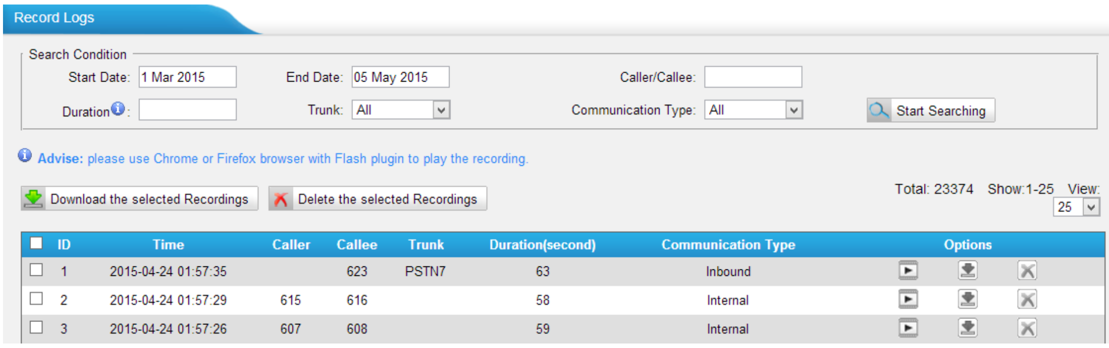


Figure 18-6 Record Logs

- **Search**
  The administrator can search and filter record data by specifying the call date, caller/callee, trunk, duration, and communication type.
- **Play**
  Click to play the recording file.
- **Download**
  Click to download the chosen recording file.
- **Delete**
  Click to delete the chosen record.
- **Download the Selected Recordings**
  Click Download the selected Recordings to download the selected recording files.
- **Delete the Selected Recordings**
  Click Delete the selected Recordings to delete the selected recording files.

System Maintenance

This chapter describes system maintenance settings including the followings:

- Firmware Update
- Backup and Restore
- Reset and Reboot
- System Logs
- Packet Tool

Firmware Update

N412 provides automatic updates, new firmware file will be checked via a cloud server. In addition, firmware upgrade can be done via HTTP and TFTP manually on N412. Please go to **System→ System Preferences→ Firmware Update** to do upgrade.

Note:

1. If "Reset configuration to Factory Defaults" is enabled, the system will restore to factory default settings.
2. When update the firmware, please don't turn off the power. Or the system will get damaged.
3. If you are trying to upgrade through HTTP, please make sure that your N412 is able to visit external network, or it cannot access Yeastar website to get the firmware file, causing the upgrade fail.

Automatic Updates

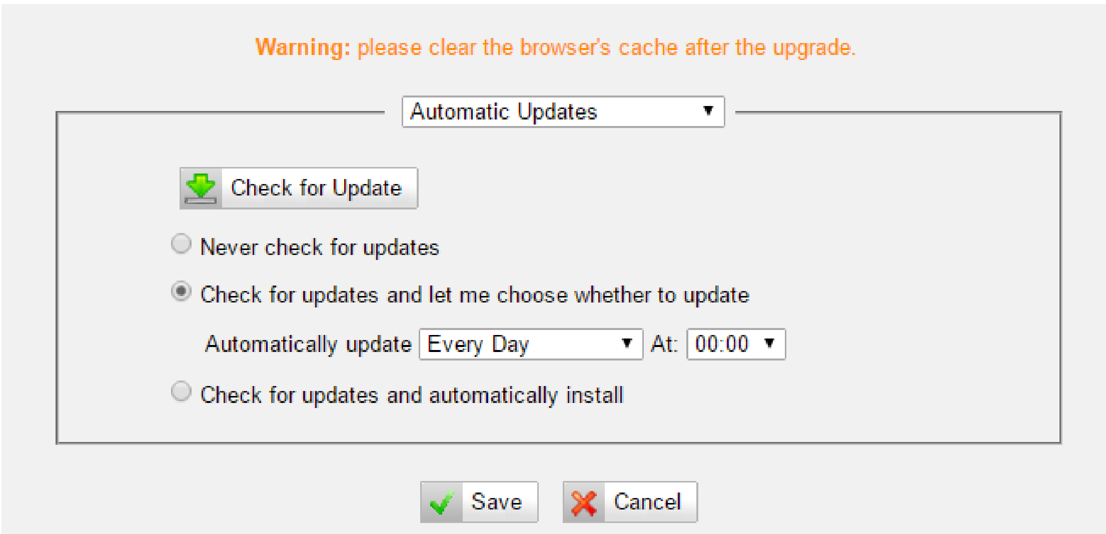


Figure 19-1 Automatic Updates

- **Never Check for Updates:** never check updates from the cloud server.
- **Check for Updates and let me choose whether to update:** set when to do check the updates automatically from the cloud server.
- **Check for updates and automatically install:** automatically downloads and installs firmware updates without even asking.

A notification will be send via a pre-configured email address for the send and third items of Automatic Updates.

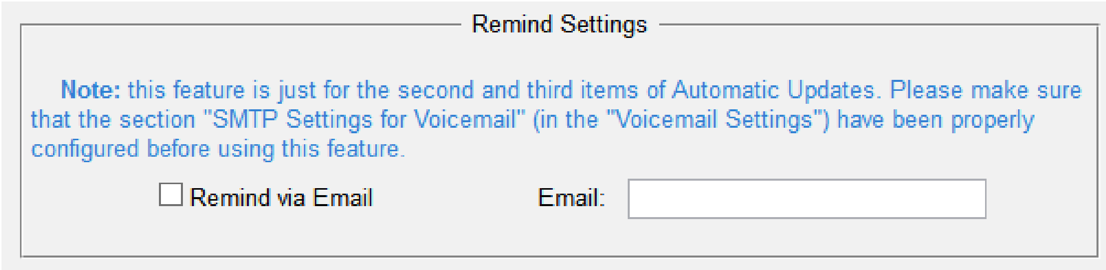


Figure 19-2 Email Notification for Automatic Updates

Upgrade through HTTP

On the Firmware Upgrade page, choose **HTTP URL**.
Step1. Enter the update image file download link.
Note: the HTTP URL should be a **BIN** file download link.
Step2. Click “Start” to upgrade.

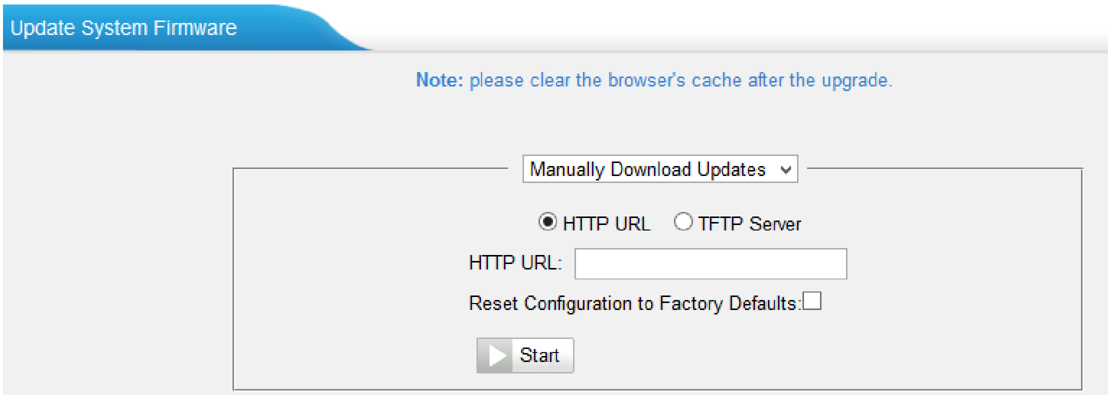


Figure 19-3 Upgrade through HTTP

Upgrade through TFTP

Step1. Download firmware file from Yeastar website.
Step2. Create a tftp Server (For example, tftpd on Windows).

1) Install tftpd32 software on computer.
   Download link: http://tftpd32.jounin.net/tftpd32_download.html
2) Configure tftpd32.
   On option “**Current Directory**”, click “**Browse**” button, choose the firmware file (BIN file) upgraded patch.

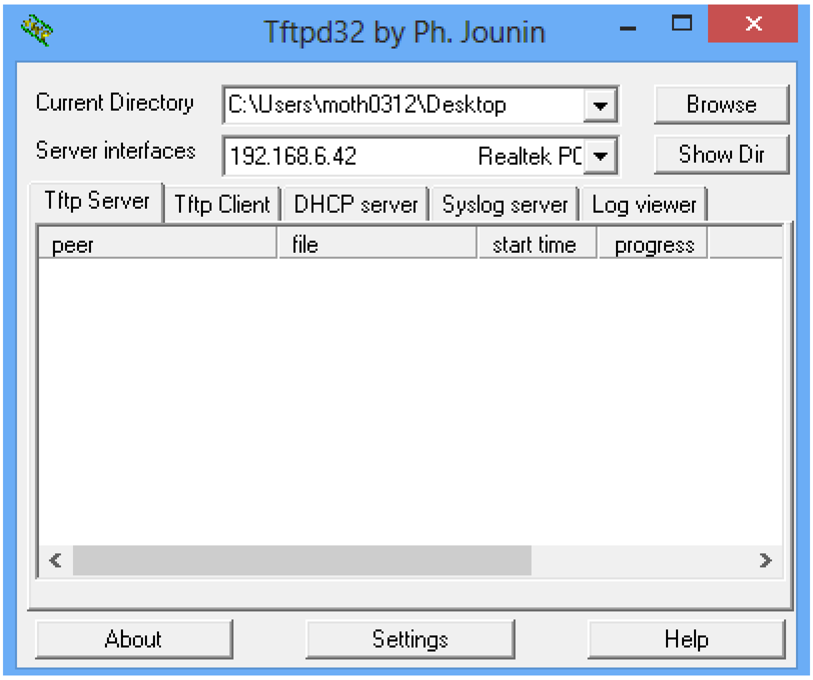


Figure 19-4 Configure Tftpd32

Step3. Logon the N412's Web page and **Firmware Update** page, choose "**TFTP Server**".

1) TFTP Server: fill in IP address of tftpd32 server (your PC's IP address).
2) File Name: enter the name of firmware update. It should be a BIN file name.
3) Click "Start" to upgrade.

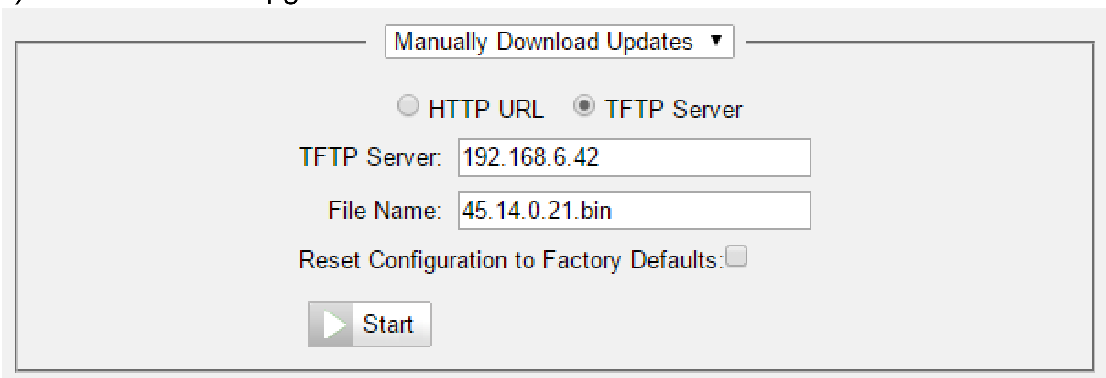


Figure 19-5 Upgrade through HTTP

Backup and Restore

N412 provides Backup and Restore feature, which allows you to create a complete backup of N412 configurations to a file.

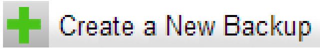
Notes:

1. The backup file only covers the configurations but not the CDR, voicemail and call recordings.
2. When you have updated the firmware version, it's not recommended to restore

using old package.

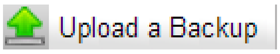
3. Backup from an earlier version cannot be restored on N412 of a later version.

- **Create a New Backup**

  Click 

  to create a new backup.

- **Upload a Backup**

  Click 

  to upload a backup.

- **Restore**

  To restore N412 configuration data, upload the backup file to N412 and click .

  Reboot the system to take effect.
  Please note the current configurations will be OVERWRITTEN with the backup data.

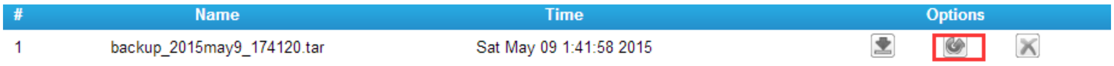


Figure 19-6 Restore Backup

Reset and Reboot

Users could reset and reboot the system under **System→ System Preferences→ Reset and Reboot**.

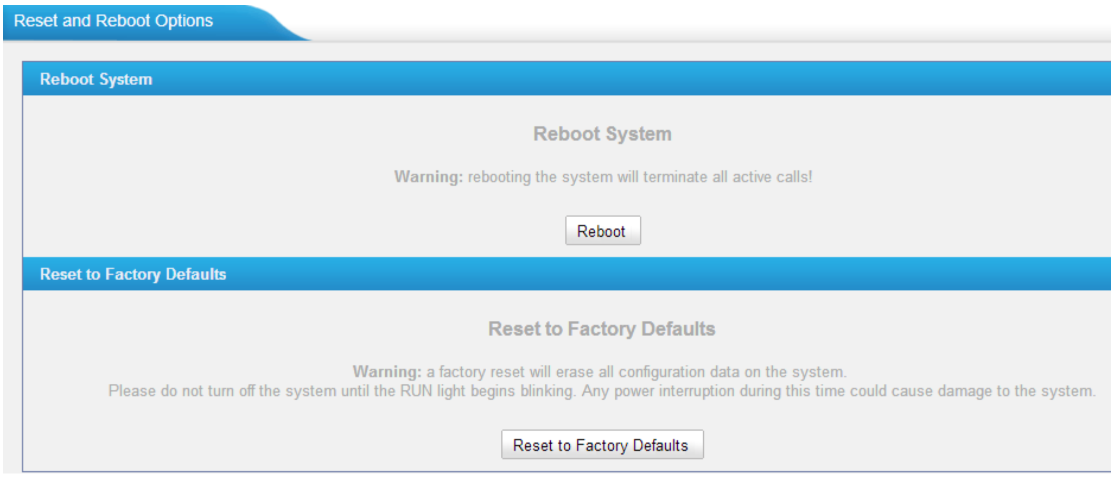


Figure 19-7 Reset and Reboot

System Logs

The N412 supports to monitor important system logs, including hardware log, web log and debug log.

Go to **Status→ Reports→ System Logs** to check the system logs.

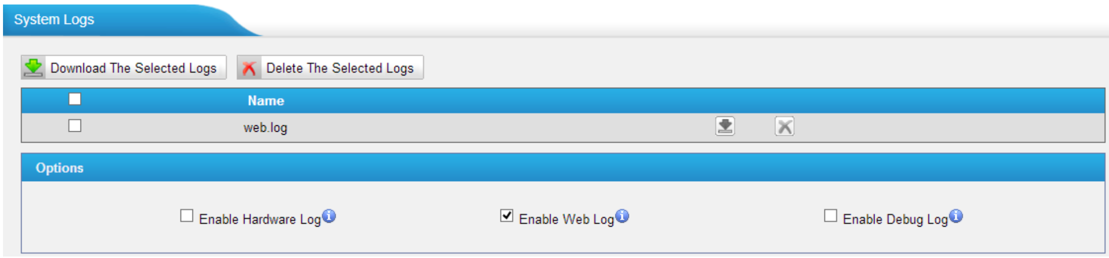


Figure 19-8 System Logs

- **Enable Hardware Log**
  Save the information of hardware; (up to 4 log files)
- **Enable Web Log**
  Save the history of web operations (up to 2 log files)
- **Enable Debug Log**
  Save debug information (up to 2 log files)
  Tick the option, the following picture shows. Set the debug level and which IP address to monitor to capture the debug logs.

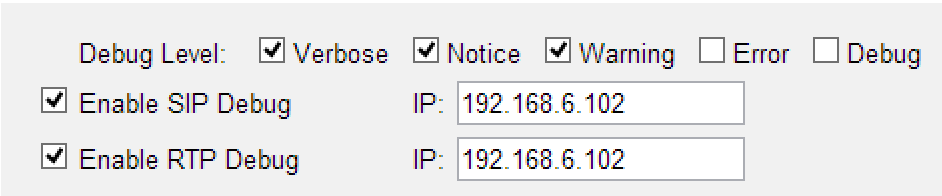


Figure 19-9 Debug Preferences

Packet Tool

Packet Capture Tool

This feature is used to capture packets for technician. Integrate packet capture tool “Wireshark” in N412. The Packet Tool can be found under **Status →Reports→ Packet Tool**.

Users could specify the destination IP address and port to get the packets.

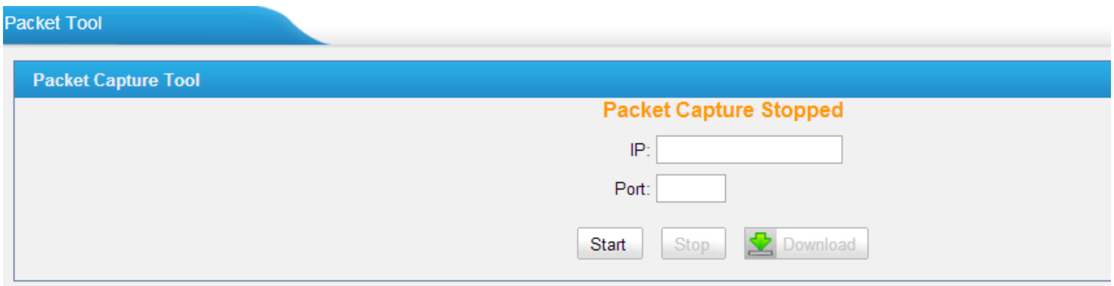


Figure 19-10 Packet Tool

DAHDI Monitor Tool

This feature is used to monitor CO lines on N412. Users could choose a CO line:

- Click **Start** to monitor the trunk.
- Click **Stop** to stop monitoring.
- Click **Download** to download the files to the local PC and analysis the files.

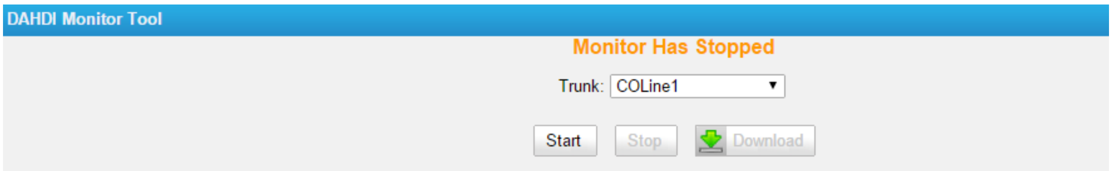


Figure 19-11 DAHDI Monitor Tool

[END]