

mPBX-100 User Manual

Version 2.12.0.93

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1. Introduction

MPBX –An IP-PBX for Small Businesses/Home Office

MPBX is a standalone embedded hybrid PBX for small businesses and remote branch offices of larger organizations (1-100 users per site). MPBX 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.

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)
• Conference	• Three-way Calling
• Define Office Time	• T.38
• Direct Inward System Access(DISA)	• Voicemail
• DIDs	• VLAN
• Distinctive Ringtone	• WAN
• Do Not Disturb(DND)	• PPPoE

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

1.2 Hardware Specifications

1.2.1 Exterior Appearance

Front Side



Figure 1-1 MPBX Front Panel Picture

No.	Identifying
<input type="checkbox"/>	Green LED: Indicates correct power is being supplied to the unit
<input type="checkbox"/>	Green LED: Indicates the MPBX is fully functional.
<input type="checkbox"/>	Green LED: Indicates stable WAN Port connection
<input type="checkbox"/>	Green LED: Indicates stable LAN Port connection
<input type="checkbox"/>	<p>Red LED: Indicates presence of an FXO/GSM port.</p> <p>Orange LED: Indicates presence of a BRI port.</p> <p>Green LED: Indicates presence of an FXS port.</p> <p>LED Blinking- Red blinking: No connection between FXO port and PSTN</p> <p>LED Alternating - Red and Green (slow blink): FXO port is receiving an incoming call.</p> <p>LED Alternating - Red and Green (fast blink): FXO port is in use</p> <p>LED Dual - Green and Red (slow blink): FXS port is ringing</p> <p>LED Dual - Green and Red (fast blink): FXS port is in use</p>

2. System set up

2.1 Connection Drawing

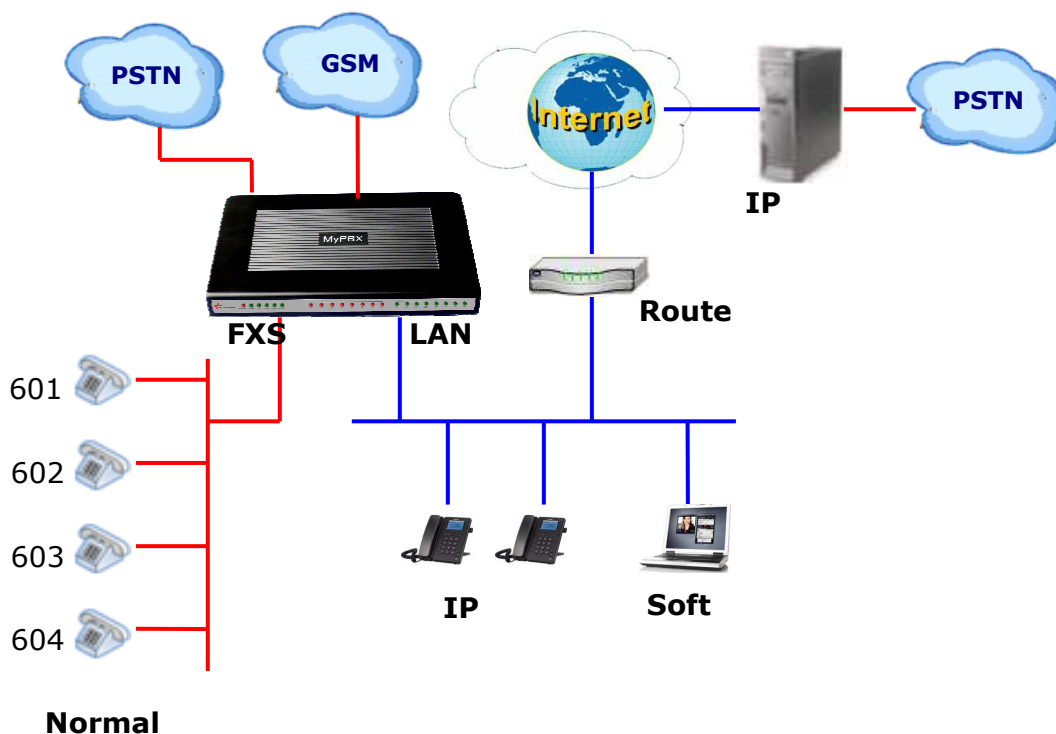


Figure 2-1

2.2 Connecting Ethernet Line

MPBX provides two 10/100M Ethernet ports with RJ45 interface and LED indicator. Plug Ethernet line into MPBX'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

MPBX 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 the MPBX unit to a power outlet:

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

3. Managing MPBX

3.1 Administrator Login

From your web browser, input the IP address of the MPBX server.

If this is the first time you are configuring MPBX, please use the default settings below:

IP Address: <http://192.168.5.150>

Username: **admin**

Password: **password**

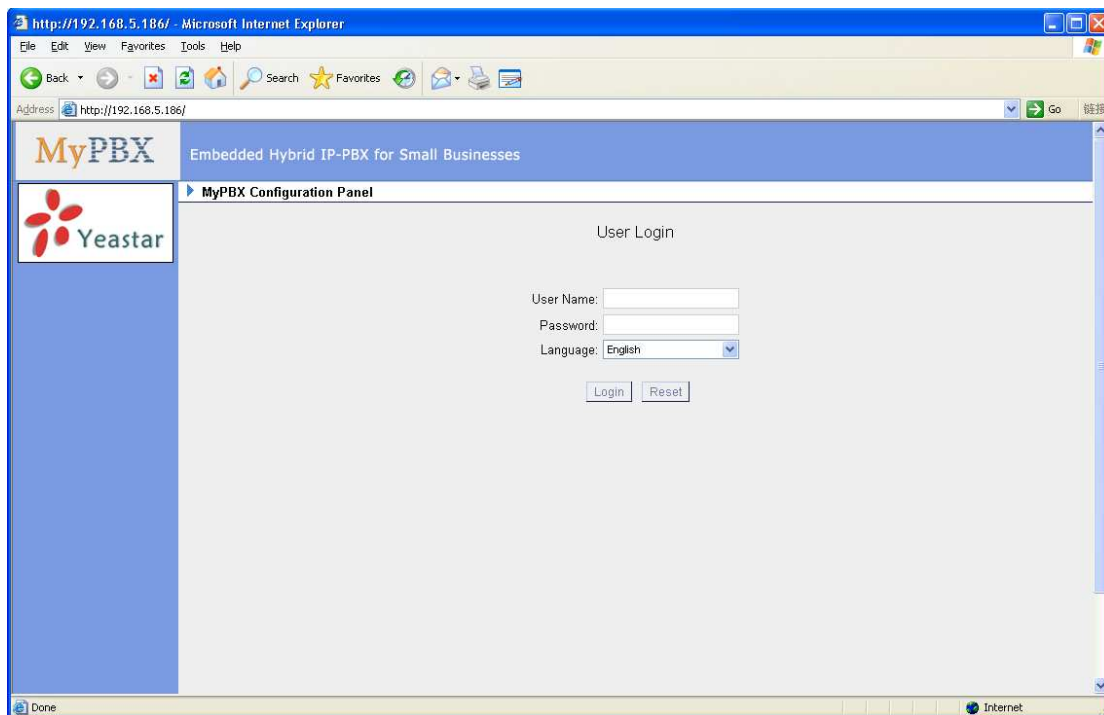


Figure 3-1

3.2 Status Monitor

3.2.1 Line Status

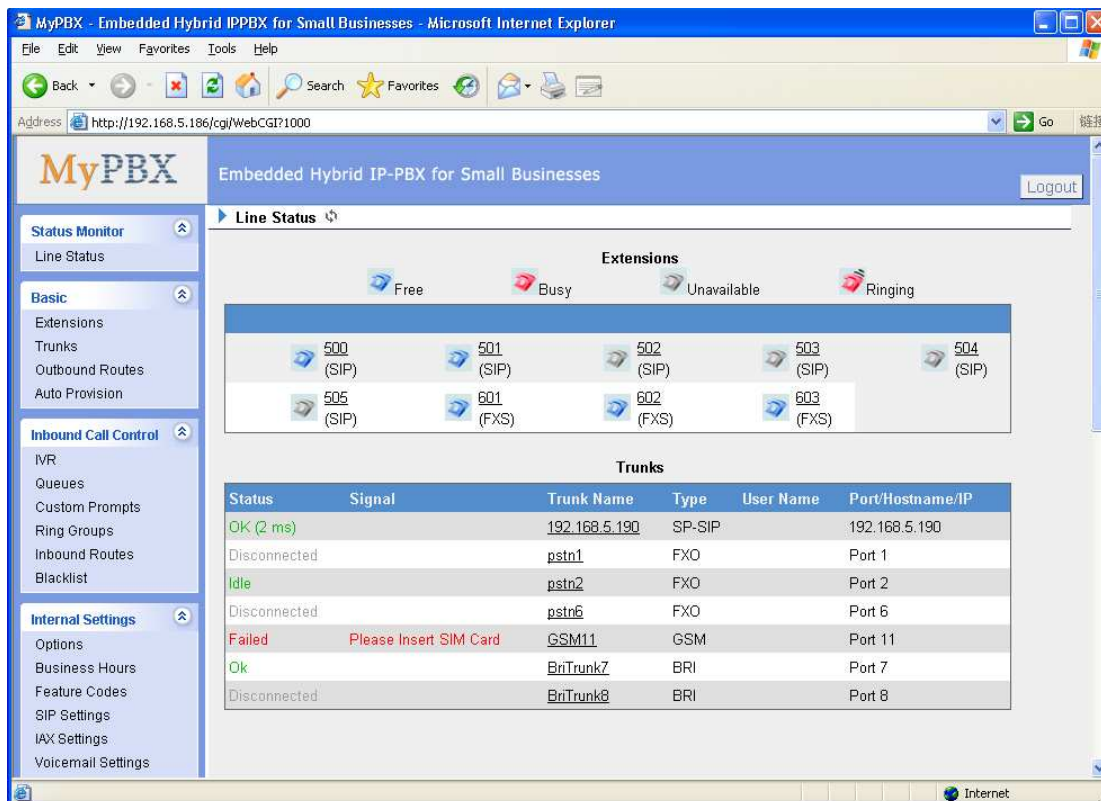






Figure 3-2

MPBX Status Description:

Extensions:

- 1)  : Extension is unregistered
- 2)  : Extension is idle
- 3)  : Extension is ringing
- 4)  : Extension is busy in a call

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.

Error: The port is error.

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

Service Provider:**Status**

Registered: Succeed registration, trunk is ready for use.

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).

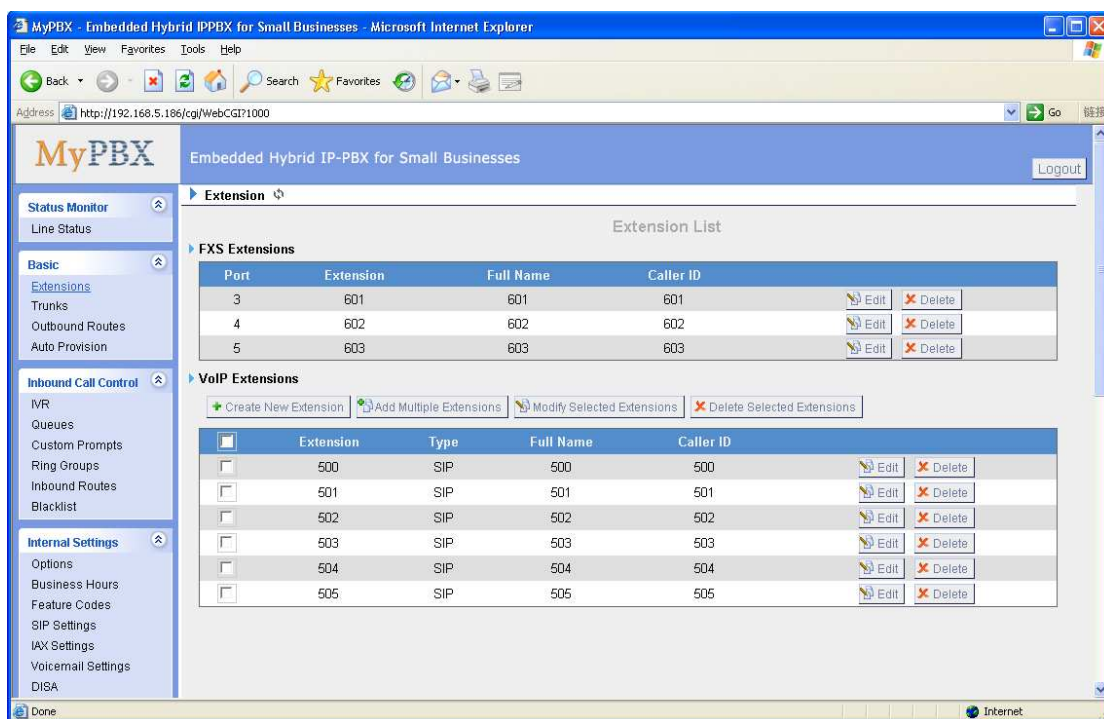


Figure 3-3

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 sending to 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 MPBX before it will recognize a second event. If a second event occurs in less time than defined by Hook Flash Detection, then MPBX 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 MPBX User Web interface, which can be used to access voicemail and extension recordings. Users may login to the MPBX 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 custom the ring time. Tone will stop over the time defined.

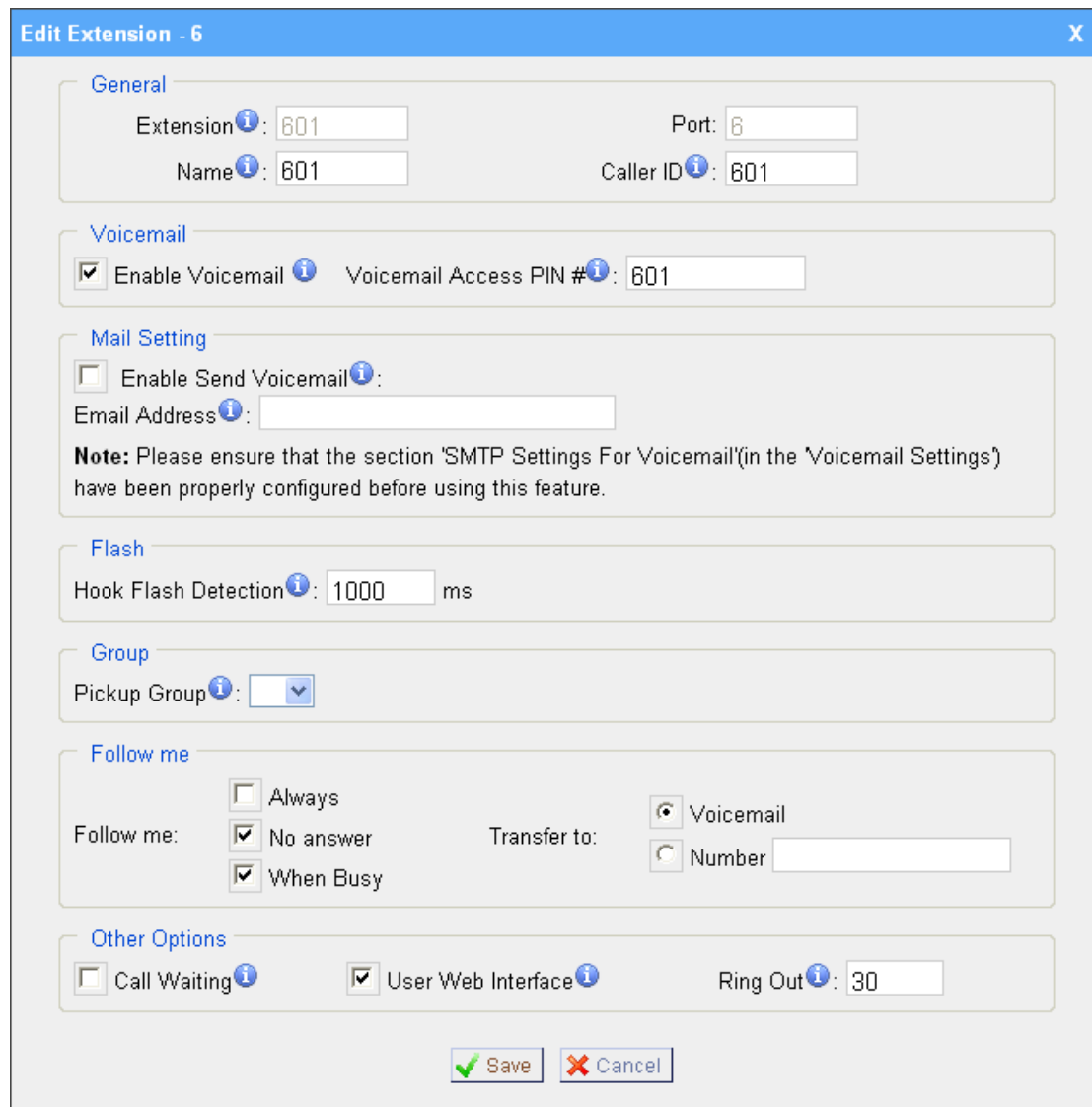


Figure 3-4

3.3.1.2 VOIP Extension

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

1. Add VOIP Extension

Go to Extensions → VOIP Extensions → Create New Extension

1) General

•Type

Extension type: SIP,IAX or SIP/IAX.

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

IAX - Identifies whether 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.

•SIP Transport

This will be the transport method used by the Extension. The options are UDP (default) or TCP or TLS.

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 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.

4) VoIP Settings

•NAT

This setting should be used when the system is using a public IP address, communicating with devices hidden behind a NAT device (such as a broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP and/or RTP ports.

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

•Enable SRTP

Enable extension for SRTP.

5) IP Restriction

•Enable IP Restriction

Check this option to enhance the VoIP security for MPBX. If this option is enabled, only the permitted IP or 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.

6) 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.

7) Follow me (Call Forwarding)

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

8) 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 MPBX User Web interface, which can be used to access voicemail and extension recordings. Users may login to the MPBX User Web interface by using their extension number and

voicemail PIN # as the login and password respectively.

.Ring Out

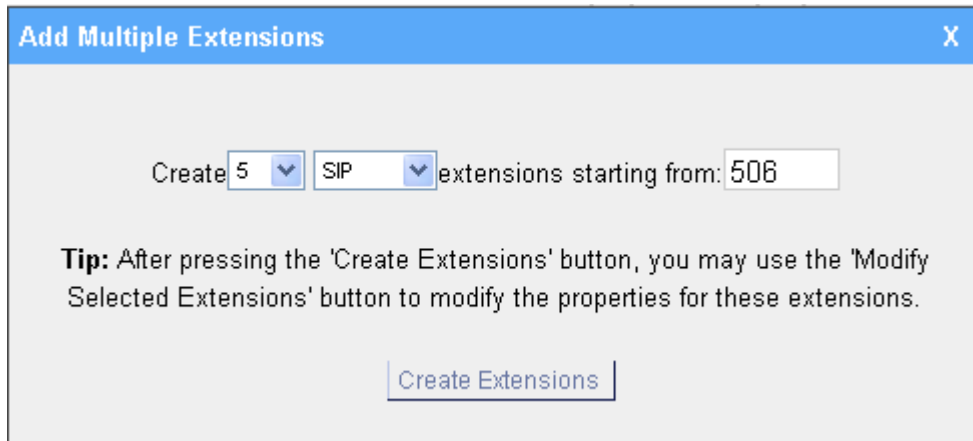
Check this option if you want to custom the ring time. Tone will stop over the time defined.

Figure 3-5

2. Add Multiple Extensions

Go to Extensions → VOIP Extensions → Add Multiple Extensions.

- 1) Choose the number of extensions that you would like to create.
- 2) Choose the type of extension that you would like to create.
- 3) Fill in the starting extension number.



Add Multiple Extensions X

Create 5 SIP extensions starting from: 506

Tip: After pressing the 'Create Extensions' button, you may use the 'Modify Selected Extensions' button to modify the properties for these extensions.

Create Extensions

Figure 3-6

3. Edit VOIP Extension

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

3.3.2 Trunk

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

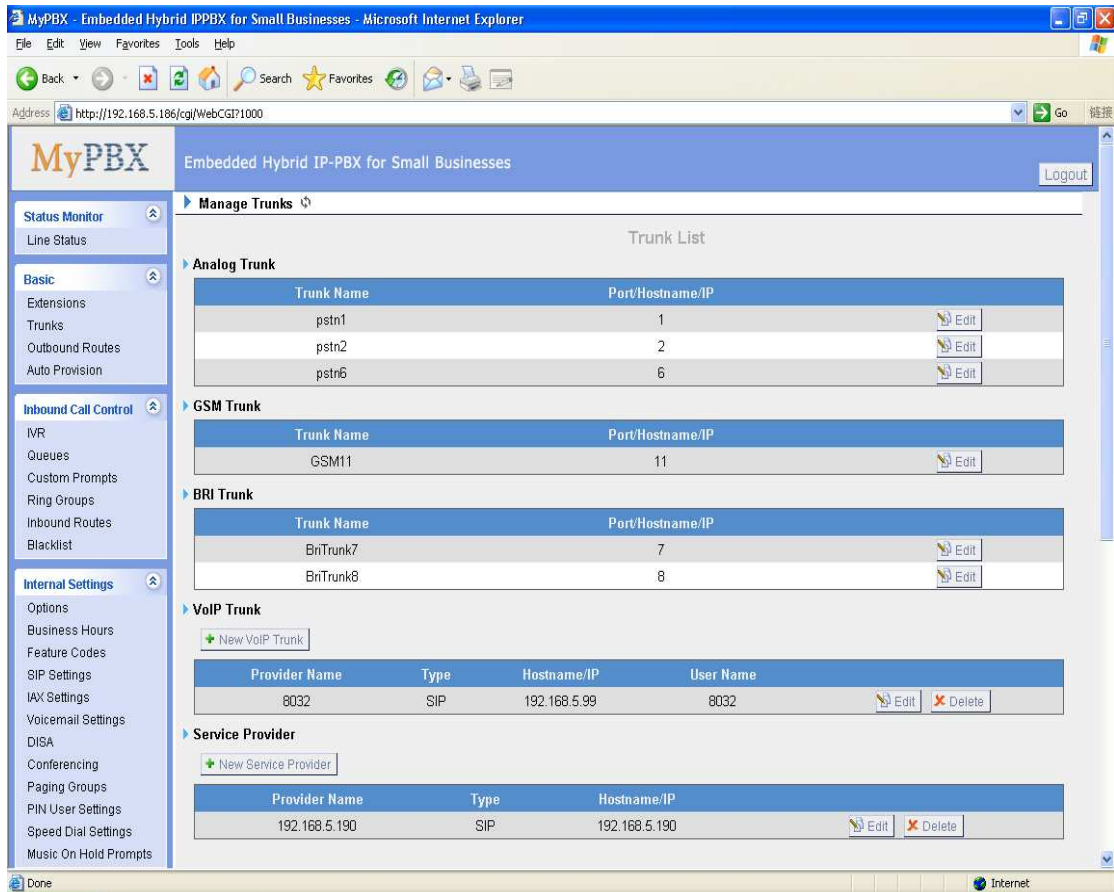


Figure 3-7

3.3.2.1 Analog Trunk (FXO)

1. Edit Analog Trunk (FXO)

On the analog 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: 'Port 5'

•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 default is 4, but better results can be achieved if set to 6 or even 8. Remember, the higher the number, the more time required to release a channel. A higher setting lowers the probability that you will encounter random hang-ups

•Busy Pattern

If Busy Detection is enabled, it is also possible to specify the cadence of your busy signal. In many countries, it is 500msec on, 500msec off. Without Busy Pattern specified, MPBX will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then MPBX 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 (the caller id signal).

•Busy Frequency

If the Frequency Detection is enabled, you must specify the local frequency.

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: Used in the UK

v23_JP: Used in Japan

DTMF: Used in Denmark, Sweden, and Holland

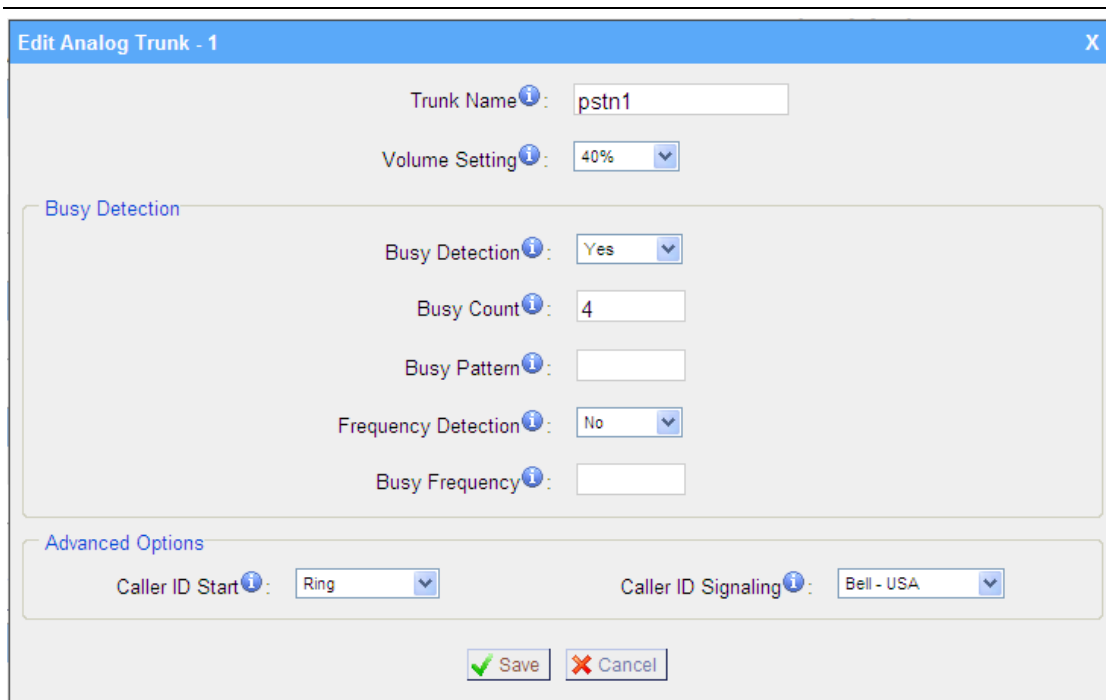


Figure 3-8

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: 'gsm1'

•Volume Setting

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

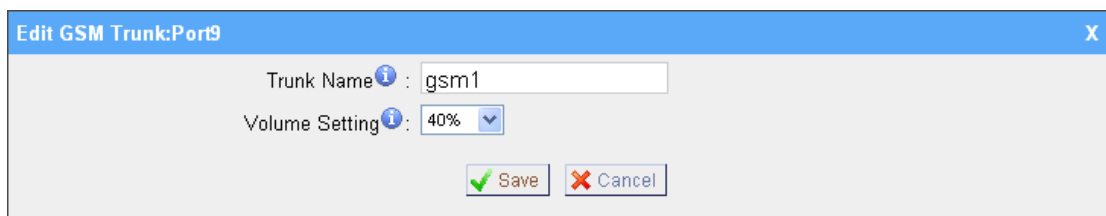


Figure 3-9

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: Minimalistic protocol to build a "network" with two or more PBX of different vendors.

•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 MPBX can dial this switch using overlap digits . If you need Direct

Dial-in (DDI; in German "Durchwahl") you should change this to yes, then MPBX 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().

•Facility Enable

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.

•Echocancel

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.

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.

Edit BRI Trunk - 7

Trunk Name: BriTrunk7

Signaling: BRI-CPE

Switch Type: euroisdn

PRI Dialplan: unknown

Reset Interval: never s

PRI Local Dialplan: unknown

Overlap Dial: no

PRI Indication: Inband

Enable Facility: Enabled

Nsf: none

Echo Cancellation: Off

Hide CallerID: No

CallerID Prefix

International Prefix:

National Prefix:

Local Prefix:

Private Prefix:

Unknown Prefix:

DOD Setting

Global DOD:

DOD:

Associated Extension: 500

Figure 3-10

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. Leave blank to specify no maximum.

•Outbound Proxy Server

A proxy that receives requests from a client , even though it may not be the server resolved by the Request-URI.

•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

Enable extension for SRTP.

•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.

Created New VoIP trunk

Type: SIP

Provider Name:

Hostname/IP: : 5060

Domain:

User Name:

Authorization Name:

Password:

From User:

Online Number ⁱ:

Maximum Channels ⁱ: 1

Enable Outbound Proxy Server

Transport: UDP ⁱ Enable SRTP ⁱ:

Caller ID ⁱ:

DOD Setting

DOD: Associated Extension: 500

Figure 3-11

2. Add IAX Trunk

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

•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. Leave blank 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.

Created New VoIP trunk X

Type: IAX

Provider Name:

Hostname/IP: : 4569

User Name:

Password:

Online Number *i*:

Maximum Channels *i*: 1

Caller ID *i*:

Save Cancel

Figure 3-12

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.

2) DOD Setting

•Global DOD

Global direct outward dialing number.

•DOD

Direct Outward Dialing Number.

•Bind Extension.

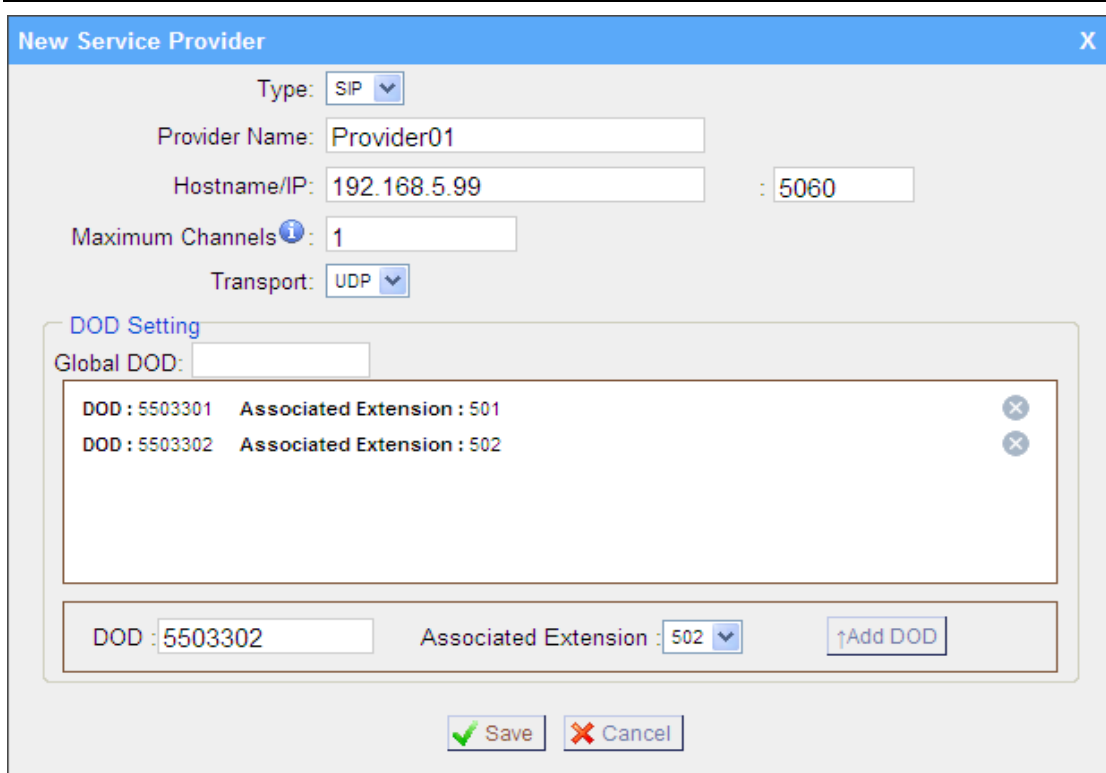


Figure 3-13

3.3.3 Outbound Routes

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

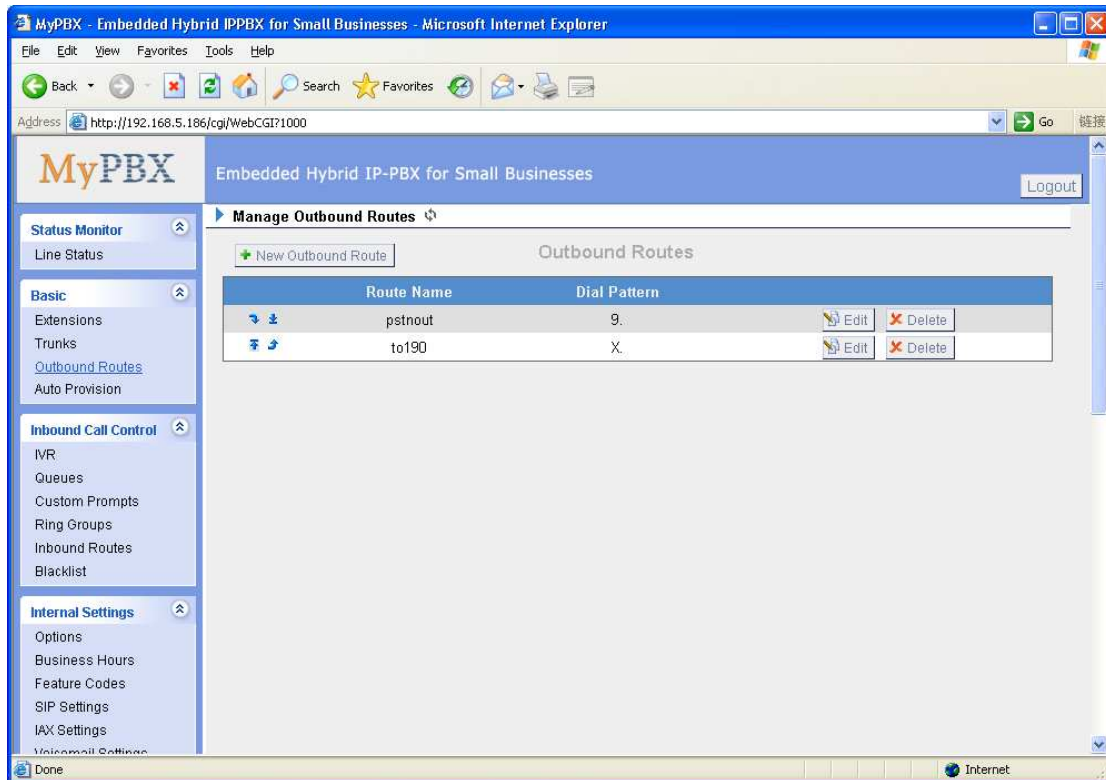


Figure 3-14

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: **1NXXNXXXXXX** 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.

•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 .

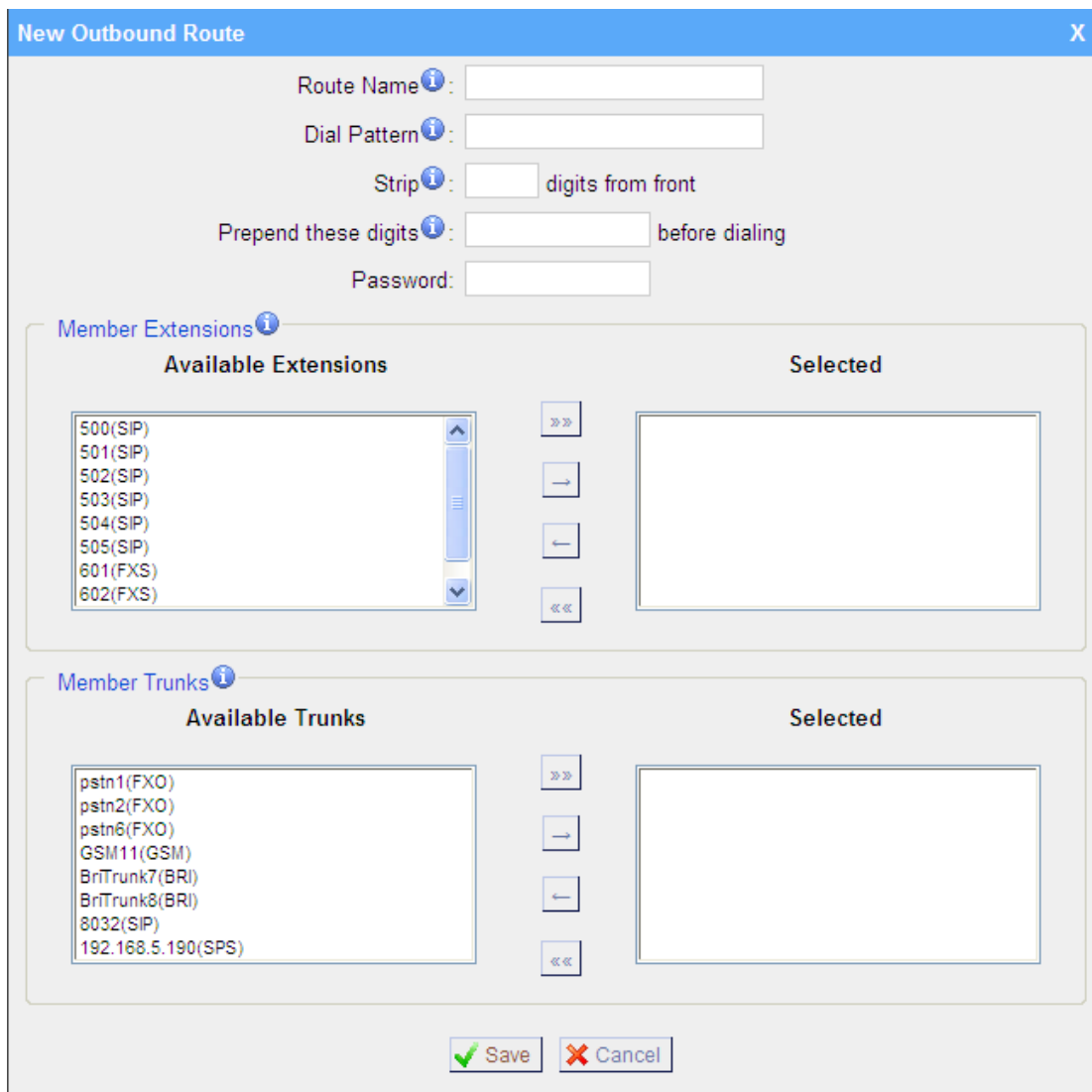


Figure 3-15

3.3.3.2 Delete Outbound Route

Click 'Delete' on selected trunk to delete it.

3.3.4 Auto Provision

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 Yealink IP Phone.

3.3.4.1 New Phone

Add New Phone have two modes,
Add New phone in webpage and Upload the IP Phone's configure file.

1. Add new phone via webpage

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

1) General

•MAC

IP Phone's MAC address.

•Name

Name of this 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

•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.

General			
Mac:	<input type="text"/>	Name:	<input type="text"/>
Call Waiting :	Enabled <input type="button" value="v"/>		
Key As Send :	# <input type="button" value="v"/>		

Line			
<input type="checkbox"/>	Line1	Extension: <input type="text"/>	Label: <input type="text"/>
<input type="checkbox"/>	Line2	Extension: <input type="text"/>	Label: <input type="text"/>
<input type="checkbox"/>	Line3	Extension: <input type="text"/>	Label: <input type="text"/>
<input type="checkbox"/>	Line4	Extension: <input type="text"/>	Label: <input type="text"/>
<input type="checkbox"/>	Line5	Extension: <input type="text"/>	Label: <input type="text"/>
<input type="checkbox"/>	Line6	Extension: <input type="text"/>	Label: <input type="text"/>

Figure 3-16

2. Upload the IP Phone's configure file to add new phone

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

Note: the file format must be .cfg

3.3.4.2 Scan Phone

Click 'Scan Yealink Phone', System will scan Yealink's IP phone in the local network, and list IP phone's MAC and IP address in this page.

Click 'MAC address', and fill in the corresponding information in the popup window. Like 3.3.4.1

3.4 Inbound Call Control

3.4.1 IVR

When there's an inbound call aims at Auto Attendant, MPBX 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.

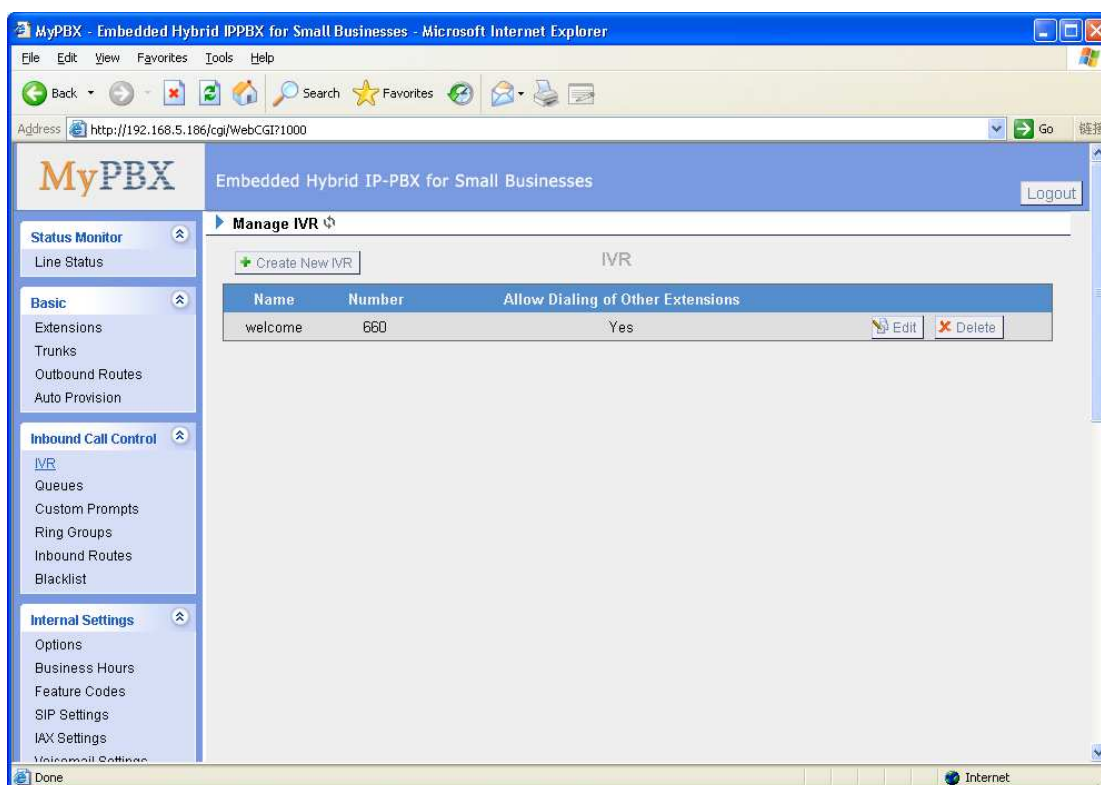


Figure 3-17

3.4.1.1 Create IVR

Click 'Create New IVR'.

1) General

•Number

MPBX 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 .

•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.

Number **i**: 660

Name **i**: welcome

Prompt **i**: default [Custom IVR Prompts](#)

Repeat Count: 3

WaitExten **i**: 3

i Allow Dialing Other Extensions

i KeyPress Events

Key	Action	Destination
0	Connect to Extension	User Extension -- 500
1	Connect to Extension	User Extension -- 501
2	Connect to RingGroup	RingGroup -- ringgroup_defe
3	No Action	
4	No Action	
5	No Action	
6	No Action	
7	No Action	
8	No Action	
9	No Action	
#	No Action	
*	No Action	
TimeOut i	Connect to Extension	User Extension -- 500
Invalid i	Hangup	

Figure 3-18

3.4.1.2 Delete IVR

Select the item that you would like to remove and press the 'Delete' button.

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 MPBX 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.

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
X

Queue Name ?:

Queue Number ?:

Queue Password ?:

Queue Agent Timeout ?:

Queue Max Wait Time ?:

Queue Ring Strategy ?:

Agents ?

Available Agents		Selected
<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> 500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) 505(SIP) 506(SIP) 507(SIP) </div>	<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>	<div style="border: 1px solid #ccc; height: 100px;"></div>

Caller Position Announcements

Announce Position ?:

Announce Hold Time ?:

Frequency ?:

Periodic Announcements

Prompt ?:

Frequency ?:

Events

Key:

Action:

Destination:

Failover-Destination

Action:

Destination:

Others

Music On Hold ?:

Leave When Empty ?:

Join Empty ?:

Agent Announcement ?:

Join Announcement ?:

Retry ?:

Wrap-up time ?:

Figure 3-19

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.

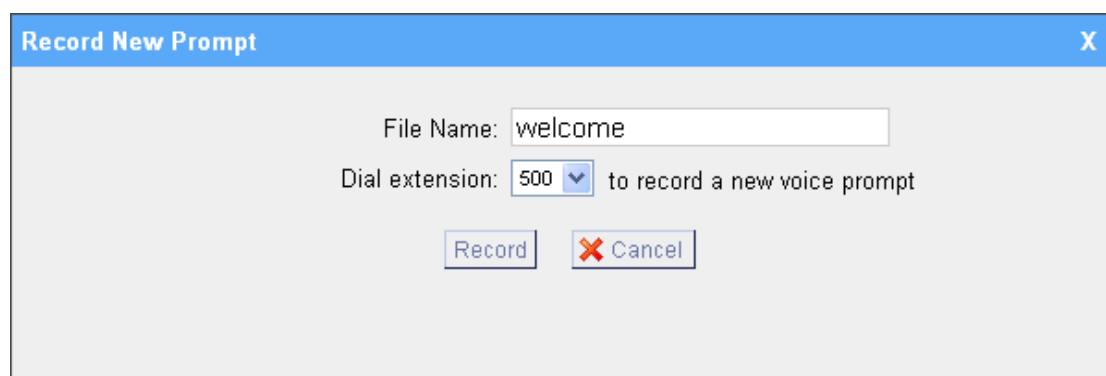


Figure 3-20

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.

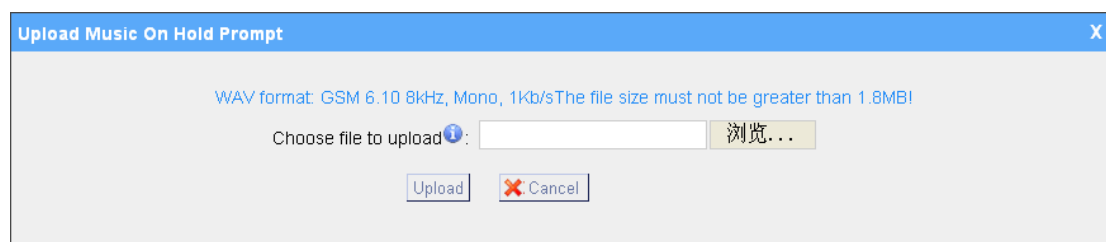


Figure 3-21

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.

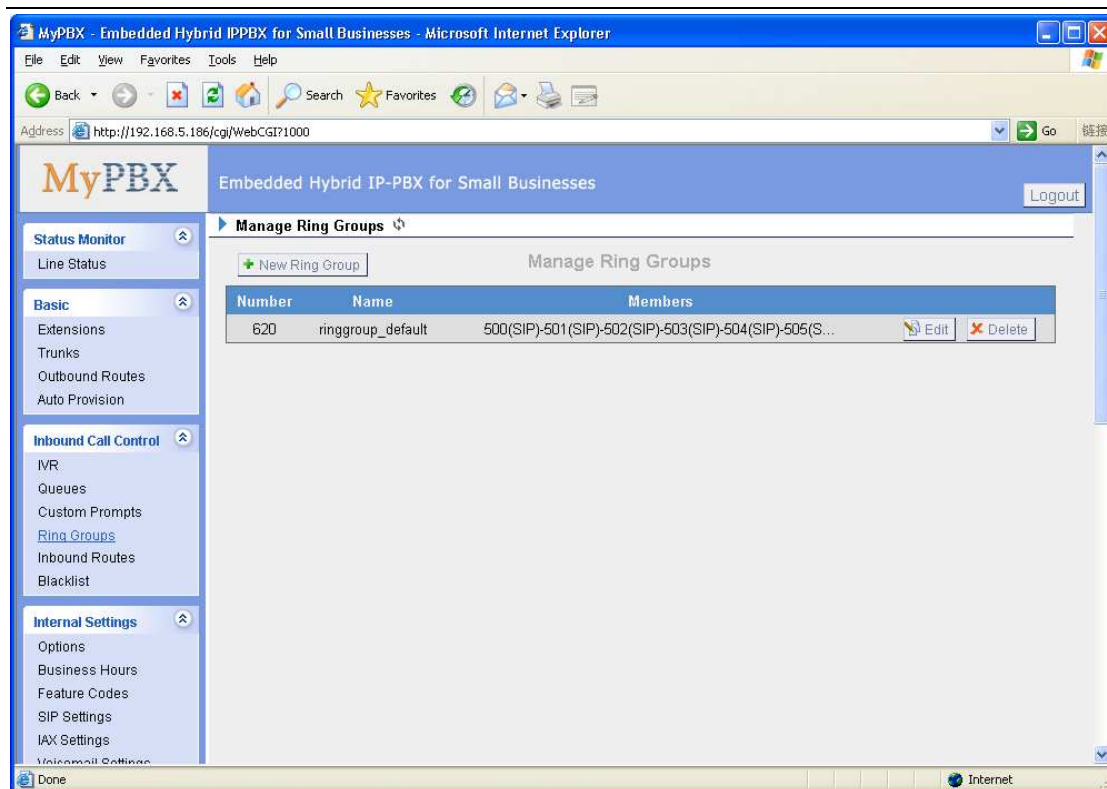


Figure 3-22

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.

New Ring Group X

Ring Group Name *i*: 625

Ring Group Number *i*: 625

Strategy *i*: Ring all simultaneously ▼

Seconds to ring each member *i*: 60

Ring Group members *i*

Available Extensions		Selected
5001(SIP)	>>	
5002(SIP)	→	
5003(SIP)	←	
5004(SIP)	<<	
5005(SIP)		
5006(SIP)		
5007(SIP)		
5008(SIP)		

Destination If No Answer:

Destination:

- End Call
- Extension: Extension -- 5001 ▼
- Voicemail: Voicemail -- 5001 ▼
- IVR: IVR -- welcome ▼
- RingGroup: RingGroup -- NationalService ▼
- Conference Room: Conference Room -- 640 ▼

Save Cancel

Figure 3-23

3.4.4.2 Delete Ring Group

Select the item you'd like to remove and press the 'Delete' button.

3.4.5 Inbound Routes

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

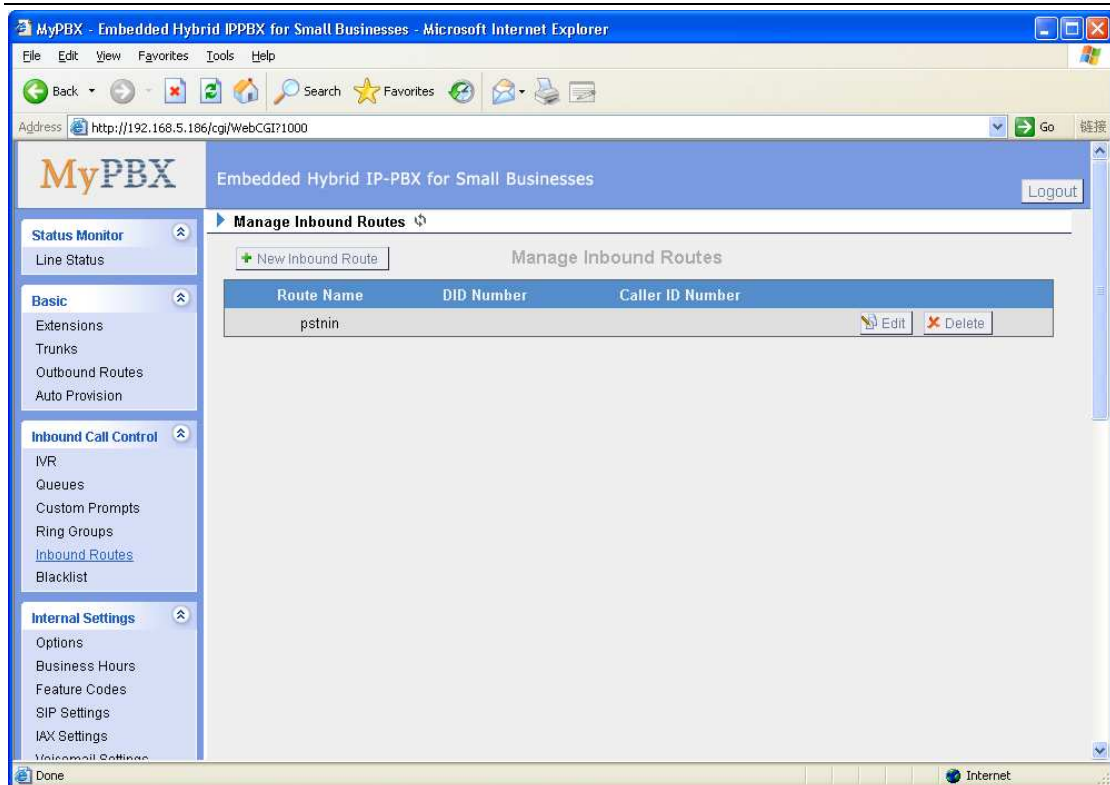


Figure 3-24

3.4.5.1 Create Inbound Route

Click 'New Inbound Route' to enter to the Manage Inbound Routes page.

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.

•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

MPBX 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.

How do I configure distinctive ring tones? Please see [APPENDIX E](#).

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.

3) During Office Hours

You can choose how to route 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.

Note: This function can only be used after the configuration of SIP Trunk in Service Provider.

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 MPBX 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.

4) Outside Office Hours

You can choose how to route incoming calls outside office hours.

create New Inbound Route
X

General

Route Name i :

DID Number i :

Extension i :

Caller ID Number i :

Distinctive Ringtone i :

Member Trunks i

Available Trunks		Selected
pstn1(FXO) pstn2(FXO) pstn6(FXO) GSM11(GSM) BriTrunk7(BRI) BriTrunk8(BRI) 8032(SIP) 192.168.5.190(SPS)	<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>	

During Office Hours

Destination:

<input checked="" type="radio"/> End Call	
<input type="radio"/> Extension	<input type="text" value="Extension -- 500"/>
<input type="radio"/> Voicemail	<input type="text" value="Voicemail -- 500"/>
<input type="radio"/> IVR	<input type="text" value="IVR -- welcome"/>
<input type="radio"/> RingGroup	<input type="text" value="RingGroup -- ringgroup_defi"/>
<input type="radio"/> Conference Room	<input type="text" value="Conference Room -- 640"/>
<input type="radio"/> DISA	<input type="text" value="DISA --"/>
<input type="radio"/> Queues	<input type="text" value="Queues --"/>
<input type="radio"/> Faxes i	<input type="text" value="Faxes -- 500"/>
<input type="radio"/> Outbound Routes i	<input type="text" value="Route Name -- pstnout"/>

Outside Office Hours

Destination:

<input checked="" type="radio"/> End Call	
<input type="radio"/> Extension	<input type="text" value="Extension -- 500"/>
<input type="radio"/> Voicemail	<input type="text" value="Voicemail -- 500"/>
<input type="radio"/> IVR	<input type="text" value="IVR -- welcome"/>
<input type="radio"/> RingGroup	<input type="text" value="RingGroup -- ringgroup_defi"/>
<input type="radio"/> Conference Room	<input type="text" value="Conference Room -- 640"/>
<input type="radio"/> DISA	<input type="text" value="DISA --"/>
<input type="radio"/> Queues	<input type="text" value="Queues --"/>
<input type="radio"/> Faxes i	<input type="text" value="Faxes -- 500"/>
<input type="radio"/> Outbound Routes i	<input type="text" value="Route Name -- pstnout"/>

Figure 3-25

3.4.5.2 Delete Routes

Select the item you would like to remove and click 'Delete' key.

3.4.6 Blacklist

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

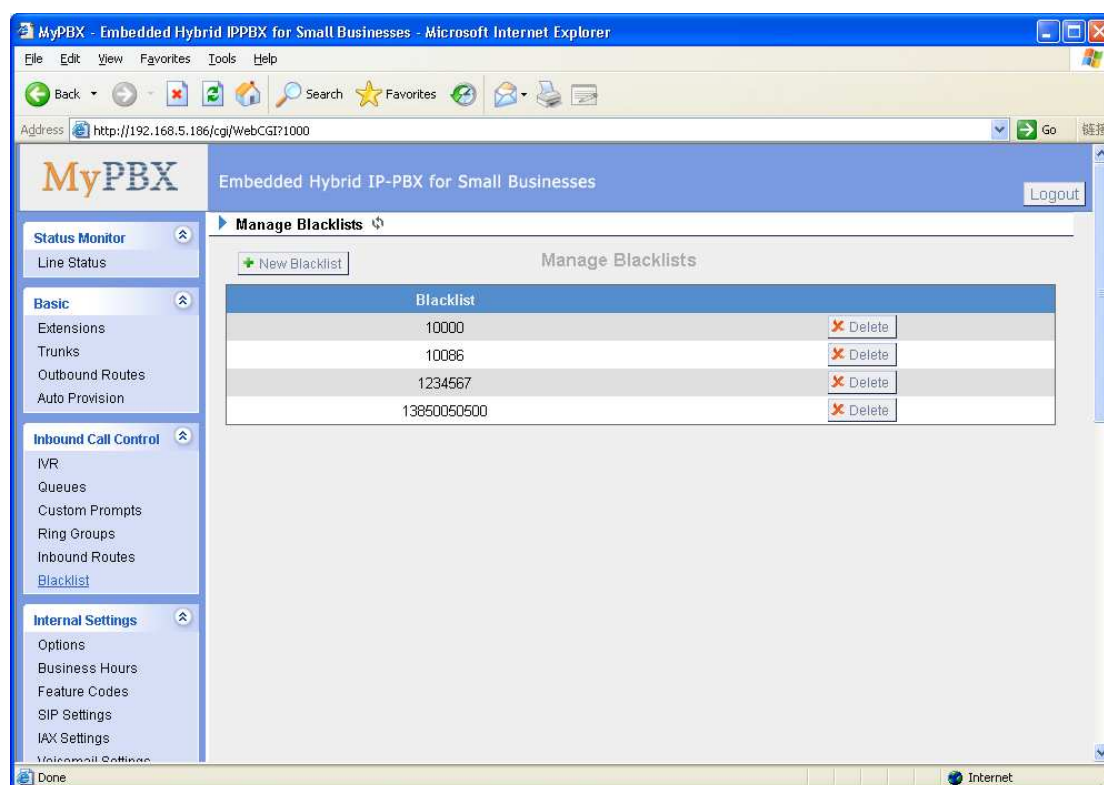


Figure 3-26

3.4.6.1 Create Blacklist

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

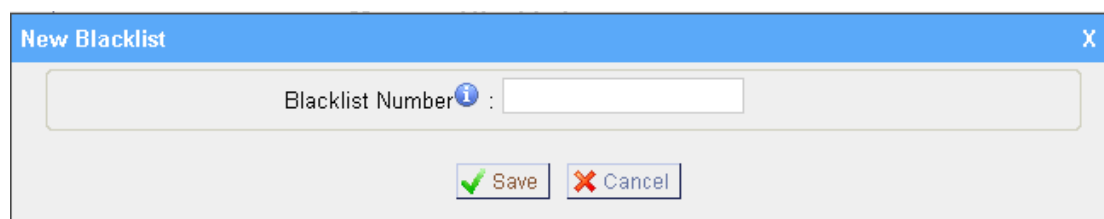


Figure 3-27

•Blacklist number

Enter the number you would like to block.

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.

•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.

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

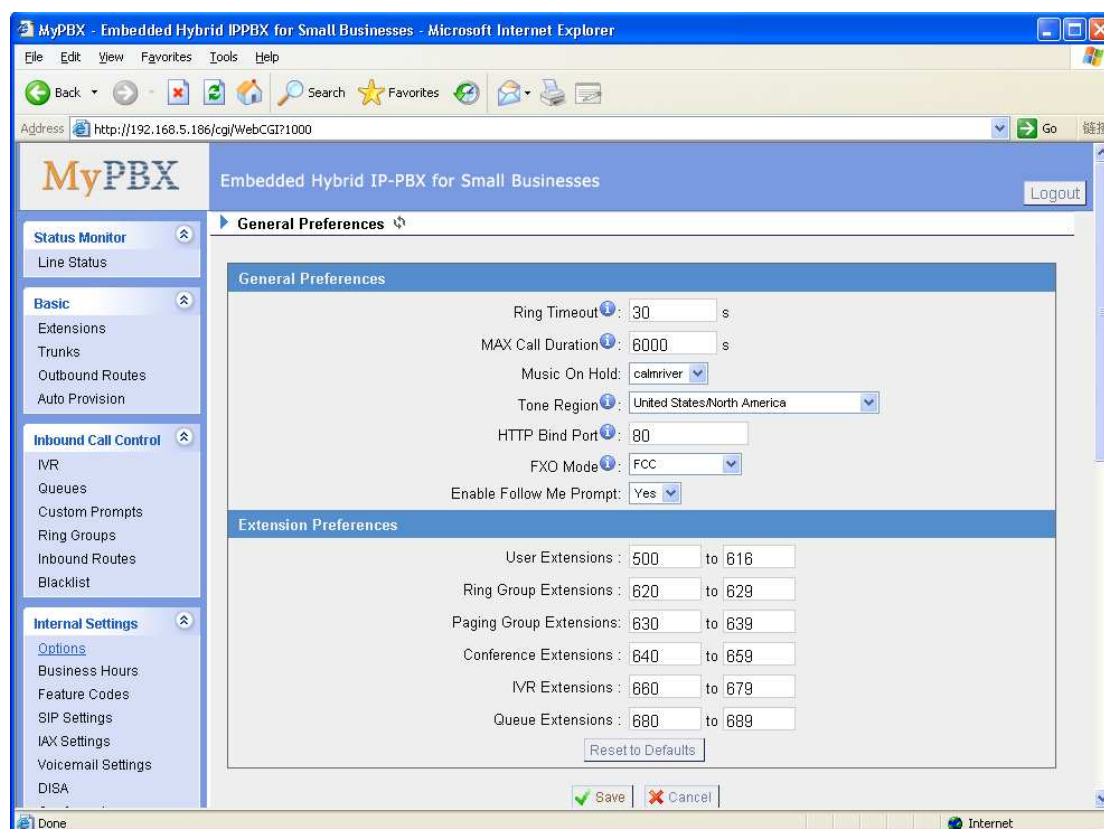


Figure 3-28

3.5.2 Business Hours

1) General

•Enable or Disable Business Hours

2) Others

•Enable Office Closed Timing

By dialing *89 (*89 is default setting) on an extension will force the office time closed for the device whatever the general setting is.

-Disable Office closed timing

By dialing *089 (*089 is default setting) on an extension will disable the Office Closed Timing.

3) Define Office Hours

You can setup the business hours here.



Figure 3-29

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).

-Attended Transfer

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

•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.

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)

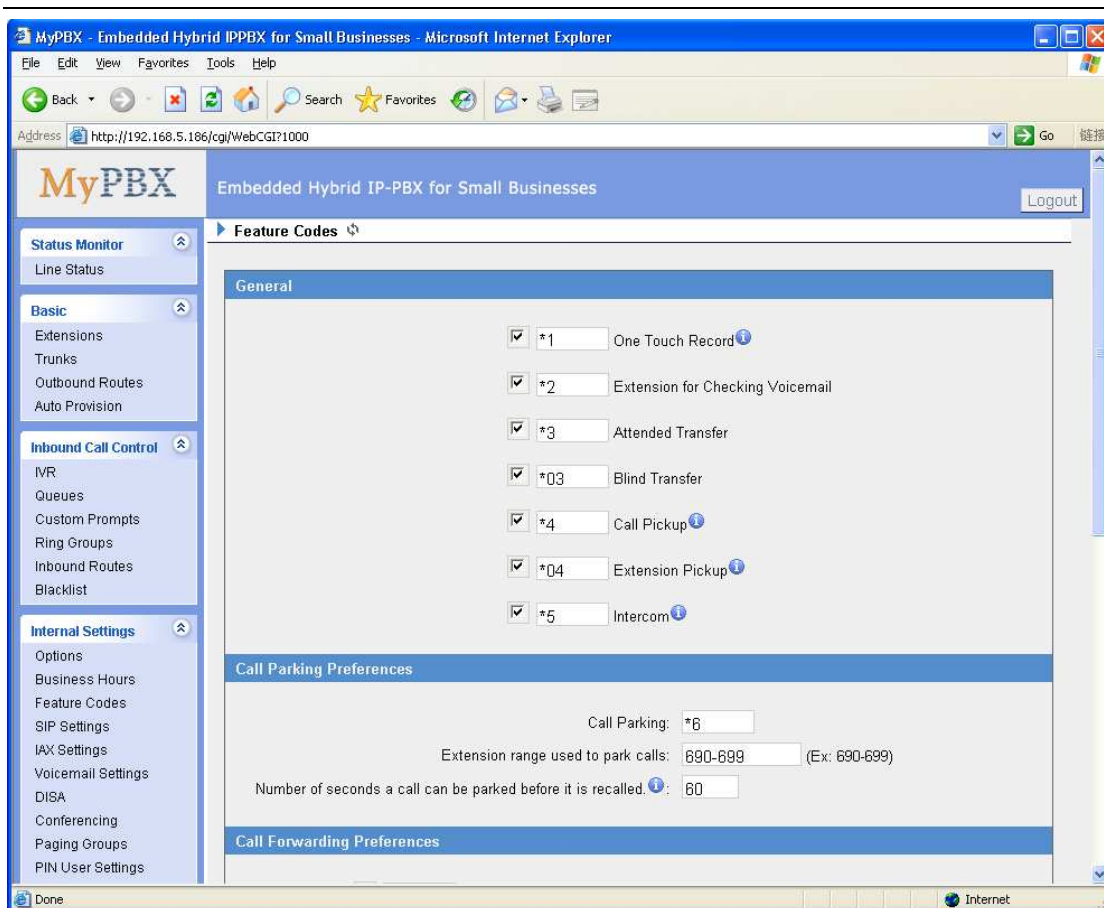


Figure 3-30

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.

•Video Support

Support for SIP video or no. Default is yes.

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 streams 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.

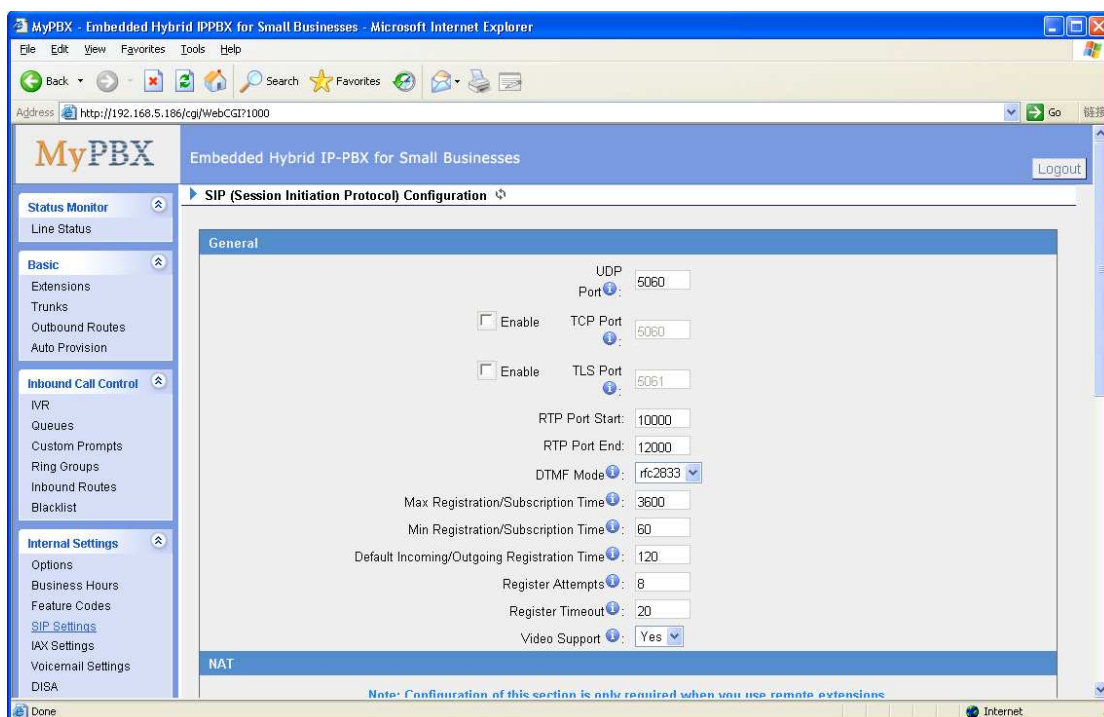


Figure 3-32

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.

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.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.

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

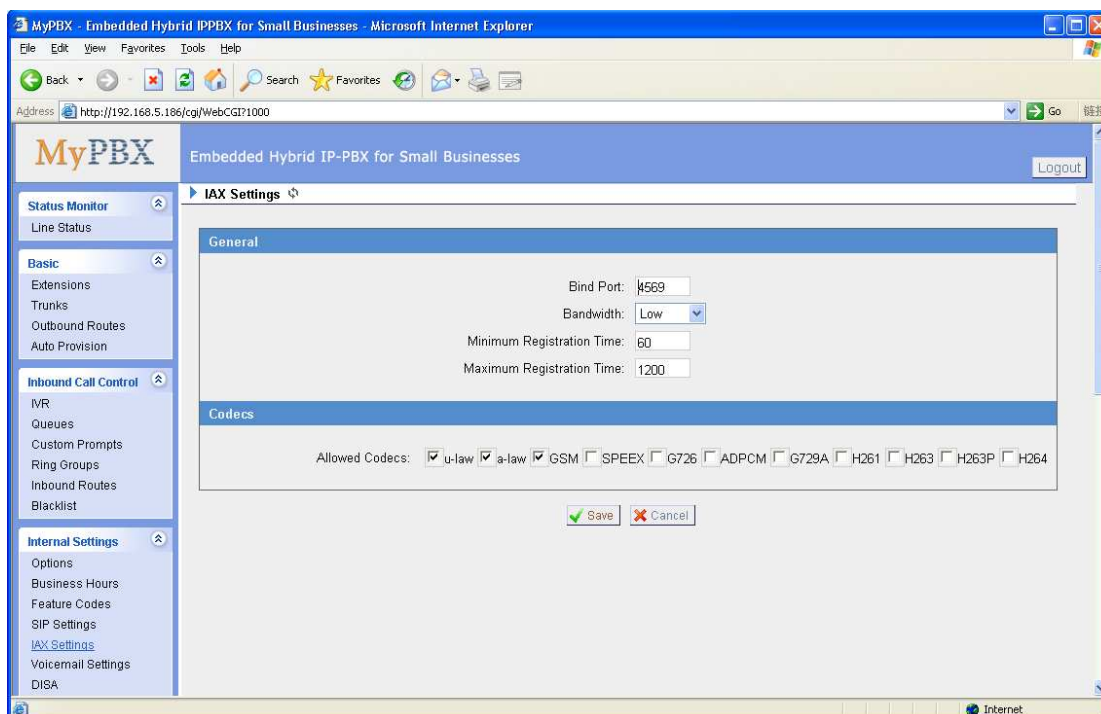


Figure 3-33

3.5.6 Voicemail Settings

1) General Voicemail Settings

•Max Messages per Folder

Sets 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.

•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 MPBX 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 MPBX 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.

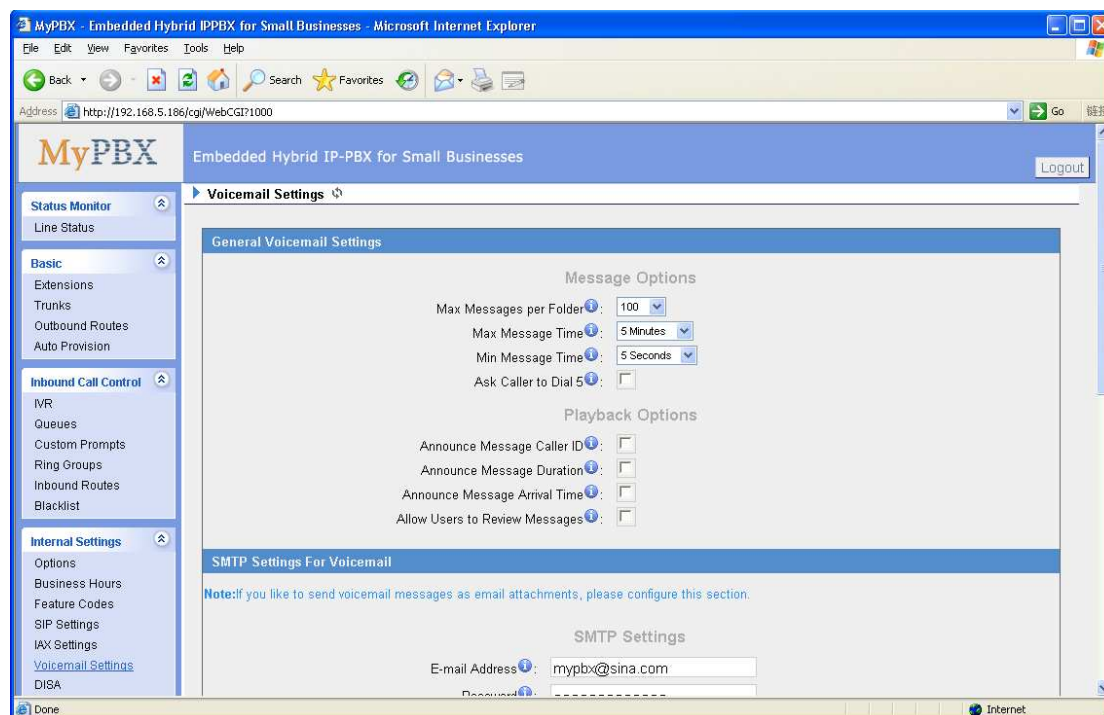


Figure 3-34

3.5.7 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.

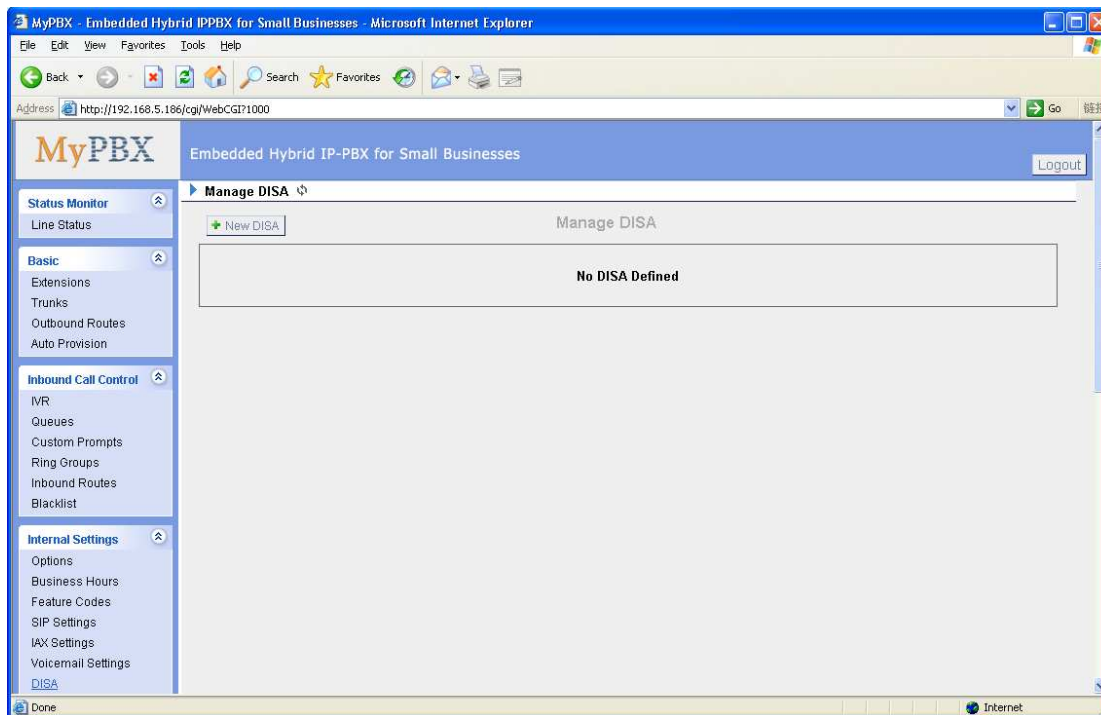


Figure 3-35

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

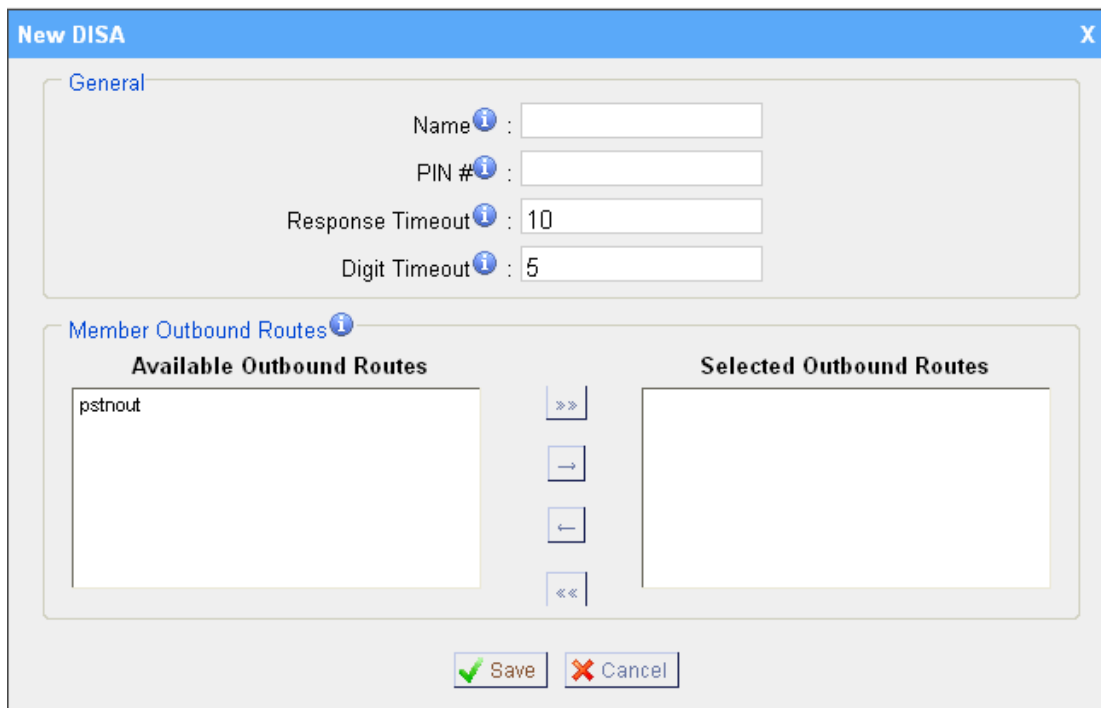


Figure 3-36

- 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 .

3.5.8 Conferencing

Conference Calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings.

•Extension

This is the number dialed to reach this Conference Room.

•Pin #

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

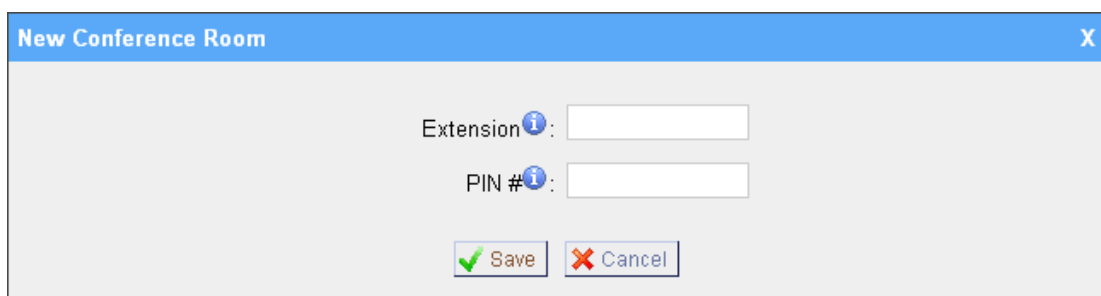


Figure 3-37

3.5.9 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.

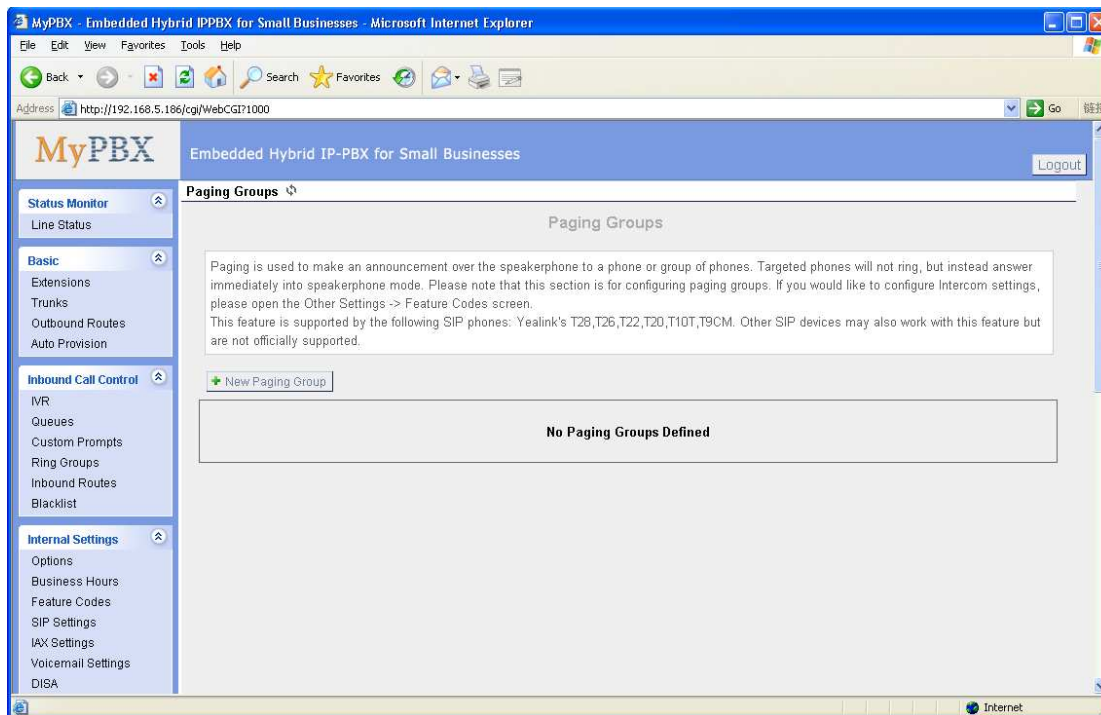


Figure 3-38

-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.

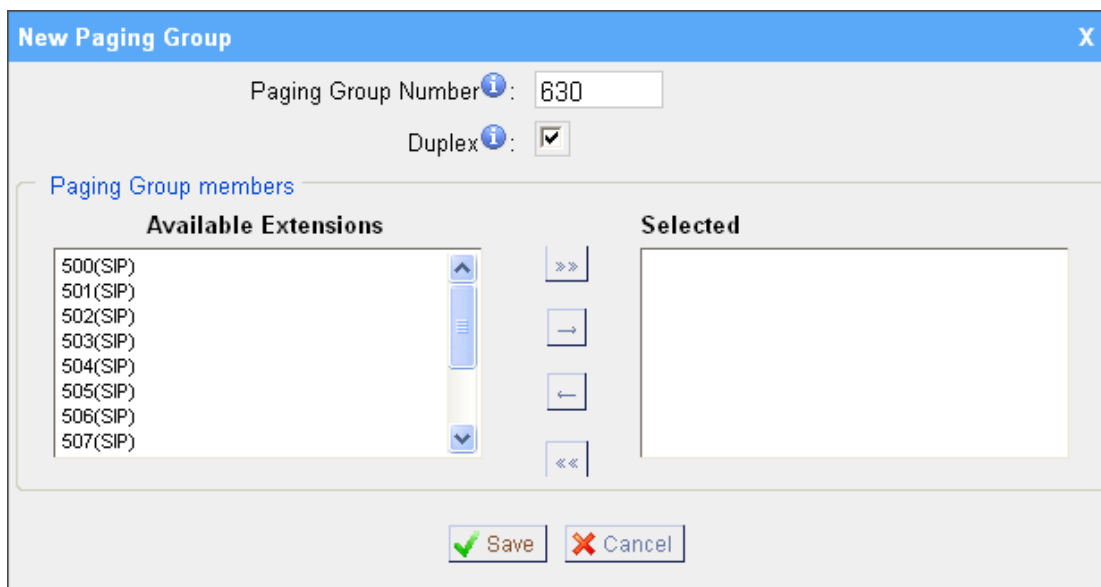


Figure 3-39

3.5.10 PIN User

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

MPBX 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.

- When a PIN is entered while making a call or during a call, that PIN is included in the call records output by the system.
- 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.

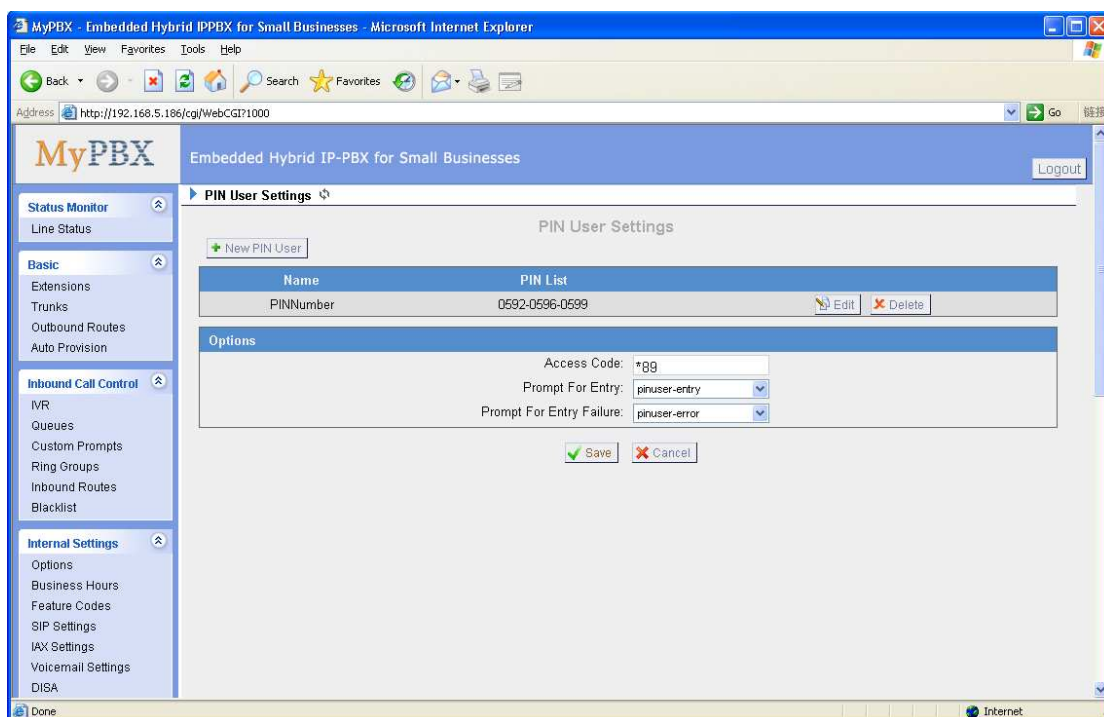


Figure 3-40

3.5.11 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.

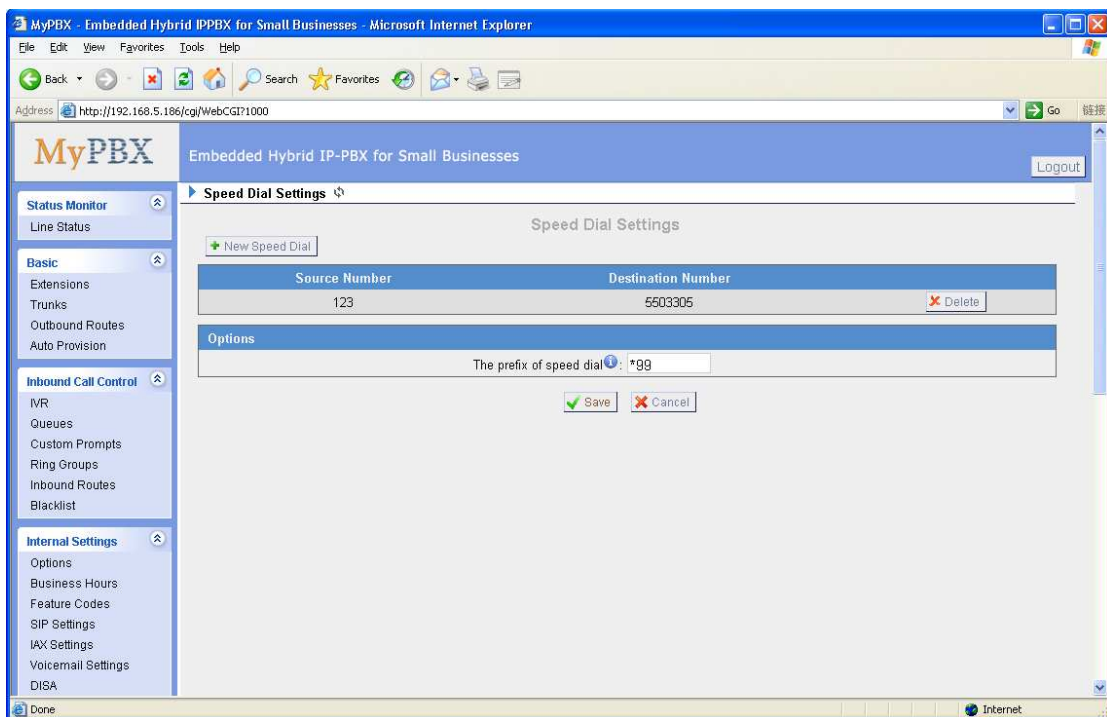


Figure 3-41

3.5.12 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.

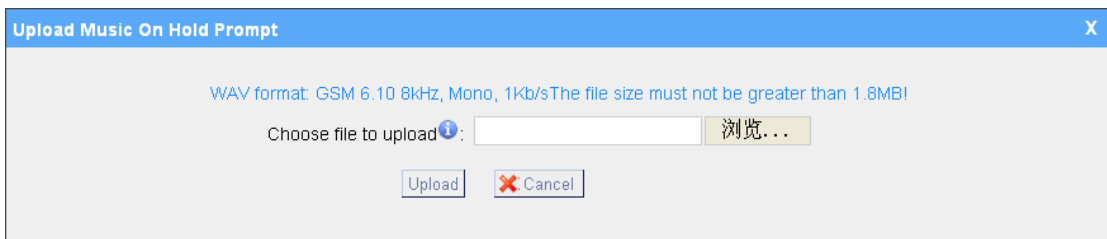


Figure 3-42

3.6 Network Settings

3.6.1 LAN Settings

-DHCP

If this option is set, MPBX 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 MPBX.

•IP Address

Set the IP Address for MPBX.

•Subnet Mask

Set the subnet mask for MPBX.

•Gateway

Set the gateway for MPBX.

•Primary DNS

Set the primary DNS for MPBX.

•Secondary DNS

Set the secondary DNS for MPBX.

•IP Address2

Set the second IP Address for MPBX.

•Subnet Mask2

Set the second subnet mask for MPBX.

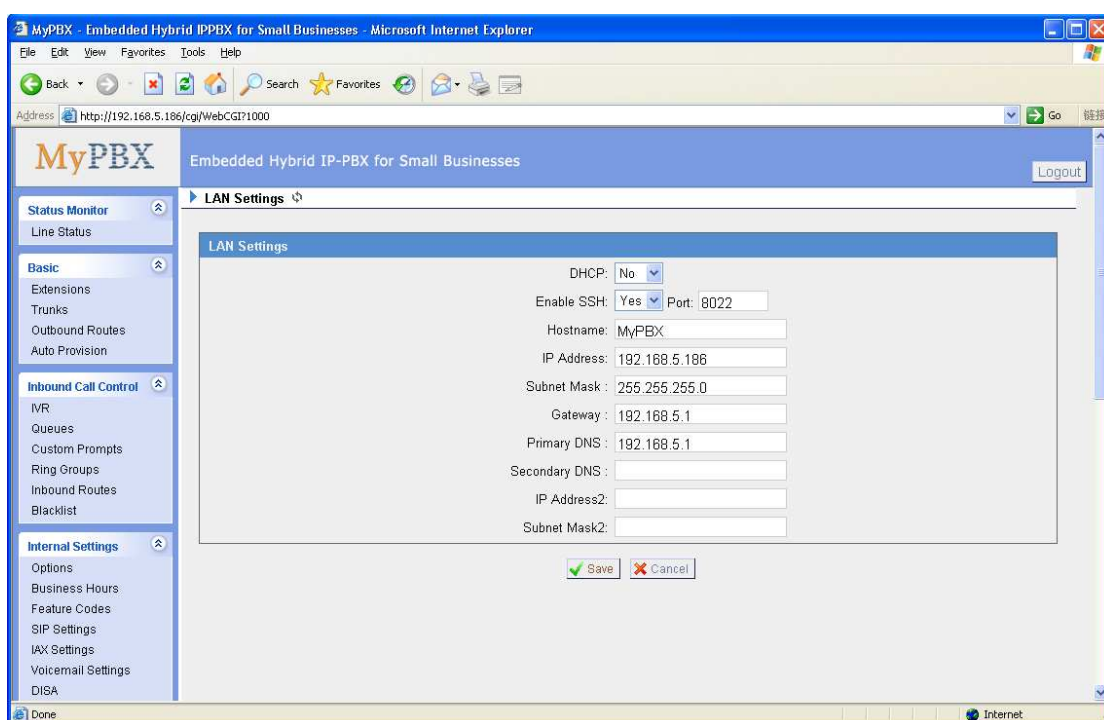


Figure 3-43

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).

3.6.3 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.

1) Enable Firewall

Enable the firewall to protect the device.

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.

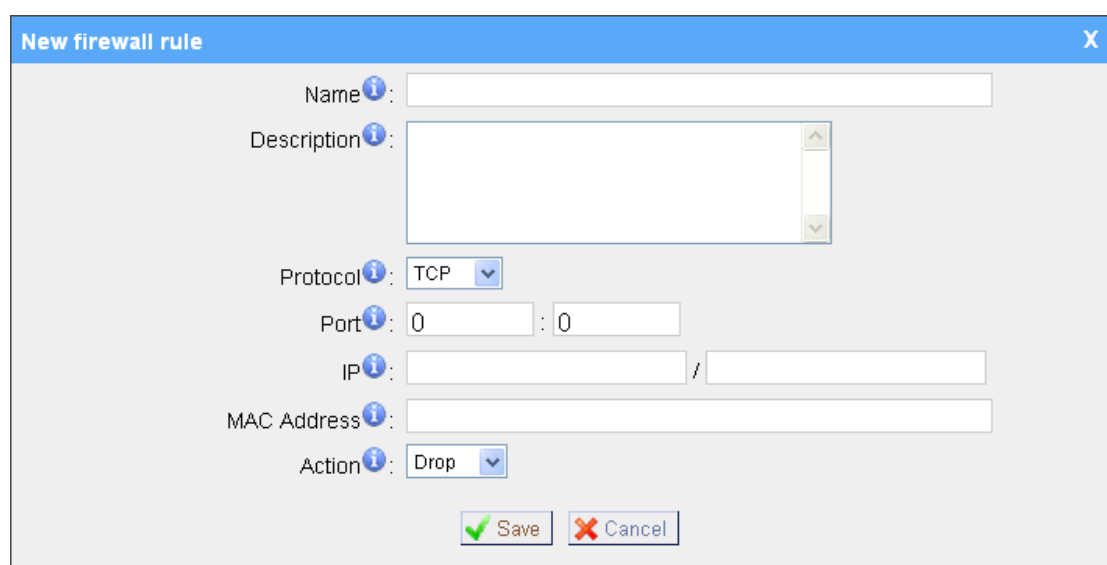


Figure 3-44

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.

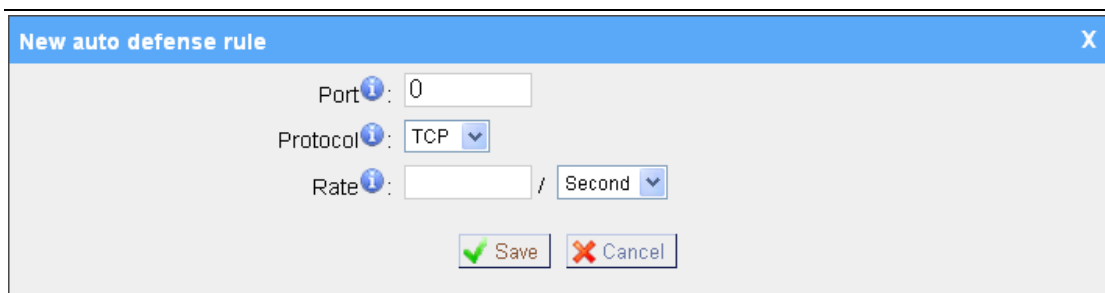


Figure 3-45

4) SIP Defense

•Port

The port used for SIP protocol.

•Protocol

Choose the protocol need to be protect, etc: UDP.

•SIP Packets

The SIP packets allowed in specific time interval .

•Time Interval

The time interval to receive SIP packets .

For example, SIP packets 90, time interval 60 means 90 SIP packets are allowed in 60 seconds.

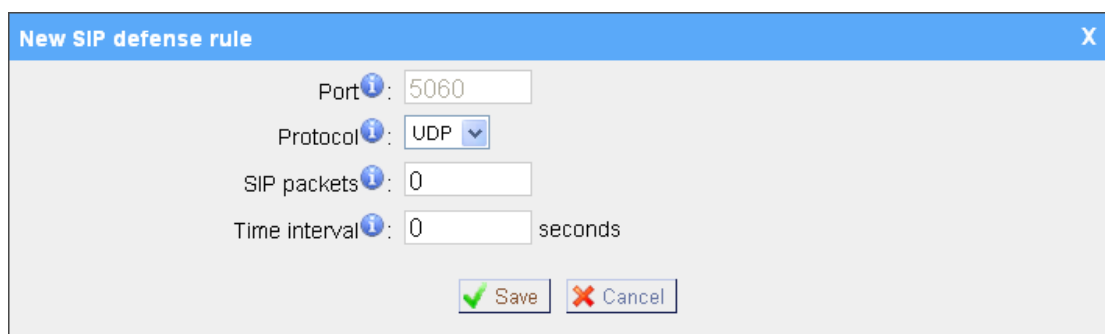


Figure 3-46

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.

3.6.4 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.

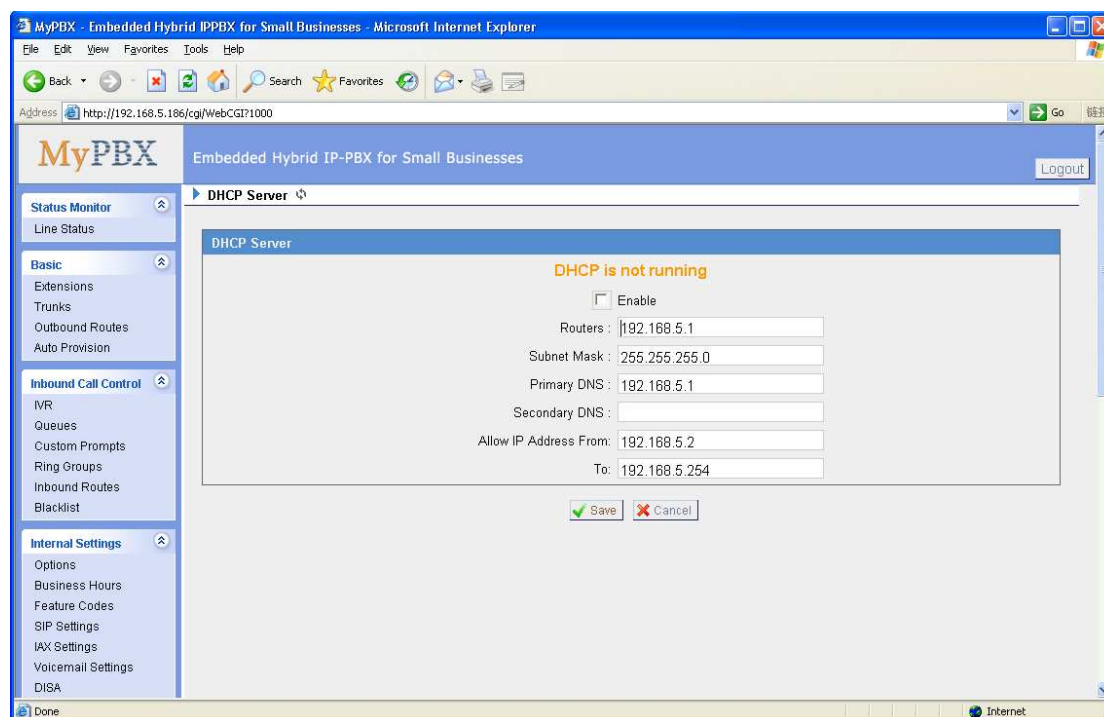


Figure 3-47

3.6.5 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 MPBX VLAN over Lan.

•VLAN Subnet Mask

Set the Subnet Mask for MPBX VLAN over Lan.

•Default Gateway

Set the Default Gateway for MPBX 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 MPBX VLAN over Lan.

•VLAN Subnet Mask

Set the Subnet Mask for MPBX VLAN over Lan.

•Default Gateway

Set the Default Gateway for MPBX 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 MPBX VLAN over Wan.

•VLAN Subnet Mask

Set the Subnet Mask for MPBX VLAN over Wan.

•Default Gateway

Set the Default Gateway for MPBX 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 MPBX VLAN over Wan.

•VLAN Subnet Mask

Set the Subnet Mask for MPBX VLAN over Wan.

•Default Gateway

Set the Default Gateway for MPBX VLAN over Wan.

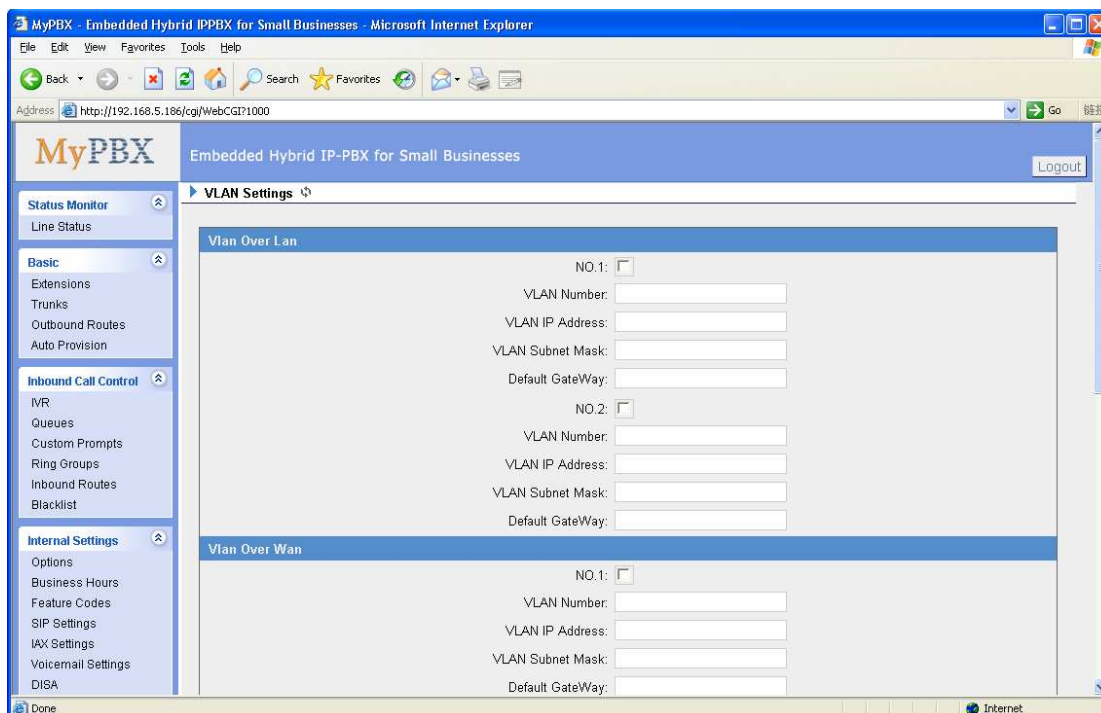


Figure 3-48

3.6.6 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

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, MPBX will move all qualified files to a Windows PC every 30 minutes. For the voicemail files and recording files, they must be created before Auto-Backup has been configured.

How to configure External Storage, please see [APPENDIX B How to Configure Autobackup](#).

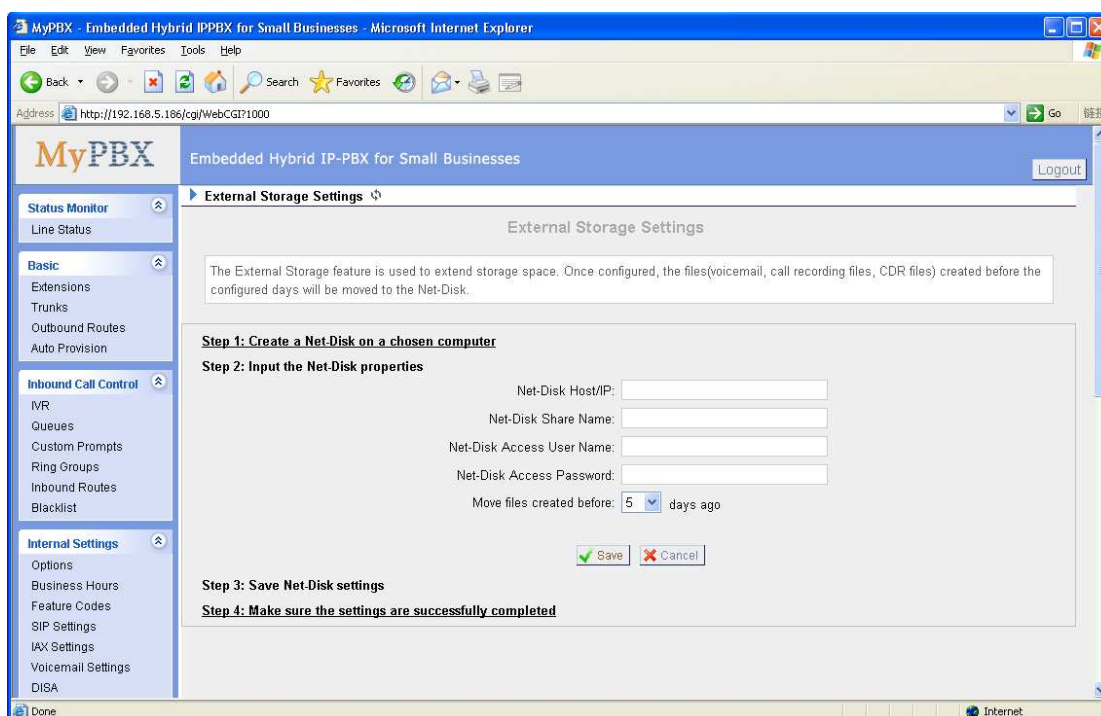


Figure 3-49

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.

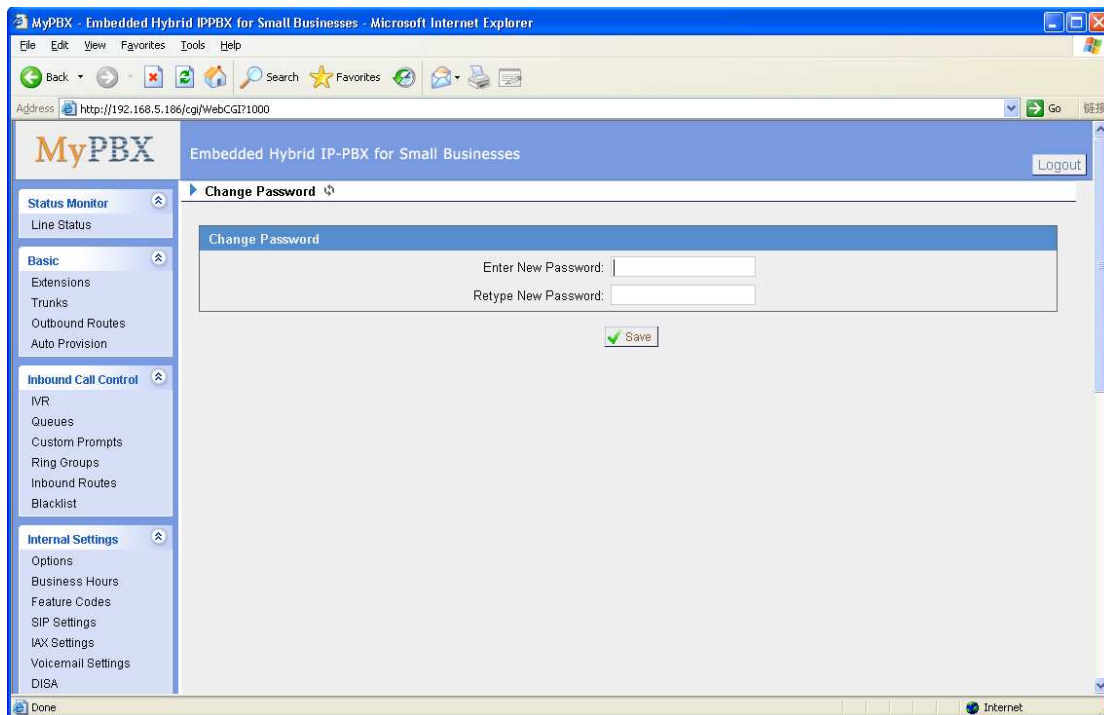


Figure 3-50

3.7.3 Date and Time

Set the date and time for MPBX.

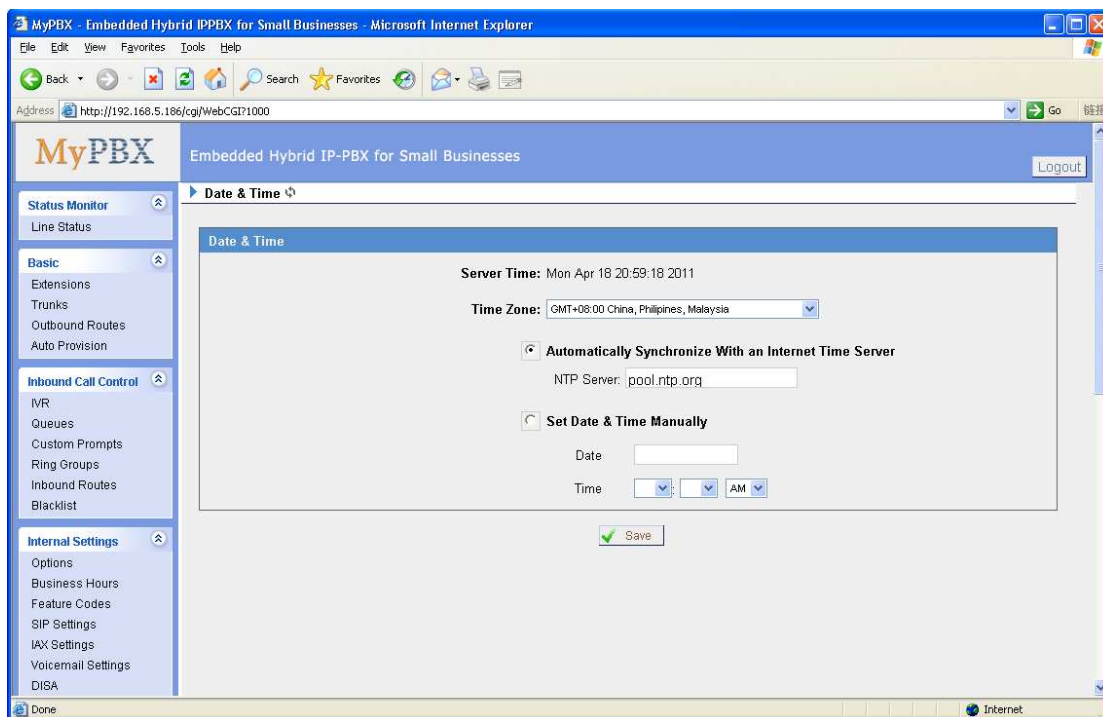


Figure 3-51

3.7.4 Backup and Restore

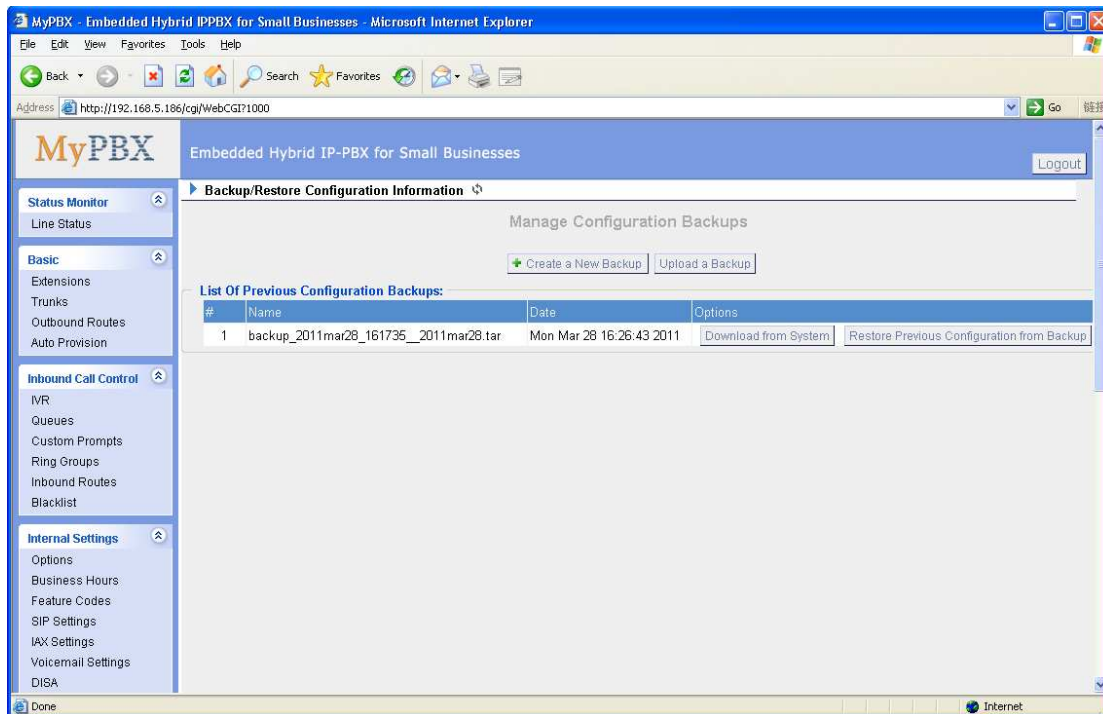


Figure 3-52

3.7.5 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.

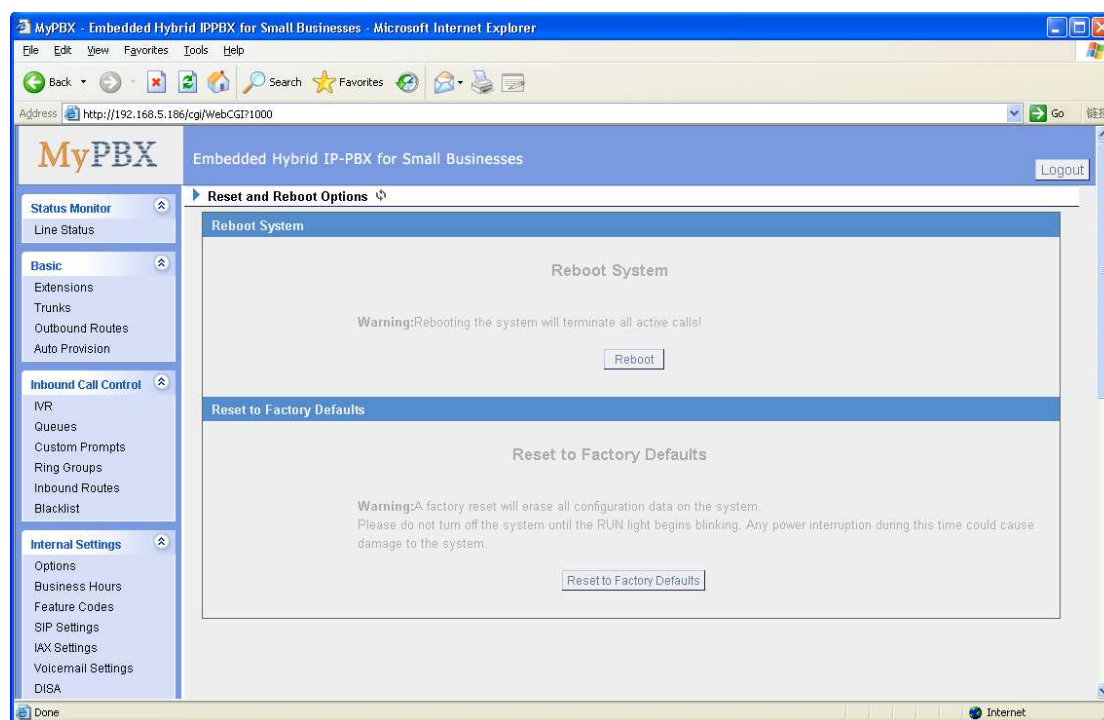


Figure 3-53

3.7.6 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/MPBX/>

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.

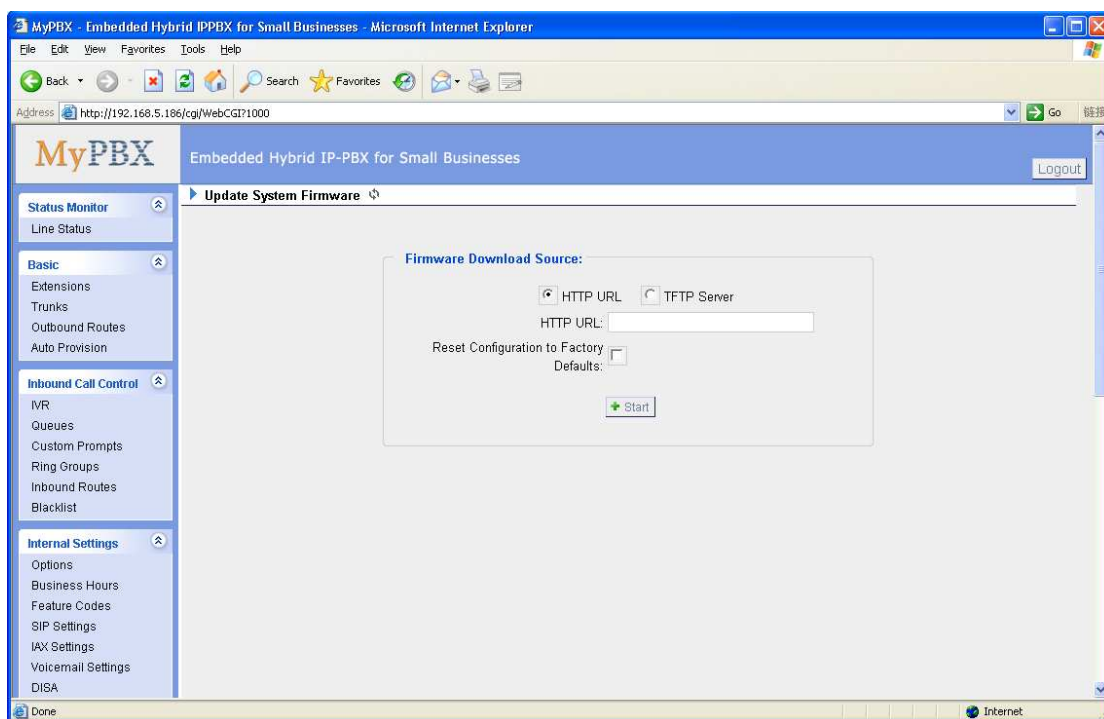


Figure 3-54

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 call time, call type and call duration and can also export CDR data to a CSV file.

MyPBX - Embedded Hybrid IPPBX for Small Businesses - Microsoft Internet Explorer

Address: http://192.168.5.186/cgi/WebCGI?1000

MyPBX Embedded Hybrid IP-PBX for Small Businesses

Status Monitor
Line Status

Basic
Extensions
Trunks
Outbound Routes
Auto Provision

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
DISA

CDR Viewer
Viewing 1-25 (most recent first)
Call Log File: Master.csv
Download Selected CDR
Delete Selected CDR
Delete All CDR

ID	Source	Destination	Start time	End Time	Duration	Billable Duration	Disposition	Communication Type
1	501	*081	2011-04-14 19:16:13	2011-04-14 19:16:15	2	2	ANSWERED	Internal
2	402	501	2011-04-14 17:55:28	2011-04-14 17:55:33	5	3	ANSWERED	Inbound
3	501	*081	2011-04-14 17:55:16	2011-04-14 17:55:19	3	3	ANSWERED	Internal
4	402	501	2011-04-14 17:54:49	2011-04-14 17:55:05	16	13	ANSWERED	Inbound
5	501	*81	2011-04-14 17:55:00	2011-04-14 17:55:02	2	2	ANSWERED	Internal
6	402	501	2011-04-14 17:54:23	2011-04-14 17:54:39	16	13	ANSWERED	Inbound
7	501	500	2011-04-14 17:52:57	2011-04-14 17:53:01	4	4	ANSWERED	Internal
8	402	500	2011-04-14 17:52:06	2011-04-14 17:52:19	13	12	ANSWERED	Inbound
9	402	500	2011-04-14 17:51:01	2011-04-14 17:51:13	12	11	ANSWERED	Inbound
10	402	500	2011-04-14 01:47:39	2011-04-14 01:47:45	6	3	ANSWERED	Inbound
11	402	501	2011-04-14 01:46:07	2011-04-14 01:46:14	7	5	ANSWERED	Inbound
12	Unknown	501	2011-04-14 01:44:37	2011-04-14 01:44:52	15	13	ANSWERED	Inbound
13	Unknown	501	2011-04-14 01:32:32	2011-04-14 01:32:40	8	0	NO ANSWER	Inbound
14	501	*081	2011-04-14 01:32:23	2011-04-14 01:32:26	3	3	ANSWERED	Internal
15	500	*81	2011-04-14 01:31:04	2011-04-14 01:31:07	3	3	ANSWERED	Internal
16	Unknown	500	2011-04-14 01:30:45	2011-04-14 01:31:02	17	14	ANSWERED	Inbound
17	Unknown	501	2011-04-14 01:22:59	2011-04-14 01:23:19	20	0	NO ANSWER	Inbound

Figure 3-55

3.8.2 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 .

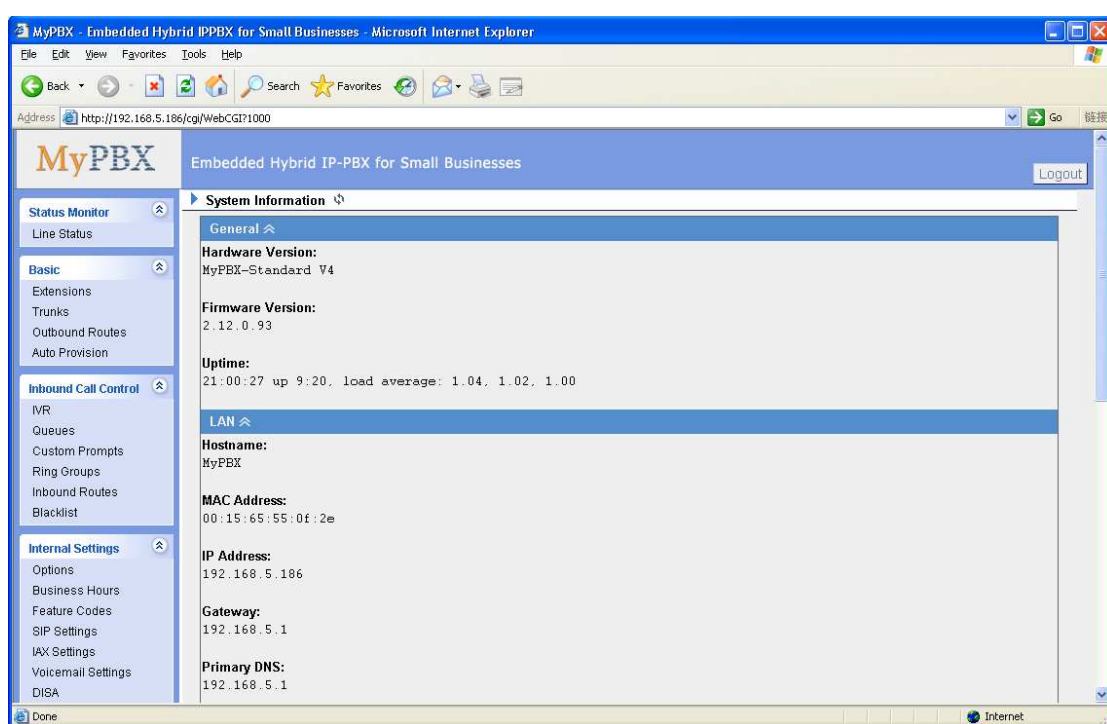


Figure 3-56

4. Access MRI

MRI (MPBX 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.

Edit Extension - 500 X

General

Type: SIP Extension: 500 Password: pincode500
 Name: 500 Caller ID: 500 Transport: UDP

Voicemail

Enable Voicemail Voicemail Access PIN #: 500

Mail Setting

Enable Send Voicemail
 Email Address:

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

VoIP Settings

NAT: DTMF Mode: RFC2833 Enable SRTP:

IP Restriction

Enable IP Restriction

Permitted 'IP address/Subnet mask' 1:
 Permitted 'IP address/Subnet mask' 2:
 Permitted 'IP address/Subnet mask' 3:
 Permitted 'IP address/Subnet mask' 4:

Group

Pickup Group:

Follow me

Follow me: Always No answer When Busy
 Transfer to: Voicemail Number

Other Options

Call Waiting User Web Interface Ring Out: 30

Figure 4-1

4.2 User login

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

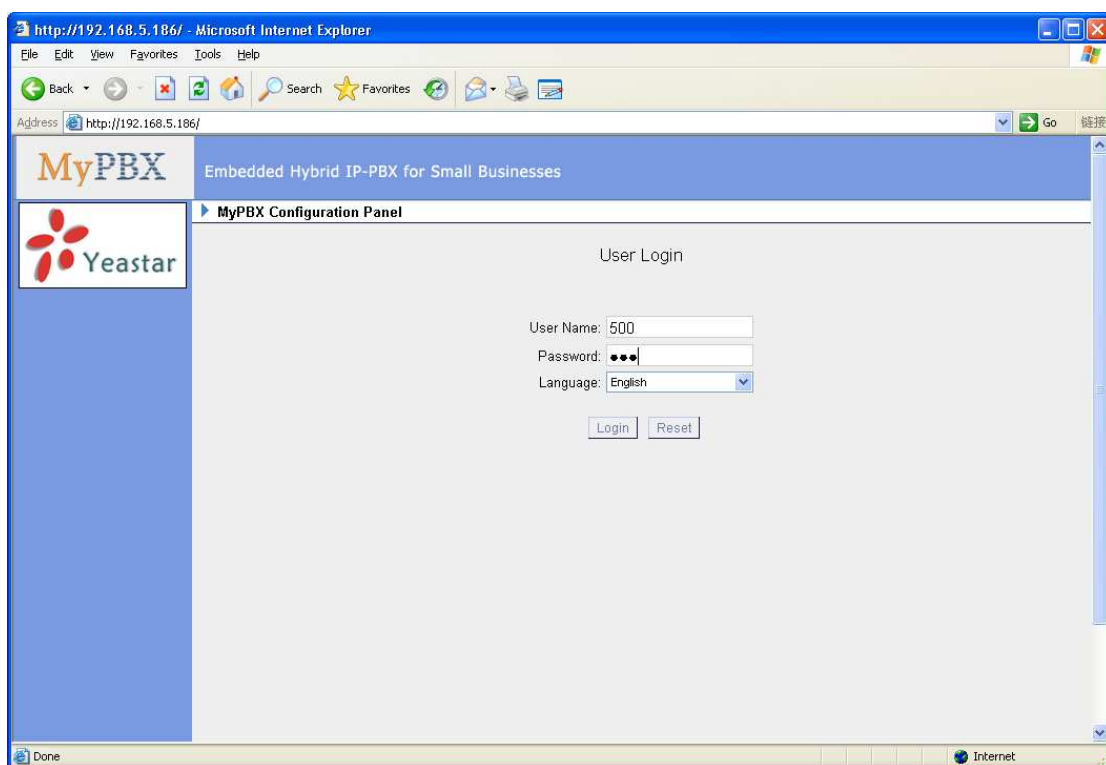


Figure 4-2

4.3 Voicemail

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

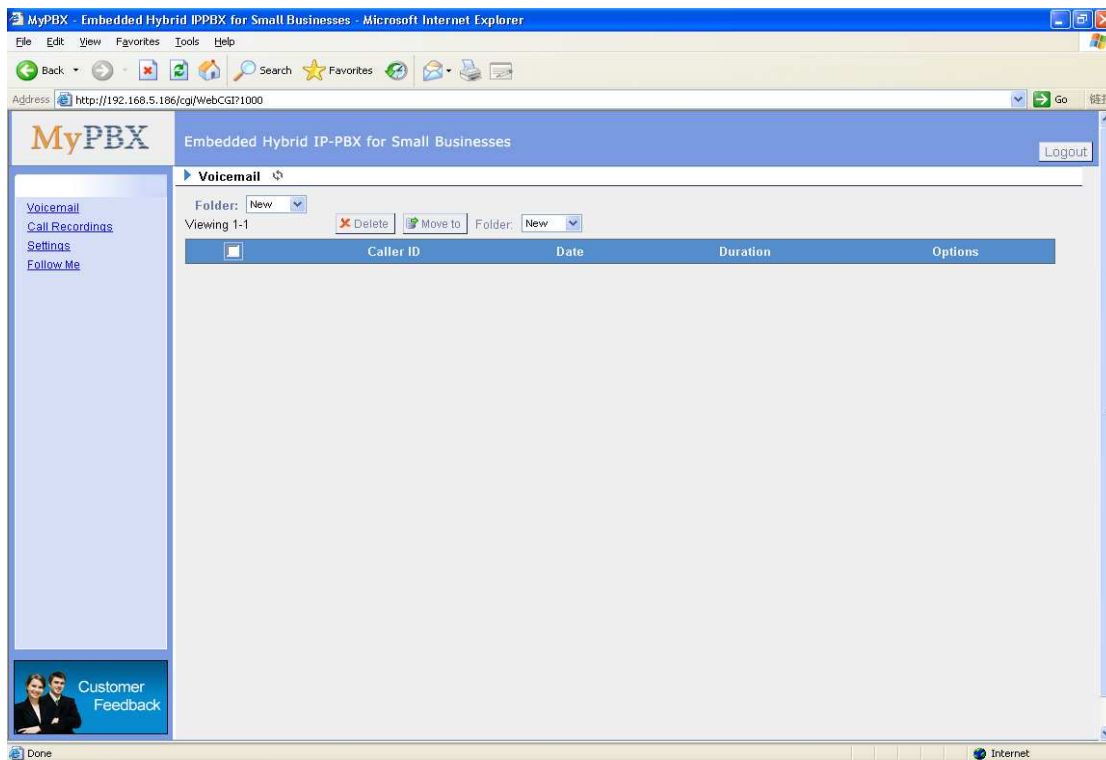


Figure 4-3

4.4 Record

Users can play, delete and download recorded files here.

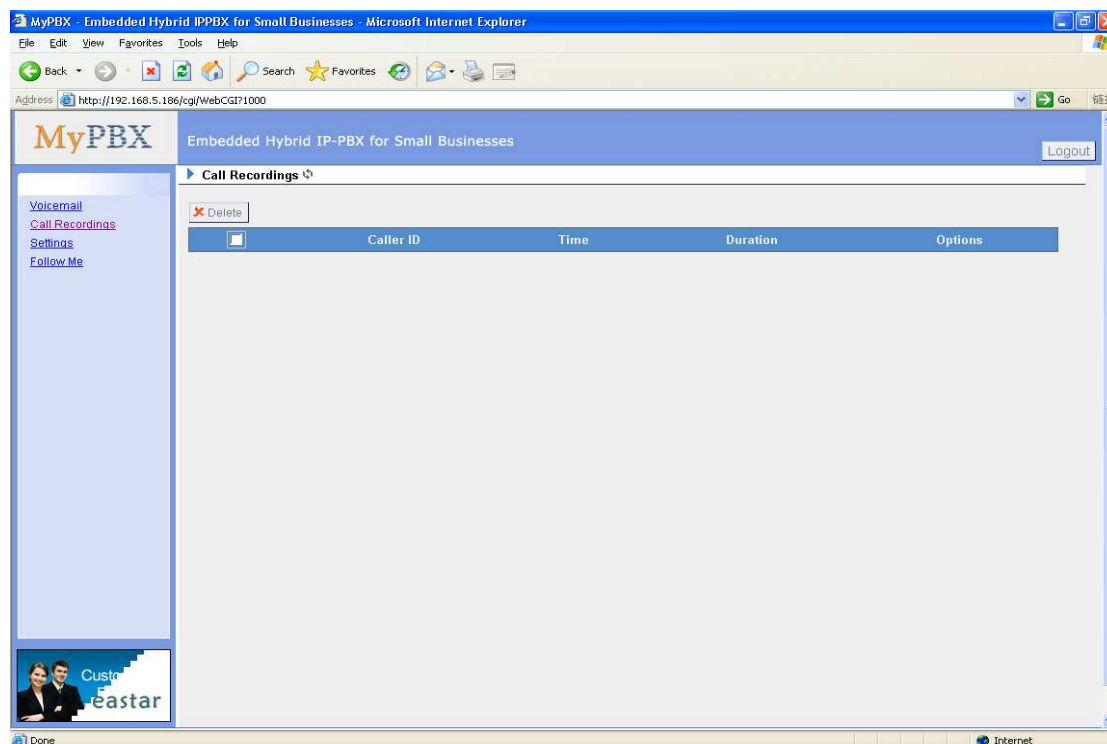


Figure 4-4

4.5 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.

•**Ring Out**

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

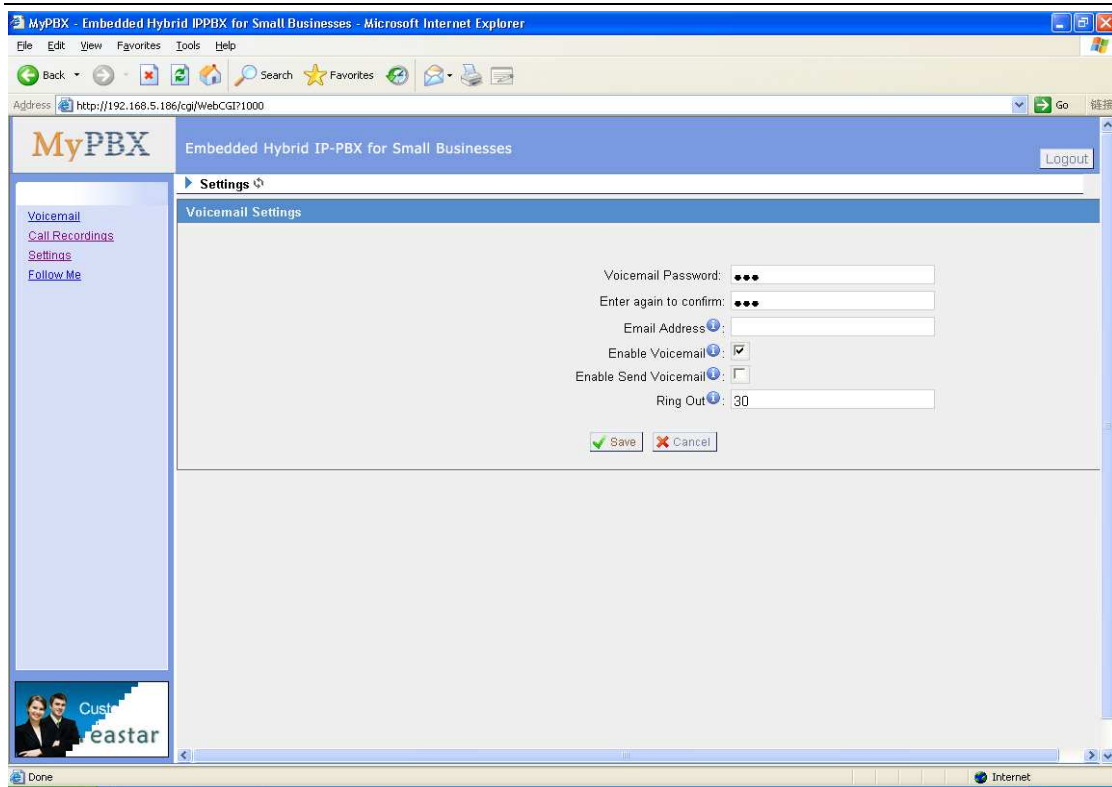


Figure 4-5

4.6 Follow me

Set the mode of call forwarding when there is an incoming call to this extension.

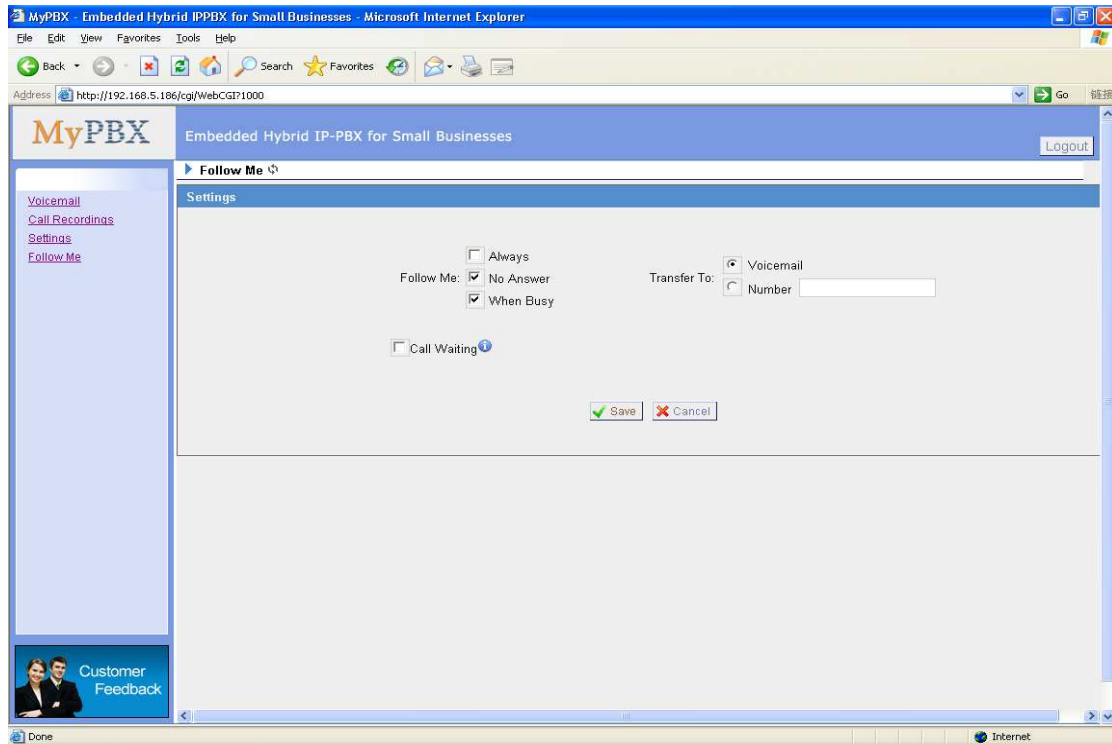
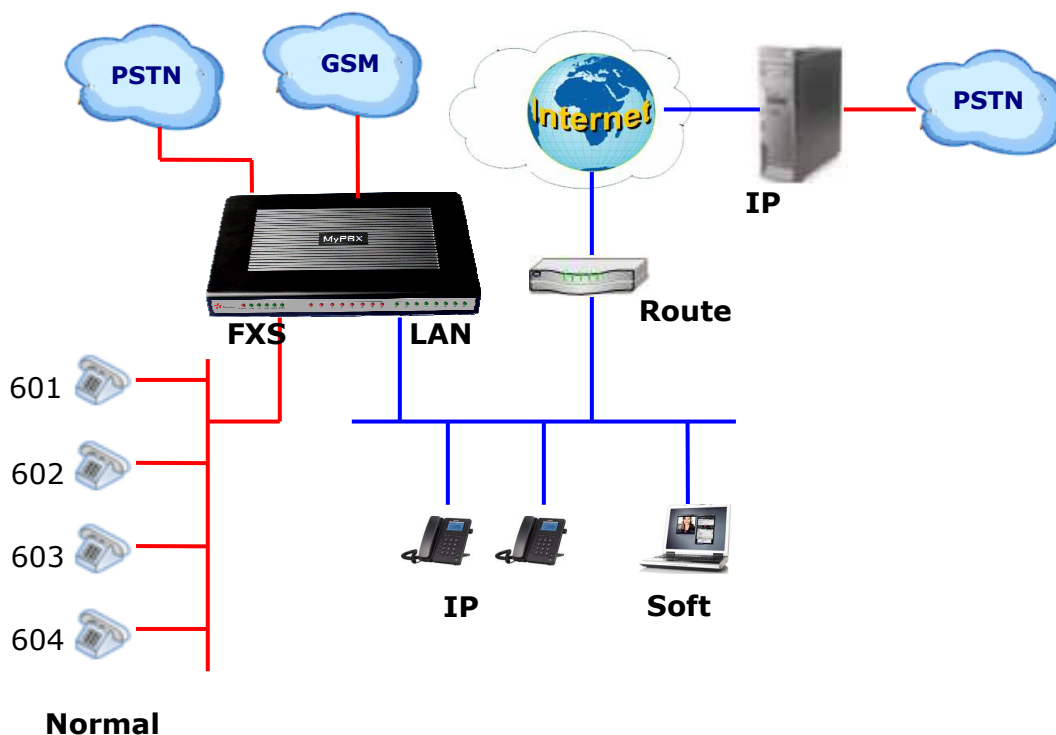


Figure 4-6

5. Use MPBX



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 MPBX, connected to a local PSTN.
- GSM Trunk:** GSM ports of MPBX, connected to GSM Network.
- BRI Trunk:** BRI ports of MPBX, 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.

The screenshot shows the 'Edit Outbound Route' configuration window. The route name is 'pstnout' and the dial pattern is '9.'. The configuration includes a 'Strip' field set to '1' digits from front, and a 'Prepend these digits' field which is empty. The 'Member Extensions' section shows a list of available extensions (empty) and a list of selected extensions: 500(SIP), 501(SIP), 502(SIP), 503(SIP), 504(SIP), and 505(SIP). The 'Member Trunks' section shows a list of available trunks: 8032(SIP), 192.168.5.190(SPS), pstn6(FXO), GSM11(GSM), BriTrunk8(BRI), and BriTrunk7(BRI). The selected trunks are pstn1(FXO) and pstn2(FXO). At the bottom, there are 'Save' and 'Cancel' buttons.

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.

Created New VoIP trunk

Type: SIP

Provider Name: voipprovider

Hostname/IP: voip.6699.org : 5060

Domain: voip.6699.org

User Name: 16885885

Authorization Name: 16885885

Password: ●●●●●●●●

From User: 16885885

Online Number:

Maximum Channels: 1

Enable Outbound Proxy Server

Transport: UDP Enable SRTP:

Caller ID:

DOD Setting

DOD: Associated Extension: 500 Add DOD

Save Cancel

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

Edit IVR welcome
✕

Number i :

Name i :

Prompt i : default ▼ [Custom IVR Prompts](#)

Play times i : 3 ▼

WaitExten i : 3 ▼

i Allow Dialing Other Extensions

i KeyPress Events

Key	Action	Destination
0	Connect to Extension ▼	User Extension -- 500 ▼
1	Connect to Extension ▼	User Extension -- 501 ▼
2	No Action ▼	 ▼
3	No Action ▼	 ▼
4	No Action ▼	 ▼
5	No Action ▼	 ▼
6	No Action ▼	 ▼
7	No Action ▼	 ▼
8	No Action ▼	 ▼
9	No Action ▼	 ▼
#	No Action ▼	 ▼
*	No Action ▼	 ▼
TimeOut i	Connect to Extension ▼	User Extension -- 500 ▼
Invalid i	Connect to Extension ▼	User Extension -- 500 ▼

✔ Save
✕ Cancel

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 ⓘ :

DID Number ⓘ :

Extension ⓘ :

Caller ID Number ⓘ :

Distinctive Ringtone ⓘ :

Member Trunks ⓘ

Available Trunks		Selected
	<input type="button" value="»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="«»"/>	<p>pstn1(FXO)</p> <p>pstn2(FXO)</p> <p>pstn6(FXO)</p> <p>GSM11(GSM)</p> <p>BrTrunk7(BRI)</p> <p>BrTrunk8(BRI)</p> <p>8032(SIP)</p> <p>voipprovider(SIP)</p>

During Office Hours

Destination:

<input type="checkbox"/> End Call	
<input type="checkbox"/> Extension	<input type="text" value="Extension -- 500"/>
<input type="checkbox"/> Voicemail	<input type="text" value="Voicemail -- 500"/>
<input checked="" type="checkbox"/> IVR	<input type="text" value="IVR -- welcome"/>
<input type="checkbox"/> RingGroup	<input type="text" value="RingGroup -- ringgroup_defi"/>
<input type="checkbox"/> Conference Room	<input type="text" value="Conference Room -- 640"/>
<input type="checkbox"/> DISA	<input type="text" value="DISA --"/>
<input type="checkbox"/> Queues	<input type="text" value="Queues --"/>
<input type="checkbox"/> Faxes ⓘ	<input type="text" value="Faxes -- 500"/>
<input type="checkbox"/> Outbound Routes ⓘ	<input type="text" value="Route Name -- pstnout"/>

Outside Office Hours

Destination:

<input type="checkbox"/> End Call	
<input type="checkbox"/> Extension	<input type="text" value="Extension -- 500"/>
<input type="checkbox"/> Voicemail	<input type="text" value="Voicemail -- 500"/>
<input checked="" type="checkbox"/> IVR	<input type="text" value="IVR -- welcome"/>
<input type="checkbox"/> RingGroup	<input type="text" value="RingGroup -- ringgroup_defi"/>
<input type="checkbox"/> Conference Room	<input type="text" value="Conference Room -- 640"/>
<input type="checkbox"/> DISA	<input type="text" value="DISA --"/>
<input type="checkbox"/> Queues	<input type="text" value="Queues --"/>
<input type="checkbox"/> Faxes ⓘ	<input type="text" value="Faxes -- 500"/>
<input type="checkbox"/> Outbound Routes ⓘ	<input type="text" value="Route Name -- pstnout"/>

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 MPBX 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 MPBX. When the status LED turns red, let go of the reset button.

Step 2: When the RUN status LED starts blinking, MPBX 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 MPBX 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

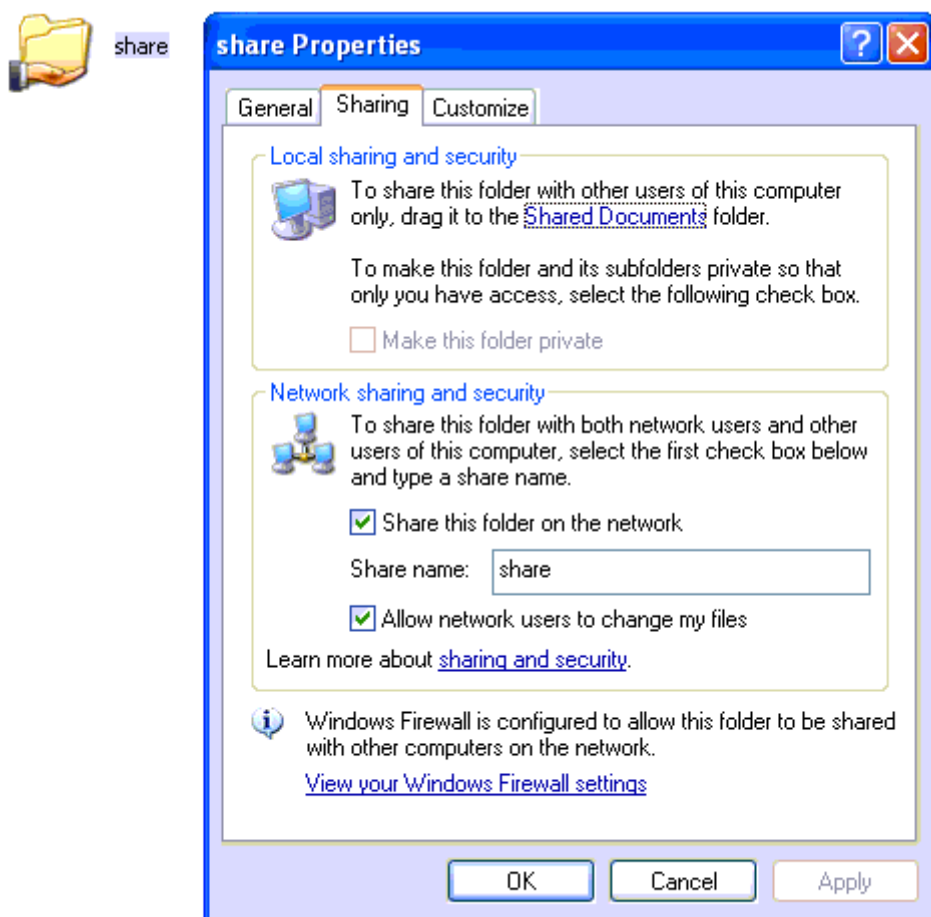


Figure B-1 Set up share Properties

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 MPBX to Figure B-2

External Storage Settings

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.

Step 1: Create a Net-Disk on a chosen computer

Step 2: Input the Net-Disk properties

Net-Disk Host/IP:

Net-Disk Share Name:

Net-Disk Access Username:

Net-Disk Access Password:

Move files created before: days ago

Step 3: Save Net-Disk settings

Step 4: Make sure the settings are successfully completed

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 MPBX 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 MPBX unit.

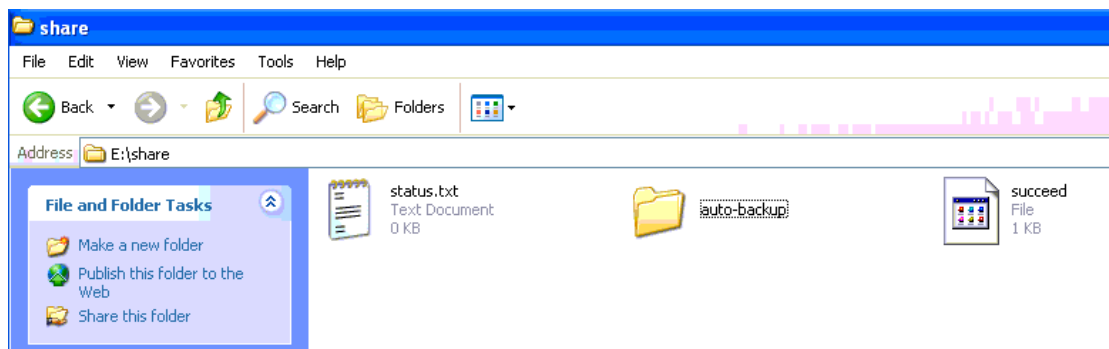


Figure B-3 Autobackup setting succeed

APPENDIX C How to Configure NAT setting

When MPBX is behind a NAT(firewall),you need to configure NAT setting on MPBX if you want to use a remote extension.

Please follow section **1** or **2** below depending on your network configuration.

1. If MPBX 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 MPBX 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

NAT

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

Enable STUN:

STUN Address:

STUN Port:

External IP Address:

External Host:

External Refresh Interval:

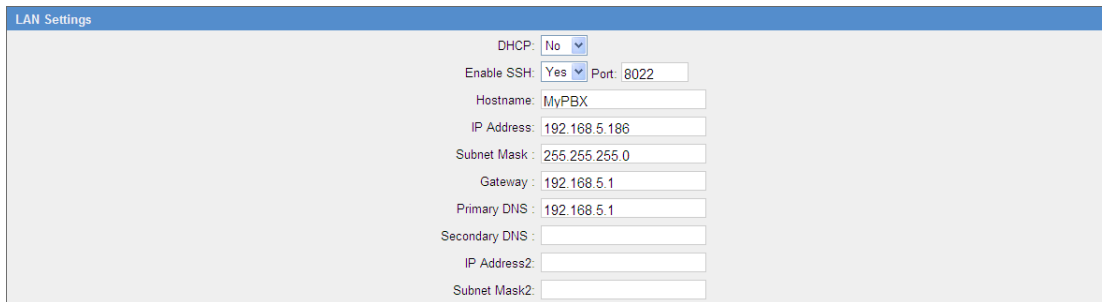
Local Network Identification:

NAT Mode:

Allow RTP Reinvite:

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 MPBX network settings should be configured like Figure C-2



LAN Settings

DHCP: No

Enable SSH: Yes Port: 8022

Hostname: MyPBX

IP Address: 192.168.5.186

Subnet Mask: 255.255.255.0

Gateway: 192.168.5.1

Primary DNS: 192.168.5.1

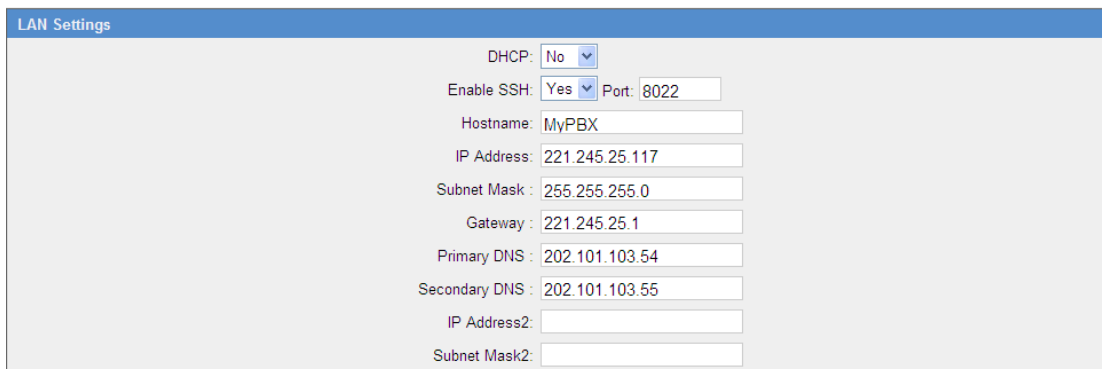
Secondary DNS:

IP Address2:

Subnet Mask2:

Figure C-2 MPBX Network setting

2. If MPBX 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:



LAN Settings

DHCP: No

Enable SSH: Yes Port: 8022

Hostname: MyPBX

IP Address: 221.245.25.117

Subnet Mask: 255.255.255.0

Gateway: 221.245.25.1

Primary DNS: 202.101.103.54

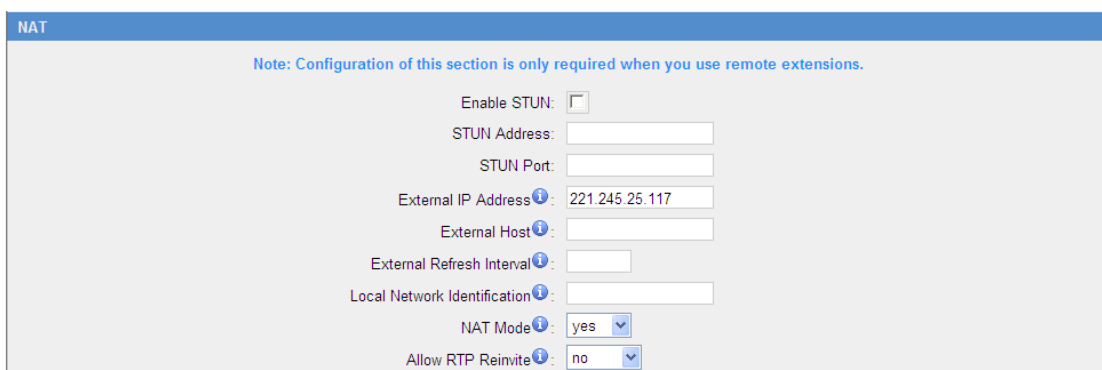
Secondary DNS: 202.101.103.55

IP Address2:

Subnet Mask2:

Figure C-3

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



NAT

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

Enable STUN:

STUN Address:

STUN Port:

External IP Address: 221.245.25.117

External Host:

External Refresh Interval:

Local Network Identification:

NAT Mode: yes

Allow RTP Reinvite: no

Figure C-4

External IP Address: The public IP address of MPBX

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.

The screenshot shows the Linksys Basic Setup page for an Etherfast Cable/DSL Router (BEFSR41). The page is divided into several sections: Internet Setup, Network Setup, and Basic Setup. The Local DHCP Server setting is highlighted with a red box, showing it is currently disabled. The Start IP Address is 192.168.1.100, and the Number of Address is 50. The DHCP Address Range is 192.168.1.100 to 192.168.1.149. The Client Lease Time is 0 minutes. The Static DNS and WINS settings are all set to 0.0.0.0.

LINKSYS
A Division of Cisco Systems, Inc. Firmware Version: 1.04.06

Etherfast® Cable/DSL Router BEFSR41

Setup

Setup Security Applications & Gaming Administration Status

Basic Setup DDNS MAC Address Clone Advanced Routing

Internet Setup

Internet Connection Type

Optional Settings (required by some ISPs)

Network Setup

Router IP

Network Address Server Settings (DHCP)

Obtain an IP automatically

Host Name:

Domain Name:

MTU: Enable Disable Size:

Local IP Address: . . .

Subnet Mask: . . .

Local DHCP Server: Enable Disable

Start IP Address:

Number of Address:

DHCP Address Range: to

Client Lease Time: minutes (0 means one day)

Static DNS 1: . . .

Static DNS 2: . . .

Static DNS 3: . . .

WINS: . . .

Basic Setup

The Basic Setup screen is where basic configuration is performed. Some ISPs (Internet Service Providers) will require that you enter the DNS information. These settings can be obtained from your ISP. After you have configured these settings, you should set a router password from the Administration->Management screen.

Completing the Internet Setup section is all that is required to set up for your specific ISP. Please look at the table below to configure the Router for your Internet connection.

[More...](#)

Save Settings **Cancel Changes**

CISCO SYSTEMS

Figure D-1

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

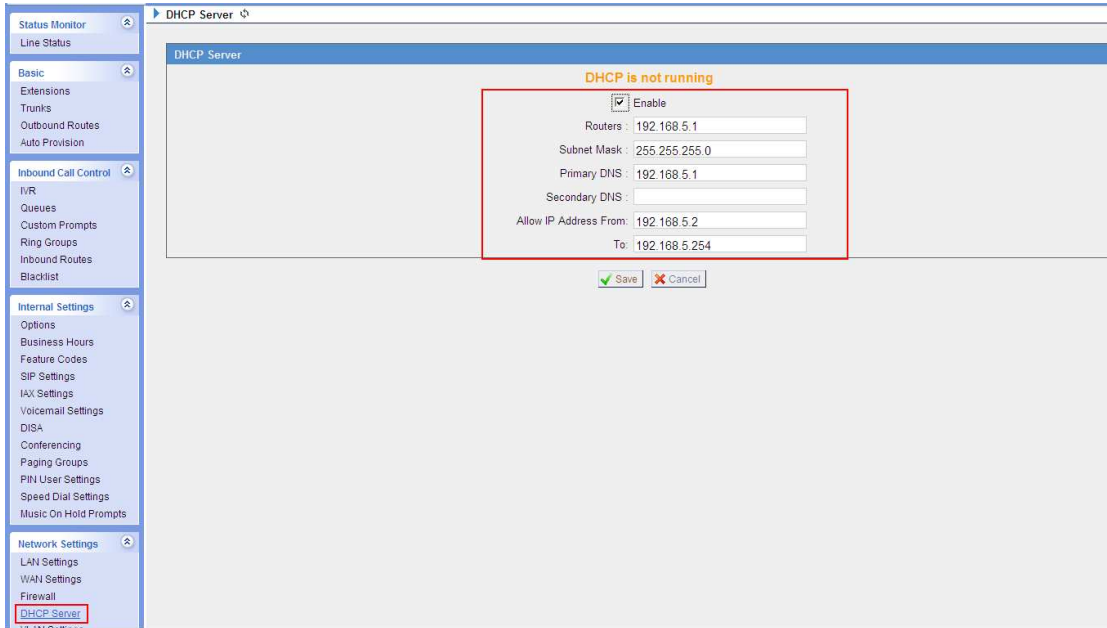


Figure D-2

Step3. Configure phones on MPBX auto-provision page.

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

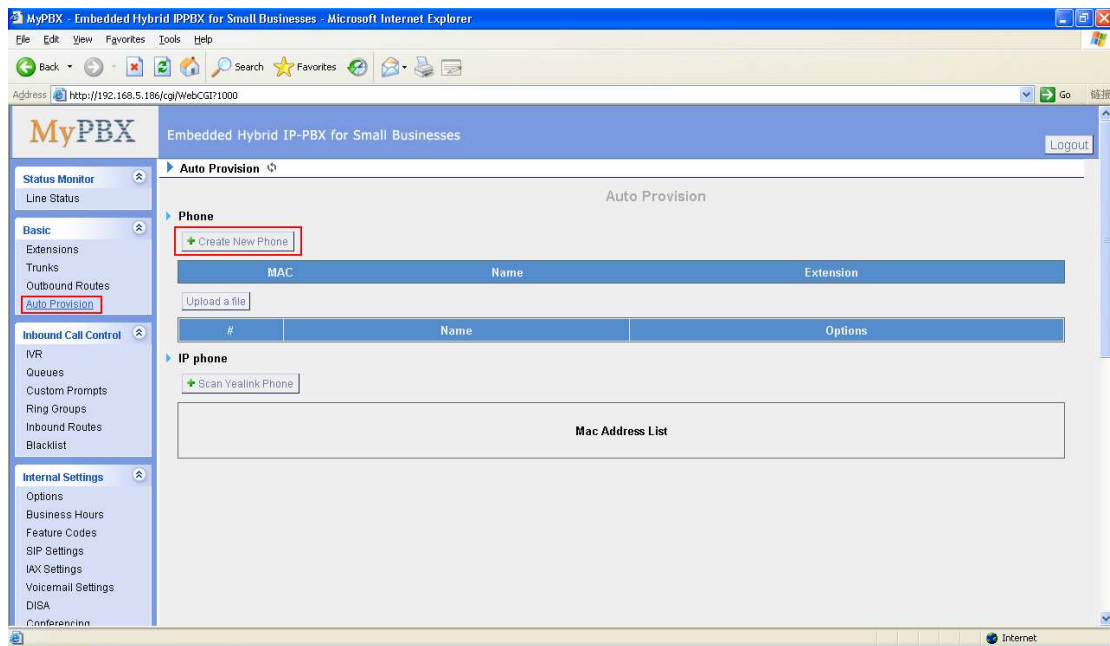


Figure D-3

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.

Internal Ringer Text	Internal Ringer File
Family	Ring1.wav
	Ring2.wav
	Ring3.wav
	Ring4.wav
	Ring5.wav
	Ring6.wav
	Ring7.wav
	Ring8.wav
	Ring1.wav
	Ring1.wav

Figure E-1

Step2. Configure the 'Distinctive Ringtone' on MPBX.

MPBX web interface, Inbound Routes → Edit Inbound Route, fill in the Internal Ringer Text on 'Distinctive Ringtone'.

Edit Inbound Route: allin

General

Route Name ⓘ : allin

DID Number ⓘ :

Extension ⓘ :

Caller ID Number ⓘ :

Distinctive Ringtone ⓘ : Family

Member Trunks ⓘ

Available Trunks	Selected
<input type="text"/>	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_defi
- Conference Room: Conference Room -- 640

Figure E-2

Step3. Finish.

<Finish>