

Manual

PBX IP 2008



Version: 1.0

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Chapter 1 the Introduction of PBX-IP 2008

Overview of the PBX-IP 2008

The PBX-IP 2008 is a complete Asterisk Appliance with four Dual ports FXO or FXS modules. It is an embedded open source Linux system with built-in SIP/IAX2 proxy server and NAT features. It provides a solid, uniform platform for traditional PSTN communications as well as VoIP communications.

Targeting for SOHO user and SMB market with an easy to use graphical interface, PBX-IP 2008 provides a cost-saving solution on their telecommunication/data needs. With PBX-IP 2008, company with branch offices in different countries can be easily combined together to work like a virtual single office through internet.

Features

- Open Source Asterisk IP PBX
- High performance OSLEC (Open Source Line Echo Celler)
- Configurable IVR menu Voice Mail,
- Voicemail to Email Call forward,
- Call waiting,
- Call transfer
- Call conference
- Call queues, Ring group
- SIP trunk, IAX trunk, PSTN analog trunk
- Call Detail Record
- Access via: SSH/telnet/web
- Firmware upgradable via web page
- 50+ available SIP/IAX2 extensions
- 20 concurrent calls

Applications

- SOHO/SMB telephony system
- Hosted service
- FAX terminal
- IVR system

Interface

- 2 * RJ45 port
- 1 * Power port
- 1 * MMC/SD slot
- 8 * RJ11 port (FXS/FXO interchangeable)
- 4 * Dual port FXO/FXS module slot
- 1 * USB port

Hardware

CPU: 400MHz Blackfin 532 Chip

Agricultura 111 Piso 1, Colonia Escandón, C.P 11800 México DF 55 52714421, 52774459, 52719163
Fax: 52718216. ventas@ansel.com.mx

Eight analog (FXO/FXS) module interface.
NAND flash 256 MB
SDRAM 64M

System

Open Source uClinux

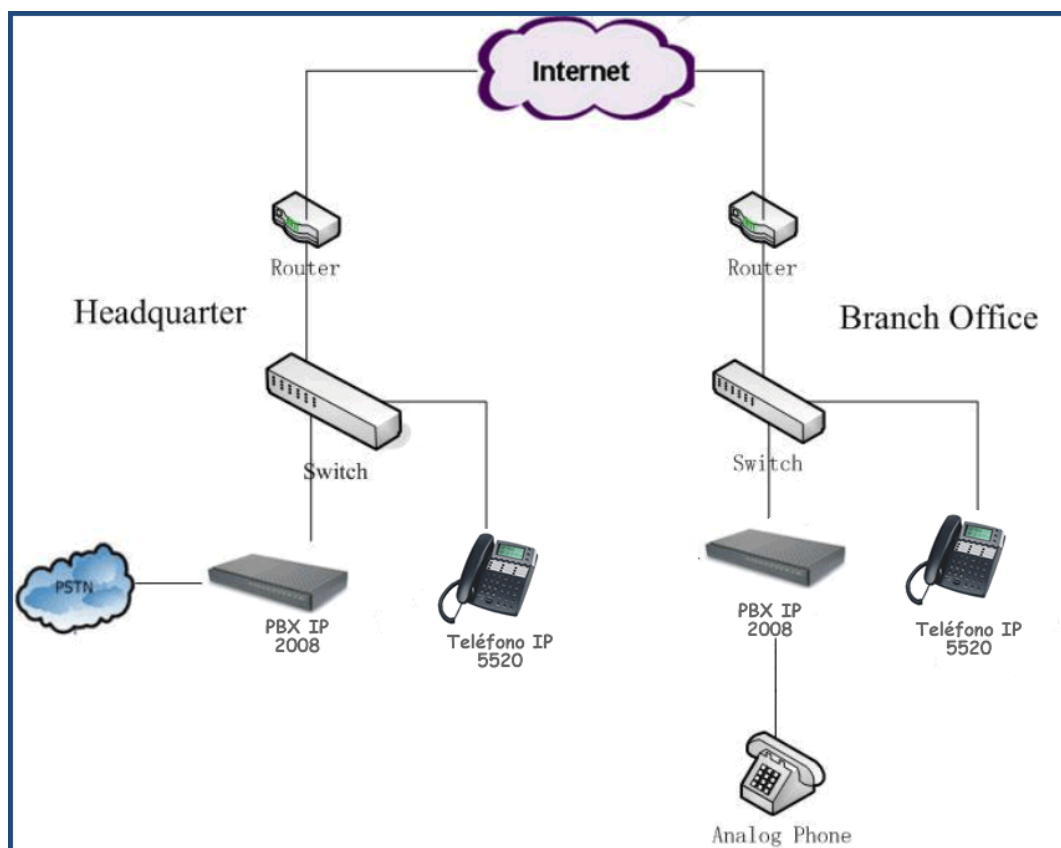
Measurement and Weight

Inner box	225 * 120 * 30mm
G.W./unit	0.79KG
Carton MEAS	456 * 442 * 362 mm
Units per Carton	21 units/ CTN
G.W./CTN	18 KG/CTN

Package

Item	Quantity
PBX-IP 2008	1
RS232 module	1
Power Adapter	1
Manual (disk)	1

For the usage of PBX-IP 2008 in VoIP field, you can refer to the following network topology.



Chapter 2 Access to the PBX-IP 2008

You need a PC to access to the PBX-IP 2008, there are four ways for you to access the PBX-IP 2008:

1. Web page access by browser
2. SSH access by putty
3. Access by browser with Fallback IP Address
4. Console port access by RS232 console cable

In order to access to PBX-IP 2008 by the first three ways, you have to check that if your network connection between PBX-IP 2008 and PC is OK. If you do not have network connection between PBX-IP 2008 and PC, you can try to use the last way to access to PBX-IP 2008 and change the IP address for IP-08.

2.1 Web Page Access by Browser

It is the most convenient and common way to access the PBX-IP 2008, you just need to open your browser and input the IP address of PBX-IP 2008 WAN port (the default IP address is 192.168.1.100). You would better use Firefox instead of IE, because there are compatible issues. Then input the default Username: admin; Password: atcom (the password of old version is mysecret or could be also anselmex) in the presented screen like the following:

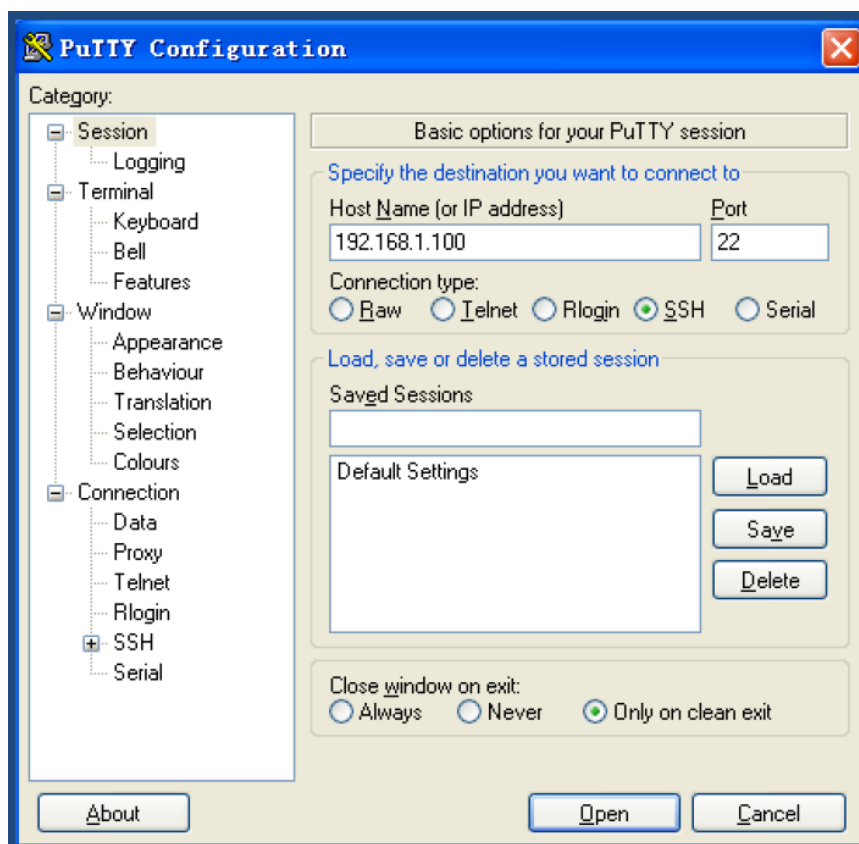
When you login successfully, you can get the configuration web page as below:

Extension	Name/Label	Status	Type
4150	Check Voicemails		VoiceMailMain
-- No Extension assigned			
	Dial by Name		Directory

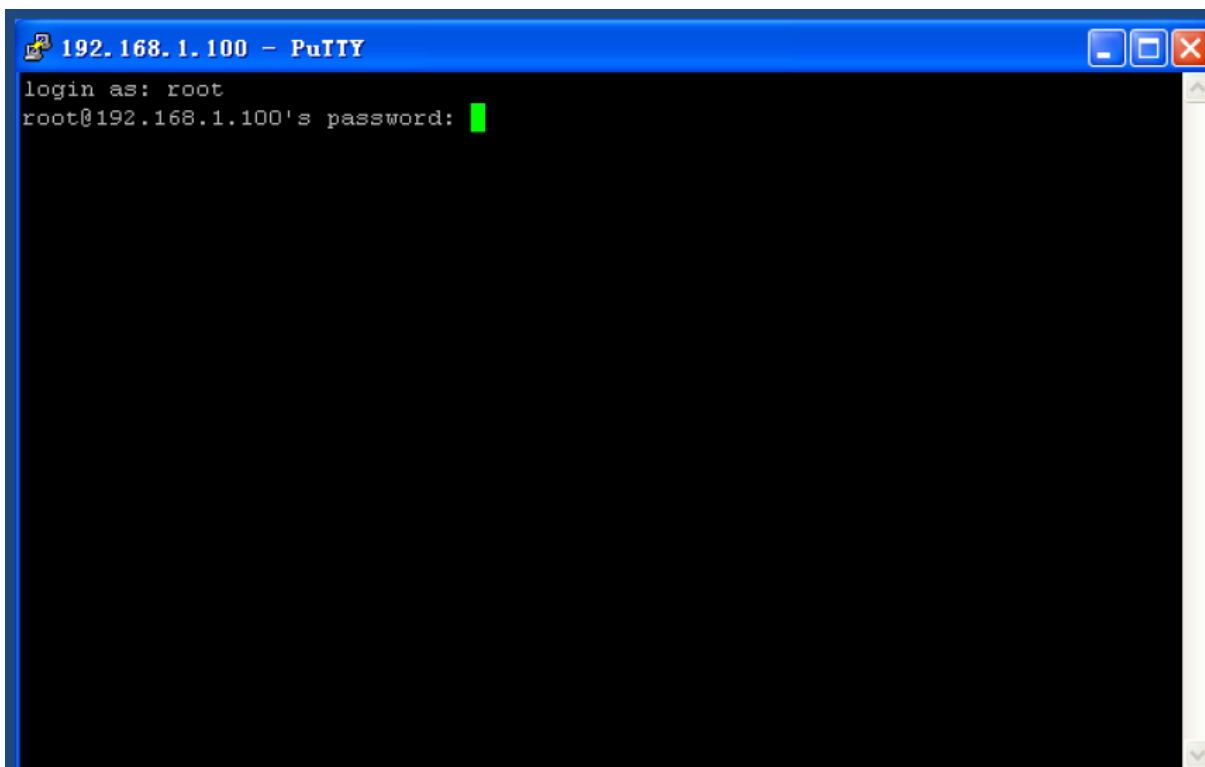
2.2 SSH Access by Putty

Logging into PBX-IP 2008 by SSH, you can configure PBX-IP 2008 by Linux command.

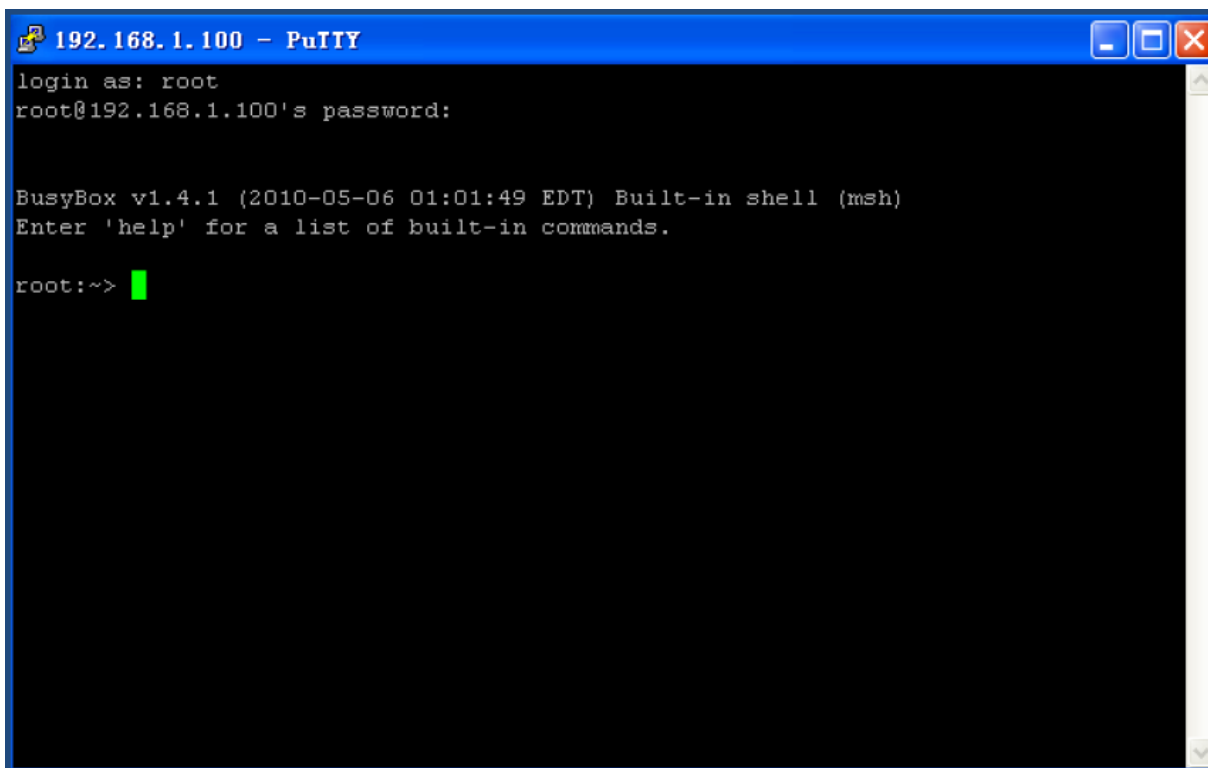
1) Please open your putty software, and input the PBX-IP 2008 IP address in the **Host Name** textbox, input port number in the **Port** textbox, click the **SSH** Connection type, then click **open** button. Please refer to the following screen:



2) Please input username: root, and the default password: 12xerXes16 in the following screen, you can access to PBX-IP 2008 successfully.



When you log into PBX-IP 2008 successfully, you can get the following illustration:



The image shows a PuTTY terminal window titled "192.168.1.100 - PuTTY". The terminal output is as follows:

```
login as: root
root@192.168.1.100's password:

BusyBox v1.4.1 (2010-05-06 01:01:49 EDT) Built-in shell (msh)
Enter 'help' for a list of built-in commands.

root:~> █
```

2.3 Access by Browser with Fallback IP Address

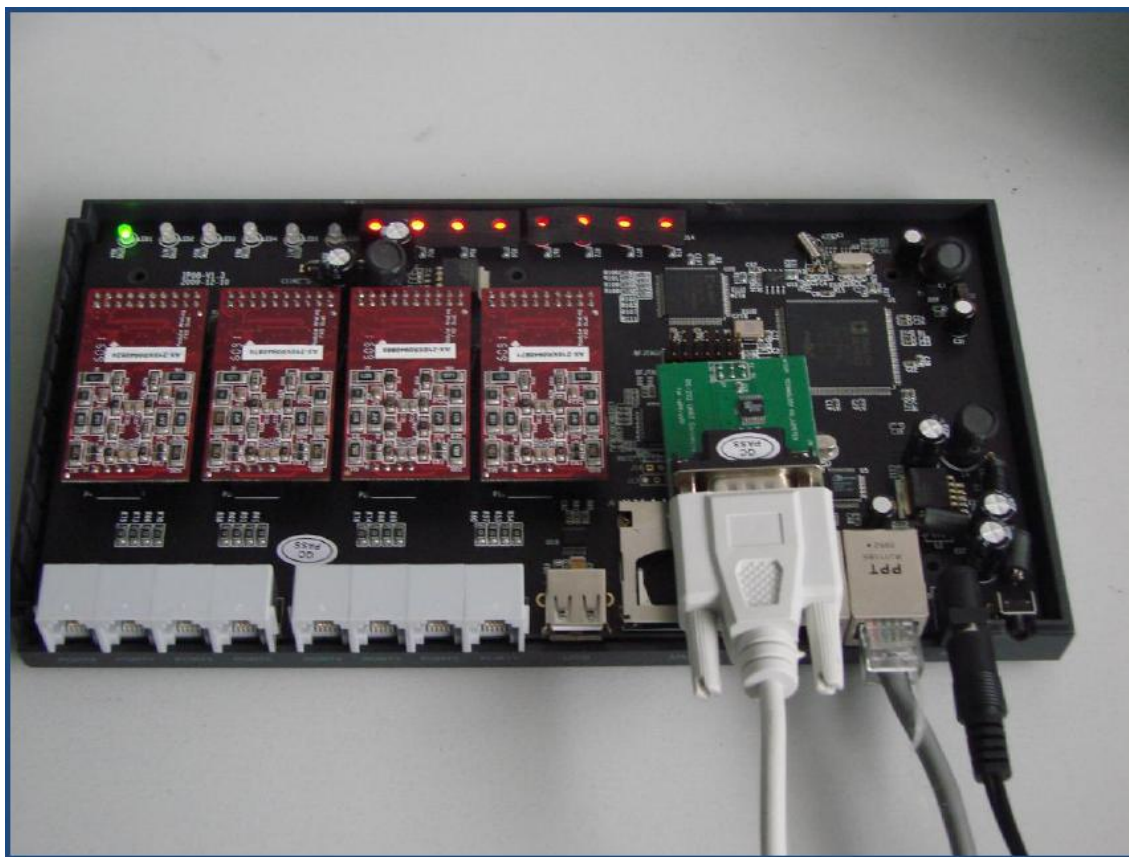
This way only be supported by the latest version (PBX-IP 2008-0.3.6) of PBX-IP 2008. If you forget the IP Address of PBX-IP 2008 you have set up, you can use the fallback IP Address: 172.31.255.254/30. Before logging into PBX-IP 2008, please set up the IP Address of your PC: 172.31.255.253 and SubMask:

255.255.255.252. At last, you can open your browser and enter:172.31.255.254 to log into the web page of PBX-IP 2008.

2.4 Console Port Access to PBX-IP 2008

If you do not have network connection between PBX-IP 2008 and PC, you can try to access to PBX-IP 2008 by console port. Please try to do as the following steps:

1. Please connect the console port of PBX-IP 2008 to your PC's console port with RS232 console cable, you can refer to the following illustration:



2. Please run your Hyper Terminal, and set up the console port like the following:

- Bits per second: 115200
- Data bits : 8
- Parity: None
- Stop bits: 1
- Flow control: None

Change the IP Address by Hyper Terminal

The default IP address of IP-08 is 192.168.1.100. Your network may have a different IP address range such as 192.168.10.xx. In this situation, you can not access to IP-08 by putty and browser if you do not change the IP-08 IP address. So you have to change the IP address for IP-08 by Hyper Terminal to make it in the same network segment as your LAN.

After you have accessed to IP-08 by Hyper Terminal, please use the following command to change the IP address for IP-08.

```
root:~> ifconfig eth0 192.168.1.151(the IP address you want to set for IP-08)
```

By this way, the IP address you set for IP-08 is temporary, it will recover to the original default IP address after rebooting. If you want to give a static and permanent IP address for IP-08, you can try to set it in web GUI, for detail steps please refer to chapter 3.

Chapter 3 Configure PBX-IP 2008 by Web GUI

3.1 System Status

In the system status screen, it displays the functions you configured, such as: trunks, extensions, conference and so on like the following screen:

ATCOM Uptime : 05:17:03 up 2 min, load average: 0.18, 0.06, 0.02 Logout

System Status
Please click on a panel to manage related features

- Configure Hardware
- Trunks
- Outgoing Calling Rules
- Dial Plans
- Users
- Ring Groups
- Music On Hold
- Call Queues
- Voice Menus
- Time Intervals
- Incoming Calling Rules
- VoiceMail
- Conferencing
- Follow Me

Trunks

Status	Trunk	Type	Username	Port/Username/IP
Unregistered	siptrunk1	sip	6036	192.168.1.20
	trunk2	Analog		Ports 2

Agents

6001
Login

6002
Login

6003
Login

6005
Login

6020
Login

Conference Rooms

6300
Not in use

Extensions

Free Busy Unavailable Ringing

Extension	Name/Label	Status	Type
6001	6001	Messages : 1/0	SIP/IM User
6002	6002	Messages : 0/0	SIP User
6003	6003	Messages : 0/0	SIP User
6004	6004	Messages : 0/0	Analog User (Port 3)

3.2 Configure Hardware

In the configure hardware page, it includes the following components: analog hardware, tone region, advanced settings.

Analog Hardware

When you boot the PBX-IP 2008, which will detect the FXO and FXS modules automatically, the analog hardware component displays the modules which are detected correctly.

Tone Region

You should select the tone region according to your country, if it does not have your country's name in the dropdown list, please ask your service operator which kind of tone region is used in your area.

3.3 Trunks

To receive calls from PSTN and make calls to the outside world, you have to use trunk. Please select the **Trunks** option from the vertical menu on the left of the main page, then you can get the following screen:


3.3.1 Create Analog Trunks

Analog trunk is associated with FXO port, and it will call outside by PSTN line. Click on **New Analog Trunk** in the illustration above, the pop-up screen is where you create and set up trunk.

There are many parameters for you to set up, I just set the following two parameters:

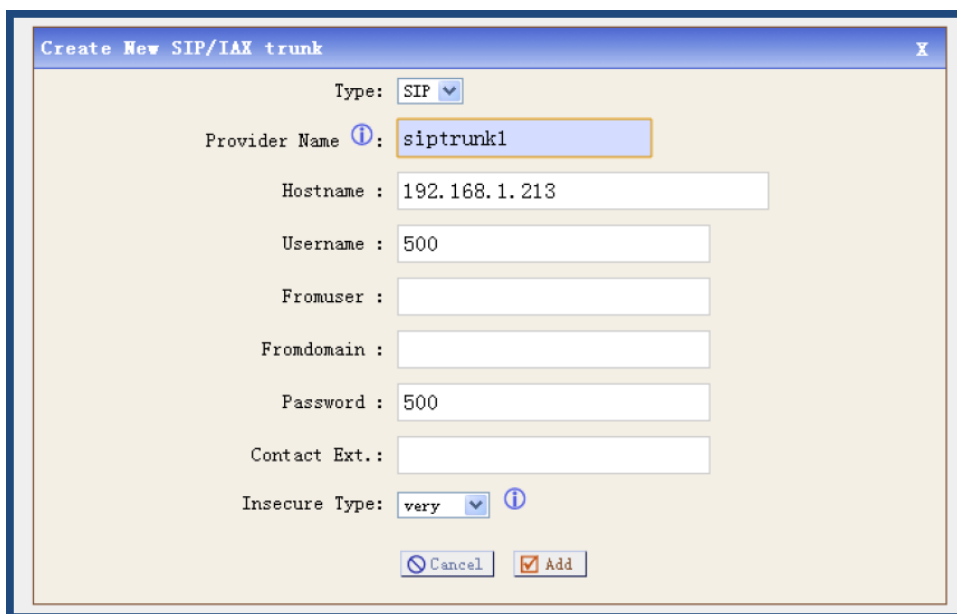
- **Channels:** select the FXO port you want to use. Here I use the port 2.

- **Trunk Name:** a unique label to help you identify the trunk when listed in outgoing calling rules and incoming calling rules. Here I use the trunk2 as my trunk name.

For the advanced options, you can put your cursor on the  label, you can get the information of the parameter, customers have to set these parameters according to your service provider and your need.

3.3.2 VoIP Trunks

A VoIP service provider (VSP) that you have signed up with is also a trunk. Via the VoIP trunk you can dial via the VoIP service to reduce your cost when making international calls. You can set up the VoIP trunk to make calls to the PSTN or other VoIP network depends on the service you use. You can also use the VoIP trunk to link headquarter and branch offices for free internal calls. Click on **New SIP/IAX Trunk**, the following screen is where you create and set up VoIP trunk:

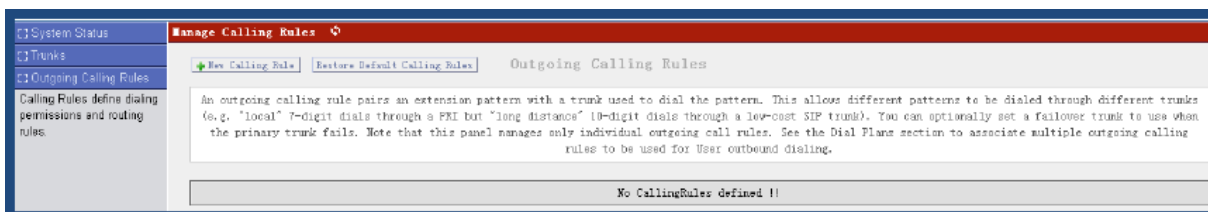


The important parameters are:

- **Type:** You can select SIP or IAX type to meet your need.
- **Provide Name:** a unique label to help you identify the trunk when listed in outgoing calling rules and incoming calling rules.
- **Hostname:** the IP address or domain name of your service provider's server.
- **Username:** the username that your service provider configured.
- **Password:** the password that your service provider configured for the user.

3.4 Outgoing Calling Rules


Outgoing calling rules is used to route an outgoing call, when you make an external call, which trunk and what dial-pattern the call used are configured in outgoing calling rules. Please select the Outgoing Calling Rules option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New Calling Rule** button on the illustration above, the following screen is where you create and set up outgoing calling rule:

The important parameters I configured are below:

- **Calling Rule Name:** a unique label to help you identify the outgoing calling rule when listed in dial plans, I use outgoing1 as the calling rule name here.
- **Pattern:** it acts like a filter for marching numbers you dialed, here I set up _2X., it means any number you dial out with prefix 2 will use this outgoing call rule.
- **Use Trunk:** select the trunk for outgoing calling rule, here I select the trunk2 I set up before.
- **Strip:** I press 1 here, it will strip the first number of the number string you dialed.

You can get the detail information about every single parameter by putting your cursor on the  label.

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

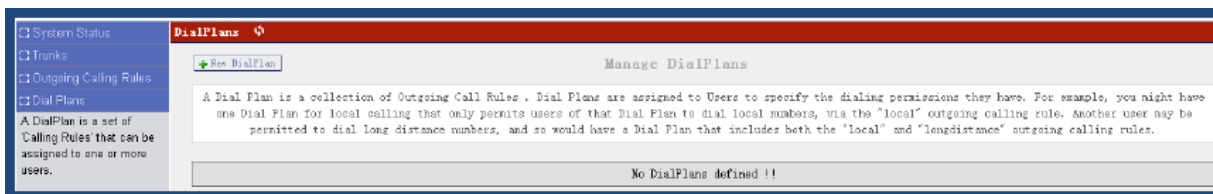
The way of outgoing calling rules works:

Every time you dial a number, asterisk will do the following in strict order:

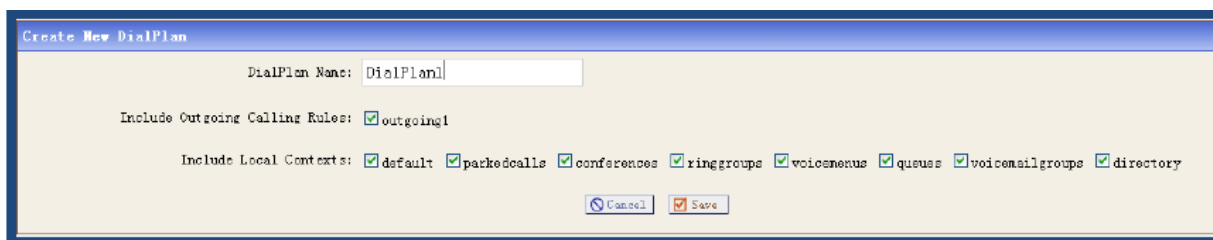
- Examine the number you dialed.
- Compare the number with the pattern that you have defined in your first outgoing rule and if
- matches, it will initiate the call using that trunk. If it does not match, it will compare the number with the pattern that you have defined in the second outgoing rule and so on.
- Pass the number to the appropriate trunk to make the call.

3.5 Dial Plans

A DialPlan is a set of Calling Rules that can be assigned to one or more users. Please select the **Dial Plans** option from the vertical menu on the left of the main page, then you can get the following screen:



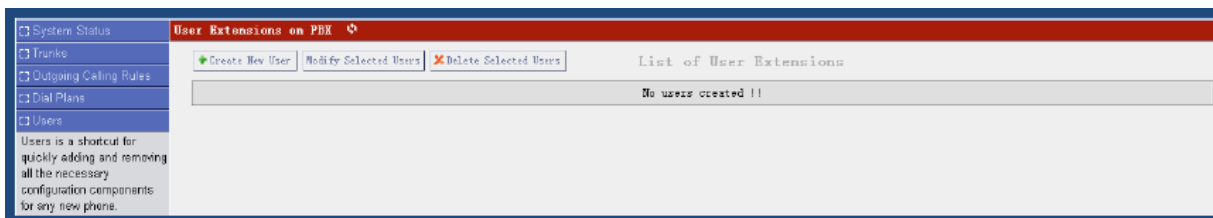
Click on **New DialPlan** button on the illustration above, the following screen is where you create and set up dial plan:



DialPlan Name: a unique label to help you identify the dial plan when listed in user component, you have to set up a dial plan name and select outgoing call rule and local context that you want to use.

3.6 Users

Users component is used to add or remove Analog, SIP, IAX extension. Please select the **Users** option from the vertical menu on the left, then you can get the following screen:



3.6.1 Create SIP/IAX User

Click on **Create New User** button on the illustration above, the following screen is where you create and set up user:

Edit User Extension - 6003

General :

Extension: 6003 Name: 6003 DialPlan: DialPlan1

CallerID: 6003 OutBound CallerID: 6003

☒ Enable Voicemail for this User

VoiceMail Access PIN code: Mailbox: 6003 Email Address:

Technology

☒ SIP ☐ IAX Analog Station: None flash: rxflash:

Codec Preference : First : u-law Second : GSM Third : None Fourth : None Fifth : None

VoIP Settings

MAC Address : Line Number : 1 SIP/IAX Password:

NAT: ☒ Can Reinvite: ☐ DTMF Mode: RFC2833 insecure: very

Other Options

☐ 3-Way Calling ☐ In Directory ☐ Call Waiting ☐ CTI ☒ Is Agent Pickup

Group: 1

In General component, you have to set up Extension, CallerID, Name, OutBound CallerID parameters, and choose a DialPlan for the extensions. Here I set up user 6003, and select DialPlan1 for the user.

I select **Enable Voicemail for this User option**, so the user has voicemail function. In the Technology component, you have to select SIP or IAX. Here I want to configure a SIP user, so I select SIP. For the Codec Preference, only the first two types of code you set are available. In the **Other Options** component, I select **Is Agent** which will be listed in Call Queues as a selectable member for call queue.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.6.2 Create Analog User

Click on **Create New User** button, the following screen is where you create and set up user:

Edit User Extension - 6005

General :

Extension: 6005 Name: 6005 DialPlan: DialPlan1

CallerID: 6005 OutBound CallerID: 6005

☒ Enable Voicemail for this User

VoiceMail Access PIN code: Mailbox: 6005 Email Address: robert.ao@atcom.

Technology

☐ SIP ☐ IAX Analog Station: Port 4 flash: 750 rxflash:

Codec Preference : First : u-law Second : GSM Third : None Fourth : None Fifth : None

VoIP Settings

MAC Address : 6005 Line Number : 1 SIP/IAX Password:

NAT: ☒ Can Reinvite: ☐ DTMF Mode: RFC2833 insecure: very

Other Options

☐ 3-Way Calling ☐ In Directory ☐ Call Waiting ☐ CTI ☒ Is Agent Pickup Group: 1

In the General component, you have to setup Extension, CallerID, Name, OutBound CallerID parameters, and choose a dialplan for the phone. Here I set up user 6005, and select DialPlan1 for the user.

I select **Enable Voicemail for this User option**, so the user has voicemail function. In the Technology componet, you have to select the port in which the analog phone will be plugged from the drop-down list of **Analog Station**. I select **Enable Voicemail for this User option**, so the user have voicemail function.

In the **Other Options** component, I select **Is Agent** which will be listed in Call Queues as a selectable member for call queue.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

Attention: in the textbox of Extension, the value you set is limited to a range, you can adjust the range in the following screen to meet your requirement. Please select the **Options** option from the vertical menu on the left, then you can get the following screen:

General Preferences | Language | Change Password | Factory Reset | Reboot | Advanced Options

Global OutBound CID :

Operator Extension :

Ring Timeout :

Extension preferences:

User Extensions : to

Conference Extensions : to

VoiceMenu Extensions : to

RingGroup Extensions : to

Queue Extensions : to

VoiceMail Group Extensions : to

3.7 Ring Groups

Define Ring groups to dial more than one extension simultaneously, or to ring more than one phone sequentially. This feature may also be called Hunt groups. Please select the **Ring Groups** option from the vertical menu on the left of the main page, then you can get the following screen:

System Status | Trunks | Outgoing Calling Rules | Dial Plans | Users | **Ring Groups**

Define Ringgroups to dial more than one extension simultaneously, or to ring more than one phone sequentially. This feature may also be called Huntgroups.

Manage RingGroups

No RingGroups defined !!

Click on **New RingGroup** button on the illustration above, the following screen is where you create and set up ring group:

New RingGroup

RingGroup Name : ringgroup1

Extension for this ring group : 6400

Ring Group Members

- 6003 (SIP) 6003
- 6002 (SIP) 6002

Available Users

- 6001 (SIP) 6001
- 6001 (TAX2) 6001

Ring Group Options :

Strategy : Ring in Order

Seconds to ring each member : 20

If not answered Goto : Hangup

Cancel Save

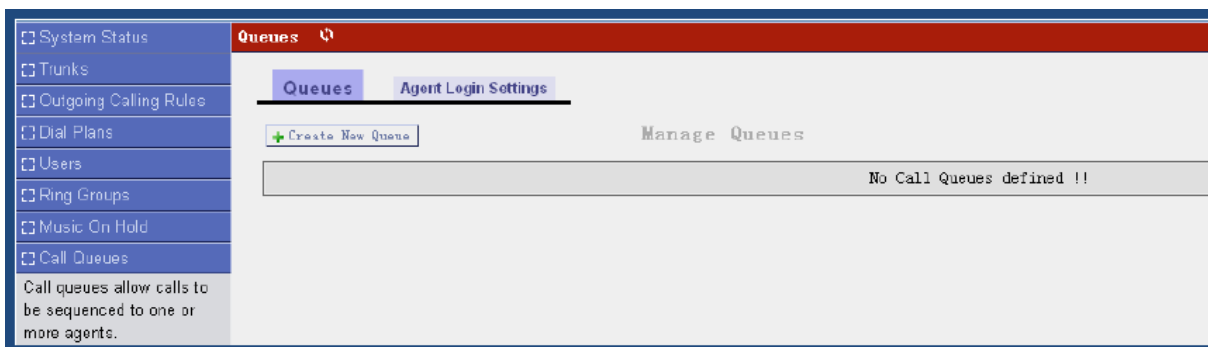
Set the ring group name and extension for the ring group, select ring group members from available users. Select strategy for ring group:

- **Ring in Order:** when someone calls the ring group, the ring group member will ring in order.
- **Ring all simultaneously:** when someone calls the ring group, all of the ring group member will ring at the same time.
- **If not answered Goto:** choose a destination from the drop-down list, when no one in the ring group answers the call.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.8 Call Queues


Please select the **Call Queues** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **Create New Queue** button on the illustration above, the following screen is where you create and set up call queue:

Extension: a unique label to help you identify the call queue when listed in **outgoing calling rules** component.

- **Agents:** select the users which you want them to be queue member.

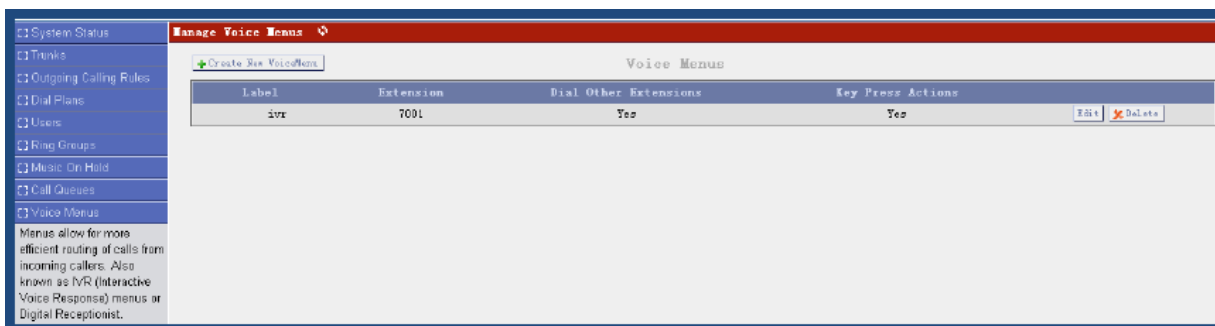
You can get information of other parameters by putting your mouse on the  label. At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.9 Voice Menus

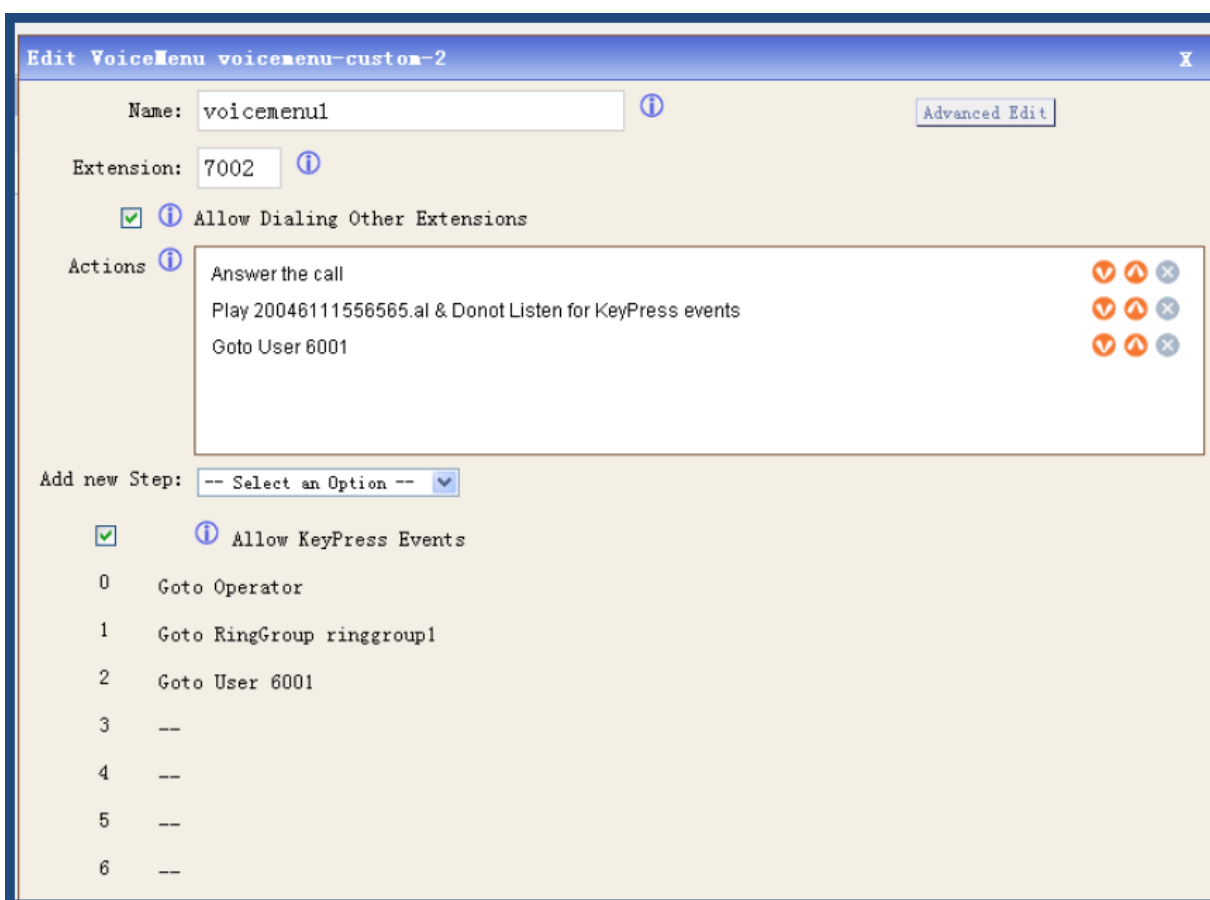
Like most organization, we would like to redirect all of the incoming calls automatically. The voice menu is very handy for these sorts of things. The system should allow callers to make the selection according to the

voice menu.

Please select the **Voice Menus** option from the vertical menu on the left, then you can get the following screen:



Click on **Create New VoiceMenu** button on the illustration above, the following screen is where you create and set up voice menu:



Name: a unique label to help you identify the voice menu when listed in incoming calling rules.

- **Add new Step:** select an action from the drop-down list. I add three steps above, so it will answer the call, and play a sound file, at last go to user 6001.

Click on **Allow KeyPress Events**: when the caller is in voice menu, they can press some specific numbers which are defined here to enter other destination. Here I define three numbers for going to operator, ringgroup,

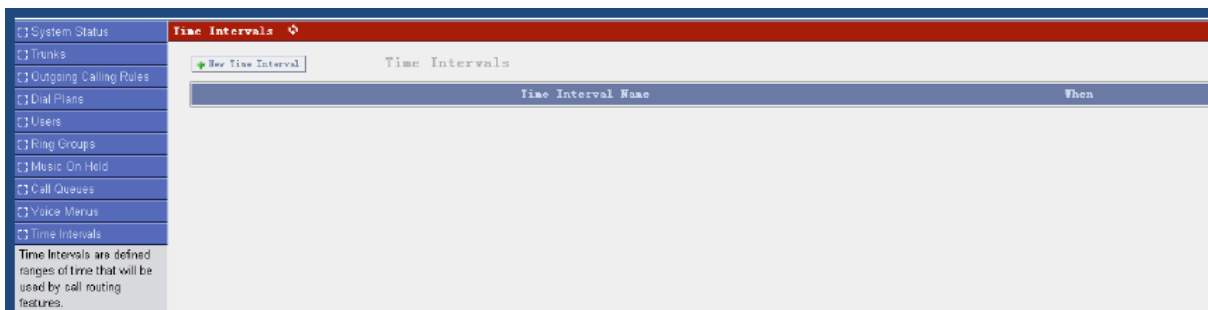
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Fax: 52718216. ventas@ansel.com.mx**

and user respectively.

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

3.10 Time Intervals

Time Intervals defines ranges of working time that will be used by call routing features. Please select the **Time Intervals** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New Time Interval** button on the illustration above, the following screen is where you create and set up time interval:

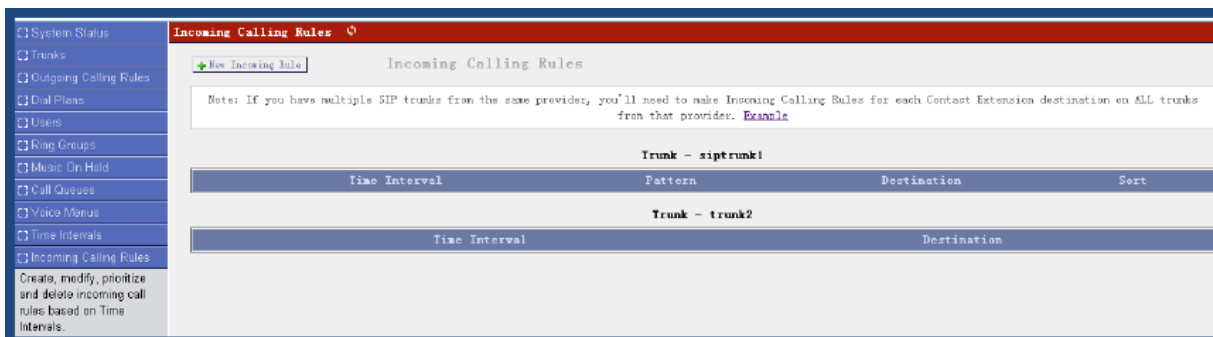
- **Time Interval Name:** a unique label to help you identify the time interval when listed in incoming calling rules. I set up timeinterval1 as time interval name.
- **By day of week:** I select it from Monday to Friday, the incoming call rule only works from Monday to Friday.
- **Time:** I set up it from 09:00 AM to 06:30 PM, the incoming call rule only works from 09:00 AM to 06:30 PM.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

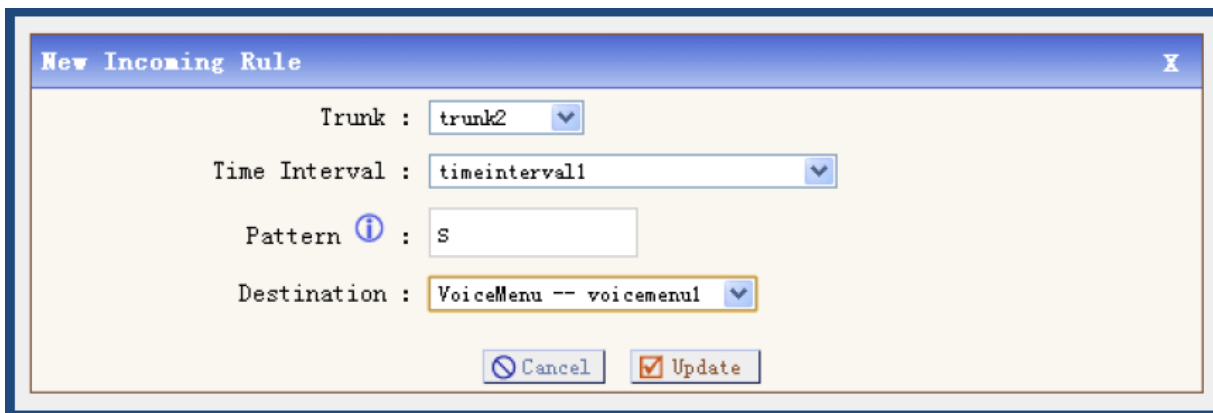
3.11 Incoming Calling Rules

This is where the behavior of incoming calls from all trunks is being handled. When an incoming call from PSTN or VoIP trunk is received, asterisk needs to know where to direct it. It can be directed to a ring group, an extension, digital receptionist, voice menu or queue. For this purpose, Incoming Calling Rules need to be set up.

Please select the **Incoming Calling Rules** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New Incoming Rule** button on the illustration above, the following screen is where you create and set up time interval:



- **Trunk:** select trunk for incoming call to use. I select trunk2 I set up before.
- **Time Interval:** determine the time when the incoming call rule works, I select timeinterval1 I set up before.
- **Pattern:** match the destination number, I use S which will match any destination number.
- **Destination:** I select voicemenu1, so the call will be ruled to voice menu.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.12 Voicemail


When you call someone who does not answer the call, you can leave a voice message for the called party if the called party supports voice mail.

Please select the **Voicemail** option from the vertical menu on the left of the main page, then you can get the

following screen:

Click on **General Settings** button on the illustration above. You can see the following screen:

Extension for checking messages: when you dial 6750, you will hear the voice message other people left for you.

You can get information of parameters by putting your cursor on the  label. If you want to set

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voicemail function for the user, you have to enable voicemail component when you set up a user. Please refer to the following illustration:

Edit User Extension - 6005

General :

Extension: 6005 Name: 6005 DialPlan: DialPlan1

CallerID: 6005 OutBound CallerID: 6005

☒ Enable Voicemail for this User

VoiceMail Access PIN code: Mailbox: 6005 Email Address: robert.ao@atcom.

Technology

☐ SIP ☐ IAX Analog Station: Port 4 flash: 750 rxflash:

Codec Preference : First : u-law Second : GSM Third : None Fourth : None Fifth : None

VoIP Settings

MAC Address : 6005 Line Number : 1 SIP/IAX Password:

NAT: ☒ Can Reinvite: ☐ DTMF Mode: RFC2833 insecure: very

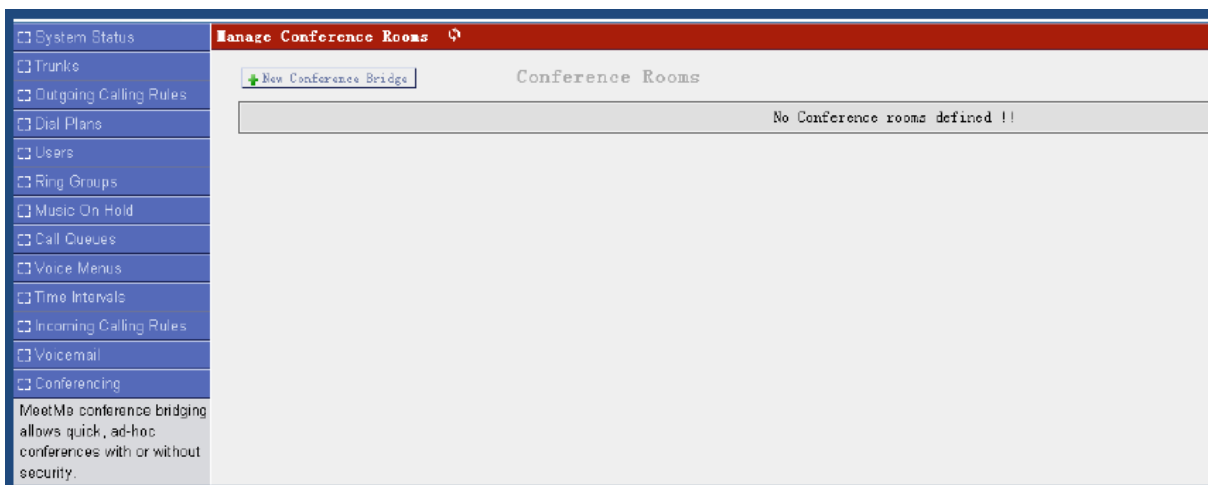
Other Options

☐ 3-Way Calling ☐ In Directory ☐ Call Waiting ☐ CTI ☒ Is Agent Pickup Group: 1

3.13 Conferencing

The conferencing function of Asterisk is similar to a Tele-conference call where multiple callers can call in and participate in a two-way conference like in a party room where everyone can talk and listen to one another or just to listen to a Tele-presentation.

Please select the **Conferencing** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New Conference Bridge** button on the illustration above. Below is what my conference configuration page looks like:

Naturally there are some options that you may wish to have for the conference room. They are entirely up to you. The main important things are for you to create the conference room number and the conference pin code for you to know how to enter into the conference.

The rest of the fields are optional. You can get information of other parameters by putting your mouse on the



label.

This conference number is 6300, the Pin Code is 123 for common member, the Pin Code is 456 for Admin. So you have to dial 6300 then, press the Pin Code, if you want to enter the conference. I enable the play hold music for option and announce callers option, so the first member who enter the conference will listen to a music and the online members will be informed when someone enter the conference.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.14 Follow Me

If A calls B, B does not answer, the call will be transferred to C who is set up in follow me. Please select the **Follow Me** option from the vertical menu on the left, then you can get the following screen:

Extension	Follow Me	Follow Order	
6001	Disabled	Not Configured	Edit
6002	Disabled	Not Configured	Edit
6003	Disabled	Not Configured	Edit
6004	Disabled	Not Configured	Edit
6005	Disabled	Not Configured	Edit
6006	Enabled	6001	Edit

You can choose user for which you want to setup follow me function, Here taking the user 6006 for an example, click on the **edit** button at the same line as 6006, you can get the following screen:

Status ⓘ : ☐ Enable ☒ Disable

'Music On Hold' Class ⓘ :

DialPlan ⓘ :

Destinations ⓘ :

[Add FollowMe Number](#)

Select the **enable** status, and click on **Add FollowMe Number** button to add a destination phone.

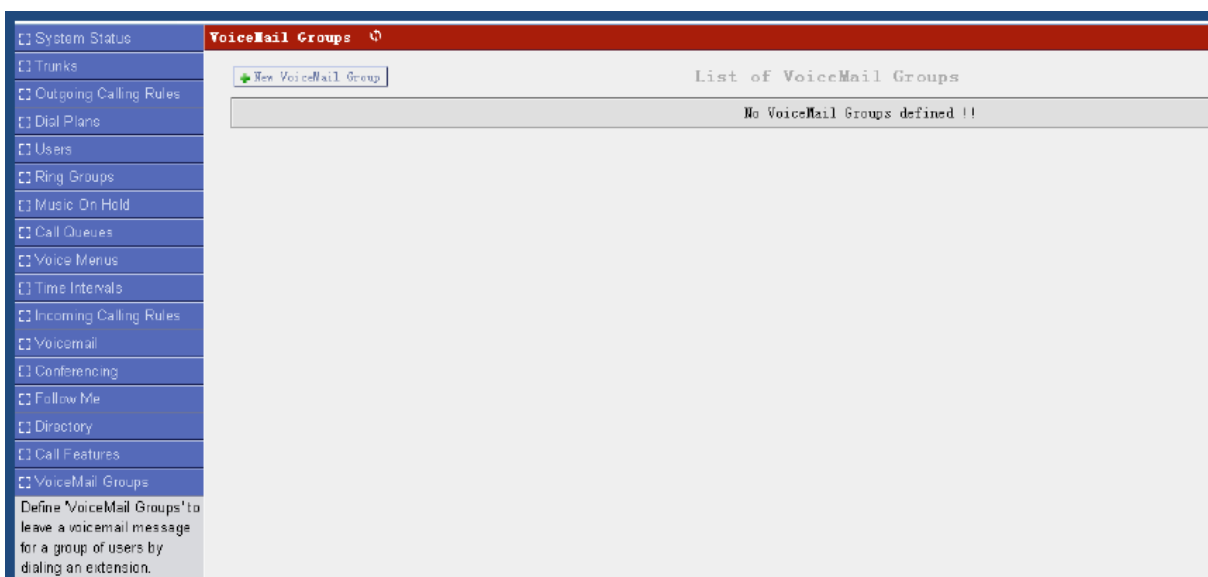
Click on **Dial Local Extension** and select 6001. Click on **Add** button and click on **Apply Changes** button in up right corner of the main page.

Through the above settings, someone calls 6006, but 6006 does not answer, the call will be transferred to 6001 automatically.

3.15 VoiceMail Groups

Define VoiceMail Groups to leave a voicemail message for a group of users by dialing an extension.

Please select the **VoiceMail Groups** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on **New VoiceMail Group** button on the illustration above. Below is what my VoiceMail Group configuration page looks like:

From the above settings, I can dial 6600 to leave message for user 6005 and 6006.

3.16 Voice Menu Prompts

This component is used for recording custom voice menu. Please select the **Voice Menu Prompts** option from the vertical menu on the left of the main page, then you can get the following screen:

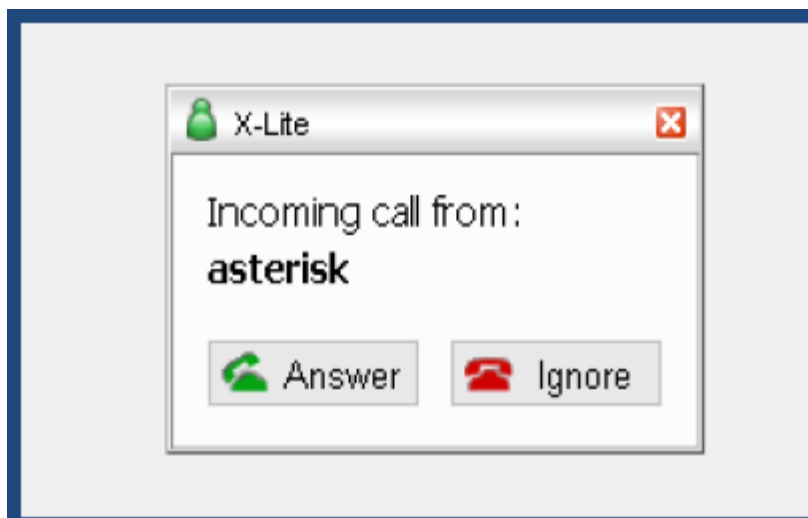


Click on **Record a new Voice Menu prompt** button on the illustration above. Below is what my Record a new Voice Menu prompt configuration page looks like:

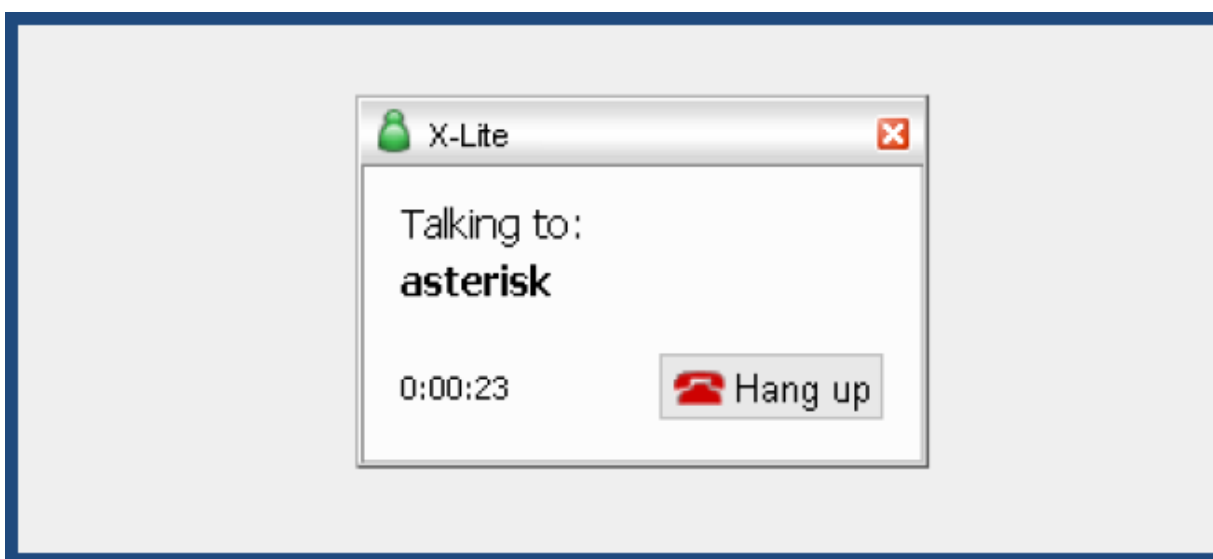
File Name: give a filename for the record sound file, here I give a name: WelcomToATCOM

Dial this User Extension to record a new voice: dial to a user, then the user pick up the phone and speak the voice menu which will be recorded. Here I select 6001 I set up before.

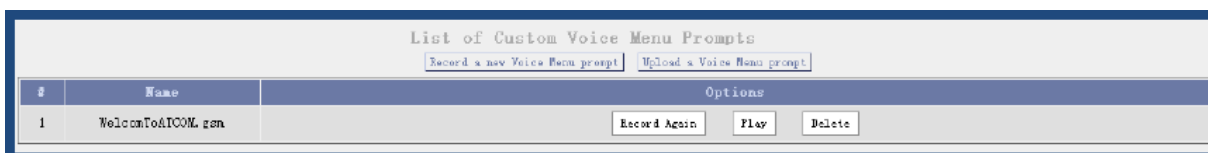
Click on **Record** button, the asterisk will call to 6001, 6001 will show like the following:



Click on **Answer** button, then you call speak and start to record what you say. The following illustration will be presented after you click on the **Answer** button.



When you want to finish the record, please click on **Hang up** button.



After you finish the recording, please refresh you webpage, and enter into **voice menu prompts** component again, you can see you have had a sound file like the above.

3.17 System Info

From this component, you can easily get the basic system information, it includes:

General Information:

The screenshot shows the 'System Information' window with the 'General' tab selected. The window has a red header bar with the title 'System Information' and a small icon. Below the header, there are four tabs: 'General', 'Network', 'Disk Usage', and 'Memory Usage'. The 'General' tab is active, displaying the following information:

```

OS Version:
Linux IPOx 2.6.22.18-ADI-2008R1astfin-svn #5 Thu Apr 29 23:10:13 EDT 2010 blackfin unknown

Uptime:
05:04:05 up 13 min,
Load Average: 0.05, 0.18, 0.08

Version Details:
Asterisk/1.4.21.2
VoIPtel GUI version: 2.0.2-ce
Firmware version: atcom_ce_ip04-0.3.6

Server Date & TimeZone: Wed May 5 05:04:05 EDT 2010

Hostname:
IPOx
  
```

The latest version of PBX-IP 2008 is atcom_ce_PBX-IP 2008-0.3.6. You can see the version that you are using from **Version Details** in the above illustration.

Network Information:

The screenshot shows the 'System Information' window with the 'Network' tab selected. The window has a red header bar with the title 'System Information' and a small icon. Below the header, there are four tabs: 'General', 'Network', 'Disk Usage', and 'Memory Usage'. The 'Network' tab is active, displaying the following information for the 'eth0' and 'lo' interfaces:

```

eth0      Link encap:Ethernet  HWaddr 00:09:45:76:89:78
          inet addr:192.168.1.151  Bcast:192.168.1.255  Mask:255.255.255.0
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:122794 errors:0 dropped:0 overruns:0 frame:0
          TX packets:68843 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:1000
          RX bytes:14130939 (13.4 MiB)  TX bytes:31815783 (30.3 MiB)
          Interrupt:48

lo        Link encap:Local Loopback
          inet addr:127.0.0.1  Mask:255.0.0.0
          UP LOOPBACK RUNNING  MTU:16436  Metric:1
          RX packets:2942 errors:0 dropped:0 overruns:0 frame:0
          TX packets:2942 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0
          RX bytes:1272712 (1.2 MiB)  TX bytes:1272712 (1.2 MiB)
  
```

Disk Usage Information:

System Information					
<div>General Network Disk Usage Memory Usage</div>					
Disk Usage:					
Filesystem	1k-blocks	Used	Available	Use%	Mounted on
/dev/mtdblock0	14327	13874	453	97%	/
/dev/mtdblock2	253952	75960	177992	30%	/persistent

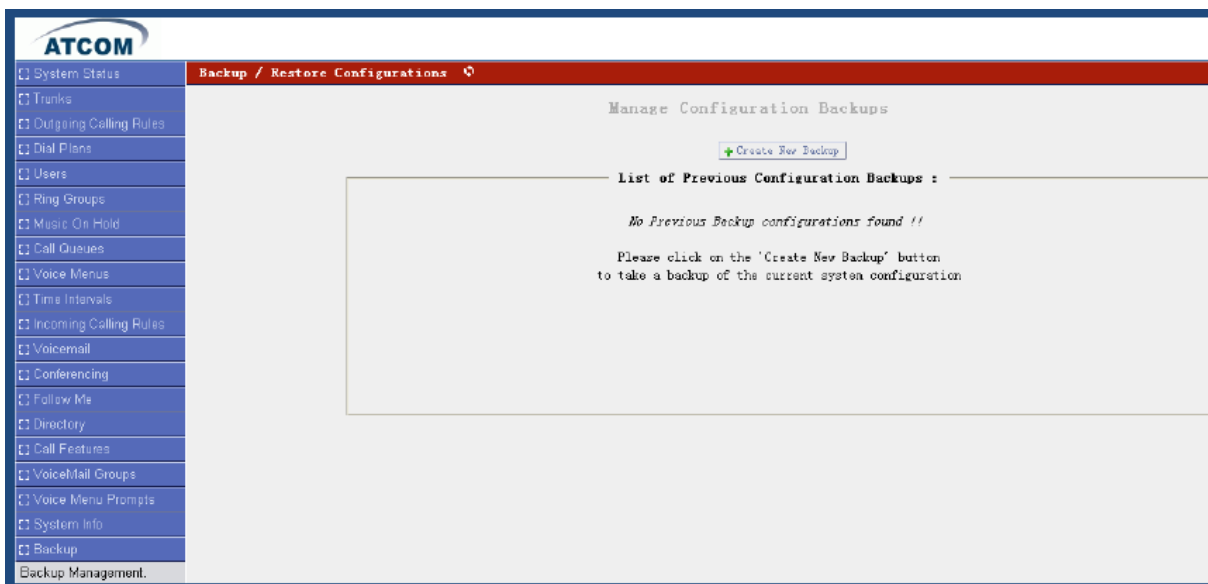
Memory Usage Information:

System Information					
<div>General Network Disk Usage Memory Usage</div>					
Memory Usage:					
	total	used	free	shared	buffers
Mem:	45928	41504	4424	0	812

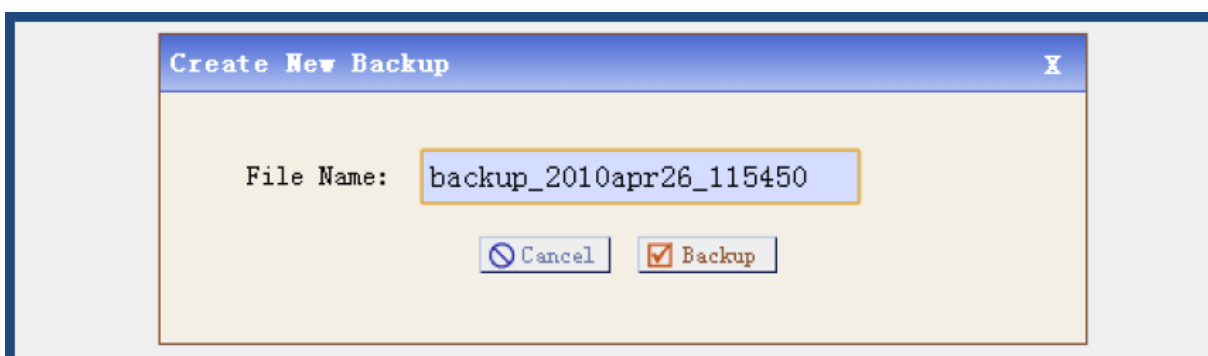
3.18 Backup

Backup and Restore are two of the mandatory functions of any application. PBX-IP 2008 is no exception. Customers can backup all the files under the /etc/asterisk/ directory and restore them.

Please select the **Backup** option from the vertical menu on the left of the main page, then you can get the following screen:

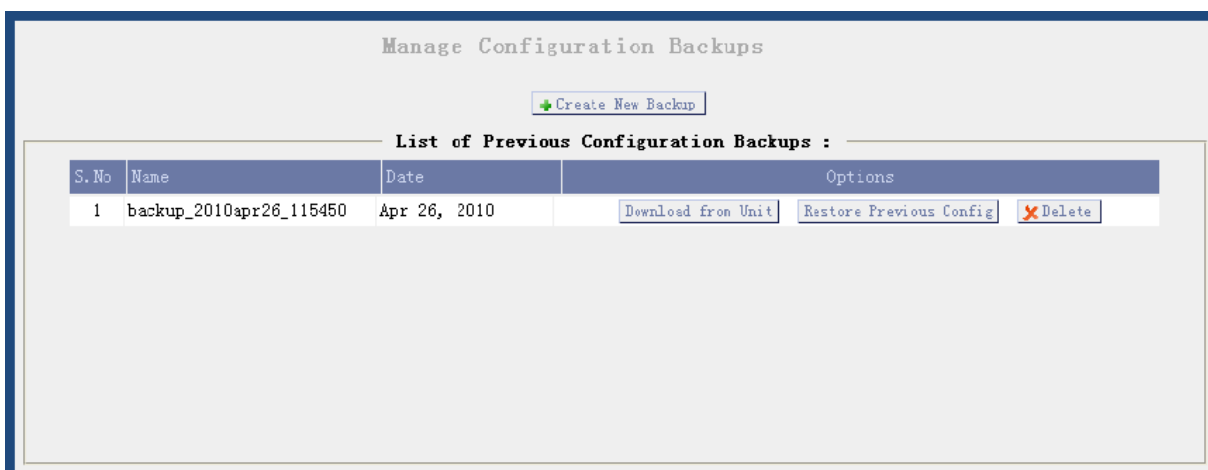


Click on **Create New Backup** button on the illustration above, you can get the following illustration:



- **File Name:** give a file name for the backed up file.

Click on **Backup** button, once the backup process is completed, you will see a screen with the backup filename displayed in illustration below.



Backup itself is not useful if it cannot be restored, PBX-IP 2008 also has this function. This is a very simple procedure. All you need to do is to click on the **Restore Previous Config** option.

3.19 Active Channels

The channels which are in communication status will be displayed in this component. Please select the **Active Channels** option from the vertical menu on the left, then you can get the following screen:

The screenshot shows the ATCOM Channel Management interface. On the left is a vertical menu with options: System Status, Trunks, Outgoing Calling Rules, Dial Plans, Users, Ring Groups, Music On Hold, Call Queues, Voice Menus, Time Intervals, Incoming Calling Rules, Voicemail, Conferencing, Follow Me, Directory, Call Features, VoiceMail Groups, Voice Menu Prompts, System Info, Backup, and Active Channels (highlighted in yellow). The main area is titled 'Channel Management' and shows 'Active Channels - 0'. It includes a 'Refresh Now' button and a status message 'Refreshing Active Channels in 2 Seconds'. Below this, a message states 'No Channels Open !!'.

Here my PBX-IP 2008 is using 2,3,4 channels, so I get the following information :

The screenshot shows the ATCOM Channel Management interface with 3 active channels. The main area displays 'Active Channels - 3' and 'Refreshing Active Channels in 5 Seconds'. Below this is a table with the following data:

Channel	State	Seconds	Application	Transfer	Hangup
Zap/2-1	Up	34		Transfer	Hangup
Zap/3-1	Up	undefined		Transfer	Hangup
Zap/4-1	Up	41	Dial(\$1ARGE), \$1RINGTIME\$, \$1DIALOPTIONS\$)	Transfer	Hangup

3.20 Options

This component is used for administrator to manage the system, it includes the following modules:

- General Preferences
- Language
- Change Password
- Factory Reset Reboot
- Advanced Options

General Preferences: you can set up a user to be the operator and the range of extension number for different types' extensions like the following screen:

The screenshot shows the 'General Preferences' tab selected in a web interface. The interface has a top navigation bar with tabs: 'General Preferences', 'Language', 'Change Password', 'Factory Reset', 'Reboot', and 'Advanced Options'. The main content area is titled 'General Preferences' and contains the following fields:

- Global OutBound CID: [text input field]
- Operator Extension: [dropdown menu showing 'User 6001']
- Ring Timeout: [text input field with value '20']

Below these fields is a section titled 'Extension preferences:' which contains a list of extension ranges, each with a label, a start value, and an end value in a text input field:

- User Extensions: 6001 to 6299
- Conference Extensions: 6300 to 6399
- VoiceMenu Extensions: 7001 to 7100
- RingGroup Extensions: 6400 to 6499
- Queue Extensions: 6500 to 6599
- VoiceMail Group Extensions: 6600 to 6699

At the bottom of the 'Extension preferences' section is a button labeled 'Reset to defaults'. At the very bottom of the form are two buttons: 'Cancel' and 'Save'.

Language: change the sound file language in which they play.

The screenshot shows the 'Language' tab selected in the web interface. The top navigation bar is the same as in the previous screenshot. The main content area is titled 'Language Settings' and contains the following field:

- Language: [dropdown menu showing 'English', 'Spanish', and 'French']

At the bottom of the form are two buttons: 'Cancel' and 'Save'.

Change Password: it is used for customers to change the admin password, click on the **Change Password** button, the following illustration will be presented below:

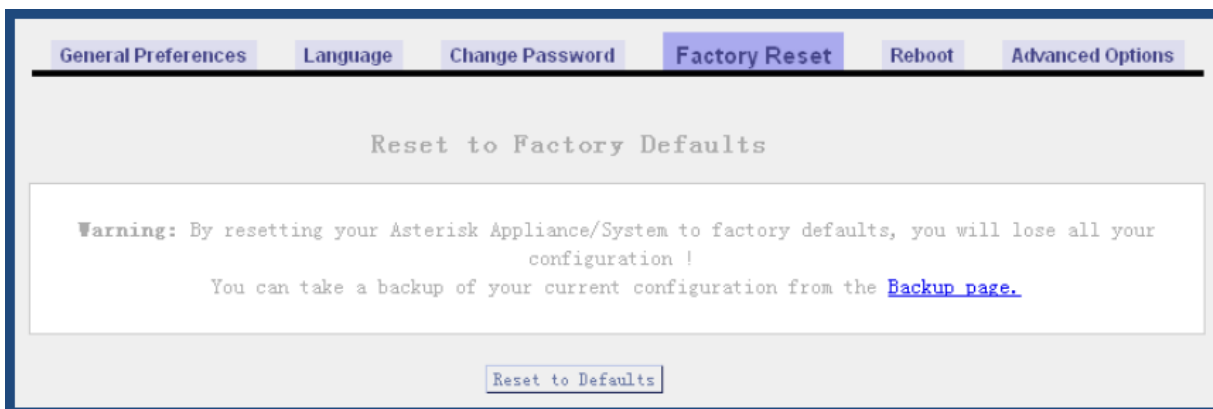
The screenshot shows the 'Change Password' tab selected in the web interface. The top navigation bar is the same as in the previous screenshots. The main content area is titled 'Change Password' and contains the following fields:

- Enter New Password: [text input field]
- Retype New Password: [text input field]

At the bottom of the form is a button labeled 'Update'.

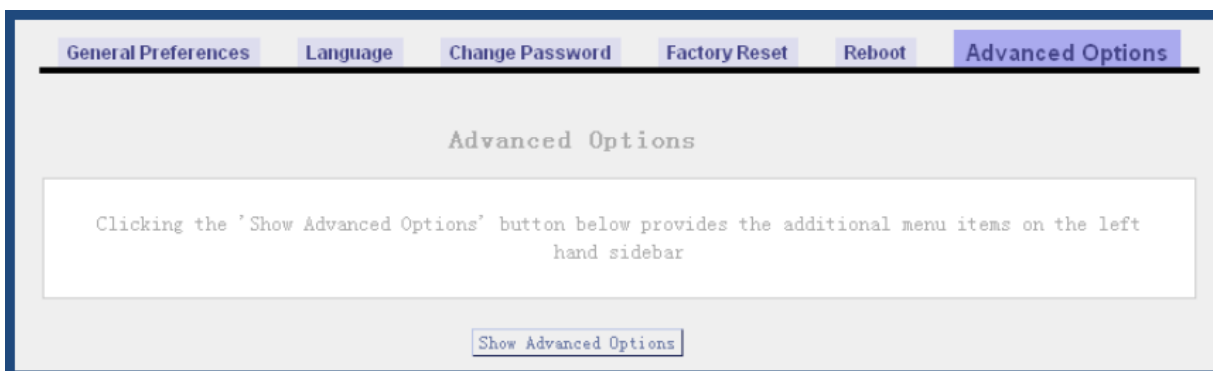
After inputting your new password, please click on **Update** button, then click on **Apply Changes** button on the up right corner of the main page.

Factory Reset: it will help you to recover to the default factory settings. Click on **Factory Reset** button, the following illustration will be presented below:

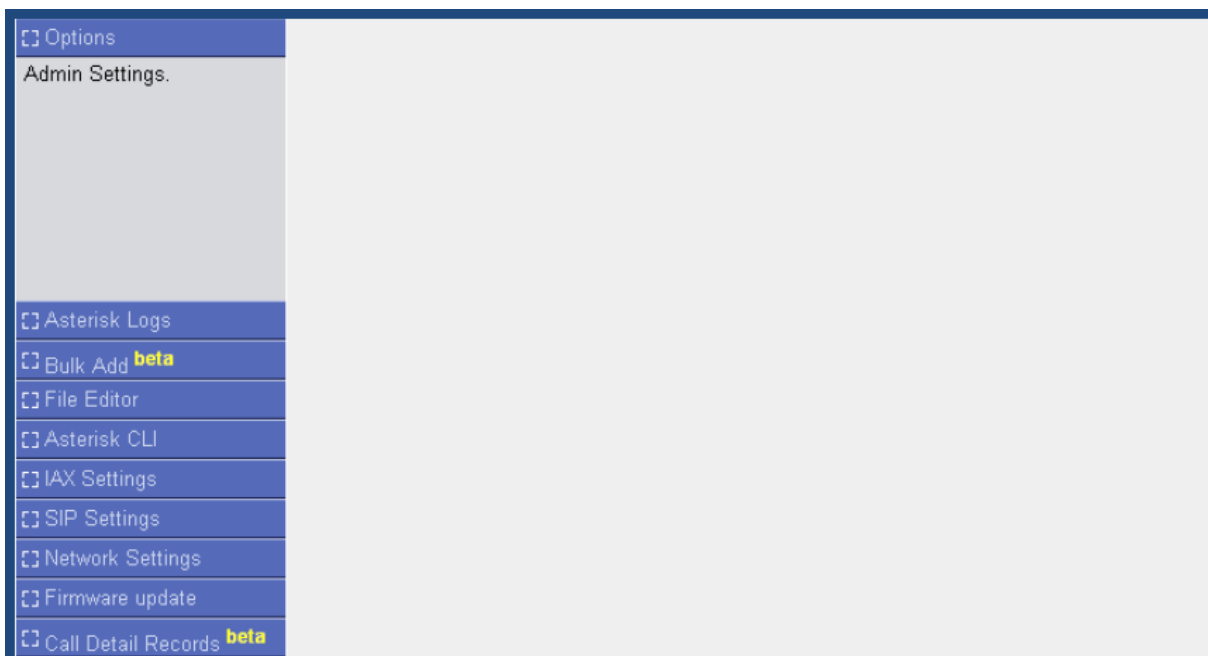


Please click on **Reset to Defaults** button to recover to default factory setting, then click on **Apply Changes** button on the up right corner of the main page.

- **Reboot:** you can click on **Reboot** button→**Reboot Now** button to reboot your system.
- **Advanced Options:** in default, PBX-IP 2008 web page hides several advanced options in the vertical menu on the left, if you need to use them, you have to display the options by clicking on **Show Advanced Options** in the following illustration:



After click on **Show Advanced Options** in the illustration above, you can see the advanced options in the vertical menu on the left of the main page like the following:

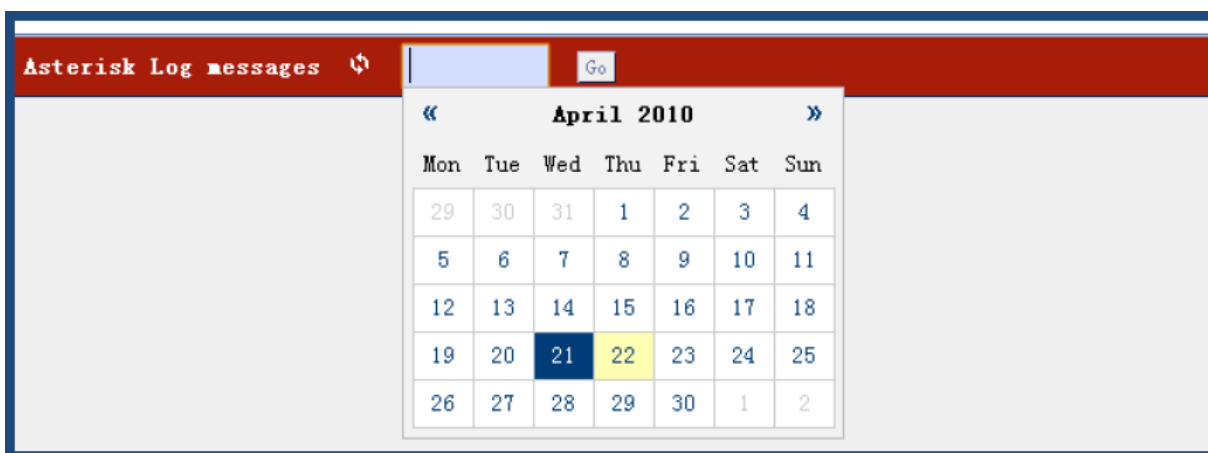


3.21 Asterisk Logs

After click on **Options**→**Advanced Options**→**Show Advanced Options**, please select the **Asterisk Logs** option from the vertical menu on the left of the main page, then you can get the following screen:



Click on the textbox, you can get the following screen:



You can see a date table, and you can select the log to watch by clicking on the date. After choosing the date, please click on **Go** button, you can see the asterisk log of the day you choosed. Here I need to see the asterisk log of April 21st,2010, I click on 21 in the date table, I get the following screen:

Asterisk Log messages 21 Apr 2010 Go

I click on **Go** button, then I get the log in the following screen:

```

Asterisk Log messages 21 Apr 2010 Go

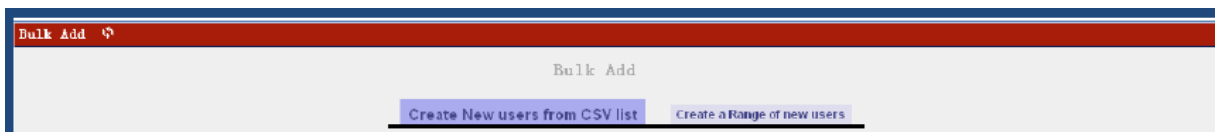
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring insecure
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring signalling
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring macaddress
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring autoprox
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring label
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring linenummer
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring flash
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring disallow
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring allow
[Apr 21 03:45:16] WARNING[19680] app_dial.c: Unable to create channel of type 'IAX2' (cause 3 - No route to destination)
[Apr 21 03:45:36] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #1)
[Apr 21 03:45:40] WARNING[19691] ast_expr2.fl: ast_yyerror(): syntax error: syntax error, unexpected '=', expecting $end; Input:
[Apr 21 03:45:40] WARNING[19691] ast_expr2.fl: If you have questions, please refer to doc/channelvariables.txt in the asterisk source.
[Apr 21 03:46:06] WARNING[19691] app_dial.c: Unable to create channel of type 'IAX2' (cause 3 - No route to destination)
[Apr 21 03:46:26] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #2)
[Apr 21 03:47:16] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #3)
[Apr 21 03:47:46] WARNING[211] chan_sip.c: Maximum retries exceeded on transmission 24806208277904-200421191943618192.168.1.3 for seqno 1 (Critical Response)
[Apr 21 03:47:46] WARNING[211] chan_sip.c: Hanging up call 24806208277904-200421191943618192.168.1.3 - no reply to our critical packet.
[Apr 21 03:48:06] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #4)
[Apr 21 03:48:56] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #5)
[Apr 21 03:49:46] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #6)
[Apr 21 03:50:36] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #7)
[Apr 21 03:51:26] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #8)
[Apr 21 03:52:16] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #9)
[Apr 21 03:53:06] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #10)
[Apr 21 03:53:56] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #11)
[Apr 21 03:54:46] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #12)
[Apr 21 03:55:36] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #13)
[Apr 21 03:56:26] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #14)
[Apr 21 03:57:16] NOTICE[211] chan_sip.c: -- Registration for '5008192.168.1.213' timed out, trying again (Attempt #15)

```

3.22 Bulk Add

Using bulk add, you can add multi-users one time. You can define the number of the users you want to create.

After click on **Options→Advanced Options→Show Advanced Options**, please select the **Bulk Add** option from the vertical menu on the left, then you can get the following screen:



Click on the **Create a Range of new users** button in the illustration above, the following screen is where you create bulk users.

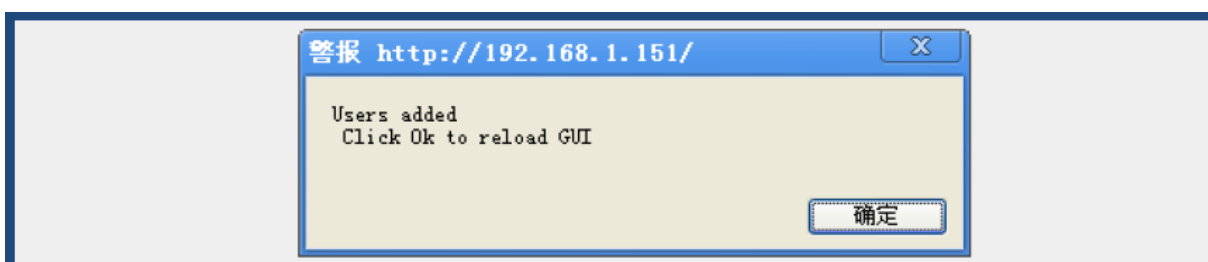
Bulk Add

[Create New users from CSV list](#)
[Create a Range of new users](#)

Create Users Starting from Extension

Tip: Use the 'Modify Selected Users' button from the Users page to edit any options for the created users.

Here I want to create five users, and the extensions starts from 6100, so I select 5 in the **Create** drop-down list, and I set 6100 in the textbox of **User Starting from Extension**.



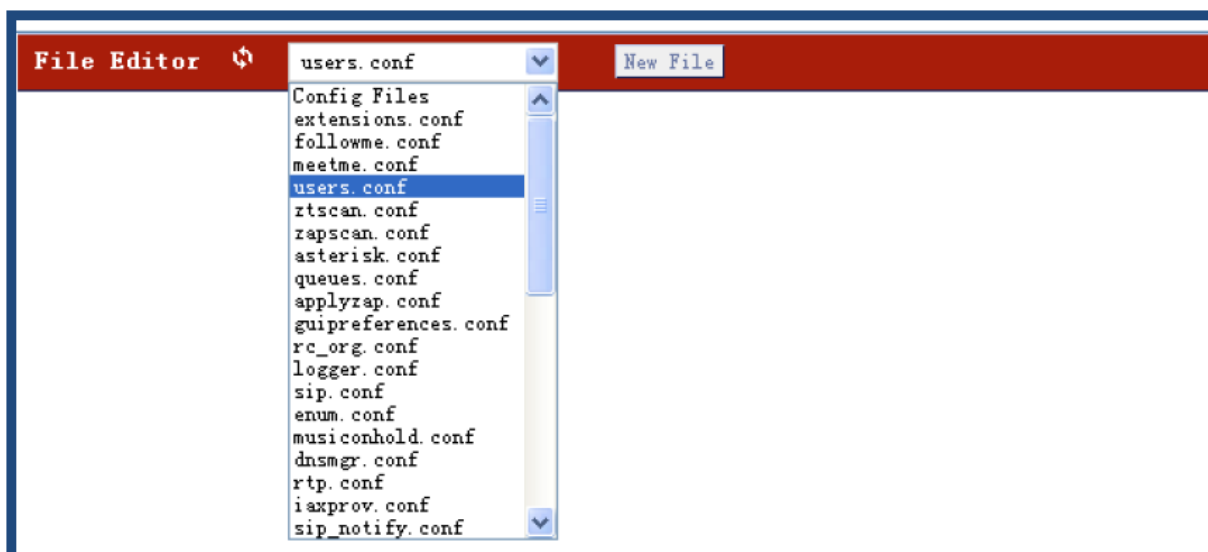
At last, click on button in the pop-up screen, then click on **Apply Changes** button on the up right corner of the main page. Please select the **System Status** option in the vertical menu on the left of the main page, you can see you have added five users: 6100, 6101, 6102, 6103, 6104.

3.23 File Editor

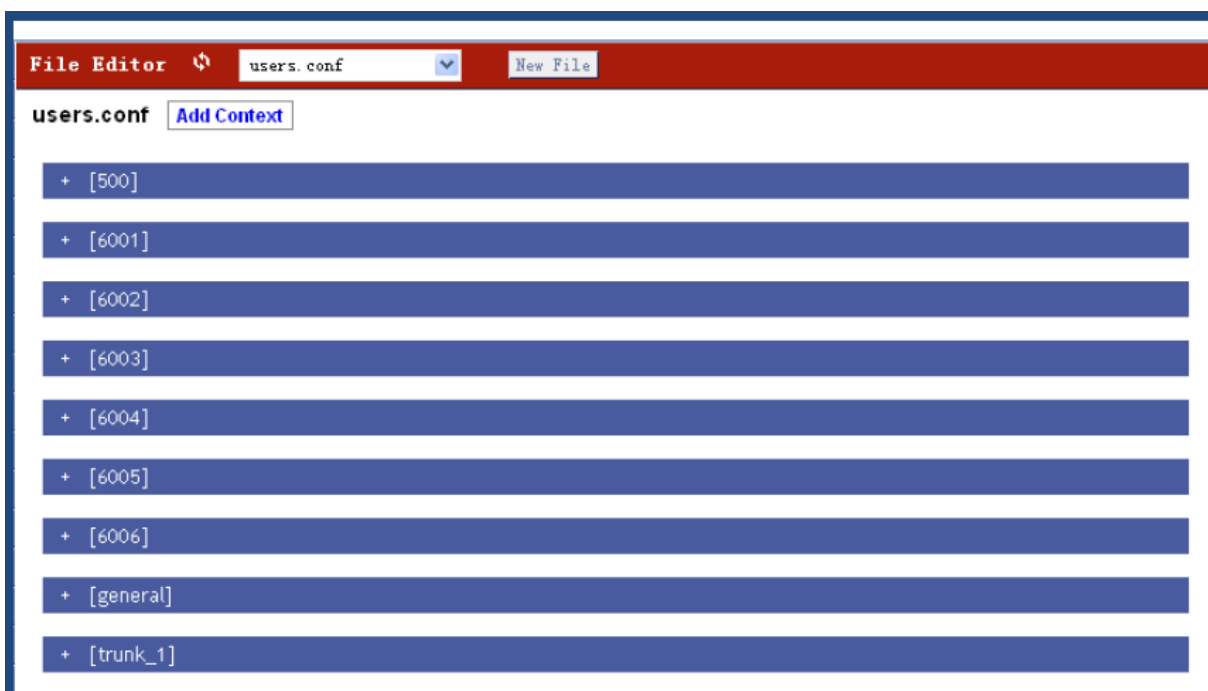
After click on **Options**→**Advanced Options**→**Show Advanced Options**, please select the **File Editor** option from the vertical menu on the left, then you can get the following screen:



From the drop-down list of config files, you can select the file you want to edit or read.



Here I select users.conf file, so I can see the file and edit to meet my requirement.



3.24 Asterisk CLI

These are some of the available CLI commands that can be executed from the console, you can input the asterisk CLI commands from the web page directly.

After click on **Options→Advanced Options→Show Advanced Options**, please select the **Asterisk CLI** option from the vertical menu on the left, then you can get the following screen:

```

Asterisk CLI> help

Command>help

! Execute a shell command
abort halt Cancel a running halt
agent logoff Sets an agent offline
agent show Show status of agents
agent show online Show all online agents
agi debug Enable AGI debugging
agi debug off Disable AGI debugging
agi dumphtml Dumps a list of agi commands in html format
agi show List AGI commands or specific help
cdr status Display the CDR status
core set debug channel Enable/disable debugging on a channel
core set debug Set level of debug chattiness
core set debug off Turns off debug chattiness

```

Here I input help command in the textbox, so I can get all the command which I can use in CLI mode.

3.25 Network Settings

In order to give a static and permanent IP address for IP-08, you have to set it in web GUI. After you enter into the web GUI of IP-08, you can try to configure IP address according to the following steps:

After click on **Options→Advanced Options→Show Advanced Options**, please select **Network Settings** option from the vertical menu on the left of main page, the following screen is where you configure the network:

eth0 Interface

DHCP:

Hostname:

Domain:

IP address:

Subnet mask:

Gateway:

DNS:

NTP:

In the drop-down list of **DHCP**, you can see the following three options:

1. DHCP: yes: PBX-IP 2008 will obtain the dynamic IP address from your router.
2. DHCP: auto: PBX-IP 2008 will use the static IP specified below and ping the default gateway. When there is no response from the default gateway, the IP-08 will switch to dynamically obtain the IP address from your router.

3. DHCP: no: PBX-IP 2008 will use the static IP address set below.

If you want to get static and permanent IP address, please do not select “yes”, after configure other parameters, please click “save” in the bottom of your page to save your setting.

3.26 Firmware Update

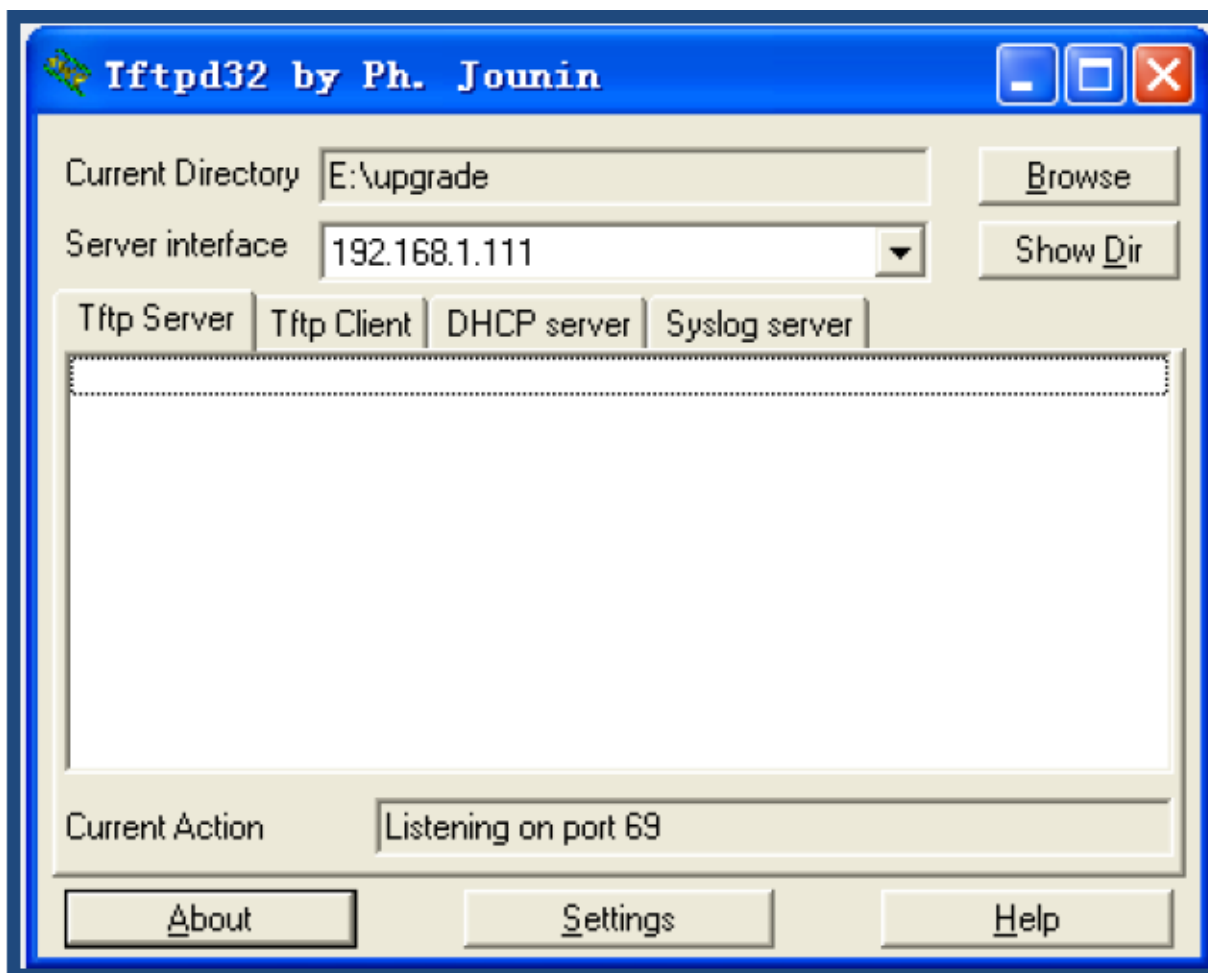
You can update to the latest version for PBX-IP 2008 by TFTP.

3.26.1 Download the Latest Firmware File and Set up TFTP Server.

- 1) Download the md5 file from

<http://www.atcom.cn/downloads/index.php?folder=SVBQQIgvZmlybXdhcmU=> , then put it in your TFTP server root directory.

- 2) Run your TFTP server, and I set up it like the following:



“E:\upgrade” is the root directory of my TFTP server, “192.168.1.111” is the IP Address of my TFTP server.

3.26.2 Update for PBX-IP 2008 from Web Page

Agricultura 111 Piso 1, Colonia Escandón, C.P 11800 México DF 55 52714421, 52774459, 52719163
Fax: 52718216. ventas@ansel.com.mx

After click on **Options→Advanced Options→Show Advanced Options**, please select **Firmware update** option from the vertical menu on the left of main page, the following screen is where you update for PBX-IP 2008:

TFTP Server: enter the IP Address of your TFTP server in this textbox.

- **File Name:** enter the update file name
- **Reset Configs:** if you choose reset Configs, it will delete all of your configuration you have done before.

After setting up, please click on **Go** button to update for PBX-IP 2008.

Power off and power on the IP-08, wait for several minutes. When the TEL port LEDs light up, it means the update is finished and you have the latest firmware.

3.27 Call Detail Records

This component provides the record of all incoming and outgoing calls including the channels used and duration of calls. After click on **Options→Advanced Options→Show Advanced Options**, please select the **Call Detail Records** option from the vertical menu on the left, then you can get the following screen:

CDR Viewer (CDR-CSV)

CDR viewer << prev next >>

Viewing 1-25 of 357
(most recent first)

View: 25

Account Code	Source	Destination	Dest. Context	Caller ID	Channel	Dest. Channel	Last app.	Last data	Start time	Answer Time	End Time	Duration	Billable seconds	Disposition	
1		6001	default		Local/6001@default-2567	2	SIP/6001-011ab340	Dial	SIP/6001@AX2/6001 [20]	2010-04-21 05:34:42		2010-04-21 05:35:12	30	0	
3		6001	default		Local/6001@default-8553	2	SIP/6001-012ab004	Dial	SIP/6001@AX2/6001 [20]	2010-04-21 05:34:10	2010-04-21 05:34:25	2010-04-21 05:34:30	18	1	
4	6005	6001	ELFR_DialPlan	**6005** 6005>	Zap/4-1		SIP/6001-012ab004	Dial	SIP/6001@AX2/6001 [20]	2010-04-21 05:31:21	2010-04-21 05:32:01	2010-04-21 05:32:04	43	3	ANSWER
6		s	default		Local/6001@default-7c11	2	SIP/6001-005ab004	Voicemail		2010-04-21 05:31:23	2010-04-21 05:31:23	2010-04-21 05:31:47	36	18	
7	6006	6750	ELFR_DialPlan	**6006** 6006>	SIP/6006-011ab340		Voicemail		2010-04-21 05:24:16	2010-04-21 05:24:16	2010-04-21 05:25:07	51	51	ANSWER	
8	6005	6600	ELFR_DialPlan	**6005** 6005>	Zap/4-1		Voicemail	6005@default/6006@default	2010-04-21 05:23:47	2010-04-21 05:23:50	2010-04-21 05:24:05	18	15	ANSWER	
9	6005	6300	ELFR_DialPlan	**6005** 6005>	Zap/4-1		Nextive	6300 [Rs	2010-04-21 04:51:41	2010-04-21 04:51:43	2010-04-21 04:52:56	75	73	ANSWER	

You can click on the **prev** to look up the last page for call record, and click on the **next** to look up the **next** page for call record, you can also set the value from the drop-down list of **view** which means how many calls will be displayed in one page.

Chapter 4 an Application Case of PBX-IP 2008

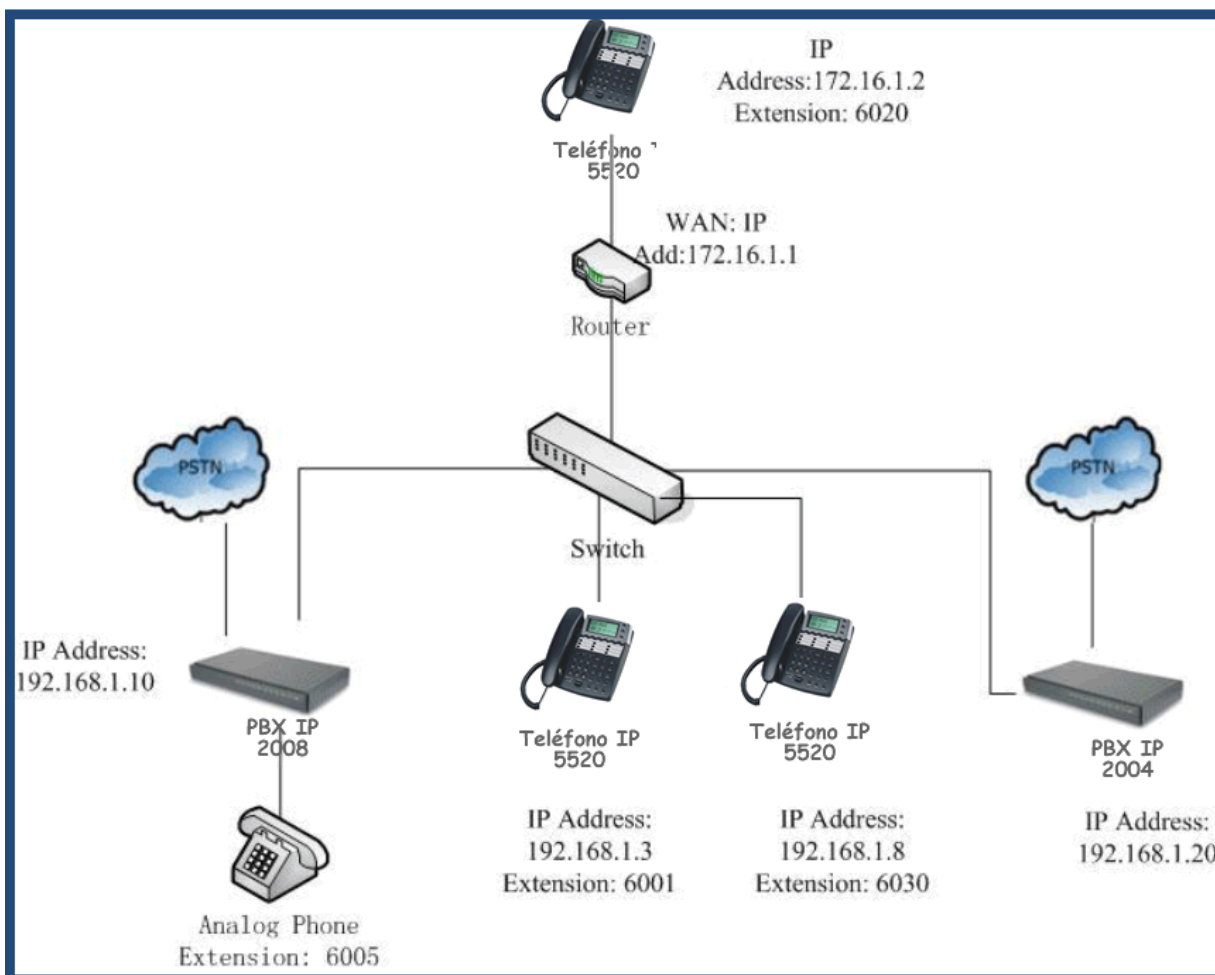


Figure: Network Topology In the network topology above: user 6020 and user 6001 will be registered to PBX-IP 2008, user 6030 will be registered to IP04, analog phone 6005 is connected to FXS port of PBX-IP 2008. After configuration, it will realize the following function:

- 1) The internal user 6005 and user 6001 can call each other directly.
- 2) 6005 and 6001 can dial-out through PBX-IP 2008 to PSTN.
- 3) 6005 and 6001 can get incoming calls from PSTN by PBX-IP 2008.
- 4) 6030 can call-out to PSTN and get incoming call from PSTN through IP04.
- 5) User 6001 and 6030 can call each other through VoIP trunk, although they are registered to different IP PBX.
- 6) User 6020, 6005 and 6001 can call each other directly, although they are not in the same network segment.

4.1 How to Make Internal Calls through PBX-IP 2008

4.1.1 Access to the Web Page of PBX-IP 2008 by Browser

After connecting PBX-IP 2008 to LAN, please open your browser of PC with windows OS and input the IP Address of PBX-IP 2008 (the default IP address is 192.168.1.100), then you can get the following screen:

Please input the default Username: admin; Password: atcom in the presented screen above. When you login successfully, you can get the configuration web page as below:

4.1.2 Add up Users from Web Page of PBX-IP 2008

- 1) Add up a DialPlan Before you add up user, you have to add up a DialPlan, please click on **Dial Plans**→**New DialPlan**, I add up a DialPlan like the following:

After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

2) Add up SIP user 6001

After logging into the web page of PBX-IP 2008, please click on **Users** → **Create New User**, I configure user 6001 like the following:

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3) Add up an Analog user 6005

After logging into the web page of PBX-IP 2008, please click on **Users** → **Create New User**, I add a user 6005 like the following:

Edit User Extension - 6005

General :

Extension: 6005 Name: 6005 DialPlan: DialPlan

CallerID: 6005 OutBound CallerID: 6005

☒ Enable Voicemail for this User

VoiceMail Access PIN code: Mailbox: 6005 Email Address: robert.ao@atcom.

Technology

☐ SIP ☐ IAX Analog Station: Port 4 flash: 750 rxflash:

Codec Preference : First : u-law Second : GSM Third : None Fourth : None Fifth : None

VoIP Settings

MAC Address : 6005 Line Number : 1 SIP/IAX Password:

NAT: ☒ Can Reininvite: ☐ DTMF Mode: RFC2833 insecure: very

Other Options

☐ 3-Way Calling ☒ In Directory ☐ Call Waiting ☐ CTI ☒ Is Agent Pickup Group: 1

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page. Please pay attention to the **Technology** component, there is an **Analog Station** drop-down list, I choose port 4 in which port the analog phone plugs.

4.1.3 Register a SIP user 6001 in AT610

After logging into the web page of IP Phone AT-610, please select VOIP option, I register the 6001 as the following illustration:



IP Phone

Current Status
Network
VOIP
Advanced
Dial-peer
Config Manage
Update
System Manage

Public SIP Configuration

Basic Setting			
Register status	Registered	Proxy Server Address	<input type="text"/>
Server Address	192.168.1.10	Proxy Server Port	<input type="text"/>
Server Port	5060	Proxy Username	<input type="text"/>
Account Name	6001	Proxy Password	<input type="text"/>
Password	****	Domain Realm	<input type="text"/>
Phone Number	6001	Enable Register	<input checked="" type="checkbox"/>
Display Name	6001		

After configuring, please click on the **APPLY** button. Now you can call each other directly between user 6001 and 6005.

4.2 How to Make a Call to Outside through PSTN

In order to dial out to PSTN with PBX-IP 2008, you need an analog trunk, an outgoing calling rule, a dial plan and a user. Here I will give the simple configuration steps which show how to make a call to outside, for detail configuration, you can refer to chapter 3.

4.2.1 Create an Analog Trunk

After logging into the web page of PBX-IP 2008, please click on **Trunks**→ **Analog Trunks**, I configure an analog trunk like the following:

Edit Analog Trunk

Channels: ☒ 2

Trunk Name ⓘ : trunk2

CallerID :

Normally you should not have to adjust your analog ports beyond the initial calibration. Should you still need to fine tune your audio settings, please use the adjustments at the right:

Port 2 Soft

Advanced Options

Busy Detection ⓘ : Yes

Busy Count ⓘ : 3

Ring Timeout ⓘ : 8000

Answer on No

Polarity Switch ⓘ :

Hangup on No

Call Progress ⓘ : No

Progress Zone ⓘ : CA

Use CallerID ⓘ : Yes

Caller ID Start ⓘ : Ring

CallerID ⓘ : As Received

Pulse Dial ⓘ : No

CID Signalling ⓘ : DTMF (Denmark, Sweden, Holland)

mailbox :

Flash Timing ⓘ : 750

Receive Flash Timing ⓘ : 1250

Cancel

Update

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.2 Create an Outgoing Calling Rule

After logging into the web page of PBX-IP 2008, please click on **Outgoing Calling Rules**→ **New Calling Rule**, I configure an outgoing calling rule like the following:

Edit Calling Rule

Calling Rule Name : outgoing1

Pattern : _2X.

☐ Send to Local Destination

Destination :

Send this call through trunk:

Use Trunk : trunk2

Strip : 1 digits from front

and Prepend these digits : before dialing

☐ Use FailOver Trunk :

fail over Trunk : siptrunk1

Strip : digits from front

and Prepend these digits : before dialing

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.3 Create a Dial Plan

After logging into the web page of PBX-IP 2008, please click on **Dial Plans**→ **New DialPlan**, I configure a dial plan like the following:

Edit DialPlan

DialPlan Name: DialPlan2

Include Outgoing Calling Rules: ☒ outgoing1

Include Local Contexts: ☒ default ☒ parkedcalls ☒ conferences ☒ ringgroups ☒ voicemenus ☒ queues ☒ voicemailgroups ☒ directory

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.4 Create a User

I will use the user 6001 I created before, here I need to reselect a dial plan for 6001, here I need to use DialPlan2, so I select DialPlan2 in the DialPlan drop-down list. Now I can call out with prefix 2, if the caller

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number is 10086, I will dial 210086.

4.3 How to Get an Incoming Call from outside

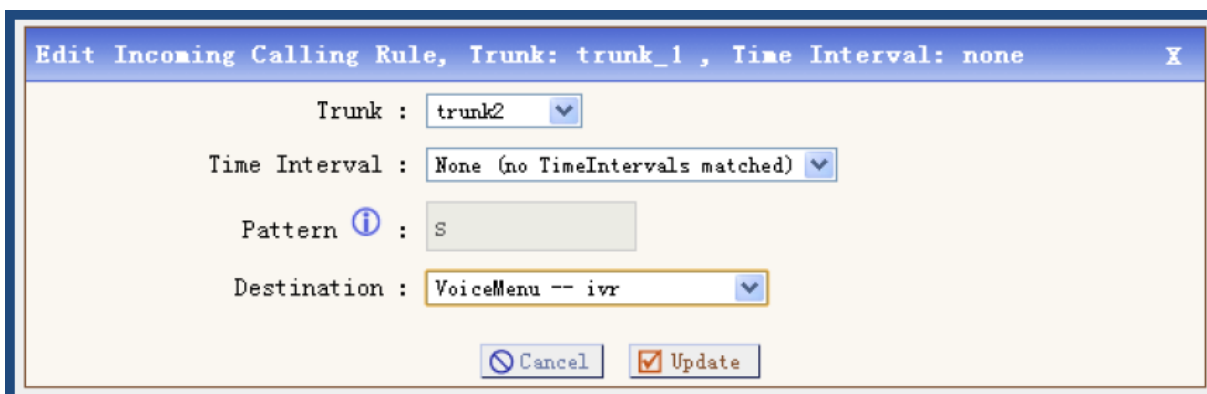
In order to get an incoming call from outside with PBX-IP 2008, you need an analog trunk, an incoming calling rule, a destination (here I use IVR). Here I will give the simple configuration steps which show how to get an incoming call from outside, for detail configuration, you can refer to chapter 3.

4.3.1 Create an Analog Trunk

I use the trunk2 I created in 4.2.1

4.3.2 Create an Incoming Calling Rule

After logging into the web page of PBX-IP 2008, please click on **Incoming Calling Rules**→ **New Incoming Rule**, I configure an incoming calling rule like the following:



At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

4.3.3 Create a Voice Menu

After logging into the web page of PBX-IP 2008, please click on **Voice Menus**→ **Create New VoiceMenu**, I create a voice menu like the following:

When the call comes from port 2, the system will play a record sound file, if the caller presses 1, user 6001 will ring, if the caller presses 2, user 6005 will ring. If the caller does not press any key, the call will go to 6001. You can also configure IP04 to let 6030 call outside and get incoming call by IP04, the steps are the same as PBX-IP 2008, you can refer to configuration of PBX-IP 2008.

4.4 How to Call Each Other Directly from Different Network Segment.

Take the user 6020, 6005 and 6001 for example, I will configure router, users and PBX-IP 2008, then the three users can call each other directly.

1) Set up router

From the web page of your router, please configure the IP address, subnet mask and default gateway of WAN port, I configured a static IP Address 172.16.1.1; Subnet Mask: 255.255.0.0; Default Gateway: 172.16.1.254. You can refer to the following:

LINKSYS
A Division of Cisco Systems, Inc.

Firmware Version: 1.05.00

Etherfast® Cable/DSL Router BEFSR41 V3

Setup

Setup | Security | Applications & Gaming | Administration | Status

Basic Setup | DDNS | MAC Address Clone | Advanced Routing

Internet Setup

Internet Connection Type: Static IP

IP Address: 172 . 16 . 1 . 1

Subnet Mask: 255 . 255 . 0 . 0

Default Gateway: 172 . 16 . 1 . 254

Static DNS 1: 0 . 0 . 0 . 0

Static DNS 2: 0 . 0 . 0 . 0

Static DNS 3: 0 . 0 . 0 . 0

Optional Settings (required by some ISPs)

Host Name:

Domain Name:

MTU: ☐ Enable ☒ Disable Size: 1492

Basic Setup

The Basic Setup screen is where basic configuration is performed. Some ISPs (Internet Service Providers) will require that you enter the DNS information. These settings can be obtained from your ISP. After you have configured these settings, you should set a router password from the Administration->Management screen.

Completing the **Internet Setup** section is all that is required to set up for your specific ISP. Please look at the table below to configure the Router for your Internet connection.

From the web page of your router, please configure the IP address, subnet mask and DHCP, I configure them like the following:

Network Setup

Router IP

Local IP Address: 192 . 168 . 1 . 254

Subnet Mask: 255 . 255 . 255 . 0

Network Address Server Settings (DHCP)

Local DHCP Server: ☒ Enable ☐ Disable

Start IP Address: 192.168.1.1

Number of Address: 254

DHCP Address Range: 192.168.1.1 to 192.168.1.254

Client Lease Time: 0 minutes (0 means one day)

WINS: 0 . 0 . 0 . 0

[More...](#)

Save Settings **Cancel Changes**

CISCO SYSTEMS

From the webpage of your router, please configure port range forwarding like the following:

LINKSYS
A Division of Cisco Systems, Inc.

Firmware Version: 1.05.00

Etherfast® Cable/DSL Router **BEFSR41 V3**

Applications & Gaming

Setup | Security | **Applications & Gaming** | Administration | Status

Port Range Forwarding | Port Triggering | UPnP Forwarding | DMZ

Port Range Forwarding

Port Range Forwarding can be used to set up public services on your network. When users from the Internet make certain requests on your network, the Router can forward those requests to computers equipped to handle the requests. If for

Port Range					
Application	Start	End	Protocol	IP Address	Enabled
IAX2	4569	to 4569	Both	192.168.1.10	<input checked="" type="checkbox"/>
	0	to 0	Both	192.168.1.0	<input type="checkbox"/>

The user 6020 uses IAX2, the port number is 4569, 192.168.1.10 is the IP address of PBX-IP 2008.

2) Add an IAX user 6020 in PBX-IP 2008

After logging into the web page of PBX-IP 2008, please click on **Users**→ **Create New User**, I configure 6020 like the following:

Edit User Extension - 6020

General :

Extension: 6020 Name: 6020 DialPlan: DialPlan1
CallerID: 6020 OutBound CallerID: 6020

☒ Enable Voicemail for this User
VoiceMail Access PIN code: Mailbox: 6020 Email Address:

Technology

☒ SIP ☒ IAX Analog Station: None flash: rxflash:
Codec Preference : First : u-law Second : GSM Third : None Fourth : None Fifth : None

VoIP Settings

MAC Address : Line Number : 1 SIP/IAX Password: 6020
NAT: ☒ Can Reininvite: ☐ DTMF Mode: RFC2833 insecure: very

Other Options

☐ 3-Way Calling ☐ In Directory ☐ Call Waiting ☐ CTI ☒ Is Agent Pickup
Group: 1

Cancel Update

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3) Set up AT-620 and register an IAX2 user 6020

After logging into web page of IP Phone AT-620, please select **Network** option to enter the screen of configuring IP Address. I set up a static IP Address: 172.16.1.2; Netmask: 255.255.0.0; Gateway: 172.16.1.254. After finishing the configuration, please click on the **Apply** button. You can refer to the following screen:

ATCOM IP Phone

Current Status Network **VOIP** Advanced Dial-peer Config Manage Update System Manage

- WAN Config
- LAN Config

WAN Status	
Active IP	172.16.1.2
Current Netmask	255.255.0.0
Current Gateway	172.16.1.254
MAC Address	00:09:45:56:fd:ce
Get MAC Time	20090915
WAN Setting	
Static <input checked="" type="radio"/>	DHCP <input type="radio"/> PPPoE <input type="radio"/>
Auto DNS	<input checked="" type="checkbox"/>
Static IP Address	172.16.1.2
Netmask	255.255.0.0
Gateway	172.16.1.254
DNS Domain	
Primary DNS	202.96.134.133
Alter DNS	202.96.128.88

APPLY

Please select the **VOIP** option, then select the **IAX2** option, I register the IAX2 user 6020 as the following illustration:

ATCOM IP Phone

Current Status Network **VOIP** Advanced Dial-peer Config Manage Update System Manage

- SIP_1
- SIP_2
- **IAX_2**

IAX2	
Register Status	Registered
IAX2 Server Addr	172.16.1.1
IAX2 Server Port	4569
Account Name	6020
Account Password	****
Phone Number	6020
Local Port	4569
Voice Mail Number	0
Voice Mail Text	mail
Echo Test Number	1
Echo Test Text	echo
Refresh Time	60 Seconds
Enable Register	<input checked="" type="checkbox"/>
Enable G.729	<input type="checkbox"/>

APPLY

After configuring, please click on the **APPLY** button.

Attention: here you must register IAX2 user instead of SIP user, because the user 6020 is not in the same network segment as PBX-IP 2008. If you use SIP user, you can not get sound when the communication is established. Now you can call each other among 6020,6001 and 6005 directly.

4.5 How to Call through VoIP Trunk

4.5.1 Call from PBX-IP 2008 to IP04

In order to call from PBX-IP 2008 to IP04, I will create a SIP user in IP04 for the SIP trunk in PBX-IP 2008, create a SIP trunk, an outgoing call rule and a dial plan in PBX-IP 2008.

1) Add an SIP user 6035(it will be used as SIP trunk in PBX-IP 2008) in IP04, after logging into the web page of IP04, please click on **Users→ Create New User**, I add the user 6035 like the following:

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page. Add a SIP user 6030 in IP04 for AT-620, the way is the same as adding 6035.

2) Add a VoIP trunk in PBX-IP 2008, after logging into the webpage of PBX-IP 2008, please click on **Trunks→VOIP Trunks→New SIP/IAX Trunk**, I configure a SIPtrunk1 like the following:

Edit SIP trunk 6035

Provider Name ⓘ: siptrunk1

Hostname : 192.168.1.20

Username : 6035

Fromuser :

Fromdomain : 6035

Password : 6035

Contact Ext.: s

Insecure Type: very ⓘ

Codecs : First : u-law Second : a-law Third : GSM

Fourth : G.726 Fifth : None

CallerID ⓘ :

☐ Enable Remote MWI :

After configuring, please click on **Add** button, and click on **Apply Changes** button in up right corner of the main page.

3) Create an outgoing calling rule in PBX-IP 2008, after logging into the webpage of PBX-IP 2008, please click on **Outgoing Calling Rules**→**New Calling Rule**, I configure an outgoing2 rule like the following:

Edit Calling Rule

Calling Rule Name : outgoing2

Pattern : _9.

☐ Send to Local Destination

Destination :

Send this call through trunk:

Use Trunk : siptrunk1

Strip : 1 digits from front

and Prepend these digits : before dialing

☐ Use FailOver Trunk :

fail over Trunk : siptrunk1

Strip : digits from front

and Prepend these digits : before dialing

Cancel Save

After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4) Create a dial plan in PBX-IP 2008, after logging into the webpage of PBX-IP 2008, please click on **Dial Plans**→**New DialPlan**, I configure a dialplan2 like the following:

Edit DialPlan

DialPlan Name: DialPlan2

Include Outgoing Calling Rules: ☒ outgoing1 ☒ outgoing2

Include Local Contexts: ☒ default ☒ parkedcalls ☒ conferences ☒ ringgroups ☒ voicemail ☒ queues ☒ voicemailgroups ☒ directory

Cancel Save

After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page. In configuration screens of 6001 and 6005, please select dialplan2 in the **DialPlan** drop-down list. Now you can call from 6001 and 6005 to 6030 by dialing 96030

4.5.2 Call from IP04 to PBX-IP 2008

In order to call from IP04 to PBX-IP 2008, I will create a SIP user in PBX-IP 2008 for the SIP trunk in IP04, create a SIP trunk, an outgoing call rule and a dial plan in IP04.

1) Add a user 6008 in PBX-IP 2008

Add a SIP user: 6008, after logging into the web page of PBX-IP 2008, please click on **Users→ Create New User**, I add a user 6008 like the following:

Create New User

General :

Extension: 6008 Name: 6008 DialPlan: DialPlan2

CallerID: 6008 OutBound CallerID: 6008

☐ Enable Voicemail for this User

VoiceMail Access PIN code: Mailbox: 6008 Email Address:

Technology

☒ SIP ☒ IAX Analog Station: None flash: 750 rxflash: 1250

Codec Preference : First : u-law Second : GSM Third : None Fourth : None Fifth : None

VoIP Settings

MAC Address : Line Number : 1 SIP/IAX Password: 6008

NAT: ☒ Can Reinvite: ☐ DTMF Mode: RFC2833 insecure: very

Other Options

☐ 3-Way Calling ☐ In Directory ☐ Call Waiting ☐ CTI ☐ Is Agent Pickup

Group: 1

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

2) Create a SIP trunk in IP04

Add a VoIP trunk in IP04, after logging into the webpage of IP04, please click on **Trunks→VOIP Trunks→New SIP/IAX Trunk**, I configure a SIP trunk like the following:

Create New SIP/IAX trunk

Type: SIP

Provider Name ⓘ: siptrunktoIP08

Hostname : 192.168.1.10

Username : 6008

Fromuser :

Fromdomain :

Password : 6008

Contact Ext.:

Insecure Type: very ⓘ

After configuring, please click on **Add** button, and click on **Apply Changes** button in up right corner of the main page.

3) Create an outgoing calling rule in IP04

After logging into the webpage of IP04, please click on **Outgoing Calling Rules**→**New Calling Rule**, I configure an outgoing1 rule like the following:

New CallingRule

Calling Rule Name : outgoing1

Pattern : _9.

☐ Send to Local Destination

Destination :

Send this call through trunk:

Use Trunk : siptrunktoIP08

Strip 1 digits from front

and Prepend these digits before dialing

☐ Use FailOver Trunk :

fail over Trunk

Strip digits from front

and Prepend these digits before dialing

Cancel Save

After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4) Create a dial plan in IP04

After logging into the webpage of IP04, please click on **Dial Plans**→**New DialPlan**, I configure a dialplan1 like the following:

Edit DialPlan

DialPlan Name: DialPlan1

Include Outgoing Calling Rules: ☒ outgoing1

Include Local Contexts: ☒ default ☒ parkedcalls ☒ conferences ☒ ringgroups ☒ voicemenus ☒ queues ☒ voicemailgroups ☒ directory

Cancel Save

After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page. In configuration screens of 6030, please select dialplan1 in the **DialPlan** drop-down list.

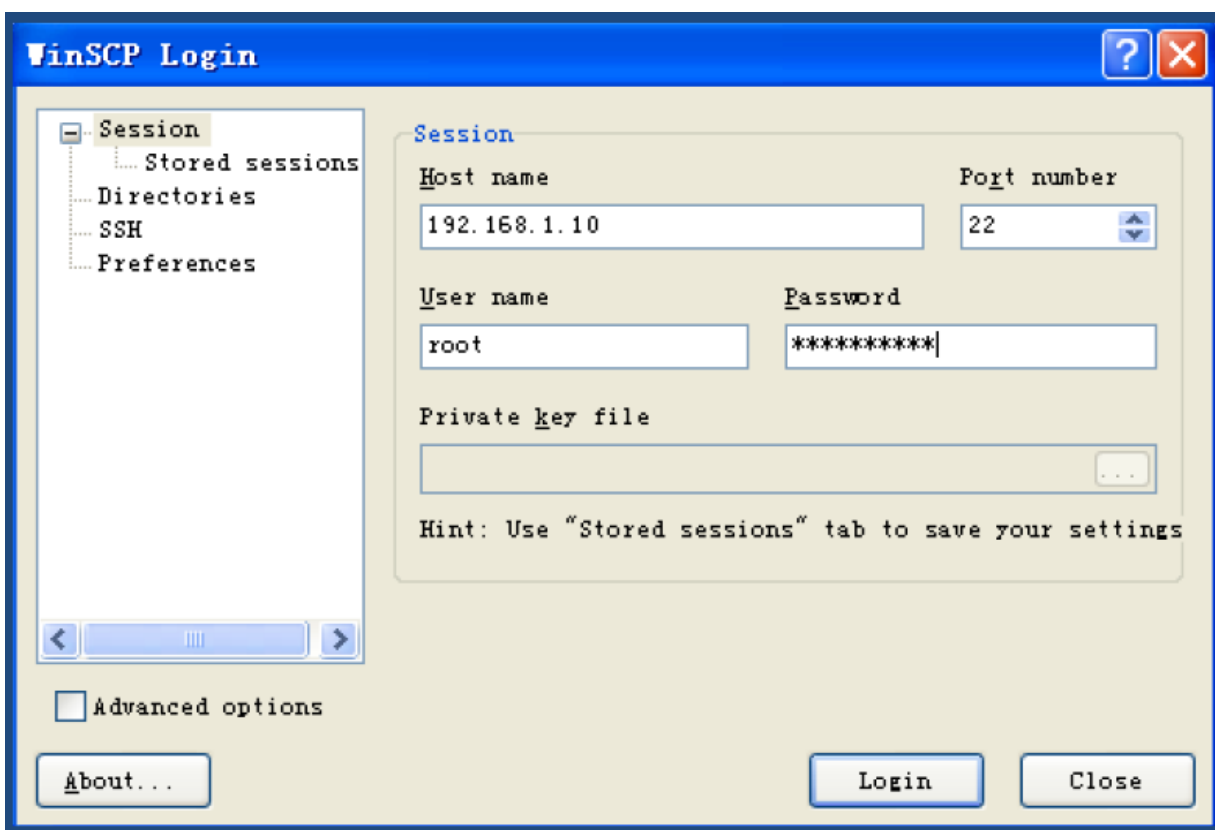
Now you can call from 6030 to 6001 and 6005 by dialing with prefix 9.

4.6 How to Transfer Files between Windows PC and PBX-IP 2008

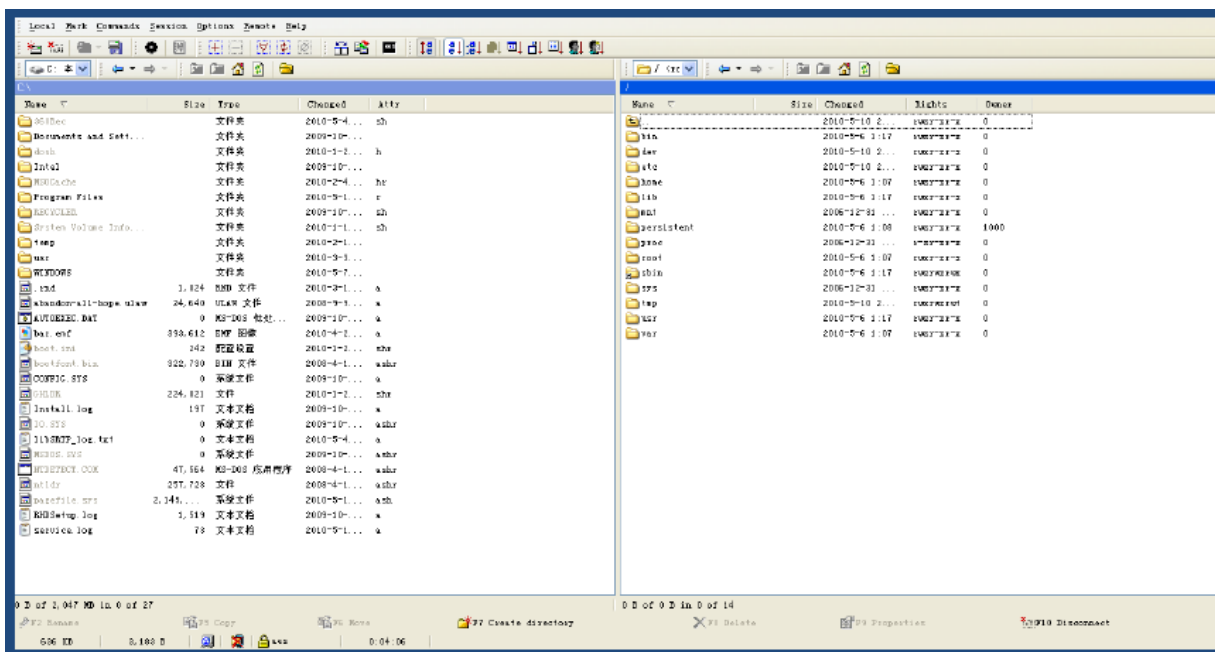
Using WinSCP software, it is the most convenient way to transfer files between windows PC and PBX-IP 2008.

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Fax: 52718216. ventas@ansel.com.mx

Open your WinSCP software , enter the IP Address, username, password of PBX-IP 2008 like the following screen:



At last, click on **Login** button, then you can get the following screen:

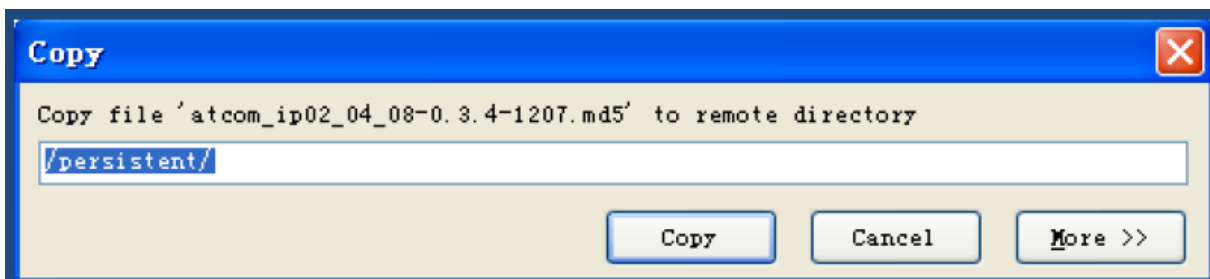


The left part of the screen displays directories and files of your windows PC, the right part of the screen displays

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directories of PBX-IP 2008.

If you want to transfer a file from windows PC to PBX-IP 2008, you just need to choose the file and drag it to the directory of PBX-IP 2008, at last, click on **copy** button in the popping-up screen like the following:



Chapter 5 Reference

<http://www.ansel.com.mx/>