# ATCOM® IPPBX IP-2G4A

## Product Guide

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# Content

# Contact ATCOM

## The Introduction of ATCOM

Founded in 1998, ATCOM technology has been always endeavoring in the R&D and manufacturing of the internet communication terminals. The product line of ATCOM includes IP Phone, USB Phone, IP PBX, VoIP gateway and Asterisk card.

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## Download Center: http://www.atcom.cn/download.html

# Chapter 1 the Introduction of IP-2G4A

## 1. Overview of the IP-2G4A

The IP-2G4A is a complete Asterisk Appliance with combination of GSM and Ananlog channels. It is an embedded open source Linux system with built-in SIP/IAX2 proxy server and NAT functions. It provides a solid, uniform platform for Mobile and VoIP communications.
Targeting for SOHO user and SMB market with an easy to use graphical interface, ATCOM GSM IP PBX provides a cost-saving solution on their telecommunication/data needs. With these devices, company with branch offices in different countries can be easily combined together to work like a virtual single office through internet, GSM and PSTN network.

## 2．Hardware

CPU: 400MHz Blackfin 532 Chip
2 x GSM ports and four analog ports
NAND flash 256 M
SDRAM 64M

## 3．System

Open Source uClinux

## 4. Function features

PSTN, GSM,ISDN
Support g711/g729/gsm codec
Voicemail
Voicemail groups
3-way Calling
Conferencing
Follow Me
Call Feature
In directory
Call Waiting
Call Queues
Pickup Group
Ring Group
Is Agent
Music On Hold
Voice Menus
Voice menus Prompts
Time intervals
Backup
Update

# 5．Applications

SOHO/SMB telephony system
Hosted service
IVR system

# 6. Interface

1 X RJ45 port.
1 X Power port.
1 X RS232 port.
4 X FXO/FXS ports.
2 X GSM channels.

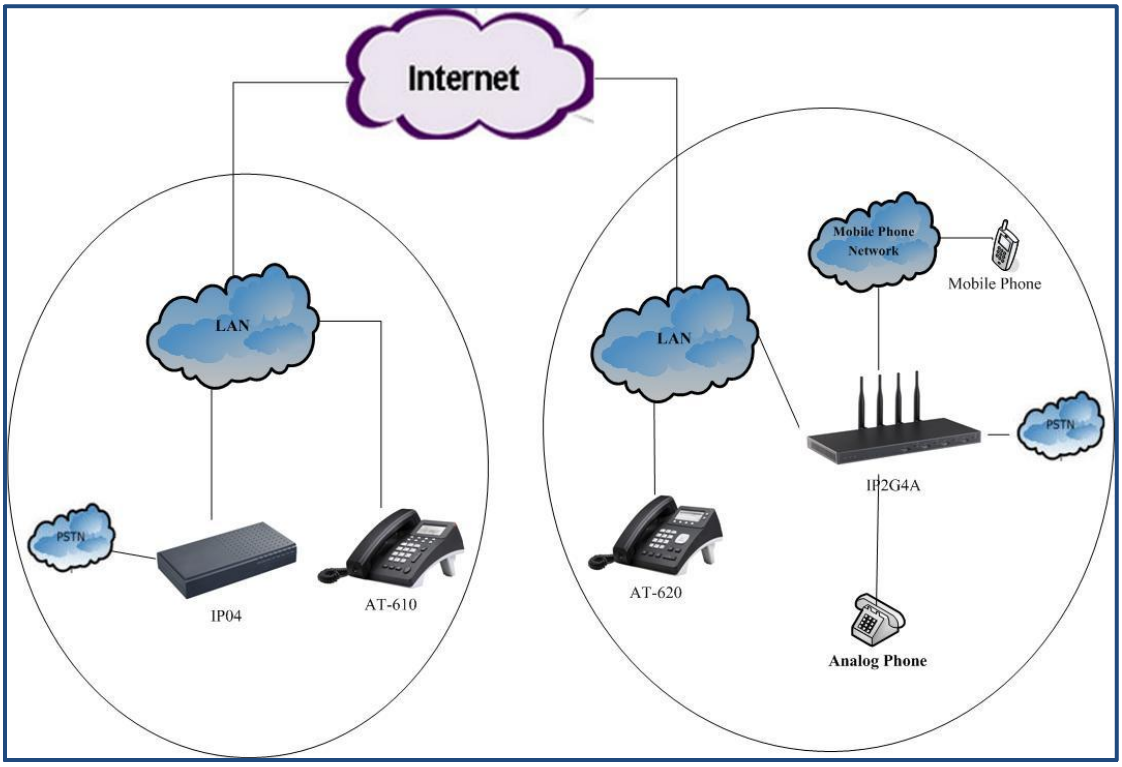
# 7．Measurement and Weight

| | |
|---|---|
| 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 |

# 8．Package

| Item | Quantity |
|---|---|
| IP-2G4A | 1 |
| RS232 module | 1 |
| Power Adapter | 1 |
| Manual (disk) | 1 |

For the usage of IP-2G4A in VoIP field, you can refer to the following network topology.




# Chapter 2 Access to the IP-2G4A

You need a PC to access to the IP-2G4A, there are four ways for you to access the IP-2G4A:

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 IP-2G4A by the first three ways, you have to check that if your network connection between IP-2G4A and PC is OK. If you do not have network connection between IP-2G4A and PC, you can try to use the last way to access to IP-2G4A and change the IP address for IP-2G4A.

## 2.1 WebPage Access by Browser

It is the most convenient and common way to access the IP-2G4A, you just need to open your browser and input the IP address of IP-2G4A WAN port (the default IP address is 192.168.1.100). You'd better use Firefox instead of IE, because there are compatible issues.

Then input the default Username: admin; Password: atcom .

## 2.2 Support SSH protocal

Logging into IP-2G4A by SSH, you can configure IP-2G4A by Linux command.

## 2.3 Console Port Access to IP-2G4A

If you do not have network connection between IP-2G4A and PC, you can try to access to IP-2G4A by console port. Please try to do as the following steps:

1. Connect the console port of IP-2G4A to your PC's console port with RS232 console cable.
2. Run your HyperTerminal, 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 HyperTerminal

The default IP address of IP-2G4A is 192.168.1.100. Your network may have a different IP address segment such as 192.168.10.xx. In this situation, you cann't access to IP-2G4A by putty and browser if you do not change the IP-2G4A IP address. So you have to change the IP address for IP-2G4A by HyperTerminal to make it in the same network segment as your LAN.

After you have accessed to IP-2G4A by HyperTerminal, please use the following command to change the IP address for IP-2G4A.

root:~> ifconfig eth0 192.168.1.151(the IP address what you want to set for IP-2G4A)

By this way, the IP address you set for IP-2G4A 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-2G4A, you can try to set it in web GUI, for detail steps please refer to chapter 3.

# Chapter3 General Operation of IP-2G4A

## 1. Backup

When you log in the web of IP-2G4A, Click on **Backup**, you can see the button of **Create New Backup**, then you can Backup the current system.

## 2. System Update

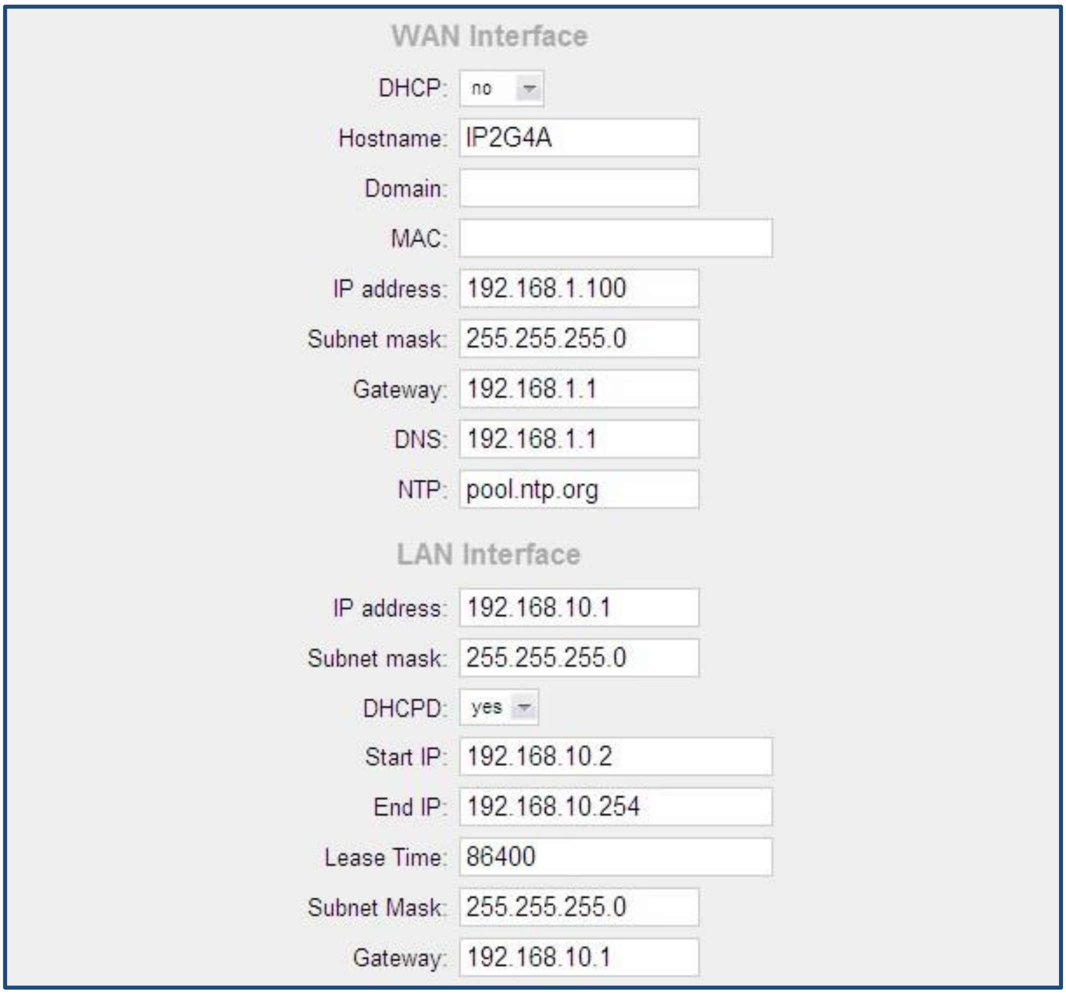
When you log in the web of IP-2G4A, click on **Options → Advanced Options → Show Advansed Options ,** After click on Show Advanced Options in the web, you can see the advanced options in the vertical menu on the left of the main page. Click on **Firmware update** , you can see the following parameters in the table.

| Parameter Name | Description | Type | Default |
|---|---|---|---|
| **HTTP URL** | The http path of the firmware file | Textbox | Null |
| **TFTP Server** | The IP address of TFTP Server where the firmware file in. | Textbox | Null |
| **File Name** | Specify the name of your uImage-md5 firmware file, make sure to use md5 version only | Textbox | Null |
| **Reset Configs** | Select this box if you wish to reset to factory defaults. This will ensure a clean update, and is highly recommended | Selected | Not selected |

If you want to upload sound file, upload backup files and so on , you can refer to the link: http://www.atcom.cn/downloads/IPPBX/ATCOM%20IPPBX%20Series%20Product%20Upgrade%20Guide-V1.0-EN.pdf

## 3. Network

After click on **Options→Advanced Options→Show Advanced Options**, please select **Network Settings** option from the vertical menu on the left of main page. You can set IP address, Subnet mask, Gateway, DNS what you want like the following:




Please click **save** button in your page to save your setting and reboot the IP-2G4A.

Attention: you need configure the IP address, Subnet mask, Gateway and DNS at WAN Interface so that the network connects successfully. The option of LAN Interface is used for Routing functions, here you needn't configure it.

# Chapter 4 Configure IP-2G4A by Web GUI

## 4.1 System Status

In the system status screen, it displays the functions you configured, such as: trunks, extensions, conference and so on. The following table is the options description of trunks.

| Name | Description |
|---|---|
| Status | The register status of trunks |
| Trunk | The name of trunks |
| Type | The type of trunks |
| Username | The username of SIP/IAX trunk |
| Port/Hostname/IP | IP Address/port |

1.The register status of trunks include three kinds: Unregistered, Request Sent, Registered.
2.The type of trunks : VoIP trunk including SIP and IAX; Analog trunk; Service Provider.

The parameter of extensions in the following table:

| Name | Description |
|---|---|
| Extension | The status of users |
| Name/label | The name of users |
| Status | Display voice message |
| Type | SIP users/IAX users/Analog users |

1. There are four kinds status of users, when the light of "Extension" list displays gray , means the user does not register that is Unavailable; when the light of "Extension" list displays green , means the user is Free; when the light of "Extension" list displays orange , means the user is Ringing; when the light of "Extension" list displays red , means the user is Busy.
2. Status: This parameter displays if other users leave messages, Messages : 0/0, the figure front of " / " displays the new messages amount; the figure behind of displays the old messages amount.

## 4.2 **Configure Hardware**

In the configure hardware page, it includes the following components: analog hardware, tone region, advanced settings. Pay attention that some browsers do not display the configure, it is unimportant.

**Analog Hardware**

When you boot the IP-2G4A, which will detect the FXO and FXS modules automatically, the analog hardware component displays the modules which are detected correctly.

| Name | Description | Type | Default |
|---|---|---|---|
| Tone Region | 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 | ComboBox | United Status/North America |
| Module Name | The name of Module | Textbox | wctdm24xxp |
| Opermode | Specifies On Hook Speed, Ringer Impedance, Ringer Threshold, current Limiting ,TIP/RING voltage adjustment, minimum Operational Look Current and so on. Please choose your country or your nearest neighboring country | ComboBox | USA |
| a-law override | Specifies the codec to be used for analog line. | ComboBox | ulaw |
| fxs honor mode | This option allows the user to determine if they would like opermode characteristics applied to trunk(FXO) modules only, or both trunk (FXO) and station(FXS) modules. | ComboBox | FXO modules |
| boostringer | This option allows the user to define whether they require normal ringing voltage(40v) or maximum ringing voltage(89v) or analog phones attached to station(FXS) modoules | ComboBox | nomal |
| fastringer | This option sometimes used in conjunction with the Low Power Option ,allows the user to increase the ringing speed to 25HZ | ComboBox | nomal |
| lowpower | This option generally used in conjunction with the Fast Ringer Option ,allows the user to set the peak voltage during Fast Ringer Operation to 50V. | ComboBox | nomal |
| ring detect | This option allows the user to choose from normal ring detection or a full wave detection | ComboBox | standard |
| MWI mode | This option allows the user to specify the type of Message Waiting indicator detection to be done on trunk(FXO) interfaces | ComboBox | none |

## 4.3 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.

## 4.3.1 Create Analog Trunks

Analog trunk is associated with FXO port, and it will call outside by PSTN line. Click on New Analog Trunk , then we can see the parameters which are in the following table in the web.

| Name | Description | Type | Default |
|---|---|---|---|
| Channels | Display the FXO or GSM modules | selected | no select |
| Trunk Name | The name you want to set for the trunk | Textbox | null |
| Busy Detection | Busy detection is used to detect far end hang up or for detecting busy signal. | Boolean | Yes |
| busycount | If Busy Detection is enabled,it is also possible to specify how many busy tones to wait for before hanging up. | Int | 3 |
| Ring Timeout | Thrunk(FXO) devices must have a timeout to determine if there was a hangup before the line was answered. | Int | 8000 |
| answeronpolarityswitch | If this option is enabled, the reception of a polarity reversal will mark when a outgoing call is answered by the remote party. | Boolean | no |
| hanguponpolarityswitch | In some countries ,a polarity reversal is used to single the disconnect of a phone line. | Boolean | no |
| Use CallerID | Enabling this option enabled CallerId detection. | Boolean | yes |
| Caller ID Start | This option allows one to define the start of a CallerID Signal. | ComboBox | Ring |
| CallerID | This option allows the lines to report the Caller ID string as received from the telco, or as a fixed value by using the custom option. | select box | As Received |
| Pulse Dial | If this option is enabled ,pulse mode dialing instead of DTMF,wil be enable. | Boolean | No |
| CID Signalling | This option defines the type of caller ID signaling to use :bell,v23,v23_jp,or dtmf. | ComboBox | Bell-USA |
| Flash Timing | Flash Time defines the time ,in millseconds,that is generated for a flash operation. | Textbox | 750 |
| Receive Flash Timing | Flash Time defines the time,in milliseconds, that is generated for a flash operation. | Textbox | 1250 |

1.Trunk name: unique label to help you identify the trunk when listed in outgoing calling rules and incoming calling rules.

## 4.3.2 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 table is the parameter of VoIP trunk:

| Name | Description | Type | Default |
|---|---|---|---|
| Type | You can select SIP or IAX type to meet your need. | ComboBox | SIP |
| Provider Name | A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. | Textbox | Null |
| Hostname | The IP Address of the server which you want to connect | Textbox | Null |
| Username | the username that your service provider configured | Textbox | Null |
| Fromdomain | The domain of the server which you want to connect | Textbox | Null |
| Password | the password that your service provider configured for the user. | Textbox | Null |
| Contact Ext. | | Textbox | s |
| Insecure Type | The insecure type of the trunk transferring data. | ComboBox | very |

1.Notice Provider Name must be unique label , especially do not the same with Username .
2.Insecure Type: insecure=very ; To allow registered hosts to call without re-authenticating
insecure=port ; Allow matching of peer by IP address without matching port number.
insecure=invite; removes the requirement for authentication of incoming INVITE messages.

# 4.4 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, then Click on New Calling Rule button, the parameters of the Outgoing Calling Rules are in the following table:

| Name | Description | Type | Default |
|---|---|---|---|
| Calling Rule Name | The name of the Calling rule | Textbox | Null |
| Pattern | The dialing rule | Textbox | Null |
| Send to Local Destination | If this option is checked and Destination is defined, calls matching the specified pattern may be sent to a local extension." | selected | no select |
| Destination | Choose the Local Destionation:User/VoiceMenu/Hungup… | ComboBox | Null |
| Use trunk | Defines the Trunk that calls, matching the specified pattern, will be placed through. | ComboBox | Null |
| Strip | Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the trunk selected in \"Use Trunk.\" One might. | Textbox | Null |
| Prepend these digits | Allows the user to specify digits that are prepended before the call is placed via the trunk. If a user's trunk required 10 digit dialing, but users were more comfortable performing 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are placed to the trunk. User may also prepend a 'w' character for analog trunks to provide a slight delay before dialing | Textbox | Null |
| Use Failover Trunk | Failover trunks can be used to make sure that a call goes through an alternate route, when the primary trunk is busy or down If "Use Failover Trunk" is checked and "Failover trunk" is defined, then calls that cannot be placed via the regular trunk may have a secondary trunk defined. If a user's primary trunk is a VoIP trunk, but one wants calls to use the PSTN when the VoIP trunk isn't available, this option is a good idea. | selected | no select |
| Fail over trunk | Choose the trunk | ComboBox | ComboBox |

1. Pattern: X ... Any Digit from 0-9; Z ... Any Digit from 1-9; N ... Any Digit from 2-9; [12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9); .... Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself); ! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible. For example, the extension _NXXXXXX would match normal 7 digit dialings, while _1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one.
2. Strip: Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the trunk selected in Use Trunk. For example, want users to dial 9 before their long distance calls; however one does not dial 9 before those calls

are placed onto analog lines and the PSTN, so one should strip 1 digit from the front before the call is placed.

3. 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.

## 4.5 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, Click on New DialPlan button, the following table displays the parameters of Dial Plans .

| Name | Description | Type | Default |
|---|---|---|---|
| DialPlan Name | The name of DialPlan, which is a unique label to help you identify the dial plan | Textbox | DialPlan1 |
| Include Outgoing Calling Rules | Select outgoing call rule which you use | selected | Not seclect |
| Include Local Contexts | Local context is used for general using configuration. | check box | Select all |

## 4.6 Users

Users component is used to add or remove Analog, SIP, IAX extension.

Click on Create New User button in the web of IP-2G4A, you can create SIP/IAX User and Analog User.

| Name | Description | Type | Default |
|---|---|---|---|
| Extension | The numbered extension | Textbox | 6001 |
| Name | A character-based name for this user | Textbox | Null |
| DialPlan | DialPlans are sets of calling rules and can be managed form the \"Dial Plans\" panel | ComboBox | Null |
| CallerID | The Caller ID (CID) string used when this user calls another internal user. | Textbox | Null |
| OutBound CallerID | Caller ID that would be applied for out bound calls from this user. Note that your ability to manipulate your outbound Caller ID may be limited by your VoIP provider. | Textbox | Null |
| Enable | Check this box if the user should have a voicemail | Selected | Not |

| | | | |
|---|---|---|---|
| Voicemail for this User | account | | selected |
| VoiceMail Access PIN code | Voicemail Password for this user | Textbox | Null |
| Mailbox | Voicemail Mailbox for this user | Textbox | Null |
| Email Address | The e-mail address for this user | Textbox | Null |
| SIP | Check this option if the User or Phone is using SIP or is a SIP device | selected | selected |
| IAX | Check this option if the User or Phone is using IAX or is an IAX device | selected | selected |
| Analog Station | If this user is attached to an analog port on the system, please choose the port number here | ComboBox | Null |
| Codec Preference | Choose priority codec | ComboBox | u-law/GSM |
| NAT | Try this setting when Asterisk is on a public IP, communicating with devices hidden behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP+RTP ports. | selected | selected |
| Can Reinvite | By default, Asterisk will route the media steams from SIP endpoints through itself. Enabling this option causes asterisk to attempt to negotiate the endpoints to route the media stream directly, bypassing asterisk. It is not always possible for asterisk to negotiate endpoint-to-endpoint media routing. | selected | Not selected |
| DTMF Mode | Set default dtmfmode for sending DTMF. info : SIP INFO messages;inband : Inband audio (requires 64 kbit codec -alaw, ulaw); auto : Use rfc2833 if offered, inband otherwise. | ComboBox | rfc2833 |
| 3-Way Calling | Check this option if the User or Phone should have 3-Way Calling capability. | selected | Not select |
| In Directory | Check this option if the user is to be listed in the system telephone directory. | selected | Not select |
| Call Waiting | Check this option if the User or Phone should have Call-Waiting capability | selected | Not select |
| Is Agent | Check this option if this User or Phone is a Call Queue Member (Agent) | selected | Not selected |
| Pickup Group | If a user called A and another user called B in the same group,A can pick up the phone taking the place of B. | selected | Not selected |

1. Analog Station: When you want to create Analog Users, please choose the FXS ports.
2. Codec Preference: Support g711u-law/g711a-law/g729/GSM

3. Attension: in the textbox of Extension, the value you set is limited to a range, you can adjust the range in the Options option to meet your requirement.

## 4.7 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: insecure=invite

| Name | Description | Type | default |
|---|---|---|---|
| Ring Group Name | Ring group name use in pbx | Str* | |
| Extension for this ring group | Ring group No. , dial the No. if you want to join , change boundary value in options | Int | 6400 |
| Ring Group Members | The ring group of numbers | {EXT1,EXT2,EXT3,…} | |
| Available Users | The entire Users | {EXT1,EXT2,EXT3,…} | |
| Strategy | Ring all simultaneously:<br>Ring in order | { Ring in Order ,ting all Extensions } | Ring in Order |
| Seconds to ring each member | Seconds to ring each member | Time | 20 |
| If not answered Goto | If not answered go to, hang up: hang up the calling channel。Operator：Go to operator 。Extension: a call to user. Voicemail: Go to IVR 。Conference: join a conference. Call queue: Go to a call queue. | {Hang-up, Operator….} | Hang up |

1. ring group application: Dial(channel type/${EXTEN}| channel type/${EXTEN}|20|i)
2. ring group up after please a call
3. non-ring if ring group user off hook or non-user registered
4. only one man can connected in coming call

## 4.8 Music on hold

'Music On Hold' lets you customize audio tracks for different queues, parked calls etc.

| Name | Description | Type | default |
|---|---|---|---|
| Upload an 8 KHz Mono Music file | Support codec: g711a/g711u | Upload | |
| New music on hold | Add a new music on hold | | |

1. Music on hold Dir: /persistent/sounds/moh/

2. Sounds : LICENSE-asterisk-moh-freeplay-ulaw
   LICENSE-asterisk-moh-freeplay-ulaw
   fpm-world-mix.ulaw
   fpm-world-mix.alaw
   fpm-sunshine.ulaw
   fpm-sunshine.alaw
   fpm-calm-river.ulaw
   fpm-calm-river.ulaw
3. Music on hold after holding status Status: busy
4. Music on hold non-rtp stream

# 4.9 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:

| Name | Description | Type | default |
|---|---|---|---|
| Extension | Extension for call queue: may be dialed to reach the call queue | Int | 6500 |
| Name | Name for call queue | Str* | |
| Strategy | Strategy: this option sets the ringing strategy for this queue, the options are<br>1. Ring all: ring all available agents simultaneously until one answers.<br>2. RoundRbin: Take turns ringing each available agent.<br>3. LeastRecent: Ring the agent which was least recently called<br>4. FewestCalls: Ring the agent with the fewest completed calls<br>5. Random: Ring a Random agent<br>6. RRmemory: RoundRobin with Memoryn Remember where it left in the last ring pass | {ringall,Roundrobin,leastrecent,Fewest calls,Random,Rrmomery} | ring all |
| Music On Hold | Select the 'Music on Hold' Class for this Queue. 'Music on Hold' classes can be managed from the the 'Music On Hold' panel on the left | Choice | default |
| LeaveWhen Empty | This option controls whether callers already on hold are forced out of a queue that has no agents. There are three options.<br>Yes: Callers are forced out of a queue when no agents are logged in.<br>No: Callers will remain in a queue with no agents.<br>Strict: Callers are forced out of a queue with no agents logged in, or if all logged in agents are unavailable.<br>The default option is Strict.<br>After a caller has left the queue, a caller will hear a busy tone and advance to the next calling rule after attempting to enter the queue | {yes,strict, No,} | strict |
| JoinEmpty | This option controls whether callers can join a call queue that has no agents. There are three options,<br>Yes: Callers can join a call queue with no agents or only unavailable agents<br>No: Callers cannot join a queue with no agents<br>Strict: Callers cannot join a queue with no agents or if all agents are unavailable. | {yes,strict, No,} | no |

| | | | |
|---|---|---|---|
| TimeOut | How many seconds an Agent's phone will ring before the Queue tries to ring the next Agent | Time | 15 |
| Wrapup Time | How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 0, which is no delay | Time | 0 |
| Max Len | How many calls can be queued at once. This count does not include calls that have been connected with Agents, it only includes calls that have not yet been connected. Default is 0, which is no limit. When the limit has been reached, a caller will hear a busy tone and advance to the next calling rule after attempting to enter the queue | Int | 0 |
| Auto full | Defining this option causes the Queue, when multiple calls are in it at the same time, to push them to Agents simultaneously. Thus, instead of completing one call to an Agent at a time, the Queue will complete as many calls simultaneously to the available Agents | checkbox | |
| Auto pause | Enabling this option pauses an agent if they fail to answer a call. This means that the agent is still logged into the queue, but they will not receive calls from the queue. Once paused, an agent can unpause by logging into the queue using the regular agent login extension | checkbox | |
| Report Hold Time | Enabling this option causes Asterisk to report, to the Agent, the hold time of the caller before the caller is connected to the Agent. | checkbox | |
| KeyPress Events | If a caller presses a key while waiting in the queue, this setting selects which voice menu should process the key press | choice | |
| Agent | This selection shows all Users defined as Agents in their User conf. Checking a User here makes them a member of the current Queue | checkbox | |

1. Call queue application: Queue(${EXTEN})
2. Change agents status:Login / Login out agents in System Info
3. Hear the music if all agents are busy, until non-conversation busy.

## 4.10 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.

| Name | Description | Type | default |
|---|---|---|---|
| Name | A name for the voice menus | Str* | |
| Extension | If you want this Voicemenu to be accessible by dialing an extension, then enter that extension number | No. | 7001 |
| Actions | A sequence of actions performed when a call enters the menu | Dial plan script | |
| Add new Step | Add additional steps performed during the menu | Dial plan script | |
| Allow KeyPress Events | Allow key press events will cause the system to listen for DTMF input from the caller and define the actions that occur when a user presses the corresponding digit | checkbox | |
| Advance edit | Advance edit for the voice menu | Dial plan script | |

1. Menus allow for more efficient routing of calls from incoming callers. Also known as IVR (Interactive Voice Response) menus or Digital Receptionist.
2. Step :
    a) Answer: Answer a channel if ringing
    b) Authenticate: This application asks the caller to enter a given password in order to continue dialplan execution.
    c) Background: Play an audio file while waiting for digits of an extension to go to.
    d) Busy Tone: Indicate the Busy condition
    e) Congestion: Indicate the congestion condition to the calling channel.
    f) Digit Timeout: set digit timeout
    g) DISA Password: Allow someone from outside the telephone switch (PBX) to obtain an internal system dialtone and to place calls from it as if they were placing a call from within the switch.
    h) Response Timeout: set response timeout
    i) Macro: macroname|arg1|arg2 .... Executes a macro using the context 'macro-<macroname>'
    j) Play Sound: Plays back given file
    k) Ringing: Indicate ringing tone
    l) Set MusicOhHold Class: select a music on hold
    m) SayAlpha: Say each character in the string including letters, numbers and other characters, one by one
    n) SayDigits: Say the digits, one by one
    o) SayNumber: Say a number (e.g. 'six thousand, five hundred and seventy two')
    p) Wait: Pause dialplan execution for a specified number of seconds
    q) WaitExten: Wait for the user to enter a new extension for a specified number of seconds
    r) To Destination: go to destination
    s) Set Language: set language (English/Spanish/French)
    t) To Directory: go to directory
    u) Dial an external Number: Place a call outside the pbx using the selected trunk
    v) AGI: Executes an AGI compliant application
    w) User Event: Send an arbitrary event to the manager interface
    x) Hangup: Hang up the calling channel
3. Allow keypress events:

    Must be voice menus have application: Background(file)

    e.x

    Background a music when keypress events
4. Advance edit

    Change dialplan for voice menus

    e.x.

```
include = default
exten = s,1,NoOp(Incoming DID)
exten = s,2,Answer()
exten = s,3,Background(record/GreetingNew)
exten = s,4,Background(record/MakeYourSelection)
exten = s,5,Background(fpm-sunshine)
exten = s,8,Voicemail(6002,u)
exten = 1,1,Goto(voicemenu-custom-2|s|1)
exten = 2,1,Voicemail(6002,u)
exten = 5,1,Goto(voicemenu-custom-3|s|1)
```

Want to control music on hold play time

```
include = default
exten = s,1,NoOp(Incoming DID)
exten = s,2,Answer()
exten = s,3,Background(record/GreetingNew)
exten = s,4,Background(record/MakeYourSelection)
exten = s,5,Set(TIMEOUT(absolute)=8)
exten = s,6,Background(fpm-sunshine)
exten = s,7,Set(TIMEOUT(absolute)=60)
exten = s,8,Voicemail(6002,u)
exten = 1,1,Goto(voicemenu-custom-2|s|1)
exten = 2,1,Voicemail(6002,u)
exten = 5,1,Goto(voicemenu-custom-3|s|1)
```

## 4.11 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,

| Name | Description | Type | default |
|---|---|---|---|
| Time Interval Name | A name for the time interval | Str* | |
| By day of week | Choice an available day of week for the time interval | {Mon,Tue, Wed,Thu,Fri ,Sat,Sun } | |
| By Days of a Month | Choice some available days of month for the time interval | {Dateof January/February/March/April/May/June/july/August/September/October/november/December/all} | |
| Time | Choice an available time slot for the time interval | {00:00-24:00} | |

1. Time intervals using in incoming call
2. Time intervals application rule:
00:00-24:00|mon-sum|1-31|January/February/March/April/May/June/july/August/September/October/november/December/all
time intervals:

```
timeinterval_date = *|mon-tue|*|*                Monday to Tuesday of weekly
```

## 4.12 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.

| Name | Description | Type | default |
|---|---|---|---|
| Trunk | Choice the trunk for the incoming rule | {analog, server provider, voip} | |
| Time Interval | Choice the time interval for the incoming rule | Choice | Non timeinterval matched |
| Pattern | Pattern of the incoming rule | Dialplan matched | S |
| Destination | Incoming to destination | {users, voice mail, ring group…} | IVR |

1. A trunk support a number of this time intervals, to support a number of Destination
2. Pattern:

All patterns are prefixed by the "_" character. In patterns, some characters have special meanings:

X ... Any Digit from 0-9

Z ... Any Digit from 1-9

N ... Any Digit from 2-9

[12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

. Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself)

! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible.

For example, the extension _NXXXXXX would match normal 7 digit dialings, while _1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one.

3.Note:You'll most likely need to add a rule with the pattern "s" (without the quotation marks) for each trunk. This signifies 'catch all', meaning all calls with a DID not matching any other rules will match this.

If you have multiple SIP trunks from the same provider, you'll want to set this pattern to whatever you specified as Contact Extension.

## 4.13 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.

| Name | Description | Type | Default |
|---|---|---|---|
| Extension for checking messages | defines the extension that Users call in order to access their voicemail accounts | NO. | 6750 |
| Direct Voicemail Dial | Check this to enable direct voicemail dial. For instance, if John's extension is 6001, you would be able to directly dial into John's voicemailbox by dialing #6001 to leave him a message | Check box | unCheck |

| | | | |
|---|---|---|---|
| Max greeting (in seconds) | Set the maximum number of seconds for a User's voicemail greeting | No. | 30 |
| Dial '0' for Operator | Enable Callers to exit the voicemail application and connect to an operator extension. The operator extension must be defined from the 'Options' panel | Check box | Check |
| Maximum messages per folder | This select box sets the maximum number of messages that a user may have in any of their folders | {10,25,100,200,500,1000} | 25 |
| Max message time | This select box sets the maximum duration of a voicemail message in seconds. Message recording will not occur for times greater than this amount | {1 minute ,2 minutes,5 minutes,15 minutes,30 minutes,umlimited} | 2 minutes |
| Min message time | This select box sets the minimum duration of a voicemail message in seconds. Messages below this threshold will be automatically deleted. | {no minimum,1 seconds,2 seconds,3 seconds,4 seconds,5 seconds} | 1 seconds |
| Say message Caller-ID | If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing. | Check box | Check |
| Say message duration | If this option is set, the duration of the message in mintues will be played back before the voicemail message begins playing | Check box | unCheck |
| Play envelope | Turn on/off playing introductions about each message when accessing them from the voicemail application. | Check box | unCheck |
| Allow users to review | Checking this option allows the caller to review their message before it is submitted as a new voicemail message | Check box | Check |

1. Voice mail application: ,Voicemail(${ARG},u)
2. Automatically generated configuration file (/etc/asterisk/voicemail.conf)

```
mailbox_number => password, name, email
mailbox_number : the number you use in extension.conf for VoiceMail() command and to register a user in sip.conf or iax.conf
password : the pass used to register a user in sip.conf or iax.conf
name : the name which to be associated with the mailbox
email : where a notification for the voicemail will come
```

3. IPPBX Max messages data: 150M

a) Email Settings for Voice mails

| Name | Description | Type | default |
|---|---|---|---|
| Send messages by e-mail only | If this option is set, then voicemails will not be checkable using a Phone. Messages will be sent via e-mail, only. Note: You need to have an smtp server configured for this functionality | Check box | unCheck |
| Attach recordings to e-mail | This option defines whether or not voicemails are sent to the Users' e-mail addresses as attachments. Note: You need to have an smtp server configured for this functionality | Check box | Check |
| Template for | From | Str* | asterisk@y |

| | | | |
|---|---|---|---|
| Voicemail Emails | | | ourcompany.null |
| | Subject | New voicemail from ${VM_CALLERID} for ${VM_MAILBOX} | |
| | Template Variables:<br>\t : TAB<br>${VM_NAME} : Recipient's firstname and lastname<br>${VM_DUR} : The duration of the voicemail message<br>${VM_MAILBOX} : The recipient's extension<br>${VM_CALLERID} : The caller id of the person who left the message<br>${VM_MSGNUM} : The message number in your mailbox<br>${VM_DATE} : The date and time the message was left | Hello ${VM_NAME}, you received a message lasting ${VM_DUR} at ${VM_DATE} from, (${VM_CALLERID}). This is message ${VM_MSGNUM} in your voicemail Inbox. | |

b) SMTP Settings

| Name | Description | Type | default |
|---|---|---|---|
| STMP server | The IP address or hostname of an SMTP server that your box may connect to, without authentication, in order to send e-mail notifications of your voicemails; i.e. mail.yourcompany.com | Str* | |
| Port | The port number on which the SMTP server is running; generally port 25 | Str* | |
| Use TLS? | Use TLS(Transport Layer Security) when communicating with the SMTP server? | Check box | unCheck |
| Authentication? | Does the SMTP Server requite authentication? | Check box | unCheck |
| Username | The username of a valid account on the STMP server | Str* | |
| Password | The password of a valid account on the STMP server | Str* | |

1. Config file: /etc/ssmtp/ssmtp.conf
2. Note:
   Firmware after that starts support Gmail

## 4.14 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.

| Name | Description | Type | default |
|---|---|---|---|
| Extension | This is the number dialed to reach this Conference Bridge | Int | 6300 |
| Marked/Admin user Extension | If the conference bridge is to have marked users or admin users, then those users should enter the conference bridge using a separate extension. Admin conference users can lock and unlock the conference and can kick the most recent conference participant. Marked users are special users whose entrance and exit, if the Wait for Marked user or Close conference when last marked user exits can | Int | |

| | | | |
|---|---|---|---|
| | either begin or end the conference altogether | | |
| Pin Code | set an optional pin code, Ex: "1234" that must be entered in order to access the Conference Bridge | Str* | |
| Admin PinCode | Defining this option sets a PIN for Conference Administrators | Str* | |
| Play music for the first caller | Checking this option causes Asterisk to play Hold Music to the first user in a conference, until another user has joined the same conference | Check box | unCheck |
| Close conference for the list call exit | Close the conference bridge when the last marked user logs out of the conference call | Check box | unCheck |
| Enable call menu | Checking this option allows a user to access the Conference Bridge menu by pressing the * "Asterisk" key on their dialpad | Check box | unCheck |
| Announces callers | Checking this option announces, to all Bridge participants, the joining of any other participants | Check box | unCheck |
| Quiet mode | Do not play enter/leave sounds | Check box | unCheck |
| Wait for marked user | Prevent conference participants from hearing each other until the marked user has joined | Check box | unCheck |

1.Conferencing application:

MeetMe([confno][,[options][,pin]]): Enters the user into a specified MeetMe conference

ex.: MeetMe(${EXTEN}|MsIqwxaA)

'1' — disable "you are currently the only person in this conference" message for first member

' a' — set admin mode

'A' — set marked mode

'b ' — run AGI script specified in ${MEETME_AGI_BACKGROUND}

'c' — announce user(s) count on joining a conference

'd' — dynamically add conference

'D' — dynamically add conference, prompting for a PIN

At the pin prompt, if the user does NOT want a pin assigned to the conference, they should hit the # key.

'e' — select an empty conference

'E' — select an empty pinless conference

'F' — Pass DTMF through the conference.

'i' — announce user join/leave with review

'I' --announce user join/leave without review

'M' — enable music on hold when the conference has a single caller

'm' — set monitor only mode (Listen only, no talking)

'p' — allow user to exit the conference by pressing '#'

'P' — always prompt for the pin even if it is specified

'q' — quiet mode (don't play enter/leave sounds)

'r' — Record conference (records as ${MEETME_RECORDINGFILE} using format ${MEETME_RECORDINGFORMAT}).

's' — Present menu (user or admin) when '*' is received ('send' to menu)

't' — set talk only mode. (Talk only, no listening)

'T' — set talker detectio

'v' — video mode
'w' — wait until the marked user enters the conference (plays music on hold until marked user enters if M is used)
All other connected users will hear MusicOnHold until the marked user enters.
'X' — allow user to exit the conference by entering a valid single digit extension of the context specified in ${MEETME_EXIT_CONTEXT} or the current context if that variable is not defined.
'x' — close the conference when last marked user exits

## 4.15 Follow Me

If A calls B, B does not answer, the call will be transferred to C who is set up in follow me.

| Name | Description | Type | default |
|---|---|---|---|
| Status | Enable/Disable FollowMe for this user | Choice | Disable |
| 'Music On Hold' Class | Music On Hold class that the caller would hear while tracking the user | Choice | Default |
| DialPlan | DialPlan that would be used for dialing the FollowMe numbers. By default this would be the same dialplan as that of the user | Choice | |
| Destinations | List of extensions/numbers that would be dialed to reach the user during FollowMe | Destinations | |
| New FollowMe Number | Add a new FollowMe number which could be a 'Local Extension' or an 'Outside Number'. The selected dialplan should have permissions to dial any outside numbers defined | {Dial Local Extension, Dial Outside Number | |
| Dial Order | This is the order in which the FollowMe destinations are dialed to reach the user | {Ring after Trying previous extension/number , Ring along with previous extension/number} | Ring after Trying previous extension/numbe |
| Follow me Option | Playback the unreachable status message if we've run out of steps to reach the or the callee has elected not to be reachable | Check box | Uncheck |
| | Playback the unreachable status message if we've run out of steps to reach the or the callee has elected not to be reachable | Check box | Uncheck |
| | Playback the unreachable status message if we've run out of steps to reach the or the callee has elected not to be reachable | Check box | Uncheck |

1.General config file : /etc/asterisk/followme.conf

# 4.16 Directory

Dialing the 'Directory Extension' would present to the caller, a directory of users listed in the system telephone directory - from which they can search by First or Last Name. To add or remove a user from the system telephone directory, edit the 'In Directory' field of the user. Preferences for 'Dialing by Name Directory'.

Directory setting:

| Name | Description | Type | default |
|---|---|---|---|
| Directory Extension | Extension to dial for accessing the Name Directory | Int | |
| Also read the extension number | In addition to the name, also read the extension number to the caller before presenting dialing options | Check box | Uncheck |
| Use first name instead of last name | Allow the caller to enter the first name of a user in the directory instead of using the last name | Check box | Uncheck |

1.Directory application: Directory(default|default|ef)

# 4.17 Call Features

Feature Codes and Call parking preferences

Features Codes

| Name | Description | Type | default |
|---|---|---|---|
| Features Codes | Blind Transfer (default is #) | Check box&&Int | # |
| | Disconnect (default is *) | Check box&&Int | * |
| | Attended transfer | Check box&&Int | |
| | Call Parking (Packing a call) | Check box&&Int | |

Call Parking Preferences

| Name | Description | Type | default |
|---|---|---|---|
| Call Parking Preferences | Extension to Dial to Park a call | Int | 700 |
| | What extensions to park calls on | Int | 701-720 |
| | Number of seconds a call can be parked for | Time | |

Application Map

| Name | Description | Type | default |
|---|---|---|---|
| Application Map | Add an application for PBX | | |

Dial Options

| Name | Description | Type | default |
|---|---|---|---|
| Dial Options | (t-Option) Allow the called party to transfer the calling party by sending the DTMF sequence defined on the Feature Codes page | Check box | Uncheck |
| | (T-Option) Allow the calling party to transfer the called party by sending the DTMF sequence defined on the Feature Codes page. | Check box | Uncheck |
| | (h-Option) Allow the called party to hang up by sending the DTMF sequence defined on the Feature Codes page. | Check box | Uncheck |
| | (H-Option) Allow the calling party to hang up by sending the DTMF sequence defined on the Feature Codes page. | Check box | Uncheck |
| | (k-Option) Allow the called party to enable parking of the call by sending the DTMF sequence defined on the Feature Codes page. | Check box | Uncheck |
| | (K-Option) Allow the calling party to enable parking of the call by sending the DTMF sequence defined on the Feature Codes page. | Check box | Uncheck |

## 4.18 VoiceMail Groups

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

| Name | Description | Type | default |
|---|---|---|---|
| VoiceMail Group's Extension | Default Voicemail Group's Extension | Int | 6601 |
| Label | The label of Voicemail Group's Extension | Str* | |
| User MailBoxes | The entire user Mailboxes | Check boxs | |

## 4.19 Voice Menu Prompts

This component is used for recording custom voice menu.

| Name | Description | Type | default |
|---|---|---|---|
| Voice menu prompts | File Name | Str* | |
| | dial this User Extension to record a new voice prompt | Choice | 6001 |
| | Upload a Voice menu prompt | Choice | |

## 4.20 System Info

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

a) General
OS Version: Linux version for PBX
Uptime: uptime for PBX

Version Details: asterisk/GUI/Firmware version for PBX
Server Date & TimeZone: time now for PBX
Hostname:name for PBX

b) Network: network message for PBX
Eth0:9----- fill back IP for PBX(vlan IP)

c) Disk Usage
Filesystem: File system of PBX
1k-blocks： A total of system modules
Used ： Used of system modules
Available ： Available of system modules
Use% ： Percentage
Mounted on： The specified directory

d) Memory Usage
Total： Memory quantity
Used： Used of Memory
Free： Free of Memory
Shared： Shared of Memory
Buffers： Buffers quantity

# 4.21 Backup

Backup and Restore are two of the mandatory functions of any application. IP-2G4A is no exception. Customers can backup all the files under the /etc/asterisk/ directory and restore them.

| Name | Description | Type | default |
|---|---|---|---|
| Backup | Create new backup | | |
| | Download from Unit | | |
| | Restore Previous config | | |

# 4.22 Active Channels

The channels which are in communication status will be displayed in this component.

| Refresh Now | Description |
|---|---|
| Status | Upload message for asterisk channels<br>Hangup：hang-up channel<br>Transfer：transfer channel |

# 4.23 Options

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

General Preferences

| Name | Description | Type | default |
|---|---|---|---|
| Global OutBound CID | This is default global CallerID that is used for all outgoing calls when no other CallerID is defined that has a higher priority. | Int | |

| | | | |
|---|---|---|---|
| | When making outgoing calls the following rules are used to determine which CallerID will be used, if they exist:<br>The first CallerID used is a CallerID set for the user making the call defined in the 'Users' tab.<br>The second CallerID is the one that is set in the 'VoIP Trunks' configuration, if applicable<br>The last CallerID used for outgoing calls is the Global CID defined in the 'Options' tab. | | |
| Operator Extension | The Operator Extension is the extension which will be dialed when a caller presses '0' to exit Voicemail. It is also available as a Voice Menu option | Chioce | |
| Ring Timeout | Number of seconds to ring a device before sending to the user's Voicemail Box | Time | 20 |
| Call Record Dir | Call Record Dir | Str* | /tmp |
| Call Record Format | Call Record Format | Choice | gsm |
| Extension preferences | User Extensions | Int | 6001-6299 |
| | Conference Extensions | Int | 6300-6399 |
| | VoiceMenu Extensions | Int | 7001-7100 |
| | RingGroup Extensions | Int | 6400-6499 |
| | Queue Extensions | Int | 6500-6599 |
| | VoiceMail Group Extensions | Int | 6600-6699 |
| | Resert to default | | |

Languages

| Name | Description | Type | default |
|---|---|---|---|
| Languages | The Language setting allows the user to specify the default prompts language for phone to phone, inbound, and outbound calls. If a soundpack selection is made but not already installed, then the pack will be downloaded from Digium | Chioce | English |

Change Password

| Name | Description | Type | default |
|---|---|---|---|
| Change Password | Enter New Password | Str* | |
| | Retype New Password | Str* | |

Factory reset

| Name | Description |
|---|---|
| Factory reset | Reset to defaults include network settings |
| | Reset to defaults but keep network settings |

# Chapter 5 an Application Case of IP-2G4A

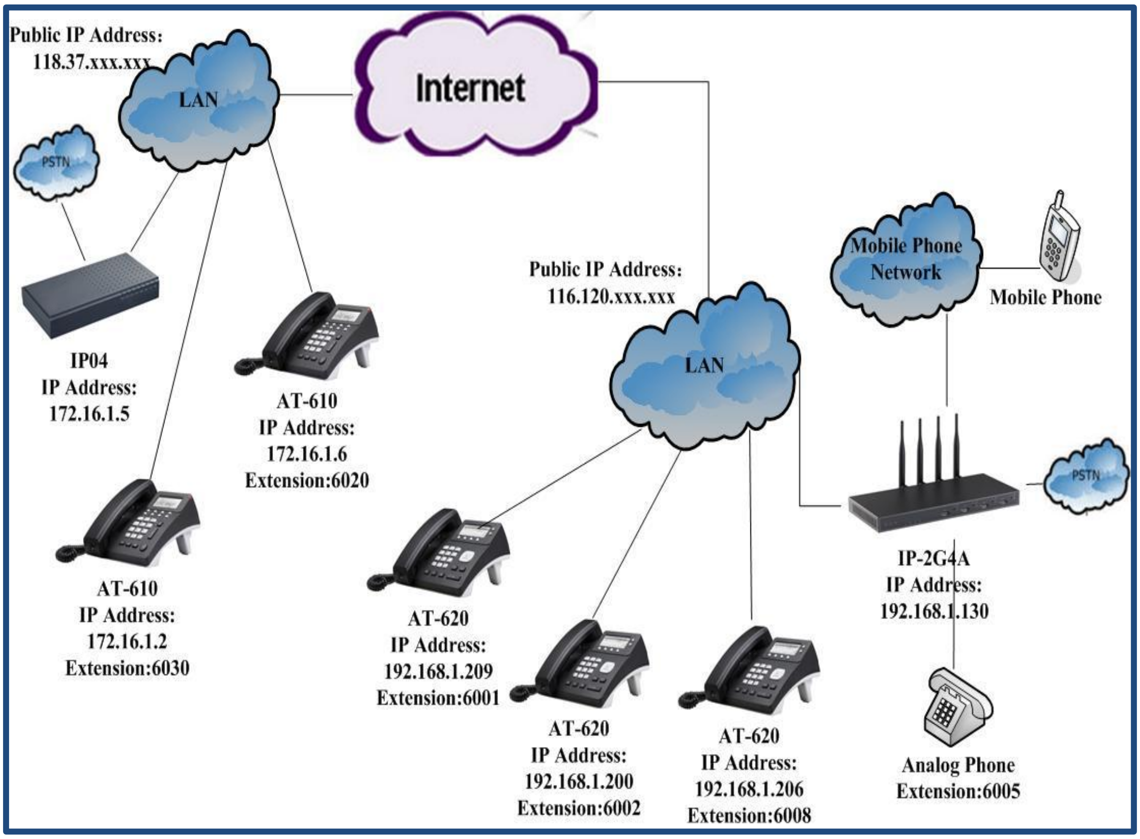



Figure: Network Topology

In the network topology above: user 6020,6001,6002,6008 will be registered to IP-2G4A, analog phone 6005 is connected to FXS port of IP-2G4A. After configuration, it will realize the following function:

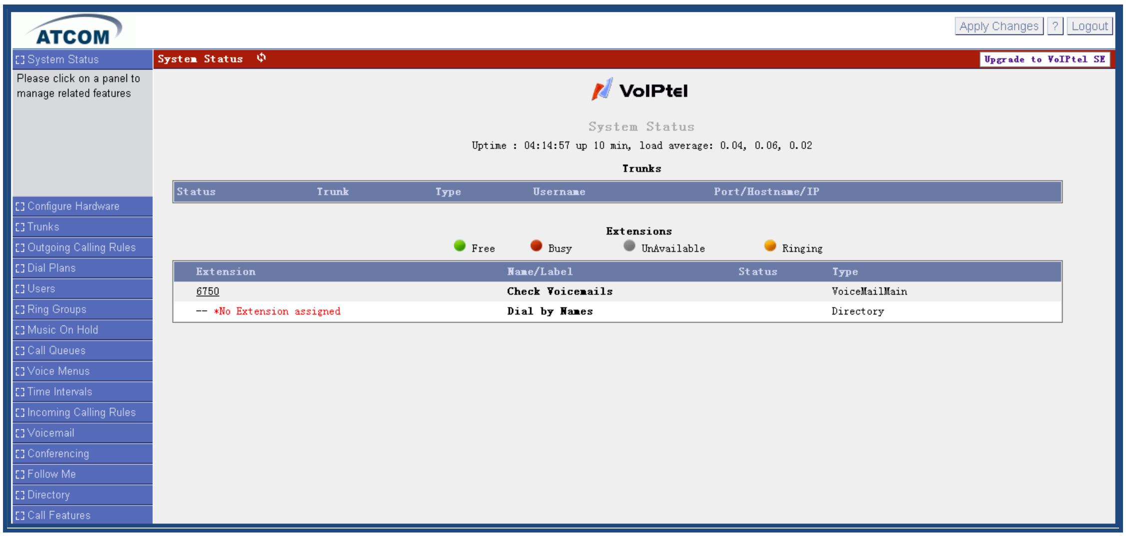
1) The internal user 6005 and user 6001, or user 6002 and user 6001 can call each other directly.
2) 6005 and 6001 can dial out through IP-2G4A to PSTN.
3) 6005 and 6001 can get incoming calls from PSTN by IP-2G4A.
4) 6001, 6002, 6008 or 6005 are all communication with the mobile phone by IP-2G4A.
5) 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.
7) Voicemail
8) IVR
9) Conference
10) Ring Groups
11) Agents
12) Follow me
13) Call pickup

# 5.1 How to Make Internal Calls through IP-2G4A

## 5.1.1 Access to the Web Page of IP-2G4A by Browser

After connecting IP-2G4A to LAN, please open your browser of PC with windows OS and input the IP Address of IP-2G4A (the default IP address is 192.168.1.100), then you can get the following screen:

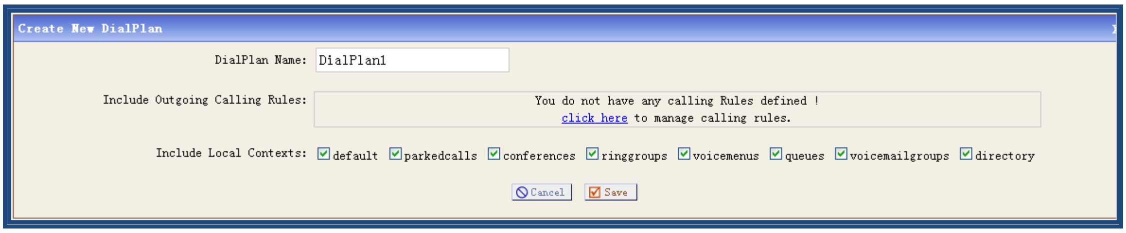

Please input the default Username: admin; Password: atcom in the presented screen above.
When you login successfully, you can get the configuration web page as below:




## 5.1.2 Add up Users from Web Page of IP-2G4A

1) Add up a DialPlan

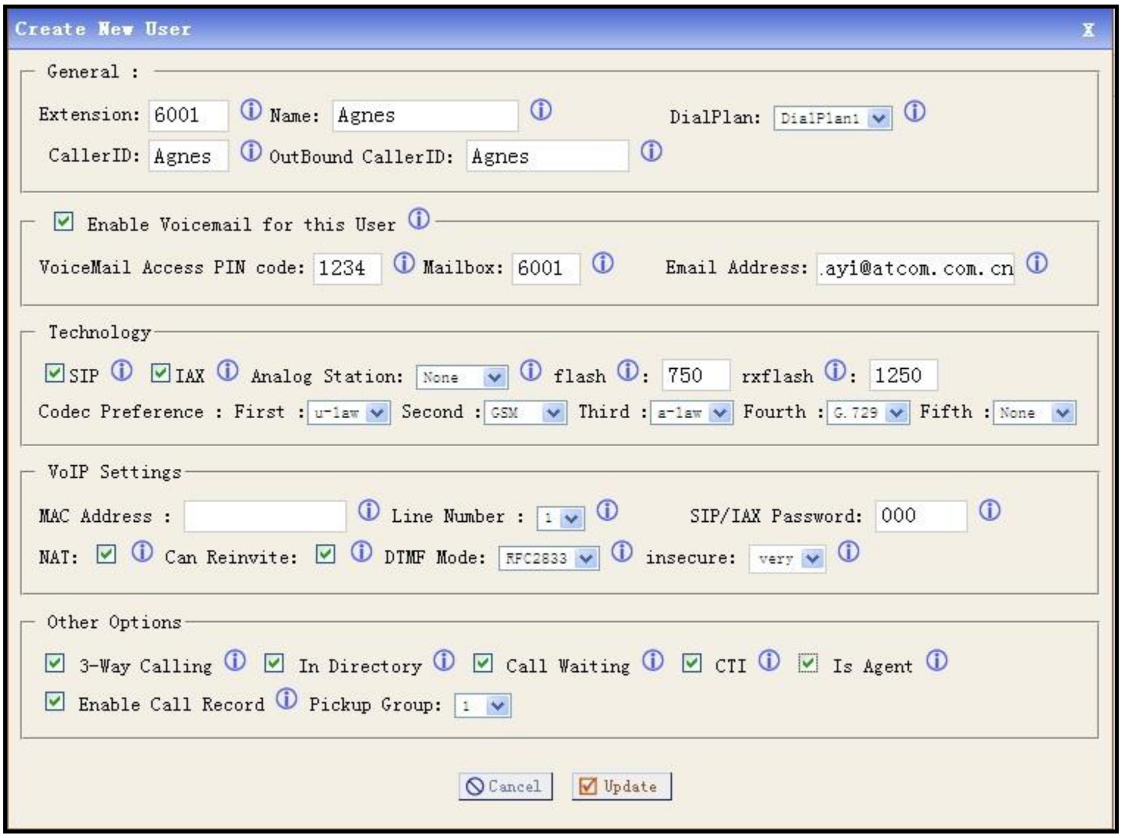
Before you add up user, you have to add up a DialPlan, please click on Dial Plans→New DialPlan, I add up a DialPlan like the following:




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

2) Add up SIP user 6001

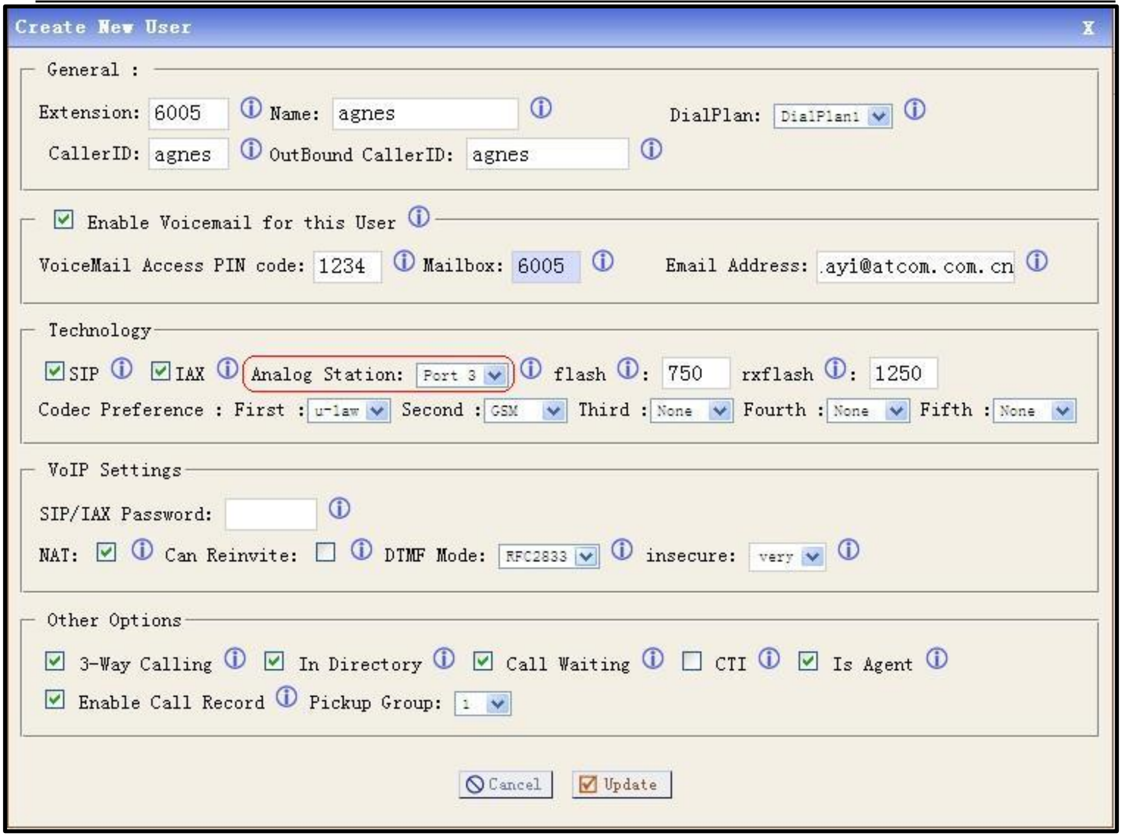
After logging into the web page of IP-2G4A, please click on **Users→ Create New User**, I configure user 6001 like the following:




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

3) Add up an Analog user 6005

After logging into the web page of IP-2G4A, please click on **Users→ Create New User**, I add a user 6005 like the following:

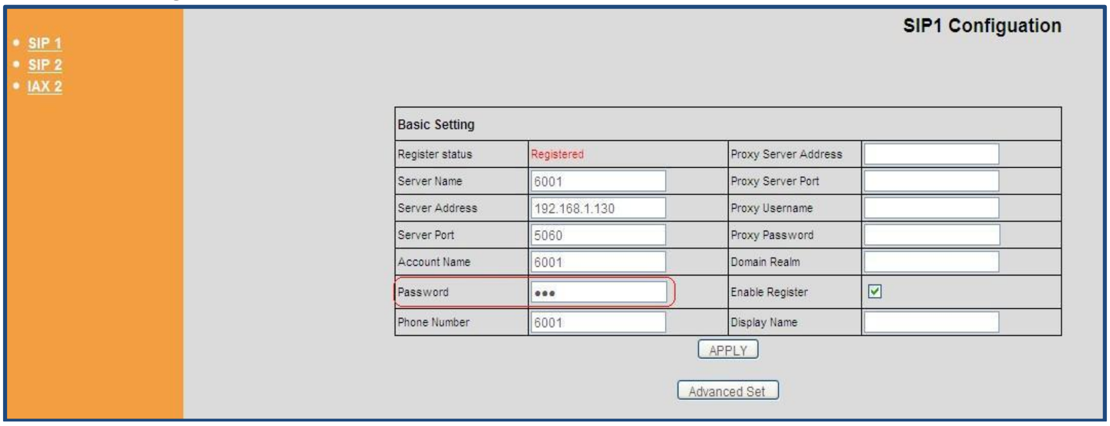



You must pay attention to the red ellipse frame on **Analog Station** drop-down list and choose the port which you need. At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

Attention: then you must reboot your IP-2G4A. The configuration can go into effect.

## 5.1.3 Register a SIP user 6001 in AT-620

After logging into the web page of IP Phone AT-620, please select VOIP option, I register the 6001 as the following illustration:




After configuring, please click on the **APPLY** button. You can see the "Register status" is Registered, if you do not register successfully, please pay attention to the Password in the red ellipse frame , which must be the same with the SIP/IAX Password of the user 6001 in IP-2G4A.

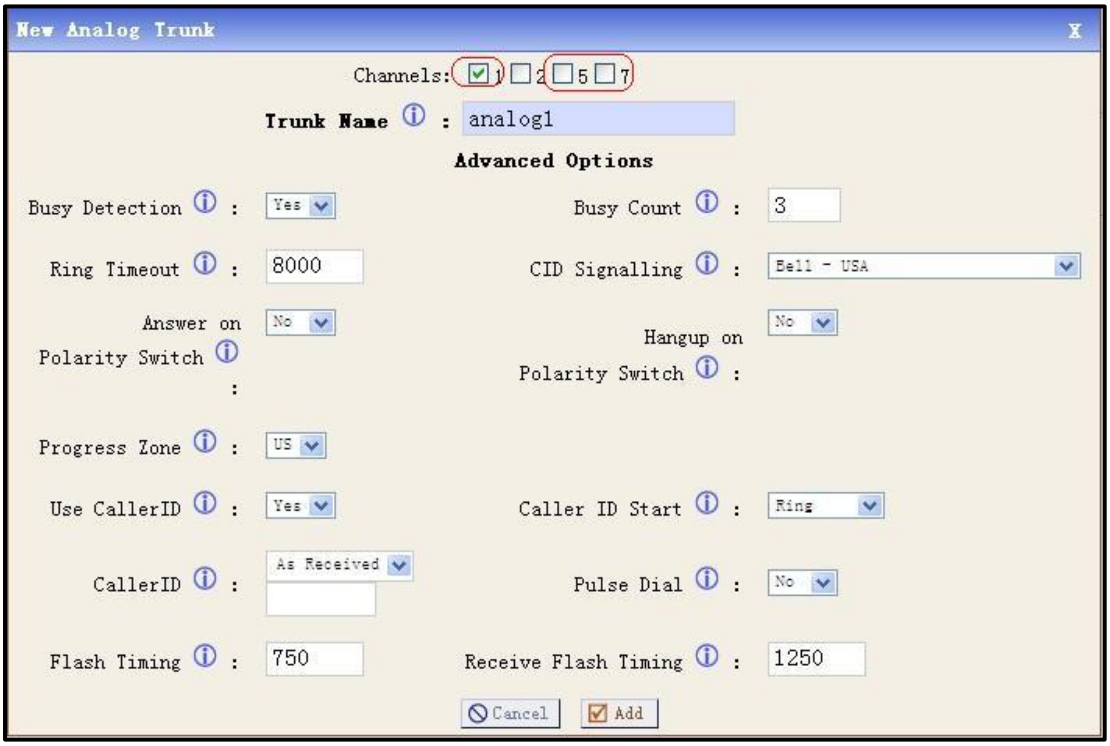
Now you can call each other directly between user 6001 and 6005.

# 5.2 How to Communicate with Outside through PSTN

In order to communicate with outside through PSTN with IP-2G4A, you need an analog trunk, an outgoing calling rule, a dial plan ,a incoming calling rule 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.

## 5.2.1 Create an Analog Trunk

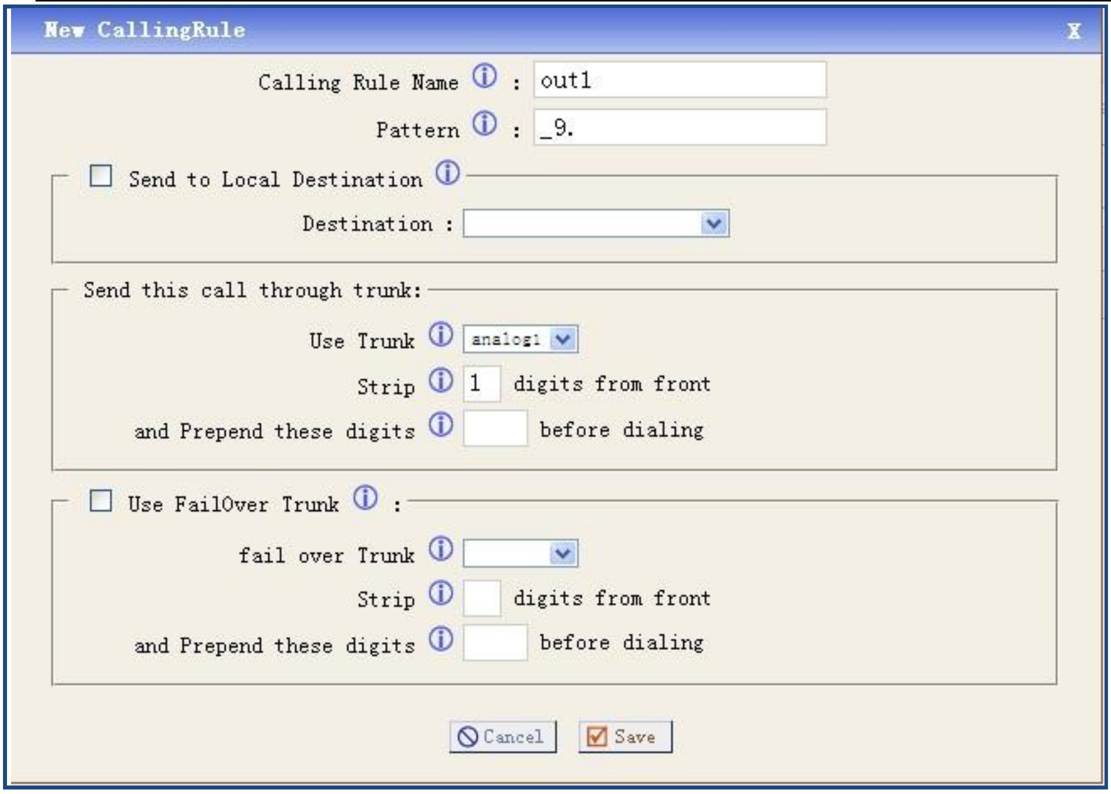
After logging into the web page of IP-2G4A, please click on **Trunks→ Analog Trunks**, I configure an analog trunk like the following:




You should hook on the Channels you need , input the name of the trunk. Others are default. At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page. Then you must restart the IP-2G4A. Please pay attention to the red ellipse frame in the screenshot above, channels five and seven are used for GSM.

## 5.2.2 Create an Outgoing Calling Rule

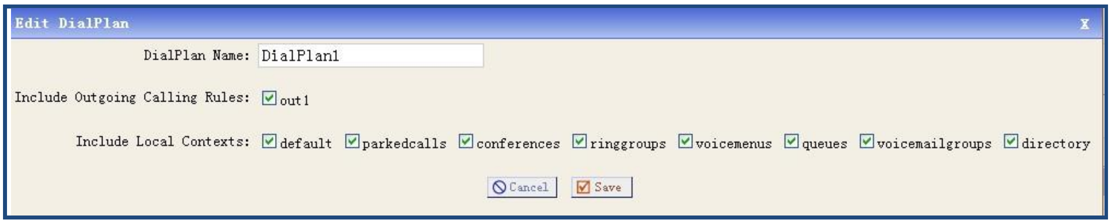
After logging into the web page of IP-2G4A, please click **Outgoing Calling Rules→ New Calling Rule**, I configure an outgoing calling rule like the following:




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

## 5.2.3 Selected the Outgoing Calling Rules in a Dial Plan

After logging into the web page of IP-2G4A, please click on **Dial Plans→Edit DialPlan1**, then selected the name of the outgoing calling rules like the following:




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

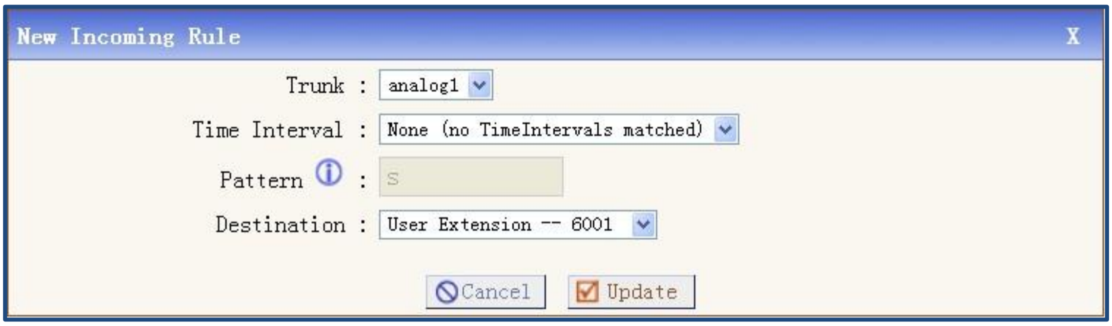
## 5.2.4 Create a User

I will use the user 6001 I created before. Now I can call out with prefix 9, if the caller number is 10086, I will dial 910086. If you use GSM ports, we will communicate with outside by Mobile Phone Network.

## 5.2.5 Create Incoming Calling Rules

In order to get an incoming call from outside with IP-2G4A, you need set Incoming Calling Rules. Of course the precondition is that you have set up a trunk, a destination which include Voice Menu、Voice mail、 a User Extension etc and a Time Interval.

After logging into the web page of IP-2G4A, please click on I**ncoming Calling Rules→ New Incoming Rule**, I configure an incoming calling rule like the following:




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

Here I use analog trunk 1, you can choose you need. Then when the outside makes a incoming call , it will be sent to user 6001 through analog 1. you may configure the analog trunk for GSM by wireless in a similar way.

Attention: Here if you choose the five channel, Outgoing Calling Rule and Incoming Calling Rules are both use the channel 5. Then you can communication with the mobile phone. For example:

I configure the Outgoing Calling Rule as _5. Then use the channel 5 and the number is 158xxxxxxx2. Incoming Calling Rules be pointed to 6001. Then I can dial a mobile phone number with prefix 5, others can dial 158xxxxxxx2 to connect us.

# 5.3 How to Call Each Other Directly from Different Network Segment.

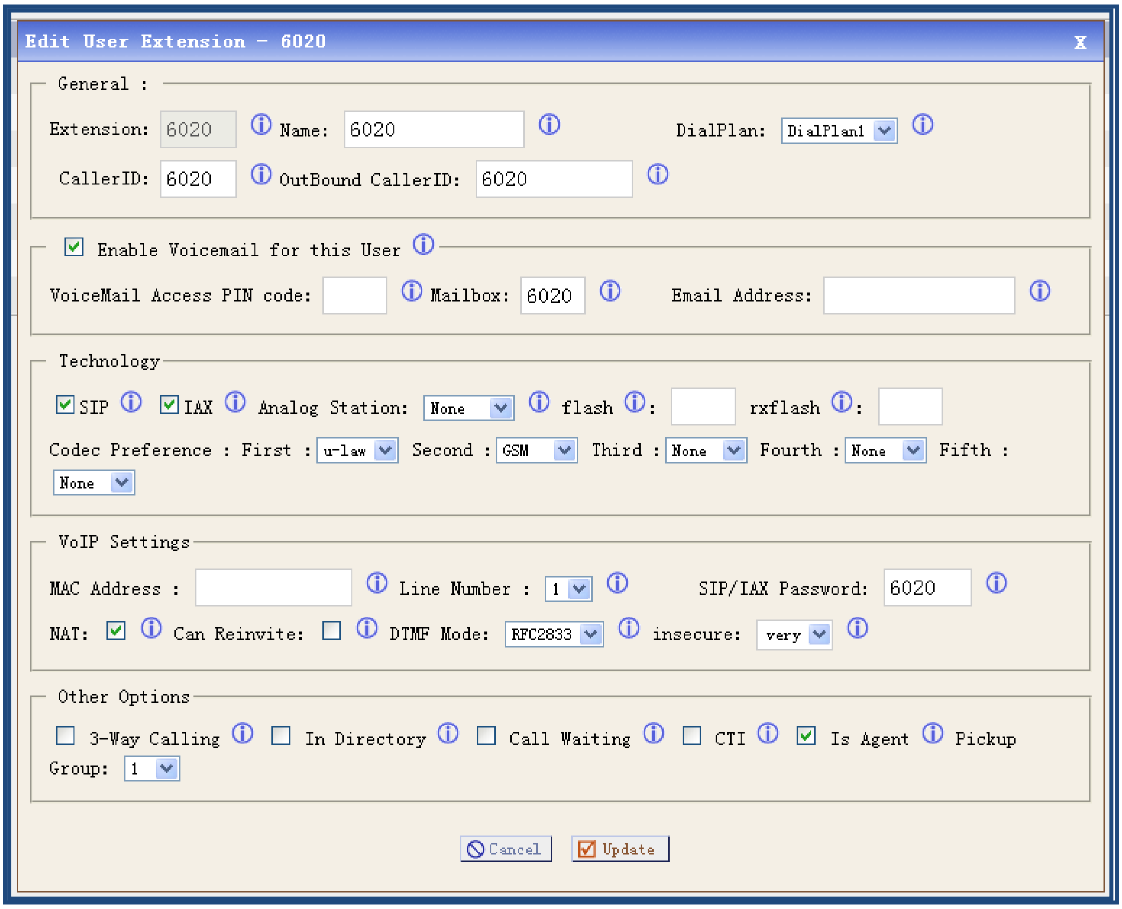
Take the user 6020, 6005 and 6001 for example, I will configure router, users and IP-2G4A, then the three users can call each other directly.

1) Set up router

   Please configure the router 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. ; DHCP option and so on. Configure **Port Range Forwarding** , you can use IAX2 , you can configure Port Range Forwarding as 4569. IP address is 192.168.1.130(the IP Address of IP-2G4A). Here I use IAX2, so I create a IAX2 user named 6020.

2) Add an IAX user 6020 in IP-2G4A

   After logging into the web page of IP-2G4A, please click on **Users→ Create New User**, I configure 6020 like the following:



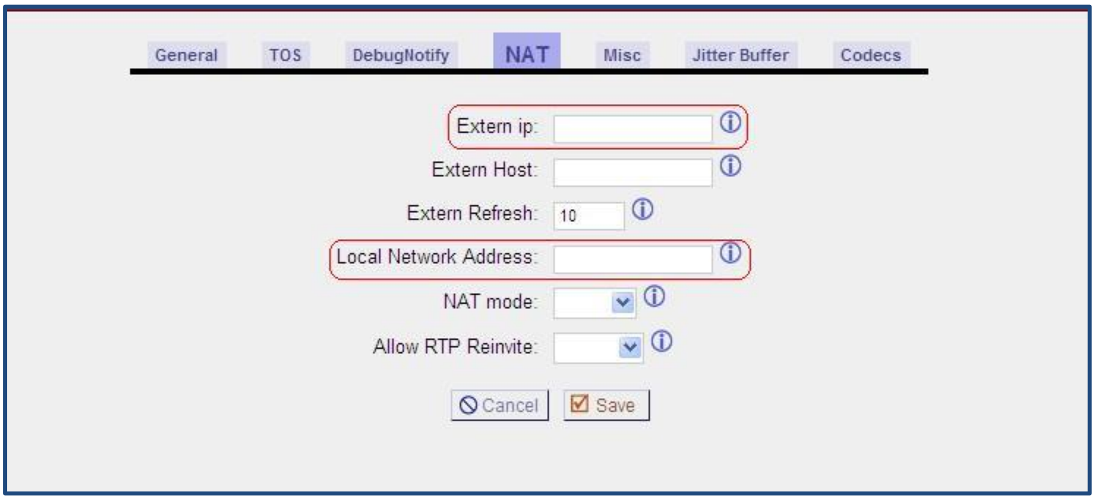

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3) Set up AT-610 and register an IAX2 user 6020

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

Please pay attention to the red ellipse frame in the screenshot above, it is the IP address of the router. 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 IP-2G4A. If you use SIP user, you need configure the **SIP Setting** , where is in **Options → Adance Options → Show Advance Options**, you can configure the two options in the red ellipse frame in the screenshot like this:



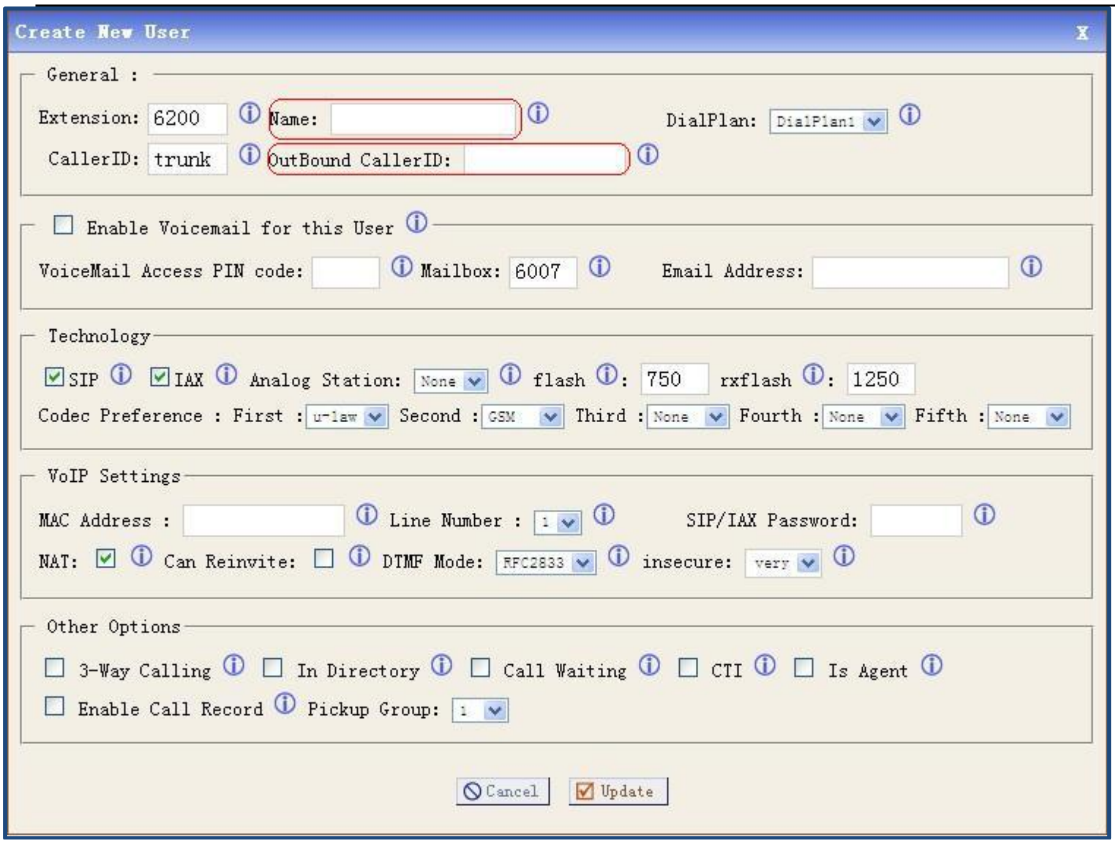

Now you can call each other among 6020,6001 and 6005 directly.

# 5.4 How to Call through VoIP Trunk

## 5.4.1 Call from IP-2G4A to IP04

In order to call from IP-2G4A to IP04, I will create a user in IP04 for the SIP/IAX trunk in IP-2G4A, create a SIP/IAX trunk, an outgoing call rule and a dial plan in IP-2G4A. But pay a attention that at the same time a port of the router where the IP04 in must be directed to the IP04.

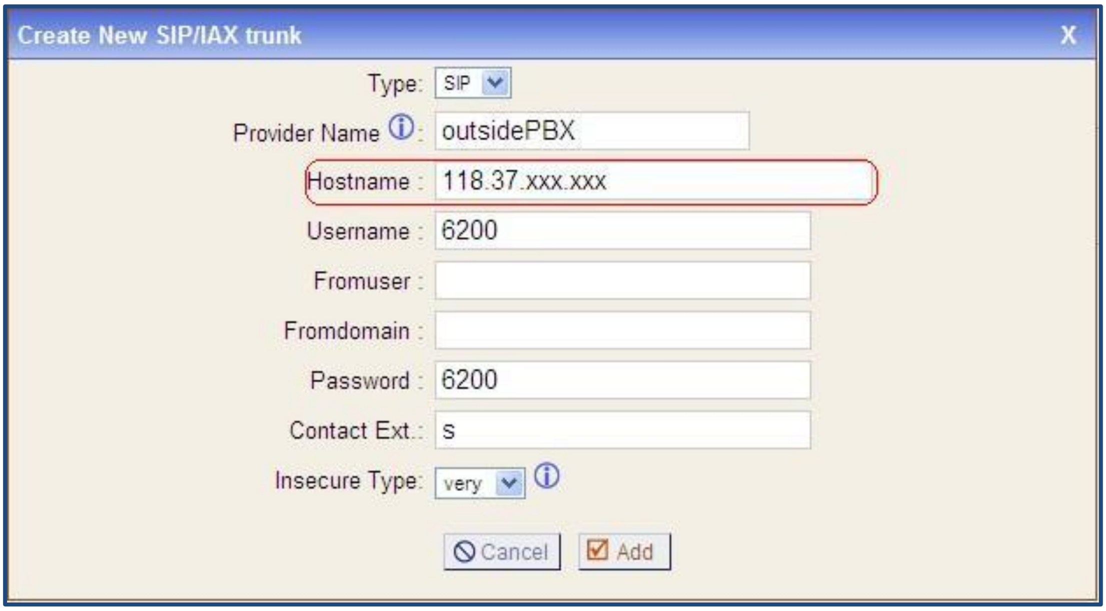
1) Add an user 6200(it will be used as SIP trunk in IP-2G4A) in IP04, after logging into the web page of IP04, please click on **Users→ Create New User**, I add the user 6200 like the following:




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 “Name” and “OutBound CallerID” in the red ellipse frame, if the user uses for a trunk ,the two options are null so that the caller ID on the phone is the calling party.

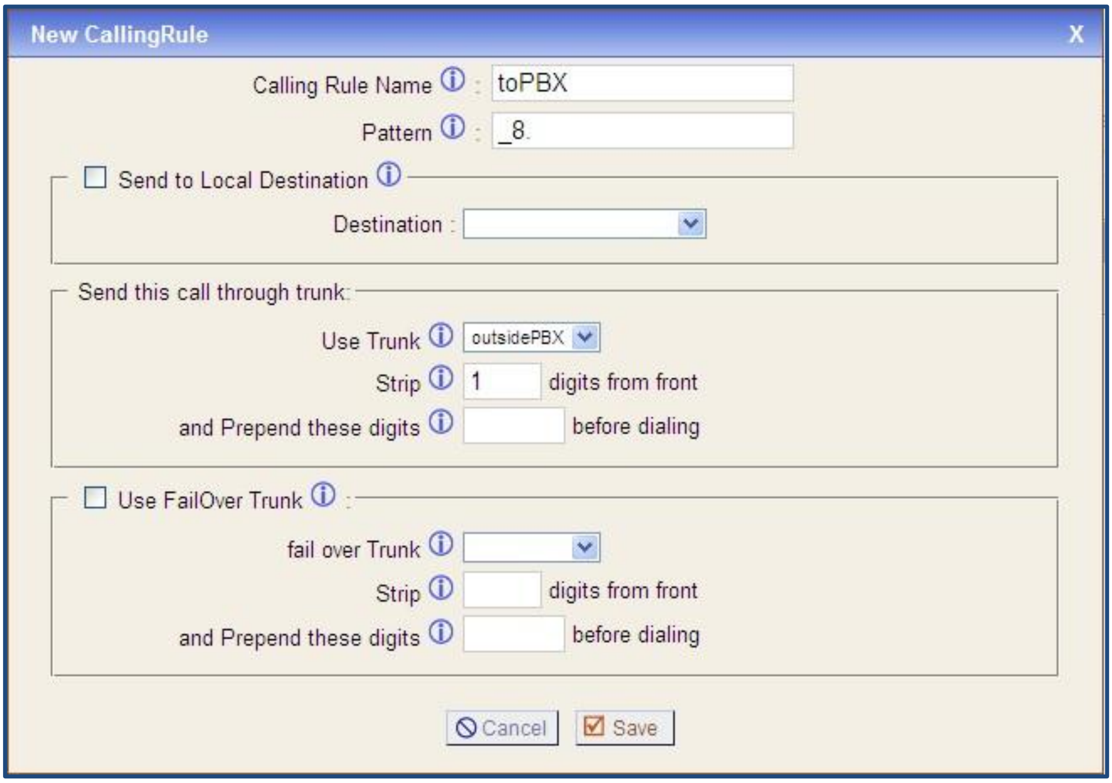
Then Add a user 6030 in IP04 for AT-620, the way is the same as adding 6001.

2) Add a VoIP trunk in IP-2G4A, after logging into the webpage of IP-2G4A, please click on **Trunks→VOIP Trunks→New SIP/IAX Trunk**, I configure a SIPtrunk1 like the following:



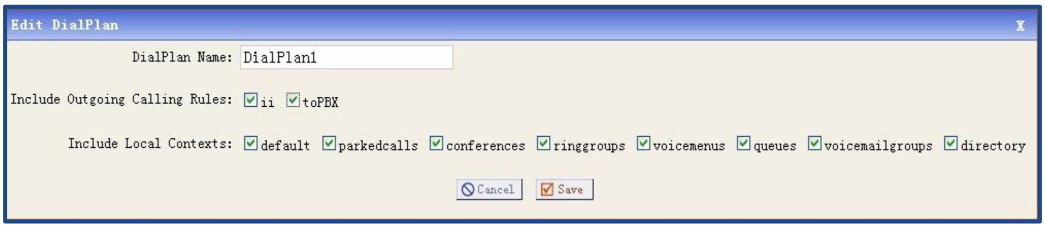

Please pay attention to the red ellipse frame, the Hostname is the public IP address where the IP04 is. After configuring, please click on **Add** button, and click on **Apply Changes** button in up right corner of the main page. Attention : the option of Fromuser :default is null.

3) Create an outgoing calling rule in IP-2G4A, after logging into the webpage of IP-2G4A, please click on **Outgoing Calling Rules→New Calling Rule**, I configure an outgoing call rule like the following:




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

4) Hook on the outgoing calling rules in dial plan in IP-2G4A, after logging into the webpage of IP-2G4A, please click on Dial Plans→Edit DialPlan, and then hook on the outgoing calling rules:




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, Now you can call from 6001 or 6005 to 6030 by dialing 86030

## 5.4.2 Call from IP04 to IP-2G4A

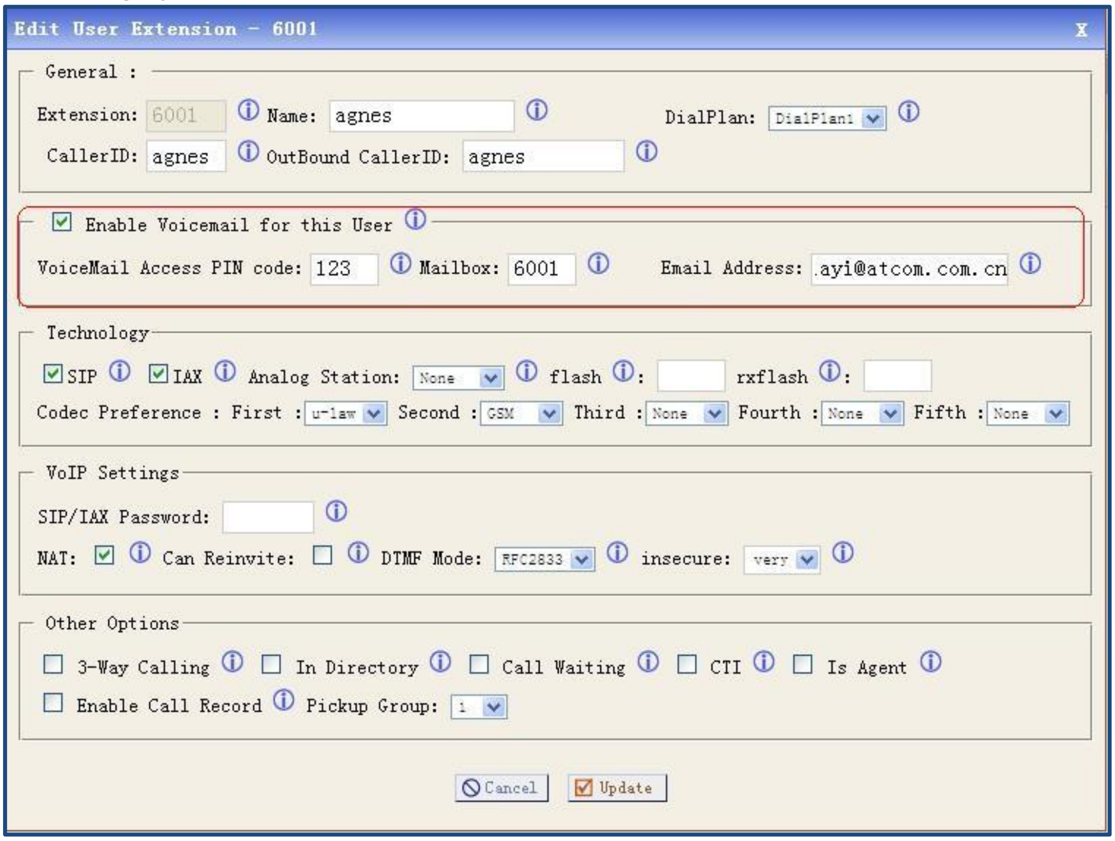
In order to call from IP04 to IP-2G4A, I will create a SIP user in IP-2G4A for the SIP trunk in IP04 like 4.5.1, and then create a SIP trunk, an outgoing call rule and a dial plan in IP04.

1) Add a user 6008 in IP-2G4A like 5.1.2.
2) Create a SIP trunk in IP04 named out .
3) Configure the router.
4) Create an outgoing calling rule in IP04 named toIP-2G4A . Here I use Pattern: _4. .
5) Hook on the "toIP-2G4A" option in DialPlan.

After configuring, please click on Save button, and click on Apply Changes button in up right corner of the main page. Now you can call from 6030 to 6001 and 6005 by dialing with prefix 4. You can communication between IP04 and IP-2G4A.

## 5.5 Voicemail

You can configure Voicemail in Users ,for example 6005 which we have configured in 4.1.2. please click on **Users→Edit** on 6001 , you can see the configuration in the following picture, especially pay attention to the configuration in the red ellipse frame. Then when you want to listen to a message, you can dial 6750 or the Mailbox 6001.




## 5.6 How to realize the IVR

IVR is Interactive Voice Response. Voice Menus allow for more efficient routing of calls from incoming callers. Also known as IVR menus or Digital Receptionist.

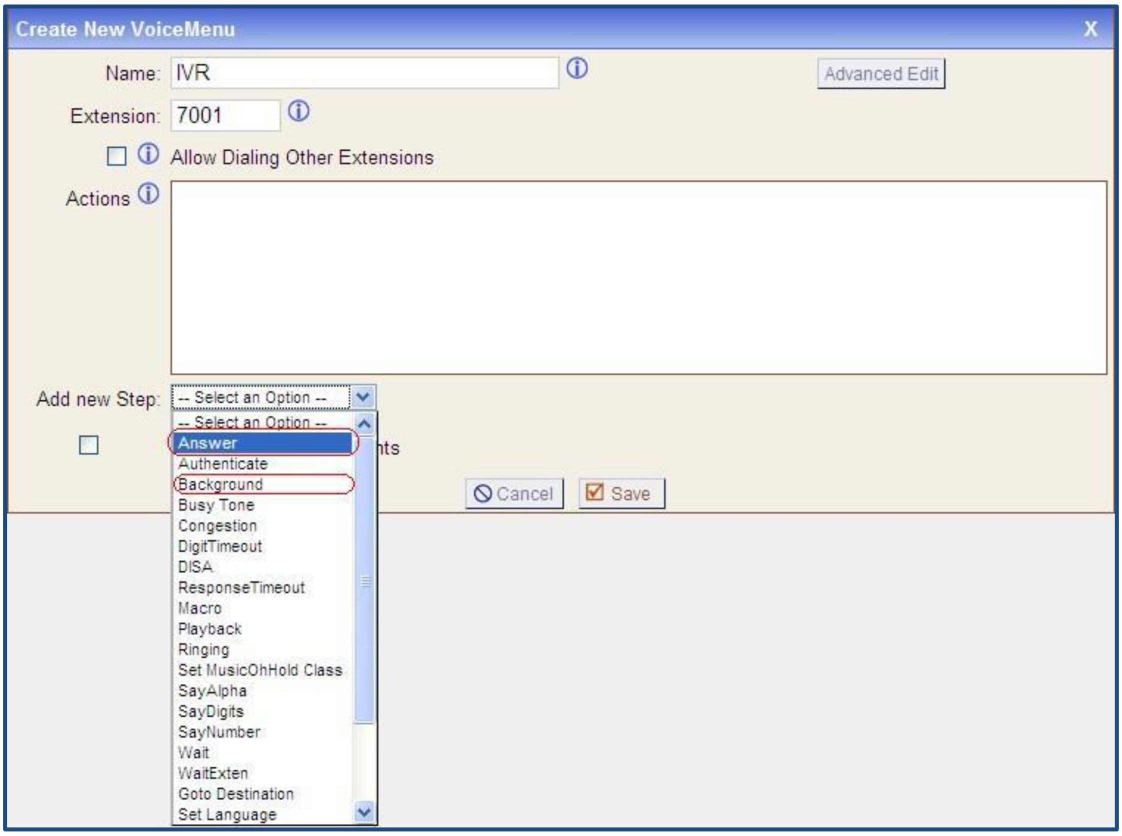
## 5.6.1 Upload Voice Menu Prompts

If you want to configure the IVR which you need , you must upload your voice prompt. You can click on **Voice Menu Prompts** , you can see the screen like this screenshots:

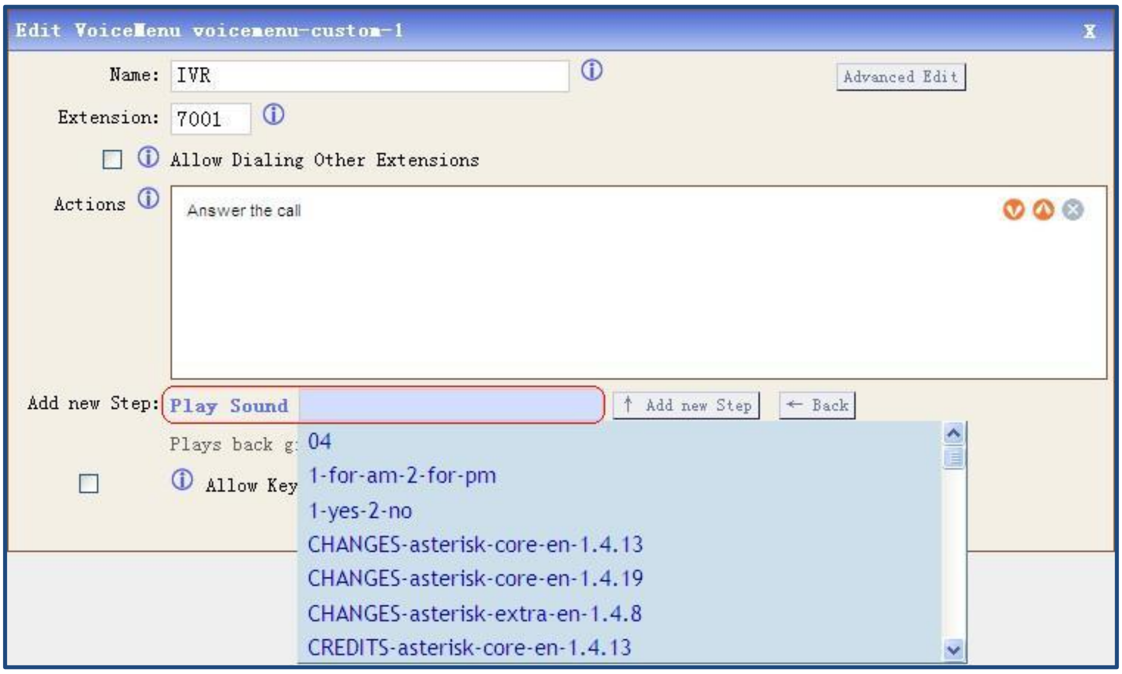

You can click the button of "Record a new Voice Menu prompt" to record a voice prompt, or you can click the button of "Upload a Voice Menu prompt" to upload your voice prompt.

## 5.6.2 Create Voice Menu

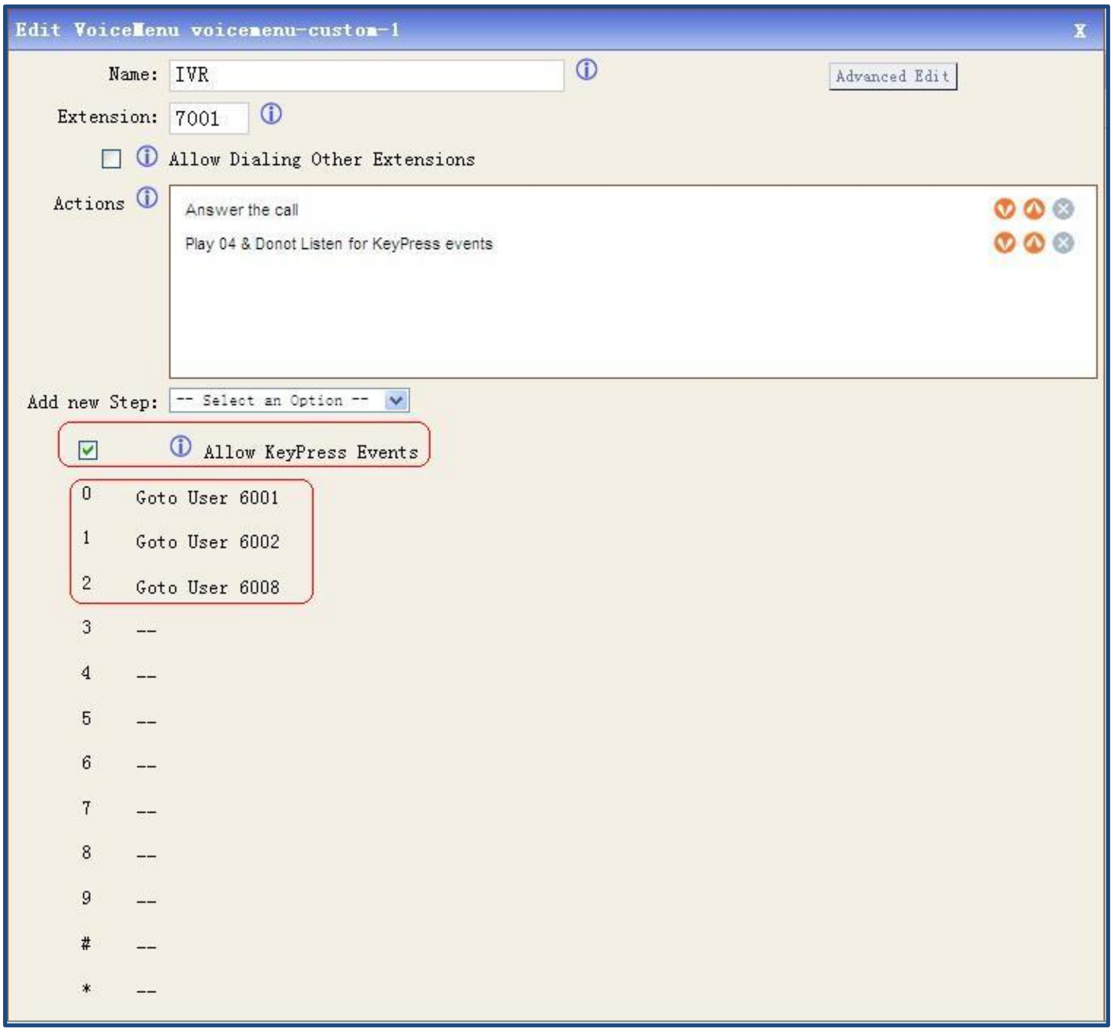
You can configure IVR like this: click on **Voice Menus → Create Voice Menus**, then you can configure the IVR like the following pictures. First, selected the option "Answer" on the "Add new step" then click the **Add new step** .




Second , selected the option "Background" on the "Add new step" then click the **Add new step** .you can see the screen display like the following screenshots, then you can select your own voice prompt. Here I use the voice prompt named 04. You can upload the voice prompt like 5.6.1.

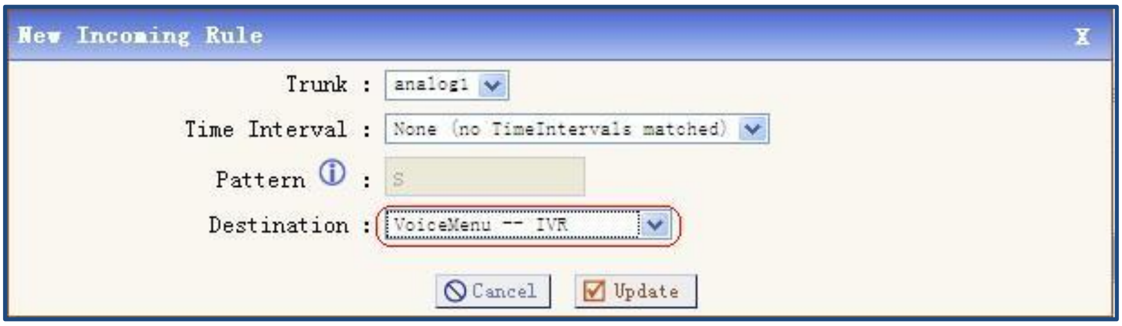



Third, hook on the option : **Allow KeyPress Envents**, then you can configure the operation from "0" to "*", which you need. Please click on save button, and click on Apply Changes button in up right corner of the main page. Here I configure that press "0" then call "6001", press "1" then call "6002", press "2" then call "6008". Of course 6001, 6002, 6008 have registered.




## 5.6.3 Add Incoming Calling Rules

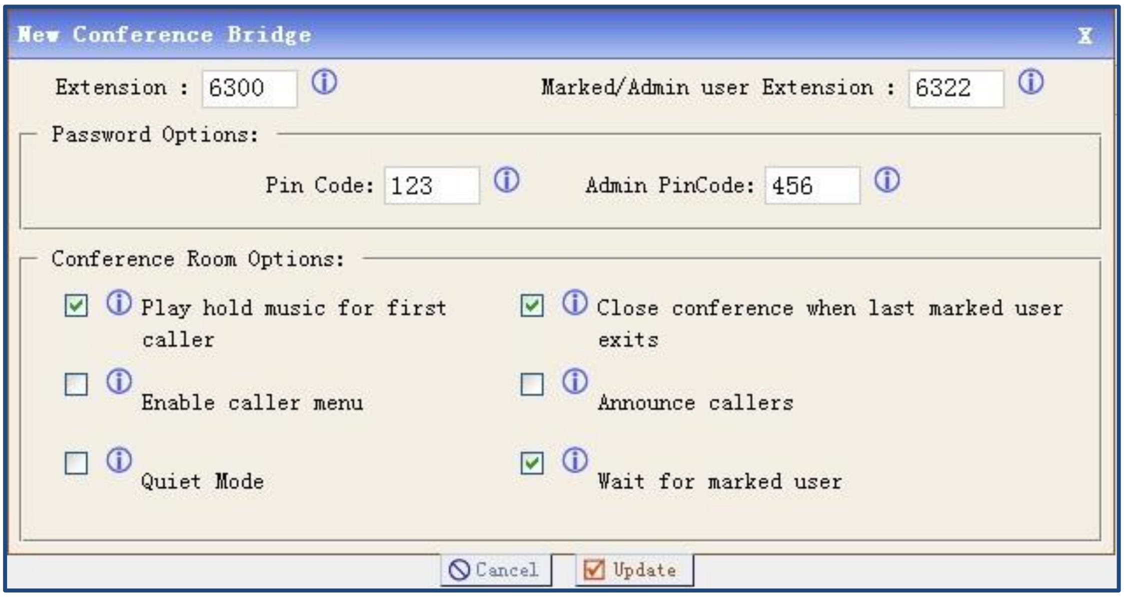
After configure the Voice Menu, you must configure the Incoming Calling Rules. Click **Incoming Calling Rules → New Incoming Calling Rules**, you can configure it like this:




Then when others call you through the analog1, they can here the IVR and do the operation which they need.

## 5.7 Conference

In order to realize the conference option, the users which will attend to the conference must have registered. Here I use 6001, 6002, 6005. Now please click **Conferencing → New conference Bridge**, you can see the screen like the following screenshots:




Then please click on Update button, and click on Apply Changes button in up right corner of the main page. Here I configure it like the screenshots above. Then 6001 dial 6300, and input Pin Code. you can here a voice promt means you are the fist user and wait oters, then you can here the music. 6002 does the same operation. 6005 dial 6322 and input Admin PinCode. Now all the users are in the conference. You can see the detail in 4.14 and configure it as your need.

## 5.8 Ring Groups

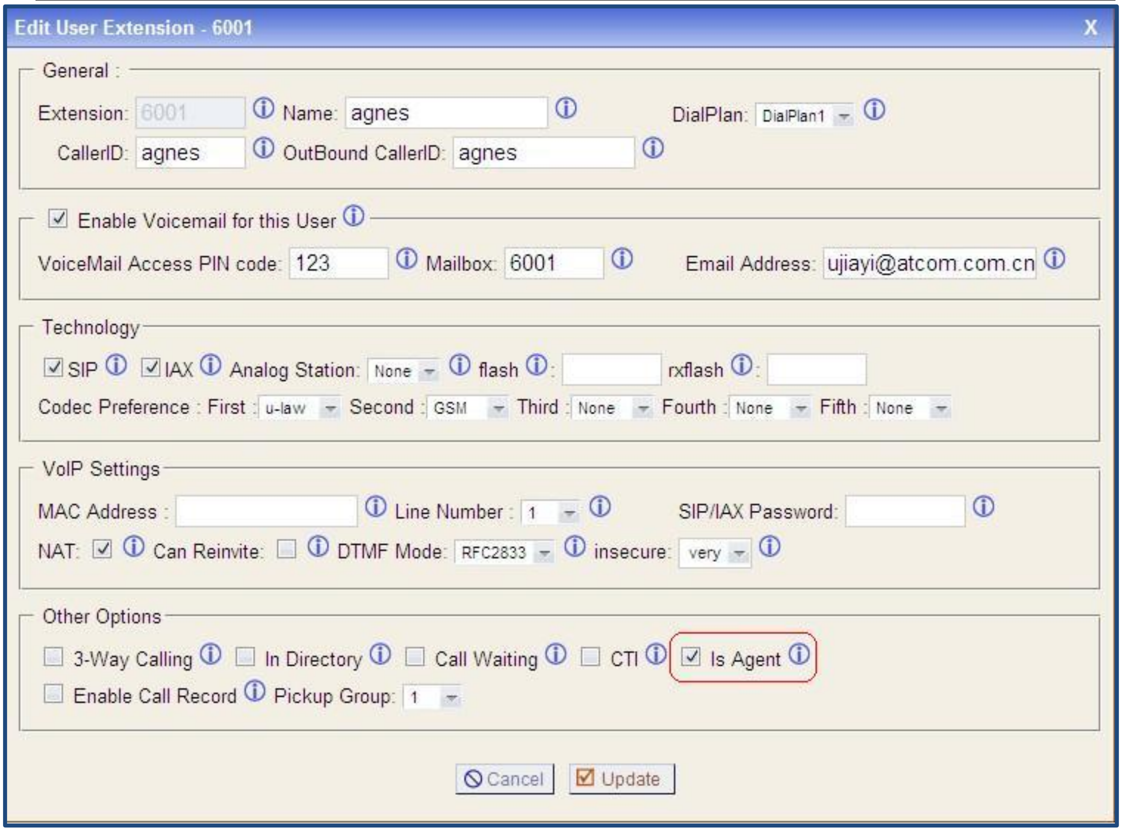
Define Ring groups to dial more than one extension simultaneously, or to ring more than one phone sequentially. This feature may also be called Hunt groups. You can click **Ring Groups→New Ring Group**, then you can configure it like the following screenshots. Of course 6001, 6002 have registered. 6008 have registered. Then 6008 dial 6400, you can hear 6001,6002 are ringing simultaneously. If you want the users are ringing sequentially, you can configure the strategy as Ring in Order.


## 5.9 Agents

When you need the function of Agents , you need complete the following two steps .

### 5.9.1 Create Users as Agents

You can create users like 5.1.2, but hook on the option of " Is Agent" like the following screenshots: please pay attention to the red ellipse frame.

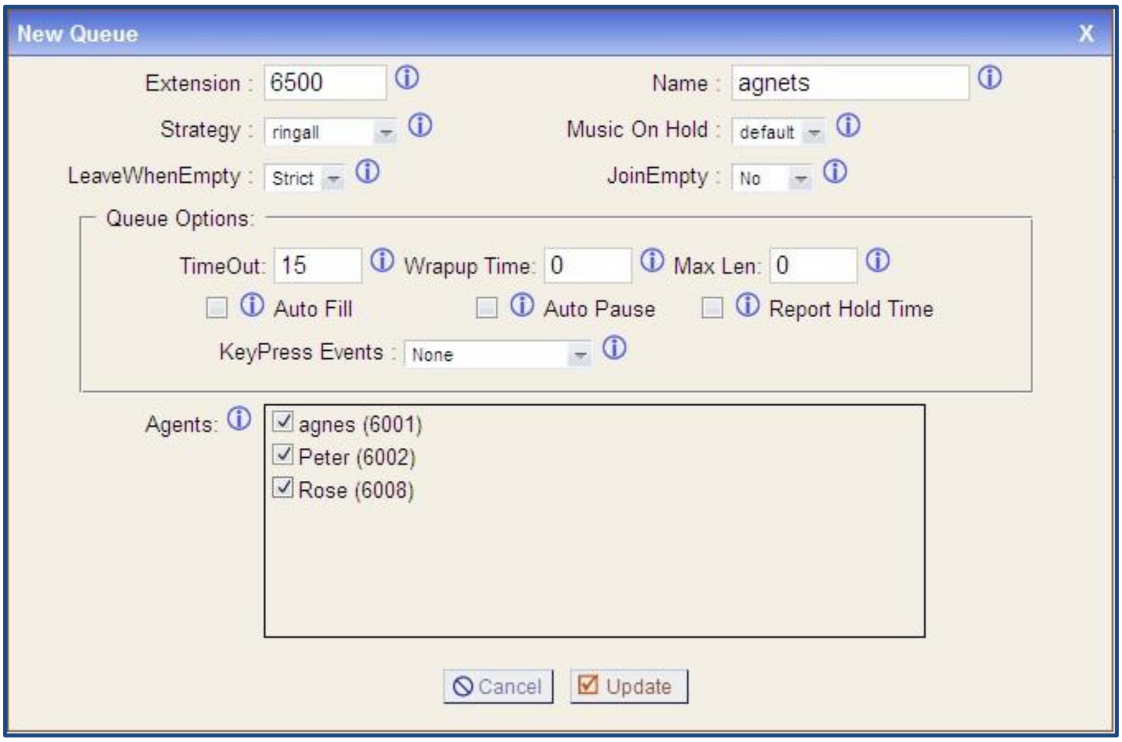



Like this I have also created 6002, 6008.Then you must click **System Status**, then you can see the following screenshots:


Click the button of "Login" so that all the Agents have logined . Then refresh the web, you can see the page that all the agents have logined like the following screenshots:


## 5.9.2 Create a Call Queue

Please click **Call Queues → Create New Queue**, then you can configure the options like this screenshots:



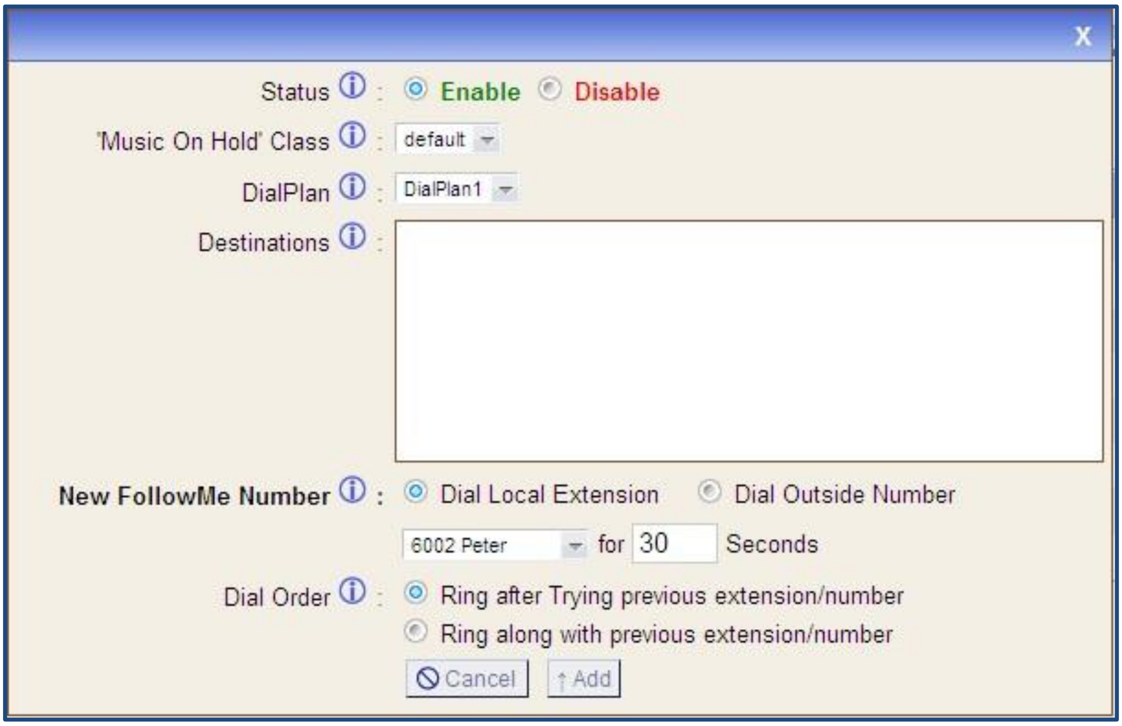

Then 6008(have registered) can call 6500, then 6001, 6002 are ringing together. Of course , if you want to configure it in detail , you can refer to 4.9.

## 5.10 Follow Me

Here 6001, 6002, 6005 have already registered and they can communicate with each other. First please click **Follow Me →Edit** on 6001, you can configure the options like the following screenshots:


Then click the button of "Add FollowMe Number", you can see the screen like the following, you can configure it like this, here I selected the Dial Local Extension number is 6005. Then click on

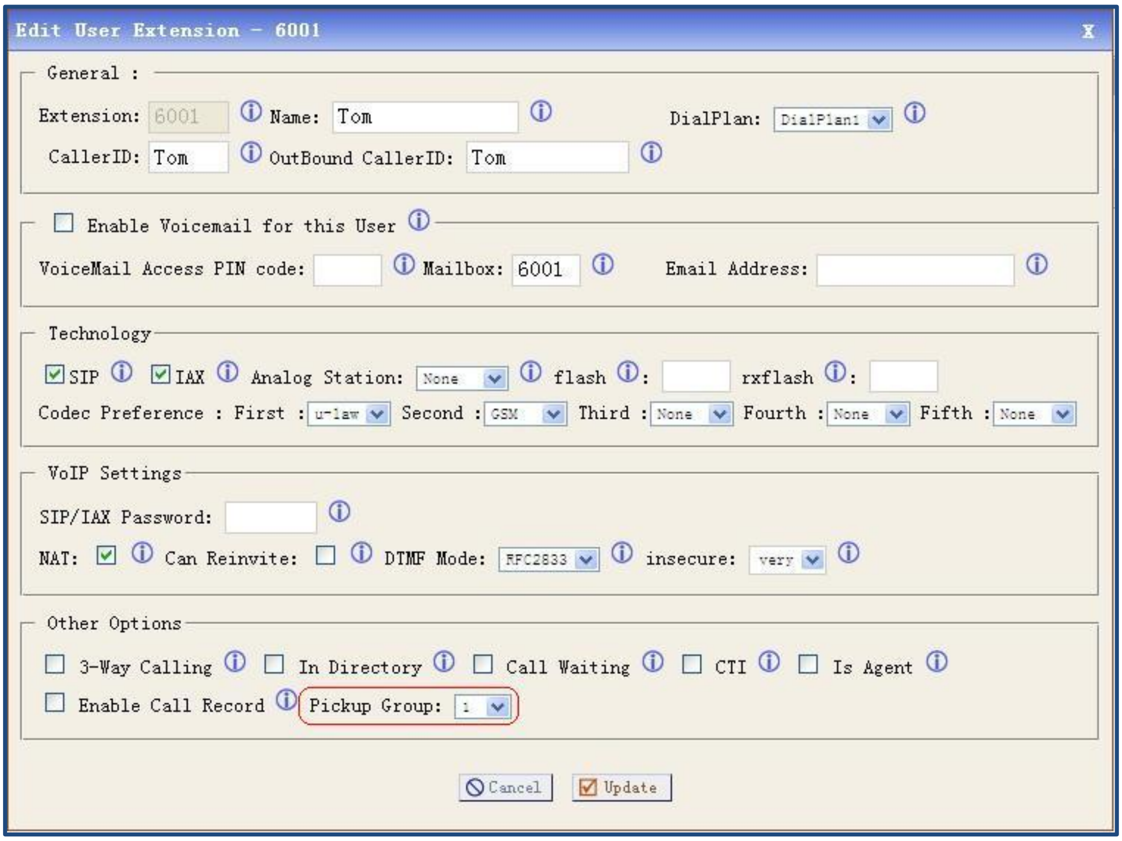
Add button, then please click on Update button, and click on Apply Changes button in up right corner of the main page.




Now when 6008 dial 6001, but nobody pick up it, after 30 seconds 6002 is ringing.

## 5.11 Group Call Pickup

This allows you to collect a call from any ringing phone that is in the same pickup group as you. There are two kinds of methods, one is that the phone itself has the function of pickup. The other is that we can configure it in the GUI of IP-2G4A. You can create users like the following:




please pay attention to the red ellipse frame, all the users must in the same group. Here I have created 6001, 6002 which are both in the group 1. I have also created 6008, it can be in group1, also can not. Then 6008 dial 6001, but we do not answer it , at the same time 6002 dial *8 then 6002 can connect with 6008. Now we completed the Group Call pickup function.

# Acronyms

**VoIP:** Voice over Internet Protocol

**FXO:** Foreign eXchange Office interface is the port that receives the analog line.

**FXS:** Foreign eXchange Subscriber interface is the port that actually delivers the analog line to the subscriber.

**SIP:** Session Initiation Protocol, SIP is a signalling protocol used for establishing sessions in an IP network.

**IAX:** Inter-Asterisk Exchange Protocol, is a communications protocol for setting up interactive user sessions. IAX is similar to SIP.

**RTP:** Real-Time Transport Protocol, RTP is used to encapsulate VoIP data packets inside UDP packets. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services.

**UDP:** User Datagram Protocol, UDP is a communications protocol that offers a limited amount of service when messages are exchanged between computers in a network that uses the Internet Protocol (IP).

**TCP:** Transmission Control Protocol , TCP is a set of rules (protocol) used along with the Internet Protocol (IP) to send data in the form of message units between computers over the Internet.

SMTP: Simple Mail Transfer Protocol, SMTP is the de facto standard for electronic mail transport across the Internet.

**TOS:** Terms of service, the "ToS" or "TOS" are rules by which one must agree to abide by in order to use a service. Unless in violation of consumer protection laws, such terms are usually legally binding.

**DTMF:** Dual-tone multi-frequency, DTMF signaling is used for telephone signaling over the line in the voice-frequency band to the call switching center. The version of DTMF used for telephone tone dialing is known by the trademarked term Touch-Tone, and is standardised by ITU-T Recommendation Q.23. Other multi-frequency systems are used for signaling internal to the telephone network.

**DHCP:** Dynamic Host Configuration Protocol, DHCP is an auto configuration protocol used on IP networks. DHCP allows a computer to be configured automatically, eliminating the need for intervention by a network administrator. It also provides a central database for keeping track of computers that have been connected to the network. This prevents two computers from accidentally being configured with the same IP address.

**NTP:** Network Time Procotol, NTP is a protocol for synchronizing the clocks of computer systems over packet-switched, variable-latency data networks. It is designed particularly to resist the effects of variable latency by using a jitter buffer.

**Vlan:** Virtual Local Area Network, is a group of hosts with a common set of requirements that communicate as if they were attached to the same broadcast domain, regardless of their physical location. A VLAN has the same attributes as a physical LAN, but it allows for end stations to be grouped together even if they are not located on the same network switch. Network reconfiguration can be done through software instead of physically relocating devices.

**HTTP:** Hypertext Transfer Protocol, The HTTP is a networking protocol for distributed, collaborative, hypermedia information systems. HTTP is the foundation of data communication for the World Wide Web. HTTP functions as a request-response protocol in the client-server computing model. TFTP: Trivial File Transfer Protocol, TFTP is a file transfer protocol, with the functionality of a very basic form of File Transfer Protocol (FTP). TFTP could be implemented using a very small amount of memory. It was therefore useful for booting computers such as routers which did not have any data storage devices. It is still used to transfer small amounts of data between hosts on a network, such as IP phone firmware or operating system images when a remote X Window System terminal or any other thin client boots from a network host or server.

**DNS:** Domain Name System, The DNS is a distributed hierarchical naming system for computers, services, or any resource connected to the Internet or a private network. It associates various information with domain names assigned to each of the participants. Most importantly, it translates domain names meaningful to humans into the numerical (binary) identifiers associated with networking equipment for the purpose of locating and addressing these devices worldwide.

**MAC:** Media Access Control address, The MAC is a unique identifier assigned to network adapters or network interface cards (NICs) usually by the manufacturer for identification. If assigned by the manufacturer, a MAC address usually encodes the manufacturer's registered identification number.

**IPv4:** Internet Protocol version 4, The IPv4 is the fourth revision in the development of the Internet Protocol (IP) and it is the first version of the protocol to be widely deployed.

**NAT:** Network Address Translation

**DTMF:** Dual Tone Multi Frequency

**GSM:** Global System for Mobile Communications

# Glossary

**Zaptel:** Zaptel refers to Jim Dixon's open computer telephony hardware driver API. Zaptel drivers were first released for BSD and Jim's Tormenta series of DIY T1 interface cards. Digium later produced interface cards from Jim's designs and improved the Zaptel drivers on the Linux platform. Digium then added further drivers also following the Zaptel API for other telephony hardware.

**Asterisk:** Asterisk is a software implementation of a telephone private branch exchange (PBX) originally created in 1999 by Mark Spencer of Digium. Like any PBX, it allows attached telephones to make calls to one another, and to connect to other telephone services including the public switched telephone network (PSTN) and Voice over Internet Protocol (VoIP) services.

**Voice Codec:**

G.711 is a high bit rate (64 Kbps) ITU standard codec. It is the native language of the modern digital telephone network. There are two versions: A-law and U-law.

G.711 A-law is indigenous to the E1 standard used in the rest of the world. G.711 U-law is indigenous to the T1 standard used in North America and Japan**.** The difference is in the method the analog signal being sampled. In both schemes, the signal is not sampled linearly, but in a logarithmic fashion. A-law provides more dynamic range as opposed to U-law. The result is a less 'fuzzy' sound as sampling artifacts are better supressed.

**Pick up:** the ability to pull a ringing call to the phone you are currently on.

There are two main types:-

**a.**Group call pickup, this allows you to collect a call from any ringing phone that is in the same pickup group as you, if there were more than one phone ringing then you would have no control over which call you collected.

**b.**Directed pickup, this allows you to pickup a call at a specific extension, maybe you're in another office and you hear a phone ringing and wonder if it's yours. You dial the pickup number and your extension, and the call will only transfer if it is your extension.

Group call pickup is typically invoked by dialing *8# or *8 from another phone in the call pickup group.

**Syslog:** Syslog is a standard for logging program messages. It allows separation of the software that generates messages from the system that stores them and the software that reports and analyzes them. It also provides devices, which would otherwise be unable to communicate, a means to notify administrators of problems or performance.

Time Zone: A Time Zone is a region on Earth, more or less bounded by lines of longitude, that has a uniform, legally mandated standard time, usually referred to as the local time.

# Reference

http://atcom.cn/download.html
http://www.asteriskguru.com/
http://www.openippbx.org/index.php?title=Main_Page
http://www.atcom.cn/