ZX50 IP PBX

User's Manual

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Chapter1 Brief Introduction

Thank you for your purchasing the ZX50 series of IP PBX.The all-in-one ZX50 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 visual voice mail,music on hold ,and auto attendant.In addition,the ZX50 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,SMEs 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 ZX50 IP PBX,an SME 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.

- Please use the external power supply which is included in the package. Other power supplies may cause damage to the device, affect the performance or induce noise.
- 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 ZX50 Specification

3.1 Apearance&Model

ZYCOO ZX50 Series IPPBX product line include ***ZX50-A4,ZX50-A8,ZX50-AG42, ZX50-G4,ZX50-AE41,***so far, since they have almost the same software and structure so we will use ZX50-AG42 as the demo unit on this article.

Model		FXS	FXO	GSM	E1
ZX50-A4	A404		4		
	A422	2	2		
ZX50-A8	A808		8		
	A826	2	6		
ZX50-AG42	AG4204		4	2	
	AG4222	2	2	2	
ZX50-G4	G4			4	
ZX50-AE41	AE4104		4		1
	AE4122	2	2		1

3.2 System Features

ZX50 series of IPPBX is an embedded ippbx based on industry standard for Home&SMEs, which is not only a PBX, but also as a voice mail Server, IVR server, conferencing server. With excellent echo cancellation function, it can meet most of the customers' requirement.

- Up to 30 concurrent calls.
- Above 100 registers
- 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 attendand
- Multimedia Music On Hold and Ring Back
- Call Monitoring
- 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)
- Codec Negotiation/Echo cancelation/VAD.etc
- FAX T.38

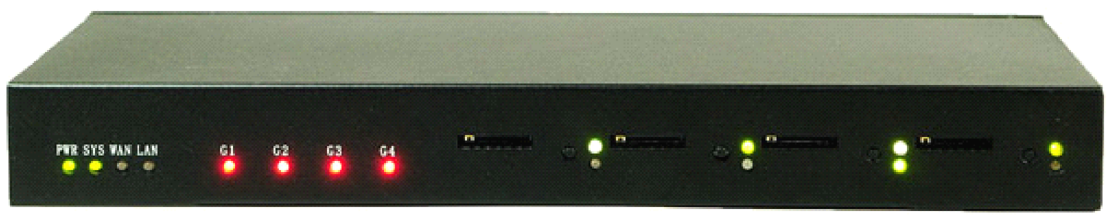
3.3 Interface&Panel

Here,we take ZX50-G4 as the sample to show the interface and the indicators at the back and frond panel.

1) Back panel

- 4 * GSM Antenna
- 2 * Network Interface (RJ45)
- 1 * Power port（DC 12V 2A）
- 1 * Reboot Button

2) Frond 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
G1~G4	GSM Modules Status	Red	GSM channel
		Off	Failed
*1-4	Analog Modules Status	Green	FXS channels
		Red	FXO channels
		Off	Failed

3) Hardware
- 32bit embedded RISC DSP
- 1G Onboard Nand Flash
- 128M Onboard SDRAM

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
- GSM Antennas 4 Unit
- Power Adapter 1 Unit

3.4 Default configuration

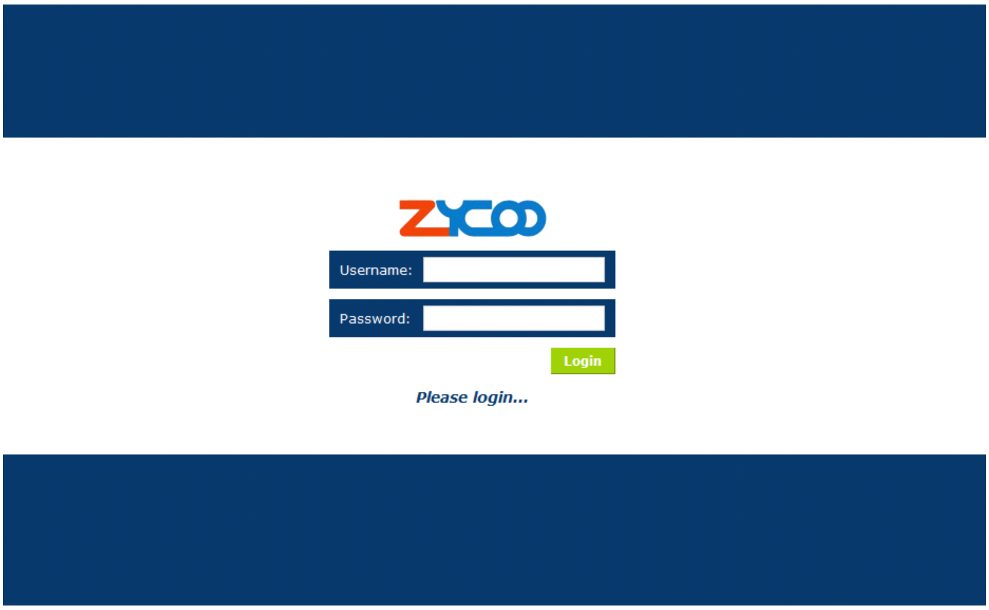
1. WAN port IP address: http://192.168.1.100:9999
2. LAN port IP address: http://192.168.10.100:9999
3. LAN port super IP: 169.254.1.254/255.255.0.0
4. Web GUI username: admin
5. Web GUI password: admin

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

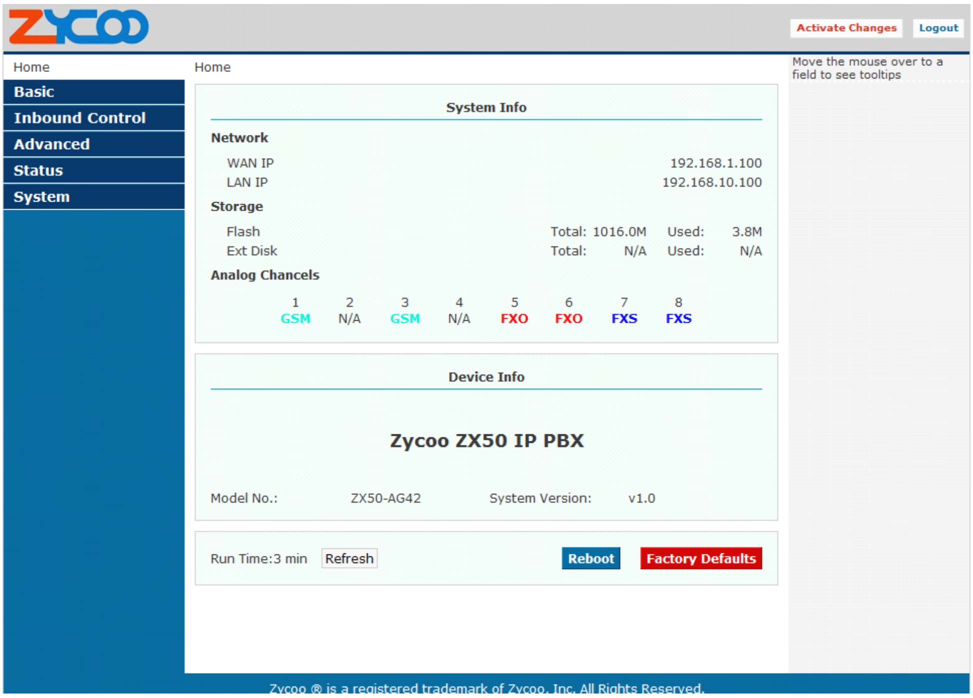
Chapter4 Login in Home Page

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:



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.



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

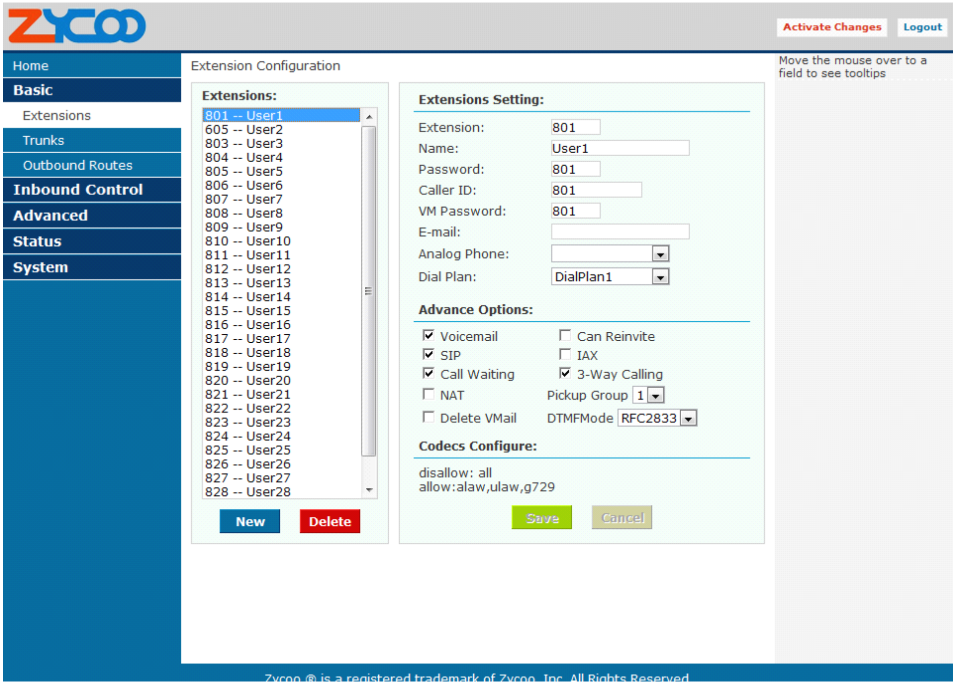
The home page is used for logout, Reboot and Factory Defaults.

- 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 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:

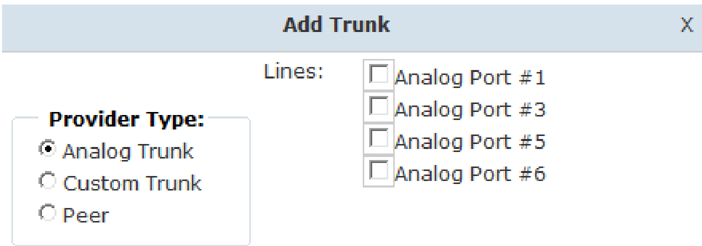
- Voicemail — The extension support voicemail
- SIP — The extension support SIP protocol
- IAX — The extension support IAX protocol
- Call Waiting — The extension support Call Waiting function
- 3-Way Calling — The extension support 3-Way Calling functions

- Pickup Group Select pickup group of the extension
- 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.
- 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**.

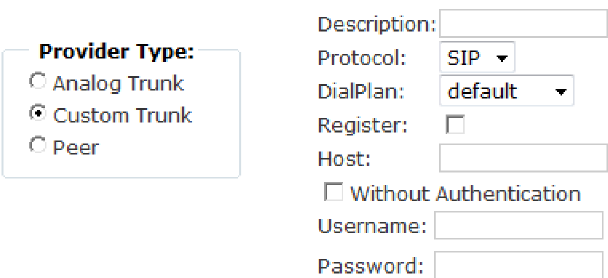


Analog Trunk

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:

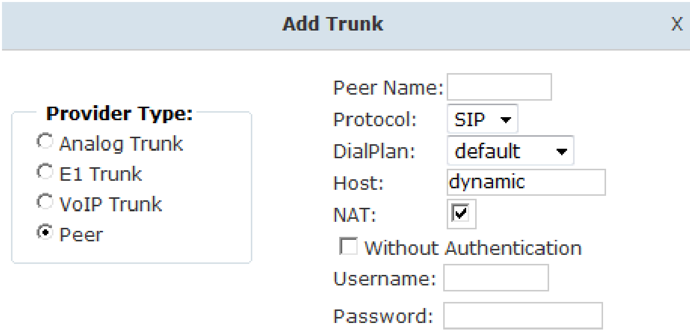


- Description The description should be used as the name of the custom VoIP definition
- Protocol Specify either a IAX or SIP protocol
- DialPlan Select a DialPlan for this trunk.
- Register Enable/Disable server register. Registering is not required for all providers
- Host The IP address of your service 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

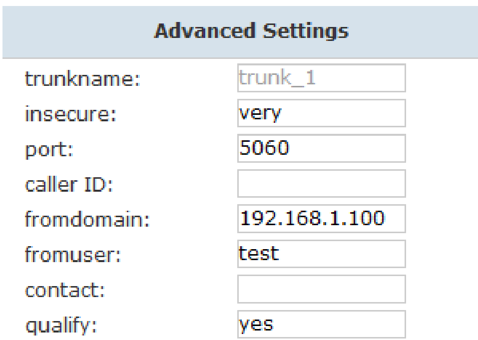
The Peer option allows you to create a custom VoIP Peer.



- 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.



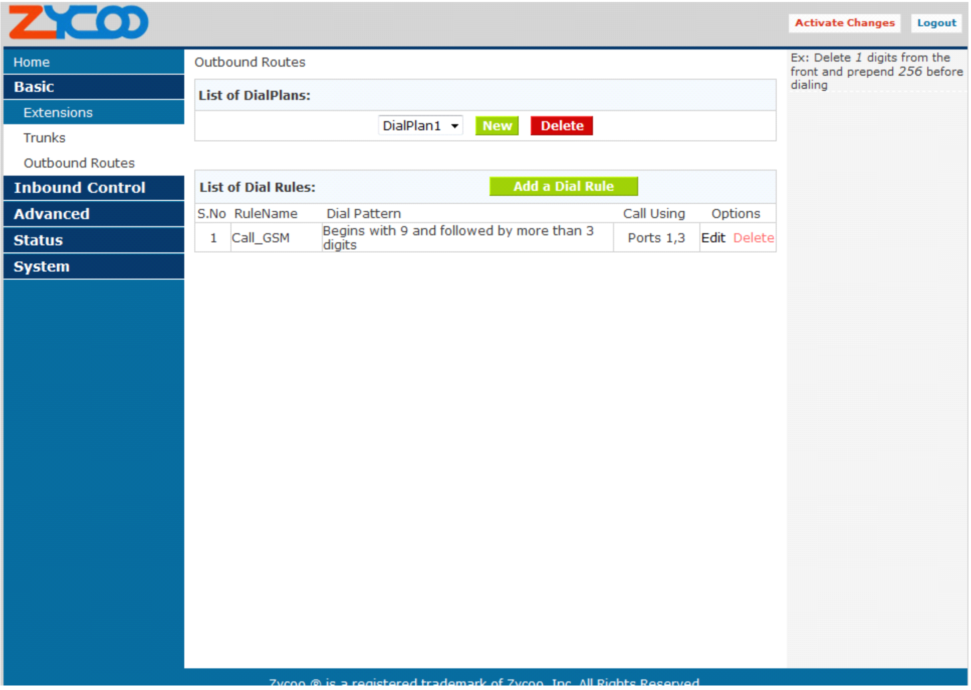
- Trunkname Specify a trunk name if you want to refer to the service provider definition as something other than specified in Comment
- Insecure This option specifies how connects to a service provider (host)

should be handled. Valid options are very/yes/no/invite/port. (Default is “very”)

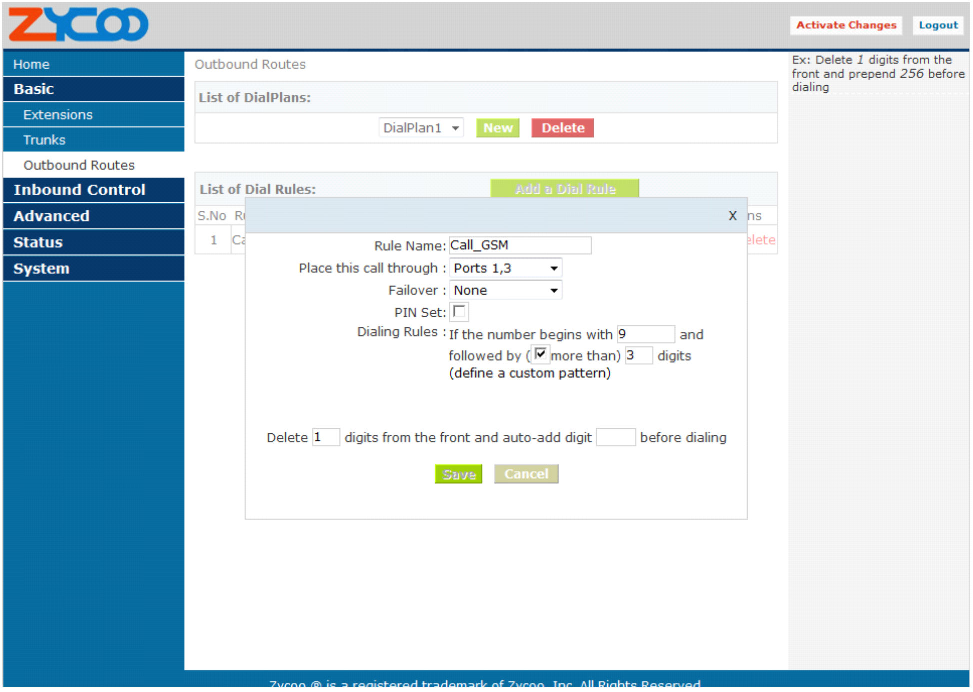
- Port The register request is sent through the port. (Default is SIP:5060,IAX:4569)
- Caller ID The caller ID will be set to the value specified in this field
- Fromdomain Sets default from: domain in SIP messages when acting as a SIP client.
- Fromuser Sets default from: user in SIP messages when acting 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).

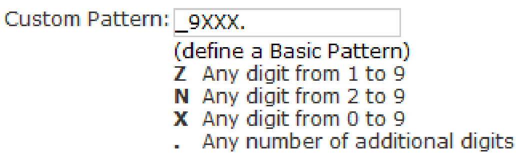


Click on Add a Dial Rule to define a new DialPlan. The following dialog will be displayed.



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.



N Any digit from 2 to 9

Z Any digit from 1 to 9

X Any digit from 0 to 9

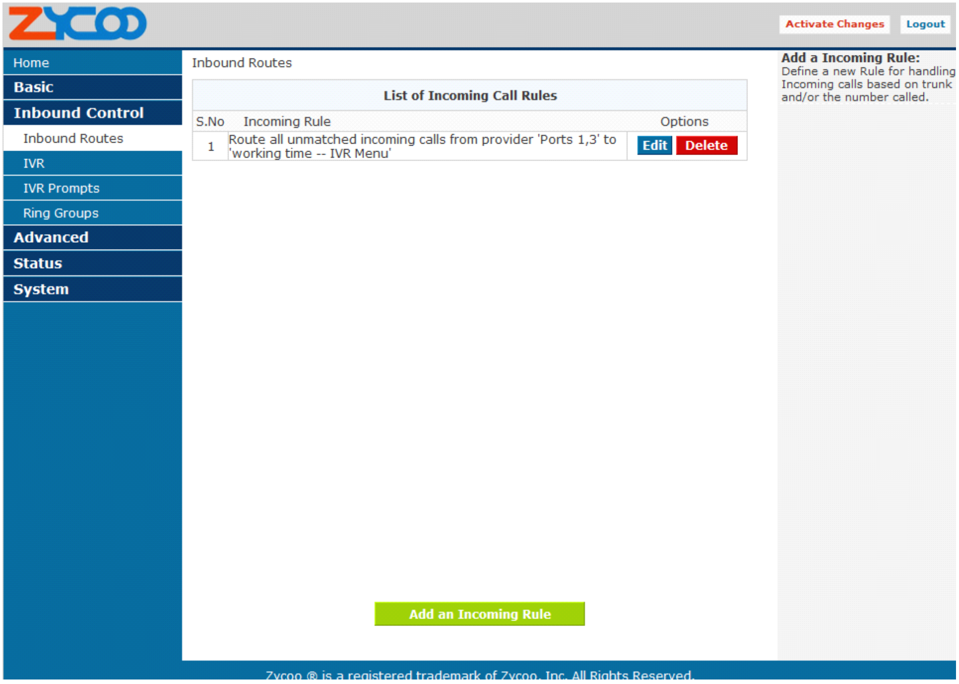
. Any number of additional digits

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.

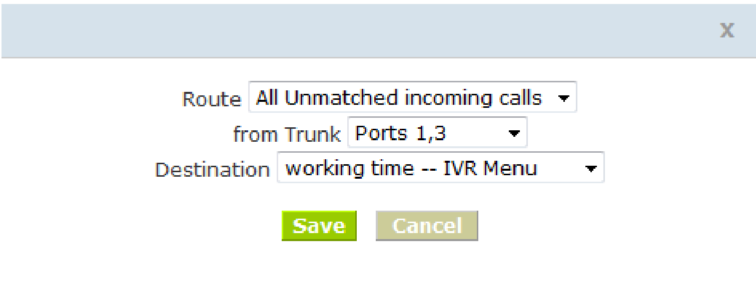
- Delete This option gives you the opportunity to remove specified digits from the call being dialed and replace them with the digits needed to make the call. You can also prepend digits to the beginning.

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.

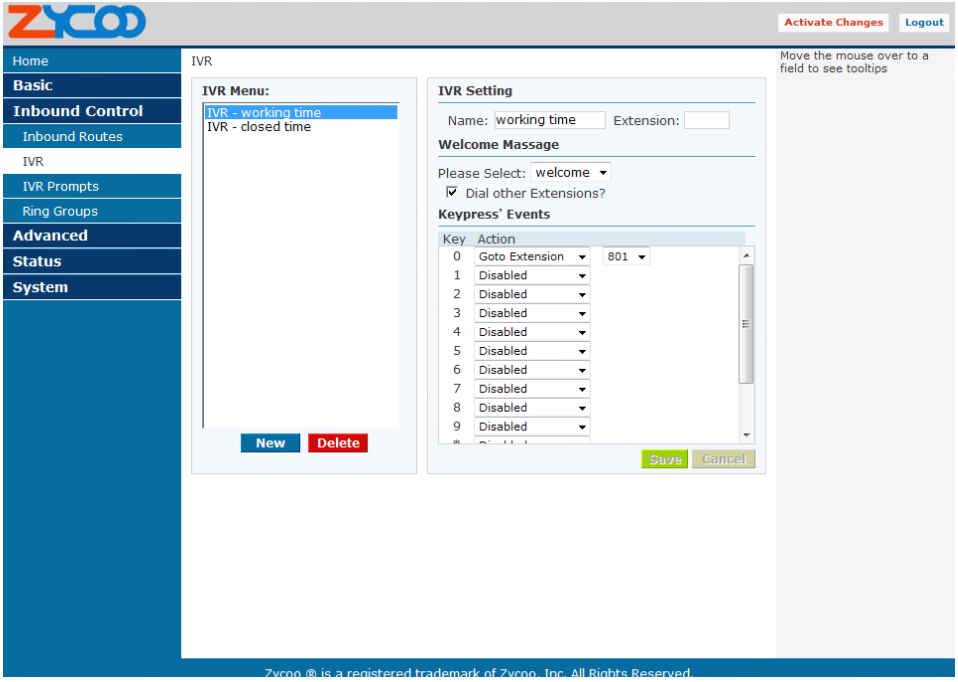


There are only a few options you need to configure

- 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.

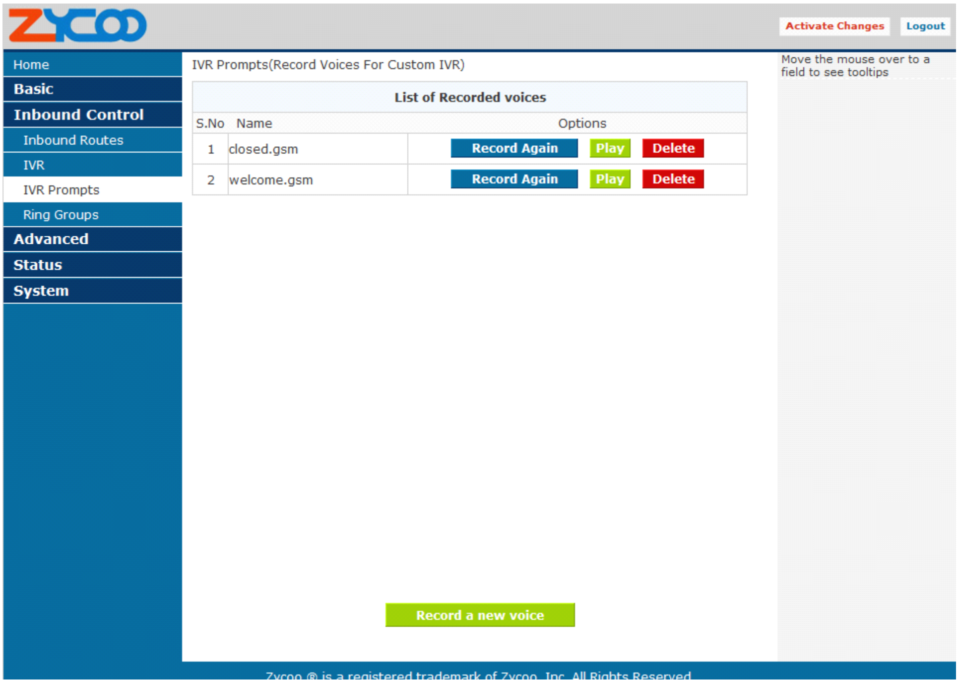


Voice menus are constructed depending on your needs. Just like your business you need to create the solution best suited to your customers.

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

6.3 IVR Prompts

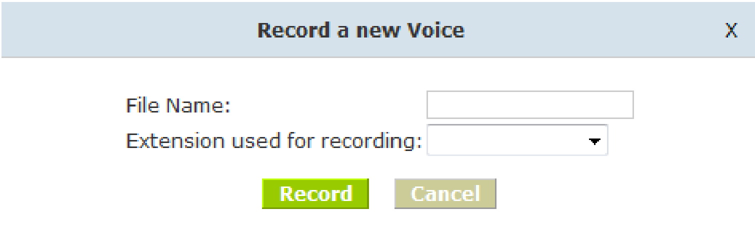
In the event that one wants to record custom IVR prompts for the IP PBX, which can be used in a IVR, the Record may be used.



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

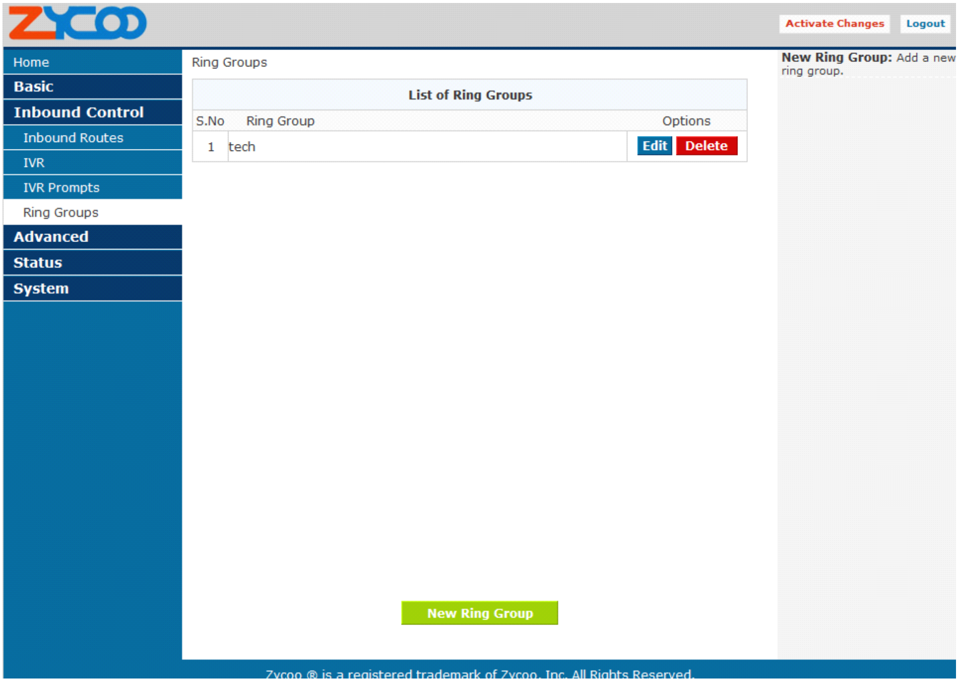
There are two options under “Record a new voice”



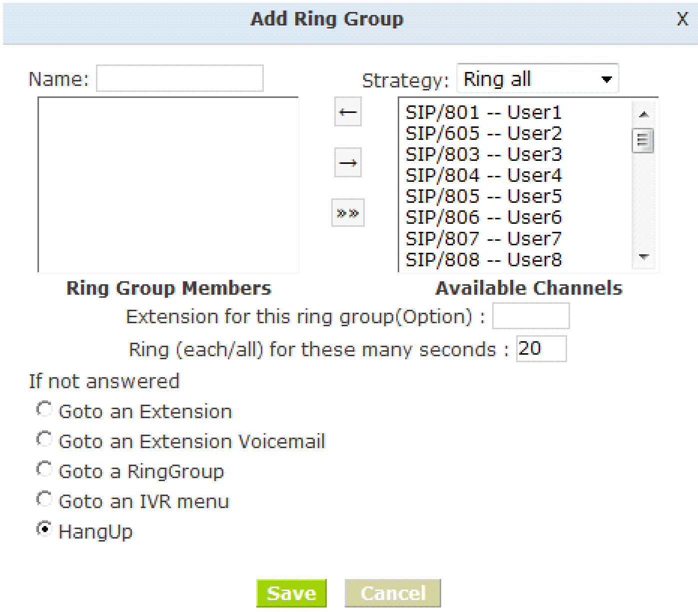
- File Name This text entry box specifies the saved name of the file that is to be recorded.
- Extension Used for Recording This drop-down select box allows the user to choose which extension will dial to wait for the user to speak the prompt

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



Define Ring Groups to Dial more than one extension

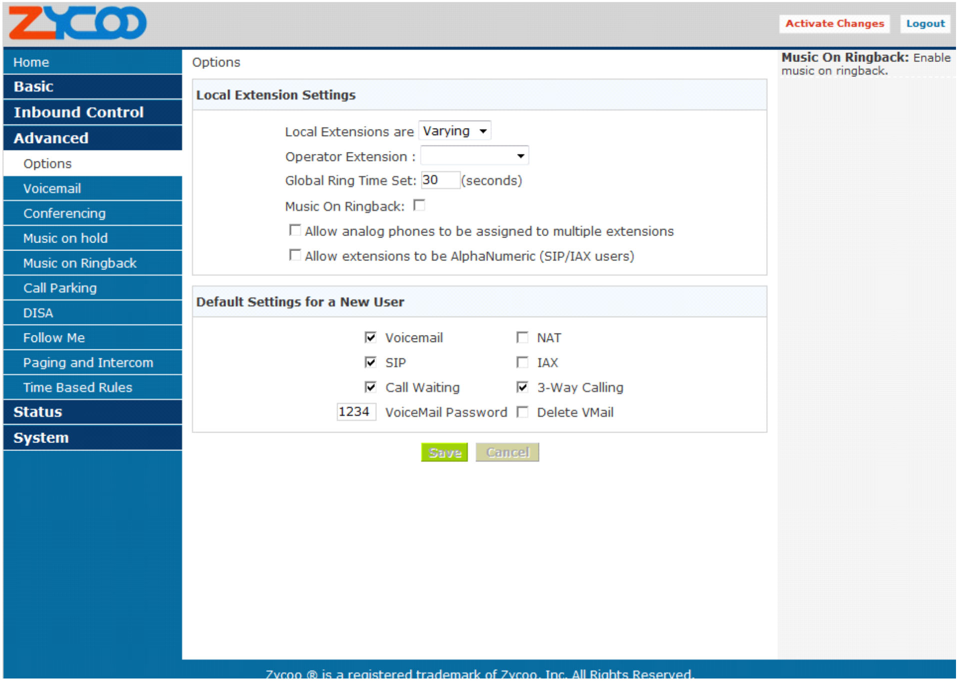


- **Name** Set a Ring Group name
- **Strategy** There is a drop-down list, you can choose Ring all or Ring in order.
- **Ring Group Members** Add Ring Group member from Available channels.

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.

Chapter7 Advanced Configuration

7.1 Options

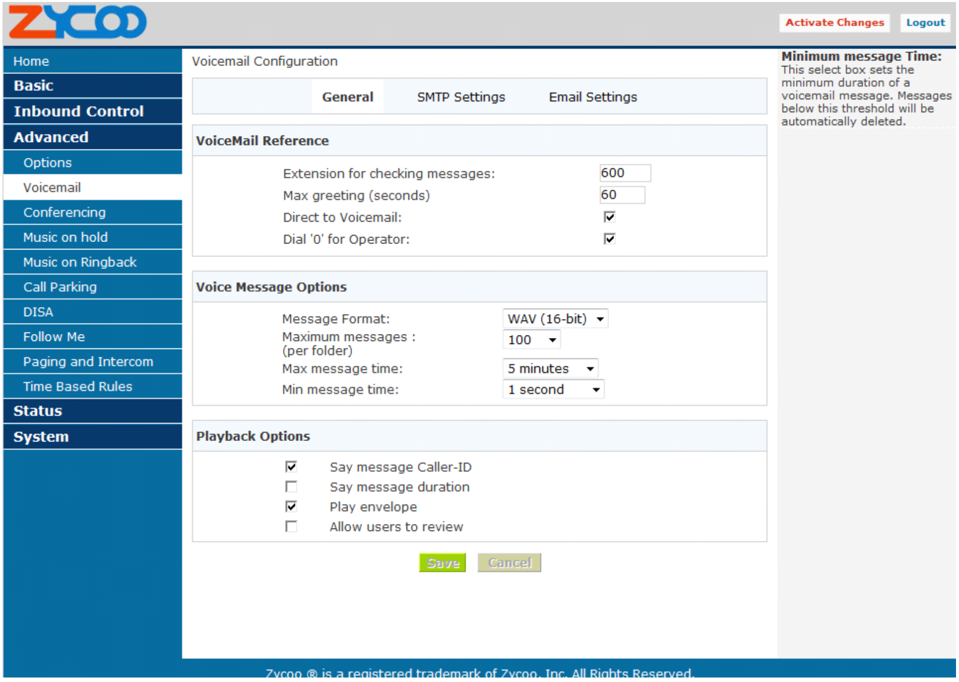


- Local Extensions are — Set up the digit of local extensions
- Operator Extension — Set up Operator Extension. (you can dial "0" go to the extension at any time)
- Global Ring Time Set — Set default each extension ring time.
- Music On Ringback — Enable/Disable the Music On Ringback function
- Default 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 ZX50 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.

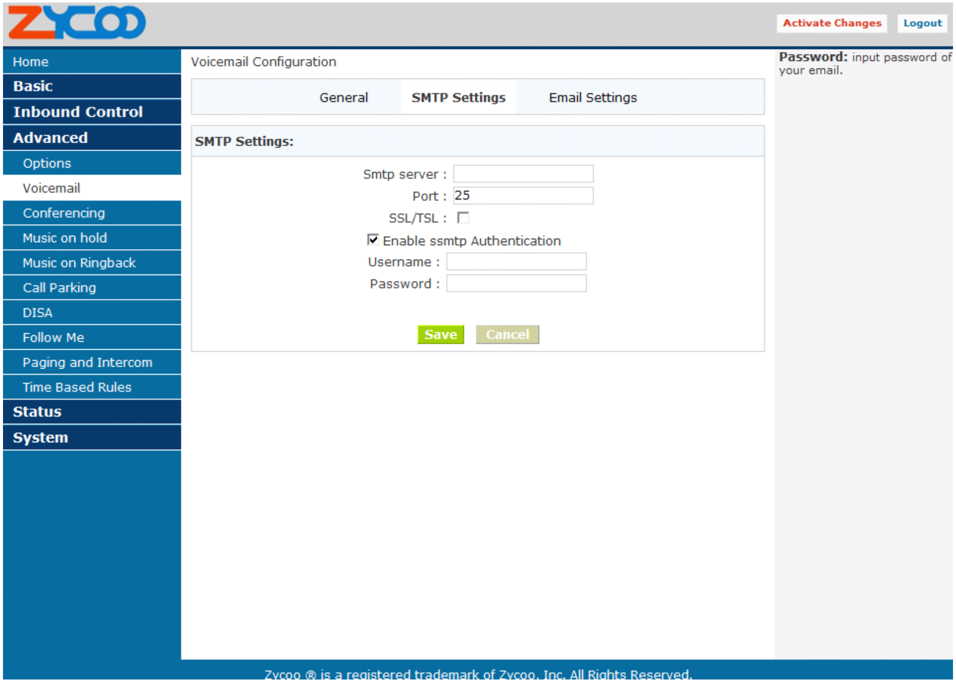
- Message Format — This option gives you the ability to choose the format in which messages will be mailed.
- Maximum Messages — The maximum number of messages per voice mail box is set here.
- Maximum Message Time — The maximum duration of a message left by a caller is set here
- Minimum Message Time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

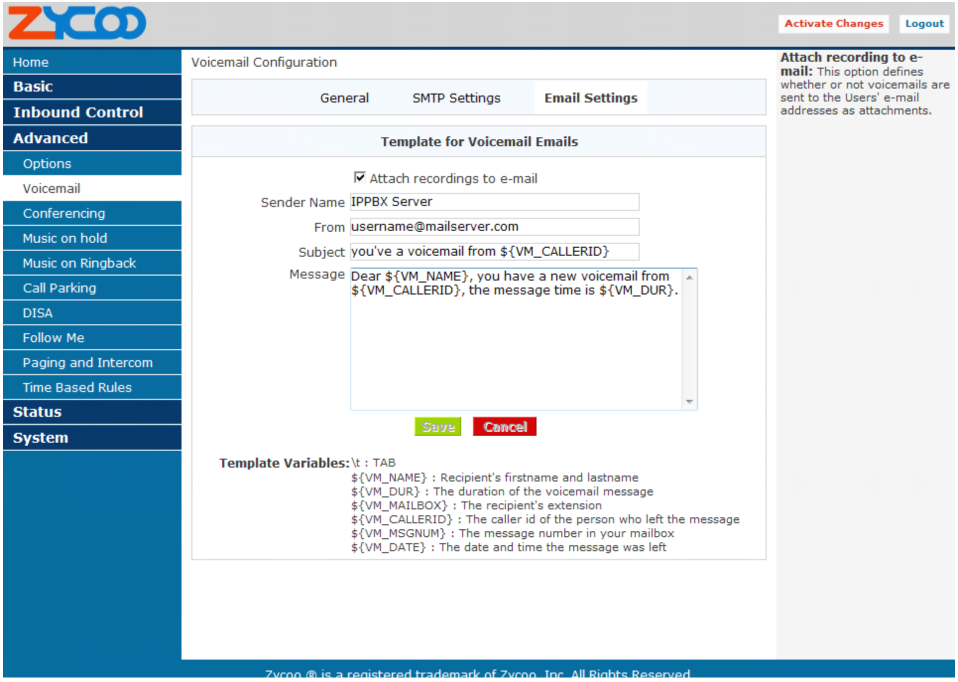
- Say Message Duration This option identifies exactly how long the message lasted.
- Play Envelop The envelope provides the date, time, and caller ID related to a voice mail.
- Allow Users to Review This option provides incoming callers the option to review their message before it is saved and can be 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



- 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.

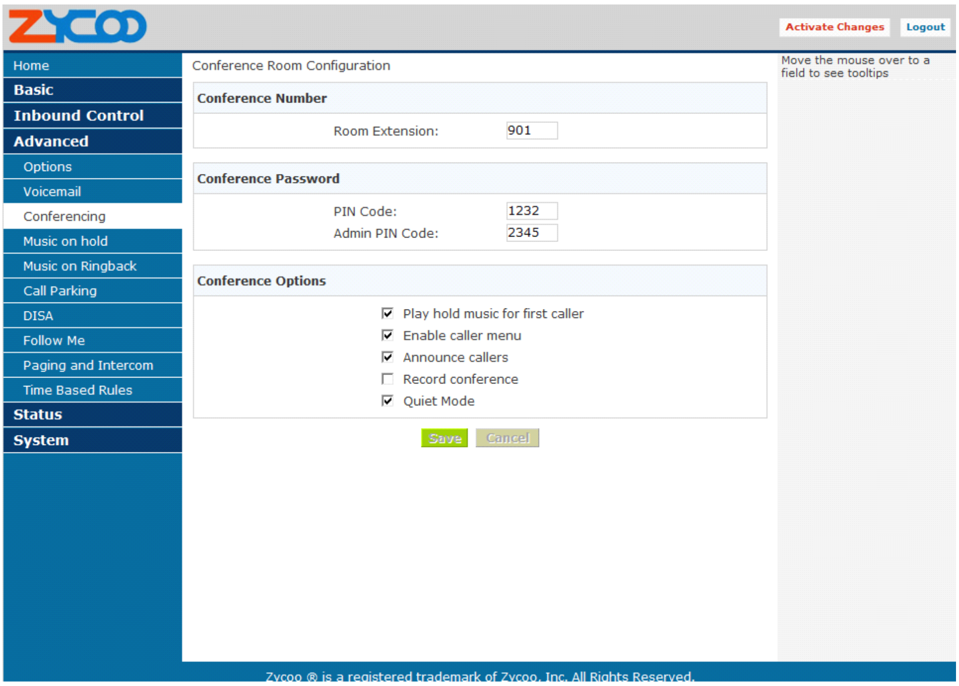
7.2.3 Email settings



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

7.3 Conferencing

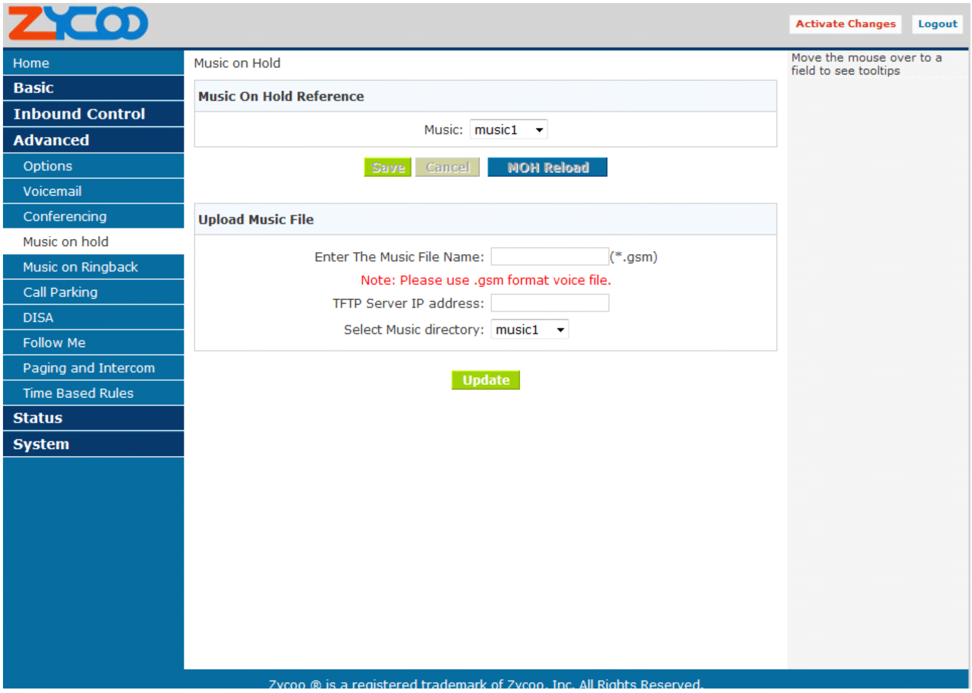
Every company reaches the point of needing more people on a call than it can effectively include through three-way calling. conference bridges allow you to include more people as well as project a professional image.



The configuration of the conference room and standard features is very straightforward.

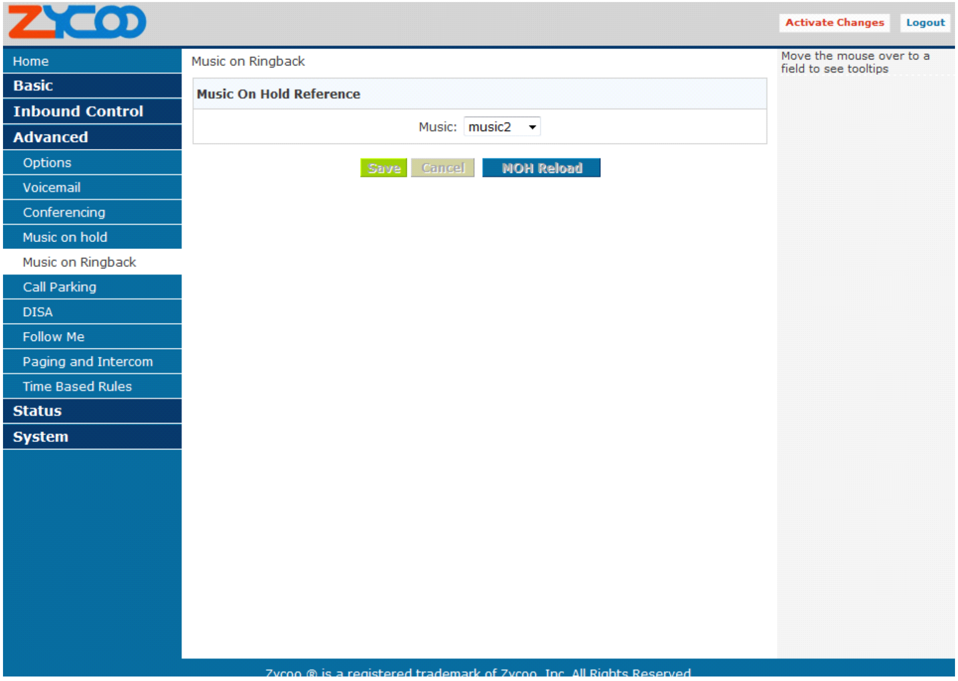
The conference room use default extension 900 , but you can always change it to any extension number you want. After establishing the extension for the room, you need to specify the password settings for the conference. Assign the PIN Code used by participants to enter the conference as well as the Administrator 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.

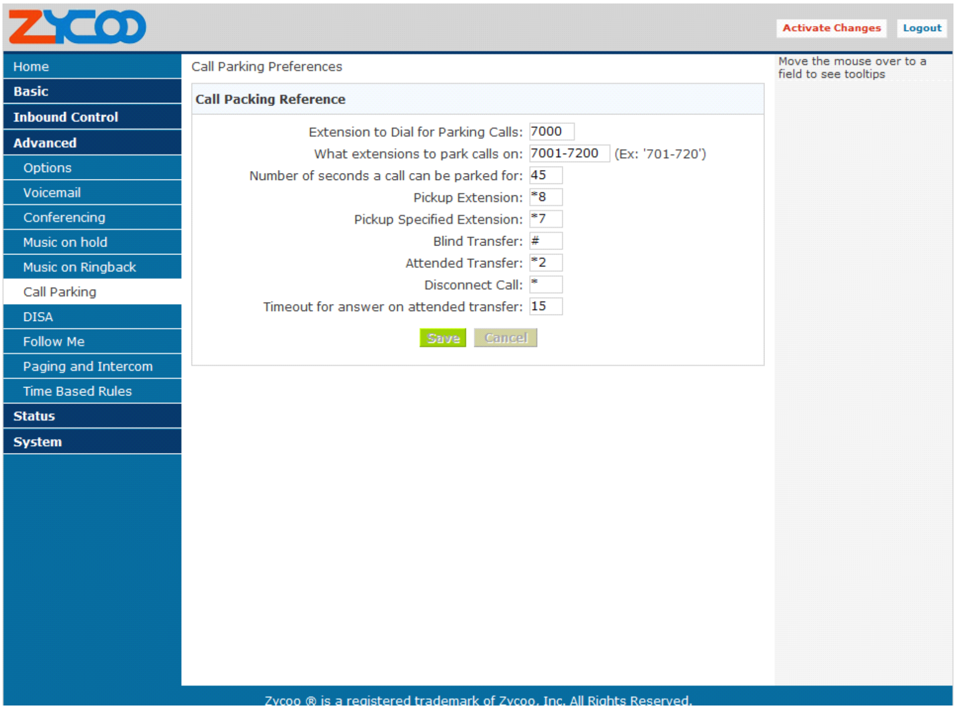
7.5 Music On Ringback



- Music Select a music for Music On Ringback

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

7.6 Call Parking

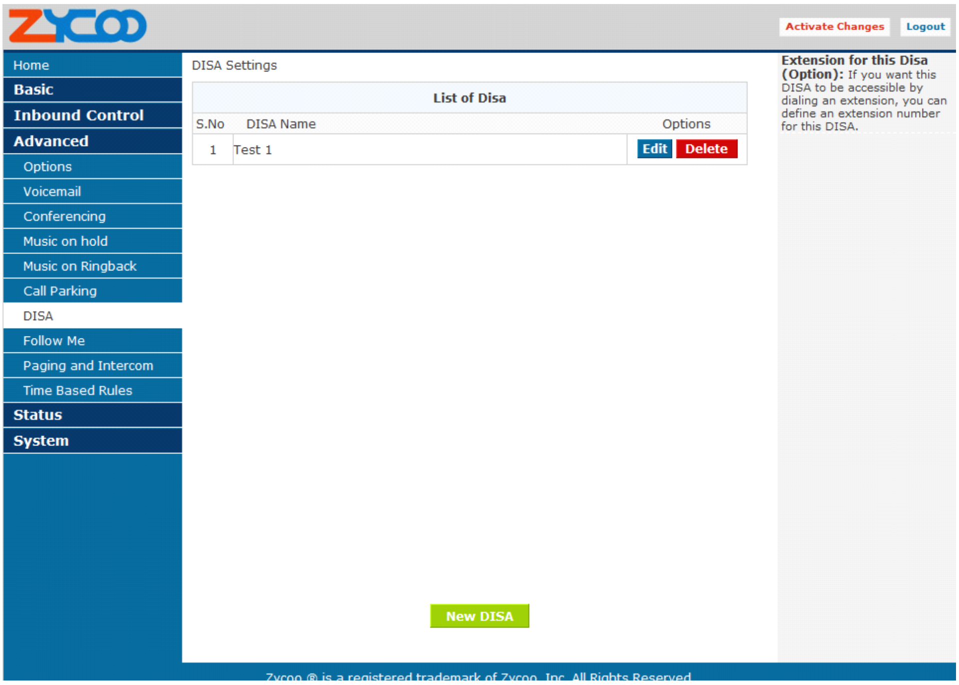


- Extension to Dial for Parking Calls: Set Call Parking number
- What extensions to park call on: Set the Call Parking get number (eg:701-720)
- Number of seconds a call can be parked for: Set the second call time
- Pickup Extension: Set Pickup Extension
- Pickup Specified Extension Set Pickup Specified Extension
- Blind Transfer allows unattended or blind transfers. It works like this:

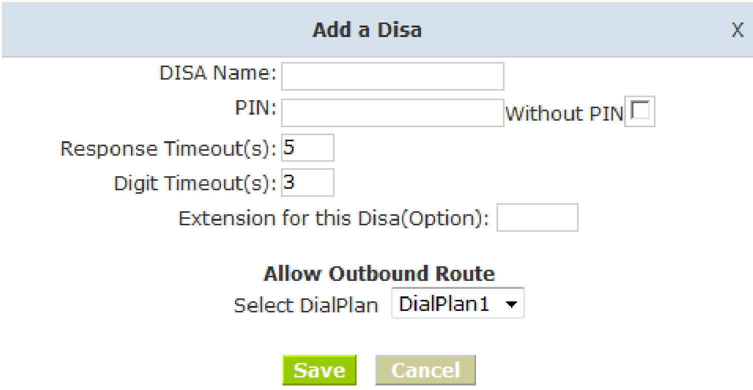
While on a conversation with another party, you dial the blindxfer sequence. the system says "Transfer" then gives you a dial tone, while putting the other party on hold. You dial the transferee number and the caller is put through to that number immediately. Your line drops. The caller ID displayed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



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

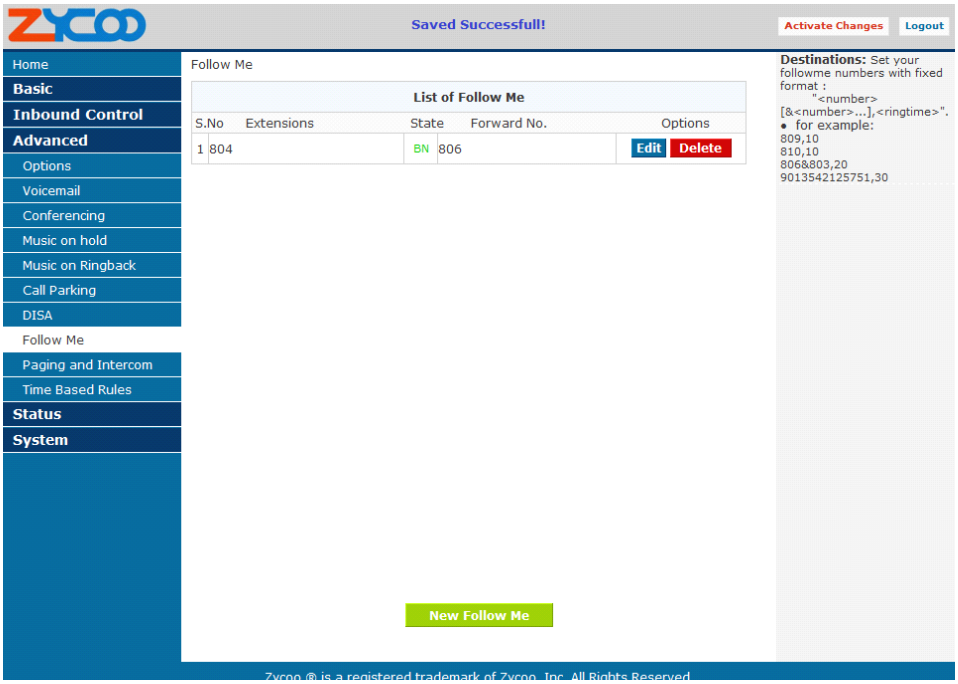


- DISA Name — Set a name for DISA
- PIN — Set a password for DISA
- Response Timeout(s) — Set effective time for inputing a password
- Digit Timeout(s) — After you input the right password, the interval between digits that you need dial.

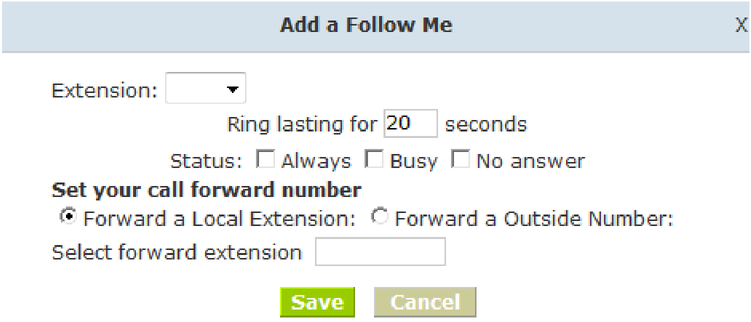
Extension for this DISA(Option) — Set a number connect DISA

Select DialPlan — Select your DialPlan for calling out

7.8 Follow Me

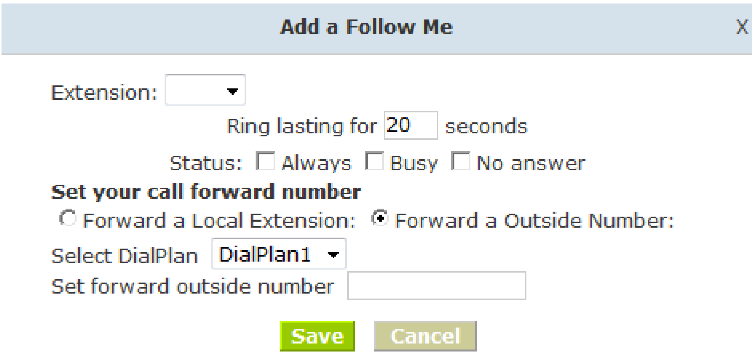


- List of Follow Me — Call Forward extensions are listed in the table.
- New Follow Me — Create a new Call Forward



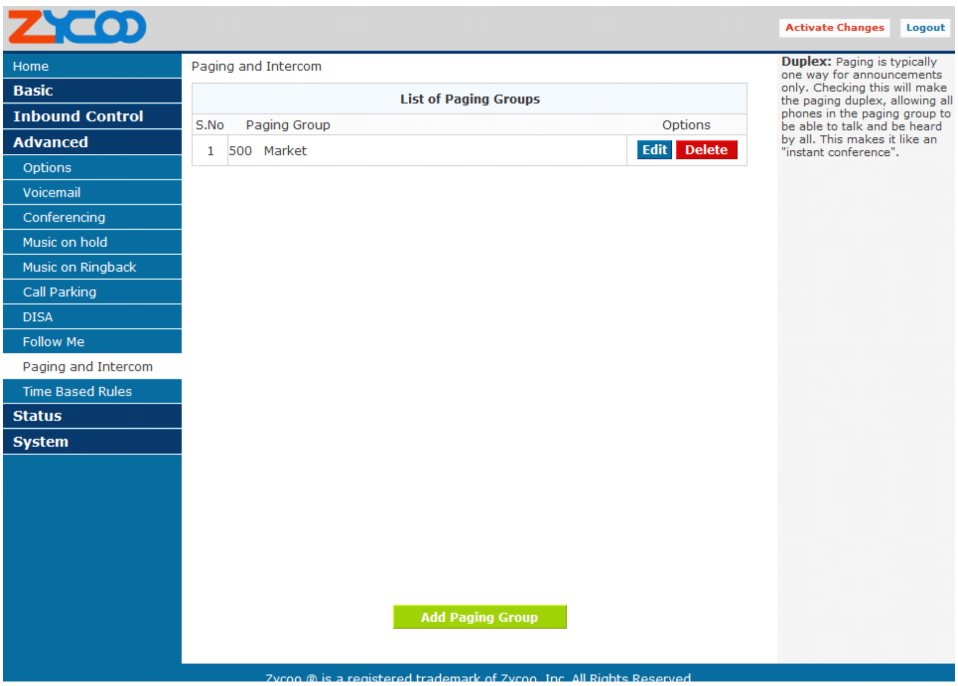
- Extension Select a need to call forward extension
- Ring Time Set the extension ring time
- State Set state of the extension.(Disable, Always, Busy, No answer)
- Select forward extension Select a call forward to extension

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

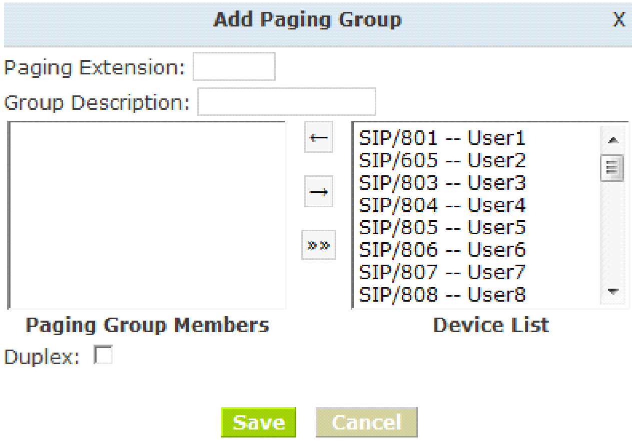


- Select DialPlan Select a Call forward to outside number using dialingrules
- Set forward outside number Input a Call forward to outside number. (Notice: This number must be consistent with the corresponding DialPlan)

7.9 Paging and Intercom

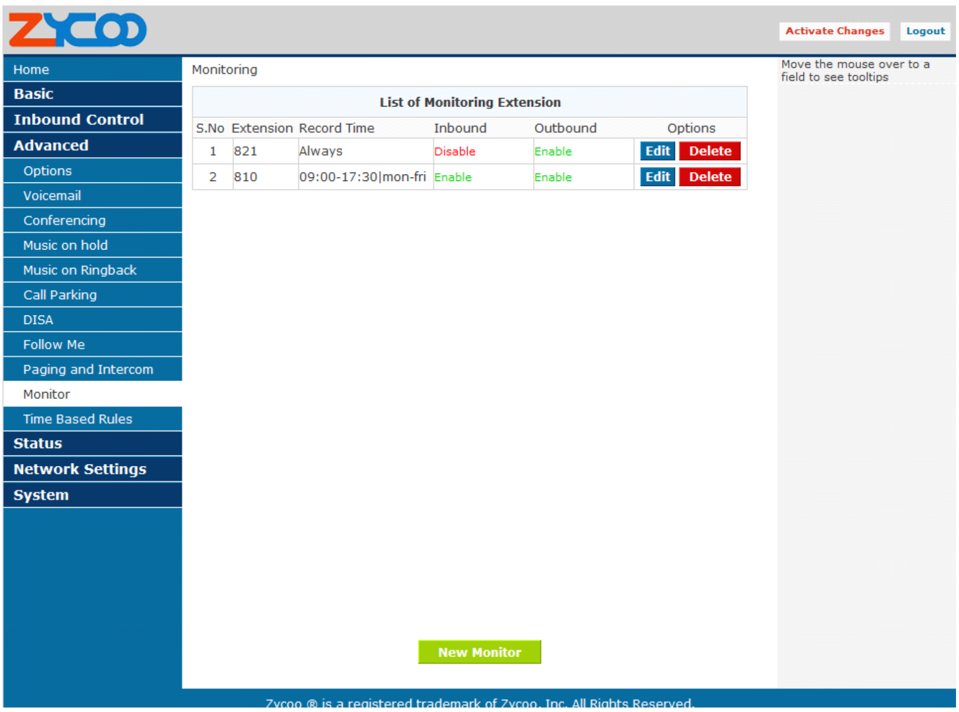


- List of Paging Groups Call Forward extensions are listed in the table.
- Add Paging Group Create a new Call Forward

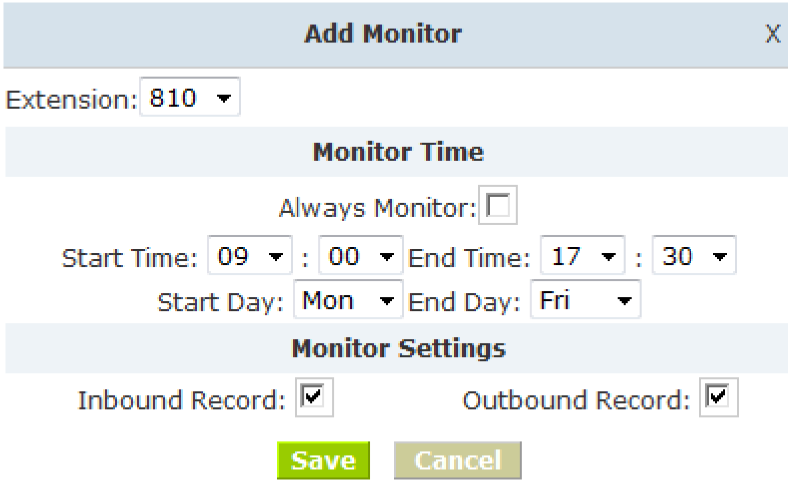


- Paging Extension Set a extension for the Paging Group.
- Group Description Provide a descriptive title for this Page Group.
- Paging Group Members Selected Device(s) in this Page.
- Device List Select Device(s) to Page.
- Duplex Paging is typically one way for announcements only. Checking this will 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

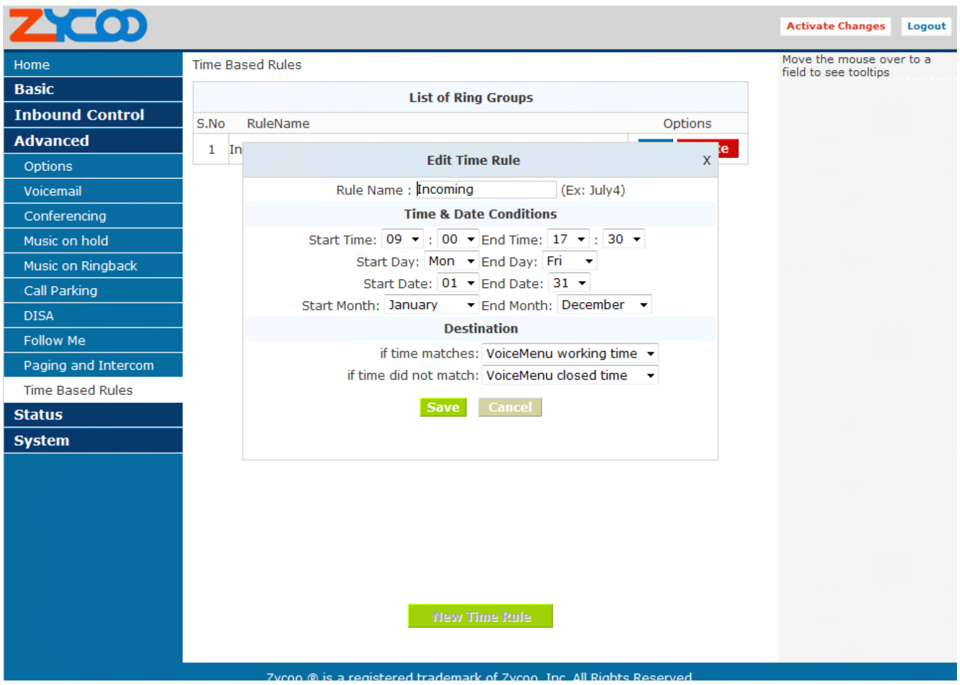


- List of Monitoring Extension Monitoring extensions are listed in the table.
- Add Monitor Create a new Monitor



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

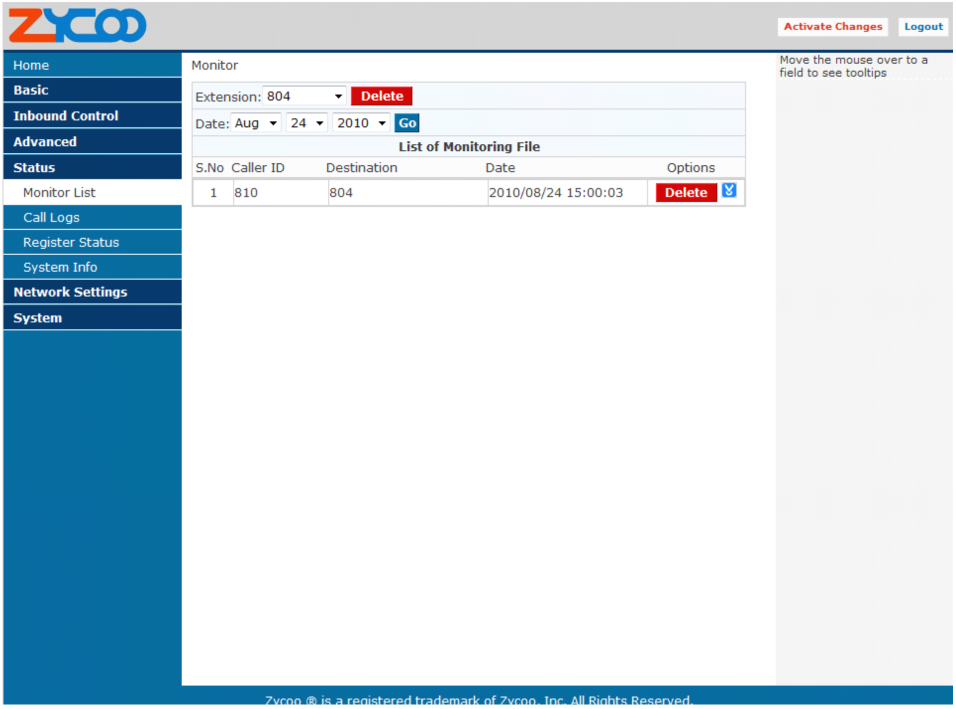
7.11 Time Based Rules



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

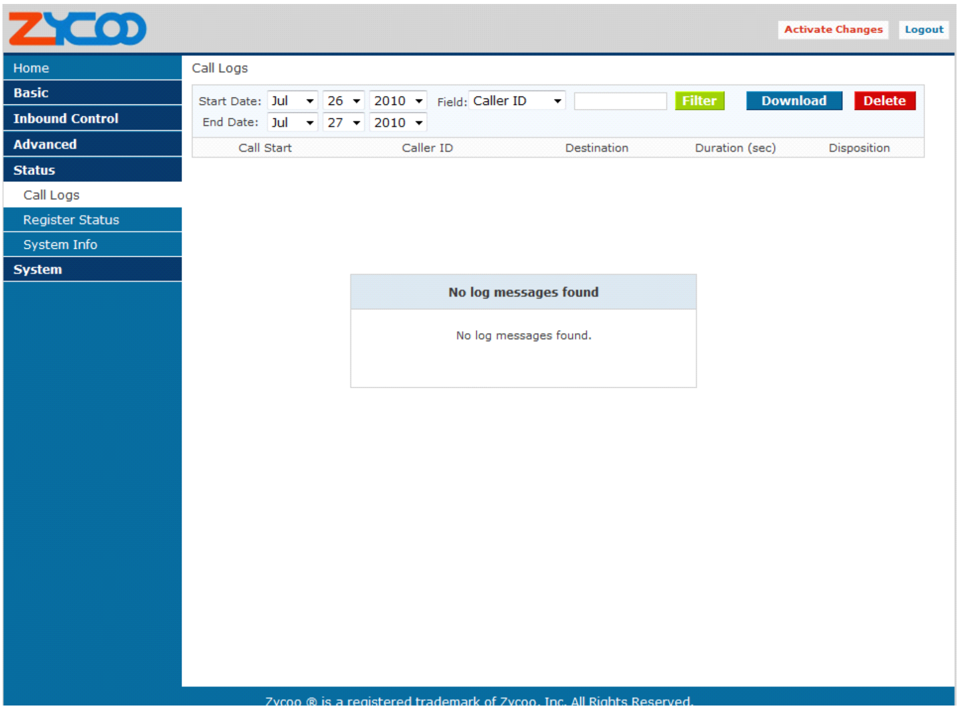
Chapter8 Status Display

8.1 Monitor List



This web page will display Monitor info for each extension

8.2 Call Logs

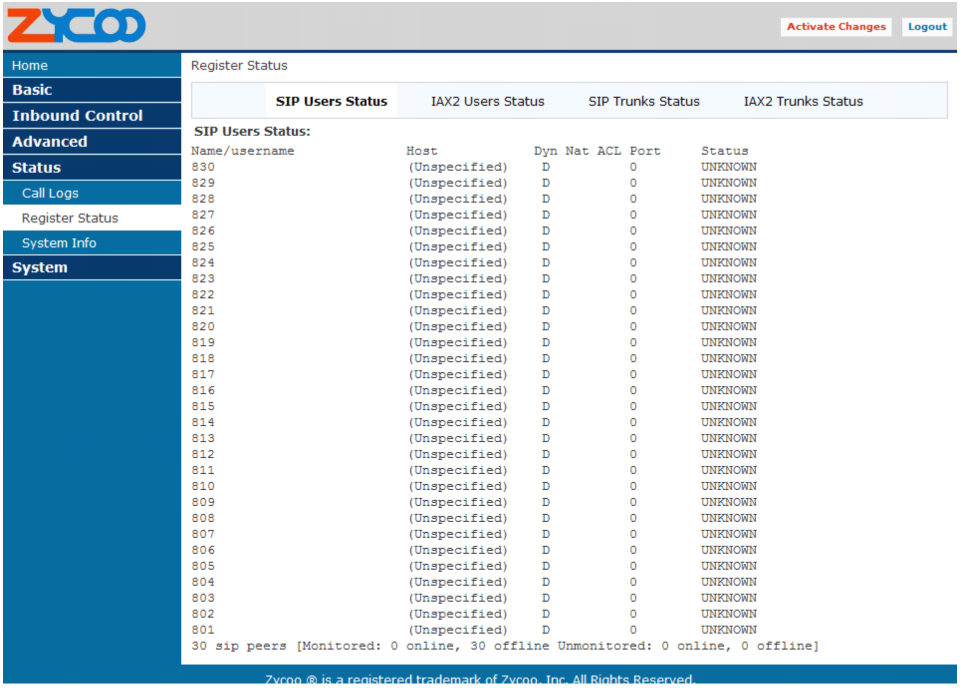


This web page will display call logs

- Download download the call logs file
- Delete delete the call logs file

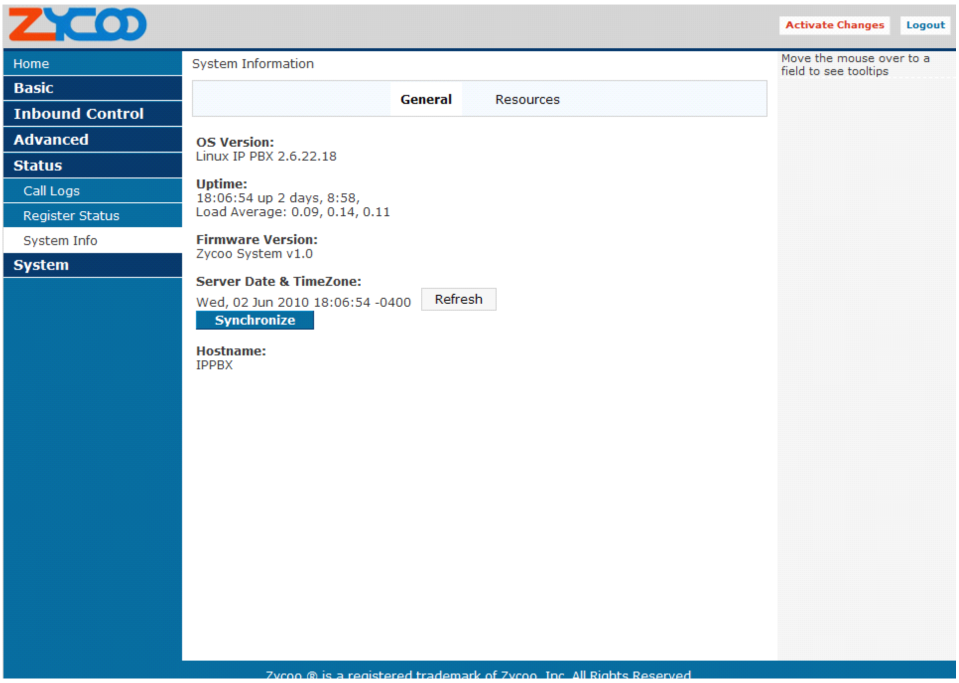
8.3 Register Status

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



8.4 System Info

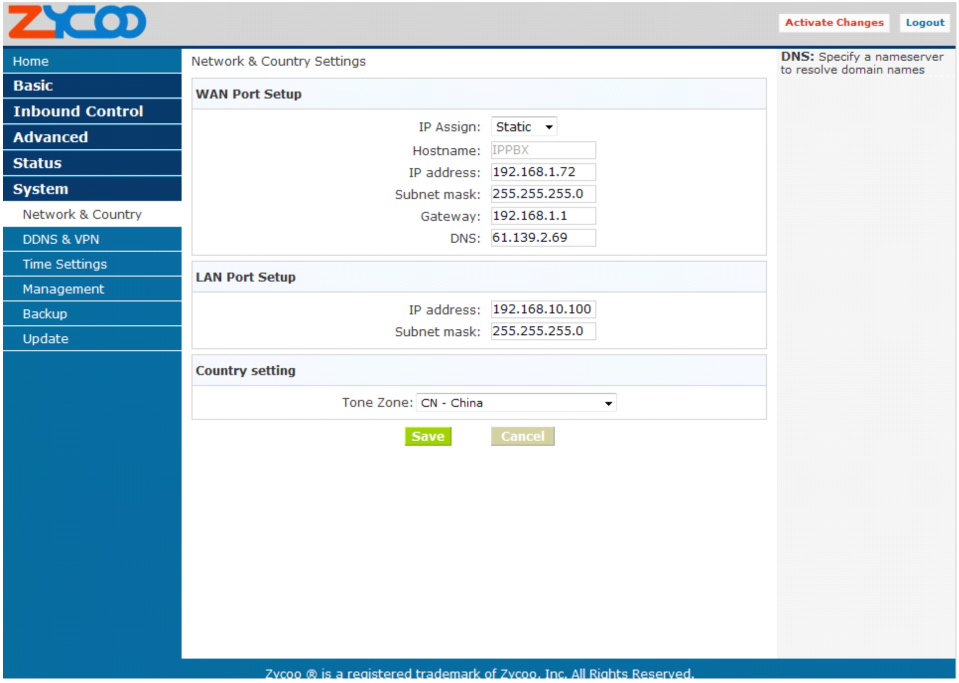
In this page it will display nonce system info



Chapter9 System Management

9.1 Network and Country

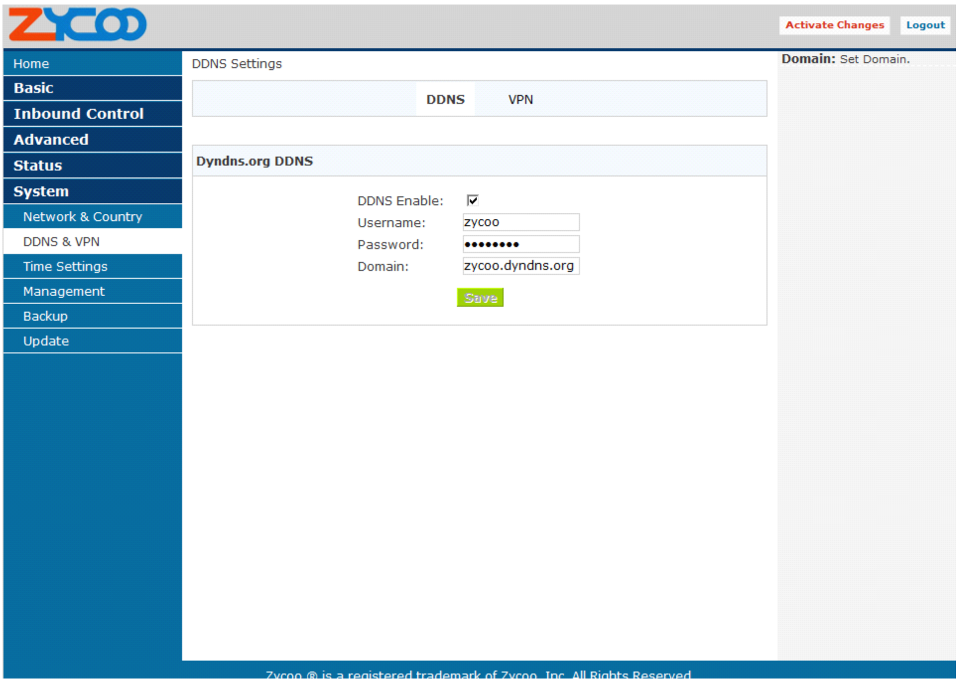
On this page you can set WAN, LAN interface information and the country of Tone Zone.



- IP Assign: you can select STATIC, DHCP and PPPoE three mode
- Tone Zone: Set your Country, and use the Country Tone

9.2 DDNS&VPN

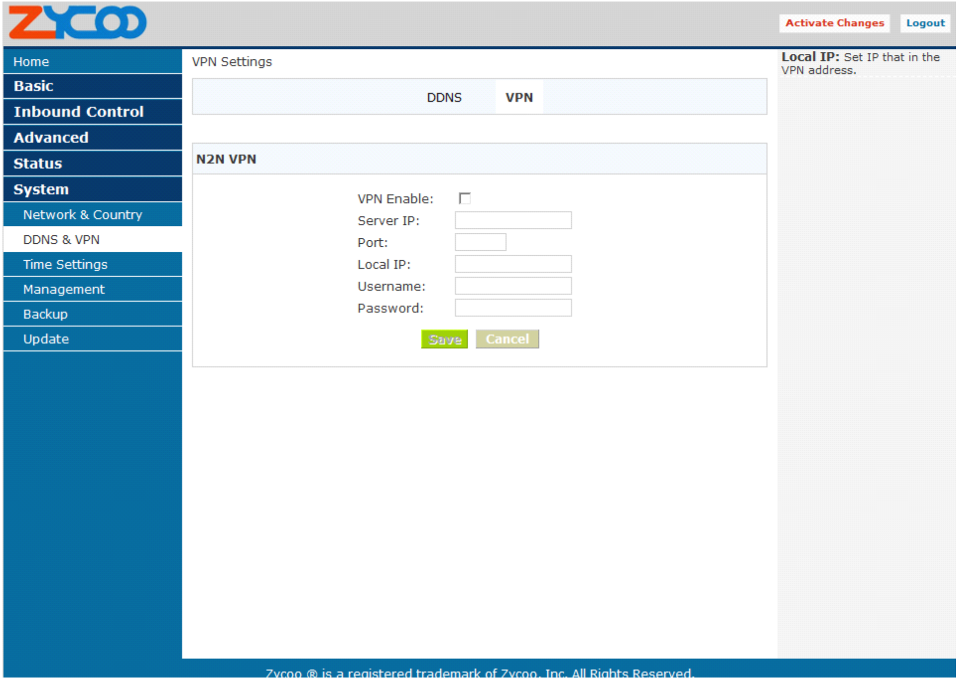
9.2.1 DDNS Settings



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

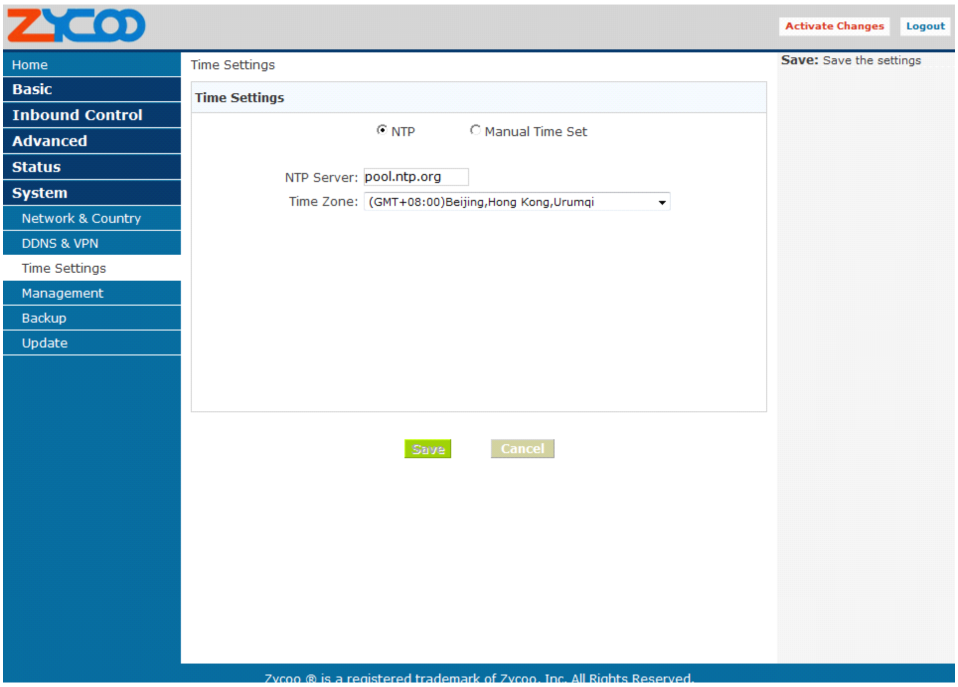
9.2.2 VPN Settings



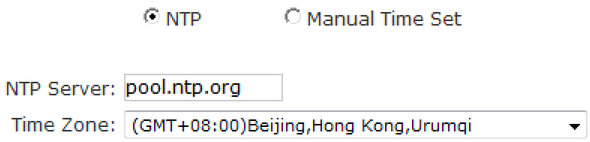
On this page, you can set VPN reference.

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

9.3Time Settings

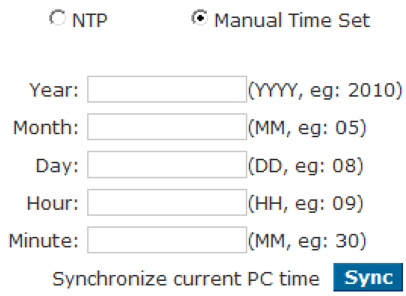


9.3.1 NTP Settings



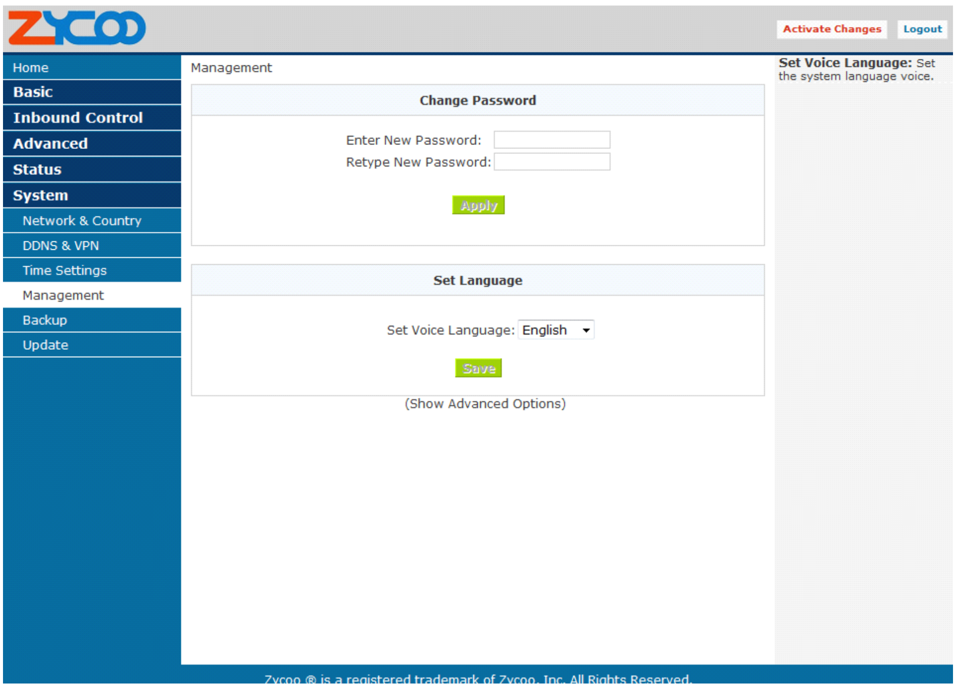
- NTP Server Specify the NTP server that you wish to use. You may type either the domain name or the IP address of the server, and it may be either remote or local. The default server is pool.ntp.org. Be aware that the PBX needs to be able to access a NTP server in order to function properly.
- Time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

9.3.2 Manual Time Settings



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

9.4 Management

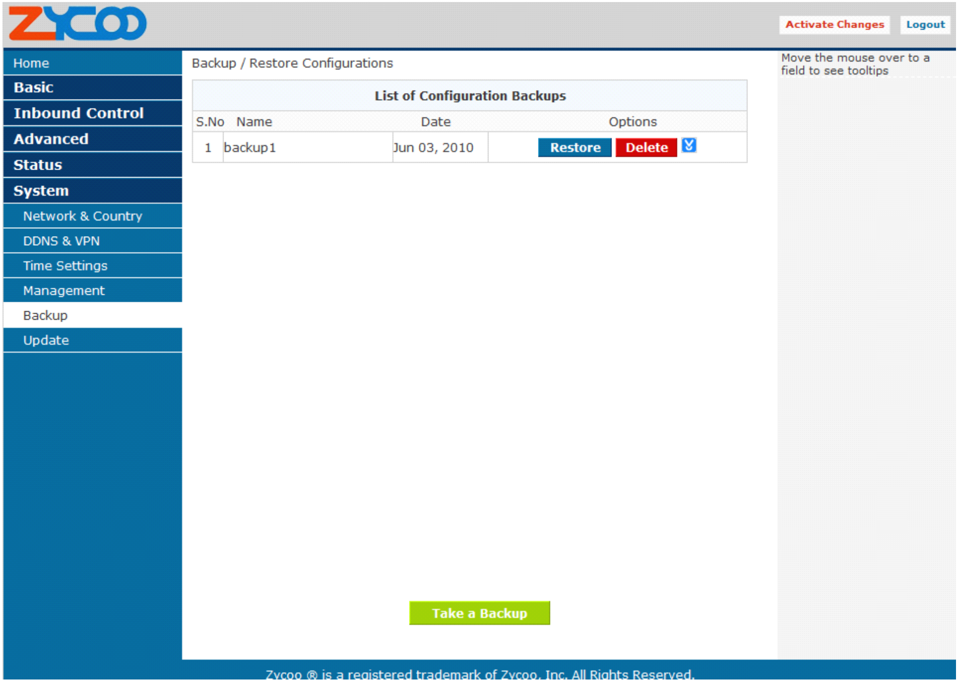


- Change Password On this page, you can change the administrator password (Default password: admin)
- Set Language Set the system language voice

And you can also set the advanced options about SIP and Zap protocol in the "Show

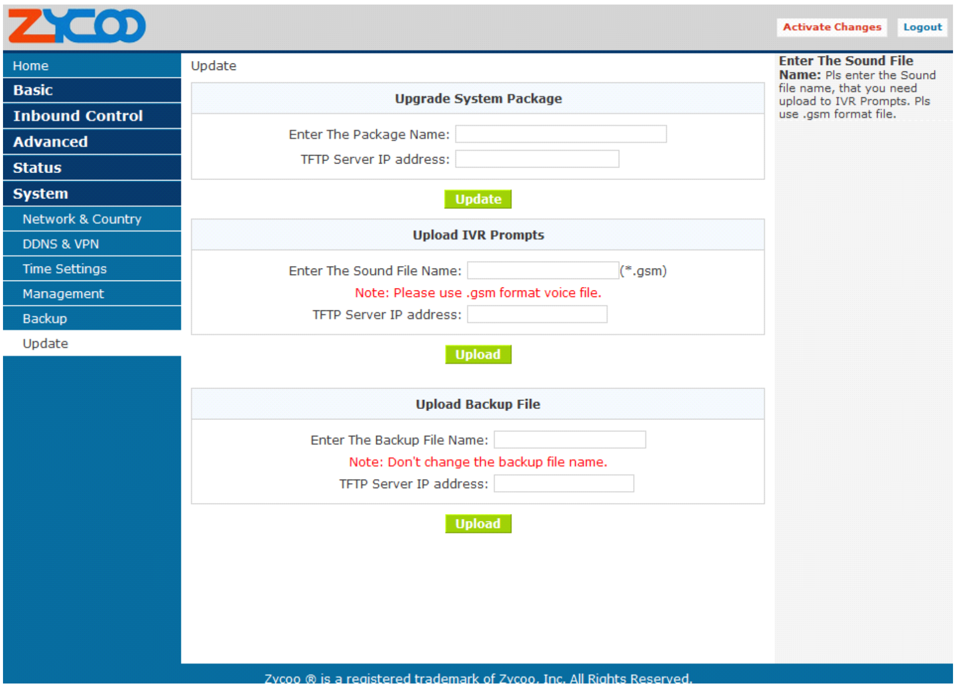
Advanced Options" list, that is useful when you set connect two ippbx in different network.

9.5 Backup



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

9.6 Upgrade

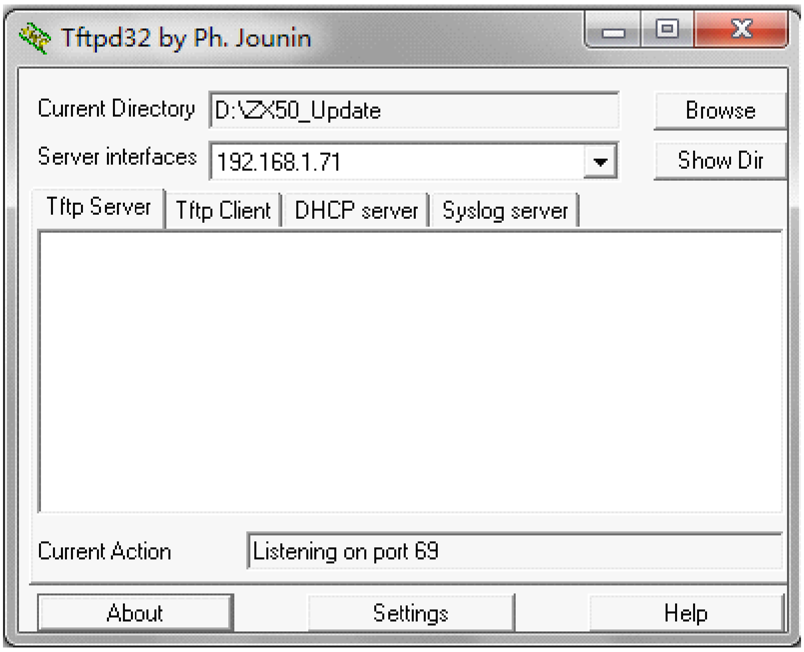


In this page you can upgrade system package

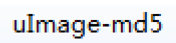
- Enter The Package Name　　Set system package name
- TFTP Server IP address　　Set TFTP server IP

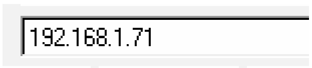
Unzip the file you download, you will get a TFTP server and an upgrading packet.

Run the TFTP server, you will see below:



Enter the configuration page, then upgrading page;

Enter The Package Name, hereby it's 

Enter TFTP Server IP address, hereby it's 

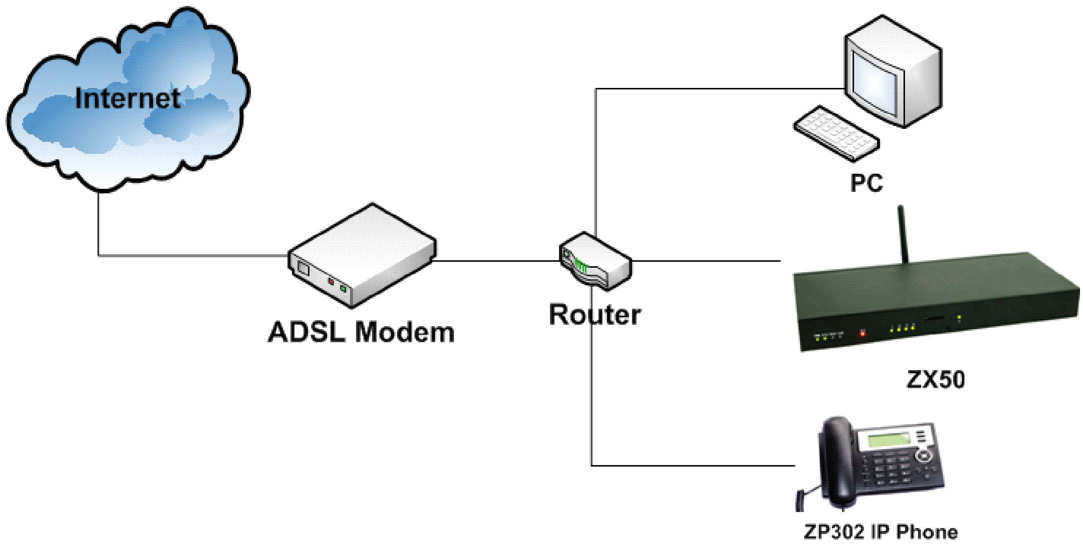
After done, click Update to update, then the system will reboot automatically.
(Note: the upgrading will set your system as default, please make backup before you do it.)

Chapter10 Operating Instruction

10.1 How to link the ZX50 IP PBX to the interwork

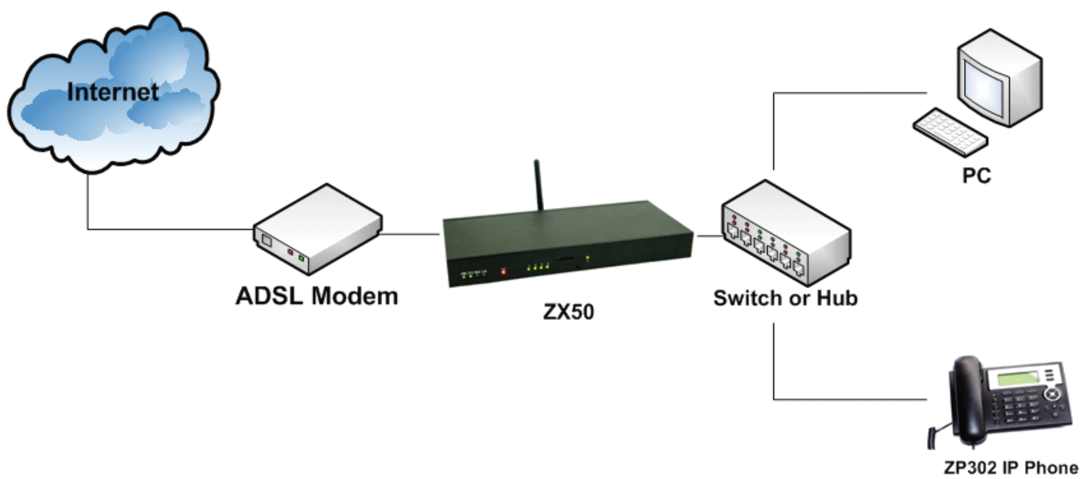
10.1.1 IP PBX behind the Router

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..



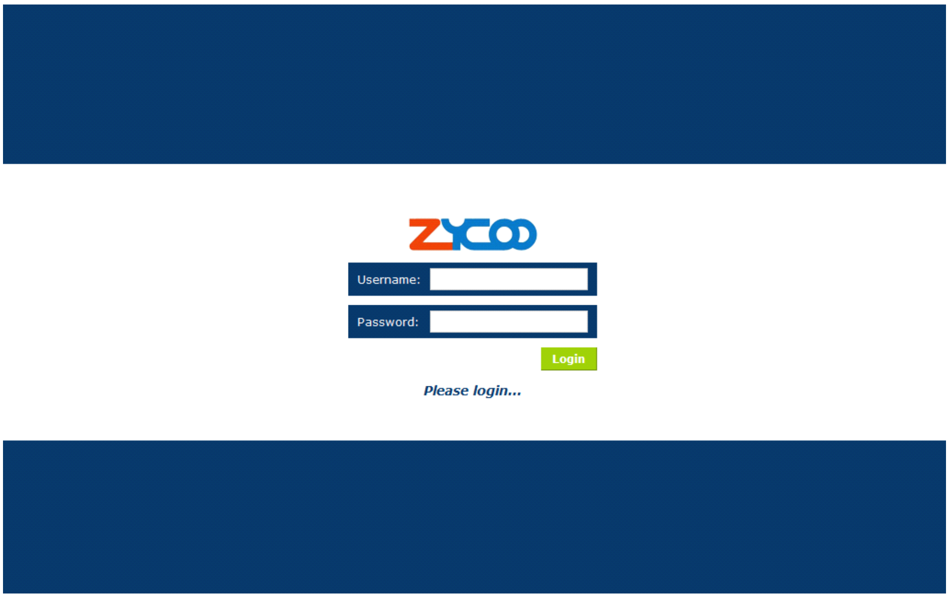
10.1.2 IP PBX behind the Modem

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)

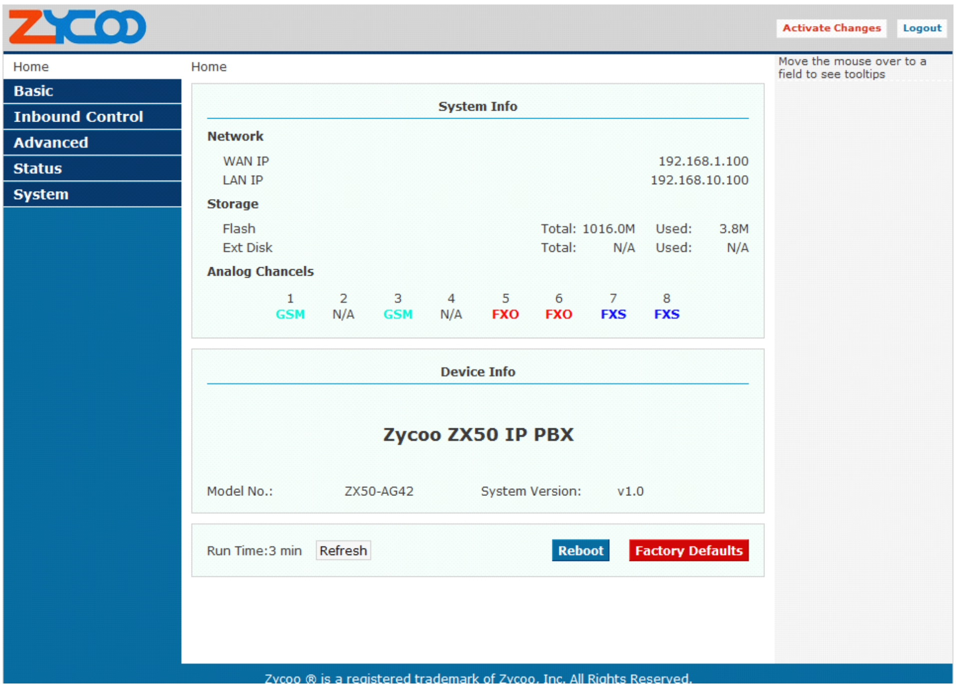


10.2 How to log in the IP PBX system

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:



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.

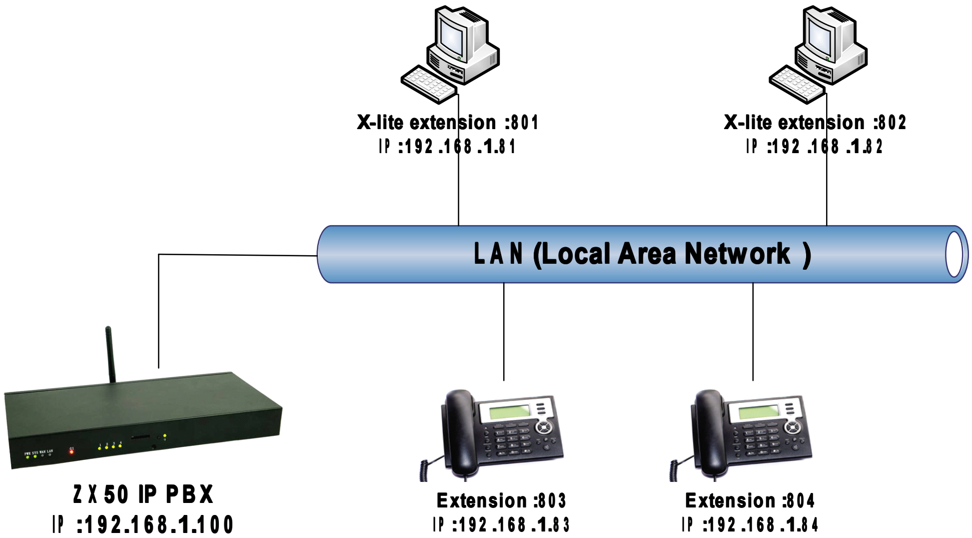
The right column of the user interface contains Tooltips. This area provides brief description for any options of the GUI

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

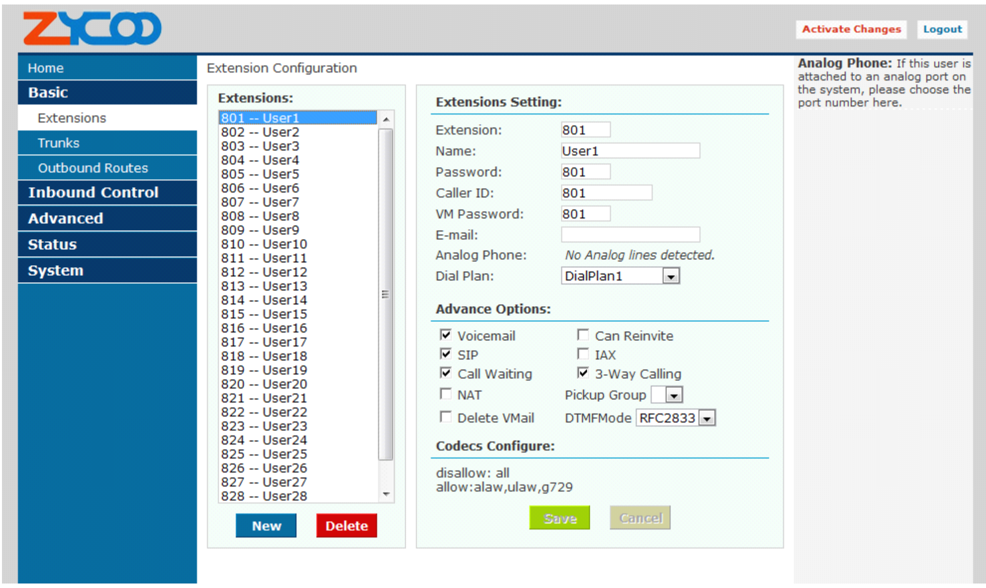
- Logout: To log out the zycoo GUI.
- Reboot: Reboot the ZX50 system
- Factory Defaults: Restore all settings to factory default.
- Activate change: Made the change active for the current configuration after you make a configuration change on some page.

10.3 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 ZX50,but setting is the same in other ZX50 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.

10.4 How to make an outbound call

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

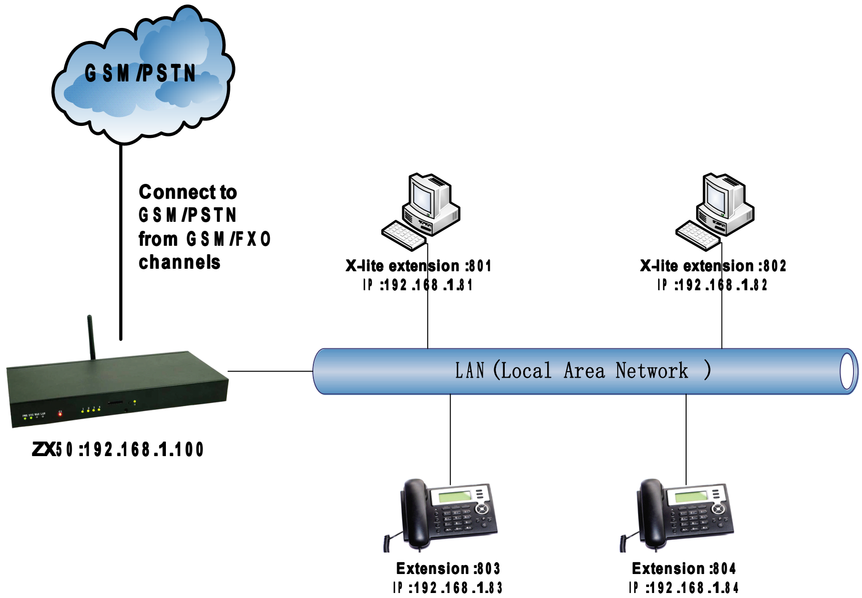
Analog Ports: GSM/FXO ports of ZX50, connect to GSM/PSTN lines.

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

I am using ZX50-G4, the port1-4 are configured as GSM ports. When a port is configured as GSM/FXO port, the corresponding LED shows RED. When a port is configured as FXS port, the corresponding LED shows GREEN.

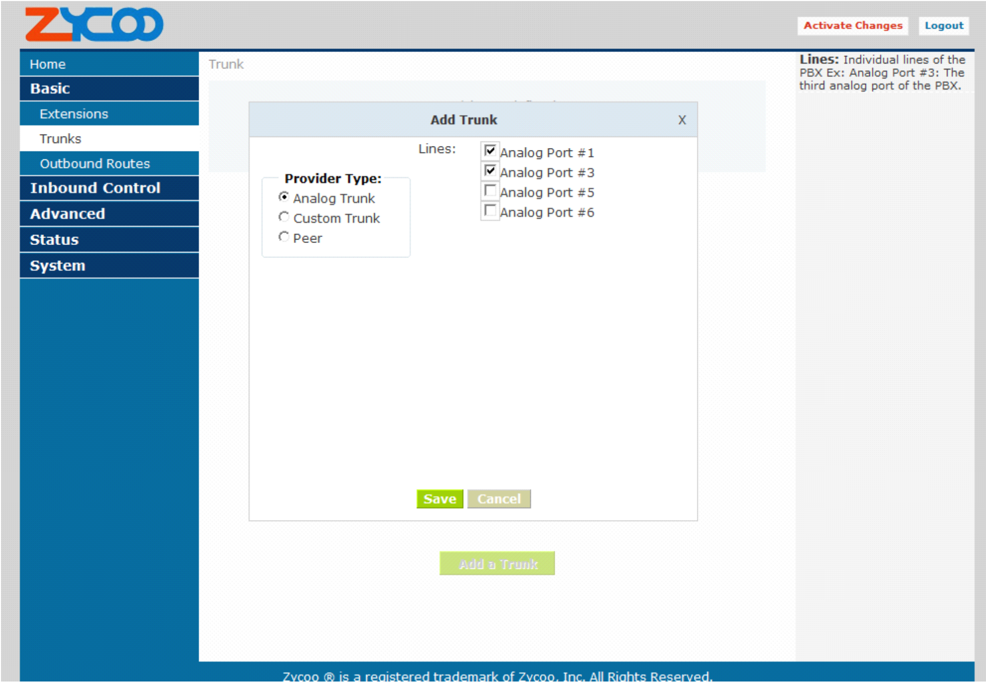
10.4.1 Make call via GSM trunk

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



<u>Add Analog Trunk</u>

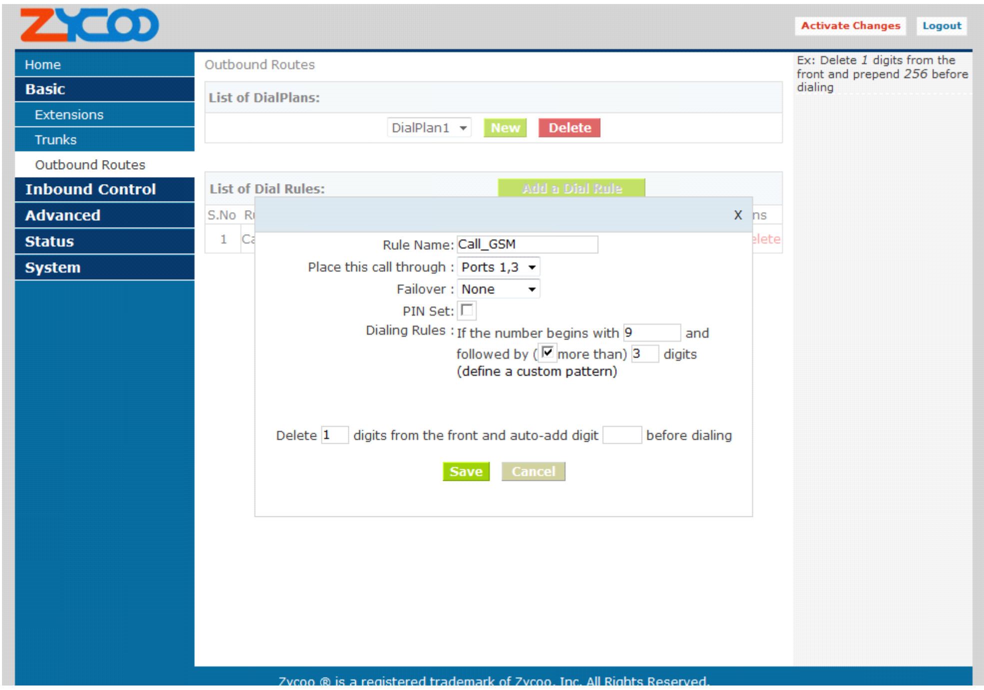
Trunks -> Add a Trunk:



Add Outbound Routers

In Outbound Routers -> add a Dial rule as below

Dial Rules

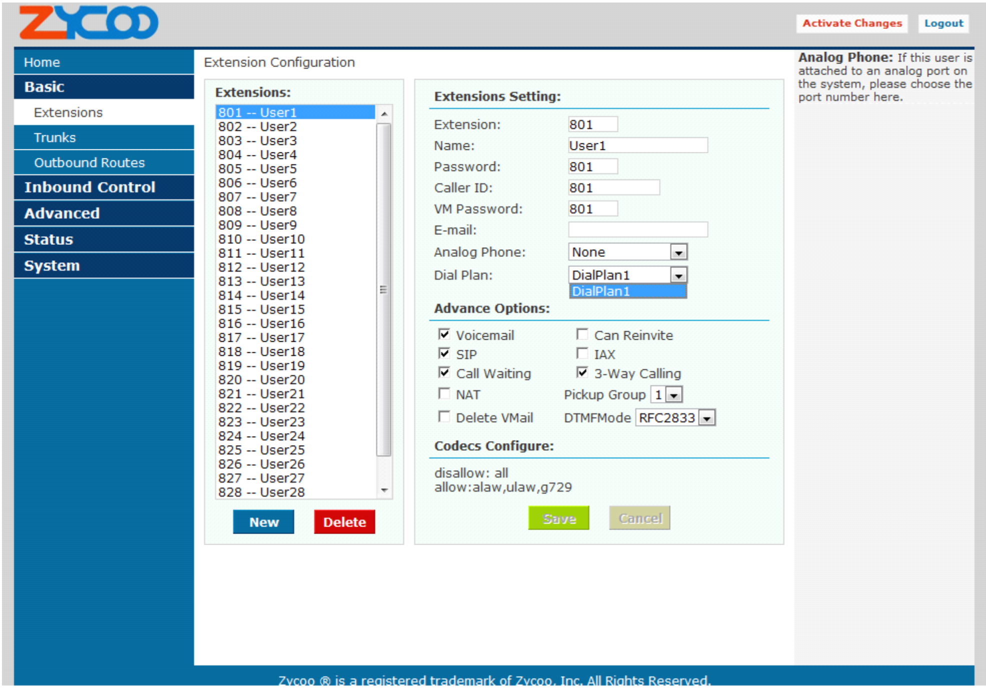


We have now added a Dial rule “OUT_GSM” in the “DialPlan1”.

As we can see from the dialing rule of “OUT_GSM”, all numbers start with 9 will be cut the first digit (‘9’) and sent to GSM (port1 or port3).

Choose Dial Plan for extensions:

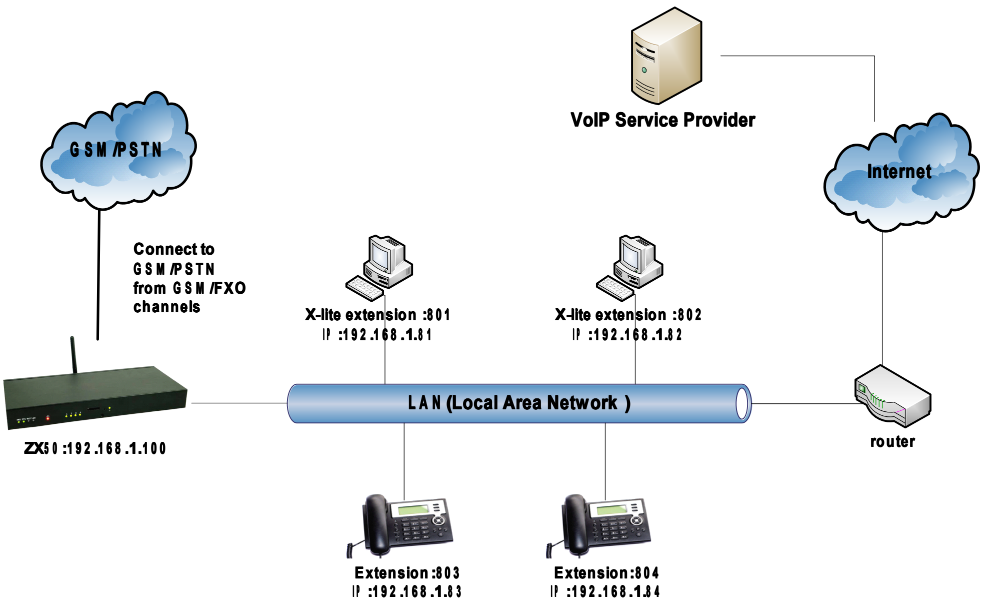
On the User page, edit the extensions to choose DialPlan1.



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

10.4.2 Make call via VoIP trunk

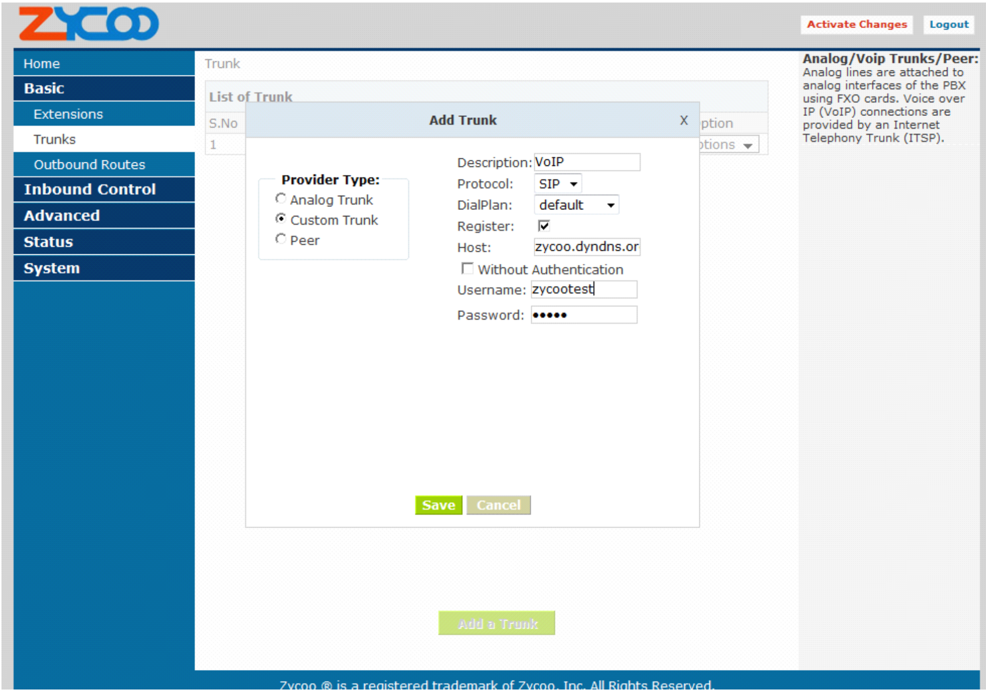
Via the voip trunking we can dial call via the voip service to reduce our cost when making international calls.



Add VoIP service provider

Trunk -> Add a Trunk:

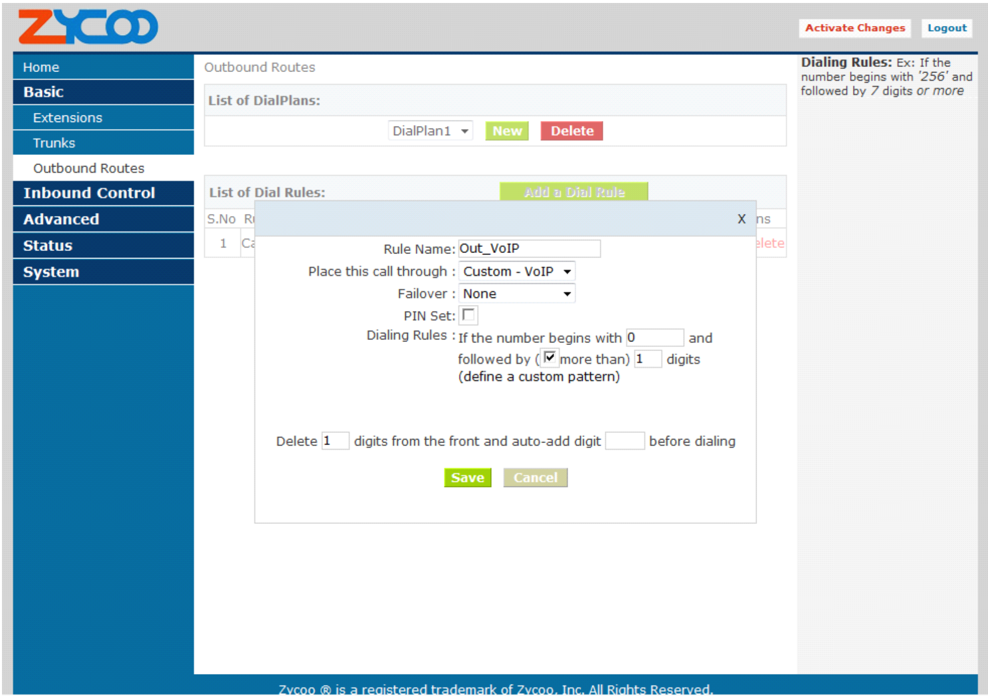
Add a Custom Trunk.



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".

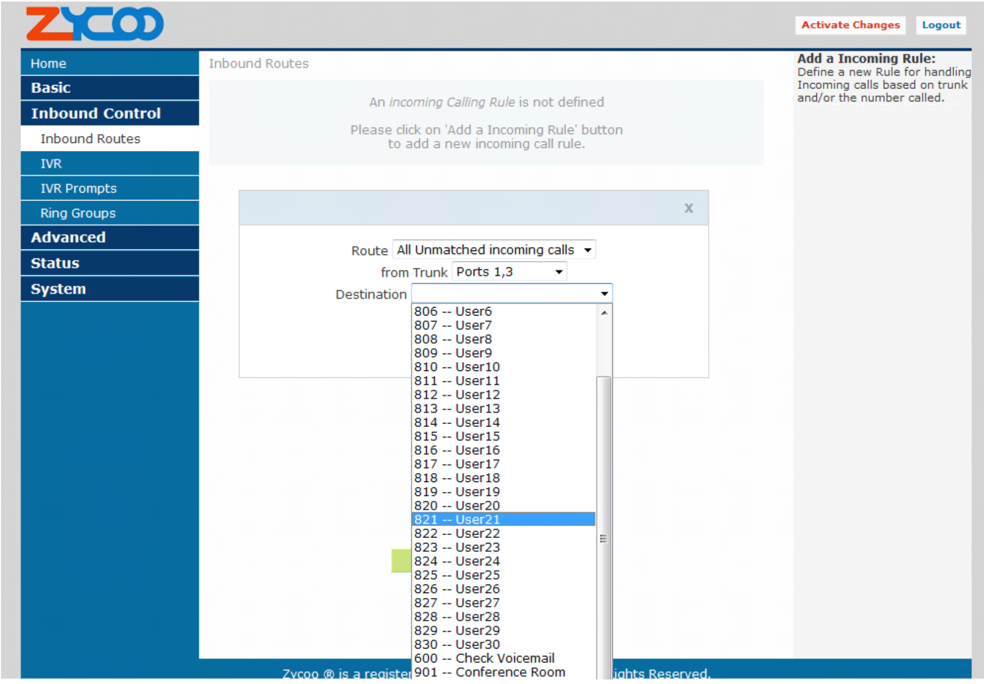
As we can see from the "Out_VoIP" dialing rule, all numbers start with 0 will be cut the first one digits ('0') and sent to my sip service provider.

The Out_GSM 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 GSM, and call starts with 0 will be route to VoIP.

10.5 How to make an incoming call

Add an Incoming call.



Select Route “All Unmatched incoming calls”

From provider “Port 1, 3”

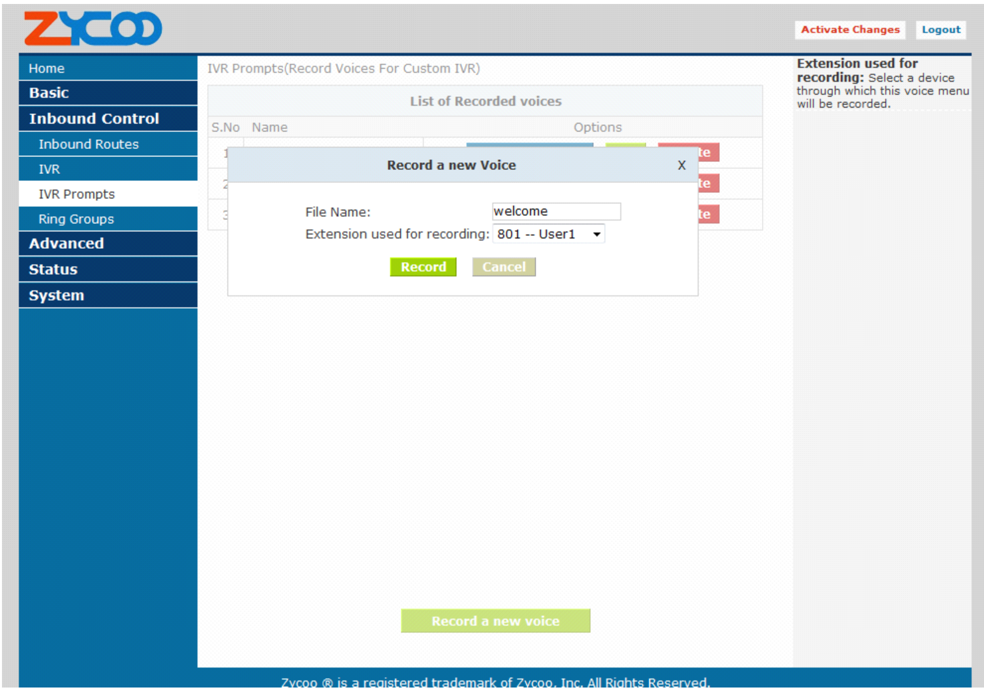
To extension “801 – User1” (here, you can select a extension, a IVR or others)

Then, if there is incoming call from Port1 or port3 channel, the extension 801 will ring.

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

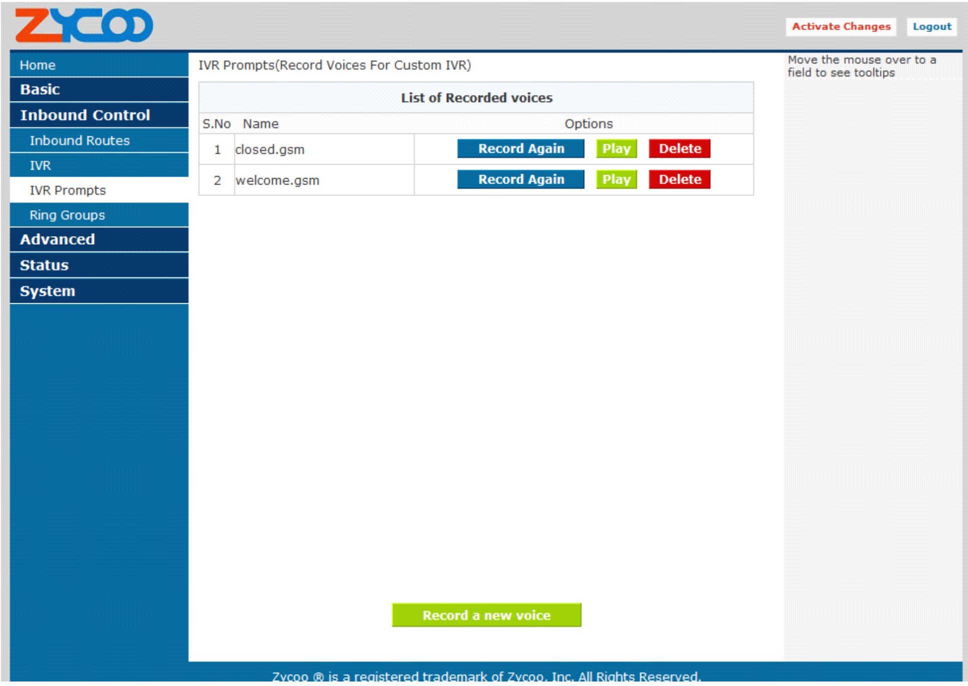
<u>Add record a custom voice</u>

Record -> Record a new voice



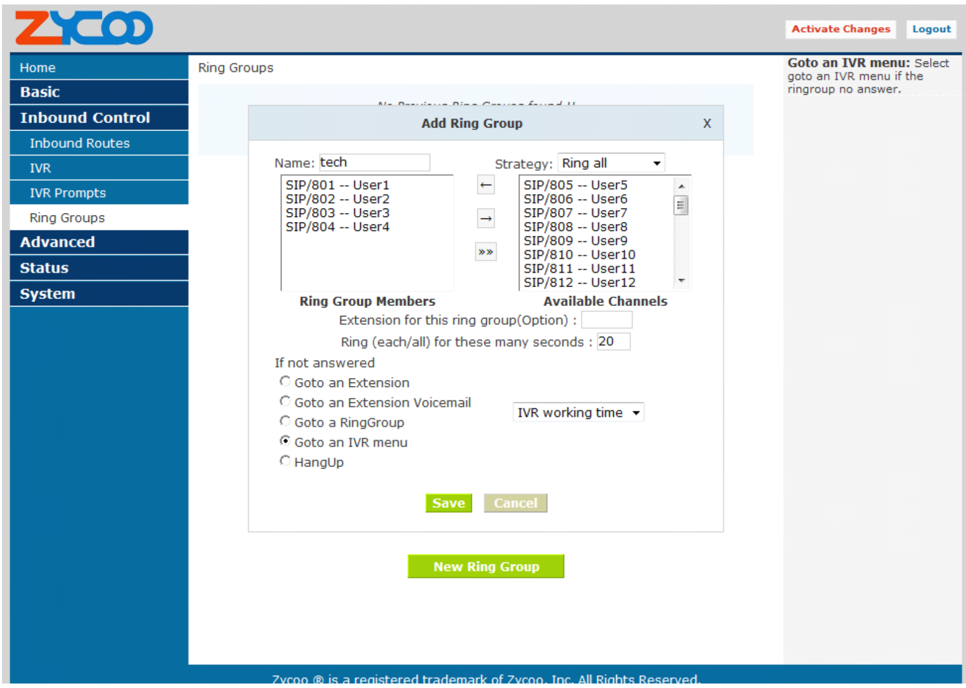
Set the record name is “Welcome”

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:

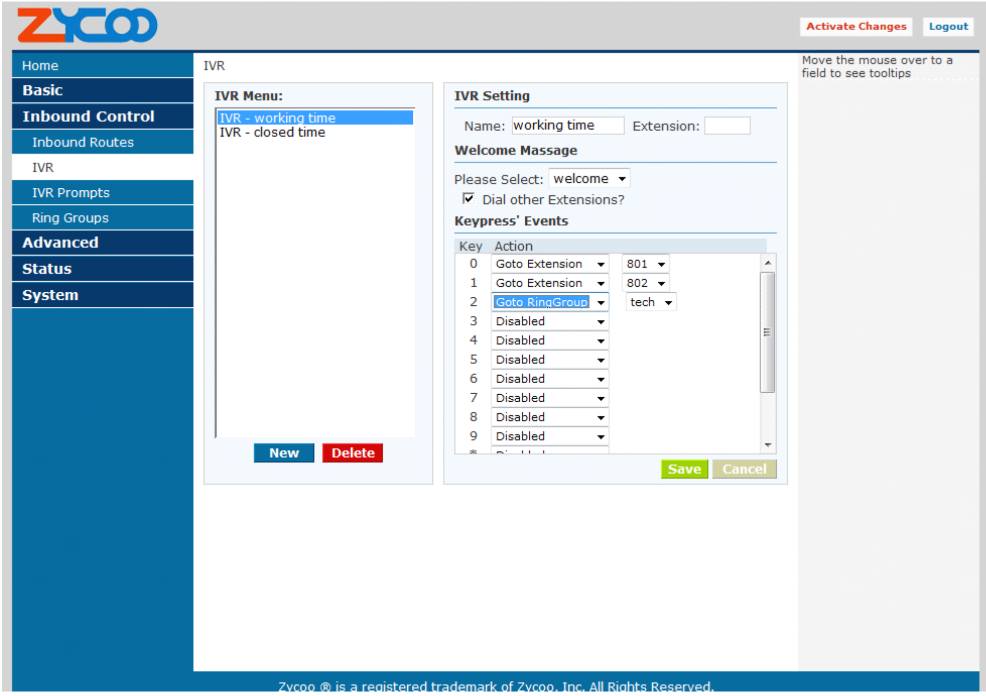
Name the ring group “tech”

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 massage is "Welcome"

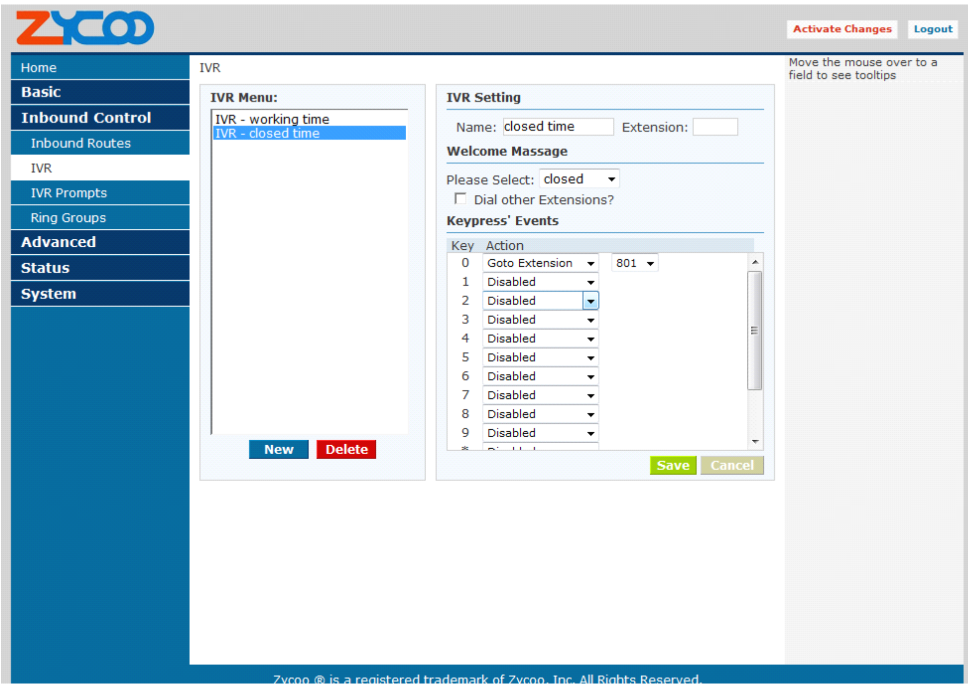
Set keypress' Events

Dial "0" go to extension 805

Dial "1" go to extension 806

Dial "2" go to ringgroup tech

Click Save button

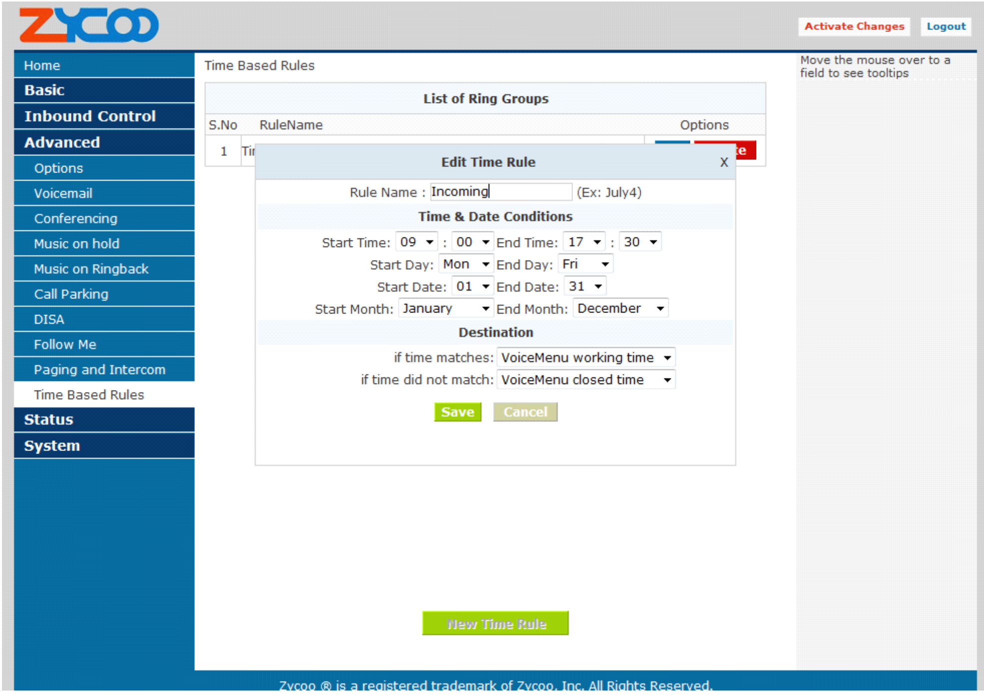


Then set IVR-closed time

Set welcome massage is “Closing”

Add a Time Rule

Time Based Rules -> New Time Rule



Set a Rule Name, eg: incoming

Set the Time & Date Conditions

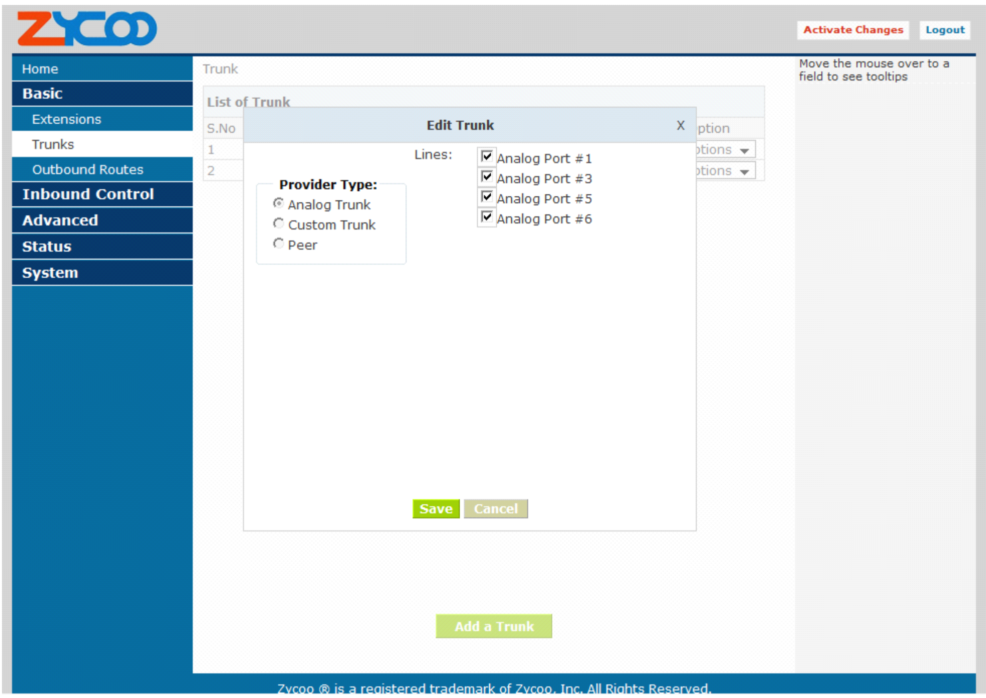
“If time matches” --- go to “working time”

“If time not match” --- go to “closed time”

Click the save button, saved the configuration

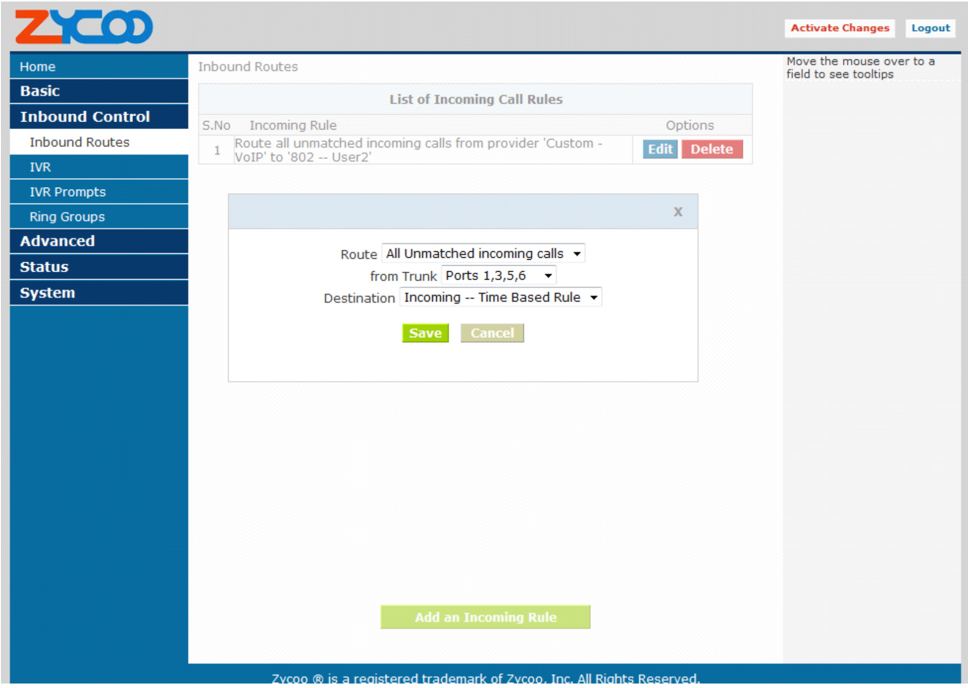
Add a Trunk

Trunks -> add a Trunk



Add an incoming router

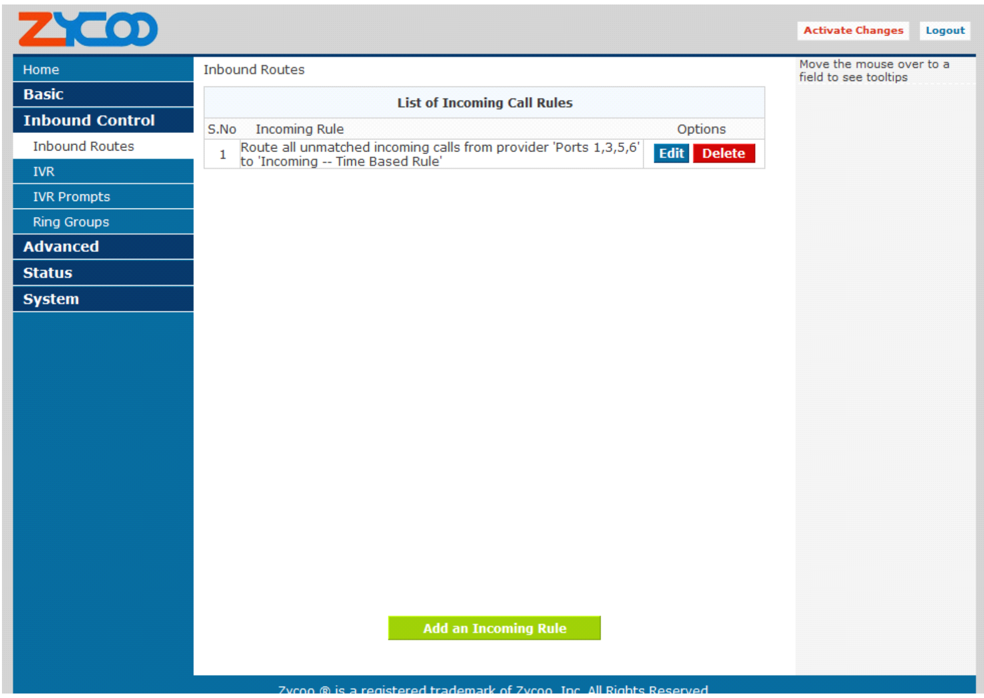
Inbound routers -> add an incoming rule



Select Route: All Unmatched incoming calls

From provider: Ports 3, 4

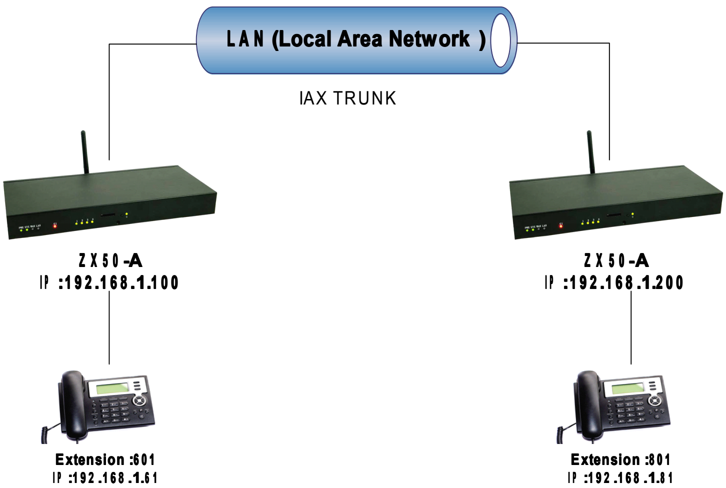
To extension: incoming—Time Based Rule



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

10.7 How to link two ZX50 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:



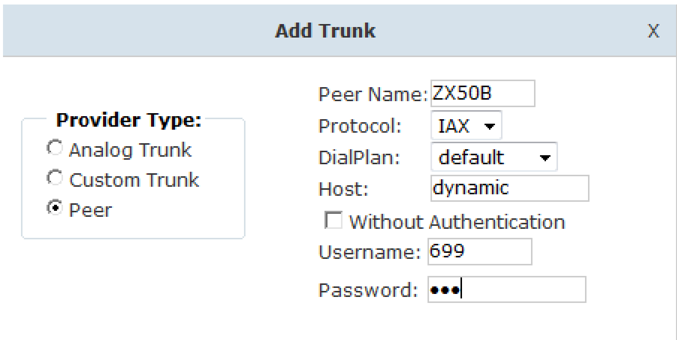
Register the ZX50-A as an peer in ZX50-B(via IAX2 trunk),so the extensions in ZX50-A can make calls to ZX50-B's extensions via this "special" trunk.

In above structure:

1. ZP302A registers toZX50-A as an extension 601.
2. ZP302B registers to ZX50-B as an extension 801.
3. All the extensions under ZX50-Aare in the format 6XX.
4. All the extensions under ZX50-B are in the format 8XX
5. Extensions under ZX50-A can make calls to extension under ZX50-B use format 8XX.
6. Extensions under ZX50-B can make calls to extension underZX50-A use format 6XX.

Step 1: *Set up a peer 699 in ZX50-A*

In the page Trunks→ Add a Trunk



Peer Name: ZX50B ;

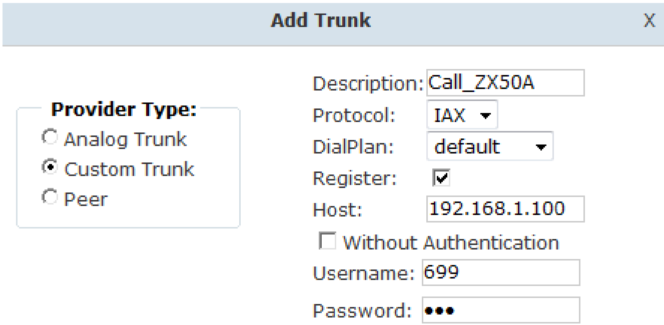
Peer Username: 699 Account of this Peer

Password: 699 IAX2 Log on password

Advance Options: Select IAX protocol

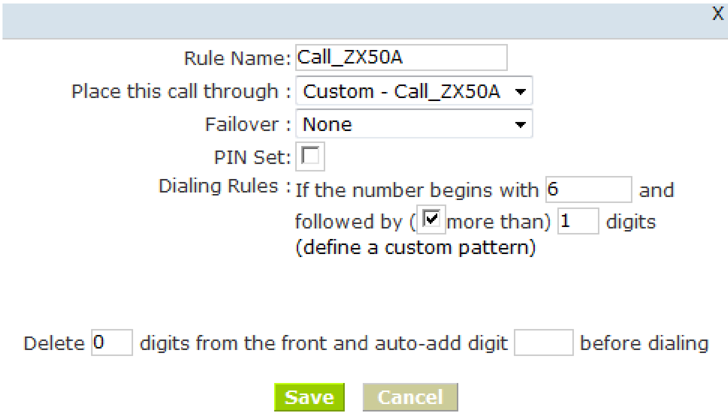
Step 2: *Set up an IAX trunk in ZX50-B to link to ZX50-A via this ZX50B Peer.*

In the page Trunks--> Add a Trunk



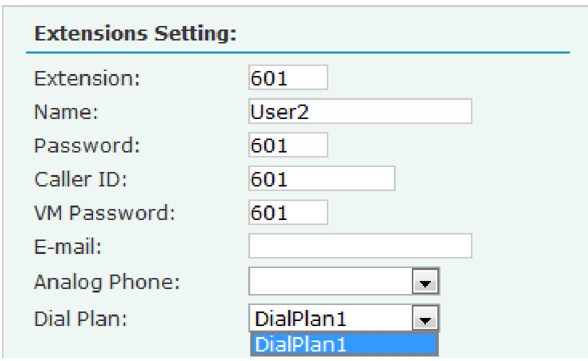
Step 3: *Set Dial Rule in ZX50-B, all calls start with 6 will be sent to ZX50-A.*

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



Step 4: *Set the user 601 and Dial Plan in ZX50-A.*

In the page: Extensions → Dial Plan



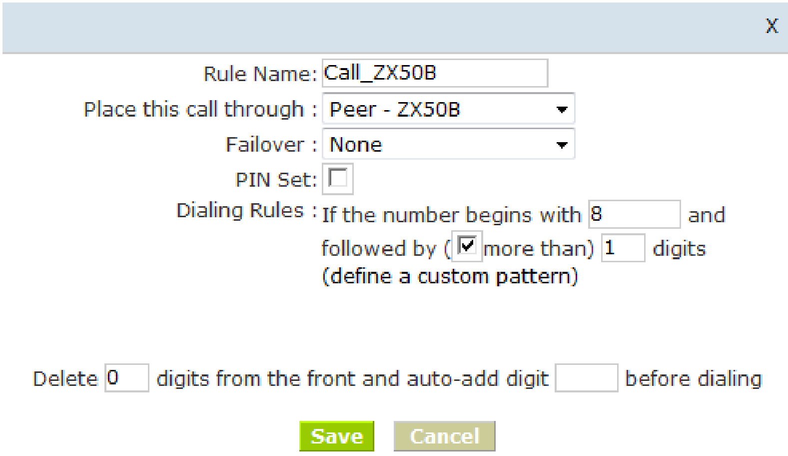
Active the change and apply the test:

1. Register an IP phone ZP302B to ZX50-B with 801 extension.
2. Register an IP phone ZP302A to ZX50-A with 601 extension.
3. Use 801 to dial 601. And you can see 601 will ring and you can pick up the calls.

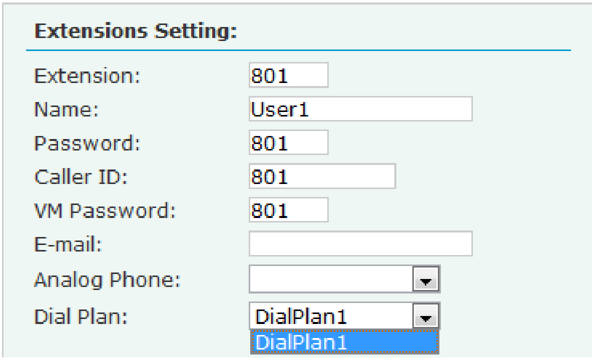
Above is the way to router ZX50-B's call to ZX50-A,

Accordingly, if you want to call from ZX50-A to ZX50-B, continue as follow:

Step 5: Set Dial Rule in ZX50-A all calls start with 8 will be sent to ZX50-B.



Step 6: *Set the user 801 and Dial Plan in ZX50-B*

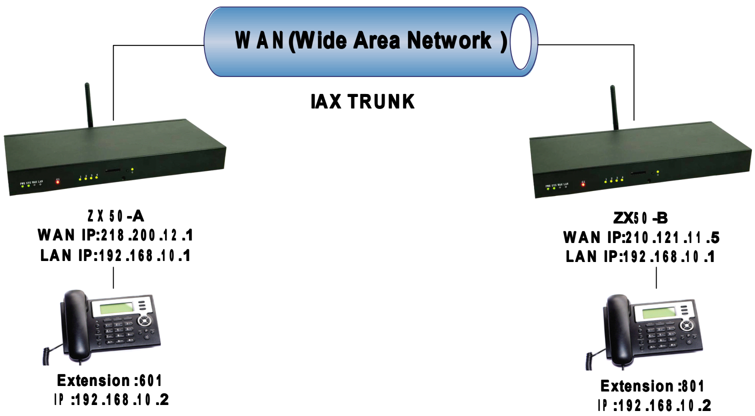


Active the change and apply the test:

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

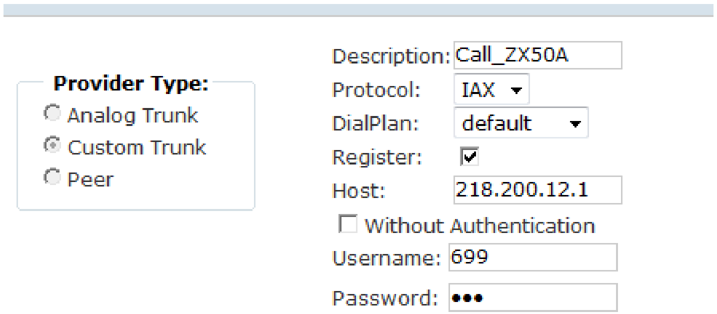
10.8 How to link two IPPBX in different network

The generally environment for two ZX50 in different location is: two the ZX50 IP PBX are both in the internet and using the public IP.

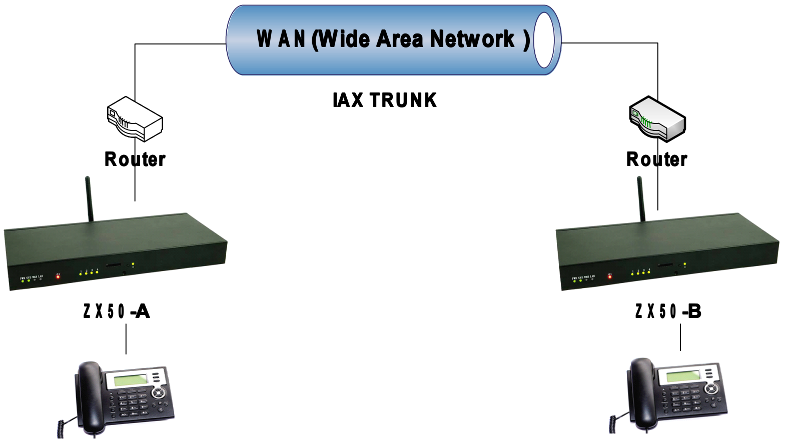


The configuration is same with above guide(10.7) "Link two ZX50 IP pbx in the same network but use the public IP address as the "HOST" settings, like the bellow:

In the page Trunks of *ZX50-B*--> Add a Trunk

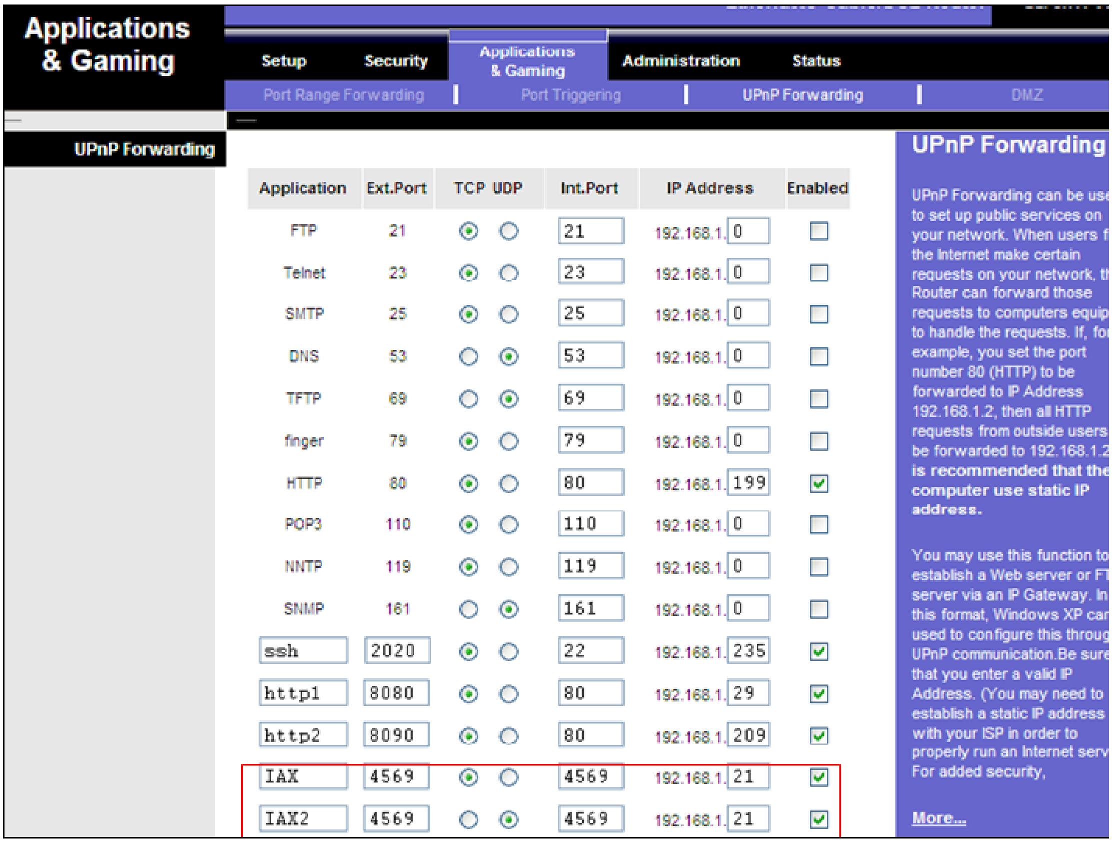


The generally environment for two ZPX50 IP PBX in different location and one or both two are both behind router and using the private IP.So, we need to do port forwarding in the router and make ZX50 IP PBX can reach to each other.



Step 1: *Set port forwarding in the router for ZX50-A*

For the ZX50-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 ZX50-A (192.168.1.21:4569). Below is the setting page in a linksys router:



<u>Step 2</u>: *Set up the Provider Host in ZX50-B*

Set up the service provider and calling rule in ZX50-B to make it register to ZX50-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.

<u>Step 3</u>: *Set port forwarding in the router for ZX50-B*

Use the same method as Step 1 do port forwarding in router-B for ZX50-B as above.

<u>Setp4:</u>*Link two ZX50 and make calls*

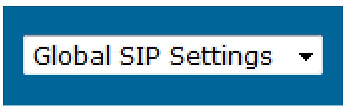
Accordingly, set the 601 users in ZX50-A and 801 users in ZX50-B, and build the correct dial rules as above, you can make calls between two the ZX50 IP PBX.

Noted: You can also apply a DDNS to get one fixed domain for both ZX50 IP PBX and connect to each other rather than use the Port Forwarding in the router.

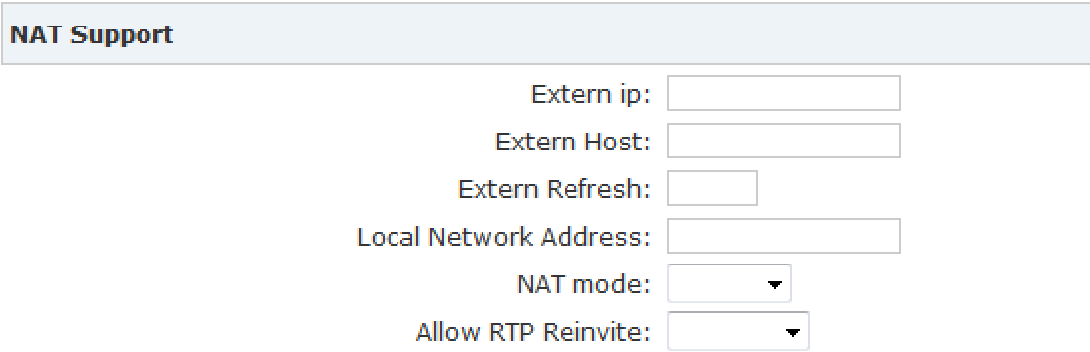
10.9 How to resolve problems about hearing only on one side

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

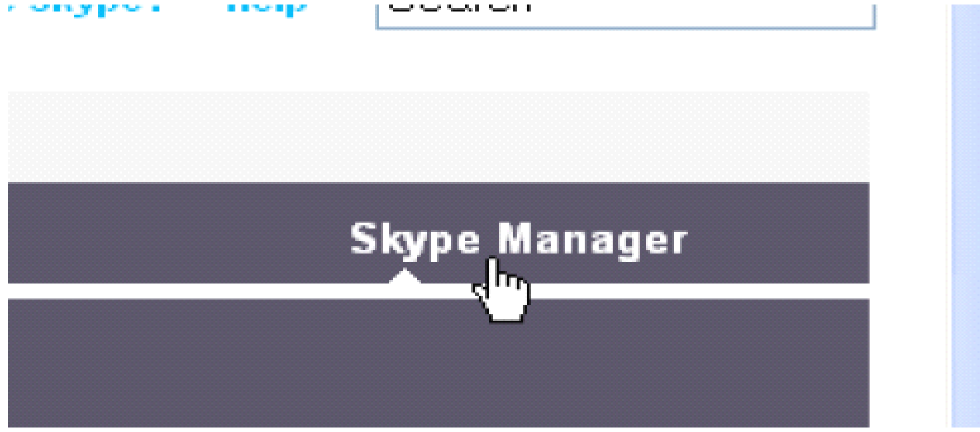


- Extern IP Replace with your external IP address this your public IP or domain
- Extern Host Replace with your external IP address this your public IP or domain
- Extern Refresh Set time for fresh, default 10
- Local Network Address Replace with your local network address and mask
- NAT mode If your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

Chapter11 How to use Skype account in ZX50

11.1 Register for Skype Manager

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



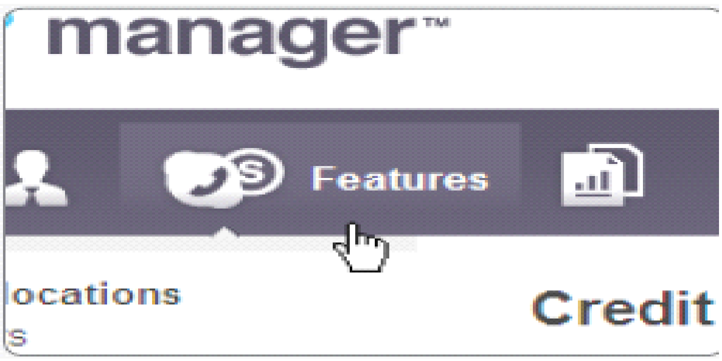
2. Complete the on-screen instructions to register for Skype Manager. You can either use your existing personal account or create a new one specifically for your Skype Manager.

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.

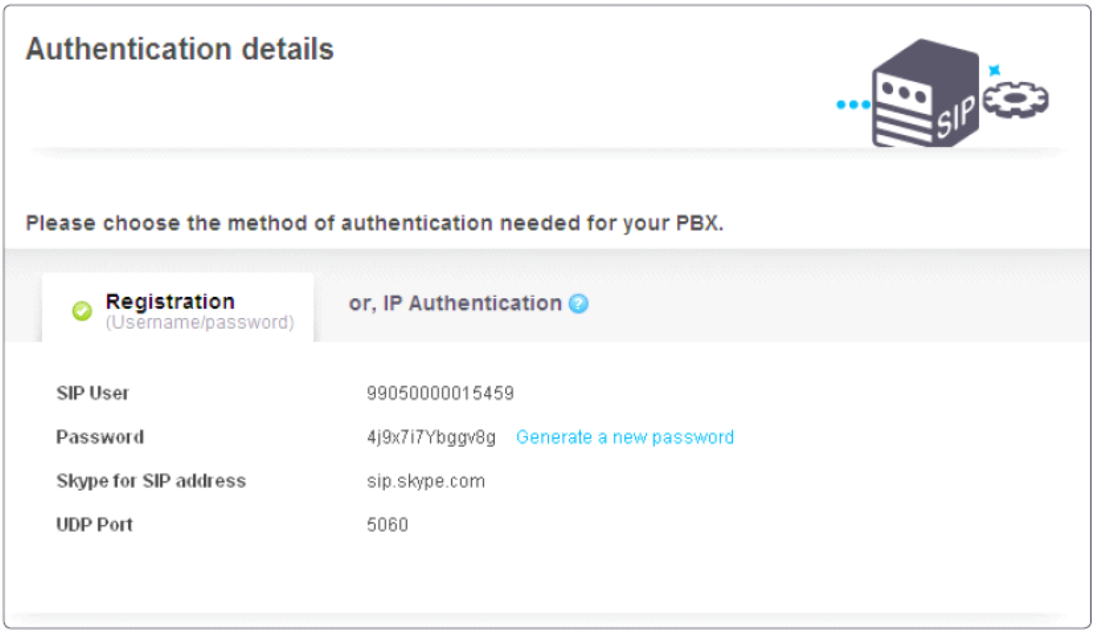
11.2 Create a SIP Profile and buy a Channel Subscription

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

1. Click **Features** in the toolbar



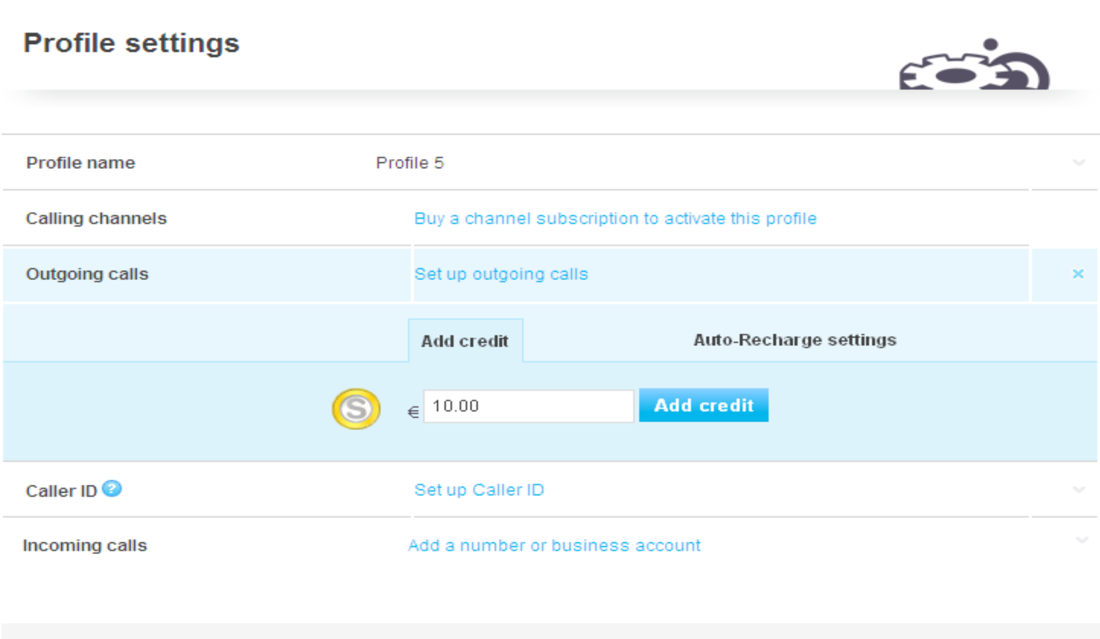
2. In the **Features** menu on the left, click **Skype for SIP**.
3. Click **Create a new profile**.
4. Give your SIP Profile a friendly name so it's easier to remember and click on Next. Your Profile's registration details, including its username and password are displayed. Make a note of these details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



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.

11.3 Allocate Skype Credit to the SIP Profile

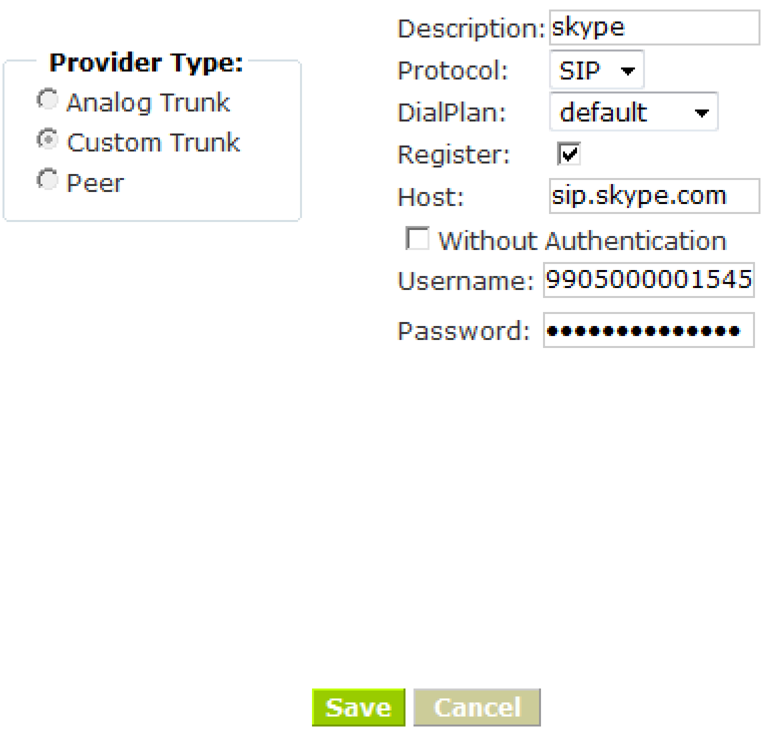
1. Click **View profile** next to the name of the SIP Profile to which you want to allocate credit.
2. Click **Set up outgoing calls**.
3. Enter the amount of Skype Credit you want to allocate to the SIP Profile and click **Add credit**.



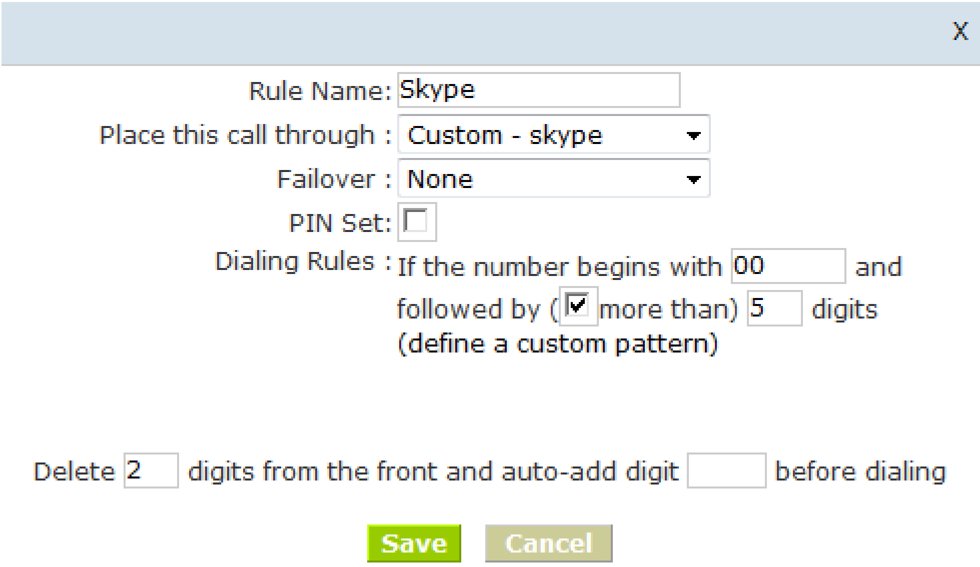
4. If you want to enable **auto-recharging**, click on the Auto-Recharge settings tab, enter the recharge amount and the minimum balance required before recharging, then click **Save changes**.

11.4 Configure your Skype for SIP certified PBX for outbound calls

In the trunk of our IPPBX setting:



Outbound setting of our IPPBX:



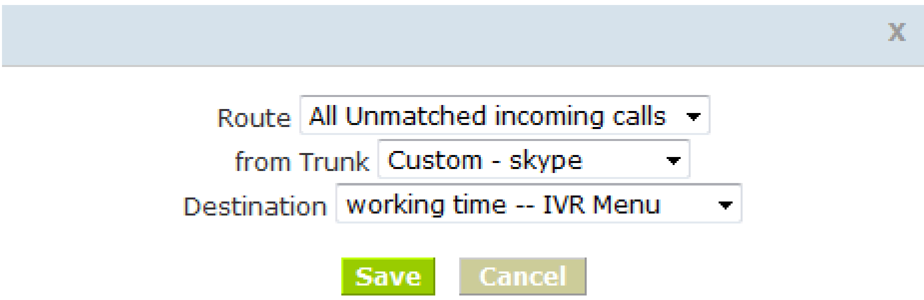
11.5 Make an outbound call

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.

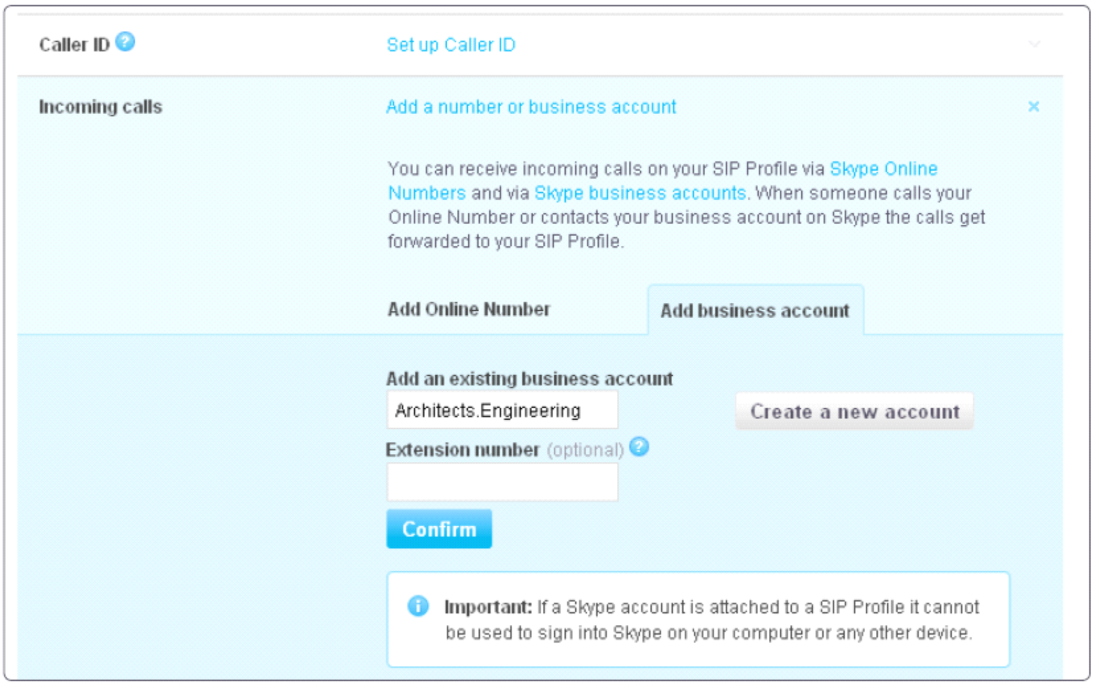
11.6 Configure your Skype for SIP certified PBX for inbound calling

Inbound Routing of our IPPBX:



11.7 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.



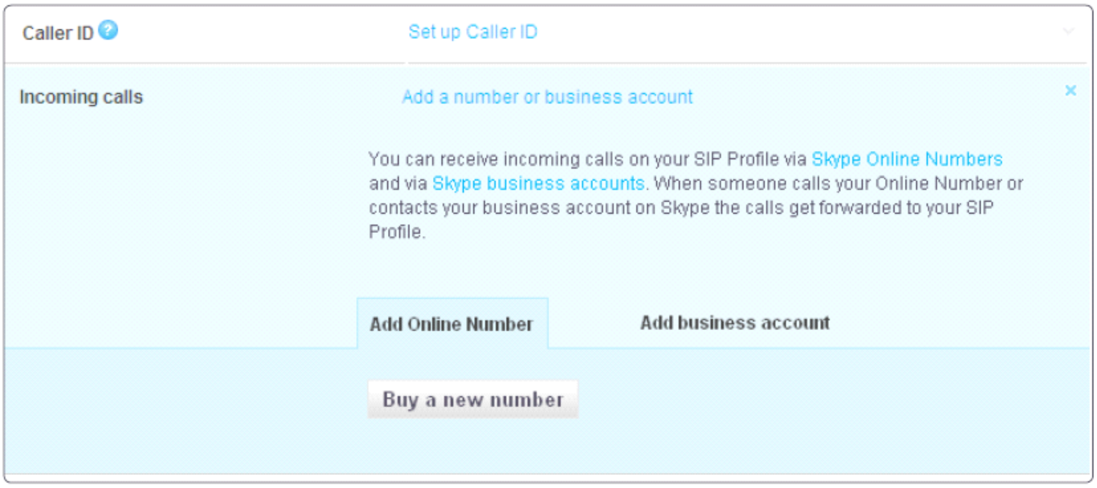
5. Click Confirm.

11.8 Make a test inbound call from Skype

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

11.9 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**



11.10 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**