

MyPBX

E1

# User Manual

Version 1.0.0.1

Yeastar Technology Co., Ltd

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# Table of Contents

1. Introduction.....	4
1.1 IP PBX Features.....	4
1.2 Hardware Specifications.....	4
1.2.1 Exterior Appearance.....	4
1.3 Package Contents.....	6
2. System set up.....	7
2.1 Connection Drawing.....	7
2.2 Connecting Ethernet Line.....	7
2.3 Supplying Power.....	7
3. Managing MyPBX.....	9
3.1 Administrator Login.....	9
3.2 Status Monitor.....	10
3.2.1 Line Status.....	10
3.3 Basic.....	11
3.3.1 Extension.....	11
3.3.1.1 Analog Station (FXS)Extensions.....	12
3.3.1.2 VOIP Extension.....	13
3.3.2 Trunk.....	16
3.3.2.1 Digital Trunk.....	16
3.3.2.2 Analog Trunk (FXO).....	17
3.3.2.3 VOIP Trunk.....	19
3.3.2.4 Service Provider.....	20
3.3.3 Outbound Routes.....	21
3.3.3.1 Create Outbound Route.....	22
3.3.3.2 Delete Outbound Route.....	23
3.3.4 Auto Provision.....	24
3.3.4.1 Create New Phone.....	24
3.4 Inbound Call Control.....	25
3.4.1 IVR.....	25
3.4.1.1 Create IVR.....	25
3.4.1.2 Delete IVR.....	27
3.4.2 IVR Prompts.....	27
3.4.3 Ring Groups.....	28
3.4.3.1 Create Ring Group.....	28
3.4.3.2 Delete Ring Group.....	29
3.4.4 Inbound Routes.....	29
3.4.4.1 Create Inbound Route.....	30
3.4.4.2 Delete Routes.....	32
3.4.5 Blacklist.....	33
3.4.5.1 Create Blacklist.....	33

---

3.5 Other Settings.....	34
3.5.1 DISA .....	34
3.5.2 Conferencing .....	35
3.5.3 Time conditions .....	36
3.5.4 Feature Codes .....	36
3.5.5 Paging and Intercom .....	39
3.6 System Settings .....	41
3.6.1 Options.....	41
3.6.2 Music on Hold Prompts/Audio Files .....	42
3.6.3 SIP Settings .....	43
3.6.4 IAX Settings .....	45
3.6.5 Voicemail Settings.....	47
3.6.6 Network Setting.....	49
3.6.7 DHCP Server .....	50
3.6.8 Autobackup setting .....	51
3.6.9 Password Settings.....	52
3.6.10 Date and Time .....	53
3.6.11 Backup and Restore.....	54
3.6.12 Reset and Reboot .....	55
3.6.13 Firmware Update.....	56
3.7 Reports .....	57
3.7.1 Call Logs.....	57
3.7.2 System Info .....	58
4. Access MRI .....	60
5. Use MyPBX .....	66
5.1 Make outbound call .....	66
5.1.1 Sample Routing via PSTN Trunk .....	67
5.1.2 Sample Routing via a VoIP Trunk .....	68
5.2 Incoming call .....	70
5.2.1 Sample Routing to an IVR .....	70
APPENDIX A FAQ.....	72
APPENDIX B How to Configure Autobackup.....	73
APPENDIX C How to configure NAT setting .....	75
APPENDIX D How to Use Auto Provision .....	78

# 1. Introduction

## MyPBX —An IP-PBX for Small Businesses/Home Office

MyPBX is a standalone embedded hybrid PBX for small businesses and remote branch offices of larger organizations (2-100 users per site). MyPBX also offers a hybrid solution (a combination of VoIP applications using legacy telecom equipment) alternative for enterprises who are not yet ready to migrate to a complete VoIP solution.

## Applications

### 1.1 IP PBX Features

• Auto-provision	• Call Transfer
• IVR(Automated Attendant)	• Call Waiting
• Blind Transfer	• Caller ID
• Call Detail Records (CDR) Search	• Do Not Disturb(DND)
• Call Forwarding	• Music On Hold
• Call Parking	• Ring Group
• Call Pickup	• Caller ID Routing
• Call Recording	• Three-way Conference Calling
• Inbound/Outbound Call Routes	• MRI(MyPBX Recording Interface)
• Paging and Intercom Groups	• Voicemail
• Direct Inward Subscriber Access (DISA)	• Blacklist
• DHCP	• TFTP,NTP server







### 1.2 Hardware Specifications

#### 1.2.1 Exterior Appearance

##### 1) Front Side



Figure 1-1 MyPBX Front Panel Picture

No.	Identifying
①	Green LED: Indicates correct power is being supplied to the unit
②	Green LED: Indicates the MyPBX is fully functional.
③	Green LED: Indicates stable Ethernet0 connection.
④	Green LED: Indicates stable Ethernet1 connection.
⑤⑥	⑤  ⑥  : The port hasn't connected the E1 line. ⑤  ⑥  : E1 line has connected, PRI is in corresponding or can't communicate. ⑤  ⑥  : Succeed.
⑦	<b>Red LED:</b> Indicates presence of an FXO module <b>Green LED:</b> Indicates presence of an FXS module LED Blinking- <b>Red</b> blinking: No connection between FXO port and PSTN LED Alternating - <b>Red</b> and <b>Green</b> (slow blink): FXO port is receiving an incoming call. LED Alternating - <b>Red</b> and <b>Green</b> (fast blink): FXO port is in use LED Dual - <b>Green</b> and <b>Red</b> blinking: FXS port is ringing LED Dual - <b>Green</b> and <b>Red</b> blinking: FXS port is in use

## 2) Rear View

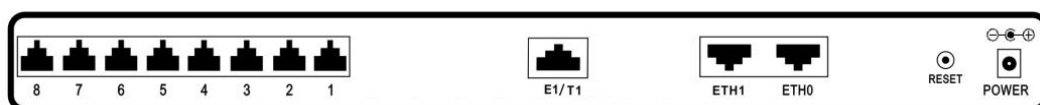


Figure 1-2 MyPBX Rear View

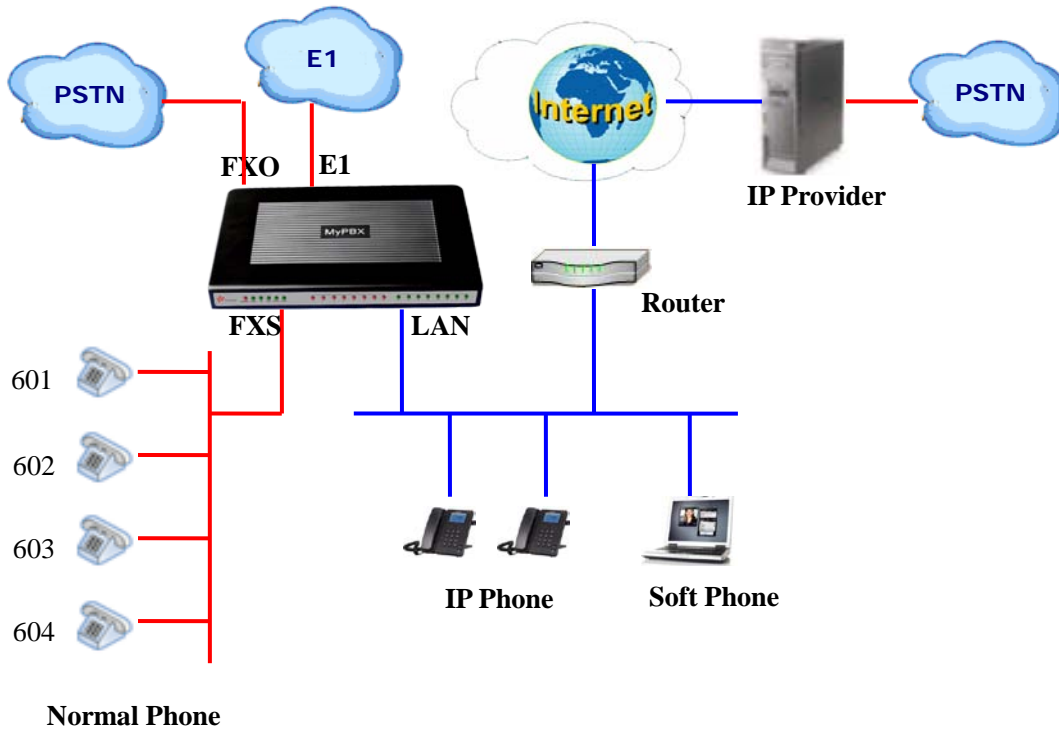
## 1.3 Package Contents

- ▲ MyPBX Main Unit x1
- ▲ Power Supply x1
- ▲ Ethernet Cable x1
- ▲ RJ11 PSTN Line Cord x1

**Note:** If any of the above items is damaged or missing, please contact your reseller.

## 2. System set up

### 2.1 Connection Drawing



### 2.2 Connecting Ethernet Line

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

### 2.3 Supplying Power

MyPBX utilizes the high-performance switch power supply, which supplies the required power for the unit.

AC Input: 100~240V

DC Output: 12V, 3A

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

1. Connect the small end of the power cable to the power input port on the MyPBX back panel, and plug the other end of the cable into a 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 MyPBX

### 3.1 Administrator Login

From your web browser, input the IP address of the MyPBX server.  
If this is the first time you are configuring MyPBX, please use the default settings below:

IP Address: <http://192.168.5.150>

Username: **admin**

Password: **password**

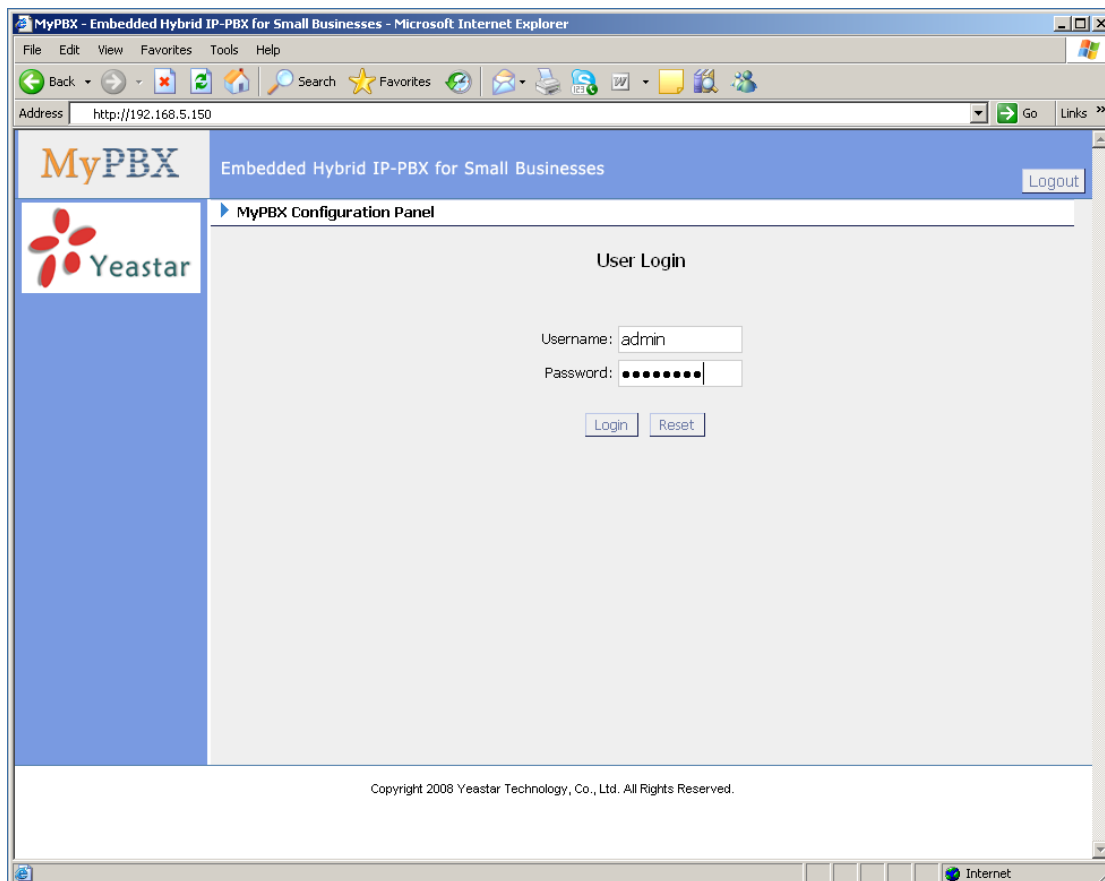


Figure 3-1

## 3.2 Status Monitor

### 3.2.1 Line Status

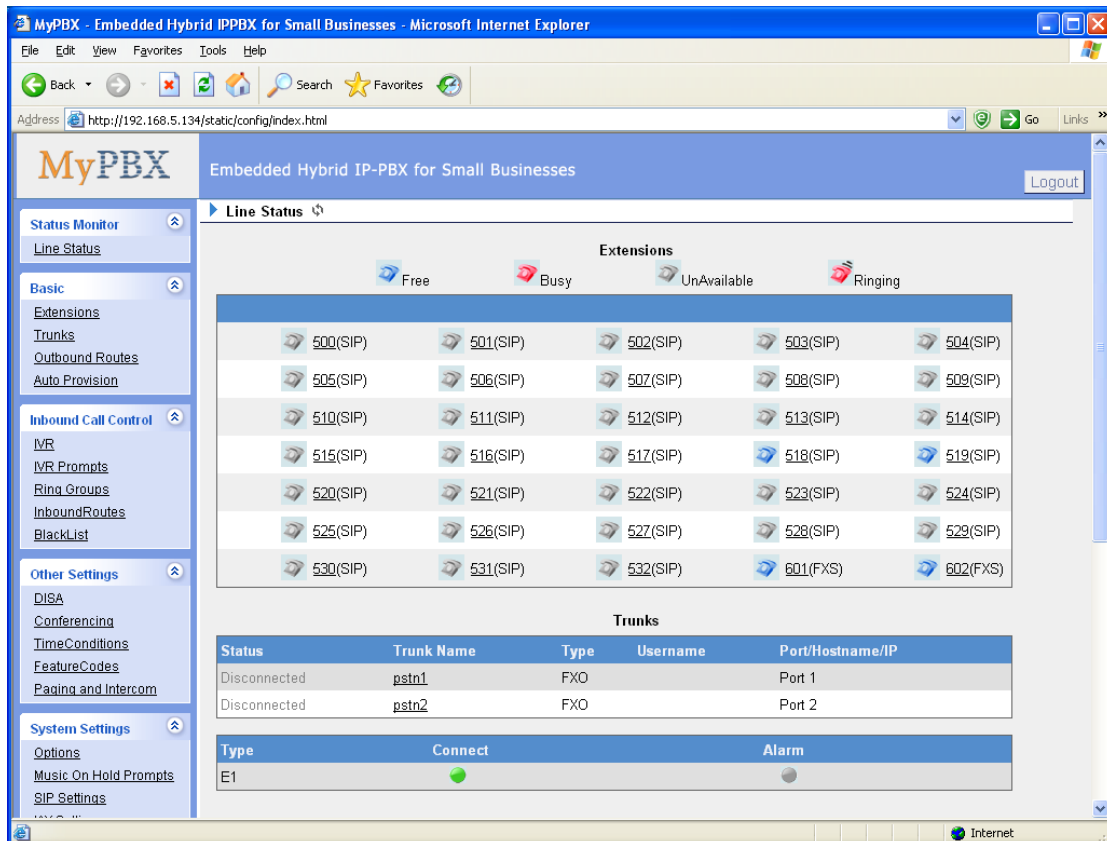






Figure 3-2

#### MyPBX Status Description:

##### Extensions:

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

##### Trunks:

###### SIP Trunk:

- 1) Unregistered — trunk registration failed.
- 2) Registered — succeed registration, trunk is ready for use.
- 3) Request Send — Registering.

4) Waiting — Waiting for authentication.

**FXO Trunk:**







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.

**E1 Trunk:**

Connect	Alarm	Comment
		The port hasn't connected the E1 line.
		E1 line has connected, PRI is in corresponding or can't communicate.
		Succeed.

### 3.3 Basic

#### 3.3.1 Extension

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

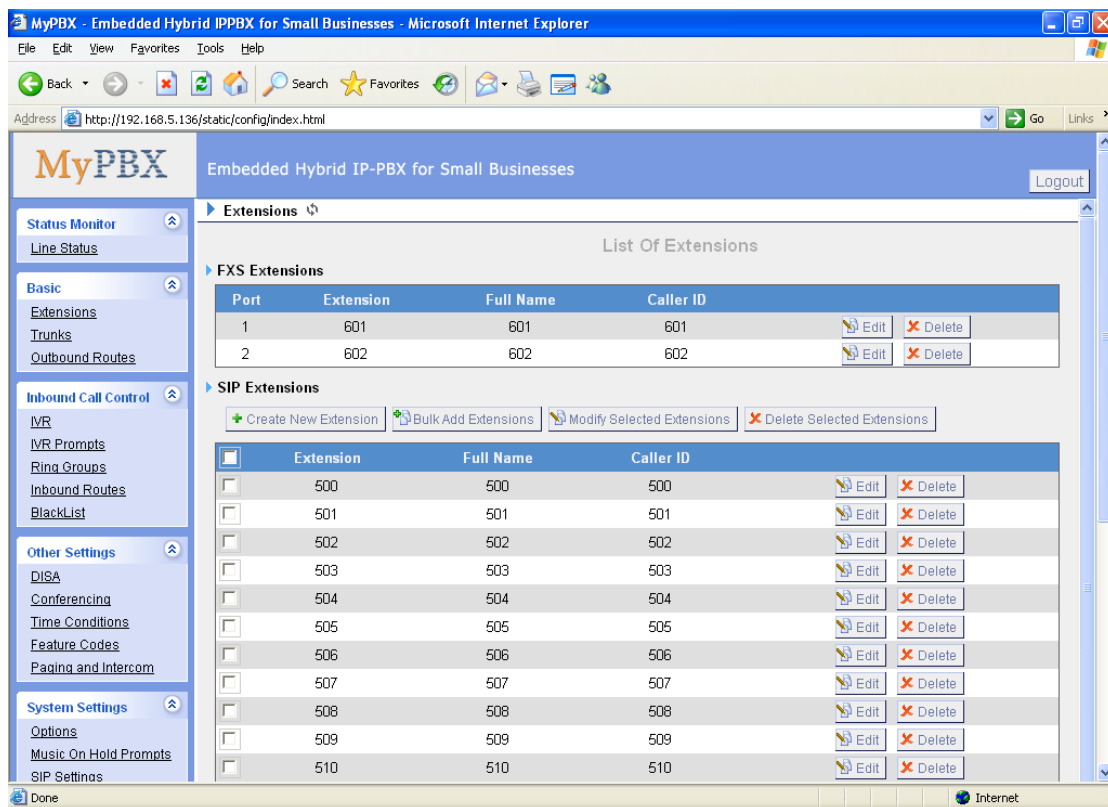


Figure 3-3

### 3.3.1.1 Analog Station (FXS) Extensions

#### Edit Analog Station Phone 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.

##### •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) Voice Mail

##### •Voice Mail Access PIN Code

Voicemail Password for this user, i.e. '1234'.

##### •Send Voice Mail to Email Address

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

**Note:** You need to have an SMTP server configured for this functionality.

#### 3) Flash

##### •rxflash

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

#### 4) Group

##### •Pickup Group

If a phone is assigned to a pickup group that is shared with a ringing phone, that phone may pick up the incoming call by calling \*4 . \*4 is default setting, but it can be changed under Feature Codes -> General -> Call Pickup.

#### 5) Follow me (Call Forwarding)

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

#### 6) Other Options

##### •Call Waiting

Check this option if the User or Phone should have Call-Waiting capability. If this

option is checked, busy transfer will not be available for this extension.

**.Hasmanager/Web Access**

Check this option to allow user to access the MyPBX Web Interface, which allows a user to manage their saved Voicemail messages and recordings. Users may access MRI by login to MyPBX web interface with their extension number and voicemail password.

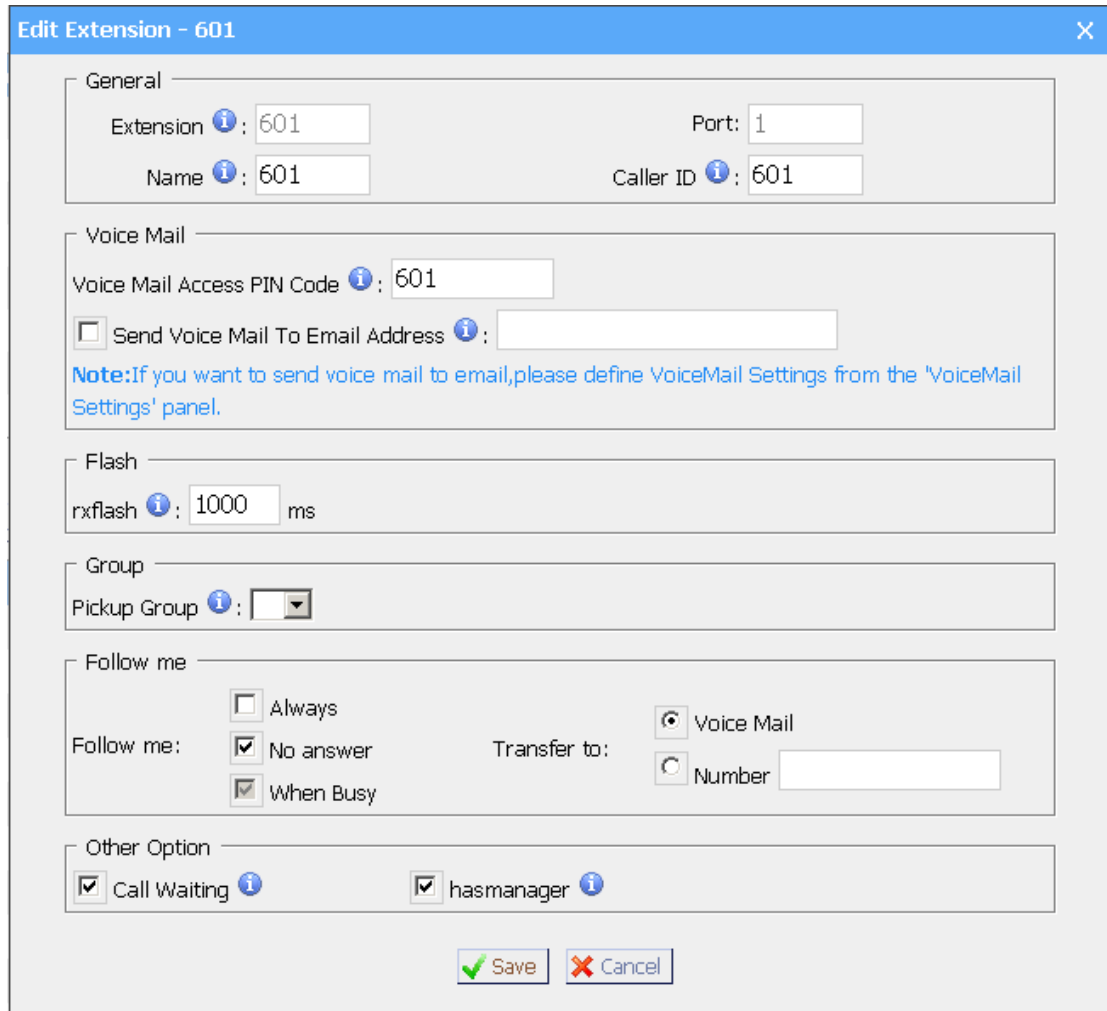


Figure 3-4

**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 MyPBX.

**1. Add VOIP Extension**

Go to Extensions → VOIP Extensions → Create New Extension

1) General

**•Type**

Extension type: SIP or 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 the user's sip account, Ex: '12t3f6'.

•**Name**

A character-based name for this user, EX: 'Bob Jones'.

•**Caller ID**

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

2) Voice Mail

•**Voice Mail Access PIN Code**

The voicemail Password for this user, i.e. '1234'.

•**Send Voice Mail to Email Address**

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

**Note:** You need to have an SMTP server configured for this functionality.

3) VoIP Setting

•**NAT**

Try this setting when MyPBX is on a public IP and communicating with devices hidden behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP or RTP ports.

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

4) Group

•**Pickup Group**

If a phone is assigned to a pickup group that is shared with a ringing phone, that phone may pick up the incoming call by pressing \*4. \*4 is the default setting, but it can be changed under Feature Codes -> General -> Call Pickup.

5) Follow me (Call Forwarding)

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

6) Other Options

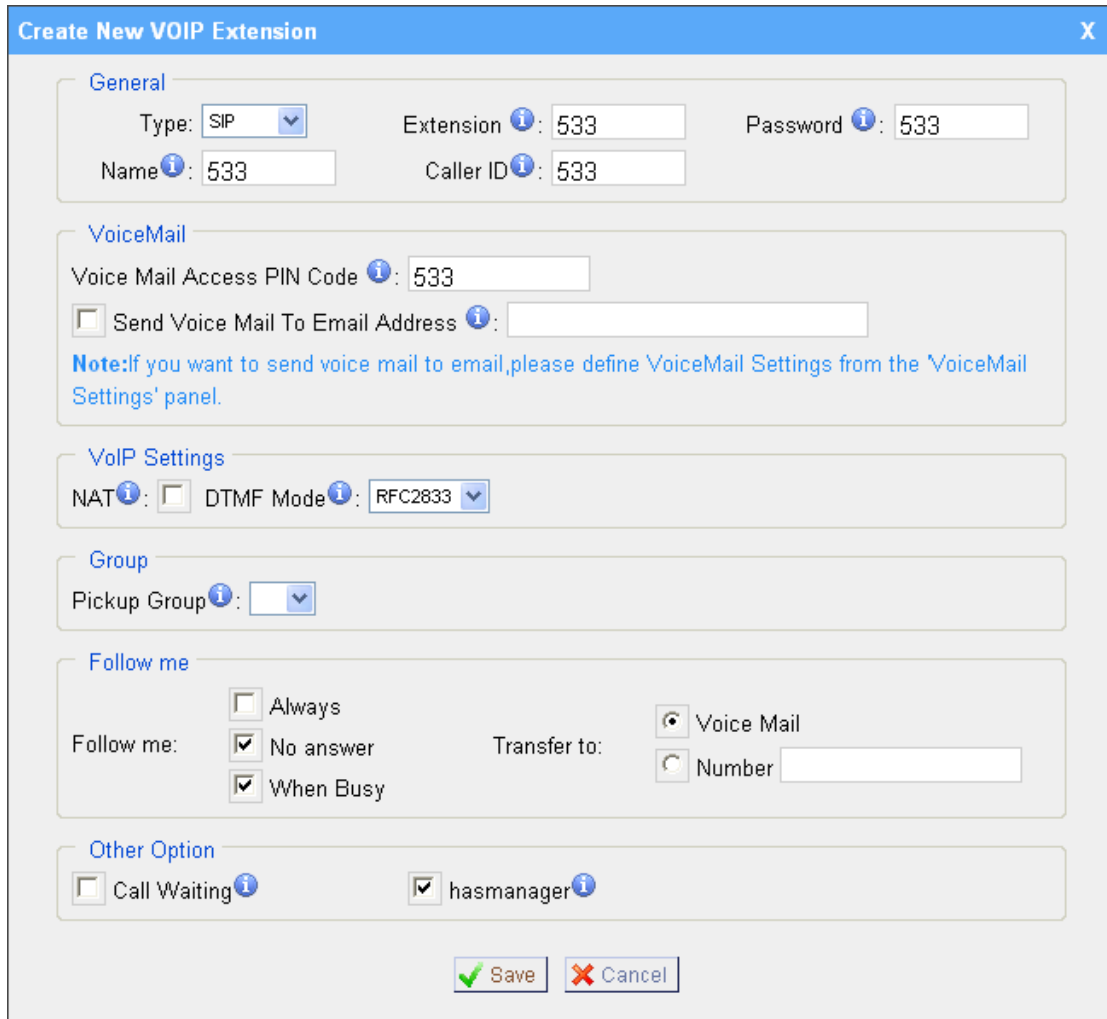
•**Call Waiting**

Check this option if the User or Phone should have Call Waiting capability. If this option is checked, busy transfer will not be available.

•**Hasmanager/Web Access**

Check this option to allow a user to access the MRI web interface, which allows

a user to manage their saved voicemail messages and recordings. Users may access MRI by logging in with their extension number and voicemail password.



The screenshot shows a web form titled "Create New VOIP Extension" with a close button (X) in the top right corner. The form is organized into several sections:

- General:** Includes fields for "Type" (set to SIP), "Extension" (533), "Password" (533), "Name" (533), and "Caller ID" (533).
- VoiceMail:** Includes a "Voice Mail Access PIN Code" field (533), a checkbox for "Send Voice Mail To Email Address", and a note: "Note: If you want to send voice mail to email, please define VoiceMail Settings from the 'VoiceMail Settings' panel."
- VoIP Settings:** Includes "NAT" (checkbox) and "DTMF Mode" (set to RFC2833).
- Group:** Includes a "Pickup Group" dropdown menu.
- Follow me:** Includes "Follow me:" options (Always, No answer, When Busy) and "Transfer to:" options (Voice Mail, Number).
- Other Option:** Includes checkboxes for "Call Waiting" and "hasmanager".

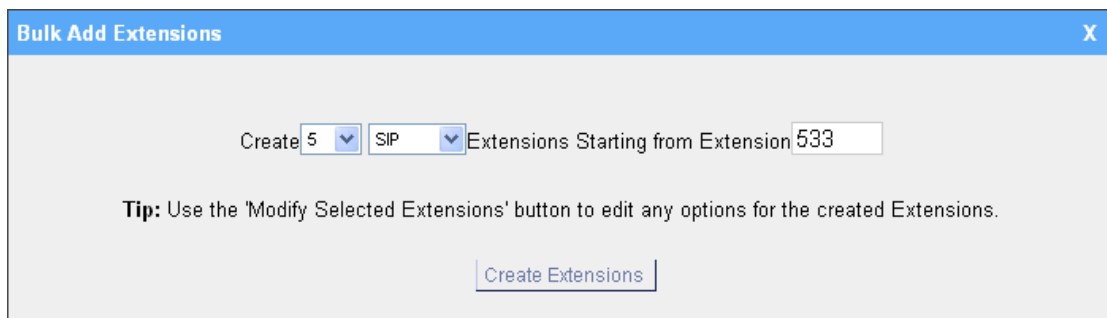
At the bottom of the form are "Save" and "Cancel" buttons.

Figure 3-5

## 2. Bulk add SIP Extensions

Go to Extensions → VOIP Extensions → Bulk add 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.



The screenshot shows a web form titled "Bulk Add Extensions" with a close button (X) in the top right corner. The form contains:

- A row of controls: "Create" (dropdown set to 5), "SIP" (dropdown), and "Extensions Starting from Extension" (text input set to 533).
- A tip: "Tip: Use the 'Modify Selected Extensions' button to edit any options for the created Extensions."
- A "Create Extensions" button.

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 four types of trunks: Digital Trunk, Analog Trunk (FXO), VOIP Trunk (SIP/IAX) and Service Provider trunks.

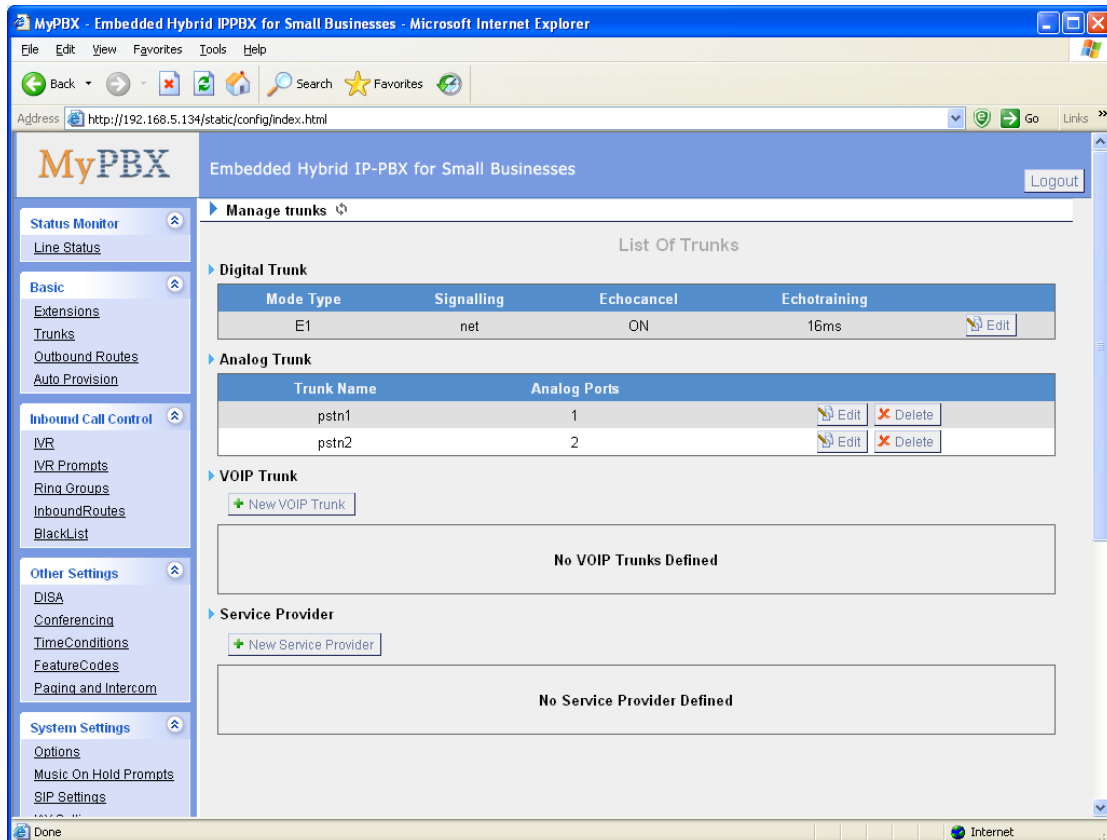


Figure 3-7

#### 3.3.2.1 Digital Trunk

##### 1. Edit Digital Trunk

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

1) General

**•Mode Type**

E1 / T1 / J1

**•Signalling**

Net /Cpe

**•Switchtype**

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!

**•Echocancel**

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

**•Echotraining**

Echotraining A common echo symptom is that at the beginning of a conversation there is a good deal of echo and then it fades away as it trains itself.

**•DOD Setting**

DOD – Direct Outward Dialing Number.

Bind Extension – This extension make call out it will display the DOD Number.

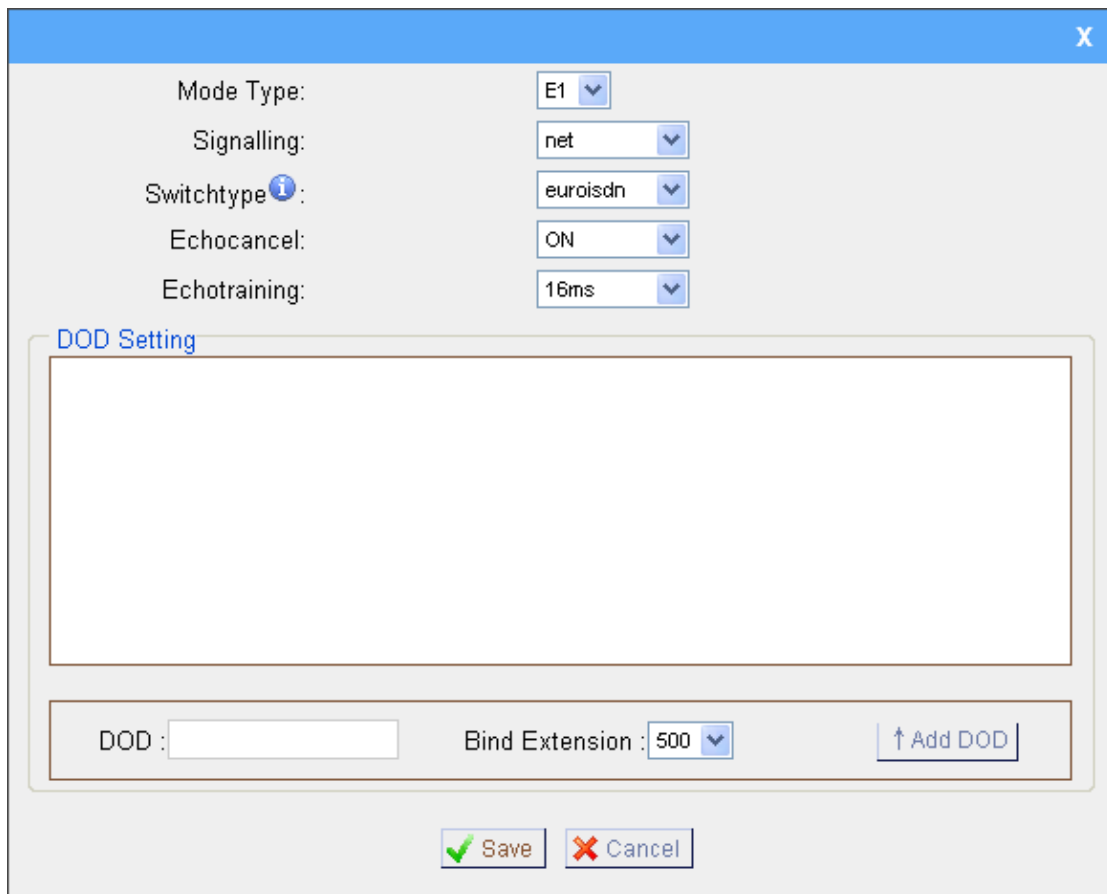


Figure 3-8

**3.3.2.2 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'.

#### •Audio Setting

Normally you should not have to adjust your analog ports beyond the initial calibration. Should you still need to fine tune your audio settings, please make adjustments as necessary.

### 2) Advanced Options

#### •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, MyPBX will accept any regular sound-silence pattern that repeats <Busy Count> times as a busy signal. If you specify Busy Pattern, then MyPBX will further check the length of the tone and silence, which will further reduce the chance of a false positive disconnect.

#### •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

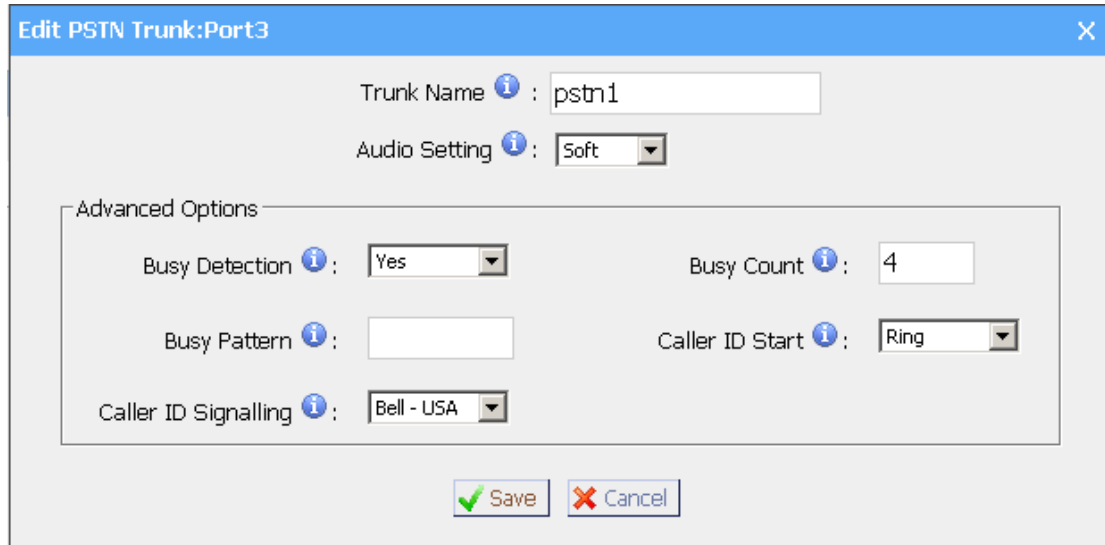


Figure 3-9

### 3.3.2.3 VOIP Trunk

#### 1. Add VOIP Trunk

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

#### •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: 'yeostar'.

#### •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.

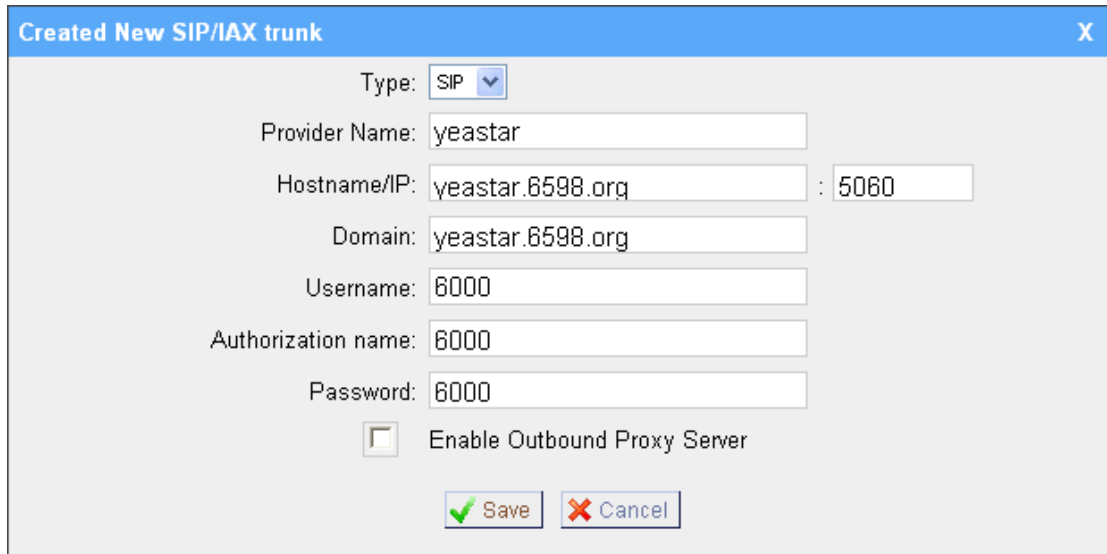
#### •Authorization name

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

#### •Password

Password of SIP account.

#### •Enable Outbound Proxy Server



The screenshot shows a dialog box titled "Created New SIP/IAX trunk" with a close button (X) in the top right corner. The form contains the following fields and options:

- Type: SIP (dropdown menu)
- Provider Name: yeastar
- Hostname/IP: yeastar.8598.org : 5060
- Domain: yeastar.8598.org
- Username: 6000
- Authorization name: 6000
- Password: 6000
- Enable Outbound Proxy Server
- Buttons: Save (with green checkmark icon), Cancel (with red X icon)

Figure 3-10

## 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

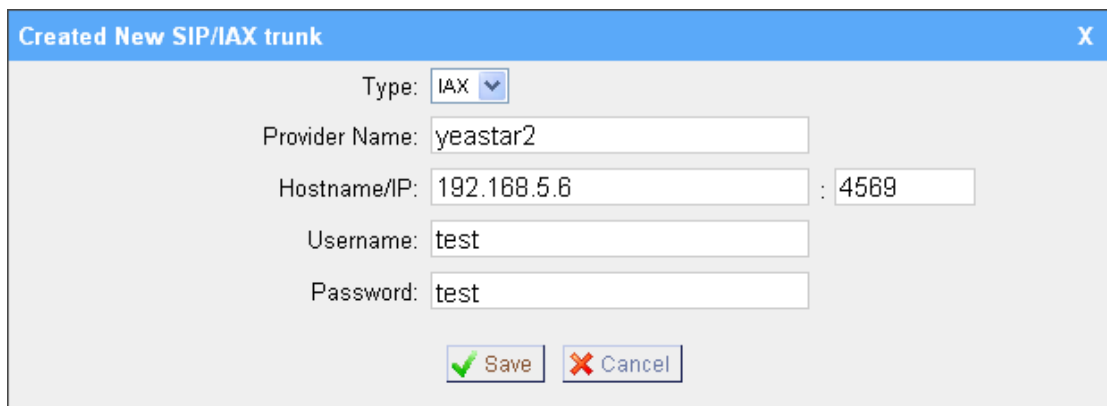
Service provider's hostname or IP address. 4569 is the standard port number used by IAX protocol. Don't change this part if it is not required.

### •Username

Username of IAX account. Used for IAX trunk registration.

### •Password

Password of IAX account.



The screenshot shows a dialog box titled "Created New SIP/IAX trunk" with a close button (X) in the top right corner. The form contains the following fields and options:

- Type: IAX (dropdown menu)
- Provider Name: yeastar2
- Hostname/IP: 192.168.5.6 : 4569
- Username: test
- Password: test
- Buttons: Save (with green checkmark icon), Cancel (with red X icon)

Figure 3-11

## 3.3.2.4 Service Provider

### 1. Add Service Provider

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

**•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.

**.DOD setting**

- 1) DOD – Direct Outward Dialing Number.
- 2) Bind Extension.

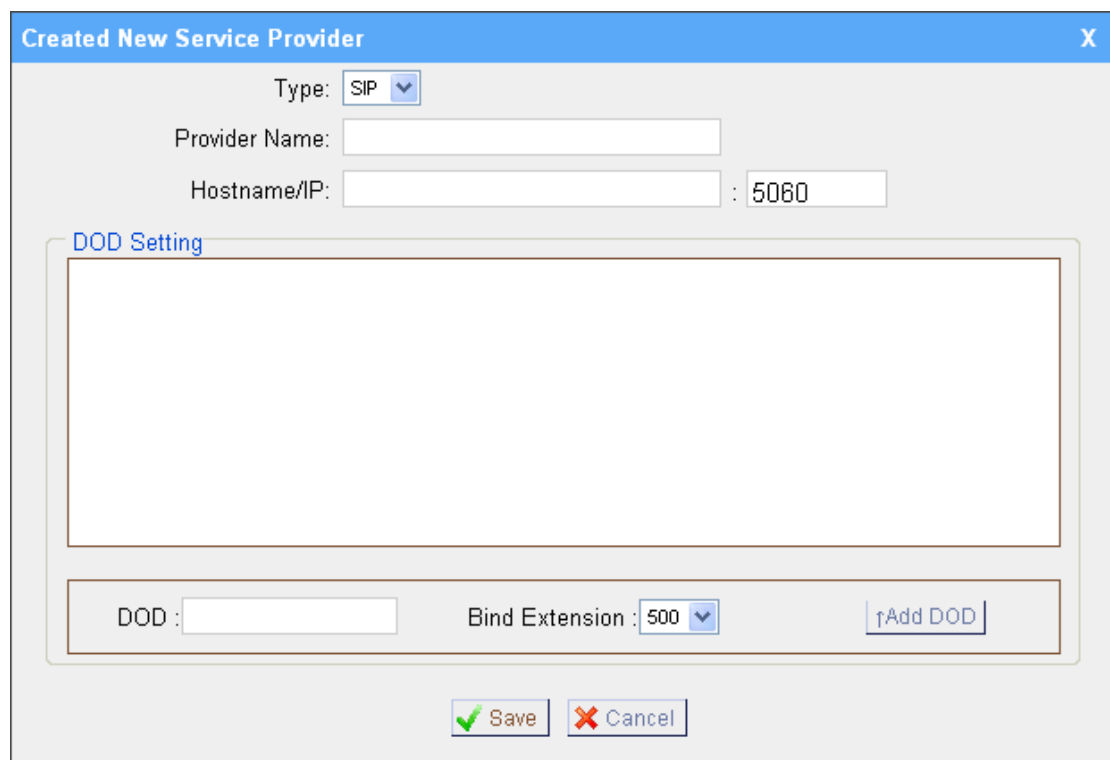


Figure 3-12

### 3.3.3 Outbound Routes

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

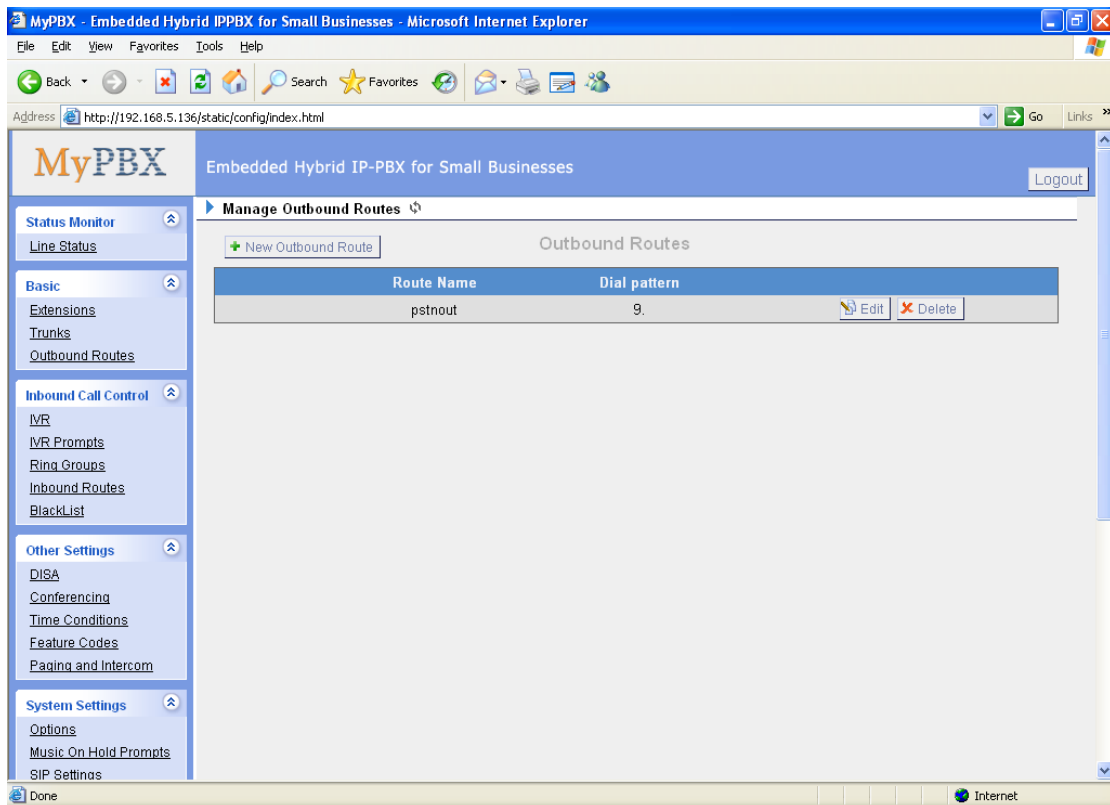


Figure 3-13

### 3.3.3.1 Create 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

In dial patterns, some characters 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)

**.** ... Wildcard: Matches Anything Remaining; i.e. 9011. Matches anything starting with 9011 (excluding 9011 itself)

**!** ... Wildcard: Causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible.

For example, **NXXXXXX** would match 7 digits; while **1NXXNXXXXXX** would represent a three digit area code plus phone number preceded by a one.

##### •Strip

Allows the user to specify the number of digits that will be stripped from the

front of the dialing string before the call is placed via one of the selected trunks  
 For example, to dial 9 before a long distance call, one digit should be stripped from the front before the call is placed.

**•Prepend these digits**

Allow the user to specify digits that are prepended to a number before the call is placed over a trunk. If a trunk requires 10 digit dialing, but users prefer 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are routed to a trunk. User may also prepend a 'w' character for analog trunks to provide a LED delay before dialing

**•Calls From Extensions**

Define the Extensions that are allowed to use the selected trunks

**•Make Outbound Calls on Trunk**

Define the Trunks that calls, matching the specified pattern, will be placed through.

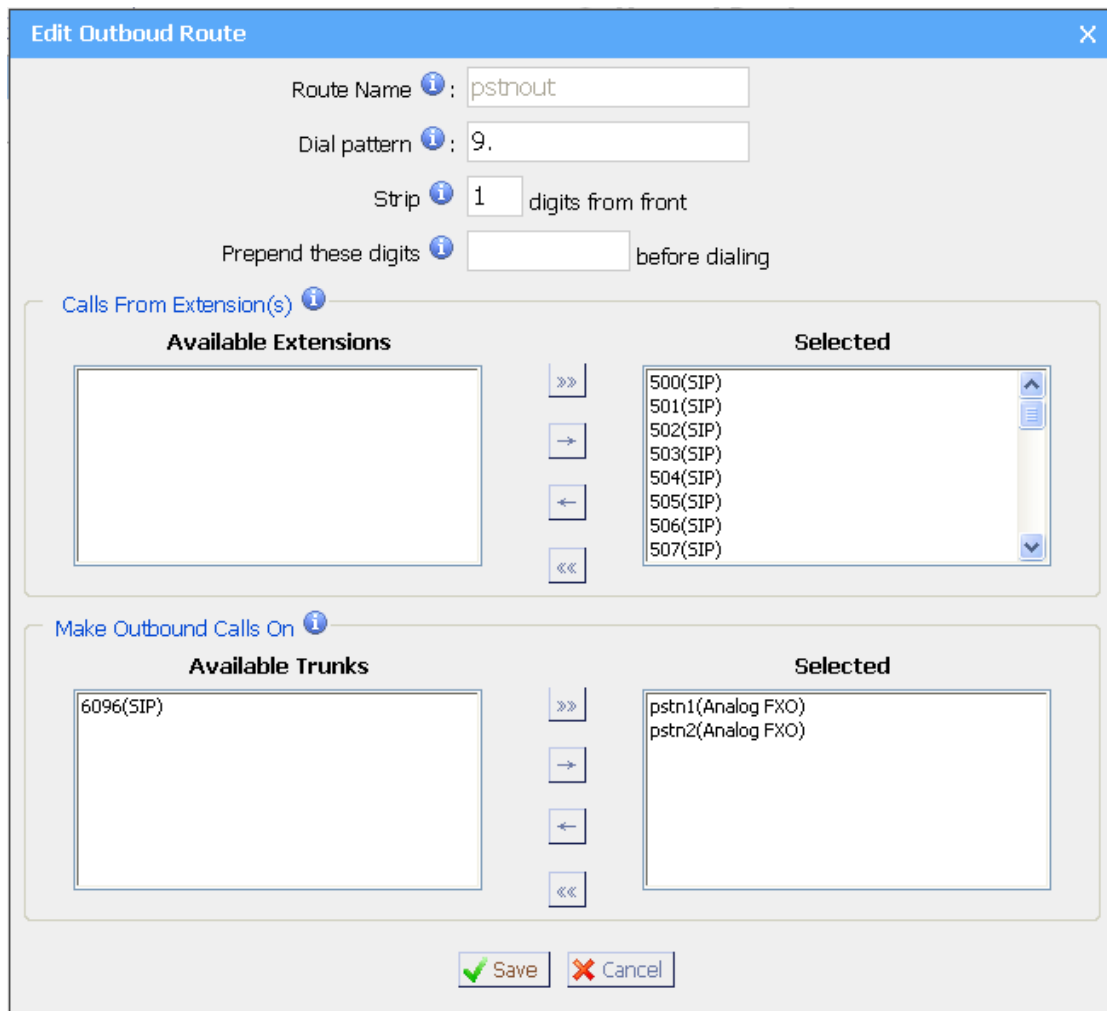


Figure 3-14

**3.3.3.2 Delete Outbound Route**

Click 'Delete' on selected trunk to delete it.

### 3.3.4 Auto Provision

#### 3.3.4.1 Create New Phone

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.

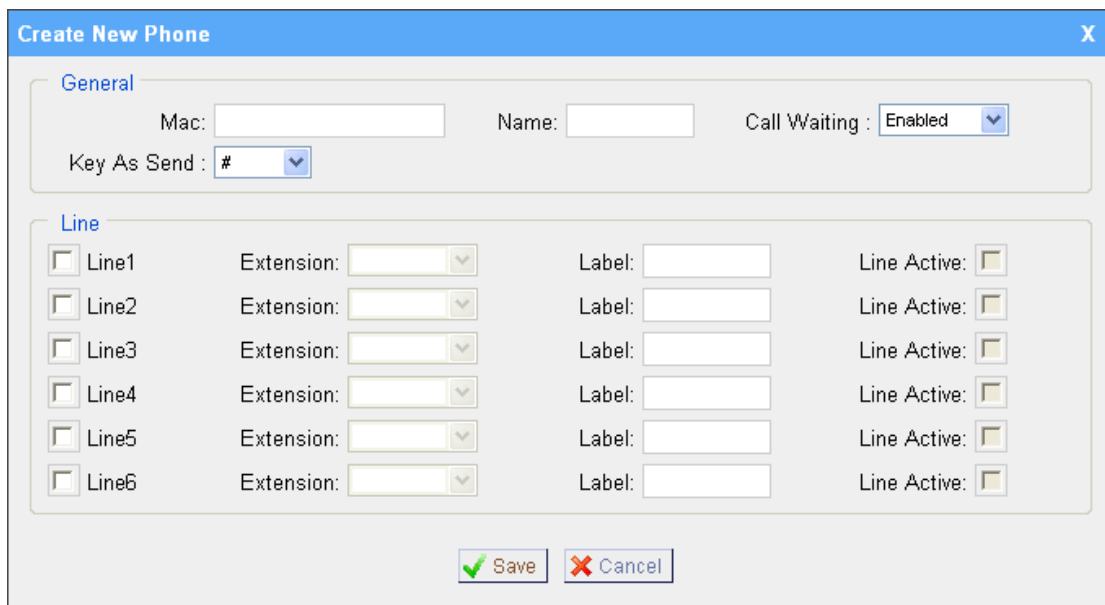


Figure 3-15



## 3.4 Inbound Call Control

### 3.4.1 IVR

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

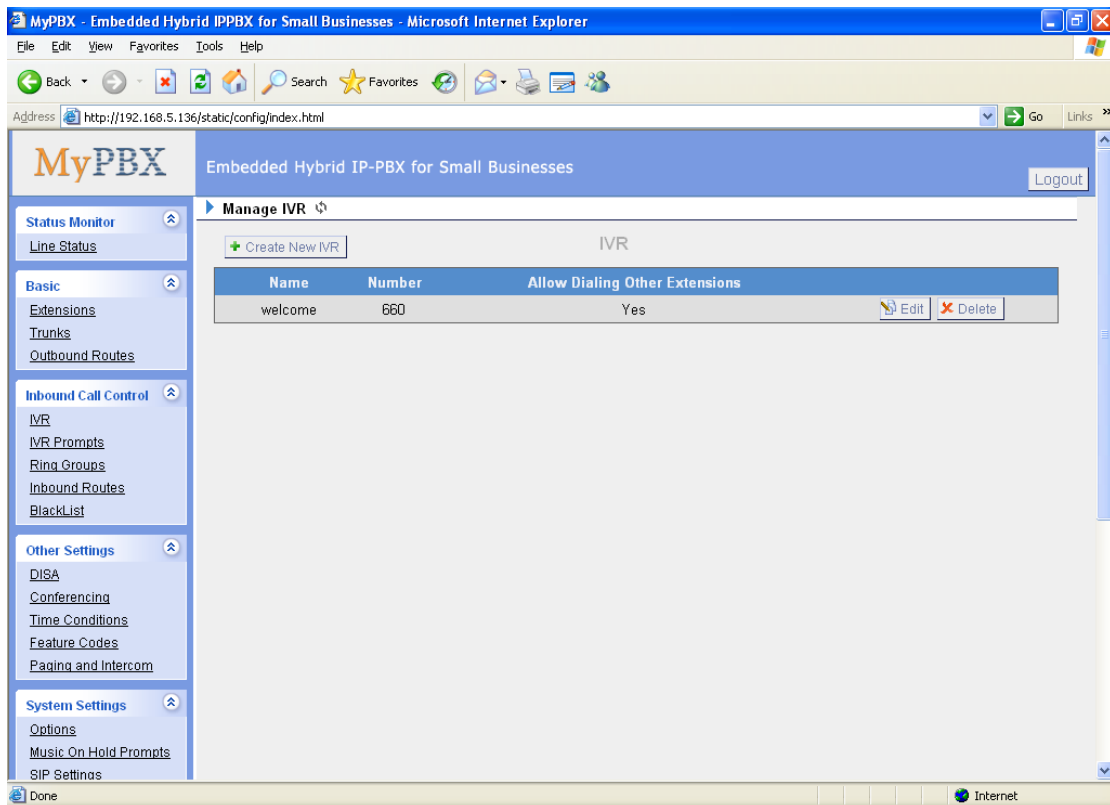


Figure 3-16

#### 3.4.1.1 Create IVR

Click 'Create New IVR'.

1) General

**•Number**

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

**•Name**

A name for the IVR

**•Prompt**

A Prompt file for the IVR

**•Play Times/Repeat Count**

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

**•WaitExten/Digit Timeout**

The number of seconds to wait before routing to the 'Time Out' condition of an IVR node.

**•Allow Dialing Other Extensions**

Check this option if a user wants to dial extension directly.

2) Key Press Events

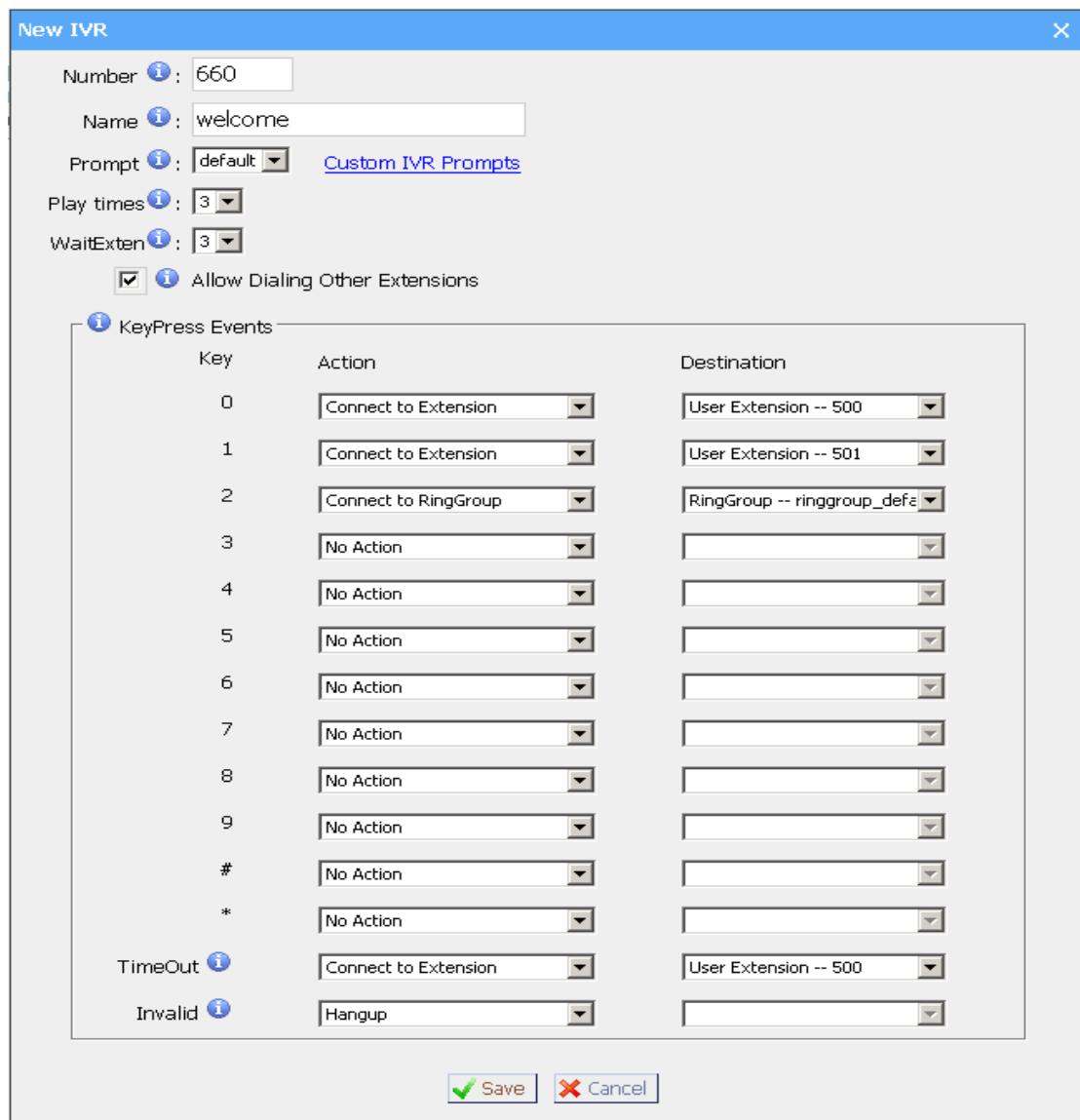
Key press events will cause the system to listen for DTMF input from the caller and define the actions that occur when a user presses the corresponding digit.

**•Time Out**

Define the timeout action. Timeout occurs when no DTMF entry is detected for 3 Seconds after the IVR has finished playing its prompts.

**•Invalid**

Define the invalid action. The invalid action is triggered if the user enters a DTMF that is not otherwise defined for the IVR.



**New IVR**

Number: 660  
 Name: welcome  
 Prompt: default [Custom IVR Prompts](#)  
 Play times: 3  
 WaitExten: 3  
 Allow Dialing Other Extensions

Key	Action	Destination
0	Connect to Extension	User Extension -- 500
1	Connect to Extension	User Extension -- 501
2	Connect to RingGroup	RingGroup -- ringgroup_defa
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	Connect to Extension	User Extension -- 500
Invalid	Hangup	

Figure 3-17

### 3.4.1.2 Delete IVR

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

### 3.4.2 IVR Prompts

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

- 1) Click 'Record New IVR 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.

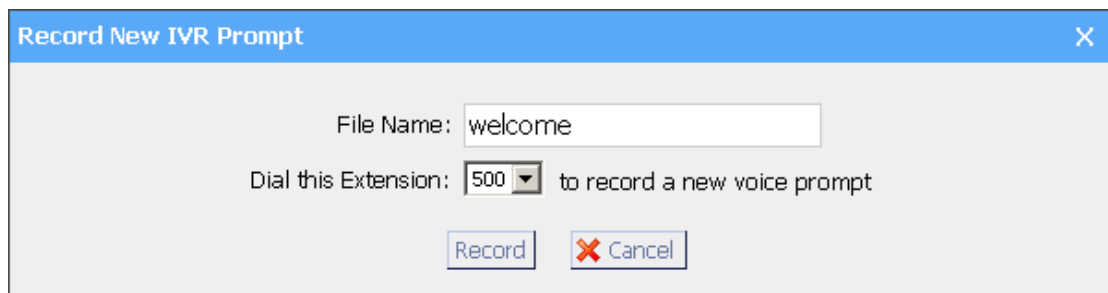


Figure 3-18

The administrator can also upload IVR prompts by doing the following:

- 1) Click 'Upload an IVR Prompt'.
- 2) Click 'Browse' to choose the desired IVR prompt.
- 3) Click 'Upload' to upload the selected prompt.

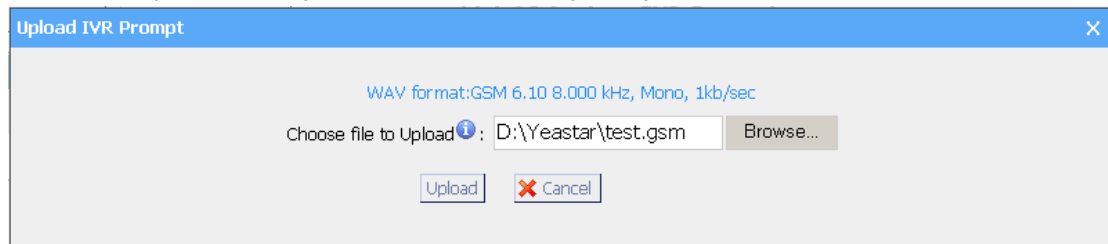


Figure 3-19

### 3.4.3 Ring Groups

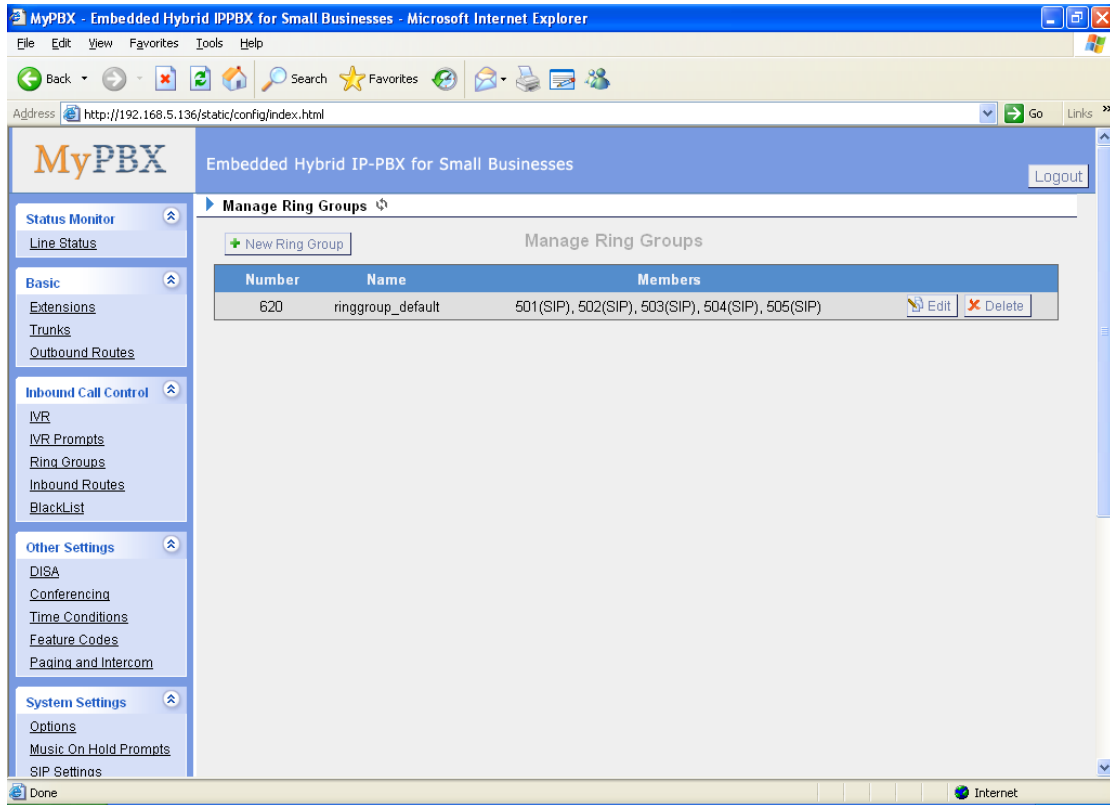


Figure 3-20

#### 3.4.3.1 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.

###### •Number for this ring group

This option defines the numbered extension that may be dialed to reach this Group.

###### •Strategy

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

1. Ring All: Ring all available Extensions simultaneously.
2. Ring in order: Ring each extension in the group sequentially.

###### •Time Out / Seconds to ring each member

###### 1) Seconds to ring each member

How many seconds an Extension will ring before MyPBX tries to ring the next Extension.

###### 2) Time out

How many seconds the extension will ring before routing to the 'No Answer' destination

**•Ring Group Members**

This selection shows all users in the group. Selecting a user here makes them a member of the current group.

**•Destination If No Answer**

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

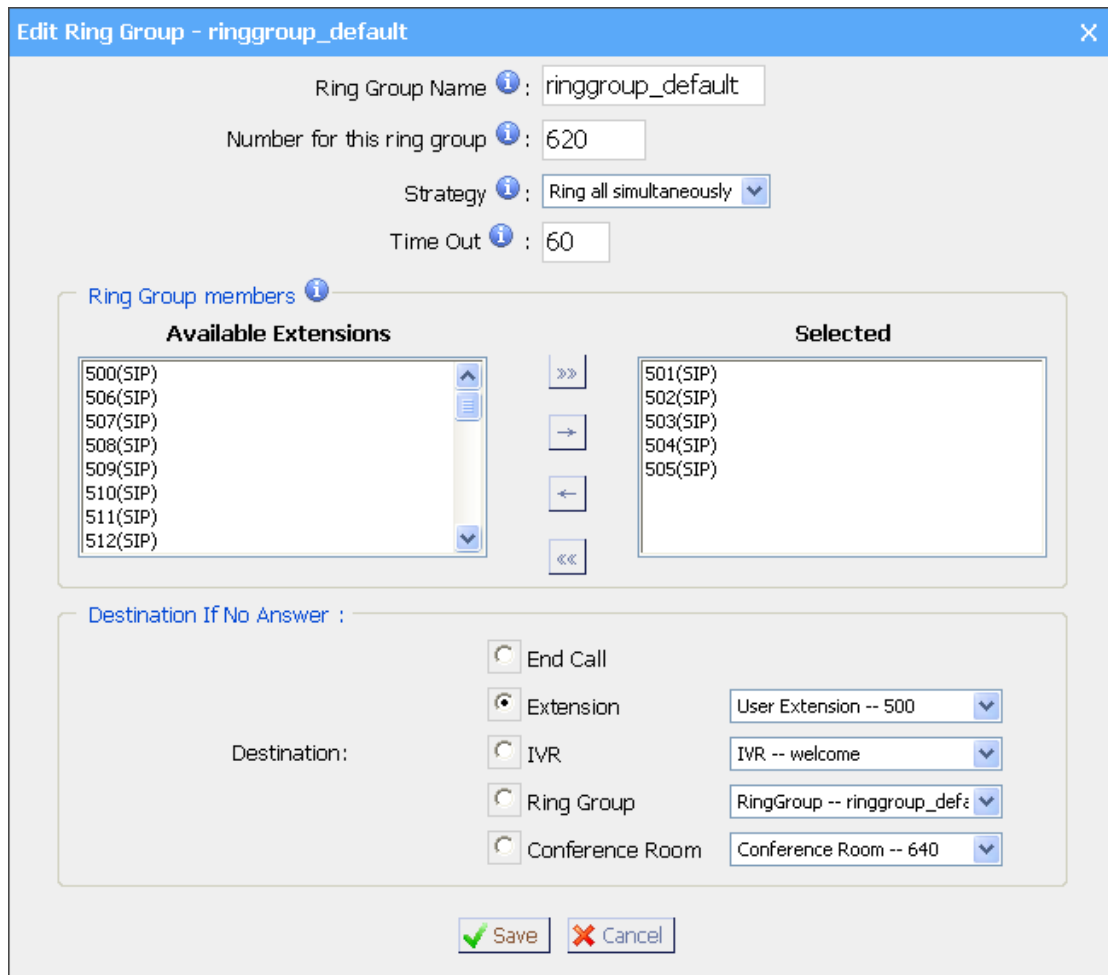


Figure 3-21

**3.4.3.2 Delete Ring Group**

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

**3.4.4 Inbound Routes**

Inbound routing processes incoming call traffic to destination extensions during business hours, or outside of working hours

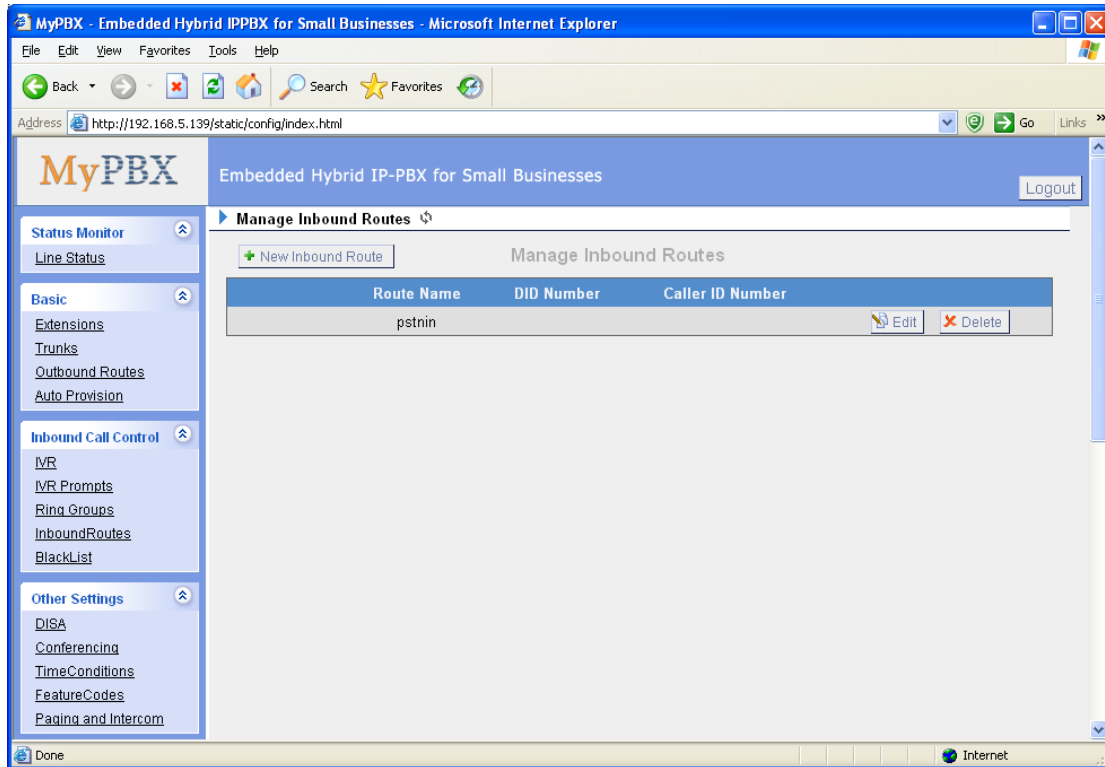


Figure 3-22

### 3.4.4.1 Create Inbound Route

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

#### 1) General

##### •Route Name

Name of this inbound Route. Ex: 'pstncallin' etc.

##### •DID Number

Define the expected DID Number if your trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. Only Service or E1 trunks need to be configured.

You can also use a pattern match (e.g. 2[345]X) to match a range of numbers.

##### •Pattern

In patterns, some characters 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)

**.** ... Wildcard: Matches anything remaining; i.e. 9011. Matches anything starting with 9011 (excluding 9011 itself)

**!** ... Wildcard: Causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible.

For example, **NXXXXXX** would match 7 digits while **1NXXNXXXXXX** would represent a three digit area code plus phone number proceeded by a one.

##### •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.

·Pattern

Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no CID info.

In patterns, some characters 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)

**.** ... Wildcard: Matches anything remaining; i.e. 9011. Matches anything starting with 9011 (excluding 9011 itself)

**!** ... Wildcard: Causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible.

For example, **NXXXXXX** would match 7 digits while **1NXXNXXXXXX** would represent a three digit area code plus phone number preceded by a one.

## 2) Trunk Sequence

This selection shows all Trunks. Moving a trunk to the 'Selected' box makes it a member of the inbound route.

## 3) During Office Hours

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

## 4) Outside Office Hours

You can choose how to route incoming calls outside of business hours.

Edit Inbound Route: pstnin
X

**General**

Route Name i :

DID Number i :

Caller ID Number i :

**Trunk Sequence** i

Available Trunks		Selected
test(SIP) test01(SIP) test02(SIP)	<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>	pstn1(Analog FXO) pstn2(Analog FXO) yeostar(SIP) testsig(SIP)

**During Office Hours**

Destination:

<input type="radio"/> End Call	
<input type="radio"/> Extension	<input type="text" value="User Extension -- 500"/> <span style="font-size: small;">v</span>
<input type="radio"/> VoiceMail	<input type="text" value="VoiceMail -- 500"/> <span style="font-size: small;">v</span>
<input checked="" type="radio"/> IVR	<input type="text" value="IVR -- welcome"/> <span style="font-size: small;">v</span>
<input type="radio"/> RingGroup	<input type="text" value="RingGroup -- ringgroup_def:"/> <span style="font-size: small;">v</span>
<input type="radio"/> Conference Room	<input type="text" value="Conference Room -- 640"/> <span style="font-size: small;">v</span>
<input type="radio"/> DISA	<input type="text"/> <span style="font-size: small;">v</span>

**Outside Office Hours**

Destination:

<input type="radio"/> End Call	
<input type="radio"/> Extension	<input type="text" value="User Extension -- 500"/> <span style="font-size: small;">v</span>
<input type="radio"/> VoiceMail	<input type="text" value="VoiceMail -- 500"/> <span style="font-size: small;">v</span>
<input checked="" type="radio"/> IVR	<input type="text" value="IVR -- welcome"/> <span style="font-size: small;">v</span>
<input type="radio"/> RingGroup	<input type="text" value="RingGroup -- ringgroup_def:"/> <span style="font-size: small;">v</span>
<input type="radio"/> Conference Room	<input type="text" value="Conference Room -- 640"/> <span style="font-size: small;">v</span>
<input type="radio"/> DISA	<input type="text"/> <span style="font-size: small;">v</span>

Figure 3-23

### 3.4.4.2 Delete Routes

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



### 3.4.5 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.

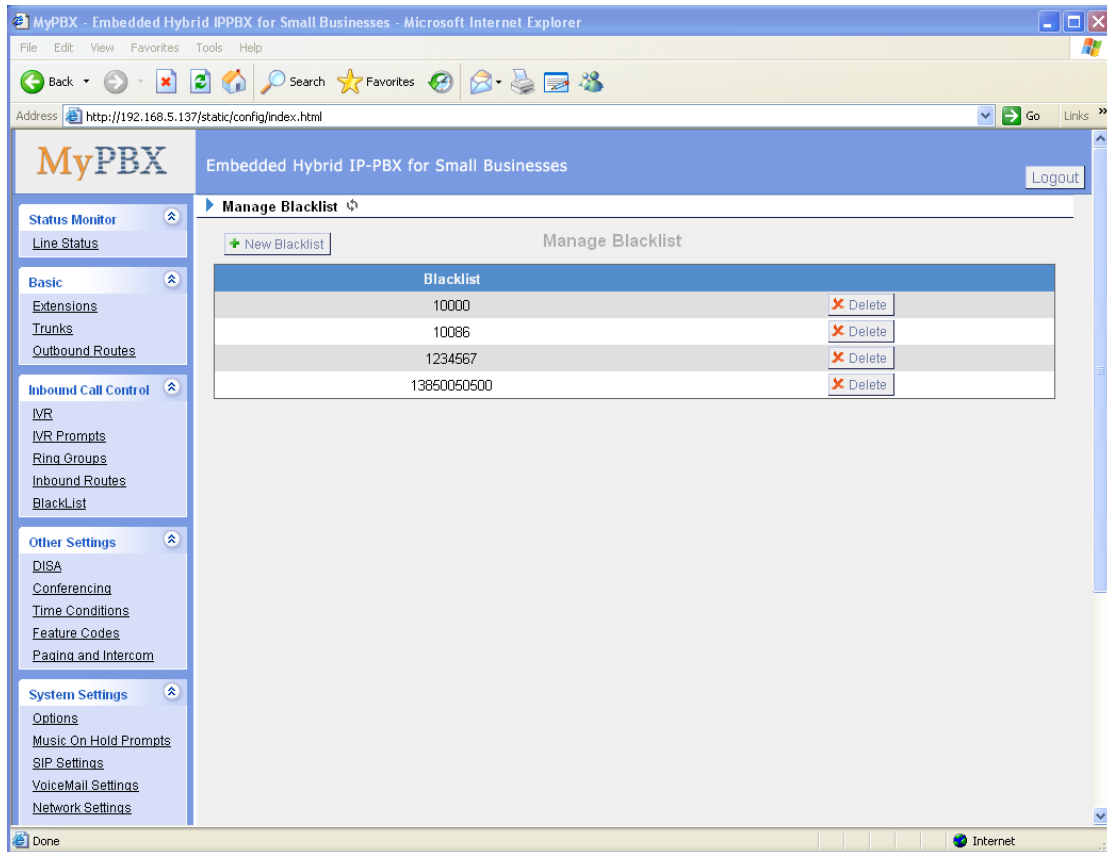


Figure 3-24

#### 3.4.5.1 Create Blacklist

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

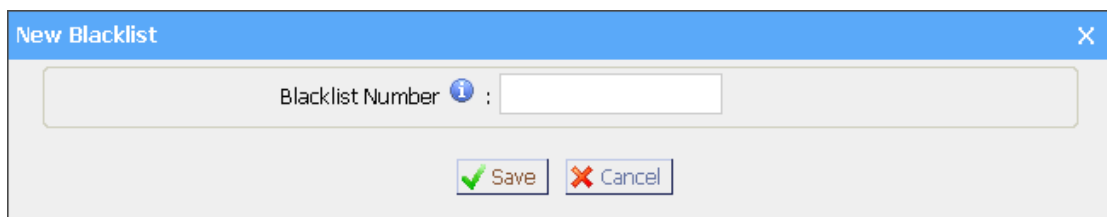


Figure 3-25

**•Blacklist number**

Enter the number you would like to block.

## 3.5 Other Settings

### 3.5.1 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.

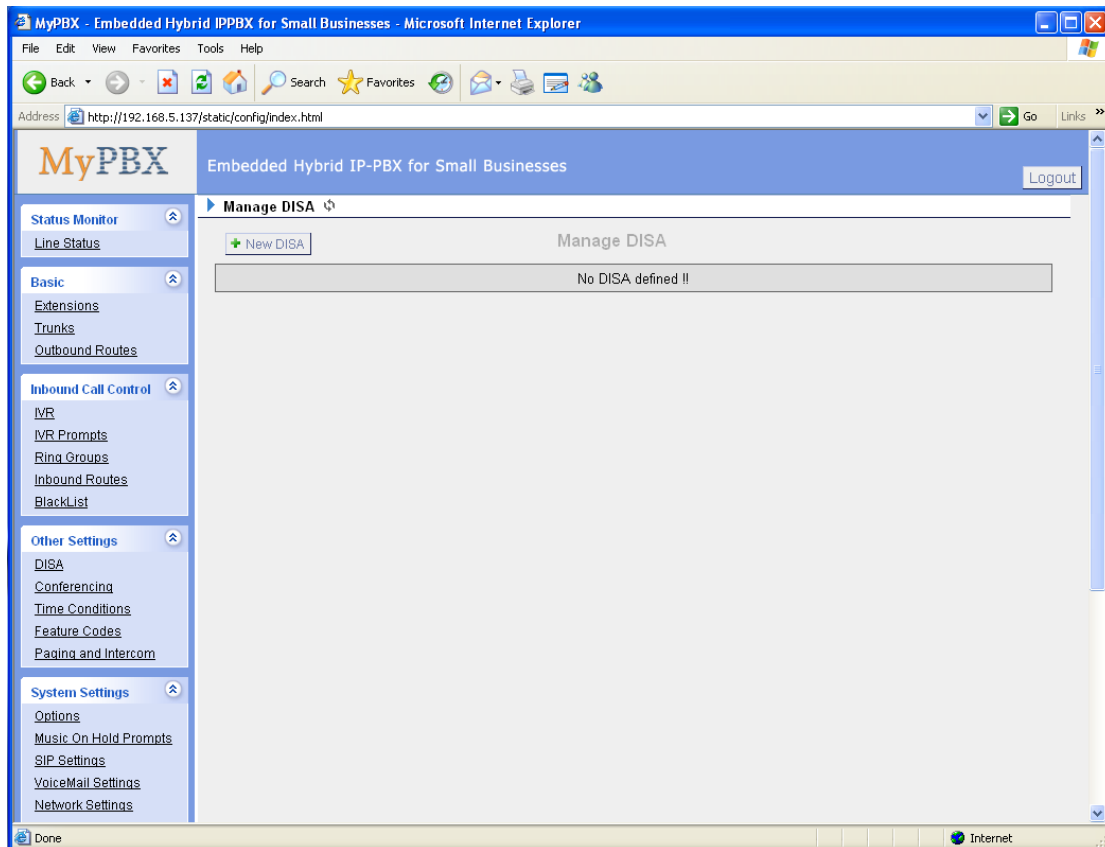


Figure 3-26

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

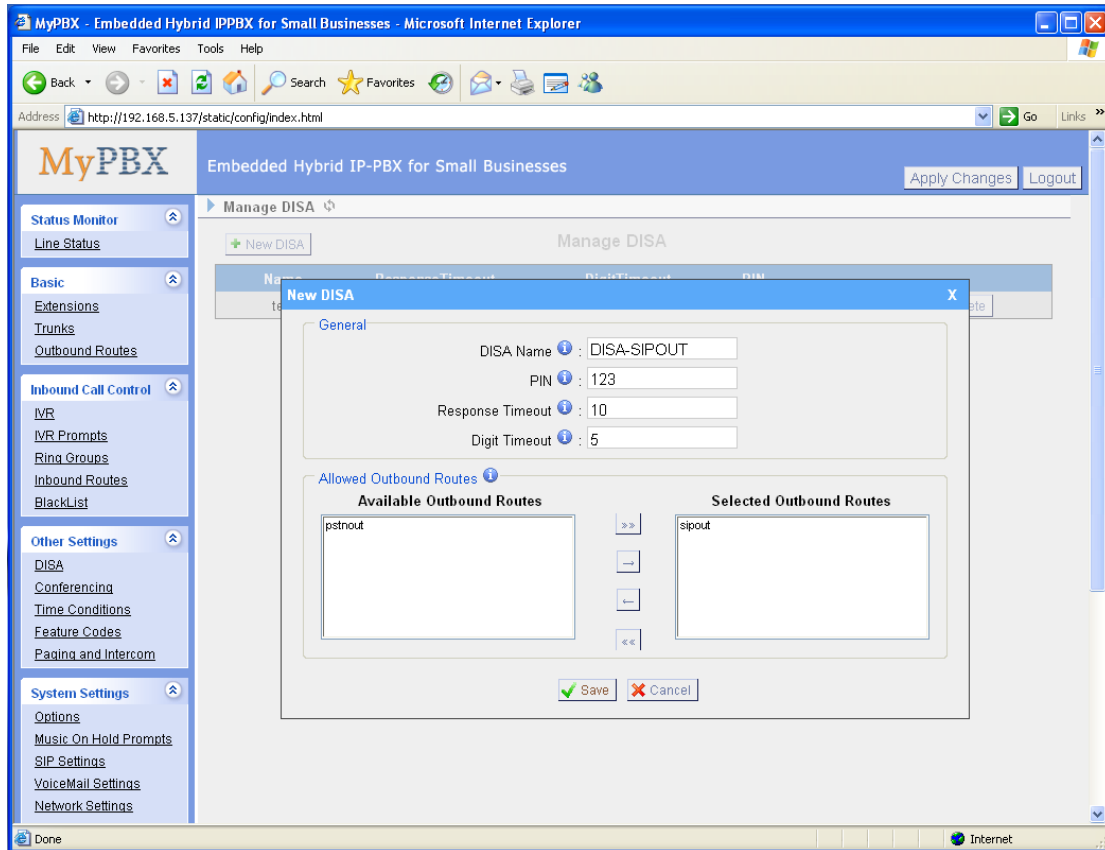


Figure 3-27

1) General

• **DISA Name**

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

• **PIN**

The user will be prompted for this number.

• **Response Timeout**

The maximum amount of time DISA will wait before disconnecting if the user has dialed an incomplete or invalid PIN number. Default is 10 seconds

• **Digit Timeout**

The maximum amount of time permitted between digits when the user is inputting in an extension. Default is 5 seconds.

2) Allowed Outbound Routes

This selection shows all outbound routes. Selected outbound routes will be available for users who log into this DISA application.

### 3.5.2 Conferencing

To create a new conference bridge, click the 'New Conference Bridge' button

• **Extension**

This is the number dialed to reach this Conference Room.

**•Pin Code**

Set an optional pin code for accessing the bridge.

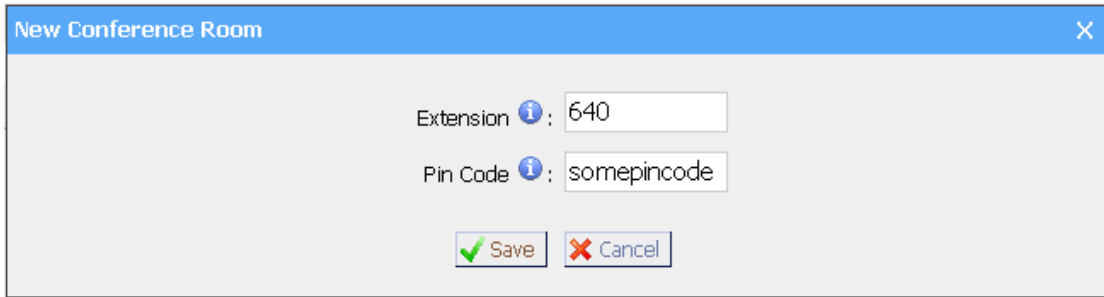


Figure 3-28

### 3.5.3 Time conditions

You can setup the business hours here.

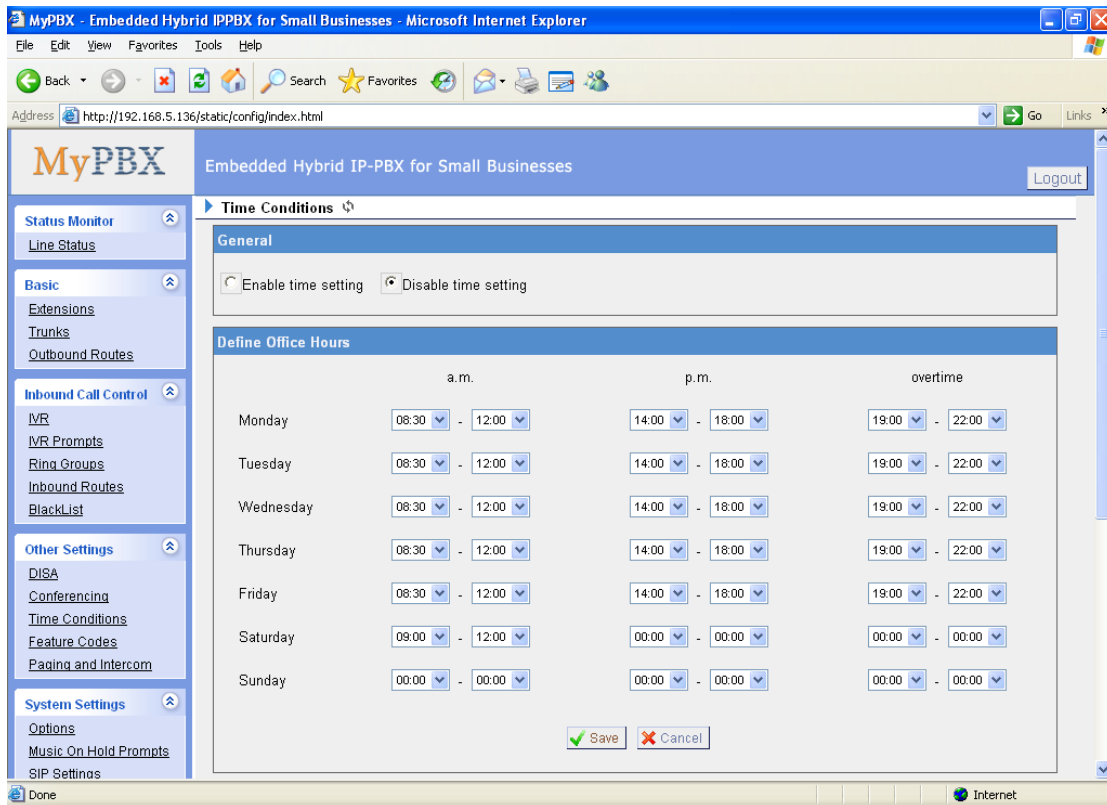


Figure 3-29

### 3.5.4 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, but it can be changed on the Feature Codes page)

**•Extension for Checking Voice Mail**

Users can check their voice mail by dialing \*2 on their phone (\*2 is default setting, but it can be changed on the Feature Codes page).

•**Attended Transfer**

Users may transfer an incoming call by dialing \*3 on their phone (\*3 is default setting, but it can be changed on the Feature Codes page).

•**Blind Transfer**

Users may blind transfer an incoming call by dialing \*03 on their phone (\*03 is default setting, but it can be changed on the Feature Codes page).

•**Call Pickup**

Users may pick up an incoming call by dialing \*4 on their phone (\*4 is default setting, but it can be changed on the Feature Codes page)

•**Pickup a specific extension**

Users may pick up a specific extension's incoming call by dialing \*04+extension number on their phone (\*04 is default setting, but it can be changed on the Feature Codes page)

•**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)

•**Call Parking on other extension**

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

•**Call Parking Duration**

The maximum duration that allowed parking a call, in seconds.

## 3. Call Forward Preferences

•**Reset to defaults**

Users may reset all call forward defaults by calling \*70 on their phone (\*70 is default setting, but it can be changed on the Feature Codes page).

**Note:** When the Reset to defaults feature is used, the call forward default values will be set as follows:

Always forward: Disabled

Busy forward to voice mail: Enabled

No answer forward to voice mail: Enabled

Do not disturb: Disabled

•**Enable always forward**

Users may enable always forward by calling \*71 on their phone (\*71 is default setting, but it can be changed on the Feature Codes page)

**·Disable always forward**

Users may disable always forward by calling \*071 on their phone (\*071 is default setting, but it can be changed on the Feature Codes page)

**·Enable busy forward**

Users may enable busy forward by dialing \*72 on their phone (\*72 is default setting, but it can be changed on the Feature Codes page)

**·Disable busy forward**

Users may disable busy forward by calling \*072 on their phone (\*072 is default setting, but it can be changed on the Feature Codes page)

**·Enable no answer forward**

Users may enable no answer forward by calling \*73 on their phone (\*73 is default setting, but it can be changed on the Feature Codes page)

**·Disable no answer forward**

Users may disable no answer forward by calling \*073 on their phone (\*073 is default setting, but it can be changed on the Feature Codes page)

**·Forward to number**

Users may forward the call to number by calling \*74 on their phone (\*74 is default setting, but it can be changed on the Feature Codes page)

**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 voice mail by calling \*074 on their phone (\*074 is default setting, but it can be changed on the Feature Codes page)

**·Enable do not disturb**

Users may enable do not disturb by calling \*75 on their phone (\*75 is default setting, but it can be changed on the Feature Codes page)

**·Disable do not disturb**

Users may disable do not disturb by calling \*075 on their phone (\*075 is default setting, but it can be changed on the Feature Codes page)

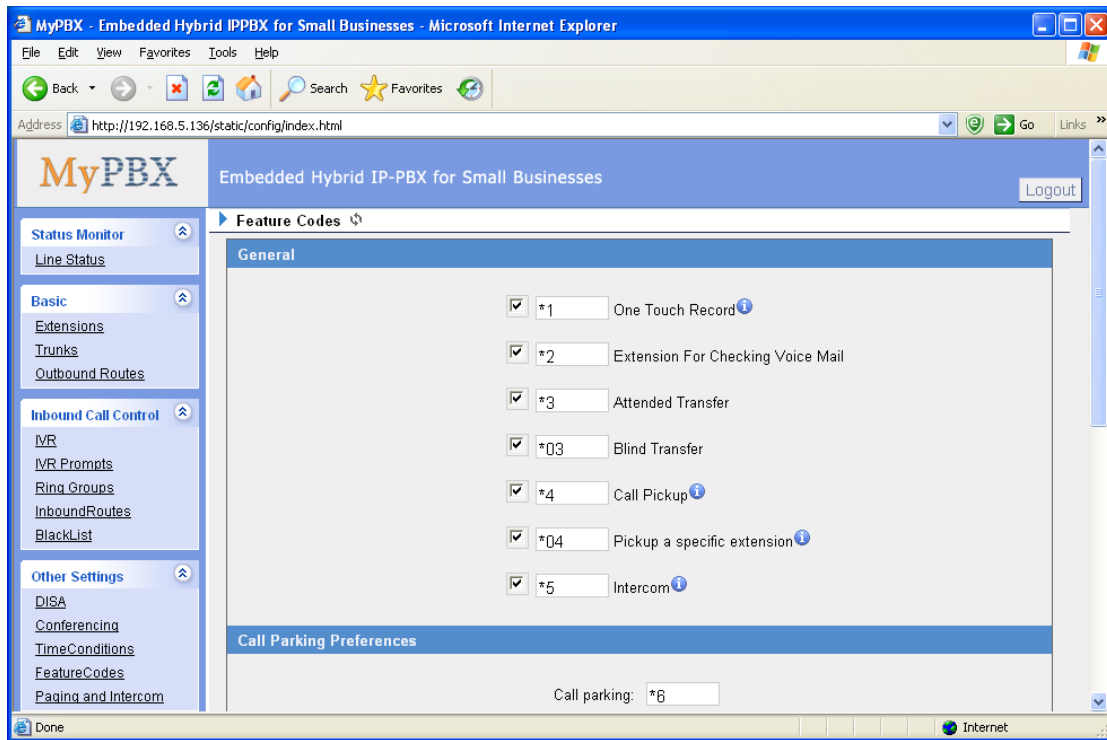


Figure 3-30

### 3.5.5 Paging and Intercom

Paging and Intercom is a feature reserved for SIP phones that support these features. It can be used to make an announcement over the speakerphone to a phone or group of phones. Targeted phones will not ring, but answer immediately into speakerphone mode. This section is for configuring group paging and group intercom. System-wide intercom access is configured through the Feature Codes page.

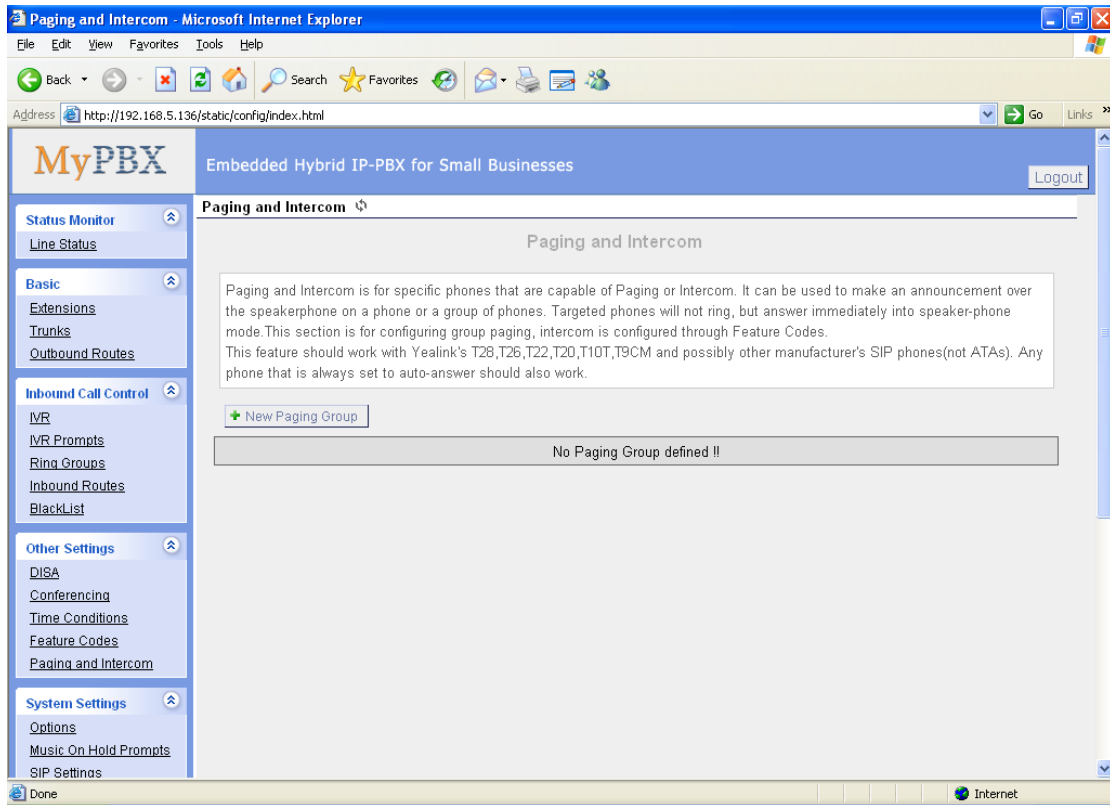


Figure 3-31

To create a new paging group, click the 'New Paging Group' button.

**·Number for this Paging group**

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

**·Duplex**

Paging is typically one way for announcements only. Checking this will make the paging duplex, allowing all phones in the paging group to be able to talk and be heard by all. This makes it like an "instant conference".



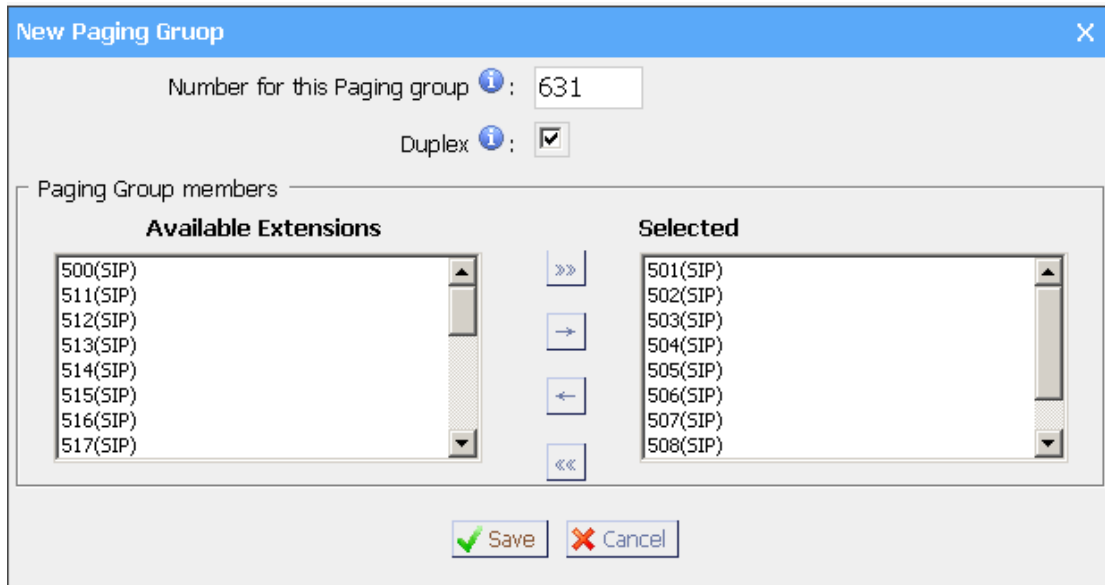


Figure 3-32

## 3.6 System Settings

### 3.6.1 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 choose your country or you nearest neighboring country for default Tones (Ex: dial tone, busy tone, ring tone etc.)

**Note:** please reboot the system to take it effect.

##### •HTTP bindport/Web Access Port

Port to bind to for HTTP sessions (default is 80)

#### 2) Extension Preferences

##### •User Extensions

The default value is 500 to 616

**•Paging Group Extensions**

The default value is 620 to 629

**•Ring 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 689

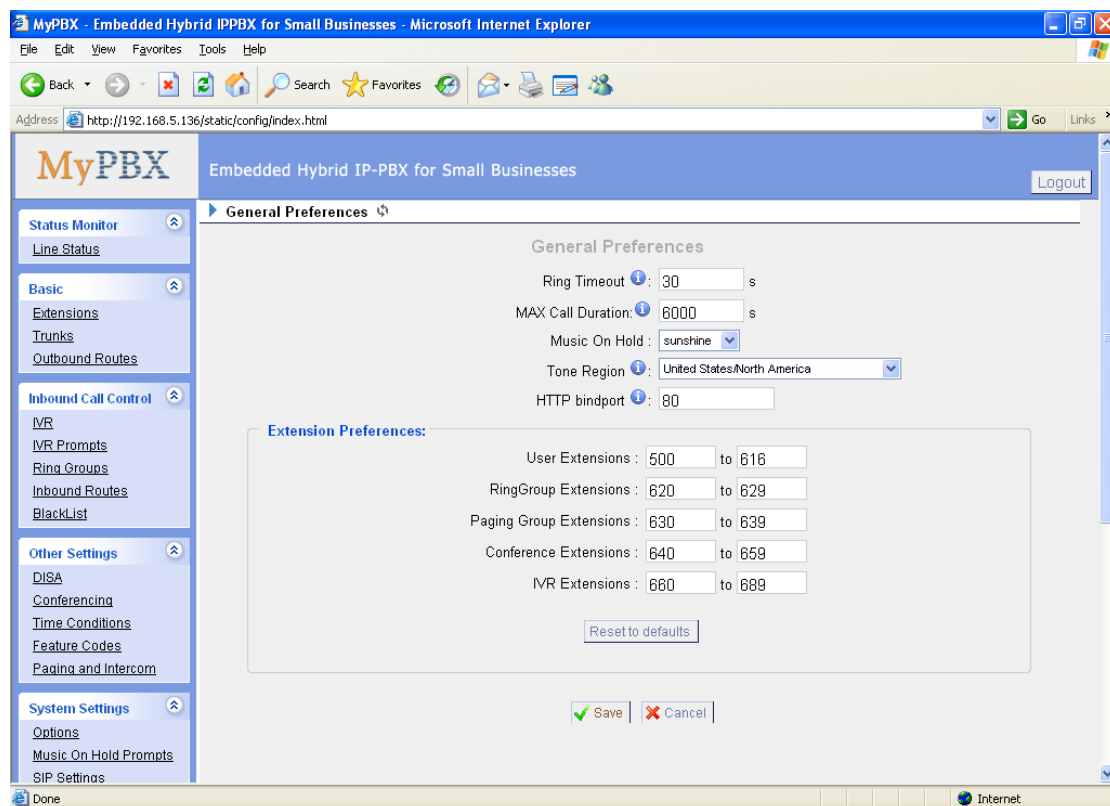


Figure 3-33

**3.6.2 Music on Hold Prompts/Audio Files**

The administrator can upload on hold music as follows:

- 1) Click 'Upload Music On hold File
- 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

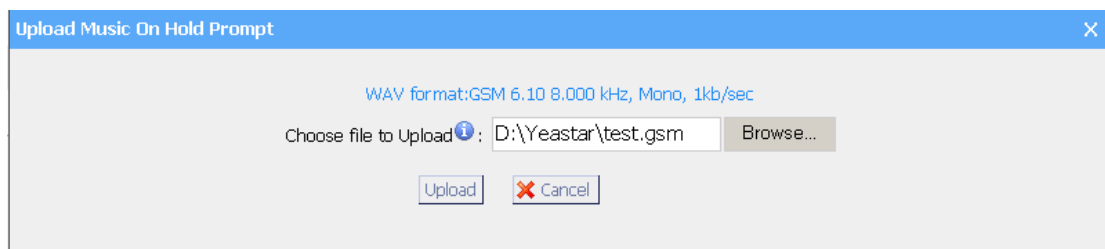


Figure 3-34

### 3.6.3 SIP Settings

#### 1) General

##### •UDP Port

UDP Port to bind to, SIP standard port is 5060.

##### •RTP Port Start

Beginning of RTP port range

##### •RTP Port End

End of RTP port range

##### •DTMF Mode

Set default DTMF mode for sending DTMF. Default setting: rfc2833

##### •Max Registration/Subscription Time

Maximum duration (in seconds) of incoming registration allowed. Default is 3600 seconds.

##### •Min Registration/Subscription Time

Minimum duration (in seconds) of incoming registration allowed. Default is 60 seconds.

##### •Default Incoming/Outgoing Registration Time

Default duration (in seconds) of incoming/outgoing registration

##### •Registerattempts

Number of SIP REGISTER messages to send to a SIP Registrar before giving up. Default 0 (no limit).

##### •Registertimeout

Number of seconds to wait for a response from a SIP Registrar before classifying the SIP REGISTER has timed out. Default is 20 seconds.

##### •Videosupport

Support for SIP video or no. Default is yes.

## 2) NAT

### •External IP

Address displayed in outbound SIP messages when behind a NAT.

### •External Host

If you specify an external host name, MyPBX will perform DNS queries on the host name periodically. Not recommended for production environments! Use External IP instead.

### •External Refresh

Set how often to refresh the external host if used. You may specify a local network in the field below.

### •Local Network Address

'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.

### •NAT Mode

Global NAT settings (Affects all peers and users);

Yes = Always ignore info and assume NAT;

No = Use NAT mode only according to RFC3581;

Never = Never attempt NAT mode or RFC3581 support;

Route = Assume NAT, don't send report.

### •Allow RTP Reinvite

MyPBX by default tries to redirect the RTP media stream (audio) directly from the caller to the callee. Some devices do not support this (especially if one of them is behind a NAT).

## 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.

**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.729, H261, H263, H263p, H264.**

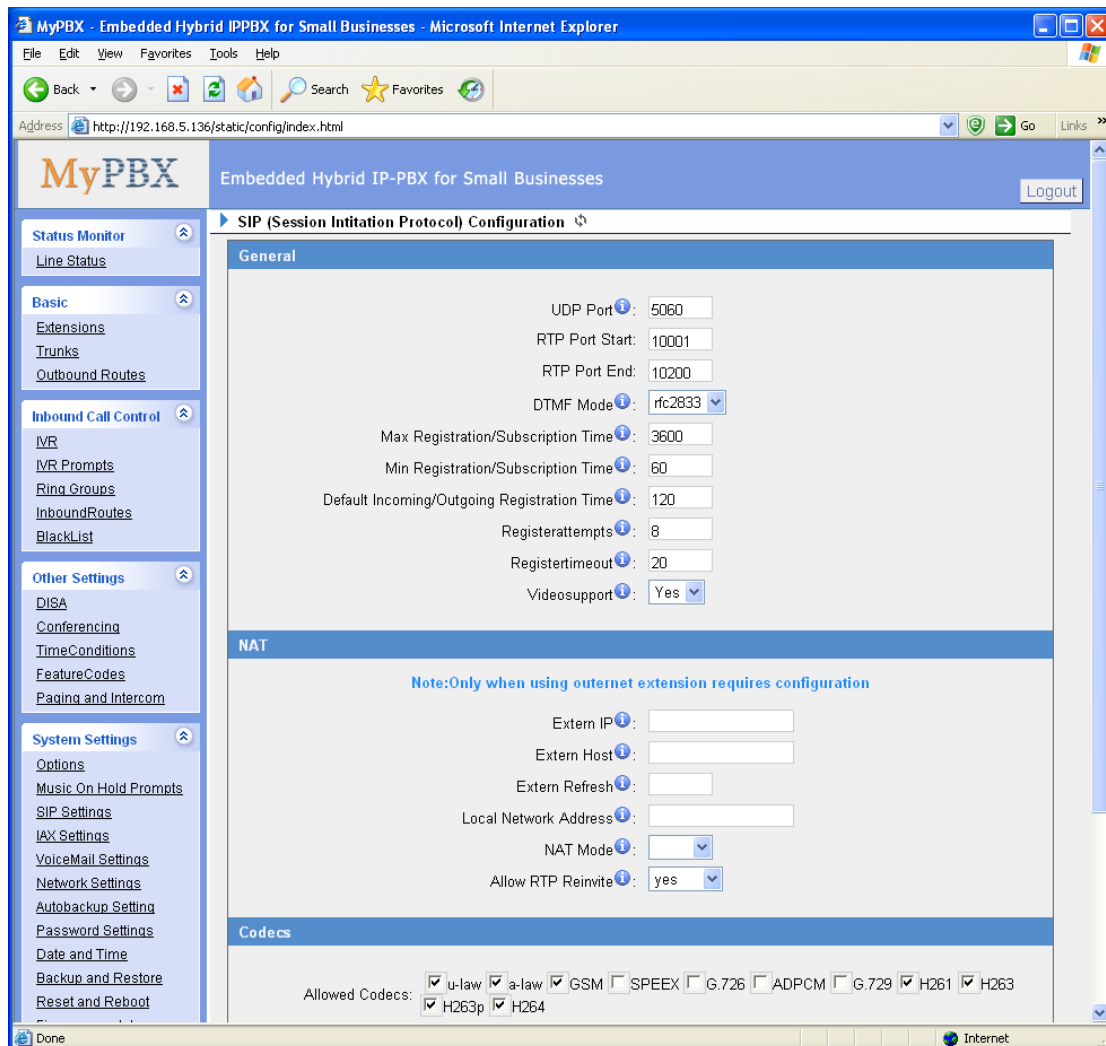


Figure 3-35

## 3.6.4 IAX Settings

### 1) General

#### • Bind Port

UDP Port to bind to, IAX2 standard port is 4569.

#### • Bandwidth

Low/medium/high with this option you can control which codec to be used.

#### • Min Registration Expire

Minimum duration (in seconds) of incoming registration allowed. Default is 60

seconds.

•**Max Registration Expire**

Maximum duration (in seconds) of incoming registration allowed. 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.

**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.729, H261, H263, H263p, H264.**

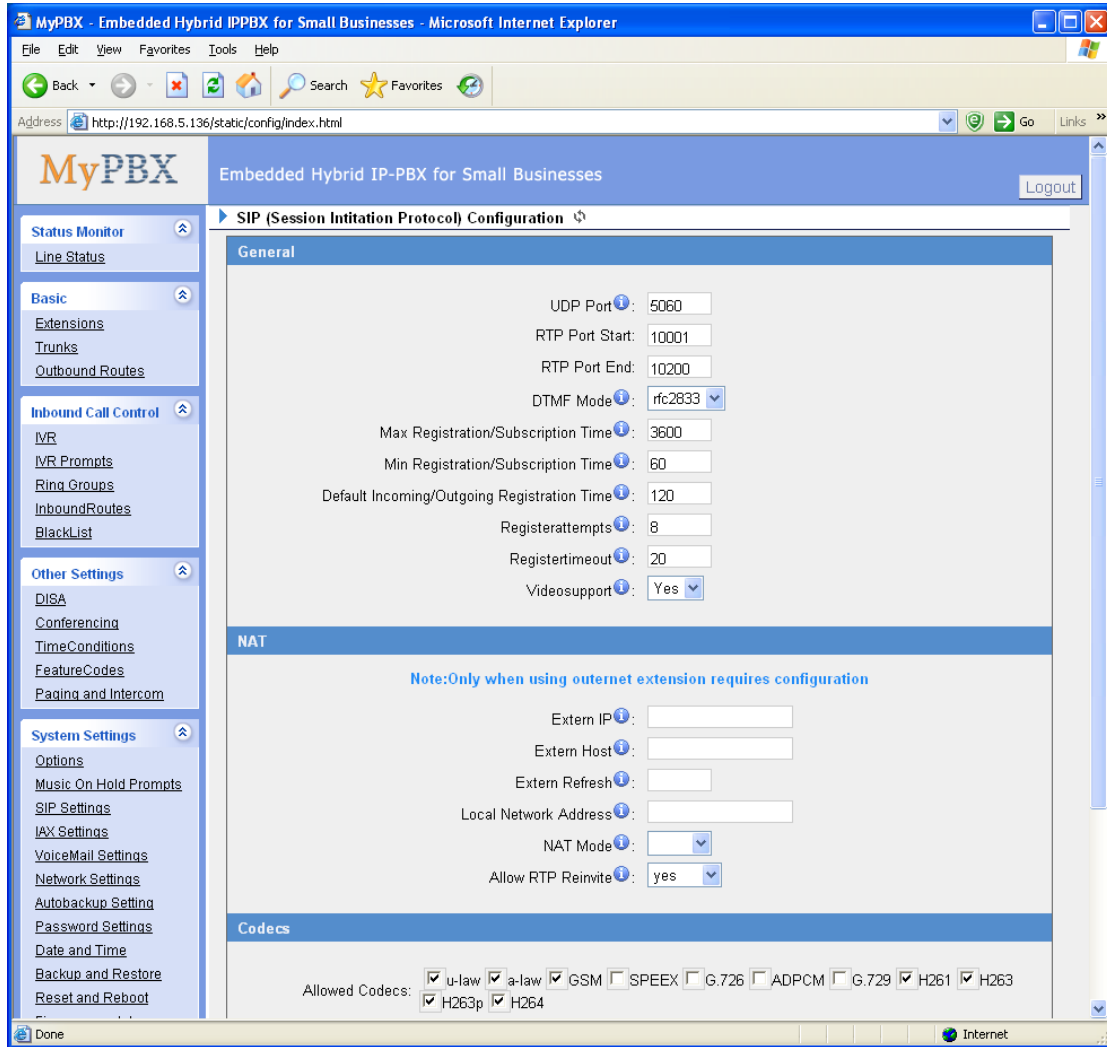


Figure 3-36

### 3.6.5 Voicemail Settings

#### 1) General Voicemail Settings

**·Maximum messages per folder**

This select box sets the maximum number of messages that a user may have in any of their folders.

**·Max message time**

This select box sets the maximum duration of a voicemail message in minutes. Message recording will not occur for times greater than this amount.

**·Min message time**

This select box sets the minimum duration of a voicemail message in seconds. Messages below this threshold will not be saved.

**·Ask caller to press 5**

If this option is set, the caller will be asked to press 5 before he can leave a message. Setting this option will minimize auto-dialer telemarketing machines from accessing voice mail.

**·Say 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.

**·Say message duration**

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

**·Say 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 callers to Review**

Checking this option allows the caller to review their message before it is submitted as a new voicemail message.

**2) SMTP Settings for Voicemail****·E-mail Address**

The E-mail Address that MyPBX will use to send voicemail.

**·Password**

he password for the email address used above

**·SMTP Server**

The IP address or hostname of an SMTP server that the MyPBX will connect to in order to send voicemail messages via email, i.e. mail.yourcompany.com.

**·Port**

SMTP Port: the default value is 25.

**·Use SSL/TLS to send secure message to server**

If the server of sending email needs to authenticate the sender, you need to select the check box.

**Note:** Must be selected for Gmail or exchange server.

After filling out the above information, you can click on the 'Test Account Settings' button to check whether the setup is OK.

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



2) If test failed, please check the above information is correct or network is proper.

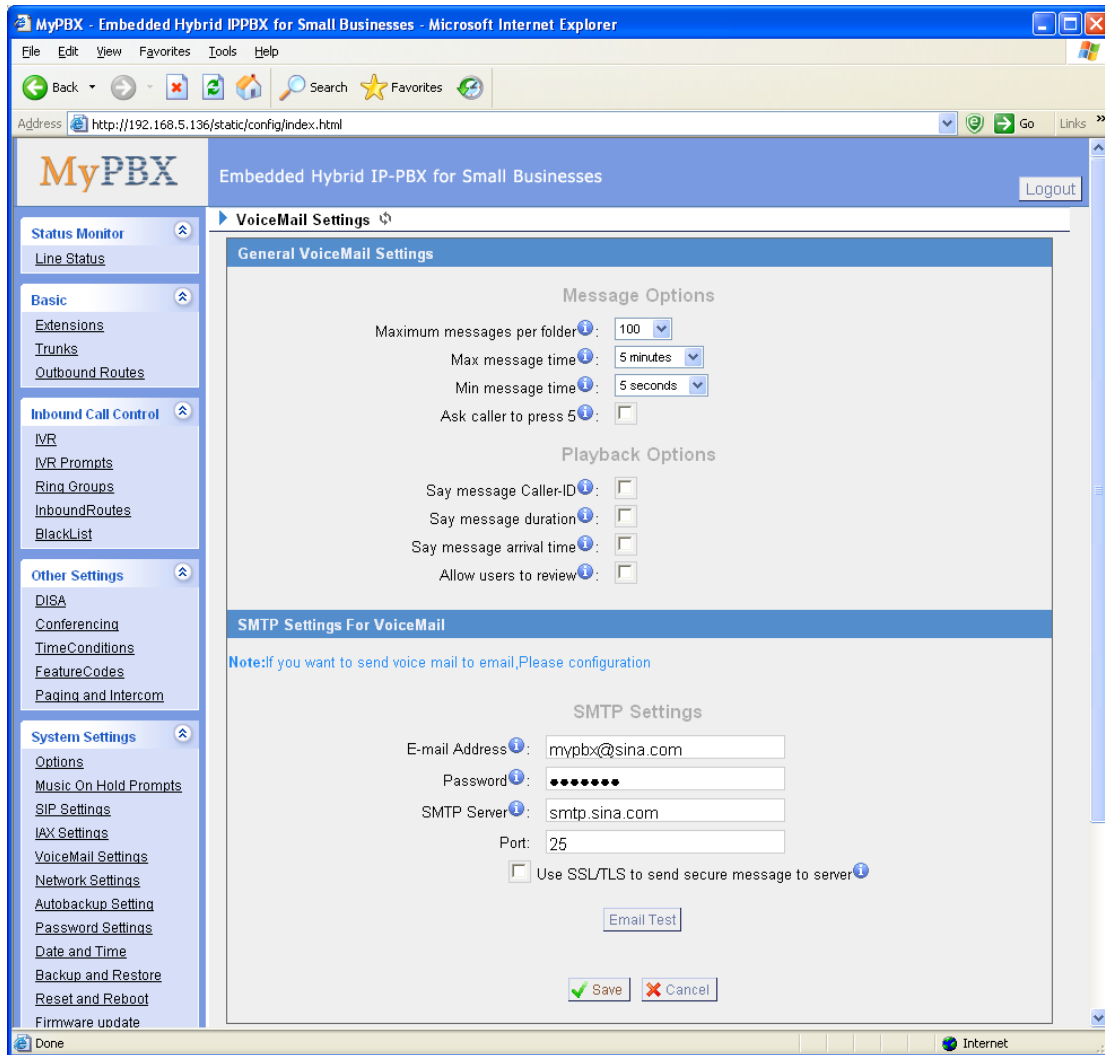


Figure 3-37

### 3.6.6 Network Setting

#### •DHCP

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

#### •Hostname

Set the host name for MyPBX.

#### •IP Address

Set the IP Address for MyPBX.

#### •Subnet Mask

Set the subnet mask for MyPBX.

**•Gateway**

Set the gateway for MyPBX.

**•Primary DNS**

Set the primary DNS for MyPBX.

**•Secondary DNS**

Set the secondary DNS for MyPBX.

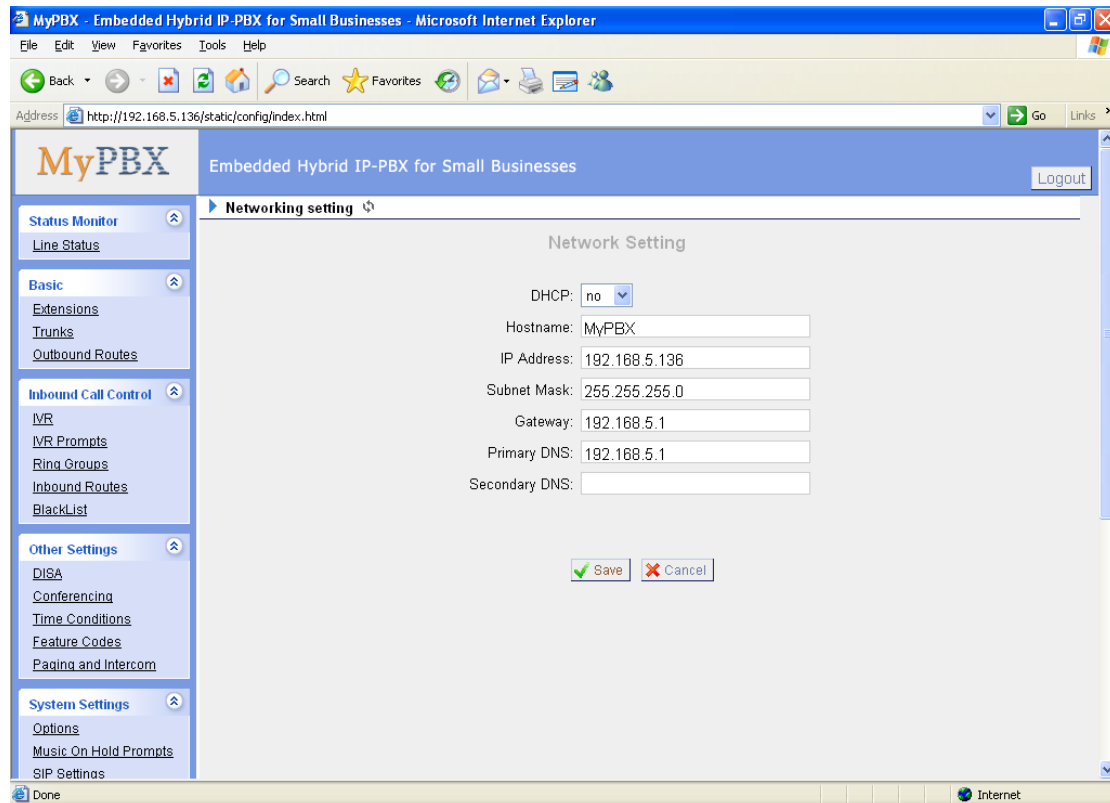


Figure 3-38

**3.6.7 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.

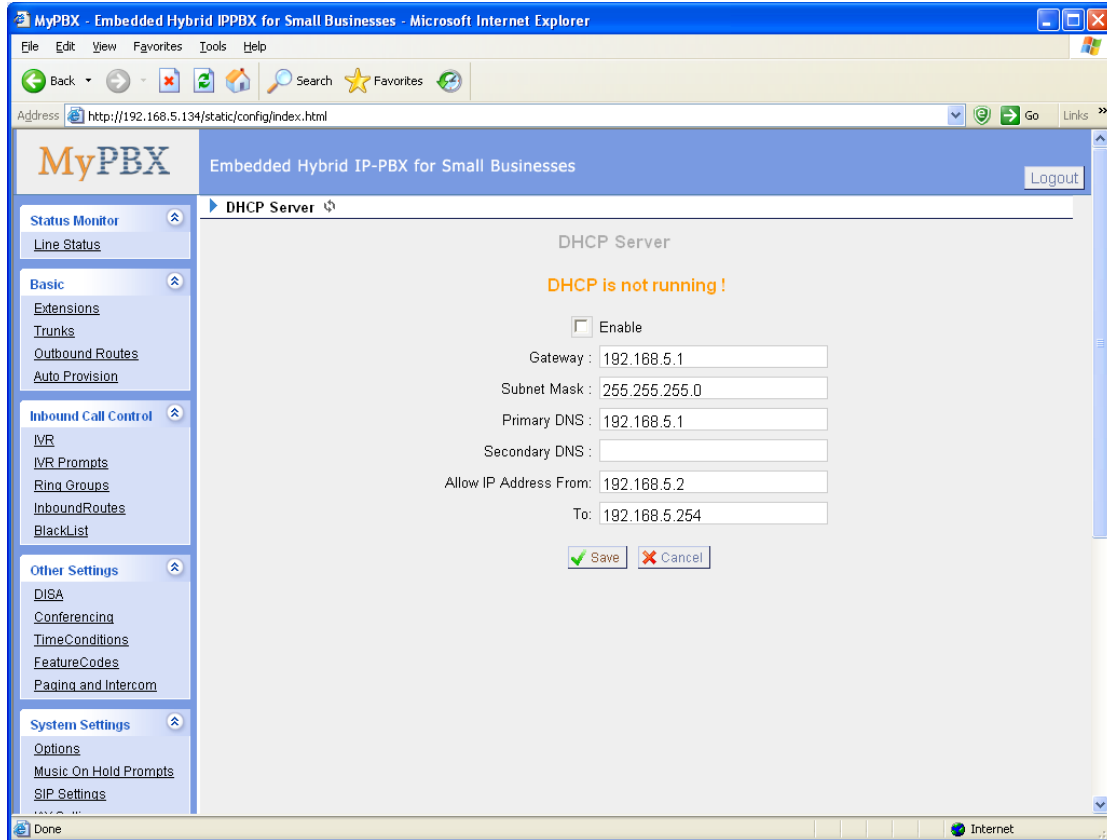


Figure 3-39

### 3.6.8 Autobackup setting

Autobackup extends the allocated disk space for backing up critical files. When properly configured, MyPBX will move all qualified files to a Windows PC every 30 minutes. For the voicemail files and recoding files, they must be created before Autobackup has been configured. For the CDR files, the call log file size must be more than 8MB.

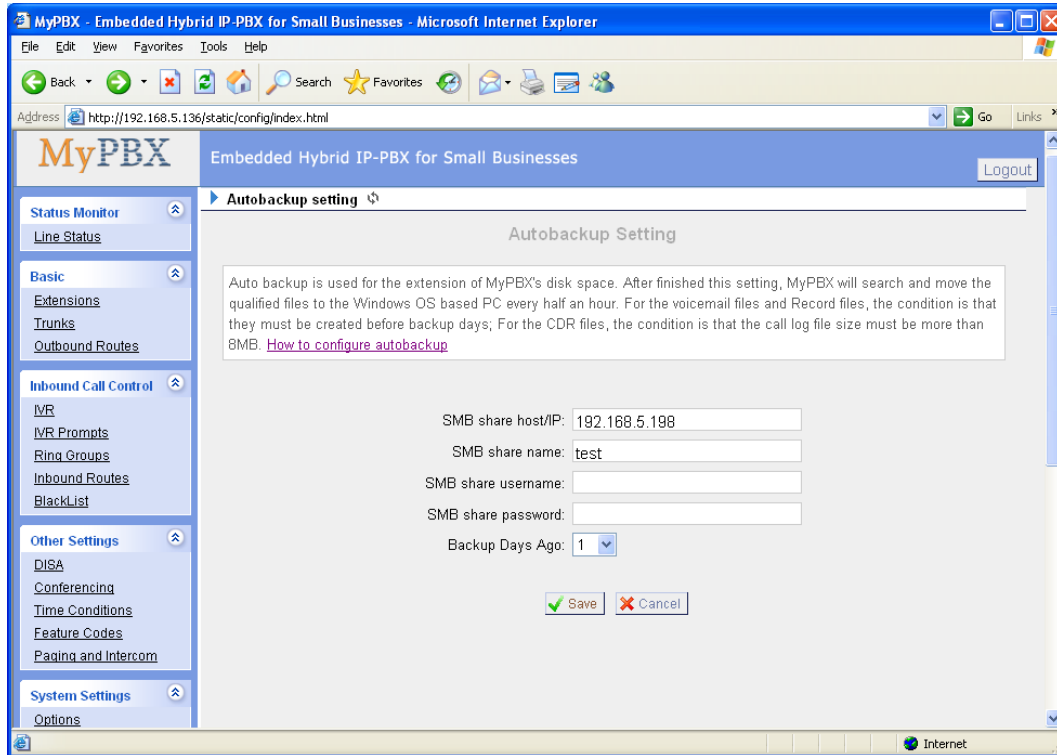


Figure 3-40

### 3.6.9 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.

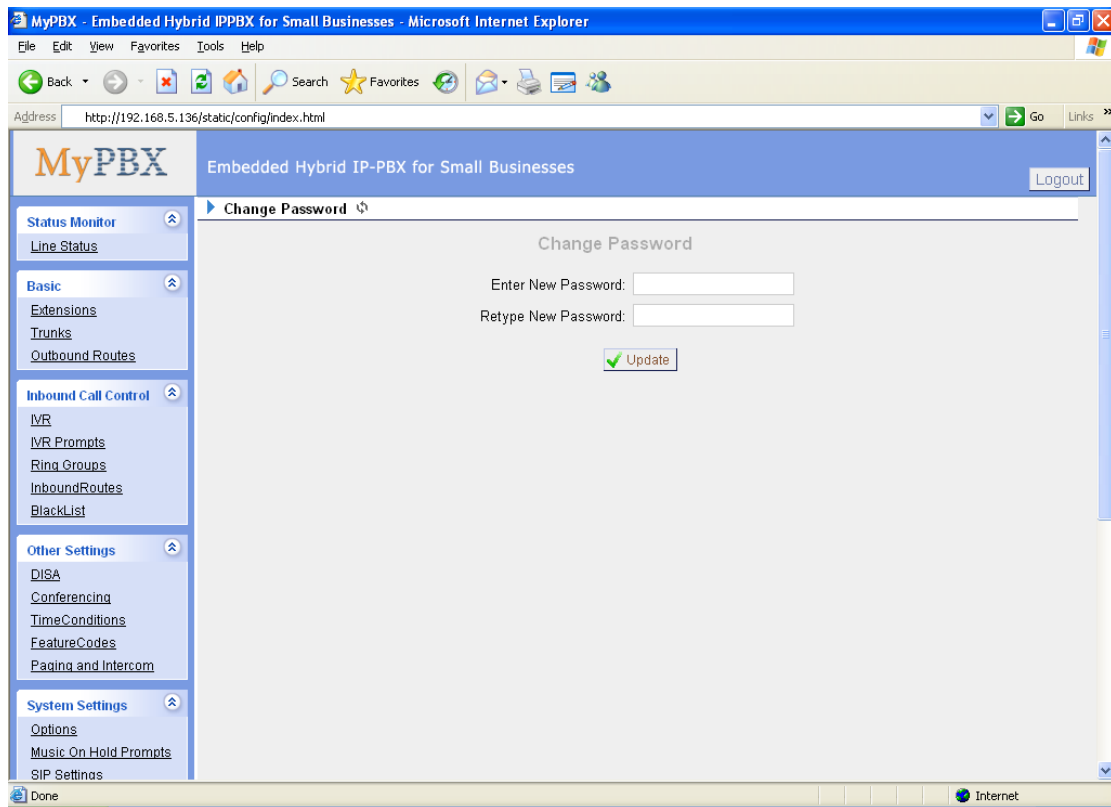


Figure 3-41

### 3.6.10 Date and Time

Set the date and time for MyPBX.

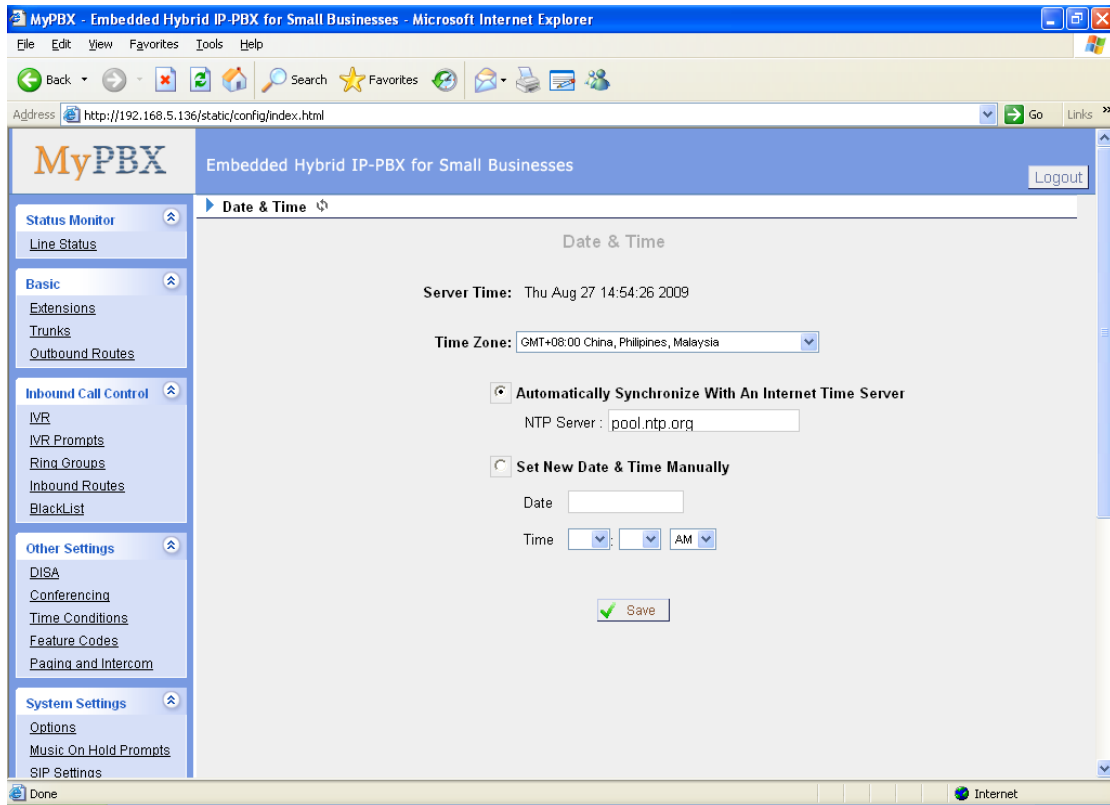


Figure 3-42

### 3.6.11 Backup and Restore

Backup / Restore MyPBX's Configuration.

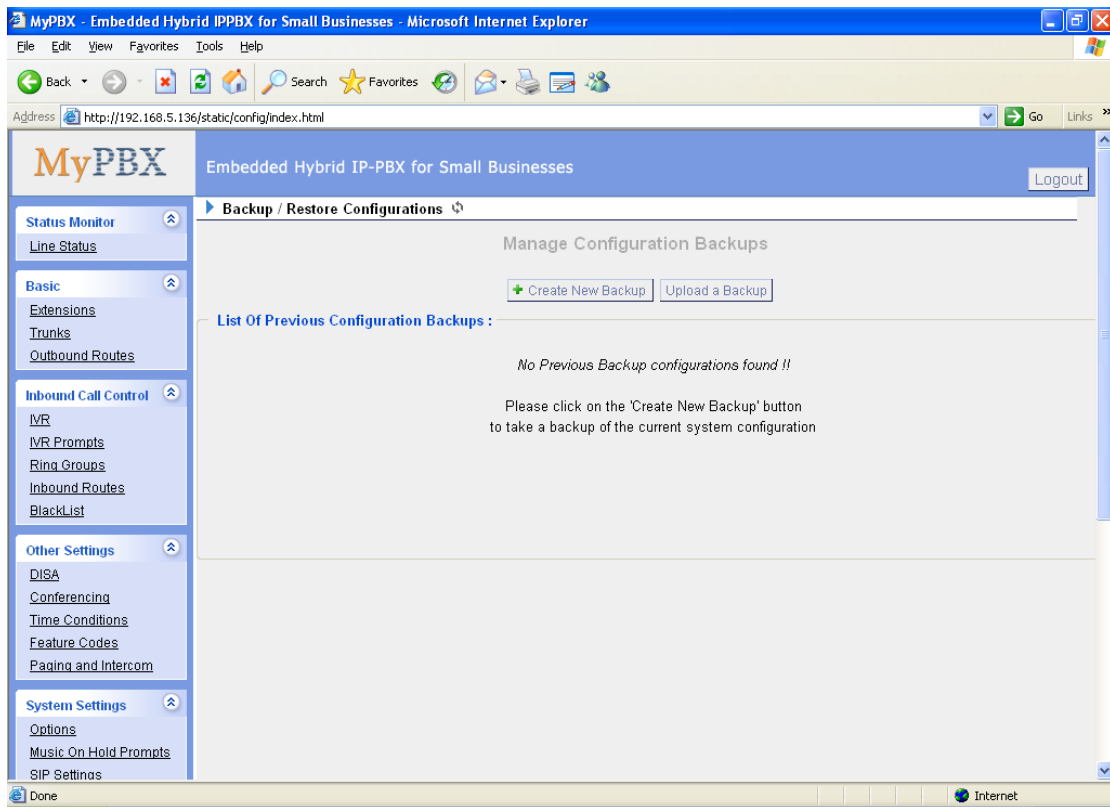


Figure 3-43

## 3.6.12 Reset and Reboot

### ·Reboot Appliance

**Warning:** Rebooting the unit will terminate all active calls.

### ·Reset to Factory Defaults

**Warning:** By resetting your System to factory defaults, you will lose all your configurations! Do not power down the MyPBX during factory default reset operations. Failure to heed this warning may result in critical damage to the unit. The unit is successfully reset when the “run” lamp begins to flash.

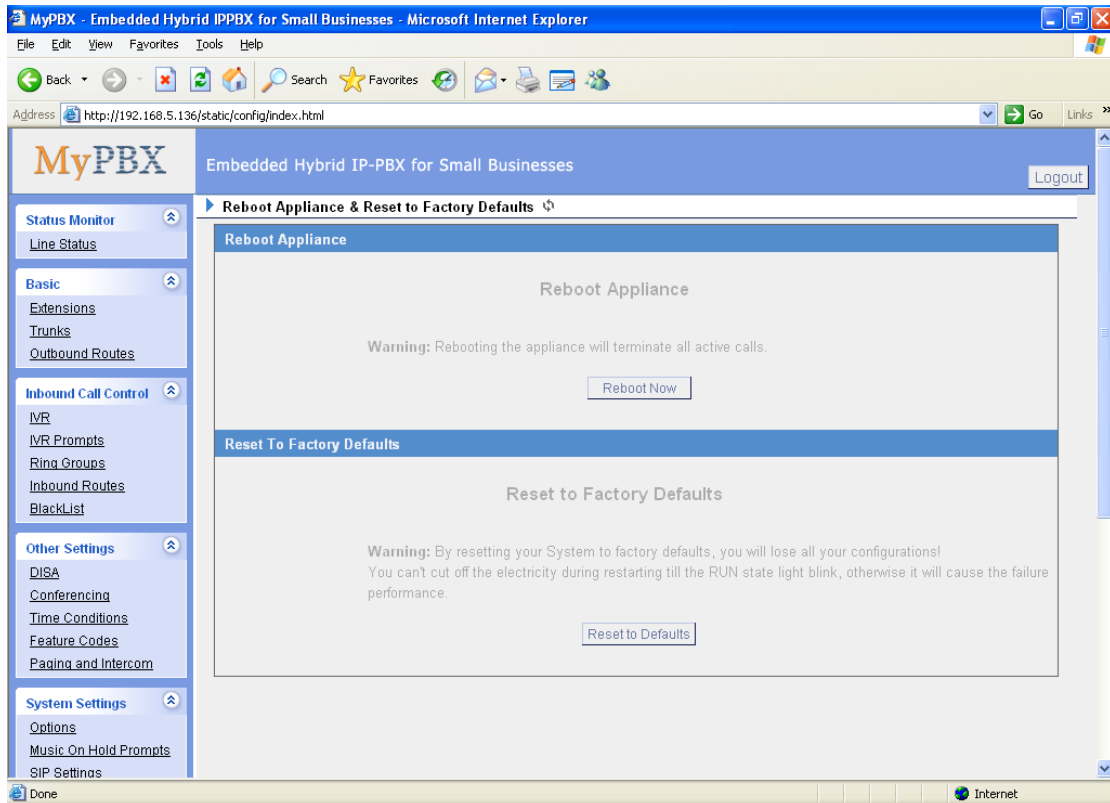


Figure 3-44

### 3.6.13 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.



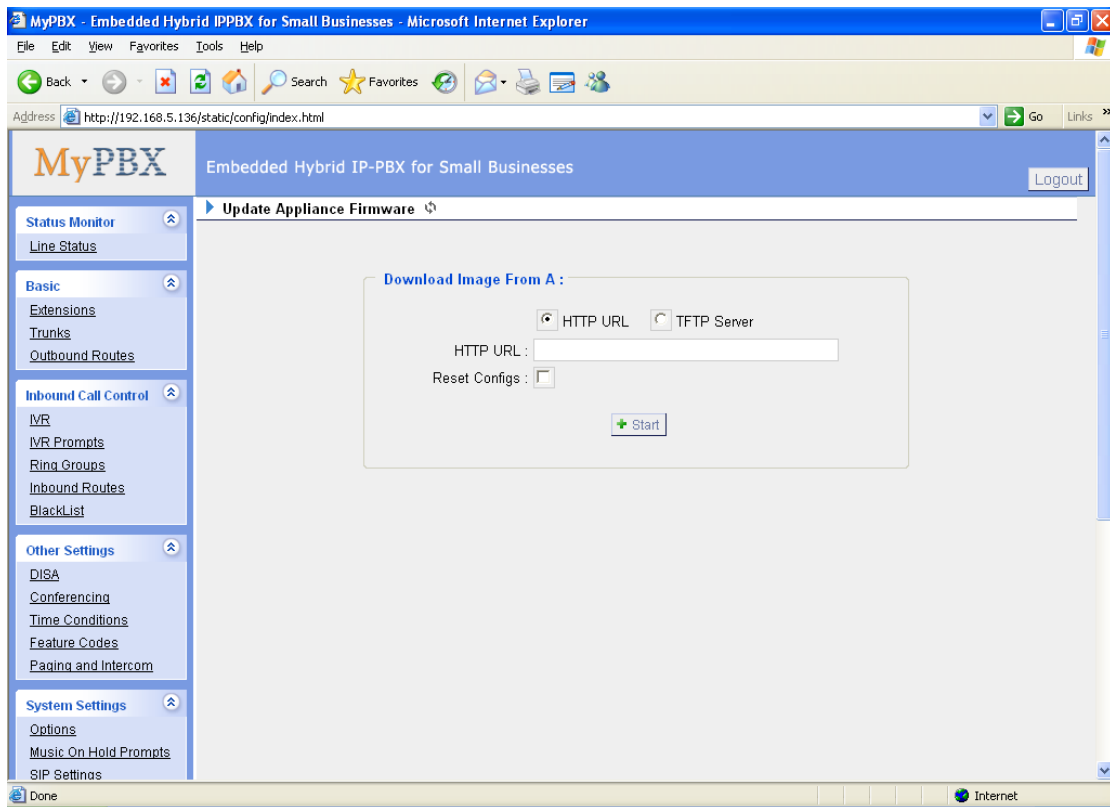


Figure 3-45

## 3.7 Reports

### 3.7.1 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.

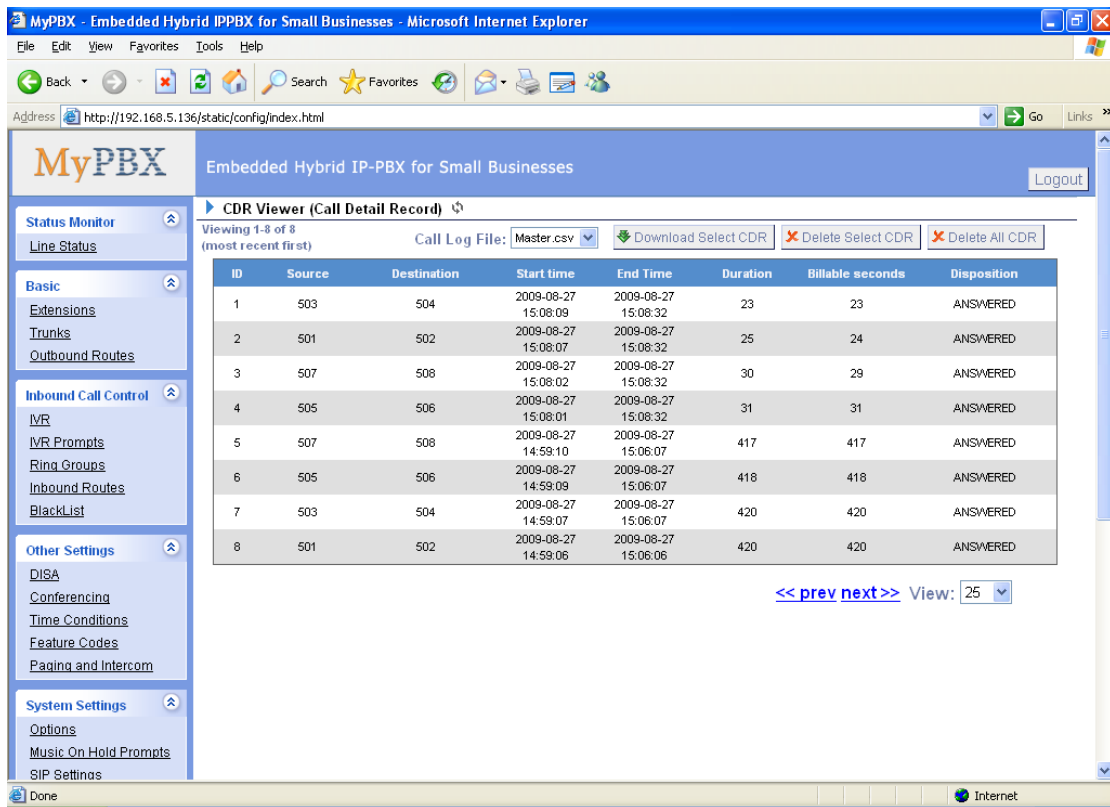


Figure 3-46

### 3.7.2 System Info

#### General:

Information about hardware version, firmware version and system uptime.

#### Network:

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

#### Disk Usage:

Disk usage information.

#### Memory Usage:

Memory usage information.

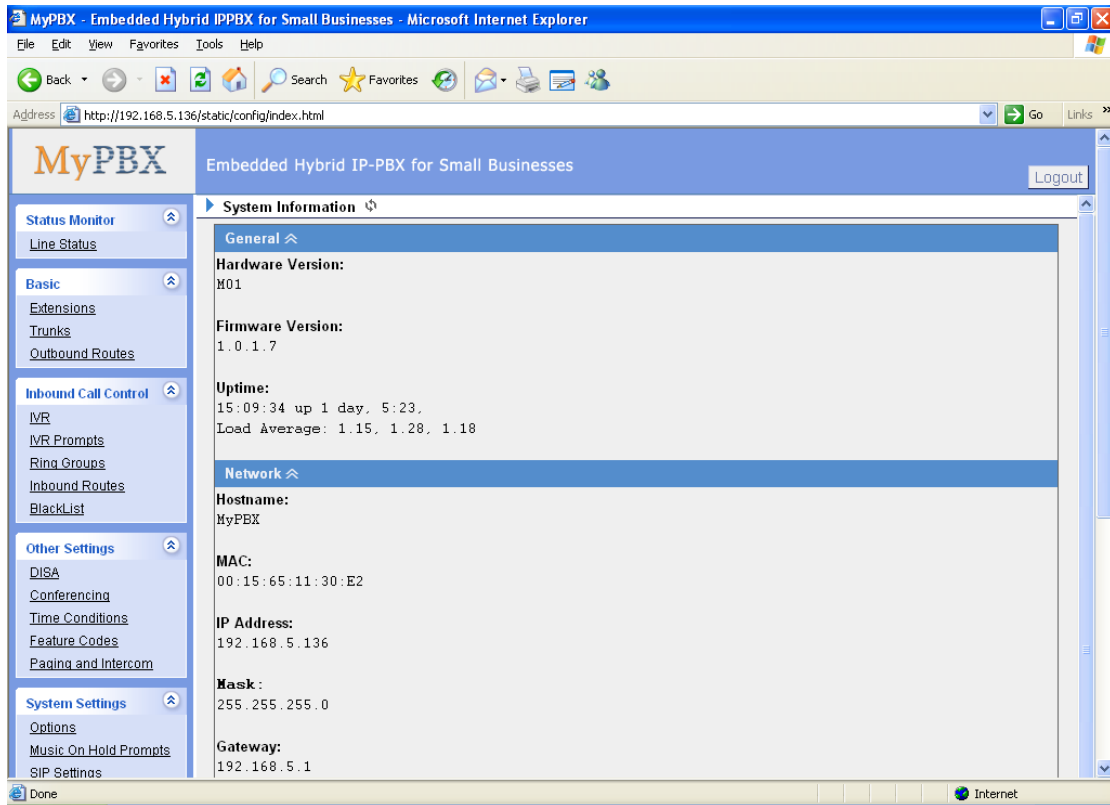


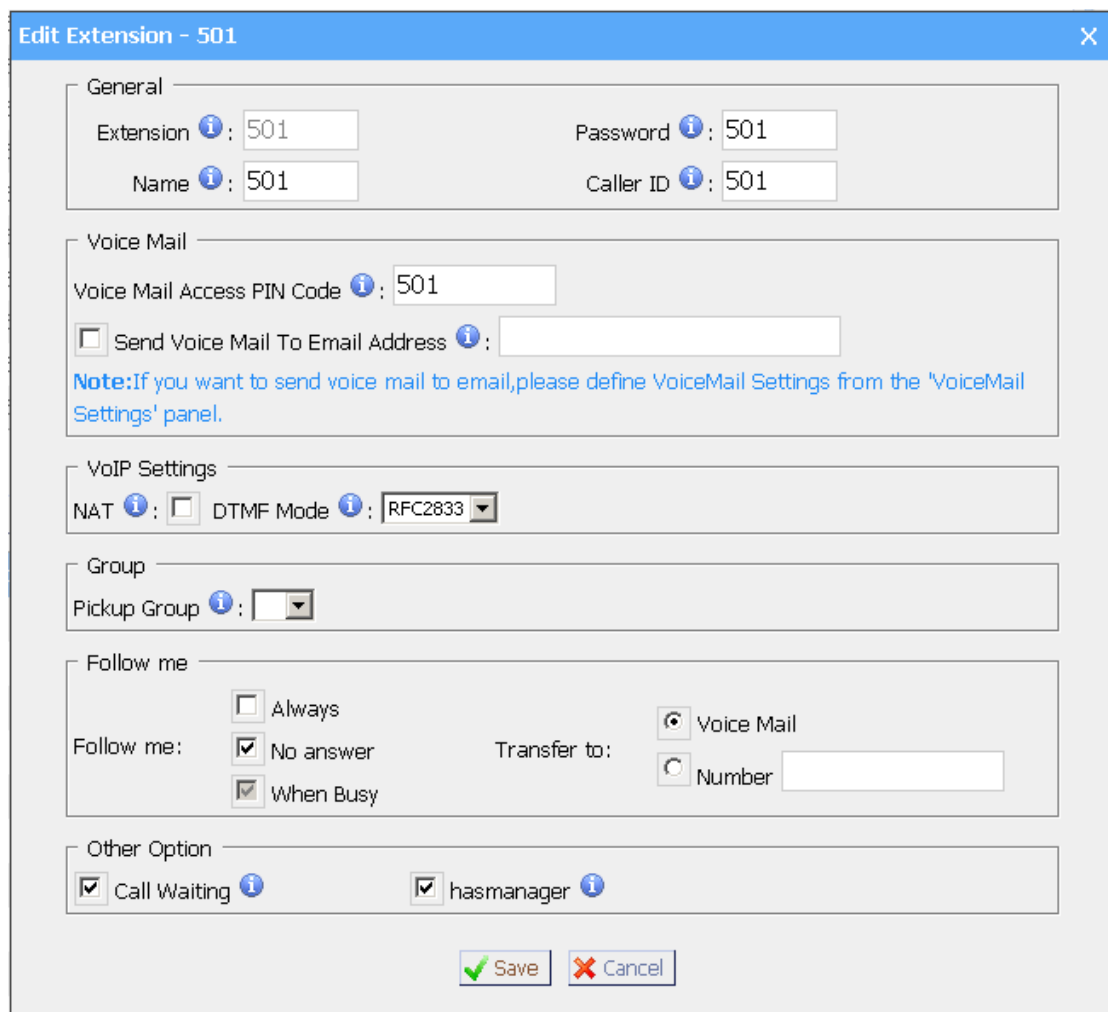
Figure 3-47

## 4. Access MRI

MRI (MyPBX Recording Interface: Check Voice Mail).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 hasmanager(web access) option must be checked before the associated user can log into MRI.



The screenshot shows the 'Edit Extension - 501' configuration page. It is divided into several sections:

- General:** Extension: 501, Name: 501, Password: 501, Caller ID: 501.
- Voice Mail:** Voice Mail Access PIN Code: 501. There is an unchecked checkbox for 'Send Voice Mail To Email Address' with an empty text field next to it. A note below states: "Note: If you want to send voice mail to email, please define VoiceMail Settings from the 'VoiceMail Settings' panel."
- VoIP Settings:** NAT:  DTMF Mode: RFC2833 (dropdown menu).
- Group:** Pickup Group: (empty dropdown menu).
- Follow me:** Follow me:  Always,  No answer,  When Busy. Transfer to:  Voice Mail,  Number (empty text field).
- Other Option:**  Call Waiting,  hasmanager.

At the bottom, there are 'Save' and 'Cancel' buttons.

Figure 4-1

## 4.2 User login

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

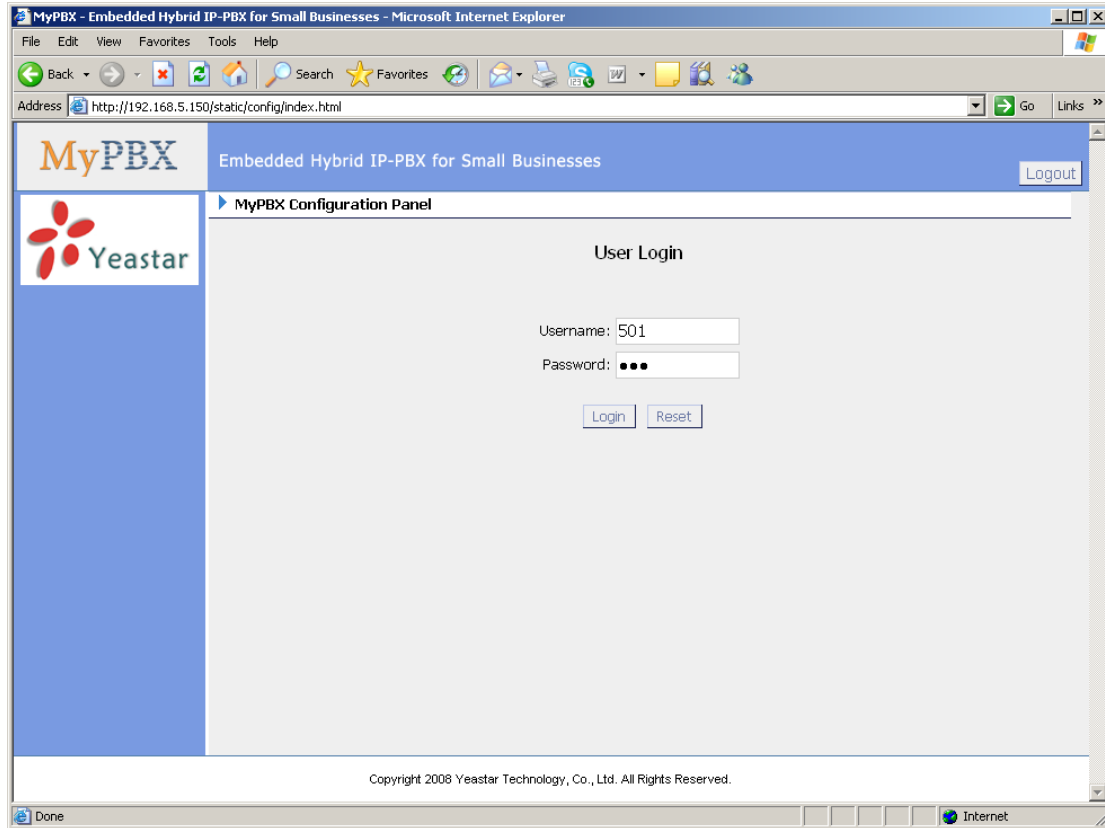


Figure 4-2

### 4.3 Voicemail

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

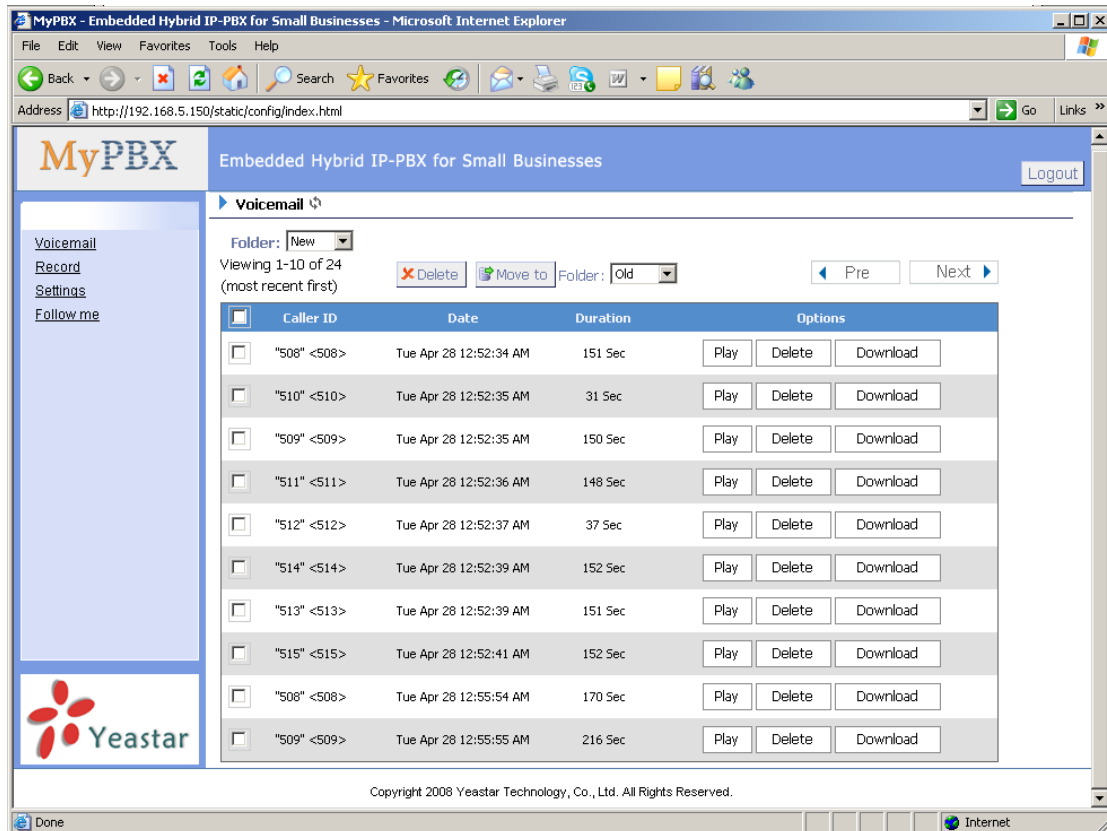
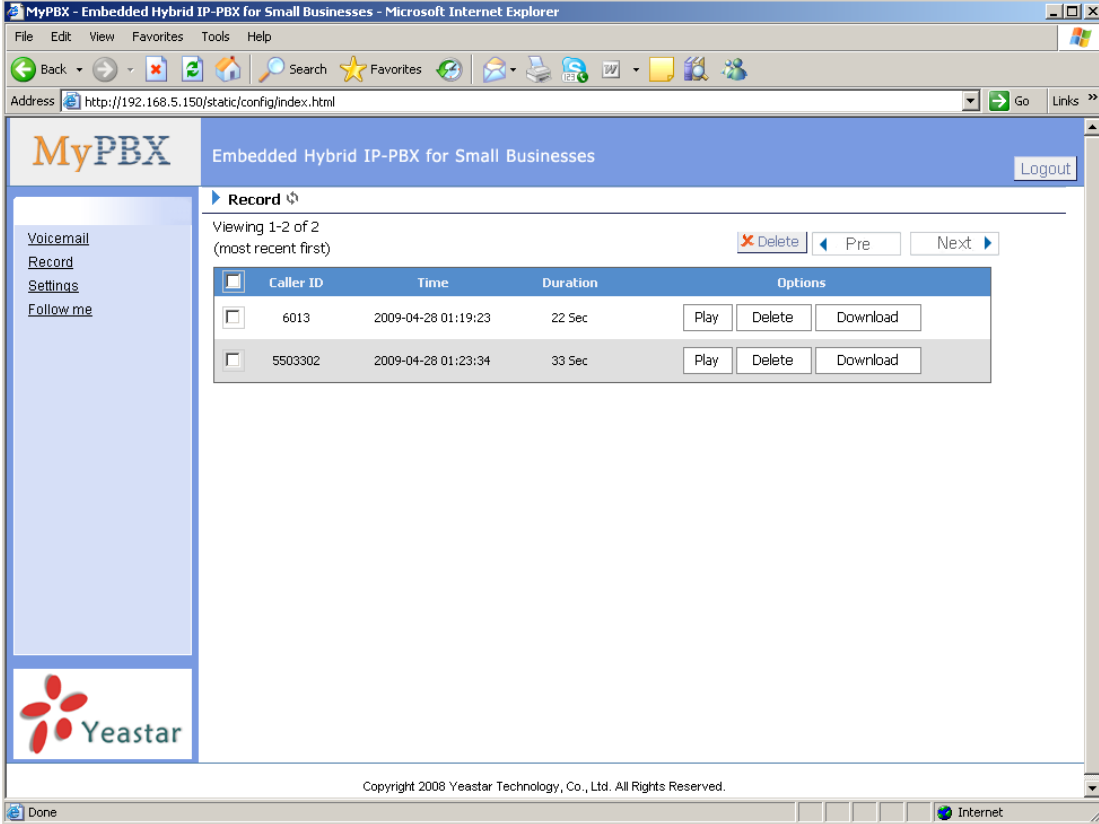


Figure 4-3

## 4.4 Record

Users can play, delete and download recorded files here



The screenshot shows the MyPBX web interface in Microsoft Internet Explorer. The browser address bar shows `http://192.168.5.150/static/config/index.html`. The page title is "MyPBX - Embedded Hybrid IP-PBX for Small Businesses". The main content area is titled "Record" and shows "Viewing 1-2 of 2 (most recent first)". There are navigation buttons for "Delete", "Pre", and "Next". A table lists the recorded calls:

<input type="checkbox"/>	Caller ID	Time	Duration	Options
<input type="checkbox"/>	6013	2009-04-28 01:19:23	22 Sec	Play Delete Download
<input type="checkbox"/>	5503302	2009-04-28 01:23:34	33 Sec	Play Delete Download

The footer contains the text "Copyright 2008 Yeastar Technology, Co., Ltd. All Rights Reserved." and the Yeastar logo.

Figure 4-4

## 4.5 Settings

Voicemail password: new voicemail box password.

Enter again to confirm: confirm new voicemail box password.

Send Voice Mail To Email Address: Set the email address that voicemail messages will be sent to.

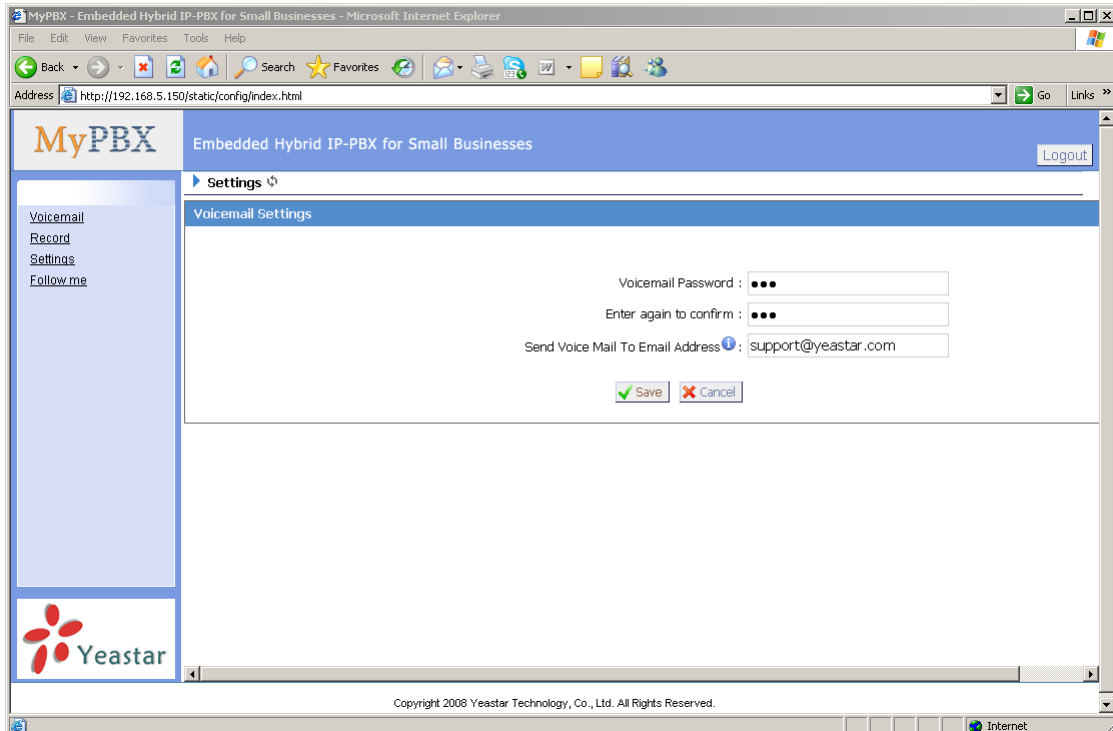


Figure 4-5

## 4.6 Follow me

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



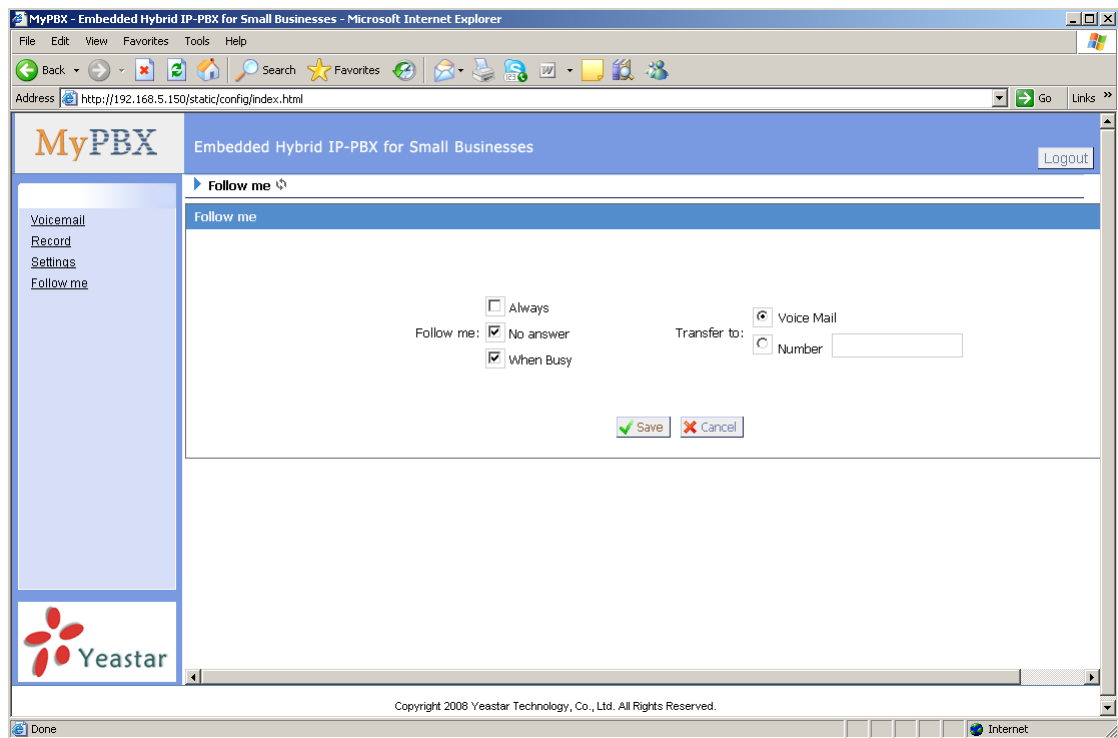
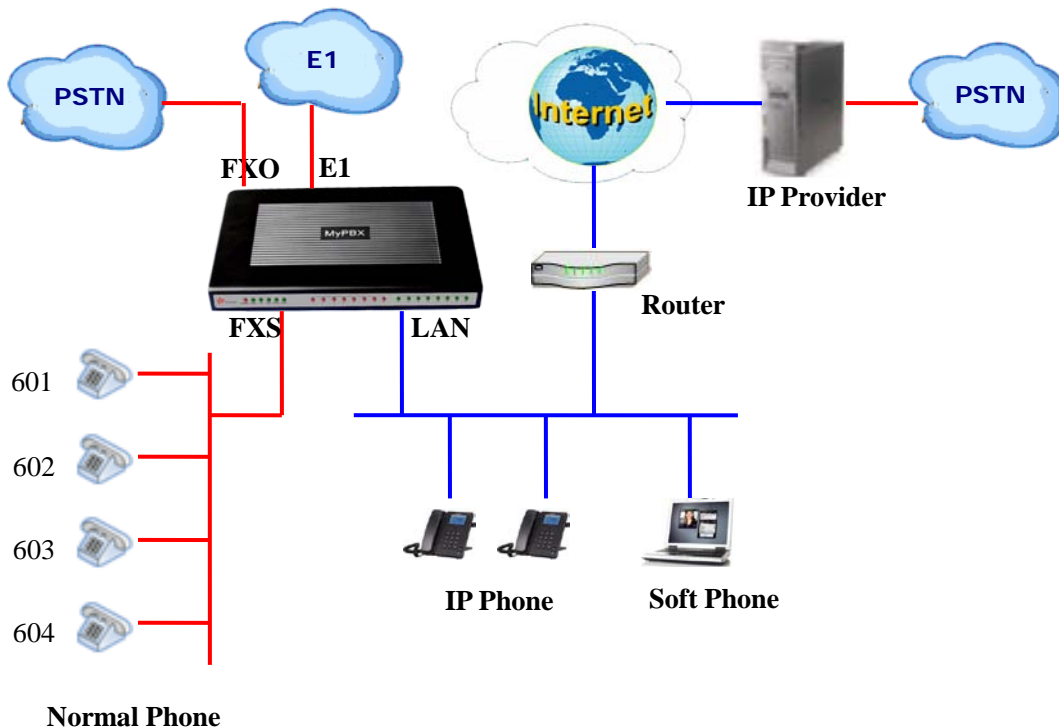


Figure 4-6

## 5. Use MyPBX



### 5.1 Make outbound call

To make an outbound call, we need to add trunk/service provider first. There are two types of service provider:

- **Analog Trunk Provider:** FXO ports of MyPBX, connected to a local PSTN
- **SIP Trunk Provider:** Connected to remote SIP VOIP 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.

#### What are E1, T1?

**E1:** E1 is the European format for digital transmission.

E1 carries signals at 2 Mbps (32 channels at 64Kbps, with 2 channels reserved for signaling and controlling), versus the T1, which carries signals at 1.544 Mbps (24 channels at 64Kbps). E1 and T1 lines may be interconnected for international use.

**T1:** A dedicated phone connection supporting data rates of 1.544Mbps per second. A T-1 line actually consists of 24 individual channels, each of which supports 64Kbits per second. Each 64Kbit/second channel can be configured to carry voice or data traffic. Most telephone companies allow you to buy just some of these individual channels, known as fractional T-1 access.

### 5.1.1 Sample Routing via PSTN Trunk

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

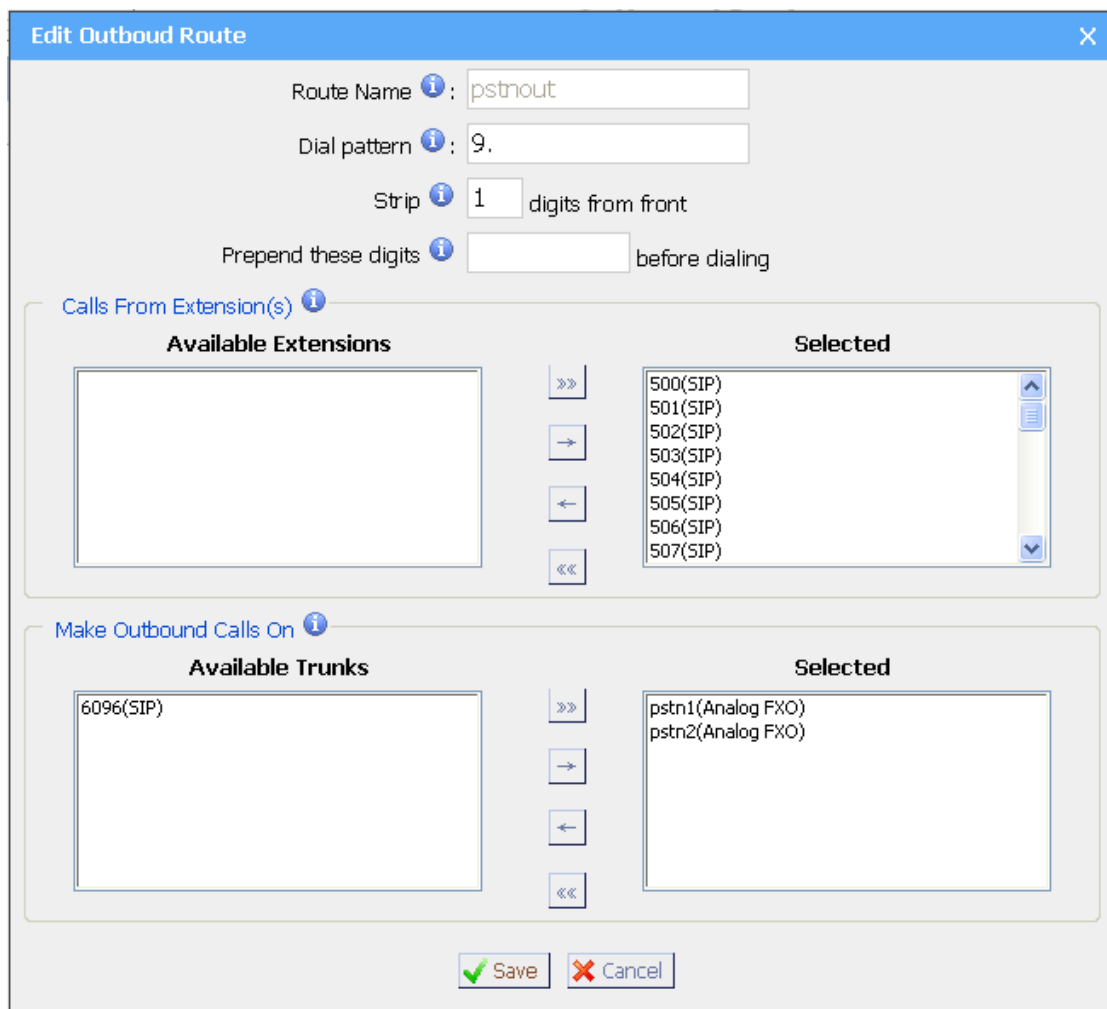


Figure 5-1

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

to the PSTN (port 1-2).

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

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

### 5.1.2 Sample Routing via a VoIP Trunk

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

#### 1. Add VoIP service provider

Before we do add this, please make sure you have a VoIP service provider account.

Trunks → SIP Trunk → Add SIP Trunk

Enter your account information on this page, and click Save.

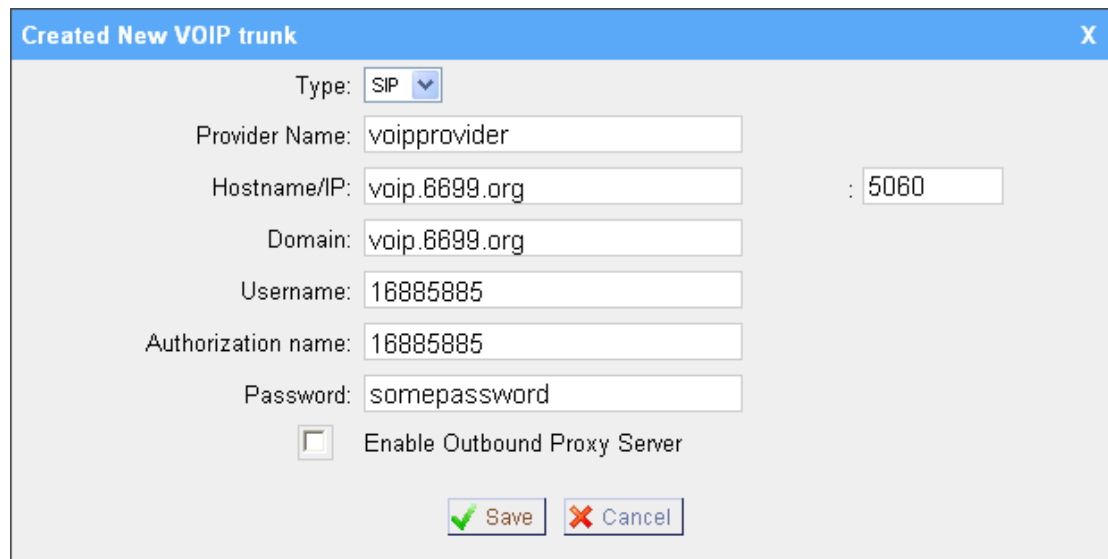


Figure 5-2

#### 2. Add Outbound Routes

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

New Outbound Route
X

Route Name i:

Dial pattern i:

Strip i  digits from front

Prepend these digits i  before dialing

i Calls From Extension(s) i

Available Extensions		Selected
	<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>	500(SIP) 501(SIP) 502(SIP) 503(SIP) 504(SIP) 505(SIP) 506(SIP) 507(SIP)

i Make Outbound Calls On i

Available Trunks		Selected
pstn1(Analog FXO) pstn2(Analog FXO)	<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>	voipprovider(SIP) yeostar(SIP)

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 provider.

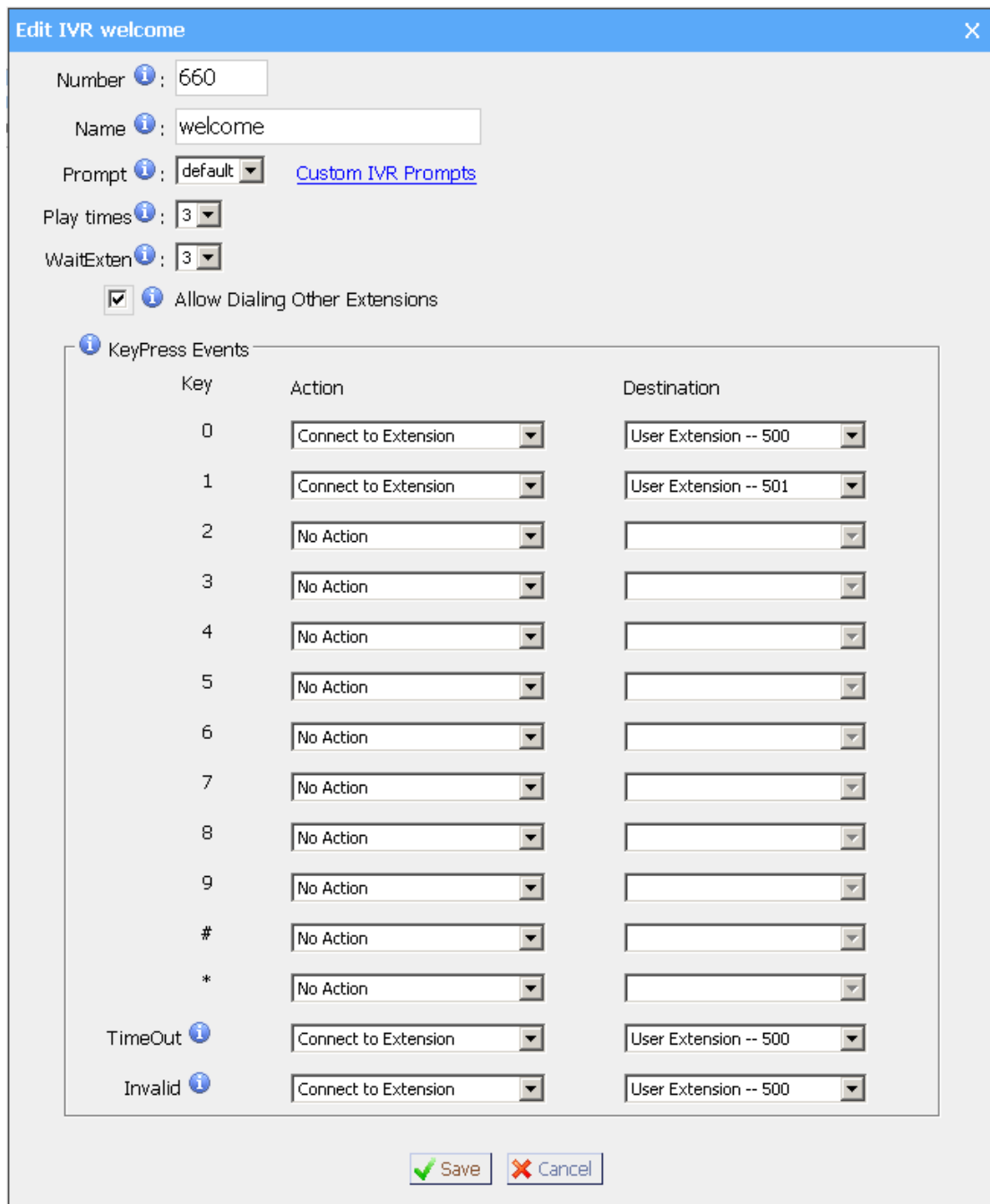
## 5.2 Incoming call

### 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*: 660

Name *i*: welcome

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>i</i>	Connect to Extension	User Extension -- 500
Invalid <i>i</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.

**Edit Inbound Route**
X

**General**

Route Name i :

DID Number i :

Caller ID Number i :

**Trunk Sequence** i

Available Trunks		Selected
	<input type="button" value="»»"/> <input type="button" value="→"/> <input type="button" value="←"/> <input type="button" value="««"/>	pstn1 (Analog FXO) pstn2 (Analog FXO) Digital Trunk(E1)

**During Office Hours**

Destination:

<input type="radio"/> End Call	
<input type="radio"/> Extension	<input type="text" value="User Extension -- 500"/>
<input type="radio"/> VoiceMail	<input type="text" value="VoiceMail -- 500"/>
<input checked="" type="radio"/> IVR	<input type="text" value="IVR -- welcome"/>
<input type="radio"/> RingGroup	<input type="text" value="RingGroup -- ringgroup_def"/>
<input type="radio"/> Conference Room	<input type="text" value="Conference Room -- 640"/>
<input type="radio"/> DISA	<input type="text"/>

**Outside Office Hours**

Destination:

<input type="radio"/> End Call	
<input type="radio"/> Extension	<input type="text" value="User Extension -- 500"/>
<input type="radio"/> VoiceMail	<input type="text" value="VoiceMail -- 500"/>
<input checked="" type="radio"/> IVR	<input type="text" value="IVR -- welcome"/>
<input type="radio"/> RingGroup	<input type="text" value="RingGroup -- ringgroup_def"/>
<input type="radio"/> Conference Room	<input type="text" value="Conference Room -- 640"/>
<input type="radio"/> DISA	<input type="text"/>

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](http://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 WAN port to the switch. And it can get the IP from your route.

b) Press the 'OK' key on T28 to get the IP of T28.

c) Put the IP on web browser then you can enter the T28 configure page through this IP.

d) Put the SIP extensions info on the T28 IP phones.

**Display Name:** 501

**User Name:** 501

**Register Name:** 501

**Password:** 501

**SIP Server:** 192.168.5.150

Use the same method register another T28 to other extension.

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

**A2:** To perform a reset, please follow steps below:

**Step 1:** Hold down the 'Reset' button on the back of the unit for 5 seconds and watch the LEDs on the front of the MyPBX. When the status LED turns red, let go of the reset button.

**Step 2:** When the RUN status LED starts blinking, MyPBX will be set back to factory defaults.

**Step 3:** To access the configuration page, navigate to **192.168.5.150** using a web browser. Make sure that you are on the 192.168.5.0 subnet before doing this.

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



# APPENDIX B How to Configure Autobackup

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

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

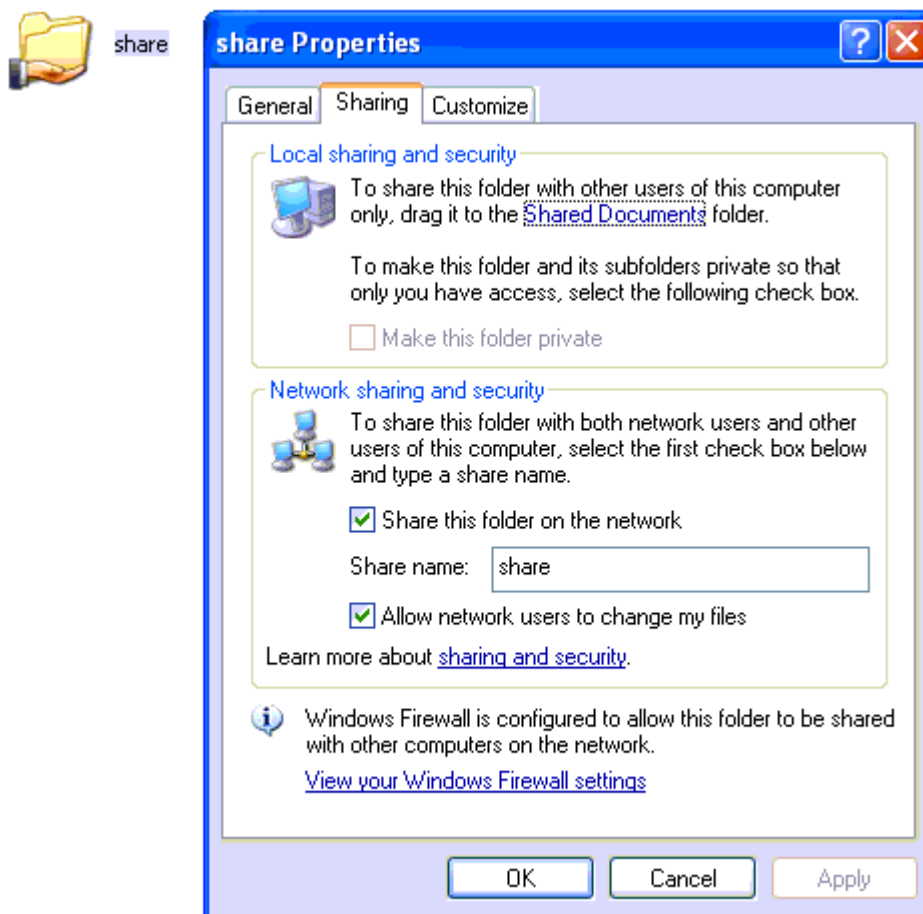


Figure B-1 Set up share Properties

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

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

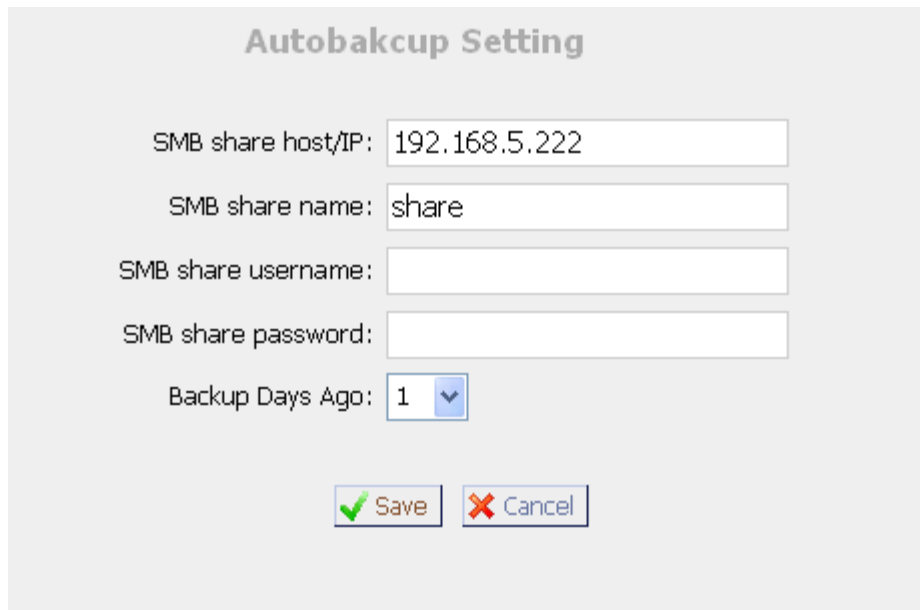


Figure B-2 Autobackup Setting

**SMB share host/IP:** Change this to the IP address of the computer where backup files will be stored.

**SMB share name:** Change this to the name of the shared folder where backups will be stored.

**SMB share username:** The user name used to log into the network share. Leave this blank if it is not required

**SMB share password:** The password used to log into the network share. Leave this blank if it is not required

If configuring Autobackup for the first time, the system will prompt you to restart MyPBX once your settings have been saved. After rebooting, open your Windows share folder to see if the MyPBX backup files and folders has been created. If the contents of the backup folder look similar to Figure B-3, then you have successfully configured Autobackup on the MyPBX unit.

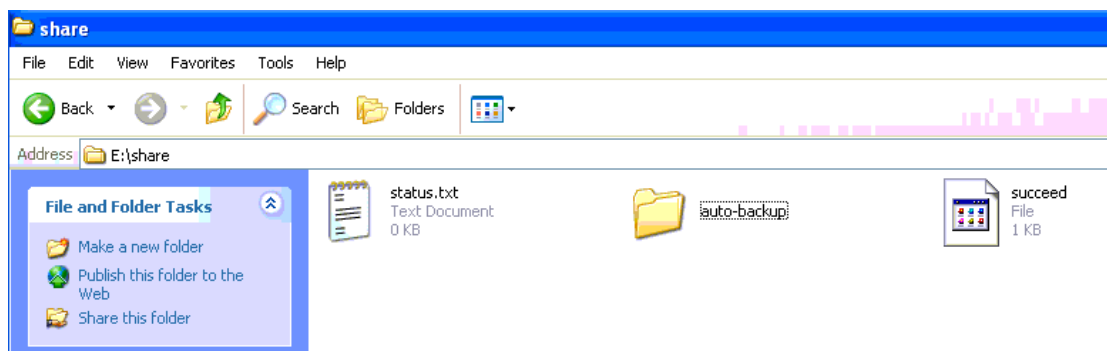


Figure B-3 Autobackup setting succeed

# APPENDIX C How to configure NAT setting

When MyPBX is behind a NAT(firewall),you need to configure NAT setting on MyPBX if you want to use a remote extension. Please follow section 1 or 2 below depending on your network configuration.

1. If MyPBX is connected to a local network, you must set up port forwarding on your router. Specifically, you must map port 5060 (default SIP port) and port 10001-10200 (default RTP port range) as UDP ports.

Next, go to the MyPBX web interface and configure the SIP settings according to Figure C-1:

**External IP:** your router's public IP address

**External Host:** your router's domain

**External refresh:** 20 seconds

**Local Network Address:** 192.168.5.0/255.255.255.0 (change this according to your network configuration)

**NAT mode:** Yes

**Allow RTP Reinvite:** No

**Note:Only when using outernet extension requires configuration**









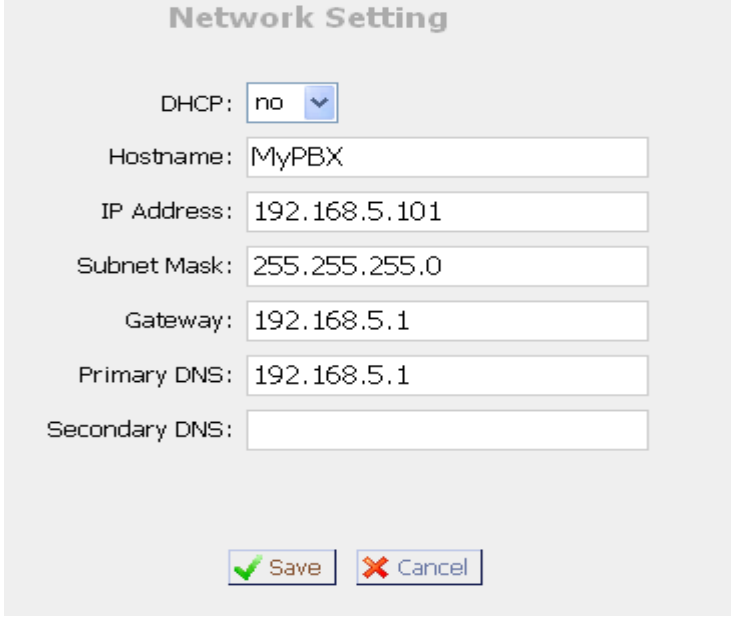
Extern IP 	<input type="text"/>
Extern Host 	<input type="text" value="yeastar.3322.org"/>
Extern Refresh 	<input type="text" value="20"/>
Local Network Address 	<input type="text" value="192.168.5.0/255.255"/>
NAT Mode 	<input type="text" value="yes"/> 
Allow RTP Reinvite 	<input type="text" value="no"/> 

Figure C-1

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



**Network Setting**

DHCP:

Hostname:

IP Address:

Subnet Mask:

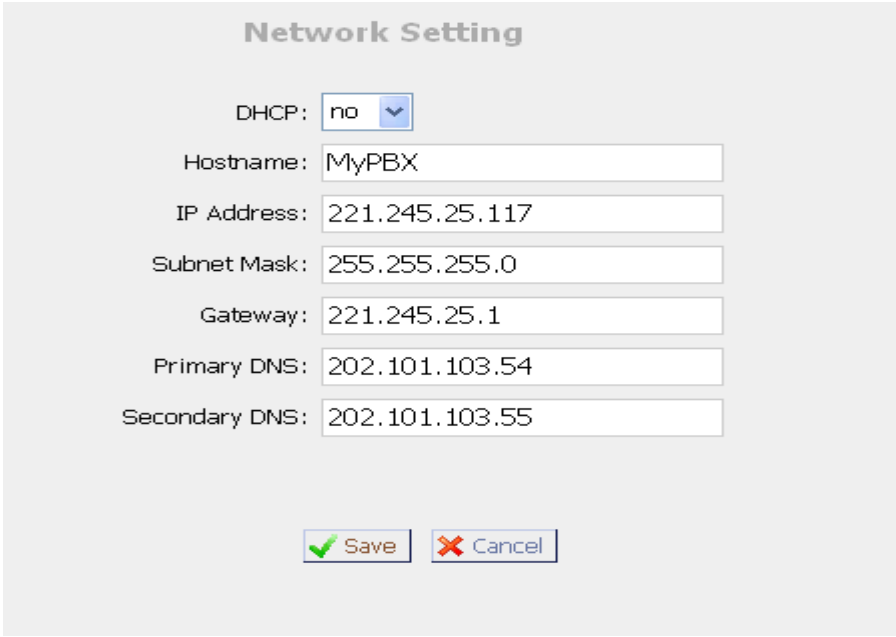
Gateway:

Primary DNS:

Secondary DNS:

Figure C-2 MyPBX Network setting

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



**Network Setting**

DHCP:

Hostname:

IP Address:

Subnet Mask:

Gateway:

Primary DNS:

Secondary DNS:

Figure C-3

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

**Note: Only when using outernet extension requires configuration**

Extern IP ⓘ :	<input type="text" value="221.245.25.117"/>
Extern Host ⓘ :	<input type="text"/>
Extern Refresh ⓘ :	<input type="text"/>
Local Network Address ⓘ :	<input type="text"/>
NAT Mode ⓘ :	yes <input type="button" value="v"/>
Allow RTP Reinvite ⓘ :	no <input type="button" value="v"/>

Figure C-4

**External IP:** The public IP address of MyPBX

**External Host:** Leave this blank if no domain has been configured

**External refresh:** Leave this blank

Local Network Address: Leave this blank

**NAT mode:** Yes

**Allow RTP Reinvite:** No

# APPENDIX D How to Use Auto Provision

**Step 1.** Disable DHCP Server on your local network.

E.g. Disable DHCP Server on Linksys Router.

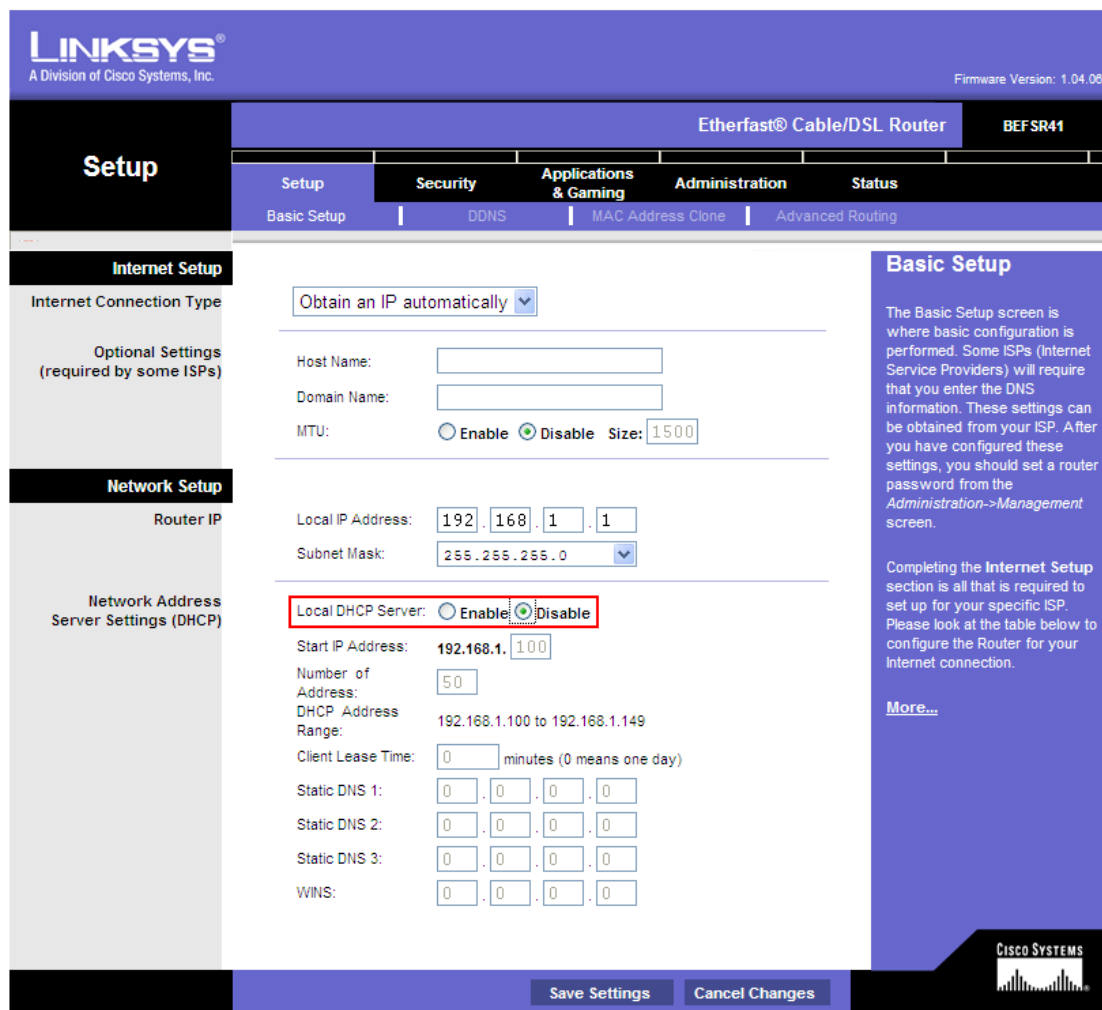


Figure D-1

**Step 2.** Enable DHCP Server on MyPBX.

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



Figure D-2

**Step 3.** Configure phones on MyPBX auto-provision page.

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

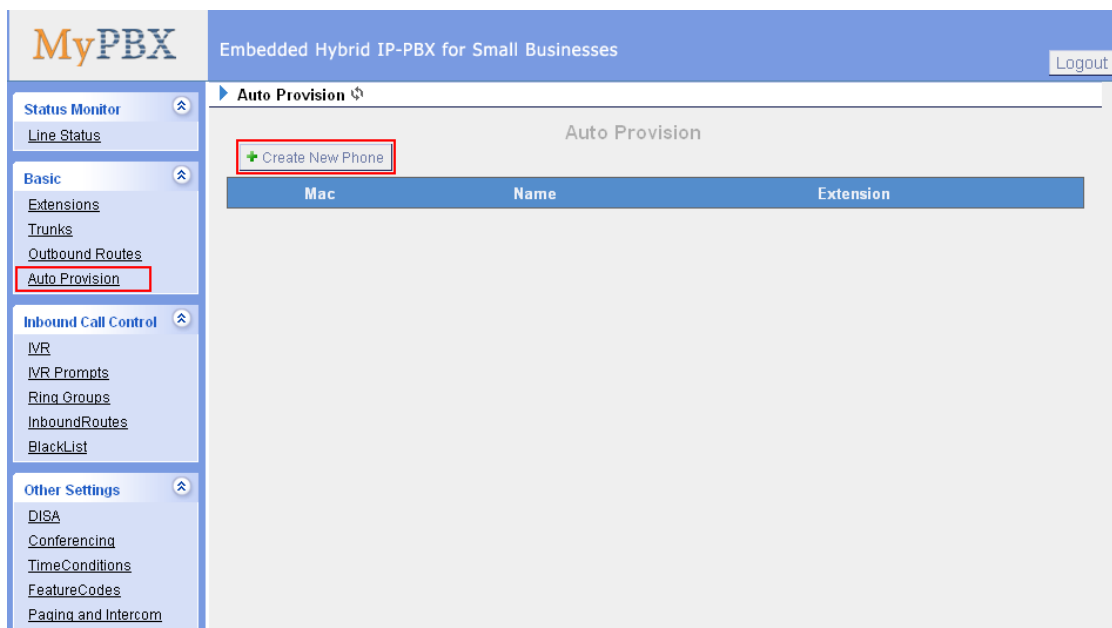
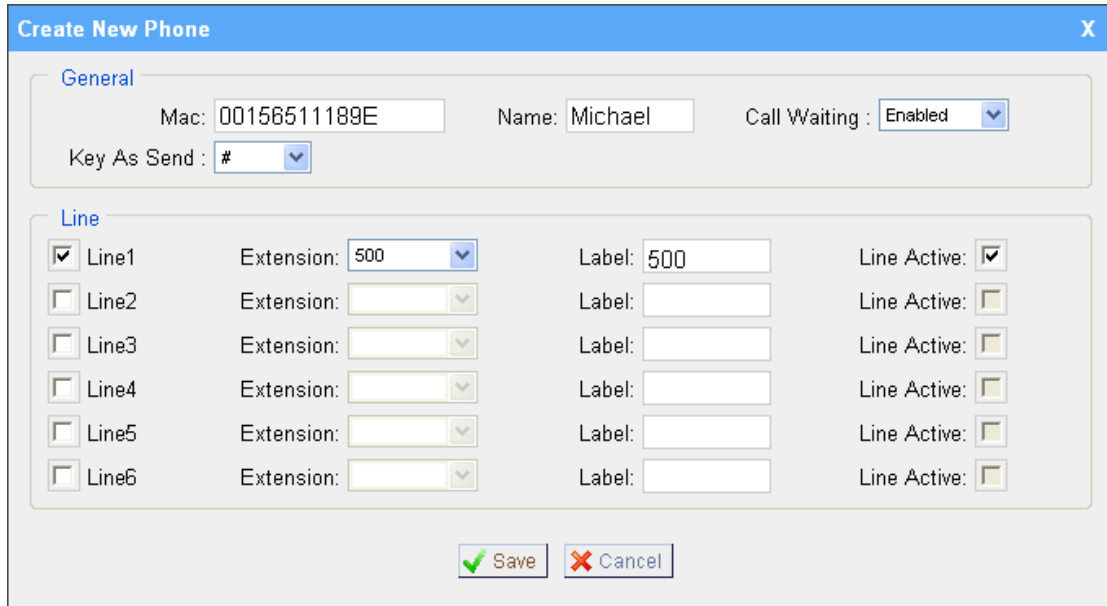


Figure D-3

- 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.



The screenshot shows a 'Create New Phone' dialog box with two main sections: 'General' and 'Line'.  
**General Section:**  
 - Mac: 00156511189E  
 - Name: Michael  
 - Call Waiting: Enabled (dropdown menu)  
 - Key As Send: # (dropdown menu)  
**Line Section:**  
 - Line1:  Extension: 500 (dropdown), Label: 500, Line Active:   
 - Line2:  Extension: (empty dropdown), Label: (empty), Line Active:   
 - Line3:  Extension: (empty dropdown), Label: (empty), Line Active:   
 - Line4:  Extension: (empty dropdown), Label: (empty), Line Active:   
 - Line5:  Extension: (empty dropdown), Label: (empty), Line Active:   
 - Line6:  Extension: (empty dropdown), Label: (empty), Line Active:   
 At the bottom are 'Save' and 'Cancel' buttons.

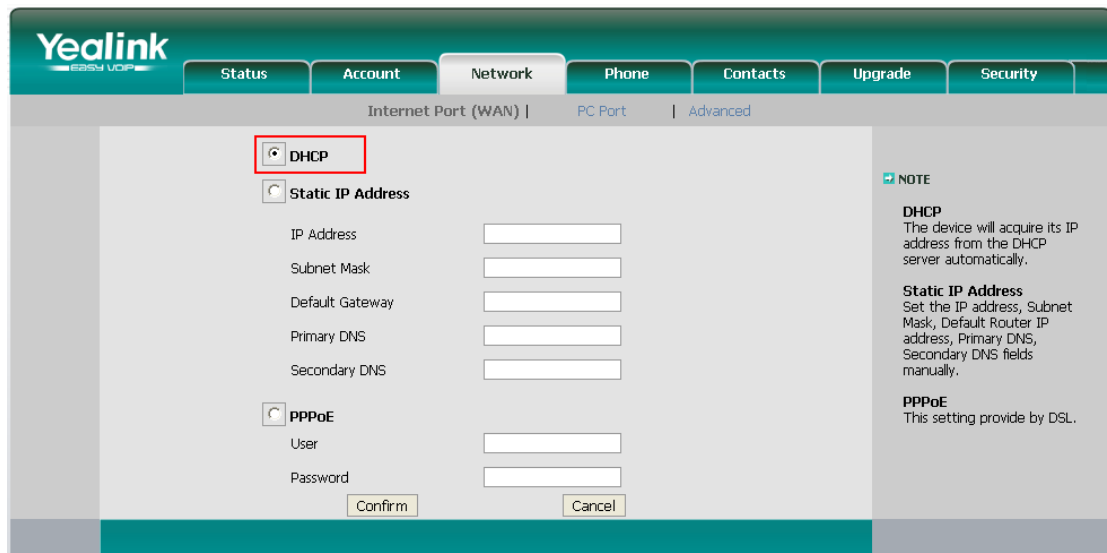
Figure D-4

**Step 4.** 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).

- Login IP phone's web page.
- Enable DHCP.



The screenshot shows the 'Network' tab in the Yealink web interface. The 'DHCP' option is selected and highlighted with a red box. Below it are fields for 'Static IP Address' (IP Address, Subnet Mask, Default Gateway, Primary DNS, Secondary DNS) and 'PPPoE' (User, Password). A 'NOTE' section on the right explains DHCP and Static IP Address settings. 'Confirm' and 'Cancel' buttons are at the bottom.

Figure D-5

**Step 5.** Finish.

<Finish>