



IPPBX User Manual

(V1.4 ,Applied to ZPX04**, ZPX08**,ZPXP**)

version	date	author	description
V1.1	2009-1-10	Yu	Editor
V1.2	2009-7-07	Yu	Editor
V1.3	2009-10-21	Yu	Editor
V1.4	2010-3-24	Yu	Editor

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Chapter1 Brief Introduction

1.1 Appearance&Model

This article is the user manual for ZYCOO IPPBX series products. It also includes the application notes for how to use ZYCOO products to build a telephony system for small office. Our IPPBX productline include ZPX04**,ZPX08** and ZPXP8** so far, since they have almost the same software and structure so we will use ZPX8** as the demo unit on this article.

ZPX04** Appearance&Model



Model	ZPX0404	ZPX0413	ZPX0422
FXS	0	1	2
FXO	4	3	2

ZPX08** Appearance&Model



Model	ZPX0808	ZPX0826	ZPX0844
FXS	0	2	4
FXO	8	6	4

ZPXP** Appearance&Model



Model	ZPXP08	ZPXP26
FXS	0	2
FXO	8	6

1.2 System Features

ZYCOO's series of IPPBX is an embedded ipbx based on standard asterisk for Home&SMEs, which is not only a PBX, but also as a voice mail Server, IVR server, conferencing server. With 4 or 8 analog interface which can be configured as FXS or FXO

ports (made in factory), and 1 Wan and 1Lan with router function. With excellent echo cancellation function, it can meet most of the customers' requirement.

- Based on Asterisk
- Configuration by Web
- Built-in SIP/IAX Server
- Static/DHCP/PPPoE network access
- Codec: G.711-Ulaw, G.711-Alaw, G.726, G.729, GSM, SPEEX
- SIP/IAX Trunk(use with VoIP Trunk operator)
- Zap Trunk(Use with PSTN)
- SIP/IAX Extensions(connect with IP Phone)
- Zap Extensions(connect with Analog Phone)
- Voice Mail Server
- Flexible Dial Plan
- Call Conference
- IVR Server
- Music On Hold
- Call Logs
- Support IP Phone with Key function
- FAX T.38
- Other basic function:
 - 1.Three way calling
 - 2.Call Forward(on Busy or on Unanswered Call or on Unregistered Extension)
 - 3.Call Hold
 - 4.Call Transfer
 - 5.Call Waiting
 - 6.Caller ID

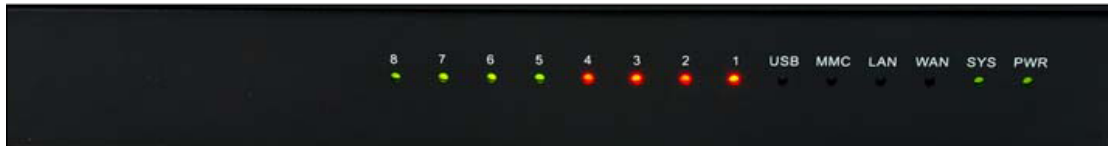
1.3 Interface&Panel

1) Interface



- 8 * Analog Ports can be FXO or FXS (RJ11)
- 1 * USB Interface(optional)
- 1 * SD MMC Interface
- 2 * Network Interface (RJ45)
- 1 * Power port (DC 12V 2A)
- 1 * Reboot Button

2) Indication Panel



Mark	Function	Status	Description
PWR	Power Status	On	Power On
		Off	Power Off
SYS	System Status	On	System working
		Off	System Failed
WAN	WAN interface Status	Wink	Data exchanging
		Off	No Data exchanging
LAN	LAN Interface Status	Wink	Data exchanging
		Off	No Data exchanging
MMC	SD card Status	On	MMC connect successfully
		Off	MMC connect failed
USB	Optional		
1-8	Analog Modules Status	Red	FXO channel
		Green	FXS channel
		Off	Failed

3) Hardware

- 32bit embedded RISC DSP
- 256M Onboard Nand Flash
- 2M Onboard Nor Flash
- 64M Onboard SDRAM
- 2G MMC/SD Storage

4) environmental requirements:

- temperature: -10 °C -45 °C
- Storage temperature: -30 °C -65 °C
- humidity: 10-80% no dew
- Power: AC 100~240V

5) Packing List

- IPPBX 1 Unit
- Power Adapter 1 Unit

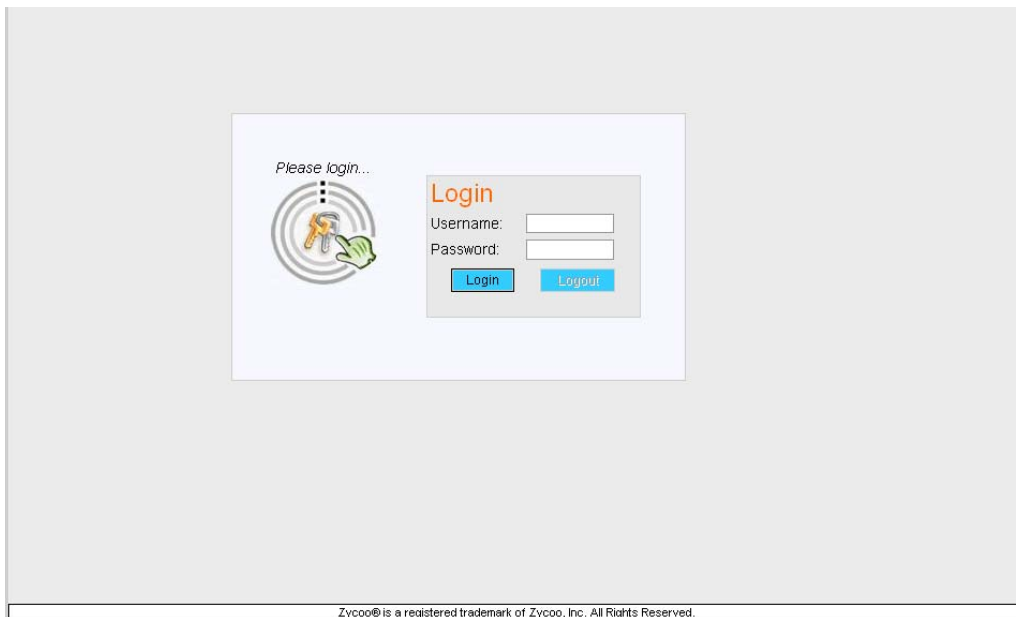
1.4 Default configuration

1. WAN port IP address: <http://192.168.1.100:9999>
2. LAN port IP address: <http://192.168.10.100:9999>
3. LAN port super IP: 169.254.1.254/255.255.0.0
4. Web GUI username: **admin**
5. Web GUI password: **admin**

Chapter2 Home Page

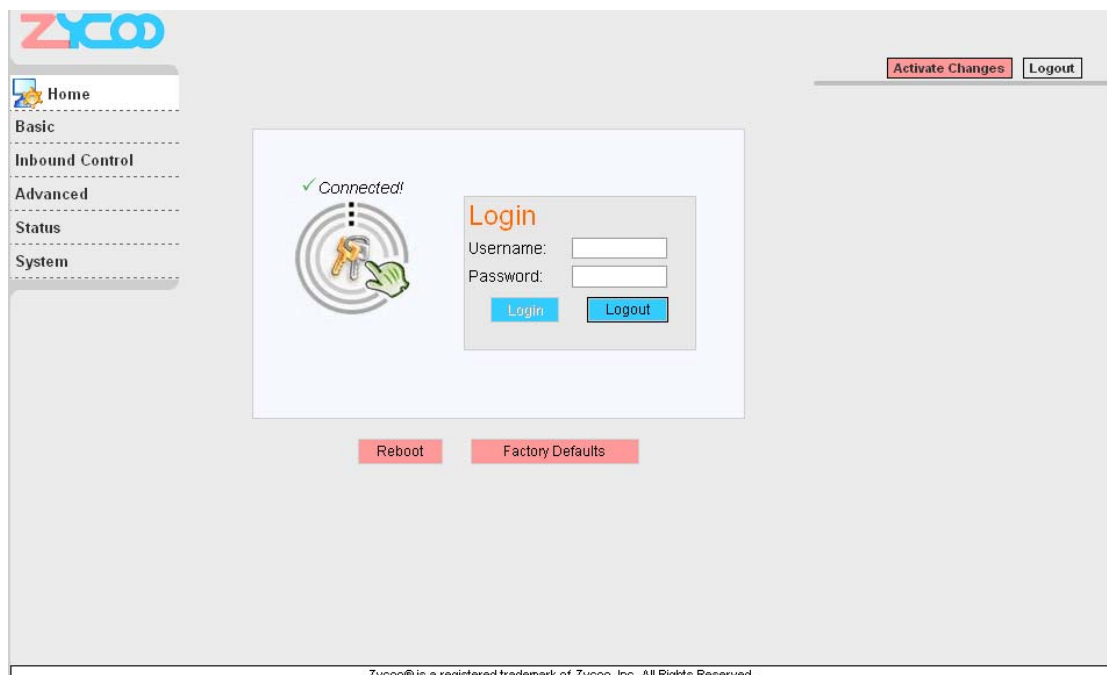
2.1. Log on to the system

After connecting the IP PBX to the local area network. Launch the web browser on a computer which is in this local area network. Enter the IP address for the system (Wan port IP address http://192.168.1.100:9999, Lan port IP address http://192.168.10.100:9999) . The start web page will appear like this:



Enter Username and password (default username is **admin**, password is **admin**), then click "login". Once the login is successful, the home page will be display:

Noted: you have to add a network segment same with the Wan ports if your PC is not at 192.168.1.***.



With the zycoo GUI, you can configure extensions, conference, voicemail, Dial Plan

and etc. Each page of the GUI has three columns:

The left column present all the options tab that you can program the system. Click the tab to go to setting page of different options.

The middle column contains the primary content for each page.

The right column of the user interface contains Tooltips. This area provides brief description for any options of the GUI

The home page is used for logout, Reboot and Factory Defaults.

- **Logout:** To log out the zycoo GUI.
- **Reboot:** Reboot the IP PBX system
- **Factory Defaults:** Restore all settings to factory default.
- **Activate change:** Made the change active for the current configuration after you make a configuration on some page.

Chapter3 Basic Configuration

3.1. Configure Extensions

Click the Extension tab and you will see the extensions setting, your created users are in this page. There are 30 users in your extensions list as default setting, you can add new extensions or remove the existing extensions.

Extensions Setting include:

- **Extension** The extension is assigned to the defined user.
- **Name** The full name of the individual assigned to this extension.
- **Password** The password is used to Extension registered
- **VM Password** The password is used to access voicemail for the specified Extension
- **E-mail** Set the user's E-mail

- **Caller ID** Identifies the Caller ID presented when the listed extension dials out
- **Analog Phone** A drop-down menu is available to identify the analog phone port which this extension will access.
- **Dial Plan** You can choice dial plan based on the extensions' need, this option references the Dial Rules option on the left tool bar.

There are also several advanced extension options available. The advanced options establish the connections from the listed extension to other systems within the IPPBX system server. These advanced options include the following:

- **Voicemail** The extension support voicemail
- **SIP** The extension support SIP protocol
- **IAX** The extension support IAX protocol
- **Call Waiting** The extension support Call Waiting function
- **3-Way Calling** The extension support 3-Way Calling functions
- **Pickup Group** Select pickup group of the extension
- **Codecs** Click here, you can set the extension's codec (default support: alaw, ulaw and G.729).

3.2. Trunk

If you want to make external call, you must register with a Trunk in order to connect to the Public Switched Telephone Network (PSTN) or other VoIP service provider. Through the web page you can add a trunk.

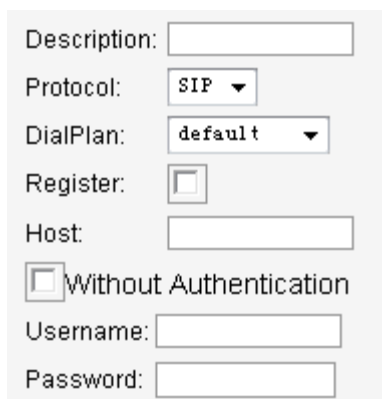
There are three Trunk categories: Analog Trunk, VoIP Providers, Custom VoIP Trunk.

Analog Trunk

Select the Analog radio button to define the analog ports you have access to as a service provider. This will give you the ability to place calls through the IP PBX utilizing analog lines. The analog ports available will be displayed when you select this option. Choose one or more analog ports by selecting their associated checkbox. You will not be able to create an analog service provider if you do not have any analog ports available.

Custom Trunk

The Custom VoIP option allows you to create a custom VoIP definition. To create the custom VoIP provider definition you will need to complete the following:



The screenshot shows a configuration form for a Custom VoIP provider. It includes the following fields and options:

- Description: [Text input field]
- Protocol: [SIP dropdown menu]
- DialPlan: [default dropdown menu]
- Register: [checkbox]
- Host: [Text input field]
- Without Authentication
- Username: [Text input field]
- Password: [Text input field]

- **Description** The description should be used as the name of the custom VoIP definition
- **Protocol** Specify either a IAX or SIP protocol
- **DialPlan** Select a DialPlan for this trunk.
- **Register** Enable/Disable server register. Registering is not required for all providers
- **Host** The IP address of your service provider
- **Username** The user name associated with your provider account
- **Password** The password associated with your provider account
- **Without Authentication** if you connect to Voip server without Authentication, pls selected

Peer

The Peer option allows you to create a custom VoIP Peer.



The screenshot shows a configuration form for a custom VoIP Peer. It includes the following fields and options:

- Peer Name: [Text input field]
- Protocol: [SIP dropdown menu]
- DialPlan: [default dropdown menu]
- Host: [dynamic text input field]
- Without Authentication
- Username: [Text input field]
- Password: [Text input field]

- **Peer Name** Defines a peer name for this peer.
- **Protocol** Specify either a IAX or SIP protocol

- **DialPlan** Select a DialPlan for this peer
- **Host** dynamic | hostname | IP Address
- **Without Authentication** if you connect to the PBX without Authentication, pls selected
- **Username** Defines the peer username
- **Password** Defines the peer password

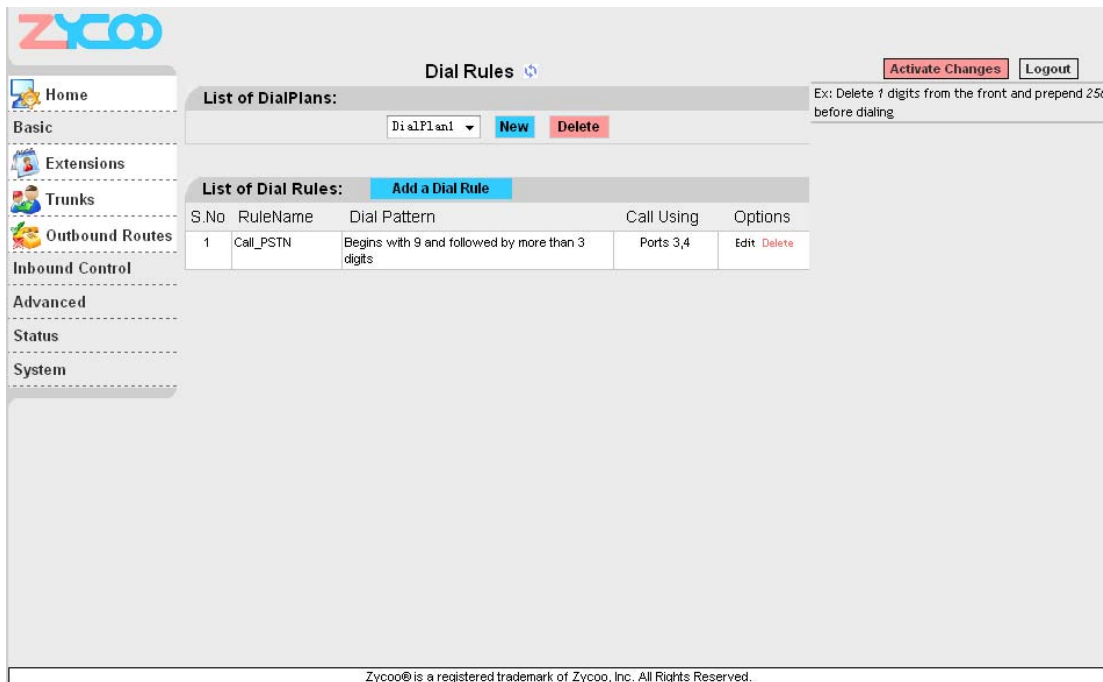
Once you have added a VoIP Trunk it will appear on the list of Trunk on the Trunk page. There is an Options drop-down list associated with each Trunk listing. The Options drop-down list allows you to edit or delete the Trunk definition, as well as further refine the definition by choosing several advance options. Select either Codecs or Advanced to further refine the definition.

- **Edit** Edit you select the trunk.
- **Codecs** Codecs provide the ability for your voice to be converted to a digital signal and transmitted across the internet.
- **Advanced** The following advanced options are available to further refine your trunk.

- **Trunkname** Specify a trunk name if you want to refer to the service provider definition as something other than specified in Comment
- **Insecure** This option specifies how connects to a service provider (host) should be handled. Valid options are very/yes/no/invite/port. (Default is “very”)
- **Port** The register request is sent through the port. (Default is SIP:5060,IAX:4569)
- **Caller ID** The caller ID will be set to the value specified in this field
- **Fromdomain** Sets default from: domain in SIP messages when acting as a SIP client.
- **Fromuser** Sets default from: user in SIP messages when acting as a SIP client
- **Contact** Specifies a primary extension for call routing

3.3. Outbound Routers

The Dial Rules tab on the left toolbar allows you to use basic pattern matching to differentiate outbound calls and route them accordingly (create different DialPlan).



Click on Add a Dial Rule to define a new DialPlan. The following dialog will be displayed.

A DialPlan is comprised of the following items:

- **Rule Name** Set a rule name
- **Place this call through** Select a Trunk through which the call should be made
- **Failover** Select a trunk Failover
- **PIN Set** Set a password when you dial base the Dial rule.
- **Dialing Rules** The Dialing Rule gives you the ability to use basic pattern matching to differentiate calls and route them accordingly. For instance, if a number begins with 9256 followed by 7 or more digits, that would define a call within the state of Alabama. If a call began with 9 followed by 7 digits, it would be a local call that probably didn't require a long distance charge. Instead of adding a rule for every

extension or phone number you call, specify the pattern in this rule similar to the example.

- **Define a custom pattern** Set a custom pattern by yourself.

Custom Pattern:

(define a Basic Pattern)

N Any digit from 2 to 9
X Any digit from 0 to 9
. Any number of additional digits

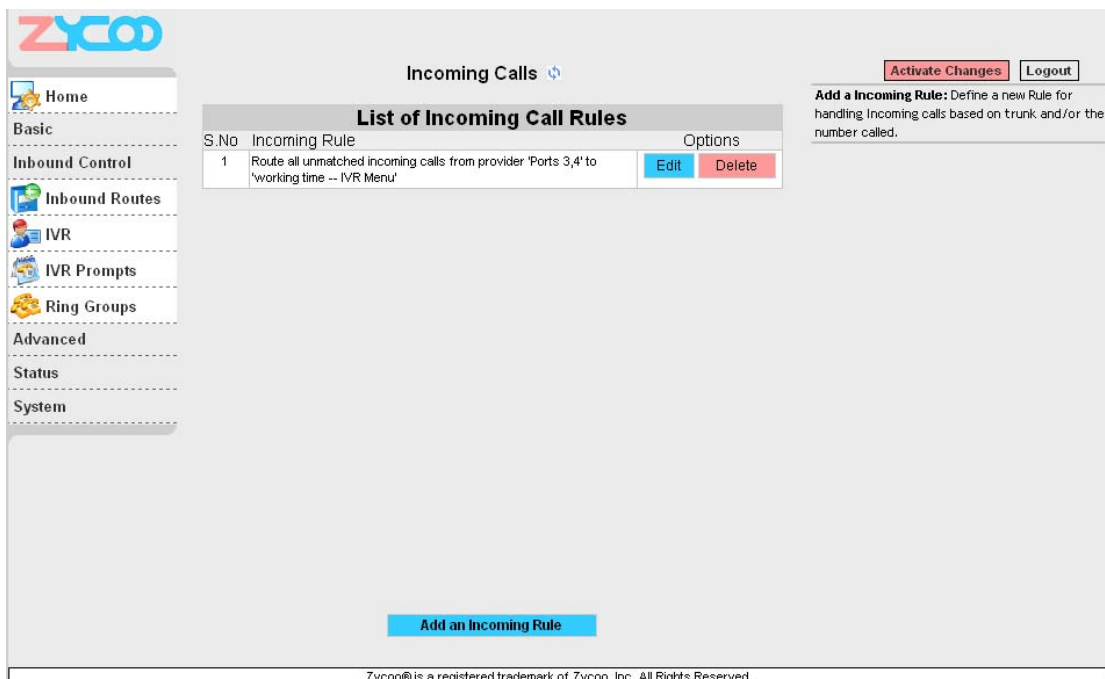
- N** Any digit from 2 to 9
- Z** Any digit from 1 to 9
- X** Any digit from 0 to 9
- .** Any number of additional digits

Example: “_9ZNXXX.” mean first number is 9, second number is any digit from 1 to 9, third number is any digit from 2 to 9 and each “X” is any digit from 0 to 9. The “.” is more.

- **Delete** This option gives you the opportunity to remove specified digits from the call being dialed and replace them with the digits needed to make the call. You can also prepend digits to the beginning.

Chapter4 Inbound Control

4.1. Inbound Routers



Incoming Calls

[Activate Changes](#) [Logout](#)

List of Incoming Call Rules

S.No	Incoming Rule	Options
1	Route all unmatched incoming calls from provider 'Ports 3,4' to 'working time -- IVR Menu'	Edit Delete

Add a Incoming Rule: Define a new Rule for handling Incoming calls based on trunk and/or the number called.

[Add an Incoming Rule](#)

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The same pattern-matching logic used for processing outbound calls can also be employed for inbound calls. The two defaults define routing based on whether an incoming call matches or doesn't match a pattern you define.

There are only a few options you need to configure

- **Route** Make a selection from the drop-down list to choose how the calls will be routed. You can select from All Unmatched Calls or Calls which Match
- **From Provider** Select from the list of providers which you previously configuration
- **To Extension** The previously configuration extension which should receive the call.

4.2. IVR (Interactive Voice Response)

Through the web page, you can create Interactive Voice Response (IVR). IVR are designed to allow for more efficient routing of calls from incoming callers.

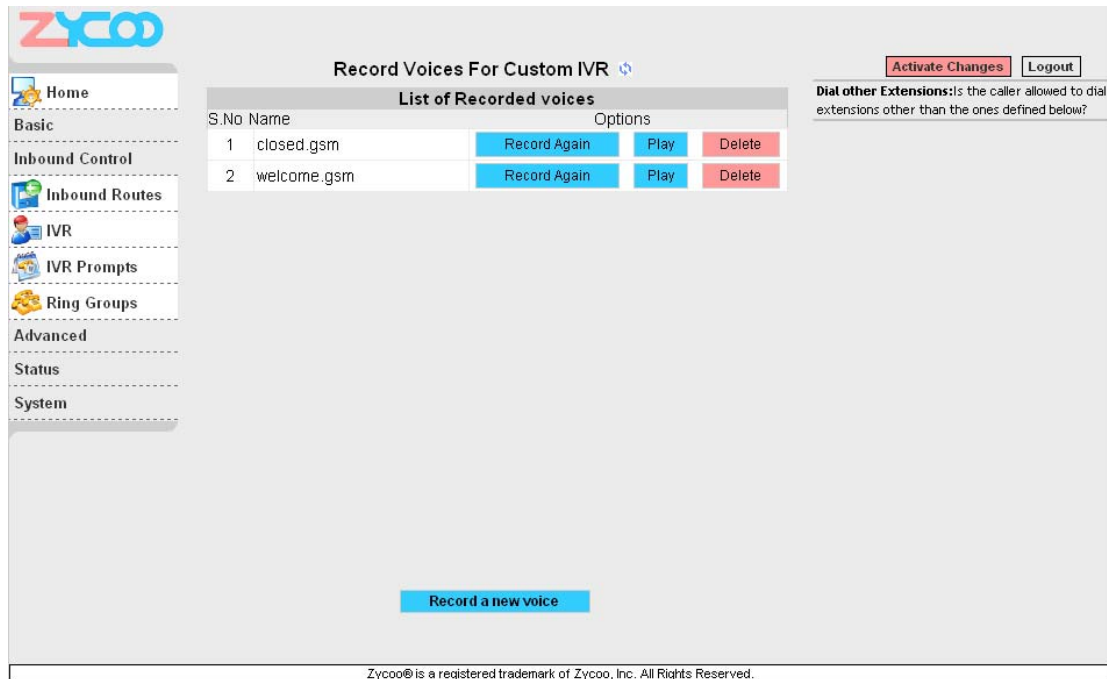
Key	Action	Extension
0	Goto Extension	801
1	Disabled	
2	Disabled	
3	Disabled	
4	Disabled	
5	Disabled	
6	Disabled	
7	Disabled	

Voice menus are constructed depending on your needs. Just like your business you need to create the solution best suited to your customers.

- **Name** Set a IVR name
- **Extension** Set a IVR connect number
- **Welcome Message** Select a welcome message voice from record
- **Dial other Extensions** Enable/Disable allow dial other extensions.

4.3. IVR Prompts

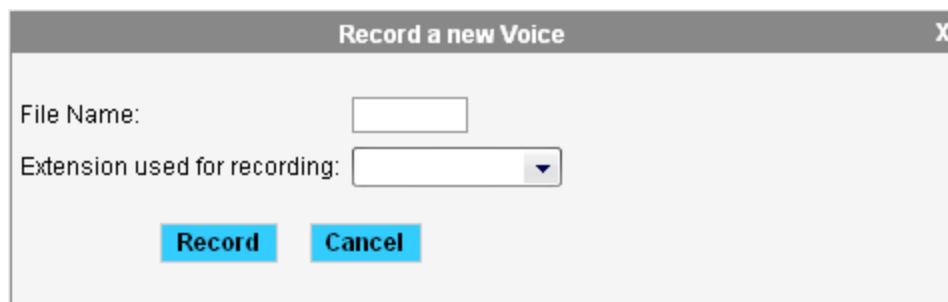
In the event that one wants to record custom IVR prompts for the IP PBX, which can be used in a IVR, the Record may be used.



A list of previously recorded menus is displayed. Here, the user may modify several options

- **Record Again** Clicking this button allows the user to make another attempt at recording and replacing an existing custom sound file
- **Play** Clicking this button brings up a dialog entry box to allow the input of an extension that System will dial and play the prompt over
- **Delete** Clicking this button will delete the selected prompt

There are two options under "Record a new voice"



- **File Name** This text entry box specifies the saved name of the file that is to be recorded.
- **Extension Used for Recording** This drop-down select box allows the user to choose which extension will dial to wait for the user to speak the prompt

4.4. Ring Groups

You can configure Ring Groups through the web page

Ring Groups

Add Ring Group

Name:

Strategy: Ring all

← SIP/801 -- User1
SIP/802 -- User2
SIP/803 -- User3
SIP/804 -- User4
SIP/805 -- User5
SIP/806 -- User6
SIP/807 -- User7
SIP/808 -- User8 →

Ring Group Members

Extension for this ring group (Option) :

Ring (each/all) for these many seconds :

If not answered

Goto an Extension

Goto an Extension Voicemail

Goto a RingGroup

Goto an IVR menu

HangUp

Save Cancel

New Ring Group

Activate Changes Logout

Goto an Extension Voicemail: Select goto an extension voicemail if the ringgroup no answer.

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Define Ring Groups to Dial more than one extension

- **Name** Set a Ring Group name
- **Strategy** There is a drop-down list, you can choose Ring all or Ring in order.
- **Ring Group Members** Add Ring Group member from Available channels.

If the Ring Group no answered you can choose to

[Goto an Extension](#), [Goto an Extension Voicemail](#), [Goto a RingGroup](#), [Goto an IVR menu](#), [HangUp](#)

Chapter5 Advanced Configuration

5.1. Operator

Local Extensions are Operator Extension

Set up the digit of local extensions

Set up Operator Extension. (you can dial "0" go to the extension at any time)

Global Ring Time Set

Set default each extension ring time.(Default settings 30s)

Default Settings for a New User

Set up the Default Settings for a New User, when you create a new extension will use the configuration.

5.2. Voicemail

Voicemail is an option available for every extension in the IP PBX system. The relationship between the extension and the voice mail is established in the User Extension section of the GUI. You can configure the voicemail through this page.

Standard configuration information is also present, allowing you to confirm the extension used to check messages as well as general parameters such as the following:

- [Extension for Checking Messages](#) This option defines the extension which Users call in order to access their voicemail account.
- [Max greeting\(Seconds\)](#) With this option, you specify the maximum amount of time available to record your voicemail greeting.
- [Attach recordings to e-mail](#) Enable/Disable send recording file to you email by attachment
- [Dial "0" for Operator](#) Callers who are sent to voice mail can press "0" for the operator and be transferred either during the voice mail salutation, or after recording the message. If this option is not enabled, a caller's pressing "0" will be ignored.

There are several options that can be specified to define the voicemail message in the system.

- [Message Format](#) This option gives you the ability to choose the format in which messages will be mailed.
- [Maximum Messages](#) The maximum number of messages per voice mail box is set here.
- [Maximum Message Time](#) The maximum duration of a message left by a caller is set here
- [Minimum Message Time](#) The minimum duration of a message is dictated here.

There are several playback options that can be specified.

- [Say Message Caller-ID](#) The Say Message Caller ID option reads the caller ID before the voice mail message is played
- [Say Message Duration](#) This option identifies exactly how long the message lasted.
- [Play Envelop](#) The envelope provides the date, time, and caller ID related to a voice mail.
- [Allow Users to Review](#) This option provides incoming callers the option to review their message before it is saved and can be played back by the owner of the voice mail extension. Standard options are presented to you, allowing you to discard the message or re-record it if you aren't happy with it.

Voice mail to email set: SMTP settings

**Voice mail to email set:
SMTP settings**

SMTP Settings:

SmtP server:

Port:

SSL/TSL:

Enable ssmtp Authentication

Username:

Password:

Port: the port number on which the SMTP server is running, generally port 25.

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- **SmtP server** The IP address or hostname of an SMTP server that your IP PBX may connect to, in order to send e-mail notifications of your voicemail; eg:mail.yourcompany.com
- **Port** The port number which the SMTP server is running is generally port 25. For SSL encrypted connection use port 465 instead.
- **SSL/TSL** Enable use SSL/TLS to send secure messages to server.
- **Enable SMTP Authentication** if your SSMTP server needs Authentication, please enable SSMTP Authentication set, and configure the follow information
- **Username** input username of your email.
- **Password** input password of your email.

Email settings

Template for Voicemail Emails

Sender Name:

From:

Subject:

Message:

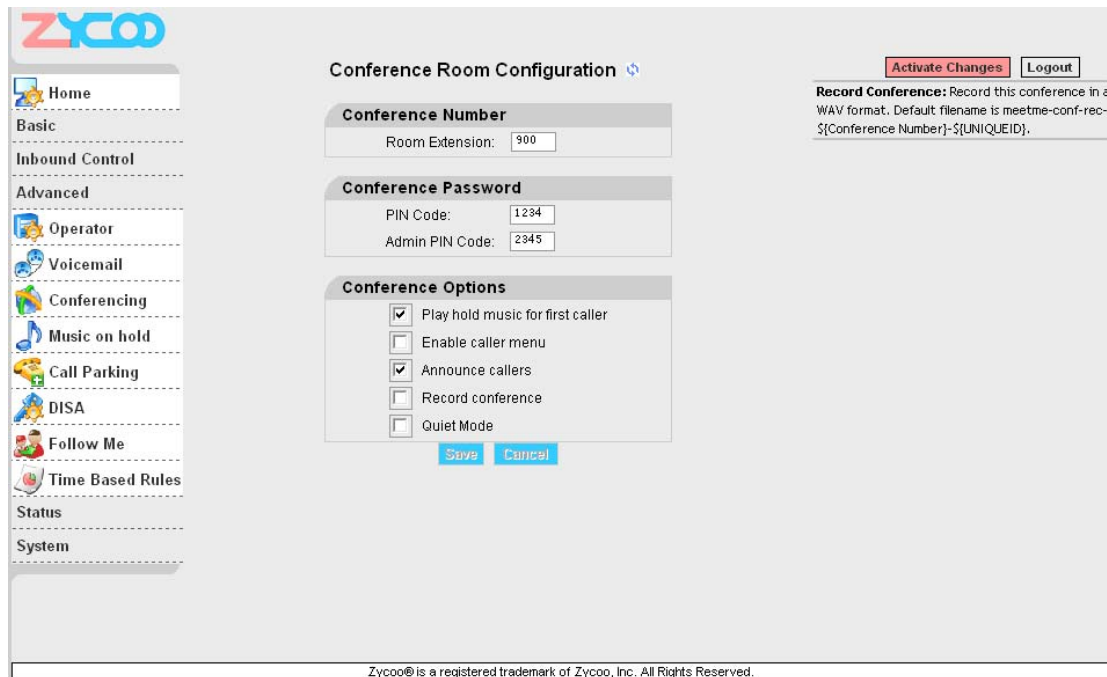
Template Variables: ⌨ : TAB

- \${VM_NAME} : Recipient's firstname and lastname
- \${VM_DUR} : The duration of the voicemail message
- \${VM_MAILBOX} : The recipient's extension
- \${VM_CALLERID} : The caller id of the person who left the message
- \${VM_MSGNUM} : The message number in your mailbox
- \${VM_DATE} : The date and time the message was left

- **Sender Name** Set the name for sender
- **From** Set the from email
- **Subject** Set the email title
- **Message** Input the matter in your email.

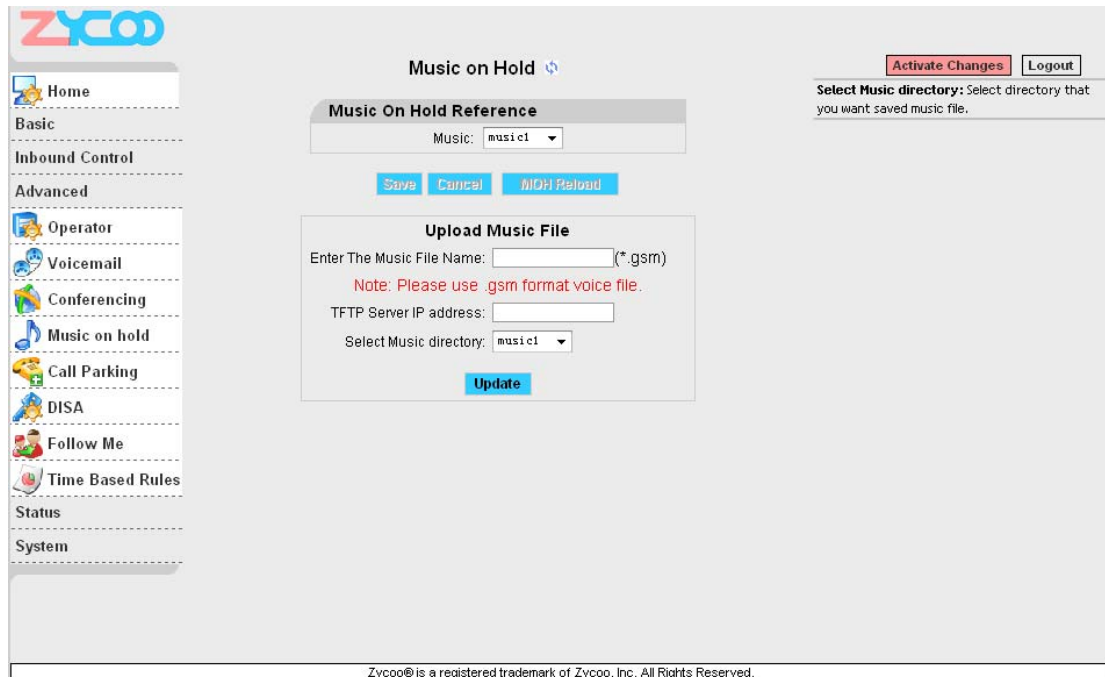
5.3. Conferencing

Every company reaches the point of needing more people on a call than it can effectively include through three-way calling. conference bridges allow you to include more people as well as project a professional image.



The configuration of the conference room and standard features is very straightforward. The conference room use default extension 900 , but you can always change it to any extension number you want. After establishing the extension for the room, you need to specify the password settings for the conference. Assign the PIN Code used by participants to enter the conference as well as the Administrator PIN Code used by the moderator of the conference to open the conference room.

5.4. Music On Hold



Music on Hold

Music On Hold Reference

Music: music1

Save Cancel MOH Reload

Upload Music File

Enter The Music File Name: (*.gsm)

Note: Please use .gsm format voice file.

TFTP Server IP address:

Select Music directory: music1

Update

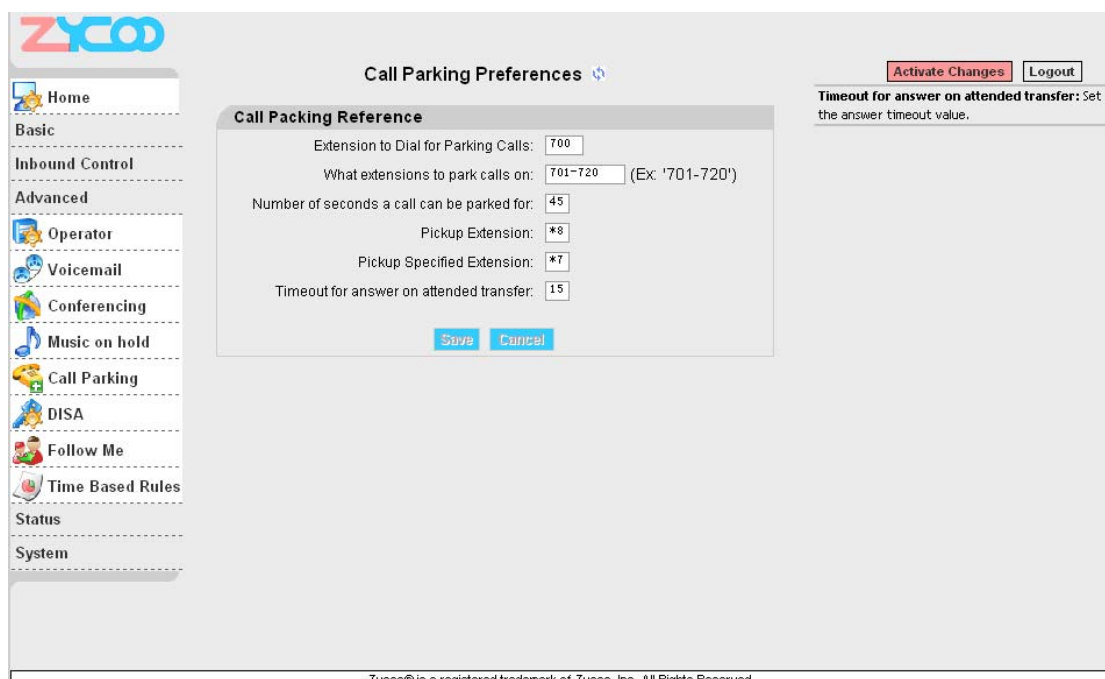
Activate Changes Logout

Select Music directory: Select directory that you want saved music file.

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- [List of Music On Hold](#) Display Music On Hold class list
- [Class](#) Set Music On Hold class name
- [Music](#) Select music. (you can replace music file through the update page.)
- [Enter The Music File Name](#) Set you want upgrade music file name
- [TFTP Server IP address](#) Set the TFTP server IP
- [Select Music directory](#) Select directory that you want saved music file.

5.5. Call Parking



Call Parking Preferences

Call Parking Reference

Extension to Dial for Parking Calls: 700

What extensions to park calls on: 701-720 (Ex: '701-720')

Number of seconds a call can be parked for: 45

Pickup Extension: *8

Pickup Specified Extension: *7

Timeout for answer on attended transfer: 15

Save Cancel

Activate Changes Logout

Timeout for answer on attended transfer: Set the answer timeout value.

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- Extension to Dial for Parking Calls: Set Call Parking number
- What extensions to park call on: Set the Call Parking get number (eg: 701-720)
- Number of seconds a call can be parked for: Set the second call time
- Pickup Extension: Set Pickup Extension
- Pickup Specified Extension Set Pickup Specified Extension
- Timeout for answer on attended transfer: Set the answer timeout value.

5.6. DISA Settings

DISA Settings

Select DialPlan: Sets the DialPlan that calls will originate from.

S.No	Disa Name	Options
1	test1	<input type="button" value="Edit"/> <input type="button" value="Delete"/>
2	test2	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

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List of Disa Disa name are listed in the table.

New Disa Create a new Disa.

Edit Disa X

DISA Name:

PIN:

Response Timeout(s):

Digit Timeout(s):

Extension for this Disa(Optional):

Allow Outbound Route

Select DialPlan

DISA Name Set a name for Disa

PIN Set a password for Disa

Response Timeout(s) Set effective time for inputting a password

Digit Timeout(s) After you input the right password, the interval between digits that you need dial.

[Extension for this Disa\(Optional\)](#) Set a number connect Disa
 Select DialPlan Select your DialPlan for calling out

5.7. Follow Me

Follow Me

[Activate Changes](#) [Logout](#)

List of Follow Me

S.No	Extensions	State	Forward No.	Options
1	802	EN	805	Edit Delete
2	810	B	803	Edit Delete
3	809	N	85337096	Edit Delete
4	818	A	805	Edit Delete

[New Follow Me](#)

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Set your FollowMe number: Set a FollowMe number which could be a 'Local Extension' or an 'Outside Number'. Pls select a DialPlan for dial outside number

[List of Forward](#) Call Forward extensions are listed in the table.
[New Forward](#) Create a new Call Forward

Add a Follow Me [X]

Extension:

Ring lasting for seconds

Status: Always Busy No answer

Set your call forward number

Forward a Local Extension: Forward a Outside Number:

Select forward extension

[Save](#) [Cancel](#)

[Extension](#) Select a need to call forward extension
[Ring Time](#) Set the extension ring time
[State](#) Set state of the extension.(Disable, Always, Busy, No answer)
[Select forward extension](#) Select a call forward to extension

When you select "Forward a Outside Number" the follow page will be displayed.

X
Add a Follow Me

Extension:

Ring lasting for seconds

Status: Always Busy No answer

Set your call forward number

Forward a Local Extension: Forward a Outside Number:

Select DialPlan

Set forward outside number

[Select DialPlan](#) Select a Call forward to outside number using dialing rules

[Set forward outside number](#) Input a Call forward to outside number. (Notice: This number must be consistent with the corresponding DialPlan)

5.8. Time Based Rules

Time Based Rules
⌵

- Home
- Basic
- Inbound Control
- Advanced
- Operator
- Voicemail
- Conferencing
- Music on hold
- Call Parking
- DISA
- Follow Me
- Time Based Rules**
- Status
- System

X
Add new Time Rule

Rule Name : (Ex: July4)

Time & Date Conditions

Start Time: : : End Time: : :

Start Day: End Day:

Start Date: End Date:

Start Month: End Month:

Destination

if time matches:

if time did not match:

Set your FollowMe number: Set a FollowMe number which could be a 'Local Extension' or an 'Outside Number'. Pls select a DialPlan for dial outside number

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On this page, Define call routing rules based on date and time

Chapter6 Status Display

6.1. Call Logs

Call Logs Feb 20 2010 Go Call Logs Download Call Logs Delete

Caller ID	Destination	Call Start	Answered	Call End	Duration (sec)	Disposition
User3 <803>	814	2010-02-20 18:03:54	2010-02-20 18:03:54	2010-02-20 18:03:55	1	ANSWERED
User3 <803>	809	2010-02-20 18:03:30	2010-02-20 18:03:30	2010-02-20 18:03:33	3	ANSWERED
User3 <803>	814	2010-02-20 17:51:14	2010-02-20 17:51:14	2010-02-20 17:51:15	1	ANSWERED
User3 <803>	803	2010-02-20 17:51:02	2010-02-20 17:51:02	2010-02-20 17:51:09	7	ANSWERED
	s	2010-02-20 17:51:02		2010-02-20 17:51:02	0	NO ANSWER
User3 <803>	809	2010-02-20 17:50:53	2010-02-20 17:50:53	2010-02-20 17:50:59	6	ANSWERED
User3 <803>	809	2010-02-20 17:50:24	2010-02-20 17:50:24	2010-02-20 17:50:30	6	ANSWERED

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This web page will display call logs

- [Call Logs Download](#) download the call logs file
- [Call Logs Delete](#) delete the call logs file

6.2. Register Status

Register Status SIP Users Status IAX2 Users Status SIP Trunks Status IAX2 Trunks Status

SIP Users Status:

Name/username	Host	Dyn	Nat	ACL	Port	Status
830	(Unspecified)	D			0	Unmonitored
829	(Unspecified)	D			0	Unmonitored
828	(Unspecified)	D			0	Unmonitored
827	(Unspecified)	D			0	Unmonitored
826	(Unspecified)	D			0	Unmonitored
825	(Unspecified)	D			0	Unmonitored
824	(Unspecified)	D			0	Unmonitored
823	(Unspecified)	D			0	Unmonitored
822	(Unspecified)	D			0	Unmonitored
821	(Unspecified)	D			0	Unmonitored
820	(Unspecified)	D			0	Unmonitored
819	(Unspecified)	D			0	Unmonitored
818	(Unspecified)	D			0	Unmonitored
817	(Unspecified)	D			0	Unmonitored
816	(Unspecified)	D			0	Unmonitored
815	(Unspecified)	D			0	Unmonitored
814	(Unspecified)	D			0	Unmonitored
813	(Unspecified)	D			0	Unmonitored
812	(Unspecified)	D			0	Unmonitored
811	(Unspecified)	D			0	Unmonitored
810	(Unspecified)	D			0	Unmonitored
809	(Unspecified)	D			0	Unmonitored
808	(Unspecified)	D			0	Unmonitored
807	(Unspecified)	D			0	Unmonitored
806/806	(Unspecified)	D			0	Unmonitored
805	(Unspecified)	D			0	Unmonitored

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In this page, you can check SIP/IAX Users or Trunks Status.

6.3. System Info

ZYCOO

System Information

[Activate Changes](#) [Logout](#)

General [Resources](#)

General: Information about OS, Uptime, Asterisk, Date, Timezone and Hostname

Home

Basic

Inbound Control

Advanced

Status

Call Logs

Register Status

System Info

System

OS Version:
Linux IP PBX 2.6.22.18

Uptime:
17:16:35 up 10 days, 50 min.
Load Average: 0.03, 0.01, 0.00

Asterisk & GUI Build:
Asterisk 1.4.4
Zycoo System v3.0.7

Server Date & TimeZone:
Sun, 21 Feb 2010 17:16:36 +0800

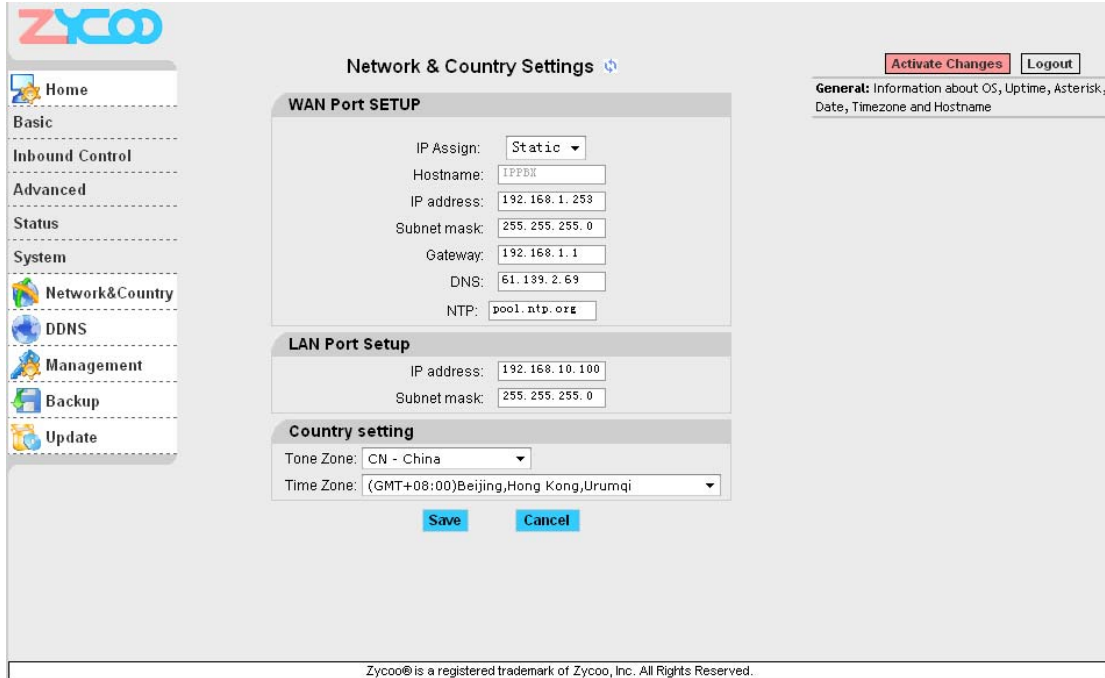
Hostname:
IPPBX

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In this page it will display nonce system info

Chapter7 System Management

7.1. Network and Country



On this page you can set WAN, LAN interface information and country.

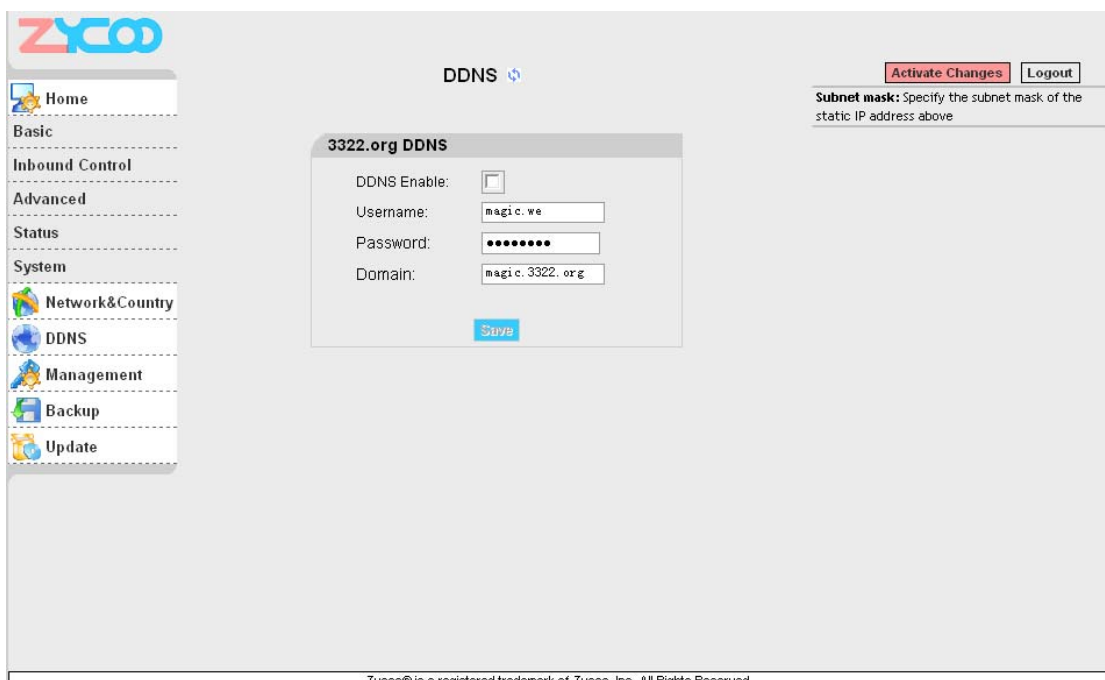
IP Assign: you can select STATIC, DHCP and PPPoE three mode

NTP: Set NTP server address.

Tone Zone: Set your Country, and use the Country Tone

Time Zone: Set your Time Zone

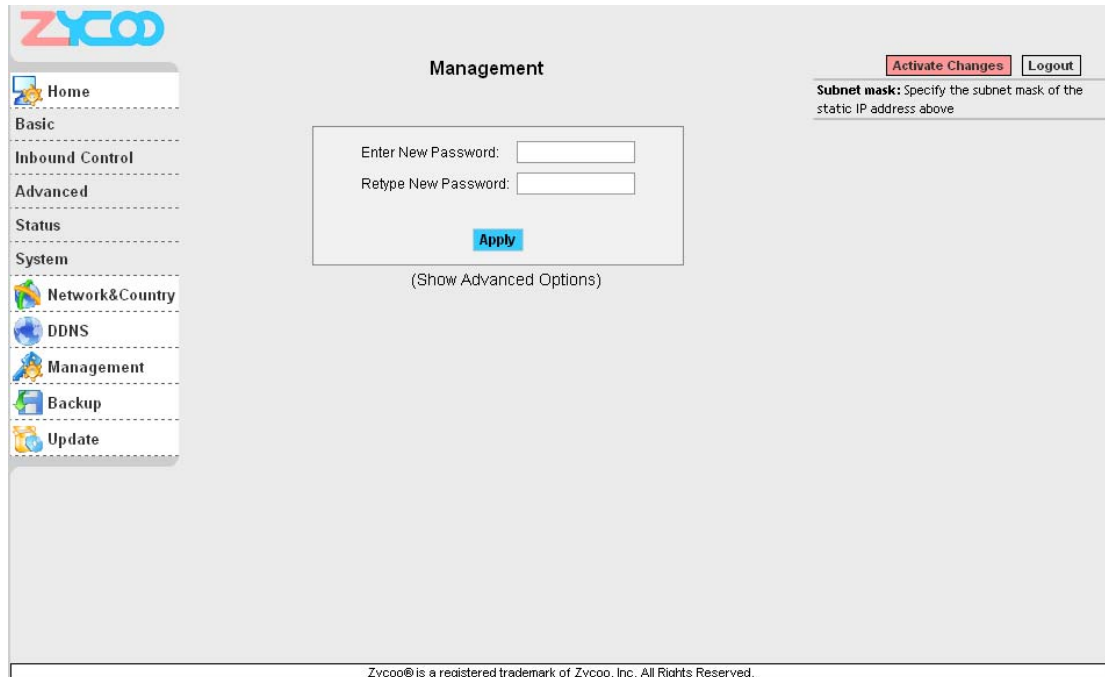
7.2. DDNS



On this page, you can set DDNS reference.

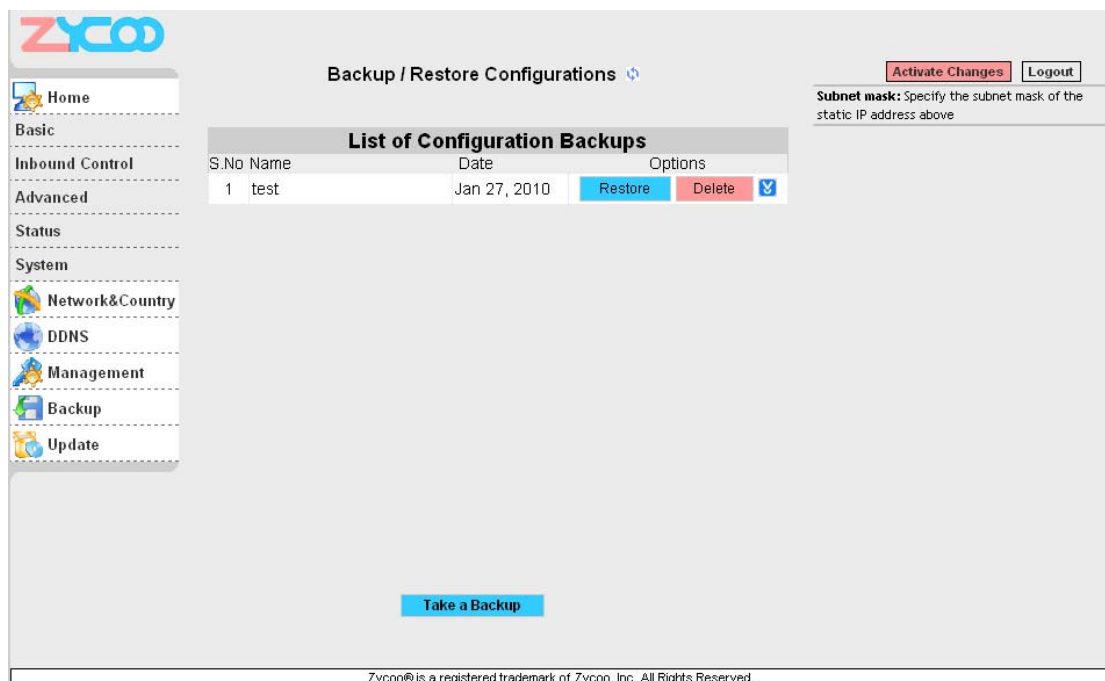
Notice: Now, it only supports 3322.org server. More other servers, you can customize based on your requirement

7.3. Management



On this page, you can change the administrator password (Default password: admin) And you can also set the advanced options about SIP and Zap protocol in the "Show Advanced Options" list, that is useful when you set connect two ipbx in different network.

7.4. Backup



On this page, clicking the "Take a Backup" button, you can backup once configuration

7.5. Upgrade

Zycoo

Home
Basic
Inbound Control
Advanced
Status
System
Network&Country
DDNS
Management
Backup
Update

Upgrade Package

Upgrade System Package
Enter The Package Name:
TFTP Server IP address:
Update

Upload IVR Prompts
Enter The Sound File Name: (*.gsm)
Note: Please use .gsm format voice file.
TFTP Server IP address:
Upload

Upload Backup File
Enter The Backup File Name:
Note: Don't change the backup file name.
TFTP Server IP address:
Upload

Activate Changes **Logout**

Subnet mask: Specify the subnet mask of the static IP address above

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In this page you can upgrade system package

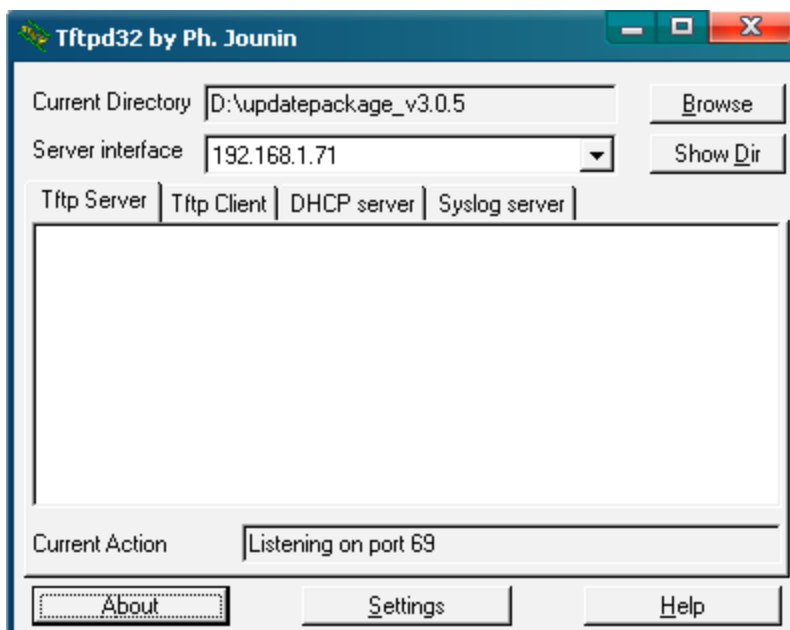
- **Enter The Package Name** Set system package name
- **TFTP Server IP address** Set TFTP server IP

How to upgrade:

Unzip the file you download, you will get a TFTP server and an upgrading packet.



Run the TFTP server, you will see below:



Enter the configuration page, then upgrading page;

Enter **The Package Name**, hereby it's zycoo-pbx-v3.0.5

Enter **TFTP Server IP address**, hereby it's

After done, click **Update** to update, then the system will reboot automatically.

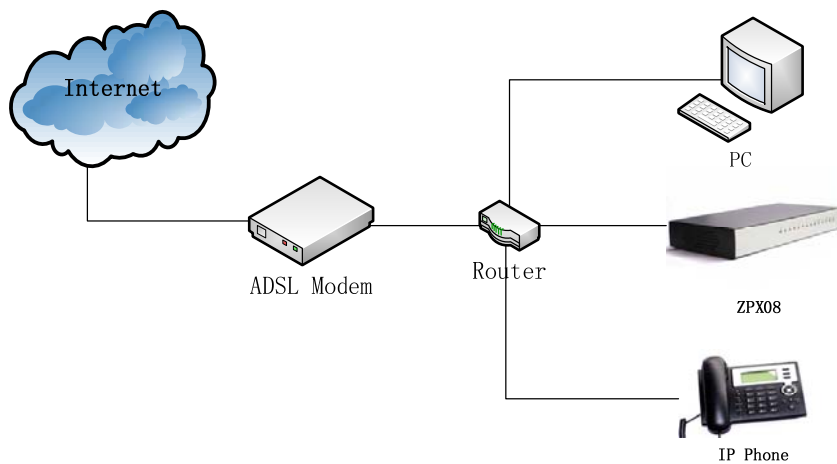
(Note: the upgrading will set your system as default, please make backup before you do it.)

Chapter8 Operating Instruction

8.1 How to link the IP PBX to the interwork

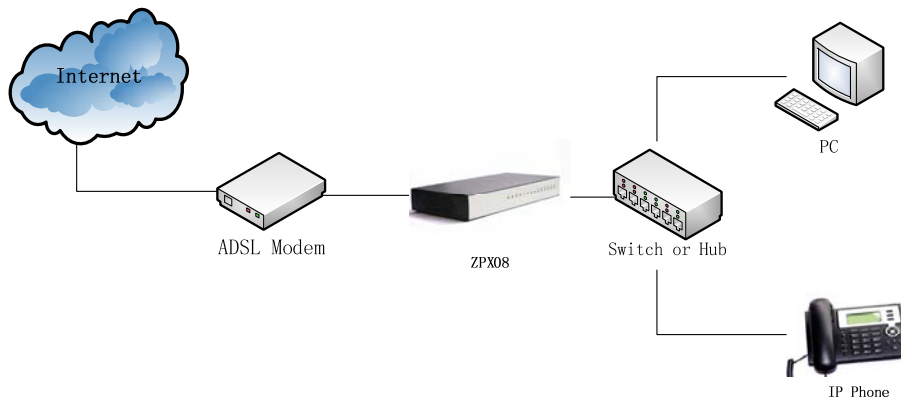
With Router

If your office access the public network with router, you can put the IPPBX behind the router. You should connect the Wan port of the IPPBX to the Lan ports of the router, and you also can connect HUB or Switch to the Lan ports of the IPPBX to let some PC or IP Phone to access the public network..



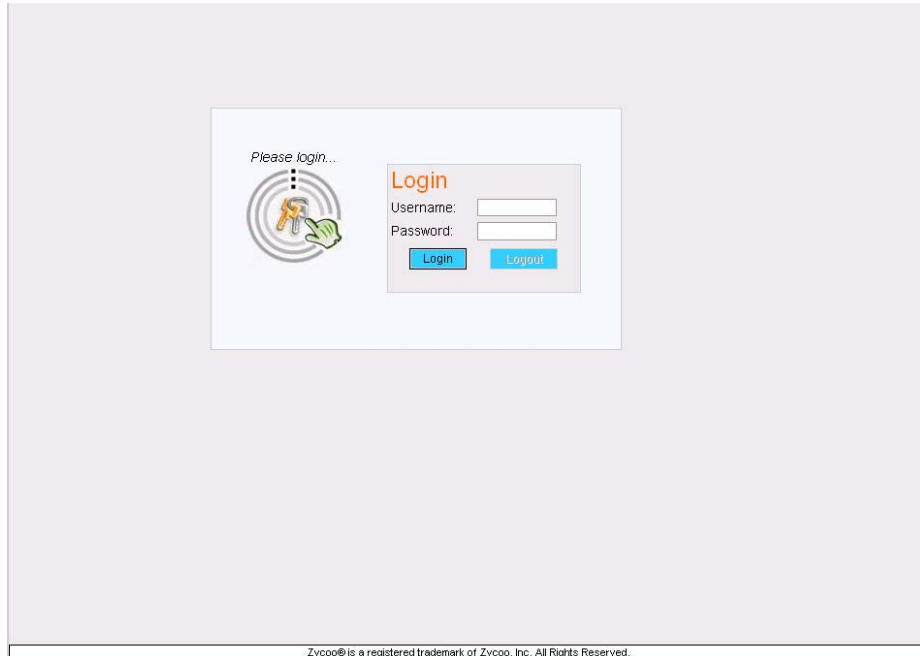
Without Router

If you have the public IP and want the IPPBX access the public network directly without router, then you should connect the Wan port of the IPPBX to the public network and connect HUB or Switch to the Lan ports of the IPPBX to let your PC access the public network..(If you want to access the public network through Modem, then you should use the PPPOE function of the IPPBX and let the IPPBX dial-up to connect the public network)

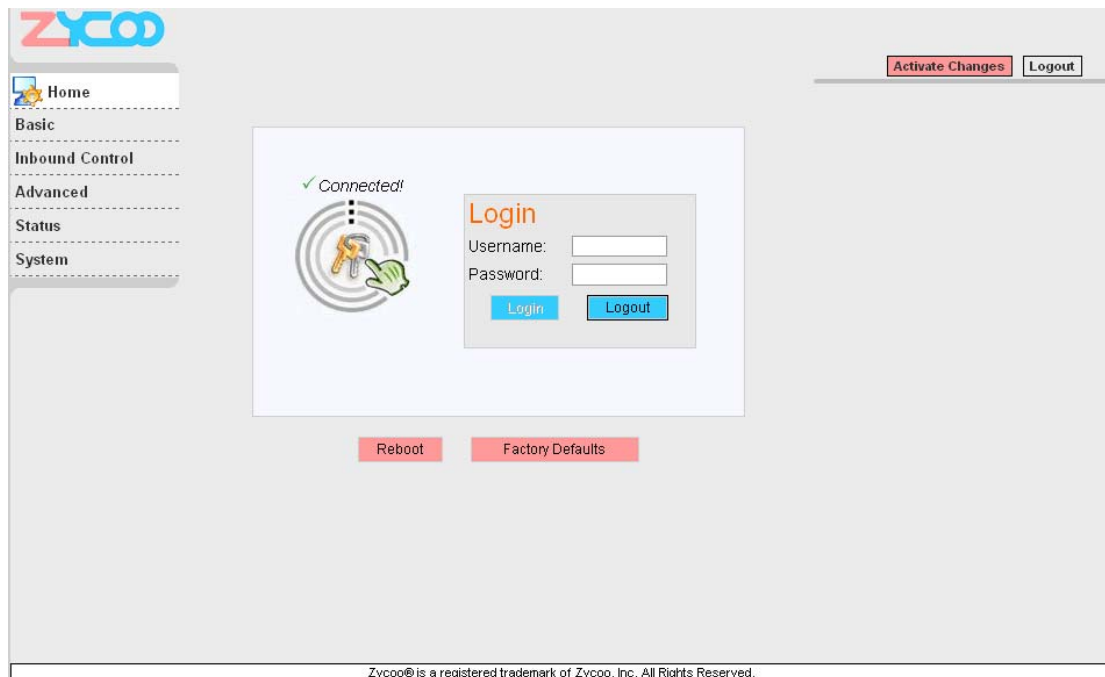


8.2 Log in to the system

After connecting the ippbx to the local area network. Launch the web browser on a computer that is in this local area network. Enter the IP address for the system (default: Wan port IP address is <http://192.168.1.100:9999>, Lan port IP address is <http://192.168.10.100:9999>) . The start web page will appear like this:



Enter Username and password (default username is **admin**, password is **admin**), then click login. Once the login is successful, the home page will be display:



With the zycoo GUI, you can configure extensions, conference, voicemail, Outbound Routers and etc. Each page of the GUI has three columns:

The left column present all the options tab that you can program the system. Click the

tab to go this kind of option setting page.

The middle column contains the primary content for each page.

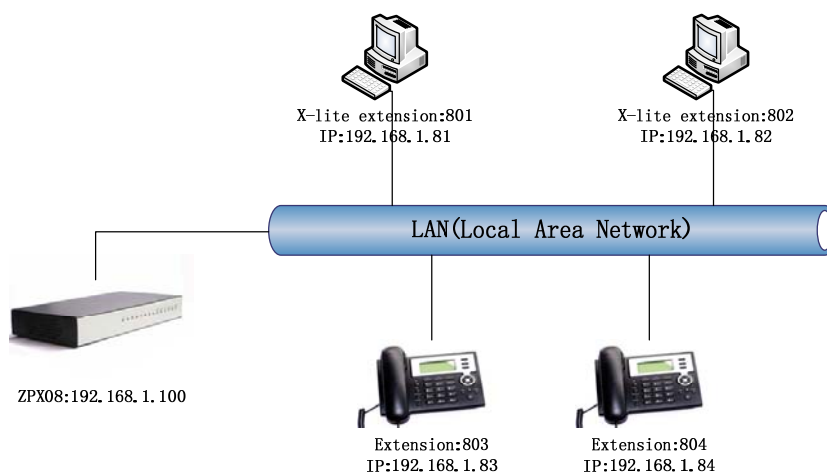
The right column of the user interface contains Tooltips. This area provides brief description for any options of the GUI

The home page is used for logoff, Reboot and Factory Defaults.

- **Logout:** To log out the zycoo GUI.
- **Reboot:** Reboot the ZPX08 system
- **Factory Defaults:** Restore all settings to factory default.
- **Activate change:** Made the change active for the current configuration after you make a configuration change on some page.

8.3 How to make a internal call

Making internal calls are the base requirement for a telephony system. Below are the settings for this usage. It is base on ZPX08, but setting is the same in other ZPX products.



Set User

Users:

The screenshot displays the Zycoo Extension Configuration interface. On the left, a navigation menu includes Home, Basic, Extensions, Trunks, Outbound Routes, Inbound Control, Advanced, Status, and System. The main area is titled 'Extension Configuration' and features a list of 25 users (801-825) on the left. The central panel is for configuring a selected extension, with fields for Extension, Name, Password, Caller ID, VM Password, E-mail, and Analog Phone. Below these are 'Advance Options' (Voicemail, SIP, Call Waiting, NAT, Can Reinvite, IAX, 3-Way Calling) and 'Codecs Configure'. Buttons for 'New', 'Delete', 'Save', and 'Cancel' are at the bottom. The right sidebar contains a warning: 'Extension: The numbered extension, eg. 888, that will be associated with this particular User / Phone.' The footer reads 'Zycoo® is a registered trademark of Zycoo, Inc. All Rights Reserved.'

There are 30 default users, the extensions number are 801~830

Set user, Extension is 803

Name, Password and Caller ID, etc.....

Select Dial Plan is DialPlan1

Set Extension 804 as the same way

Use a IP Phone based SIP protocol registered with the user.

Then you can use 803 call 804 successfully.

8.4 How to make a outbound call

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

Analog Ports: FXO ports of ZPX08, connect to local PSTN

VoIP Trunk: SIP or IAX trunk, connect to remote SIP/IAX server

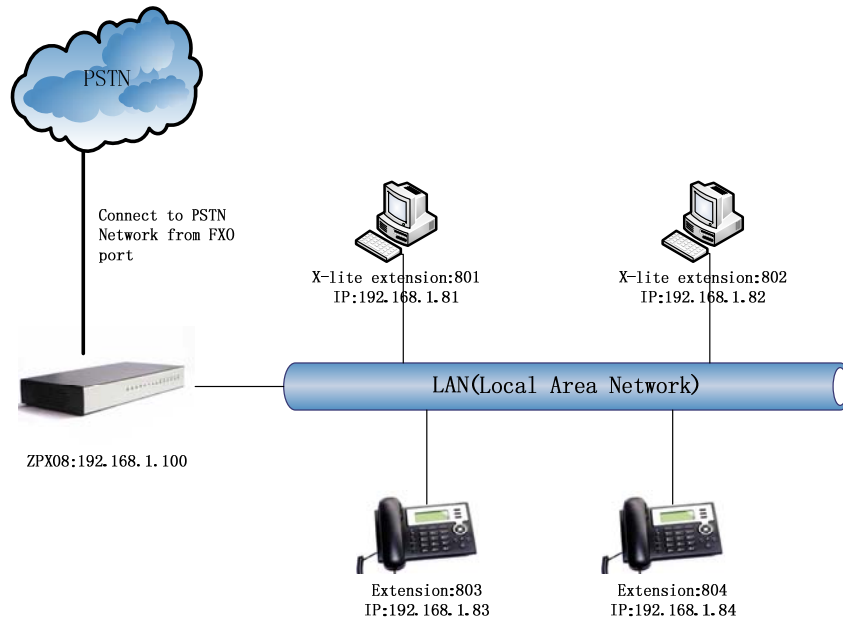
I am using ZPX08, the port1-4 are configured as FXO ports, port5-8 are configured as FXS ports. When a port is configured as FXO port, the corresponding LED shows **RED**. When a port is configured as FXS port, the corresponding LED shows **GREEN**.

What are FXO and FXS?

FXS (Foreign eXchange Station) is an interface which drives a telephone or FAX machine. FXS interfaces get phones plugged into them, delivery battery, and provide ringing. FXS interfaces are signalled with FXO signalling. **FXO** (Foreign eXchange Office) is an interface that connects to a phone line. They supply your PBX with access to the public telephone network. FXO interfaces use FXS signalling. FXS interfaces allow you to hook telephones to your PBX, and FXO interfaces allow you to connect your PBX to real analog phone lines.

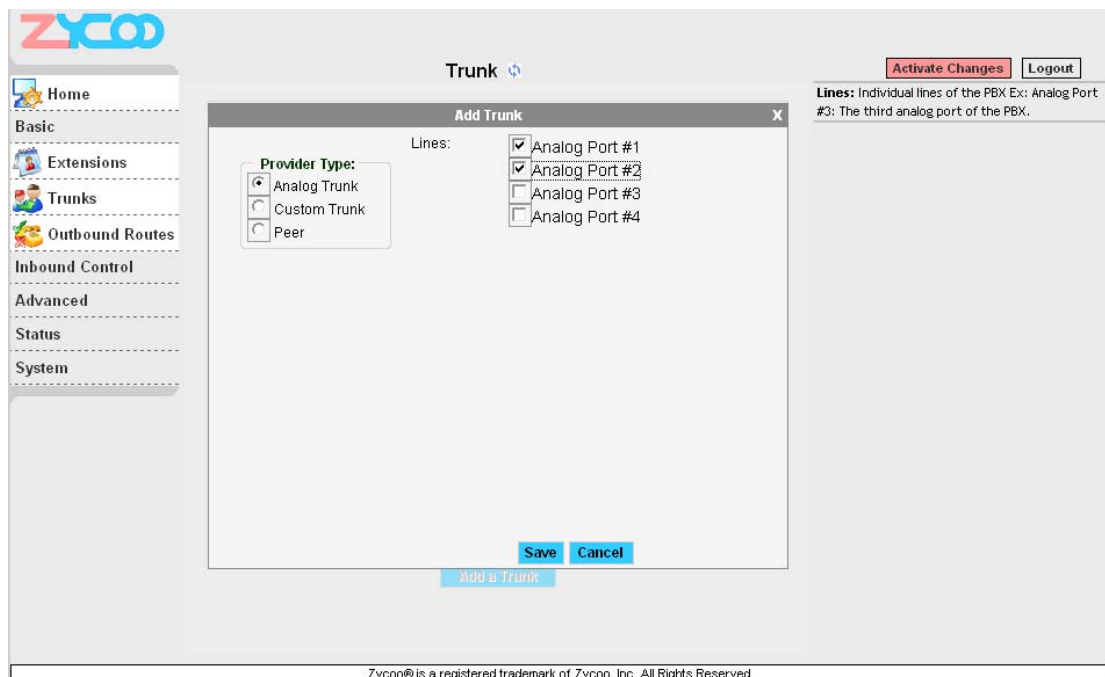
Make call via PSTN trunk

You can use the FXO trunking to make outgoing call via your local PSTN line. The set up is as per below:



Add Analog Trunk

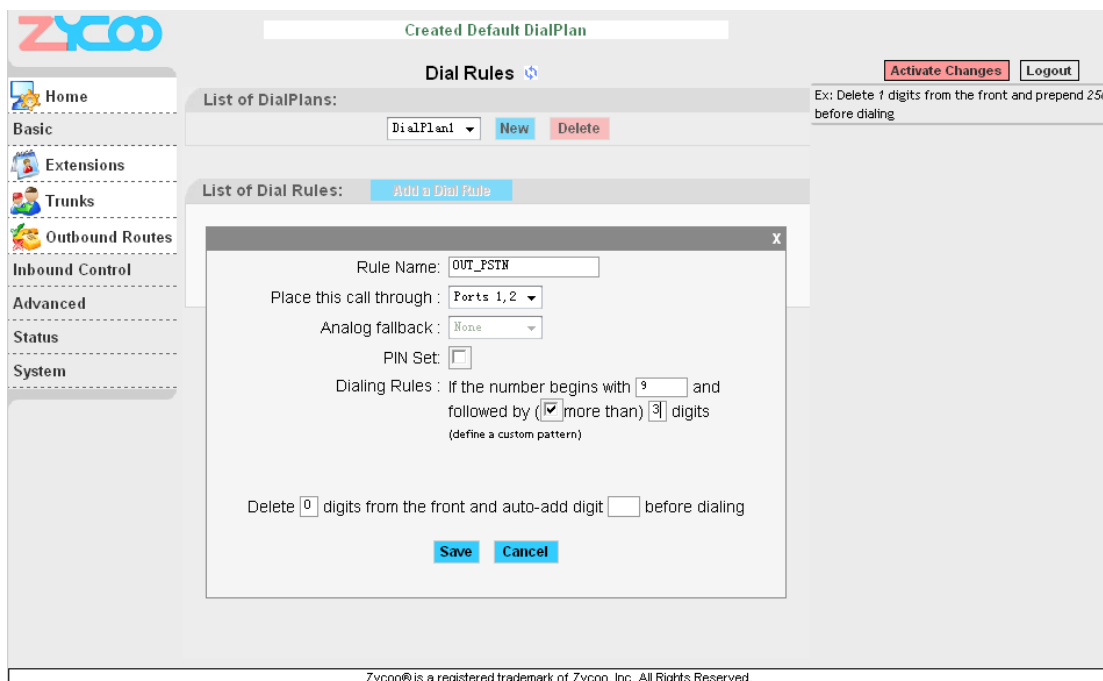
Trunks -> Add a Trunk:



Add Outbound Routers

In Outbound Routers -> add a Dial rule as below

Dial Rules

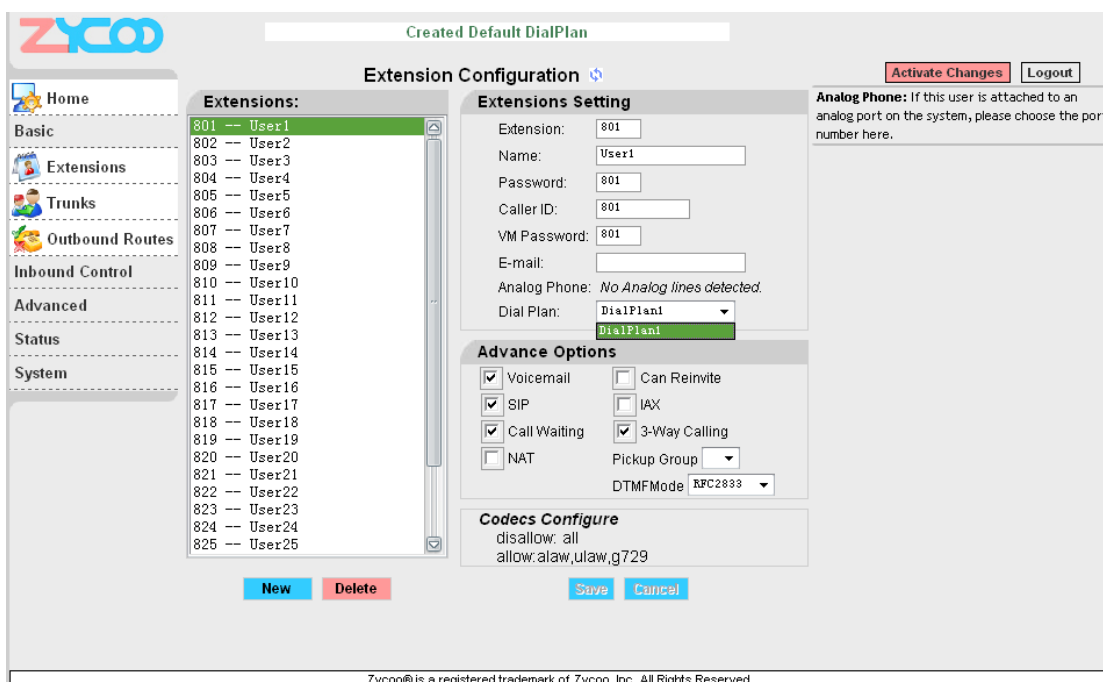


We have now added a Dial rule “OUT_PSTN” in the “DialPlan1”.

As we can see from the dialing rule of “OUT_PSTN”, all numbers start with 9 will be cut the first digit (‘9’) and sent to PSTN (port3 or port4).

Choose Dial Plan for extensions:

On the User page, edit the extensions to choose DialPlan1.

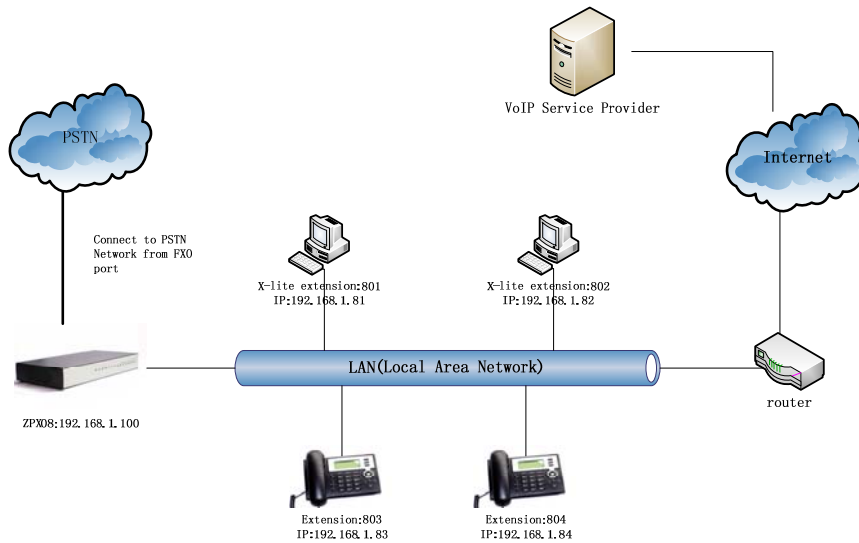


After we have done above, in the extension we can dial 9 + local number to dial out via PSTN line.

Make call via VoIP trunk

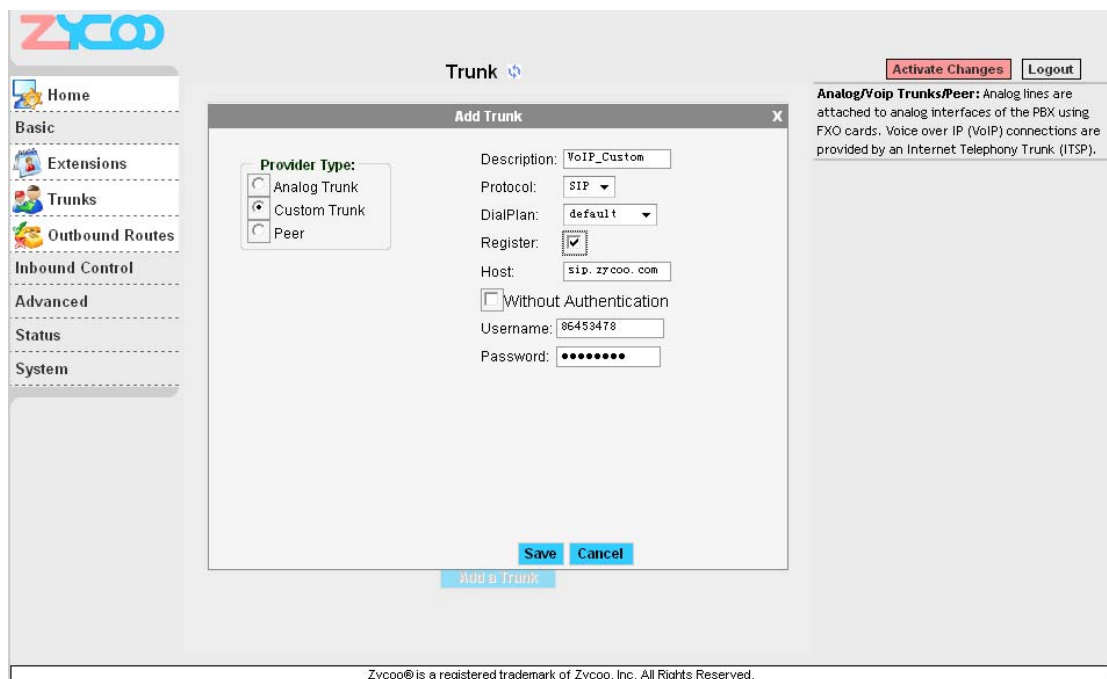
Via the voip trunking we can dial call via the voip service to reduce our cost when making

international calls.



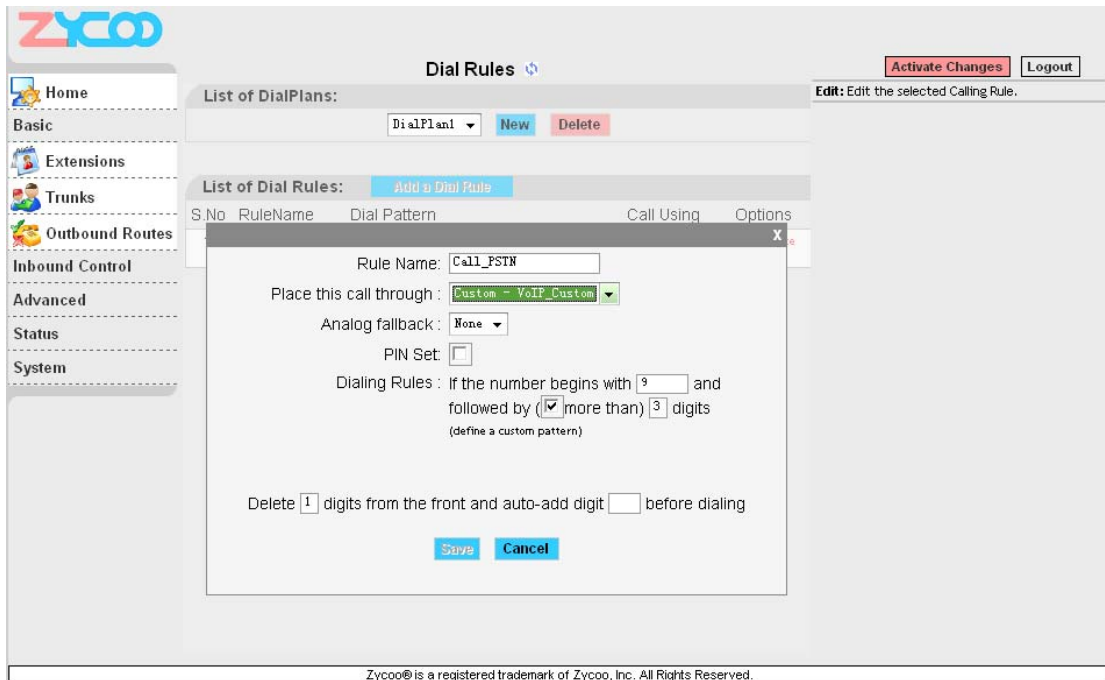
Add VoIP service provider

Trunk -> Add a Trunk:
Add a Custom Trunk.



Add Dial Rule

In Dial Rules -> add a new calling rule as below
Dial Rules



Now we have added a new calling rule “Out_VoIP” in the “DialPlan1”.

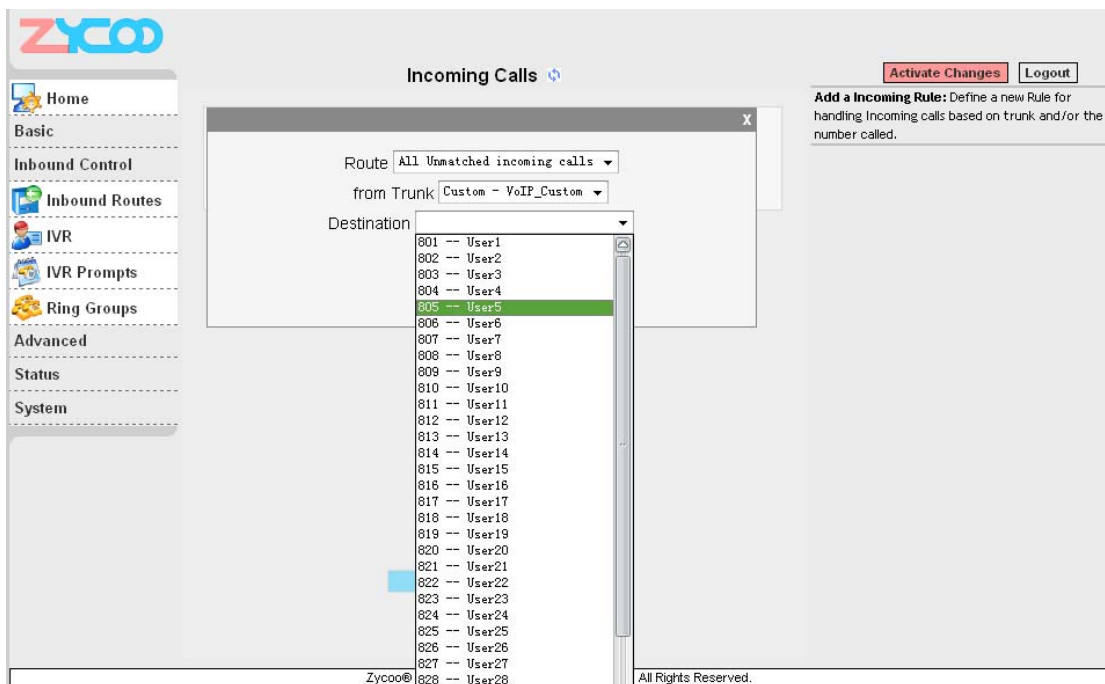
As we can see from the “Out_VoIP” dialing rule, all numbers start with 0 will be cut the first one digits (‘0’) and sent to my sip service provider.

The Out_PSTN is in the same DialPlan1. Since we have added this dial plan to the extensions in above, we don’t need to add dial plan again.

So when we have added two calling rules, any call start with 9 will be route to PSTN, and call starts with 0 will be route to VoIP.

8.5 How to make an incoming call

Add an Incoming call.



Select Route “All Unmatched incoming calls”

From provider “Port 3, 4”

To extension “801 – User1” (here, you can select a extension, a IVR or others)

Then, if there is incoming call from Port3 or port4 channel, the extension 801 will ring.

8.6 How to Set an incoming call to IVR based time rule

Add record a custom voice

Record -> Record a new voice

The screenshot displays the Zycoo web interface. On the left is a navigation menu with categories: Home, Basic, Inbound Control, Inbound Routes, IVR, IVR Prompts, Ring Groups, Advanced, Status, and System. The main content area is titled 'Record Voices For Custom IVR' and features a table of recorded voices. The table has columns for S.No, Name, and Options. Two rows are visible: one for 'closed.gsm' and one for 'welcome.gsm'. A modal dialog box titled 'Record a new Voice' is open in the foreground. It contains a 'File Name' input field with 'welcome' typed in, an 'Extension used for recording' dropdown menu currently showing '801 -- User1', and two buttons: 'Record' and 'Cancel'. At the bottom of the main interface, there is a 'Record a new voice' button. The footer of the page reads 'Zycoo® is a registered trademark of Zycoo, Inc. All Rights Reserved.'

Set the record name is “Welcome”

Choose a extension used for recording, here we use EXT 801

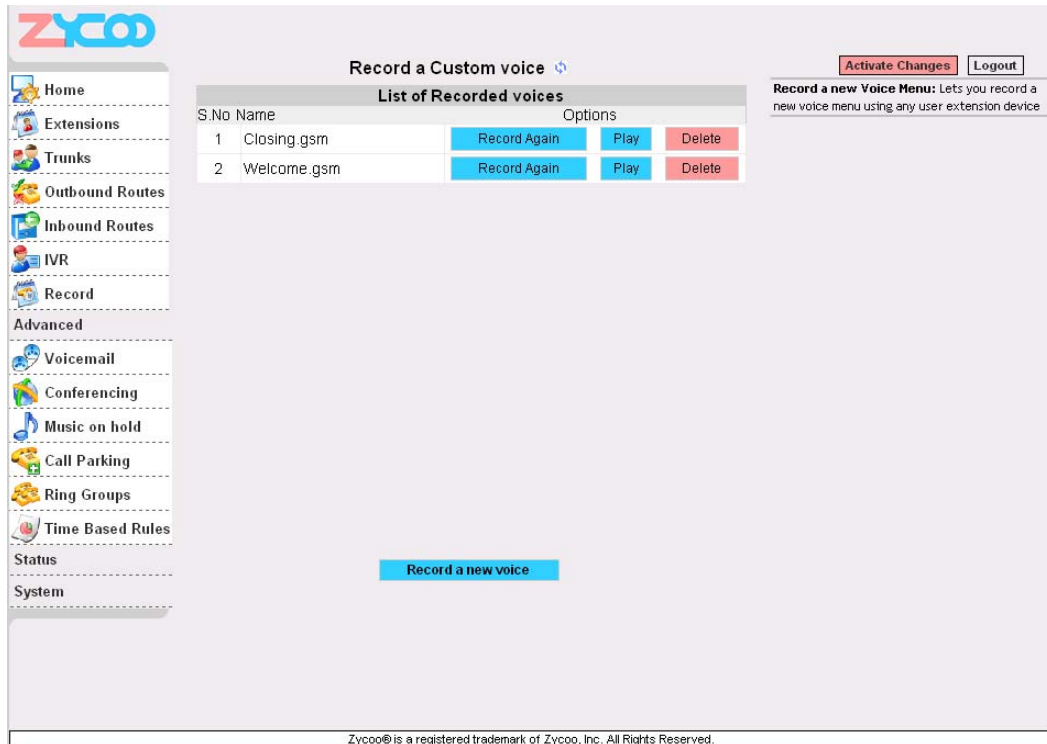
Click Record button

Then, the extension 801 will ring

Pick up the phone record “Welcome” message

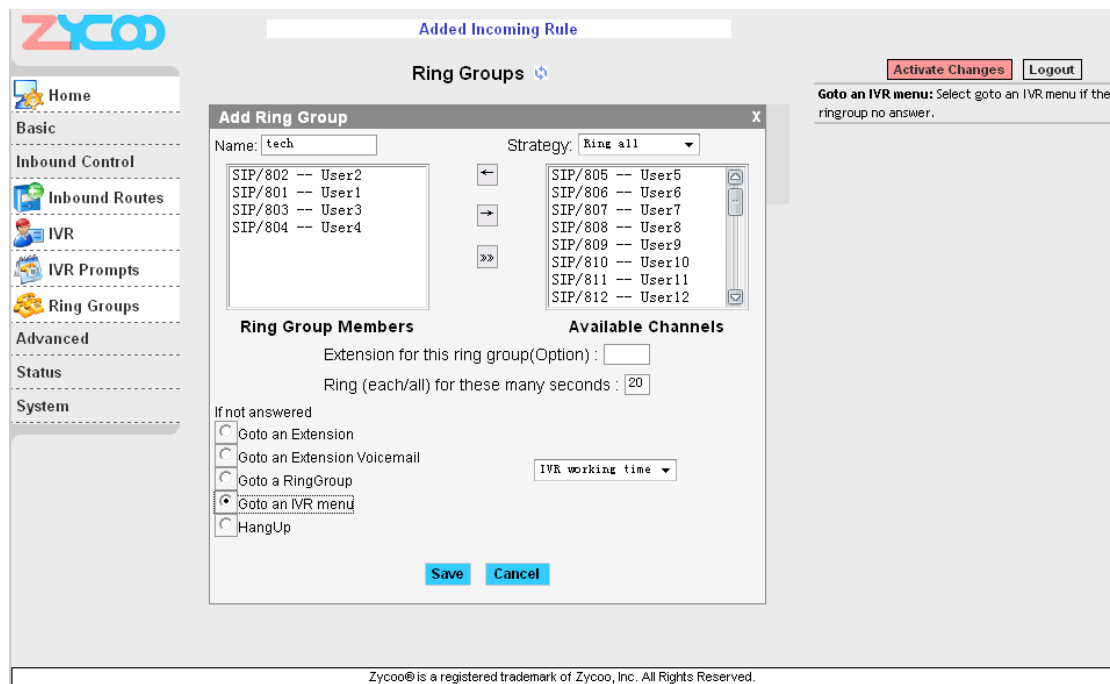
Then hangup and finish the record .

Use the same way to record “Closing” message



Add a Ring Group

Ring Group -> New Ring Group



Example:

Name the ring group "tech"

Choose the group members whose extensions are "801. 802, 803, 804"

"if no answered", choose "goto IVR"-- "working time"

Click "Save" button

Ring Groups

[Activate Changes](#) [Logout](#)

S.No	Ring Group	Options
1	tech	Edit Delete

[New Ring Group](#)

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Set IVR

IVR

IVR

[Activate Changes](#) [Logout](#)

IVR Menu:

- IVR - working time
- IVR - closed time

IVR Setting

Name: Extension:

Welcome Message

Please Select:

Dial other Extensions?

Keypress' Events

Key	Action
0	Goto Extension 801
1	Goto Extension 802
2	Goto RingGroup tech
3	Disabled
4	Disabled
5	Disabled
6	Disabled
7	Disabled

[New](#) [Delete](#) [Save](#) [Cancel](#)

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Select IVR-working time, Set welcome message is "Welcome"

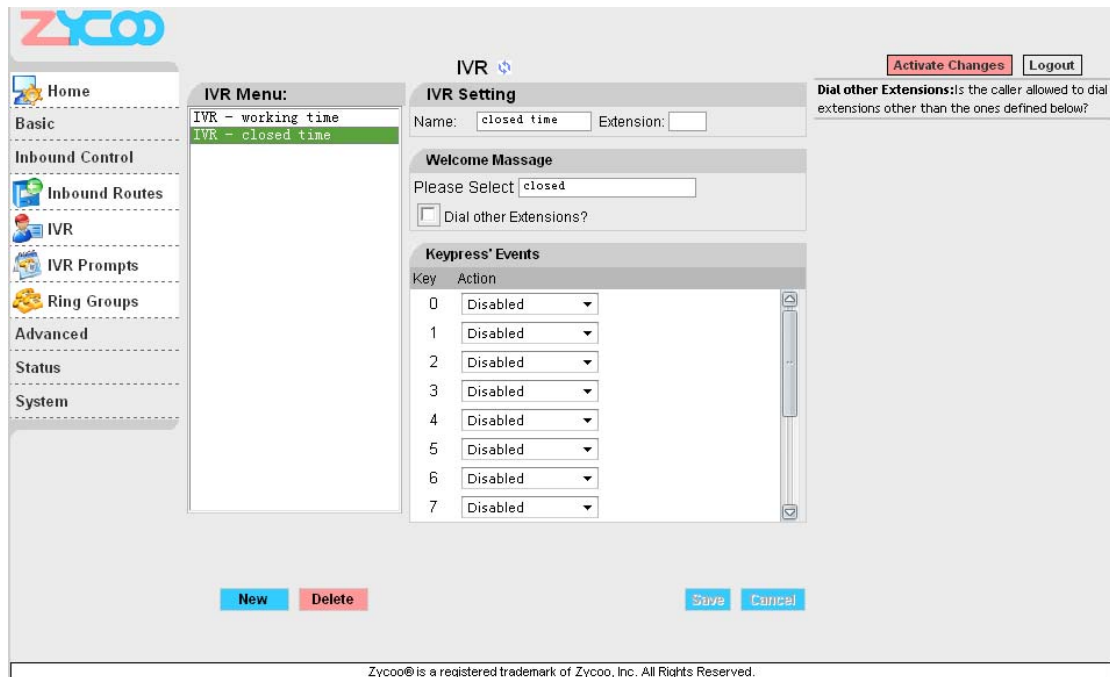
Set keypress' Events

Dial "0" go to extension 805

Dial "1" go to extension 806

Dial "2" go to ringgroup tech

Click Save button

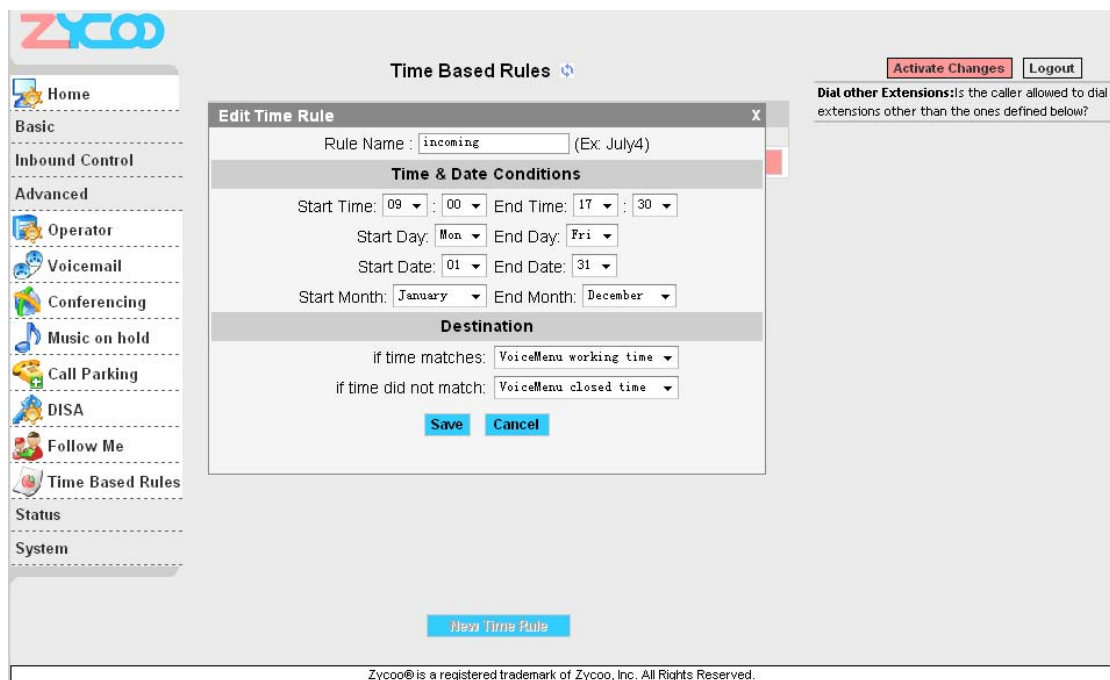


Then set IVR-closed time

Set welcome message is "Closing"

Add a Time Rule

Time Based Rules -> New Time Rule



Set a Rule Name, eg: incoming

Set the Time & Date Conditions

"If time matches" --- go to "working time"

"If time not match" --- go to "closed time"

Click the save button, saved the configuration

Zycoo

Time Based Rules

Activate Changes Logout

Keypress Events: Define the actions that occur when a user presses the corresponding digit.

S.No	RuleName	Options
1	incoming	Edit Delete

New Time Rule

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Add a Trunk

Trunks -> add a Trunk

Zycoo

Trunk

Activate Changes Logout

Lines: Individual lines of the PBX Ex: Analog Port #3: The third analog port of the PBX.

Add Trunk

Provider Type:

Analog Trunk

Custom Trunk

Peer

Lines:

Analog Port #1

Analog Port #2

Analog Port #3

Analog Port #4

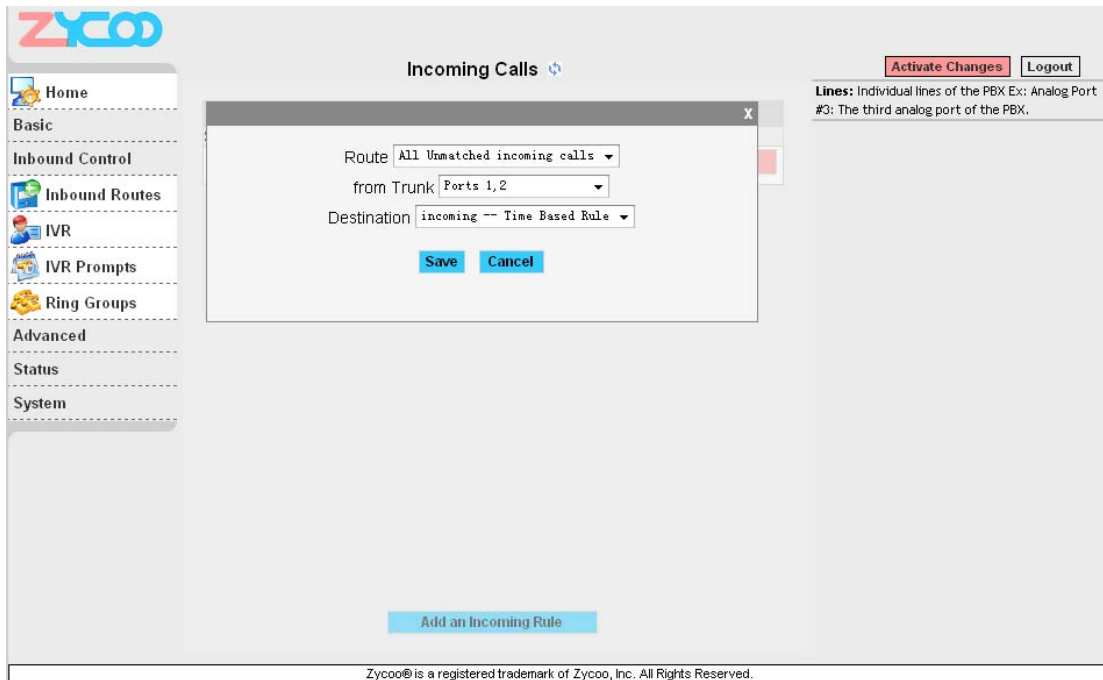
Save Cancel

Add a Trunk

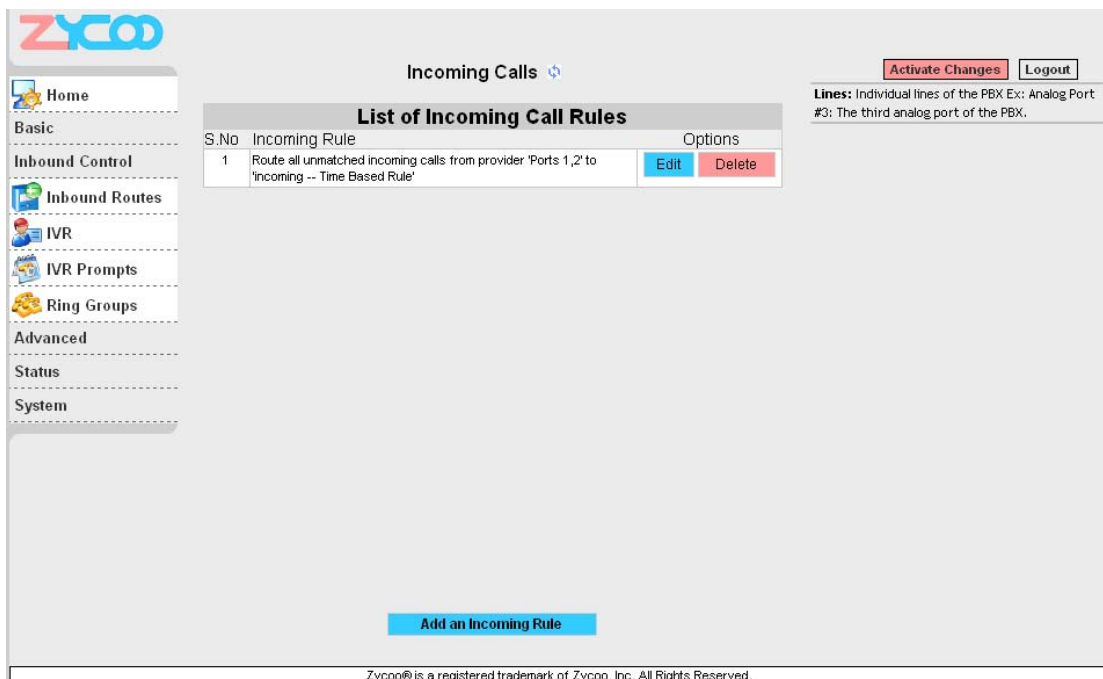
Zycoo® is a registered trademark of Zycoo, Inc. All Rights Reserved.

Add an incoming router

Inbound routers -> add an incoming rule



Select Route: All Unmatched incoming calls
 From provider: Ports 3, 4
 To extension: incoming—Time Based Rule

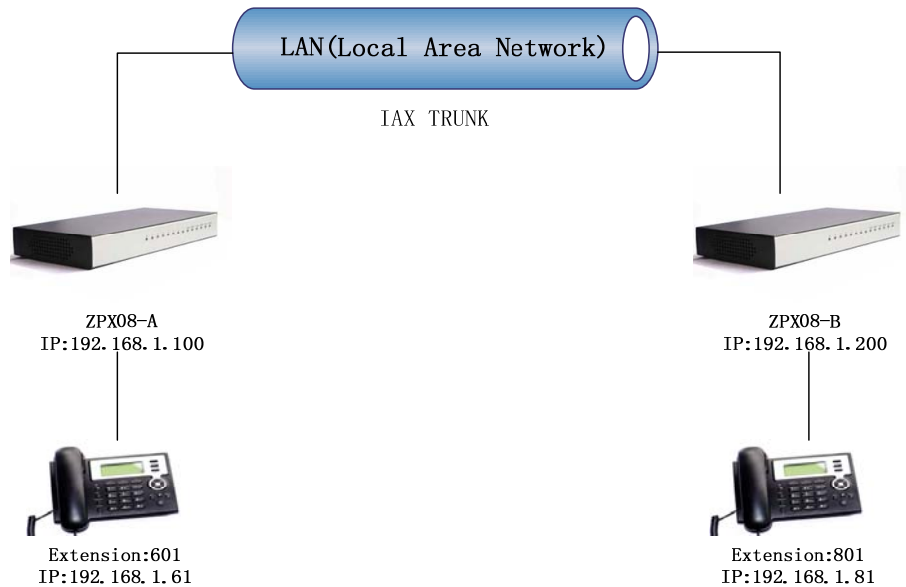


Then click Activate Changes, Made the change active for the current configuration

8.7 Link two IPPBX in the same network

The simplest case to link two IPPBX together in the same network. We start from this and then try to expand to different network. We use ZPX08 here, same method for other model IPPBX products.

Below is the structure of how to link two IPPBX in the same LAN:



The method of connecting two ZPX08 in different location is:

- 1) Register the ZPX08-A as an peer in ZPX08-B(via IAX2 trunk),so the extensions in ZPX08-A can make calls to ZPX08-B's extensions via this "special" trunk.
- 2) Use the reverse method in ZPX08-B to register to ZPX08-A.

In above structure:

- 1) ZP302A registers to ZPX08-A as an extension 601.
- 2) ZP302B registers to ZPX08-B as an extension 801.
- 3) All the extensions under ZPX08-A are in the format 6XX.
- 4) All the extensions under ZPX08-B are in the format 8XX
- 5) Extensions under ZPX08-A can make calls to extension under ZPX08-B use format 8XX.
- 6) Extensions under ZPX08-B can make calls to extension under ZPX08-A use format 6XX.
- 7) The two ZPX08 links each other via IAX2 trunk.

Step 1: Set up a peer 699 in ZPX08-A

In the page Trunks→ Add a Trunk

The screenshot shows the configuration interface for adding a trunk. On the left, under 'Provider Type', the 'Peer' option is selected. On the right, the following fields are filled: Peer Name: PXB_users, Protocol: IAX, DialPlan: default, Host: dynamic. There is an unchecked checkbox for 'Without Authentication'. The Username field contains '699' and the Password field contains three dots.

Peer Name: ZPXB_Users ;
 Peer Username: 699 ; Account of this Peer
 Password: 699 ;IAX2 Log on password
 Advance Options:Select IAX protocol

Step 2: Set up an IAX trunk in ZPX08-B to link to ZPX08-A via this ZPX08B_Users extension.

In the page Trunks--> Add a Trunk

Add Trunk

Provider Type:

- Analog Trunk
- Custom Trunk
- Peer

Description:

Protocol:

DialPlan:

Register:

Host:

Without Authentication

Username:

Password:

Step 3: Set Dial Rule in ZPX08-B, all calls start with 6 will be sent to ZPX08-A.

In the page: Outbound Routers --> Add a Dial Rule

List of Dial Plans:

List of Dial Rules:

Rule Name:

Place this call through:

Analog fallback:

Dialing Rules: If the number begins with and followed by digits or more
(define a custom pattern)

Strip digits from the front and prepend before dialing

Step 4: Set the user Dial Plan in ZPX08-A,

In the page: Extensions → Dial Plan

Extensions Setting

Extension: 601
Name: test
Password: 601
VM Password: 1234
Caller ID: 601
Analog Phone:
Dial Plan: DialPlan

Advance Options

Voicemail Can Reinwrite
 SIP IAX
 Call Waiting 3-Way Calling
 NAT

rfc2833 DTMFMode

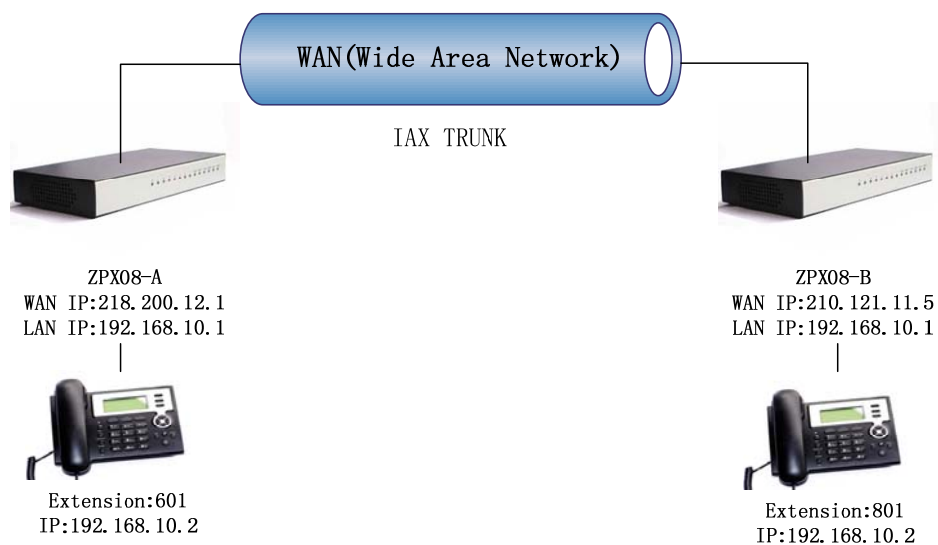
Active the change and apply the test:

1. Register an IP phone ZP302B to ZPX08-B with 801 extension.
 2. Register an IP phone ZP302A to ZPX08-A with 601 extension.
 3. Use 801 to dial 601. And you can see 601 will ring and you can pick up the calls.
- Above is the way to router ZPX08-B's call to ZPX08-A, the method to link ZPX08-A to ZPX08-B is the same as above.

8.8 Link two IPPBX in different network

Two ZPX08 are in internet

The generally environment for two ZPX08 in different location is: two ZPX08 are both in the internet and using the public IP.



The configuration is same with "Link two ZPX08 in the same network".

But when you set the trunk, you must use the public ip.

Like the follow:

Set up an IAX trunk in ZPX08-B to link to ZPX08-A via this ZPX08B_Users extension.

In the page Trunks--> Add a Trunk

Provider Type:

Analog Trunk

Custom Trunk

Peer

Description:

Protocol:

DialPlan:

Register:

Host:

Without Authentication

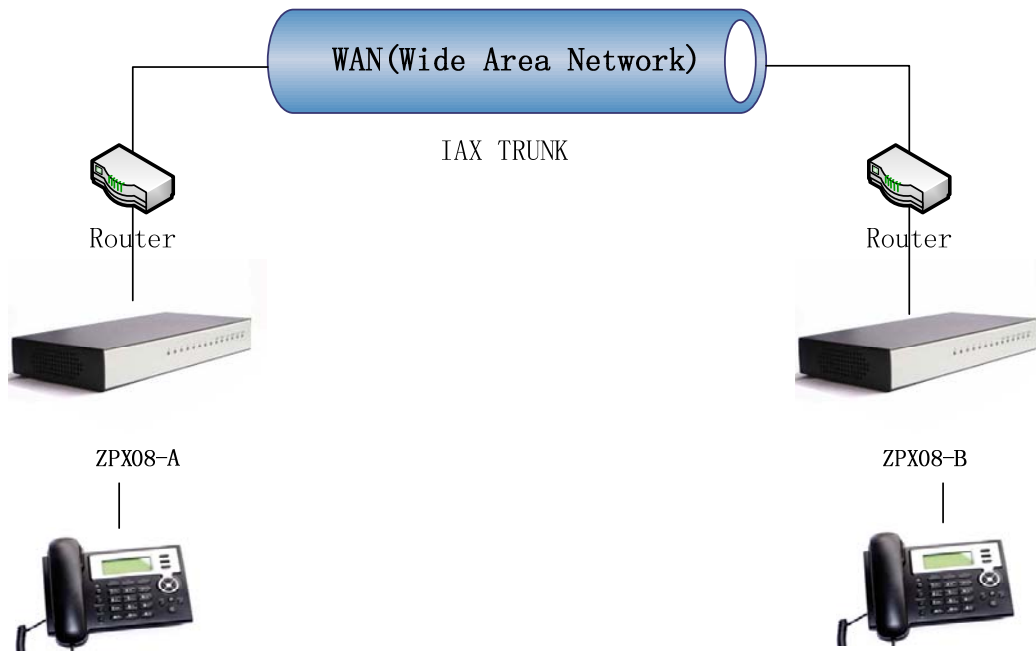
Username:

Password:

Two ZPX08s are behind router.

The generally environment for two ZPX08 in different location is: two ZPX08 are both behind router and using the private IP.

Since the ZPX08 doesn't have the public IP, we need to do port forwarding in the router and make ZPX08 is reachable to others.



Step 1: Set port forwarding in the router for ZPX08-A

The ZPX08-A is behind the router, to register to ZPX08-A via the internet, you need to forward the IAX2 port in your router, so all the packets received on the router WAN port (210.11.25.127:4569) will be forwarded to the ZPX08-A (192.168.1.21:4569). Below is the setting page in a linksys router:

Applications & Gaming						
Setup	Security	Applications & Gaming	Administration	Status		
Port Range Forwarding	Port Triggering	UPnP Forwarding	DMZ			
UPnP Forwarding						
Application	Ext.Port	TCP	UDP	Int.Port	IP Address	Enabled
FTP	21	<input checked="" type="radio"/>	<input type="radio"/>	21	192.168.1.0	<input type="checkbox"/>
Teinet	23	<input checked="" type="radio"/>	<input type="radio"/>	23	192.168.1.0	<input type="checkbox"/>
SMTP	25	<input checked="" type="radio"/>	<input type="radio"/>	25	192.168.1.0	<input type="checkbox"/>
DNS	53	<input type="radio"/>	<input checked="" type="radio"/>	53	192.168.1.0	<input type="checkbox"/>
TFTP	69	<input type="radio"/>	<input checked="" type="radio"/>	69	192.168.1.0	<input type="checkbox"/>
finger	79	<input checked="" type="radio"/>	<input type="radio"/>	79	192.168.1.0	<input type="checkbox"/>
HTTP	80	<input checked="" type="radio"/>	<input type="radio"/>	80	192.168.1.199	<input checked="" type="checkbox"/>
POP3	110	<input checked="" type="radio"/>	<input type="radio"/>	110	192.168.1.0	<input type="checkbox"/>
NNTP	119	<input checked="" type="radio"/>	<input type="radio"/>	119	192.168.1.0	<input type="checkbox"/>
SNMP	161	<input type="radio"/>	<input checked="" type="radio"/>	161	192.168.1.0	<input type="checkbox"/>
ssh	2020	<input checked="" type="radio"/>	<input type="radio"/>	22	192.168.1.235	<input checked="" type="checkbox"/>
http1	8080	<input checked="" type="radio"/>	<input type="radio"/>	80	192.168.1.29	<input checked="" type="checkbox"/>
http2	8090	<input checked="" type="radio"/>	<input type="radio"/>	80	192.168.1.209	<input checked="" type="checkbox"/>
IAX	4569	<input checked="" type="radio"/>	<input type="radio"/>	4569	192.168.1.21	<input checked="" type="checkbox"/>
IAX2	4569	<input type="radio"/>	<input checked="" type="radio"/>	4569	192.168.1.21	<input checked="" type="checkbox"/>

Step 2. Set up the service provider and calling rule in ZPX08-B to make it register to ZPX08-A. This method is almost the same as above, EXCEPT you need to use the 210.11.25.127 as the service provider instead of 192.168.1.21.

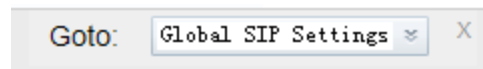
Step 3. Use the same method do port forwarding in router-B for ZPX08-B. Your public address from network provider maybe a dynamic ip which will be changed periodically. To overcome the problem of dynamic ip, you may need to use the DDNS service , for more info please google the internet.

Step 4. How to resolve problems about hearing only on one side:

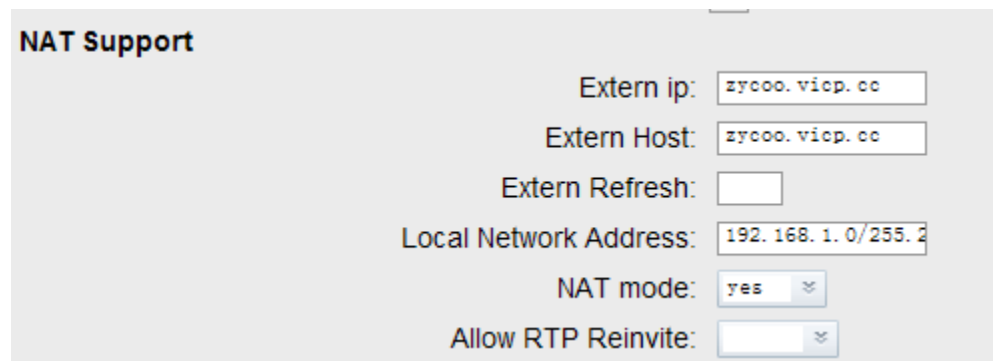
If your IPPBX behind the Router,you should build a IP Address Map:

As follow:

Mangerment---->Show Advanced Options ---->GOTO : Golbal SIP Settings



--->NAT Support

A screenshot of the 'NAT Support' configuration page. The page has a title 'NAT Support' and several configuration fields: 'Extern ip:' with the value 'zycoo.vicp.cc', 'Extern Host:' with the value 'zycoo.vicp.cc', 'Extern Refresh:' with an empty text box, 'Local Network Address:' with the value '192.168.1.0/255.255.255.252', 'NAT mode:' with a dropdown menu set to 'yes', and 'Allow RTP Reinvite:' with a dropdown menu set to 'yes'.

- Extern ip** Replace with your external ip address this your public IP or domain
- Extern Host** Replace with your external ip address this your public IP or domain
- Extern Refresh** Set time for fresh,default 10
- Local Network Address** Replace with your local network address and mask
- NAT mode** If your IPPBX behind the Router, set default yes