Manual

PBX IP 2008



Version: 1.0



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CHAPTER 5 REFERENCE



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 Canceller) 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



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:



ATCOM	
[] Home	Velcome to VoIPtel CE
Please login	Asterisk™ Configuration Engine Username:
	Password:

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

ATCOM								Apply Changes 7	Logout
(] System Status	System Status 🔅 Ф							Upgrade to Vol	Ptel SE
Please click on a panel to manage related features				,	🛿 VolPtel				
			Uptine		stem Status nin, load average:	0.04, 0.06, 0.02			
					Trunks				
	Status	I runk	Туре	Username	P	ort/Hostname/IP			
CC Configure Hardware									
C Trunks					Extensions				
C Outgoing Calling Rules			I Free	😑 Busy	 UnAvailable 	🥚 Ringing			
🕄 Dial Plans	Extension		ħ	lame/Label		Status	Туре		
() Users	6750		c	heck Voicemail	ls .		VoiceMailMain		
🔁 Ring Groups	«No Extens	ion assigned	I	ial by Mamez			Directory		
() Music On Hold									
23 Call Queues									
El Voice Menus									
23 Time Intervals									
23 Incoming Calling Rules									
C3 Voicemail									
C Conferencing									
C3 Follow Me									
C Directory									
C3 Call Features									

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:

😵 PuTTY Configurat	tion 🛛 🕅
PullY Configurat Category: Session Generation Session Generation Selection Colours Colours Connection Proxy Telnet Rlogin SSH Serial	Basic options for your PuTTY session Specify the destination you want to connect to Host Name (or IP address) Port 192.168.1.100 22 Connection type: Baw Basic options for your PuTTY session Serial Load, save or delete a stored session Serial Load, save or delete a stored session Save Default Settings Load Default Settings Delete Close window on exit: Only on clean exit
About	<u>Open</u>

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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:



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:~> if config 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						? Lago
🗆 System Status		υ	ptime : 05:17:03 u	p 2 min, load average: 0.1	8, 0.08, 0.02	
Please click on a panel to manage related features				Irunks		
nonage toneso tonisioe	Status	Irunk	Iype	Username	Port/Hostname/IP	
	Unregistered	siptrunkl	sip	6035	192.168.1.20	
		t runk 2	Analog		Portz 2	
		Agent s	2	1 1	a	
Configure Hardware			6001 Login	6002 6003 Login Login	6005 Login	
🖸 Trunks			2			
□ Outgoing Calling Rules □ Dial Plans			6020			
🗆 Users			Login			
🗆 Ring Groups			Confe	rence Rooms		
E Music On Hold				6300		
Call Queues				Not in use		
C Voice Menus			-	Extensions		
Time Intervals		🔵 F1	ree 🗢 Busy	Unavailable	Ringing	
Incoming Calling Rules	Extension		ame/Label	Status	Туре	
🗆 Veicemail	<u>e001</u>		001	Mossages :		
Conferencing	6002		002	Wessages :		
🗆 Follow Me	6003		003	Messages :		
	<u>6004</u>	6	004	Messages :	0/0 Analog Uzer (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:

] System Status	∎anage Analog trunks	¢		
Configure Hardware				
] Trunks	Analog Trunks	Service Providers	VOIP Trunks	T1/E1/BRI Trunks
runks are outbound lines	🔸 New Analog Trunk			
sed to allow the system to				
nake calls to the real world.				
Trunks can be VoIP lines or			No Analog	g Trunks Defined.
raditional telephony lines.				

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.

Edit Analog Trunk					X
	Channels:	2			
Irv	unk Name 🛈 :	trunk2			
	CallerID :				
Normally you should not b calibration. Should you s use		ine tune your aud	-	FOIL	2 Soft 💌
		Advanced Optio	ns		
Busy Detection $①$:	Yes 💙		1	Busy Count 🛈 :	3
Ring Timeout 🛈 :	8000				
Answer on Polarity Switch 🛈 :	No 💙		Polar	Hangup on ity Switch 🛈 :	No 💌
Call Progress 🛈 :	No 💌		Pro	gress Zone 🛈 :	CA 💌
Use CallerID 🛈 :	Yes 💙		Calle	r ID Start 🛈 :	Ring 💌
CallerID 🛈 :	As Received 💌		1	Pulse Dial 🛈 :	No 💙
CID Signalling 🛈 :	DTMF (Denmark, Sw	eden, Holland) 💌		mailbox :	~
Flash Timing 🛈 :	750		Receive Fl	ash Timing 🛈 :	1250
		🛇 Cancel 🛛 🗹 Upd	ate		

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 **W** 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:

Create New SIP/IAX trunk	X
Туре:	SIP V
Provider Name 🛈:	siptrunk1
Hostname :	192. 168. 1. 213
Username :	500
Fromuser :	
Fromdomain :	
Password :	500
Contact Ext.:	
Insecure Type:	very 💌 🛈
	Cancel

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:

Edit Calling Rule X
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 🛈 siptrunki 🖂
Strip 🛈 📃 digits from front
and Prepend these digits 🛈 📃 before dialing
Cancel

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:

DisiFians 🚸
♣ Rex DialPlans
A Dial Flam is a collection of Outgoing Call Rules . Dial Flams are assigned to Users to specify the dialing permissions they have. For emample, you night have
one Dial Flan for local calling that only permits users of that Dial Flan to dial local numbers, via the "local" outgoing calling rule. Another user may be permitted to dial long distance numbers, and so would have a Dial Flan that includes both the "local" and "longdistance" outgoing calling rules.
No DialPlans defined !!

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

Create Hew DialPlan	
DialPlan Nanc:	DialPlanl
Include Outgoing Calling Rules:	♥ outgoing1
Include Local Contexts:	edafault Parkedcalls Conferences Fringgroups Voiceneous Voicensilgroups Edirectory
	Stancel Save

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:

🕄 System Status	User Extensions on PBX 🔅	
C) Trunke C) Outgoing Calling Rules	✤ Dreate Hew User Nodify Selected Users X Delate Selected Users	List of User Extensions
CI Dial Plans		No users created !!
C3 Users		
Users is a shortcut for		
quickly adding and removing		
all the necessary		
configuration components		
for any new phone.		



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 X
General : Extension: 6003 ① Name: 6003 ① DialPlan: DialPlan ①
CallerID: 6003 ① OutBound CallerID: 6003 ①
Image: Second state of the second s
Technology SIP ① IAX ① Analog Station: None ① Codec Preference : First : u-law Second : GSM Mone V
VoIP Settings MAC Address : ① Line Number : 1 ② ③ SIP/IAX Password: ① NAT: ⑦ ① Can Reinvite: ① DIMF Mode: RFC2833 ♥ ④ insecure: very ♥ ④
Other Options 3-Way Calling (I) In Directory (I) Call Waiting (I) CTI (I) V Is Agent (I) Pickup Group: 1 V
Cancel ↓ Update

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 ① Name: 6005 ② DialPlan: DialPlan ③ ③ CallerID: 6005 ③ OutBound CallerID: 6005 ③
Image: Second structure Image: Second structure Image: Second structure Image: Second structure
Technology SIP ① IAX ① Analog Station: Port 4 ♥ ① flash ①: 750 rxflash ①: Codec Preference : First : u-law ♥ Second : GSM ♥ Third : None ♥
VoIP Settings MAC Address : 6005 ① Line Number : 1 Image: Can Reinvite: ① DIMF Mode: RFC2833 ① insecure: Very
Other Options 3-Way Calling (1) In Directory (1) Call Waiting (1) CTI (1) V Is Agent (1) Pickup Group: 1 V
S Cancel ☑ Update

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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.

Attension: 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	Fact	tory R	leset	Reboot	Advanced Options
	Globa	l OutBound CID 🛈	:				
	Oper	ator Extension 🛈	~				
		Ring Timeout 🛈	: 20				
		Extension prefer	ences:				
		User Extensions :	6001	to	6299		
	Confe	rence Extensions :	6300	to	6399		
	Voic	eMenu Extensions :	7001	to	7100		
	Ring	Group Extensions :	6400	to	6499		
	(Queue Extensions :	6500	to	6599		
	VoiceMail (Group Extensions :	6600	to	6699		
		Reset to defau	lts				
		♥ Cancel Save]				

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:

age RingGroupα Φ
* Res Ringfrom Manage KingGroups
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	X
RingGroup Name :	ringgroup1
Extension for this ring group :	6400
Ring Group Members	Available Users
6003 (SIP) 6003 6002 (SIP) 6002	≪≪ 6001 (SIP) 6001 6001 (IAX2) 6001
~	
Ring Group Options :	
Strate	egy : Ring in Order 💌
Seconds to ring each memb	ber : 20
If not answered Go	
	© 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:

System Status	Queues 🗘		
[] Trunks			
[] Outgoing Calling Rules	Queues Agont Login Sottings		
[] Dial Plans	🔶 Create New Queue	Manage Queues	
[] Users			No Call Queues defined !!
🖸 Ring Groups			No carl gledes dernied ::
🕻 Music On Hold			
[] Call Queues			
Call queues allow calls to			
be sequenced to one or more agents.			

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

Edit Queue 6500 X
Extension : 6500 () Name : queue1 ()
Strategy : ringall 💌 🛈 Music On Hold : default 🗹 🛈
LeaveWhenEmpty : Strict 🕶 🛈 JoinEmpty : No 💌 🛈
Queue Options:
TimeOut: 15 (1) Wrapup Time: 0 (1) Max Len: 0 (1)
🗋 🛈 Auto Fill 📄 🕕 Auto Pause 📄 🕕 Report Hold Time
KeyPress Events : None 🔽 🛈
Agents: 1 0 002 (6002) 0 6003 (6003)
© Cancel ☑ Vpdate

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 **W** 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:

C1 System Status	Manage Voice Menus 🔌				
C1 Trunks	+ Create New VoiceMerry		Voice Menus		
23 Outgoing Calling Rules					
C) Dial Plans	Label	Extension	Dial Other Extensions	Key Press Actions	
🕃 Users	ivr	7001	Yes	Yes	Edit 🗶 Dalata
🕃 Ring Groups					
() Music On Hold					
Call Queues					
() Voice Menus					
Menus allow for more					
efficient routing of calls from					
incoming callers. Also					
known as IVR (Interactive					
Voice Response) menus or Digital Receptionist.					

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

Edit VoiceMen	u ⊽oicemenu-custom-2			X		
Name:	voicemenul	٦	Advanced Edit			
Extension:	7002					
V ()	Allow Dialing Other Extensions					
Actions 🛈	Answer the call Play 20046111556565.al & Donot Listen for Ke Goto User 6001	vPress events		© @ © © @ ©		
Add new Step:	Select an Option 💌					
	Illow KeyPress Events					
0 Got	o Operator					
1 Got	1 Goto RingGroup ringgroup1					
2 Got	o User 6001					
3						
4						
5						
6						

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,



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:

[] System Status	Tine Intervals 9
[]Trunks	
Clutgoing Calling Rules	+ Her Max Interval Time Intervals
	line Interval Name Thom
Time Intervals	
Time Intervals are defined ranges of time that will be used by call routing features.	

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

New Time Interval	X
Time Interval Name :	timeinterval1
•	By day of week
	Mon 💌 to Fri 💌
0	By Days of a Month
	Date : Month :
Time:	Entire Day
	Start Time : 09:00 AM End Time : 06:30 PM
	S Cancel ☑ Update

- **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:

🕄 System Status	Incoming Calling Rules Ø			
 Trunks Outgoing Calling Rules 	+ New Incoming Lule Incoming Calling H	tules		
() Dial Plans () Users	Note: If you have multiple SIP trunks from the same p	rovider, you'll need to make Incoming (from that provider. <u>Example</u>		destination on ALL trunks
C Ring Groups		Irunk – siptrunkl		
C Music On Hold	Time Interval	Pattern	Destination	Sort
C3 Voice Menus		Trunk - trunk2		
Time Intervals	Time Interval		Destination	
C3 Incoming Calling Rules Create, modify, prioritize and delete incoming call rules based on Time Intervals.				

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

New Incoming Rule
Trunk : trunk2 💙
Time Interval : timeinterval1 🗸
Pattern 🛈 : s
Destination : VoiceMenu voicemenul 💌
S Cancel ☑ Update

- 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:

[] System Status	General VoiceMail Settings 单
[] Trunks	
[] Outgoing Calling Rules	General Settings Email Settings for VoiceMails SMTP Settings
🖸 Dial Plans	
[] Users	General VoiceMail Settings
[] Ring Groups	Extension for checking messages () 6750
[] Music On Hold	Direct Voicenail Dial (1) :
[] Call Queues	
[] Voice Menus	Max greeting (in seconds) 🛈 : 30
[] Time Intervals	Dial 'O' for Operator 🛈 : 🗹
[] Incoming Calling Rules	
53 Voicemail	Tessage Options
General settings for	Maximun nessages per folder 🛈 : 🗵 🔽
voicemail.	Max message time 🛈 : 2 minutes 🛩
	Min message time 🛈 : 1 second 🛩

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

General Settings	Email Settings for VoiceMails	SMTP Settings
General	VoiceMail Settings	
	for checking messages () 6750	
	: 0100	
	rect Voicemail Dial 🛈 : 🗆	
	eeting (in seconds) ① : 30	
Di	al 'O' for Operator 🛈 : 🗹	
Tessage Op		_
Maximum	messages per folder 🛈 : 25 💌	
	Max message time (1) : 2 minu Min message time (1) : 1 secon	
	min message (ime 🤍 . 📑	
Playback 0	-	
	y message Caller-ID 🛈 : 🗹	
S	ay message duration 🛈 : 🗌	
	Play envelope 🛈 : 🔲	
Al	low users to review 🛈 : 🗹	
	Save ∑ Cancel	

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





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: DialPlani 🗸 🛈			
CallerID: 6005 (1) OutBound CallerID: 6005 (1)			
Enable Voicemail for this User ()			
VoiceMail Access PIN code: 1 Mailbox: 6005 1 Email Address: robert.ao@atcom.			
Technology			
SIP (IAX (Analog Station: Port 4 💌 (flash (): 750 rxflash ():			
Codec Preference : First : u-law V Second : GSM V Third : None V Fourth : None V Fifth : None V			
VoIP Settings			
MAC Address : 6005 ① Line Number : 1 🕑 ① SIP/IAX Password: ①			
NAT: 🗹 🛈 Can Reinvite: 🗌 🛈 DTMF Mode: RFC2833 📝 🛈 insecure: 👽 🐨 🛈			
Other Options			
□ 3-Way Calling ① □ In Directory ① □ Call Waiting ① □ CTI ① ☑ Is Agent ① Pickup Group: 1 ☑			
○ Cancel Vpdate			

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:



🖸 System Status	Manage Conference Rooms 9
C] Trunks	+ New Conference Bridge Conference Rooms
Clutgoing Calling Rules	
🔁 Dial Plans	No Conference rooms defined !!
[] Users	
🖸 Ring Groups	
[] Music On Hold	
C3 Call Queues	
C⊒ Voide Menus	
C3 Time Intervals	
Calling Rules	
[] Voicemail	
C Conferencing	
MeetMe conference bridging allows quick, ad-hoc conferences with or without security.	

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

Edit Conference Bridge 6300 X			
Extension: 6300 🛈	Marked/Admin user Extension :		
	- Password Options:		
Pin Code: 12	3 1 Admin PinCode: 456		
·	Conference Room Options:		
Play hold music for find caller	rst 🔲 🛈 Close conference when last marked user exits		
🗆 🛈 Enable caller menu	Announce callers		
🗖 🛈 Quiet Mode	🔲 🛈 Wait for marked user		
🚫 Cancel 🗹 Update			

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



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:

	Follow Me 🔅			
		Tollouida Destances dest	The second second	
		FollowMe Preferences for U	Jsers FollowMe Options	
		'Follow Me' preference	es for users	
C Ring Groups	Extension	Follow Me	Follow Order	
3 Music On Hold	6001	Disabled	Not Configured	Edit
[] Call Queues	6002	Disabled	Not Configured	Edit
C) Voice Menus	6003	Disabled	Sot Configured	Edit
[] Time Intervals	6004	Disabled	Not Configured	Edit
🕄 Incoming Calling Rules	6005	Disabled	Not Configured	Edit
[] Voicemail	6006	Enabled 6	001	Edit
Conferencing				
[] Follow Me				

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:

	X
Status 🛈 :	○ Enable ⊙ Disable
'Music On Hold' Class 🛈 :	×
DialPlan 🛈 :	DialPlan1 💙
Destinations 🛈 :	
	Add Followie Humber
	Save Save

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



	X
Status 🛈 :	💿 Enable 🔘 Disable
'Music On Hold' Class 🛈 :	
DialPlan 🛈 :	DialPlan1 💟
Destinations 🛈 :	6001 (10 seconds) 📀 🙆 😵
New FollowNe Number 🕕 :	 Dial Local Extension Dial Outside Number
Dial Order 🛈 :	for 30 Seconds 6001 6001 6002 6002 6003 6003 6004 6004 6005 6005 6006 6006 Add

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:



[] System Status	VoiceMail Groups 🖑
13 Trunks	Ter VoiceMail Group List of VoiceMail Groups
Clothering Calling Rules	
[] Dial Plans	No VoiceMail Groups defined !!
[] Users	
[] Ring Groups	
] Music On Hold	
[] Call Queues	
[] Voice Menus	
[] Time Intervals	
🖸 Incoming Calling Rules	
🖸 Voicemail	
Conferencing	
[] Fallow Me	
Directory	
🖸 Call Features	
∐VoiceMail Groups	
Define VoiceMail Groups' to	
leave a voicemail message for a mount of uppers by	
for a group of users by dialing an extension.	

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

New Voice Mail Group		x
VoiceMail Group's Extension:	6600	
Label:	notice	
User MailBoxes:	☑ 6005 ☑ 6006	
	Cancel Save	

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:



ATCOM	
🖸 System Status	Custom Voice Henu Prompts 🔅
C3 Trunks	List of Custom Voice Menu Prompts
C) Outgoing Calling Rules	Record a new Yords Waru prompt Upload a Voice Nanu prompt
53 Dial Plans	
C) Users	He custer Vaice Heru prospts found ()
C Ring Groups	
[] Music On Hold	You can record a new VoiceRenu Prompt by clicking on the 'Record a new Yoice Menu prompt' or click on the 'Upload a Yoice Menu prompt' button to upload a custom voice menu.
🖸 Call Queues	
⊡Maice Menus	
C3 Time Intervals	
53 Incoming Calling Rules	
C3 Voicemail	
Conferencing	
C3 Follow Me	
⊡ Directory	
CC Call Features	
C3 VoideMail Groups	
C3 Vaide Menu Prompts	
Record or Upload custom VoiceMenu prompts.	

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:	WelcomToATCOM
dial this User E prompt:	extension to record a new voice	6001 💌

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:

9
🏅 X-Lite 🛛 🔀
Incoming call from : asterisk
🚄 Answer 🛛 🕿 Ignore

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.

🍐 X-Lite	×
Talking to: asterisk	
0:00:23	🕿 Hang up

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

List of Custom Voice Menu Prompts Record a new Voice Menu prompt Upload a Voice Menu prompt			
2	Name	Options	
1	WelcomToAICOM.gsm	Eccord Again Play Delete	

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

COMMUNICA

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



General Information:

System Information Ø
General Network Disk Usage Memory Usage
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:

System	Information 🗘
Gene	eral Network Disk Usage Memory Usage
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:



stem Informa	ation 🌵						
General	Network	Disk Us	sage	Memory U	Jsage		
isk Usage:							
Disk Usage: Filesystem	1k-b	locks	Used	Available	Use%	Mounted	on
-		locks 14327	Used 13874	Available 453	Use% 97%		on

Memory Usage Information:

System Info	rmation ϕ					
General	Network	Disk Usage	Memory	/ Usage		
Henory Us	age: 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:



ATCOM	
[] System Status	Backup / Restore Configurations 🔍
[] Trunks	Manage Configuration Backups
Dutgoing Calling Rules	manage constraints backaps
🖸 Dial Plans	
[] Users	List of Previous Configuration Backups :
[] Ring Groups	
23 Music On Hold	No Previous Backup configurations found //
[] Call Queues	Please click on the 'Create New Backup' button
C Voice Menus	to take a backup of the surrent system configuration
() Time Intervals	
1 Incoming Calling Rules	
[] Voicemail	
Conferencing	
C Follow Me	
53 Directory	
Call Features	
[] VoiceMail Groups	
C Voice Menu Prompts	
13 System Info	
🖸 Backup	
Backup Management.	

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

Create New Back	Create New Backup X			
File Name:	backup_2010apr26_115450			
	Scancel ☑ Backup			

• 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.

Manage Configuration Backups						
+ Create New Backup						
			 List of Previou 	us Configuration Backu	ips :	
	S.No	Name	Date		Options	
	1	backup_2010apr26_115450	Apr 26, 2010	Download from Unit	Restore Previous Config	🗶 Delete
I						



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:

ATCOM	
[] System Status	Channel Management 🌼
[] Trunks	Refresh Now
COutgoing Calling Rules	Active Channels - 0
CDial Plans	Refreshing Active Charnels in 2 Seconds
[]Users	No Channels Open !!
[] Ring Groups	
[] Music On Hold	
[] Call Queues	
EC Voice Menus	
CCTime Intervals	
[] Incoming Calling Rules	
[]Voicemail	
[] Conferencing	
C3 Follow Me	
C: Directory	
[] Call Features	
53 VoiceMail Groups	
C: Voice Menu Prompts	
[] System Info	
C) Backup	
C3 Active Channels ^{beta}	

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

hannel Manageme	nt 🔍						
Refresh For Active Channels = 3							
Refreshing Active Channels in 5 Seconds							
Channel		Seconds	Application				
Zap/2-1	Ūp	34		Transfer	Hangup		
Zap/3-1	Up	undefined		Transfor	Hangup		
Zap/4-1	Up	41	Disl(\$(ARG2),\$(NINGIINE),\$(DIALOPTIONS))	Trans fer	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:

General Preferences Language Change Password Factory Reset Reboot Advanced Options
Global OutBound CID ① :
Operator Extension 🕕 : User 6001 💌
Ring Timeout 🛈 : 20
Extension preferences:
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
Reset to defaults
Save ∑ Save

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

nced Options							
Language Settings							
Language (1) : English							
Spanish Cancel French							
Language (1) : English English Spanish							

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:

General Preferences	Language	Change Passwo	rd Factory Reset	Reboot	Advanced Options		
Change Password							
	Ente	er New Password:					
	Retyr	oe New Password:					
✓ 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:

General Preferences	Language	Change Password	Factory Reset	Reboot	Advanced Options
	Rese	et to Factory	Defaults		
		configurat	tem to factory defau ion ! configuration from t)		
		Reset to Defaul	ts		

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 \rightarrow **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:

General Preferences	Language	Change Password	Factory Reset	Reboot	Advanced Options
		Advanced Opt	ions		
Clicking the 'Sh	ow Advanced Op	tions' button below hand si	-	itional menu	items on the left
		Show Advanced Op	tions		

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:



[] Options
Admin Settings.
🖸 Asterisk Logs
C3 _{Bulk Add} beta
[] File Editor
[] Asterisk CLI
[] IAX Settings
[] SIP Settings
[] Network Settings
[] Firmware update
C Call Detail Records beta

3.21 Asterisk Logs

After click on **Options** \rightarrow **Advanced Options** \rightarrow **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:

Asterisk Log messages 🚸 🧰 😥

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

Asterisk Log messages 🔅 🌣			(7 0			
	«		Apr	il 2	010		»
	Mon	Tue	₩ed	Thu	Fri	Sat	Sun
	29		31	1	2	3	4
	5	6	7	8	9	10	11
	12	13	14	15	16	17	18
	19	20	21	22	23	24	25
	26	27	28	29	30	1	2

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 🗔

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

Asterisk Log messages 🆃 <mark>21 Apr 2010</mark> 🙃
[Apr 21 03:44:29] WARMING[19672] chan_tap.c: Ignoring insecure
[Apr 21 03:44:29] MARNING[19672] chan_rap.c: Ignoring signalling
[Apr 21 03:44:29] MARMING[19672] chan_rap.c: Ignoring macaddress
[Apr 21 03:44:29] WARKING[19672] chan_tap.c: Ignoring autoprov
[Apr 21 03:44:29] MARHING[19672] chan_rap.c: Ignoring label
[Apr 21 03:44:29] MARHING[19672] chan_rap.c: Ignoring linenunber
[Apr 21 03:44:29] WARKING[19672] chan_sep.c: Ignoring flash
[Apr 21 03:44:29] MARHING[19672] chan_rap.c: Ignoring disallow
[Apr 21 03:44:29] NARHING[19672] chan_rap.c: Ignoring allow
[Apr 21 03:45:16] WARKING[19680] app_dial.c: Unable to create channel of type 'LAX2' (cause 3 - No route to destination)
[kpr 21 03:45:36] NOTICE[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #1)
[Apr 21 03:45:40] MARKENG[19691] ast_expr2.fl: ast_yyerror(): syntax error: syntax error, unexpected '=', expecting \$end; Input:
[Apr 21 03:45:40] MARKING[19691] ast_expr2.fl: If you have questions, plaase refer to doc/channelvariables.txt in the asterisk source.
[kpr 21 03:46:06] MARMING[19691] app_dial.c: Unable to create channel of type 'IAX2' (cause 3 - No route to destination)
[Apr 21 03:48:26] NOTICI[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #2)
[Apr 21 03:47:16] NOTICE[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #3)
[kpr 21 03:47:46] MARXING[211] chan_sip.c: Naximum retries exceeded on transmission 24306203277904-20042119194361092.168.1.3 for seqno 1 (Critical Lesponse)
[Apr 21 03:47:46] MARKENG[211] chan_sip.c: Kanging up call 24806208277904-200421191943818192.168.1.3 - no reply to our critical packet.
[Apr 21 03:48:06] NOTICE[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #4)
[kpr 21 03:48:56] NOTICE[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #5)
[Apr 21 03:49:46] NOTICE[211] chan_sip.c: Registration for '5009192.168.1.213' timed out, trying again (Attempt #6)
[Apr 21 03:50:36] NOTICE[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #7)
[kpr 21 03:51:26] NOTICE[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #8)
[Apr 21 03:52:16] NOTICE[211] chan_sip.c: Registration for '5000192.168.1.213' timed out, trying again (Attempt #9)
[Apr 21 03:53:06] NOTICE[211] chan_sip.c: Hegistration for '5000192.188.1.213' timed out, trying again (Attempt #10)
[kpr 21 03:53:56] NOTICE[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #11)
[Apr 21 03:54:46] NOTICE[211] chan_sip.c: Registration for '5000192.168.1.213' timed out, trying again (Attempt #12)
[Apr 21 03:55:38] NOTICI[211] chan_sip.c: Registration for '5000192.188.1.213' timed out, trying again (Attempt #13)
[Apr 21 03:56:26] NOTICE[211] chan_sip.c: Heristration for '5000192.168.1.213' timed out, trying again (Attempt #14)
[Apr 21 03:57:16] NOTICE[211] chan_sip.c: — — Registration for '5009192.168.1.213' 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:

Bulk Add 🕈	
Bulk Add	
Create New users from CSV list	Create a Range of new users

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

MMUNICATIONS	
	Bulk Add
_	Create New users from CSV list Create a Range of new users
	Create 5 💌 Users Starting from Extension 6100
	Create Users
Tip: Use the 'l	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.**

x
定

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** \rightarrow **Advanced Options** \rightarrow **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.



ile Editor	Φ	users.conf	¥
		Config Files extensions.conf	^
		followme.conf	
		meetme.conf users.conf	
		ztscan. conf	E
		zapscan. conf asterisk. conf	
		queues. conf	
		applyzap. conf	_
		guipreferences.conf rc_org.conf	
		logger.conf	
		sip.conf enum.conf	
		musiconhold.conf	
		dnsmgr.conf rtp.conf	
		iaxprov.conf	
		sip_notify.conf	Y

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

File Editor 🌵 users.conf 💌 New File
users.conf Add Context
+ [500]
+ [6001]
+ [6002]
+ [6003]
+ [6004]
+ [6005]
+ [6006]
+ [general]
+ [trunk_1]

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

MMUNICATIONS	
🌵 Asterisk CLI> he	lp
Command)help	
1	Execute a shell command
	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** \rightarrow **Advanced Options** \rightarrow **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 Int	erface
DHCP:	auto 💌
Hostname:	IP08
Domain:	
IP address:	192.168.1.151
Subnet mask:	255. 255. 255. 0
Gateway:	192.168.1.1
DNS:	192.168.1.1
NTP:	pool.ntp.org

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=SVBQQlgvZmlybXdhcmU=, then put it in your TFTP server root directory.

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

🏘 Tftpd32 by Ph. Jounin	
Current Directory E:\upgrade	<u>B</u> rowse Show <u>D</u> ir
Tftp Server Tftp Client DHCP server Syslog server	
Current Action Listening on port 69	
<u>About</u> <u>S</u> ettings	<u>H</u> elp

"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



After click on **Options** \rightarrow **Advanced Options** \rightarrow **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:

Download image from a :	
○ HTTP URL ③ TFTP Set	ver
TFTP Server :	+ G₀
File Name 🛈 :	
Reset Configs	

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** \rightarrow **Advanced Options** \rightarrow **Show Advanced Options**, please select the **Call Detail Records** option from the vertical menu on the left, then you can get the following screen:

			DR-CSV)												
	DR V	/Iew	/er <u><<</u>	prev ne	xt >	2									
	ing 1-													View:	25
	t rece <u>Account</u>			Dest. Context	<u>Caller</u>	Channel	Dest.	Last app.	last data	Start tine	Answer	Red Time	Durstion	Billable	n:
	Code		June 111 Act of	Derc. Contert	ID.	<u>CREATE</u>	Chennel	Lact app.							br spo
ı			6001	default		Local/60010default-2567	2	SIP/8001- 011 sb340	Dial	SIF/6001&IA#2/6001 20	2010-04- 21 05:34:42		2010-04- 21 05:35:12	30	0
з			60DL	default		Local/60010default-8553	2	SIP/6001- 01250004	Diel	SIF/6001&IAX2/6001 20	21	21	2010-04- 21 05:34:28	18	1
a		6005	6001	DLFN_DialFlani	**6005** 46005>	Zap/4-1	SIP/6001- 01250004	Dial	SIF/6001&IAC2/6001 20	2010-04-21 05:31:21	21	2010-04- 21 05:32:04	43	3	ABSAB
6			5	default		Local/60010default-7c11	2	SIP/8001- 005e6004	YaitExten	6	21	21	2010-04- 21 05:31:47	38	18
7		6006	6750	DLPN_DialPlant	**6008** 660060	SIP/6006-011ab340		YoiceNailVain		2010-04-21 05:24:16	2010-04- 21 05:24:16	21	5t	51	ARSME
в		6005	6600	DLPN_DialPlant	**6005** 66105>	Zap/4-1		VoiceNail	60052 de faul t&60062 de faul t	2010-04-21 05:23:47	2010-04- 21 05:23:50	21	18	15	ARSVE
9		6005	6300	DLPN_Di al Pl ant	**6005** 661052	Zap/4-1		NeetNe	6300 Ms	2010-04-21 04:51:41	2010-04- 21 04:51:43	21	75	73	ARSYE



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:

ATCOM	
[] Home	Velcome to VoIPtel CE
Please login	Asterisk [™] Configuration Engine Username: Password: Login

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:

ATCOM								Apply Changes 7	Logout
🕄 System Status	System Status Ф							Upgrade to Vol	Ptel SE
Please click on a panel to manage related teatures				,	🛙 VolPtel				
			Uptine :		stem Status nin, load average:	0.04, 0.06, 0.02			
					Trunks				
	Status	Irunk	Туре	Username	P	ort/Hostname/IP			
C3 Configure Hardware									
C Trunks					Extensions				
Caling Rules			I Jiee	🔴 Busy	OnAvailable	🥚 Ringing			
🕄 Dial Plans	Extension		N	ans/Labs1		Status	Туре		
C) Users	6750		c	heck Voicemail	Lø		VoiceMailMain		
🕄 Ring Groups	*No Extensio	m assigned	D	ial by Mamez			Directory		
C3 Music On Hold									
C3 Call Queues									
E3 Voice Menus									
C3 Time Intervals									
C3 Incoming Calling Rules									
🕄 Voicemail									
Conferencing									
C) Follow Me									
53 Directory									
😋 Call Features									

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

 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:

• New DialPlan	
DialPlan Name:	DialPlan1
Include Outgoing Calling Rules:	You do not have any calling Rules defined !
	alish have to serve allies when
	click here to manage calling rules.

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 \rightarrow Create New User, I configure user 6001 like the following:

Edit User Extension - 6001 X								
General :								
Extension: 6001 (DialPlan: DialPlani)								
CallerID: 6001 ① OutBound CallerID: 6001 ①								
✓ Enable Voicemail for this User ①								
VoiceMail Access PIN code: 1 Mailbox: 6001 1 Email Address: 1								
Technology								
♥SIP ① ♥IAX ① Analog Station: None ♥ ① flash ①: rxflash ①:								
Codec Preference : First : u-law V Second : GSM V Third : None V Fourth : None V Fifth : None V								
VoIP Settings								
MAC Address : ① Line Number : 1 🕑 ① SIP/IAX Password: ①								
NAT: 🗹 🛈 Can Reinvite: 🗌 🛈 DIMF Mode: RFC2833 🗸 🛈 insecure: very 🗸 🛈								
Other Options								
Other Options 3-Way Calling ① In Directory ① Call Waiting ① CTI ① V Is Agent ① Pickup Group: 1 V								
Cancel ↓ Update								

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: DialPlan1 💌 🛈						
CallerID: 6005 ① OutBound CallerID: 6005 ①						
Enable Voicemail for this User (1)						
VoiceMail Access PIN code: ① Mailbox: 6005 ① Email Address: robert.ao@atcom. ①						
- Technology-						
SIP (1) LAX (1) Analog Station: Port 4 v (1) flash (1): 750 rxflash (1):						
Codec Preference : First : u-law 💙 Second : GSM 💙 Third : None 💟 Fourth : None 💟 Fifth :						
None 💌						
VoIP Settings						
MAC Address : 6005 ① Line Number : 1 💌 ① SIP/IAX Password: ①						
NAT: 🗹 🛈 Can Reinvite: 🗌 🛈 DIMF Mode: 📧 🗹 insecure: 🔽 🐨 🛈						
_ Other Options						
🗆 3-Way Calling 🛈 🗹 In Directory 🛈 🗖 Call Waiting 🛈 🗖 CII 🛈 🗹 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. 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

COMMUNICATIONS

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



\bigcirc			IP Phone			
ATCOM "	Current Status Network VOIP	Advanced Dial-pee	er Config Manage Up	date <u>s</u>	<u>iystem Manage</u>	
					F	Public SIP Configuation
		Basic Setting				
		Register status	Registered		Proxy Server Address	
		Server Address	192.168.1.10	1	Proky Server Port	
		Server Port	5060	1	Proxy Usemame	
		Account Name	6001	ł	Proxy Password	
		Password		(Domain Realm	
		Phone Number	6001	E	Enable Register	
		Display Name	6001			
		-		APP	LY	
				Advance	ed S et	

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		X
	Channels: 🔽 2	
Tr	unk Name 🛈 : trunk2	
	CallerID :	
calibration. Should you	have to adjust your analog po still need to fine tune your e the adjustments at the right	audio settings, please
	Advanced Opt	ions
Busy Detection $①$:	Yes 💌	Busy Count 🛈 : 3
Ring Timeout 🛈 :	8000	
Answer on Polarity Switch 🛈 :	No 💙	Hangup on 🛛 🛚 🔽 No 💌 Polarity Switch 🛈 :
Call Progress 🛈 :	No 💌	Progress Zone 🛈 : 🖸 🗸
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** \rightarrow **New Calling Rule**, I configure an outgoing calling rule like the following:



Edit Calling Rule	x
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 🛈 siptrunki 💟	
Strip 🛈 📃 digits from front	
and Prepend these digits 🛈 📃 before dialing	
Cancel Save	

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** \rightarrow **New DialPlan**, I configure a dial plan like the following:

Edit DialPlan	
DialPlan Name:	DialFlon2
Include Outgoing Calling Rules:	🗹 outgoing!
Include local Contexts:	V default V parkedcalls V conferences V ringgroups V voiceneaus V queues V voicenailgroups V directory
	© Cancel

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



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** \rightarrow **New Incoming Rule**, I configure an incoming calling rule like the following:

Edit Incoming Calling Rule, Trunk: trunk_1 , Time Interval: none X
Trunk : trunk2 🗸
Time Interval : None (no TimeIntervals matched) 💌
Pattern 🛈 : s
Destination : VoiceMenu ivr 💌
🚫 Cancel 🗹 Vpdate

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:

Edit VoiceMen	u voicemenu-custom-2	X
Name:	voicemenul ① Advanced Edit	
Extension:	7002	
	Allow Dialing Other Extensions	
Actions 🛈	Answer the call	♥ 🌢 😒
	Play record/WelcomToATCOM & Donot Listen for KeyPress events	V 🛯 🛛
	Goto User 6001	V 🛯 🛛
Add new Step:	Select an Option 💌	
	① Allow KeyPress Events	
0		
1 Got	o User 6001	
2 Got	o User 6005	
3		
4		

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

COMMUNICATIONS

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.				Firmwar	e Version: 1.05.00
			Etherf	ast® Cable/DSL Router	BEFSR41 V3
Setup	Setup Secu	urity Applications & Gaming	Administration	Status	
	Basic Setup	DDNS	MAC Address Clone	Advanced Routing	
Internet Setup	Static IP			Basic S	
Internet Connection Type	IP Address: Subnet Mask:	172 . 16 . 1 255 . 255 . 0	. 1 . 0	where bas performed (Internet So will require	Setup screen is ic configuration is . Some ISPs ervice Providers) e that you enter the
	Default Gatew Static DNS 1:		. 254	settings ca from your l	nation. These In be obtained SP. After you have these settings,
	Static DNS 2:	0.0.0	. 0	you should password	i set a router from the
	Static DNS 3:	0.0.0	. 0	Administra	
Optional Settings (required by some ISPs)	Host Name: Domain Nam	e:		Setup sect required to	g the Internet tion is all that is set up for your P. Please look at
	MTU:	⊂Enable ⊙Disa	able Size: 1492	the table b	elow to configure for your Internet

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: Subnet Mask:	192 . 168 . 1 . 254 255.255.255.0 ▼	<u>More</u>
Network Address Server Settings (DHCP)	Local DHCP Server: Start IP Address: Number of Address: DHCP Address Range: Client Lease Time: WINS:	 Enable Disable 192.168.1.1 192.168.1.1 to 192.168.1.254 minutes (0 means one day) .00.0.0 	Cisco Systems
		Save Settings Cancel Changes	athusathus

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



LINKSYS [®] A Division of Cisco Systems, Inc.							Firmwar	e Version: 1.05.00
Applications					Ethe	rfast® Cable	/DSL Router	BEFSR41 V3
& Gaming	l Setup	Security	Applicat & Gam		l Administration	Status		
	Port Range	Forwarding		rt Triggerin	g UP	nP Forwarding		DMZ
							Port Ran	ge Forwarding
Port Range Forwarding								e Forwarding can set up public
			Port	Range			services of When use	n your network. rs from the
	Application	Start	End	Protocol	IP Address	Enabled	Internet ma requests o	ake certain In your network,
	IAX2	4569 to	4569	Both 💌	192.168.1. 10			can forward
		0 to	0	Both 🚩	192.168.1. 0		computers	equipped to

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 \rightarrow Create New User, I configure 6020 like the following:

Edit User Extension - 6020 X
General :
Extension: 6020 🛈 Name: 6020 🛈 DialPlan: DialPlan1 🕑 🛈
CallerID: 6020 ① OutBound CallerID: 6020 ①
🖂 🗹 Enable Voicemail for this User 🕕 —
VoiceMail Access PIN code: ① Mailbox: 6020 ① Email Address: ①
Technology ✓SIP ① ✓IAX ① Analog Station: None ♥ ① flash ①: rxflash ①: Codec Preference : First : u-law ♥ Second : GSM ♥ Third : None ♥ Fourth : None ♥ Fifth : None ♥
VoIP Settings MAC Address : Image:
Other Options 3-Way Calling (1)
Cancel ✓ Update

COMMUNICATIONS

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	Current Status Network VOIP &	dvanced Dial-peer	Config I		
• WAN Config		WAN Status			
LAN Config		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 📀	рнср 🔿		PPPOE O
		Auto DNB			
		Static IP Address		172.16	1.2
		Netmask	[255.25	500
		Gateway	[172.16	1.254
		DNS Domain	[
		Primary DNS		202.96	.134.133
		Alter DNS		202.96	.128.68
					(APPLY)

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

ATCOM	Gurrent Status Network VOIP /	IP Phone	System Mana	ige
a 610.1		IAX2		
• <u>SIP 1</u> • <u>SIP 2</u>		Register Status	Registered	
• <u>IAX 2</u>		WK2 Server Addr	172.16.1.1	
		IAX2 Server Port	4569	
		Account Name	5020	
		Account Password		
		Phone Number	60.20	
		Local Port	4559	
		Voice Mail Number	D	
		Voice Mail Text	mail	
		Echo Test Number	1	
		Echo Test Text	echa	
		Refresh Time	60	Seconds
		Enable Register		
		Enable 0.729		
		AP	PLY	

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:

Edit User Extension - 6035				
General : Extension: 6035 ① Name: 6035 ① DialPlan: 💌 ①				
CallerID: 6035 (1) OutBound CallerID: 6035 (1)				
Enable Voicemail for this User VoiceMail Access PIN code: Imail Mailbox: 6001 Imail Address:				
Technology VSIP I VIAX I Analog Station: None V Item V Codec Preference : First : U-law V Second : OSM V Third : None V				
VoIP Settings MAC Address : ① Line Number : ① SIP/IAX Password: 6035 ① NAT: ⑦ Can Reinvite: ① DIMF Mode: RFC2833 ♥ ① insecure: very ♥ ①				
Other Options 3-Way Calling ① In Directory ① Call Waiting ① CTI ① Is Agent ① Enable Call Record ① 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. 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:

Provider Name 🛈:	siptrunk1
Hostname :	192.168.1.20
Username :	6035
Fromuser :	
Fromdomain :	6035
Password :	6035
Contact Ext.:	S
Insecure Type:	very V
Codecs :	
CallerID 🛈 :	Fourth : G.726 V Fifth : None V
🔲 Enable Remote MWI :	

COMMUNICATIONS

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:

ANSEL

Edit Calling Rule	X
Calling Rule Name 🛈 : outgoing2	
Pattern ① : _9.	
🗌 Send to Local Destination 🛈	
Destination :	
Send this call through trunk:	
Use Trunk 🛈 siptrunki 💌	
Strip 🛈 1 digits from front	
and Prepend these digits (1) before dialing	
🗌 Use FailOver Trunk 🛈 :	
fail over Trunk 🛈 siptrunki 💌	
Strip 🛈 📃 digits from front	
and Prepend these digits ① 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 ♥voicemenus ♥queues ♥voicenailgroups ♥directory
	Scancel Seve

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 **DialPaln** 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 \rightarrow Create New User, I add a user 6008 like the following:

Create New User X
General :
Extension: 6008 🛈 Name: 6008 🛈 DialPlan: DialPlan2 💽 🛈
CallerID: 6008 (1) OutBound CallerID: 6008 (1)
🗌 Enable Voicemail for this User 🛈
VoiceMail Access PIN code: 1 Mailbox: 6008 1 Email Address: 1
Technology ✓ SIP ① ✓ IAX ① Analog Station: None ♥ ① flash ①: 750 rxflash ①: 1250 Codec Preference : First : u-law ♥ Second : GSM ♥ Third : None ♥ Fourth : None ♥ Fifth : None ♥ None ♥
VoIP Settings
MAC Address : ① Line Number : 1 🗸 ① SIP/IAX Password: 6008 ①
NAT: 🗹 🛈 Can Reinvite: 🗌 🛈 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. 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		
Туре:	SIP 💌	
Provider Name 🛈:	siptrunktoIP08	
Hostname :	192.168.1.10	
Username :	6008	
Fromuser :		
Fromdomain :		
Password :	6008	
Contact Ext.:		
Insecure Type:	very V	

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

COM

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

COMMUNICATIONS

	Callir	ug Rule Nam	e 🛈 :	outgoing1		
		Patter	n 🛈 :	_9.		
🗖 🗖 se	and to Local Des	stination 🤇	D			
	De	stination	-	~]	
Send ·	this call throug	gh trunk:—				
	U	se Trunk 🛈	siptru	nktoIPO8 💌		
		Strip 🚺) 1 d:	igits from front		
a	nd Prepend these	e digits 🛈		before dialing		
- 🗆 U:	se FailOver Trur	uk 🛈 :				
	fail ov	er Trunk 🛈		~		
		Strip 🚺) d:	igits from front		
	nd Prepend these	o digita (Î		before dialing		

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:	DialFlani
Include Outgoing Calling Rules:	V outgoing 1
Include Local Contexts:	⊠defsult ⊠parkedcalls ⊄comferences ⊄ringgroups ⊄voicenenus ⊄queues ⊄voicenailgroups ⊄directory
	Scanel Sov

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 **DialPaln** 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.



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

WinSCP Login		? 🔀
Session Stored sessions Directories SSH Preferences	Session <u>H</u> ost name 192.168.1.10 <u>U</u> ser name root Private <u>k</u> ey file Hint: Use "Stored sessio	Po <u>r</u> t number 22 Password ********** ons" tab to save your settings
<		
Advanced options		
<u>A</u> bout		Login Close

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

Local Hark Connanda Service. Options Henote B				
	[2] [음 12] 🖬 [12] [21] (21 10) 프 신 프 영 10	<mark>□</mark> / (nt ♥ (= + ⇒ -))	- A D L D	
CV		1		
Nowe V Size Trpe	Cheoged Atty		Chenged Rights	
○ 351Dec 文作史	2010-5-4 sh	<u>e</u>	2010-5-10 2 EVENTRATE	0
Doruments and Sett 文件关	2009-10	Din	2010-5-6 1:17	0
Contact 文件奏	2010-1-2h	in the second se	2010-5-10 2 ruxr-xr-x	0
Intel 文作表	2009-10	i sta	2010-5-10 2 rver-sra	0
○ HERICa.che 文件先 ○ Frogram Files 文件先	2010-2-4 hr 2010-3-1 r	Dane Data	2010-5-6 1:07 EVENTR	0
	2010-5-1 r 2009-10 sh		2010-5-6 1:17 ruxr-zr-z	a 0
□ RECUCLER. 文件実 □ Srsten Volume Info 文件実	2009-10 20	Densi Densistent	2006-12-81 EVENTERTE 2010-5-6 1:08 EVENTERTE	1000
□ Sriten Volume Info 又样是 □ teng 文件表	2010-1-1 5h 2010-2-1	Dersistent Date	2010-5-6 1:00 EVENTER 2006-12-31 ETENTER	1000
□ Tang 又1+头 □ uxr 文件楽	2010-2-1	in the second se	2010-5-6 1:07 rwxr-zr-z	0
□uxr 又作来 □wUXD045 文化素	2010-3-1	📮 root	2010-5-6 1:17 FURST-11-1	0
□ n.1305 0 (1) 124 880 文件	2010-3-1 a	G 575	2006-12-31 1967-31-2	0
an that in the standard all hope ulaw 24,640 ULAN 文作	2000-7-1 a	0 tap	2010-5-10 2 EVERTHERE	8
AUTOEXEC. DAT 0 MS-DOS 4241	2009-10 9	- 157	2010-5-6 1:17 EVENTURE	0
bar enf 393.612 ENF	2010-4-1 a			ŝ
Abort. ini 242 FEE RE	2010-1-2 mbm			*
abootfont.bin. 822,780 BIH 文件	2008-4-1 anhr			
CONFRIG.STS 0 系統文件	2009-10 9			
■GHLOK 224, 121 文件	2010-1-2 sha			
Install.log 197 文本文档	2009-10 a			
□ 10.818 0 承載文件	2009-10 ashr			
🗐][] 31/37/7_log.tg1 0 文本文档	2010-5-4 a			
■ MERIOS. SVS 0 系統文件	2009-10 ashr			
Intragenet.com 47,564 K3-D08 広用程序 回 atldy 257,728 文件	2008-4-1 unkr			
🔤 ntidr 257,728 文件	2008-4-1 a.shr			
🗟 parefile, 573 2, 143, 系發文件	2010-5-1 a.sh.			
BHISetup.log 1,519 支本支格	2009-10 x			
🖹 service log 78 文本文档	2010-5-1 a			
0 D of 2,047 MD Lo. 0 of 27		DB of 0 D in D of 14		
#F2 Banama Bars Copy	The Roma 77 Create directory	X FI Delata	P9 Properties	7.0910 Disconnect
636 KD 8, 100 D 🔯 🗿 🗠 🗤 🕫	-			

The left part of the screen displays directories and files of your windows PC, the right part of the screen displays Agricultura 111 Piso 1, Colonia Escandón, C.P 11800 México DF 55 52714421, 52774459, 52719163 Fax: 52718216. ventas@ansel.com.mx



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:

Сору
Copy file 'atcom_ip02_04_08=0.3.4=1207.md5' to remote directory
/persistent/
Copy Cancel <u>M</u> ore >>

Chapter 5 Reference

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