Administrator Guide

MyPBX SOHO

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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

About This Guide

Thanks for choosing Yeastar MyPBX SOHO. Ideal for small offices and businesses with fewer than 32 users, MyPBX SOHO is an affordable, yet fully-featured embedded hybrid IP PBX with ISDN BRI and PSTN connectivity, providing a cost-effective solution for your business.

Audience

This guide is intended for administrators who need to prepare for, install, configure and operate MyPBX Telephone System. In this guide, we describe every details on the functionality and configuration of MyPBX. We begin by assuming that you are interested in MyPBX and familiar with networking and other IT disciplines.

Note: this guide applies to MyPBX SOHO V4/V5/V6, the hardware pictures in this document are for MyPBX SOHO V5.

Safety when working with electricity

- Do not open the device when the device is powered on.
- Do not work on the device, connect or disconnect cables when lightning strikes.

Overview

This chapter provides the following sections:

- Feature
- MyPBX SOHO Front Panel
- MyPBX SOHO Rear Panel

Features

● Alert	● Follow me
● Auto-provision	● HTTPS
● Blacklist	● Integrated built-in packet capture tools
● BLF Support	● Interactive Voice Response (IVR)
● Blind Transfer	● Intercom/Zone Intercom
● Call Back	● L2TP
● Call Detail Records(CDR)	● LDAP
● Call Forward	● Mobility Extension
● Call Parking	● Multiple administrators
● Call Pickup	● Music On Hold
● Call Recording	● Music On Transfer
● Call Routing	● Open VPN
● Call transfer	● Paging/Zone Paging
● Call Transfer	● PIN Users
● Call Waiting	● QoS
● Caller ID	● Queue
● Conference	● Ring Group
● Database Grant	● Route by Caller ID
● DDNS	● Security Center
● Define Office Time	● Skype Integration (Skype Connect)
● Dial by Name	● Speed Dial
● DIDs	● Spy functions
● Direct Inward System Access (DISA)	● Static Route
● Distinctive Ringtone	● T.38
● Do Not Disturb(DND)	● Three-way Calling
● External Storage	● VLAN
● Firewalls	● Voicemail

Learn more about MyPBX SOHO here:
http://www.yeastar.com/Products/MyPBX-SOHO

MyPBX SOHO Front Panel

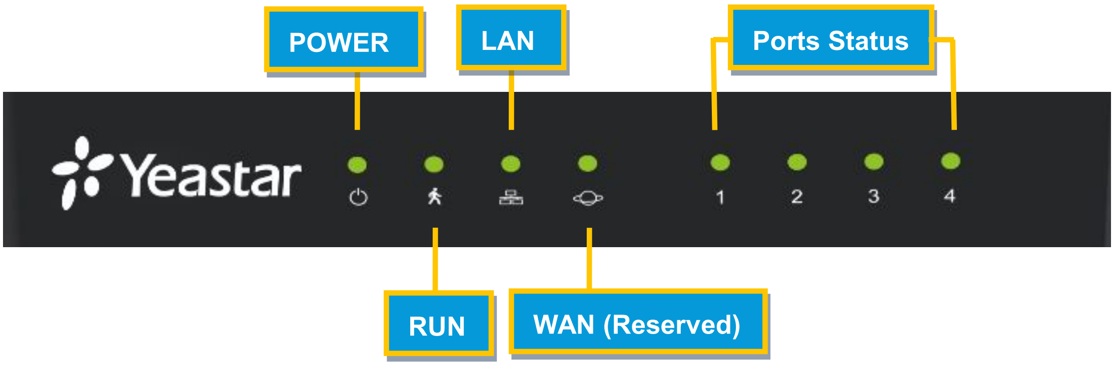


Figure 1-1 MyPBX SOHO Front Panel

Table 1-1 MyPBX SOHO 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	/	WAN port is reserved.
Ports Status	Red LED stands for FXO port. Orange LED indicates presence of BRI ports. Green LED stands for FXS port. Red LED blinks: FXO port isn't connected to PSTN line. Alternately blinks Red and Green: FXO port has an incoming call. Alternately blinks Red and Green fast: FXO port is in a call. Alternately blinks Green and Red: FXS port is ringing. Alternately blinks Green and Red fast: FXS port is in a call.	

Table 1-2 MyPBX SOHO Front Panels–Port Description

Port	Description
FXS Ports (1-24)	For connection of analog phones/fax machines.
Co Ports (1-8)	For connection of PSTN trunks.

MyPBX SOHO Rear Panel

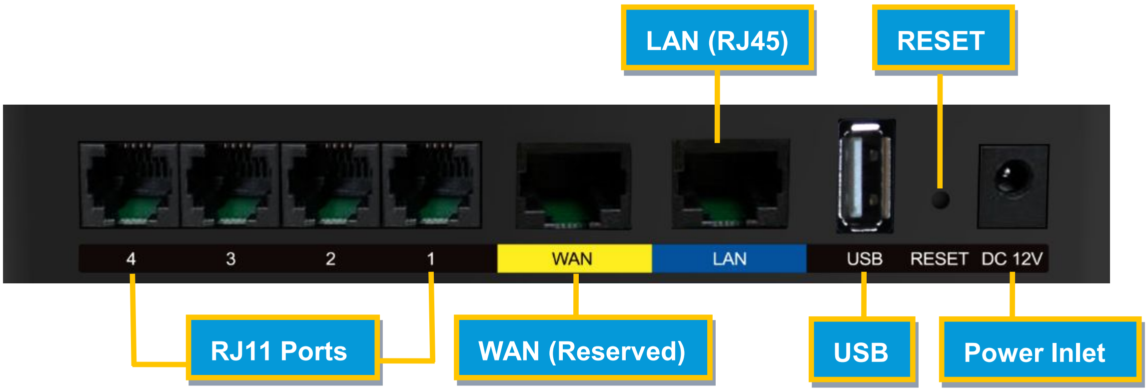


Figure 1-2 MyPBX SOHO Rear Panel

Table 1-3 MyPBX SOHO Rear Panels–Port Description

Port	Description
RJ21 Ports	• **FXO port** (red light): for the connection of PSTN lines or FXS ports of traditional PBX. • **FXS port** (green light): for the connection of analog phones. • **BRI port** (orange light): for the connection of ISDN BRI lines. **Note:** the sequence number of the ports corresponds to that of the Indicator lights in the front panel. (I.e. the LED lights in the front indicate the connection status of the corresponding ports at the front panel.)
WAN	Reserved port.
LAN	10/100Base-TX, connect one end of an RJ-45 Ethernet cable into the LAN port.
USB	Reserved port.
Reset Button	Press the reset button to restore the factory defaults. **Note:** please make sure that you want to reset, because once reset the previous configurations would be erased automatically.
Power Inlet	For connection of power supply.

Installation

Before getting started with MyPBX SOHO, you need to know how to install the device properly. This chapter gives detailed installation instructions.

- MyPBX SOHO Packing List
- Specifications and Operating Environment
- Placement Instructions
- Connecting MyPBX SOHO

MyPBX SOHO Packing List

Upon receiving Yeastar MyPBX SOHO gift box, please open the package and check if all the items are supplied as MyPBX SOHO Packing List. If there is any problem, please contact your provider.

Table 2-1 MyPBX SOHO Packing List

Item	Unit	QTY	Description
MyPBX SOHO	PC	1	MyPBX SOHO device unit
Power adapter	PC	1	For the input of 12V/1A DC power
Telephone line	PC	1	
Network cable	PC	1	
Rubber feet	PC	4	
Warranty card	PC	1	With Serial Number printed for Repair & Return.
Feature code card	PC	1	

Specifications and Operating Environment

Table 2-2 Specifications and Operating Environment

MyPBX SOHO	Description
Size (L×W×H)	160 mm ×160 mm ×30 mm
Power Supply	DC 12V,1A
Operating Temperature	0°C to 40°C, 32°F to 104°F
Storage Temperature	-20°C to 65°C, 4°F to 149°F
Humidity	10% to 90% (non-condensing)

Placement Instructions

To avoid unexpected accident, personal injury or device damage, please read the following instructions before installing MyPBX.

- Ambient Temperature: to avoid overheating, please do not run MyPBX SOHO in the place where the ambient temperature is above 104°F (40°C).
- Ventilation: please make sure that the device has good ventilation around.
- Anti-jamming: there may be some sources of interference that might affect the normal running of the Gateway. It's highly recommended that the device
    i. Should be placed away from high-power radio, radar transmitters and high frequency, and high-current devices.
    ii. Is using independent power junction box and effective anti-grid interference measures have been taken.
- Mechanical load: Please make sure that the device is placed steadily to avoid any accident that might cause damage. If placed on the desktop, please ensure it is horizontally placed.

Connecting MyPBX SOHO

Connection of Ethernet Ports

MyPBX SOHO provides two10/100M adaptive RJ45 Ethernet ports, LAN and WAN. Connect one end of a network cable to the LAN/WAN port of the MyPBX SOHO, and the other end to any port of company's switch/router.

Connection of FXO/FXS/BRI Ports

MyPBX SOHO supports various outside lines, FXO, FXS and BRI.

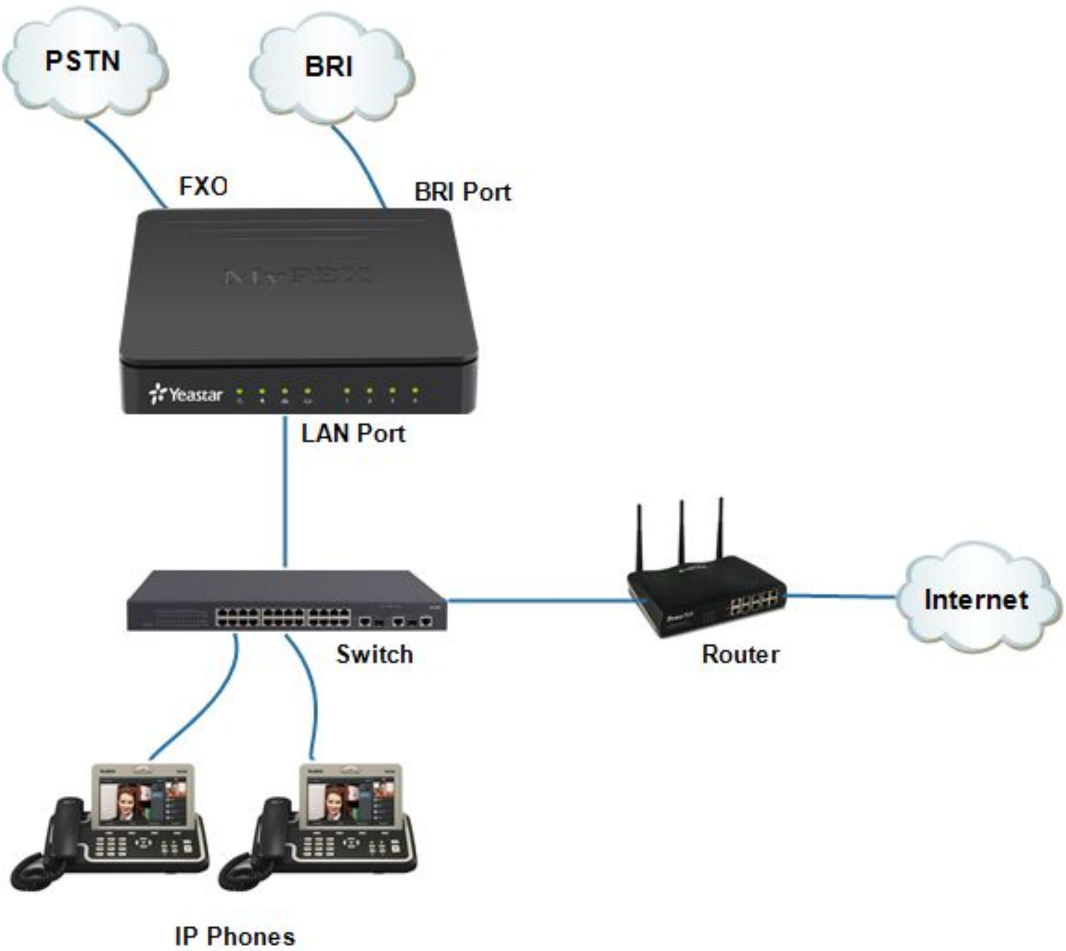
- **Connection of FXO ports**
  The FXO port could be connected to the PSTN line or the FXS port of a traditional PBX with a phone line.
- **Connection of FXS ports**
  Connect one end of a RJ11 phone cable to the port, connect the other end to the analog phone.
- **Connection of BRI ports**

The BRI port could be connected to the BRI line and the BRI port of a traditional PBX with a BRI RJ45-RJ11 cable.

Power Connection

Connect the power adapter to the MyPBX SOHO's power port, and then plug the power socket into an electrical outlet. The device will start booting. In the meantime, users would see that the "POWER" and "RUN" indicator lights turn on.
Note: please switch off the power before plugging or unplugging the cables.

Connection Diagram



Getting Started

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

- Accessing Web GUI
- Web Configuration Panel
- User Management
- Making and Receiving Calls

Accessing Web GUI

MyPBX SOHO provides 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.

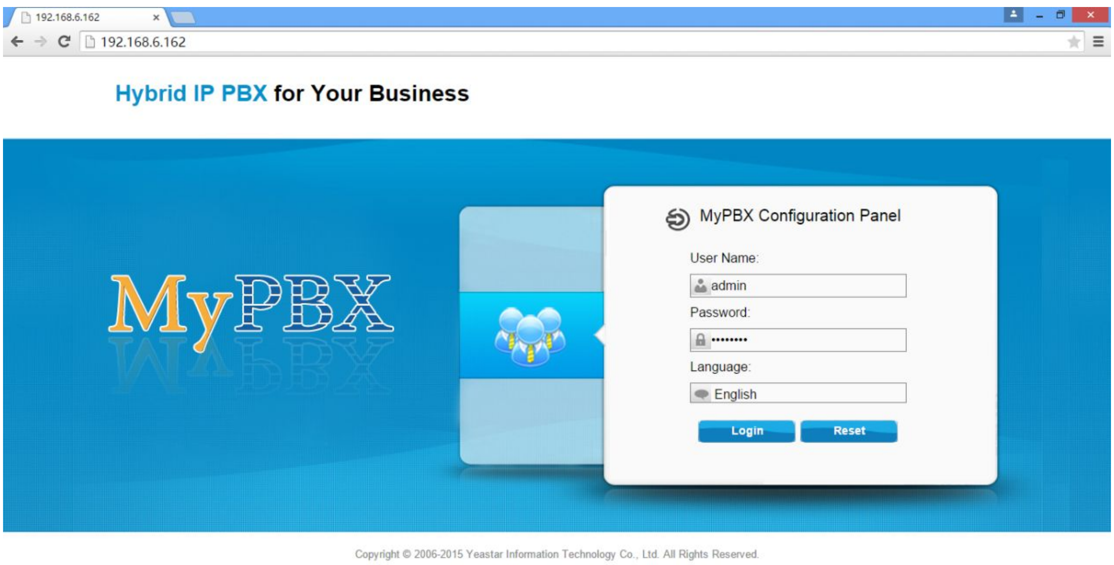


Figure 3-1 MyPBX SOHO Web Configuration Panel Login Page

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

Web Configuration Panel

There are 5 main sections on the Web Configuration Panel for users to check the MyPBX SOHO's status and configure it.

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

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

MyPBX SOHO supports 4 user types with different privileges.

User Privileges

- **Administrator** has the highest privilege. The administrator can access all pages on MyPBX SOHO Web and make all the configurations on the system.
  Username: admin
  Default Password: password
- **User Administrator** has basic privileges; do not have the permission to create VoIP trunks, reset, update, backup and restore MyPBX.
- **CDR Administrator** only has the permission to manage call logs.
- **Extension User** has the privilege to check voicemails, one-touch recordings and CDR.
  Username: Extension number (i.e.601)
  Default Password: Extension number (i.e. 601)
  The password is the same as "Voicemail Access PIN" and can be changed on extension edit page.

Making and Receiving Calls

The MyPBX default settings are sufficient to make phone calls. It is not necessary to make any changes unless the user wants to create new extensions or trunks.

- **Internal calls between extensions**
  Insert FXS module on MyPBX and connect analog phones to FXS ports. Connect analog phones to FXS ports on MyPBX SOHO, users could make calls between extensions just by dialing the other's extension number.
  If IP terminals have registered to MyPBX SOHO successfully, users could also make internal calls using SIP extensions.

- **Outbound calls**
  Firstly, please connect PSTN/BRI lines to FXS/BRI ports on MyPBX SOHO. Then the default extensions are able to make outbound calls via the trunks by simply dialing "9+phone number".
  Note: the dial pattern of the default outbound route is "9." and strip 1 digit. If you have changed it, please dial according to the outbound route settings.

- **Inbound calls**
  When the user calls the trunk number of the PSTN or BRI trunk, MyPBX SOHO would route the call to IVR (the "welcome" prompt) by default. Then the user can operate following the prompt.

Extensions

This chapter explains how to create and configure extensions on MyPBX SOHO. It supports VoIP extensions and FXS extensions, go to **PBX→ Extensions→ FXS/VoIP Extensions** page to configure the extensions.

- FXS Extensions
- VoIP Extensions

FXS Extensions

MyPBX SOHO supports up to 4 FXS extensions. To extend FXS extensions, S2 module or SO module should be installed on MyPBX.
After inserting the module, users could see FXS extensions on MyPBX web GUI. Click to edit each FXS extension.

To change a FXS extension's number, you have to click to delete the extension first, then click to edit the extension's number.

FXS Extension Configuration

The extension settings are divided into General Settings and Other Settings.

1) General Settings

Table 4-1 FXS Extension Configuration- General Settings

Items	Description
Extension	The numbered extension, which will be associated with this particular User/Phone.
Port	Corresponding port for the FXS port, depends on which slot the module was inserted.
Name	A character-based name for this user, e.g. "Bob Jones".
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 Settings	• Enable Send Voicemail Once enabled, the voicemail will be sent to a configured

	email address. • Enable Send Voicemail Wether to send voicemail to email or not. • Email Address If "Voicemail to Email" is enabled, fill in the email address where to send the voicemail. **Note:** please ensure that the section "SMTP Settings for Voicemail" (in the "Voicemail Settings") have been properly configured before using this feature.
Flash	Sets the amount of time, in milliseconds, that must pass since the last hook-flash event received by MyPBX SOHO before it will recognize a second event. If a second event occurs in less time than defined by Hook Flash Detection, then MyPBX SOHO will ignore the event. The default value of Flash is 1000 ms.
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.

2) Advanced Settings

Table 4-2 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 MyPBX SOHO's call waiting function.
DND	Don't Disturb. When DND is enabled for an extension, the extension will not be available.
User Web Interface	Check this option to allow the user to log in to the MyPBX User Web interface, which can be used to access voicemail and extension recordings. Users may log in the MyPBX User Web interface by using their extension number and voicemail PIN as the user name and password respectively.
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.
Mobility Extension	MyPBX allows you to use your mobile phone as an extension. If you set your mobile phone as a mobility extension and then you call MyPBX with this mobile phone, you will hear a dial tone. MyPBX will recognize your call as a call from an extension. You can dial the number of other extensions (your caller ID will be the number of your extension) or dial out via outbound routes just like dialing from your extension. **Note**: if callback is enabled in the inbound route, the mobility extension function of this inbound route will be disabled. • **Enable Mobility Extension** Enable this feature. • **Mobility Extension Number** When you dial the server with this number, the mobile phone gets the permission of the extension. For example: dialing the other extensions, playing the voicemail. • **Ring Simultaneously** When the extension has an incoming call, it rings its mobility extension simultaneously. **Note:** MyPBX doesn't support the PSTN line for Ring Simultaneously. • **Outbound Prefix** Fill in proper prefix of mobile number so that it can match an outbound route to dial the mobility extension. For example, if you set the prefix 9, it will send "9+ mobility extension number" to the outbound route.
Caller ID Type	Normally, you choose the "default" option except for using MyPBX SOHO 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.

VoIP Extensions

MyPBX supports SIP and IAX VoIP extensions. Users could extend VoIP extension by clicking Add Extension to add a VoIP extension.

VoIP Extension Configuration

The extension settings are divided into General Settings and Other Settings.

1) General

Table 4-3 VoIP Extension Configuration- General

Items	Description
Type	Extension type: SIP, IAX or SIP/IAX. • **SIP**: the extension sends and receives calls using the VoIP protocol SIP. • **IAX**: the extension sends and receives calls using the VoIP protocol IAX.
Extension	The numbered extension, which will be associated with this particular User/Phone.

Password	The password for this extension.
Name	A character-based name for this user, e.g. “Bob Jones”.
Caller ID	The Caller ID will be used when this user calls another internal extension.
Register Name	It is for extension registration validation. Users will not be able register the extension if the authorization name is incorrect even though the username and password are correct.
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 Settings	• Enable Send Voicemail Once enabled, the voicemail will be sent to a configured email address. • Enable Send Voicemail Wether to send voicemail to email or not. • Email Address If “Voicemail to Email” is enabled, fill in the email address where to send the voicemail. **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.

DTMF Mode	RFC233, 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.

2) Other Settings

Table 4-4 VoIP Extension Configuration—Other 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 MyPBX SOHO’s call waiting function.
DND	Don’t Disturb. When DND is enabled for an extension, the extension will not be available.
User Web Interface	Check this option to allow the user to log in to the MyPBX User Web interface, which can be used to access voicemail and extension recordings. Users may log in the MyPBX User Web interface by using their extension number and voicemail PIN as the user name and password respectively.
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 MyPBX. 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.
Mobility Extension	MyPBX allows you to use your mobile phone as an extension. If you set your mobile phone as a mobility extension and then you call MyPBX with this mobile phone, you will hear a dial tone. MyPBX will recognize your call as a call from an extension. You can dial the number of other extensions (your caller ID will be the number of your extension) or dial out via outbound routes just like dialing from your extension. **Note**: if callback is enabled in the inbound route, the mobility extension function of this inbound route will be disabled. • **Enable Mobility Extension** Enable this feature. • **Mobility Extension Number** When you dial the server with this number, the mobile phone gets the permission of the extension. For example: dialing the other extensions, playing the voicemail. • **Ring Simultaneously** When the extension has an incoming call, it rings its mobility extension simultaneously. **Note:** MyPBX doesn't support the PSTN line for Ring Simultaneously. • **Outbound Prefix** Fill in proper prefix of mobile number so that it can match an outbound route to dial the mobility extension. For example, if you set the prefix 9, it will send "9+ mobility extension number" to the outbound route.
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.

Batch Edit VoIP Extensions

Users could add VoIP extensions in bulk and batch edit the selected VoIP extensions.

- **Add Bulk Extensions**

  Click Add Bulk Extensions to add extensions in bulk. Choose protocol and define the extension number starting from a number.

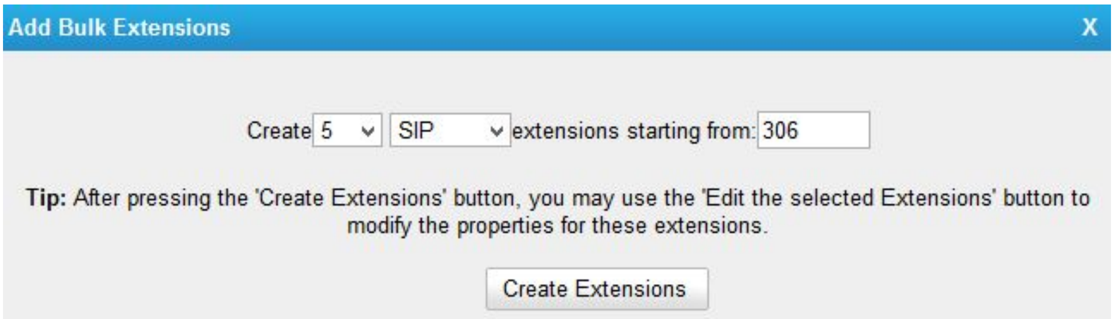


Figure 4-1 Add Bulk Extensions

- **Edit the Selected Extensions**

  Click Edit the selected Extensions, you can edit the selected extensions.

- **Delete the Selected Extensions**

  Click Delete the selected Extensions to delete selected extensions in bulk.

Provisioning

MyPBX provides Phone Provisioning and Neogate Provisioning features, which help users to configure the phones and gateways in bulk, which saves time substantially.

- Provisioning Methods
- Phone Provisioning
- Gateway Provisioning

Provisioning Methods

MyPBX supports provisioning in PnP mode and DHCP mode.

- **PnP**
  1. The IP phone or VoIP gateway broadcast SIP SUBSCRIBE message when boots up.
  2. MyPBX reply with SIP message NOTIFY and indicate the provisioning URL.
  3. The IP phone or VoIP gateway use the path to download the configuration file, resolve and apply the configurations written in the configuration file.

- **DHCP**
  Use DHCP method, you should set your SIP phone or VoIP gateway as a DHCP client and enable MyPBX DHCP server to set it as a DHCP server.
  1. The IP phone or VoIP gateway get IP address from MyPBX.
  2. MyPBX assign an IP address and indicate the provisioning URL.
  3. The IP phone or VoIP gateway download configuration file, resolve and apply the configurations written in the configuration file.

Phone Provisioning

Phone Provisioning feature is applicable to mainstream IP phone on the market, like Yealink, Polycom, Cisco, Grandstream, snom, Fanvil, Aastra, and Panasonic.

General Settings for Yealink

MyPBX provides general settings for Yealink IP phones. General configurations will apply to the selected Yealink phone no matter which device model it is. You can configure the general settings before provisioning Yealink IP phones, including general preferences, codecs, remote phone book and firmware upgrade.

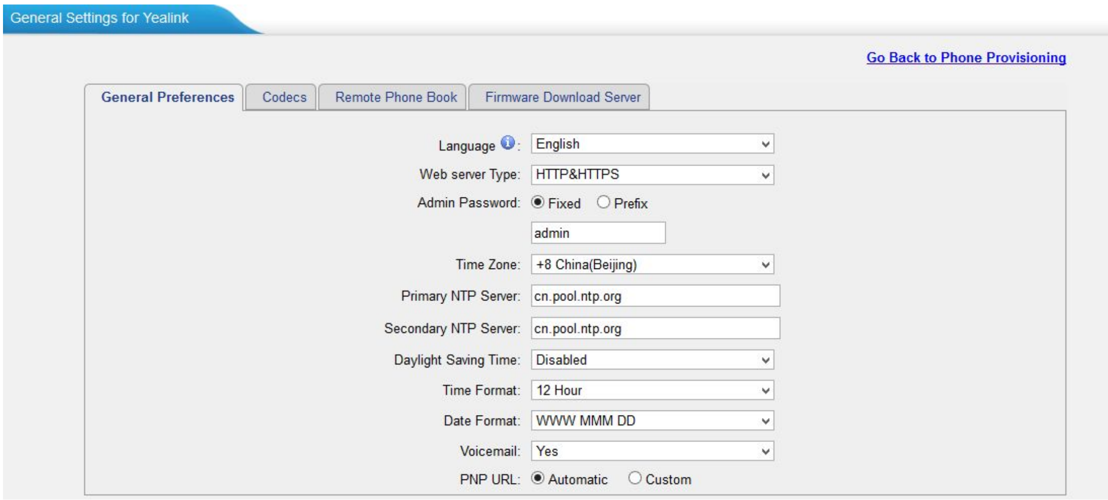


Figure 5-1 General Settings for Yealink

Table 5-1 Description of Yealink General Settings

General Preferences	
Language	Select the LCD display language on the phone.
Web Server Type	Choose the web server type. • HTTP • HTTPS • HTTP&HTTPS
Admin Password	Set the admin password for the phone.
Time Zone	Set the time zone for the phone.
Primary NTP Server	Set the primary NTP server to obtain the time.
Secondary NTP Server	Set the secondary NTP server to obtain the time.
Daylight Saving Time	Daylight Saving Time settings.
Time Format	Set the time as 12-hour or 24-hour.
Date Format	Set the date display format.
Voicemail	Whether to enable voicemail feature or not.
PNP URL	Specify the PNP URL.
Codecs	
Choose codecs for Yealink phones.	
Remote Phone Book	
Phonebook URL	Set the URL of a server where the phonebook file stored.
Phonebook Name	Fill in the phonebook name.
Firmware Download Server	
Firmware Name	A name for the firmware which you will upgrade to.
Server Type	Set the server where the phone downloads the firmware file from.

General Settings for Aastra

MyPBX provides general settings for Aastra IP phones. General configurations will apply to the selected Aastra phone no matter which device model it is. You can configure the general settings before provisioning Aastra IP phones, including general preferences, program keys configuration and soft keys configuration.

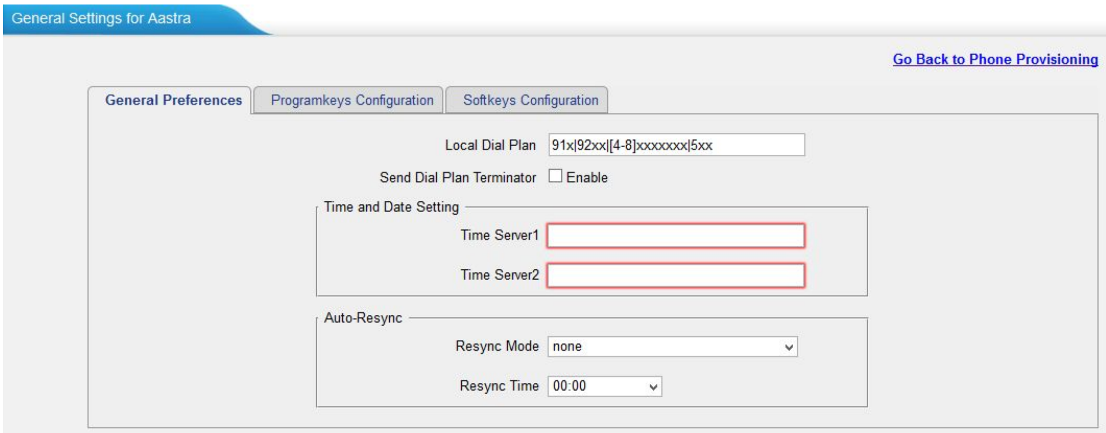


Figure 5-2 General Settings for Aastra

Phone Book

You can add contacts on MyPBX, or upload a phone book to MyPBX, the phone book on MyPBX will be applied to the phones during the phone provision process.

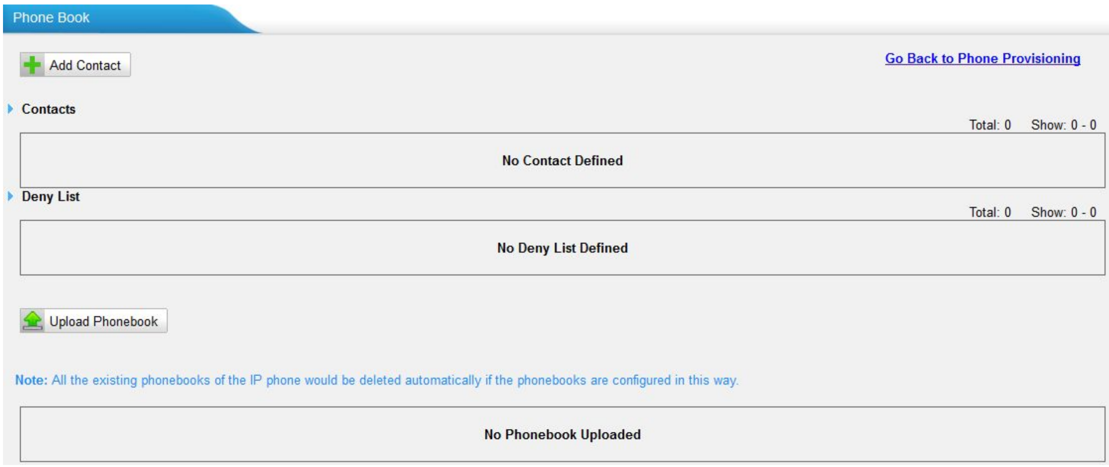


Figure 5-3 Phone Book

➢ Add Contact

Click 

to add contacts on MyPBX. If you choose the type as "Deny List", the contact will be added in the "Deny List". A contact in the Deny List could not call in the IP phone which uses the phone book.

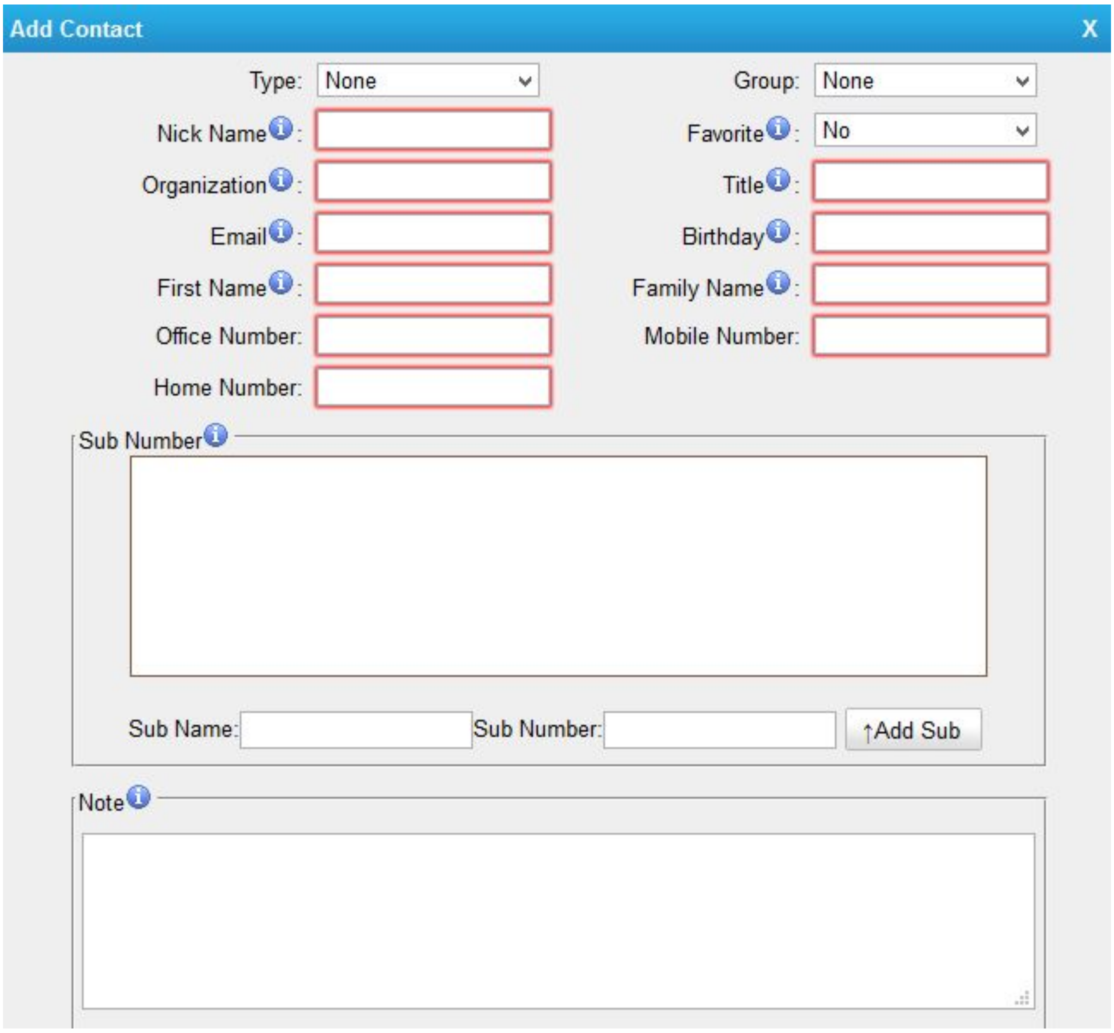


Figure 5-4 Add Contacts

Note: the following settings only work with Snom phone:

- Favorite
- Organization
- Title
- Email
- Birthday
- First Name
- Family Name
- Sub Number
- Note

➢ **Upload Phonebook**

Click Upload Phonebook to upload a phonebook to MyPBX. The file format should be *.xml.

Note: all the existing phonebooks of the IP phone would be deleted automatically if the phonebooks are configured in this way.

TA Provisioning

With this feature, you can easily configure the TA Series Analog VoIP Gateway on MyPBX, saving your time and efforts.

Click Add to add a TA gateway, configure the TA analog gateway's General settings, Extensions, Codecs, LAN Settings and Password, then click Save.

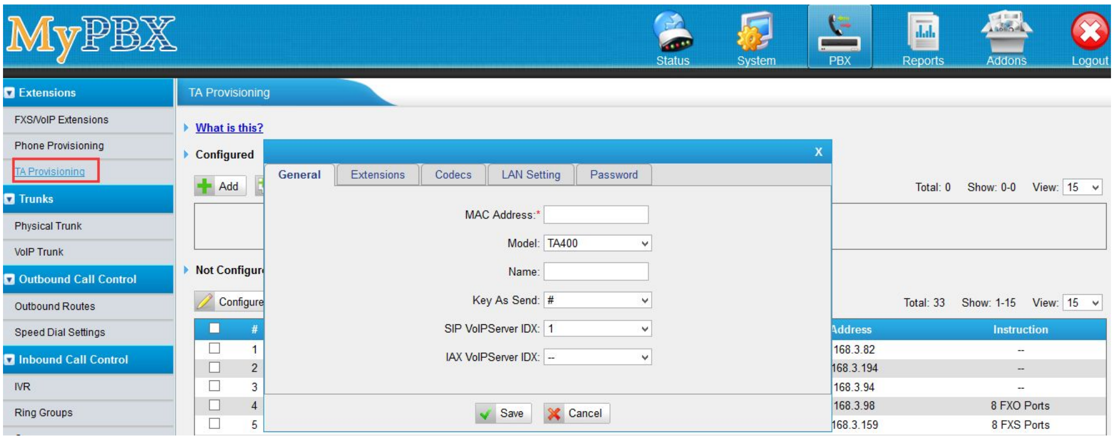


Figure 5-5 Add a Gateway

Select when to reboot the gateway automatically or save the changes only, then click "Save" again. The new configurations will be applied to the gateway after rebooting.

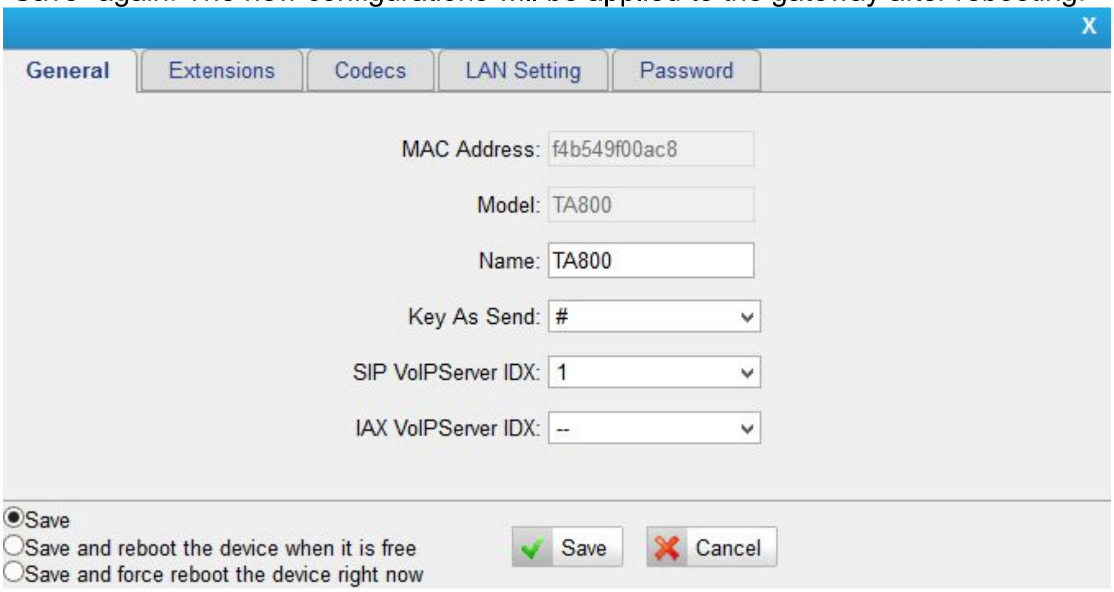


Figure 5-6 TA Provisioning

Trunks

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

- PSTN Trunks
- BRI Trunks
- VoIP Trunks

PSTN Trunks

The public switched telephone network (PSTN) is the network of the world's public circuit-switched telephone networks.

Go to **PBX→Trunks→ Physical Trunks→Analog Trunk** to edit the PSTN trunks. Before configuring a PSTN trunk, please make sure that O2 module or SO module are installed on MyPBX, and an analog line is connected to MyPBX FXO port and PSTN provider.

Click to edit the PSTN trunk.

PSTN Trunk Configuration

Please check the PSTN line configuration parameters below.

1) **General Settings**

Table 6-1 PSTN trunk-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 PSTN trunk-Hangup Detection

Hang up Detection	
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, MyPBX SOHO will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then MyPBX SOHO 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.

3) Answer Detection

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

If the PSTN trunk 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 PSTN trunk-Answer Detection

Answer Detection	
Answer Detection	Select which type to detect the call as answered. • Default: MyPBX SOHO will start to charge once you grab the PSTN trunk to call out, whether the call is answered or not. • Polarity Detection: if the PSTN trunk supports polarity, you can choose "Polarity detection". When the callee answers the call, the provider will send a polarity signal, and then MyPBX SOHO starts to bill. • Ring Detection: if you choose this option, MyPBX SOHO will charge the call according to PSTN trunk ring back tone detection. When the "ring duration" or the "ring interval duration" detected on MyPBX SOHO 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 ringback 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 PSTN trunk-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
Ring Detect Timeout	FXO (FXS signaled) devices must have a timeout to determine if there was a hangup before the line was answered. Range from 1000 to 8000. The default value is 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 MyPBX, please install B2 module on MyPBX, 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 6-5 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-CPE**: ISDN BRI in TE mode and Point to Point. • **BRI-CPE-PTMP**: ISDN BRI in TE mode and Point to multi Point. • **BRI-NET**: ISDN BRI in NET mode and Point to Point. • **BRI-NET-PTMP**: ISDN BRI in NET mode and Point to multi Point.
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. • **net5**: NET5 ISDN PRI switches (Europe).
Overlap Dial	Define whether MyPBX 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 MyPBX 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**

2) Caller ID Prefix

Table 6-6 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.

3) Dialplan

Table 6-7 BRI trunk-Dialplan

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

4) 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.

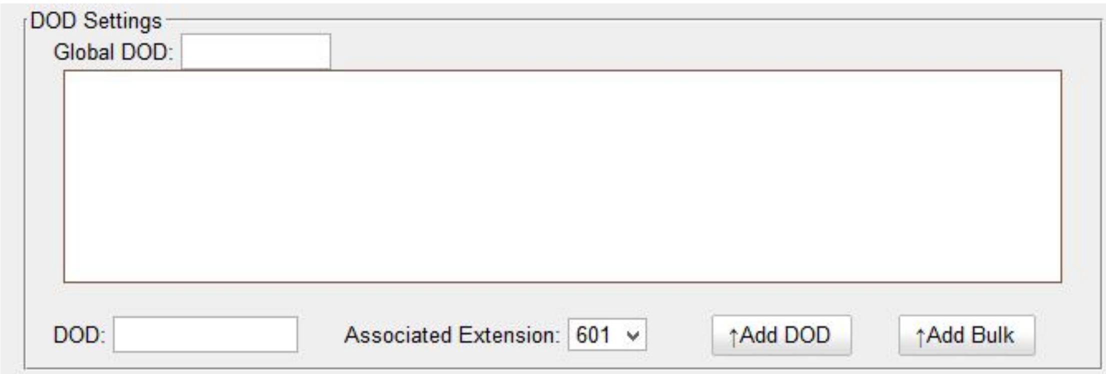


Figure 6-1 BRI trunk-DOD Settings

Table 6-8 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.

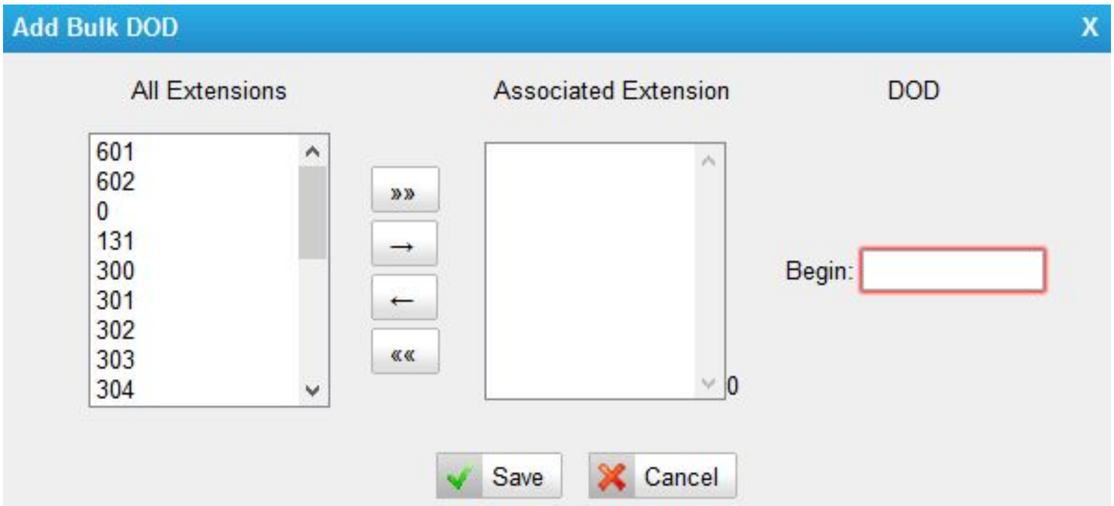


Figure 6-2 Add Bulk DOD

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

MyPBX provides 2 types of VoIP trunks:

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

VoIP Trunk

1) General Settings

Table 6-9 VoIP Trunk-General Settings

General Settings	
Type	Set the protocol. • **SIP** • **IAX**
Trunk 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 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 MyPBX SOHO and sends INVITE packet to MyPBX SOHO, MyPBX SOHO 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**: MyPBX SOHO 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: 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.

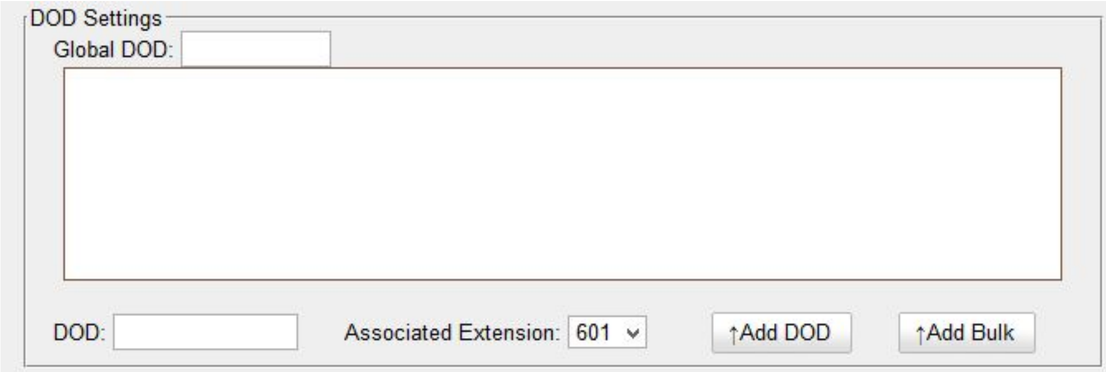


Figure 6-3 VoIP trunk-DOD Settings

Table 6-10 VoIP 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.

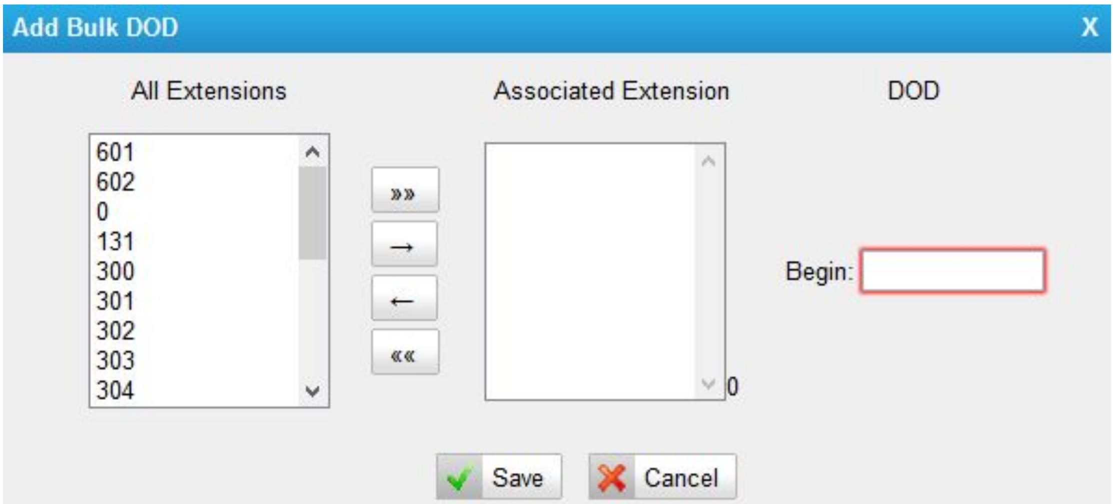


Figure 6-4 Add Bulk DOD

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.

Service Provider

1) General Settings

Table 6-11 Service Provider Trunk-General Settings

General Settings	
Type	
Trunk 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.

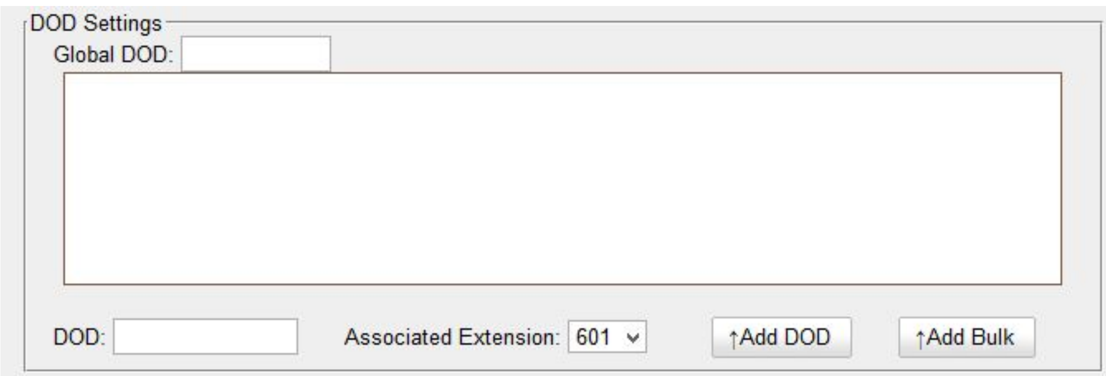


Figure 6-5 DOD Settings

Table 6-12 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.

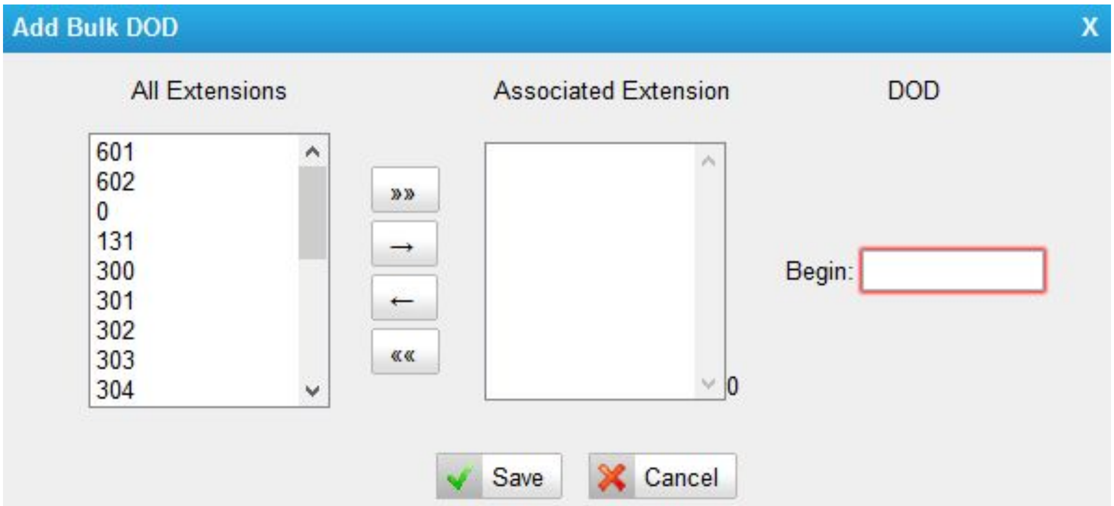


Figure 6-6 Add Bulk DOD

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 Routes

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

- Outbound Routes
- Inbound Routes
- PIN Settings
- Blacklist

Outbound Routes

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

Notes:

- MyPBX SOHO compares the number with the pattern that you have defined in your route 1. If it matches, it will initiate the call using the selected trunks. If it does not, it will compare the number with the pattern you have defined 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→Outbound Routes** 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, MyPBX SOHO 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 Add 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 dots 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.

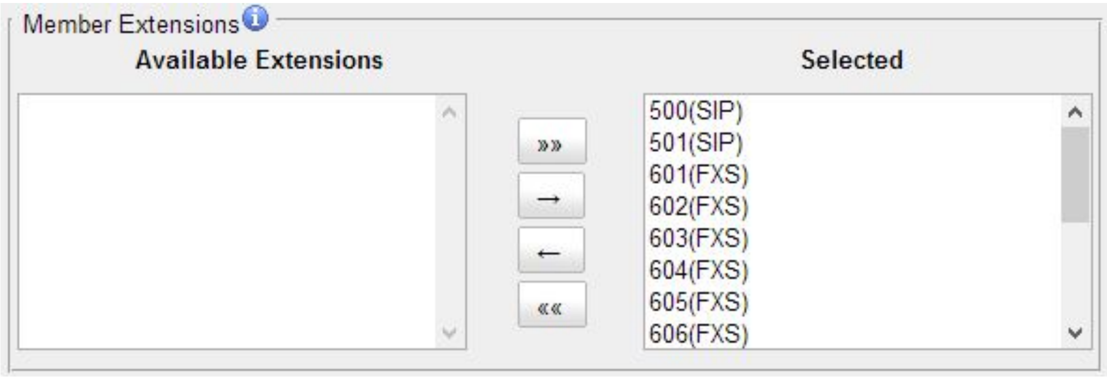


Figure 7-1 Outbound Route-Member Extensions

4) **Member Trunks**
Move the trunks that would be used on this outbound route to "Selected" Box.

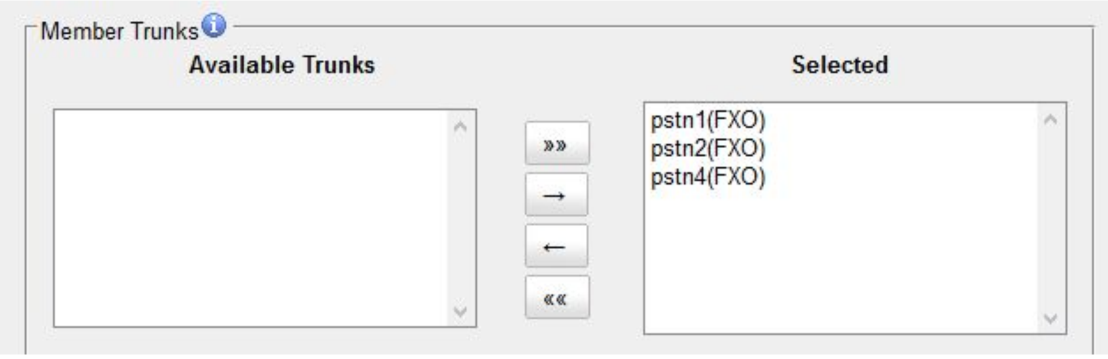


Figure 7-2 Outbound Route-Member Trunks

Inbound Routes

When a call comes into MyPBX SOHO from the outside, MyPBX SOHO 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→ Inbound Routes** to inbound routes.
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.
Extension	Define the extension for DID number. This field is only valid when you use BRI or VoIP trunk for this inbound router. You can only input number and "-" in this field, and the format can be xxx or xxx-xxx. The count of the number must be only one or equal the count of the DID number.
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.
Distinctive Ringtone	Distinctive ring tone distinguishes calls from different inbound routes. For example, if you define an IP phone's internal ringer file name as "Family", then set the Distinctive Ringtone as "Family", the ring tone will be played if the phone receives an incoming call through this inbound route. **Note**: this feature works with all IP phones which support distinctive ringtone.
Enable Callback	Choose to enable Callback feature for the inbound route or not.

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 7-3 Inbound Route-Member Trunks

3) **Time Conditions**

Time condition has two call destinations. It is used to control how MyPBX routes calls during and outside business hours.

Figure 7-4 Inbound Route-Time Conditions

- **Time Groups**
  Select a time group this condition will be checked against.
- **Day Destination**
  Select the destination when the time matches the times defined in the assigned time group.
- **Night Destination**
  Select the destination when the time does not match the time group assigned.

Table 7-4 Inbound Route-Time Conditions

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.
DISA	Route the incoming calls to a specific DISA.
Queues	Route the incoming calls to a specific Queue.

Outbound Routes	Route the incoming calls to a specific outbound route.
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.

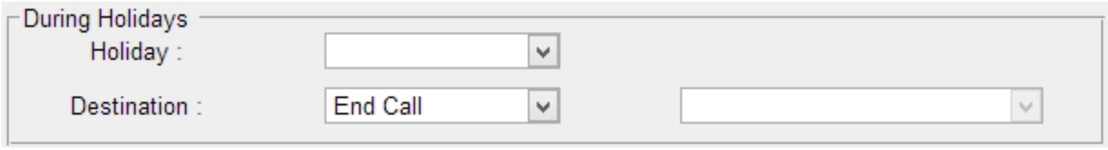


Figure 7-5 Inbound Route-During Holidays

- **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.

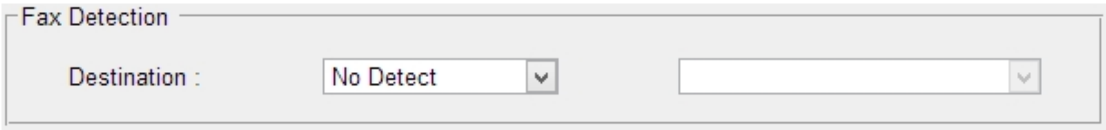


Figure 7-6 Inbound Route-Fax Detection

- **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.

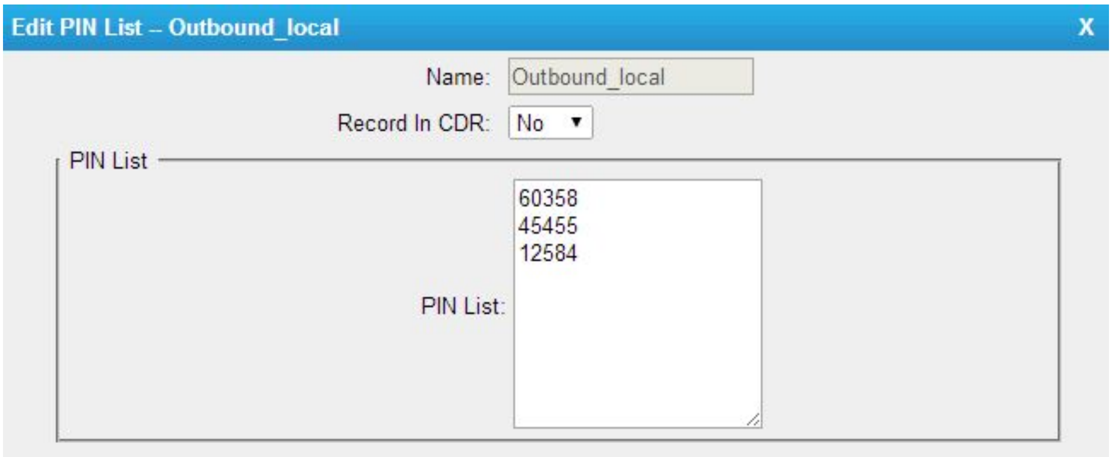


Figure 7-7 PIN Settings

Blacklist

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

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 MyPBX SOHO can't call this number.

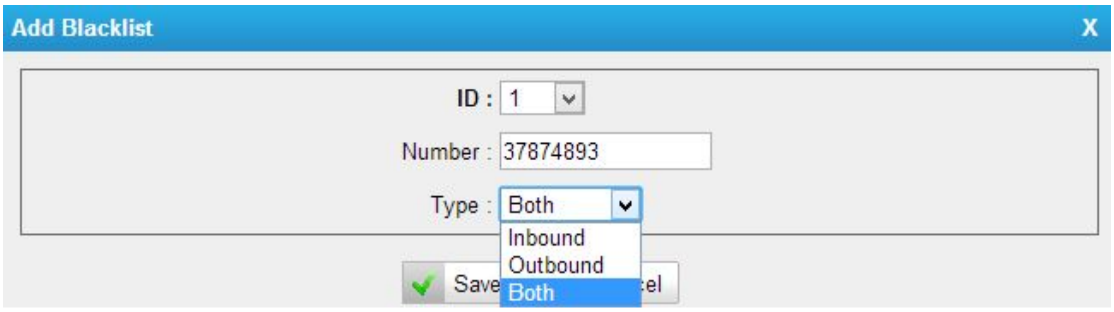


Figure 7-8 Number Blacklist

IVR

Like most organizations, where possible, we would like to route incoming calls an Auto Attendant. You can create one or more IVR (Auto Attendant) on MyPBX SOHO to achieve it. When calls are routed to an IVR, MyPBX SOHO 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	MyPBX SOHO 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 MyPBX SOHO or choose a **Custom Prompt** which is uploaded or created on MyPBX SOHO.
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.

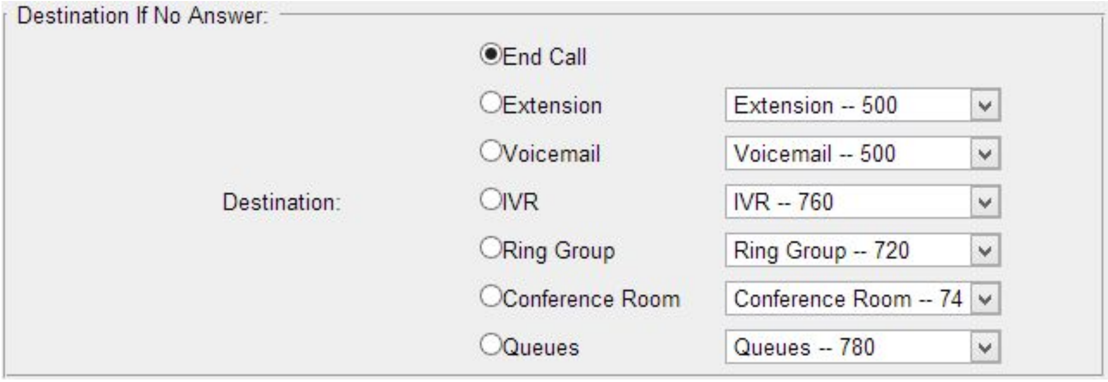


Figure 9-2 Destination if No Answer

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 MyPBX SOHO 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. • **RingAll:** ring all available agents simultaneously until one answers. • **LeastRecent:** ring the agent which was least recently called. • **FewestCalls:** ring the agent with the fewest completed calls.

	• **Randoml:** ring a random Agent. • **RRmemory:** Round Robin with Memory, remembers where it left off in the last ring pass. • **Linear:** rings agents in the order they are listed in the configuration file.

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.

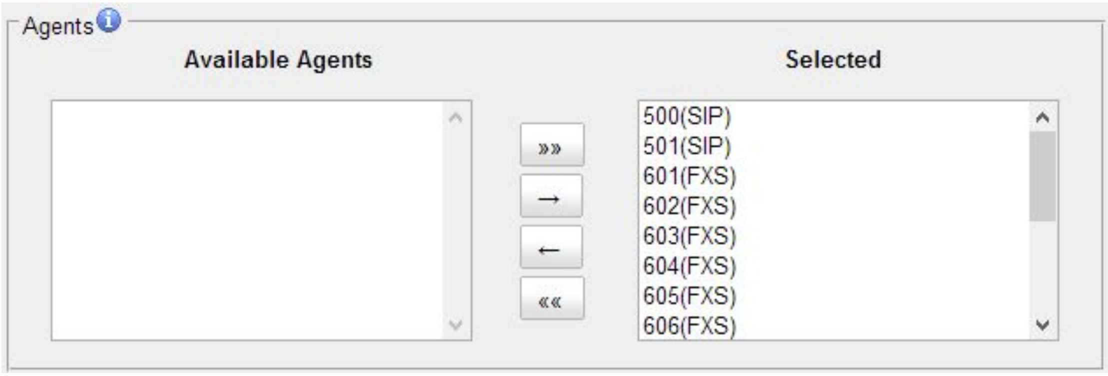


Figure 10-1 Agents

3) Caller Position Announcement

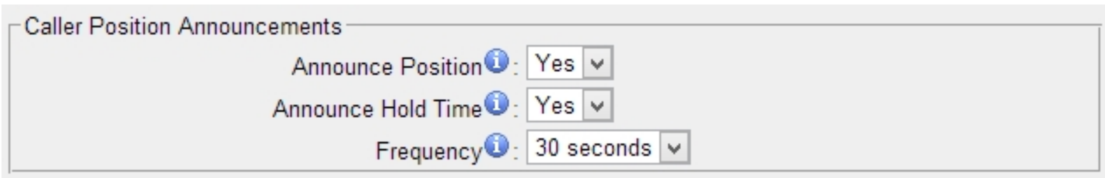


Figure 10-2 Caller Position Announcement

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 MyPBX SOHO 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.

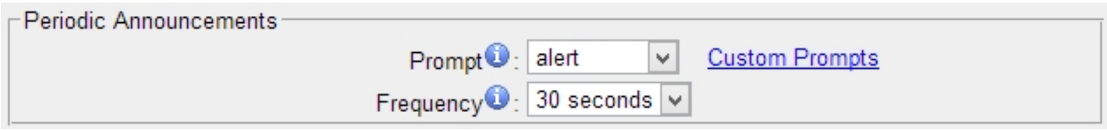


Figure 10-3 Periodic Announcements

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.

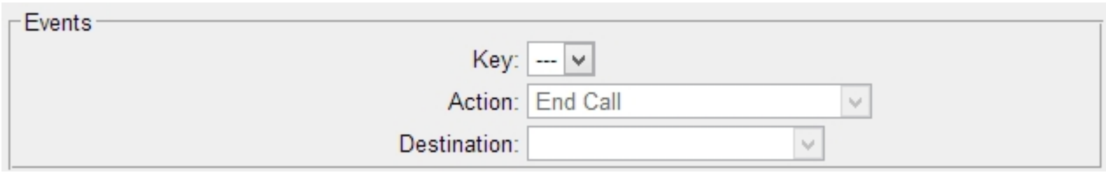


Figure 10-4 Events

6) Failover Destination

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

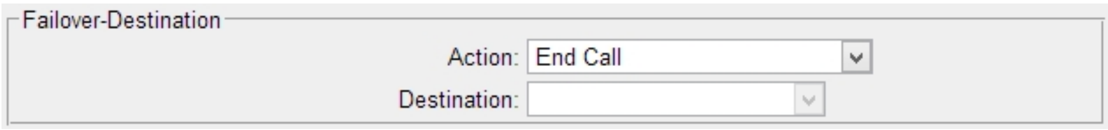


Figure 10-5 Failover Destination

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).

Intercom & Paging

MyPBX supports intercom and paging features to make an announcement to a single extension or a group. This chapter introduces the configuration of intercom and paging.

Intercom

Intercom is a feature that allows you to make an announcement to one extension via a phone speaker. The called party does not need to pick up the handset. It can be achieved by pressing the feature code on your phone and it is a two-way audio call.

The default Intercom feature code is *5. To make an announcement to a specific extension, you need to dial *5+ extension number on your phone. For example, make an announcement to extension 500, you need to dial *5500, then the extension 500 will be automatically picked up.

Paging Group

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

Paging group is supported by the following SIP phones:
Yealink T28, T26, T22, T20, T10T, T9CM. Other SIP devices may also work with this feature but are not officially supported.

Note: a paging group can have a maximum of 20 members.

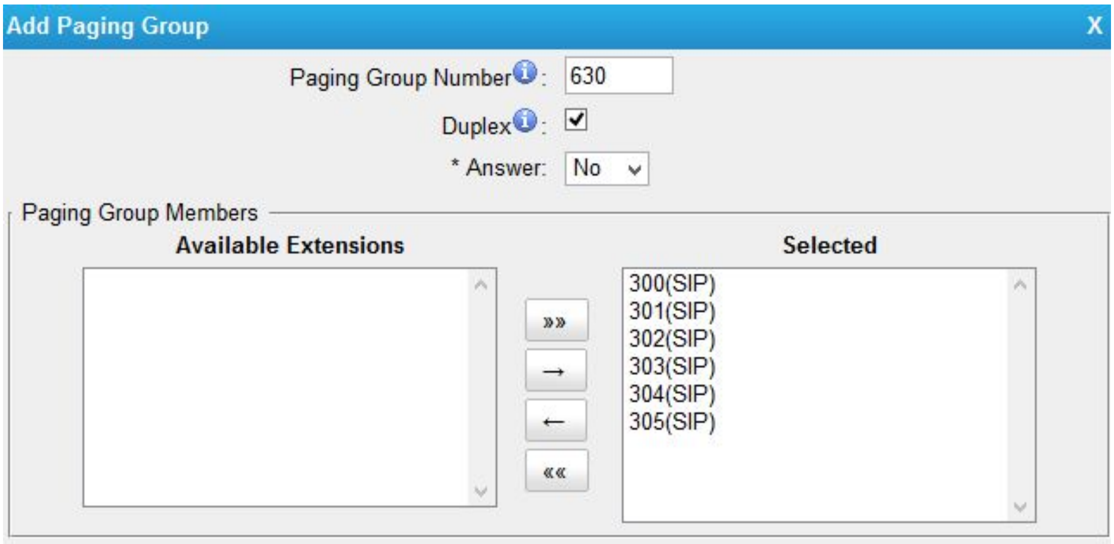


Figure 11-1 Paging Group

- **Paging Group Number**
  Configure the paging group extension.
- **Duplex**
  Paging is typically one way for announcements only. Checking this option will make paging duplex, allowing all users in the group to talk and be heard by all.
- ***Answer**
  If * Answer is enabled, users could press * to talk to the broadcaster.

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 12-1 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 MyPBX SOHO 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 MyPBX SOHO 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 MyPBX SOHO. 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 12-2 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.
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.

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

- System Prompt
- Custom Prompt
- Music on Hold

System Prompt

MyPBX SOHO 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 MyPBX SOHO 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.

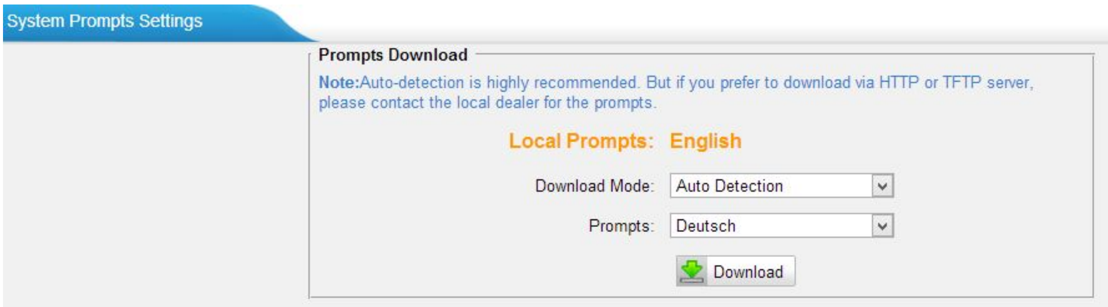


Figure 13-1 Update System Prompts- Auto Detection

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

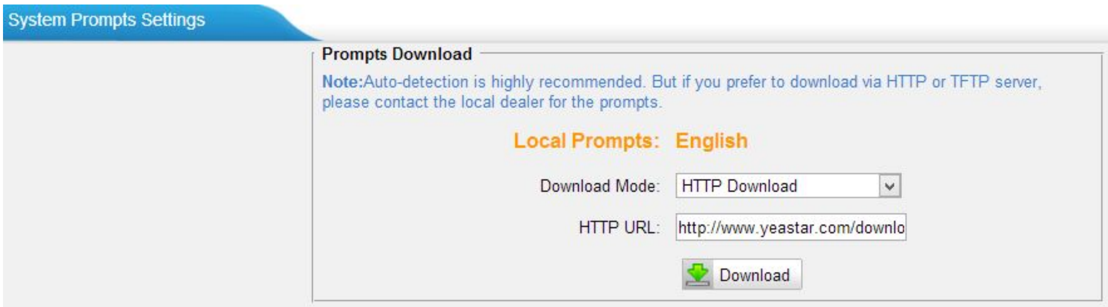


Figure 13-2 Update System Prompts- HTTP Download

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 MyPBX SOHO 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 MyPBX SOHO, such as D:\fr.tar.gz.

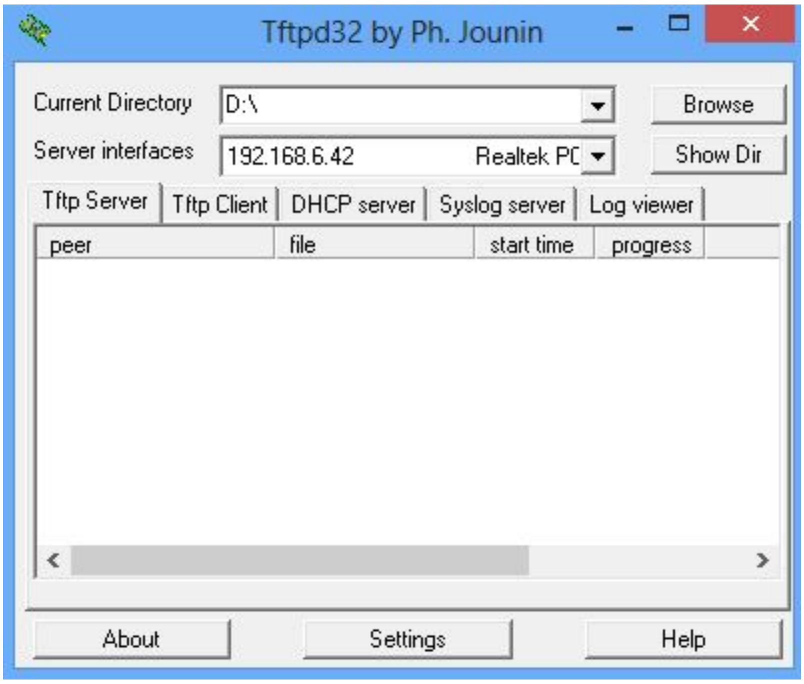


Figure 13-3 Configure Tftpd32

Step3. Update via TFTP

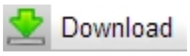
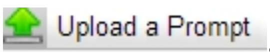
1) TFTP Server: fill in IP address of tftpd32 server, such as 192.168.6.42.
2) File Name: enter the name of voice prompt tar file name, such as "fr.tar.gz".
3) Click  to download the system prompt and update.

Figure 13-4 Update System Prompts- TFTP Download

Custom Prompt

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

Upload a custom prompt via **PBX→Audio Settings→Custom Prompts**.

1) Click the button .

2) Click Choose File to choose the desired prompt.

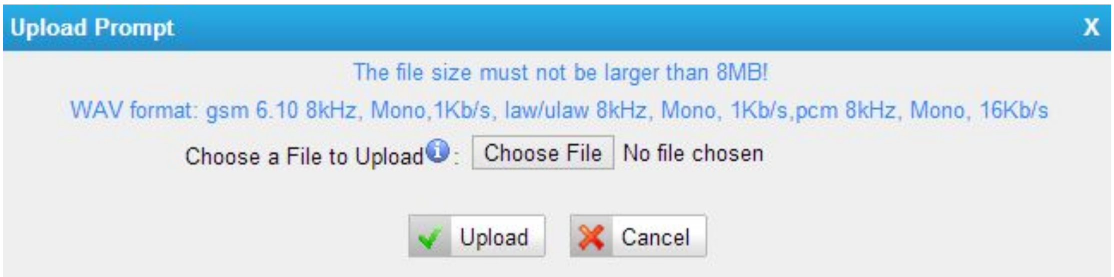


Figure 13-5 Upload Custom Prompt

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 MyPBX SOHO, you can also upload the one you want to MyPBX SOHO.

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.

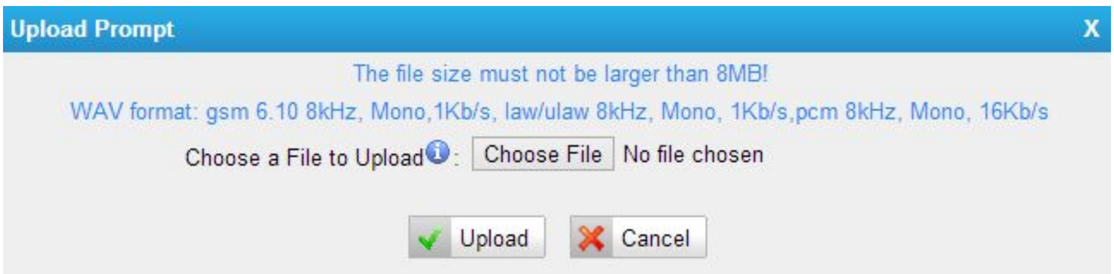


Figure 13-6 Upload Music on Hold File

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.

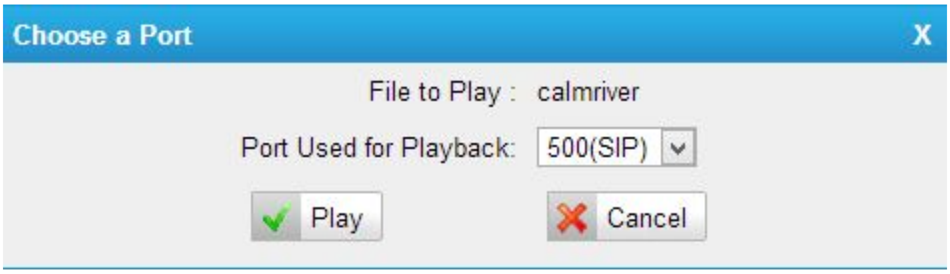


Figure 13-7 Play a Music on Hold Prompt

Voicemail

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

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 14-1 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 14-1 Voicemail-SMTP Settings

- **E-mail Address**
  The E-mail Address that MyPBX SOHO 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 MyPBX SOHO 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 MyPBX SOHO 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 MyPBX SOHO, 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.

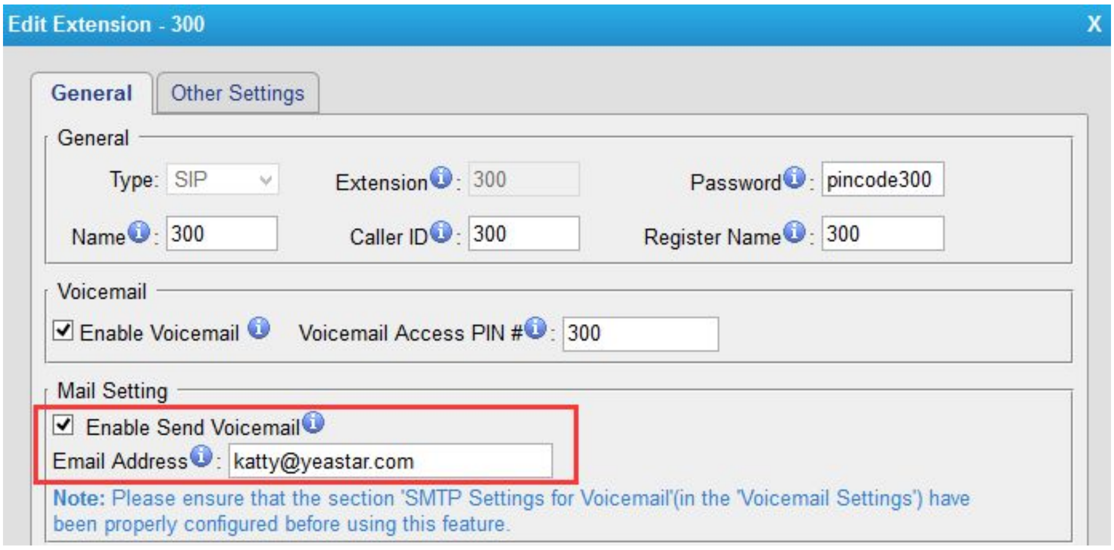


Figure 14-2 Enable Voicemail to Email

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 MyPBX SOHO. You can check the voicemail by pressing voicemail feature code on your phone or log in MyPBX SOHO 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 MyPBX SOHO by Extension User Account.
   Before logging in MyPBX SOHO Web using the extension User account, you should enable “User Web Interface” for the extension.

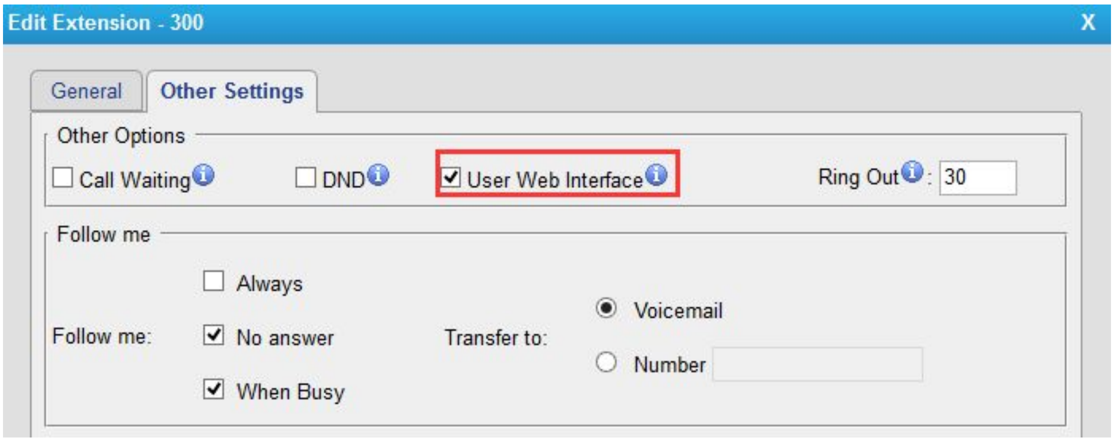


Figure 14-3 Enable User Interface

User Name: Extension Number (i.e. 601)

Password: Voicemail Access PIN, the default password is the same as the extension number. (i.e. 601)

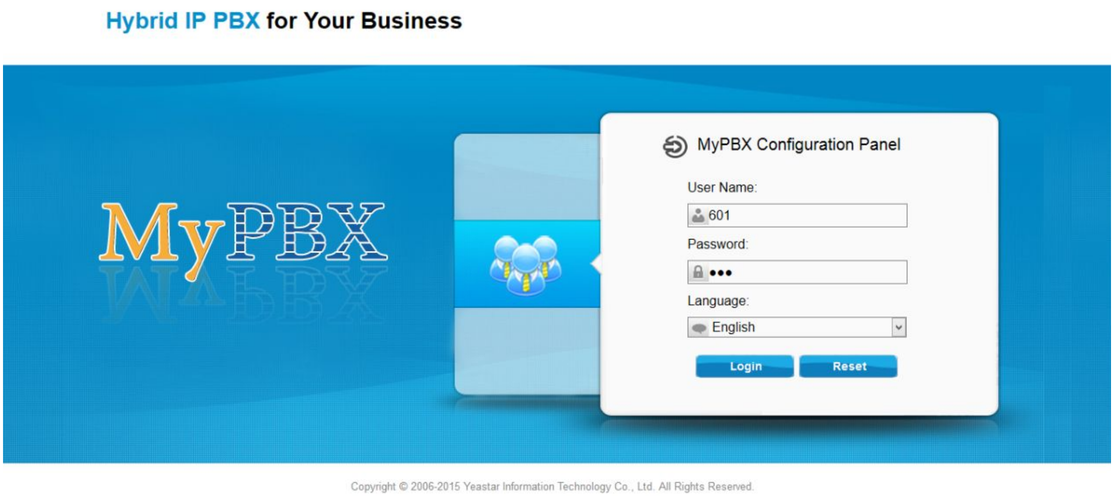


Figure 14-4 Login MyPBX SOHO via Extension Account

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 MyPBX 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 greeting.
8. Press 5 to change your password.
9. Choose the message that you want to record, press # to finish the record.
10. Press 1 to accept your message.
11. Press 2 to listen to your message.
12. Press 3 to re-record your message if you don't like the previous message.

Call Features

This chapter shows various call features on MyPBX SOHO:

- Feature Code
- Call Transfer
- Call Pickup
- Spy
- Call Parking
- Speed Dialing
- PIN User
- Callback
- DISA
- DNIS

Feature Code

Feature Codes are used to enable and disable certain features available in MyPBX SOHO. MyPBX SOHO 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

Table15-1 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: 4000 ms. The timeout to input the next digit.

2) Call Parking Preferences

Table 15-2 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 15-3 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 Forward All Calls	Default Code: *71.
Disable Forward All Calls	Default Code: *071.

Enable Forward When Busy	Default Code: *72.
Disable Forward When Busy	Default Code: *072.
Enable Forward No Answer	Default Code: *73.
Disable Forward No Answer	Default Code: *073.
Forward to Number	Default Code: *74.
Forward to Voicemail	Default Code: *074.
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 MyPBX SOHO: 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, user 501 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 MyPBX SOHO. 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.

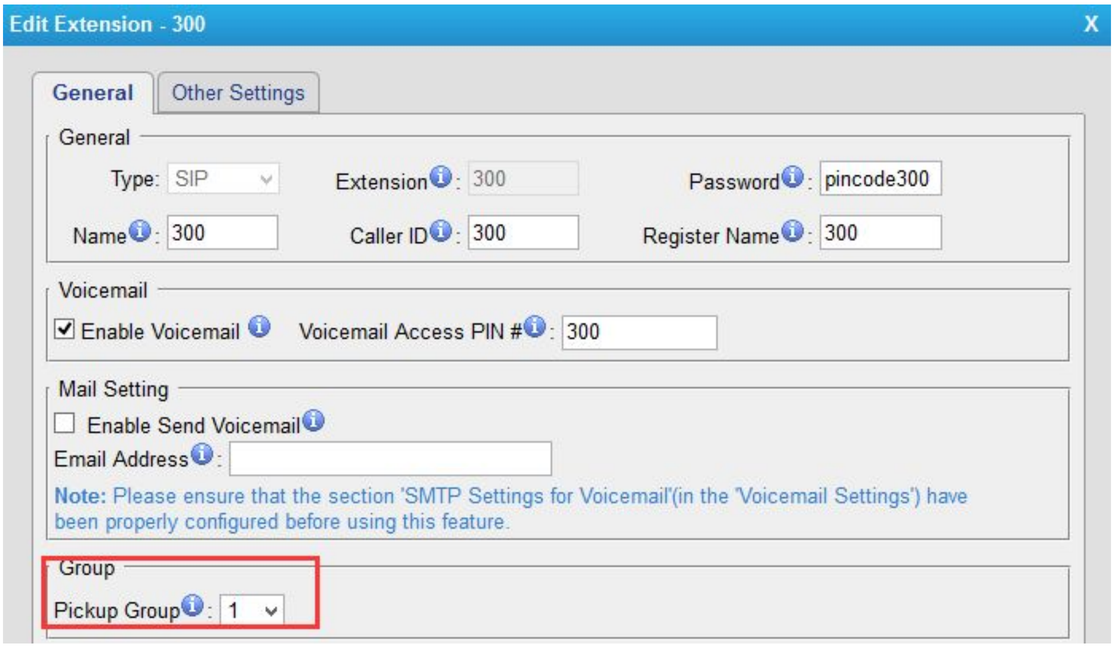


Figure 15-1 Group Extensions

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.

Spy

MyPBX SOHO 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 Extension 100 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 MyPBX SOHO 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 DialSettings**, 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 Add Speed Dial 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.

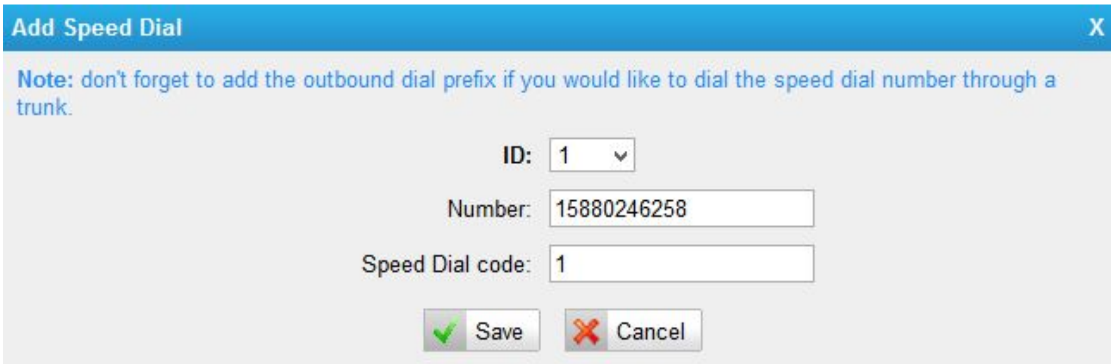


Figure 15-2 Speed Dial

To make a speed dial, e.g. you want to call 15880246258, simply dial *991. The *99 tells MyPBX SOHO 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.)

PIN User

Image the scenario:

A company's manager visits the factory and needs to make an international call to confirm something important with a foreign customer. However, all the extensions assigned to users in factory have no permission to make international calls (factory extensions are not selected in the international outbound route).

In this case, PIN User feature on MyPBX can help the manager to make international calls from an extension which is not selected on the international outbound route. He can dial PIN User feature code (default *89) on the phone first, then enter a PIN code following by the prompt. If the PIN code is correct, he could make international calls on the IP phone placed in factory.

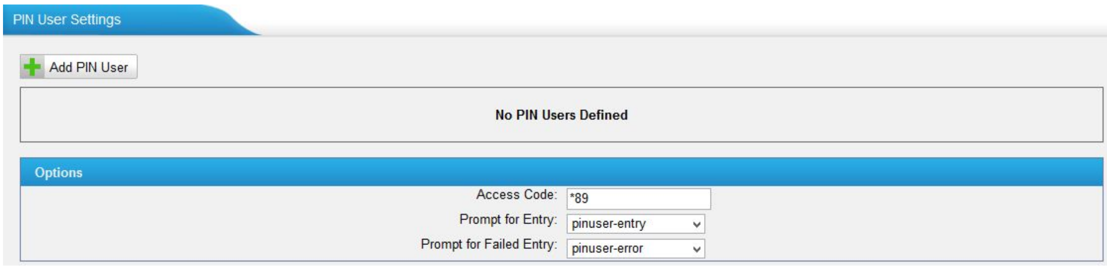


Figure 15-3 PIN User

- **Access Code**
  Dial this code to enable PIN User feature.
- **Prompt for Entry**
  Choose a prompt to ask the user to enter a PIN code.
- **Prompt for Entry Failure**
  Choose the error prompt to play when the user enters a wrong PIN code.

Click Add PIN User to add a PIN User.

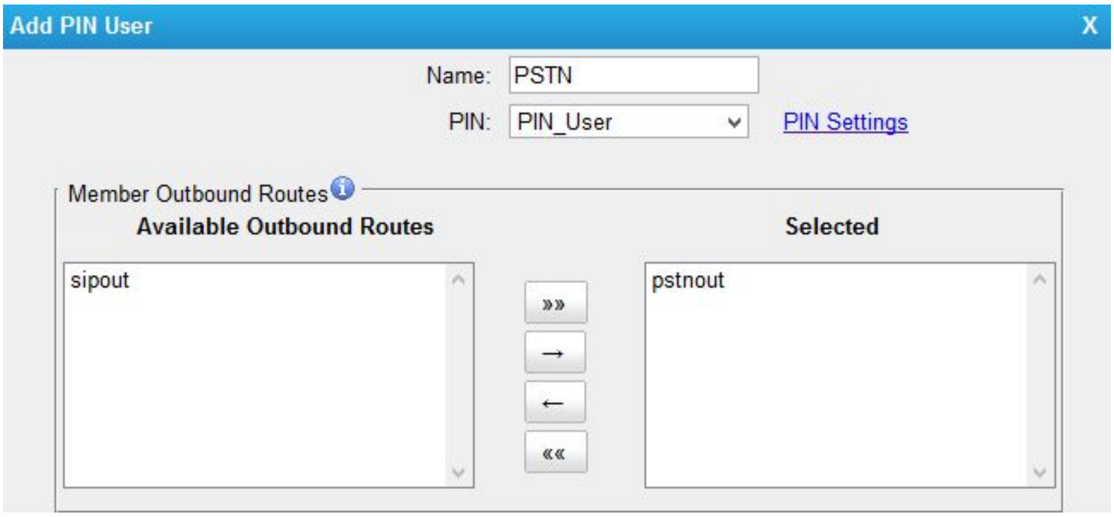


Figure 15-4 Add PIN User

- **Name**
  Set a name for the PIN user.
- **PIN**
  Select PIN lists from PIN settings.
- **Member Outbound Route**

Choose outbound routes for the PIN user. Users who are not in the outbound routes could use the outbound route to make external calls by “PIN User”.

Callback

Callback feature allows callers to hang up and get called back to MyPBX. Callback feature could reduce the cost for the users who work out of the office using their own mobile phones.

➢ **Enabling Callback**

Callback requires you to enable it on an inbound route. When you call in MyPBX through the inbound route and hang up the call. MyPBX will call you back and direct you to the selected destination on the inbound route.

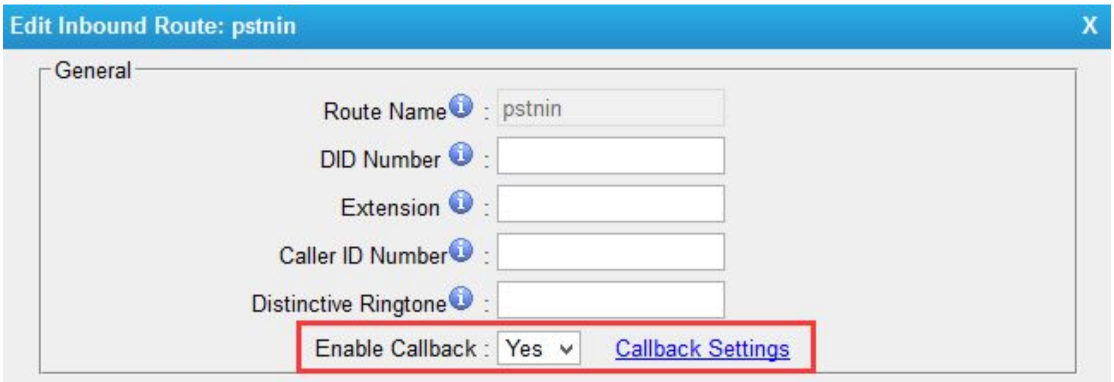


Figure 15-5 Enable Callback

➢ **Callback Settings**

Callback can be configured on **PBX→Advanced Settings→Callback Settings** page.

Notes:

1. If you'd like to use callback feature, please make sure it's enabled on the inbound route setting panel.
2. No callback rules needed to be set if the trunk supports call back with the caller ID directly.

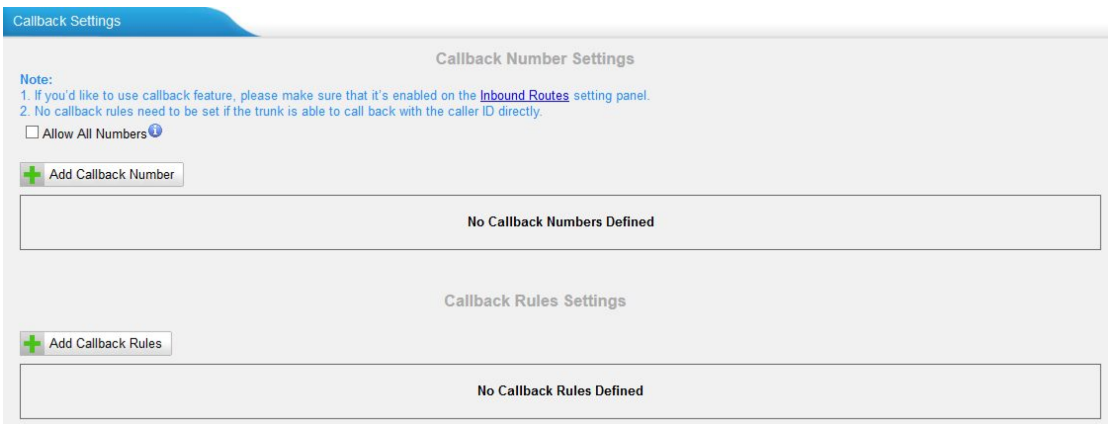


Figure 15-6 Callback Settings Page

- **Allow All Numbers**

  If you want to apply Callback function to all incoming numbers, please tick Allow All numbers.

- **Add Callback Number**

  Fill in the caller's Caller ID here, meaning the caller ID is allowed to use call back feature.

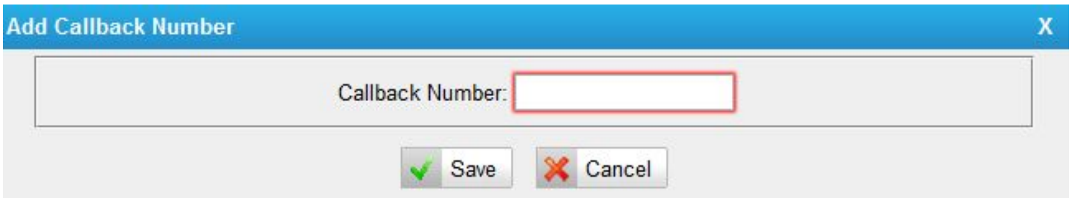


Figure 15-7 Add Callback Number

- **Add Callback Rules**

  You will need to create callback rules when the system should strip or add digits.

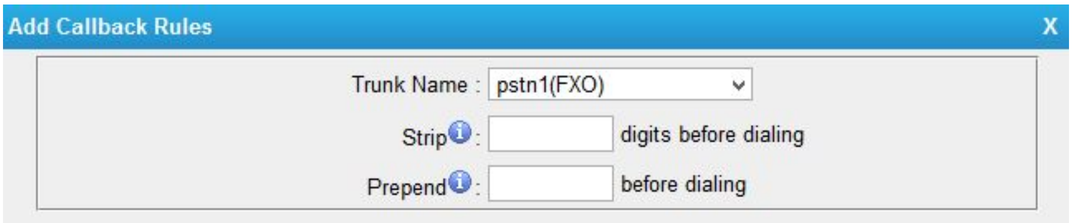


Figure 15-8 Add Callback Rules

Table 15-4 Callback Rules

Callback Rules	
Trunk Name	Choose the trunk for callback rules.
Strip	Define how many digits will be stripped from the call in number before the callback is placed. For example, when you call from number 123456789 into MyPBX, the caller ID is 0123456789, but you can only call 123456789 successfully from MyPBX trunk. You should configure number 0123456789 as the call back number and strip 1 digit before the callback is placed.
Prepend	Define digits added before a callback number before the callback is placed. For example, the call in number (Caller ID) is 123456789, MyPBX need to send 9123456789 to its trunk when calling this number. You should configure 123456789 as the call back number and add 9 before the callback is placed. You can add "w" for analog trunks for some delay too.

DISA

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

➢ **Adding a DISA**

Go to **PBX→Advanced Settings→DISA** to add one DISA.

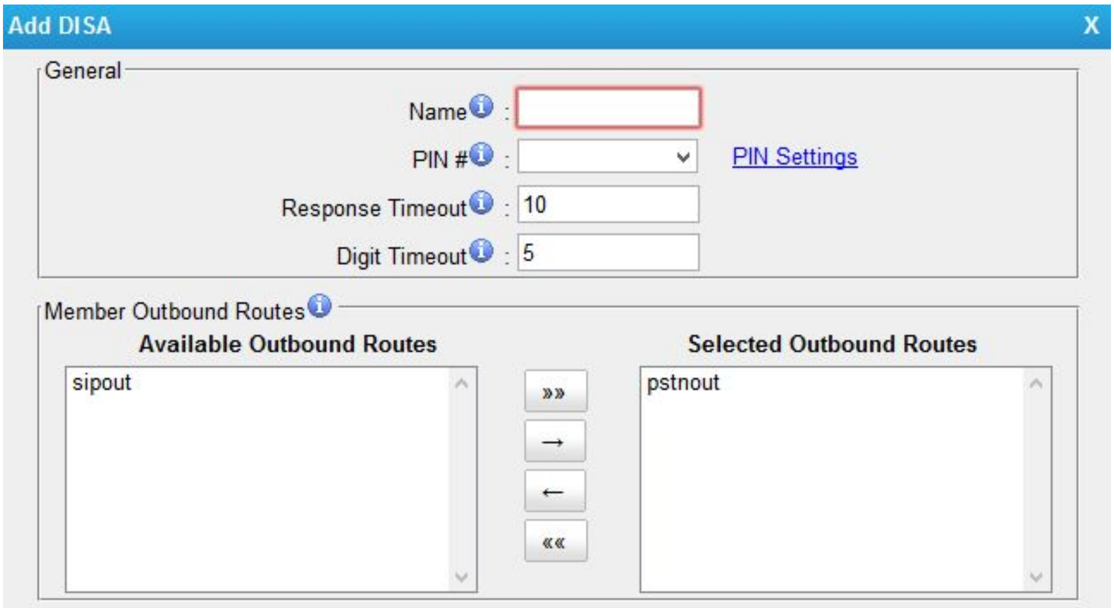


Figure 15-9 Add DISA

- **DISA Name**
  Give this DISA application a name to help you identify it.
- **PIN**
  A password is required if you want use DISA feature. Select a PIN list from PIN Settings.
- **Response Timeout**
  The maximum amount of time the system will wait before hanging up the call if the user has dialed an incomplete or invalid number. The default is 10 seconds.
- **Digit Timeout**
  The maximum amount of time permitted between each digit when the user is dialing an extension number. The default is 5 seconds.
- **Member Outbound Routes**
  Used to set the outbound routes that can be access from this DISA.

➢ Applying to an Inbound Route

A DISA can be selected as a destination in an inbound route. Once the caller call in MyPBX through the inbound route, he/she will get a dial tone or a dial tone after inputting a correct PIN code, the caller are able to use trunks on MyPBX to make external calls.

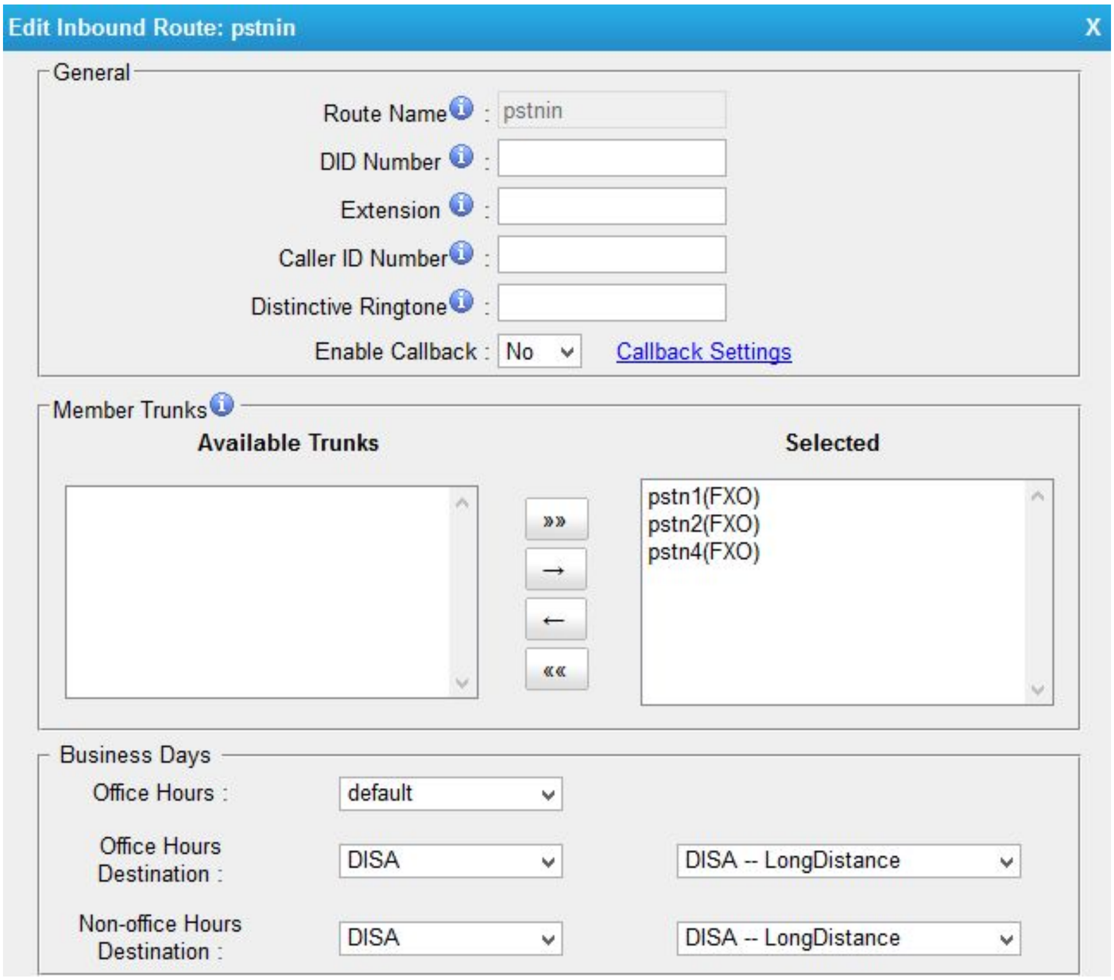
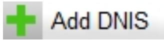


Figure 15-10 Choose DISA on Inbound Route

DNIS

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

Users could configure DNIS to allow the IP phones to display which trunk is passing the call. Suppose a company has two trunks, one for sales service, one for support service. To make a difference between two kinds of calls, we can configure two DNIS for the two trunks.

Log in MyPBX and go to **PBX→Advanced Settings→DNIS Settings**, click 

to create a DNIS.

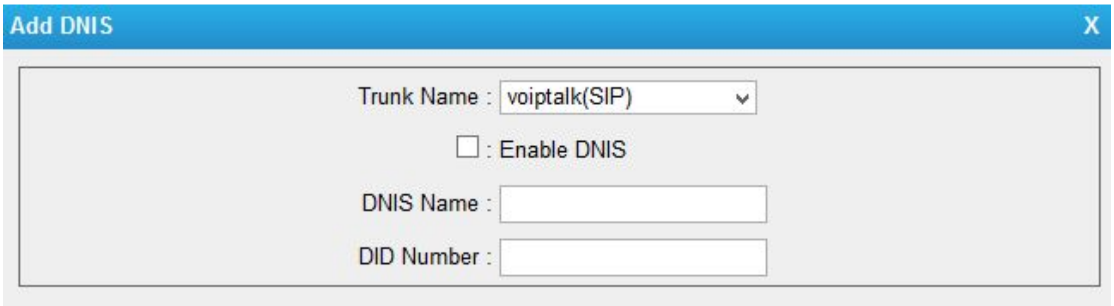


Figure 15-11 Add DNIS

- **Trunk Name :** choose a Trunk for the DINS.
- **Enable DNIS:** tick to enable DNIS feature.
- **DNIS Name:** give a name for this DNIS, when a call reaches the selected trunk, the name will be displayed on the ringing phone.
- **DNIS Number:** DID number will be needed for SIP trunk, BRI trunk, because these trunks may have multiple numbers, and multiple DNIS can be set for one SIP trunk.

Notes:

1. PSTN trunk do not need to set DID number.
2. If you do not set the DID number, all calls through selected VoIP/BRI trunk will show the DNIS name as Caller Name.

PBX Basic Settings

This chapter explains PBX basic settings, which can be applied globally to MyPBX SOHO. 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 Duration	The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout. The default value is 6000s.
Maximum Concurrent Calls	Maximum concurrent calls limits. The default value 0 means no limit.
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, 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.
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

MyPBX SOHO supports HTTP and HTTPS protocol. 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.

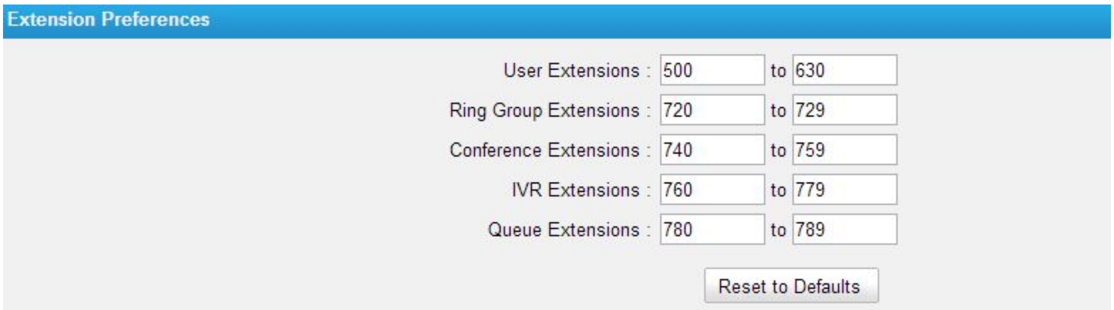


Figure 16-2 Extension Preferences

Business Hours

On Business Hours page, you can create time groups 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. Go to **PBX→Basic Settings→Business Hours** to find Business Hours settings.

- **Enable Business Hours**: the calls will be routed to the specific destinations against a time group.

- **Disable Business Hours**: the calls will always be routed to the "Day Destination".

Day/Night Control

This feature enables designated users (usually the company operator or receptionist) to override the normal routing of calls based on the time of day. For example, the system may be configured to direct calls during office hours directly to the receptionist and to route calls to voice menu after hours. Using the day/night control, the receptionist can override this time settings and force all calls to follow the setting as if the office were closed. This override is usually used if the receptionist needs to be away unexpectedly or if the office needs to close due to poor weather.

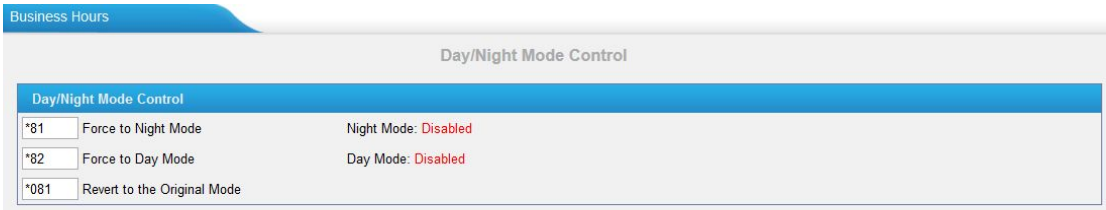


Figure 16-3 Day/Night Control

- **Force to Night Mode (*81)**
  The system starts in day. To switch to night mode, dial the feature code *81. All incoming callers will now be directed to the "Night Destination".
- **Force to Day Mode (*82)**
  Dial *82 to switch to day mode (office open mode), all calls will go to "Day Destination".
- **Revert to the Original Mode (*081)**
  Dial *081 to revert to the original mode, calls will go to the relevant destinations according to the office time.

Time Groups

Time Group is used to define periods of time that can then be selected in the Inbound Time Conditions or Outbound Routes.

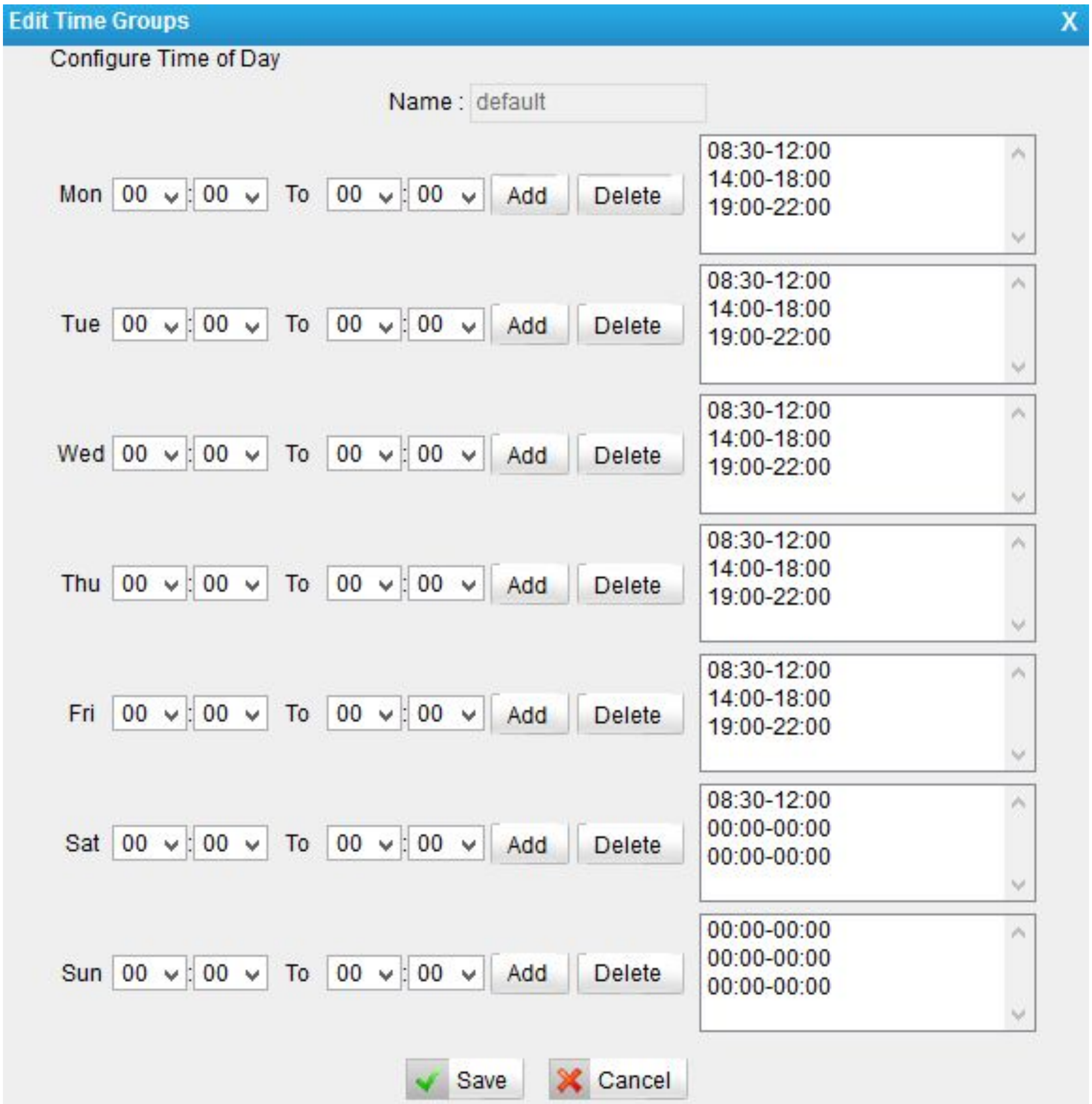


Figure 16-4 Time Groups

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 Holiday.

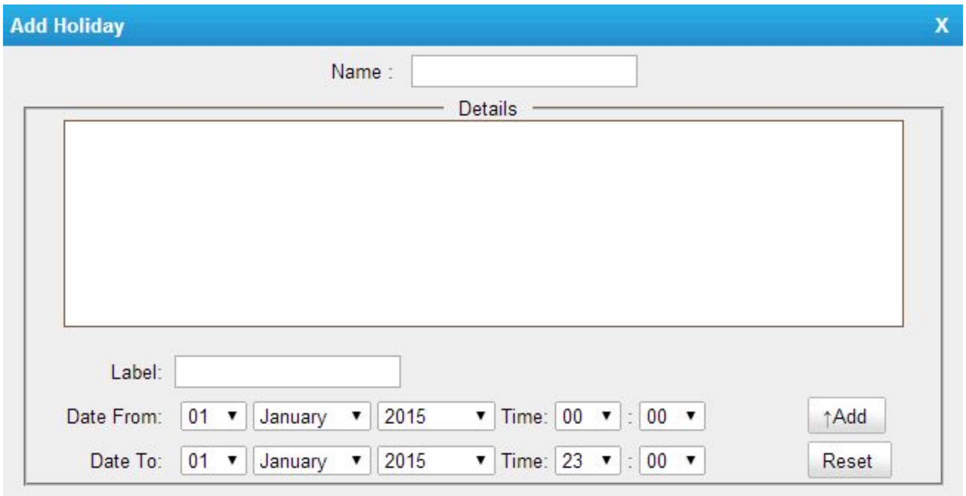


Figure 16-5 Holidays

SIP & IAX Settings

SIP settings and IAX settings can be found via **PBX→Advanced Settings**. It is wise to leave the default setting as provided on this page. However, for a few fields, you need to change them to suit your situation.

- SIP Settings
- IAX Settings

SIP Settings

1) General Settings

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 5061.
TLS Verify Server	When using MyPBX as a TLS client, whether or not to verify server's certificate. It is "No" by default.
TLS Verify Client	When using MyPBX as a TLS server, whether or not to verify client's certificate. It is "No" by default.
TLS Ignore Common Name	Set this parameter as "No", then common name must be the same with IP or domain name.
TLS Client Method	When using MyPBX as TLS client, specify the protocol for outbound TLS connections. You can select it as tlsv1, sslv2 or sslv3.
RTP Port	Set RTP port range.
DTMF Mode	Set default mode for sending DTMF. Default setting: rfc2833.
Max Registration/Subscription Time	Maximum duration (in seconds) of a SIP registration. The default is 3600 seconds.
Min Registration/Subscription Time	Minimum duration (in seconds) of a SIP registration. The default is 60 seconds.
Default Incoming/Outgoing Registration Time	Default Incoming/Outgoing Registration Time: Default duration (in seconds) of incoming/outgoing registration.
Register Attempts	The number of SIP REGISTER messages to send to a SIP Registrar before giving up. Default is 0 (no limit).
Register Timeout	Number of seconds to wait for a response from a SIP Registrar

	before considering the register has timed out. The default is 20 seconds.
Calling Channel Codec Priority	Once enabled, when dialing out via SIP/SPS trunks, the codec of calling channel will be selected in preference. If not, MyPBX will follow the priority in your SIP/SPS trunks.
Video Support	Support for SIP video or no. The default is yes.
Max Bit Rate	Configure the max bit rate for video stream. The default: 384kb/s.
DNS SRV Look Up	Please enable this option when your SIP trunk contains more than one IP address.
User Agent	To change the user agent parameter of asterisk, the default is “MyPBX”; you could change it if needed.

2) NAT Settings

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.

3) Codecs

A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet.
MyPBX SOHO supports G711 a-law, u-law, GSM, SPEEX, G722, G726, G729A, ADPCM, MPEG4, SPEEX, H261, H263, H263P, H264.

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".

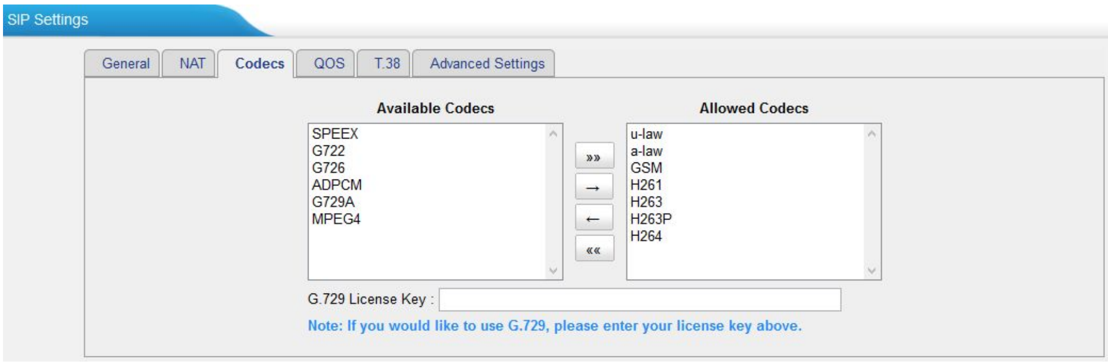


Figure 17-1 SIP Settings-Codecs

4) QoS

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

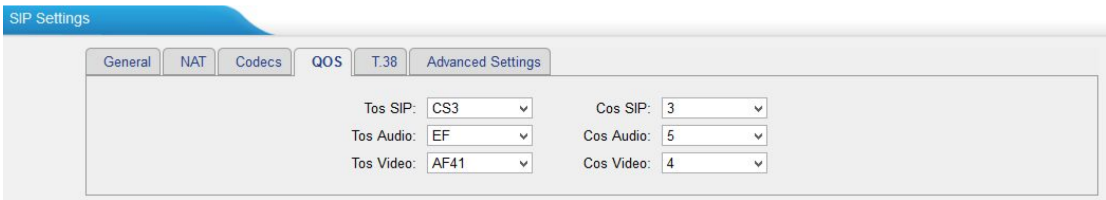


Figure 17-2 QoS Settings

5) T.38

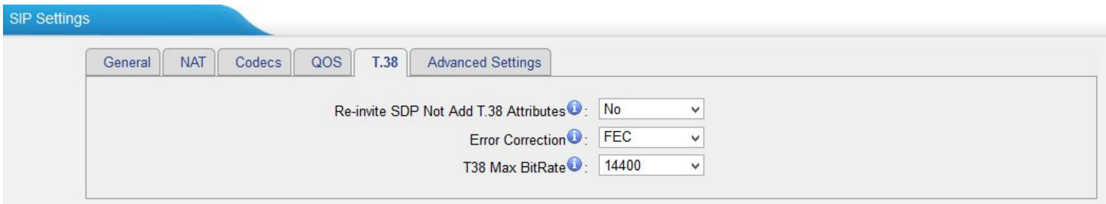


Figure 17-3 T.38 Settings

- **Re-invite SDP Not Add T.38 Attributes**
  If set to Yes, SDP in re-invite packet will not add T.38 attributes.
- **Error Correction**
  Set the Error Correction Mode (ECM) for the Fax.
- **T38 Max BitRate**
  Set the maximum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000 and 14400. The default setting is 14400.

6) Advanced Settings

Table 17-2 SIP Advanced Settings

Advanced Settings	
From Field	Where to get the caller ID in SIP packet.
To Field	Where to get the DID in SIP packet.
180 Ringing	It is set when the telecom provider needs. Usually it is not needed.
Remote Party ID	Whether to send Remote-Party-ID on SIP header or not. Default: no.
Allow Guest	Whether to allow anonymous registration extension or not. Default: no. This option is used to avoid some anonymous calls by hackers.
Pedantic	Enable pedantic parameter. Default: no.
Alwaysauthreject	If enabled, when MyPBX rejects “Register” or “Invite” packets, MyPBX always respond the packets using “SIP404 NOT FOUND”.
OPTIONS Response 200	If set to yes, the response to an OPTIONS is always 200 OK.
Session timers	Enable session-timer mode, default: yes.
Session-expires	The max refresh interval.
Session-minse	The min refresh interval, which mustn't be less than 90s.
Session-refresher	Choose session-refresher, the default is Uas.

IAX Settings

1) General

Table 17-3 IAX General Settings

General Settings	
Bind Port	Port used for IAX2 registrations. The default setting is 4569.
Bandwidth	Low/medium/high with this option you can control which codec to be used.
Min Registration Time	Minimum duration (in seconds) of an IAX2 registration. The default settings is 60 seconds.
Max Registration Time	Maximum duration (in seconds) of an IAX2 registration. The default setting is 1200 seconds.

2) Codecs

A codec is a compression or decompression algorithm used in the transmission of voice packets over a network or the Internet.
Tick the options to choose allowed codecs.

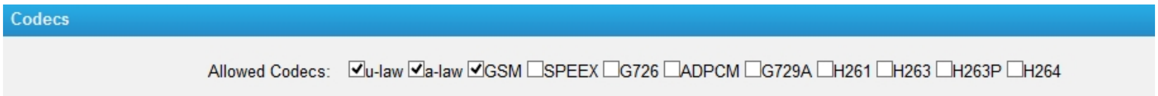


Figure 17-4 Codecs

Network Settings

This chapter explains system settings on MyPBX SOHO. Click the main menu 

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

- LAN Settings
- DHCP Server
- VLAN Settings
- VPN Settings
- DDNS Settings
- Static Route

LAN Settings

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

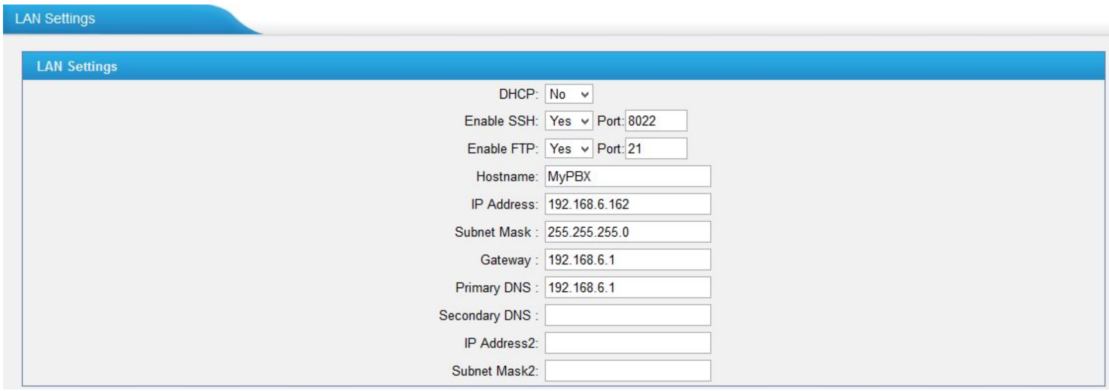


Figure 18-1 LAN Settings

Table 18-1 LAN Settings

Items	Description
DHCP	If this option is set as yes, MyPBX SOHO 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 MyPBX SOHO.
Enable SSH	By using SSH, you can log in to MyPBX SOHO 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 MyPBX SOHO via FTP if this option is enabled. Users could access FTP resource on MyPBX

	SOHO via Windows explorer or Web browser. FTP default user: **root**, password: **ys123456** Default Port: 21.
Hostname	Set the host name for MyPBX SOHO.
IP Address	Set the IP Address for MyPBX SOHO.
Subnet Mask	Set the subnet mask for MyPBX SOHO.
Gateway	Set the gateway for MyPBX SOHO.
Primary DNS	Set the primary DNS for MyPBX SOHO.
Secondary DNS	Set the secondary DNS for MyPBX SOHO.
IP Address2	Set the second IP Address for MyPBX SOHO.
Subnet Mask2	Set the second subnet mask for MyPBX SOHO.

DHCP Server

Dynamic Host Configuration Protocol (DHCP) is a network protocol that enables a server to automatically assign an IP address to a computer from a defined range of numbers (i.e., a scope) configured for a given network. You can set a local network NTP server for MyPBX here, too.

Note: MyPBX SOHO can work as a DHCP server, but cannot act as a router.

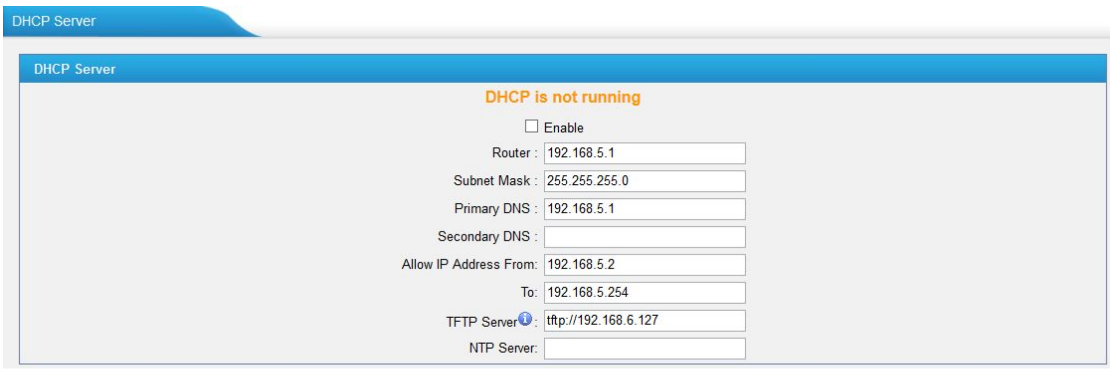


Figure 18-2 DHCP Server

VLAN Settings

VLAN (Virtual Local Area Network) is a group of hosts with a common set of requirements, which communicate as if they were attached to the same broadcast domain, regardless of their physical location.

A VLAN is a broadcast domain created by switches. This means the VLAN is configured on switches, layer 3 switches. Note that some of the switches don't support VLAN.

When do I need a VLAN?

It is important to point out that you do not have to configure a VLAN until your network gets so large and has so much traffic that you need one. You need to consider using VLAN in any of following situations:

- You have more than 200 devices on your LAN.
- You have a lot of broadcast on your LAN.
- Groups of users need more security or are being slowed down by too many broadcasts.
- Groups of users need to be on the same broadcast domain because they are running the same applications. An example would be a company that has a lot of VoIP phones connect to an IPPBX. The administrator of the network might like to separate the VoIP from the network and configure VLAN for IP phones and PCs.

Note:

MyPBX SOHO acts as a VLAN client, a 3-layer switch is needed.

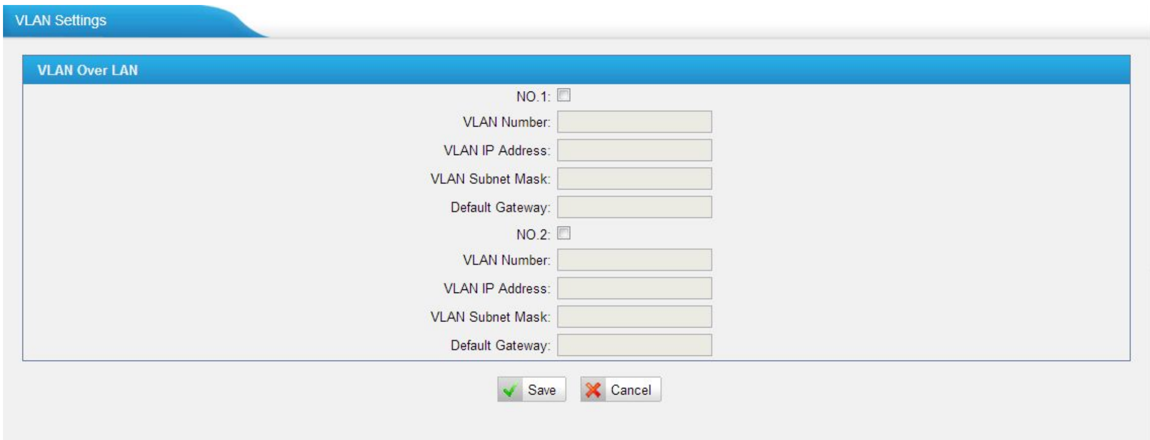


Figure 18-3 VLAN Settings

Please follow the steps below to set up VLAN on MyPBX.

Step1. Create VLANs on your switch.

Step2. Allocate a VLAN ID and IP address for MyPBX.

Step3. Configure VLAN settings page on MyPBX.

VPN Settings

OpenVPN is a free and open source software application that implements virtual private network (VPN) techniques for creating secure point-to-point or site-to-site connections in routed or bridged configurations and remote access facilities. It uses SSL/TLS security for encryption and is capable of traversing network address translators (NATs) and firewalls.

MyPBX supports OpenVPN, IPSec and L2TP.

Select one type of VPN and import VPN profile to MyPBX. If upload successfully, you could check the network status under “Status” menu. If it shows disconnect you may try reboot to take effect.

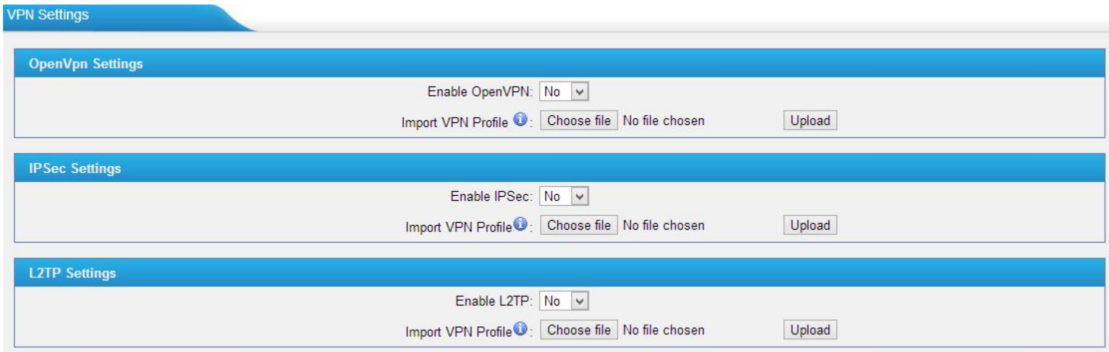


Figure 18-4 VPN Settings

Note: for more details about the above VPN settings, please contact our technical support.

DDNS Settings

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

MyPBX supports the following DDNS providers:

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

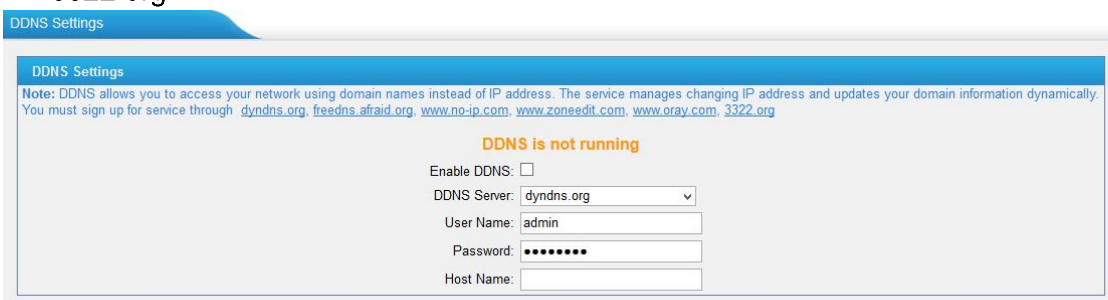


Figure 18-5 DDNS Settings

Static Route

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

The default gateway priority of MyPBX from high to low is OpenVPN → WAN port→LAN port.

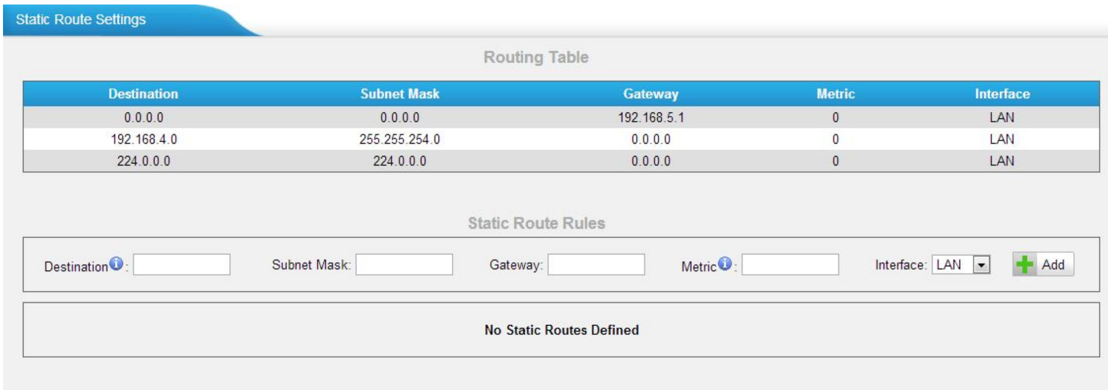


Figure 18-6 Static Route

Table 18-2 Static Route Settings

Items	Description
Destination	Set the destination IP address or IP subnet for MyPBX to reach using the static route. Example: IP address: 192.168.6.34 IP subnet: 192.168.6.0
Subnet Mask	Set the subnet mask for the destination IP address.
Gateway	Set the gateway that MyPBX will reach the destination via this gateway.
Metric	The cost of a route is calculated by using what are called routing metric. Routing metrics are assigned to routes by routing protocols to provide measurable statistic which can be used to judge how useful (how low cost) a route is.
Interface	Select which Internet port to go through.

MyPBX Security

This chapter describes how to secure MyPBX. Users are strongly recommended to configure firewall and other security options on MyPBX SOHO to prevent the attack fraud and the system failure or calls loss.

- Security Center
- Firewall Rules
- IP Blacklist
- AMI Settings
- Database Grant
- Alert Settings
- Certificates

Security Center

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

1) 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 19-1 Security Center—Firewall

2) 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.

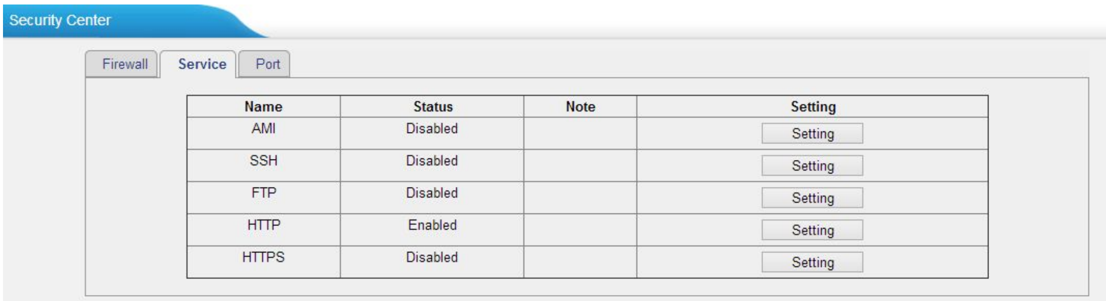


Figure 19-2 Security Center—Service

3) 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.

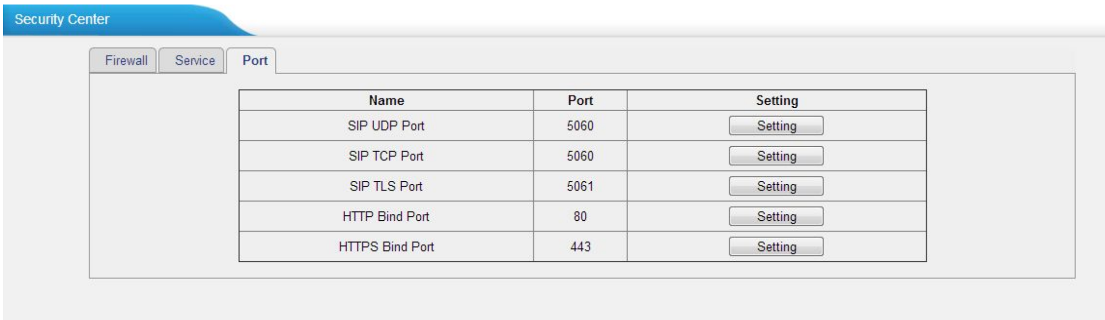


Figure 19-3 Security Center—Port

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 19-4 Firewall Settings

1) General Settings

Table 19-1 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.

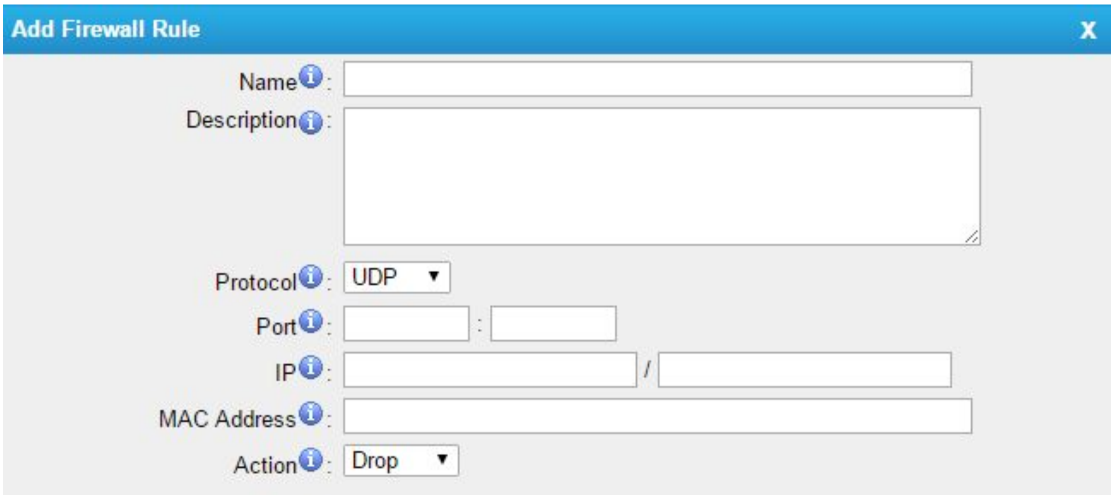


Figure 19-5 Common Rules

Table 19-2 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 192.168.5.0 to 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

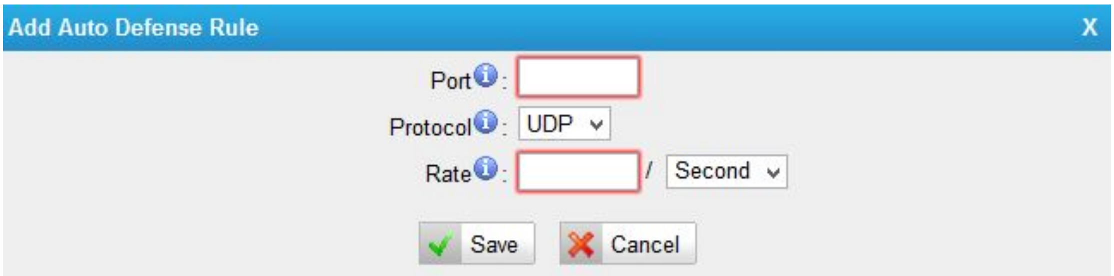


Figure 19-6 Auto Defense

Table 19-3 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.

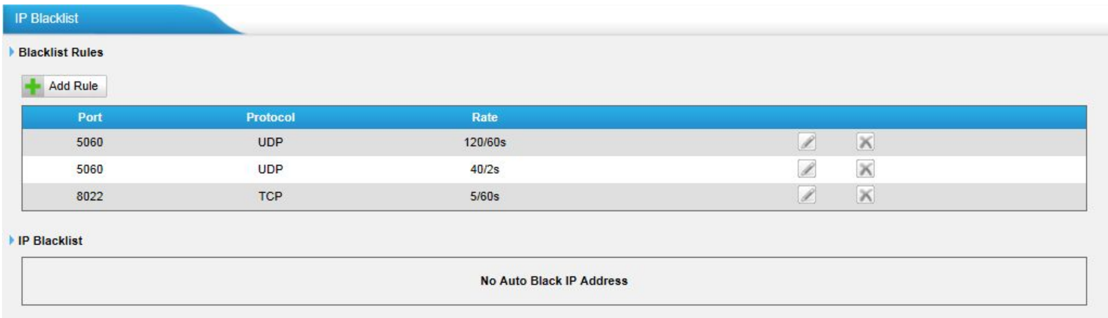


Figure 19-7 IP Blacklist Settings Page

1) Blacklist rules

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

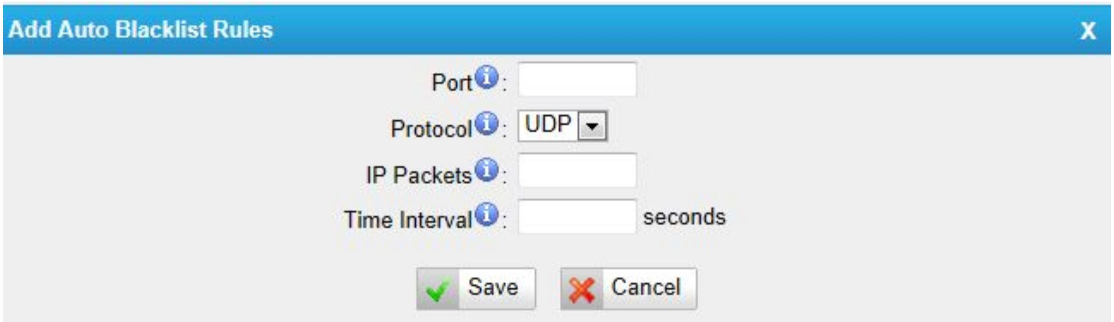


Figure 19-8 Add Blacklist Rule

Table 19-4 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 MyPBX SOHO using AMI interface. It is disabled by default. If necessary, you can enable it.

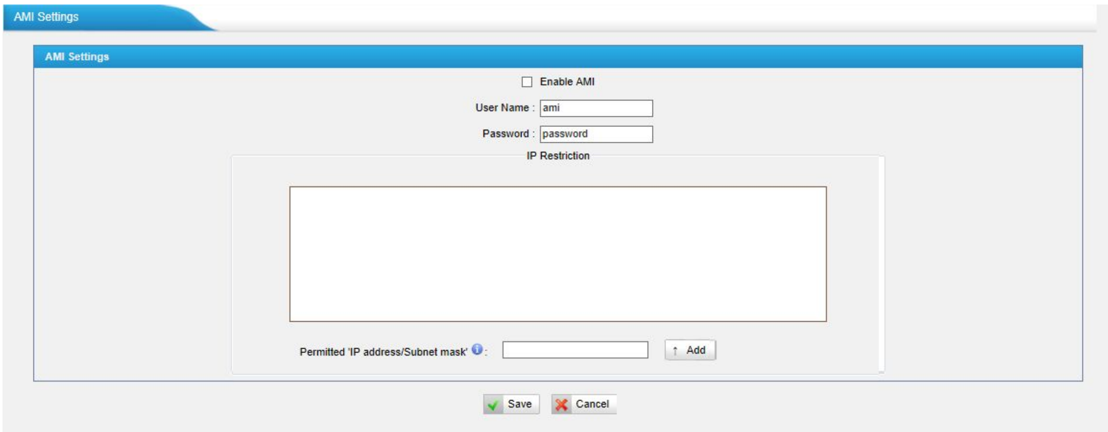


Figure 19-9 AMI Settings

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

Database Grant

MyPBX SOHO are using MySQL database. The 3rd party software can access MySQL via internet. Before that, you need to grant the authority to the database user.

Go to "Database Grant" page, click 

, you can add a database user, set user password and grant authority.

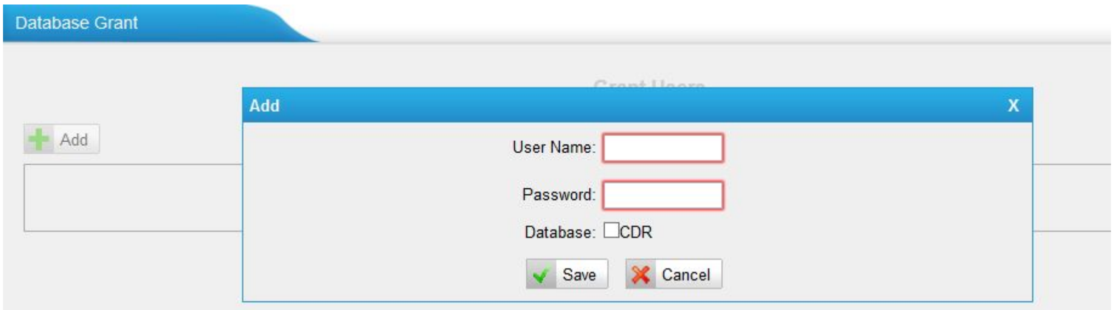


Figure 19-10 Database Grant

Username/password: The 3rd party can use this username and password to access the MySQL.

Database: check CDR, then this user has authority to check CDR database.

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 19-11 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 19-5 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 19-6 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)

Certificates

MyPBX supports TLS extension. Before you register a TLS extension on IP phone, you should upload certificates first.

Figure 19-12 Upload Certificate

- **Trusted Certificate**
  This certificate is a CA certificate. When selecting “TLS Verify Client” as “Yes”, you should upload a CA. The relevant 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 MyPBX. If IP phone enables “TLS Verify server”, you should also upload the relevant CA certificate on IP phone.

System Settings

This chapter describes system maintenance settings including the followings:

- LDAP Server
- External Storage
- Password Settings
- Date and Time
- Firmware Upgrade
- Backup and Restore
- Reset and Reboot

LDAP Server

LDAP is used as a phone book on MyPBX so that you can search a key word from your IP phone. The key word can be a name, a mobile number, an email or other key words in the phonebook.

Note: it requires that the IP phone should support LDAP feature.

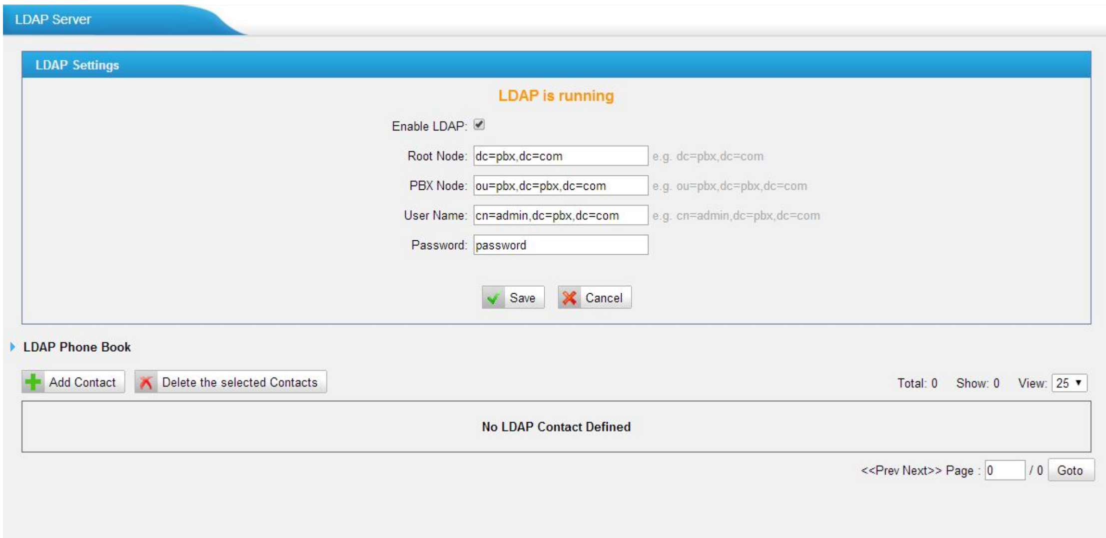


Figure 20-1 LDAP Server

- **LADP Settings**

Table 20-1 LDAP Settings

Items	Description
Enable LDAP	Enable LDAP to use LDAP on your IP phone.
Root Node	A root node for this LDAP, e.g. dc=pbx, dc=com.
PBX Node	A pbx node for this LDAP, e.g. ou=pbx, dc=pbx, dc=com.
User Name	A user for this LDAP, e.g. cn=admin, dc=pbx, dc=com.

Password	A password used to access LDAP.

- **Add Contacts**

Click **Add Contact** to add contacts to the LDAP phonebook. If LDAP is enabled, the IP phone could access the LDAP phone book.
MyPBX SOHO supports up to 1000 contacts in LDAP phone book.

External Storage

The External Storage feature is used to extend storage space. Once configured, the files (voicemail, call recording files and call logs) created before the configured days will be moved to the Net-Disk.

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.

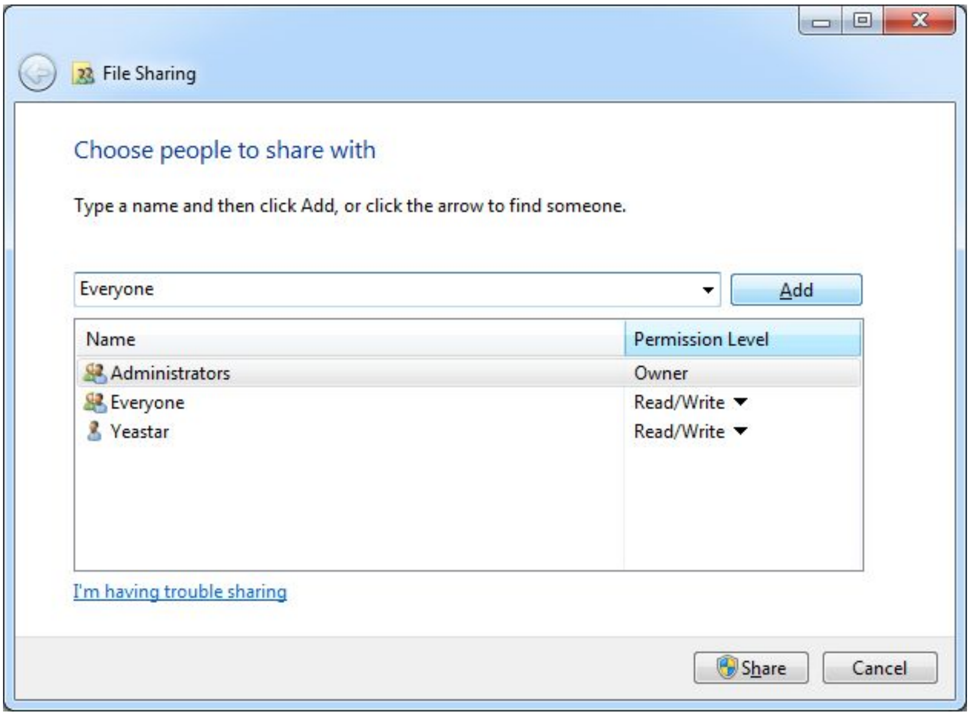


Figure 20-2 File Sharing

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

Step 1: Choose a window-based computer that is always in service.
Step 2: Create a folder.
Step 3: Create a text file named "status.txt" in the folder.

Step 4: Share this folder.

Step 5: Configure External Storage Settings on MyPBX.

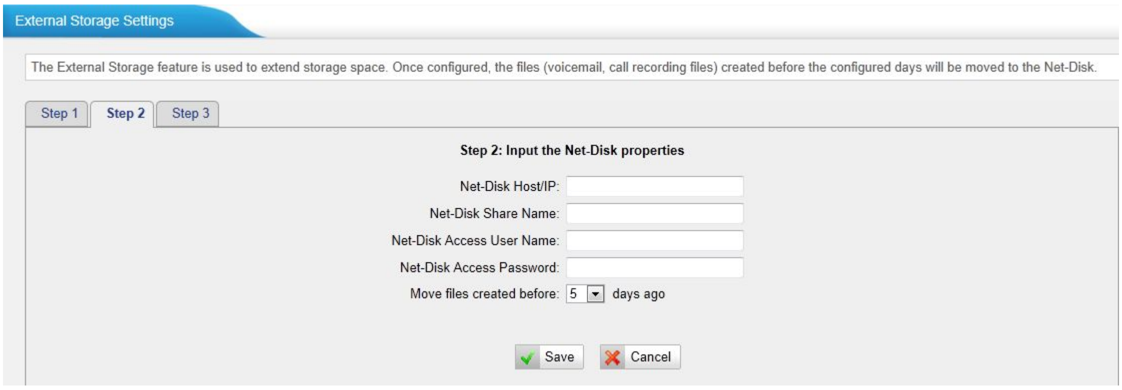


Figure 20-3 External Storage Settings

- **Net-Disk Host/IP**: set the IP address of the computer where backup files will be stored.
- **Net-Disk Share Name**: fill in the name of the shared folder where backups will be stored.
- **Net-Disk Share Username**: set the user name used to log into the network share. Leave this blank if it is not required
- **Net-Disk Share Password**: set the password used to log into the network share. Leave this blank if it is not required.

If the configuration is correct, MyPBX backup files and folders will be created in the shared folder.

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.

There are 3 accounts for MyPBX SOHO: "admin", "user" and "cdr". Accounts "user" and "cdr" are disabled by default.

Default password for the 3 accounts are all "password".

Follow the steps to 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 MyPBX SOHO 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 MyPBX SOHO using the new password.

Figure 20-4 Password Settings

Note: the administrator could change other accounts' password without entering the old password.

Date and Time

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

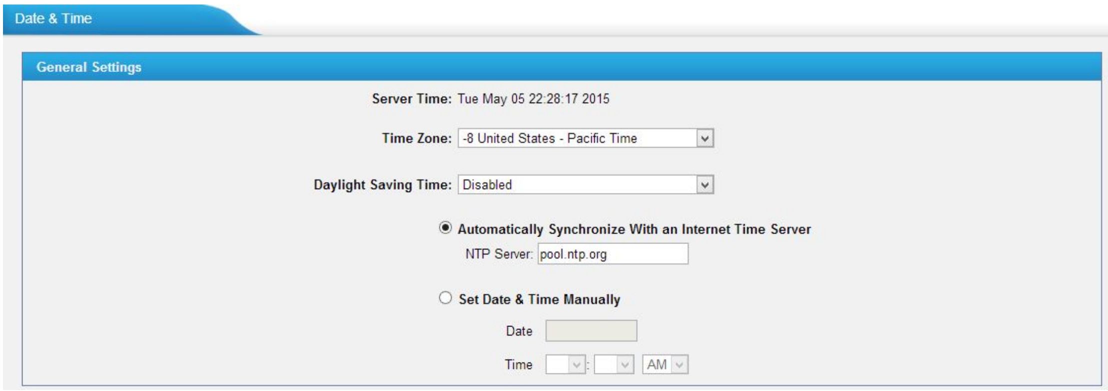


Figure 20-5 Date and Time

- **Time Zone**
  Select your current and correct time zone on MyPBX SOHO.
- **Daylight Saving Time**
  The option is disabled by default. Enable it when necessary.
- **Automatically Synchronize with an Internet Time Server**
  MyPBX SOHO will adjust its internal clock to a central network server. Please note the MyPBX SOHO 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.

Firmware Upgrade

MyPBX SOHO can be upgraded to a new firmware version via network or locally. Users could upgrade firmware via HTTP or TFTP. Please go to **System→ System Preferences→ Firmware Update** to do upgrade.

Notes:

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 be damaged.
3. If you are trying to upgrade through HTTP, please make sure that your MyPBX SOHO is able to visit external network, or it cannot access Yeastar website to get the firmware file, causing the upgrade fail.

Upgrade through HTTP

On the Firmware Upgrade page, choose **HTTP URL**.

Step1. Enter the download link of the firmware file.

Note: the HTTP URL should be a **BIN** file download link.

Step2. Click "Start" to upgrade.

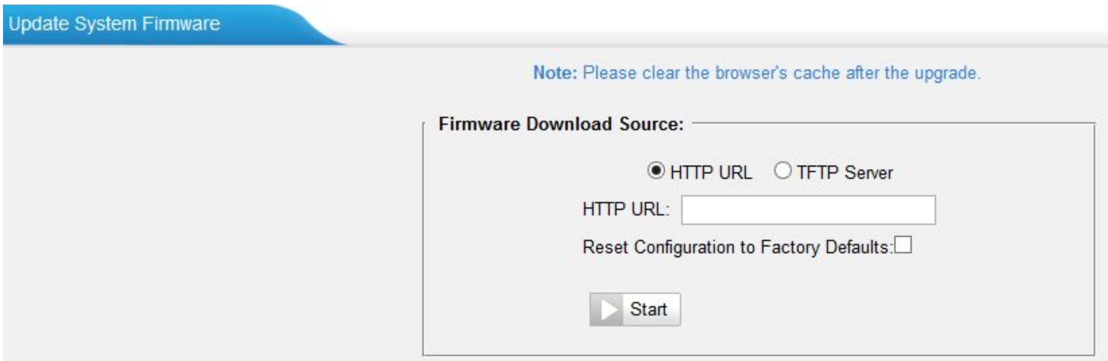


Figure 20-6 Upgrade through HTTP

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.

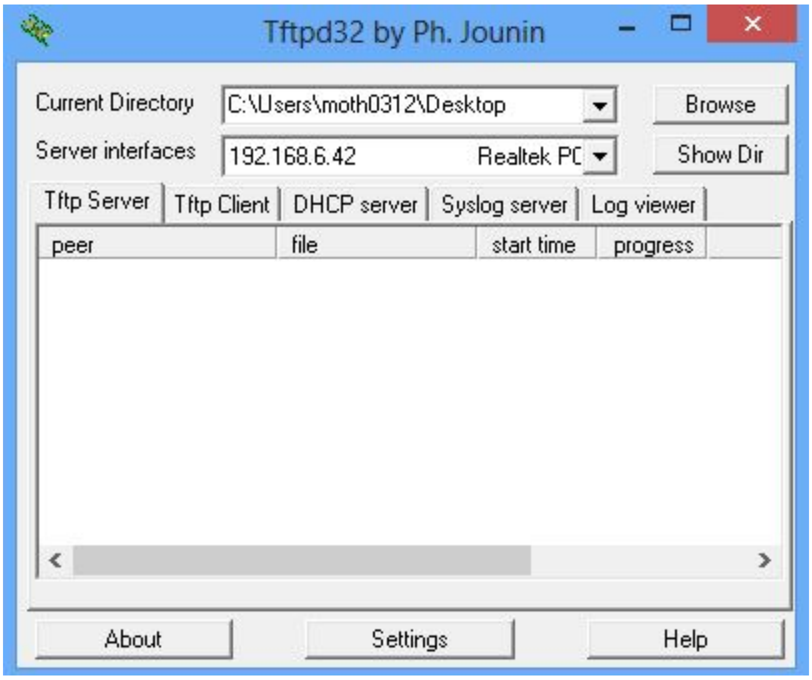


Figure 20-7 Configure Tftpd32

Step3. Logon the MyPBX SOHO's Web page and go to **System→System Preferences→ Firmware Update**, 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.

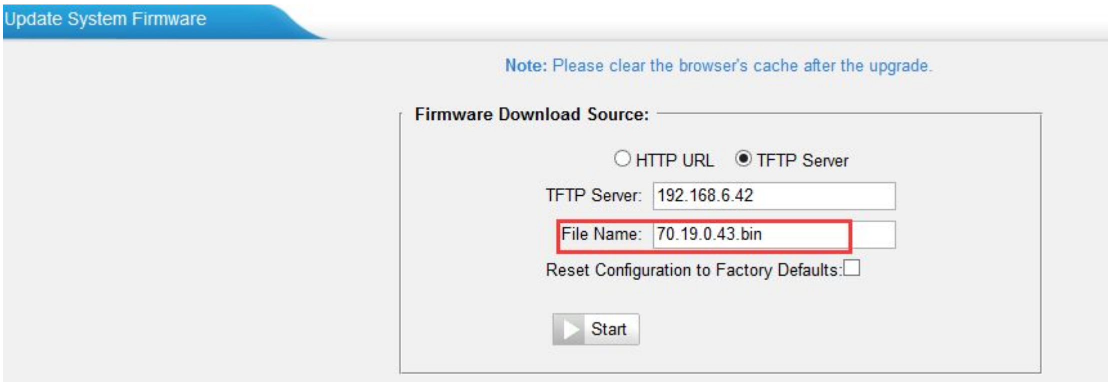


Figure 20-8 Upgrade through HTTP

Backup and Restore

MyPBX SOHO provides Backup and Restore feature, which allows you to create a complete backup of MyPBX SOHO 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 MyPBX SOHO of a later version.

- **Create a New Backup**

  Click Create a New Backup to create a new backup.
- **Upload a Backup**

  Click Upload a Backup to upload a backup.
- **Restore**

  To restore MyPBX SOHO configuration data, upload the backup file to MyPBX SOHO and click . Reboot the system to take effect.

  Please note the current configurations will be OVERWRITTEN with the backup data.

#	Name	Time	Options
1	backup_2015may9_174120.tar	Sat May 09 1:41:58 2015	

Figure 20-9 Restore Backup

Reset and Reboot

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

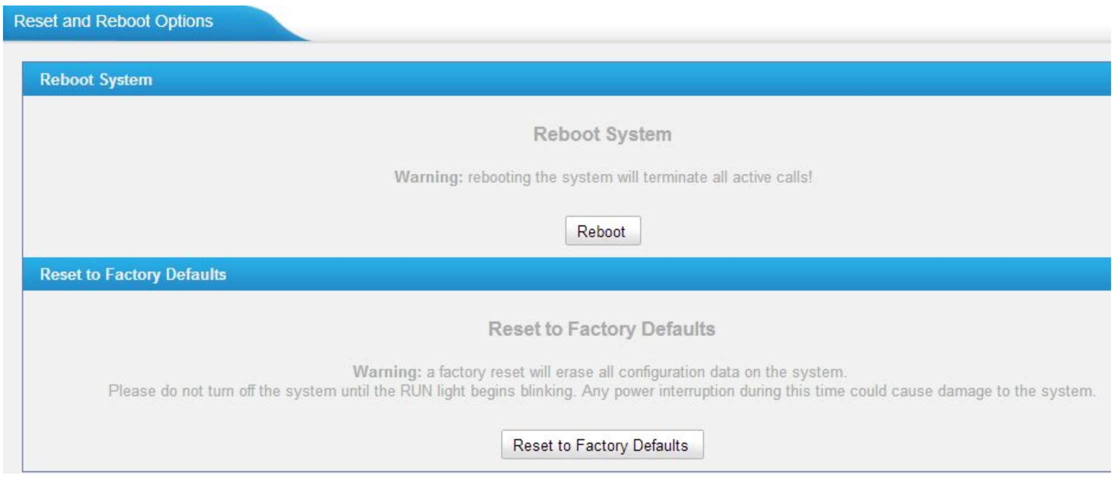


Figure 20-10 Reset and Reboot

Status

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

Extension Status

Users could view all the extension status on this page.

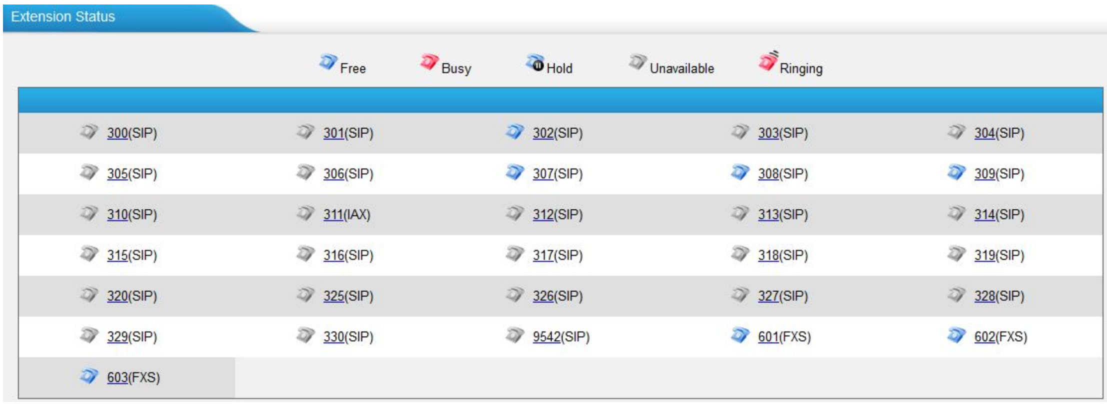


Figure 21-1 Extension Status

- Extension is unavailable
- Extension is idle
- Extension is ringing
- Extension is busy
- Extension is on hold

Trunk Status

Users could check all the trunks status on this page.

Trunk Status

Status	Signal	Trunk Name	Type	User Name	Port/Hostname/IP	Reachability
OK (19 ms)		N824	SP-SIP		192.168.6.141	OK (19 ms)
OK (1 ms)		U300	SP-SIP		192.168.6.132	OK (1 ms)
Disconnected		pstn7	FXO		Port 7	
Disconnected		pstn8	FXO		Port 8	
Disconnected		BriTrunk9	BRI		Port 9	
Disconnected		BriTrunk10	BRI		Port 10	

Figure 21-2 Trunk Status

➢ VoIP Trunk

Table 21-1 VoIP Trunk Status

Status	**VoIP trunk status:** • Registered: successful registration, trunk is ready for use. • Unregistered: trunk registration failed. • Request Send: registering. • Waiting: waiting for authentication. **Service Provider status:** • OK: successful registration, trunk is ready for use. • Unreachable: cannot reach the VoIP service provider. • Failed: trunk registry failed.
Provider Name	Display the trunk name.
Type	Display the trunk type. • **SIP:** SIP VoIP trunk. • **IAX**: IAX VoIP trunk. • **SP-SIP**: Service Provider SIP trunk. • **SP-IAX**: Service Provider IAX trunk.
User Name	Shows the trunk user name if the VoIP trunk is registration based.
Hostname Name/IP	Display the host name/IP for the VoIP trunk.
Reachability	Shows the reachability status of the VoIP provider. • OK • UNREACHABLE

➢ PSTN Trunk

Table 21-2 PSTN Trunk Status

Status	PSTN trunk status: • Idle: the port is idle. • Busy: the port is in use. • Disconnected: there is no line connected to the port.

Trunk Name	Display the trunk name.
Type	Display the trunk type: PSTN.
Port	Display the relevant physical port of the trunk.

- **BRI Trunk**

Table 21-3 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→SystemStatus→Network Status**.

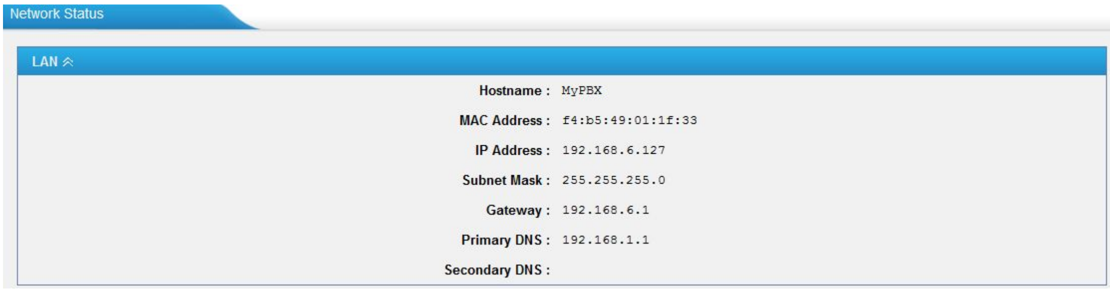


Figure 21-3 Network Status

System Info

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

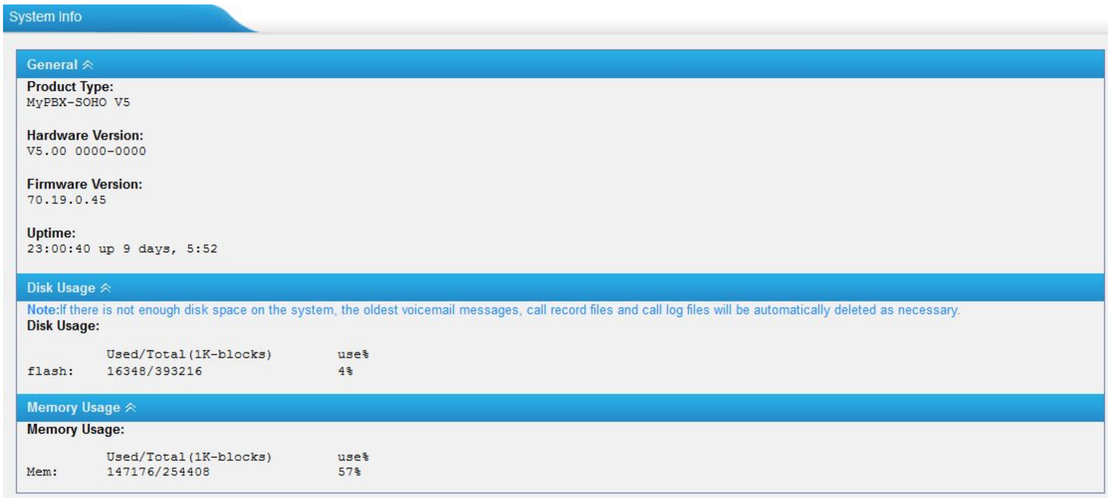


Figure 21-4 System Info

Reports

MyPBX provide reports including call logs and system logs. Important system events can be monitored on MyPBX and users could download the system logs.

- Call Logs
- System Logs
- Packet Capture Tool
- DAHDI Monitor Tool

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.

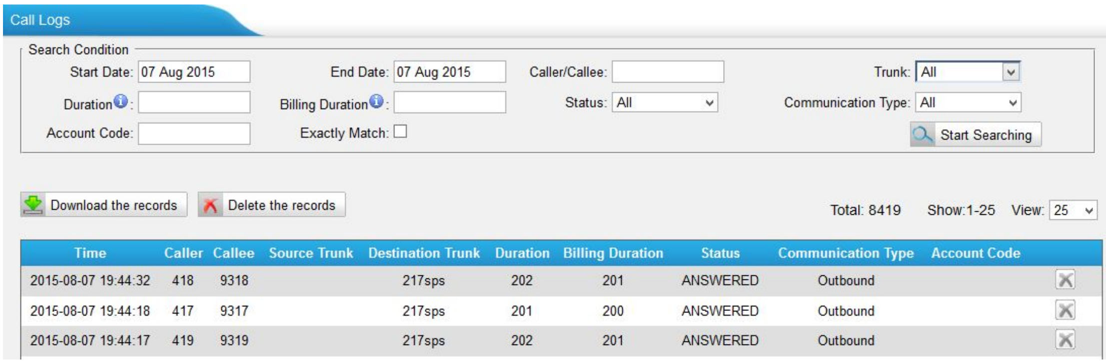
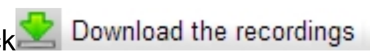
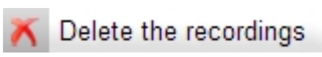


Figure 22-1 Call Log List

- **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 

  to export the filtered records to a .csv file.
- **Delete Searched Results**
  Click 

  to delete the filtered records.

System Logs

The MyPBX SOHO supports to monitor important system logs, including hardware log, web log and debug log.
Go to **Reports** → **System Logs** to check the system logs.

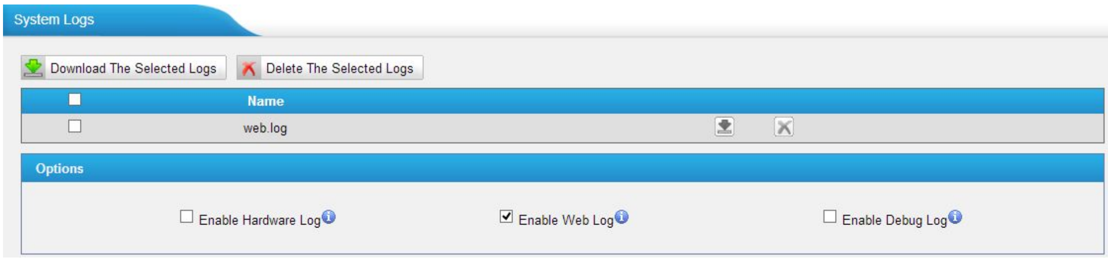


Figure 22-2 System Logs

- **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.

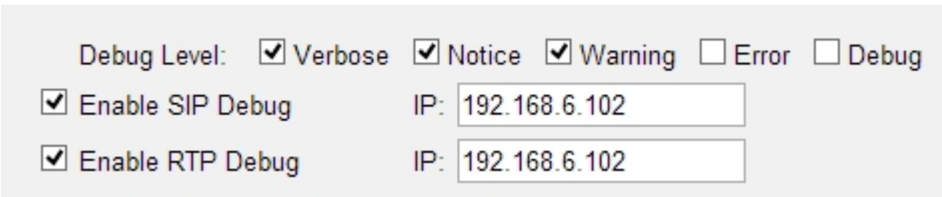


Figure 22-3 Debug Preferences

Packet Capture Tool

This feature is used to capture packets for technician. Integrate packet capture tool "Wireshark" in MyPBX SOHO. The Packet Tool can be found under **Reports→System Logs→ Packet Tool**.
Users could specify the destination IP address and port to get the packets.

Figure 22-4 Packet Tool

DAHDI Monitor Tool

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

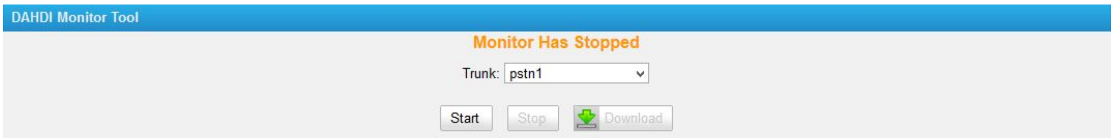


Figure 22-5 DAHDI Monitor Tool

[The End]