



Figure: ATCOM IPPBX IP01

ATCOM[®] IPPBX IP01

Product Guide

Version: 1.0

2010-05-10



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Chapter 1 the Introduction of IP01

Overview of the IP01

The IP01 is a complete Asterisk Appliance with one single port 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, IP01 provides a cost-saving solution on their telecommunication/data needs. With IP01, 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

1 * RJ45 port 1 * Power port 1* RJ11 port (FXS/FXO interchangeable) 1* Single port FXO/FXS module slot

Hardware

CPU: 400MHz Blackfin 532 Chip One analog (FXO/FXS) module interface



5

NAND flash 256 MB SDRAM 64MB

System

Open Source uClinux

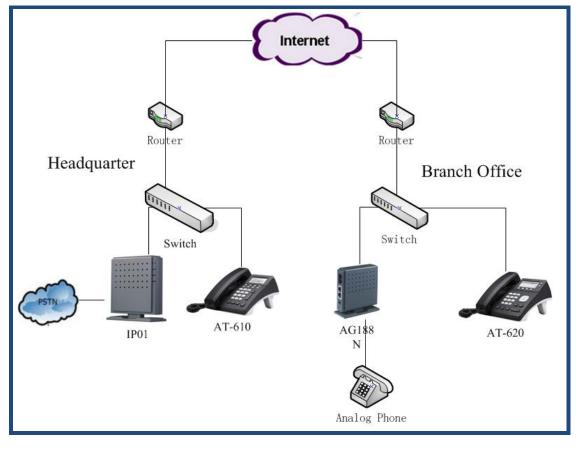
Measurement and Weight

Inner box	100 * 100 * 28mm
G.W./unit	0.36KG
Carton MEAS	456 * 442 * 362 mm
Units per Carton	40 units/ CTN
G.W./CTN	17 KG/CTN

Package

Item	Quantity
IP01	1
RS232 module	1
Power Adapter	1
Manual (disk)	1

For the usage of IP01 in VoIP field, you can refer to the following network topology.



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Chapter 2 Access to the IP01

You need a PC to access to the IP01, there are four ways for you to access the IP01:

- 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 IP01 by the first three ways, you have to check that if your network connection between IP01 and PC is OK. If you do not have network connection between IP01 and PC, you can try to use the last way to access to IP01 and change the IP address for IP01.

2.1 Web Page Access by Browser

It is the most convenient and common way to access the IP01, you just need to open your browser and input the IP address of IP01 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) in the presented screen like the following:

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

ATCOM '								Apply Changes ? Log
	System Status 🔅							Upgrade to VoIPtel
Please click on a panel to manage related features				K	VolPt∈l			
			Untine		tem Status	0.04, 0.06, 0.02		
			opeimo	. ourrenor up ro	Irunks			
	Status	Trunk	Туре	Username		Port/Hostname/IP		
[] Configure Hardware			-72-					
[] Trunks				1	ixtensions			
COutgoing Calling Rules			👄 Free	🛑 Busy	 UnAvailable 	🥮 Ringing		
[] Dial Plans	Extension		R	ame/Label		Status	Туре	
[] Users	<u>6750</u>		С	heck Voicemail	8		VoiceMailMain	
[] Ring Groups	*No Extensi	on assigned	D	ial by Names			Directory	
🖸 Music On Hold								
[] Call Queues								
🕄 Voice Menus								
🕄 Time Intervals								
[] Incoming Calling Rules								
[] Voicemail								
[] Conferencing								
[] Follow Me								
[] Directory								
[] Call Features								

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



2.2 SSH Access by Putty

Logging into IP01 by SSH, you can configure IP01 by Linux command.

 Please open your putty software, and input the IP01 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:

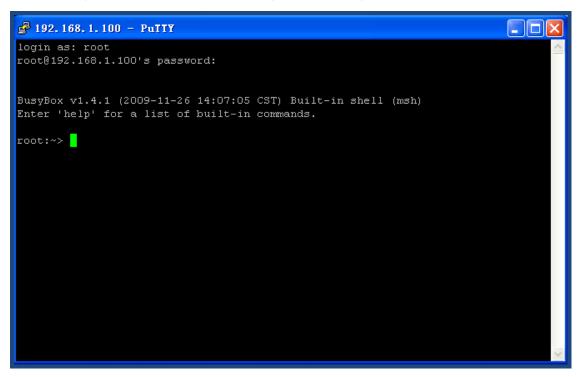
😵 PuITY Configurat	ion 🔀
Category:	
 Session Logging Terminal Keyboard Bell Features Window Appearance Behaviour Translation Selection Colours Connection Data Proxy Telnet Rlogin SSH Serial 	Basic options for your PuTTY session Specify the destination you want to connect to Host Name (or IP address) Port 192.168.1.100 22 Connection type: Image: Serial Load, save or delete a stored session Saved Sessions Default Settings Load Save Delete Close window on exit: Image: Only on clean exit
About	<u>O</u> pen <u>C</u> ancel

2) Please input username: root, and the default password: 12xerXes16 in the following screen, you can access to IP01 successfully.

₽ 192.168.1.100 - PuTTY	
login as: root root@192.168.1.100's password:	A 100
10002192.100.1.100 S passeda.	



When you log into IP01 successfully, you can get the following illustration:



2.3 Access by Browser with Fallback IP Address

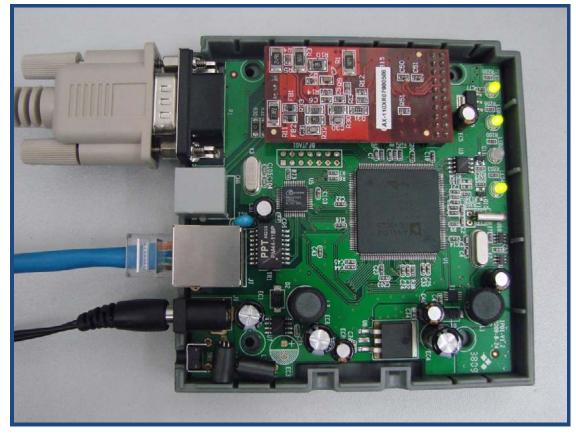
This way only be supported by the latest version (voiptel_ce_ip01-0.3.6) of IP01. If you forget the IP Address of IP01 you have set up, you can use the fallback IP Address: 172.31.255.254/30. Before logging into IP01, please set up the IP Address of your PC: 172.31.255.253 and SubMask: 255.255.255.252. At last, you can open your browser and enter:172.31.255.254 to log into the web page of IP01.

2.4 Console Port Access to IP01

If you do not have network connection between IP01 and PC, you can try to access to IP01 by console port. Please try to do as the following steps:

1. Please connect the console port of IP01 to your PC's console port with RS232 console cable, you can refer to the following illustration:





 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

3. Change the IP Address by Hyper Terminal

The default IP address of IP01 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 IP01 by putty and browser if you do not change the IP01 IP address. So you have to change the IP address for IP01 by Hyper Terminal to make it in the same network segment as your LAN.

After you have accessed to IP01 by Hyper Terminal, please use the following command to change the IP address for IP01.

root:~> ifconfig eth0 192.168.1.151(the IP address you want to set for IP01)

By this way, the IP address you set for IP01 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 IP01, you can try to set it in web GUI, for detail steps please refer to chapter 3.



Chapter 3 Configure IP01 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						? Logout
[] System Status		Up	time : 05:17:03 up	2 min, load average: 0.18,	0.08, 0.02	
Please click on a panel to manage related features				Trunks		
inanage relates leareres	Status	Irunk	Туре	Username	Port/Hostname/IP	
	Unregistered	siptrunk1	sip	6035	192.168.1.20	
		trunk2	Analog		Ports 2	
ra Casterina Handaran		Agents	4		2	
C3 Configure Hardware				002 6003 gin Login	6005 Login	
C Outgoing Calling Rules			2			
Dial Plans			6020 Login			
[] Users			Login			
[] Ring Groups			Confere	nce Rooms		
23 Music On Hold			N	6300 ot in use		
 Call Queues						
[] Voice Menus			•	Extensions		
C Time Intervals		😐 Fr		UnAvailable	Ringing	
[] Incoming Calling Rules	Extension		me/Label	Status	Туре	
[] Voicemail	• <u>6001</u>	60	01	Messages : 1/ Messages : 0/		
[] Conferencing	 6002 6003 		02	Messages : U/ Messages : O/		
C3 Follow Me	6003		04	Messages : 0/		

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 IP01, 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:

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[] System Status	Manage Analog trunks	φ		
[] Configure Hardware [] Trunks	Analog Trunks	Service Providers	VOIP Trunks	T1/E1/BRI Trunks
Trunks are outbound lines used to allow the system to	+ New Analog Trunk			
make calls to the real world. Trunks can be VoIP lines or traditional telephony lines.			No Analog	Trunks Defined.

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.

New Analog Trunk		Σ	2
	Channels:	: 🗹 1	
Tı	unk Name 🕕 :	trunk1	
		Advanced Options	
Busy Detection $①$:	Yes 💟	Busy Count 🛈 : 3	
Ring Timeout 🛈 :	8000		
Answer on	No 🗸	Hangup on 🛛 💌	
Polarity Switch 🛈 :		Polarity Switch 🛈 :	
Call Progress 🛈 :	No	Progress Zone 🛈 : 🔽 💌	
Use CallerID 🛈 :	Yes 💟	Caller ID Start 🛈 : 🛛 🕅 💌	
CallerID 🛈 :	As Received 💟	Pulse Dial 🛈 : 🕨 💌	
CID Signalling 🛈 :	Bell - USA	▼ mailbox : ▼	
Flash Timing 🛈 :	750	Receive Flash Timing 🛈 : 1250	
		♦ Cancel ✓ Add	

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 1.

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 trunk1 as my trunk name.

For the advanced options, you can put your cursor on the **(1)** 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 💌
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:

[] System Status	Tanage Calling Rules 🔍
[] Trunks [] Outgoing Calling Rules	◆ Ter Calling Rale Restore Defualt Calling Rales Outgoing Calling Rules
Calling Rules define dialing permissions and routing rules.	An outgoing calling rule pairs an extension pattern with a trunk used to dial the pattern. This allows different patterns to be dialed through different trunks (e.g. "local" 7-digit dials through a FRI but "long distance" 10-digit dials through a low-cost SIP trunk). You can optionally set a failower trunk to use when the primary trunk fails. Note that this panel manages only individual outgoing call rules. See the Dial Plans section to associate multiple outgoing calling rules to be used for User outbound dialing.
	No CallingRules defined !!

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



New CallingRule	X
Calling Rule Name 🛈 : outgoing1	
Pattern 🛈 : _2x.	
🔽 🗖 Send to Local Destination 🛈 ———————————————————————————————————	-
Destination :	
- Send this call through trunk:	7
Use Trunk 🛈 trunk1 💌	
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	

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 trunk1 I set up before. **Strip**: I press 1 here, it will strip the first digit of the number string you dialed.

You can get the detail information about every single parameter by putting your cursor on the ①

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:

[] System Status	DialPlans Ø
[] Trunks [] Outgoing Calling Rules	◆Ner DialPlans Manage DialPlans
C3 Dial Plans A DialPlan is a set of "Calling Rules" that can be	A Dial Plam is a collection of Outgoing Call Rules . Dial Plams are assigned to Users to specify the dialing permissions they have. For example, you might have one Dial Plam for local calling that only permits users of that Dial Plam to dial local numbers, via the "local" outgoing calling rule. Another users may be permitted to dial long distumen numbers, and so vould have a Dial Plam that includes both the "local" and "longdistance" outgoing rules.
assigned to one or more users.	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 New DialPlan	
DialPlan Name:	DialPlan1
Include Outgoing Calling Rules:	☑ outgoing1
Include Local Contexts:	♥default ♥parkedcalls ♥conferences ♥ringgroups ♥voicemenus ♥queues ♥voicemailgroups ♥directory
	© Cancel

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:



Create New User
General : Extension: 6001 ① DialPlan: DialPlan: DialPlan1 ① CallerID: ① ①
Image: Constraint of this User VoiceMail Access PIN code: Image: Constraint of this User Image: Constrated of this User Imag
Technology Image: SIP Image: SIP Image: Signed and Station: None Image: Image: Signed and Station: None Image: Signed and Station: Signed and Station: None Image: Signed and Station: None Image: Signed and Station: None Image: Signed and Station: Signed and Station
VoIP Settings MAC Address : Image: Construction of the set of the
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

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 6001, 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:



Create New User	x
General : Extension: 6002 ① DialPlan: DialPlan1 ① CallerID: 6002 ① OutBound CallerID: ①	
Image: Constraint of the second se	
Technology SIP ① FIAX ① Analog Station: Port 1 ♥ ① flash ①: 750 rxflash ①: 1250 Codec Preference : First : u-law ♥ Second : GSM ♥ Third : None ♥ Fourth : None ♥ Fifth : None ♥	
VolP Settings MAC Address : ① Line Number : 1 ♥ ① SIP/IAX Password: ① NAT: Image:	
Other Options Other Options 3-Way Calling ① In Directory ① Call Waiting ① Enable Call Record ① Pickup Group:	
Cancel Update	

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 6002, 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:

								AICUM
	General Preferences	Language	Change Passwo	ord	Facto	ry Reset	Reboot	Advanced Options
	Global O	utBound CID 🛈 :						
	Operat	or Extension 🛈 : [*					
	F	Ring Timeout 🛈 :	20					
		Call Record Dir :	/tmp					
	Cal	Record Format :	gsm 🖌					
C Ext	ension preferences:							
			Jser Extensions :	6001	to	6299		
		Confere	ence Extensions :	6300	to	6399		
		VoiceN	lenu Extensions :	7001	to	7100		
		RingG	roup Extensions :	6400	to	6499		
		Qu	ieue Extensions :	6500	to	6599		
		VoiceMail G	roup Extensions :	6600	to	6699		
			Reset to d	efaults				
			O Cancel	Save				

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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	Hanage RingGroups 🚯
[] Trunks	+ New Kingdroup Manage RingGroups
[] Outgoing Calling Rules	
[] Dial Plans	No RingGroups defined !!
[] 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.	

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	per : 20
If not answered Go	oto : Mangup
	Save ∑ Cancel

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 Agent Login Settings
🖸 Dial Plans	+ Create New Queue Hanage Queues
[] Users	No Call Queues defined !!
[] Ring Groups	
🖸 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 : queuel ①
Strategy : ringall	🕶 🛈 Music On Hold : default 💌 🛈
LeaveWhenEmpty : Strict 💌 🤇	JoinEmpty : No 💙 🛈
	Queue Options:
TimeOut: 15 🛈	Wrapup Time: 0 (Î) Max Len: 0 (Î)
🗌 🛈 Auto Fill	🗌 🛈 Auto Pause 🔲 🛈 Report Hold Time
KeyPress Events :	None 🕑 🛈
Agents: ① ☑ 6002 (600	
6003 (600	
	○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 ① 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:

	Lanage Voice Lenus 🗳					
[] Trunks	+ Create New Voice Menus					
COutgoing Calling Rules	Label	Extension	Dial Other Extensions	Key Press Actions		
	ivr	7001	Yes	Yes	Edit X Delete	
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 Voicelen	u voicemenu-custom-2			X
Name:	voicemenul	١	Advanced Edit	
Extension:	7002			
I	Allow Dialing Other Extensions			
Actions 🛈	Answer the call Play 20046111556565.al & Donot Listen for Key Goto User 6001	Press events		V Q 8 V Q 8 V Q 8
Add new Step:	Select an Option 💌			
	 Allow 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	Time Intervals 🔅		
[] Trunks	+ New Time Interval	Time Intervals	
[] Outgoing Calling Rules	when The Interval		
[] Dial Plans		Time Interval Name	Then
[]Users			
[] Ring Groups			
C Music On Hold			
[] Call Queues			
[] Voice Menus			
[] 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:

Time Interval Name :	timeinterval1
۲	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
	🚫 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:



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	X
Trunk :	trunk2 V
Time Interval :	timeinterval1 💌
Pattern 🛈 :	s
Destination :	VoiceMenu voicemenul 💌
	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 voicemenul, 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 🕴
[] 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 Voicemail Dial 🛈 : 🗆
[] Call Queues	
[] Voice Menus	Max greeting (in seconds) 🛈 : 30
[] Time Intervals	Dial'O' for Operator 🛈 : 🗹
[] Incoming Calling Rules	Tessage Options
[] Voicemail	
General settings for	Maximum messages per folder 🕕 : 25 💌
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 VoiceMa	ils SMTP Settings
General	VoiceMail Settings	
Extension	for checking messages $\textcircled{0}$:	6750
Di	rect Voicemail Dial 🛈 : [
Max gr	eeting (in seconds) 🛈 :	30
Di	al 'O' for Operator 🛈 : [•
Tessage Op	otions	
Maximum	messages per folder 🛈 : [25 💌
	Max message time 🛈 :	2 minutes 💌
	Min message time 🛈 : [1 second 💉
Playback (ptions	
Sa	y message Caller-ID 🛈 : [~
S	ay message duration 🛈 : [
	Play envelope 🛈 : [
Al	low users to review 🛈 : [
	○ Cancel Save	

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 ① 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:



Create New User	х	
General : Extension: 6001 ① DialPlan: DialPlan: ① CallerID: ⑥		
Image: Contract of the second seco		
Technology SIP ① FAX ① Analog Station: None ♥ ① flash ①: 750 codec Preference : First : u-law ♥ Second : GSM ♥ Third : None ♥ Fourth : None ♥ Fifth : None ♥		
VoIP Settings MAC Address : ① Line Number : Image: Standard Structure ① SIP/IAX Password: NAT: ⑦ DTMF Mode: RFC2833 ① insecure: Very ①		
Other Options Other Options 3-Way Calling ① In Directory ① Call Waiting ① Enable Call Record ① Pickup Group: 1		
Cancel Update		

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	Hanage Conference Rooms 🔅
[] Trunks	+ New Conference Bridge Conference Rooms
[] Outgoing Calling Rules	
[] Dial Plans	No Conference rooms defined !!
[] Users	
[] Ring Groups	
[] Music On Hold	
[] Call Queues	
[] Voice Menus	
[] Time Intervals	
[] Incoming Calling Rules	
[] Voicemail	
[] 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:

	ATCOM
Edit Conference Bridge 6300	X
Extension : 6300 ()	Marked/Admin user Extension :
	- Password Options:
Pin Code: 12	3 1 Admin PinCode: 456 1
c	Conference Room Options:
Image: Play hold music for fir caller	st 🔲 🛈 Close conference when last marked user exits
🗆 🛈 Enable caller menu	Announce callers
🗆 🛈 Quiet Mode	🔲 🛈 Wait for marked user
	Cancel Vpdate

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:

	Follow IIe 🔅			
[] Outgoing Calling Rules		FollowMe Preferences for U	sers FollowMe Options	
		'Follow Me' preferenc	es for users	
[] Ring Groups	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 60	01	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 FollowNe Number
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 O Dial Outside Number
Dial Order 🛈 :	for 30 Seconds 6001 6001 6002 6002 6003 6003 6004 6004 g with previous extension/number 6005 6005 6006 6006

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.

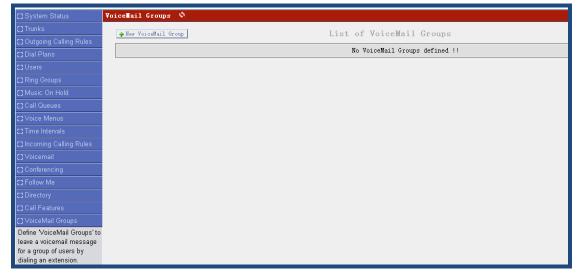
Through the above settings, someone calls 6006, but 6006 does not answer, the call will be transferred to 6001 automatically.



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:

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

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 Menu Prompts 🔌
[] Trunks	List of Custom Voice Menu Prompts
Cloutgoing Calling Rules	Record a new Voice Menu prompt Upload a Voice Menu prompt
[] Dial Plans	
[] Users	No custom Voice Menu prompts found //
[] Ring Groups	
C3 Music On Hold	You can record a new VoiceMenu Prompt by clicking on the 'Record a new Voice Menu prompt ' or click on the 'Upload a Voice Menu prompt' button to upload a custom voice menu.
[] Call Queues	
C Voice Menus	
[] Time Intervals	
[] Incoming Calling Rules	
[] Voicemail	
[] Conferencing	
[] Follow Me	
[] Directory	
[] Call Features	
[] VoiceMail Groups	
[] Voice 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 Extens: prompt:	ion 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:

A via	
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 asteri			
0:00:23	🕿 Har	ig 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 Vpload a Voice Menu prompt				
#	Name	Options		
1	WelcomToATCOM.gsm	Record 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

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

```
System Information 🔅
   General
               Network
                          Disk Usage
                                      Memory Usage
 OS Version:
 Linux IPOx 2.6.22.18-ADI-2008R1astfin-svn #2 Wed May 5 23:59:04 EDT 2010 blackfin unknown
 Uptime:
 02:01:35 \quad {\tt up}=3:19,\\
 Load Average: 0.64, 0.29, 0.10
 Version Details:
 Asterisk/1.4.21.2
 VoIPtel GUI version: 2.0.2-ce
 Firmware version: voiptel_ce_ip01-0.3.6
 Server Date & TimeZone: Sat May 15 02:01:36 EDT 2010
 Hostname:
  IPOx
```

The latest version of IP01 is voiptel_ce_ip01-0.3.6. You can see the version that you are using from **Version Details** in the above illustration.



Network Information:

Stor 1	Information \$
Gene	ral Network Disk Usage Memory Usage
ethO	Link encap:Ethernet HWaddr 00:09:45:54:49:7E inet addr:192.168.1.184 Bcast:192.168.1.255 Mask:255.255.255.0 UP BROADCAST RUNNING MULTICAST MTU:1500 Metric:1 RX packets:9552 errors:0 dropped:0 overruns:0 frame:0 TX packets:9427 errors:0 dropped:0 overruns:0 carrier:0 collisions:0 txqueuelen:1000 RX bytes:1187671 (1.1 MiB) TX bytes:6930187 (6.6 MiB) Interrupt:48
eth0:9	Link encap:Ethernet HWaddr 00:09:45:54:49:7E inet addr:172.31.255.254 Beast:172.31.255.255 Mask:255.255.255.252 UP BROADCAST RUNNING MULTICAST MTU:1500 Metric:1 Interrupt:48
10	Link encap:Local Loopback inet addr:127.0.0.1 Mask:255.0.0.0 UP LOOPBACK RUNNING MTU:16436 Metric:1 RX packets:44 errors:0 dropped:0 overruns:0 frame:0 TX packets:44 errors:0 dropped:0 overruns:0 carrier:0 collisions:0 txqueuelen:0 RX bytes:4607 (4.4 KiB) TX bytes:4607 (4.4 KiB)

Disk Usage Information:

ysten Inforna	tion Ø						
General	Network	Disk U	sage	Memory U	Jsage		
Disk Usage:							
Filesystem	1k-b)	locks	Used	Available	Use%	Mounted on	
/dev/mtdblock0		14327	13874	453	97%	1	
/dev/mtdblock2	2	53952	75960	177992	30%	/persistent	

Memory Usage Information:

System Info	rmation ϕ					
General	Network	Disk Usage	Memory	Usage		
Tenory Us	age:					
Mem:	total 45928	used 41504	free 4424	shared O	buffers 812	



3.18 Backup

Backup and Restore are two of the mandatory functions of any application. IP01 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:

Create New Back	Create New Backup		
File Name:	backup_2010apr26_115450		
	S Cancel		

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 Configur	ation Backups		
	🔶 Cre	ate New Backup		
	— List of Previous	Configuration Backu	ps :	
S.No Name	Date		Options	
1 backup_2010apr26_115450	Apr 26, 2010	Download from Unit	Restore Previous Config	🗶 Delete



Backup itself is not useful if it cannot be restored, IP01 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
[] Outgoing Calling Rules	Active Channels - 0
[] Dial Plans	Refreshing Active Channels in 2 Seconds
[] Users	No Channels Open !!
C Ring Groups	
🖸 Music On Hold	
[] Call Queues	
[] Voice Menus	
[] Time Intervals	
🖸 Incoming Calling Rules	
[] Voicemail	
Conferencing	
[] Follow Me	
C Directory	
[] Call Features	
[] VoiceMail Groups	
[] Voice Menu Prompts	
[] System Info	
[] Backup	
C Active Channels beta	

Here my IP01 is using 1 channel, so I get the following information :

Channel Managemen					
Refresh Now			Active Channels - 1		
			Refreshing Active Channels in 4 Seconds		
Channel	State	Seconds	Application		
Zap/1-1	Up	5	VoiceMailMain(\$[CALLERID(num)]@default)	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:



General Preferences	Language	Change Password	Facto	ry Re	set	Reboot	Advanced Options	
(1-1-1)	outBound CID 🛈							
Operato	or Extension 🛈	:						
Ring Timeout 🛈 : 20								
	Call Record Dir	: /tmp						
Cal	1 Record Format	: Esm 💙						
 Extension preferent 	ices:							
		User Extensions :	6001	to	6299			
	Confei	rence Extensions :	6300	to	6399			
	Voice	eMenu Extensions :	7001	to	7100			
	Ring	Group Extensions :	6400	to	6499			
	Ģ	Queue Extensions :	6500	to	6599			
	VoiceMail (Group Extensions :	6600	to	6699			
		Reset to defau	lts					
		Q Cancel Sav	e					

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

General Preferences	Language	Change Password	Factory Reset	Reboot	Advanced Options
	L	anguage Settin	gs		
		Language 🛈 : 🛯 En	glish 💙		
		En s Spe	lish mish		
		○ Cancel	nch		

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	ord	Factory Reset	Reboot	Advanced Options
		Change Pass	word			
	Ente	er New Password:				
	Retyr	pe New Password:				
		Vp	date			

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:



Reset to Factory Defaults Warning: By resetting your Asterisk Appliance/System to factory defaults, you will lose all your configuration ! You can take a backup of your current configuration from the <u>Backup page.</u> You can also only reset the PBX config but keep the netowrk setting.	General Preferences	Language	Change Password	Factory Reset	Reboot	Advanced Options
Warning: By resetting your Asterisk Appliance/System to factory defaults, you will lose all your configuration ! You can take a backup of your current configuration from the <u>Backup page.</u>		Re	set to Factory	Defaults		
configuration ! You can take a backup of your current configuration from the <u>Backup page.</u>						
	Warning: By rese	tting your As		-	aults, you v	will lose all your
				-		
			set to Defaults but keep			

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, IP01 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 Opti	ons			
Clicking the 'Show	/ Advanced Opt	ions' button below p hand side		tional menu	items on the left	
		Show Advanced Optic	ns			

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:

Admin Settings.
[] Asterisk Logs
C3 Bulk Add beta
C3 File Editor
[] Asterisk CLI
[] IAX Settings
[] SIP Settings
[] Network Settings
[] Firmware update
Call Detail Records beta



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:

Asterisk Log messages 🌵 😽 🚱							
Click on the textbox, you can get	t the f	ollov	ving	scree	n:		
Asterisk Log messages ϕ				Go			
	«		Арт	il 2	010		>>
	Mon	Tue	₩ed	Thu	Fri	Sat	Sun
	29	30	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 21^{st} ,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] WARNING[19672] chan_zep.c: Ignoring insecure
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring signalling
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring macaddress
[Apr 21 03:44:29] WARNING[19672] chan_rap.c: Ignoring autoprov
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring label
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring linenumber
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring flash
[Apr 21 03:44:29] WARNING[19672] chan_zap.c: Ignoring disallow
[Apr 21 03:44:29] WARNING[19672] chan_zap.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 '5000192.168.1.213' 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[19891] ast_expr2.fl: If you have questions, please refer to doc/channelvariables.txt in the asterisk source.
[Apr 21 03:46:05] WARNING[19891] app_dial. c: Unable to create channel of type 'LAX2' (cause 3 - No route to destination)
[Apr 21 03:46:25] NOTICE[211] chan_sip.c: Registration for '50004192.168.1.213' timed out, trying again (Attempt #2)
[Apr 21 03:47:16] NOTICE[211] chan_sip.c: Registration for '5000192.168.1.213' timed out, trying again (Attempt #3)
[Apr 21 03:47:46] WARING[211] chan_sip.c: Maximum retries exceeded on transmission 24806208277904-200421191943818192.188.1.3 for seqno 1 (Critical Response)
[Apr 21 03:47:46] WARING[211] chan_sip.c: Hanging up call 24006208277804-20021191943610192.168.1.3 - no reply to our critical packet.
[Apr 21 03:48:06] NOTICE[211] chan_sip.c: Registration for '5008192.188.1.213' timed out, trying again (Attempt #4)
[Apr 21 03:48:56] NOTICE[211] chan_sip.c: Registration for '5000192.168.1.213' timed out, trying again (Attempt #5) [Apr 21 03:49:46] NOTICE[211] chan sip.c: Registration for '5000192.168.1.213' timed out, trying again (Attempt #6)
(Apr 21 05:49:40] MULLE[211] char_sip.c. — Registration for Soughts. Iso. 1.215 timed out, trying again (Attempt #0) [Apr 21 05:058] MULLE[211] chan sip.c. — Registration for 'Sought2.188.1.213' timed out, trying again (Attempt #7)
(Apr 21 05:00:00] MOILE[211] char_sip.c. — Registration for Soucies.ico (inclusion) (inclu
Apr 21 05.51.69 MOLEE[211] char_sip.c. — Registration for Souche.cost.clis timed out, trying again (attempt #0) [Apr 21 05.52:16] MOLEE[211] chan sip.c. — Registration for Souche.cost.clis timed out, trying again (attempt #9)
(Apr 21 05.02.10) MOLEC[21] char_sip.c. — Registration for Sourise. Ioo: 1.215 timed out, (rying again (attempt #5) [Apr 21 05.35:06] MOLEC[21] char_sip.c. — Registration for 'Sourise. 1061.215 timed out, trying again (attempt #10)
Apr 21 05:55:55 [MICE[21] chan_sip.c:Registration for Souch2.105 (timed out, trying again (Attempt #1))
Char 21 03:53:46] MOTICE[211] chan sin c: Registration for Source 21, 123 timed out, trying again (Attempt #12)
Apr 21 03:55:36] MOTICE[211] chan_sip.c: Registration for 50000192.168.1.213' timed out, trying again (Attempt #13)
Apr 21 03:56:26] NOTICE[211] cham sip.c: Registration for '5000192.168.1.213' timed out, trying again (Attempt #14)
Apr 21 03:57:16] NOTICE[211] chan.sip.c: Registration for '5000192.168.1.213' 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.

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 '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**.

警报 http://192.168.1.151/		
Vsers added Click Ok to reload GVI		
	确定	

At last, click on **Example** 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:

File Editor 🌵 Config Files 💌 New File

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



File Editor	Φ	users.conf	*	New File
		Config Files	~	
		extensions.conf		
		followme.conf		
		meetme.conf		
		users.conf		
		ztscan. conf		
		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	×.	

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:

	ATCOM
🌵 Asterisk CLI> he	lp
Command> help	
	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 IP01, you have to set it in web GUI. After you enter into the web GUI of IP01, 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 In	eth0 Interface			
DHCP:	auto 🗸			
Hostname:	IPOx			
Domain:	openippbx.org			
IP address:	192.168.1.184			
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: IP01 will obtain the dynamic IP address from your router.
- 2. DHCP: auto: IP01 will use the static IP specified below and ping the default gateway. When there is no response from the default gateway, the IP-04 will switch to dynamically obtain the IP address from your router.
- 3. DHCP: no: IP01 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 IP01 by TFTP.

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

- Download the md5 file from <u>http://www.atcom.cn/downloads/IPPBX/Firmware/IP01-v0.3.6-100506.md5</u>, then put it in your TFTP server root directory.
- 2) Run your TFTP server, and I set up it like the following:

🏘 Iftpd32 by Ph. Jounin	
Current Directory E:\upgrade Server interface 192.168.1.111	<u>B</u> rowse Show <u>D</u> ir
Current Action Listening on port 69	
About <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 IP01 from Web Page

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 IP01:

C HTTP URL	TFTP Server	
TFTP Server :		+ Go
File Name 🛈 :		
Reset Configs ①:	1	



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 IP01.

Power off and power on the IP01, wait for several minutes. When the TEL port LED 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**, please select the **Call Detail Records** option from the vertical menu on the left, then you can get the following screen:

CI	DR ¥iever (CDR-CS¥) ↔															
С	CDR viewer << prev next >>															
		ing 1-2													View:	25 🗸
(mo		t recen						_								
	1	<u>Account</u> <u>Code</u>	Source	Destination	<u>Dest. Context</u>	<u>Caller</u> ID	Channel .	<u>Dest.</u> <u>Channel</u>	<u>Last app.</u>	Last data	<u>Start time</u>	<u>Answer</u> Time	<u>End Time</u>	Duration	<u>Billable</u> <u>seconds</u>	² <u>Disposit</u>
1	1			6001	default		Local/6001@default=2567	2	SIP/6001- 011ab340	Dial	SIF/6001&IAX2/6001 20	2010-04- 21 05:34:42		2010-04- 21 05:35:12	30	0
3	3			6001	default		Local/6001@default=6553	2	SIP/6001- 012Ъ0004	Dial	SIF/6001&IAX2/6001 20	21	2010-04- 21 05:34:27	21	18	1
4	4		6005	6001	DLPN_DialPlan1	""6005"" <6005>	Zap/4-1	SIP/6001- 012Ъ0004	Dial	SIP/6001&IAX2/6001 20	2010-04-21 05:31:21	21	2010-04- 21 05:32:04	43	3	ANSWERED
6	5			s	default		Local/6001@default=7c11	2	SIP/6001- 005eb004	WaitExten	6	21	2010-04- 21 05:31:29	21	38	18
7	7		6006	6750	DLPN_DialPlan1	""6006"" <6006>	SIP/6006-011ab340		VoiceMailMain		2010-04-21 05:24:16	21	2010-04- 21 05:25:07		51	ANSWERED
8	3		6005	6600	DLPN_DialPlan1	""6005"" <6005>	Zap/4-1		VoiceMail	6005@default&6006@default	2010-04-21 05:23:47	21	2010-04- 21 05:24:05		15	ANSWERED
9	9		6005	6300	DLPN_DialPlani	″″6005″″ ≺6005>	Zap/4-1		MeetMe	6300 Ms	2010-04-21 04:51:41	21	2010-04- 21 04:52:56	75	73	ANSWERED

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 IP01

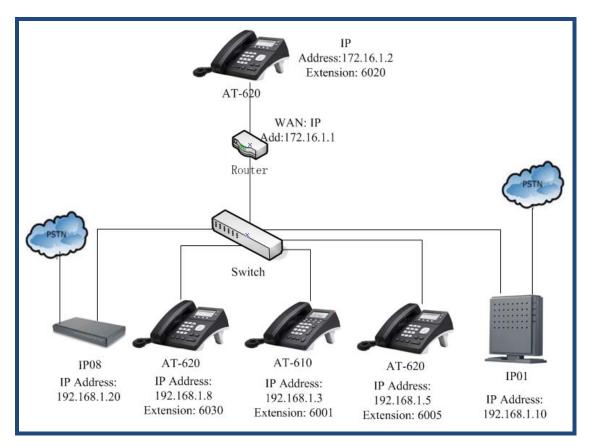


Figure: Network Topology

In the network topology above: user 6020, user 6001 and user 6005 will be registered to IP01, user 6030 will be registered to IP08. 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 IP01 to PSTN.
- 3) 6005 and 6001 can get incoming calls from PSTN by IP01.
- 4) 6030 can call-out to PSTN and get incoming call from PSTN through IP01.
- 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 IP01

4.1.1 Access to the Web Page of IP01 by Browser

After connecting IP01 to LAN, please open your browser of PC with windows OS and input the IP Address of IP01 (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:

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 ?	Logout
[] System Status	System Status 🔅							Upgrade to VoII	Ptel SE
Please click on a panel to manage related features				/	🗸 VolPtel				
			Uptime		ystem Status O min, load average:	0.04, 0.06, 0.02			
					Irunks				
	Status	Trunk	Туре	Username	F	ort/Hostname/IP			
[] Configure Hardware	_								
[] Trunks				-	Extensions				
[] Outgoing Calling Rules			🔵 Free	🔴 Busy	UnAvailable	🥚 Ringing	:		
[] Dial Plans	Extension		J.	lame/Label		Status	Туре		
[] Users	<u>6750</u>		c	Check Voicemai	.1s		VoiceMailMain		
[] Ring Groups	*No Extensio	on assigned	п	Dial by Names			Directory		
C Music On Hold									
[] Call Queues									
[] Voice Menus									
C Time Intervals									
[] Incoming Calling Rules									
[] Voicemail									
Conferencing									
[] Follow Me									
E3 Directory									
[] Call Features									

4.1.2 Add up Users from Web Page of IP01

1) Add up a DialPlan

Before you add up user, you have to add up a DialPlan, please click on **Dial Plans** \rightarrow **New DialPlan**, I add up a DialPlan like the following:

Create New DialPlan	
DialPlan Name:	DialPlan1
Include Outgoing Calling Rules:	You do not have any calling Rules defined ! click here to manage calling rules.
Include Local Contexts:	Vdefault Vparkedcalls Vconferences Vringgroups Vvoicemenus Vqueues Vvoicemailgroups Vdirectory
	Scancel Zseve

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 and SIP user 6005

After logging into the web page of IP01, please click on Users \rightarrow Create New User, I configure user 6001 like the following:



Create New User X
General : Extension: 6001 ① Name: 6001 ① DialPlan: DialPlan1 ♥ ① CallerID: 6001 ① OutBound CallerID: 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: 750 rxflash ①: 1250 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

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

4.1.3 Register 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:

\frown			IP Phone	1		
ATCOM'	Current Status Network VOIP	Advanced Dial-pe	er <u>Config Manage</u>	<u>Update</u>	<u>System Manage</u>	
					I	Public SIP Configuation
		Basic Setting				
		Register status	Registered		Proxy Server Address	
		Server Address	192.168.1.10		Proxy Server Port	
		Server Port	5060		Proxy Username	
		Account Name	6001		Proxy Password	
		Password			Domain Realm	
		Phone Number	6001		Enable Register	
		Display Name	6001			
				AP	PLY	
				Advan	ed Set	

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



4.1.4 Register SIP user 6005 in AT620

After logging into the web page of IP Phone AT-620, please select VOIP option, I register the 6005 as the following illustration:

			IP Phon	е		
АТСОМ"	Current Status Network VOIP A	dvanced Dial-peer	Config Manage Upo	date <u>Syste</u>	em Manage	
• <u>SIP 1</u> • <u>SIP 2</u> • <u>IAX 2</u>						SIP1 Configuation
		Basic Setting				
		Register status	Registered		Proxy Server Address	
		Server Address	192.168.1.10		Proxy Server Port	
		Server Port	5060		Proxy Username	
		Account Name	6005		Proxy Password	
		Password	••••		Domain Realm	
		Phone Number	6005		Enable Register	
		Display Name	6005			
				AP	PLY	
				Advan	ced Set	

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 IP01, 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 IP01, please click on **Trunks** \rightarrow **Analog Trunks**, I configure an analog trunk like the following:

		/	ATCOM
New Analog Trunk			X
	Channels: 🗹 1		
Tr	unk Name 🛈 : trunk1		
	Advanced	Options	
Busy Detection $\textcircled{1}$:	Yes 💙	Busy Count 🛈 :	3
Ring Timeout 🛈 :	8000		
Answer on Polarity Switch $\textcircled{1}$:	No 💌	Hangup on Polarity Switch (i) :	No 💟
Call Progress 🛈 :	No 🗸	Progress Zone 🛈 :	US 💌
Use CallerID 🛈 :	Yes 🗸	Caller ID Start 🛈 :	Ring 💟
CallerID 🛈 :	As Received 💙	Pulse Dial 🛈 :	No 💌
CID Signalling 🛈 :	Bell - USA	▼ mailbox :	V
Flash Timing 🛈 :	750	Receive Flash Timing 🛈 :	1250
	○ Cancel	Add	

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 IP01, please click on **Outgoing Calling Rules** >> New Calling Rule, I configure an outgoing calling rule like the following:



New CallingRule	X
Calling Rule Name 🛈 : outgoing1	
Pattern 🛈 : _2x.	
🔽 🔲 Send to Local Destination 🛈 ———————————————————————————————————	1
Destination :	
- Send this call through trunk:	-
Use Trunk 🛈 trunki 💌	
Strip 🛈 1 digits from front	
and Prepend these digits 🛈 🛛 before dialing	
🗌 Use FailOver Trunk 🛈 :	Ī
fail over Trunk 🛈 🔛	
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 IP01, please click on **Dial Plans** \rightarrow **New DialPlan**, I configure a dial plan like the following:

Edit DialPlan	
DialPlan Name:	DialPlan2
Include Outgoing Calling Rules:	🗹 outgoing 1
Include Local Contexts:	🗹 default 🗹 parkedcalls 🗹 conferences 🗹 ringgroups 🗹 voicemenus 🗹 queues 🗹 voicemailgroups 🗹 directory
	Save Save

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 IP01, 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 trunk1 I created in 4.2.1

4.3.2 Create an Incoming Calling Rule

After logging into the web page of IP01, please click on **Incoming Calling Rules**→ **New Incoming Rule**, I configure an incoming calling rule like the following:

New Incoming Rule		x
Trunk	trunki 💌	
Time Interval	: None (no TimeIntervals matched) 🗸	
Pattern 🛈	s	
Destination	: VoiceMenu ivr	
	S 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 IP01, please click on Voice Menus→ Create New VoiceMenu, I create a voice menu like the following:



Edit Voice l en	u voicemenu-custom-2			X
Name:	voicemenul	١	Advanced Edit	
Extension:	7002			
☑ ①	Allow Dialing Other Extensions			
Actions 🛈	Answer the call		(0 40 80
	Play record/WelcomToATCOM & Donot Listen fo	r KeyPress events	(୭ୣୣୖୖୖୖୖ ⊗ ∥
	Goto User 6001		(♥ ❹ ⊗ ∥
Add new Step:	Select an Option 💌]
	🛈 Allow KeyPress Events			
0				
1 Got	o User 6001			
2 Got	o User 6005			
3				
4				

When the call comes from port 1, 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 IP01 to let 6030 call outside and get incoming call by IP01, the steps are the same as IP01, you can refer to configuration of IP01.

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 IP01, 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
				Ether	fast® Cable/	DSL Router	BEFSR41 V3
Setup	Setup Basic Setu	Security	Applications & Gaming DDNS	Administration	Status Advanced F	Courting	
Internet Setup Internet Connection Type		ress: t Mask: t Gateway: DNS 1: DNS 2:	172 . 16 . 1 255 . 255 . 0 172 . 16 . 1 0 . 0 . 0 0 . 0 . 0 0 . 0 . 0 0 . 0 . 0 0 . 0 . 0 0 . 0 . 0	. 1 . 0 . 254 . 0 . 0		where basi performed. (Internet Se will require DNS inform settings car from your IS	Setup screen is configuration is Some ISP's rvice Providers) that you enter the lation. These lation. These lation in the attion in the set a router set a router for-
Optional Settings (required by some ISPs)	Host N Domai MTU:	lame: n Name:	 ○Enable ⊙Disa	able Size: 1492		Setup secti required to specific ISP the table be	the Internet on is all that is set up for your . Please look at low to configure or 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) . 0 . 0 . 0 	<u>Cisco Systems</u>
		Save Settings Cancel Changes	«مىللەسمىيالەس

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					Eth	erfast® Cable	/DSL Router	BEFSR41 V3
& Gaming	Setup	Security	Applicat & Gam		Administration	Status		
	Port Range	Forwarding	Po	rt Triggerin	g UF	PnP Forwarding	3	DMZ
							Port Ran	ge Forwarding
Port Range Forwarding								e Forwarding can set up public
			Port	Range			services o When use	n your network. rs from the
	Application	Start	End	Protocol	IP Address	Enabled	Internet ma requests c	ake certain n your network,
	IAX2	4569 to	4569	Both 💌	192.168.1. 10	V		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 IP01.

2) Add an IAX user 6020 in IP01

After logging into the web page of IP01, please click on Users \rightarrow Create New User, I configure 6020 like the following:

Create New User X
General : Extension: 6020 ① Name: 6020 ① DialPlan: DialPlan1 ♥ ① CallerID: 6020 ① OutBound CallerID: 6020 ①
Image: Second state of the second s
Technology ✓ SIP ① ✓ IAX ① Analog Station: None ① flash ①: 750 rxflash ①: 1250 Codec Preference : First : u-law Second : GSX Third : None Fourth : None Fifth : None
VoIP Settings MAC Address : ① Line Number : 1 v ① SIP/IAX Password: ① NAT: ⑦ Can Reinvite: ① DIMF Mode: RFC2833 v ① insecure: very v 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.

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:

Атсом	Current Status Network VOIP 4	Advanced Dial-peer	IP P		
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 💿	DHCP O		PPPOE O
		Auto DNS			
		Static IP Address		172.16	1.2
		Netmask		255.25	5.0.0
		Gateway		172.16	.1.254
		DNS Domain			
		Primary DNS		202.96	.134.133
		Alter DNS		202.96	.128.68
		L			APPLY

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

470011		IP Phone		
ATCOM '	Current Status Network VOIP	Advanced Dial-peer Config Manage Update	<u>System Mana</u>	uge
		IAX2		
 <u>SIP 1</u> <u>SIP 2</u> 		Register Status	Registered	
• <u>IAX 2</u>		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		
		Enable G.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 IP01. 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 IP01 to IP08

In order to call from IP01 to IP08, I will create a SIP user in IP08 for the SIP trunk in IP01, create



a SIP trunk, an outgoing call rule and a dial plan in IP01.

Add an SIP user 6035(it will be used as SIP trunk in IP01) in IP08, after logging into the web page of IP08, please click on Users→ Create New User, I add the user 6035 like the following:

Create New User		
General : Extension: 6035 ① Name: 6035 ① DialPlan: DialPlani ① CallerID: 6035 ③ OutBound CallerID: 6035 ③		
✓ Enable Voicemail for this User ① VoiceMail Access PIN code: ① Mailbox: 6035 ① Email Address: ①		
Technology ✓ SIP ① ✓ IAX ① Analog Station: None ♥ ① flash ①: 750 rxflash ①: 1250 Codec Preference : First : □ Second : GSM ♥ None ♥ Fifth : None		
VoIP Settings MAC Address : Image:		
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 IP08 for AT-620, the way is the same as adding 6035.

 Add a VoIP trunk in IP01, after logging into the webpage of IP01, please click on Trunks→VOIP Trunks→New SIP/IAX Trunk, I configure a SIP trunk1 like the following:

ATCOM	")
ATCON	

Edit SIP trunk 6035	X	
Provider Name 🛈:	siptrunk1	
Hostname :	192. 168. 1. 20	
Username :	6035	
Fromuser :		
Fromdomain :	6035	
Password :	6035	
Contact Ext.:	S	
Insecure Type:	very V	
	First : u-law Second : a-law Third : GSM V	
CallerID 🛈 :	Fourth : G.726 V Fifth : None V	
🗌 Enable Remote MWI :		
	○ Cancel Add	

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 IP01, after logging into the webpage of IP01, please click on **Outgoing Calling Rules→New Calling Rule**, I configure an outgoing2 rule like the following:

-				
A	ТС	0	M	1

R	Edit Calling Rule	K
	Calling Rule Name 🛈 : outgoing2	
	Pattern ① : _9.	
	- 🗆 Send to Local Destination 🛈	
	Destination :	
	Send this call through trunk:	
	Use Trunk 🛈 siptrunkl 💌	
	Strip 🛈 1 digits from front	
	and Prepend these digits () before dialing	
	Use FailOver Trunk 🛈 :	
	fail over Trunk 🛈 siptrunkl 🗸	
	Strip 🛈 📃 digits from front	
	and Prepend these digits 🛈 📃 before dialing	
	Save Save	

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

Create a dial plan in IP01, after logging into the webpage of IP01, 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:	Vdefault Vparkedcalls Vconferences Vringgroups Vvoicemenus Vqueues Vvoicemailgroups Vdirectory
	Save 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 IP08 to IP01

In order to call from IP08 to IP01, I will create a SIP user in IP01 for the SIP trunk in IP08, create a SIP trunk, an outgoing call rule and a dial plan in IP08.

1) Add a user 6008 in IP01

Add a SIP user: 6008, after logging into the web page of IP01, please click on Users→ Create New User, I add a user 6008 like the following:



Create New User
General : Extension: 6008 ① Name: 6008 ① DialPlan: DialPlan1 ♥ ① CallerID: 6008 ① OutBound CallerID: 6008 ①
Enable Voicemail for this User Image: Im
Technology ✓ SIP ① ✓ IAX ① Analog Station: None ● ① flash ①: 750 rxflash ①: 1250 Codec Preference : First : u-law ♥ Second : GSK ♥ Third : None ♥ Fifth : None ♥ Fourth : None ♥
VoIP Settings MAC Address : Image: I
Other Options 3-Way Calling (1) In Directory (1) Call Waiting (1) CTI (1) V Is Agent (1) Enable Call Record (1) Pickup Group: 1 V
○ Cancel Vpdate

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 IP08

Add a VoIP trunk in IP08, after logging into the webpage of IP08, please click on **Trunks→VOIP Trunks→New SIP/IAX Trunk**, I configure a SIP trunk like the following:

Create New SIP/IAX trunk X		
Туре:	SIP 🗸	
Provider Name 🛈:	siptrunk	
Hostname :	192.168.1.10	
Username :	6008	
Fromuser :		
Fromdomain :		
Password :	6008	
Contact Ext.:		
Insecure Type:	very 🗸 🛈	
	S Cancel ✓ Add	

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 IP08



After logging into the webpage of IP08, please click on **Outgoing Calling Rules**→**New Calling Rule**, I configure an outgoing1 rule like the following:

New CallingRule	Х
Calling Rule Name 🛈: outgoing1	
Pattern ①:_9.	
Send to Local Destination ①	7
Destination :	
Send this call through trunk:	Γ 1
Use Trunk 🛈 siptrunk 🛩	
Strip 🛈 1 digits from front	
and Prepend these digits ① before dialing	
Use FailOver Trunk ① :	Г Л
fail over Trunk 🛈 🔽 💌	
Strip 🛈 digits from front	
and Prepend these digits ① before dialing	
Save 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 IP08

After logging into the webpage of IP08, please click on **Dial Plans**->**New DialPlan**, I configure a dialplan1 like the following:

Edit DialPlan	
DialPlan Name	DialPlani
Include Outgoing Calling Rules	🗹 outgoingl
Include Local Contexts	♥default ♥parkedcalls ♥conferences ♥ringgroups ♥voicemenus ♥queues ♥voicemailgroups ♥directory
	Cancel V 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 IP01

Using WinSCP software, it is the most convenient way to transfer files between windows PC and IP01. Open your WinSCP software, enter the IP Address, username, password of IP01 like the following screen: