Model: **PX0522**
Version 1.0
User Manual

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# Chapter1 Brief Introduction

## 1.1 Apearance & Model

This article is the user manual for PX0522 series products. It also includes the application notes for how to use IPshop products to build a telephony system for small office. Our IPPBX productline includes PX09XX and PX05XX, since they have almost the same software and structure so we will use PX0522 as the demo unit on this article. The PX05XX are available with 4 ports, respective 4xFXO or 4xFXS or 2FXO+2FXS. The PX09XX are available with 8 ports configured for FXO/FXS by 4.

## 1.2 System Features

IPshop series of IPPBX are embedded ippbx based on standard asterisk for Home&SMEs, which is not only a PBX, but also as a voice mail Server, IVR server, conferencing server. With 4 or 8 analog interface which can be configured as FXS or FXO
ports (made in factory), and 1 Wan and 1Lan with router function. With excellent echo cancellation function, it can meet most of the customers' requirement.

- Based on Asterisk
- Configuration by Web
- Built-in SIP/IAX Server
- Static/DHCP/PPPoE network access
- Codec: G.711-Ulaw, G.711-Alaw, G.726, G.729, GSM, SPEEX
- SIP/IAX Trunk(use with VoIP Trunk operator)
- Zap Trunk(Use with PSTN)
- SIP/IAX Extensions(connect with IP Phone)
- Zap Extensions(connect with Analog Phone)
- Voice Mail Server
- Flexible Dial Plan
- Call Conference
- IVR Server
- Music On Hold
- Call Logs
- Support IP Phone with Key function
- FAX T.38

Other basic function:
1. Three way calling
2. Call Forward(on Busy or on Unanswered Call or on Unregistered Extension)
3. Call Hold
4. Call Transfer
5. Call Waiting
6. Caller ID

## 1.3 Interface&Panel

1) Interface


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

2) LED Indicator 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 Status | Blink | Connected/flow |
| | | Off | No connection/flow |
| LAN | LAN Status | Blink | Connected/flow |
| | | Off | No connection/flow |
| MMC | SD card Status | On | MMC Connected |
| | | Off | MMC failed |
| USB | Optional (like MMC) | On | USB Connected |
| | | Off | USB failed |
| 1-8 | FXO/FXS status | Red | FXO |
| | | Green | FXS |
| | | Off | Failed |

3) Hardware
- 32bit embedded RISC DSP
- 256MB Onboard Nand Flash
- 2MB Onboard Nor Flash
- 64MB Onboard SDRAM
- 2GB MMC/SD Storage

4) environmental requirements:
- temperature: -10 ℃ -45 ℃
- Storage temperature: -30 ℃ -65 ℃
- humidity: 10-80% no dew
- Power: AC 100~240V

5) Packing List
- IPPBX 1 Unit
- Power Adapter 1 Unit

## 1.4 Default configuration

1. Wan port IP address: http://192.168.1.100:9999
2. Lan port IP address: http://192.168.10.100:9999
3. Web GUI username: admin
4. Web GUI password: admin

The system advice to change default password at first login (or until changed). This is important to avoid hacker attempts and/or non-intended access to the system and telephone account.

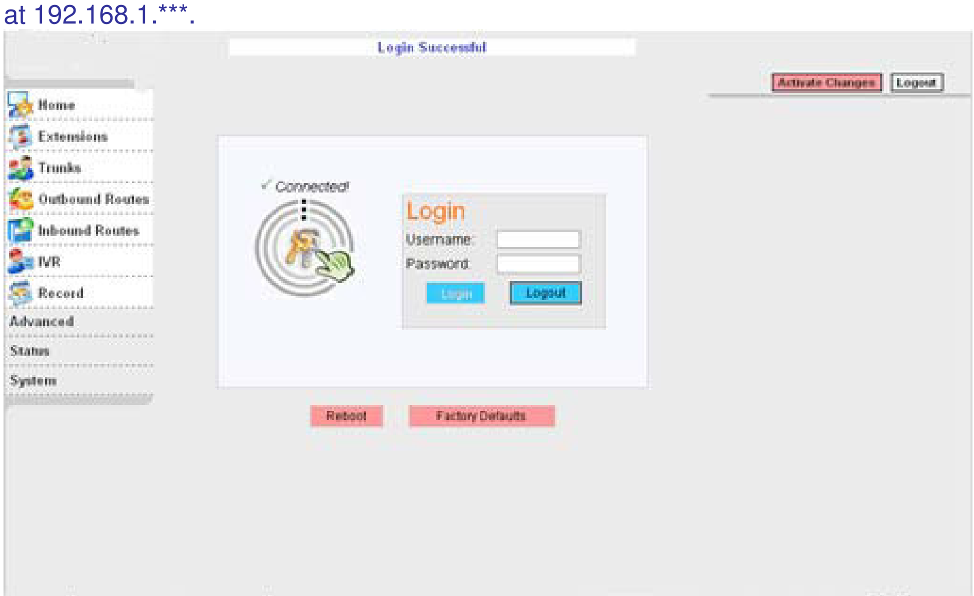
# Chapter2 Basic Configuration

## 2.1. Log on to the system

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


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

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



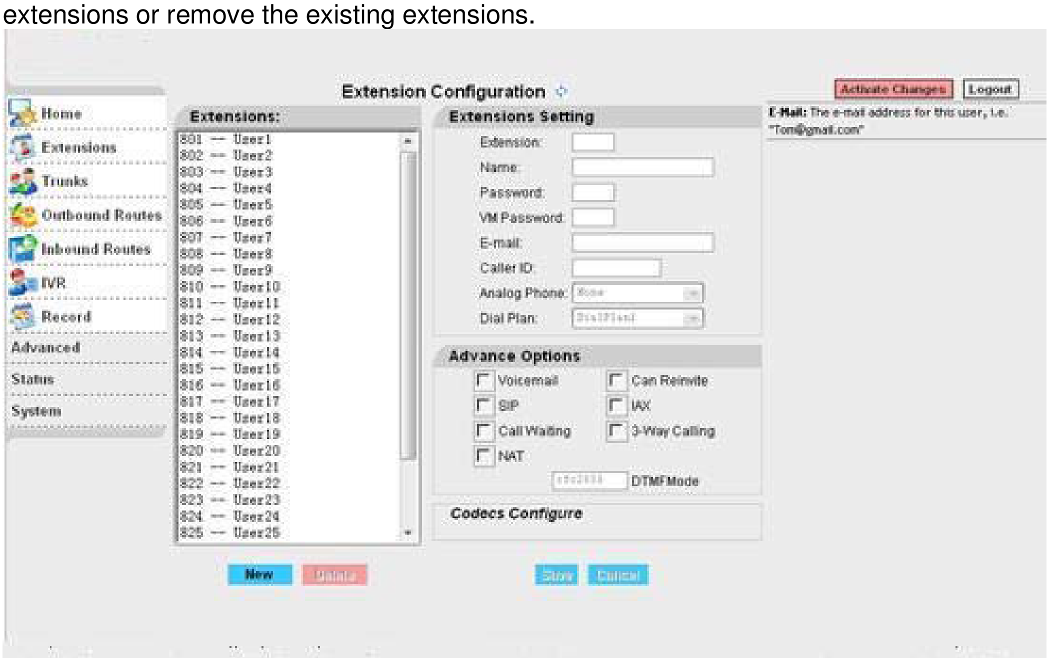

With the WEB-GUI, you can configure extensions, conference, voicemail, Dial Plan and etc. Each page of the GUI has three columns:

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 WEB 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.

## 2.2. 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:

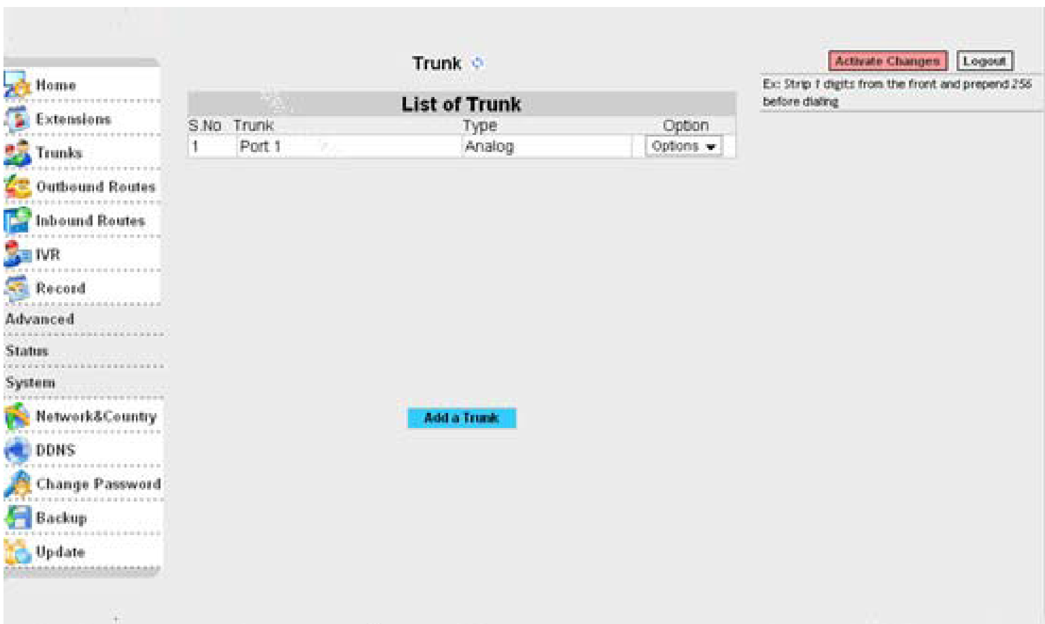
- Extension The extension is assigned to the defined user.
- Name The full name of the individual assigned to this extension.
- Password The password is used to Extension registered
- VM Password The password is used to access voicemail for the specified Extension
- E-mail Set the user's E-mail
- Caller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

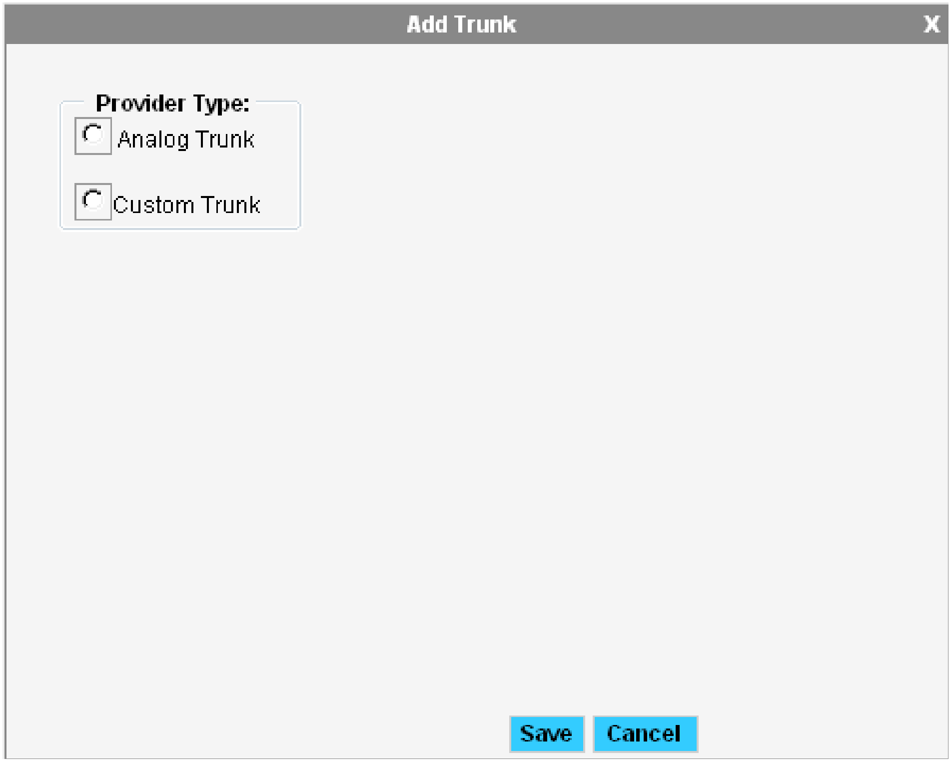
- Voicemail The extension support voicemail
- SIP The extension support SIP protocol
- IAX The extension support IAX protocol
- Call Waiting The extension support Call Waiting function
- 3-Way Calling The extension support 3-Way Calling functions
- Codecs Click here, you can set the extension's codec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

## 2.3. Trunk

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

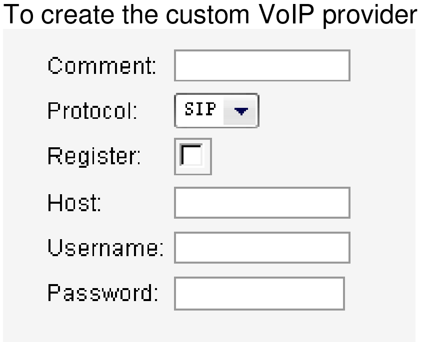



There are three Trunk categories: Analog Trunk, VoIP Providers, Custom VoIP Trunk.




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

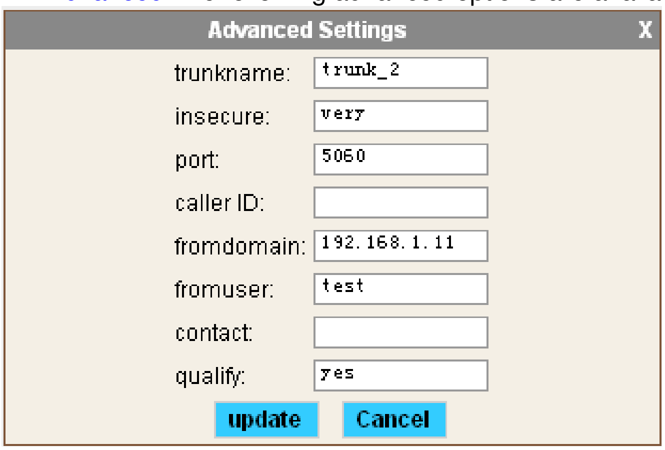
To create the custom VoIP provider definition you will need to complete the following:




- Comment The comment field should be used as the name of the custom VoIP definition
- Protocol Specify either a IAX or SIP protocol
- 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

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.

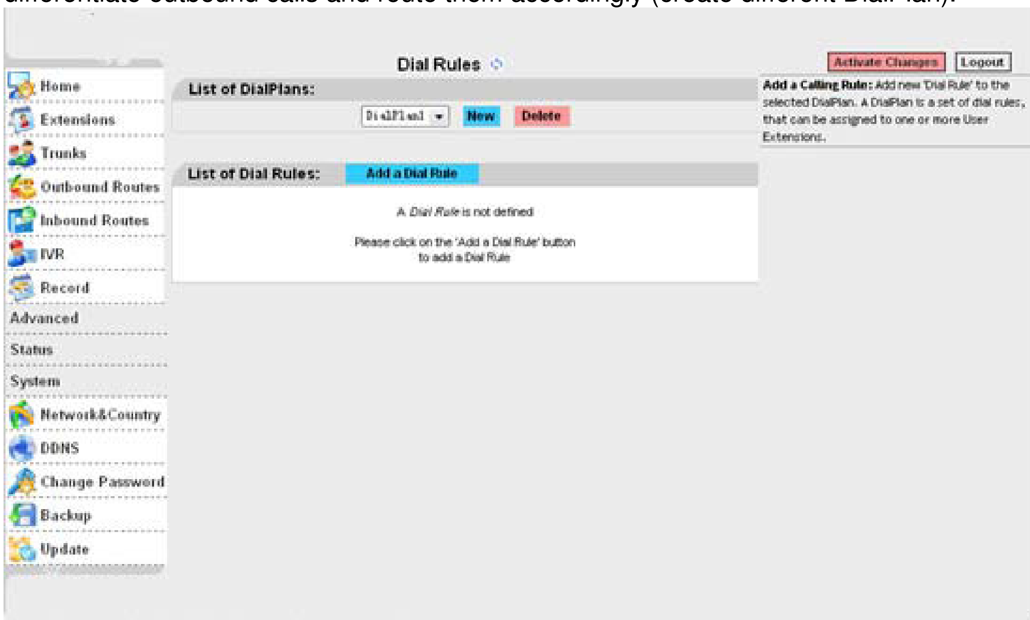
- Codecs Codecs provide the ability for your voice to be converted to a digital signal and transmitted across the internet.
- Advanced The following advanced options are available to further refine your trunk.



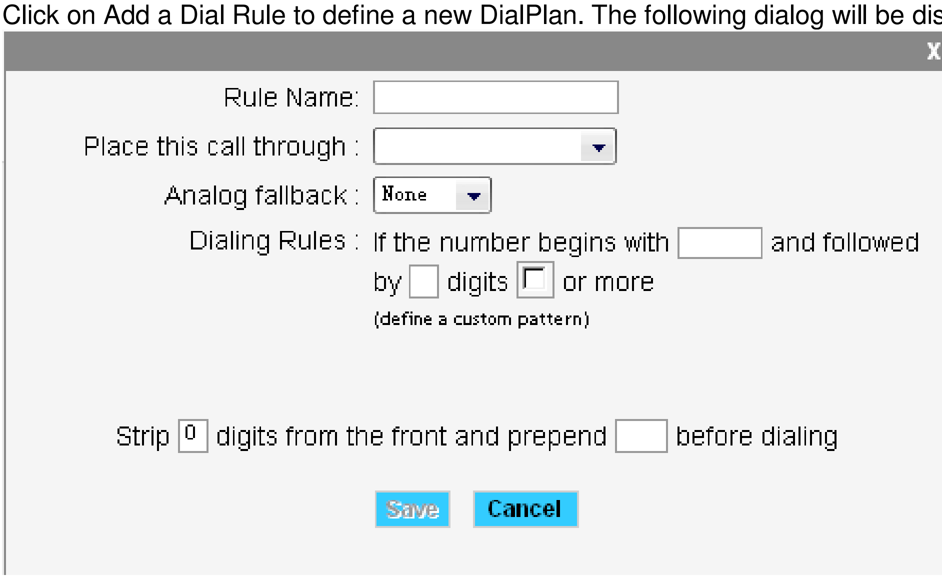

- Trunkname Specify a trunk name if you want to refer to the service provider definition as something other than specified in Comment.
- Insecure This option specifies how connects to a service provider (host) should be handled. Valid options are very/yes/no/invite/port. (Default is “very”)
- Port The 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

## 2.4. Outbound Rules

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.

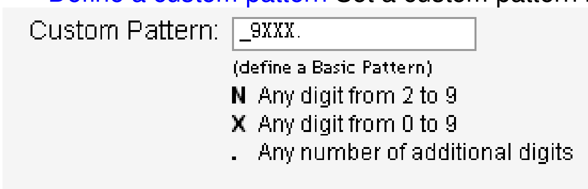



A DialPlan is comprised of the following items:

- Rule Name Set a rule name
- Place this call through Select a Trunk through which the call should be made
- Analog fallback Select a Analog fallback
- 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 256 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.



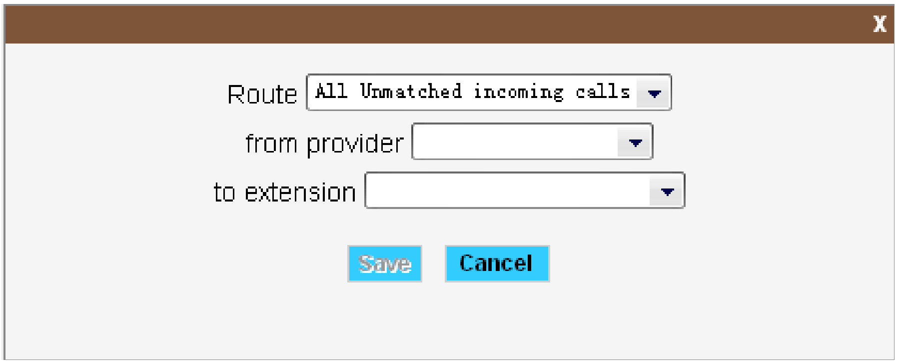

**N** Any digit from 2 to 9
**Z** Any digit from 1 to 9
**X** Any digit from 0 to 9
**.** Any number of additional digits

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.

- Strip 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.

## 2.5. Inbound Rules


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.

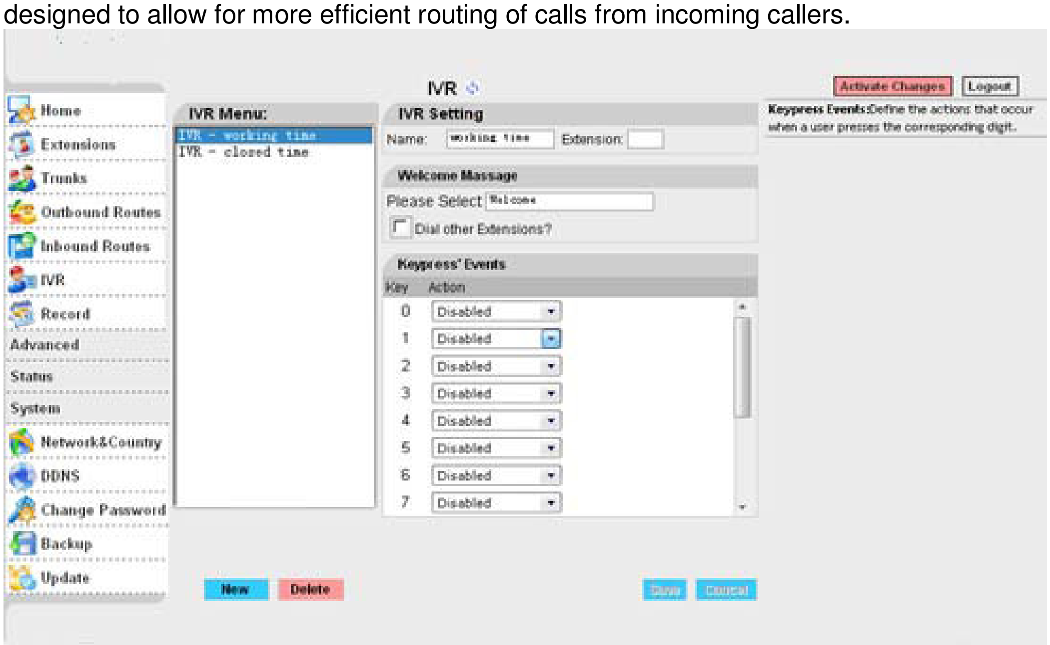



There are only a few options you need to configure

- 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.

## 2.6. 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.

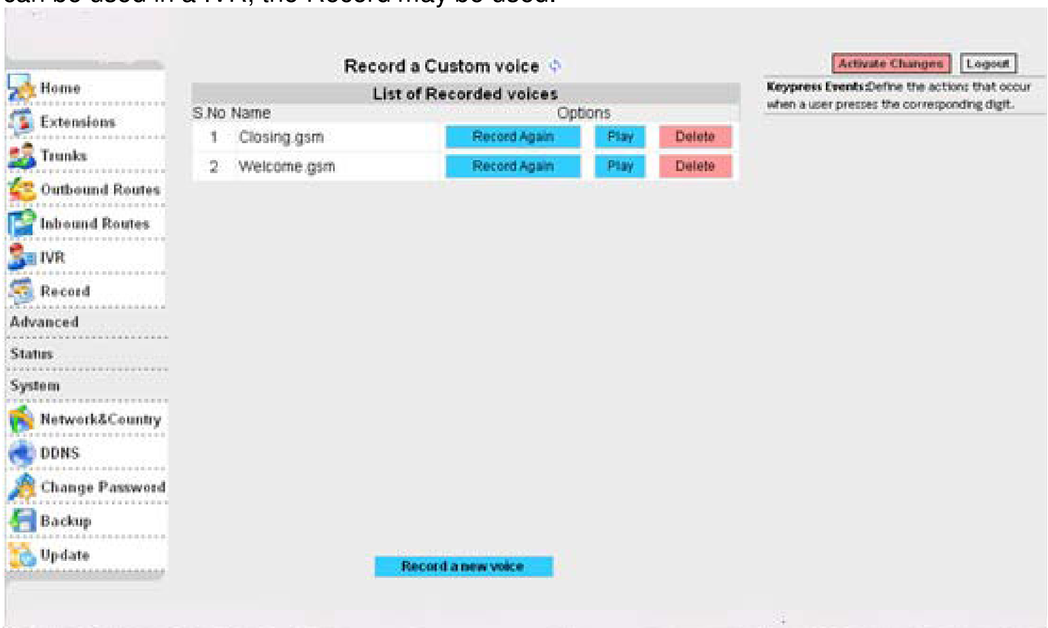



Voice menus are constructed depending on your needs. Just like your business you need to create the solution best suited to your customers.

- 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.

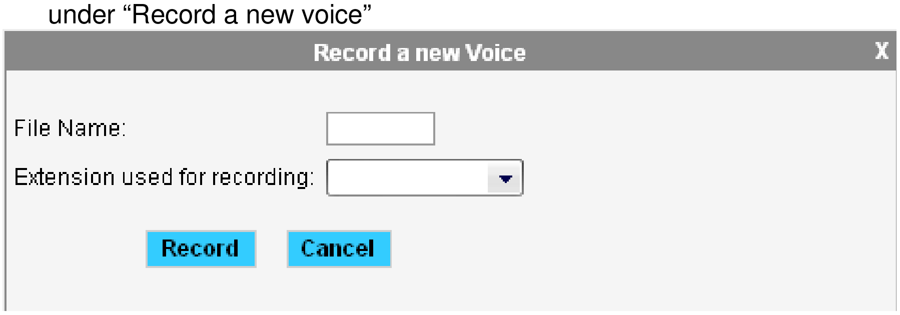
## 2.7. Record

In the event that one wants to record custom menu 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”

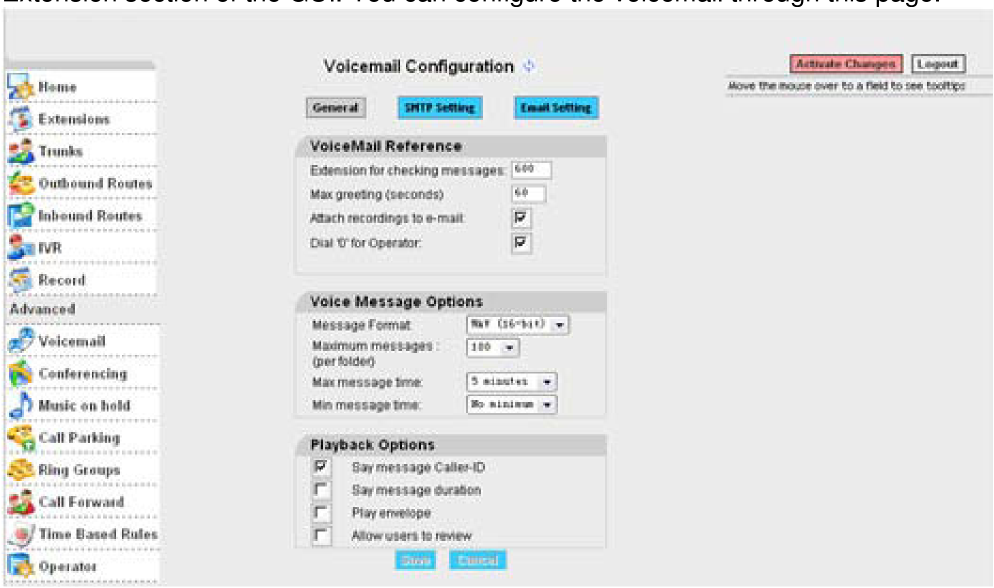



- File Name This text entry box specifies the saved name of the file that is to be recorded.
- Extension Used for Recording This drop-down select box allows the user to choose which extension will dial to wait for the user to speak the prompt

# Chapter3 Advanced Configuration

## 3.1. Voicemail

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




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

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

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

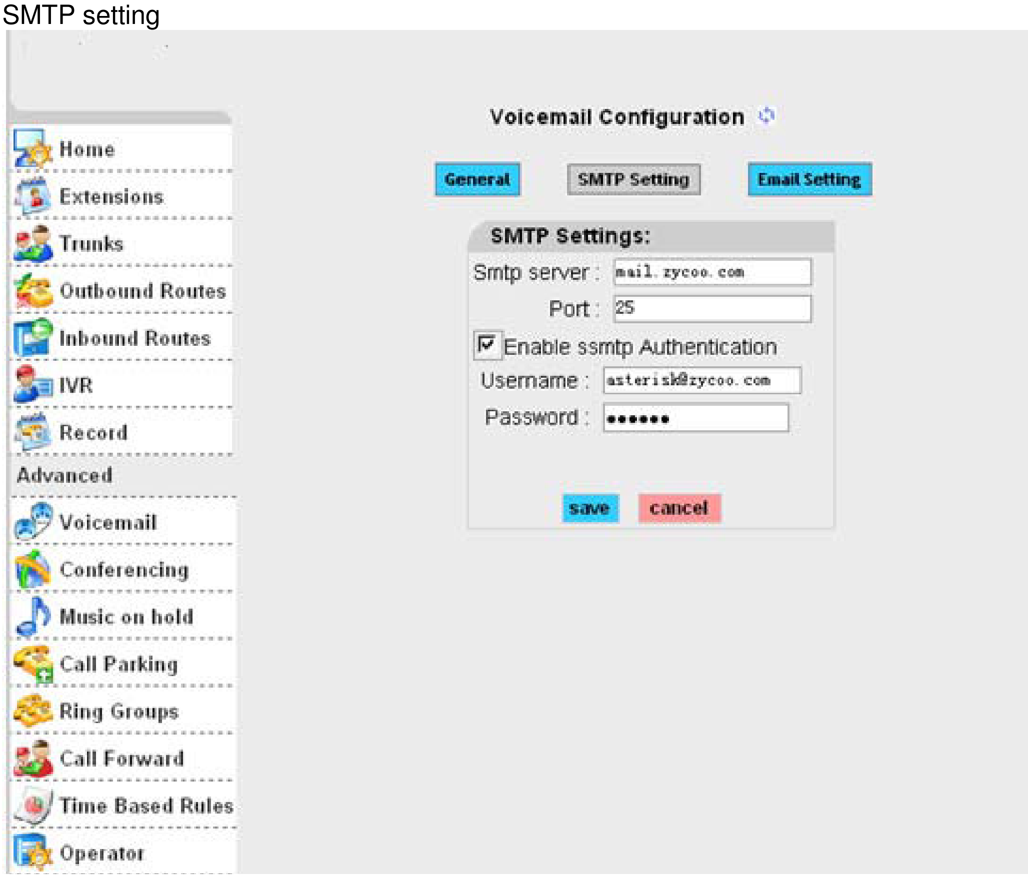
- Message Format This option gives you the ability to choose the format in which messages will be mailed.
- Maximum Messages The maximum number of messages per voice mail box is set here.
- Maximum Message Time The maximum duration of a message left by a caller is set here.
- Minimum Message Time The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

There are several playback options that can be specified.

- Say Message Caller-ID The Say Message Caller ID option reads the caller ID before the voice mail message is played.

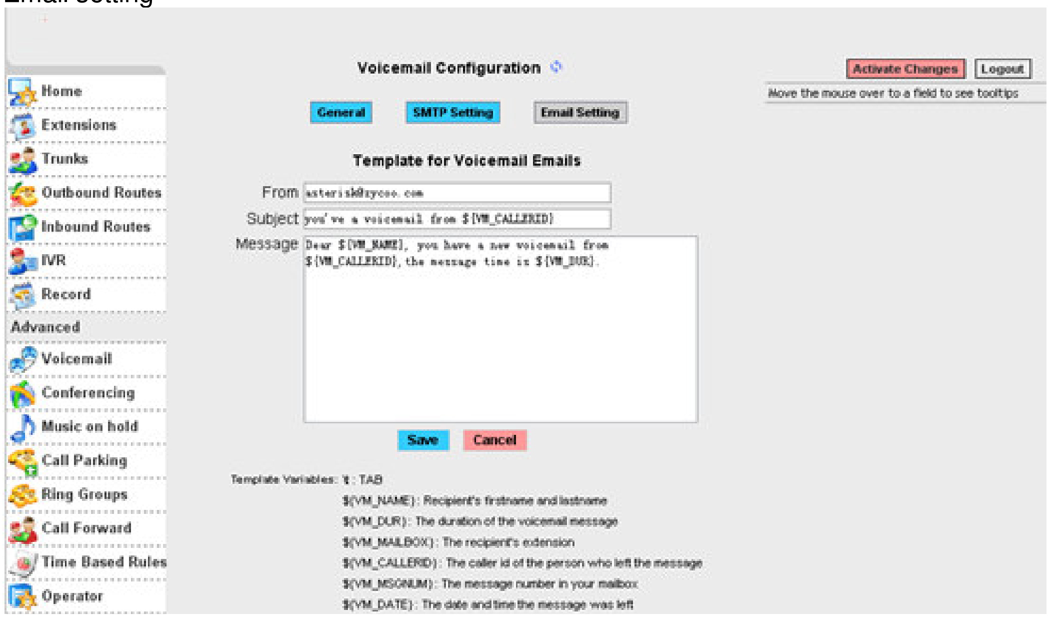
- Say Message Duration This option identifies exactly how long the message lasted.
- Play Envelop The envelope provides the date, time, and caller ID related to a voice mail.
- Allow Users to Review This option provides incoming callers the option to review their message before it is saved and can be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

**Voicemail to email set:**
SMTP setting




- 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.
- Enable SMTP Authentication if your SMTP server needs Authentication, please enable SMTP Authentication set, and configure the follow information:
- Username input username of your email.
- Password input password of your email.

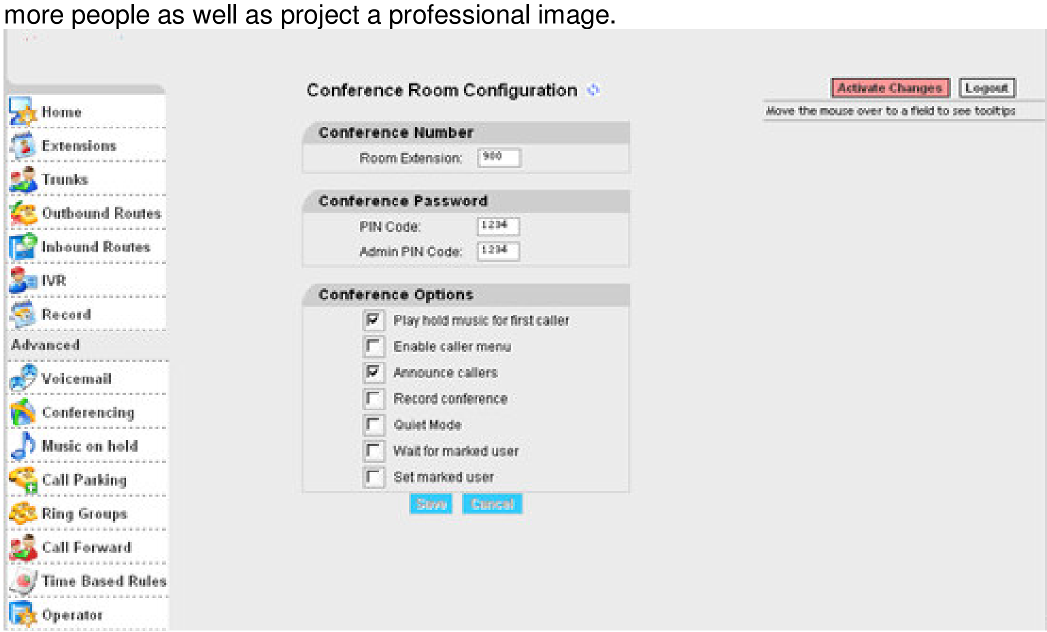
Email setting




- From Set the from email
- Subject Set the email title
- Massage Input the matter in your email.

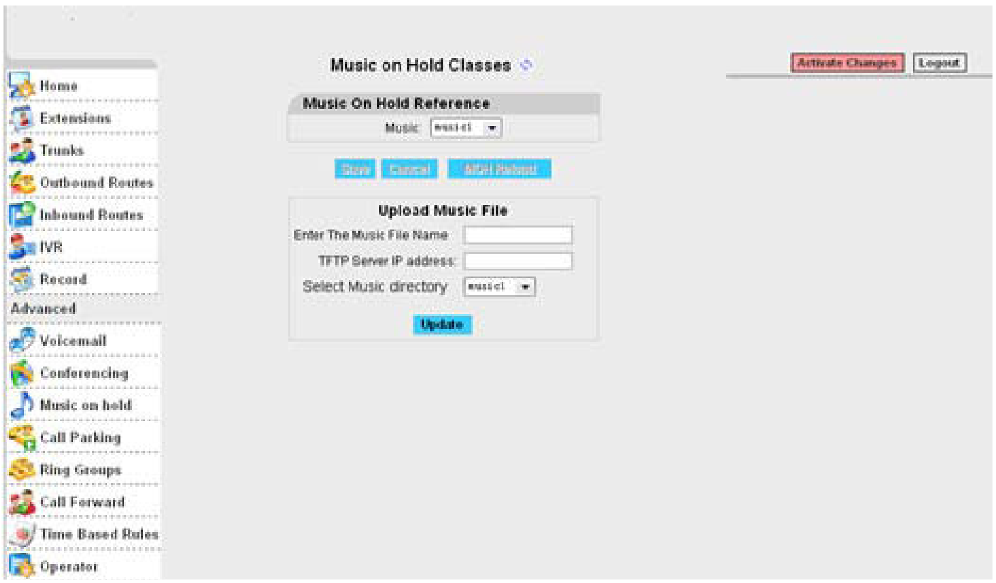
## 3.2. Conferencing

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.



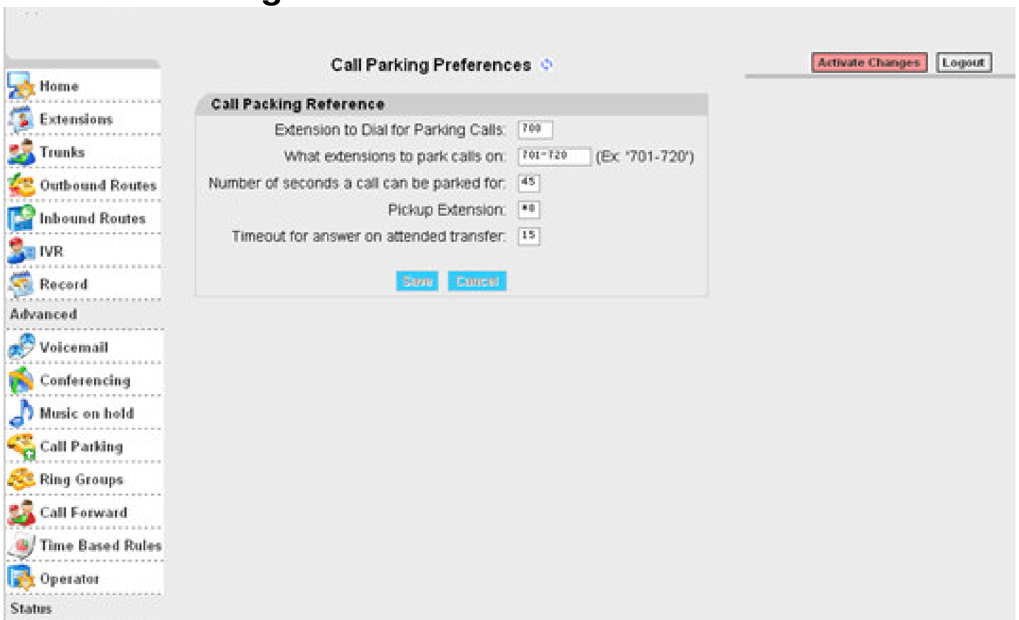

The configuration of the conference room and standard features is very straightforward. The conference room use default extension 900 , but you can always change it to any extension number you want. After establishing the extension for the room, you need to specify the password settings for the conference. 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.

## 3.3. 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.

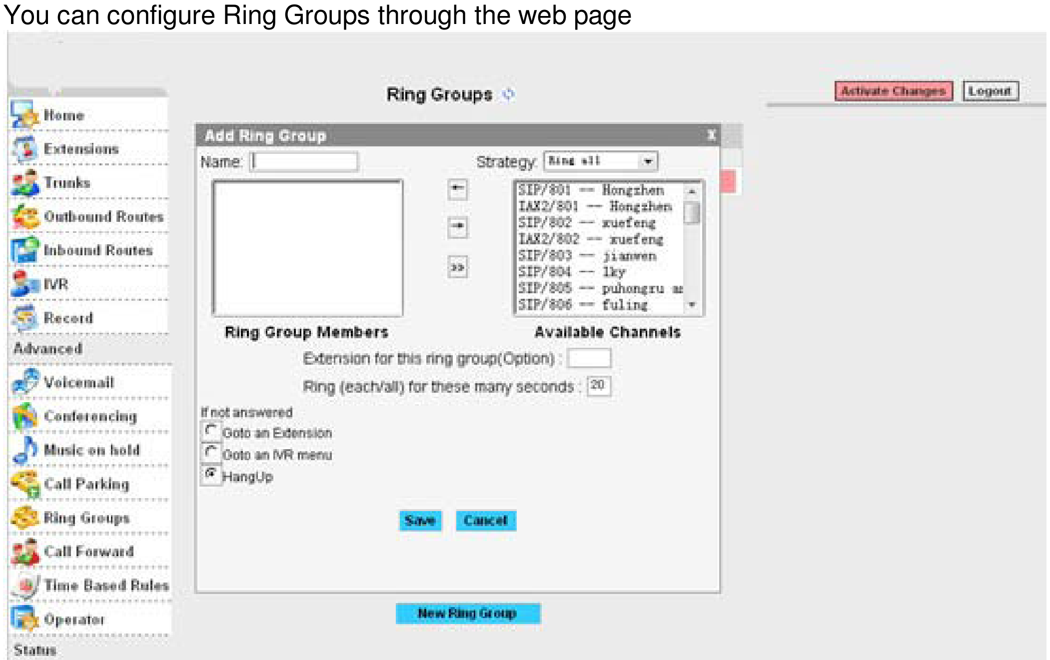
## 3.4. Call Parking




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

## 3.5. Ring Groups

You can configure Ring Groups through the web page



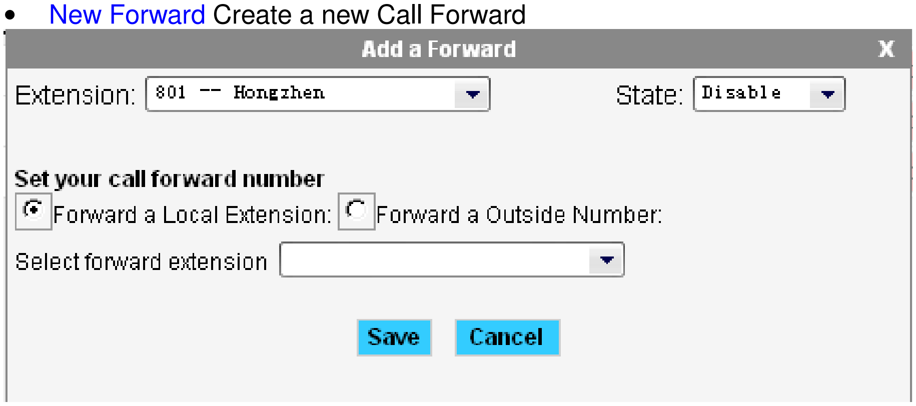

Define Ring Groups to Dial more than one extension

- 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:
  1 Goto Voicemail of this user
  2 Goto an IVR menu
  3 HangUp

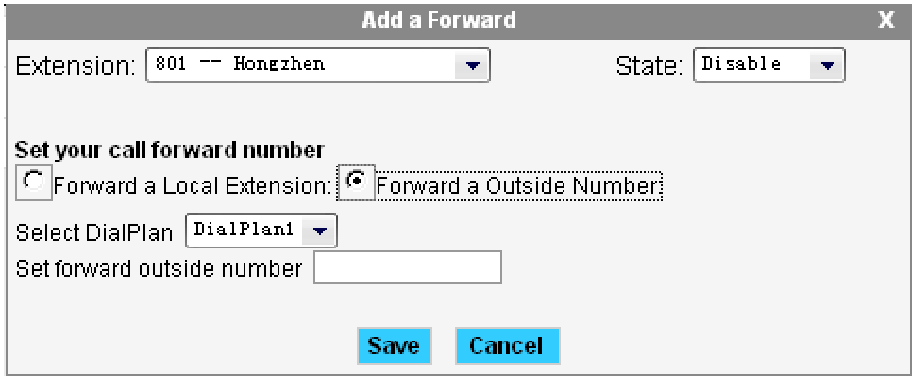
## 3.6. Call Forward


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



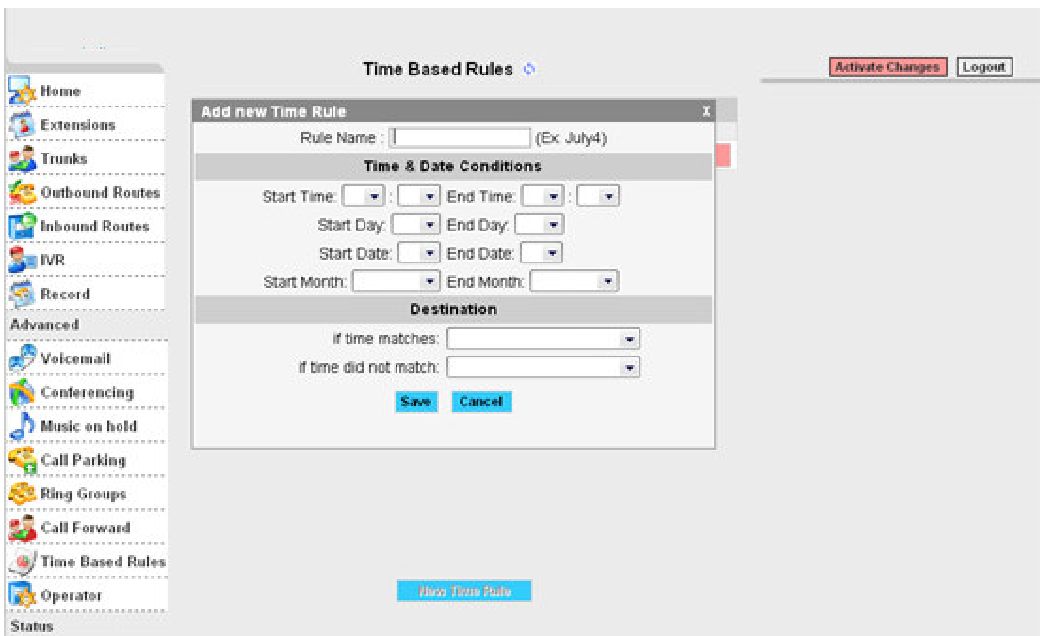

- Extension Select a need to call forward extension
- State Set state of the extension.(Disable, Always, Busy, No answer)
- Select forward extension Select a call forward to extension

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



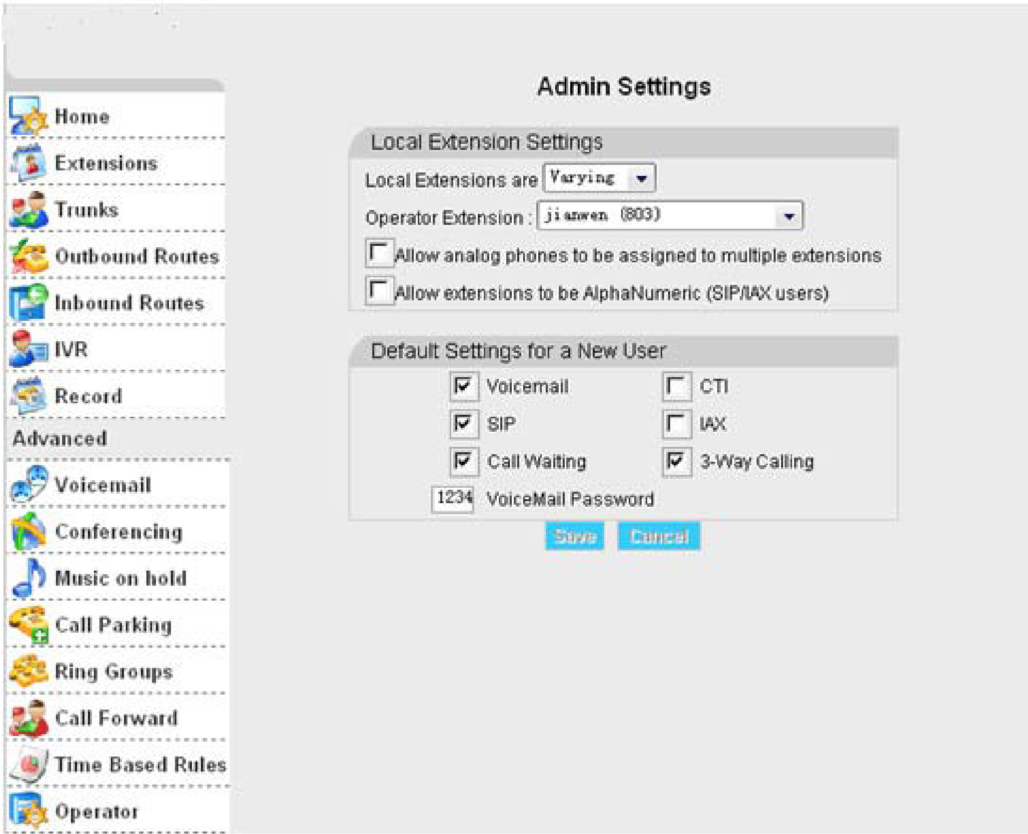

- Select DialPlan Select a Call forward to outside number using dialing rules
- Set forward outside number Input a Call forward to outside number. (Notice: This number must be consistent with the corresponding DialPlan)

## 3.7. Time Based Rules




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

## 3.8. Operator

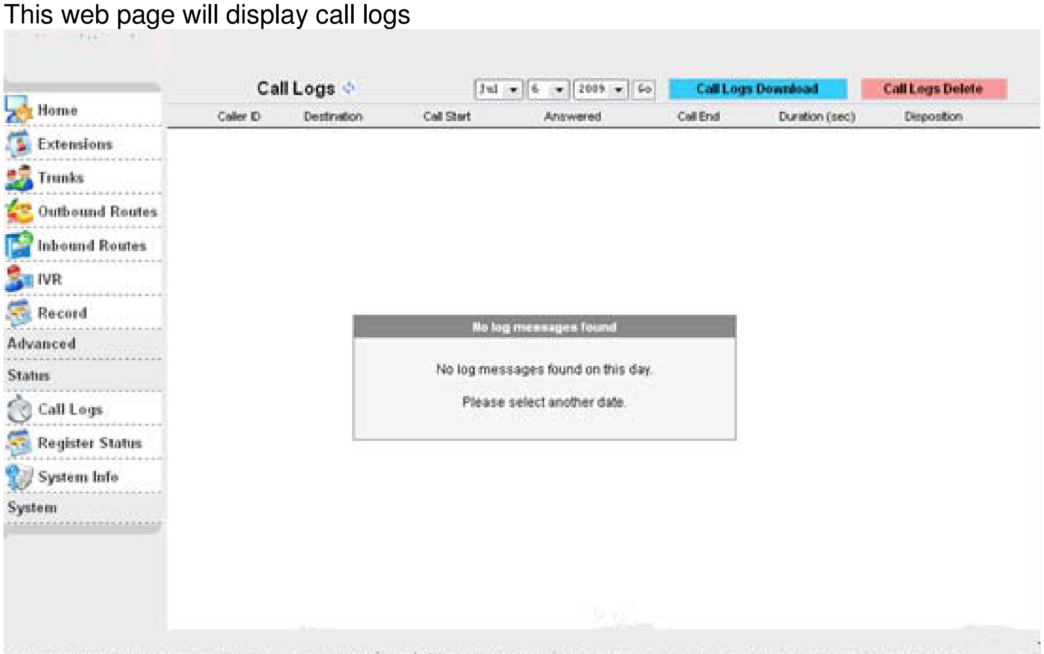



- Local Extensions are Set up the digit of local extensions
- Operator Extension Set up Operator Extension. (you can dial “0” go to the extension at any time).
- Default Settings for a New User Set up the Default Settings for a New User, when you create a new extension will use the configuration.

# Chapter4 Status Display

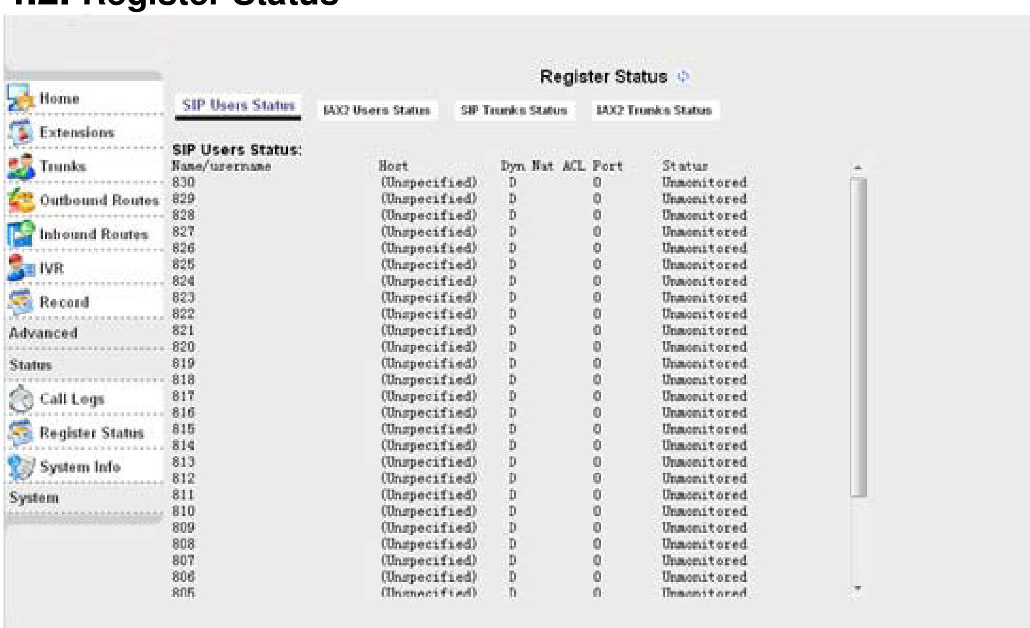
## 4.1. Call Logs

This web page will display call logs



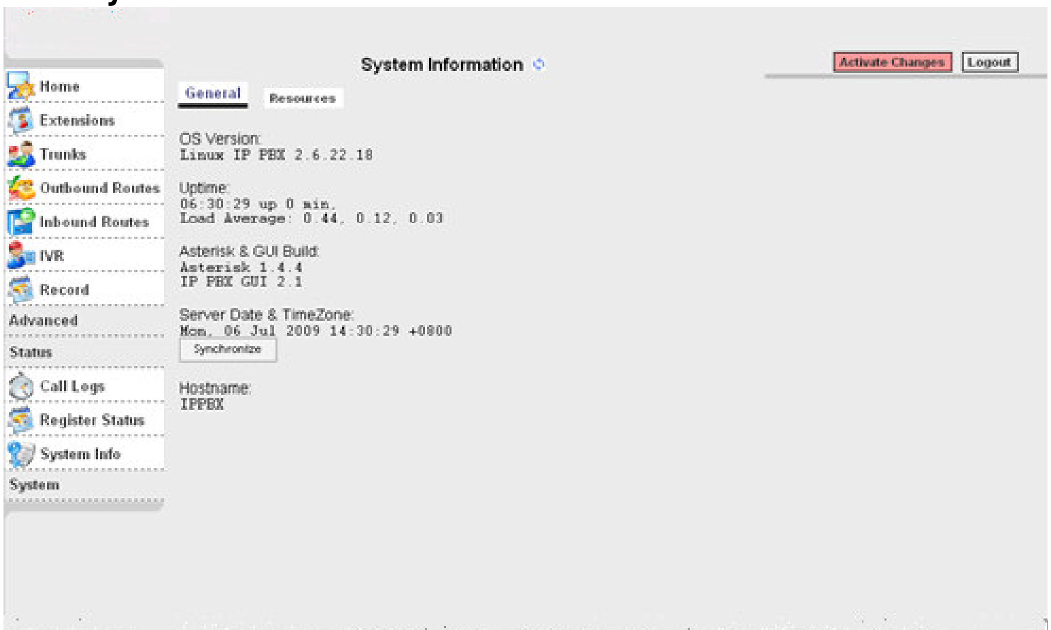

- Call Logs Download download the call logs file
- Call Logs Delete delete the call logs file

## 4.2. Register Status




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

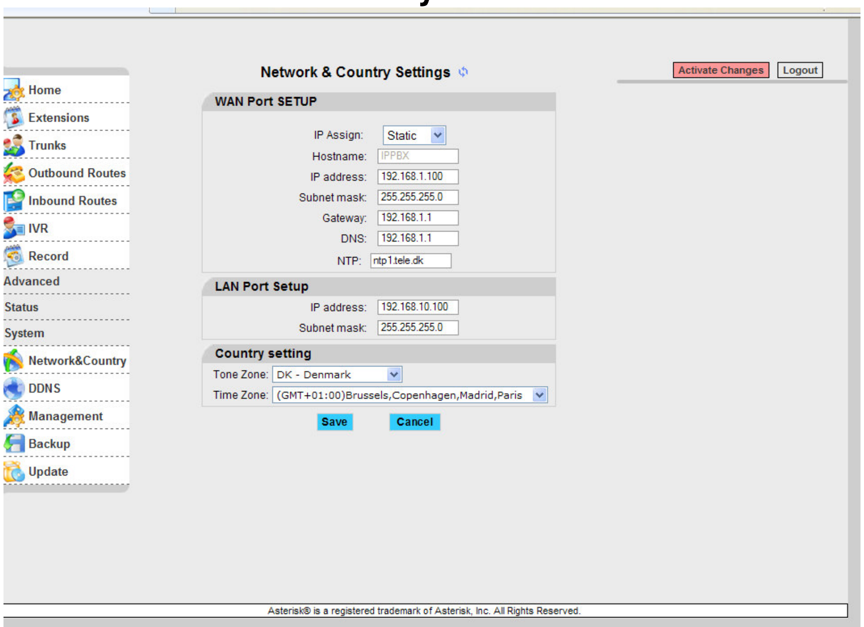
## 4.3. System Info




In this page it will display nonce system info

# Chapter5 System Management

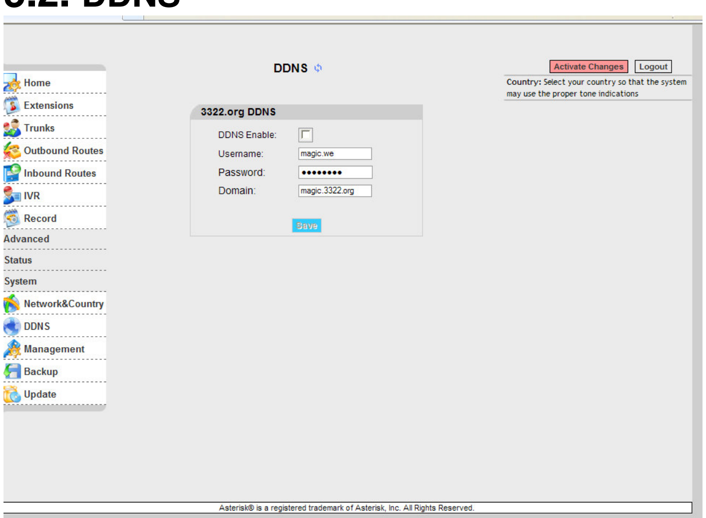
## 5.1. Network and Country




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

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

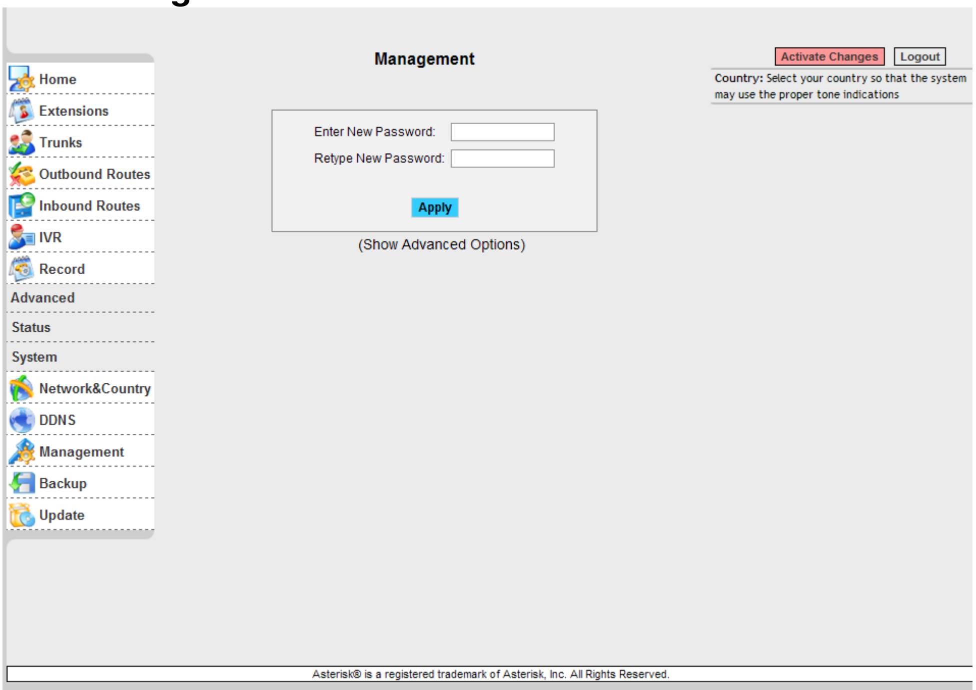
## 5.2. DDNS




On this page, you can set DDNS reference.

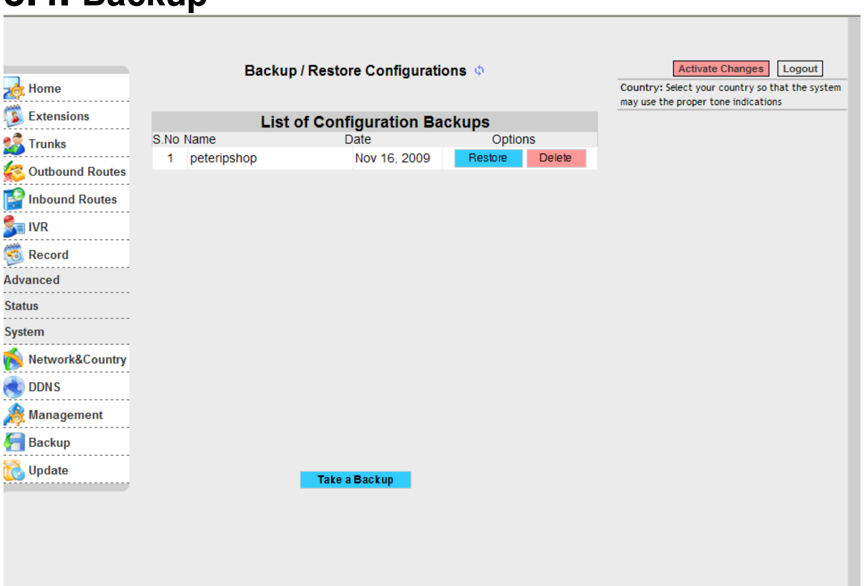
Notice: Now, it only supports 3322.org server. More other servers, you can customize based on your requirement

## 5.3. Management



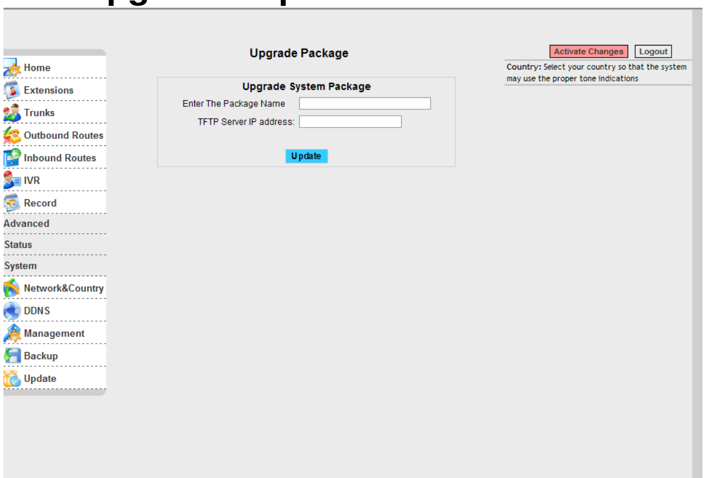

On this page, you can change the administrator password (Default password: admin)
And you can also set the advanced options about SIP and IAX2 protocol in the "Show Advanced Options" list,that is useful when you set connect two ippbx in different network.

## 5.4. Backup




On this page, clicking the “Take a Backup” button, you can backup, (delete/restore) configuration.

## 5.5. Upgrade / Update



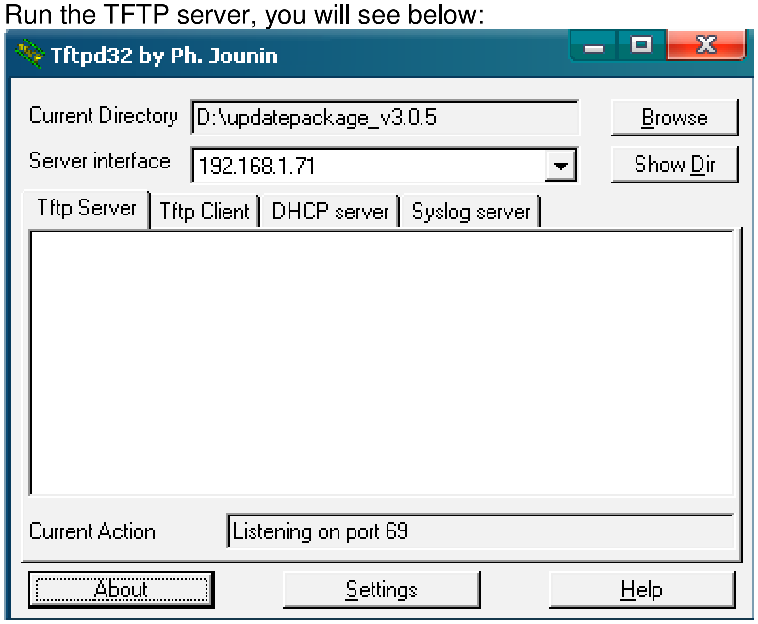

In this page you can upgrade system package

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

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

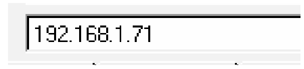


Run the TFTP server, you will see below:




Enter the configuration page, then upgrading page;

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

- Enter TFTP Server IP address, hereby it's 

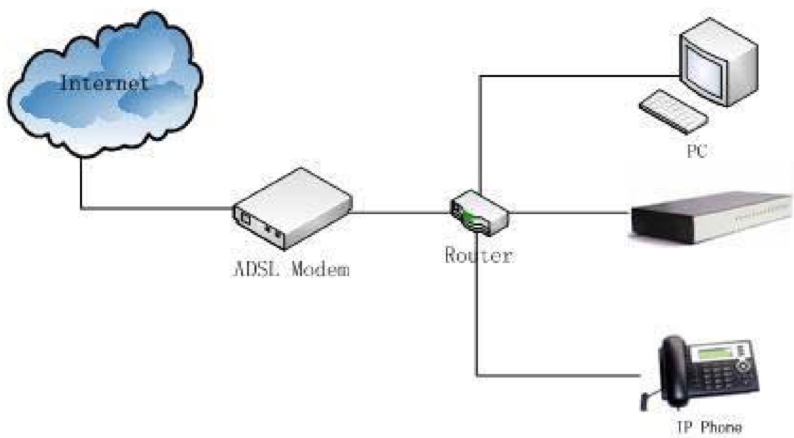

After done, click Update to update, then the system will reboot automatically.
(Note: the upgrading will set your system as default, please make backup before you do it.)

# Chapter6 Operating Instruction

## 6.1 How to link the IP PBX to the internet/network

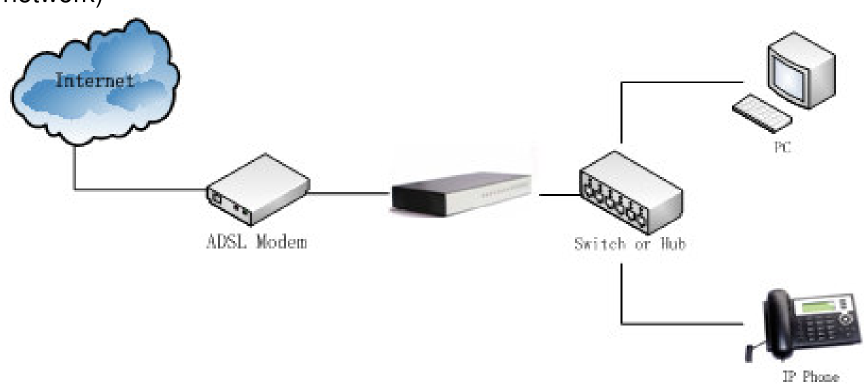
**With Router**

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




**Without Router**

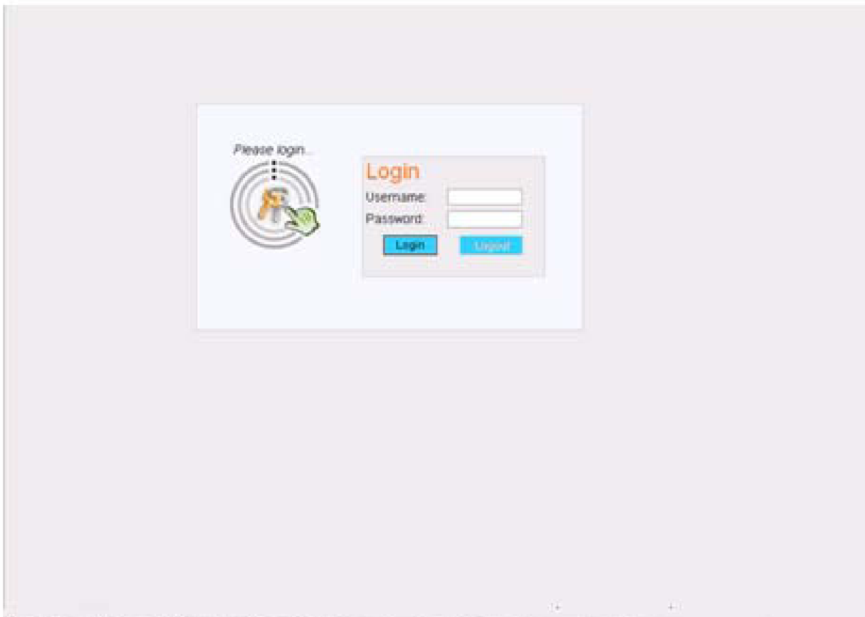
If you have the public IP and want the IPPBX access the public network directly without router,then you should connect the Wan port of the IPPBX to the public network and connect HUB or Switch to the Lan ports of the IPPBX to let your PC access the public network..(If you want to access the public network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



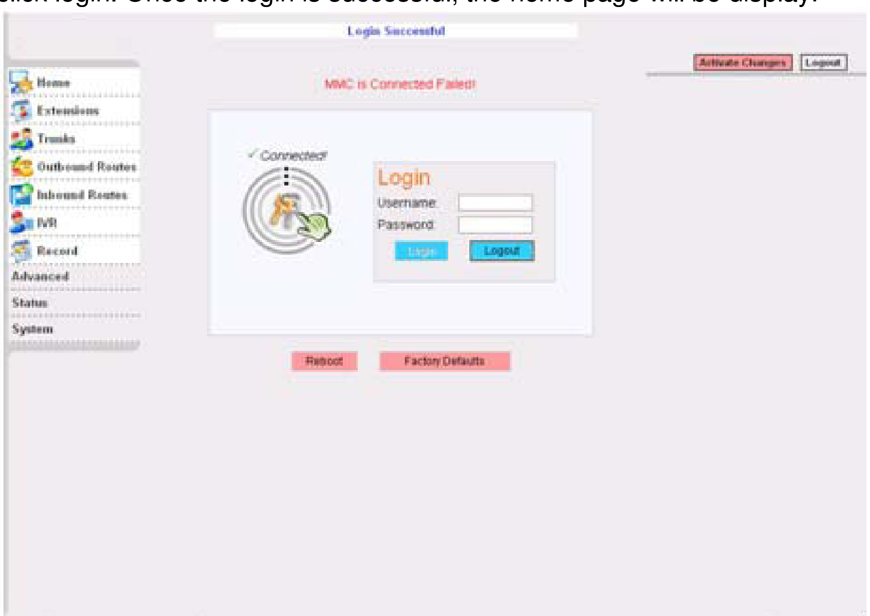

## 6.2 Log in to the system

After connecting the ippbx to the local area network. Launch the web browser on a computer that is in this local area network. Enter the IP address for the system (default: Wan port IP address is http://192.168.1.100:9999, Lan port IP address is http://192.168.

10.100:9999) . The start web page will appear like this:




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



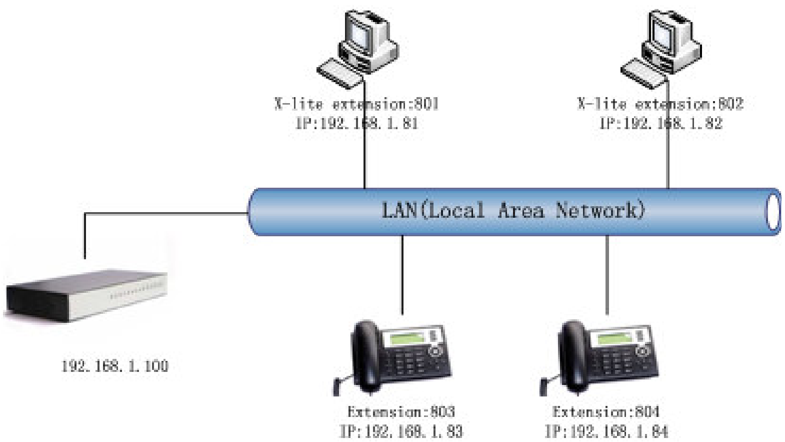

With the WEB GUI, you can configure extensions, conference, voicemail, Outbound Routers and etc. Each page of the GUI has three columns:
The left column present all the options tab that you can program the system. Click the tab to go this kind of option setting page.
The middle column contains the primary 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 WEB-GUI.

- Reboot: Reboot the PX0522 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.

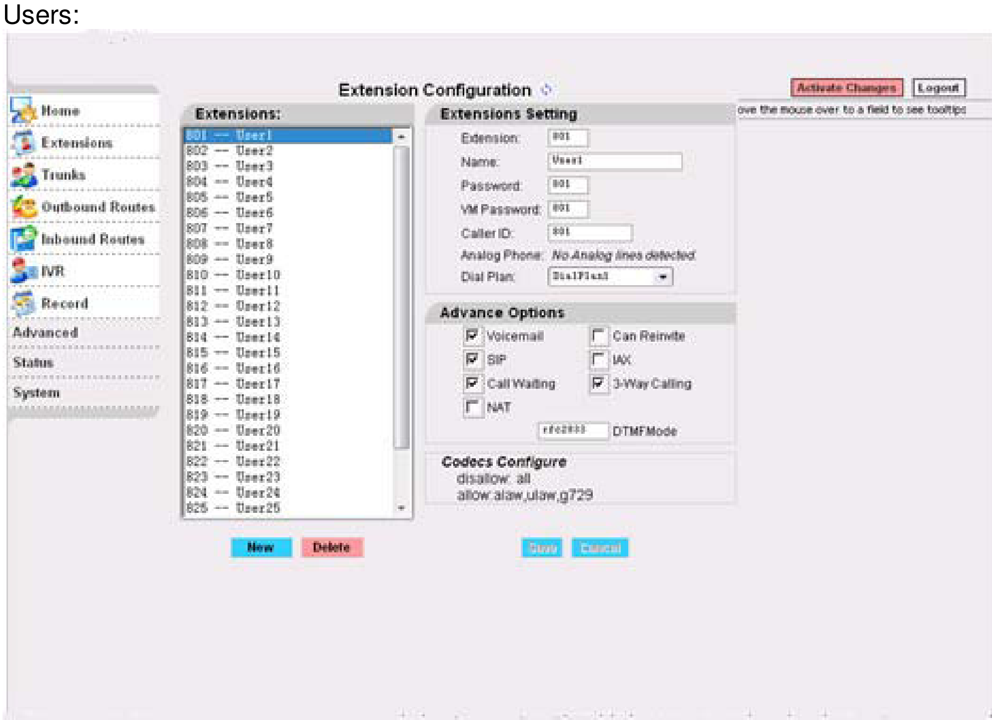
## 6.3 How to make an internal call

Making internal calls are the base requirement for a telephony system. Below are the settings for this usage. It is base on PX0522,but setting is the same in other IP PBX products.




**Set User**

Users:




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.

## 6.4 How to make a outbound call

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

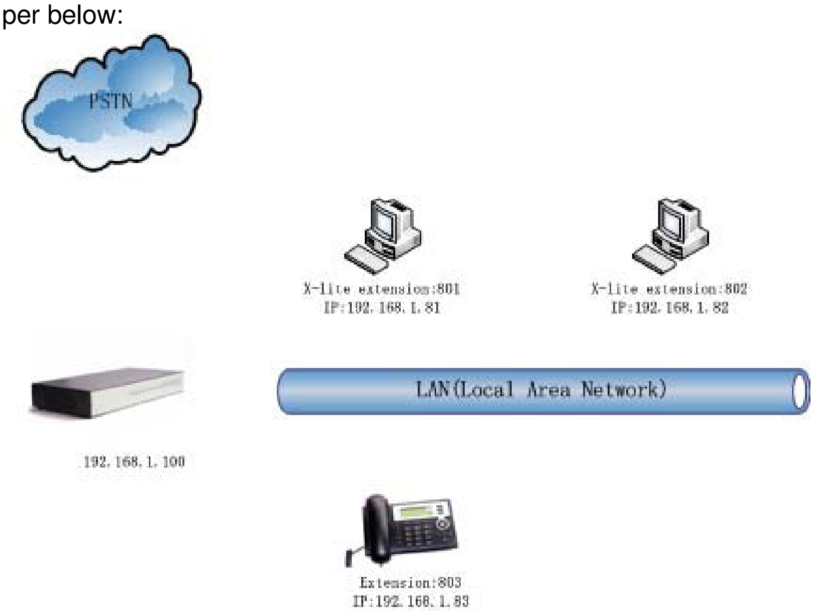
- Analog Ports: FXO ports of PX0522, connect to local PSTN
- VoIP Trunk: SIP or IAX trunk, connect to remote SIP/IAX server

I am using PX0522, the port 1-2 are configured as FXO ports, port 3-4 are configured as FXS ports. When a port is configured as FXO port, the corresponding LED shows RED.
When a port is configured as FXS port, the corresponding LED shows GREEN.
What are FXO and FXS?

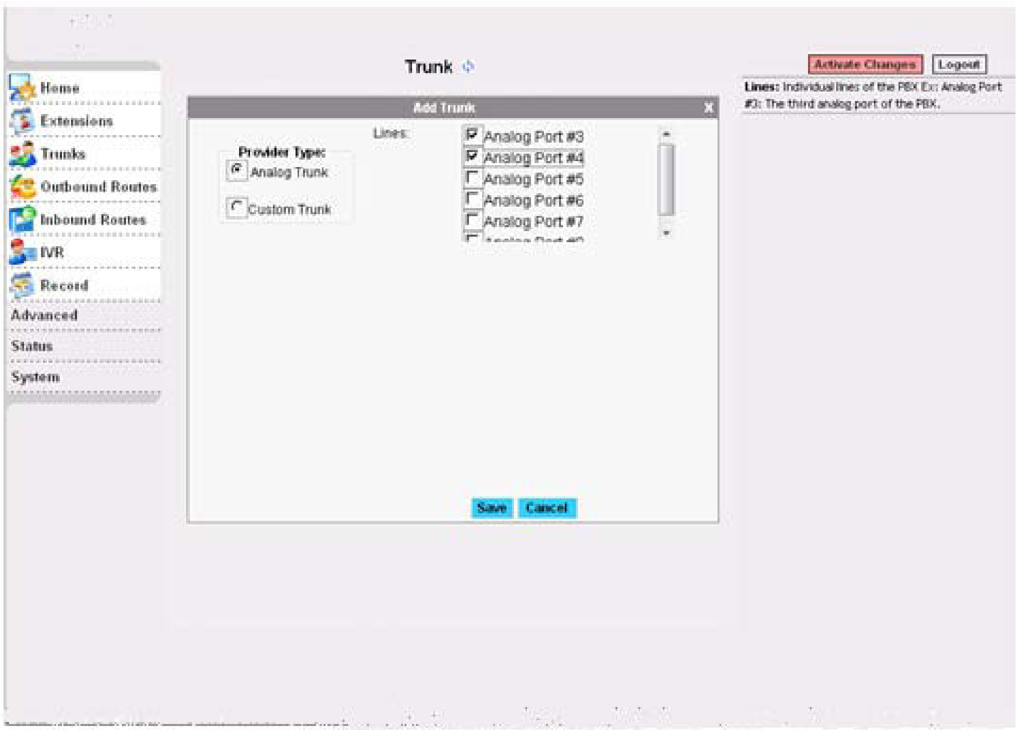
**FXS** (Foreign eXchange Station) is an interface which drives a telephone or FAX machine. FXS interfaces get phones plugged into them, delivery battery, and provide ringing. FXS interfaces are signalled with FXO signalling. **FXO** (Foreign eXchange Office) is an interface that connects to a phone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

## Make call via PSTN trunk

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

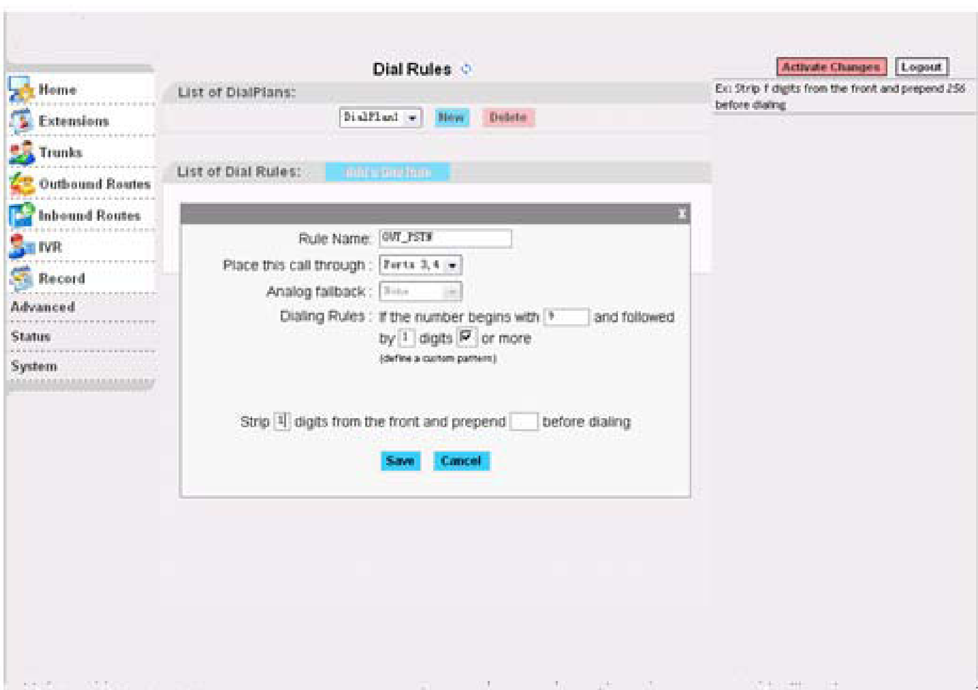



**Add Analog Trunk**
Trunks -> Add a Trunk:



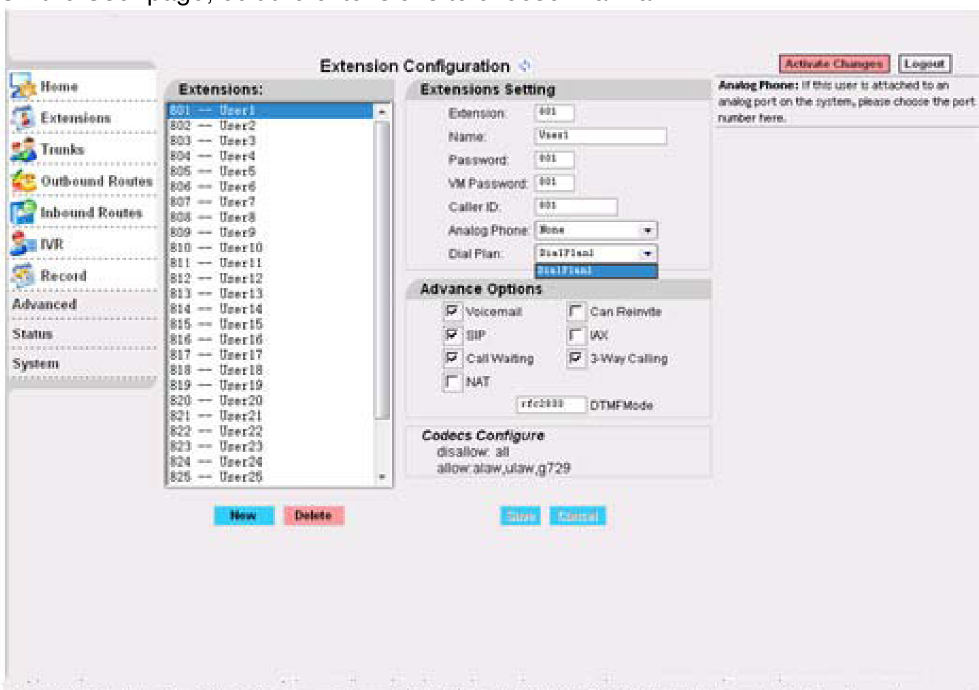

## Add Outbound Routers

In Outbound Routers -> add a Dial rule as below
Dial Rules




We have now added a Dial rule "OUT_PSTN" in the "DialPlan1".
As we can see from the dialing rule of "OUT_PSTN", all numbers start with 9 will be cut the first digit ('9') and sent to PSTN (port3 or port4).

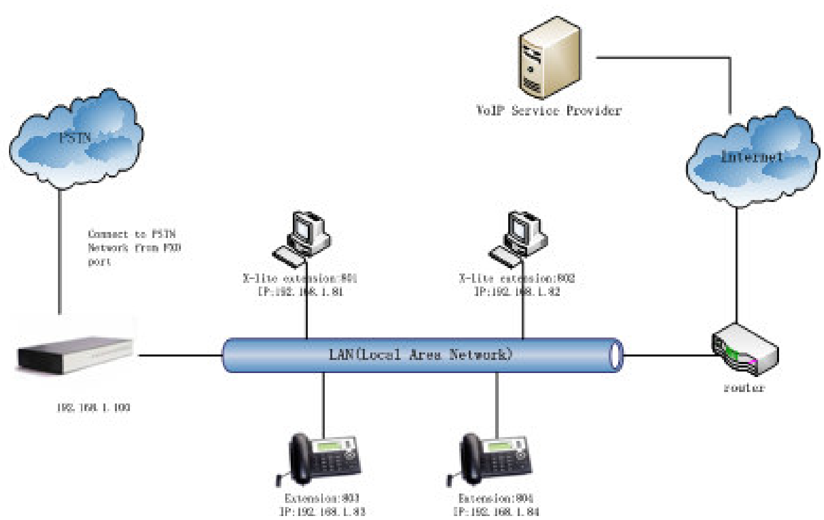
**Choose Dial Plan for extensions:**
On the User page, edit the extensions to choose DialPlan1.



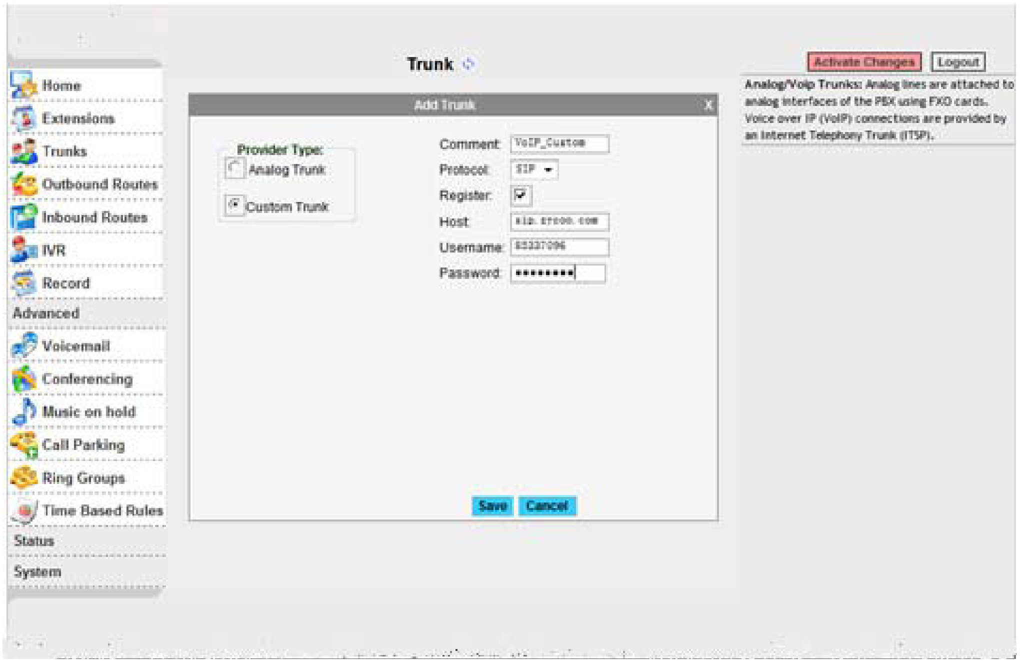

After we have done above, in the extension we can dial 9 + local number to dial out via PSTN line.

# Make call via VoIP trunk

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



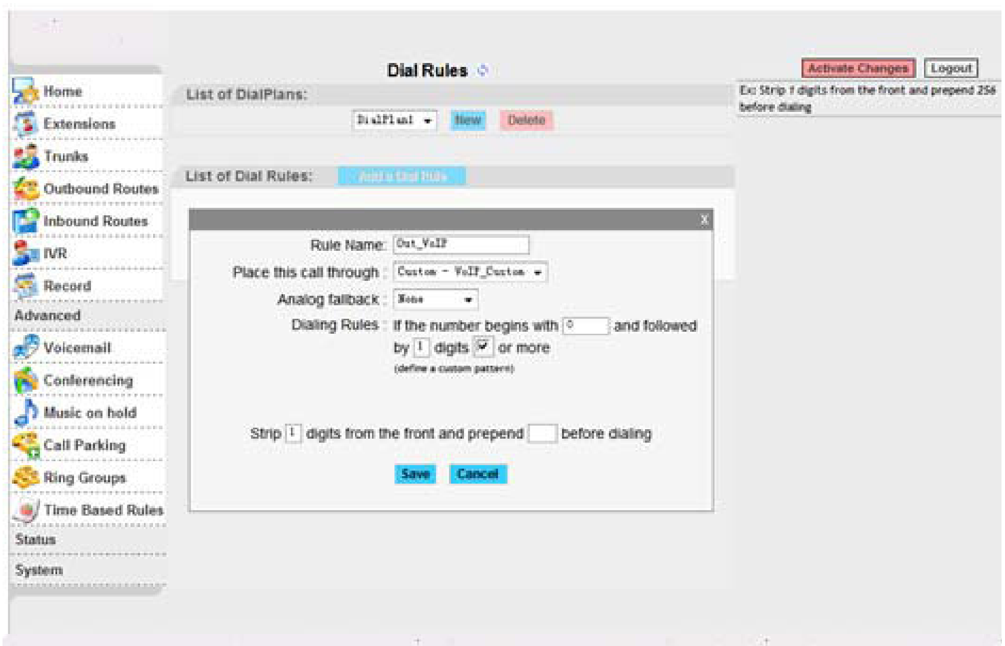

**Add VoIP service provider**
Trunk -> Add a Trunk:
Add a Custom Trunk.




## Add Dial Rule

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

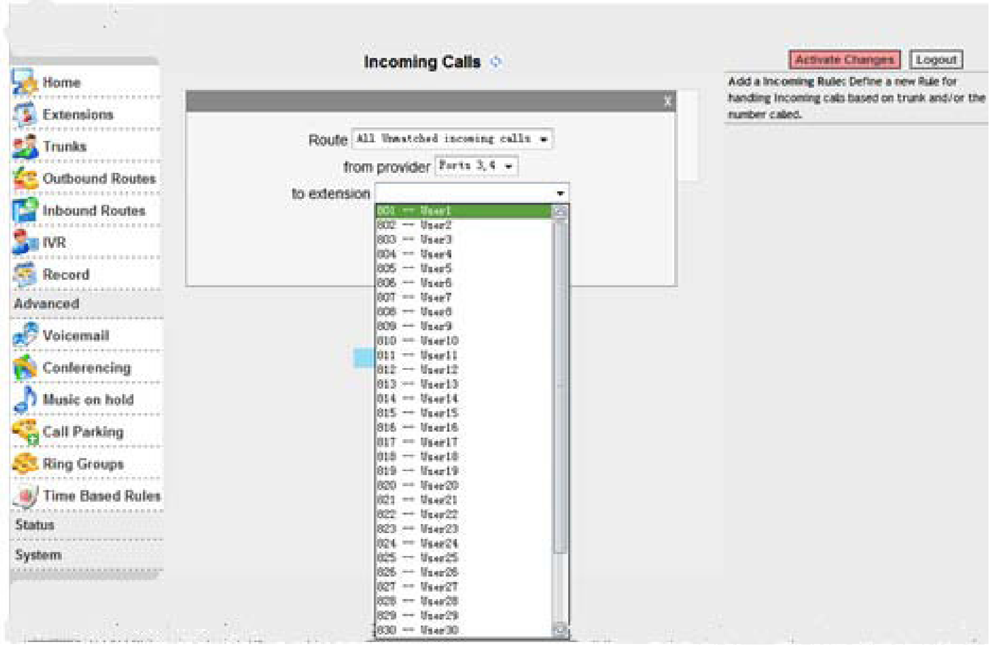
*Dial Rules*




Now we have added a new calling rule "Out_VoIP" in the "DialPlan1".
As we can see from the "Out_VoIP" dialing rule, all numbers start with 0 will be cut the first one digits ('0') and sent to my sip service provider. The Out_PSTN is in the same DialPlan1. Since we have added this dial plan to the extensions in above, we don't need to add dial plan again. So when we have added two calling rules, any call start with 9 will be route to PSTN, and call starts with 0 will be route to VoIP.

## 6.5 How to make an incoming call

Add an Incoming call.

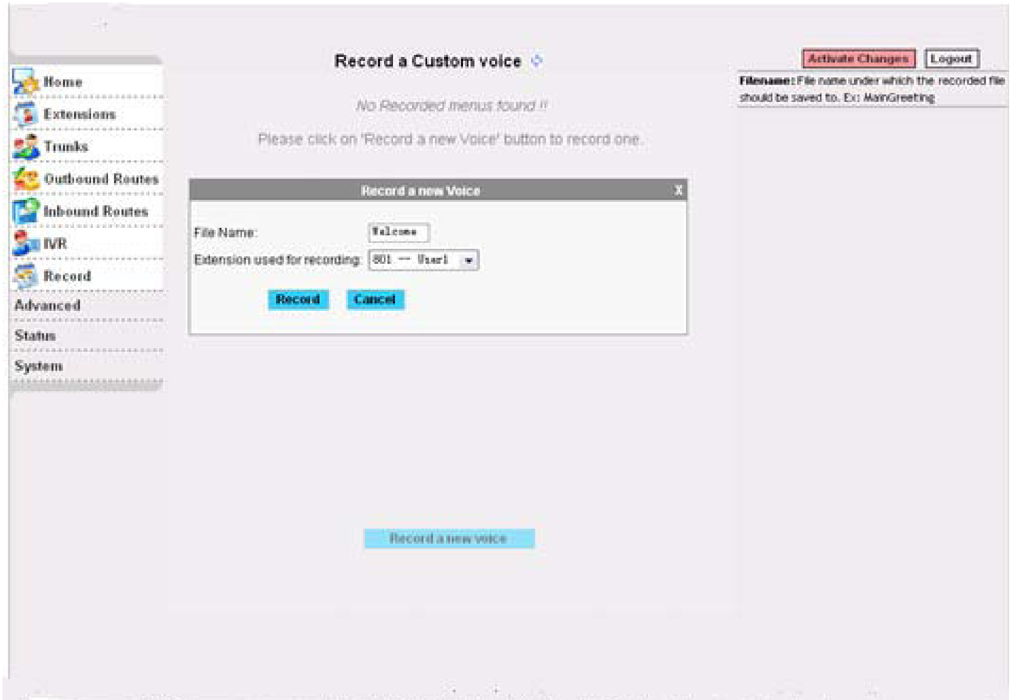



Select Route "All Unmatched incoming calls"
From provider "Port 3, 4"
To extension "801 – User1" (here, you can select a extension, a IVR or others)
Then, if there is incoming call from Port3 or port4 channel, the extension 801 will ring.

## 6.6 How to Set an incoming call to IVR based time rule

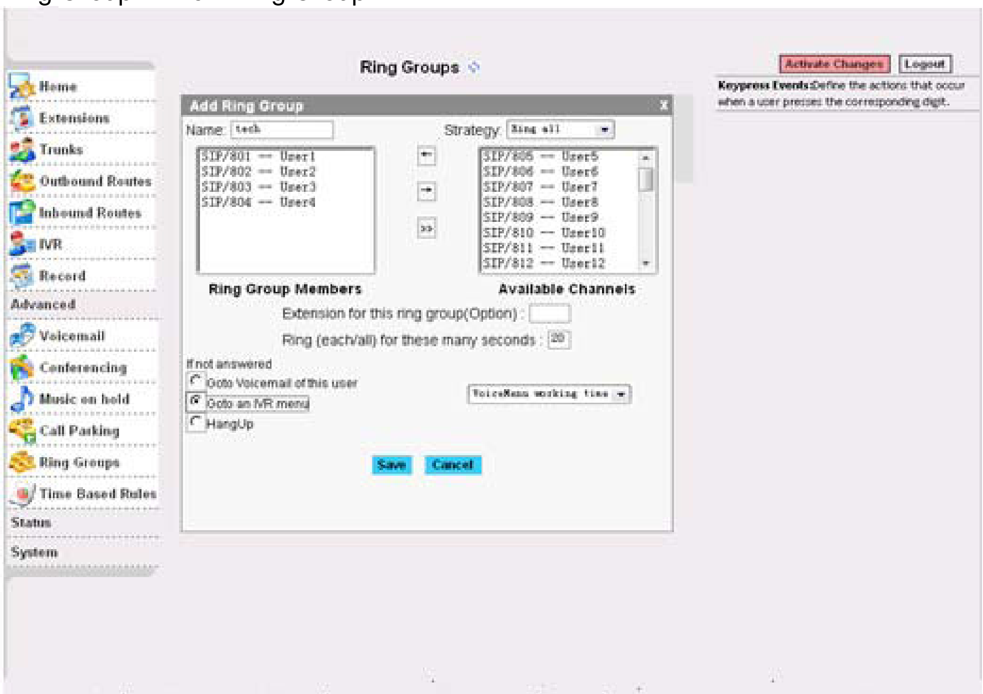
**Add record a custom voice**

Record -> Record a new voice

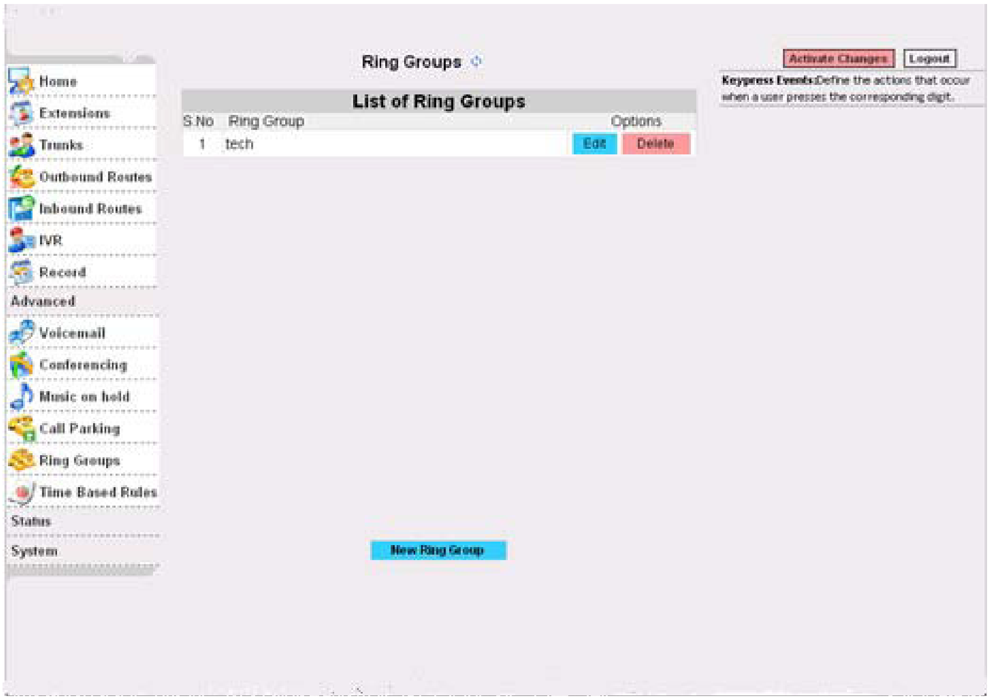



Set the record name is “Welcome”
Choose a extension used for recording, here we use EXT 801
Click Record button
Then, the extension 801 will ring
Pick up the phone record “Welcome” message
Then hangup and finish the record .
Use the same way to record “Closing” message

**Add a Ring Group**
Ring Group -> New Ring Group

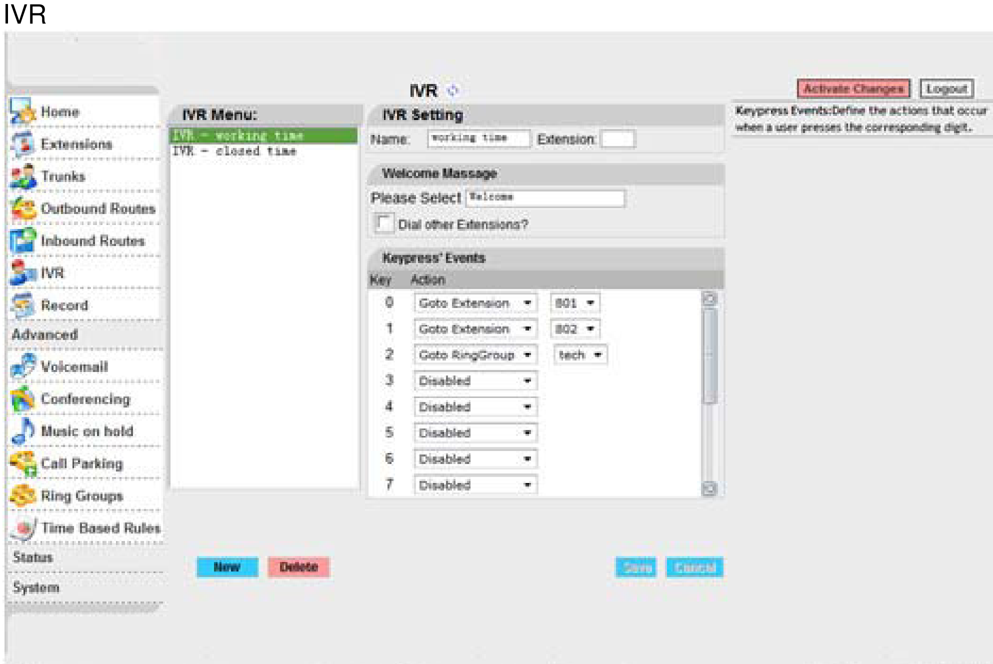



Example:
Name the ring group “tech”
Choose the group members whose extensions are “801. 802, 803, 804”
“if no answered”, choose “goto IVR”-- “working time”
Click “Save” button

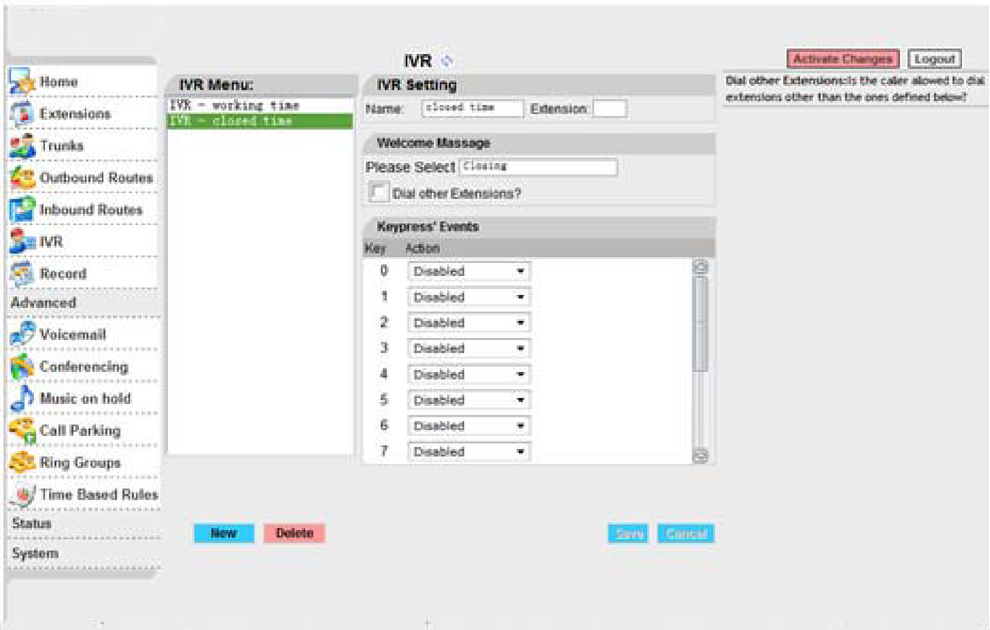



**Set IVR**

IVR



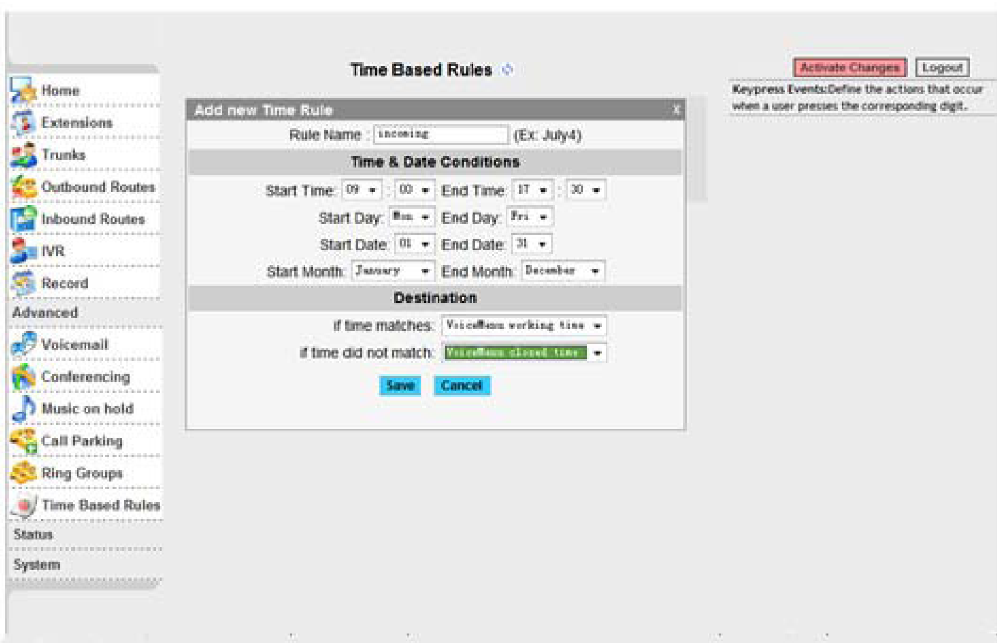

Select IVR-working time, Set welcome massage is “Welcome”
Set keypress’ Events
Dial “0” go to extension 805
Dial “1” go to extension 806
Dial “2” go to ringgroup tech
Click Save button



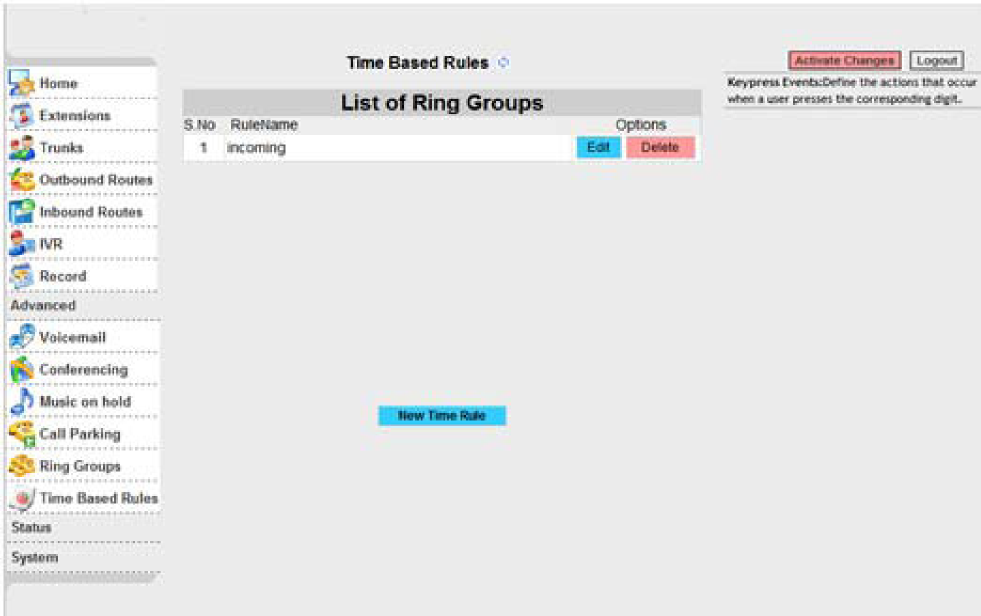

Then set IVR-closed time
Set welcome massage is “Closing”

## Add a Time Rule

Time Based Rules -> New Time Rule

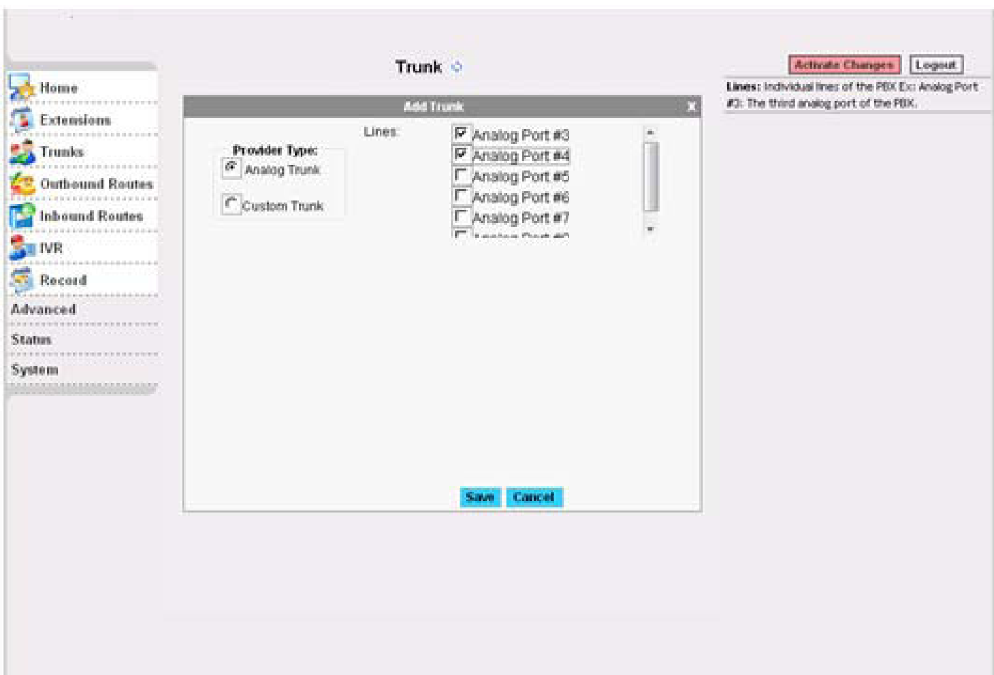



Set a Rule Name, eg: incoming
Set the Time & Date Conditions
“If time matches” --- go to “working time”
“If time not match” --- go to “closed time”
Click the save button, saved the configuration



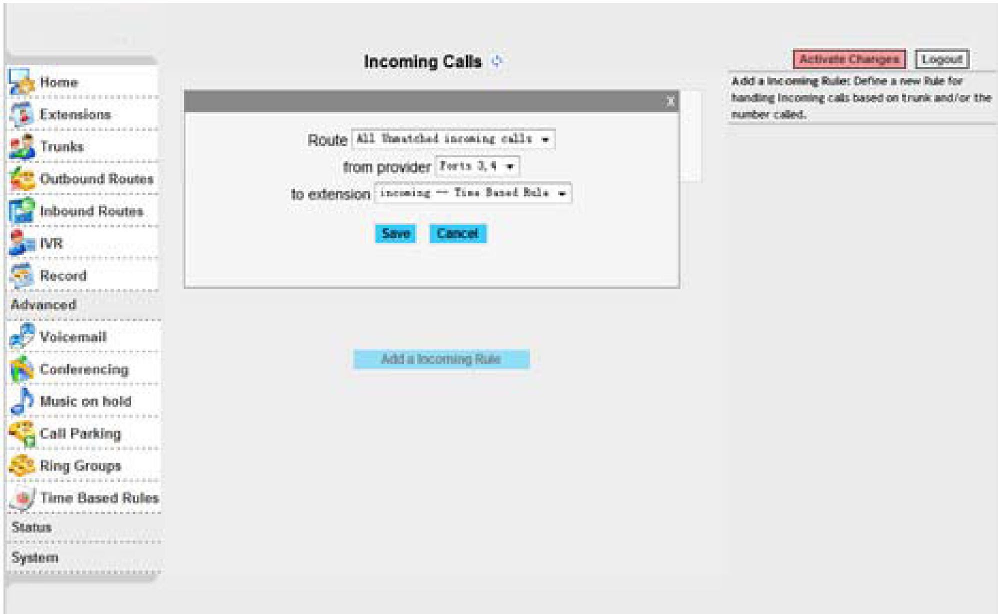

**Add a Trunk**

Trunks -> add a Trunk

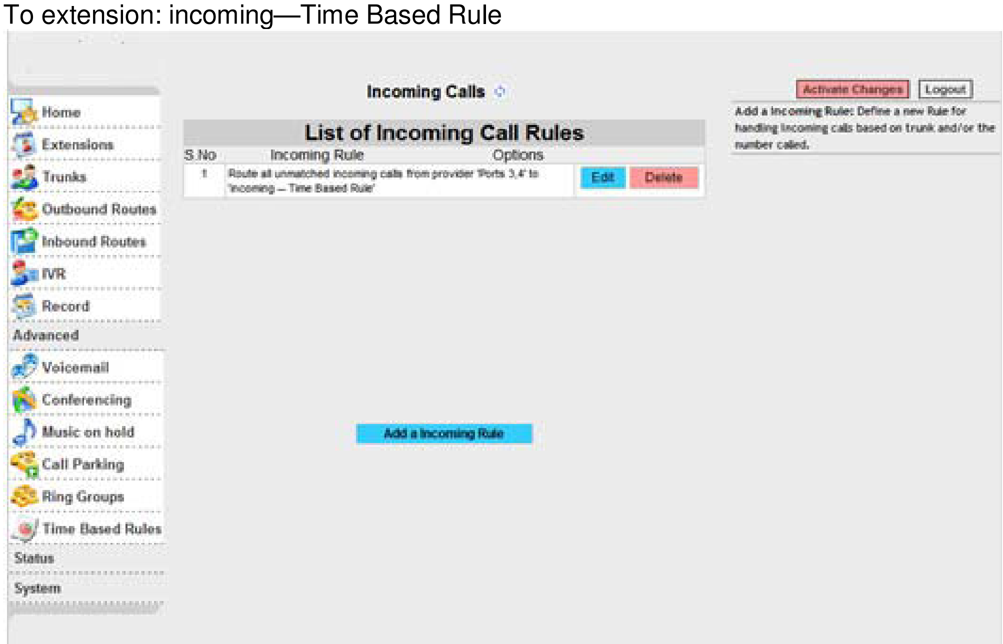



**Add an incoming router**

Inbound routers -> add an incoming rule



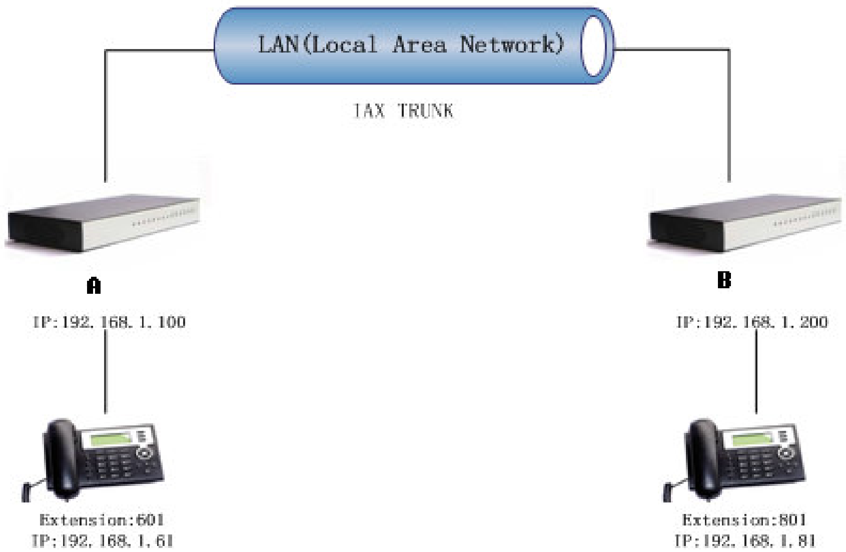

Select Route: All Unmatched incoming calls
From provider: Ports 3, 4
To extension: incoming—Time Based Rule




## 6.7 Link two IPPBX in the same network

The simplest case to link two IP PBX together in the same network. We start from this and then try to expand to different network. We use PX0522 here, same method for other model IPPBX products.

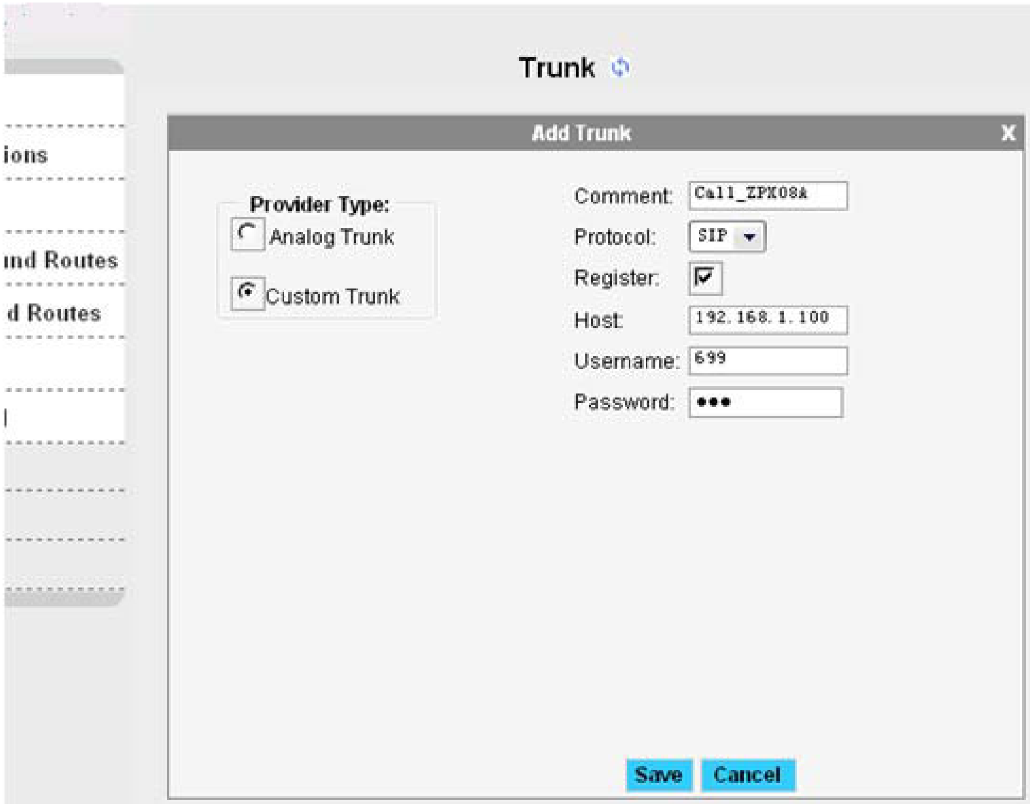
Below is the structure of how to link two IPPBX in the same LAN:



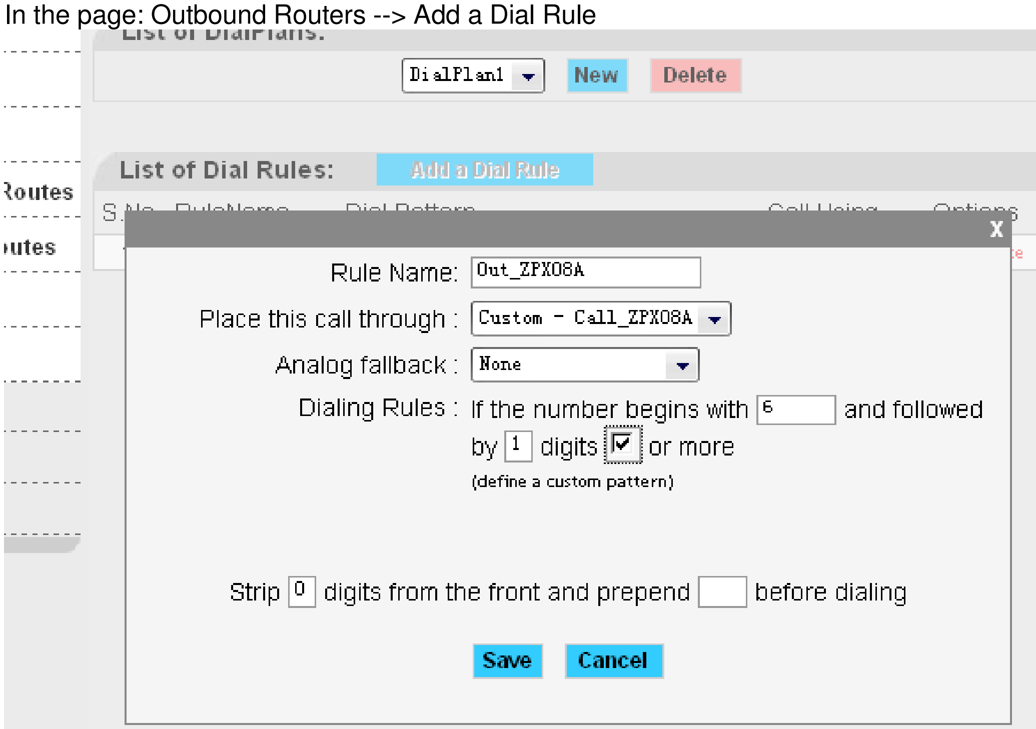

The method of connecting two PX0522 in different location is:
1) Register the PX0522-A as an extension in PX0522-B(via IAX2 trunk),so the extensions in PX0522-A can make calls to PX0522-B's extensions via this "special" trunk.
2) Use the reverse method in V PX0522-B to register to PX0522-A.
In above structure:
1) IP0021A registers to PX0522-A as an extension 601.
2) IP0021B registers to PX0522-B as an extension 801.
3) All the extensions under PX0522-A are in the format 6XX.
4) All the extensions under PX0522-B are in the format 8XX
5) Extensions under PX05220-A can make calls to extension under PX0522-B use format 8XX.
6) Extensions under PX0522-B can make calls to extension under PX0522-A use format 6XX.
7) The two PX0522 links each other via IAX2 trunk.

**Step 1: Set up a extension 699 in PX0522-A**
Extension: 699 ; Phone number of this extension
Name: PX0522B_Users ;
Password: 699 ;IAX2 Log on password
Caller ID: 699 ; Caller ID
Advance Options: Select IAX protocol

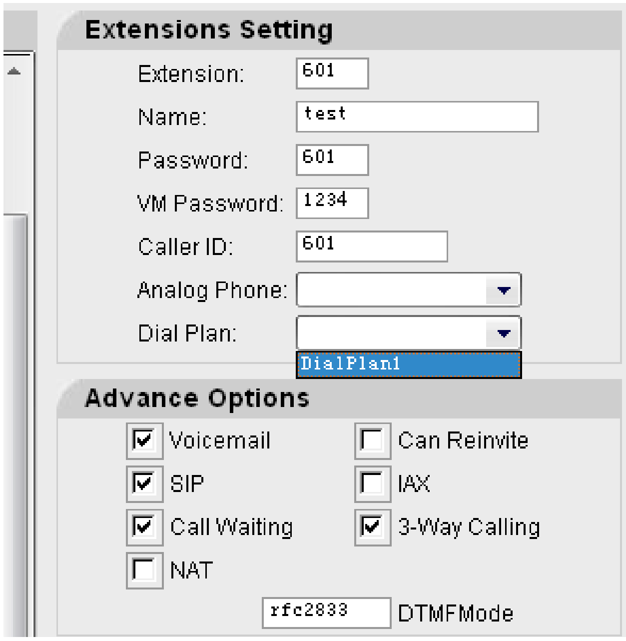
**Step 2: Set up an IAX trunk in PX0522-B to link to PX0522-A via this PX0522B_Users extension.**
In the page Trunks--> Add a Trunk




**Step 3:** Set Dial Rule in PX0522-B, all calls start with 6 will be sent to PX0522-A.
In the page: Outbound Routers --> Add a Dial Rule




**Step 4:** Set the user Dial Plan in PX0522-A,
In the page: Extensions, Dial Plan



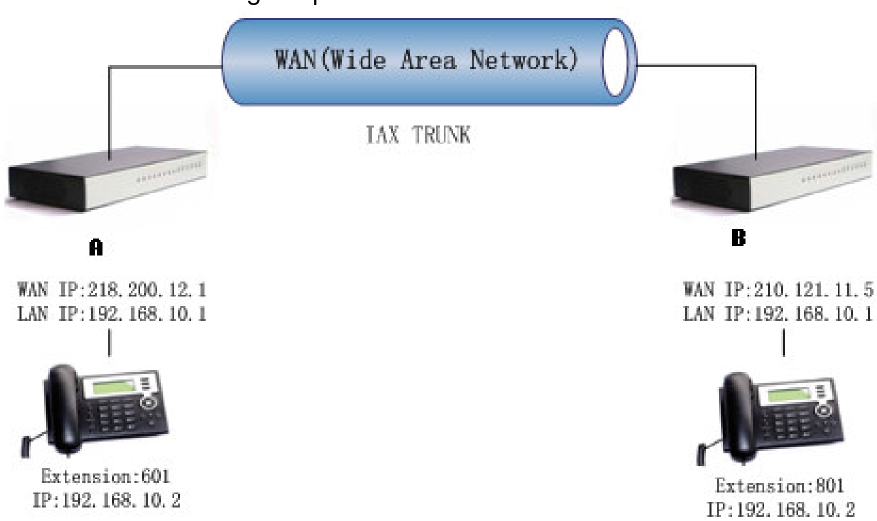

Active the change and apply the test:
1. Register an IP phone IP0021B to PX0522-B with 801 extension.
2. Register an IP phone IP0021A to PX0522-A with 601 extension.
3. Use 801 to dial 601. And you can see 601 will ring and you can pick up the calls.

Above is the way to router PX0522-B's call to PX0522-A, the method to link PX0522-A to PX0522-B is the same as above.

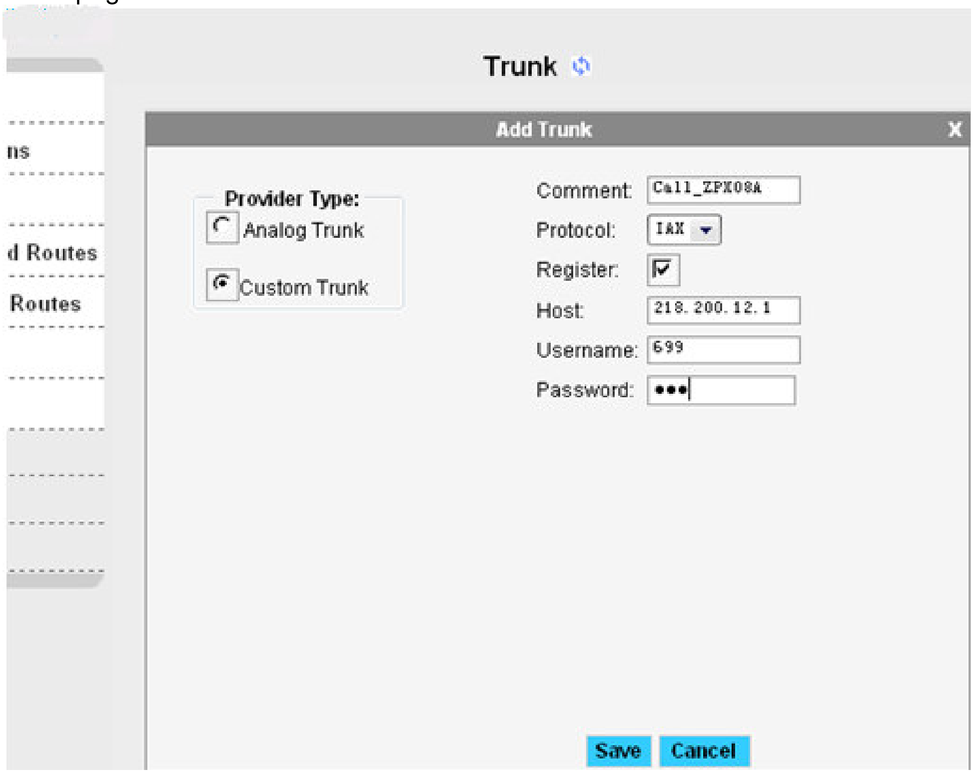
## 6.8 Link two IPPBX in different network

**Two PX0522 are connected via the internet**

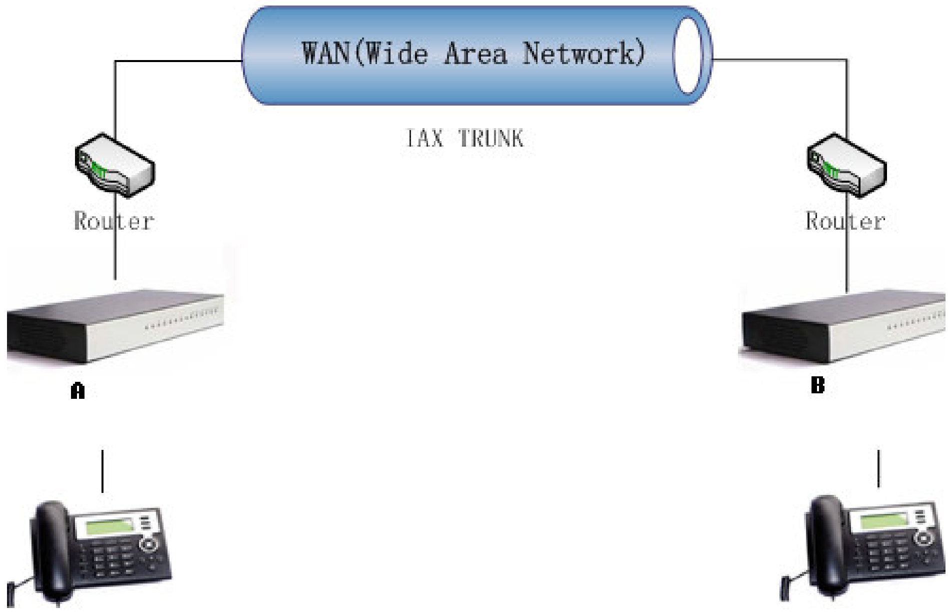
The generally environment for two PX0522 in different location is: two PX0522 are both in the internet and using the public IP.



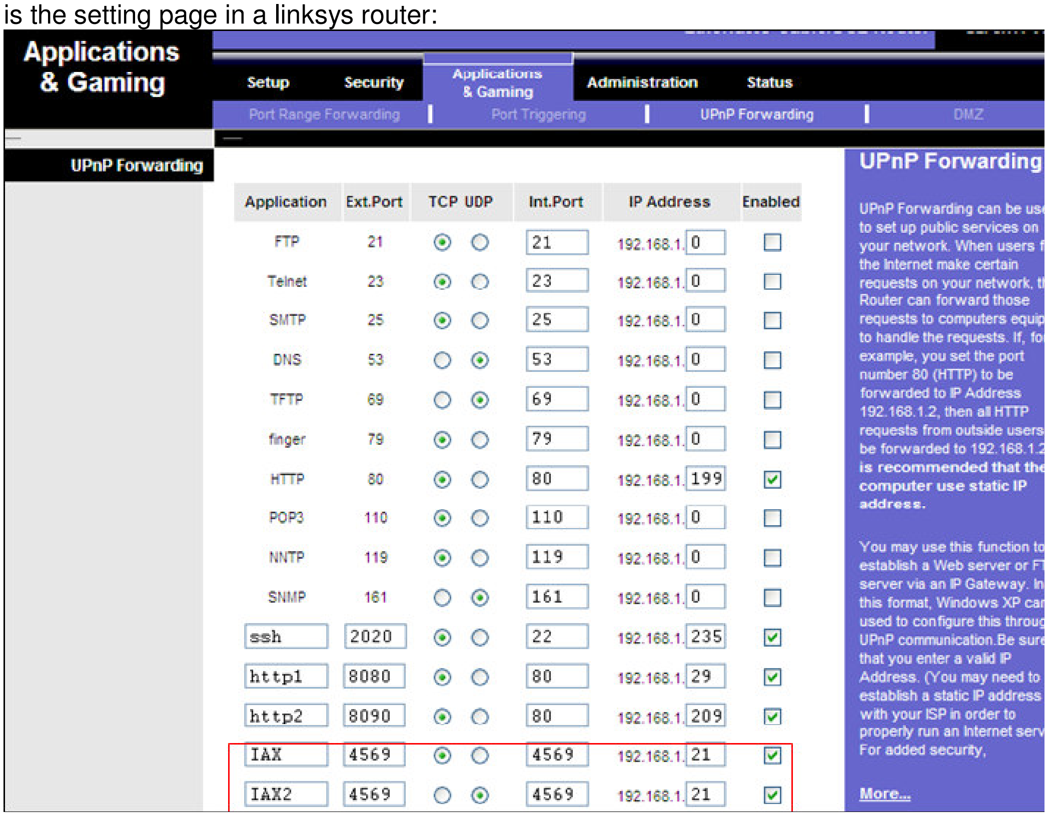
The configuration is same with “Link two V PX0522 in the same network”.
But when you set the trunk, you must use the public ip.
Like the follow:
Set up an IAX trunk in PX0522-B to link to PX0522-A via this PX0522B_Users extension.
In the page Trunks--> Add a Trunk




**Two PX0522s are behind router.**
The generally environment for two PX0522 in different location is: two PX0522 are both behind router and using the private IP.
Since the PX0522 doesn’t have the public IP, we need to do port forwarding in the router and make PX0522 is reachable to others.




**Step 1:** Set port forwarding in the router for PX0522-A
The PX0522-A is behind the router, to register to PX0522-A via the internet, you need to forward the IAX2 port in your router, so all the packets received on the router WAN port (210.11.25.127:4569) will be forwarded to the PX0522-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 | ⦿ | ○ | 21 | 192.168.1.0 | ☐ |
| Telnet | 23 | ⦿ | ○ | 23 | 192.168.1.0 | ☐ |
| SMTP | 25 | ⦿ | ○ | 25 | 192.168.1.0 | ☐ |
| DNS | 53 | ○ | ⦿ | 53 | 192.168.1.0 | ☐ |
| TFTP | 69 | ○ | ⦿ | 69 | 192.168.1.0 | ☐ |
| finger | 79 | ⦿ | ○ | 79 | 192.168.1.0 | ☐ |
| HTTP | 80 | ⦿ | ○ | 80 | 192.168.1.199 | ☑ |
| POP3 | 110 | ⦿ | ○ | 110 | 192.168.1.0 | ☐ |
| NNTP | 119 | ⦿ | ○ | 119 | 192.168.1.0 | ☐ |
| SNMP | 161 | ○ | ⦿ | 161 | 192.168.1.0 | ☐ |
| ssh | 2020 | ⦿ | ○ | 22 | 192.168.1.235 | ☑ |
| http1 | 8080 | ⦿ | ○ | 80 | 192.168.1.29 | ☑ |
| http2 | 8090 | ⦿ | ○ | 80 | 192.168.1.209 | ☑ |
| IAX | 4569 | ⦿ | ○ | 4569 | 192.168.1.21 | ☑ |
| IAX2 | 4569 | ○ | ⦿ | 4569 | 192.168.1.21 | ☑ |

**Step 2.** Set up the service provider and calling rule in PX0522-B to make it register to PX0522-A. This method is almost the same as above, EXCEPT you need to use the 210.11.25.127 as the service provider instead of 192.168.1.21.

**Step 3.** Use the same method do port forwarding in router-B for PX0522-B. Your public address from network provider maybe a dynamic ip which will be changed periodically. To overcome the problem of dynamic ip, you may need to use the DDNS service , for more info please google the internet.