MyPBX

User Manual

Version 2.16.0.58

Yeastar Technology Co., Ltd.

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1. Introduction

MyPBX —IP-PBX for Small Businesses/Home Office

MyPBX is a standalone embedded hybrid PBX for small businesses and remote branch offices of larger organizations (1-100 users per site). MyPBX also offers a hybrid solution (a combination of VoIP applications using legacy telecom equipment) alternative for enterprises who are not yet ready to migrate to a complete VoIP solution.

Caveat

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation. Any Changes or modifications not expressly approved by the party responsible for compliance could void the user‟s authority to operate the equipment.

Customer Information

1. This equipment complies with Part 68 of the FCC rules and the requirements adopted by the ACTA. On the bottom of this equipment is a label that contains, among other information, a product identifier in the format [US: X8PIS00BSTD]. If requested, this number must be provided to the telephone company.
2. A plug and jack used to connect this equipment to the premises wiring and telephone network must comply with the applicable FCC Part 68 rules and requirements adopted by the ACTA. A compliant telephone cord and modular plug is provided with this product. It is designed to be connected to a compatible modular jack that is also compliant. See installation instructions for details.
3. If this equipment [Yeastar MyPBX] causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But if advance notice isn't practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.
4. The telephone company may make changes in its facilities, equipment, operations or procedures that could affect the operation of the equipment. If this happens the telephone company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.
5. If trouble is experienced with this equipment [US: X8PIS00BSTD], for repair or warranty information, Service can be facilitated through our office at:
   U.S. Agent Company name:
   Commlogik Corporation
   Address: 1921 NW 82nd Ave Miami - FL - 33126, USA
   Tel: +1 (305) 677-7888
   Fax: +1 (305) 677-7889
   If the equipment is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.
6. Please follow instructions for repairing if any (e.g. battery replacement section); otherwise do not alternate or repair any parts of device except specified.
7. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission or corporation commission for information.
8. If your home has specially wired alarm equipment connected to the telephone line, ensure the installation of this [US: X8PIS00BSTD] does not disable your alarm equipment. If you have questions about what will disable alarm equipment, consult your telephone company or a qualified installer.
9. If the telephone company requests information on what equipment is connected to their lines, inform them of:
   a) The ringer equivalence number [0.0B]

b) The USOC jack required [RJ11C]
c) Facility Interface Codes ("FIC") [METALLIC]
d) Service Order Codes ("SOC") [9.0y]
e) The FCC Registration Number [US: X8PIS00BSTD]

10. The REN is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. The REN for this product is part of the product identifier that has the format US: AAAEQ##TXXXX. The digits represented by ## are the REN without a decimal point. For this product the FCC Registration number is [US:X8PIS00BSTD] indicates the REN would be 0.0B.

Applications

1.1 Features

● Auto-provision	● Firewalls
● Blind Transfer	● Follow me
● BLF Support	● Interactive Voice Response (IVR)
● Blacklist	● Intercom / Zone Intercom
● Call Detail Records(CDR)	● Music On Hold
● Call Forward	● Music On Transfer
● Call Parking	● Paging / Zone Paging
● Call Recording	● PIN Users
● Call Pickup	● Queue
● Call Routing	● QOS
● Call Transfer	● Ring Group
● Call Waiting	● Route by Caller ID
● Caller ID	● Skype Integration (Skype Connect)
● Call Back	● Three-way Calling
● Conference	● Mobility Extension
● SMS to Mail	● External Storage
● Mail to SMS	● DDNS
● Speed Dial	● OpenVPN
● Define Office Time	● T.38
● Direct Inward System Access(DISA)	● Voicemail
● DIDs	● VLAN

● Distinctive Ringtone	● WAN
● Do Not Disturb(DND)	● PPPoE
● Dial by Name	● Static Route

Note: the features with asterisk (*) will be supported in next version.

1.2 Hardware Specifications

1.2.1 Exterior Appearance

Front Side

①② ③④ ⑤

Figure 1-1 MyPBX Front Panel Picture

No.	Identifying
①	Green LED: Indicates correct power is being supplied to the unit
②	Green LED: Indicates the MyPBX is fully functional.
③	Green LED: Indicates stable WAN Port connection
④	Green LED: Indicates stable LAN Port connection
⑤	Red LED: Indicates presence of an FXO/GSM port. Orange LED: Indicates presence of a BRI port. Green LED: Indicates presence of an FXS port. LED Blinking- Red blinking: No connection between FXO port and PSTN LED Alternating - Red and Green (slow blink): FXO port is receiving an incoming call.

	LED Alternating - Red and Green (fast blink): FXO port is in use LED Dual - Green and Red (slow blink): FXS port is ringing LED Dual - Green and Red (fast blink): FXS port is in use

2. System set up

2.1 Connection Drawing

A. MyPBX

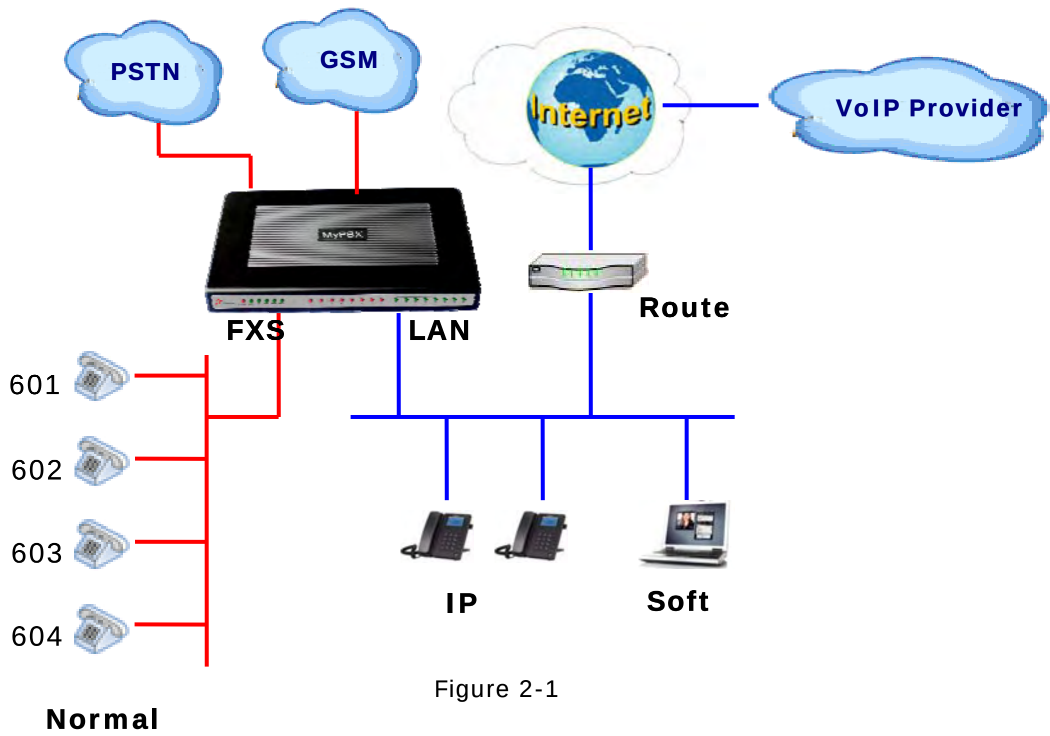


Figure 2-1

2.2 Connecting Ethernet Line

MyPBX provides two 10/100M Ethernet ports with RJ45 interface and LED indicator. Plug Ethernet line into MyPBX's Ethernet port, and then connect the other end of the Ethernet line with a hub, switch, router, LAN or WAN. Once connected, check the status of the LED indicator. A yellow LED indicates the port is in the connection process, and a green LED indicates the port is properly connected.

2.3 Supplying Power

MyPBX utilizes the high-performance switch power supply, which supplies the required power for the unit.
AC Input: 100~240V DC Output: 12V, 5A

Please follow the steps below to connect MyPBX unit to a power outlet:

1. Connect the small end of the power cable to the power input port on the MyPBX back panel, and plug the other end of the cable into a 100V AC power outlet.
2. Check the Power LED on the front panel. A solid green LED indicates that power is being supplied correctly.

3. Managing MyPBX

3.1 Administrator Login

From your web browser, input the IP address of the MyPBX server.
If this is the first time you are configuring MyPBX, please use the default settings below (your PC should be in the same local network with MyPBX):
IP Address: http://192.168.5.150
Username: admin
Password: password

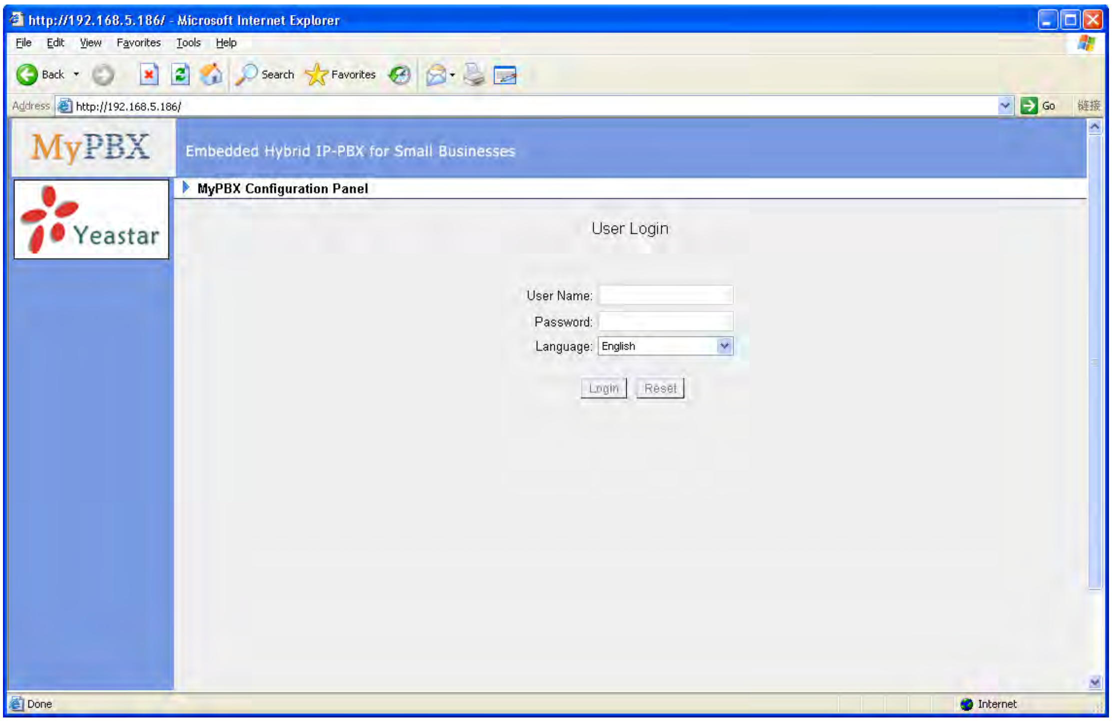


Figure 3-1

3.2 Status Monitor

3.2.1 Line Status

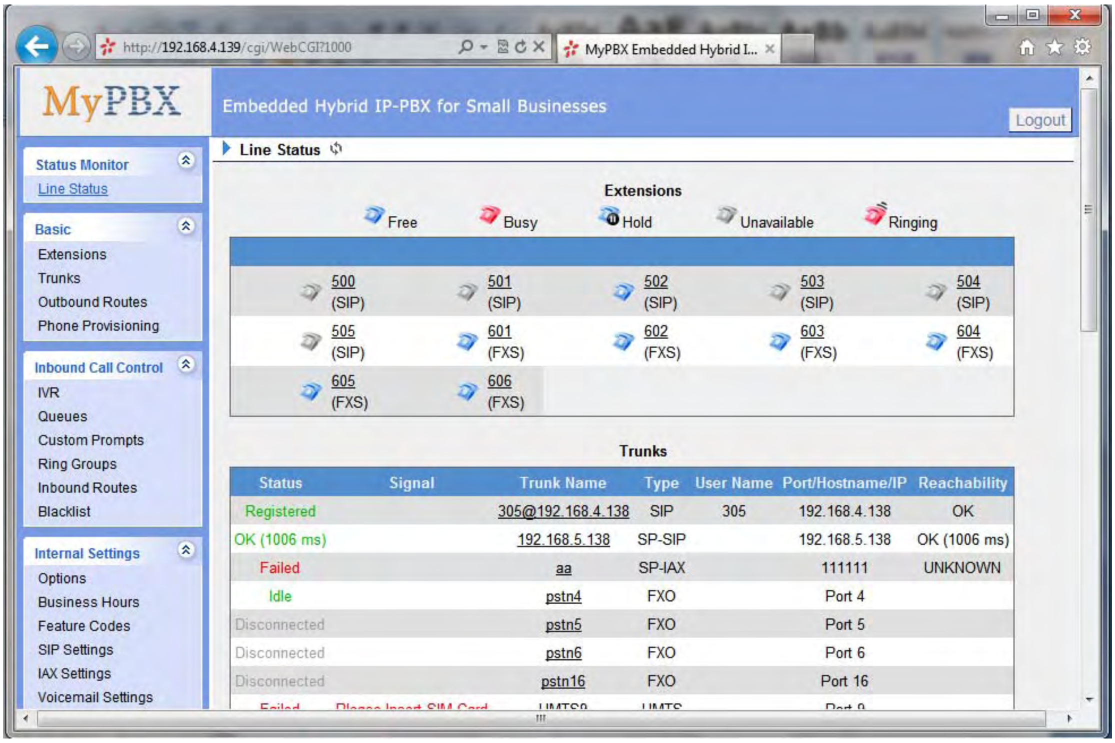
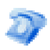


Figure 3.2.1

MyPBX Status Description:

Extensions:

1)  : Extension is unregistered

2)  : Extension is idle

3）  : Extension is ringing

4）  : Extension is busy in a call

5）  : Extension is on hold

Trunks:

VOIP Trunk:

Status

Unregistered: Trunk registration failed.

Registered: Succeed registration, trunk is ready for use.

Request Send: Registering.

Waiting: Waiting for authentication.

FXO Trunk:

Status

Idle: The port is idle.

Busy: The port is in use.

Disconnected: The port hasn't connected the PSTN line.

More detail message, please refer to the LED identifying of front panel.

GSM Trunk:

Status

Idle: The port is idle.

Busy: The port is in use.

Signal

: No signal.

: Poor.

: Average.

: Good.

: Excellent.

BRI Trunk:

Status

Ok: The ports connect correctly.

Disconnected: The port hasn't connected the PSTN line.

Service Provider:

Status

OK: Succeed registration, trunk is ready for use.

Unreachable: The trunk is unreachable.

Failed: Trunk registration failed.

3.3 Basic

3.3.1 Extension

Extension has two types: Analog extensions (FXS) and VOIP extensions (SIP extension or IAX extension).

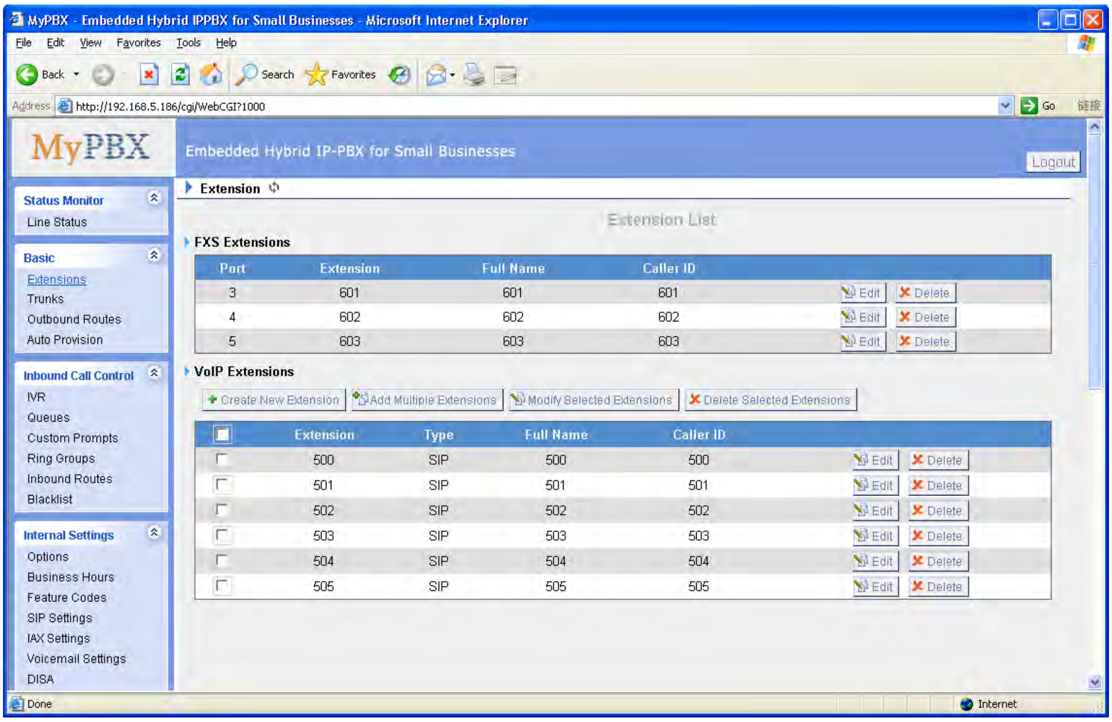


Figure 3.3.1

3.3.1.1 Analog Extensions (FXS)

Edit Analog Extensions

On the administration page of FXS extensions, click 'Edit' on the extension that you want to edit, and modify the following information on the popup window

1) General

·Extension

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

·Port

The extension correspond port.

·Name

A character-based name for this user, i.e. 'Bob Jones' .

·Caller ID

The Caller ID (CID) string will be used when this user calls another internal user.

2) Voicemail

·Enable Voicemail

Check this box if the user should have a voicemail account.

·Voicemail Access PIN #
Voicemail Password for this extension, i.e. '1234' .

3) Mail Setting

·Enable Send Voicemail

Once enabled, the voicemail will be sent to the below email address as an attachment.

·Send Voicemail to Email Address

This option defines whether or not voicemails/Fax is sent to the Email address as an attachment.

Note: Please ensure that all voicemail settings are properly configured on the System Settings -> Voicemail Settings page before using this feature.

4) Flash

·Hook Flash Detection

Sets the amount of time, in milliseconds, that must pass since the last hook-flash event received by MyPBX before it will recognize a second event. If a second event occurs in less time than defined by Hook Flash Detection, then MyPBX will ignore the event. The default value of Flash is 1000 ms, and it can be configured in 1ms increments.

5) Group

·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 (default *4).

Note: *4 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.

6) Follow me (Call Forwarding)

This function sets inbound call forwarding on an extension. An administrator can configure Follow Me for this extension.

7) Other Options

·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.

·User Web Interface

Check this option to allow the user to login to the MyPBX User Web interface, which can be used to access voicemail and extension recordings. Users may login to the MyPBX User Web interface by using their extension number and voicemail PIN # as the login and password respectively.

·DND

Don’t Disturb.

·Ring Out

Check this option if you want to custom the ring time. Tone will stop over the time defined.

8) Volume Settings

Rxgain: The Volume sent to FXS extension.

Txgain: The Volume sent out by the FXS extension.

9) Mobility Extension

MyPBX allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call MyPBX with this mobility 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) and use all outbound route that your extension can use of MyPBX.

·Mobility Extension Number

Don't forget to add the dial patterns of the outbound route at the beginning of your mobile phone number when you fill in the mobility extension number filed. E.g. if you want to set “15960XXXXXX” as mobile extension, and the dial pattern of the outbound route is “9”; you should set “915960XXXXXX” here.

Note: If callback is enabled in the inbound route, the mobility extension function of this inbound route will be disabled.

10) Spy Settings

MyPBX 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, refer to 'Feature Codes' page for more information.

·spy modes

There are 3 spy modes available for choice:

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

Note: for 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.

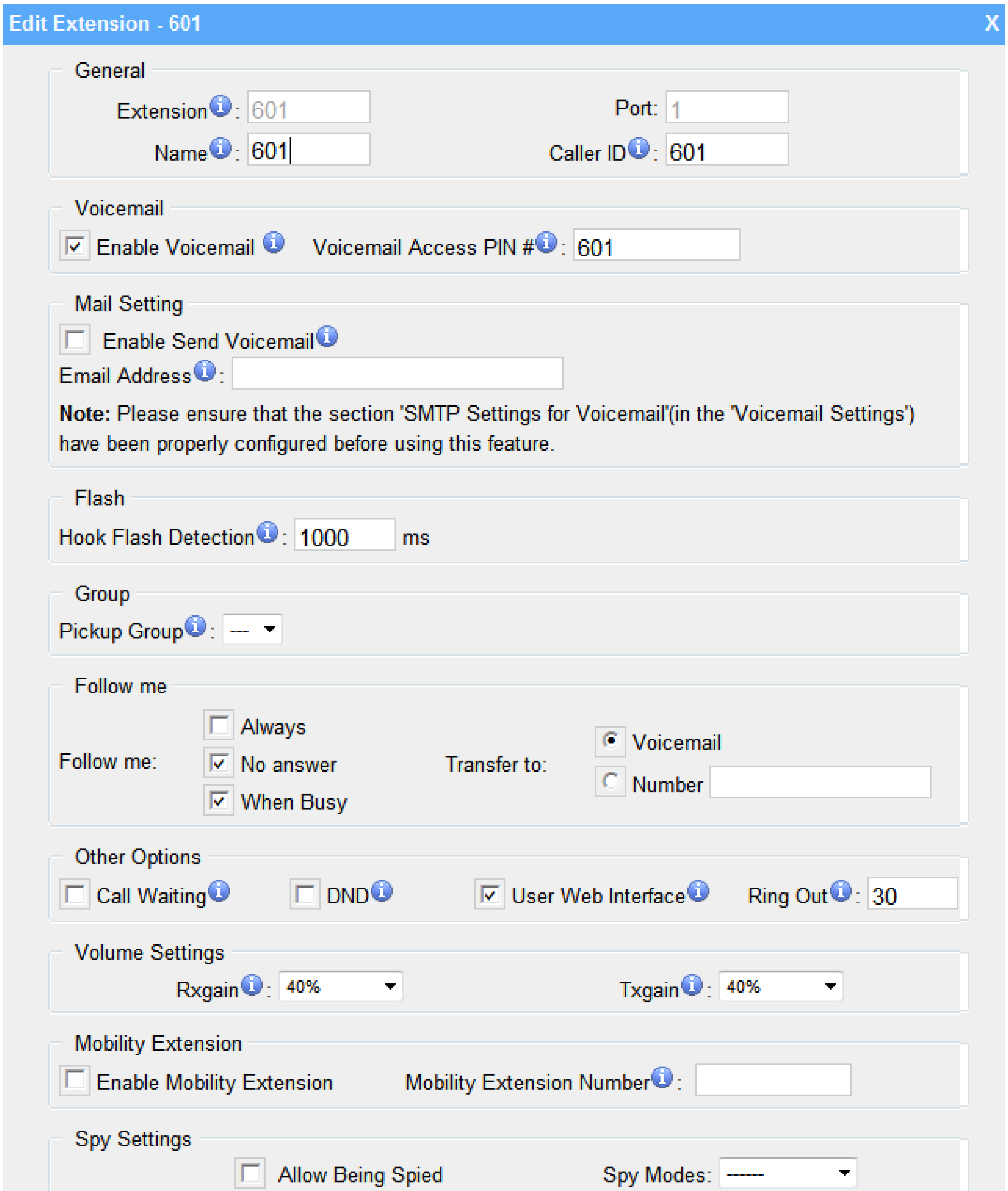


Figure 3.3.1.1

3.3.1.2 VOIP Extension

A VOIP extension is a SIP/IAX Account that allows an IP Phone or an IP Soft-Phone client to register on MyPBX.

1. Add VOIP Extension

Go to Extensions → VOIP Extensions → Create New Extension

1) General

·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, i.e. 1234, that will be associated with this particular User / Phone.

·Password

The password for this extension, Ex: ‘12t3f6’

·Name

A character-based name for this user, EX: ‘Bob Jones’

·Caller ID

The Caller ID will be used when this user calls another internal extension.

2) Voicemail

·Enable Voicemail

Check this box if the user should have a voicemail account.

·Voicemail Access PIN #

The voicemail Password for this extension, i.e. ‘1234’ .

3) Mail Setting

This option defines whether or not voicemails or faxes are sent to an Email Address as attachment.

·Enable Send Voicemail

Once enabled, the voicemail will be sent to email as an attachment.

·Email Address

Email address used to receive the voicemail or Fax.

Note: Please ensure that the section 'SMTP Settings For Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.

4) Group

·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 (default is *4).

Note: *4 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.

5) Follow me (Call Forwarding)
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.

6）Other Options
.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’s call waiting function.

.DND
Don’t Disturb. When DND is enabled for an extension, the extension will be not available.

.User Web Interface
Check this option to allow the user to login to the MyPBX User Web interface, which can be used to check voicemail and extension recordings. Users may login to MyPBX User Web interface by using their extension number and voicemail PIN # as the login and password respectively.

.Ring Out
Check this option if you want to customize the ring time. Ring tone will stop over the time defined.

7) VoIP Settings
·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.

·Enable SRTP
Enable extension for SRTP (RTP Encryption).

·Qualify
Send check alive packets to IP phones

·SIP Transport
This will be the transport method used by the extension. The options are UDP (default) or TCP or TLS.

·DTMF Mode – RFC2833, Info, Inband, Auto.

8) 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.

9) Mobility Extension
MyPBX allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call MyPBX with this mobility phone, you will hear a dial tone. MyPBX will recognize your call as a call from an extension. You can dial other extension numbers (Your caller ID will be the number of your extension) and dial out using all outbound route that your extension can use on MyPBX.

·Mobility Extension Number
Don't forget to add the dial patterns of the outbound route at the beginning of your mobile phone number when you fill in the mobility extension number filed. E.g. if you want to set "15960XXXXXX" as mobile extension, and the dial pattern of the outbound route is "9"; you should set "915960XXXXXX" here.

Note: If callback is enabled on the inbound route, the mobility extension function of this inbound route will be disabled.

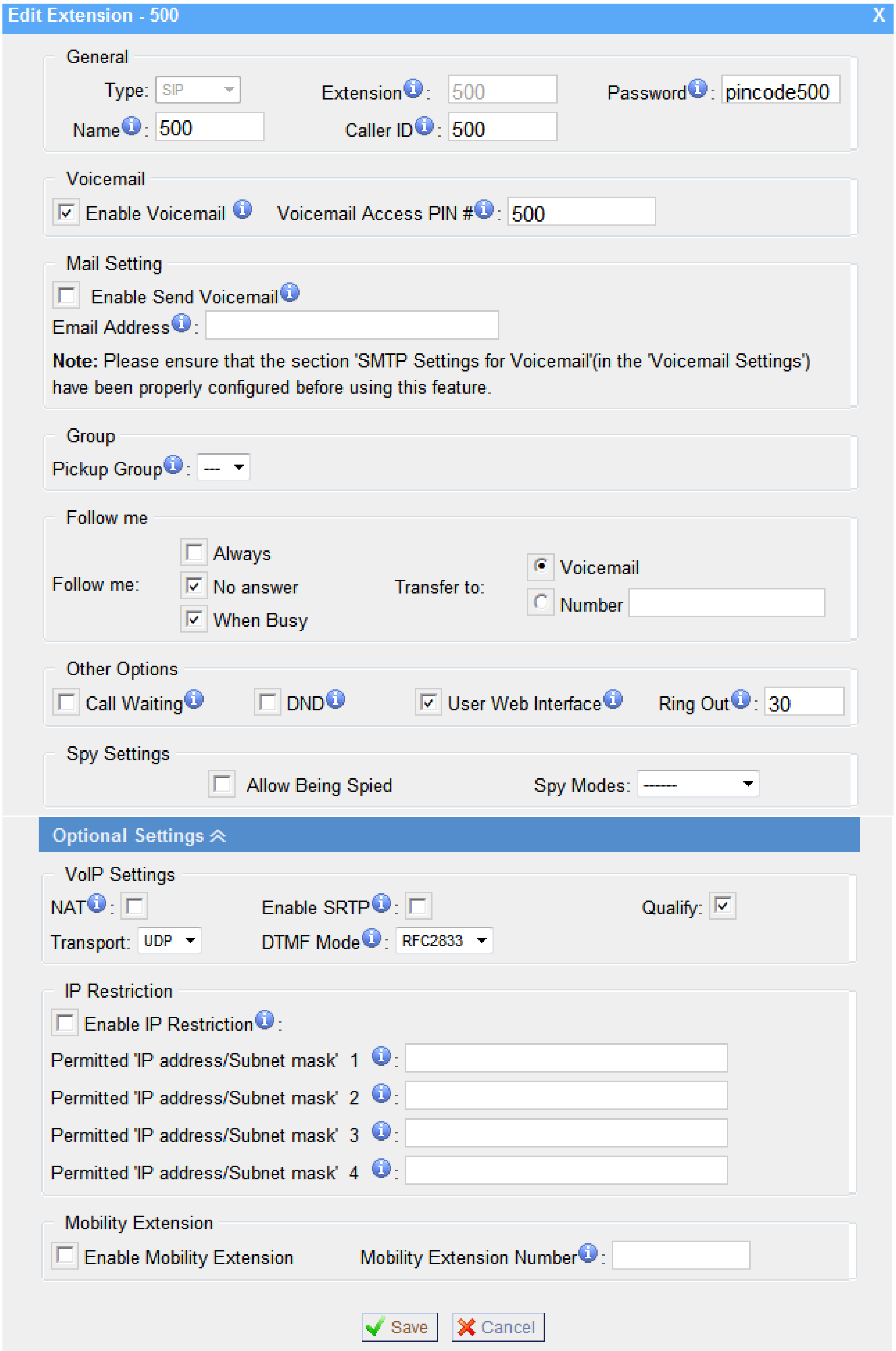


Figure 3.3.1.2

2. Add Multiple Extensions

Go to Extensions → VOIP Extensions → Add Multiple Extensions.
1) Select the number of extensions that you would like to create.
2) Select the type of extension that you would like to create.
3) Fill in the starting extension number.

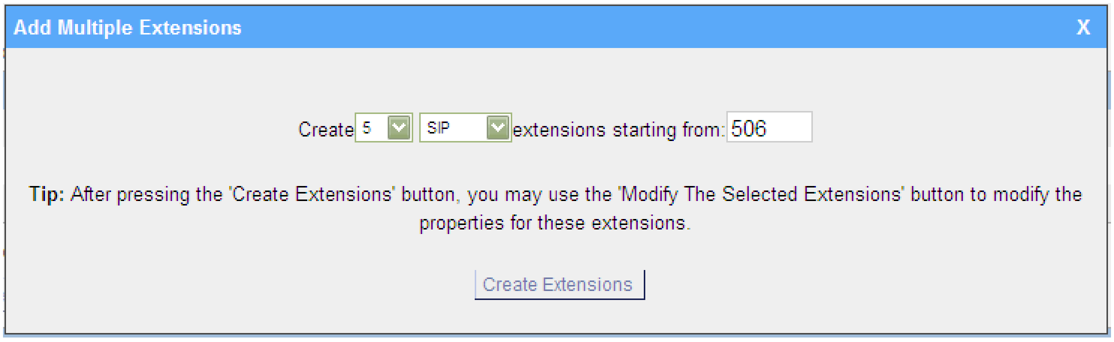


Figure 3.3.1.3

3. Edit VOIP Extension

Click 'Edit' on VOIP Extension administration page or click 'Modify Selected Extensions' to edit extensions.

3.3.2 Trunk

There are five types of trunks: Analog trunks (FXO), GSM Trunk, BRI Trunk, VoIP Trunk and Service Provider trunk.

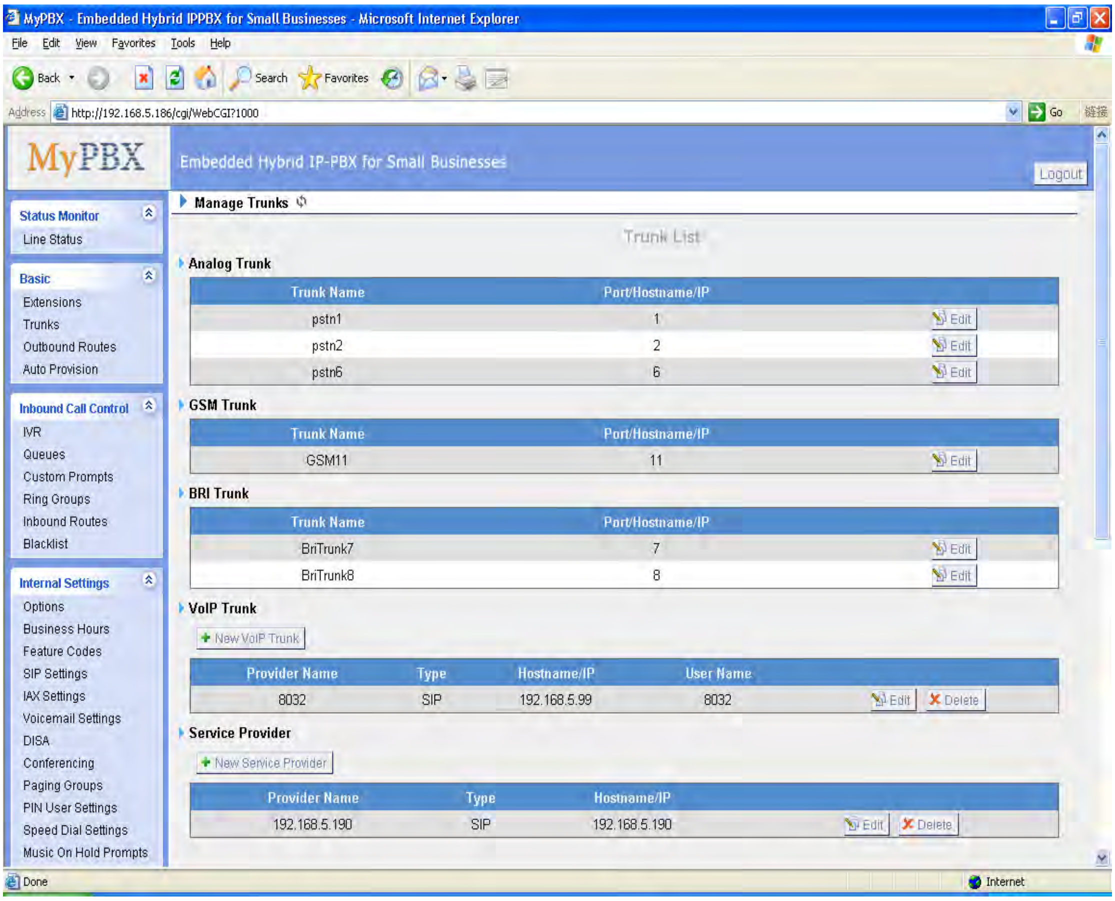


Figure 3.3.2

3.3.2.1 Analog Trunk (FXO)

1. Edit Analog Trunk (FXO)

On the Trunk administration page, click ‘Edit’ on the selected trunk and modify its properties in the popup window:

1）General

·Trunk Name

A unique label used to identify this trunk when listed in outbound rules, incoming rules, etc. Ex: 'pstn5'

·Volume Setting

Used to modify the volume level of this trunk . Normally, this setting does not need to be changed.

2）Busy 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 500 msec on, 500 msec off. Without Busy Pattern specified, MyPBX will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then MyPBX will further check the length of the tone and silence, which will further reduce the chance of a false positive disconnect.

·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.

·Polarity Detection

Configure if the call needs to be hung up when a polarity signal arrived

3）Advanced Options

·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

.Caller ID Detection

For fxo trunks, this option forces MyPBX to clarify Caller ID incoming calls.

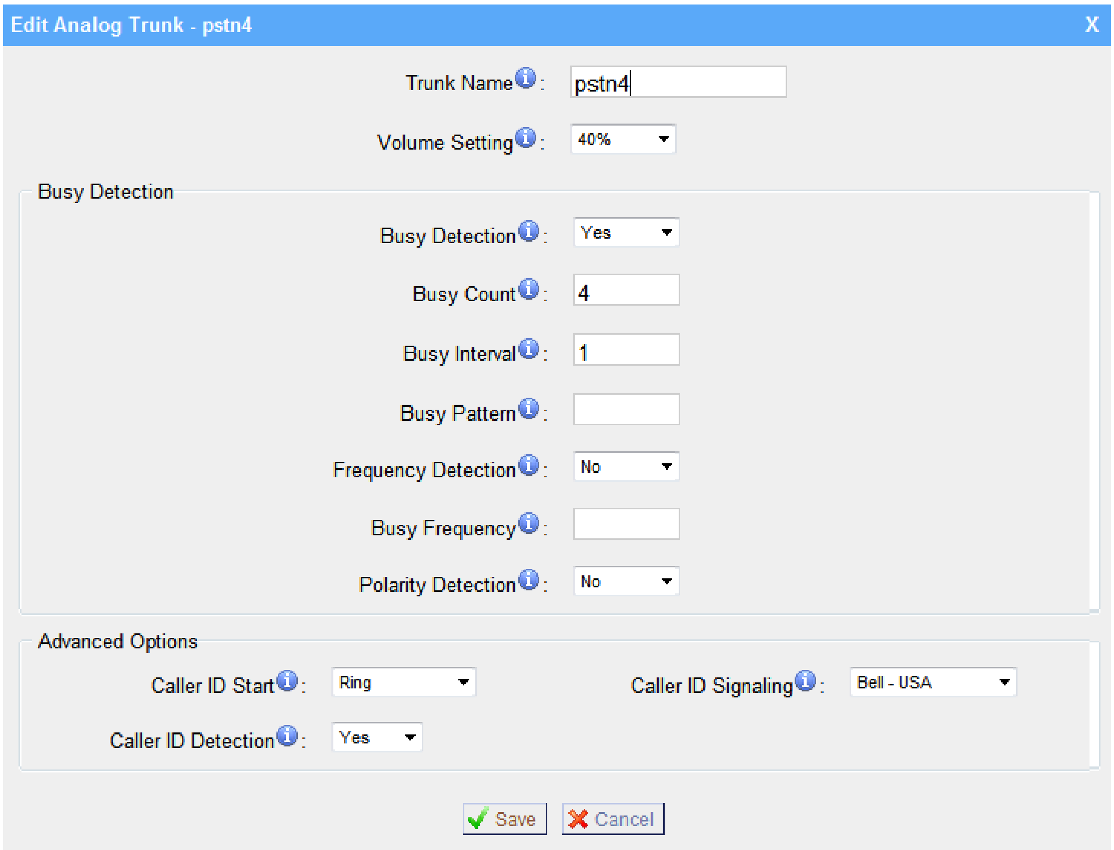


Figure 3.3.2.1

3.3.2.2 GSM Trunk

1. Edit GSM Trunk

1）General

·Trunk Name

A unique label used to identify this trunk when listed in outbound rules, incoming rules, etc. Ex: 'GSM9'

·Volume Setting

Used to modify the volume level of this trunk. Normally, this setting does not need to be changed.

·Pin Code

Please enter your SIM card pin code here if your card has a pin code.

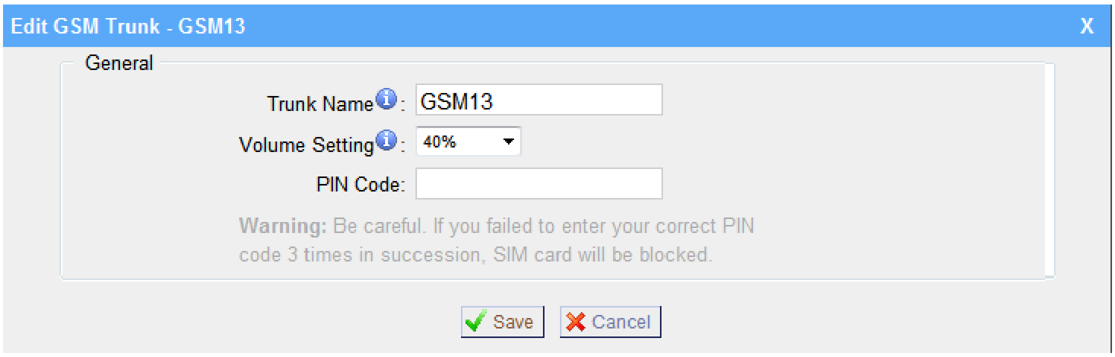


Figure 3.3.2.2

3.3.2.3 BRI Trunk

1. Edit BRI Trunk

1）General

·Trunk Name

A unique label used to identify this trunk when listed in outbound rules, incoming rules, etc. Ex: 'BriTrunk1'

·Signaling

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

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.

·PRI Dial Plan

Sets an option required for some (rare) switches that require a dial plan parameter to be passed. This option is ignored by most BRI switches. It may be necessary on a few pieces of hardware. This option can almost always be left unchanged from the default.

·Reset interval

Sets 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 Local Dial Plan

Sets an option required for some (rare) switches that require a dial plan parameter to be passed. This option is ignored by most BRI switches. It may be necessary on a few pieces of hardware. This option can almost always be left unchanged from the default.

·Over Lap Dial

Whether MyPBX can dial this switch using overlap digits . 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.

·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

Echocancel Obviously this disables or enables echo cancellation, it is recommended to not turn this off.

·Hide CallerID

If you want others to see your CID, please disable this option.

·Codec

You can choose alaw or ulaw codes.

2) CallerID Prefix

·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) DOD Setting

·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.

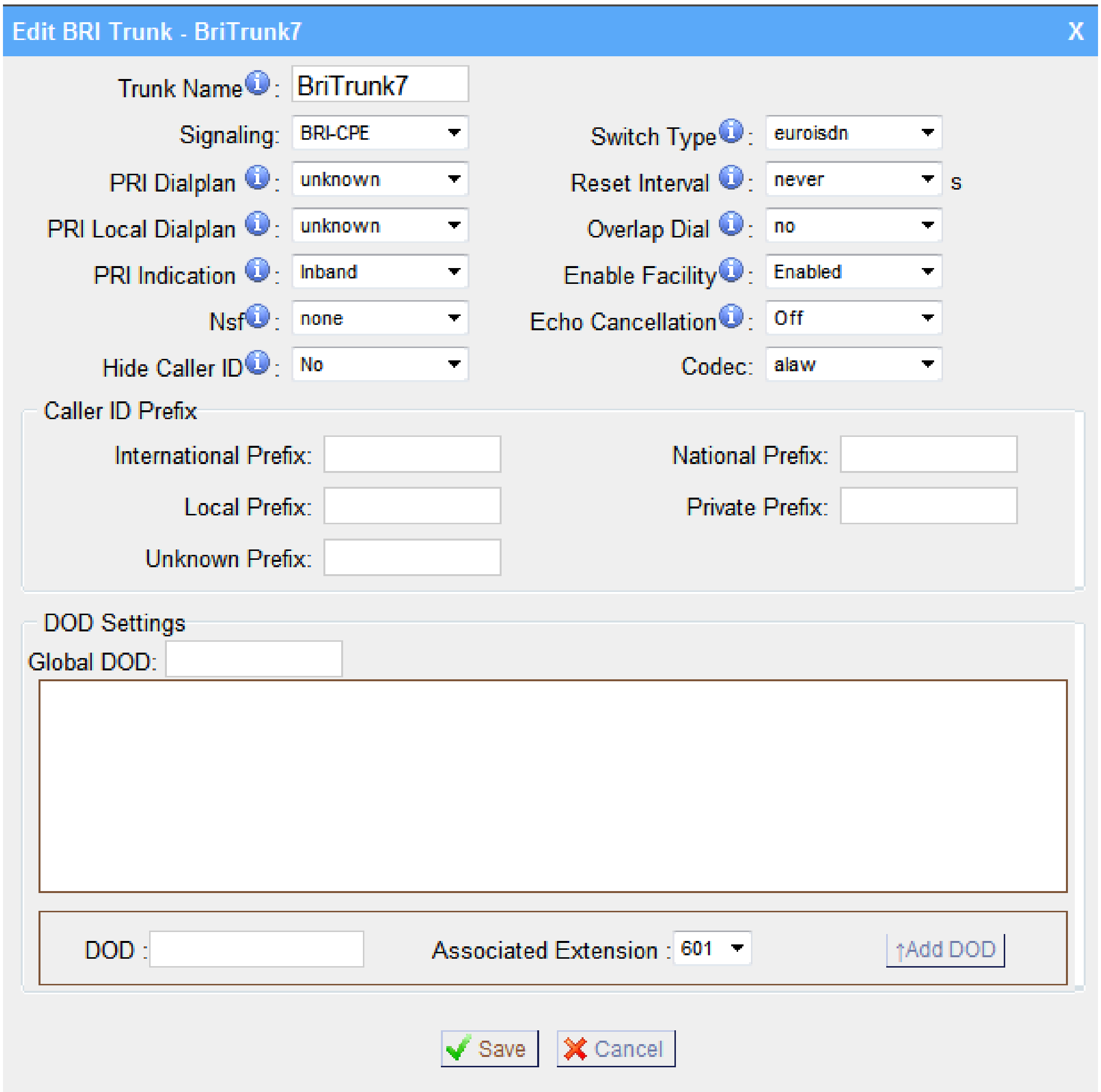


Figure 3.3.2.3

3.3.2.4 VOIP Trunk

1. Add SIP Trunk

Input correct SIP information (provide by VOIP provider). Inaccurate information will prevent the trunk from registering.

1) General setting

·Type

SIP – Identifies whether the trunk sends and receives calls using the VoIP protocol SIP

·Provider Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. Ex: 'yeastar'.

·Hostname / IP

Service provider's hostname or IP address.5060 is the standard port number used by SIP protocol. Don't change this part if it is not required.

.Domain

VoIP provider's server domain name .

·Username

Username of SIP account . Used for SIP trunk registration.

.Authorization name

Used for SIP authentication. Leave this blank if not required.

·Password

Password of SIP account .

.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.

.Online number

Define the online number that expected by 'Skype Connect' and some other SIP service providers. Leave this field blank if it's no required.

·Maximum Channels

Controls 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' screen will override the caller ID set in the 'VOIP trunk' screen. Please note that not all the service providers support this feature. Contact your service provider for more information.

·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

Note: codec can only display when it is edited after creating the trunk.

·Transport

This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.

·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 Setting

·DOD

Direct Outward Dialing Number.

·Associated Extension

The extension make call out via SIP Trunk will display the associated DOD.

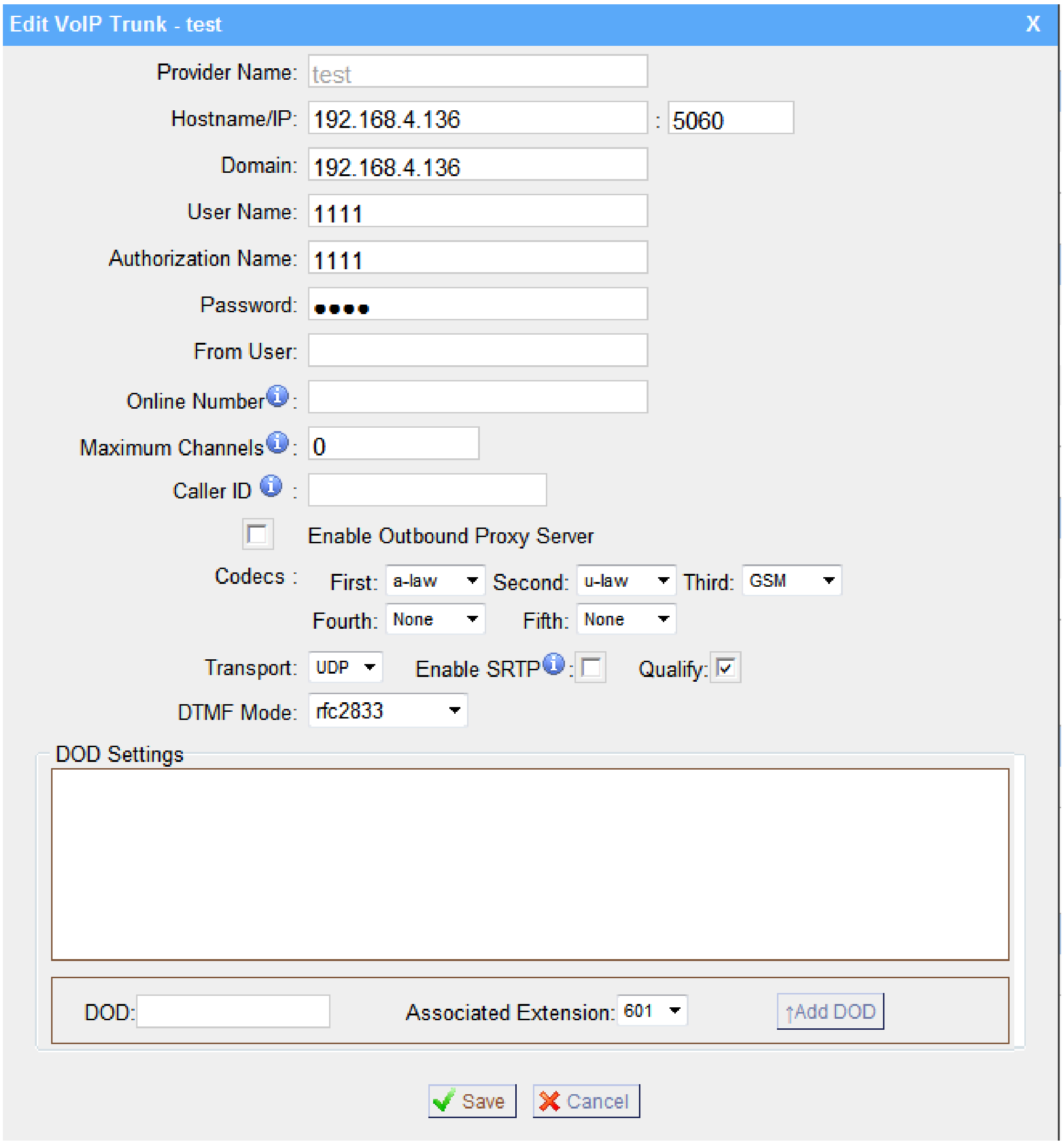


Figure 3.3.2.4.1

2. Add IAX Trunk

Input correct IAX information (provided by VOIP provider). Inaccurate information will prevent the trunk from registering.

1) General

·Type

IAX – Identifies whether the trunk sends and receives calls by using the VoIP protocol IAX.

·Provider Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. Ex: 'yeastar2'.

·Hostname / IP

Service provider's hostname or IP address. 4569 is the standard port number used by IAX protocol. Don't change this part if it is not required.

·Username

Username of IAX account . Used for IAX trunk registration.

·Password

Password of IAX account .

.Online number

Define the online number that expected by 'Skype Connect' and some other SIP service providers. Leave this field blank if it's no required.

·Maximum Channels

Controls 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' screen will override the caller ID setting in the 'VOIP trunk' screen. Please note that not all the service providers support this feature. Contact your service provider for more information.

2) DOD Setting

·DOD

Direct Outward Dialing Number.

·Associated Extension

The extension make call out via BRI Trunk will display the associated DOD.

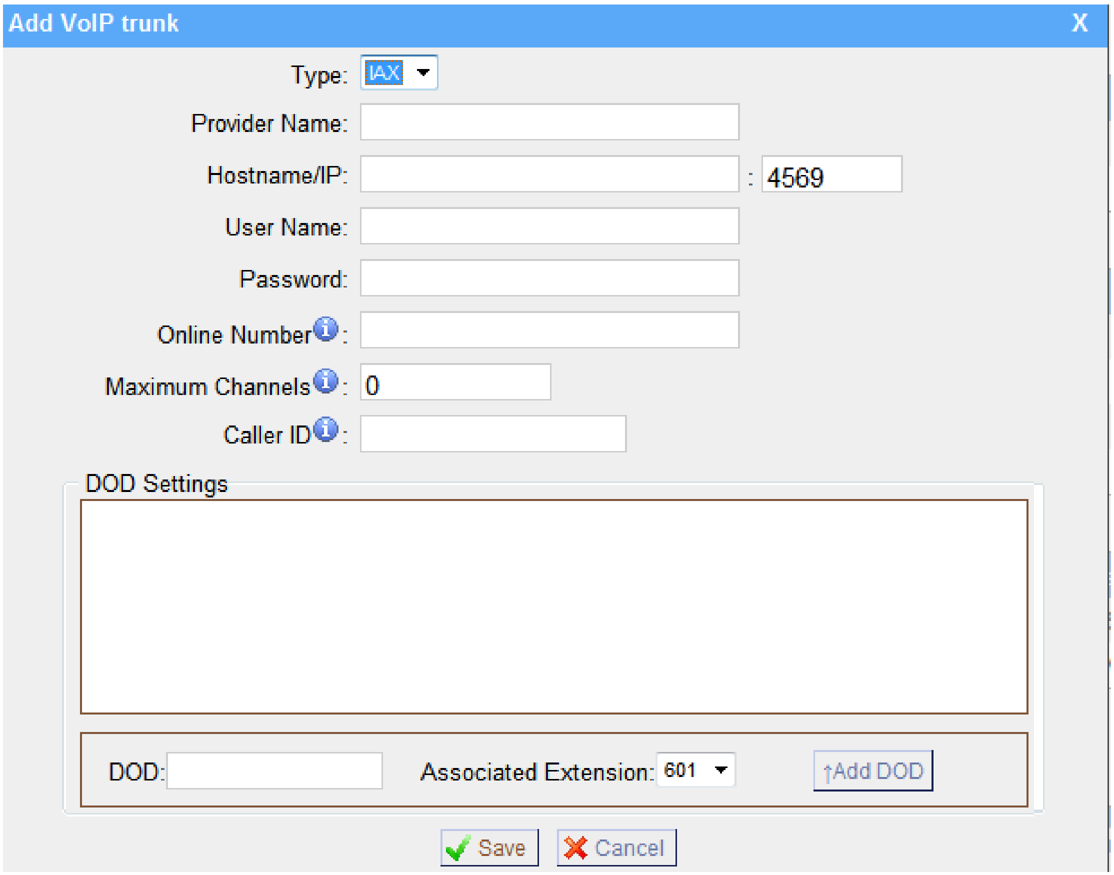


Figure 3.3.3.4.2

3.3.2.5 Service Provider

1. Add Service Provider

To Create the Service provider definition you need to complete the following VoIP fields.

1) General

·Type

SIP or IAX

SIP – Identifies whether the trunk sends and receives calls by using the VoIP protocol SIP.

IAX - Identifies whether the trunk sends and receives calls by using the VoIP protocol IAX.

·Provider Name

A unique label would help to you identify this trunk. Ex: 'Provider2'.

·Hostname / IP

Service provider's hostname or IP address.

Note: 5060 is the standard port number used by SIP protocol, 4569 is the standard port number used by IAX protocol. Don't change this part if it is not required.

·Maximum Channels

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk . Inbound calls are not counted against the maximum. Leave blank to specify no maximum.

·Codecs

Define the codec for this sip trunk and its priority

Note: codec can only display when edit it after creating the trunk.

·Transport

This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.

·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 Setting

·DOD

Direct Outward Dialing Number.

·Associated Extension

The extension make call out via BRI Trunk will display the associated DOD.

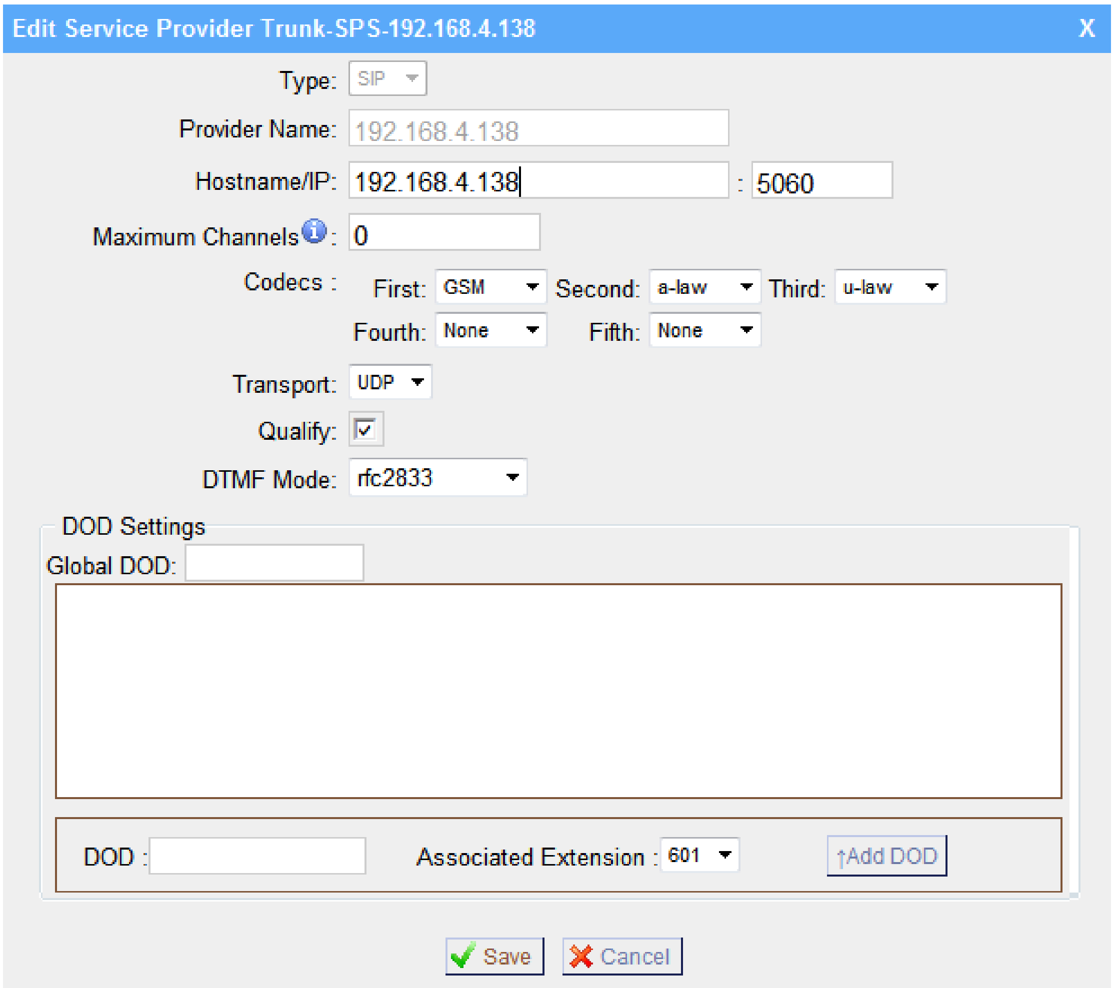


Figure 3.3.2.5

3.3.3 Outbound Routes

Outbound routing defines how outgoing calls are processed through the trunks.

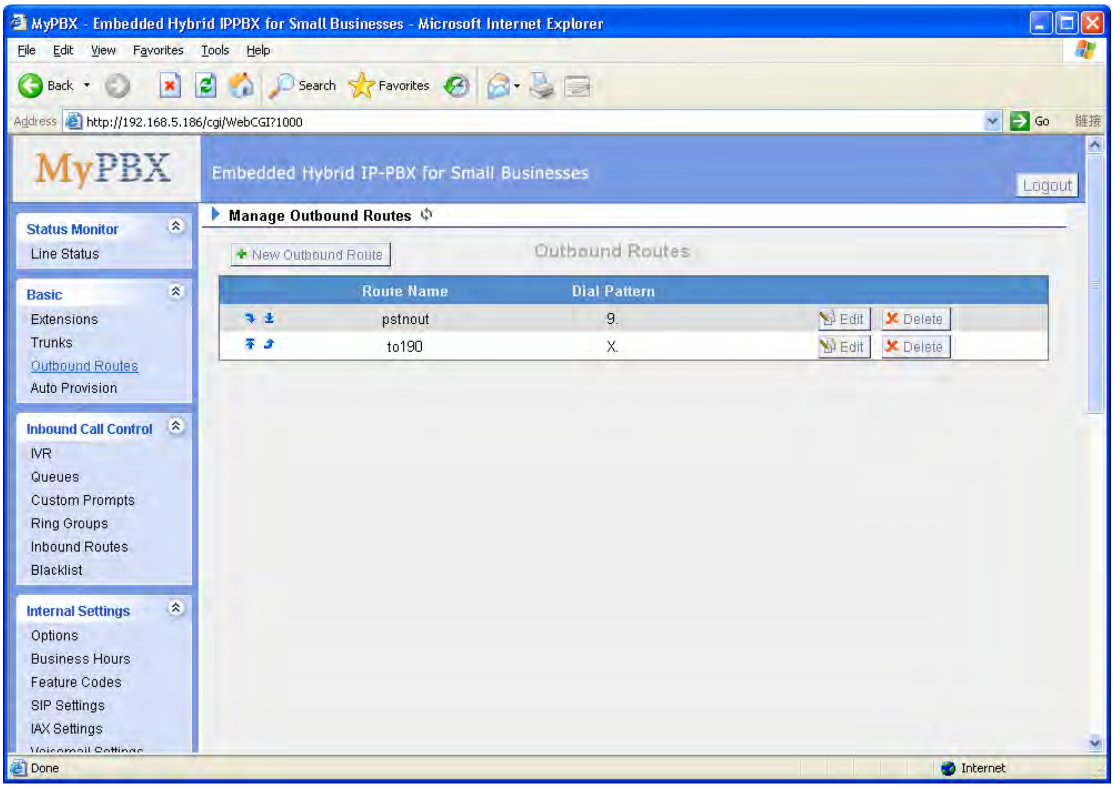


Figure 3.3.3

3.3.3.1 New Outbound Route

Click 'New Outbound Route' and fill in the corresponding information in the popup window.

1）General

·Route Name

Name of this Outbound Route . Ex: 'Local' or 'Long Distance' etc.

·Dial Pattern

Outbound calls that match this dial pattern will use this outbound route. There are a number of dial pattern characters that have special meanings:

X : Any Digit from 0-9

Z : Any Digit from 1-9

N : Any Digit from 2-9

[12345-9] : Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '**.**' Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '**!**' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.

Example 2: **1NXXNXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

·Strip digits from front

Allows the user 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 these digits before dialing

These digits will be prepended to the phone number before the call is placed. 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 calls are placed. When using analog trunks, a 'w' character may also be prepended to provide a slight delay before dialing.

·Password

The route password can be used to protect this route from being accessed without a password.

·T.38 Support:

Enable T38 fax in this outbound route (Only for SIP Trunk).

·Rrmemory Hunt

Round robin with memory, remembers which trunk was used last time, and then use the next available trunk to call out.

·Member Extensions

Defines the extensions that will be permitted to use this outbound route .

·Member Trunks

Defines the trunks that can be used for this outbound route .

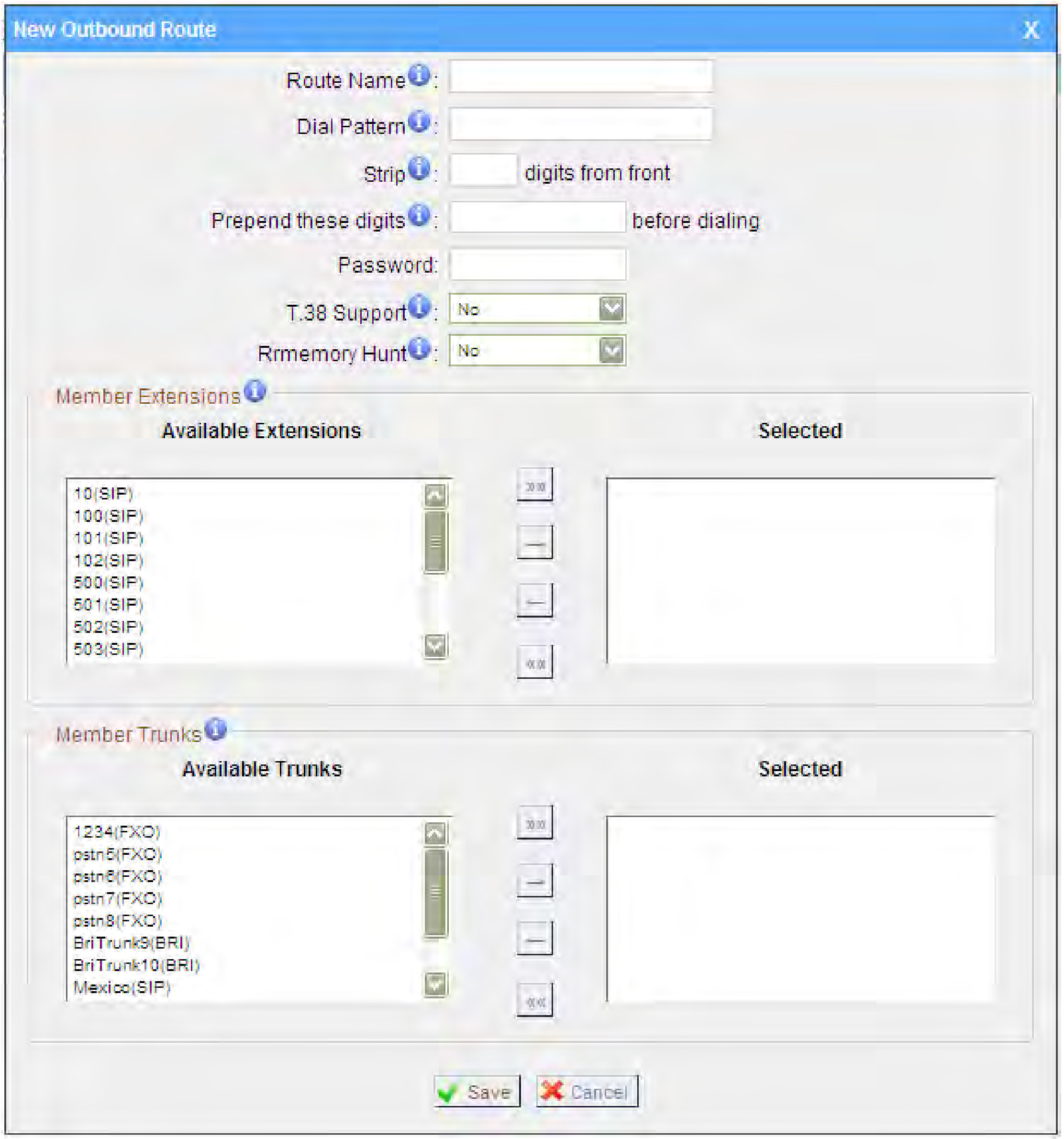


Figure 3.3.3.1

3.3.4 Phone Provisioning

The Auto Provision sub menu provides users a method to Auto Provision IP Phone after the Express Setup process.

Note: Auto Provision functions fully test with these models:

Yealink (T12,T18,T20,T22,T26,T28,T32,T38,VP530,VP-2009)

Snom (300,320,360,370)

Polycom (IP 6000,IP 7000,IP 32X,IP33X,IP430,IP450,IP550,IP560,VVX1500)

Cisco (IP7940,IP7960)

Aastra(480i,480i CT,6757i,6757i CT, 6737i)

News:

When provisioning Yealink and Snom IP phone, MyPBX is not needed to be set as the only DHCP server any more.

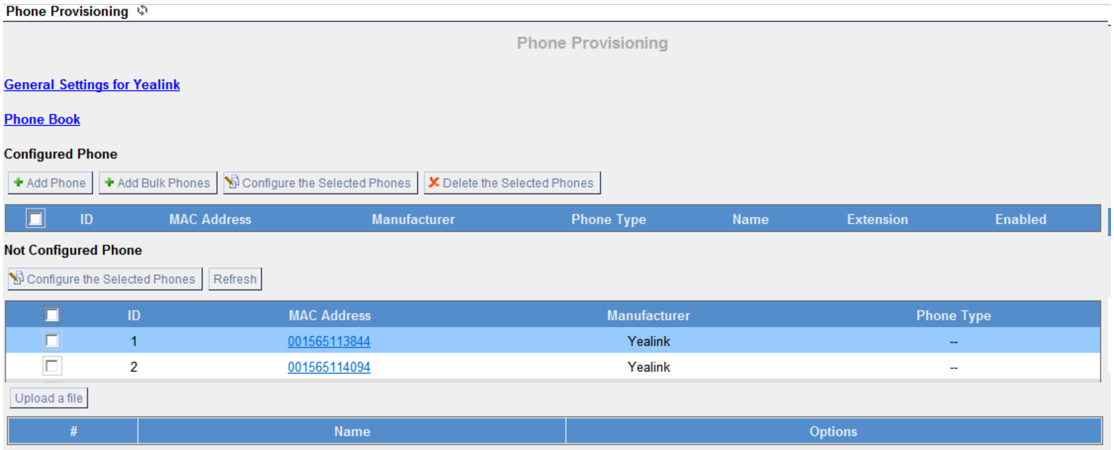
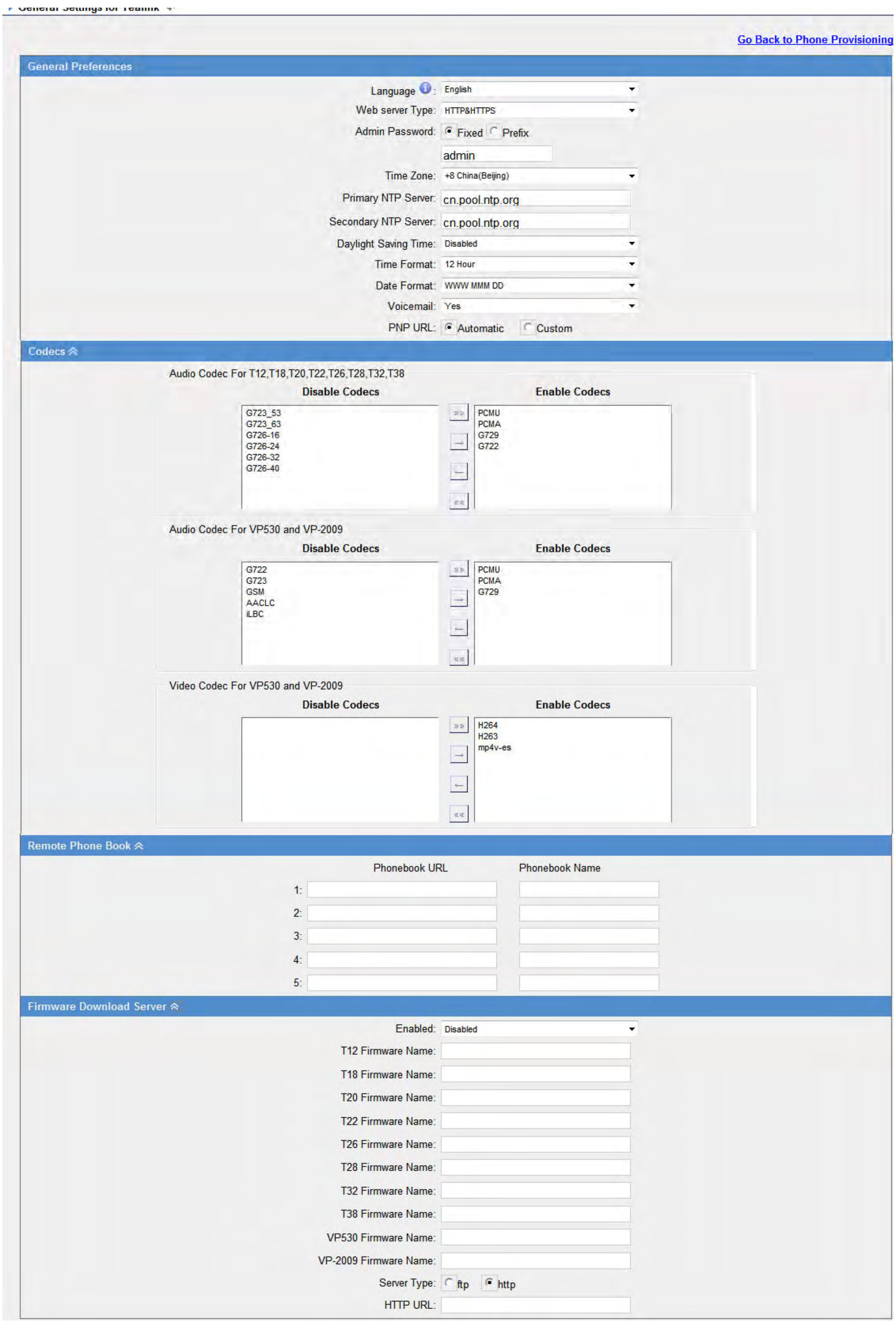


Figure 3.3.4

3.3.4.1 General Settings for Yealink

In this page, you can configure it before provisioning Yealink IP phones, including the items like general preferences, codecs, remote phone book and firmware upgrade

Note: if firmware download server is enabled, IP phone will update the firmware automatically according the version and server you have configured during the provision process.



Go Back to Phone Provisioning
General Preferences
Language: English
Web server Type: HTTP&HTTPS
Admin Password: Fixed Prefix
admin
Time Zone: +8 China(Beijing)
Primary NTP Server: cn.pool.ntp.org
Secondary NTP Server: cn.pool.ntp.org
Daylight Saving Time: Disabled
Time Format: 12 Hour
Date Format: WWW MMM DD
Voicemail: Yes
PNP URL: Automatic Custom
Codecs
Audio Codec For T12,T18,T20,T22,T26,T28,T32,T38
Disable Codecs
G723_53
G723_63
G726-16
G726-24
G726-32
G726-40
Enable Codecs
PCMU
PCMA
G729
G722
Audio Codec For VP530 and VP-2009
Disable Codecs
G722
G723
GSM
AACLC
iLBC
Enable Codecs
PCMU
PCMA
G729
Video Codec For VP530 and VP-2009
Disable Codecs
Enable Codecs
H264
H263
mp4v-es
Remote Phone Book
Phonebook URL
Phonebook Name
1:
2:
3:
4:
5:
Firmware Download Server
Enabled: Disabled
T12 Firmware Name:
T18 Firmware Name:
T20 Firmware Name:
T22 Firmware Name:
T26 Firmware Name:
T28 Firmware Name:
T32 Firmware Name:
T38 Firmware Name:
VP530 Firmware Name:
VP-2009 Firmware Name:
Server Type: ftp http
HTTP URL:

Figure 3.3.4.1

3.3.4.2 Phone book

You can add your contacts here and provision them to your IP phone.

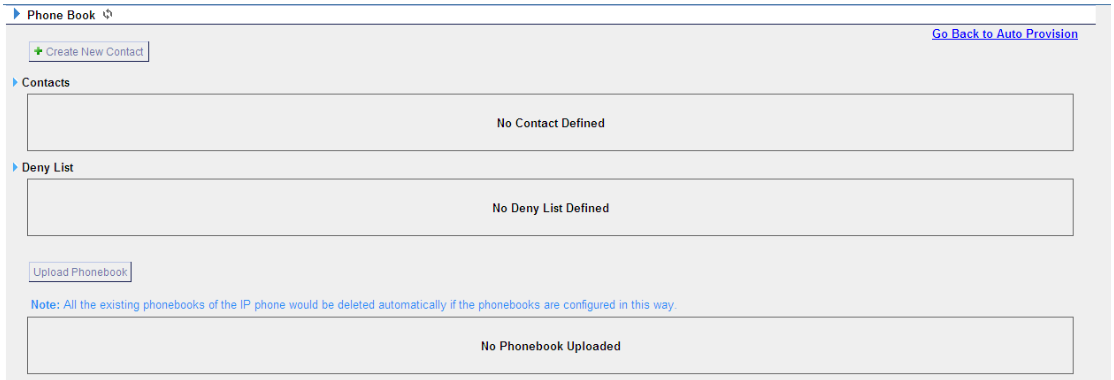


Figure 3.3.4.2

1) Add Contact

·Type

There are three types: None, VIP and Deny list (Blacklist).

·Group

There are 5 groups: None, Friends, Family, Work, Colleagues list.

·Nick Name

You can set a nick name for this number.

·Favorite

Only works with snom phone.

·Organization

Input the organization of this contact. Only works with snom phone.

·Title

Input the title of this contact. Only works with snom phone.

·Email

Input the email of this contact. Only works with snom phone.

·Birthday

Input the birthday of this contact. Only works with snom phone.

·First Name

Input the first name of this contact. Only works with snom phone.

·Family Name

Input the family of this contact. Only works with snom phone.

·Office Number

Input the office number here

·Mobile Number

Input the mobile number here

·Home Number

Input the home number here

·Sub Number

Add sub number of this contact. Only works with snom phone.

·Note

Take some note of this contact. Only works with snom phone.

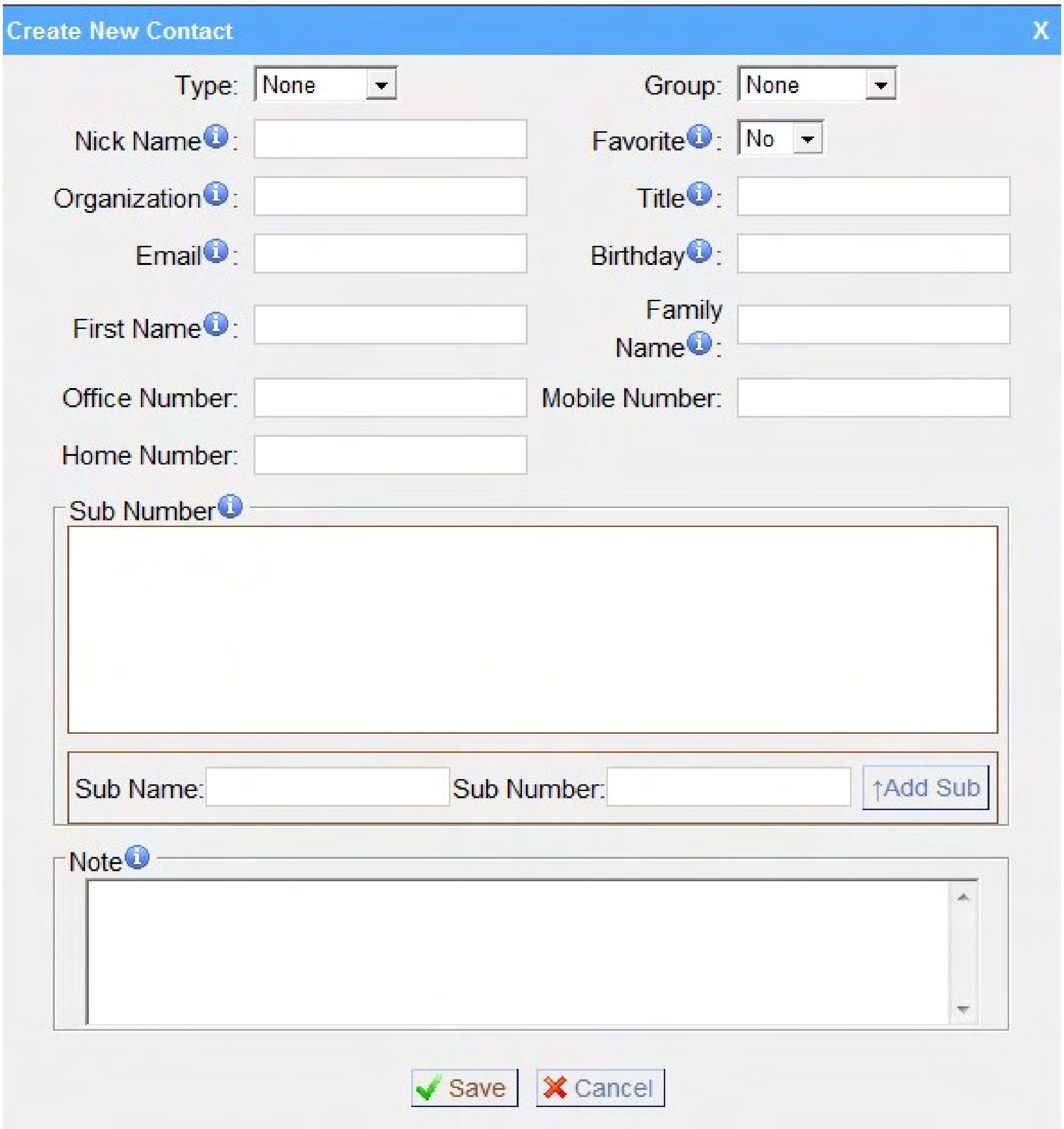


Figure 3.3.4.2.1

2) Upload Phonebook

You can upload a phonebook before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones. The format of phonebook should be *.xml.

Note: All the existing phonebooks of the IP phone will be replaced automatically if the phonebooks are configured in this way.

3.3.4.3 configure phone

Let's take provisioning yealink as an example

Create New Phone have two modes,

Create New phone in webpage and Upload the IP Phone's configure file.

1. Add new phone via webpage

Click 'Add Phone' and fill in the corresponding information in the popup window.

1) General

· Enabled

Choose yes or no to enable or disable this extension

· MAC address

Input the MAC address of IP phone

·Name

Put the name of this Phone here.

·Manufacturer

You can choose the Manufacturer of IP phone

·Phone Type

Choose the model of your phone. Only for snom phone

·Call Waiting

This call feature allows your phone to accept other incoming calls to an extension already in an active call.

·Key as Send

Configure the key as send, you choose # ,* or disable it

·Auto redial

Enable the auto redial for IP Phone

·Auto answer

Configure if auto answer is allowed for IP phone

·Phone book

Enable the feature of phone book of IP phone

·Line

Extension: Selected the extension number for IP Phone.

Label: It is shown on the LCD for users to identify the account.

Line Active: You can choose on/off to enable/disable the account respectively.

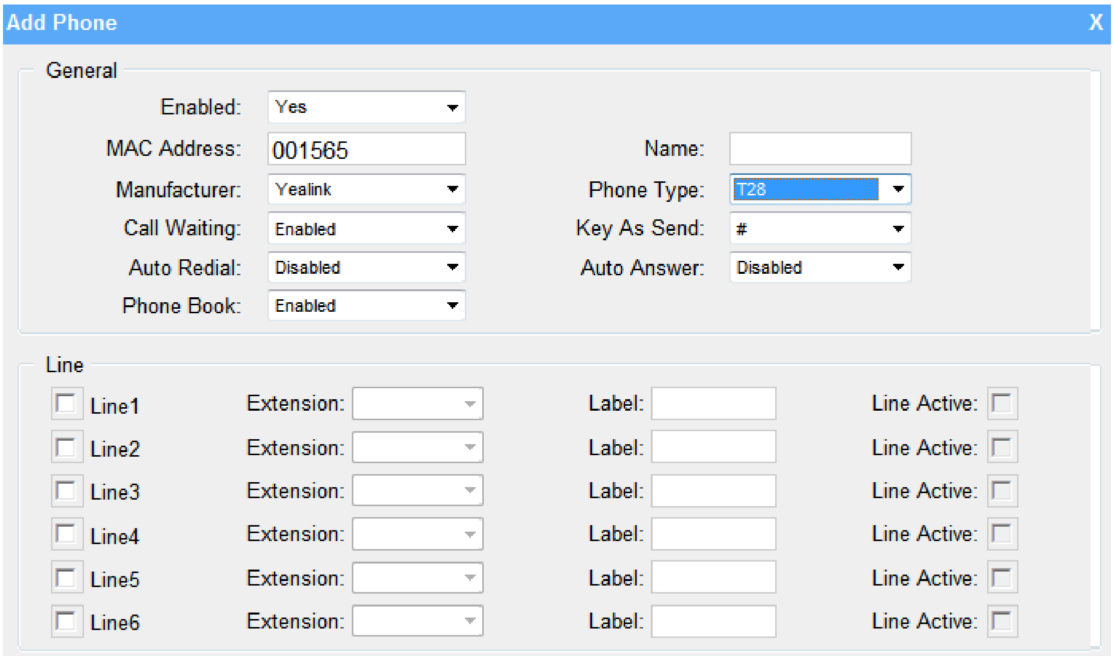


Figure 3.3.4.3

2) Audio codec

In this section, we can design the allowed codec for IP phone

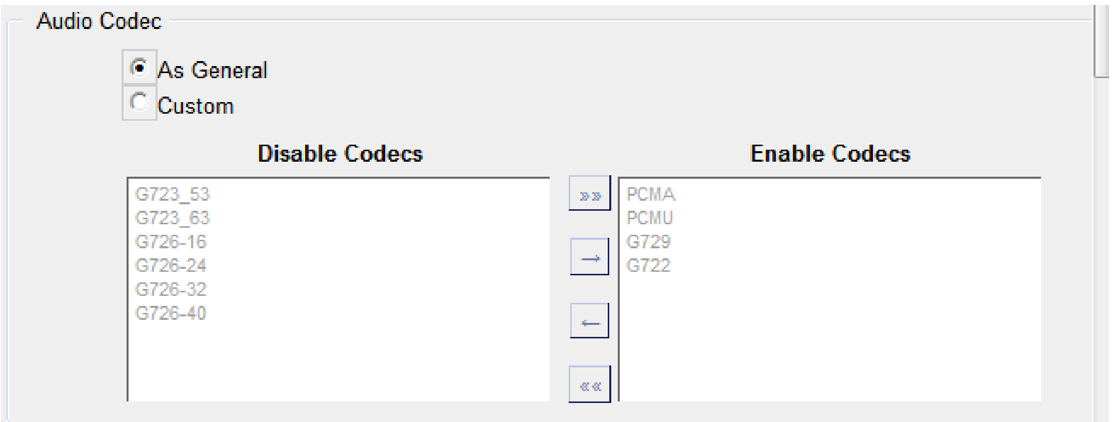


Figure 3.3.4.4

3) Line keys settings

Configure the DSS keys/Function Keys

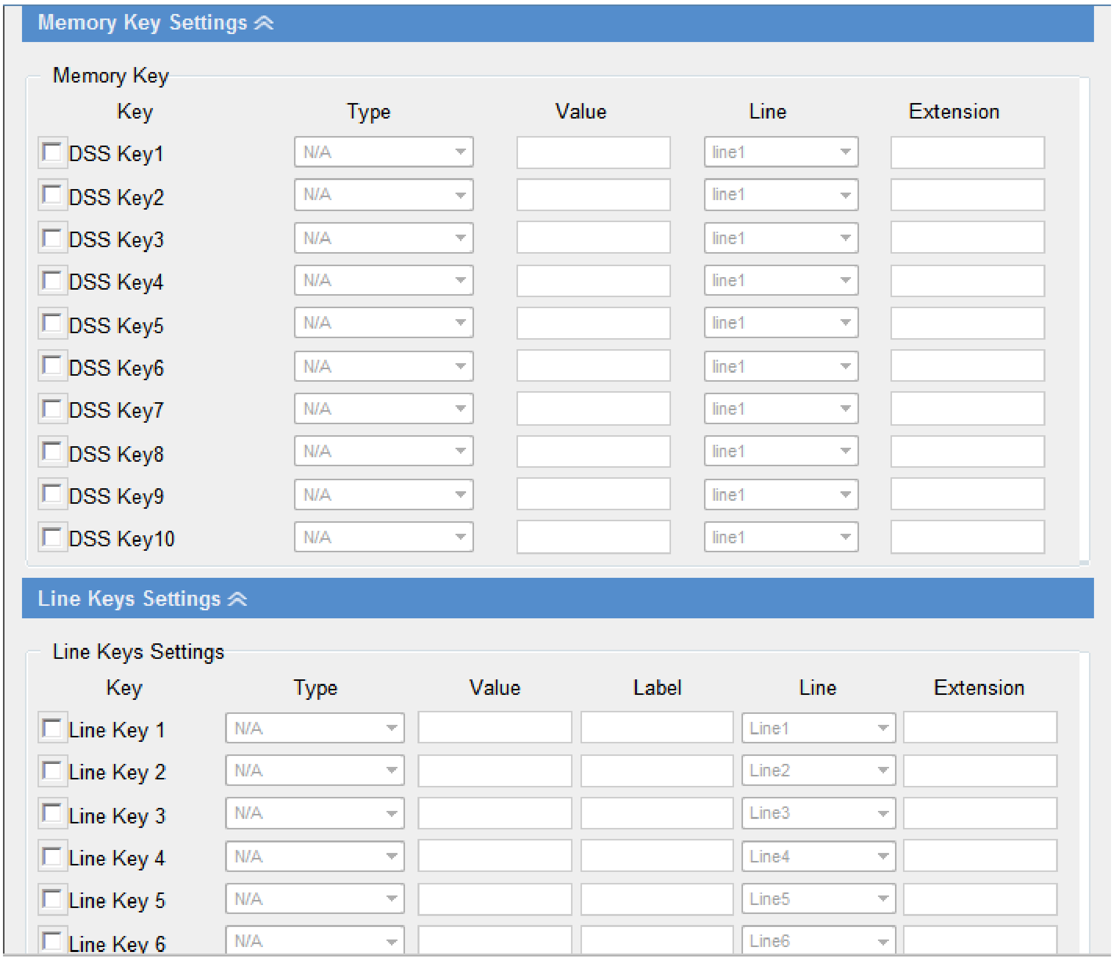


Figure 3.3.4.5

3.3.4.3 Not configured phone

In this section, MyPBX will scan all the supported IP phones and display here, we can click the 'MAC address' of IP phone and input the corresponding information in the popup window, like the picture shows below

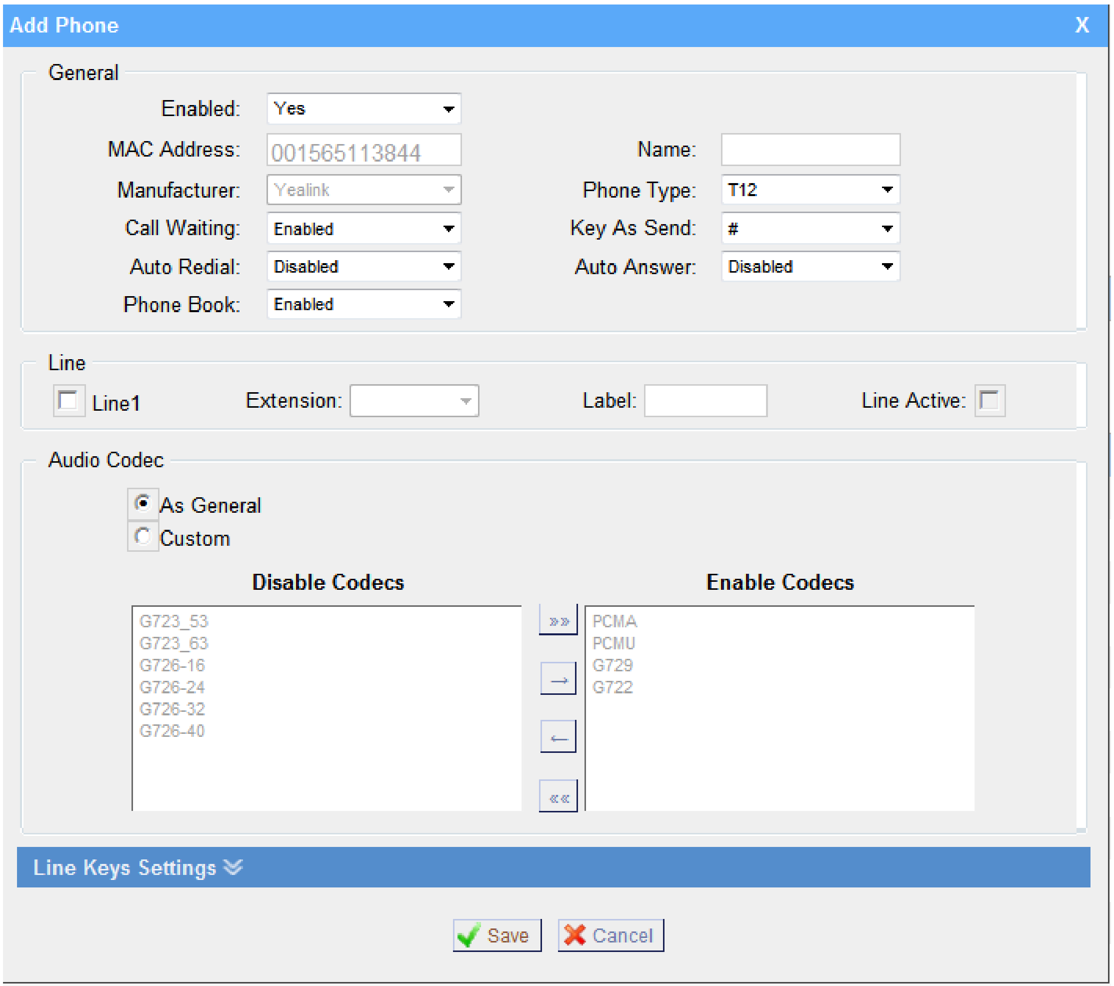


Figure 3.3.4.6

3.3.4.4 Upload a file

Click 'Upload a file' and choose the configure file of IP phone in the popup window.

Note: the file format must be .cfg

Please edit the configuration files in advance before uploading.

3.4 Inbound Call Control

3.4.1 IVR

When there's an inbound call aims at Auto Attendant, MyPBX will play an IVR recording and route the caller to the requested destination (for example, 'Welcome to XX company, for sales press 1, for technical support press 2, for operator press 0, etc'). The system will transfer the call to corresponding extension according to DTMF digits inputted by the user.

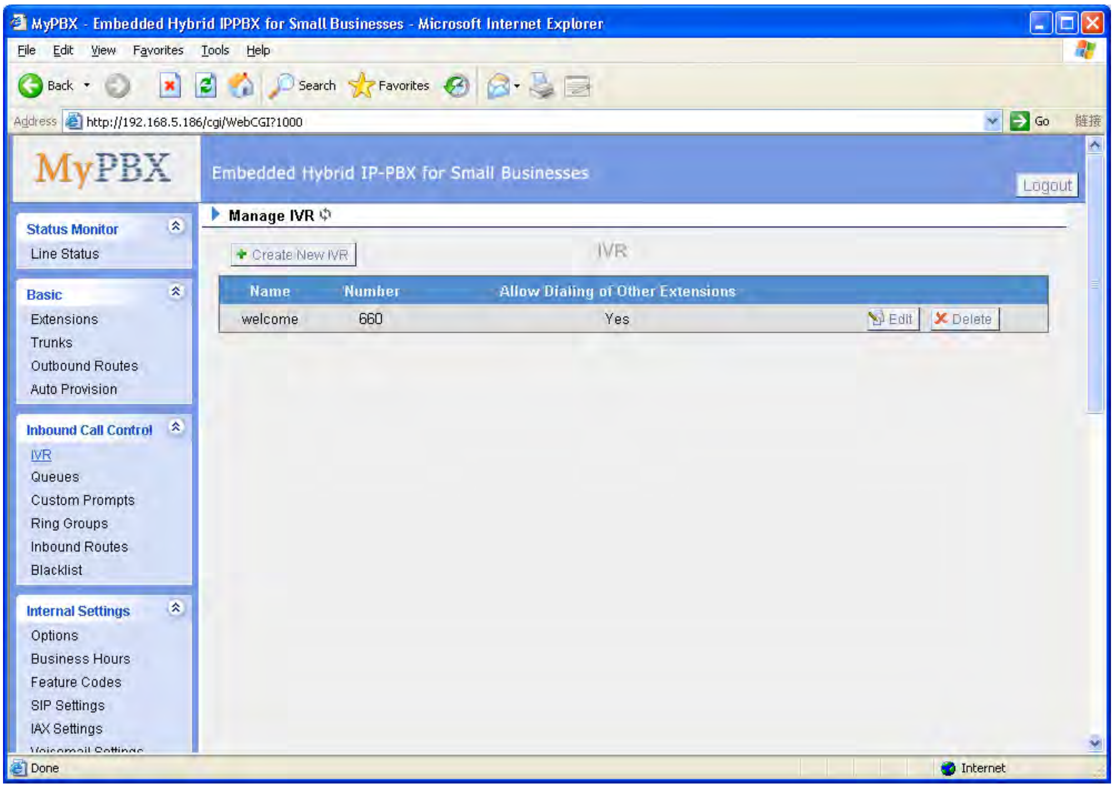


Figure 3.4.1

3.4.1.1 Create IVR

Click 'Create New IVR'.

1) General

·Number

MyPBX treats IVR as an extension; you can dial this extension number to reach the IVR.

·Name

A name for the IVR

·Prompt

The prompt recording that will be played when this IVR is reached.

·Repeat Count

The number of times that the selected IVR prompt will be played.

·Key Timeout

Wait for the user to enter a new extension for a specified number of seconds.

·Allow Dialing of Other Extensions

Allow the caller to dial other extensions other than the ones explicitly defined.

2) Key Press Events

A list of actions that can be performed depending on the digit dialed by the user .

·Key

The Key pressed when the callers hear the IVR prompt.

·Action

When the callers press the corresponding key, the action MyPBX executes.
No Action: Do nothing
Connect to Extension: Connect the call to an extension.
Connect to Voicemail: Connect the call to the voicemail of an extension
Connect to RingGroup: Connect the call to a ringgroup.
Connect to IVR:Connect the call to an IVR.
Connnect to Conference Room: Connect the call to a conference room.
Connect to DISA:Connect the call to a DISA.
Connect to Queue:Connect the call to a queue.
Connect to Faxes:Connect the call to Faxes of extensions.
Dial by Name:The callers can dial the name of an extension to connect to the corresponding extension.
Hung up: Hung up the call.

·Destination

Where will MyPBX route the call when the action occurs.

·Time Out

Defines the timeout action . A timeout occurs after the IVR prompt has finished playing for the number of times specified by the 'Repeat Count' field.

·Invalid

Defines the invalid action . The invalid action is triggered if the user enters a DTMF digit that is not defined for this IVR.

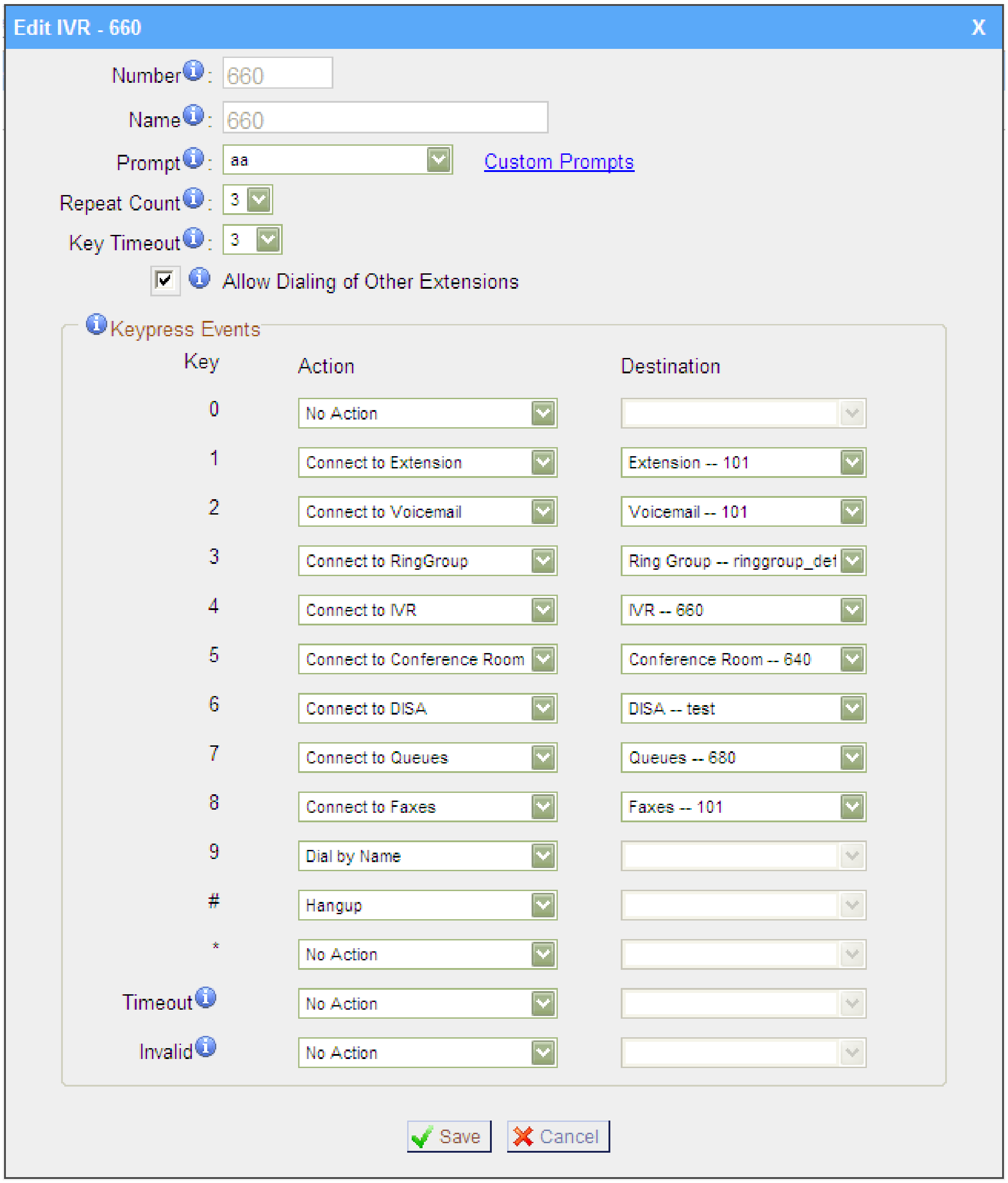


Figure 3.4.1.1

3.4.2 Queues

Call Queues give users (i.e. call centers) an efficient means to have their calls answered in the order they were received to deliver top tier customer service.

Call queues allow calls to be sequenced to one or more agents.
Note: Dial 'Queue number + '*'' to log in or 'Queue number + '**'' to log out the queue. For example, if the queue number is '680', then agent can dial '680*' to log in or '680**' to log out.

1） General

·Queue Name

A name for the Queue.

·Queue 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

This option sets the Ringing Strategy for this Queue. The options are
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.
Random: Ring a Random Agent.
RRmemory: Round Robin with Memory, Remembers where it left off in the last ring pass".

2） Agents

This selection shows all users. Selecting a user here makes them a agent of the current queue.

3) Caller Position Announcements

·Announce Position

Announce position of caller in the queue

·Announce Hold Time

Enabling this option causes MyPBX to announce the hold time to the caller periodically based on the frequency timer. Either yes or no; hold time will not be announced if <1 minute.

·Frequency

How often to announce queue position and estimated hold time.

Note: '0 seconds' means disable the announcement

4) Periodic Announcements

·Prompt

Select a prompt file to play periodically.

·Frequency

How often to announce a prompt to the caller.

5) Events

If a caller presses the key while waiting in the queue, this setting selects which action should process the key press.

6) Failover-Destination

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

7) Others

·Music On Hold

Select the 'Music on Hold' Class for this Queue.

·Leave When Empty

This option controls whether callers already on hold are forced out of a queue that has no agents. There are two options.

Yes: Callers are forced out of a queue when no agents are logged in.

No: Callers will remain in a queue with no agents.

·Join Empty

This option controls whether callers can join a call queue that has no agents. There are two options,

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 option is No.

·Agent Announcement

Announcement played to the Agent prior to bridging in the caller.

·Join Announcement

Announcement played to callers once prior to joining the queue.

·Retry

The number of seconds we wait before trying all the phones again.

·Wrap-up time

How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 30.


New Queues
Queue Name: 692
Queue Number: 692
Queue Password:
Queue Agent Timeout: 30
Queue Max Wait Time: 1800
Queue Ring Strategy: ringall
Agents
Available Agents
Selected
500(SIP)
501(SIP)
502(SIP)
503(SIP)
504(SIP)
505(SIP)
506(SIP)
507(SIP)
Caller Position Announcements
Announce Position: Yes
Announce Hold Time: Yes
Frequency: 30 seconds
Periodic Announcements
Prompt:
Frequency: 30 seconds
Events
Key:
Action: End Call
Destination:
Failover-Destination
Action: End Call
Destination:
Others
Music On Hold: calmriver
Leave When Empty: Yes
Join Empty: No
Agent Announcement:
Join Announcement:
Retry: 30
Wrap-up time: 30
Save
Cancel

Figure 3.4.2

3.4.3 Custom Prompts

1. Record new Prompt

The administrator can use this screen to record custom prompts by doing the following:

1）Click 'Record New Custom Prompt'

2) Input the desired file name on the popup window and choose an extension to call for recording (such as 500).

3) Click 'Record'. The selected extension will ring and you can pick up the phone to start recording.

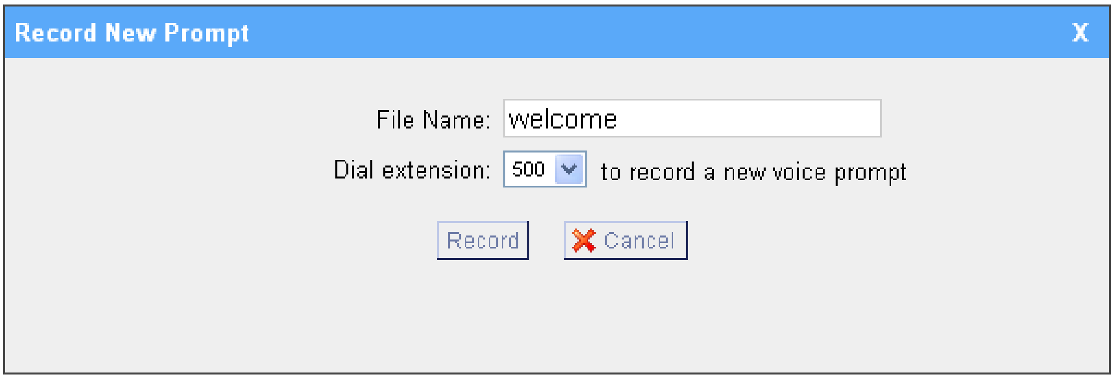


Figure 3.4.3.1

2. Upload Prompt

The administrator can also upload prompts by doing the following:

1)）Click 'Upload Prompt'.

2) Click 'Browse' to choose the desired prompt.

3) Click 'Upload' to upload the selected prompt.

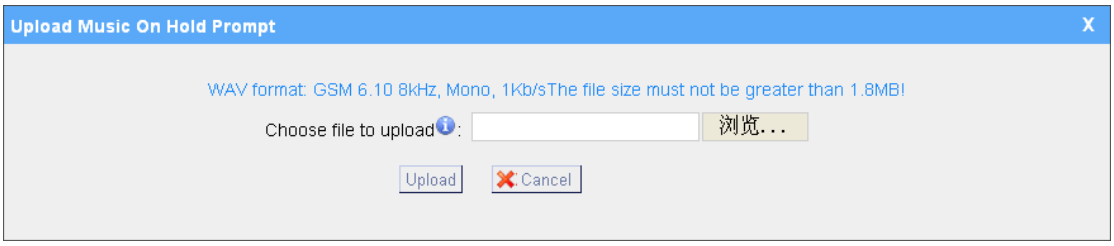


Figure 3.4.3.2

3.4.4 Ring Groups

Ring groups can be configured to balance the call traffic for multiple users and give callers a higher level of availability for incoming calls. Multiple ring methods and voicemail are supported.

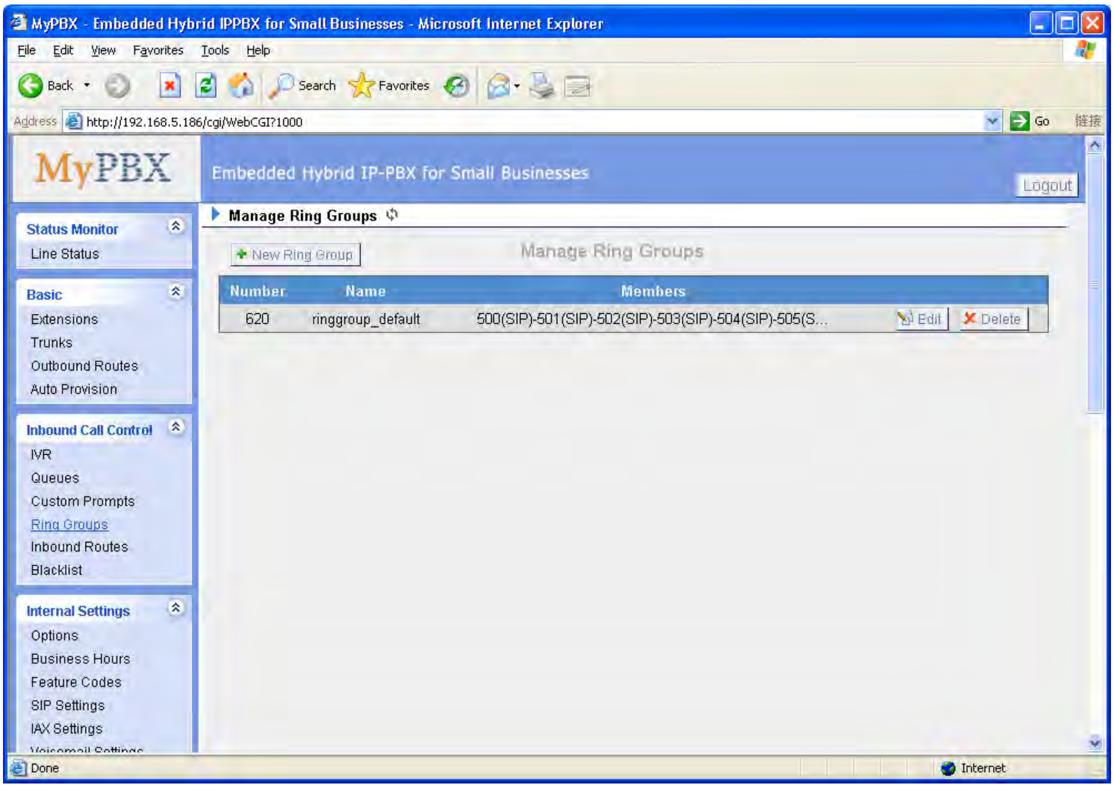


Figure 3.4.4

3.4.4.1 Create Ring Group

Click 'New Ring Group' to enter into the Manage Ring Groups page

1）General

·Ring Group Name

This option defines a name for this group, i.e. 'Sales'. 'Ring Group Name' is a label to help you identify this group in the group list.

·Ring Group Number

This option defines the numbered extension that can be dialed to reach this group.

·Strategy

This option sets the Ringing Strategy for this Group. The options are as follows:

1. Ring All Simultaneously: Ring all available Extensions simultaneously.
2. Ring Sequentially: Ring each extension in the group one at a time.

·Seconds to ring each member

1. If the strategy is 'Ring All Simultaneously', it means set the number of seconds to ring this group before routing the call according to the 'Destination if No Answer' settings.
2. If the strategy is 'Ring Sequentially', it means set the number of seconds to ring a single extension before moving onto the next one.

2) Ring Group Members

An extension can be made a member of this ring group by moving it into the 'Selected' box.

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 3.4.4.1

3.4.5 Inbound Routes

Inbound routing processes incoming call traffic to destination extensions during office hours or outside office hours.

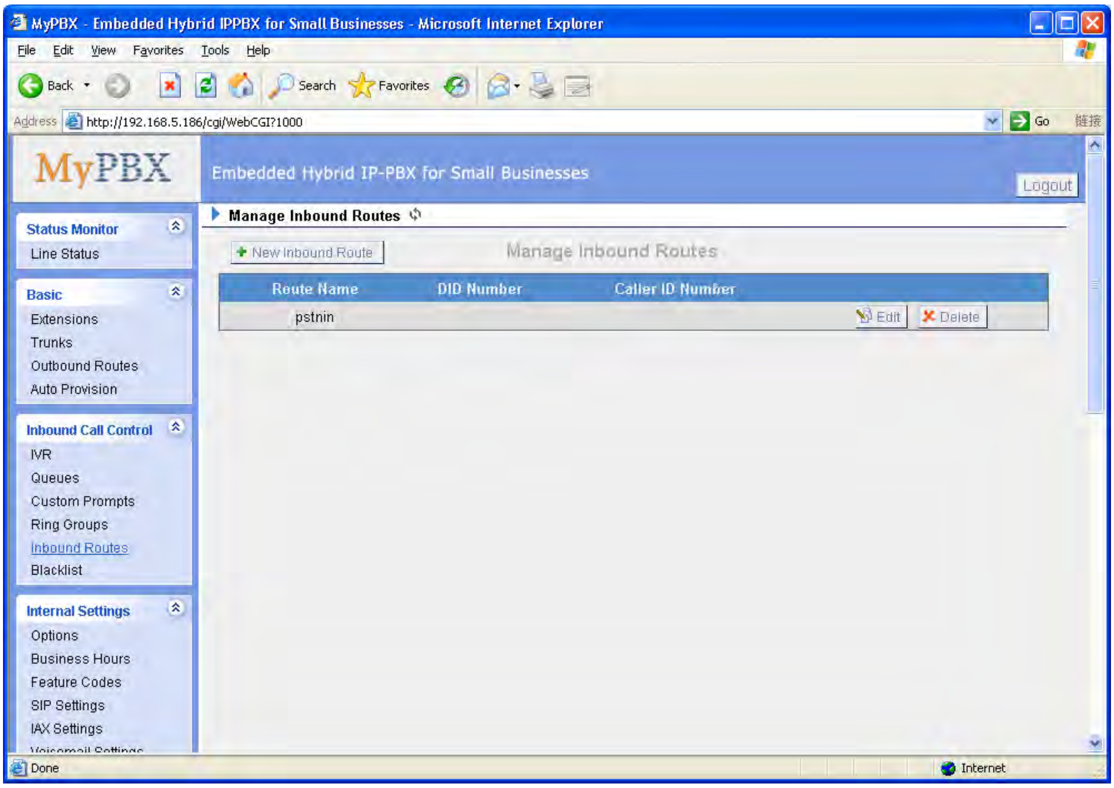


Figure 3.4.5

3.4.5.1 Create Inbound Route

Click 'New Inbound Route' to enter to the Manage Inbound Routes page.
When an incoming call arrives, the system will first check 'fax detection', then 'Holidays', at last 'Business Days'.

1) General

·Route Name

A name for this inbound route. Ex: 'pstncallin' etc.

·DID Number

Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. Only service provider, E1 trunks, BRI trunks or SIP trunks need to be configured with this setting.

You can also use pattern matching to match a range of numbers. The following patterns may be used:

X : Any Digit from 0-9

Z : Any Digit from 1-9

N : Any Digit from 2-9

[12345-9] : Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '**.**' Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '**!**' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.
Example 2: **1NXXNXXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

For more information, please refer to **Appendix G How to Use DID.**

·Extension
Define the extension for DID number. This field is only valid when you use BRI, SIP, SPS or SPX 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
Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no DID info.
You can also use a pattern match (e.g. 2[345]X) to match a range of numbers. The following patterns may be used:
X : Any Digit from 0-9
Z : Any Digit from 1-9
N : Any Digit from 2-9
[12345-9] : Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)
The '**.**' Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.
The '**!**' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.
Example 2: **1NXXNXXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

·Distinctive Ringtone
MyPBX support mapping to custom ring tone files. For example, if you configure the distinctive ringing for custom ring tone to '**Family**', the ring tone will be played if the phone receives the incoming call.

·Enable Callback
You can enable the callback function of this inbound route. If you want to configure the callback function, please refer to **chapter 3.5.12**

How do I configure distinctive ring tones? Please refer to APPENDIX E.
Currently distinctive ringtone can be compatible with Yealink and Snom phone.

2) Member Trunks

This area allows you to 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.

4) Business Days

Define where the calls will be routed during Business Days.

·Office Days

Select one defined business days office days.

·Office Hours Destination

Configure where to route the incoming calls during office hours.

·End Calls

Route the incoming calls to end calls, System will auto hang-up the call.

·Extension

Route the incoming calls to a specific extension.

·Voicemail

Route the incoming calls to 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

Route the incoming calls to a specific Conference Room.

·DISA

Route the incoming calls to a specific DISA.

·Queues

Route the incoming calls to a specific Queue.

·Faxes

Route the incoming faxes to a specific extension’s mail address.
Note: This function only supports T.38 faxes.

·Outbound Routes

Route the incoming calls to a specific outbound route.

This function is mainly used for the connection of two branches.
For example: Company A locates headquarters in the USA with a branch B in China. A and B both have MyPBX phone systems .
Now if staff of A would like to make a call to a telephone or mobile phone in China from the extension of A but via the FXS line of B, that can be done by this configuration.

·Non-office Hours Destination
Configure where to route the incoming calls during non-office hours.

5) During Holidays
Define where the calls will be routed during Holidays.

·Holiday
Select the 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.

6) Fax Detection
Configure if detecting faxes in this inbound route.

·Destination
Configure where the faxes will be routed when faxes are detected.

·No detect
Do not detect faxes.

·Custom Email
Customize an E-mail address to receive the faxes. You should first configure the ‘Voicemail Settigns->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 receive faxes with custom Email address, the ‘SMTP settings’ of ‘Voicemail Settings’ should be configured successfully in advance. If you 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.

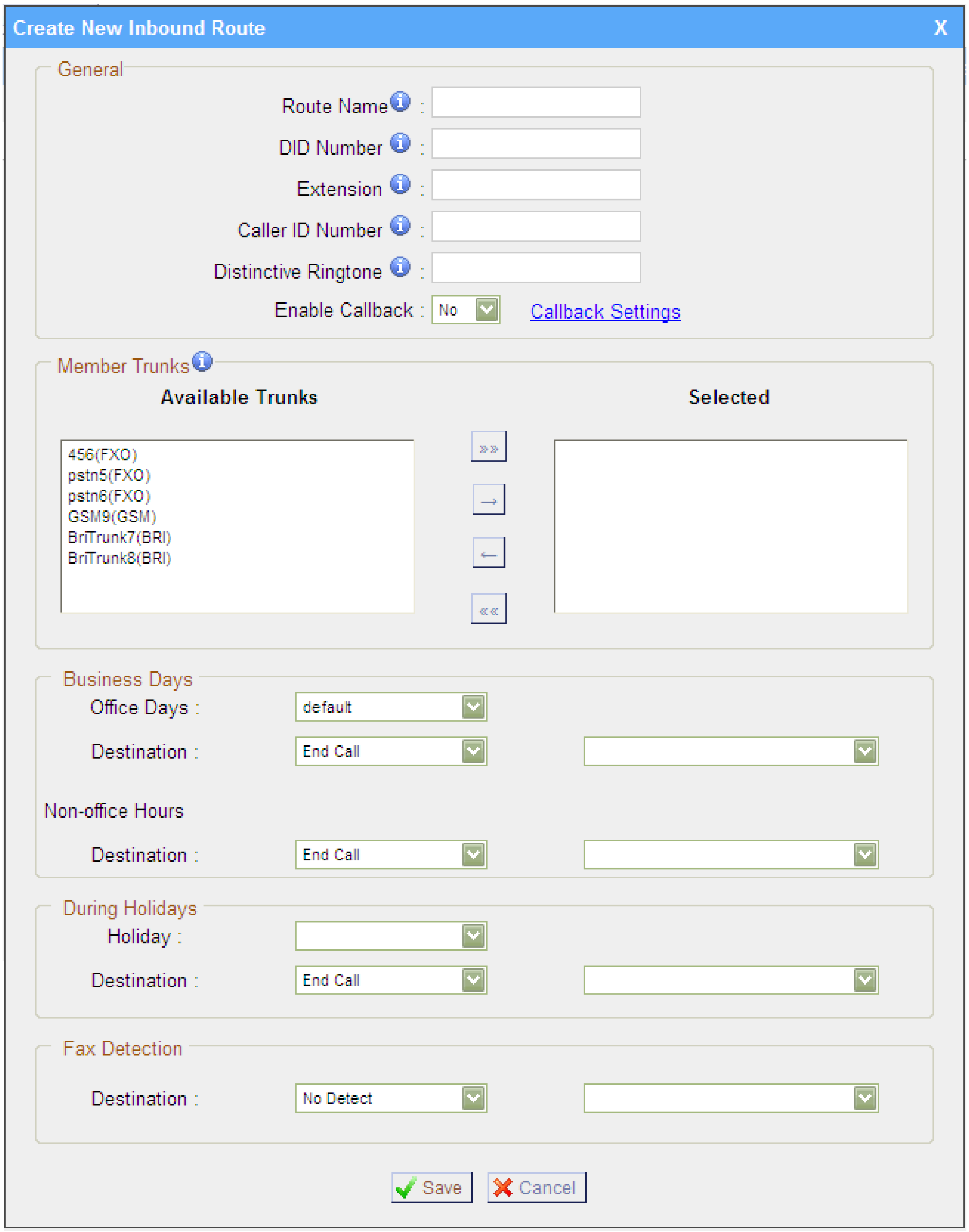


Figure 3.4.5.1

3.4.6 Blacklist

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

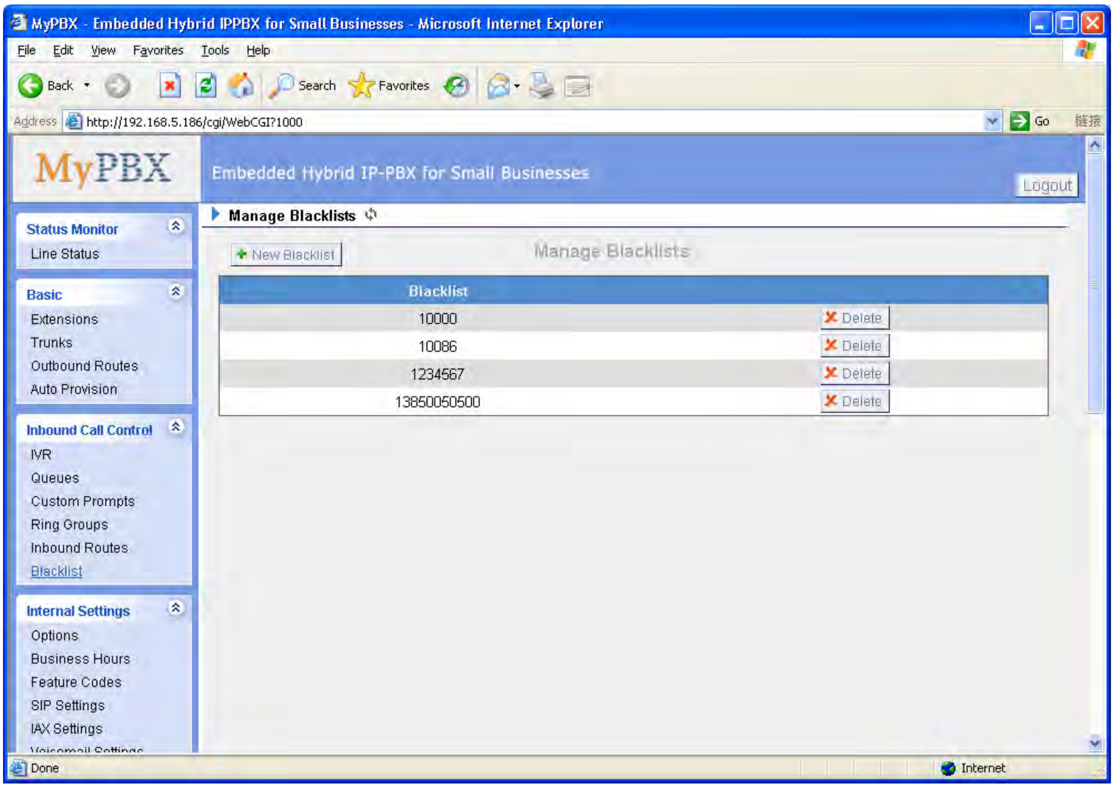


Figure 3.4.6

3.4.6.1 Create Blacklist

Click 'New Blacklist' to create a new number blacklist.

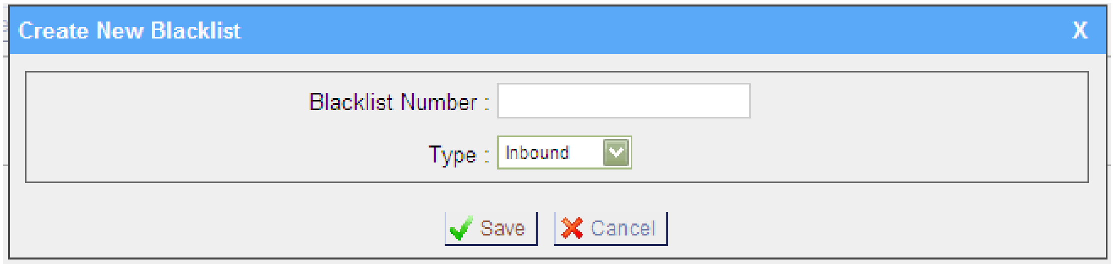


Figure 3.4.6.1

·Blacklist number

Enter the number you would like to block.

·Type

The number blocked for incoming or outgoing calls or both.

3.5 Internal Settings

3.5.1 Options

1) General

·**Ring Timeout**

Number of seconds to ring a device before handling the call as per the extension's Follow Me settings . Default value is 30s.

.**MAX call duration**

The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout. Default value is 6000s.

.**Maximum concurrent calls**

Maximum concurrent calls limits. Default value 0 means no limit

·**Music on hold**

Used to set hold music for the system.

·**Tone Region**

Please select your country or nearest neighboring country to enable the default dial tone, busy tone, and ring tone for your region.

Note: please reboot the system to take it effect.

·**HTTP bind port / Web Access Port**

Port to use for HTTP sessions . Default: 80

Note: please reboot the system to take it effect.

·**FXO Mode**

FXO port's operation mode .

·**Enable Follow Me Prompt**

When set Follow me to Transfer to number on the extension page (e.g. when 500 is busy, transfer to 501), while 500 is busy, the call will be transferred to 501. 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.

·**Virtual Ring Back Tone**

It's only for GSM trunk. Once enabled, when the caller call out with GSM trunks, the caller will only hear the virtual ring back tone generated by the system before callee answers the call.

· Distinctive Caller ID

When incoming calls are routed from ring group/queue/IVR, the caller ID displays with the name of ring group/queue/IVR , for example 5503302(ringgroup_default)

Note: To display IVR's name, please press the key instead of the extension number directly.

·Follow Me Prompt

Configure whether to play a prompt 'please wait while trying to look at 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.

2) Extension Preferences

·User Extensions

The default value is 500 to 616

·Ring Group Extensions

The default value is 620 to 629

·Paging Group Extensions

The default value is 630 to 639

·Conference Extensions

The default value is 640 to 659

·IVR Extensions

The default value is 660 to 679

·Queue Extensions

The default value is 680 to 689

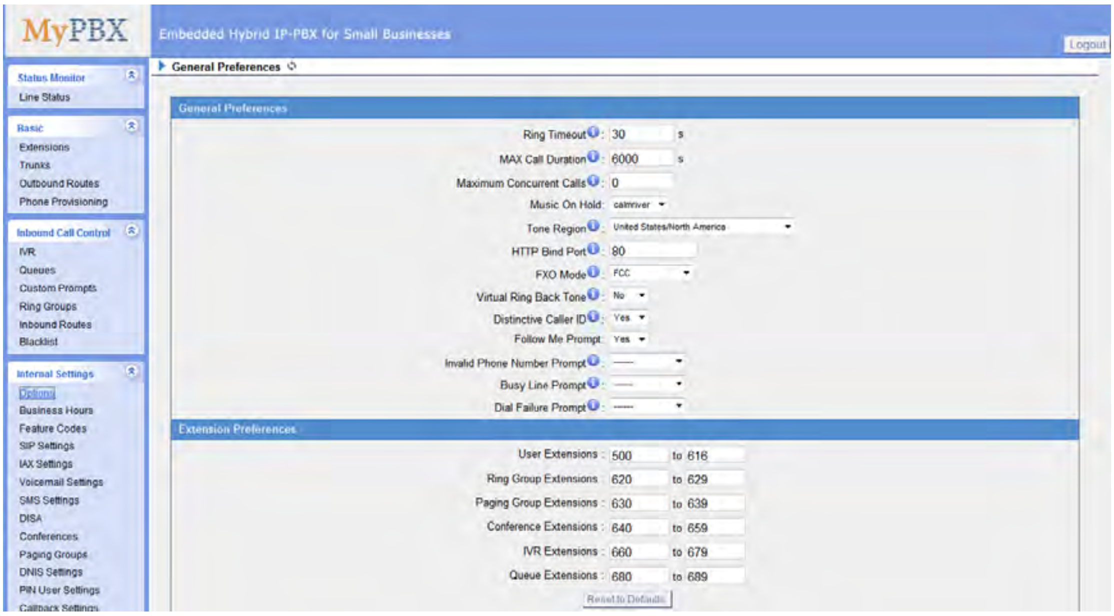


Figure 3.5.1

3.5.2 Business Hours

1) General

·Enable or Disable Business Hours

2) Others

·Enable Office Closed Timing

By dialing *81 (*81 is default) on an extension will force the office time closed for the device whatever the general setting is.

·Disable Office closed timing

By dialing *081 (*081 is default) on an extension will disable the Office Closed Timing.

3) Business Days

You can setup the business hours here.

4) Holidays

You can setup the holidays here.

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

Figure 3.5.2

3.5.3 Feature Codes

1) General

·One Touch Record

A user may initiate or stop call recording by dialing *1 during a call. (*1 is default setting)

·Extension for Checking Voicemail

Users can check their Voicemail by dialing *2 on their phone (*2 is default setting).

·Voicemail main menu

Users can go to the main menu by dialing *02 (*02 is default setting).

·Attended Transfer

Users may transfer an incoming call by dialing *3 on their phone (*3 is default setting).

·Attended Transfer Timeout

The time out of transferring a call

·Blind Transfer

Users may blind transfer an incoming call by dialing *03 on their phone (*03 is default setting).

·Call Pickup

Users may pick up an incoming call by dialing *4 on their phone (*4 is default setting)

·Extension Pickup

Users may pick up a specific extension's incoming call by dialing *04+extension number on their phone (*04 is default setting)

·Intercom

Define the feature code that is used to dial an extension in intercom mode. For instance setting this value to *5 would allow you to initiate an intercom call with extension 501 by dialing *5501.

·Normal Spy

In this mode, you can only listen to the extension being spied, for example you can dial *90501 to monitor extension 501

·Whisper Spy

In this mode you can listen/whisper to the extension being spied, for example, dialing *91501 to listen to extension 501, you can also talk with 501 too.

·Barge Spy

In this mode, you can barge in both extensions involved the call, for example dialing *92501 to barge in and talk with all the extensions inside

2) Call Park Preferences

·Call Parking

User may park an incoming call on his own telephone by pressing '*6' (*6 is default setting)

·Extension range used to park calls

User may park an incoming call on a designated extension at first and then pick up the call again on any other extension.

·Number of seconds a call can be parked before it is recalled.

Defines the number of seconds that a call can be parked before it is recalled to the station that parked it .

3) Call Forwarding Preferences

·Reset to Defaults

Users may reset all call forward defaults by calling *70 on their phone (*70 is default setting).

Note: When reset to defaults. 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
Users may enable always forward by calling *71 on their phone (*71 is default setting)

·Disable Forward All Calls
Users may disable always forward by calling *071 on their phone (*071 is default setting)

·Enable Forward When Busy
Users may enable busy forward by dialing *72 on their phone (*72 is default setting)

·Disable Forward When Busy
Users may disable busy forward by calling *072 on their phone (*072 is default setting)

·Enable Forward No Answer
Users may enable no answer forward by calling *73 on their phone (*73 is default setting)

·Disable Forward No Answer
Users may disable no answer forward by calling *073 on their phone (*072 is default setting)

·Forward to number
Users may activate call forwarding by dialing this feature code, followed by the extension or phone number to forward all calls to.

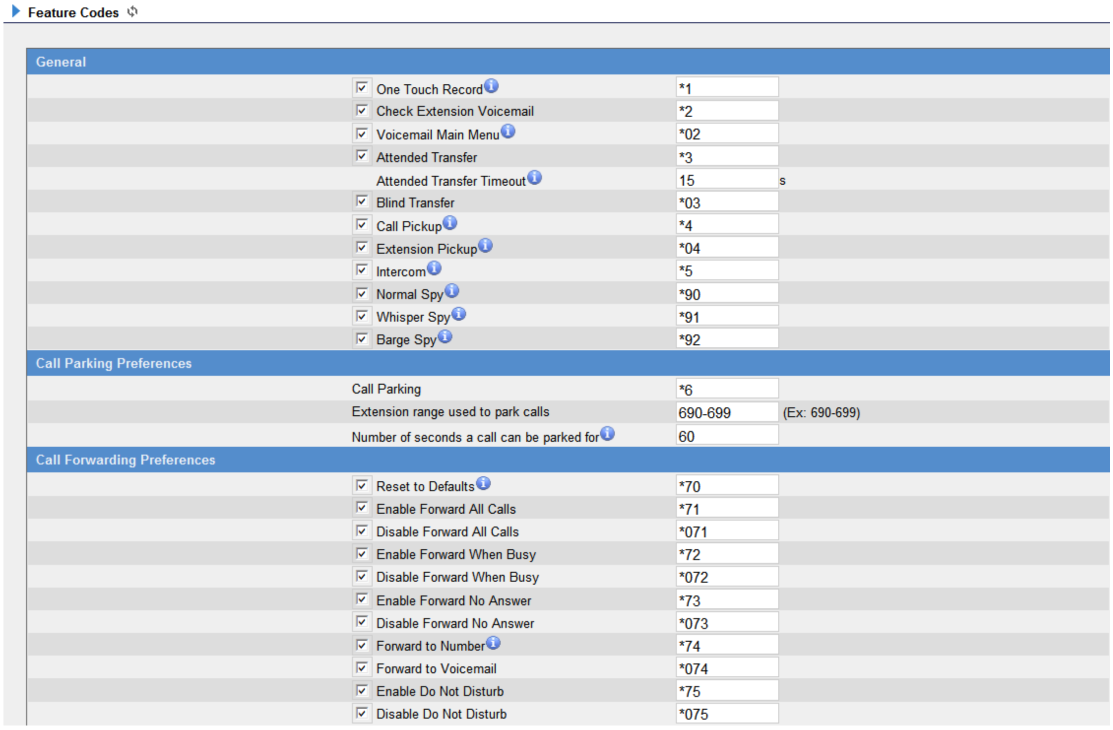
Note: Users may activate Forward to number by dialing *74 + phone number. e.g.: by dialing *74501, all calls will be forwarded to extension 501.

·Forward to Voicemail
Users may forward the call to Voicemail by calling *074 on their phone (*074 is default setting)

·Enable Do Not Disturb
Users may enable do not disturb by calling *75 on their phone (*75 is default setting)

·Disable Do Not Disturb
Users may disable do not disturb by calling *075 on their phone (*075 is default

setting)



Feature Codes

General		
☑ One Touch Record	*1	
☑ Check Extension Voicemail	*2	
☑ Voicemail Main Menu	*02	
☑ Attended Transfer	*3	
Attended Transfer Timeout	15	s
☑ Blind Transfer	*03	
☑ Call Pickup	*4	
☑ Extension Pickup	*04	
☑ Intercom	*5	
☑ Normal Spy	*90	
☑ Whisper Spy	*91	
☑ Barge Spy	*92	
Call Parking Preferences		
Call Parking	*6	
Extension range used to park calls	690-699	(Ex: 690-699)
Number of seconds a call can be parked for	60	
Call Forwarding Preferences		
☑ Reset to Defaults	*70	
☑ Enable Forward All Calls	*71	
☑ Disable Forward All Calls	*071	
☑ Enable Forward When Busy	*72	
☑ Disable Forward When Busy	*072	
☑ Enable Forward No Answer	*73	
☑ Disable Forward No Answer	*073	
☑ Forward to Number	*74	
☑ Forward to Voicemail	*074	
☑ Enable Do Not Disturb	*75	
☑ Disable Do Not Disturb	*075	

Figure 3.5.3

3.5.4 SIP Settings

1) General

·UDP Port

Port use for sip registrations, Default is 5060.

·TCP Port

Port use for sip registrations, Default is 5060.

·TLS Port

Port use for sip registrations, Default is 5061.

·RTP Port Start

Beginning of RTP port range

·RTP Port End

End of 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 . Default is 3600 seconds.

·Min Registration / Subscription Time

Minimum duration (in seconds) of a SIP registration . 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 8 times.

·Register Timeout

Number of seconds to wait for a response from a SIP Registrar before timed out . 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. 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 useragent parameter of asterisk, the default is ‘MyPBX’, you should change it if needed.

2) NAT

Note: Configuration of this section is only required when using remote extensions.

·Enable STUN

STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.

·STUN Address

The STUN server allows clients to find out their public address, the type of NAT they are behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between the client and the VOIP provider and so establish a call.

·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 Reinvite

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.

u-law: A PSTN standard codec, used in North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

a-law: A PSTN standard codec, used outside of North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

GSM: A wireless standard codec, used worldwide, that provides adequate voice quality and consumes 13.3kbit/s in each direction (receiving and transmitting) of a VoIP call. GSM is supported by many VoIP phones.

SPEEX: Speex is an Open Source/Free Software patent-free audio compression format designed for speech. The Speex Project aims to lower the barrier of entry for voice applications by providing a free alternative to expensive proprietary speech codecs. Moreover, Speex is well-adapted to Internet applications and provides useful features that are not present in most other codecs.

G.722: G.722 is a wideband speech coding algorithms which supports the bit rate of 64, 56 and 48kbps wideband. It's a broadband voice encoding of G series.

G.726: A PSTN codec, used worldwide, that provides good voice quality and consumes 32kbit/s in each direction (receiving and transmitting) of a VoIP call. G.726 is supported by some VoIP phones.

ADPCM, G.729A, H261, H263, H263p, H264,MPEG4.

Note: If you would like to use G.729, please enter your license.

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.

5) 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.

·Qualify
Send check alive packets to the sip provider.

·Remote Party ID
Whether send Remote-Party-ID on SIP header. Default no.

·Session -timers
Enable sesstion-timer mode, default: yes

·Sesstion-expires
The max refresh interval

·Sesstion-minse
The min refresh interval, which mustn't be less than 90s

·Sesstion-refresher
Choose sesstion-refersher, the default is Uas

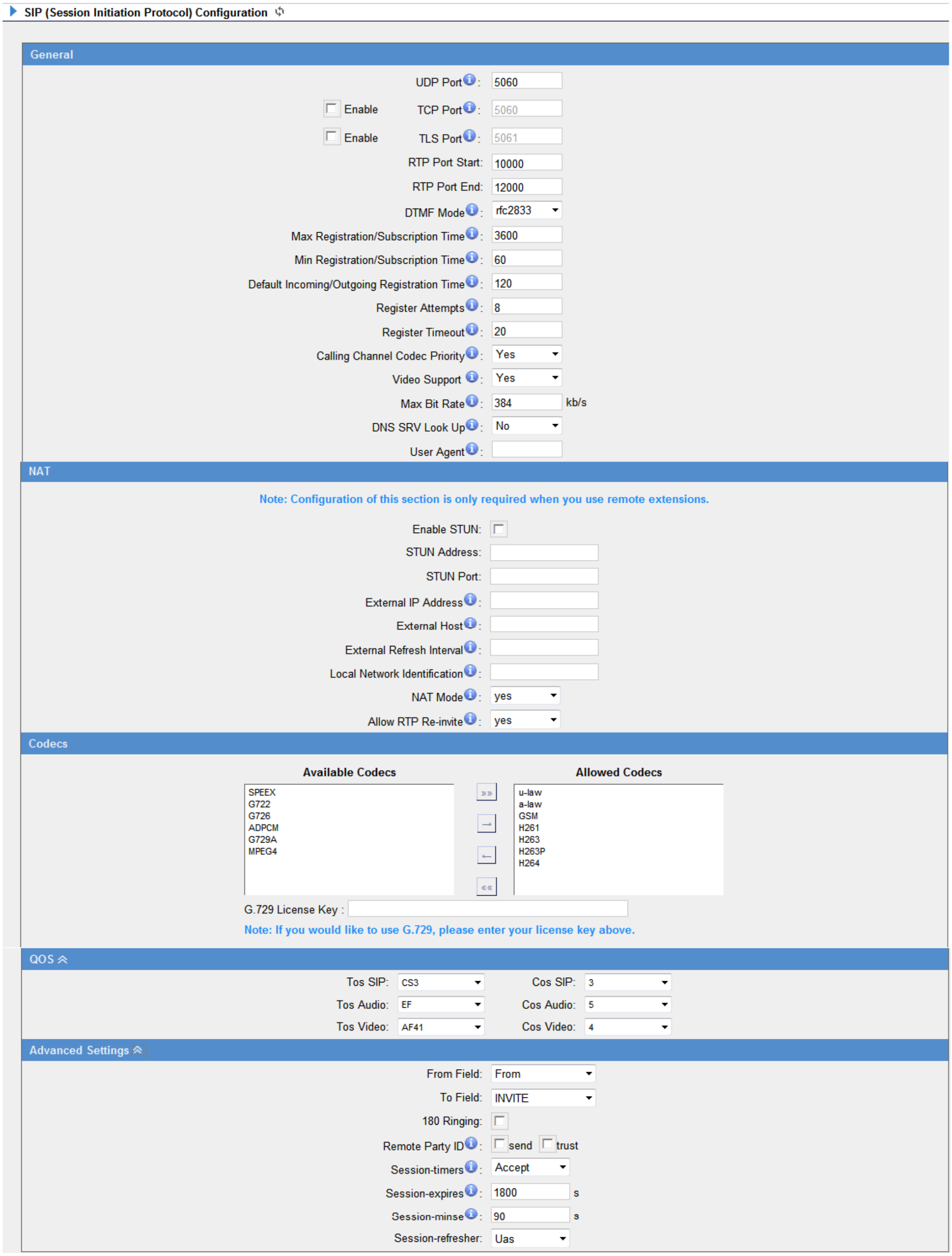


Figure 3.5.4

3.5.5 IAX Settings

1) General

·Bind Port

Port use for IAX2 registrations, Default 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 a IAX2 registration. Default is 60 seconds.

·Max Registration Time
Maximum duration (in seconds) of a IAX2 registration. Default is 1200 seconds.

2）Codecs
A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet.
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ADPCM, G.729A, H261, H263, H263p, H264.
Note: If you would like to use G.729, please enter your license.

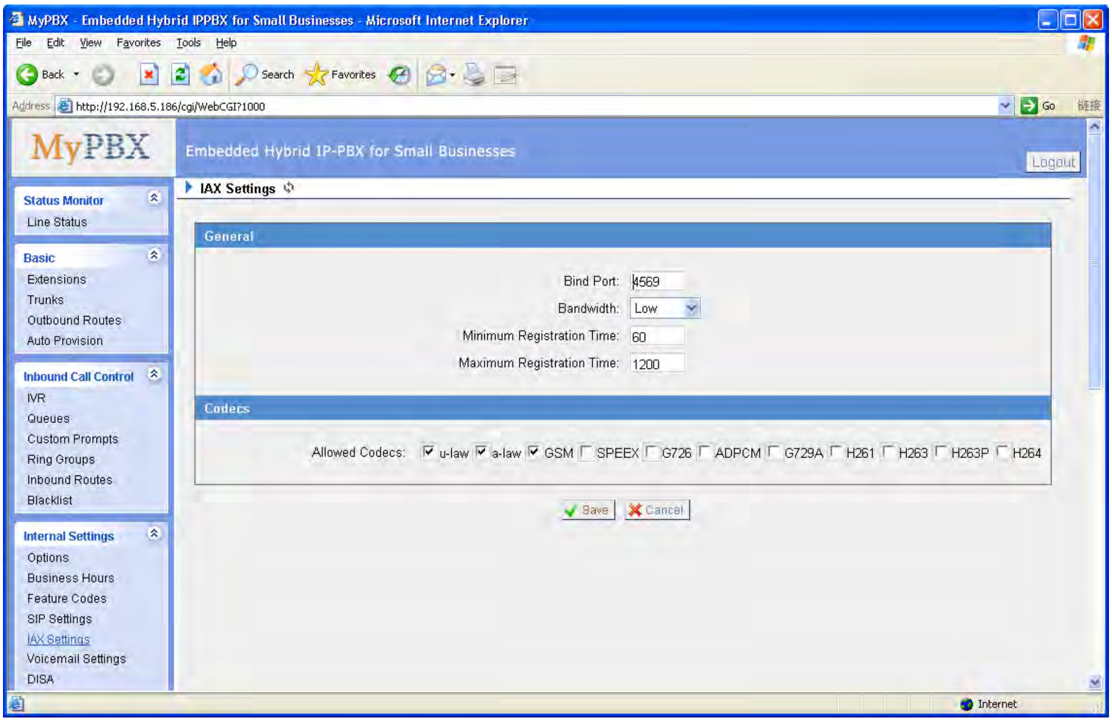


Figure 3.5.5

3.5.6 Voicemail Settings

1) General Voicemail Settings

a) Message Options

·Max Messages 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.

·Operator Breakout from Voicemail

If this option is set, the caller can jump out of the voicemail and go to the destination (IVR) you set by dialing “0”.

b) Greeting Settings

·Busy Prompt

Greeting played when the extension called is busy.
Skip greeting: Do not play a greeting.
Play busy greeting: play the extension busy greeting.

·Unavailable Prompt

Greeting played when the extension called is Unavailable.
Skip greeting: Do not play a greeting.
Play Unavailable greeting: play the extension Unavailable greeting.

·Leave a Message Prompt

Greeting when ask the caller to dial 5 to leave a message.
Skip greeting: Do not play a greeting.
Play busy greeting: play the extension busy greeting.
Play Unavailable greeting: play the extension Unavailable greeting.

c) Playback Options

·Announce Message Caller ID

If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing.

·Announce Message Duration

If this option is set, the duration of the message in minutes will be played back before the voicemail message begins playing.

. Announce Message Arrival Time

If this option is set, the arrival time of the message will be played back before the voicemail message begins playing.

. Allow Users to Review Messages

Allow callers to review their recorded message before sending it to voicemail.

2) SMTP Settings for Voicemail

Note: If you want to send voicemail messages as email attachments, please configure this section.

·E-mail Address

The E-mail Address that MyPBX 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 will connect to in order to send voicemail messages via email, i.e. mail.yourcompany.com.

·Port
SMTP Port: the default value is 25.

·Use SSL/TLS to send secure message to server
If the server of sending email needs to authenticate the sender, you need to select the check box.
Note: Must be selected for Gmail or exchange server.
After filling out the above information, you can click on the 'Test Account Settings' button to check whether the setup is OK.
1) If the test is successful, you can use the email safely.
2) If test failed, please check the above information is correct or network is proper.

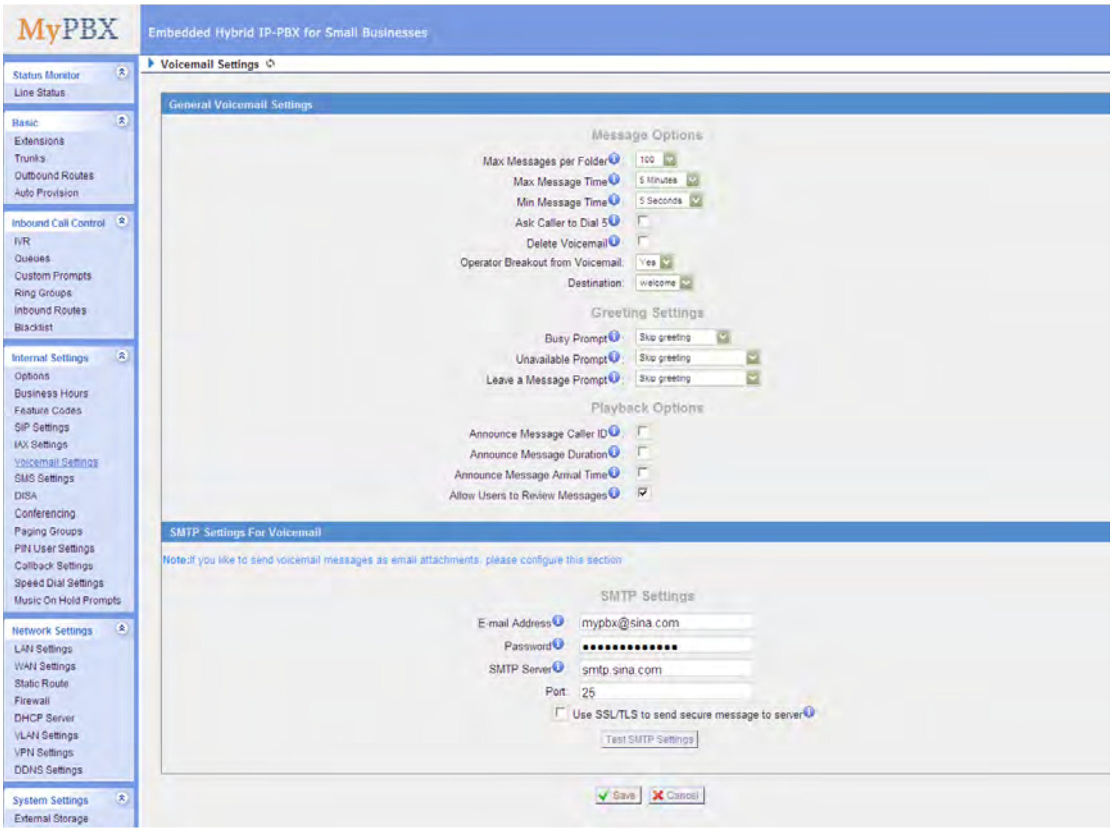


Figure 3.5.6

3.5.7 SMS Settings

MyPBX supports SMS to Email and Email to SMS. (Only for GSM trunks)

1) Enable SMS to Email
If you enable this, as soon as the GSM trunks receive small messages, MyPBX

will send the text of this message to the email addresses listed on the Email List. You can add email addresses to the Email List.

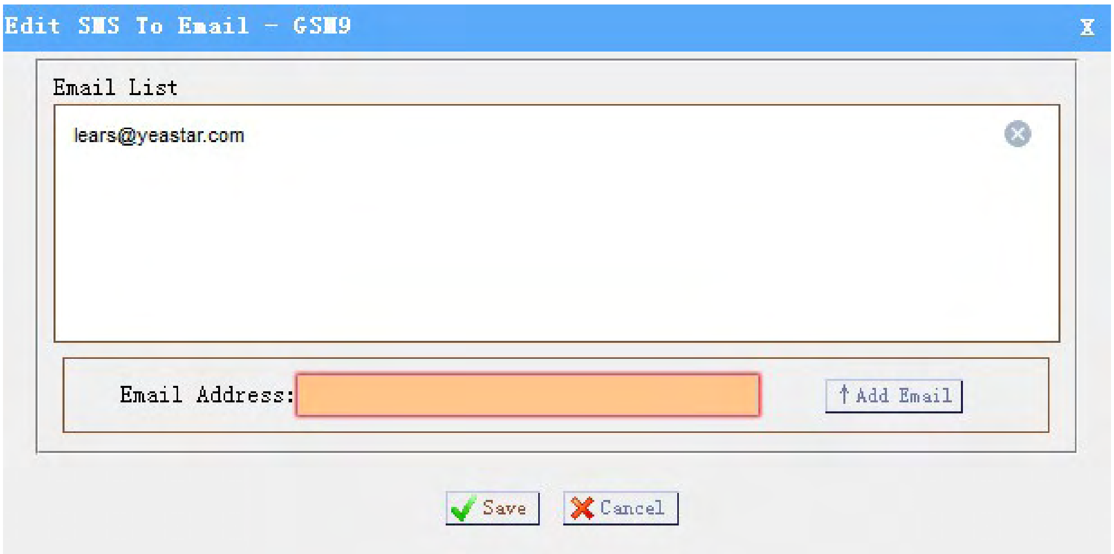


Figure 3.5.7.1

2) Enable Email to SMS

If you enable this, you can use MyPBX to send out message by sending an email to the specified address.

·Enable Country Code

If you want to add country code before the dialed numbers, please tick this.

·Country Code

The country code to be add before the dialed numbers.

·Receive mails every

The intervening time of receiving mails from POP3 server.

·Access Code

This PIN code is used to verify the subject of the emails received. If the form of email passes the verification, it will be send out by SIM card. If not, this email will be deleted immediately.

3) Email Settings

Note:

1. If you want to use “SMS to Email”, please configure POP3 setting.
2. If you configure the POP3 setting, MyPBX will download emails from the mail server regularly. Once downloaded, the emails will be deleted from the mail server.

·Email Address

This email address will be used to:

1. Send email to the addresses listed on "SMS to Email" setting.
2. Receive email and send the text of the email to the target mobile number by SMS.

Note: If you use gmail, just put your user name here. E.g. email address: test@gmail.com, you just put "test" here.

·Password

Input the password of this email here.

·SMTP Server (SMTP)

·SMTP Server Port

·Receive Server (POP3)

·Receive Server Port

If you want to know more about Email to SMS, please refer to APPENDIX F

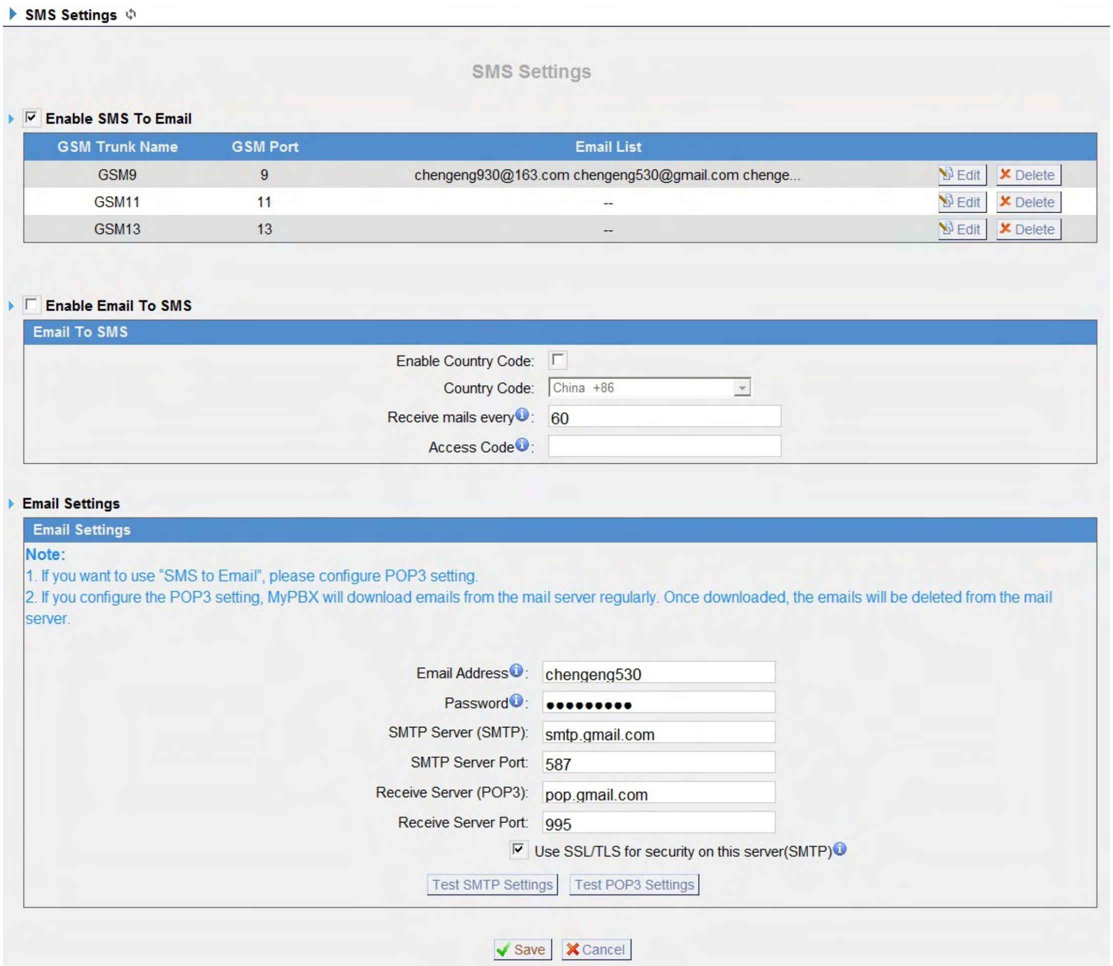


Figure 3.5.7.2

3.5.8 DISA

DISA (Direct Inward System Access) allows someone calling in from outside the telephone switch (PBX) to obtain an 'internal' system dial tone and make calls as if they were using one of the extensions attached to the telephone switch. 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. Obviously, this type of access has serious security implications, and great care must be taken not to compromise your security.

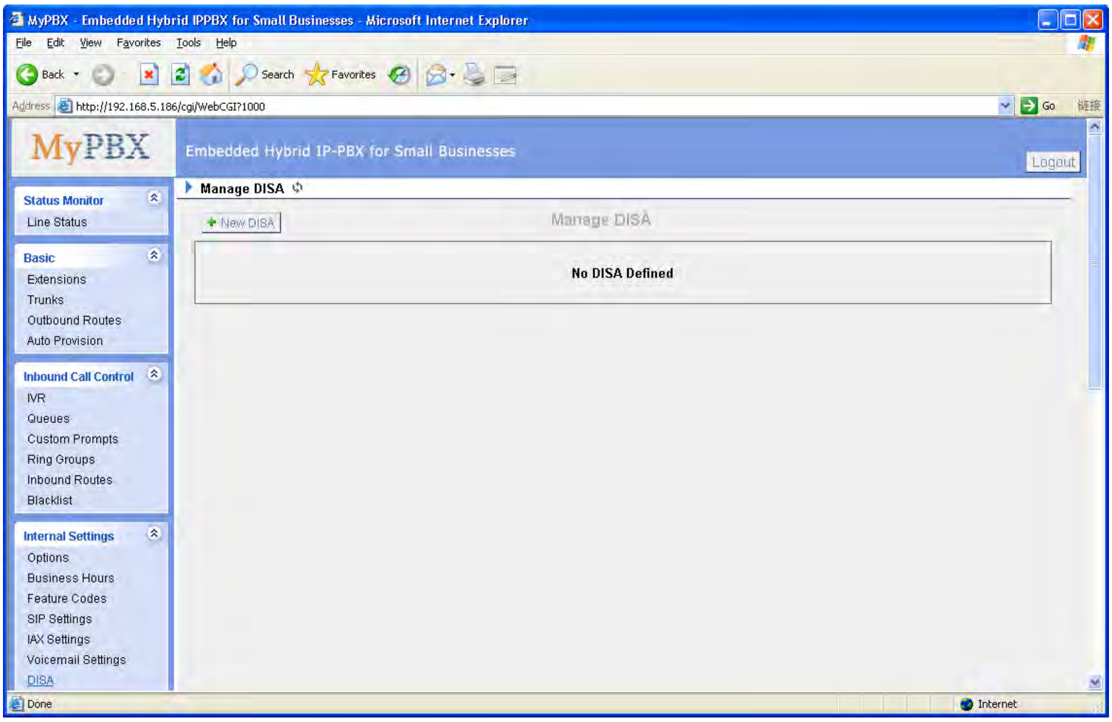


Figure 3.5.8.1

To add a new DISA application, click the New DISA button.

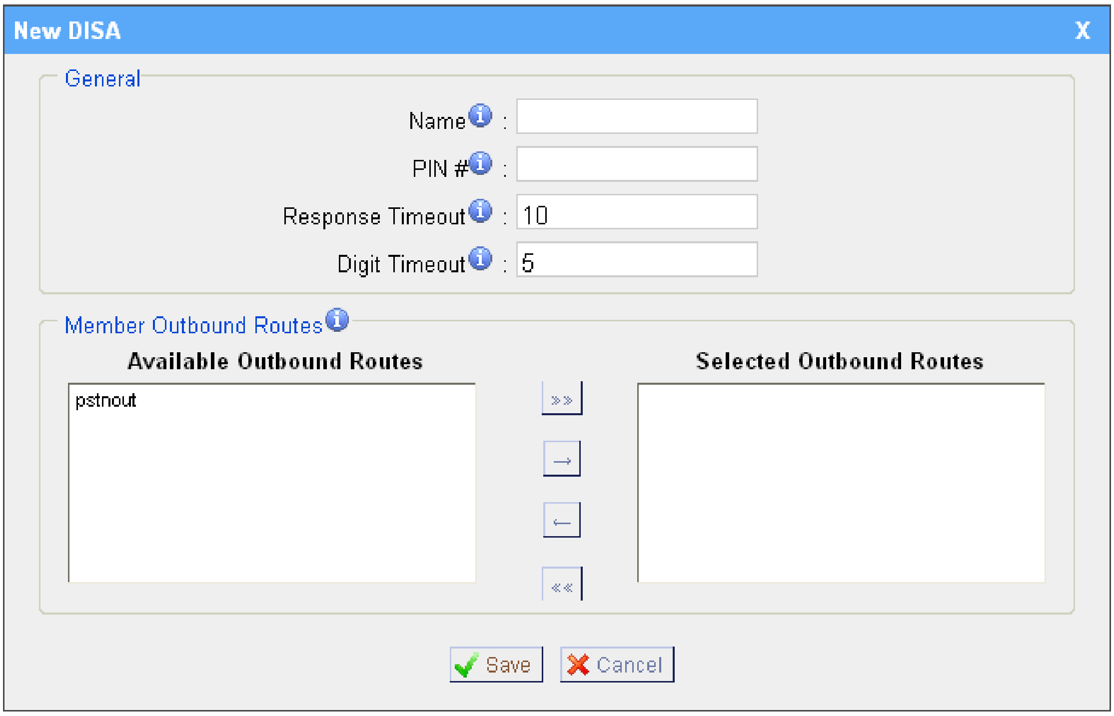


Figure 3.5.8.2

1) General

·DISA Name

Give this DISA application a name to help you identify it.

·PIN #

The password for this DISA .

·Response Timeout

The maximum amount of time the system will wait before hanging up the call if the user has dialed an incomplete or invalid number. Default is 10 seconds.

·Digit Timeout

The maximum amount of time permitted between each digit when the user is dialing an extension number. Default is 5 seconds.

2) Member Outbound Routes

Used to set the outbound routes that can be accessed from this DISA .

3.5.9 Conferencing

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.

·Extension

This is the number dialed to reach this Conference Room.

·Admin

Admin can kick a user out and can lock the conference room.

·Pin #

Set a PIN # that must be entered in order to access this conference room (i.e. 1234).

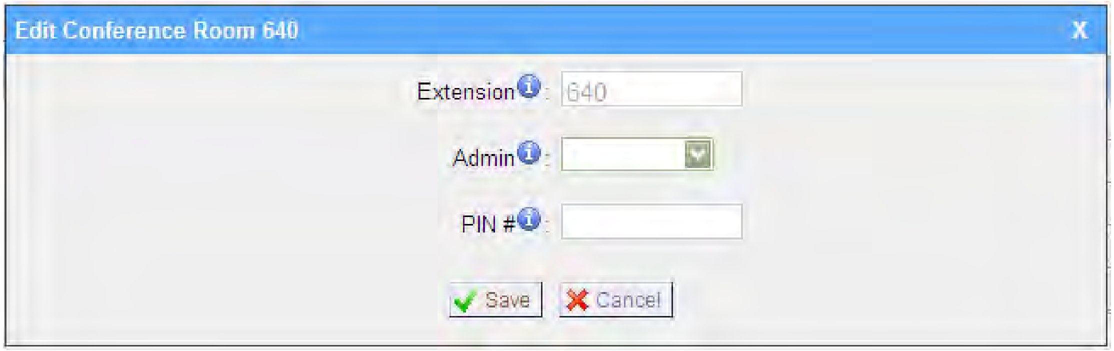


Figure 3.5.9

3.5.10 Paging Groups

Paging is used to make an announcement over the speakerphone to a phone or group of phones. Targeted phones will not ring, but instead answer immediately into speakerphone mode. Please note that this section is for configuring paging groups. If you would like to configure Intercom settings, please open the Other Settings -> Feature Codes screen.

This feature is supported by the following SIP phones:

Yealink's 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.

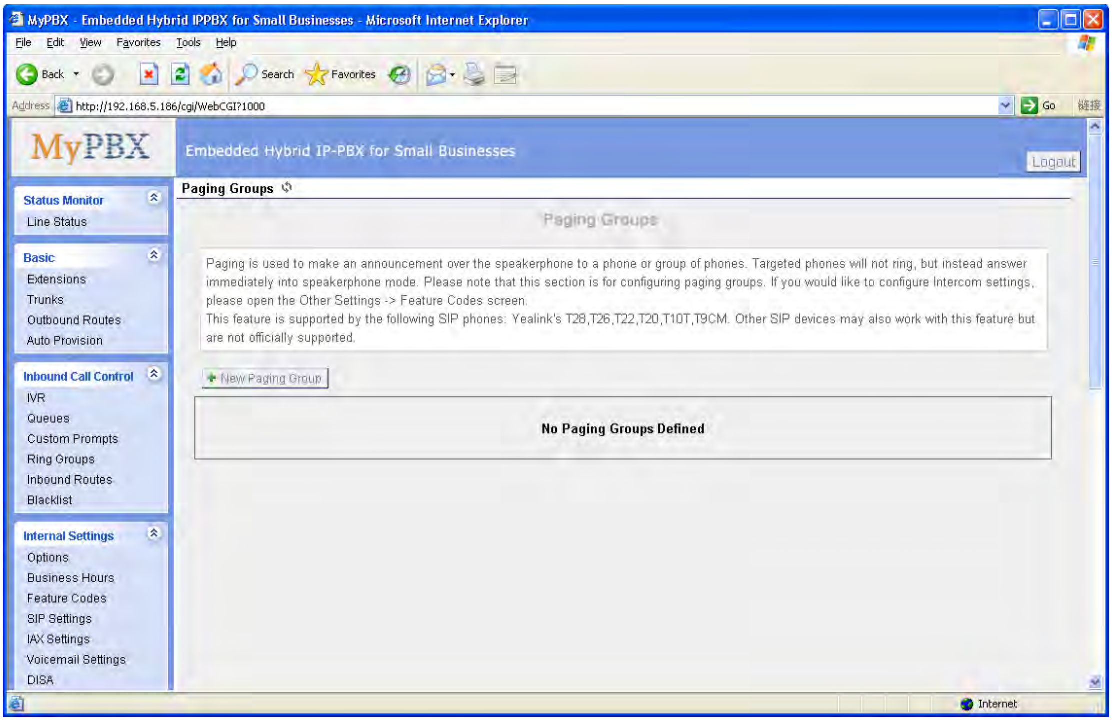


Figure 3.5.10.1

·Paging Group Number

Defines the numbered extension that may be dialed to reach this group.

·Duplex

Paging is typically one way for announcements only. Checking this will make paging duplex, allowing all users in the group to talk and be heard by all.

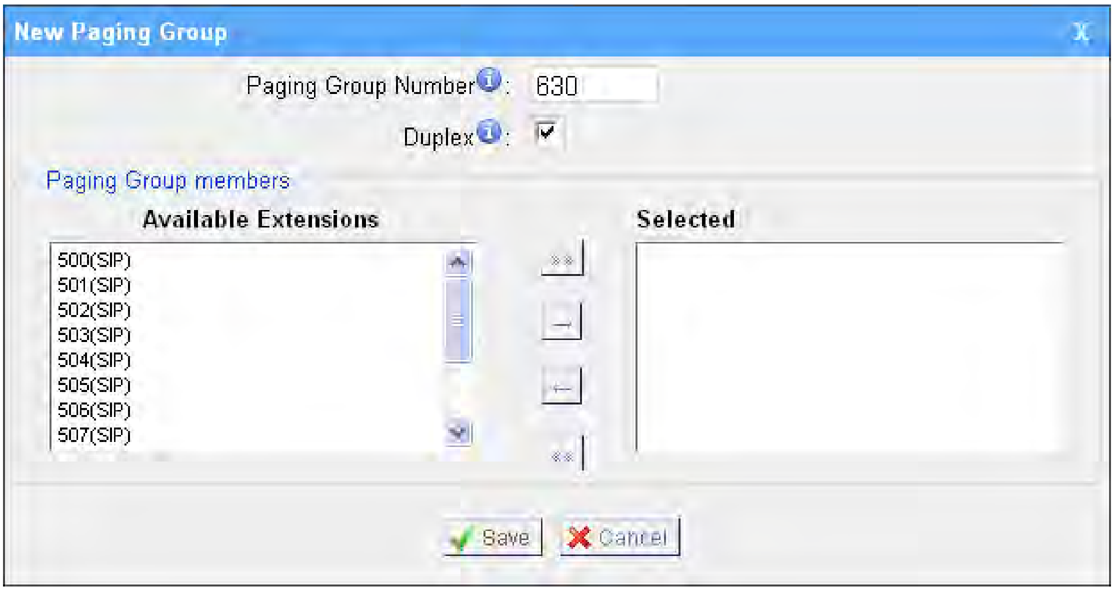


Figure 3.5.10.2

3.5.11 DNIS Settings

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

Note:
1. DID number is not available in PSTN/GSM/UMTS trunks
2. If DID is not configured here, all the calls via this trunk will show the DNIS instead of the original caller ID

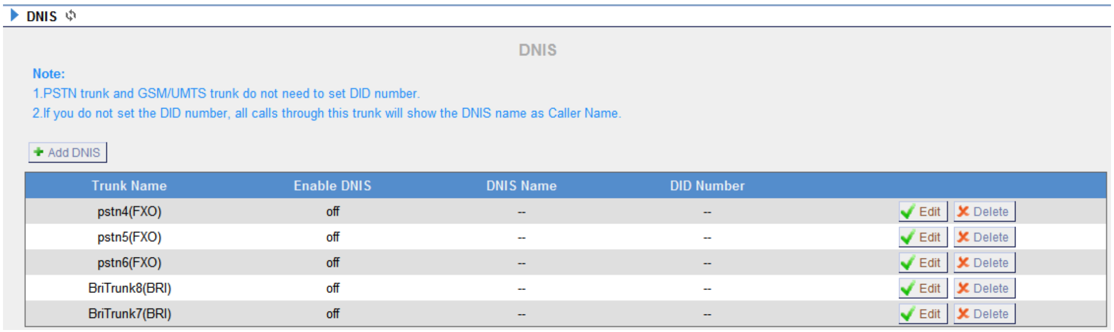


Figure 3.5.11

3.5.12 PIN User Settings

PIN User are used to manage lists of PINs that can be used to access restricted features such as Outbound Routes. If user use PIN User call out, system will auto record the PIN in the call detail records.

1) Options

·Access Code

.Dial this code to access PIN.

·Prompt for Entry

Prompt caller enter the PIN Number.

·Prompt for Entry Failure

Prompt the caller when an invalid PIN is entered.

2) PIN User

MyPBX can store a number of PIN Users. PIN Users may be used to keep track of calls in relation to particular activities or clients. They can also be used to keep track of calls by particular users or sets of users.

• PIN entered are checked against those stored by the system. If an invalid PIN is entered, the PIN is requested again.

• The system administrator can configure certain numbers or types of numbers to require entry of a PIN before you can continue making a call to such a number.

• The system administrator can also configure you to have to enter a PIN before making any external call.

·Name

A character-based name for this PIN list, i.e. 'YeastarPIN'

·PIN List

Enter a list of one or more PINs, One PIN per line.

·Outbound Route

PIN User can use those outbound route to make call out.

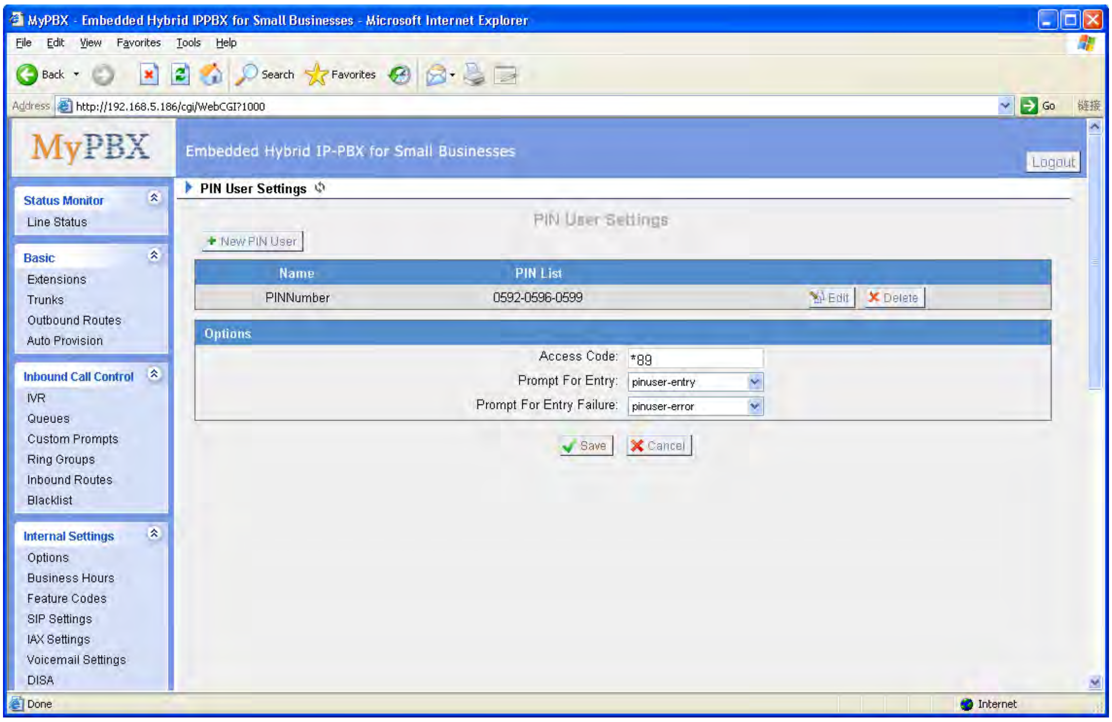


Figure 3.5.12

3.5.13 Callback Settings

MyPBX allows caller A to dial an inbound route number, and after hearing the ring, A can hang up the call or wait for MyPBX to cut off the call, then MyPBX will call A with this number. When A pick up the call, A can dial the number he wants to call; MyPBX will call the number with its outbound route.

Note:

1. If you'd like to use callback feature, please make sure if 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.

·Allow All Numbers

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

Follow the step to use this function.

Step 1: Enable Callback.
Inbound Routes – Choose "Yes" on" Enable Callback" to enable this function. Refer to **chapter 3.4.5.1**

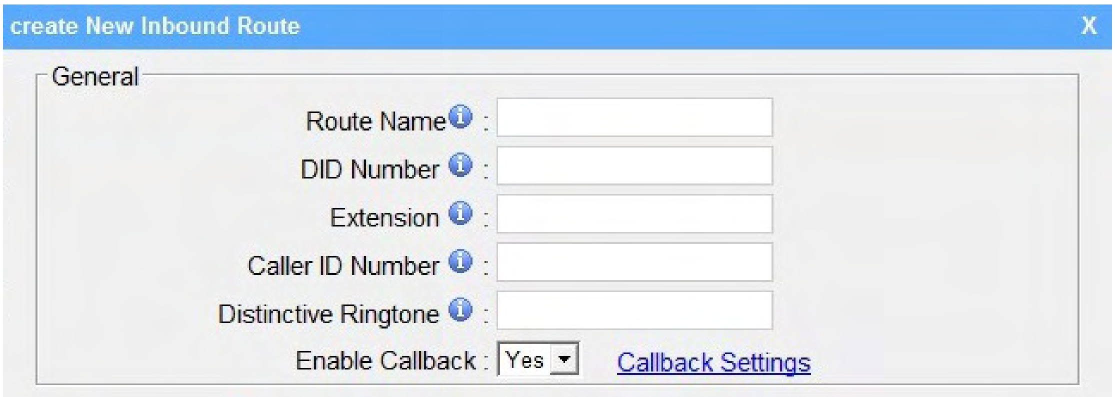


Figure 3.5.13.1

Step 2: Create Callback number

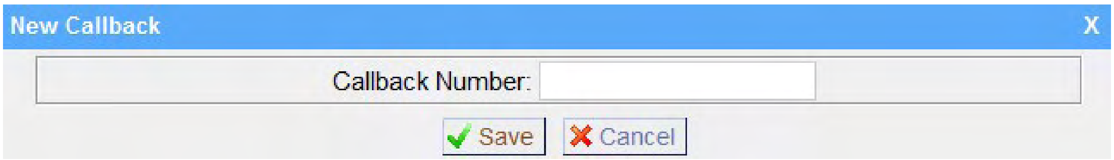


Figure 3.5.13.2

Step 3: Create Callback Rules
You will need to create callback rules when the system should strip or add digits.

·Trunk Name
Choose the trunk with callback rules

·Strip digits from front
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 before dialing
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 call to 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.

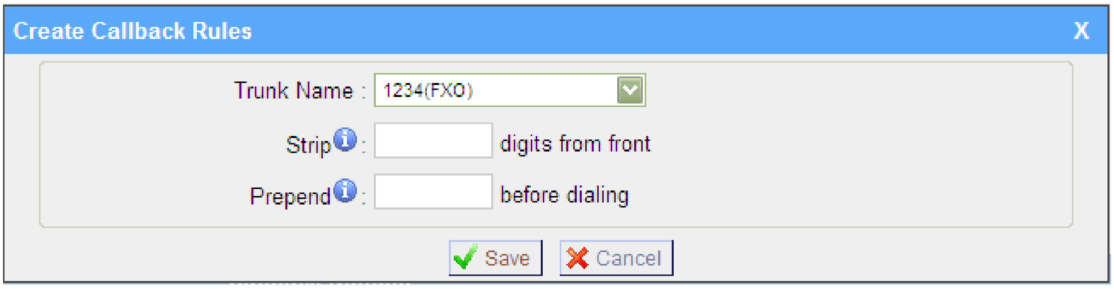


Figure 3.5.13.3

3.5.14 Speed Dial Settings

1) Options

·The prefix of speed dial

The prefix should be dialed before the speed dial number. Default is *99.

2) Add new speed dial.

·Source Number

The speed dial number.

·Destination Number

The number you want to call.

e.g. The source number is ”123”. The destination number is 5503305.The prefix number is *99.You can use an extension with any type to dial *99123, then it will call to number 5503305.

Note: Don’t forget to add the outbound dial prefix if you would like to dial the speed dial number through trunk.

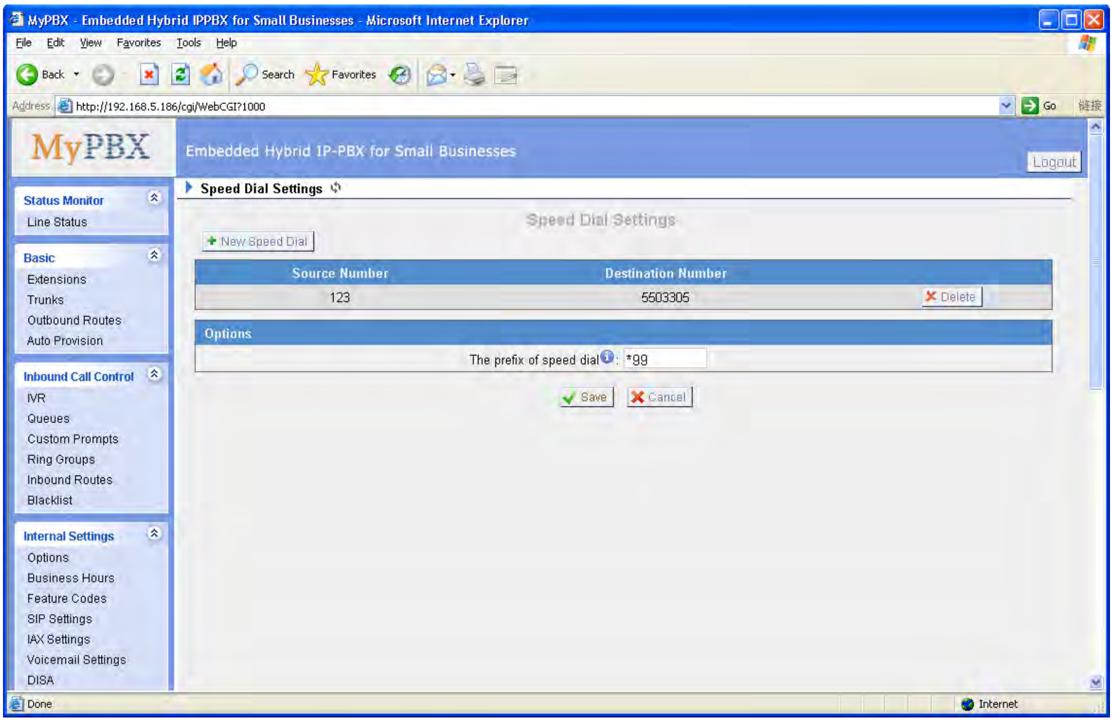


Figure 3.5.14

3.5.15 Music on Hold Prompts

The administrator can upload on hold music as follows:

1) Click 'Upload Music on Prompt '

2) Click 'Browse' to choose the desired audio file.

3) Click 'Upload' to upload the selected file.

Note: The sound file format should be as follows: GSM 6.10, 8.000kHz, Mono, 1kb/sec, The file size must not be greater than 1.8MB.

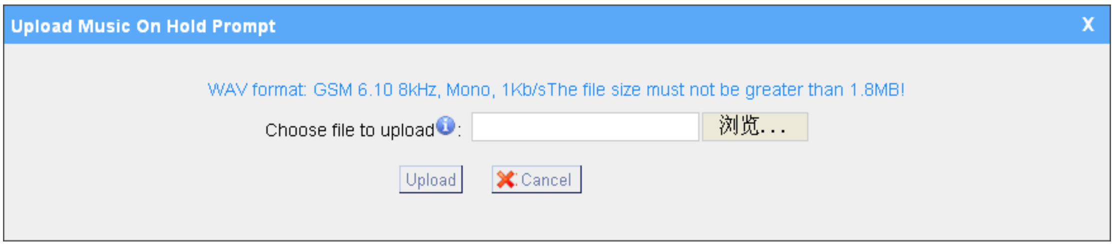


Figure 3.5.15

3.6 Network Settings

3.6.1 LAN Settings

·DHCP

If this option is set, MyPBX will use DHCP to get an available IP address from your local network. Not recommended.

·Enable SSH

This is the advance way to access the device, you can use the putty software to access the device. In the SSH access, you can do more advance setting and debug.

·Port: the default is 8022,

·Hostname

Set the host name for MyPBX.

·IP Address

Set the IP Address for MyPBX.

·Subnet Mask

Set the subnet mask for MyPBX.

·Gateway

Set the gateway for MyPBX.

·Primary DNS

Set the primary DNS for MyPBX.

·Secondary DNS

Set the secondary DNS for MyPBX.

·IP Address2

Set the second IP Address for MyPBX.

·Subnet Mask2

Set the second subnet mask for MyPBX.

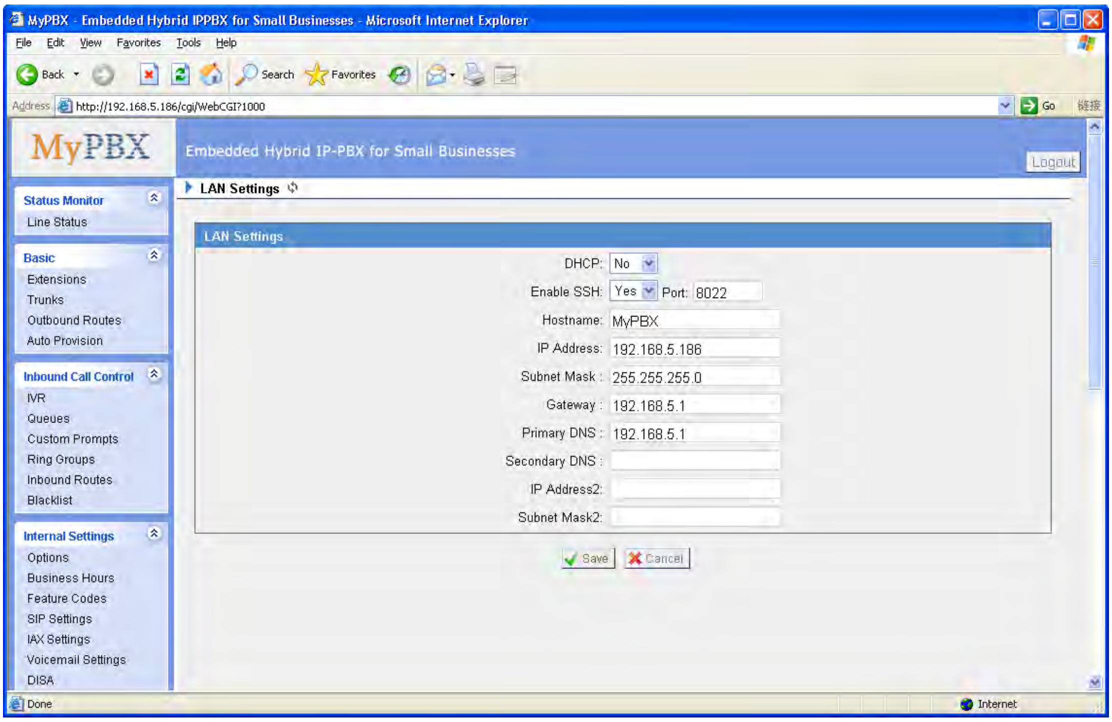


Figure 3.6.1

3.6.2 WAN Settings

Note: Only have WAN port's hardware version support this functions.

It support three connection types: DHCP (obtain an IP automatically), PPPoE, Static IP Address.

·DHCP

.If your ISP says that you are connecting through DHCP or a dynamic IP address from your ISP, perform these steps:

Step1: Select **DHCP** as the WAN Connection Type.

Step2: Click **Save** button to save the settings.

Step3: Reboot the device.

Step4: Check the WAN's Status (System Info → WAN → Status).

·Static IP Address

If your ISP says that you are connecting through a static or fixed IP address from your ISP, perform these steps:

Step1: Select **Static IP Address** as the WAN Connection Type.

Step2: Enter the IP Address.

Step3: Enter the Subnet Mask.

Step4: Enter the Gateway Address.

Step5: Enter the Primary DNS and Secondary DNS.

Step6: Click the **Save** button to save the settings.

Step7: Reboot the device.
Step8: Check the WAN's Status (System Info → WAN → Status).

·PPPoE

If your DSL provider says that you are connecting through PPPoE or if you normally enter a user name and password to access the Internet, perform these steps:

Step1: Select **PPPoE** as the WAN Connection Type.
Step2: Enter the User Name.
Step3: Enter the Password.
Step4: Click the **Save** button to save the settings.
Step5: Reboot the device.
Step6: Check the WAN's Status (System Info → WAN → Status).

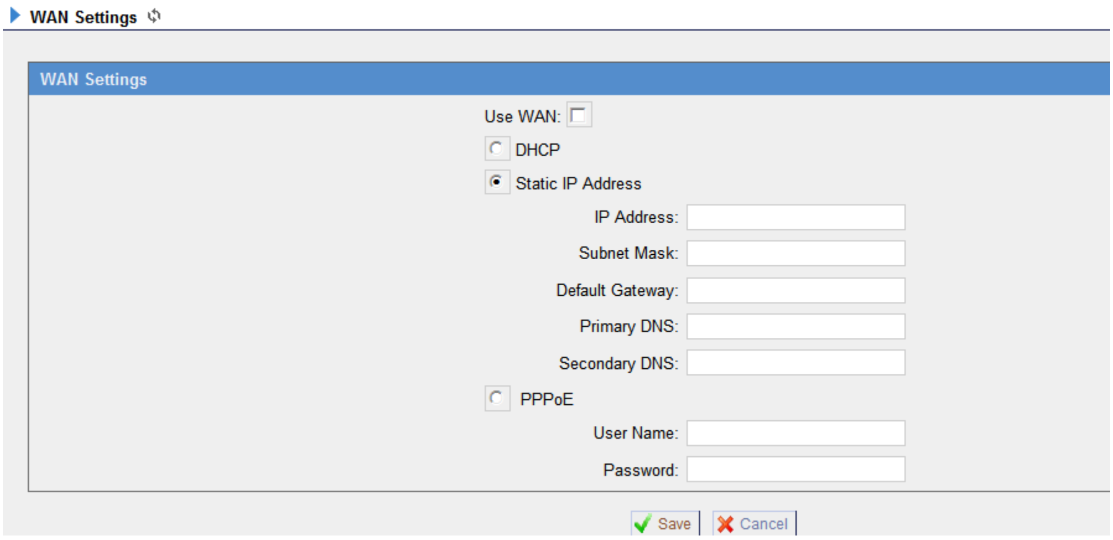


Figure 3.6.2

3.6.3 Static Route

MyPBX will have more than one internet connection in some situations but it has only one default gateway. You will need to set some Static Route for MyPBX to force it goes out through different gateway when access to different internet. The default gateway priority of MyPBX from high to low is OpenVPN→WAN port→LAN port.

1) Route table

The current route rules of MyPBX

·Destination

The destination network to be accessed to by MyPBX

·Subnet Mask

Specify the destination network portion.

·Gateway

Which gateway MyPBX will go through when access to the destination network.

·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

Define which internet port to go through.

2) Static Route Rules

You can add new static route rules here.

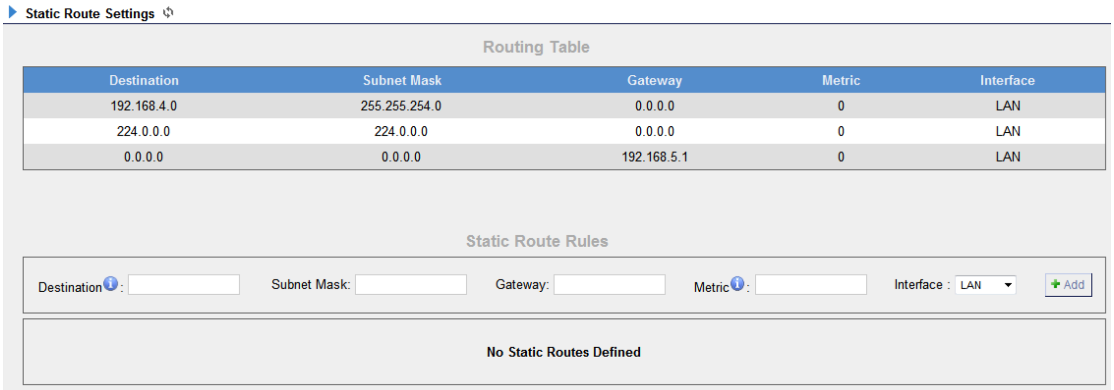


Figure 3.6.3

3.6.4 Firewall

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.

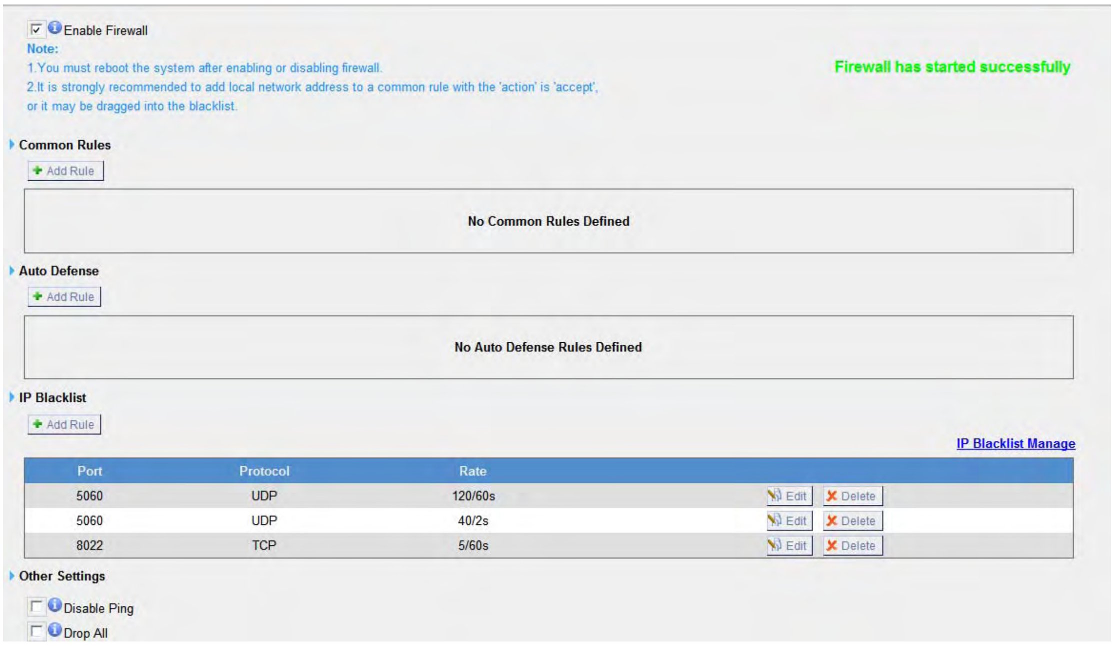


Figure 3.6.4

1) Enable Firewall

Enable the firewall to protect the device. You should reboot the device to let the firewall run successfully.

2) Common Rules

·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
Ex: 192.168.5.100/255.255.255.255 for IP 192.168.5.100
Ex: 216.207.245.47/255.255.255.255 for IP 216.207.245.47
Ex: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.

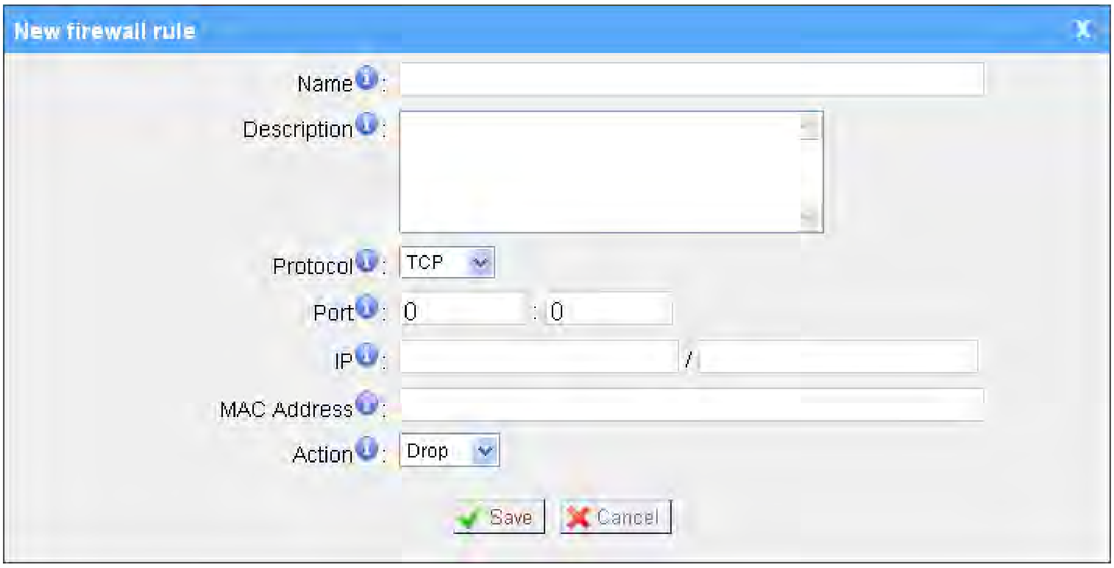


Figure 3.6.4.1

3) Auto Defense

·Port

Auto defense port, e.g.: 8022.

·Protocol

Auto defense protocol , TCP or UDP.

·Rate

The maximum packets or connections can be handled per unit time.

E.g.: (Port: 8022 Protocol: TCP Rate: 10/minute) means maximum 10 TCP connection to port 8022 can be handled per minute, the eleventh connection will be refused directly.

Figure 3.6.4.2

4) IP Blacklist

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

a) New Rule

·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.

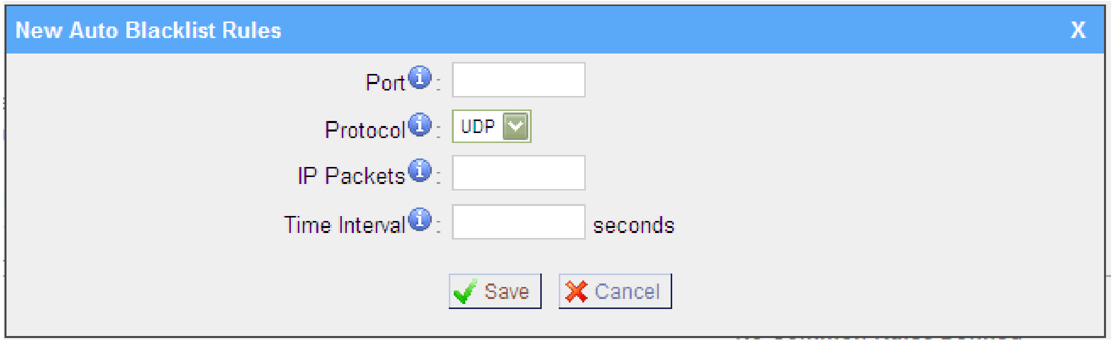
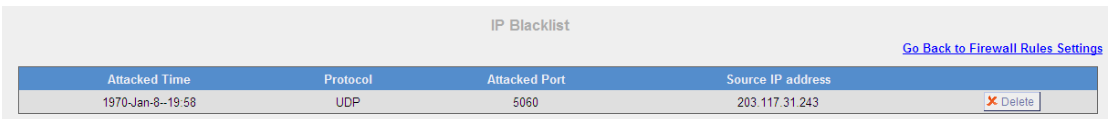


Figure 3.6.4.3

b) IP Blacklist Manage

You can manage the IP addresses which are blocked automatically here.


IP Blacklist

Go Back to Firewall Rules Settings

Attacked Time	Protocol	Attacked Port	Source IP address	
1970-Jan-8--19:58	UDP	5060	203.117.31.243	Delete

Figure 3.6.4.4

5) Other Settings

·Disable Ping

Enable this item, net ping from remote hosts will be dropped.

·Drop All

When you enable 'Drop All' feature, 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.

3.6.5 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.

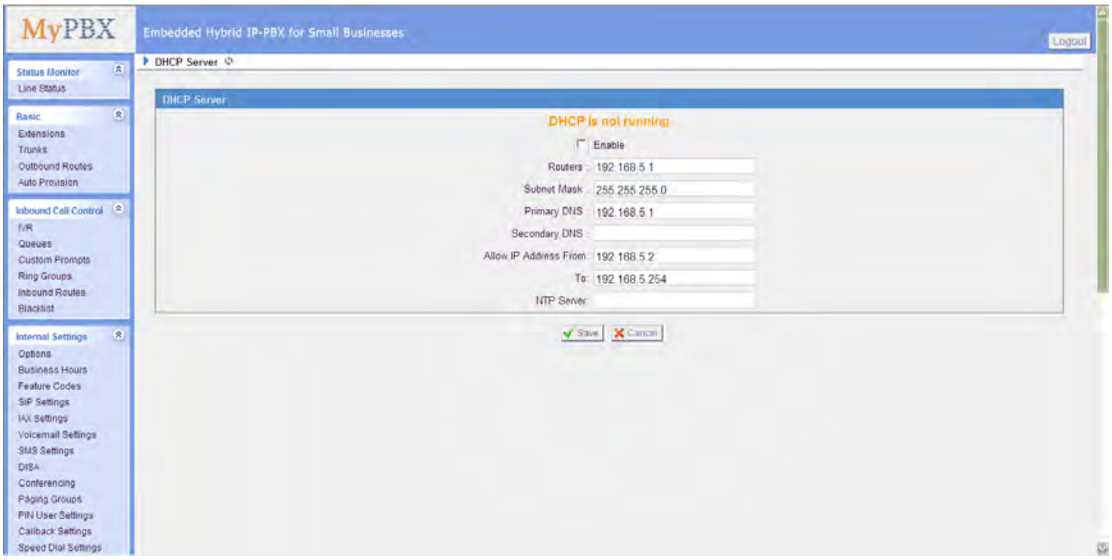


Figure 3.6.5

3.6.6 VLAN Settings

A VLAN(Virtual LAN) is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.

1) VLAN Over Lan

·NO.1

Click the NO.1 you can edit the first VLAN over Lan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for MyPBX VLAN over Lan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Lan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Lan

·NO.2

Click the NO.2 you can edit the first VLAN over Lan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for MyPBX VLAN over Lan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Lan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Lan.

2) VLAN Over Wan

·NO.1

Click the NO.1 you can edit the first VLAN over Wan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for MyPBX VLAN over Wan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Wan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Wan.

·NO.2

Click the NO.2 you can edit the first VLAN over Wan.

·VLAN Number

.The VLAN Number is a unique value you assign to each VLAN on a single device.

·VLAN IP Address

Set the IP Address for MyPBX VLAN over Wan.

·VLAN Subnet Mask

Set the Subnet Mask for MyPBX VLAN over Wan.

·Default Gateway

Set the Default Gateway for MyPBX VLAN over Wan.

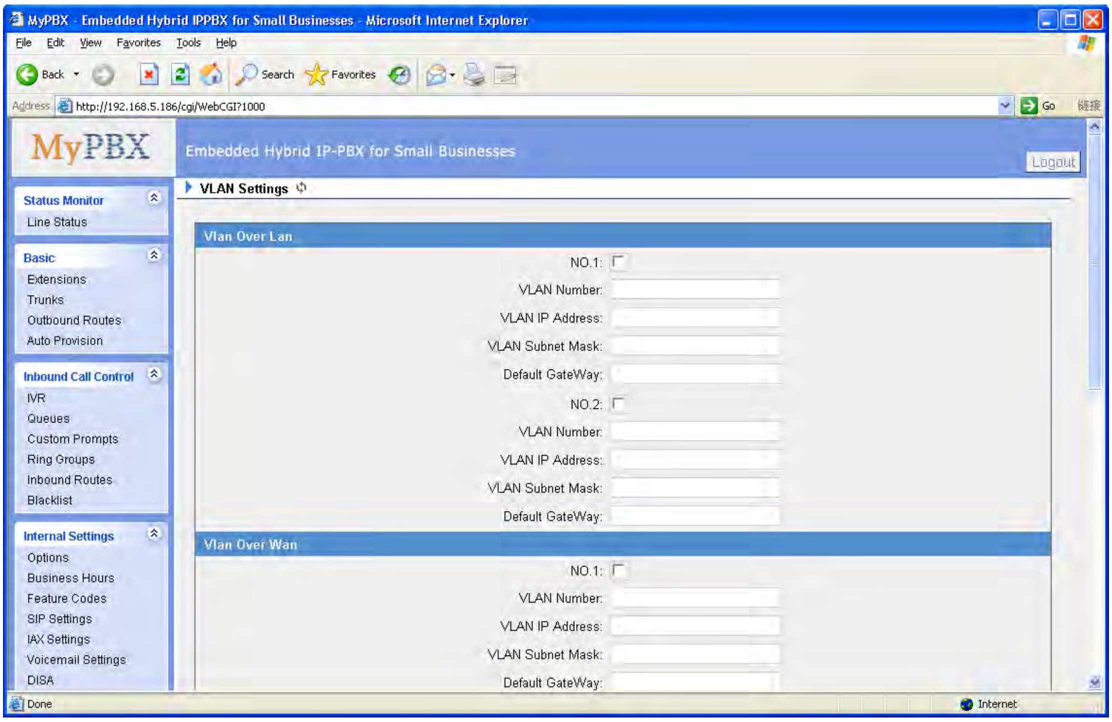


Figure 3.6.6

3.6.7 VPN Settings

A virtual private network (VPN) is a method of computer networking--typically using the public internet--that allows users to privately share information between remote locations, or between a remote location and a business' home network. A VPN can provide secure information transport by authenticating users, and encrypting data to prevent unauthorized persons from reading the information transmitted. The VPN can be used to send any kind of network traffic securely. MyPBX supports OpenVPN.

·Enable VPN

·Import VPN Config

Import configuration file of OpenVPN. Don't configure 'user' and 'group' in the 'config' file.

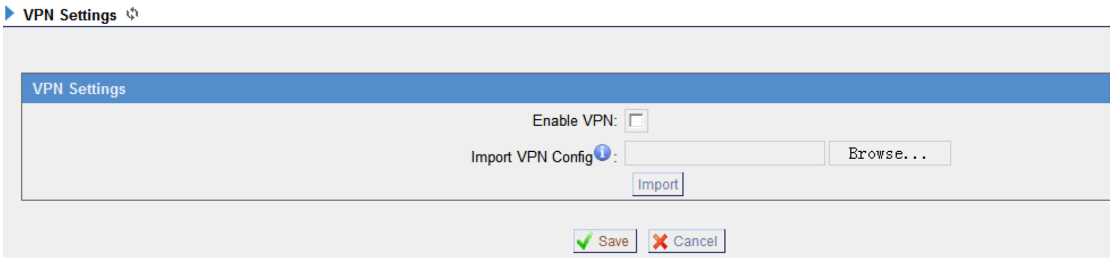


Figure 3.6.7

3.6.8 DDNS Settings

DDNS(Dynamic DNS) is a method / protocol / network service that provides the capability for a networked device, such as a router or computer system using the Internet Protocol Suite, to notify a Domain Name System (DNS) name server to change, in real time, the active DNS configuration of its configured hostnames, addresses or other information.

·Enable DDNS

·DDNS Server

Select the DDNS server you sign up for service.

·User Name

User name the DDNS server provides you.

·Password

User account's password .

·Host Name

Note: DDNS allows you to access your network using domain names instead of IP address. The service manages changing IP address and updates your domain information dynamically. You must sign up for service through dyndns.org, freedns.afraid.org, www.no-ip.com, www.zoneedit.com

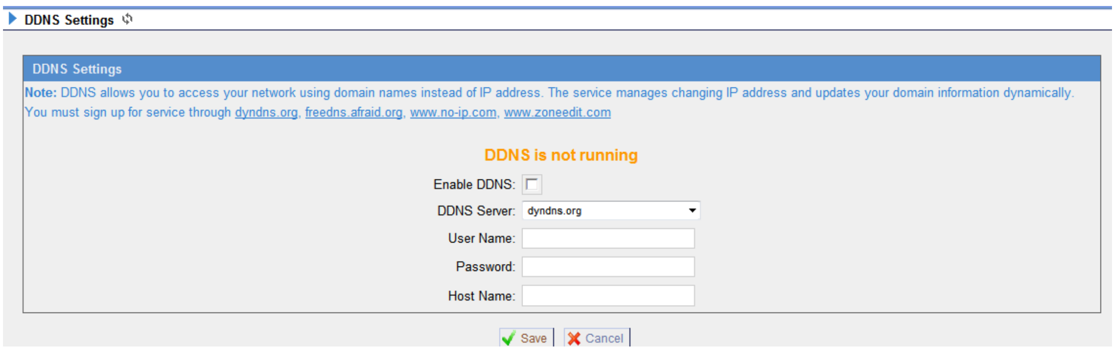


Figure 3.6.8

3.7 System Settings

3.7.1 External Storage

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

Auto-Backup extends the allocated disk space for backing up critical files. When properly configured, MyPBX will move all qualified files to a Windows PC every 30 minutes. For the voicemail files and recoding files, they must be created before Auto-Backup has been configured.

How to configure External Storage, please see APPENDIX B How to Configure Autobackup.

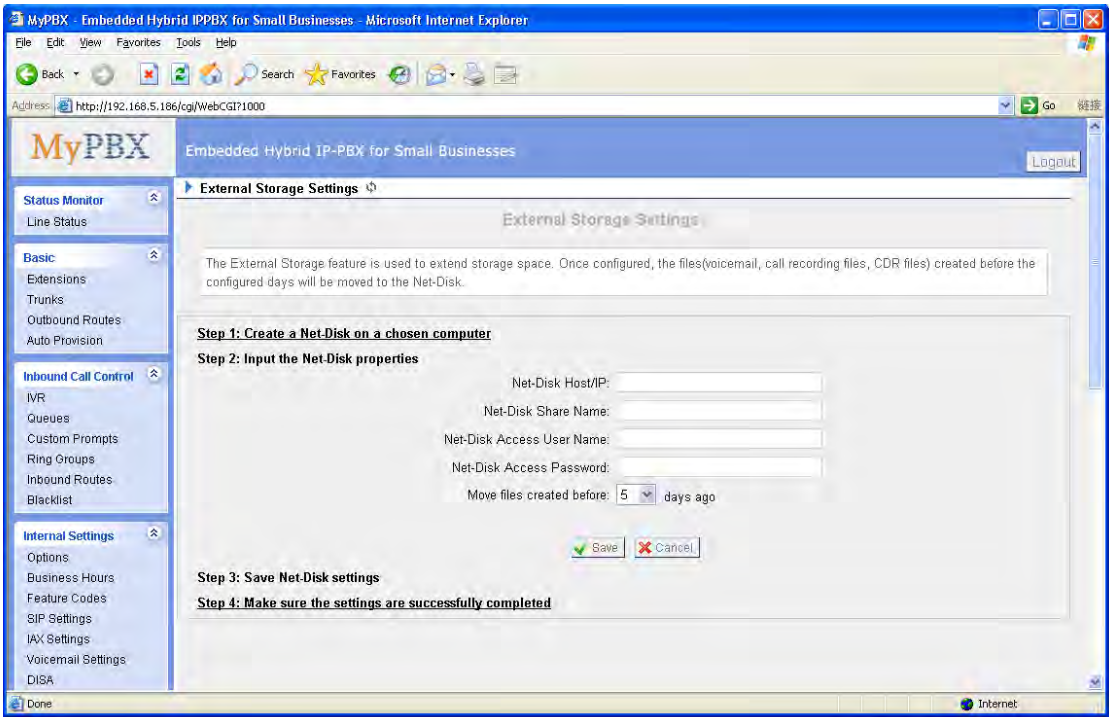


Figure 3.7.1

3.7.2 Password Settings

The default password is '**password**'. To change the password, enter the new password and click update. The system will then prompt you re-login using your new password.

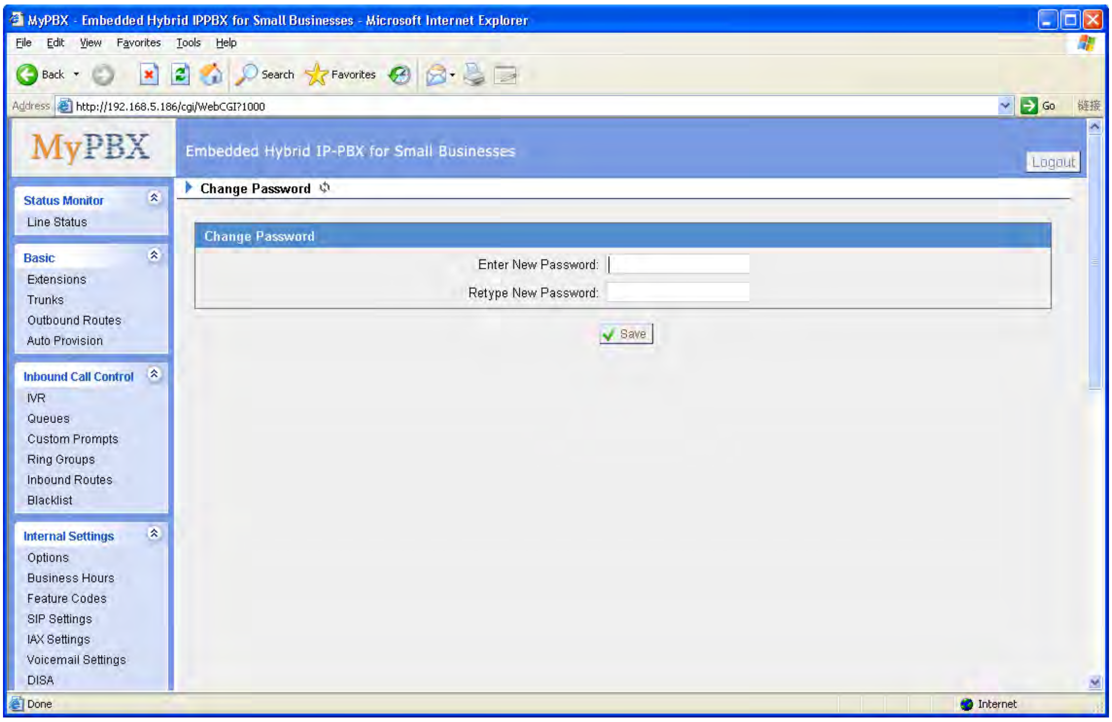


Figure 3.7.2

3.7.3 System Prompts

MyPBX have prompts of many languages. You can download the appropriate language you need. MyPBX can support American English, Australian English, Chinese, Dutch, French, Canadian French, German, Greek, Hungarian, Italian, Polish, Portuguese, Brazilian Portuguese, Russian, Spanish, Mexican Spanish, Turkish, Thai, Korean currently.

Note:
Auto-detection is highly recommended. But if you prefer to download via HTTP or TFTP server, please contact the local dealer for the prompts.

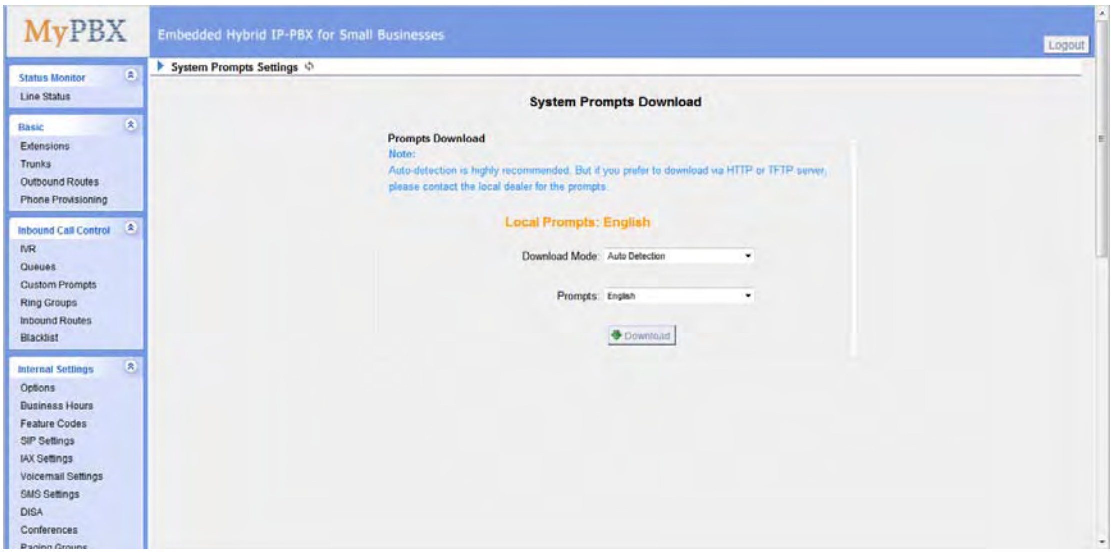


Figure 3.7.3

3.7.4 Date and Time

Set the date and time for MyPBX.

·Time Zone

You can choose your time zone here.

·Daylight Saving Time

Set the mode to Automatic or disabled

·Automatically Synchronize With an Internet Time Server

Input the NTP server so that MyPBX will update the time automatically

·Set Date & Time Manually

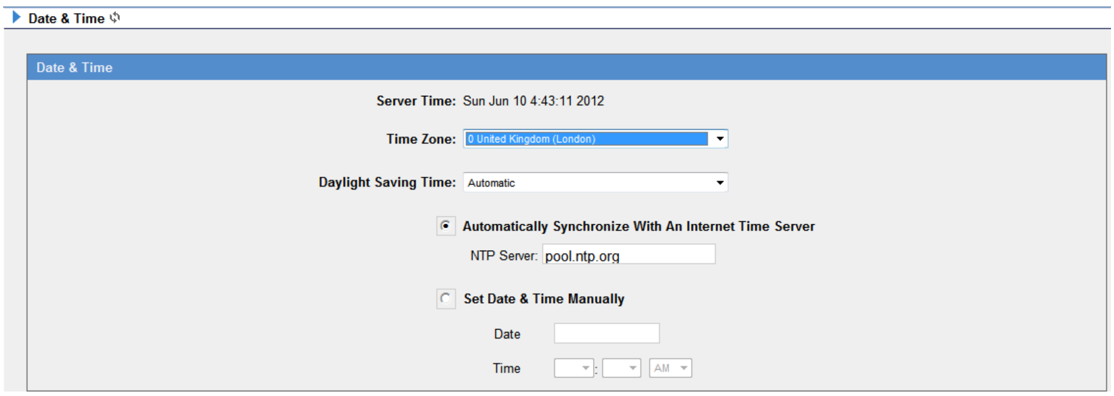


Figure 3.7.4

3.7.5 Backup and Restore

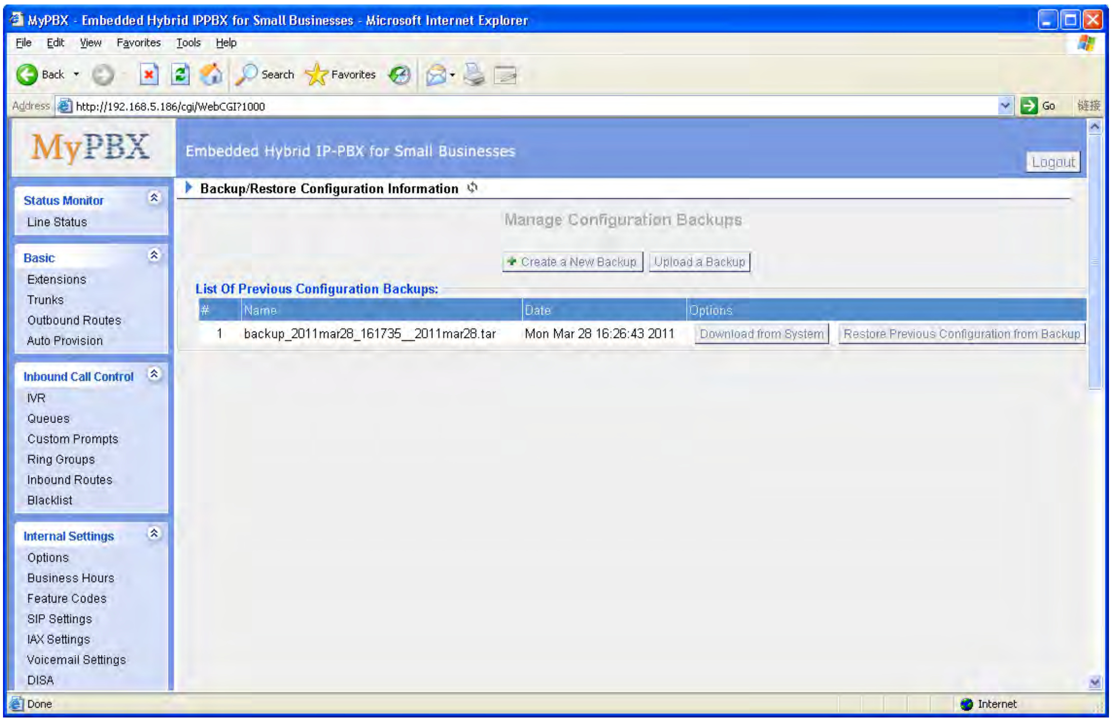


Figure 3.7.5

3.7.6 Reset and Reboot

·Reboot System

Warning: Rebooting the system will terminate all active calls!

·Reset to Factory Defaults

Warning: A factory reset will erase all configuration data on the system. Please do not turn off the system until the RUN light begins blinking. Any power interruption during this time could cause damage to the system.

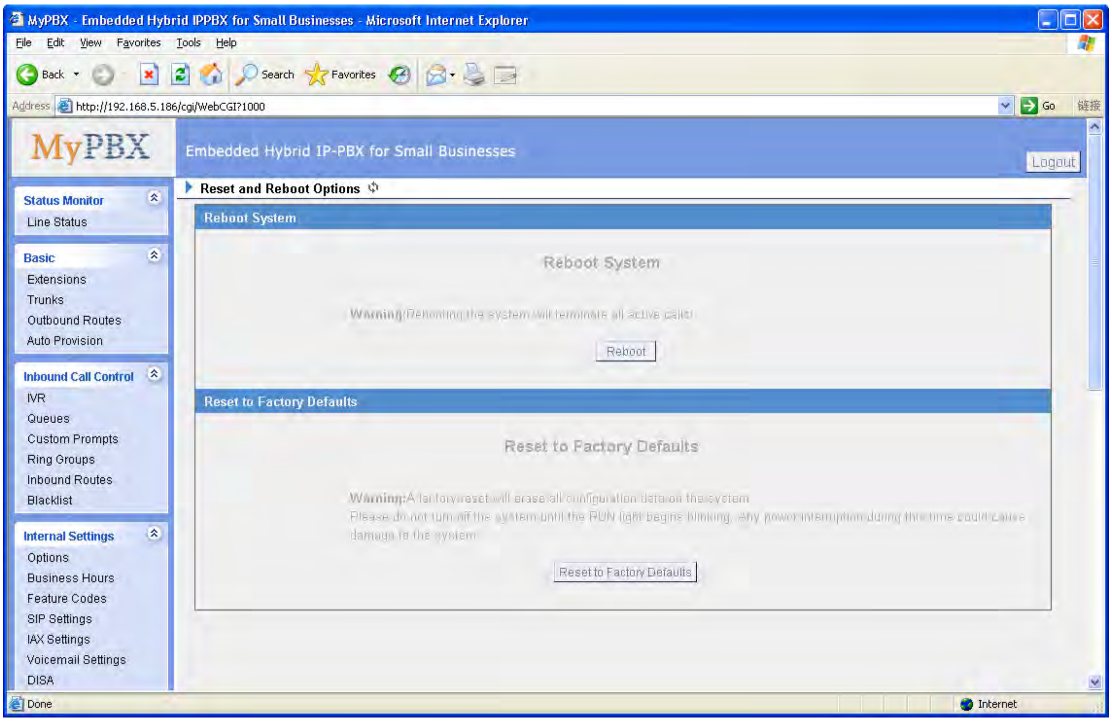


Figure 3.7.6

3.7.7 Firmware Update

Upgrading of the firmware is possible through the Administrator web interface using a TFTP Server or an HTTP URL.

Enter your TFTP Server IP address and firmware file location, then click start to update the firmware.

More Information, please see http://www.yeastar.com/download/MyPBX/

Note:

1. If enabled 'Reset configuration to Factory Defaults', System will restore to factory default settings.
2. When update the firmware, please don't turn off the power.

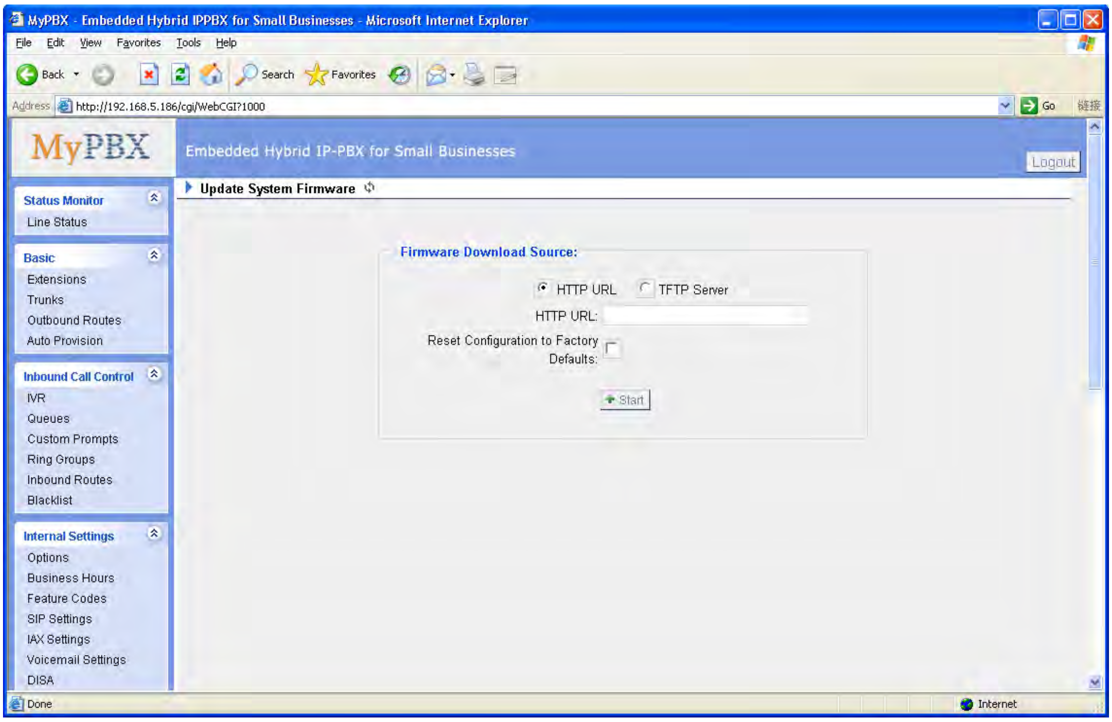


Figure 3.7.7

3.8 Reports

3.8.1 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.

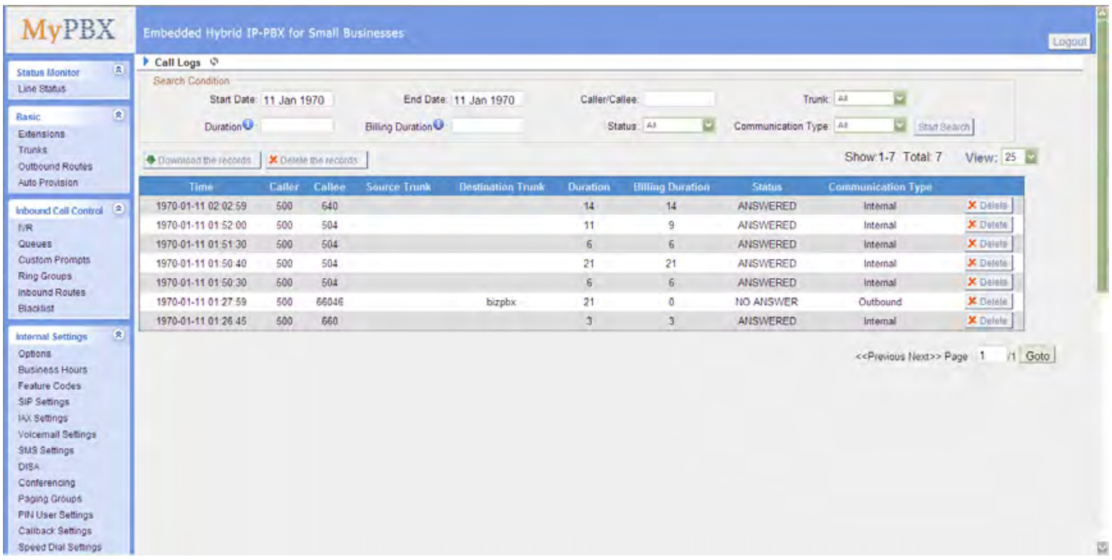


Figure 3.8.1

3.8.2 System Logs

You can download and delete the system logs of MyPBX.

Options

·Enable Hardware Log

Save the infomation of hardware; (up to 4 log files)

·Enable Normal Log

Save the prompt information; (up to 16 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)

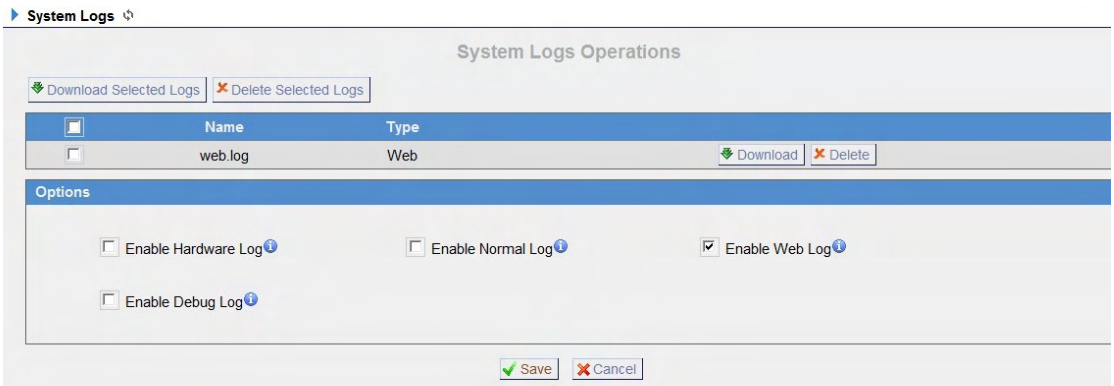


Figure 3.8.2

3.8.3 System Info

General:

Information about hardware version , firmware version and system uptime.

LAN:

Information about hostname, MAC address, IP address, gateway, Primary DNS and Secondary DNS.

Disk Usage:

Disk usage information.

Memory Usage:

Memory usage information .

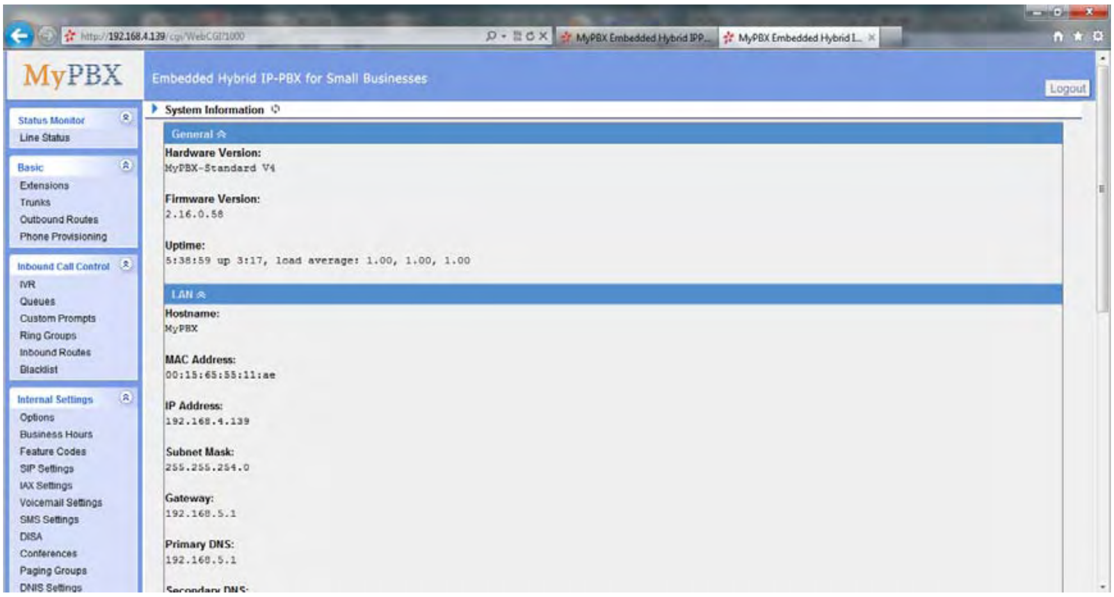

MyPBX
Embedded Hybrid IP-PBX for Small Businesses
Logout
Status Monitor
Line Status
Basic
Extensions
Trunks
Outbound Routes
Phone Provisioning
Inbound Call Control
IVR
Queues
Custom Prompts
Ring Groups
Inbound Routes
Blacklist
Internal Settings
Options
Business Hours
Feature Codes
SIP Settings
IAX Settings
Voicemail Settings
SMS Settings
DISA
Conferences
Paging Groups
DNIS Settings
System Information
General
Hardware Version:
MyPBX-Standard V4
Firmware Version:
2.16.0.58
Uptime:
5:38:59 up 3:17, load average: 1.00, 1.00, 1.00
LAN
Hostname:
MyPBX
MAC Address:
00:15:65:55:11:ae
IP Address:
192.168.4.139
Subnet Mask:
255.255.254.0
Gateway:
192.168.5.1
Primary DNS:
192.168.5.1

Figure 3.8.3

4. Access MRI

MRI (MyPBX Recording Interface).Users may access MRI by logging into the MRI web interface with their username (extension number) and voicemail password.

4.1 Allow users to access MRI

The extension's 'User Web Interface' option must be checked before the associated user can log into MRI.

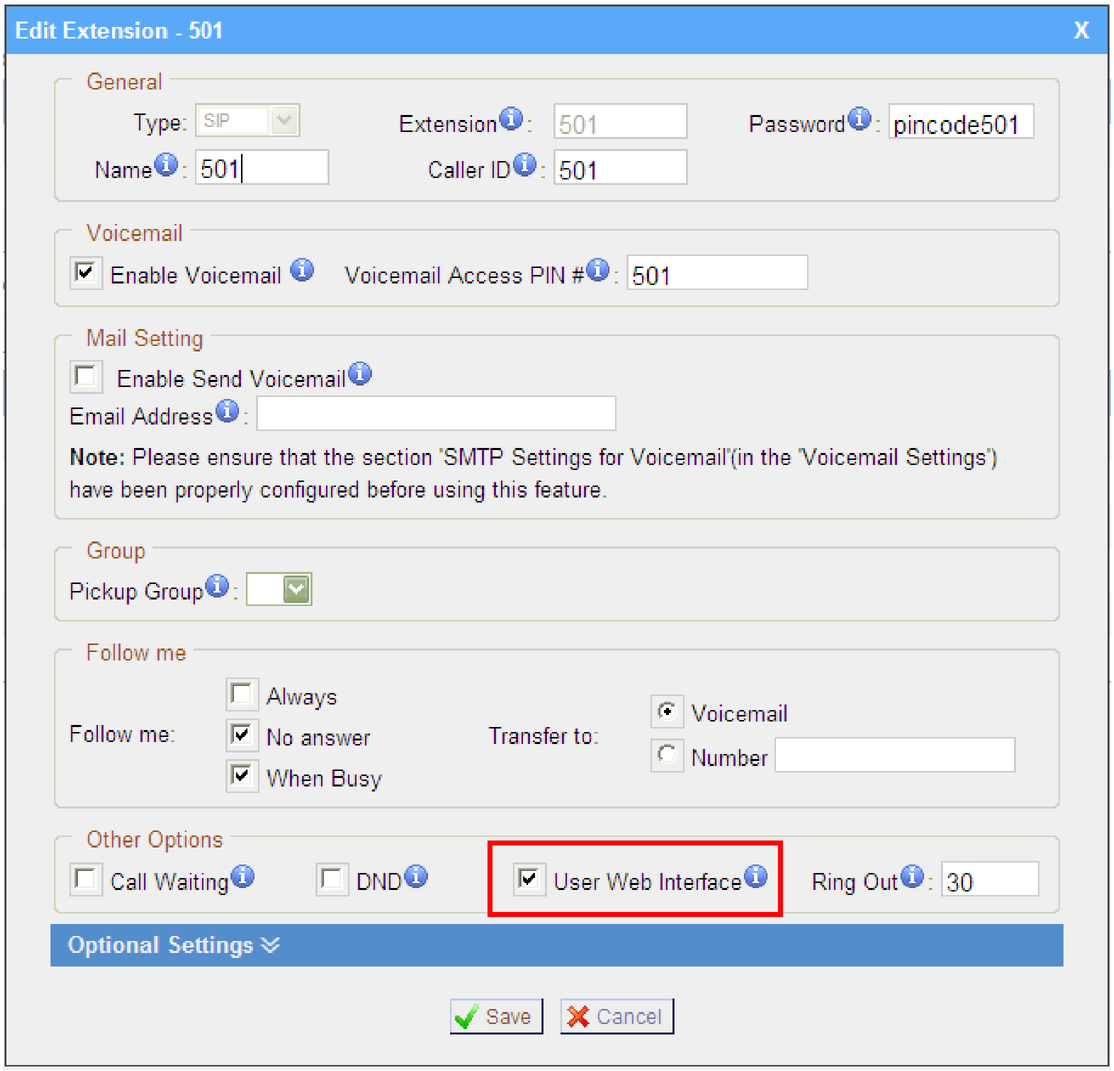


Figure 4-1

4.2 User login

Users can access the MRI web interface by navigating to the MyPBX IP address using a web browser. If you are unsure of this address, please contact your network administrator.

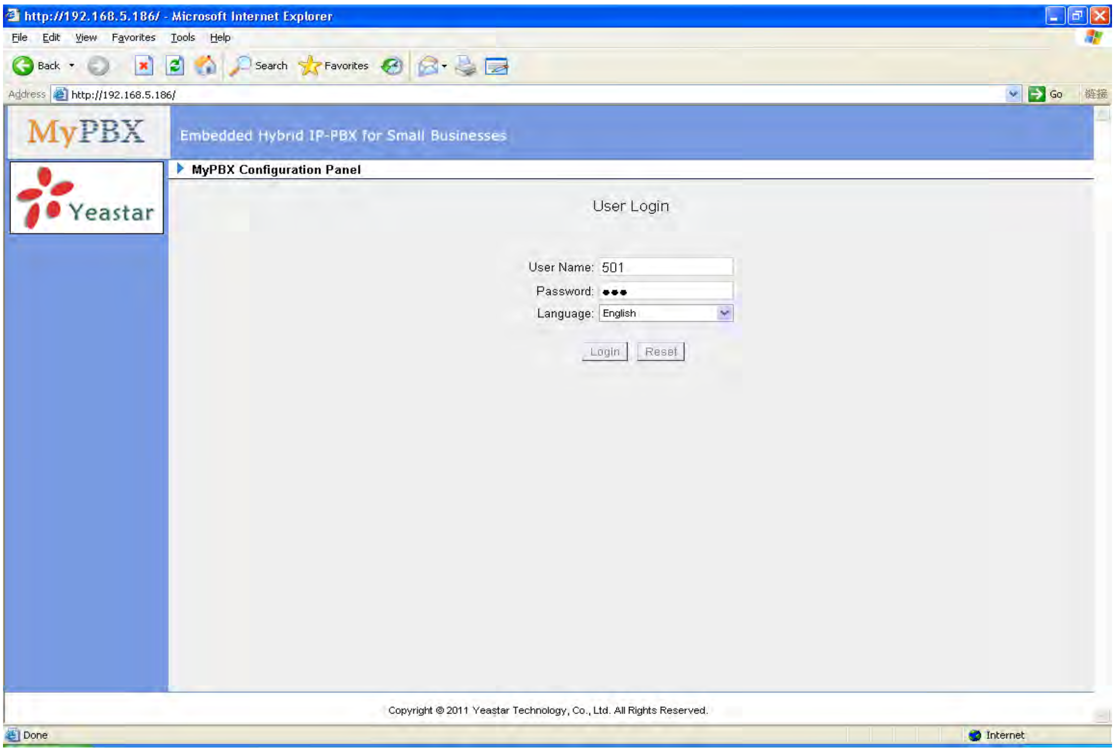


Figure 4-2

4.3 Voicemail

Users can check, delete, move and download voicemail files here.

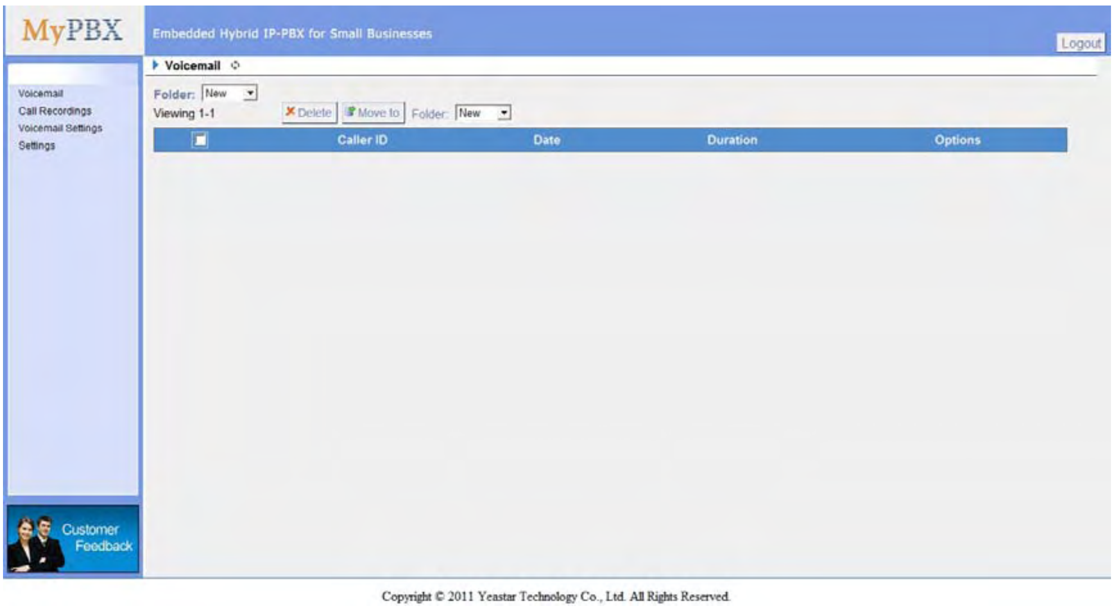


Figure 4-3

4.4 Record

Users can play, delete and download recorded files here.

Figure 4-4

4.5 Voicemail Settings

·**Voicemail password**: new voicemail box password.

·**Enter again to confirm**: confirm new voicemail box password.

·**Email Address**: Email address use to receive the voicemail or Fax.

Note: Please ensure that the section 'SMTP Settings For Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.

·**Enable Voicemail**

Check this box if the user should have a voicemail account.

·**Enable Send Voicemail**

Once enabled, the voicemail or Faxex will be sent to email as an attachment.

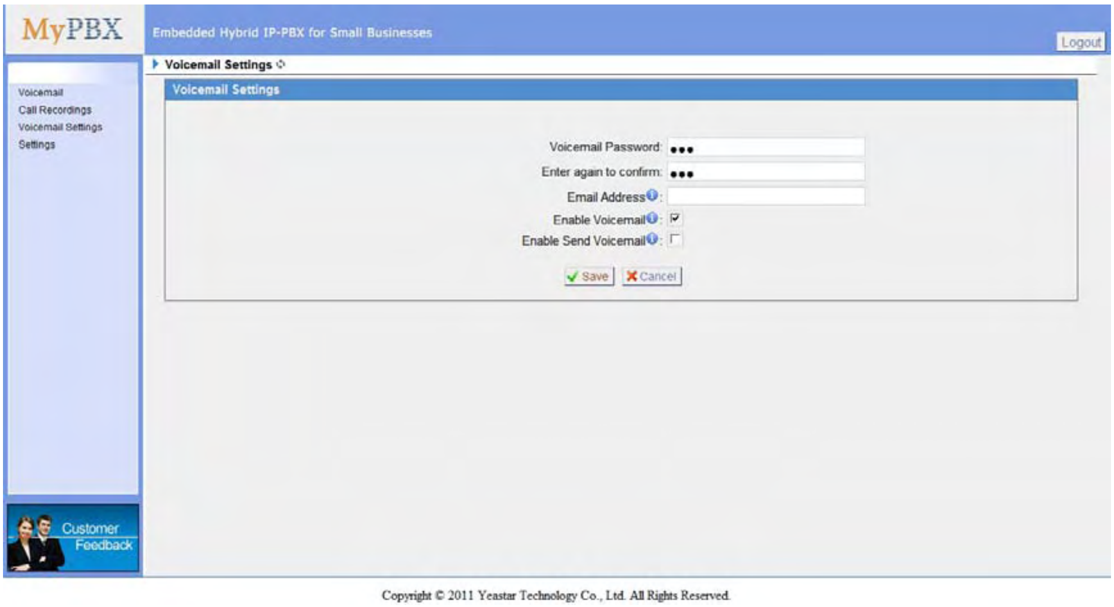


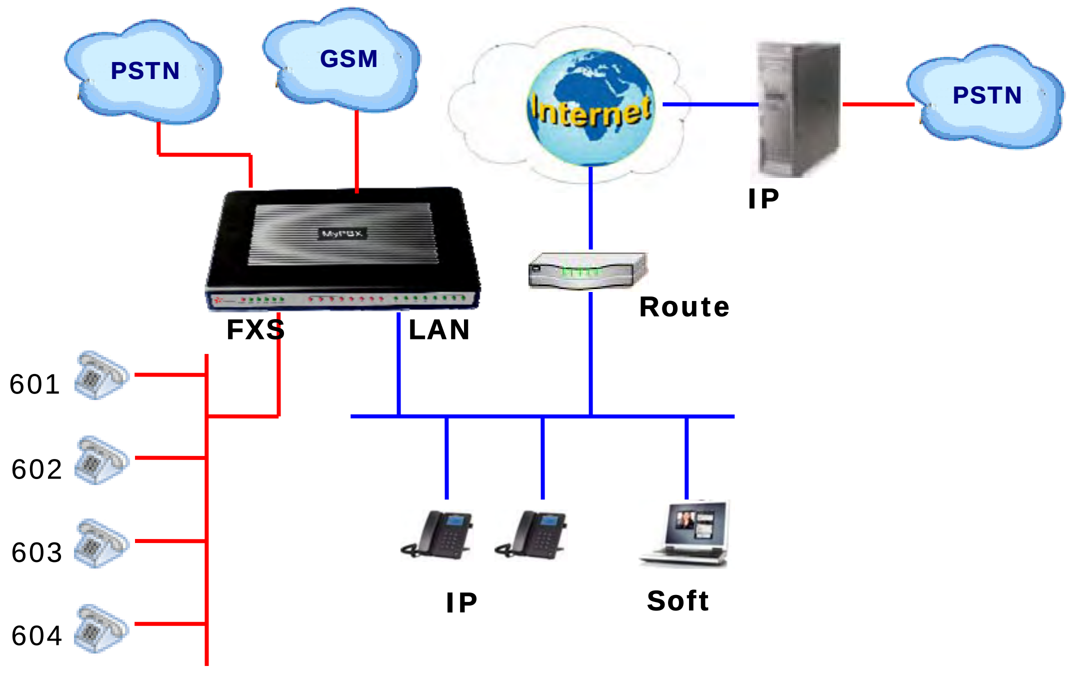
Figure 4-5

4.6 Settings

You can do some basic setting here. Such as call forwarding, DND, Mobile Extension Number.

Figure 4-6

5. Use MyPBX



5.1 Make outbound call

To make an outbound call, we need to add trunk first. There are five types of VoIP Trunk:

·**Analog Trunk:** FXO ports of MyPBX, connected to a local PSTN.
·**GSM Trunk:** GSM ports of MyPBX, connected to GSM Network.
·**BRI Trunk:** BRI ports of MyPBX, connected to a local PSTN.
·**VoIP Trunk:** Connected to remote VOIP service server.
·**Service Provider:** Connected to service provider server.

What are FXO and FXS?

FXS (Foreign exchange Station) is an interface which drives an analog telephone or FAX machine. FXS interfaces deliver power, provide ringing, and use FXO signaling. FXS interfaces are what allow you to hook telephones and other analog devices to your PBX

FXO (Foreign exchange Office) is an interface that connects to a phone line to supply your PBX with access to a public telephone network. FXO interfaces use FXS signaling. FXO interfaces allow you to connect your PBX to real analog phone lines.

5.1.1 Sample Routing via PSTN Trunk

Let's route all inside extensions through an analog trunk by dialing 9. In Outbound Routes, add a new outbound route as below.

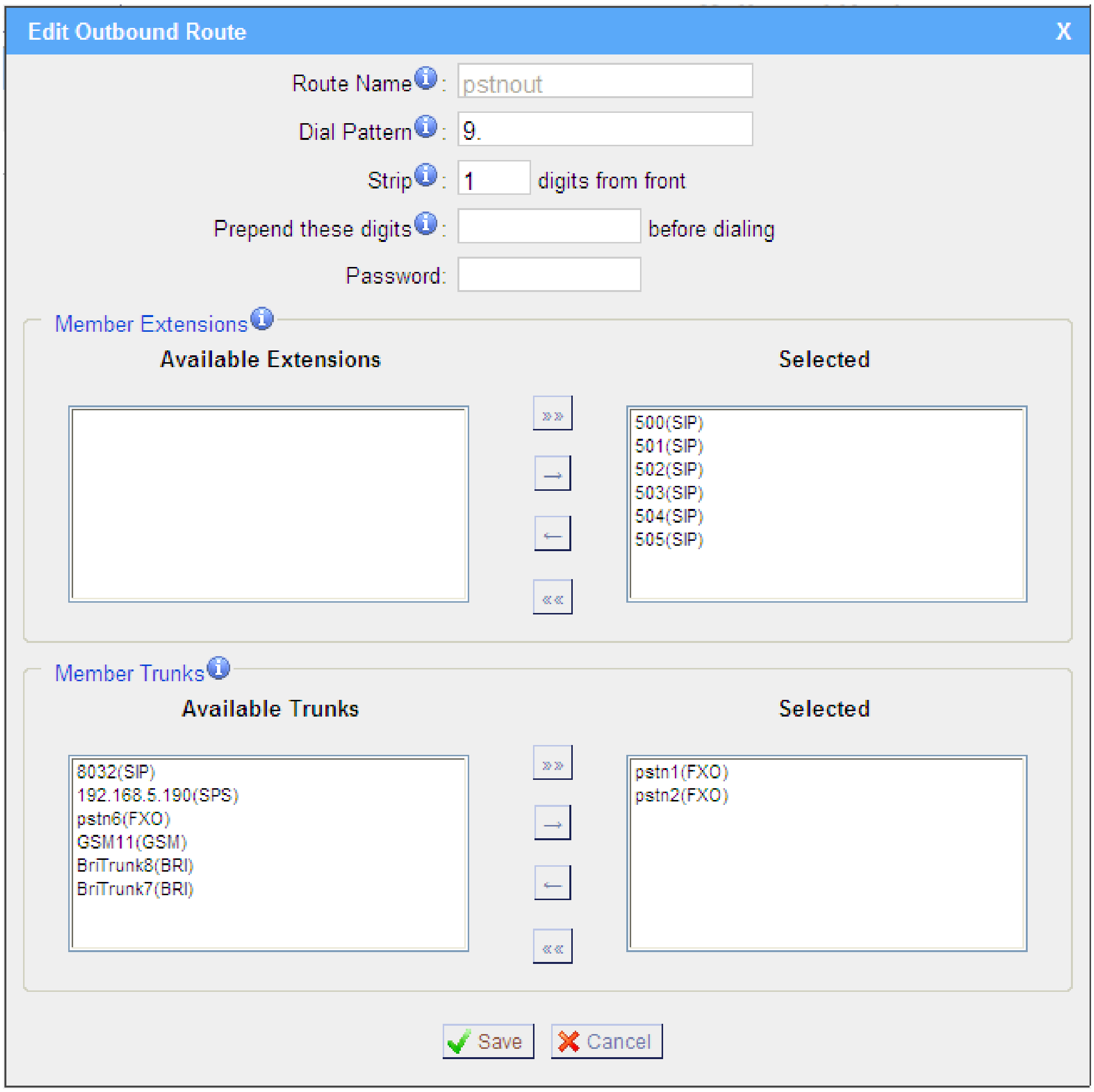


Figure 5-1

As we can see from the outbound route of 'pstnout', all phone numbers starting with 9 will have their first digit stripped off (digit 9) and will be sent to the PSTN (port 1-2).

After we have configured the above, we can dial 9 + local number to dial out via a PSTN line.

Note: Setting number prefix to wild card X and setting Strip to 0 digits from the front will allow all calls to go through this outbound route.

5.1.2 Sample Routing via VoIP Trunk

Let's configure all inside extensions to dial '0' through the VoIP Trunk.

1. Add VoIP service provider

Before we do add this, please make sure you have a VoIP Trunk account.
Trunks→ VoIP Trunk → SIP Trunk
Enter your account information on this page, and click Save.

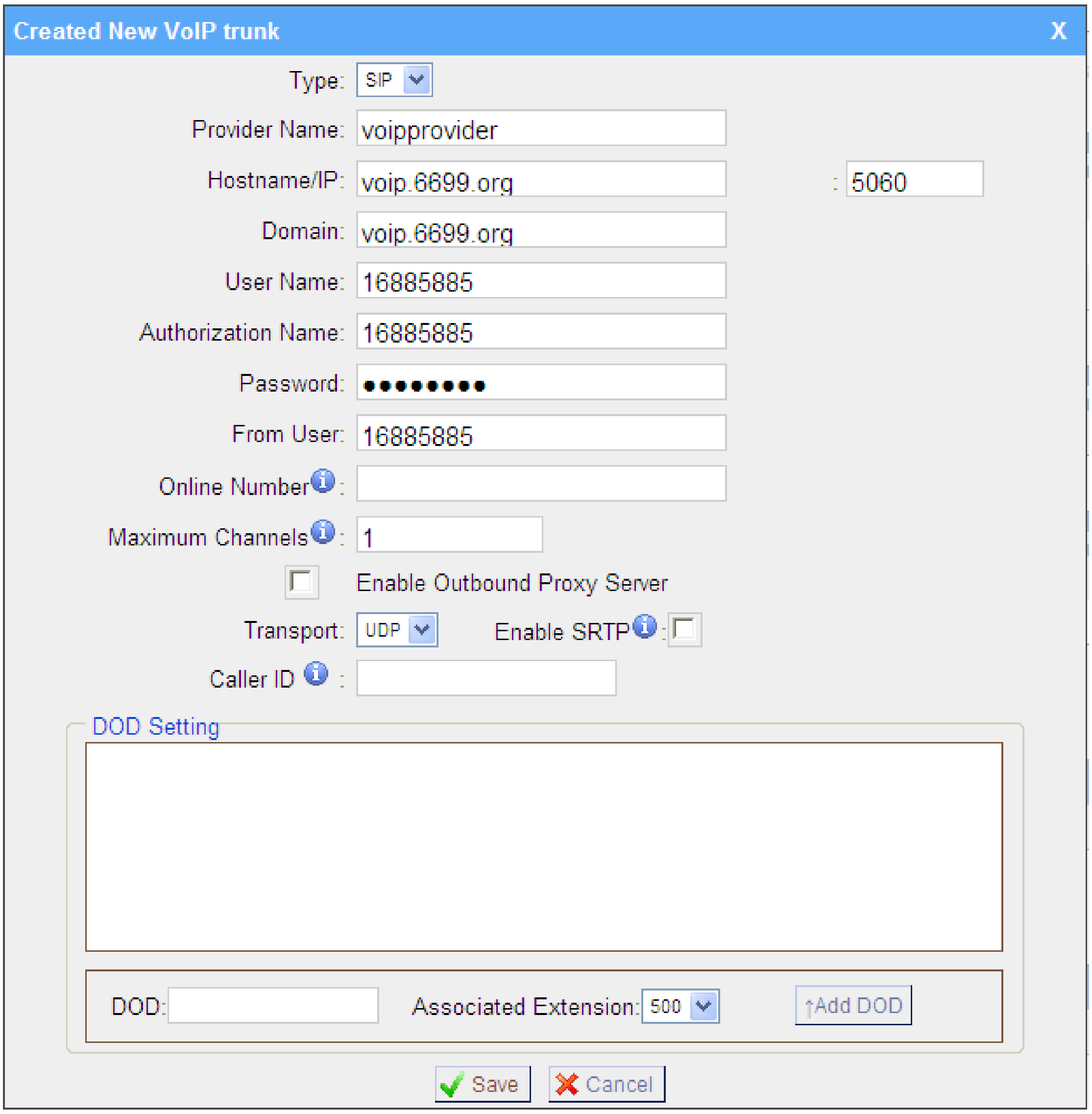


Figure 5-2

2. Add Outbound Routes

As we can see from the Outbound Route of 'voipout', all phone numbers starting with 0 will have their first digit stripped off (digit 0) and will be sent to the SIP Trunk.

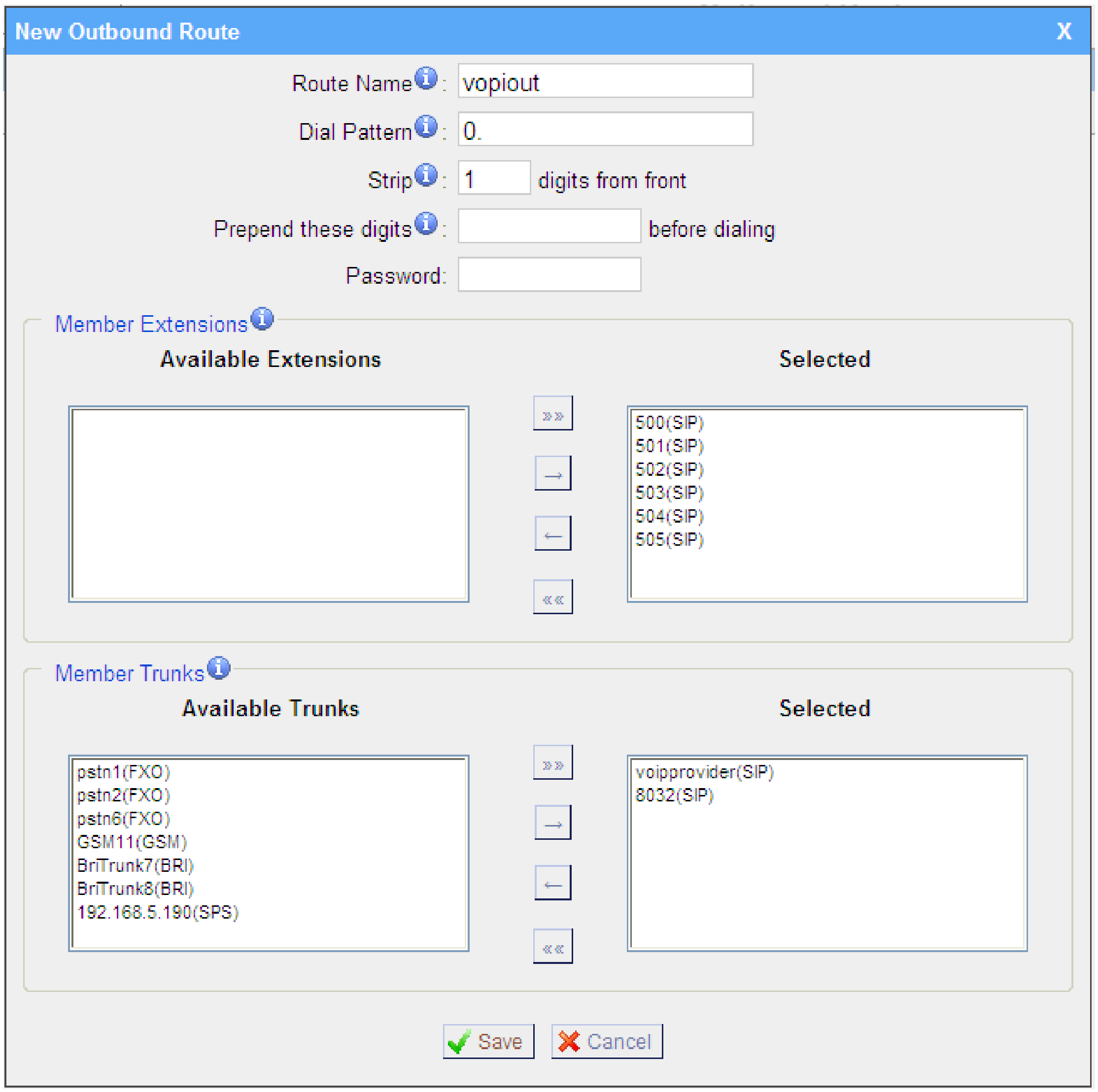


Figure 5-3

Now that we have added two outbound dialing rules, any call starting with 9 will be routed to the PSTN, and any number starting with 0 will be routed to the SIP Trunk.

5.2 Incoming call

5.2.1 Sample Routing to an IVR

Let's configure an incoming call to route to the IVR. In the IVR itself, let's configure digit 0 to route the call to extension 500, and digit 1 to route the call to extension 501.

1. Add IVR

To add a new IVR, go to IVR→ Create New IVR

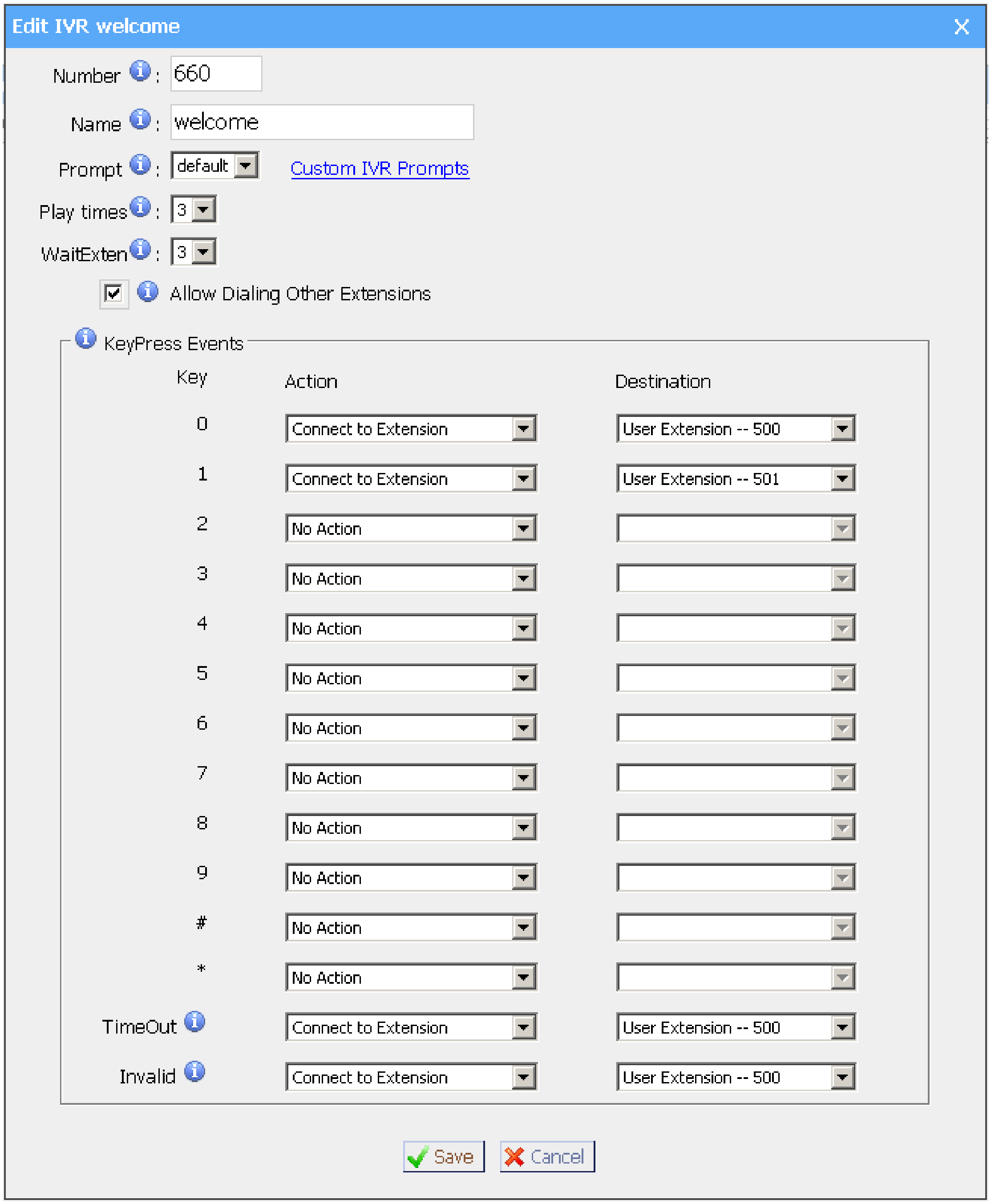


Figure 5-4

2. Add Inbound Routes

As we can see from the Inbound Route of 'allin', all incoming calls will be sent to the IVR.

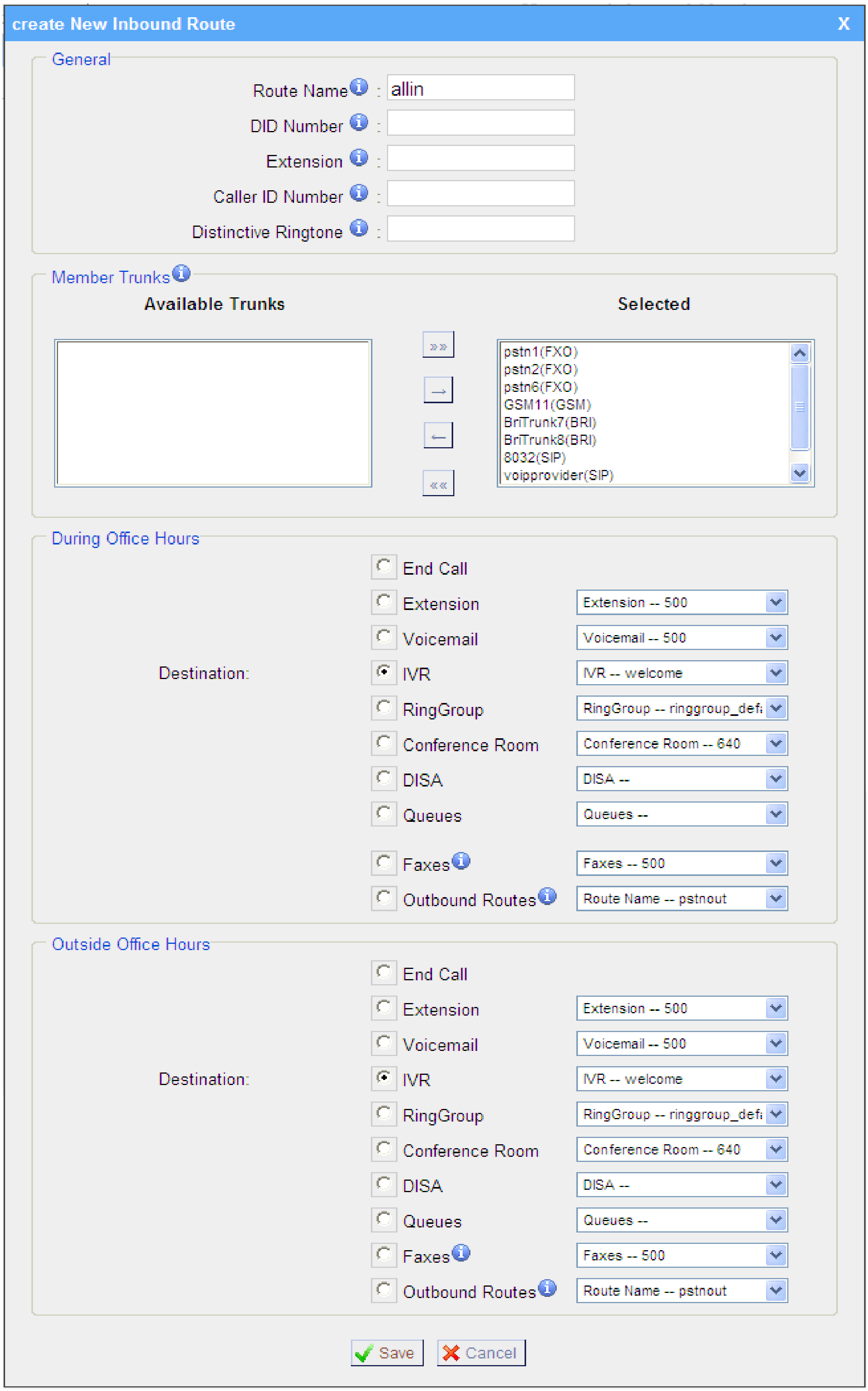

create New Inbound Route
X
General
Route Name : allin
DID Number :
Extension :
Caller ID Number :
Distinctive Ringtone :
Member Trunks
Available Trunks
Selected
pstn1(FXO)
pstn2(FXO)
pstn6(FXO)
GSM11(GSM)
BriTrunk7(BRI)
BriTrunk8(BRI)
8032(SIP)
voipprovider(SIP)
During Office Hours
Destination:
End Call
Extension
Extension -- 500
Voicemail
Voicemail -- 500
IVR
IVR -- welcome
RingGroup
RingGroup -- ringgroup_defa
Conference Room
Conference Room -- 640
DISA
DISA --
Queues
Queues --
Faxes
Faxes -- 500
Outbound Routes
Route Name -- pstnout
Outside Office Hours
Destination:
End Call
Extension
Extension -- 500
Voicemail
Voicemail -- 500
IVR
IVR -- welcome
RingGroup
RingGroup -- ringgroup_defa
Conference Room
Conference Room -- 640
DISA
DISA --
Queues
Queues --
Faxes
Faxes -- 500
Outbound Routes
Route Name -- pstnout
Save
Cancel

Figure 5-5

APPENDIX A FAQ

Q1. How to Register SIP device?

A1:
1) Register SIP soft phone
Download the x-lite softphone from counterpath website
www.counterpath.com
After install the x-lite, right click the panel and select the SIP Account setting and then configure it.
Display Name: 500
User Name: 500
Password: 500
Authorization Name: 500
Domain: 192.168.5.150

2) Register IP Phone (for example, Yealink's T28 IP Phone)
a) Connect the T28's Internet port to the switch. And it can get the IP from your route.
b) Press the 'OK' key on T28 to get the IP of T28.
c) Put the IP on web browser then you can enter the T28 configure page through this IP.
d) Put the SIP extensions info on the T28 IP phones.
Display Name: 501
User Name: 501
Register Name: 501
Password: 501
SIP Server: 192.168.5.150

Use the same method register another T28 to other extension.

Q2. How do I reset MyPBX back to the factory default settings?

A2: To perform a reset, please follow steps below:
Step 1: Hold down the 'Reset' button on the back of the unit for 5 seconds and watch the LEDs on the front of the MyPBX. When the status LED turns red, let go of the reset button.
Step 2: When the RUN status LED starts blinking, MyPBX will be set back to factory defaults.
Step 3: To access the configuration page, navigate to 192.168.5.150 using

a web browser. Make sure that you are on the 192.168.5.0 subnet before doing this.

Step 4: Login to the device with the username 'admin' and the password 'password', in order to begin reconfiguring the device.

APPENDIX B How to Configure Autobackup

Before Autobackup 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 Add a new folder, rename it, and set this new folder's share Properties according to Figure B-1

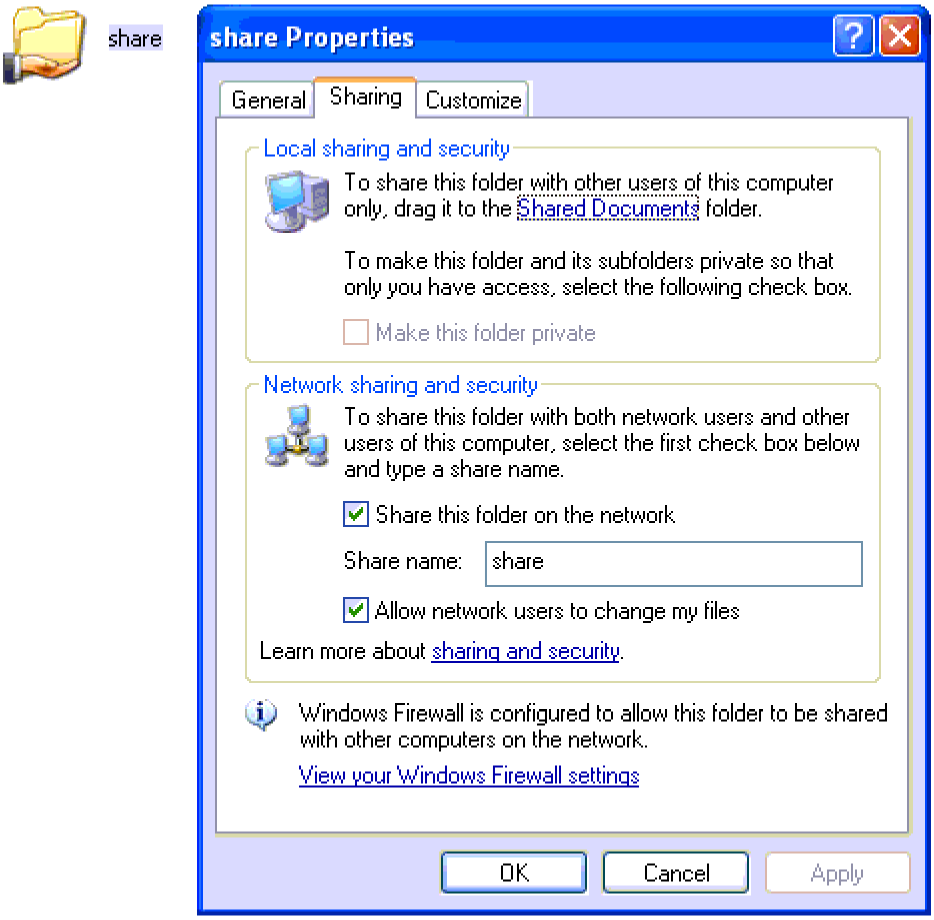


Figure B-1 Set up share Properties

Step 2 Enter the new folder and create a new text file, then rename this file to status.txt. This step is very important, DO NOT forget to create the status.txt file.

Step 3 Configure Autobackup settings on MyPBX to Figure B-2

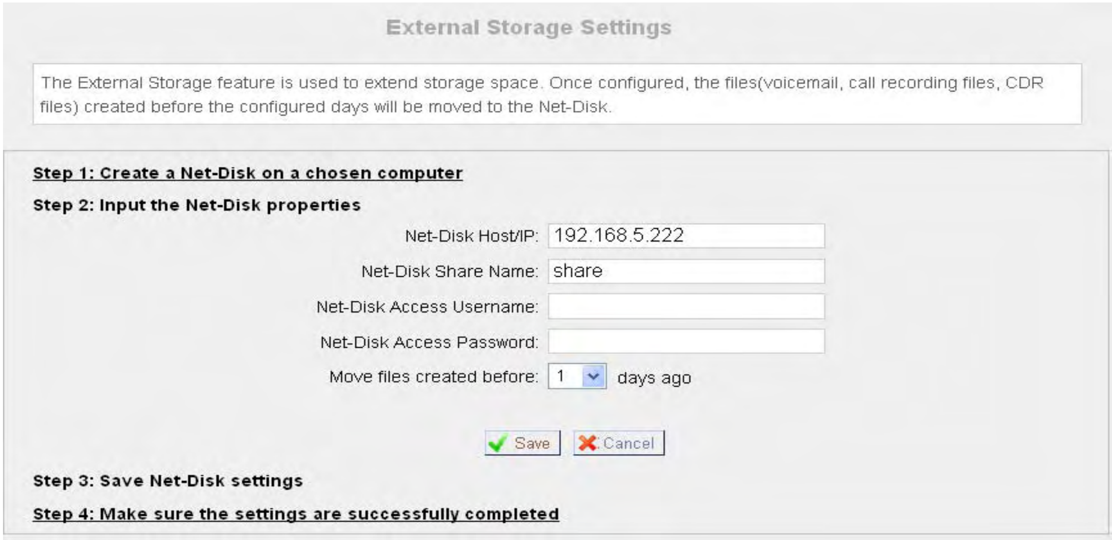


Figure B-2 Autobackup Setting

Net-Disk Host/IP: Change this to the IP address of the computer where backup files will be stored.

Net-Disk Share Name: Change this to the name of the shared folder where backups will be stored.

Net-Disk Share Username: The user name used to log into the network share. Leave this blank if it is not required

Net-Disk Share Password: The password used to log into the network share. Leave this blank if it is not required

If configuring is correctly, open your Windows share folder to see if the MyPBX backup files and folders has been created. If the contents of the backup folder look similar to Figure B-3, then you have successfully configured Autobackup on the MyPBX unit.

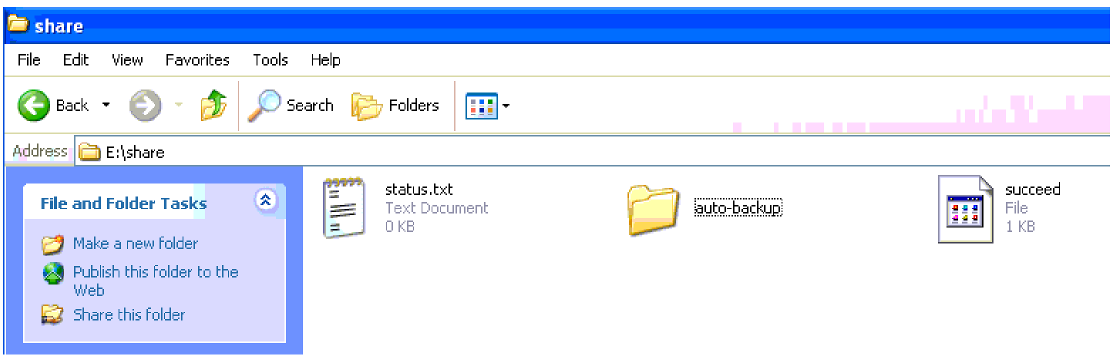


Figure B-3 Autobackup setting succeed

APPENDIX C How to Configure NAT setting

When MyPBX is behind a NAT(firewall),you need to configure NAT setting on MyPBX if you want to use a remote extension.
Please follow section **1** or **2** below depending on your network configuration.
1. If MyPBX is connected to a local network, you must set up port forwarding on your router. Specifically, you must map port 5060 (default SIP port) and port 10001-10200 (default RTP port range) as UDP ports.

Next, go to the MyPBX web interface and configure the SIP settings according to Figure C-1:
External IP Address: your router's public IP address
External Host: your router's domain
External Refresh Interval: 20 seconds
Local Network Identification:192.168.5.0/255.255.255.0 (change this according to your network configuration)
NAT mode: Yes
Allow RTP Reinvite: No

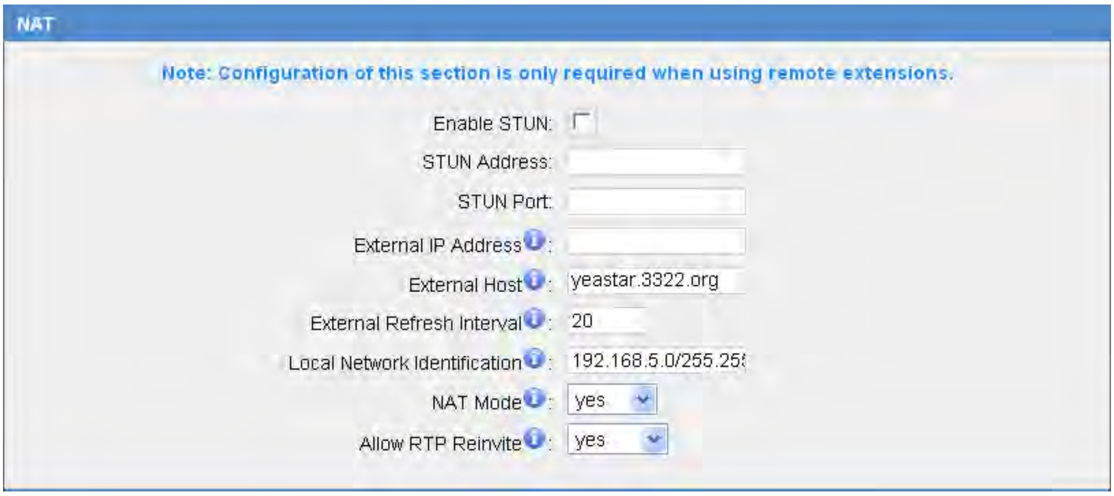


Figure C-1

Assuming that your router's host address is yeastar.3322.org, your local network is from 192.168.5.1-192.168.5.254, and the subnet Mask is 255.255.255.0, the MyPBX network settings should configured like Figure C-2

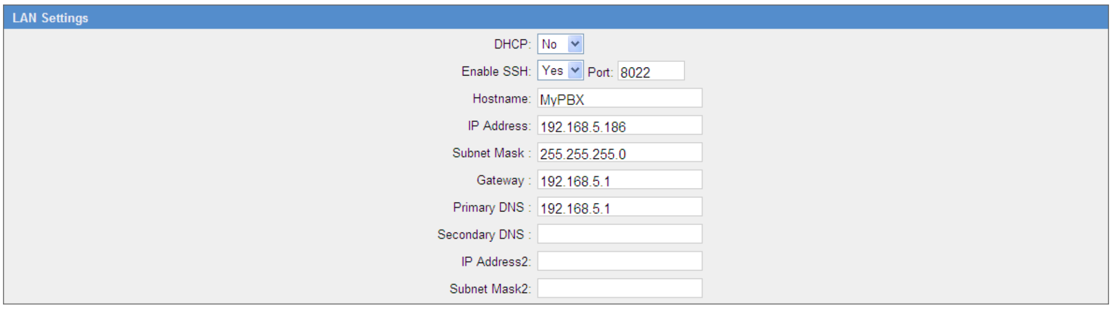


Figure C-2 MyPBX Network setting

2. If MyPBX has a public IP, (i.e. is connected directly to your internet service provider), the network settings should be configured according to Figure C-3:

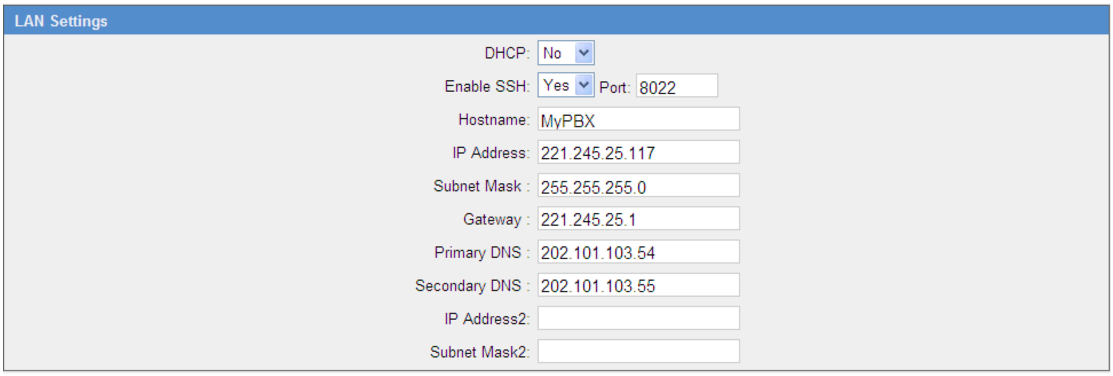


Figure C-3

Next, you should configure the NAT settings according to Figure C-4

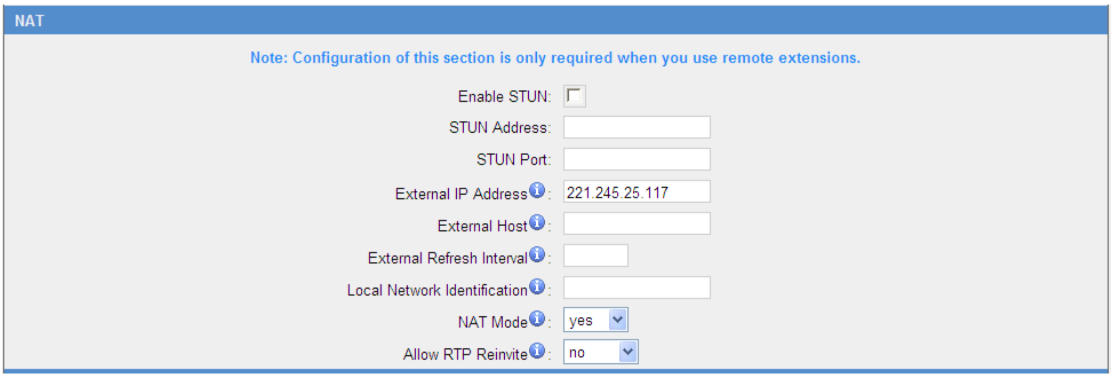


Figure C-4

External IP Address: The public IP address of MyPBX
External Host: Leave this blank if no domain has been configured
External Refresh Interval: Leave this blank
Local Network Identification: Leave this blank
NAT mode: Yes
Allow RTP Reinvite: No

APPENDIX D How to Use Auto Provision

Step1. Disable DHCP Server on your local network.
E.g. Disable DHCP Server on Linksys Router.

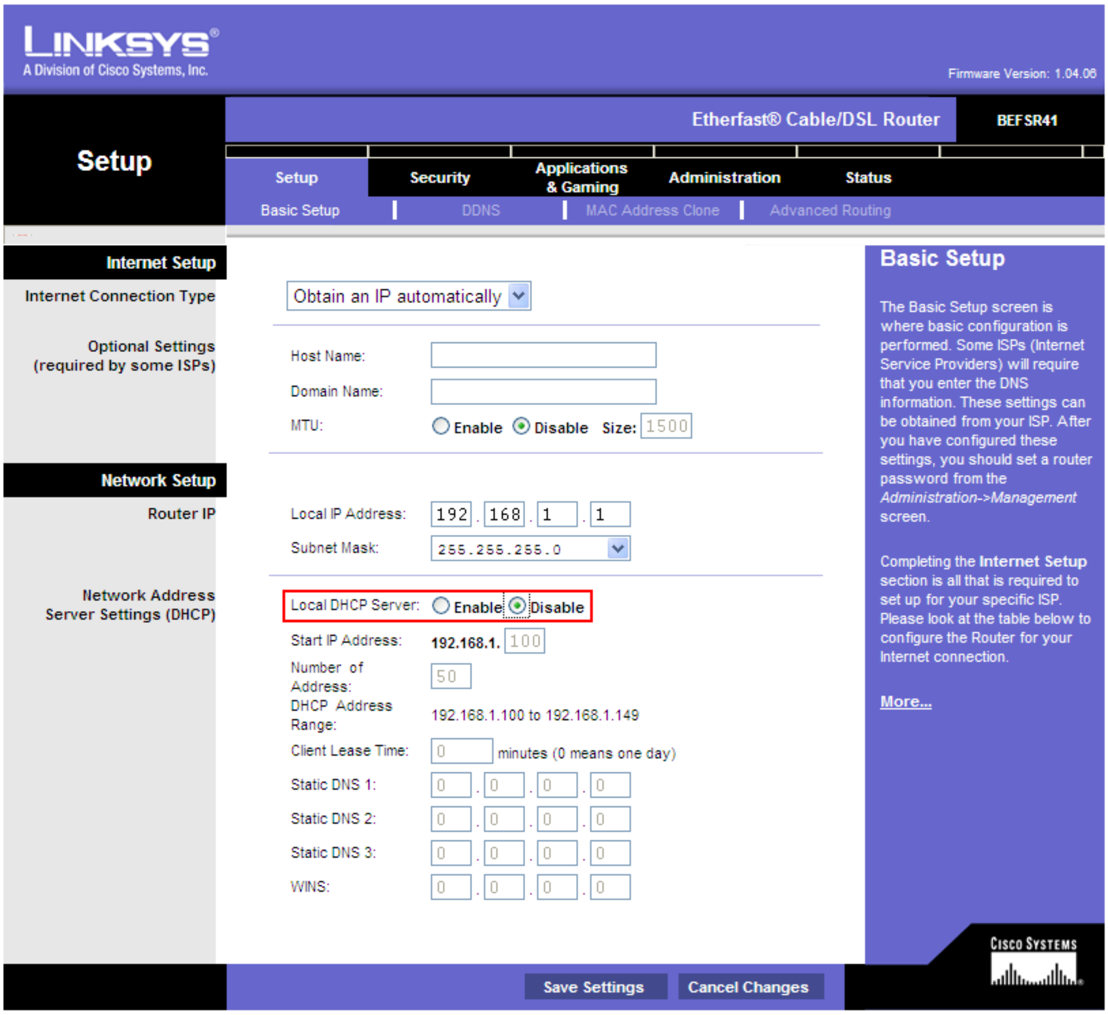


Figure D-1

Step2. Enable DHCP Server on MyPBX.
Login MyPBX web interface, System Settings → DHCP Server → Enable DHCP Server.

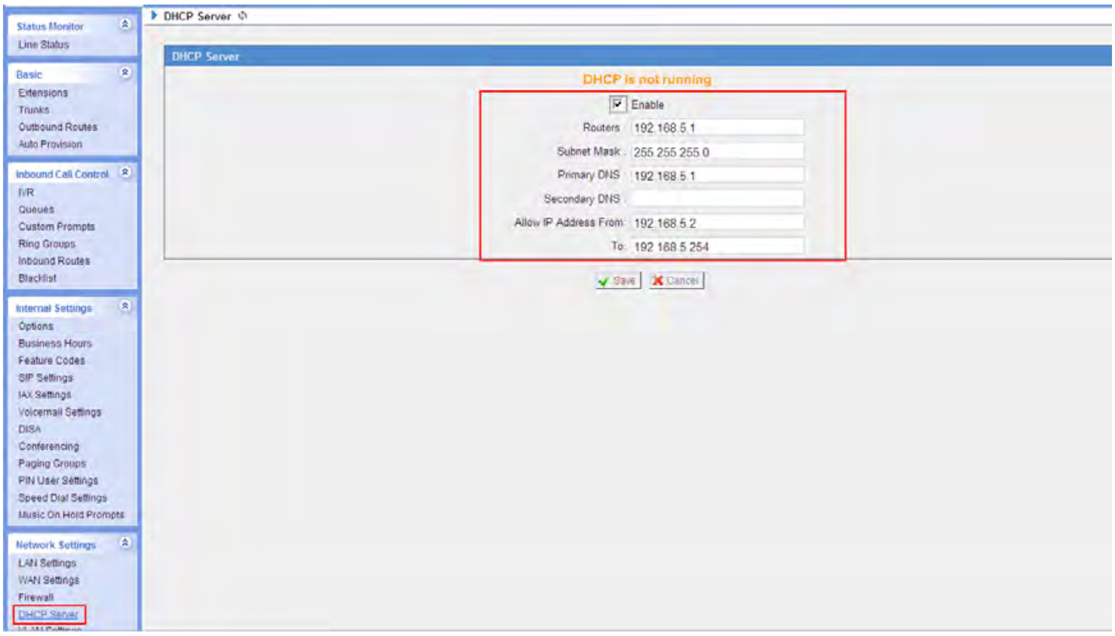


Figure D-2

Step3. Configure phones on MyPBX auto-provision page.

1. Login MyPBX web interface, Basic → Auto Provision → Create New Phone.

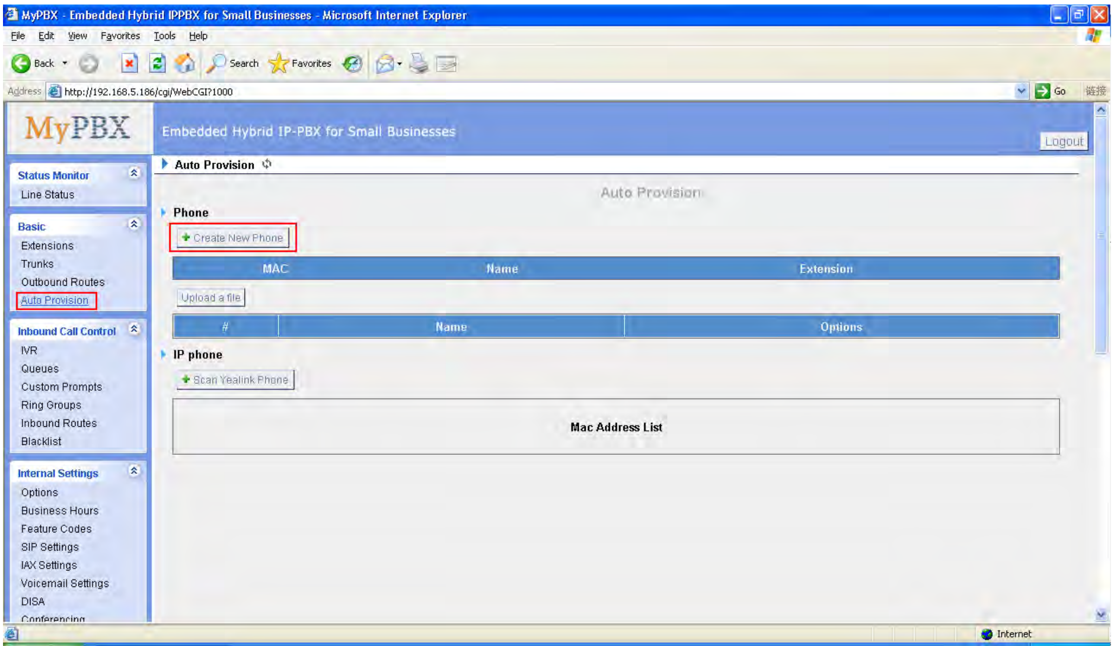


Figure D-3

2. Fill in the phone detail message on the pop-up windows.

Input IP Phone’s MAC address, configure Name, Call waiting, Line, Extension, Label, Line active for the phone.

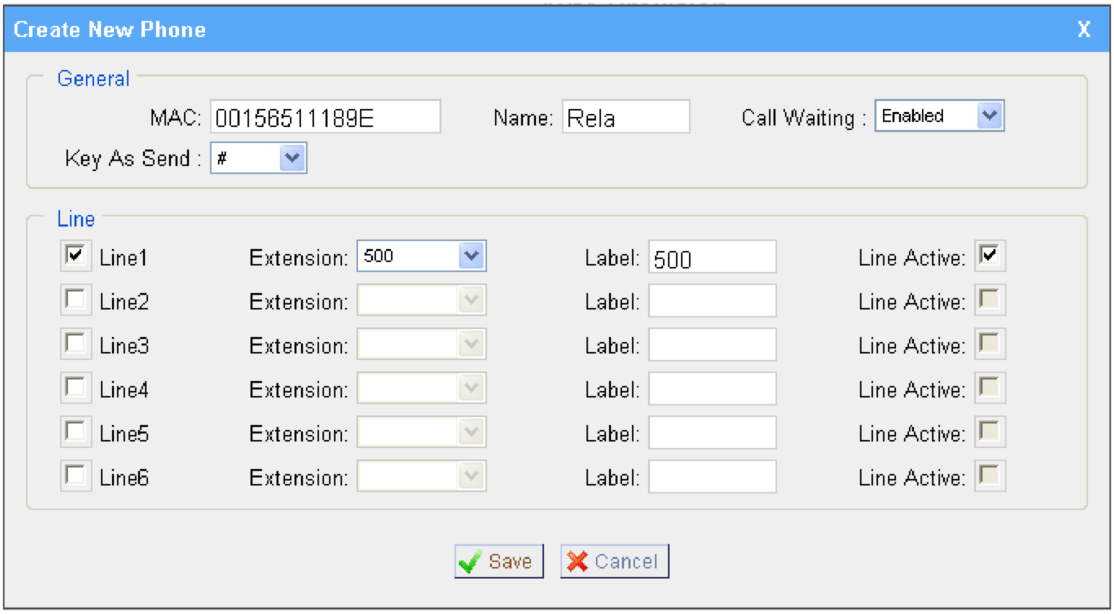


Figure D-4

Step4. Turn on the power and connect the network cable to IP Phone.

Remark: The factory default setting of DHCP for IP Phone is enable, so you can skip this step to step 5.

If the DHCP is disable, please follow below step to enable it. (e.g.: Yealink's IP Phone).

1. Login IP phone's web page.
2. Enable DHCP.

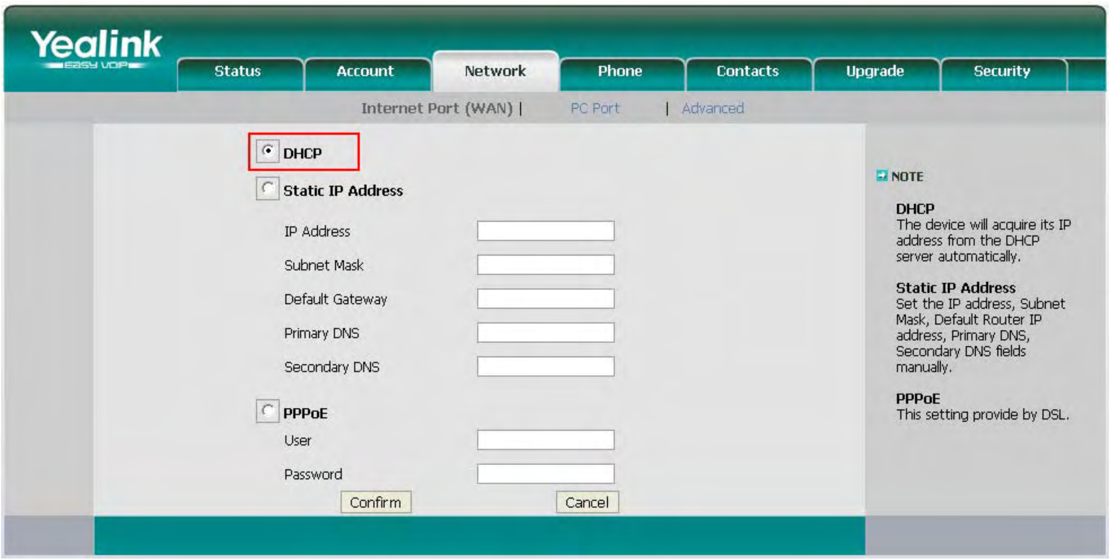


Figure D-5

Step5. Finish.

APPENDIX E How Do I Configure Distinctive Ring Tones

Step1: On your IP phone, navigate to the Phone settings web configuration page and find the Distinctive Ring Tone section.
For each custom ring tone, enter the Internal Ringer Text (can be digits or text) to trigger the ring tone. For example, you may enter “Family”.
e.g.: Yealink’s IP phone.

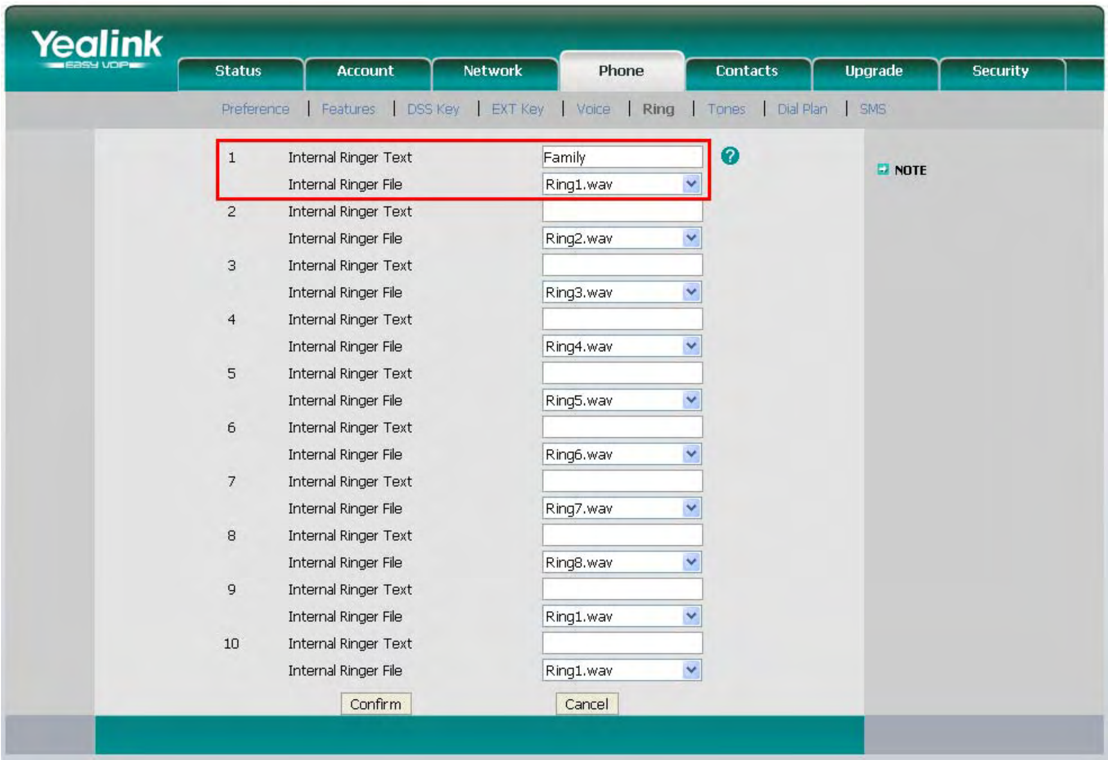


Figure E-1

Step2. Configure the ‘Distinctive Ringtone’ on MyPBX.
MyPBX web interface, Inbound Routes → Edit Inbound Route, fill in the Internal Ringer Text on ‘Distinctive Ringtone’.

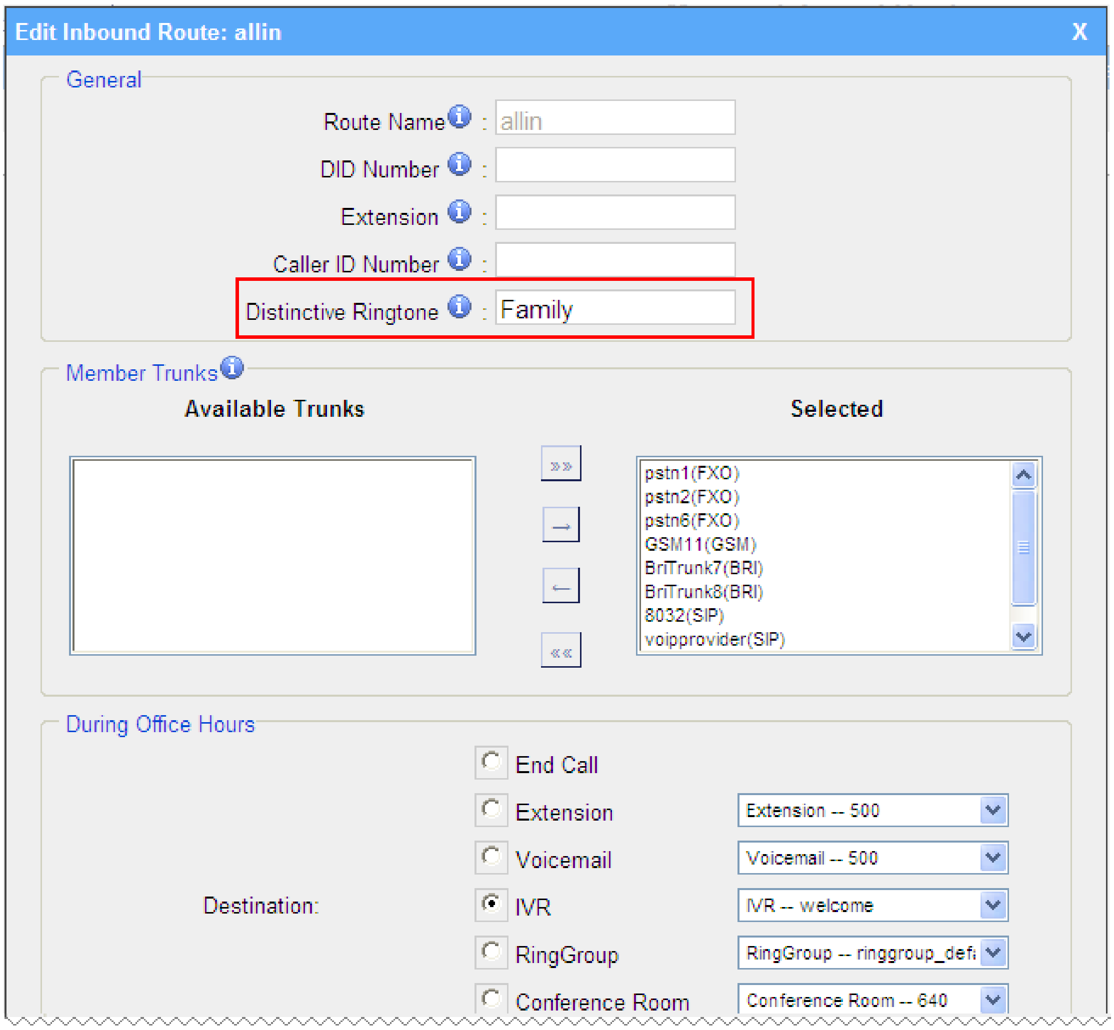


Figure E-2

Step3. Finish.

APPENDIX F How to Use Email to SMS

How to use Email to SMS

You need to send an email to the specified email address (you set in Email Settings. In this case, it is lears@yeastar.com).The content of this email will be sent to the number you want as message. The subject (title) of the email will determine the number. Here are some examples of the formats to the subject of the email.

Example:

1. Send message with no PIN code and default GSM port.

Format: phonenumber

if the subject is "12345678", the text of this email("Welcome to Yeastar!") will be sent to number "12345678" through the first available GSM trunk(No pin code should be set by administrator).

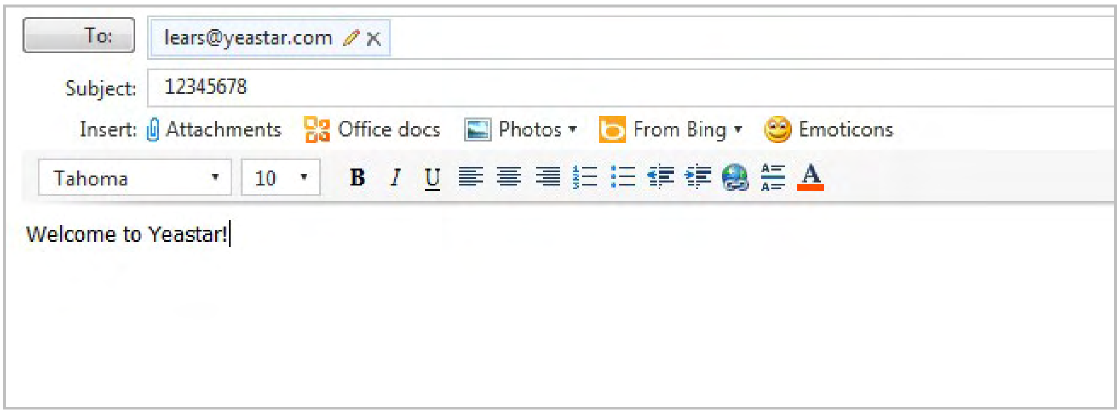


Figure F-1

2. Send message with no PIN code and specified GSM port.

Format: port:portnumber-phonenumber

if the subject is "port:9-12345678", the text of this email ("Welcome to Yeastar!") will be sent to the number "12345678" through GSM trunk 9 (No pin code should be set by administrator).

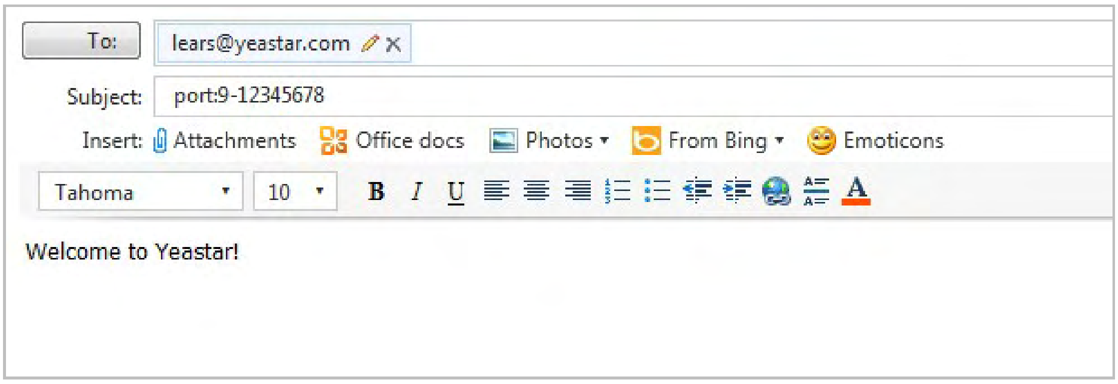


Figure F-2

3. Send message with PIN code and default GSM port.

Format: 500:pincodenumber-phonenumber

if the subject is "500:987-12345678", the text of this email(“Welcome to Yeastar!”) will be sent to number "12345678" through the first available GSM trunk("987" is the pin code set by administrator).

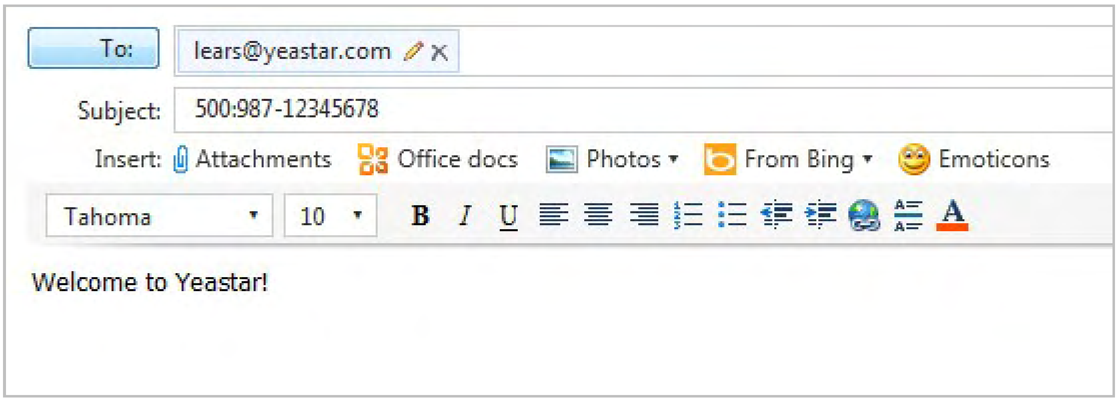


Figure F-3

4. Send message with PIN code and specified GSM port.

Format: 500:pincodenumber-port:portnumber-phonenumber

if the subject is "500:987-port:9-12345678", the text of this email(“Welcome to Yeastar!”) will be sent to number "12345678" through GSM trunk 9("987" is the pin code set by administrator).

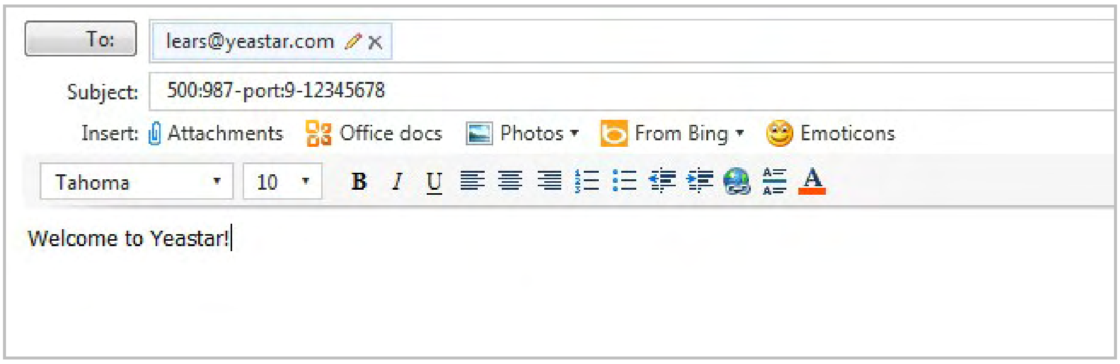


Figure F-4

APPENDIX G How to Use DID

Direct inward dialing (DID), also called direct dial-in (DDI) in Europe and Oceania, is a feature offered by telephone companies for use with their customers' private branch exchange (PBX) systems. In DID service the telephone company provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk.

MyPBX support DID, you can configure DID in inbound route. Related settings: **DID Number, Extension, Destination.**

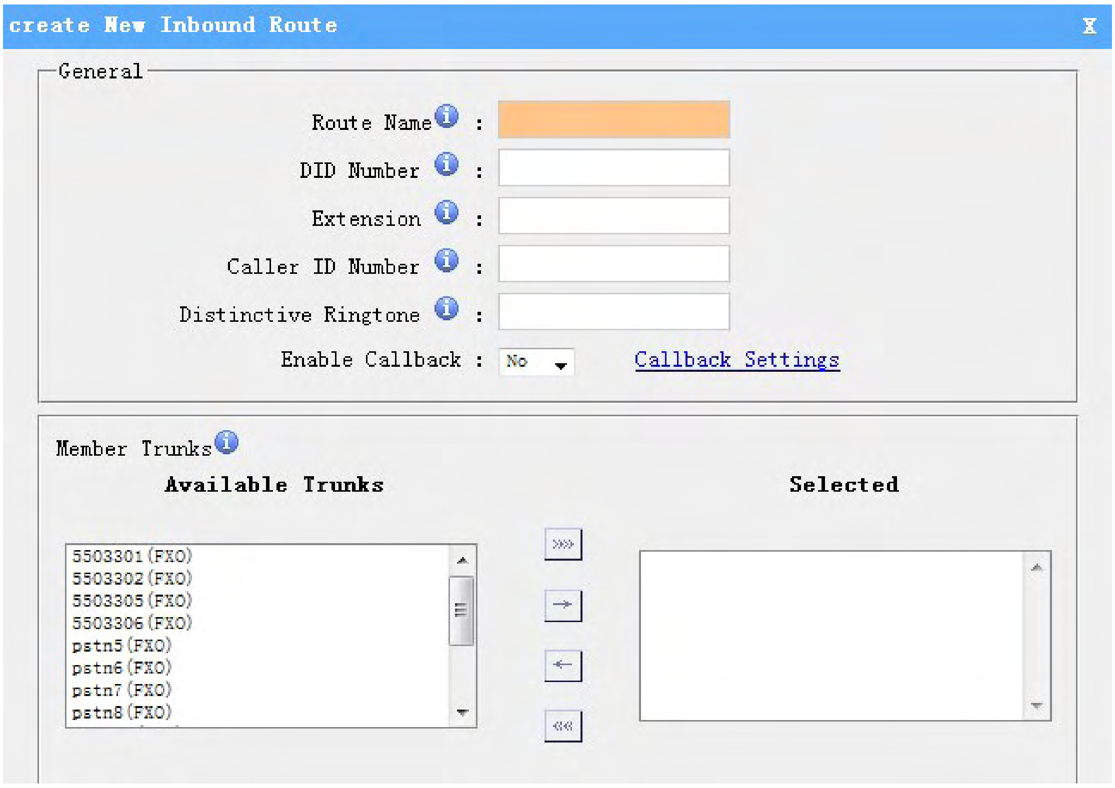


Figure G-1

·DID Number

Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. Only service provider, E1 trunks, BRI trunks or SIP trunks need to be configured with this setting.

You can also use pattern matching to match a range of numbers. The following patterns may be used:

X: Any Digit from 0-9

Z: Any Digit from 1-9

N: Any Digit from 2-9
[12345-9] : Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)
The '**.**' Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.
The '**!**' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

·Extension

Define the extension for DID number, this field only valid when use E1 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.

·Destination

If you don't set the extension, you can set the destination of the call here.

Example 1:

Step1: You set the DID number (5503XXX in this example).
Step2: You choose the destination (Ring Group in this example).

The configuration of this example means when the incoming call with DID number 5503XXX (7 digits number start with 5503) will go to the destination Ring Group.

If you choose the destination, please leave the Extension form blank.

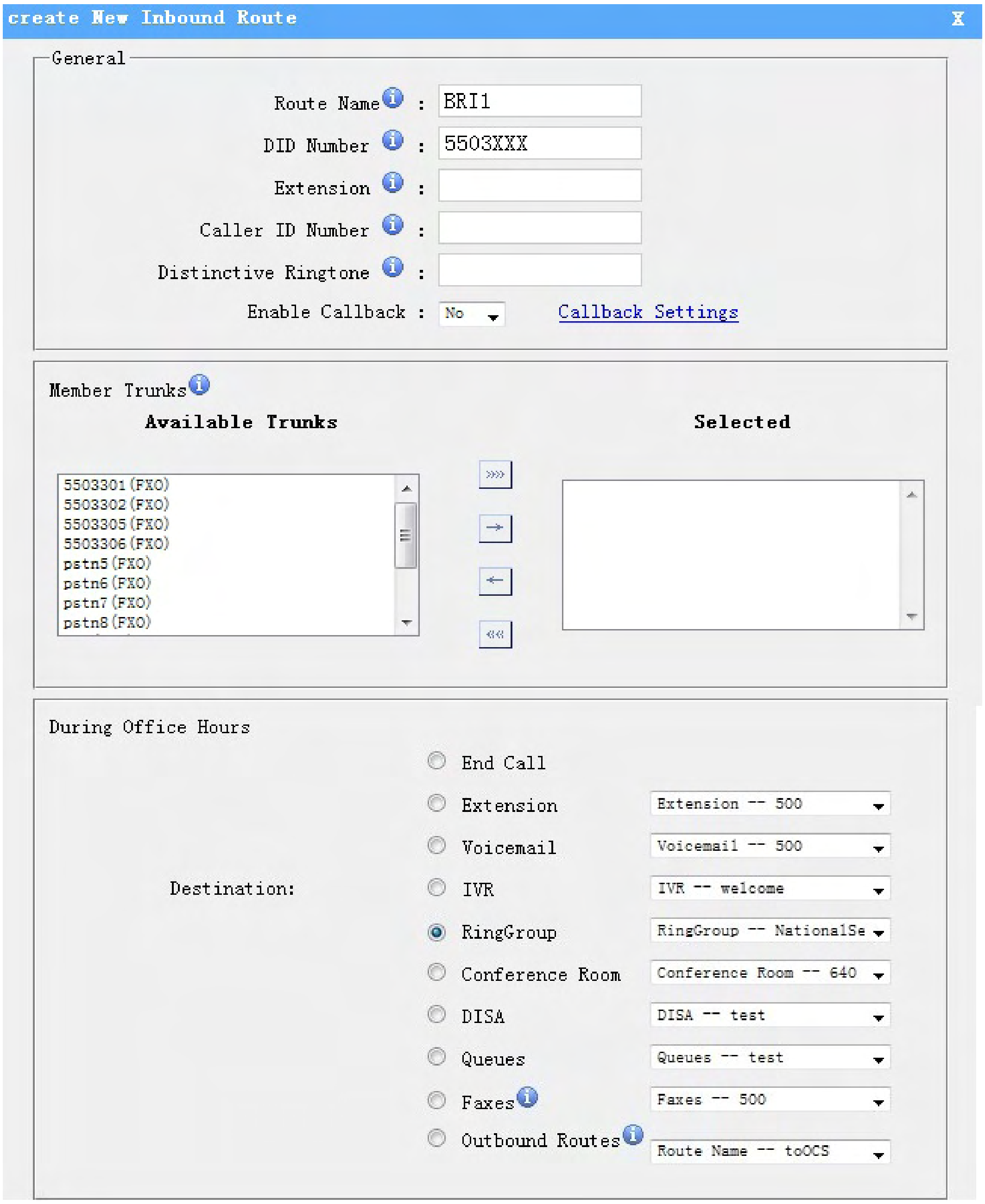


Figure G-2

Example 2:

Step1: You set the DID number (6001-6099 in this example).

Step2: You set the Extension (6001-6099 in this example).

The configuration of this example means when the incoming call with DID number 6001 to 6099 will go to the destination 6001 to 6099(number 6001 to extension 6001, number 6002 to extension 6002).

The destination you set below will be disabled if you set the Extension.

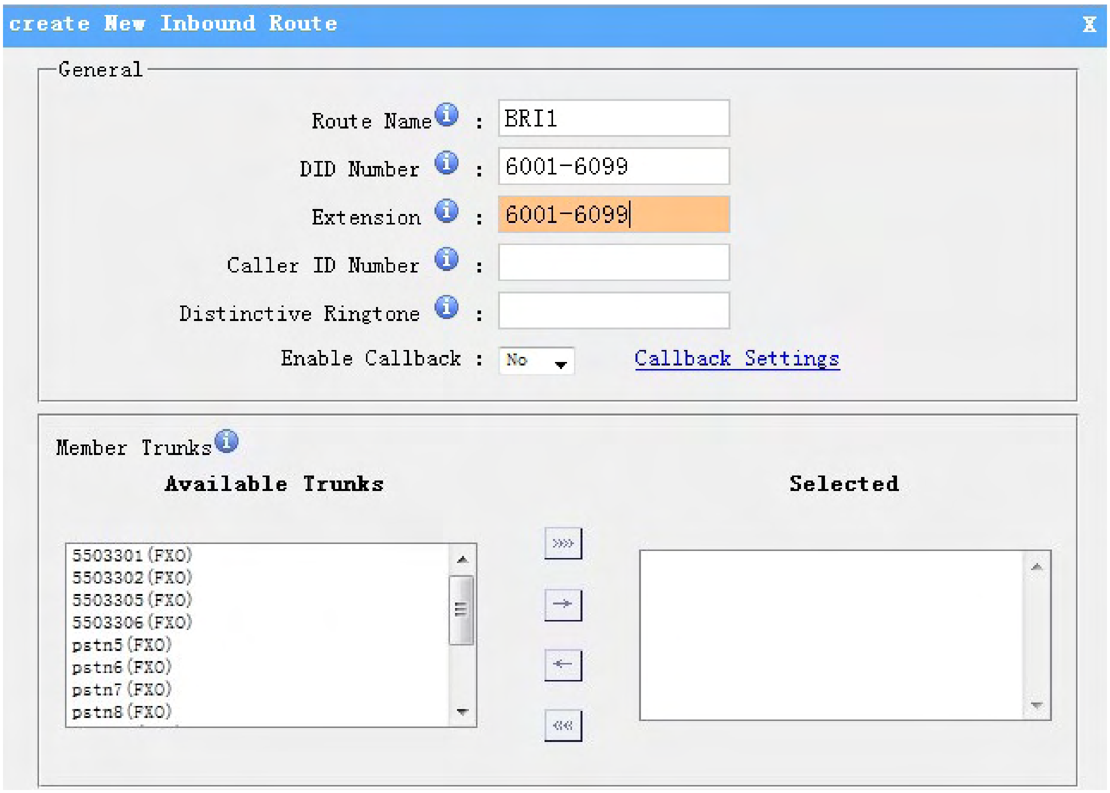

create New Inbound Route
X
General
Route Name :
BRI1
DID Number :
6001-6099
Extension :
6001-6099
Caller ID Number :
Distinctive Ringtone :
Enable Callback :
No
Callback Settings
Member Trunks
Available Trunks
Selected
5503301(FXO)
5503302(FXO)
5503305(FXO)
5503306(FXO)
pstn5(FXO)
pstn6(FXO)
pstn7(FXO)
pstn8(FXO)

Figure G-3

APPENDIX H How to Use BLF Key to Choose the PSTN line.

MyPBX allows you to choose the specific PSTN line to make outbound call by pressing the BLF key on the IP Phone.

Follow the steps to do the configuration with your Yealink phone

1. We want to choose pstn1 or pstn2 to call out.

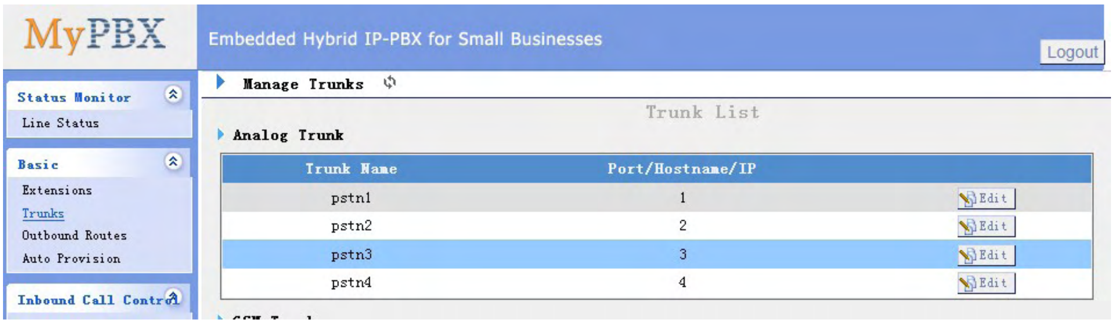


Figure H-1

2. Configure the IP Phone:

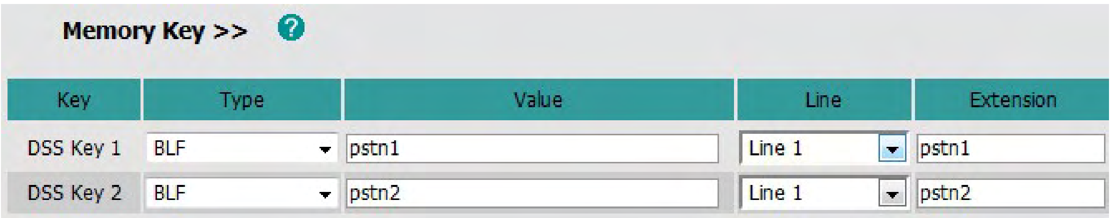


Figure H-2

Test

When you press DSS Key 1/2, the phone will connect to pstn1/pstn2 line. If pstn1/pstn2 is not busy, you will hear the dial tone. You can dial the number you want and use this line to call out then.

<Finish>