



ZX100 IP PBX User's Manual

The information contained in this document is subject to change at any time without prior notification. Specifications of the product are subject to change at any time without notice. If you want to learn more info about our product, please visit our web www.zycoo.com.

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Chapter1 Brief Introduction

Thank you for your purchasing the ZX100 series of IP PBX with powerful network function,such as NAT map,router,VPN,Firewall,DDNS,etc. The all-in-one ZX100 IP PBX can not only provide the traditional basic PBX features(call hold, call forwarding, call waiting and so on)as well as enhanced features such as recording,visual voice mail, music on hold ,and auto attendant. In addition, the ZX100 IP PBX supports innovative functionality like private VoIP networking, remote access, superior VoIP voice quality with advanced audio processing, and the revolutionary ability to traverse a NAT and firewall. With Zycoo VoIP solutions, the middle business can quickly deploy VoIP networks to connect multiple branch locations over the Internet without the need to change the current equipment or dial plan. By using the ZX100 IP PBX, the customers can take advantage of the VoIP services provided by the ITSPs(Internet Telephony Service Providers)or traditional telephony services, reduce intra-company telephony expenses, and allow VoIP remote access anywhere via the internet.

Chapter2 Safety Notice

Please read the following safety notices before installing or using this IP PBX. They are crucial for a safe and reliable operation of the device.

- Before using the external power supply in the package, please check with residential power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it, otherwise, it may cause fire or electric shock.
- The plug-socket combination must be accessible at all times because it serves as the main disconnecting device.
- Do not drop, knock or shake it. Rough handling can break internal circuit boards.
- Do not install the device in places where there is direct sunlight. Also do not place the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposing the device to high temperature, below -10°C or high humidity. Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling to the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock or breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug or phone line, it may cause an electric shock.
- Do not install this device in an ill-ventilated place.
- You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

Chapter3 ZX100 Specification

3.1 Model

ZYCOO ZX100 Series IPPBX product line include **ZX100-16,ZX100-E1**, so far, since they have almost the same software and structure so we will use ZX100-A16 as the demo unit on this article.

	Model	FXS	FXO	E1
ZX100-A16	A16016	0	16	0
	A16214	2	14	0
	A16412	4	12	0
	A16610	6	10	0
	A16808	8	8	0
ZX100-E1	E1	0	0	1

3.2 System Features

ZX100 series of IP PBX is not only the full feature phone system, but also a professional router device. The all-in-one ZX100 IP PBX can not only provide the traditional basic PBX features (call hold, call forwarding, call waiting and so on) as well as enhanced features such as recording server, voice mail server, conference server, etc. In addition, it supports powerful network management, such as NAT map, router, VPN, Firewall, DDNS, etc

- Up to 80 concurrent calls.
- Above 1000 registers
- Up to 1000 hours recording (can be extended)
- Configuration by Web
- Built-in SIP/IAX Server
- Build in Voice Mail Server
- Codec: G.711-Ulaw, G.711-Alaw, G.726, G.729, GSM, SPEEX
- SIP/IAX Extensions (connect with IP Phone)
- Zap Extensions (connect with Analog Phone)
- Call Forward/Call Hold/Call Transfer/Call Waiting/Caller ID
- Flexible Dial Plan
- Ring group
- Conference Room
- IVR and Auto attendant
- Multimedia Music On Hold and Ring Back
- Call Monitoring
- Video Call Support
- DISA setting
- Call parking
- Call Paging and intercom
- Follow me Setting

- Call Logs check and download
- Support IP Phone with Key function
- BLF(Busy Lamp Field)
- Static/DHCP/PPPoE network access
- System backup and store
- Set system time manually
- VPN Client (support N2N)
- DDNS Client (support Dyndns.org)
- Router setting
- NAT map
- IP Filter
- Codec Negotiation/Echo cancelation/VAD.etc
- FAX T.38

3.3 Interface&Panel

Here, we take ZX100-A16 as the sample to show the interface and the indicators at the back and frond panel.

1) Back panel



- 16 * Analog Ports
- 2 * Network Interface (RJ45)
- 1 * Power Interface(DC 110V~240V)
- 1 * Power Switch

2) Frond Panel



Mark	Function	Status	Description
PWR	Power Status	On	Power On
		Off	Power Off
HDD	Disk Status	Wink	System working
		Off	System Failed
PWR	Power Button		Power On/Power Off the system
RST	Reset Button		Restart the system
LCD	System Info		Display the current System info
1-16	Analog Modules Status	Green	FXS/FXO channels

		Off	Failed
--	--	-----	--------

3) Hardware

- CPU:ATOM N270, 1.6 Ghz x86 CPU
- RAM: 2 GB DDR2 RAM
- Storage: 320G mobile hard disk drive
- PSU: 200W Power Supply with industry standard
- Dimensions: W: 465 mm, H: 44.5 mm , D: 370 mm
- Weight: 7.0Kg

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

- ZX100 IPPBX 1
- Power Cable 1
- VGA Cable 1
- USB Cable 2

3.4 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. Web GUI username: **admin**
4. Web GUI password: **admin**

3.5 Default Feature Key

1. Press '**11' Playback the IP Address of WAN port
2. Press '**12' Playback the IP Address of LAN port
3. Press '600' Get into the Voicemail Box
4. Press '900' Get into the Meeting
5. Press '#' Blind Transfer
6. Press '*2' Attended Transfer
7. Press '*' Disconnect Call

Chapter4 Login in Home Page

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:

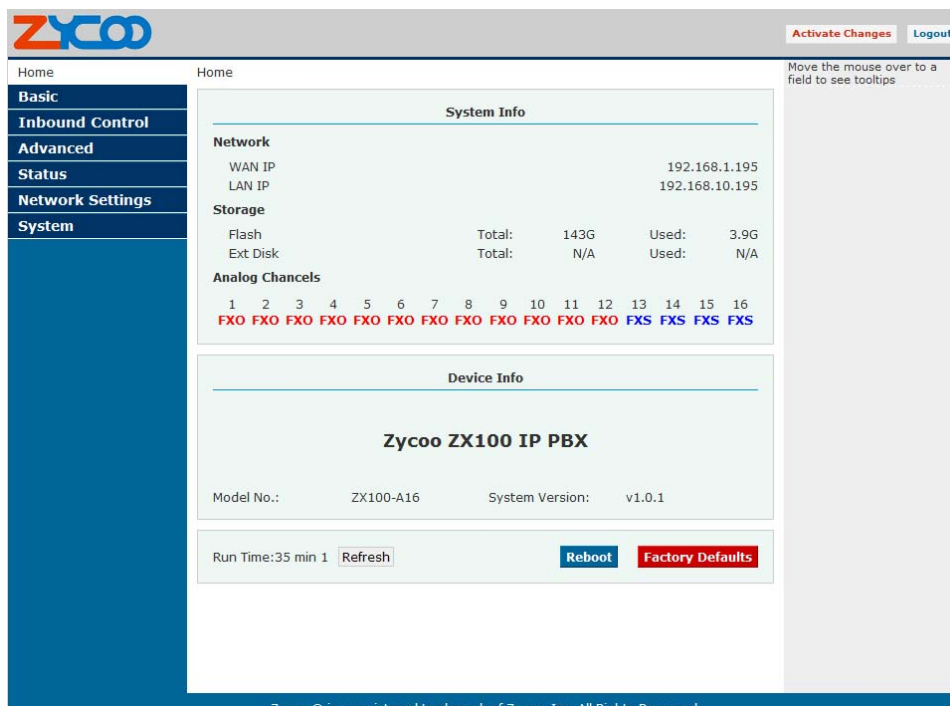



Please login...



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.XXX.



System Info

Network															
WAN IP			192.168.1.195												
LAN IP			192.168.10.195												
Storage															
Flash	Total:	143G	Used: 3.9G												
Ext Disk	Total:	N/A	Used: N/A												
Analog Chancels															
1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
FXO	FXO	FXO	FXO	FXO	FXO	FXO	FXO	FXO	FXO	FXO	FXO	FXS	FXS	FXS	FXS

Device Info

Zycoo ZX100 IP PBX

Model No.: ZX100-A16 System Version: v1.0.1

Run Time:35 min 1 Refresh Reboot Factory Defaults

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

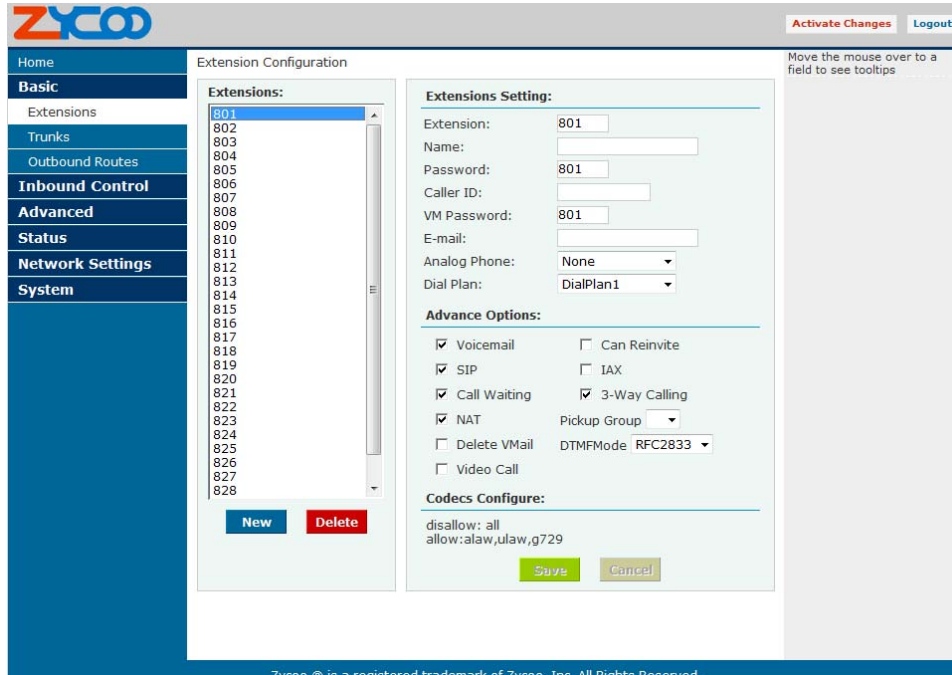
The home page is used for logout, Reboot and Factory Defaults.

- [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.

Chapter5 Basic Configuration

5.1 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 Settings 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
- [Delete VMail](#) If this option is set, then voicemails will not be checkable using a Phone. Messages will be sent via e-mail, only. Note: You need to have an smtp server configured for this functionality.
- [Video Call](#) Enable/Disable Video support function for this extension.
- [Codecs](#) Click here, you can set the extension's codec (default support: alaw, ulaw and G.729).

5.2 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, Peer.**

[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:

- [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 this.

Peer

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

Add Trunk X

Provider Type:

Analog Trunk

E1 Trunk

VoIP Trunk

Peer

Peer Name:

Protocol:

DialPlan:

Host:

NAT:

Without Authentication

Username:

Password:

- **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
- **NAT** Disable/Enable the NAT function
- **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.

Advanced Settings

trunkname:

insecure:

port:

caller ID:

fromdomain:

fromuser:

contact:

qualify:

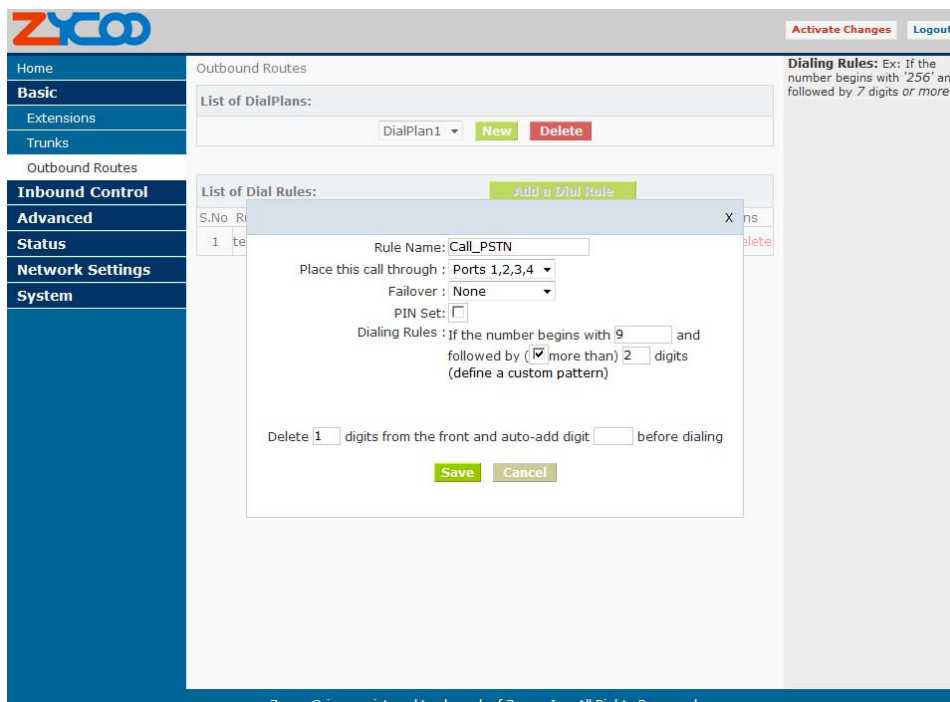
- **Trunkname** Specify a trunk name if you want to refer to the service provider definition as something other than specified in Comment

- **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

5.3 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)
Z Any digit from 1 to 9
N Any digit from 2 to 9
X Any digit from 0 to 9
 . Any number of additional digits

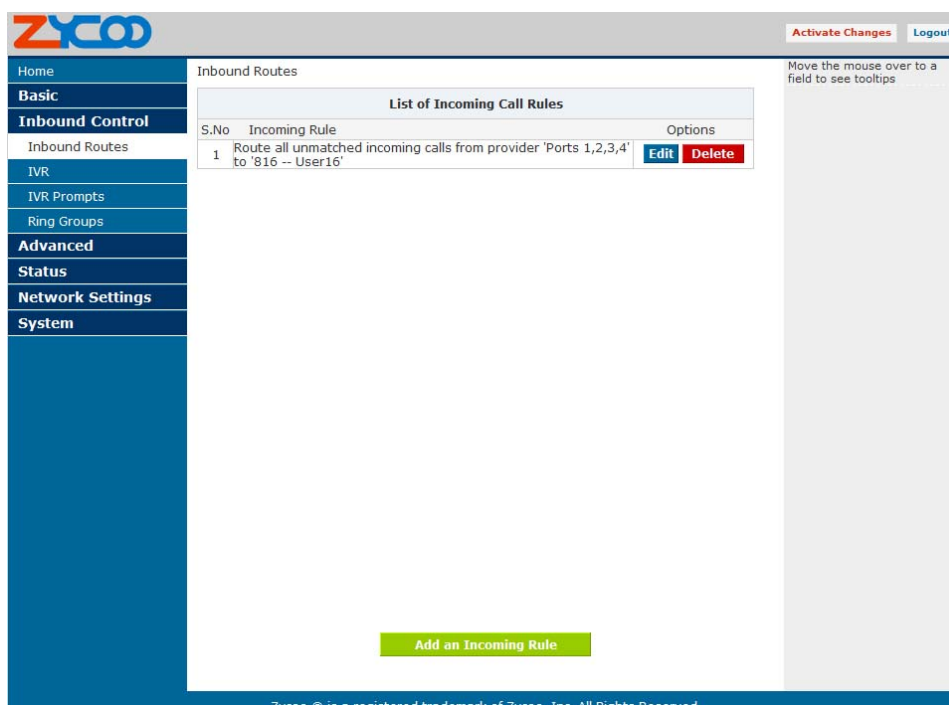
- N** Any digit from 2 to 9
- Z** Any digit from 1 to 9
- X** Any digit from 0 to 9
- . Any number of additional digits

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.

Chapter6 Inbound Control

6.1 Inbound Routers



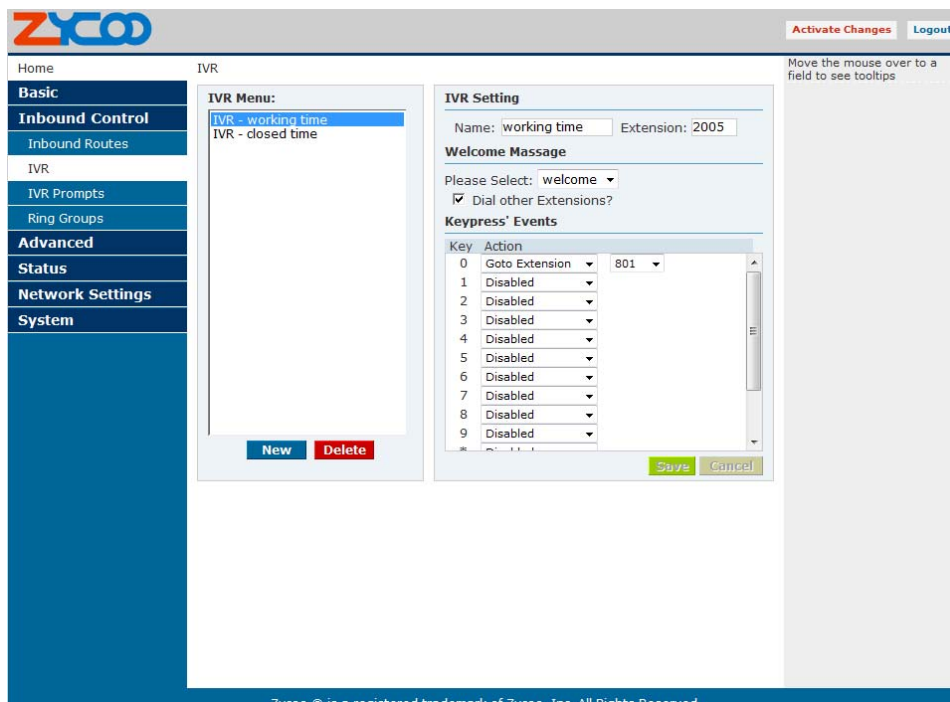
The same pattern-matching logic used for processing outbound calls can also be employed for inbound calls. The two defaults define routing based on whether an incoming call matches or doesn't match a pattern you define.

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.

6.2 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.

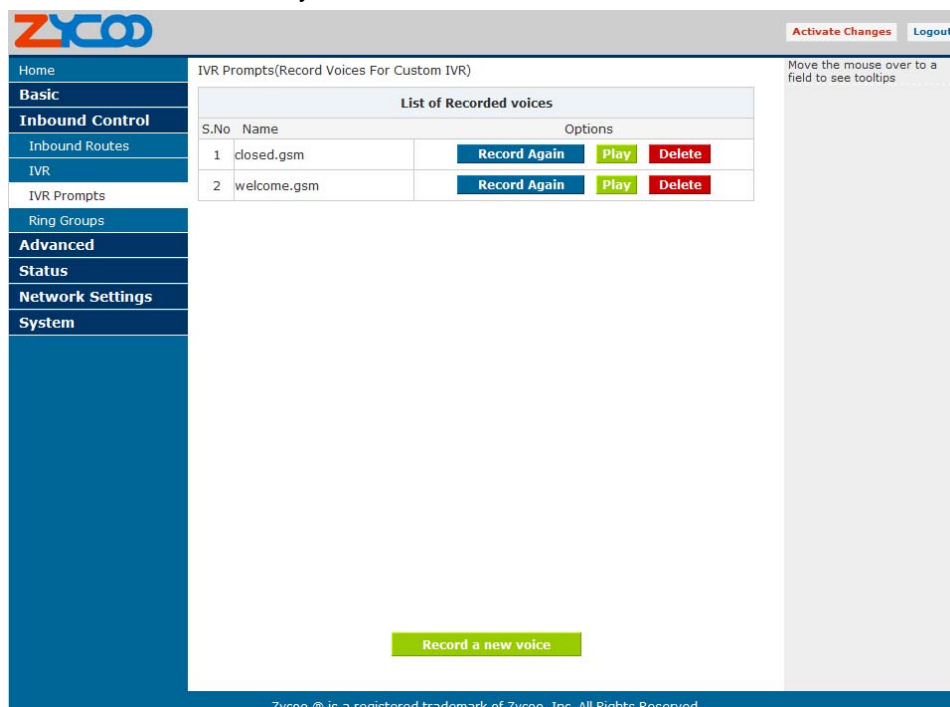


Voice menus are constructed depending on your needs. Just like your business you need to create the solution best suited to your customers.

- **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.

6.3 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”

The dialog box titled "Record a new Voice" contains the following fields and buttons:

- File Name:** A text input field.
- Extension used for recording:** A dropdown menu.
- Record:** A green button.
- Cancel:** A grey button.

- **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

6.4 Ring Groups

A ring group is a group of users assigned to answer incoming call to a single extension. When a caller dials a ring group extension, all of the phones of the users in the ring group will ring together, the call is answered when any one of the users in the group pick up the call. You can configure Ring Groups through the web page

The screenshot shows the Zycoo web interface for configuring Ring Groups. On the left is a navigation menu with categories: Home, Basic, Inbound Control, Inbound Routes, IVR, IVR Prompts, Ring Groups, Advanced, Status, Network Settings, and System. The main content area is titled "Ring Groups" and contains a "List of Ring Groups" table:

S.No	Ring Group	Options
1	tech	Edit Delete

Below the table is a green "New Ring Group" button. On the right side of the interface, there are links for "Activate Changes" and "Logout", and a section titled "Goto an Extension Voicemail" with instructions: "Select goto an extension voicemail if the ringgroup no answer."

Define Ring Groups to Dial more than one extension

X

Add Ring Group

Name:

Strategy: Ring all

←→»»

Ring Group Members

Extension for this ring group(Optional) :

Ring (each/all) for these many seconds :

↑↓

- SIP/801 -- User1
- SIP/605 -- User2
- SIP/803 -- User3
- SIP/804 -- User4
- SIP/805 -- User5
- SIP/806 -- User6
- SIP/807 -- User7
- SIP/808 -- User8

Available Channels

If not answered

- Goto an Extension
- Goto an Extension Voicemail
- Goto a RingGroup
- Goto an IVR menu
- HangUp

Save Cancel

- **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](#).

Chapter7 Advanced Configuration

7.1 Options

- [Local Extensions are](#) Set up the digit of local extensions
- [Operator Extension](#) 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.
- [Music On Ringback](#) Enable/Disable the Music On Ringback function
- [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.

7.2 Voice Mail

The ZX100 provides Voice mail for its end users as an optional feature. End users can retrieve their voice mails and change their password. 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.

7.2.1 General Settings

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.

7.2.2 SMTP settings

The screenshot shows the Zycoo Voicemail Configuration web interface. The left sidebar contains a navigation menu with categories: Home, Basic, Inbound Control, Advanced, Options, Voicemail, Conferencing, Music on hold, Music on Ringback, Call Parking, DISA, Follow Me, Paging and Intercom, Monitor, Time Based Rules, Status, Network Settings, and System. The main content area is titled 'Voicemail Configuration' and has three tabs: 'General', 'SMTP Settings' (which is selected), and 'Email Settings'. Under the 'SMTP Settings' tab, there are several input fields: 'Smtp server' (empty), 'Port' (set to 25), 'SSL/TSL' (checkbox, unchecked), 'Enable ssmtp Authentication' (checkbox, checked), 'Username' (empty), and 'Password' (empty). There are 'Save' and 'Cancel' buttons at the bottom of the form. At the top right of the interface, there are 'Activate Changes' and 'Logout' buttons. A tooltip on the right side says 'Move the mouse over to a field to see tooltips'. At the bottom of the page, there is a small copyright notice: 'Zycoo® is a registered trademark of Zycoo, Inc. All Rights Reserved.'

- **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 on which the SMTP server is running; generally port 25.
- **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.

7.2.3 Email settings

Voicemail Configuration

General SMTP Settings **Email Settings**

Template for Voicemail Emails

Attach recordings to e-mail

Sender Name: IPPBX Server

From: gang.chen@zycoo.com

Subject: you've a voicemail from \${VM_CALLERID}

Message: Dear \${VM_NAME}, you have a new voicemail from \${VM_CALLERID}, the message time is \${VM_DUR}. Receive voicemail time is \${VM_DATE}.

Save **Cancel**

Template Variables:

- \t : 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.

7.3 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.

Conference Room Configuration

Conference Number

Room Extension: 900

Conference Password

PIN Code: 1234

Admin PIN Code: 2345

Conference Options

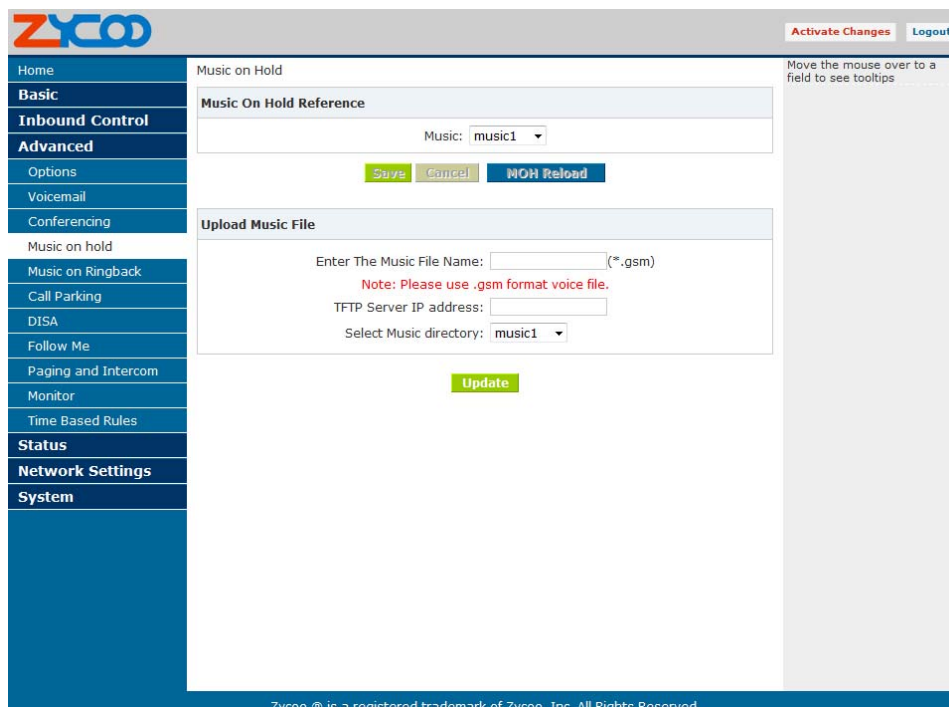
- Play hold music for first caller
- Enable caller menu
- Announce callers
- Record conference
- Quiet Mode

Save **Cancel**

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.

7.4 Music On Hold



- [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.

7.5 Music On Ringback

The screenshot shows the Zycoo web interface for configuring Music On Ringback. The left sidebar contains a navigation menu with categories: Home, Basic, Inbound Control, Advanced, Options, Voicemail, Conferencing, Music on hold, Music on Ringback, Call Parking, DISA, Follow Me, Paging and Intercom, Monitor, Time Based Rules, Status, Network Settings, and System. The main content area is titled "Music on Ringback" and contains a "Music On Ringback Reference" section. This section has a dropdown menu labeled "Music:" with "music2" selected. Below the dropdown are three buttons: "Save" (green), "Cancel" (grey), and "WOR Reload" (blue). At the top right of the interface are "Activate Changes" and "Logout" buttons. A footer note at the bottom of the page reads: "Zycoo ® is a registered trademark of Zycoo, Inc. All Rights Reserved."

- **Music** Select a music for Music On Ringback

Notice: You must enable Music On Ringback function.(In Options Page)

7.6 Call Parking

The screenshot shows the Zycoo web interface for configuring Call Parking Preferences. The left sidebar is identical to the previous screenshot. The main content area is titled "Call Parking Preferences" and contains a "Call Parking Reference" section. This section includes several input fields: "Extension to Dial for Parking Calls:" (700), "What extensions to park calls on:" (701-720) with a note "(Ex: '701-720')", "Number of seconds a call can be parked for:" (45), "Pickup Extension:" (*8), "Pickup Specified Extension:" (*7), "Blind Transfer:" (#), "Attended Transfer:" (*2), "Disconnect Call:" (*), and "Timeout for answer on attended transfer:" (15). Below these fields are "Save" (green) and "Cancel" (grey) buttons. At the top right of the interface are "Activate Changes" and "Logout" buttons. A footer note at the bottom of the page reads: "Zycoo ® is a registered trademark of Zycoo, Inc. All Rights Reserved."

- **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
- **Blind Transfer** allows unattended or blind transfers. It works like this:

While on a conversation with another party, you dial the blindxfer sequence. the system says "Transfer" then gives you a dial tone, while putting the other party on hold. You dial the transferee number and the caller is put through to that number immediately. Your line drops. The caller ID displayed to the person receiving the transferred call is exactly the same as the caller ID presented to you.

- **Attended Transfer** allows attended transfer or supervised transfer. It works like this: While on conversation with another party, you dial the atxfer key sequence. the system says "Transfer" then gives you a dial tone, while putting the other party on hold. You dial the transferee number and talk with the transferee to introduce the call, then you can hang up and the other party will be connected with the transferee. In case the transferee does not want to answer the call, he/she simply hangs up and you will be back to your original conversation. Press the disconnect key sequence, set to * by default, to return yourself to the original caller.
- **Disconnect Call** Disconnect the current transfer call(for Attended transfer).
- **Timeout for answer on attended transfer:** Set the answer timeout value.

7.7 DISA Settings

The screenshot shows the 'DISA Settings' page in the ZYCOO web interface. On the left is a navigation menu with categories like Home, Basic, Inbound Control, Advanced, Status, Network Settings, and System. The main area displays a table titled 'List of Disa' with the following data:

S.No	DISA Name	Options
1	Test	Edit Delete

Below the table is a green 'New DISA' button. To the right of the table, there is a note: 'Extension for this Disa (Option): If you want this DISA to be accessible by dialing an extension, you can define an extension number for this DISA.' At the top right of the page are 'Activate Changes' and 'Logout' buttons. The footer contains the text: 'Zycop © is a registered trademark of Zycop, Inc. All Rights Reserved.'

- **List of DISA** DISA name are listed in the table.
- **New DISA** Create a new DISA.

Add a Disa X

DISA Name:

PIN: Without PIN

Response Timeout(s):

Digit Timeout(s):

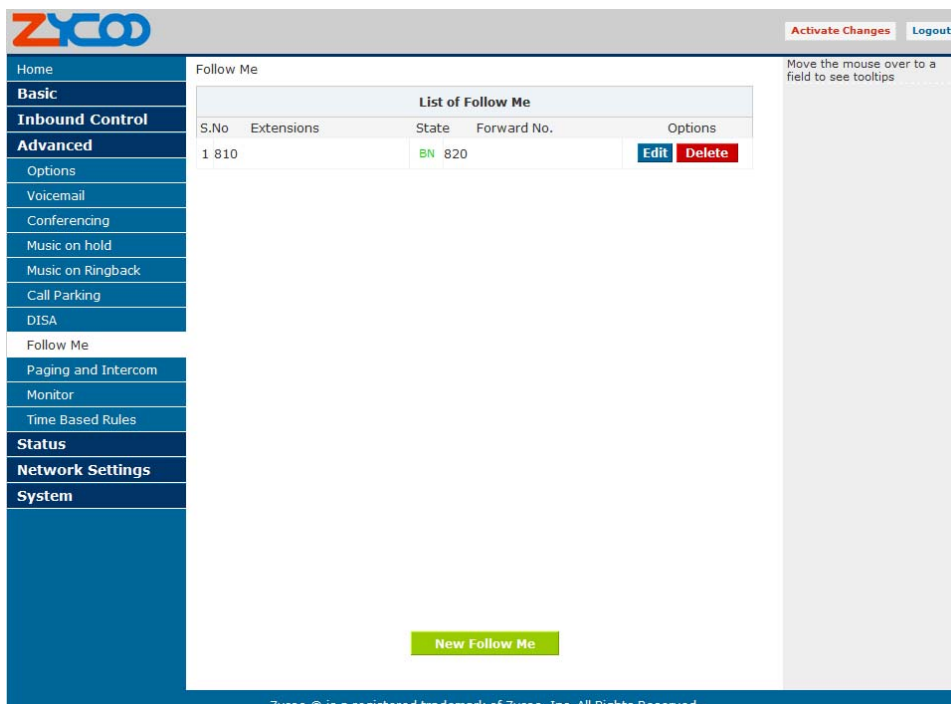
Extension for this Disa(Optional):

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

7.8 Follow Me



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

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 an Outside Number" the follow page will be displayed.

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 DialPlan

Set forward outside number

Save
Cancel

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

7.9 Paging and Intercom

Activate Changes Logout

- Home
- Basic**
- Inbound Control
- Advanced**
- Options
- Voicemail
- Conferencing
- Music on hold
- Music on Ringback
- Call Parking
- DISA
- Follow Me
- Paging and Intercom
- Monitor
- Time Based Rules
- Status**
- Network Settings
- System

List of Paging Groups

S.No	Paging Group	Options
1	5000 test	Edit Delete

Add 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".

- [List of Paging Groups](#) Paging Groups are listed in the table.
- [Add Paging Group](#) Create a new Paging Group

Add Paging Group X

Paging Extension:

Group Description:

← SIP/801 -- User1
 SIP/605 -- User2
 SIP/803 -- User3
 SIP/804 -- User4
 SIP/805 -- User5
 SIP/806 -- User6
 SIP/807 -- User7
 SIP/808 -- User8
 ↓

Paging Group Members **Device List**

Duplex:

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

- [Paging Extension](#) Set a extension for the Paging Group.
- [Group Description](#) Provide a descriptive title for this Page Group.
- [Paging Group Members](#) Selected Device(s) in this Page.
- [Device List](#) Select Device(s) to Page.
- [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".

7.10 Monitor

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

Home Monitoring

Basic

Inbound Control

Advanced

Options

Voicemail

Conferencing

Music on hold

Music on Ringback

Call Parking

DISA

Follow Me

Paging and Intercom

Monitor

Time Based Rules

Status

Network Settings

System

List of Monitoring Extension

S.No	Extension	Record Time	Inbound	Outbound	Options
1	821	Always	Disable	Enable	Edit Delete
2	810	09:00-17:30 mon-fri	Enable	Enable	Edit Delete

[New Monitor](#)

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- [List of Monitoring Extension](#) Monitoring extensions are listed in the table.
- [Add Monitor](#) Create a new Monitor

Add Monitor X

Extension:

Monitor Time

Always Monitor:

Start Time: : End Time: :

Start Day: End Day:

Monitor Settings

Inbound Record: Outbound Record:

- Extension Select a Monitoring extension
- Monitoring Time Set always Monitor or select a Monitoring time
- Monitoring Settings Set inbound record and outbound record

7.11 Time Based Rules

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

Chapter8 Status Display

8.1 Monitor List

ZYCOO Activate Changes Logout

Home Monitor Move the mouse over to a field to see tooltips

Basic Extension: 804 Delete

Inbound Control Date: Aug 24 2010 Go

Advanced **List of Monitoring File**

S.No	Caller ID	Destination	Date	Options
1	810	804	2010/08/24 15:00:03	Delete

Call Logs

Register Status

System Info

Network Settings

System

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This web page will display Monitor info for each extension

8.2 Call Logs

ZYCOO Activate Changes Logout

Home Call Logs

Basic Start Date: Aug 26 2010 Field: Caller ID Filter Download Delete

Inbound Control End Date: Aug 26 2010

Advanced

Call Start	Caller ID	Destination	Duration (sec)	Disposition
------------	-----------	-------------	----------------	-------------

Monitor List

Call Logs

Register Status

System Info

Network Settings

System

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This web page will display call logs

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

8.3 Register Status

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

Register Status

[Home](#) | [Basic](#) | [Inbound Control](#) | [Advanced](#) | [Status](#) | [Monitor List](#) | [Call Logs](#) | [Register Status](#) | [System Info](#) | [Network Settings](#) | [System](#)

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

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	
831	(Unspecified)	D	N		0	UNKNOWN	
830	(Unspecified)	D			0	UNKNOWN	
829	(Unspecified)	D			0	UNKNOWN	
828	(Unspecified)	D			0	UNKNOWN	
827	(Unspecified)	D			0	UNKNOWN	
826	(Unspecified)	D			0	UNKNOWN	
825	(Unspecified)	D			0	UNKNOWN	
824	(Unspecified)	D			0	UNKNOWN	
823	(Unspecified)	D			0	UNKNOWN	
822	(Unspecified)	D			0	UNKNOWN	
821	(Unspecified)	D			0	UNKNOWN	
820	(Unspecified)	D			0	UNKNOWN	
819	(Unspecified)	D			0	UNKNOWN	
818	(Unspecified)	D			0	UNKNOWN	
817	(Unspecified)	D			0	UNKNOWN	
816	(Unspecified)	D			0	UNKNOWN	
815	(Unspecified)	D			0	UNKNOWN	
814	(Unspecified)	D			0	UNKNOWN	
813/813	(Unspecified)	D	N		0	UNKNOWN	
812/812	(Unspecified)	D	N		0	UNKNOWN	
811	(Unspecified)	D			0	UNKNOWN	
810/810	(Unspecified)	D			0	UNKNOWN	
809/809	(Unspecified)	D	N		0	UNKNOWN	
808	(Unspecified)	D			0	UNKNOWN	
807	(Unspecified)	D			0	UNKNOWN	
806	(Unspecified)	D			0	UNKNOWN	
805	(Unspecified)	D			0	UNKNOWN	
804	(Unspecified)	D			0	UNKNOWN	
801	(Unspecified)	D			0	UNKNOWN	

29 sip peers [Monitored: 0 online, 29 offline Unmonitored: 0 online, 0 offline]

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8.4 System Info

In this page it will display nonce system info

System Information

[Home](#) | [Basic](#) | [Inbound Control](#) | [Advanced](#) | [Status](#) | [Monitor List](#) | [Call Logs](#) | [Register Status](#) | [System Info](#) | [Network Settings](#) | [System](#)

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

[General](#) | [Resources](#)

OS Version:
Linux IP PBX 2.6.22.18

Uptime:
11:39:17 up 1 day, 1:45, 1 user,
Load Average: 0.00, 0.03, 0.00

Firmware Version:
Zycoo System v1.0.1

Server Date & TimeZone:
Thu Aug 26 11:39:17 SGT 2010 [Refresh](#)

[Synchronize](#)

Hostname:
voip

Move the mouse over to a field to see tooltips

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Chapter9 Network Settings

9.1 Network and Country

On this page you can set WAN, LAN interface information and the country of Tone Zone.

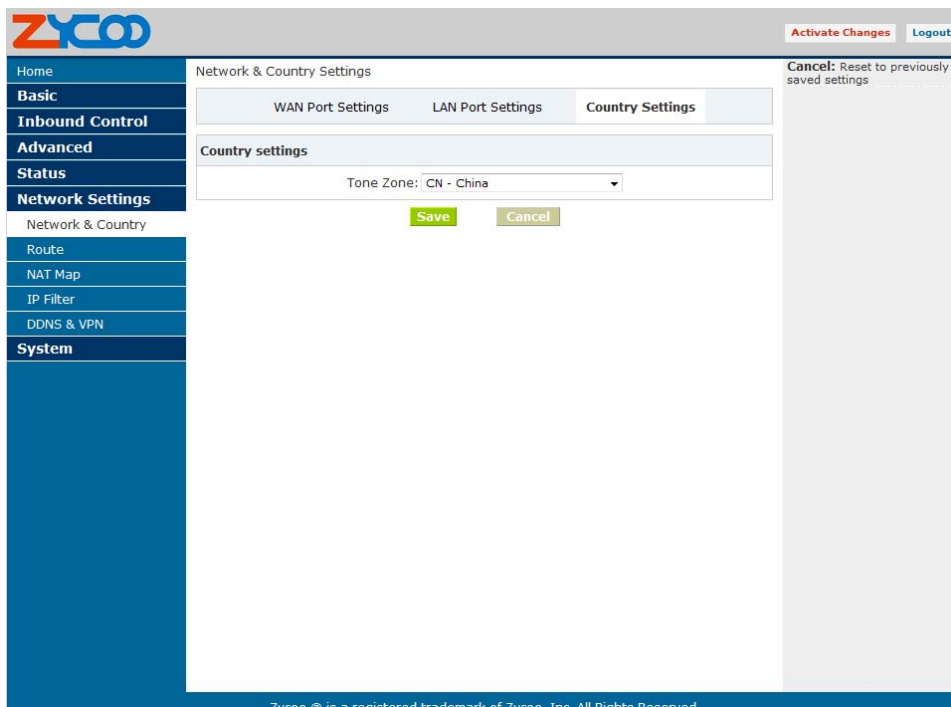
9.1.1 WAN Port Settings

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

9.1.2 LAN Port Settings

Set the LAN Port IP Address and Subnet mask.

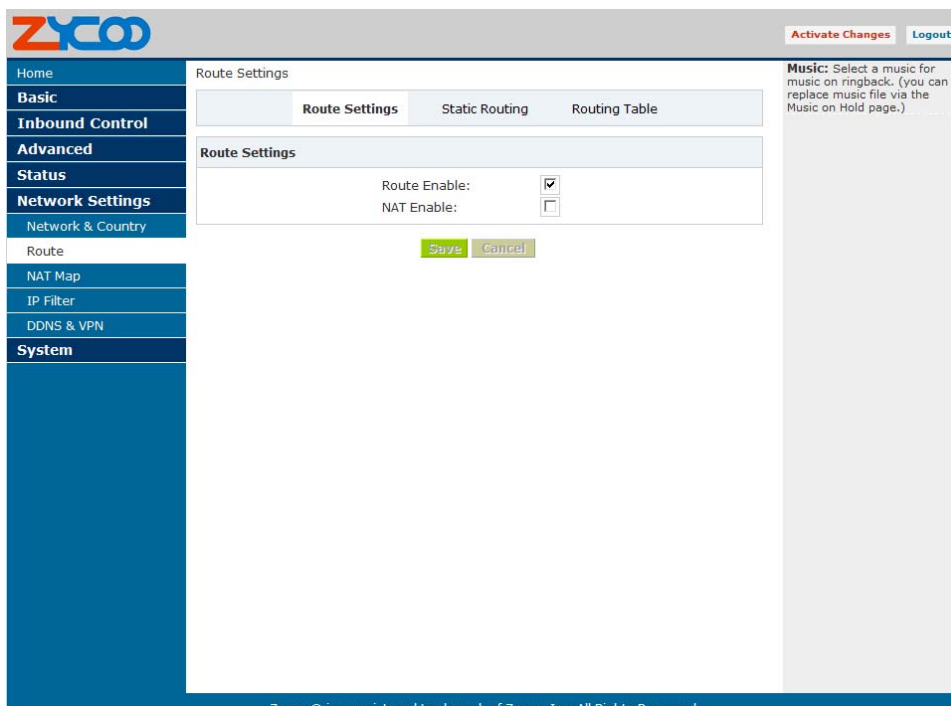
9.1.2 Country Settings



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

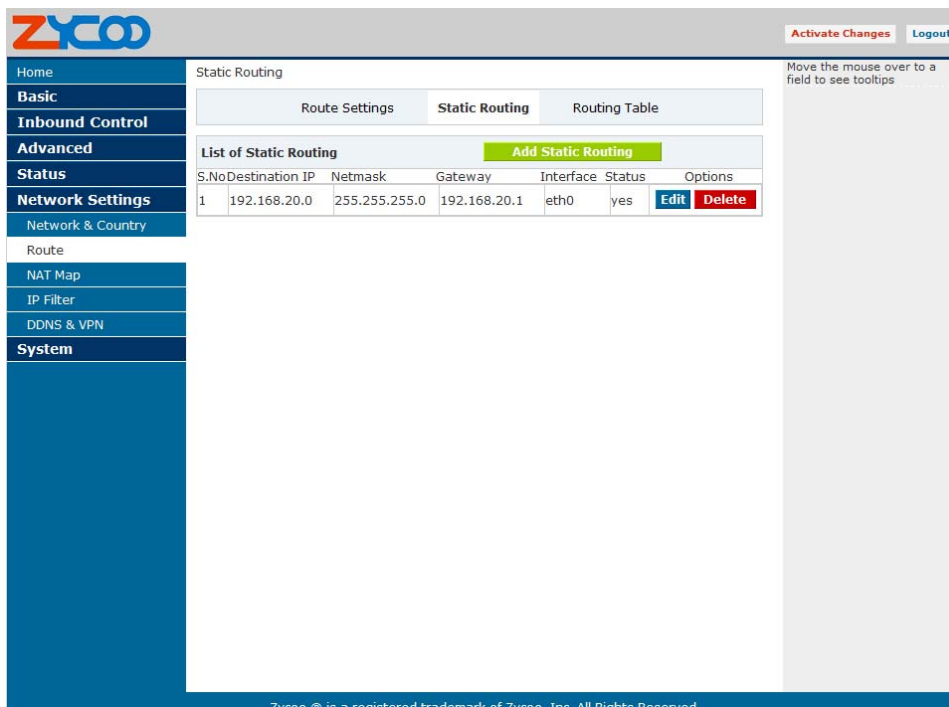
9.2 Route

9.2.1 Route Settings



- **Route Enable** Enable/Disable Route function
- **NAT Enable** Enable/Disable NAT Function

9.2.2 Static Routing



- [List of Static Routing](#) Static Routing are listed in the table.
- [Add Static Routing](#) Create a New Static Routing

Static Routing Settings X

Destination IP:	<input style="width: 90%;" type="text" value="192.168.20.0"/>
Netmask:	<input style="width: 90%;" type="text" value="255.255.255.0"/>
Gateway:	<input style="width: 90%;" type="text" value="192.168.20.1"/>
Interface:	<input style="border: none; border-bottom: 1px solid #ccc;" type="text" value="eth0"/> ▼
Availability:	<input style="border: none; border-bottom: 1px solid #ccc;" type="text" value="Yes"/> ▼

- [Destination IP](#) Set Destination IP for the Static Routing
- [Netmask](#) Set Net Mask for the Static Routing
- [Gateway](#) Set Gateway for the Static Routing
- [Interface](#) Set Interface for the Static Routing
- [Availability](#) Set Status for the Static Routing

9.2.3 Routing Table

Routing Table:
Kernel IP routing table

Destination	Gateway	Genmask	Flags	Metric	Ref	Use	Iface
192.168.1.0	0.0.0.0	255.255.255.0	U	0	0	0	eth0
192.168.10.0	0.0.0.0	255.255.255.0	U	0	0	0	eth1
169.254.0.0	0.0.0.0	255.255.0.0	U	1002	0	0	eth0
169.254.0.0	0.0.0.0	255.255.0.0	U	1003	0	0	eth1
0.0.0.0	192.168.1.1	0.0.0.0	UG	0	0	0	eth0

This page will display the current routing table.

9.4 NAT Map

- [List of NAT Map](#) NAT Map are listed in the table.
- [Add NAT Map](#) Create a New NAT Map

NAT Map Settings X

Outside Port: (XX or XX-XX)

Destination IP:

Destination Port:

Protocol: ▼

Availability: ▼

Save
Cancel

- [Outside Port](#) Set Outside Port for the NAT Map
- [Destination IP](#) Set Destination IP for the NAT Map
- [Destination Port](#) Set Destination Port for the NAT Map
- [Protocol](#) Set Protocol for the NAT Map
- [Availability](#) Set Status for the NAT Map

9.5 IP Filter

Activate Changes Logout

- Home
- Basic
- Inbound Control
- Advanced
- Status
- Network Settings
- Network & Country
- Route
- NAT Map
- IP Filter
- DDNS & VPN
- System

IP Filter

Add IP Filter

S.No	Source IP	Port	Destination IP	Port	Direction	Protocol	Interface	Action	Status	Options
1	0.0.0.0/0	all	0.0.0.0/0	22	INPUT	tcp	ppp+	ACCEPT	yes	Edit Delete
2	0.0.0.0/0	all	0.0.0.0/0	22	INPUT	tcp	eth0	ACCEPT	yes	Edit Delete
3	0.0.0.0/0	all	0.0.0.0/0	9999	INPUT	tcp	eth0	ACCEPT	yes	Edit Delete
4	0.0.0.0/0	all	0.0.0.0/0	9999	INPUT	tcp	ppp+	ACCEPT	yes	Edit Delete
5	0.0.0.0/0	all	0.0.0.0/0	5060	INPUT	udp	eth0	ACCEPT	yes	Edit Delete
6	0.0.0.0/0	all	0.0.0.0/0	5060	INPUT	udp	ppp+	ACCEPT	yes	Edit Delete
7	0.0.0.0/0	all	0.0.0.0/0	10001:1020	INPUT	udp	eth0	ACCEPT	yes	Edit Delete
8	0.0.0.0/0	all	0.0.0.0/0	10001:1020	INPUT	udp	ppp+	ACCEPT	yes	Edit Delete
9	0.0.0.0/0	all	0.0.0.0/0	all	INPUT	tcp	eth1	ACCEPT	yes	Edit Delete
10	0.0.0.0/0	all	0.0.0.0/0	all	INPUT	udp	eth1	ACCEPT	yes	Edit Delete
11	0.0.0.0/0	all	0.0.0.0/0	all	INPUT	tcp	ppp+	DROP	yes	Edit Delete

- [List of IP Filter](#) IP Filters are listed in the table.
- [Add IP Filter](#) Crate a New IP Filter

IP Filter Settings X

Source IP: /

Source Port:

Destination IP: /

Destination Port:

Interface: ▼

Protocol: ▼

Direction: ▼

Action: ▼

Availability: ▼

Save
Cancel

- [Source IP](#) Set Source IP for the IP Filter
- [Source Port](#) Set Source Port for the IP Filter
- [Destination IP](#) Set Destination IP for the IP Filter
- [Destination Port](#) Set Destination Port for the IP Filter
- [Interface](#) Select Interface for the IP Filter
- [Protocol](#) Select Protocol for the IP Filter
- [Direction](#) Select Direction for the IP Filter
- [Action](#) Select Action for the IP Filter
- [Availability](#) Select Action for the IP Filter

9.3 DDNS&VPN

9.2.1 DDNS Settings

On this page, you can set DDNS reference.

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

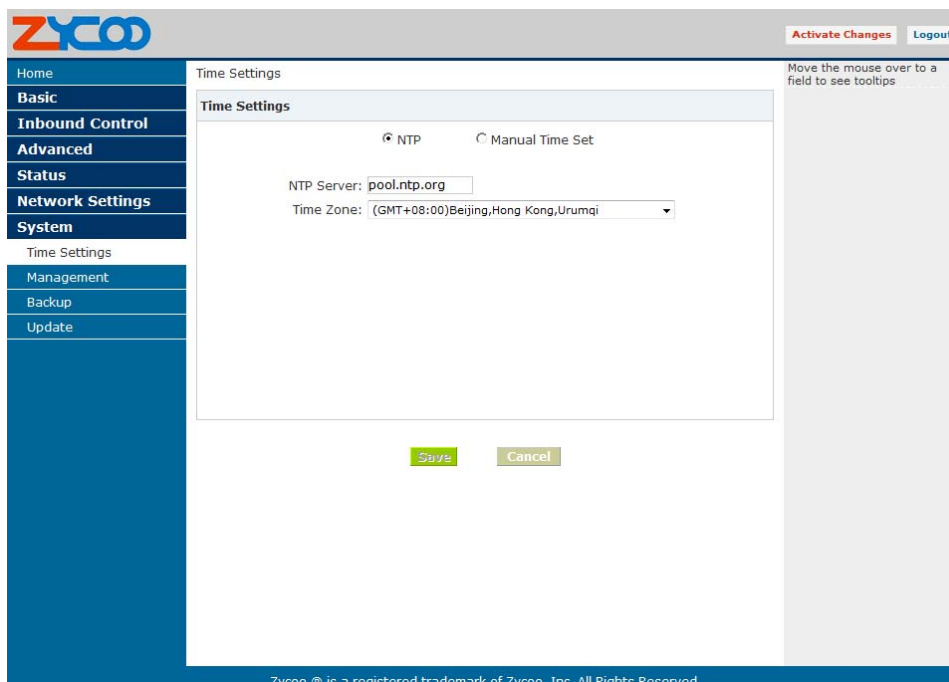
9.2.2 VPN Settings

On this page, you can set VPN reference.

Notice: Now, it only supports N2N VPN. More other VPN, you can customize based on your requirement.

Chapter10 System Management

10.1 Time Settings



10.1.1 NTP Settings

NTP Manual Time Set

NTP Server:

Time Zone:

- **NTP Server** Specify the NTP server that you wish to use. You may type either the domain name or the IP address of the server, and it may be either remote or local. The default server is pool.ntp.org. Be aware that the PBX needs to be able to access a NTP server in order to function properly.
- **Time Zone** Select your time zone so that the system will set time base on the time zone.

10.1.2 Manual Time Settings

NTP Manual Time Set

Year: (YYYY, eg: 2010)

Month: (MM, eg: 05)

Day: (DD, eg: 08)

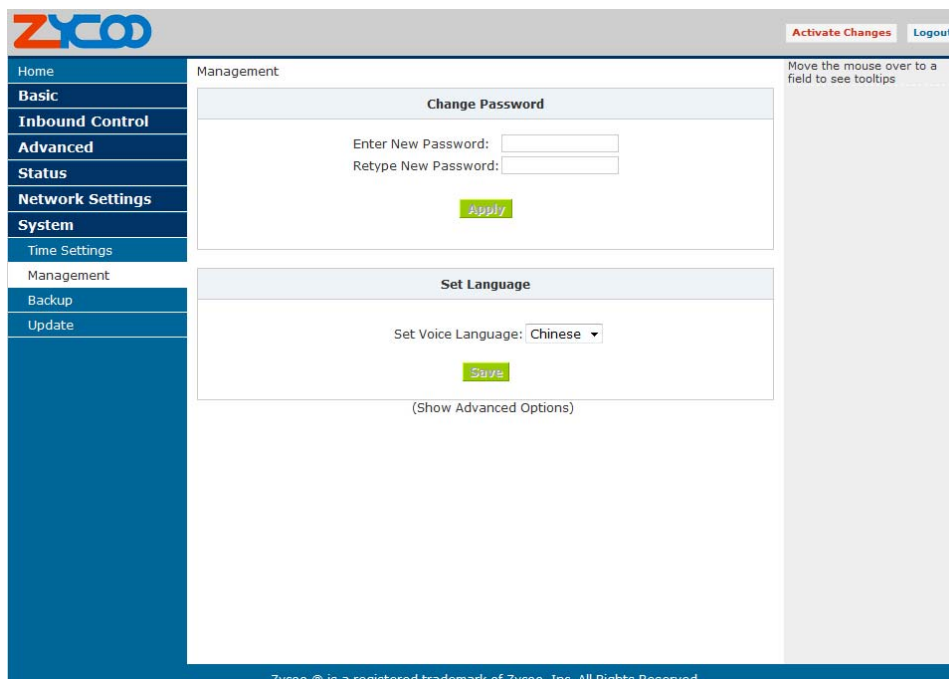
Hour: (HH, eg: 09)

Minute: (MM, eg: 30)

Synchronize current PC time

- **Synchronize current PC time** Click the button ,the current PC time synchronization.

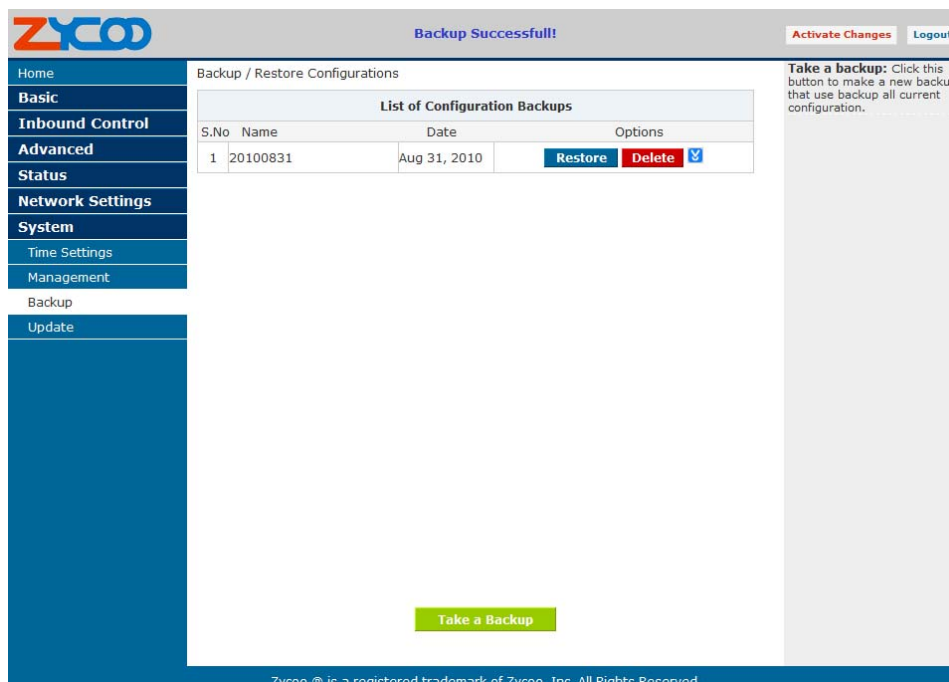
10.2 Management



- **Change Password** On this page, you can change the administrator password (Default password: admin)
- **Set Language** Set the system language voice

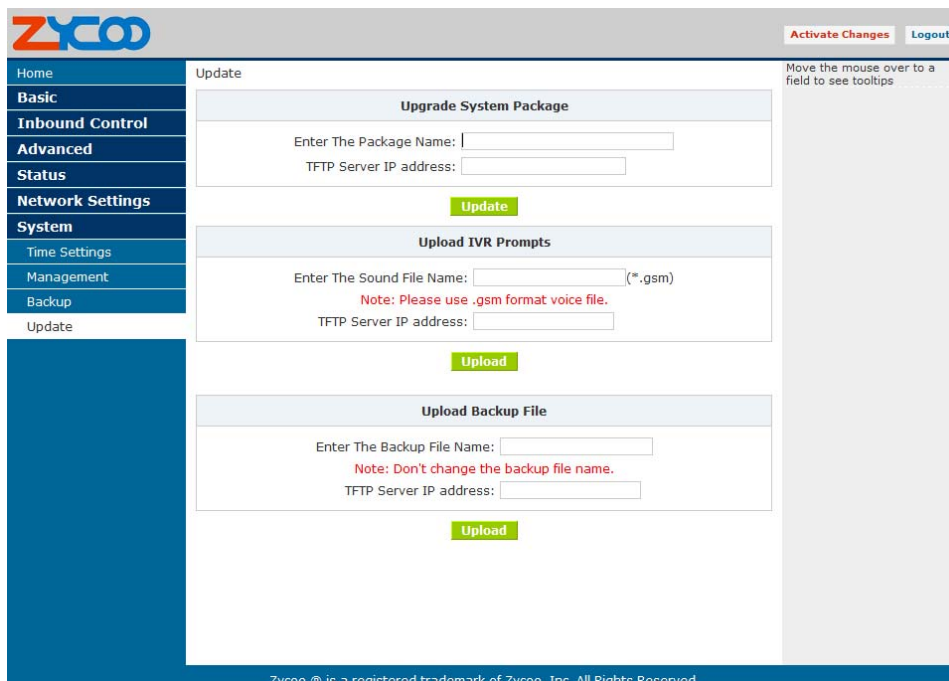
And you can also set the advanced options about SIP and Zap protocol in the "Show Advanced Options" list, that is useful when you set connect two ippbx in different network.

10.3 Backup



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

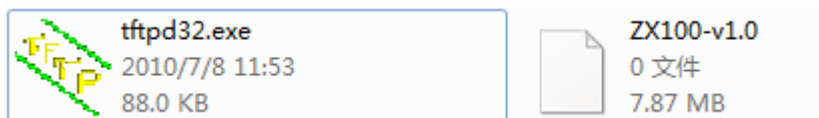
10.4 Upgrade



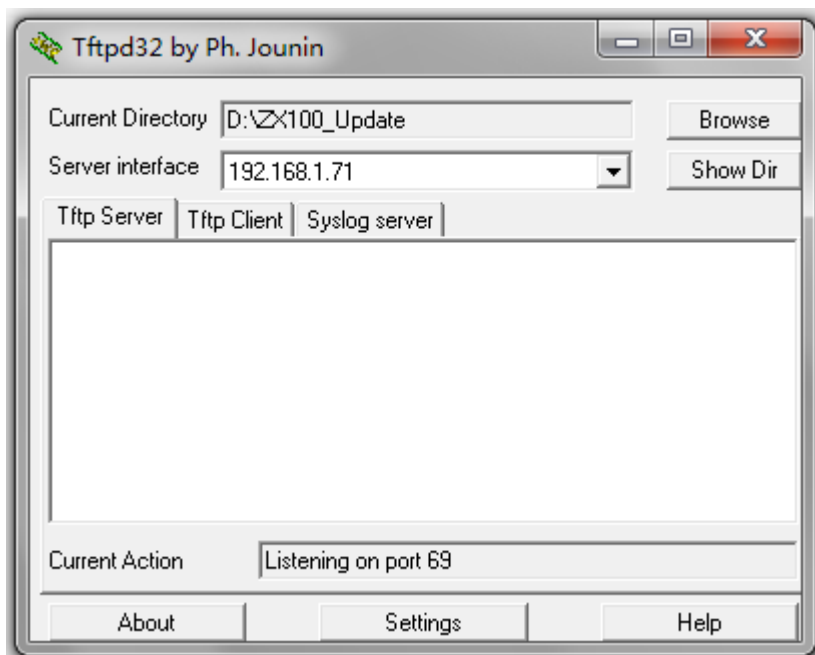
In this page you can upgrade system package

- [Enter The Package Name](#) Set system package name
- [TFTP Server IP address](#) Set TFTP server IP

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 **ZX100-v1.0**

Enter TFTP Server IP address, hereby it's

After done, click [Update](#) to update, then the system will reboot automatically.

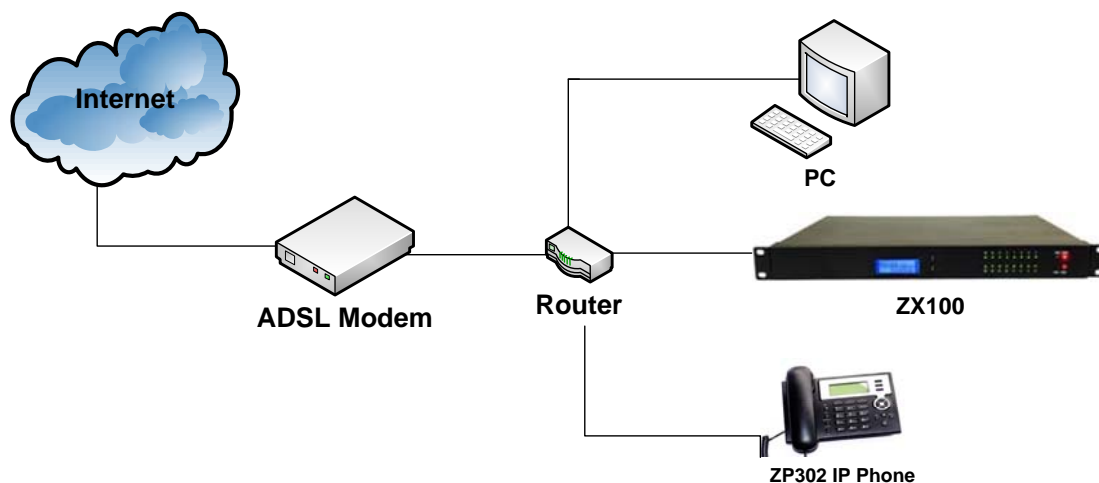
(Note: the upgrading will set your system as default, please make backup before you do it.)

Chapter11 Operating Instruction

11.1 How to link the ZX100 IP PBX to the interwork

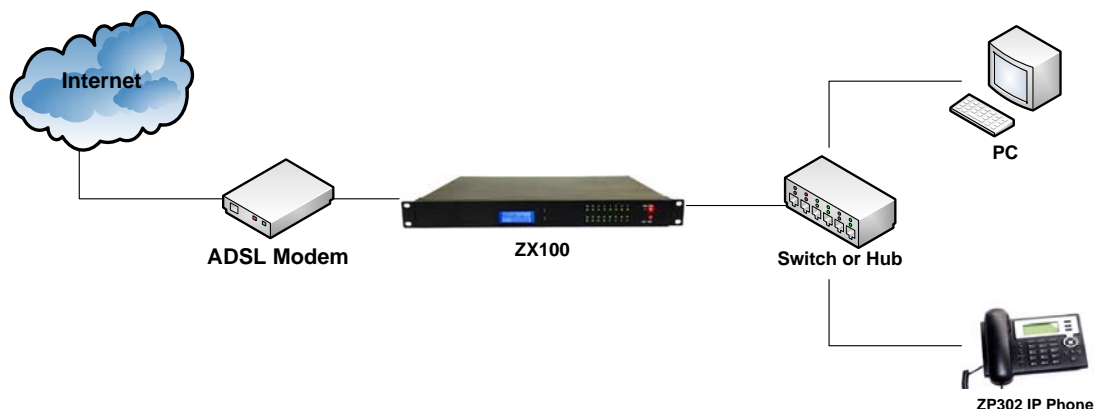
10.1.1 IP PBX behind the Router

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..



10.1.2 IP PBX behind the Modem

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)



11.2 How to log in the IP PBX system

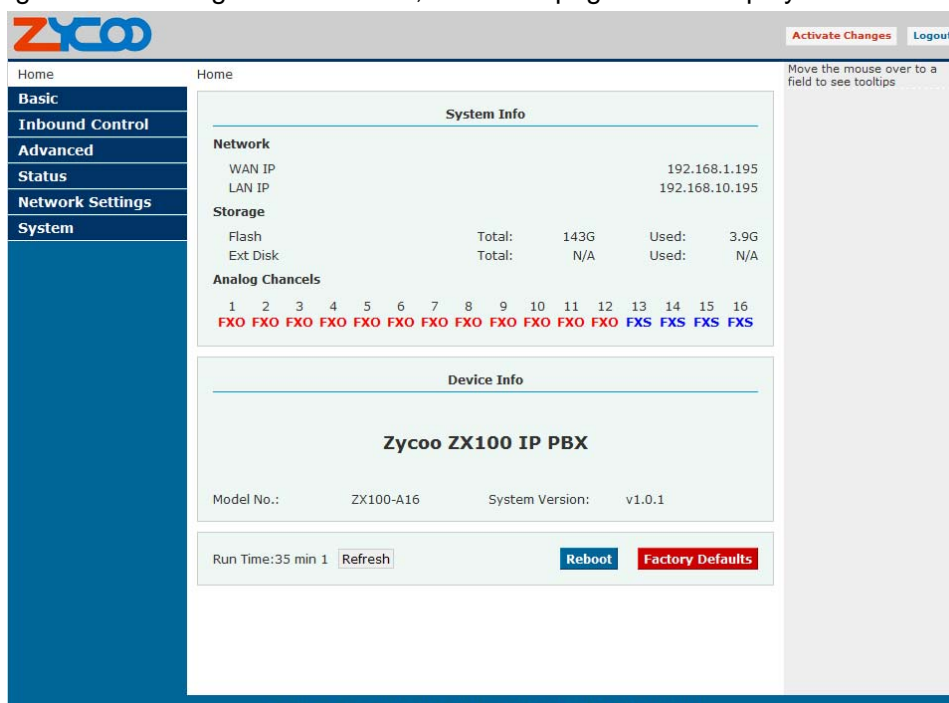
After connecting the ippbx to the local area network. Launch the web browser on a computer that is in this local area network. Enter the IP address for the system (default: Wan port IP address is <http://192.168.1.100:9999>, Lan port IP address is <http://192.168.10.100:9999>) . The start web page will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




Please login...



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:



The screenshot shows the ZYCOO web interface. On the left is a navigation menu with tabs: Home, Basic, Inbound Control, Advanced, Status, Network Settings, and System. The main content area is titled 'System Info' and includes:

- Network:** WAN IP (192.168.1.195), LAN IP (192.168.10.195)
- Storage:** Flash (Total: 143G, Used: 3.9G), Ext Disk (Total: N/A, Used: N/A)
- Analog Chancels:** A table with 16 columns (1-16). Columns 1-12 are labeled 'FXO' and columns 13-16 are labeled 'FXS'.
- Device Info:** Zycoo ZX100 IP PBX, Model No.: ZX100-A16, System Version: v1.0.1
- Run Time:** 35 min 1, with Refresh, Reboot, and Factory Defaults buttons.

At the top right of the page are 'Activate Changes' and 'Logout' buttons. A footer note at the bottom reads: 'Zycoo® is a registered trademark of Zycoo, Inc. All Rights Reserved.'

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.

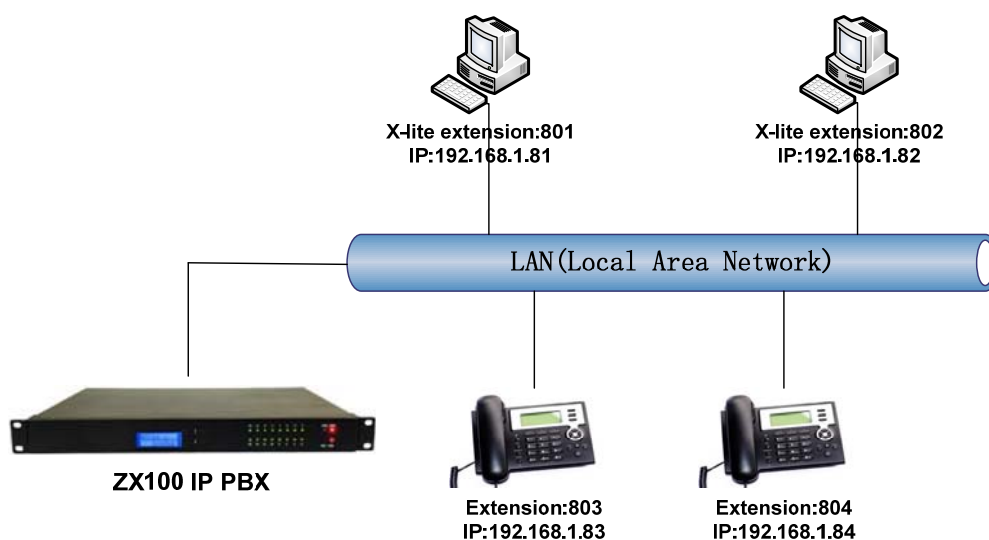
The right column of the user interface contains Tooltips. This area provides brief description for any options of the GUI

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

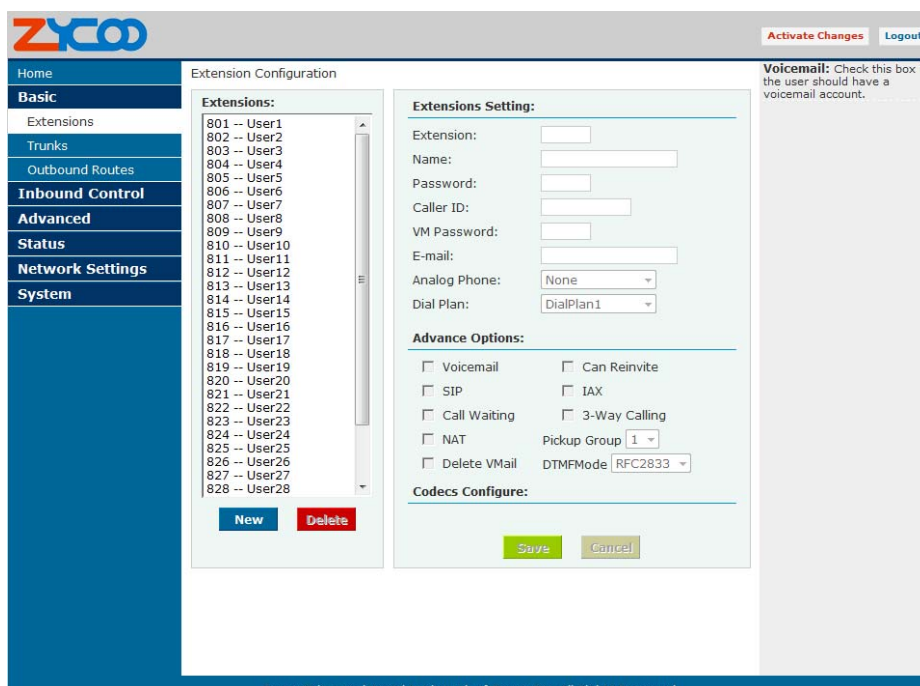
- **Logout:** To log out the zycoo GUI.
- **Reboot:** Reboot the ZX100 system
- **Factory Defaults:** Restore all settings to factory default.
- **Activate change:** Made the change active for the current configuration after you make a configuration change on some page.

11.3 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 ZX100, but setting is the same in other ZX100 products.



Set User



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.

11.4 How to make an outbound call

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

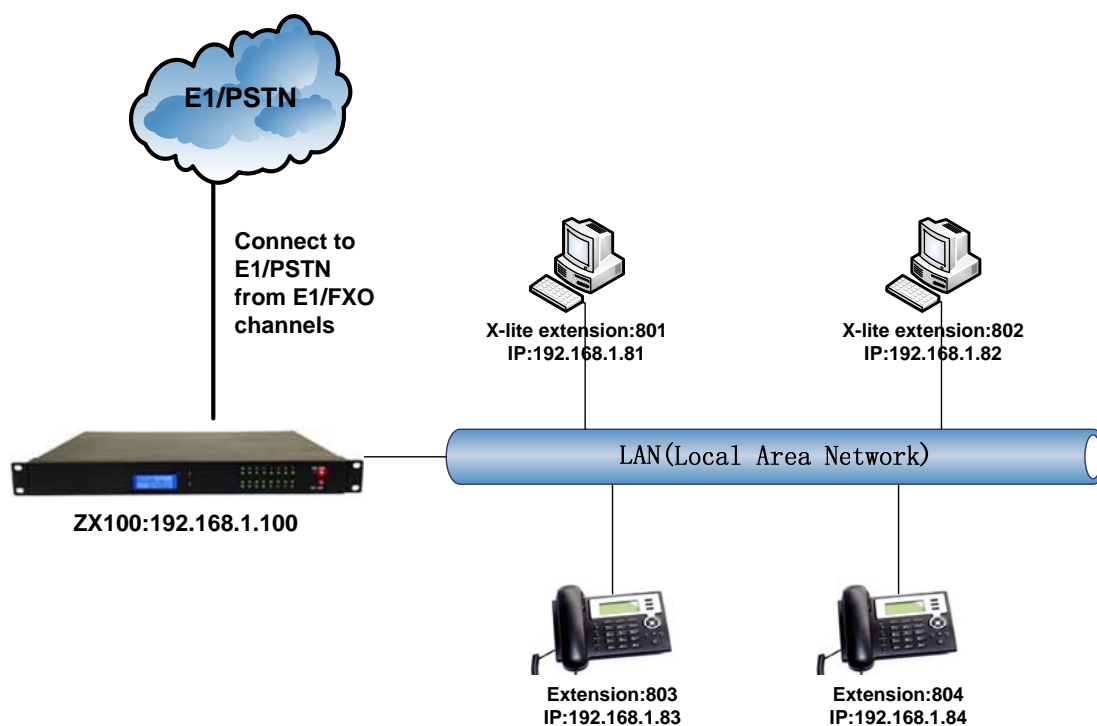
Analog Ports: FXO ports of ZX100, connect to PSTN lines.

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

I am using ZX100-A16, the port1-4 are configured as FXO 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**.

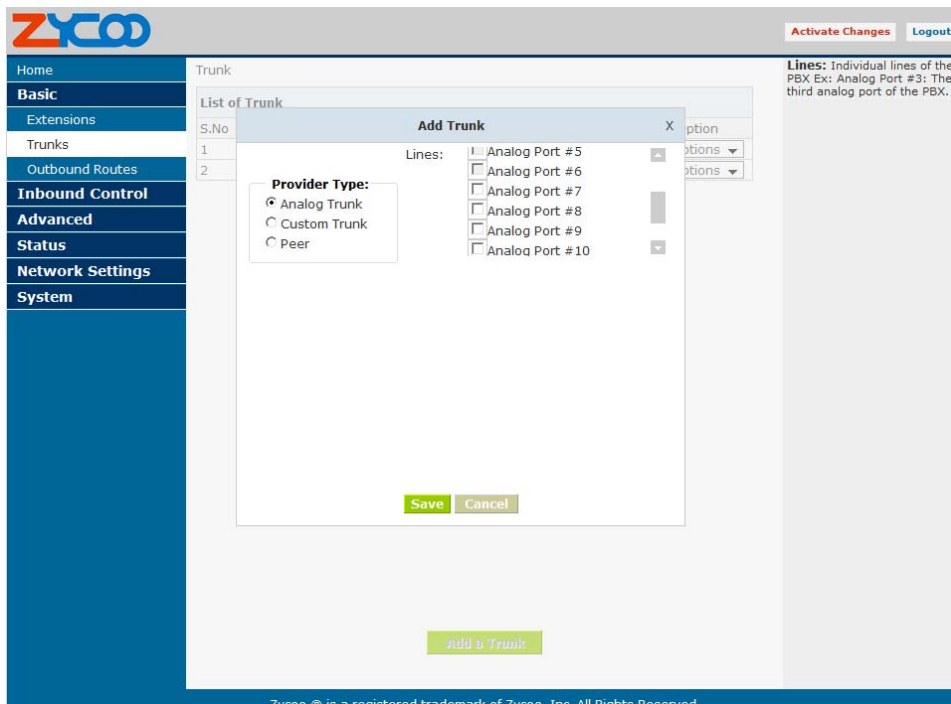
11.4.1 Make call via PSTN trunk

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



Add Analog Trunk

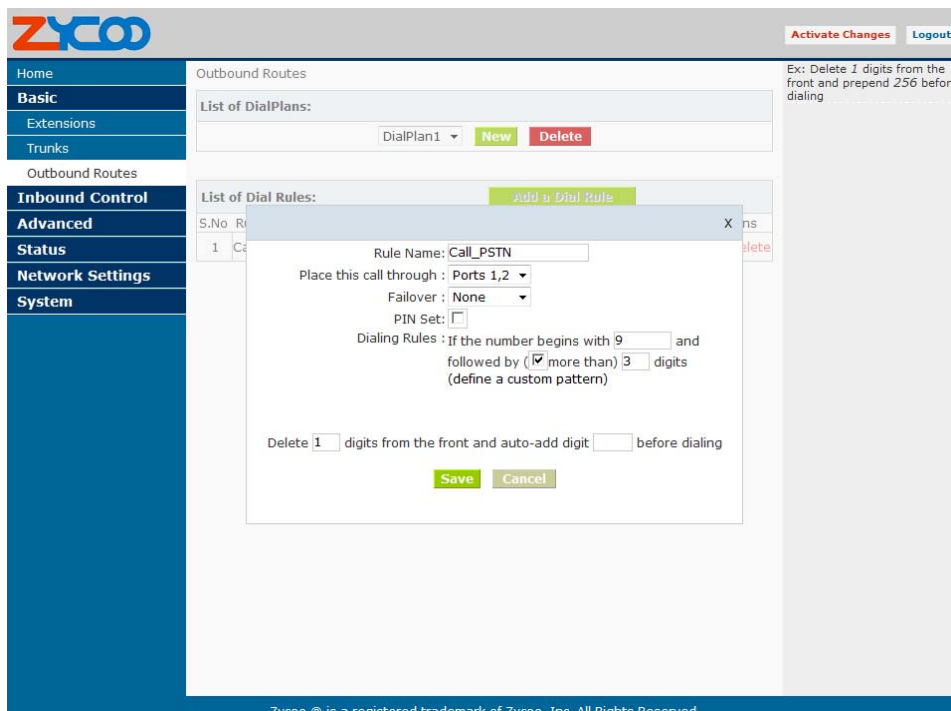
Trunks -> Add a Trunk:



Add Outbound Routers

In Outbound Routers -> add a Dial rule as below

Dial Rules

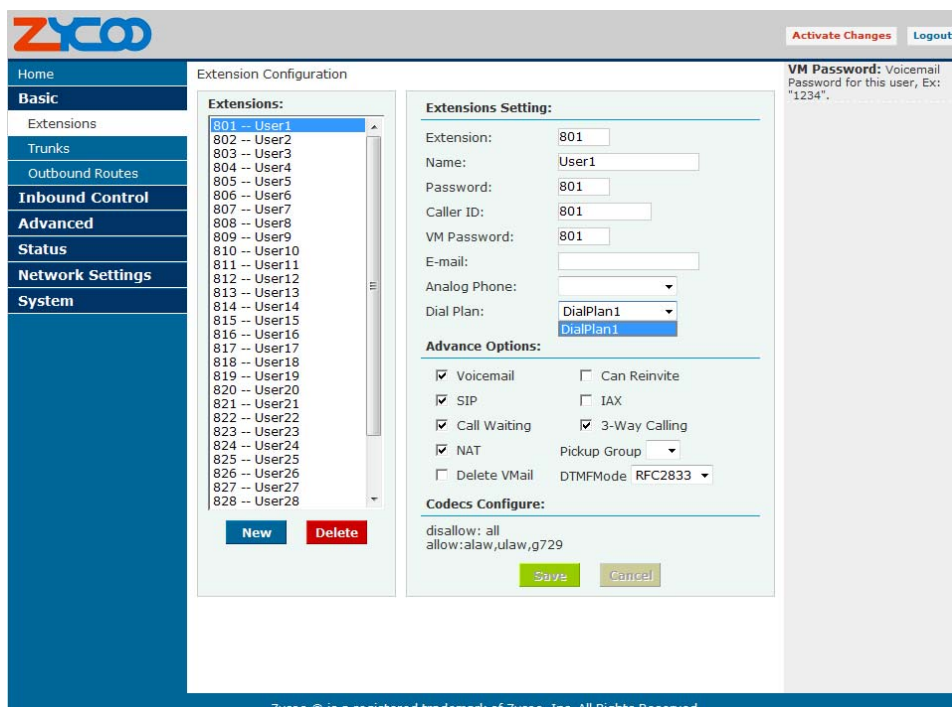


We have now added a Dial rule “OUT_PSTN” in the “DialPlan1”.

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 (port1 or port3).

Choose Dial Plan for extensions:

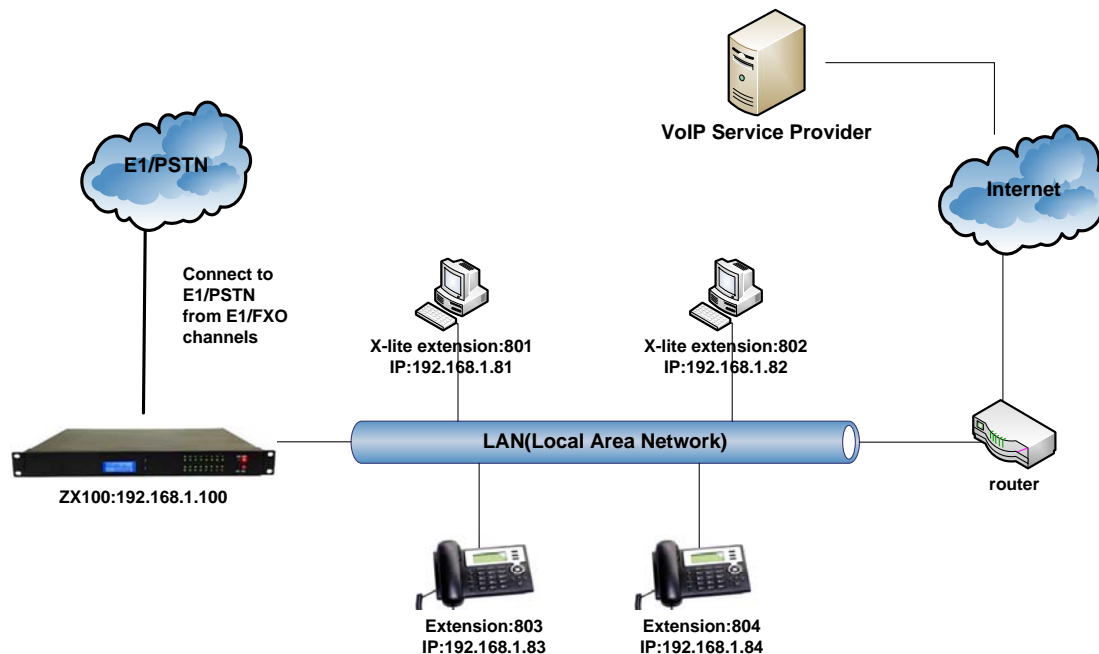
On the User page, edit the extensions to choose DialPlan1.



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

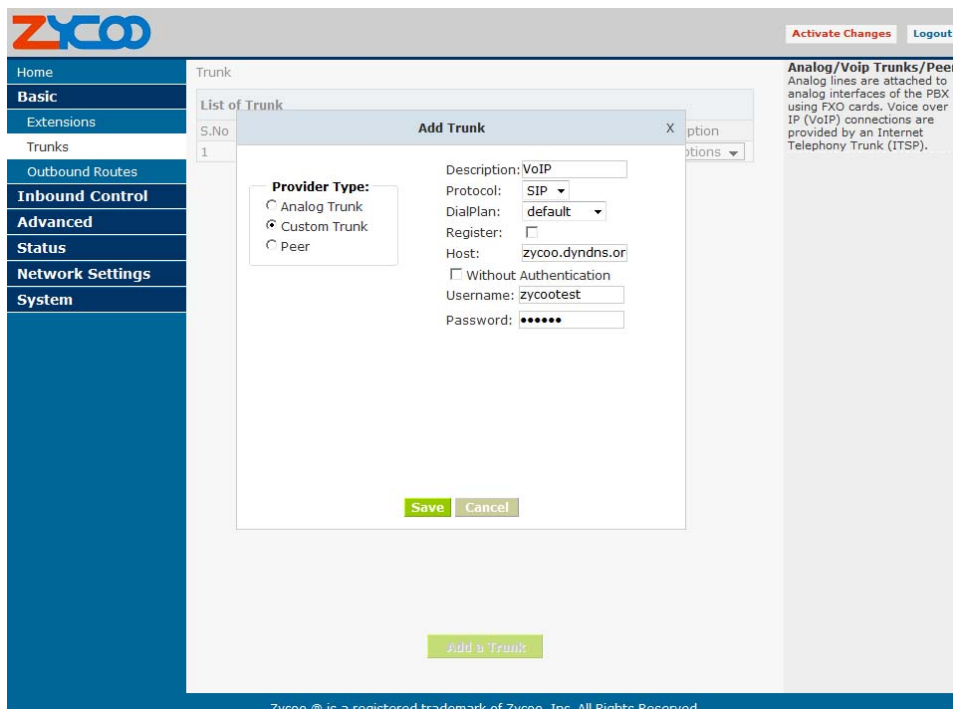
11.4.2 Make call via VoIP trunk

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



Add VoIP service provider

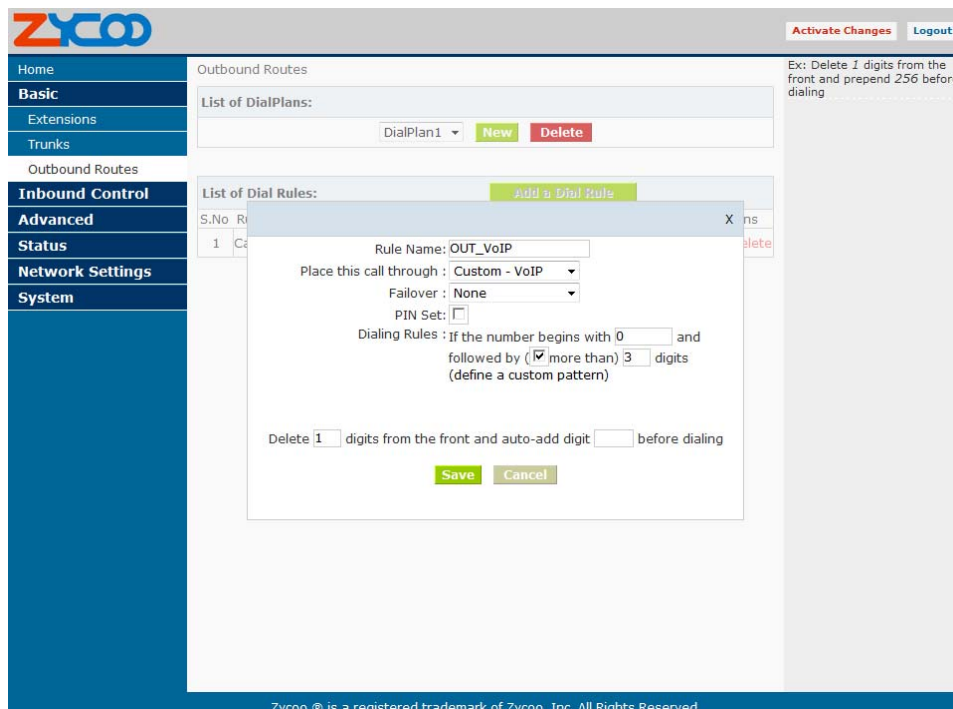
Trunk -> Add a Trunk:
Add a Custom Trunk.



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".

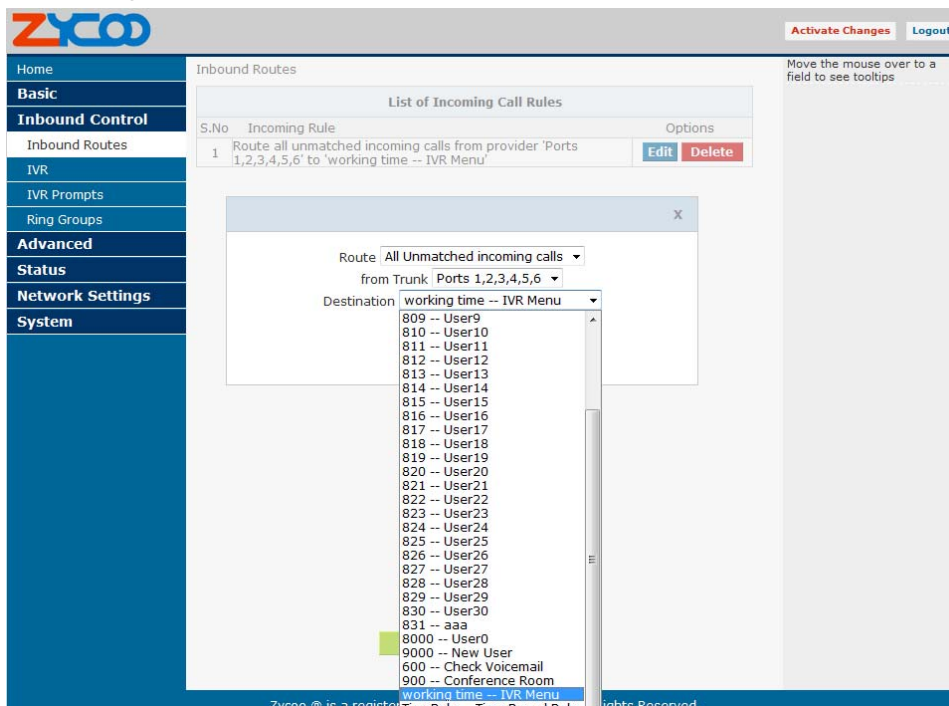
As we can see from the "Out_VoIP" dialing rule, all numbers start with 0 will be cut the first one digits ('0') and sent to my sip service provider.

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.

11.5 How to make an incoming call

Add an Incoming call.



Select Route “All Unmatched incoming calls”

From provider “Port 1,2,3,4,5,6”

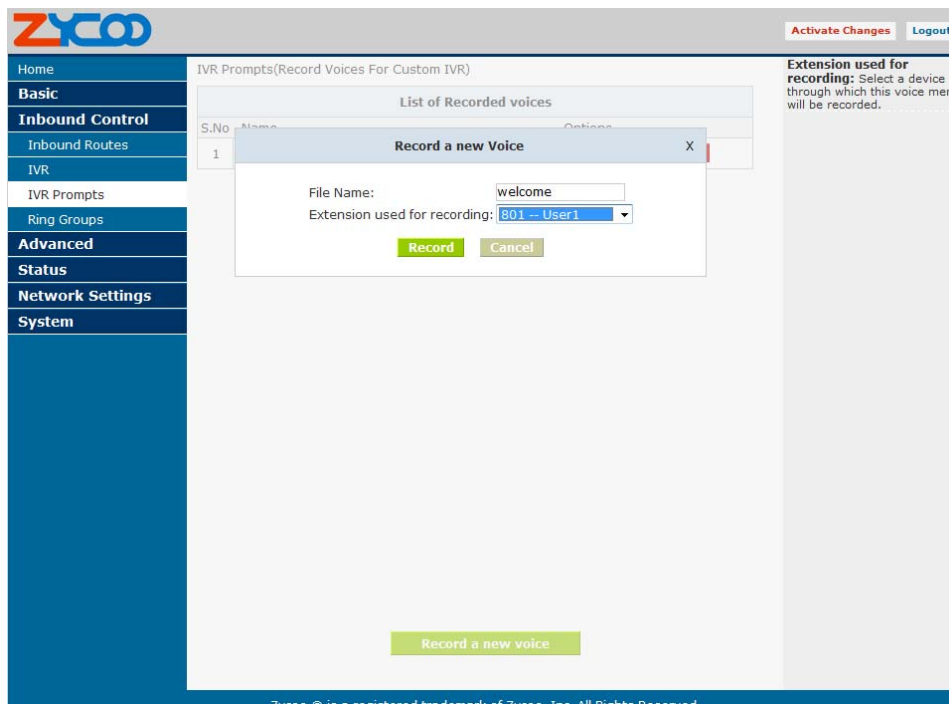
To extension “801 – User1” (here, you can select a extension, a IVR or others)

Then, if there is incoming call from Port 1,2,3,4,5,6 channel, the extension 801 will ring.

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

Add record a custom voice

Record -> Record a new voice



Set the record name is “Welcome”

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

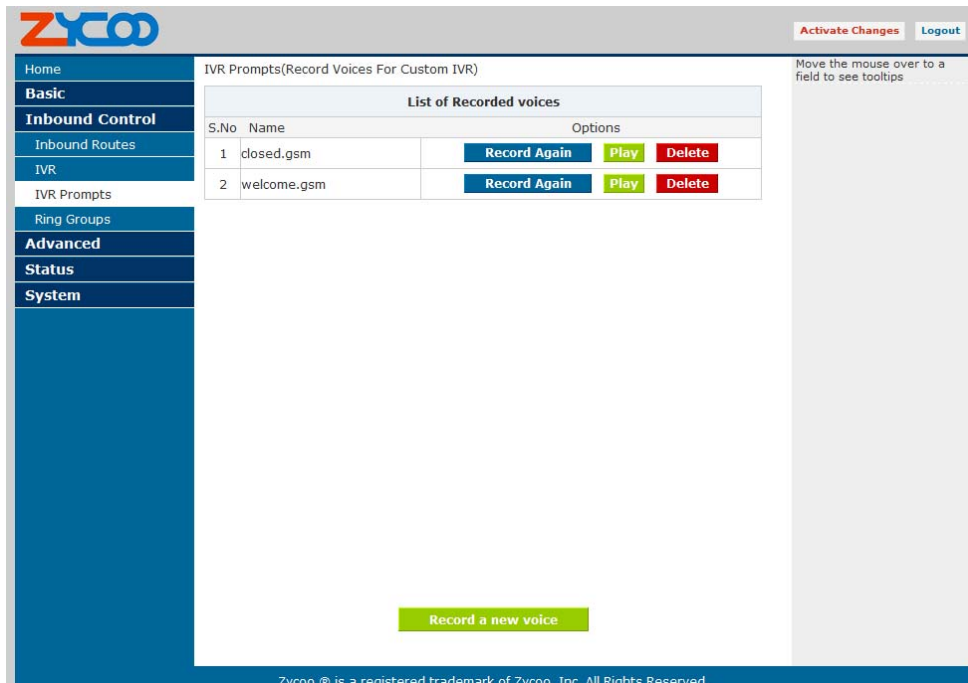
Click Record button

Then, the extension 801 will ring

Pick up the phone record “Welcome” message

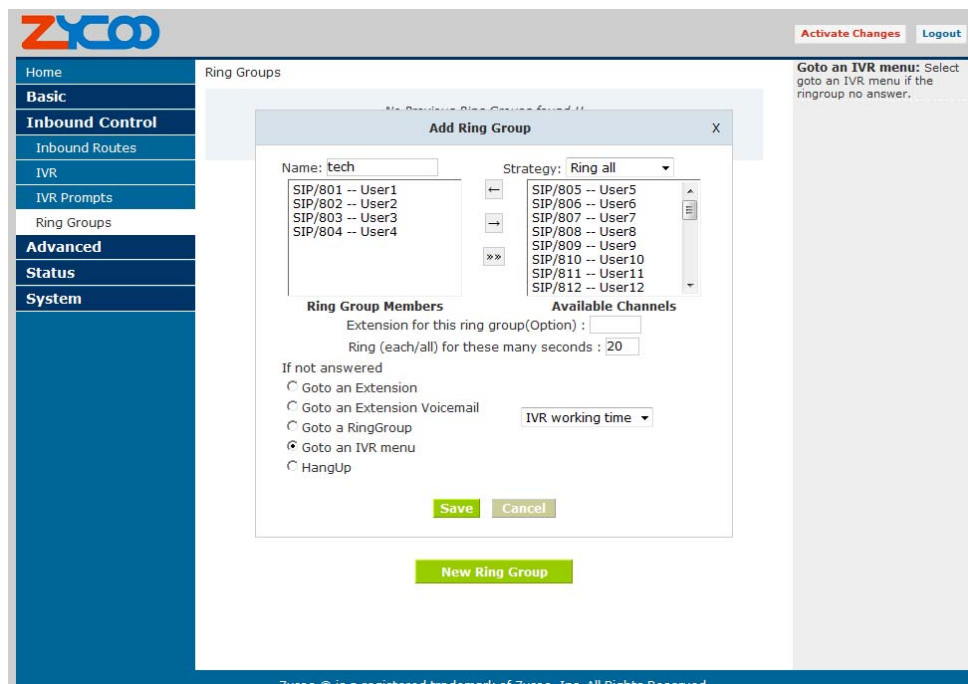
Then hangup and finish the record.

Use the same way to record “Closing” message



Add a Ring Group

Ring Group -> New Ring Group



Example:

Name the ring group “tech”

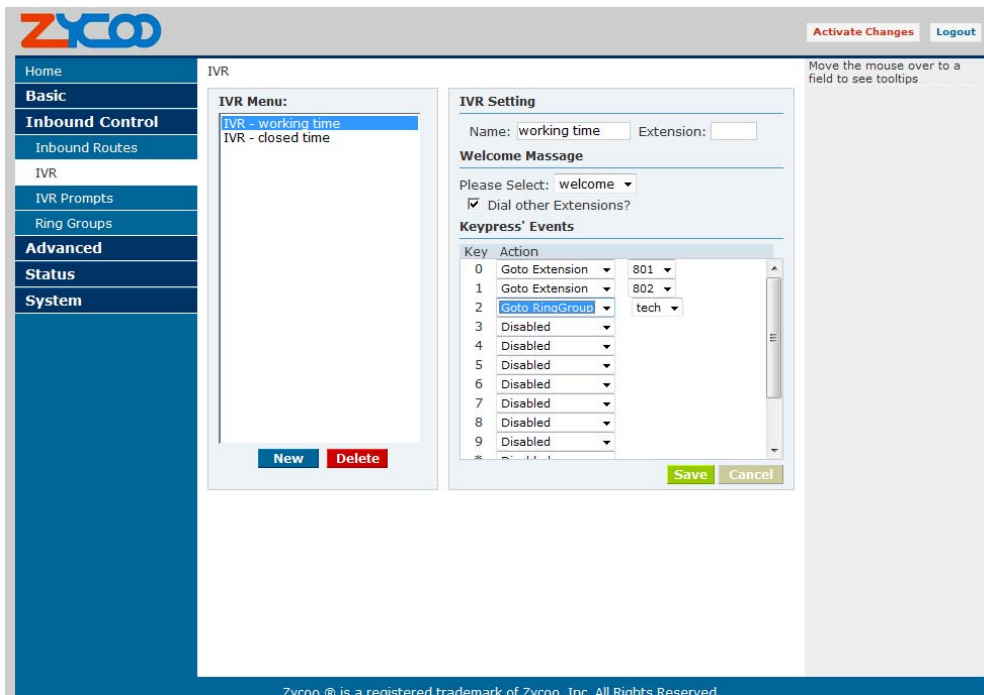
Choose the group members whose extensions are “801, 802, 803, 804”

“if no answered”, choose “goto IVR”-- “working time”

Click “Save” button

Set IVR

IVR



Select IVR-working time, Set welcome message is “Welcome”

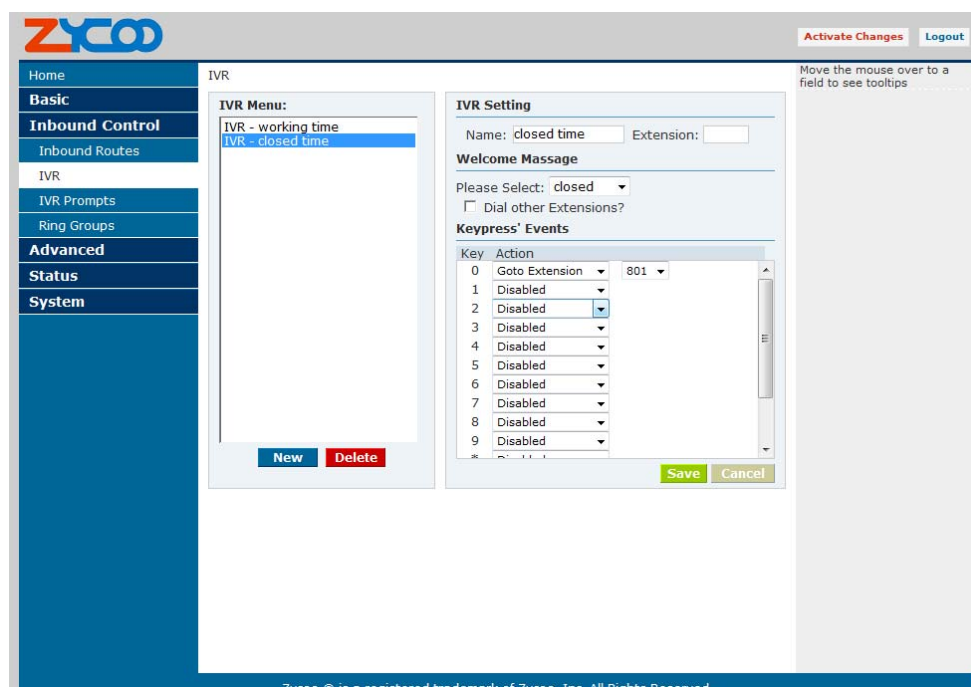
Set keypress' Events

Dial “0” go to extension 805

Dial “1” go to extension 806

Dial “2” go to ringgroup tech

Click Save button

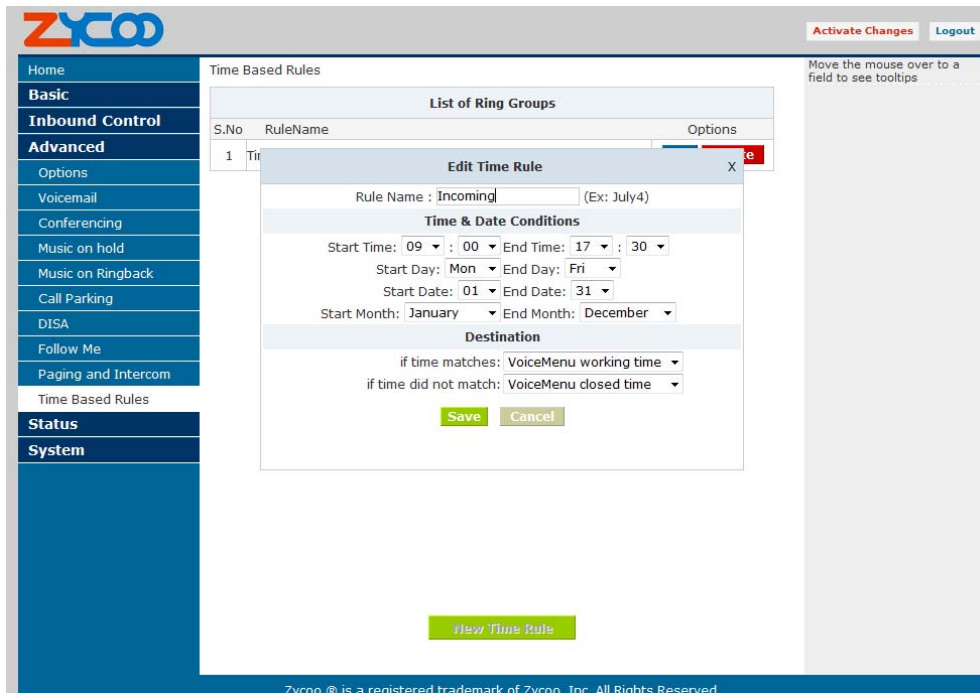


Then set IVR-closed time

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

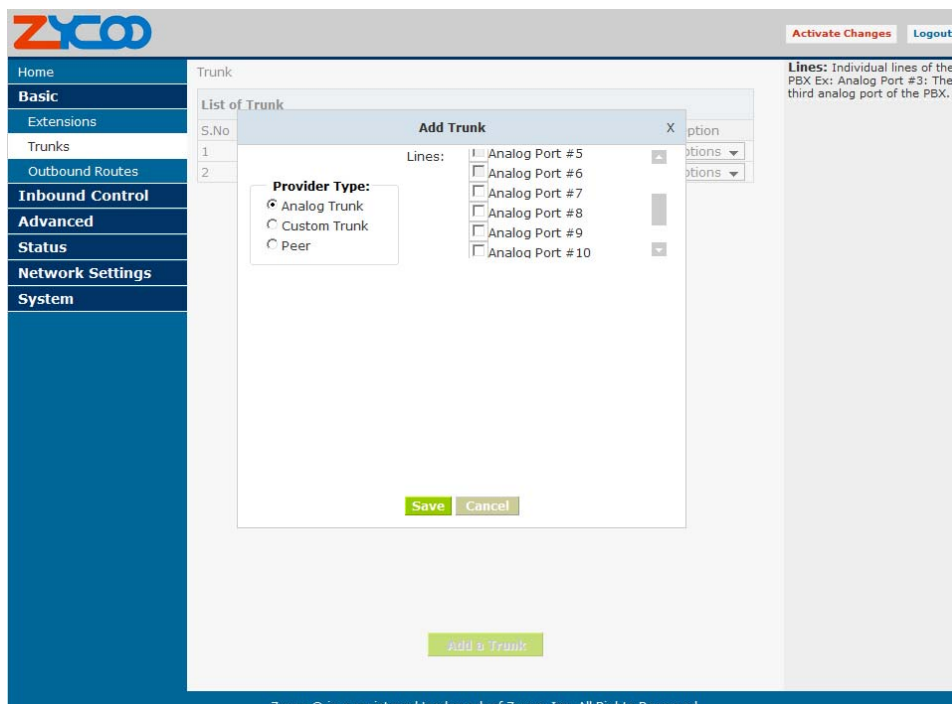
"If time matches" --- go to "working time"

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

Click the save button, saved the configuration

Add a Trunk

Trunks -> add a Trunk



Add an incoming router

Inbound routers -> add an incoming rule

The screenshot shows the ZYCOO web interface for configuring inbound routes. The left sidebar contains navigation options: Home, Basic, Inbound Control (selected), Inbound Routes, IVR, IVR Prompts, Ring Groups, Advanced, Status, Network Settings, and System. The main content area is titled 'Inbound Routes' and contains a table 'List of Incoming Call Rules' with one entry: S.No 1, Incoming Rule 'Route all unmatched incoming calls from provider 'Ports 1,2,3,4,5,6' to 'working time -- IVR Menu'', and Options 'Edit' and 'Delete'. A modal dialog is open with the following configuration: Route: 'All Unmatched incoming calls', from Trunk: 'Ports 1,2,3,4,5,6', and Destination: 'Incoming -- Time Based Rule'. There are 'Save' and 'Cancel' buttons in the dialog. At the bottom of the main area is an 'Add an Incoming Rule' button. The footer text reads: 'Zycop® is a registered trademark of Zycop, Inc. All Rights Reserved.'

Select Route: All Unmatched incoming calls

From provider: Ports 1,2,3,4,5,6

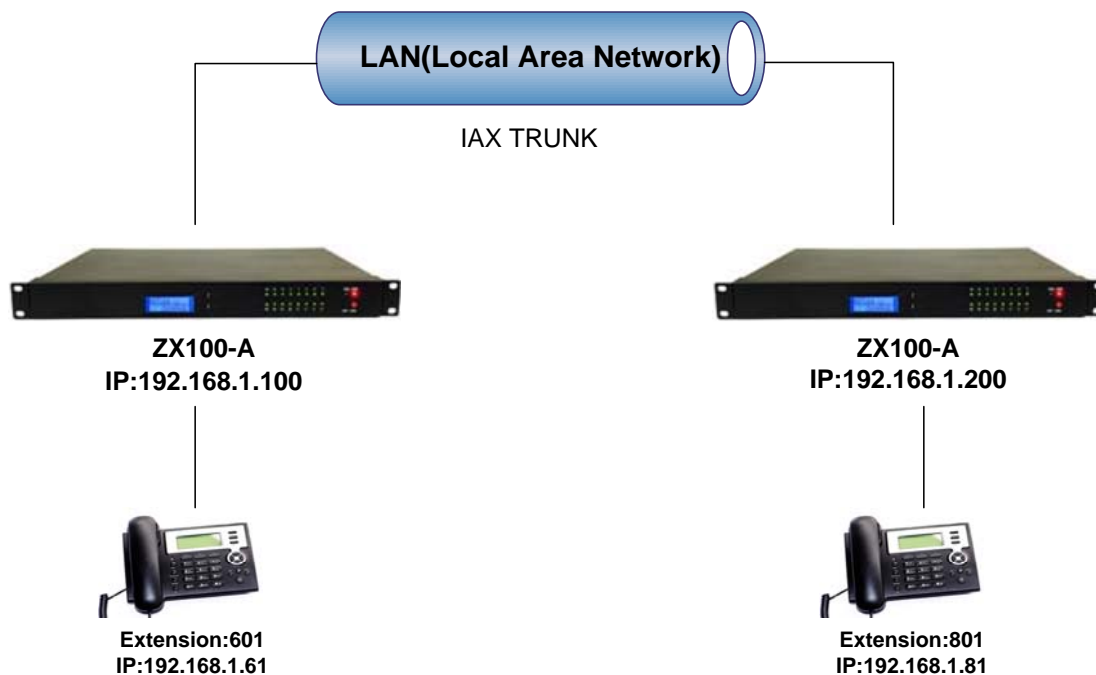
To extension: incoming—Time Based Rule

This screenshot shows the same ZYCOO web interface as the previous one, but the modal dialog is closed. The 'List of Incoming Call Rules' table now shows the rule with the destination 'Incoming -- Time Based Rule'. The 'Add an Incoming Rule' button is still visible at the bottom. The footer text remains: 'Zycop® is a registered trademark of Zycop, Inc. All Rights Reserved.'

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

11.7 How to link two ZX100 IPPBX in the same network

We start from linking two the IP PBX in the same network and then try to expand to different network. Below is the structure of how to link two IPPBX in the same LAN:



Register the ZX100-A as an peer in ZX100-B(via IAX2 trunk),so the extensions in ZX100-A can make calls to ZX100-B's extensions via this "special" trunk.

In above structure:

1. ZP302A registers to ZX100-A as an extension 601.
2. ZP302B registers to ZX100-B as an extension 801.
3. All the extensions under ZX100-A are in the format 6XX.
4. All the extensions under ZX100-B are in the format 8XX
5. Extensions under ZX100-A can make calls to extension under ZX100-B use format 8XX.
6. Extensions under ZX100-B can make calls to extension under ZX100-A use format 6XX.

Step 1: Set up a peer 699 in ZX100-A

In the page Trunks → Add a Trunk

Add Trunk X

Provider Type:

Analog Trunk

Custom Trunk

Peer

Peer Name:

Protocol:

DialPlan:

Host:

Without Authentication

Username:

Password:

Peer Name: ZX100B ;
 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 ZX100-B to link to ZX100-A via this ZX100B Peer.

In the page Trunks--> Add a Trunk

Provider Type: <input type="radio"/> Analog Trunk <input checked="" type="radio"/> Custom Trunk <input type="radio"/> Peer	Description:	Call_ZX100A
	Protocol:	IAX
	DialPlan:	default
	Register:	<input checked="" type="checkbox"/>
	Host:	192.168.1.100
	<input type="checkbox"/> Without Authentication	
	Username:	699
Password:	•••	

Step 3: Set Dial Rule in ZX100-B, all calls start with 6 will be sent to ZX100-A.

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

Rule Name: Call_ZX100A	
Place this call through :	Custom - Call_ZX100A
Failover :	None
PIN Set:	<input type="checkbox"/>
Dialing Rules :	If the number begins with 6 and followed by (<input checked="" type="checkbox"/> more than) 1 digits (define a custom pattern)
Delete 0 digits from the front and auto-add digit before dialing	
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

Step 4: Set the user 601 and Dial Plan in ZX100-A.

In the page: Extensions → Dial Plan

Extensions Setting:	
Extension:	601
Name:	User2
Password:	601
Caller ID:	601
VM Password:	601
E-mail:	
Analog Phone:	
Dial Plan:	DialPlan1

Active the change and apply the test:

1. Register an IP phone ZP302B to ZX100-B with 801 extension.
2. Register an IP phone ZP302A to ZX100-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 ZX100-B's call to ZX100-A,

Accordingly, if you want to call from ZX100-A to ZX100-B, continue as follow:

Step 5: Set Dial Rule in ZX100-A all calls start with 8 will be sent to ZX100-B.

X

Rule Name:

Place this call through :

Failover :

PIN Set:

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

Delete digits from the front and auto-add digit before dialing

Step 6: Set the user 801 and Dial Plan in ZX100-B

Extensions Setting:

Extension:

Name:

Password:

Caller ID:

VM Password:

E-mail:

Analog Phone:

Dial Plan:

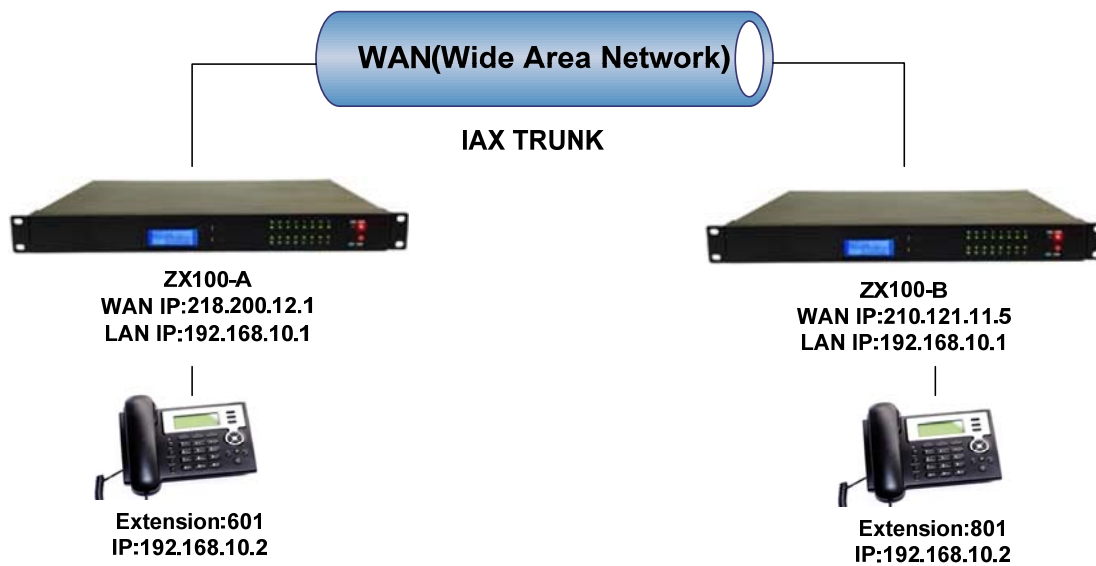
DialPlan1

Active the change and apply the test:

Use 601 to dial 801, and you can see 801 will ring and you can pick up the calls.

11.8 How to link two IPPBX in different network

The generally environment for two ZX100 in different location is: two the ZX100 IP PBX are both in the internet and using the public IP.

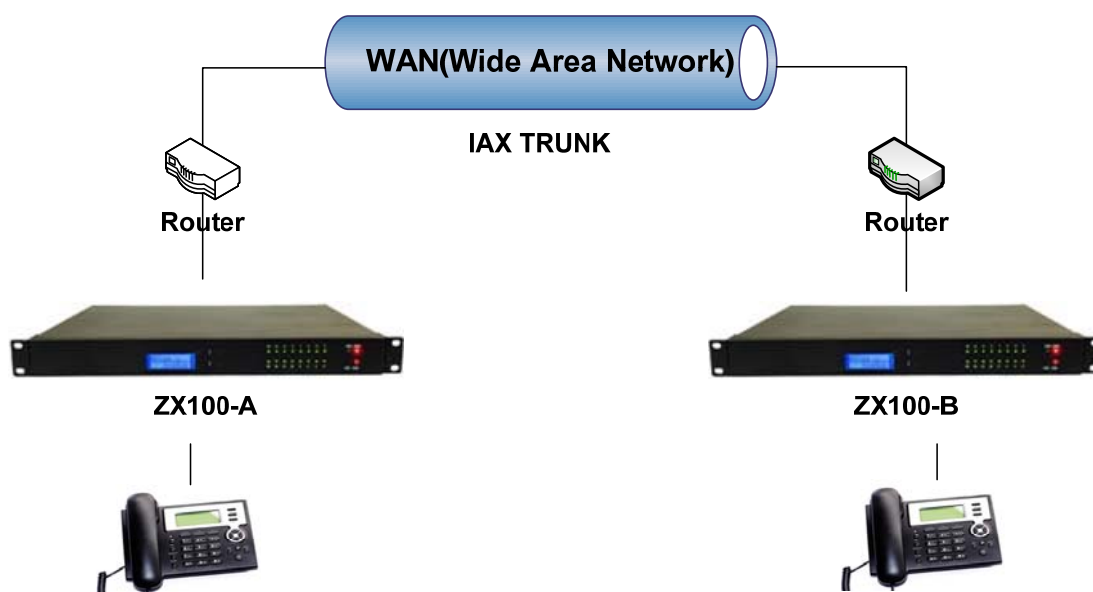


The configuration is same with above guide(10.7) "Link two ZX100 IP pbx in the same network but use the public IP address as the "HOST" settings, like the bellow:

In the page Trunks of ZX100-B-> Add a Trunk

<p>Provider Type:</p> <p><input type="radio"/> Analog Trunk</p> <p><input checked="" type="radio"/> Custom Trunk</p> <p><input type="radio"/> Peer</p>	<p>Description: <input type="text" value="Call_ZX100A"/></p> <p>Protocol: <input type="text" value="IAX"/></p> <p>DialPlan: <input type="text" value="default"/></p> <p>Register: <input checked="" type="checkbox"/></p> <p>Host: <input type="text" value="218.200.12.1"/></p> <p><input type="checkbox"/> Without Authentication</p> <p>Username: <input type="text" value="699"/></p> <p>Password: <input type="text" value="●●●"/></p>
---	---

The generally environment for two ZX100 IP PBX in different location and one or both two are both behind router and using the private IP. So, we need to do port forwarding in the router and make ZX100 IP PBX can reach to each other.



Step 1: Set port forwarding in the router for ZX100-A

For the ZX100-A is behind the router, 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 ZX100-A (192.168.1.21:4569). Below is the setting page in a linksys router:

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"/>
Telnet	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 Provider Host in ZX100-B

Set up the service provider and calling rule in ZX100-B to make it register to ZX100-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: Set port forwarding in the router for ZX100-B

Use the same method as Step 1 do port forwarding in router-B for ZX100-B as above.

Setp4: Link two ZX100 and make calls

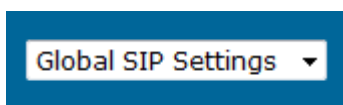
Accordingly, set the 601 users in ZX100-A and 801 users in ZX100-B, and build the correct dial rules as above, you can make calls between two the ZX100 IP PBX.

Noted: You can also apply a DDNS to get one fixed domain for both ZX100 IP PBX and connect to each other rather than use the Port Forwarding in the router.

11.9 How to resolve problems about hearing only on one side

If your IPPBX behind the Router, you should build a IP Address Map to resolve this problem as follow:

Management---->Show Advanced Options ----> Global SIP Settings



--->NAT Support

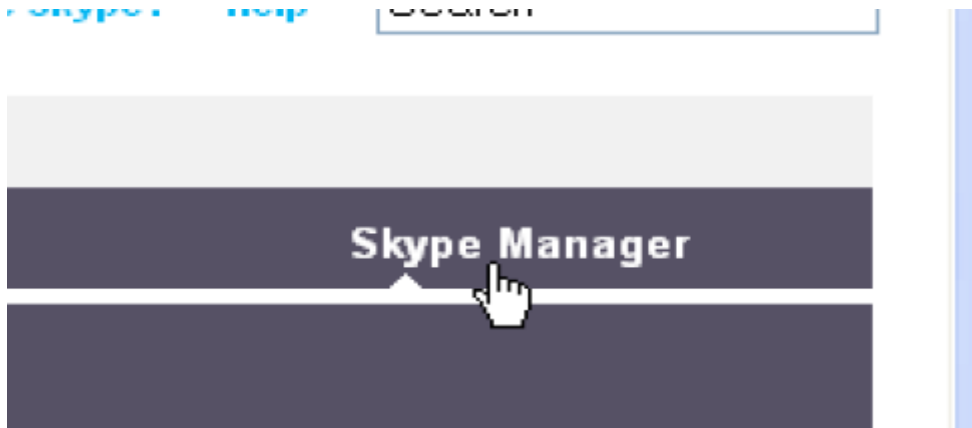
NAT SupportExtern ip: Extern Host: Extern Refresh: Local Network Address: NAT mode: Allow RTP Reinvite:

- [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

Chapter12 How to use Skype account in ZX100

12.1 Register for Skype Manager

1. Visit skype.com/business and click **Skype Manager**



2. Complete the on-screen instructions to register for Skype Manager. You can either use your existing personal account or create a new one specifically for your Skype Manager.
7. Please bear in mind that the account you use to register will be used to administer products and credit throughout your business. We therefore recommend that you create a new Skype account using your business name.

12.2 Create a SIP Profile and buy a Channel Subscription


Note: You need to be signed into Skype Manager to access the Skype for SIP settings.

1. Click **Features** in the toolbar



2. In the **Features** menu on the left, click **Skype for SIP**.
3. Click **Create a new profile**.
4. Give your SIP Profile a friendly name so it's easier to remember and click on Next. Your Profile's registration details, including its username and password are displayed. Make a note of these details so that you can set up and configure your PBX.

Authentication details



Please choose the method of authentication needed for your PBX.

Registration
(Username/password) **or, IP Authentication**

SIP User	99050000015459
Password	4j9x7i7Ybggv8g Generate a new password
Skype for SIP address	sip.skype.com
UDP Port	5060

5. Click **Profile settings**.
6. Click **Buy a channel subscription to activate this profile**.
7. Enter the number of channels you require and click **Buy now**.
8. Channel subscriptions are the amount of concurrent calls you would like to use with your SIP Profile. These channels are charged on a monthly basis.
9. If you don't want to make outbound calls with Skype for SIP, please proceed to step 6.

12.3 Allocate Skype Credit to the SIP Profile

1. Click **View profile** next to the name of the SIP Profile to which you want to allocate credit.
2. Click **Set up outgoing calls**.
3. Enter the amount of Skype Credit you want to allocate to the SIP Profile and click **Add credit**.

Profile settings



Profile name	Profile 5	▼
Calling channels	Buy a channel subscription to activate this profile	
Outgoing calls	Set up outgoing calls ×	
	Add credit	Auto-Recharge settings
	<input type="text" value="10.00"/> Add credit	
Caller ID	Set up Caller ID ▼	
Incoming calls	Add a number or business account ▼	

- If you want to enable **auto-recharging**, click on the Auto-Recharge settings tab, enter the recharge amount and the minimum balance required before recharging, then click **Save changes**.

12.4 Configure your Skype for SIP certified PBX for outbound calls

In the trunk of our IPPBX setting:

<p>Provider Type:</p> <p><input type="radio"/> Analog Trunk</p> <p><input checked="" type="radio"/> Custom Trunk</p> <p><input type="radio"/> Peer</p>	<p>Description: <input type="text" value="skype"/></p> <p>Protocol: <input style="border: none; border-bottom: 1px solid #ccc; background-color: #f0f0f0; padding: 2px 5px;" type="text" value="SIP"/></p> <p>DialPlan: <input style="border: none; border-bottom: 1px solid #ccc; background-color: #f0f0f0; padding: 2px 5px;" type="text" value="default"/></p> <p>Register: <input checked="" type="checkbox"/></p> <p>Host: <input type="text" value="sip.skype.com"/></p> <p><input type="checkbox"/> Without Authentication</p> <p>Username: <input type="text" value="9905000001545"/></p> <p>Password: <input type="password" value="••••••••••"/></p>
<p>Save Cancel</p>	

Outbound setting of our IPPBX:

X

Rule Name:

Place this call through :

Failover :

PIN Set:

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

Delete digits from the front and auto-add digit before dialing

12.5 Make an outbound call

After we have done above, in the extension we can dial 00 + Country Code + City Area Code + local number to dial out via skype line

For example: Dial number 00862885337096 will contact our company.

12.6 Configure your Skype for SIP certified PBX for inbound calling

Inbound Routing of our IPPBX:

X

Route

from Trunk

Destination

12.7 Set up a business account to test inbound calls from people with Skype

1. Create a new business account in Skype Manager. For more information on creating a new business account, please see the [Skype Manager User Guide](#).
2. Click **View profile** next to the name of the SIP Profile to which you want to add the business account.
3. Click **Add a number or business account**.
4. In the **Add business account** tab, enter the newly created business.

Caller ID ? Set up Caller ID

Incoming calls + Add a number or business account ×

You can receive incoming calls on your SIP Profile via [Skype Online Numbers](#) and via [Skype business accounts](#). When someone calls your Online Number or contacts your business account on Skype the calls get forwarded to your SIP Profile.

Add Online Number + Add business account +

Add an existing business account

Architects.Engineering + Create a new account

Extension number (optional) ?

Confirm

Important: If a Skype account is attached to a SIP Profile it cannot be used to sign into Skype on your computer or any other device.

5. Click Confirm.

12.8 Make a test inbound call from Skype

Call the business account's Skype Name you created in step 7 from Skype.

12.9 Assign an Online Number to receive calls from landlines and mobile phones

1. Click **View profile** next to the name of the SIP Profile to which you want to assign an Online Number.
2. Click **Add a number or business account**.
3. Click **Buy a new number**

Caller ID ? Set up Caller ID

Incoming calls + Add a number or business account ×

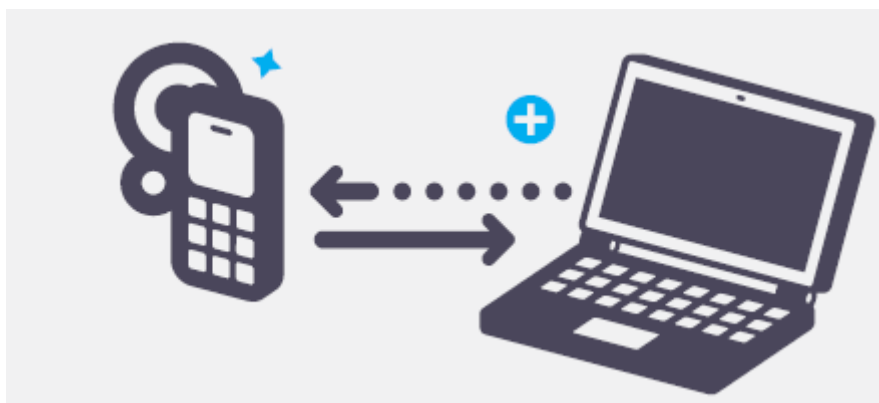
You can receive incoming calls on your SIP Profile via [Skype Online Numbers](#) and via [Skype business accounts](#). When someone calls your Online Number or contacts your business account on Skype the calls get forwarded to your SIP Profile.

Add Online Number + Add business account +

Buy a new number

12.10 Make a test inbound call from a landline or mobile phone

Call the Online Number associated with the SIP Profile from a landline or mobile phone.



You have now successfully set up Skype for SIP for use with your Skype for SIP certified PBX.

For more help with setting up and using Skype for SIP, please see support.skype.com or check the [skype for sip user guide](#)