IP-PBX Quick Start Guide

Introduce

This is the quick start guide for ATCOM IP-PBX series. It includes the procedure for how to set up the IP-PBX as a basic IP-PBX system.

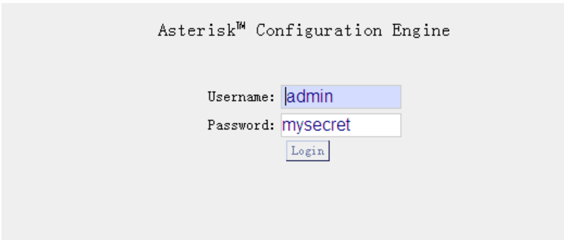
Configure and set up the IP-PBX

You need another computer to access and configure the IP-PBX. There are several ways to access the IPXX series products. Different ways has different usage. The ***web/SSH*** accesses are base on network connection which you need internet connection. And the ***console port access*** is via the RS232 console cable which allows you to access the devices when you fail to access to it via network.

➢ **Web access**

It is the most common way to access the IP-PBX. Most settings can be done through the web interface. Connect the IP-PBX's WAN port to your switch. Simply put the device's WAN port IP address (the default IP address of IP-PBX is 192.168.1.100) in your web browser (better use Firefox, there is compatible issue with IE) and enter the username and password to access the device.

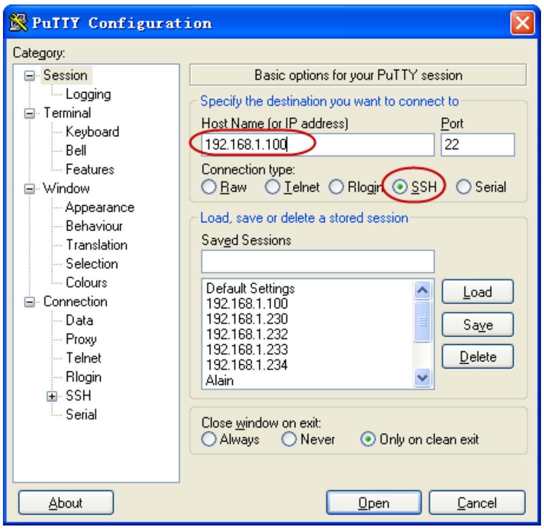
The web access username/password is: **admin/mysecret**



➢ **SSH access**

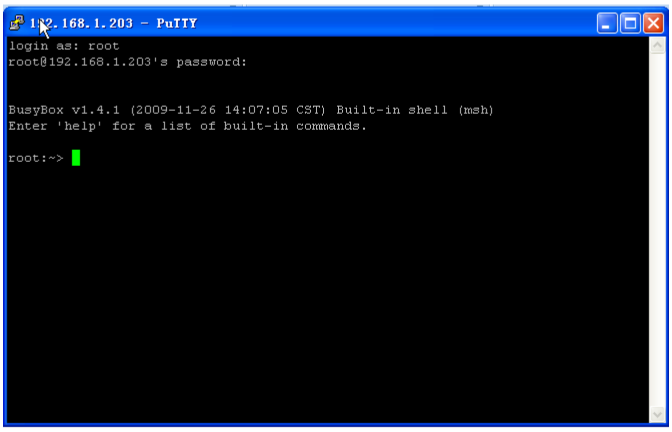
Connect the IP-PBX's WAN port to your switch. The default ip of the WAN port is 192.168.1.100. SSH is the advance way to access the device, you can use ***putty*** software to access the device. In the SSH access, you can access the Linux OS and do more advance setting and debug.

The SSH user/password is: **root/12xerXes06**



Note: when you enter the password in putty, the password won't show the password in shadow or clear text. It just show nothing.

This is what you can see from putty access. Here you can enter the Linux command and do advance configure.

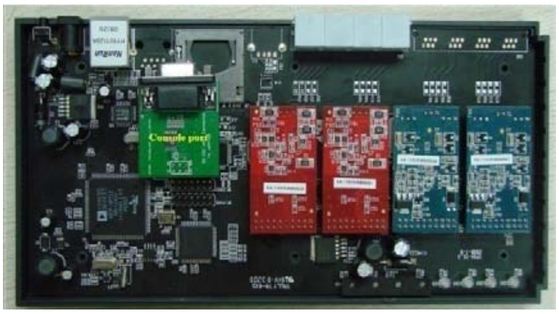


- **Console access**

The console access is very useful when you can't access to the device from your network.

There is a console port module of IP-PBX for console access. You need a RS232 console cable to connect to the IP-PBX, below are the connections pictures for console access.

IP04:



Below is the console port setting to access the IPXX

Running the Hyper Terminal, Putty or Minicom in your computer to connect the IP-PBX, the setting of the console port should be:

Bits per second to 115200;

Data bits : 8

Parity: None

Stop bits: 1

Flow control: None

How to change the IP address

The default IP the IP-PBX is 192.168.1.100. Your network may have a different IP range such as 192.168.10.xx. In this case, you will not be able to configure the IP-PBX if you put it directly into your LAN. You can change the ip address of IP-PBX to make it in the same network segment as your LAN.

Steps:

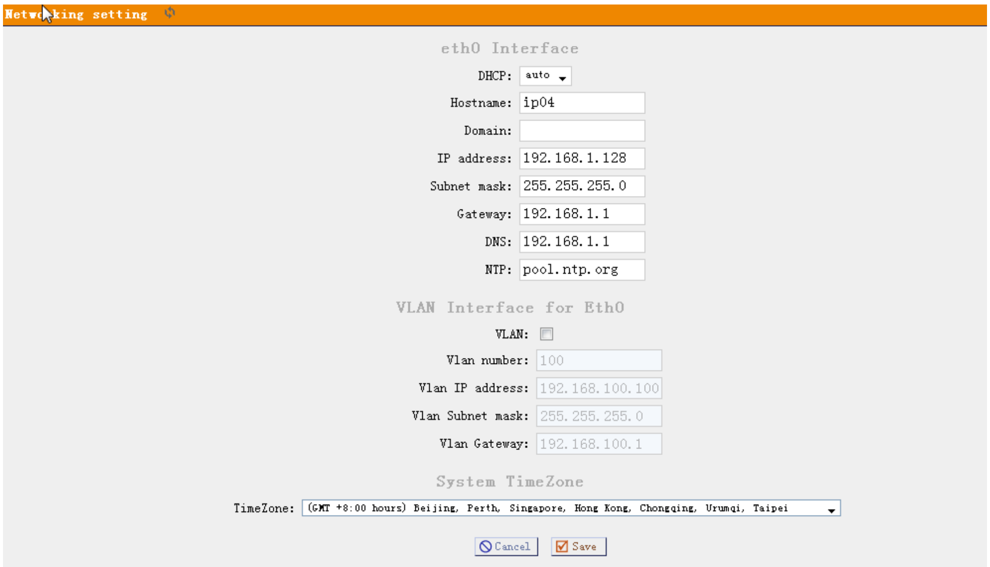
1) Connect the IP-PBX's WAN port to your PC.

2) Use putty to access to the IP-PBX via SSH.

3) Run below Linux command to change the IP-PBX's IP to fit your network.
   root:~> ifconfig eth0 IP_Address

replace IP_Address with a valid IP in your LAN (if your LAN has the ip segment 192.168.10.xx, then you will properly put 192.168.10.100 here)

4) After Step 3) is done, the IP-PBX will have a temporary IP but it will lose change after reboot. To change its IP permanently you need to change the IP from the Web GUI. The page to configure the IP address is:
   Go to **Options--> Advance Options-->Show Advance Options**. Then you can find the network settings in the main menu.



There are three types for network settings:

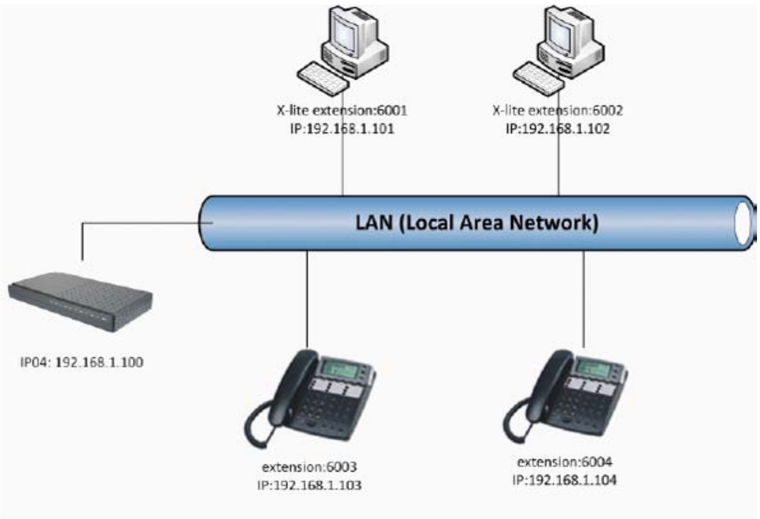
a) **DHCP: Yes** IP04 will obtain the dynamic IP from your router.
b) **DHCP: auto** IP04 will use the static IP specified below and ping the default gateway, when there is no response from the default gateway. The IP04 will switch to dynamic obtain the IP from your router.
c) **DHCP: No** IP04 will use the static IP specified below.

Set up extensions and make internal calls

Create extensions:

Making internal calls are the base requirement for a telephony system. You can create extensions in the IP-PBX and use the IP phone and softphones to register to the IP-PBX and make calls.

System Setup



Internal calls in the same office

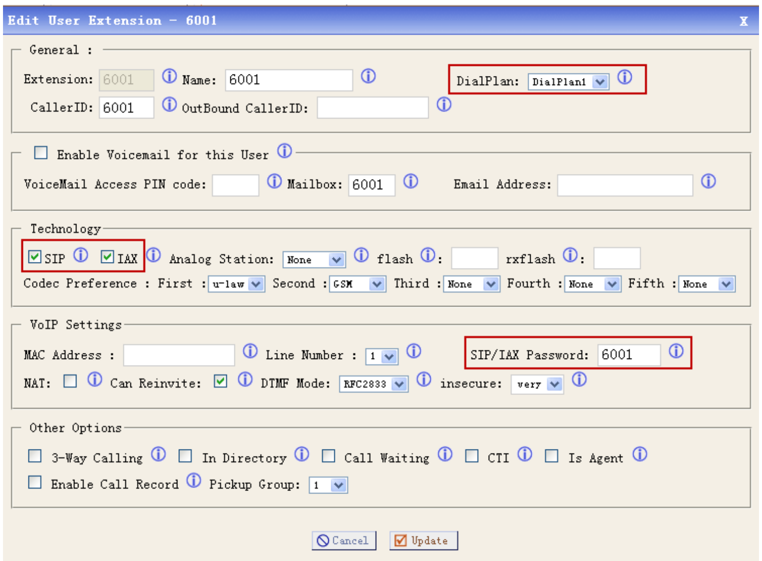
At the beginning, we need to add some extensions to make internal calls. Each extension acts as an internal number. There are three types of extensions we can add: SIP, IAX2 and ZAP.

Before set up the extensions, we need to go to the **Options --> General Preferences** to set the user extensions range. The default user extensions range is from 6001~6299.

Steps:

1 Go to page **Dial Plan-->Create New Dialplan** to create a default Dial Plan.

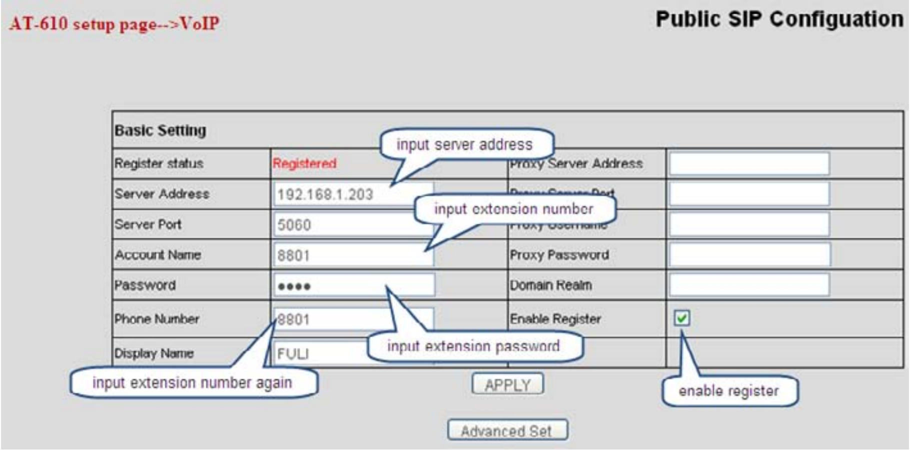
2 Go to page **Users-->Create New User** to create the extensions: 6001



After the extensions are created, you can use SIP terminal (IP phones/softphone/ATA etc.) to register to the extension to make calls.

Example: Use AT-610 to register to the IP-XX and make internal call

1) Use the ***info*** key in the AT-610 to get the ip address of AT-610.
2) Put the IP address you get in your web browser to open the AT-610, the default password is **admin/admin**
3) Click the ***Voip*** page
4) Put the account information you get from IP-PBX in the AT-610.



The field you should put is:

Server Address: put the ip of your IP-08

Account name: the extension number of your account in IP08

Password: the password of your extension number.

Click Enable register.

Apply the setting and you should see the AT-610 register status change to Registered.

Configure several AT-610 via the same method and every of them have an extension number. They can call each other via the extension number.

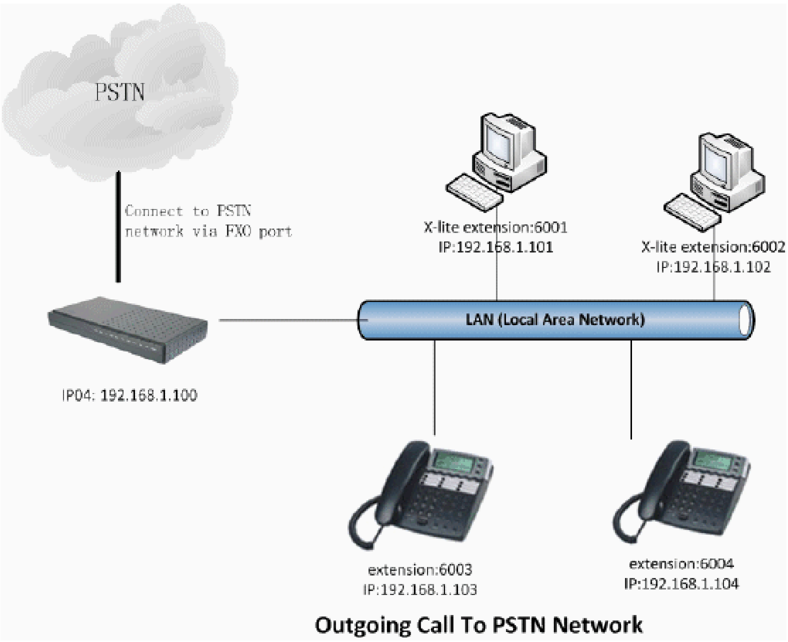
How to make calls via the FXO port

Make outbound calls to PSTN

There are many kinds of trunk you can use to make outgoing calls. It includes: Analog FXO trunk, Digital E1/T1/BRI Trunk, SIP trunk, IAX trunk etc. Here we use the FXO port to make outgoing calls.

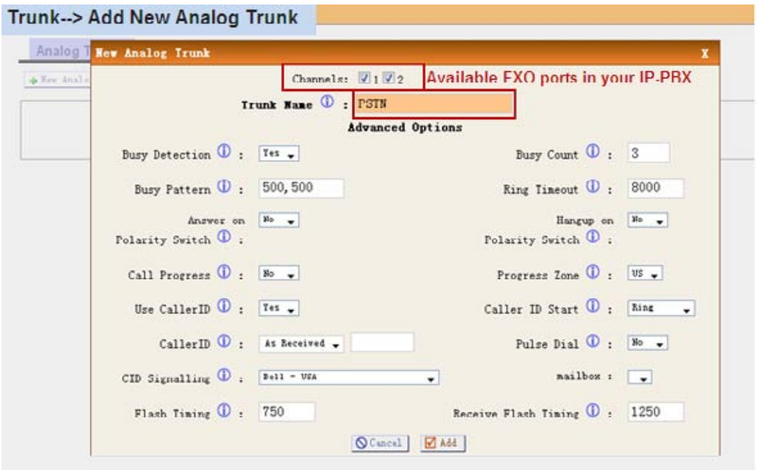
Analog/FXO trunk

For the IP01/04/08, you can install FXO module and use the FXO port to make outgoing call via your local PSTN line. The set up is as per below:



Step 1: Create FXO trunk

Go to page **Trunks--> Add New Analog Trunk**



Note: The port1 and port2 of IP-PBX are slotted with FXO modules. Always click "Apply Changes" in the right top corner when you do some changes.

Step 2: Create Outgoing Calling Rules

Go to page **Outgoing Calling Rules.**

The calling rule is the handler for every call you make. The number matched in the calling rule will go to corresponding trunk set in the calling rule.

For example, in above calling rule, the pattern **_9.** and **strip 1 digits** means all calls start with 9 will be cut the first digit and sent out via the PSTN trunk(Port1 and Port2). In this calling rule, if you dial 983018049, the IP-PBX will send the number 83018049 to port1 or port2 and calling out.

Step 3: Add New DialPlan

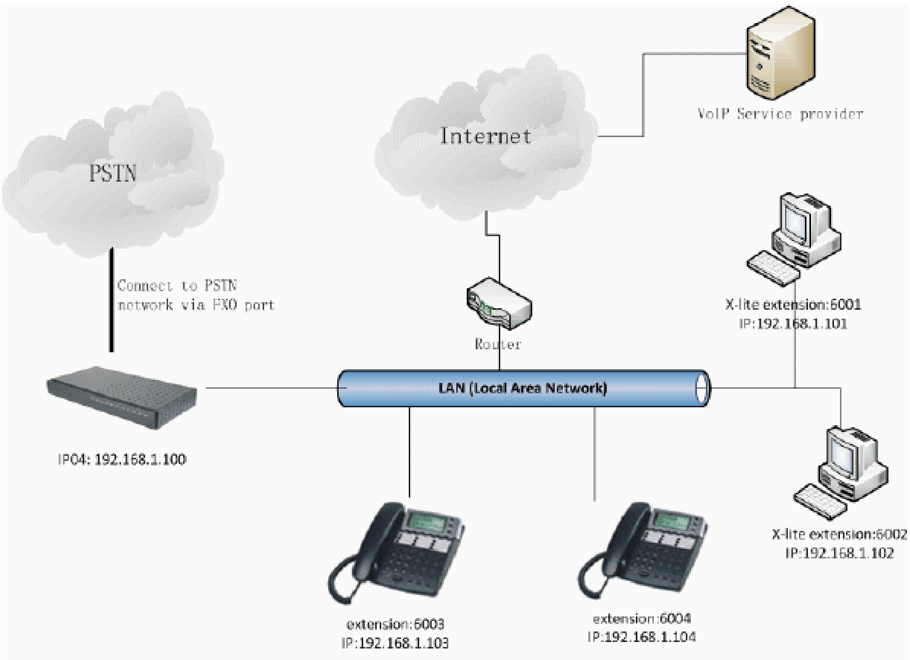
Go to page **Dial Plans**--> **Create New Dial Plans**

The Dial-plan is a set of calling rules. Every user will have its dial-plan in their user-setting. Users can use same dial-plan or different dial-plan.

When users are assigned a dial-plan, all the number this user dial will follow the calling rules included in the dial-plan.

How to make VoIP outgoing call?

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



Outgoing Calls Network

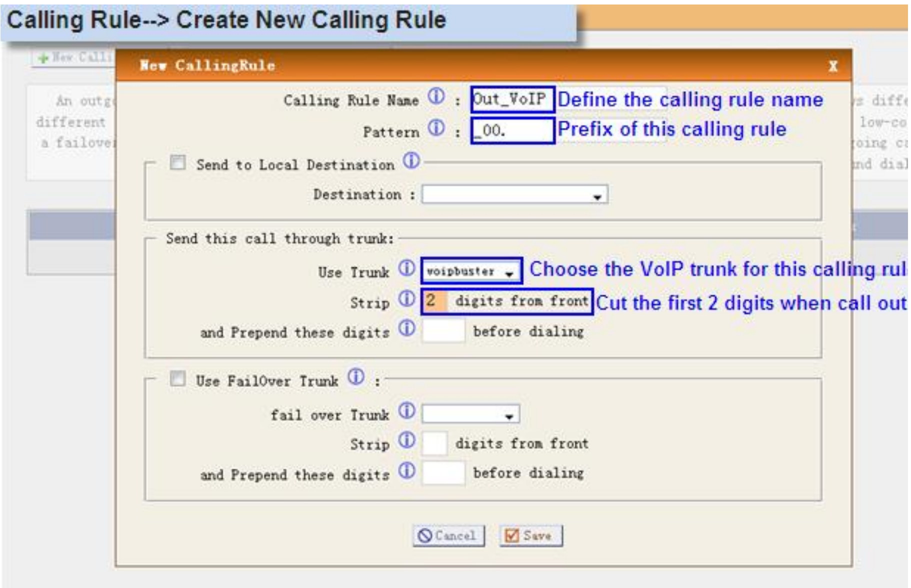
Step 1: Add Voip trunks

Go to page Trunks--> Voip Trunks--> Add New Sip trunks

There are many service provider provide voip calls. Any provider use standard Sip or IAX2 protocol should work with the IP-PBX

Step 2: Add voip calling rule

Go to page Outgoing Calling Rules.



All calls start with 00 will be sent out via our voip service provider.

Step 3: Add this new calling rule to the dial plan1

All extensions which use dialplan1 are able to use the voipbuster service now

IVR—Auto Attendant

IVR stands for Interactive Voice Response. You can set up IVR as the auto-attendant for the incoming calls.

Procedure to set up the IVR

Step1: Customize your voice prompt

You can record the IVR welcome voice from an extension. In the page **Voice Menu Prompt--> Record a New Voice Prompt**:

- **Name:** Specify the name of this voice prompt.
- **Extension:** Enter the extension here and then click record, the extension will ring. Answer the call and you will hear "Please leave the message after the tone, when down, hung up or press the pound key..du...". Then please say the message you want to record. The message will be then record as a gsm format file and you can use it in the IVR.

Step2: Set up the IVR voice menu

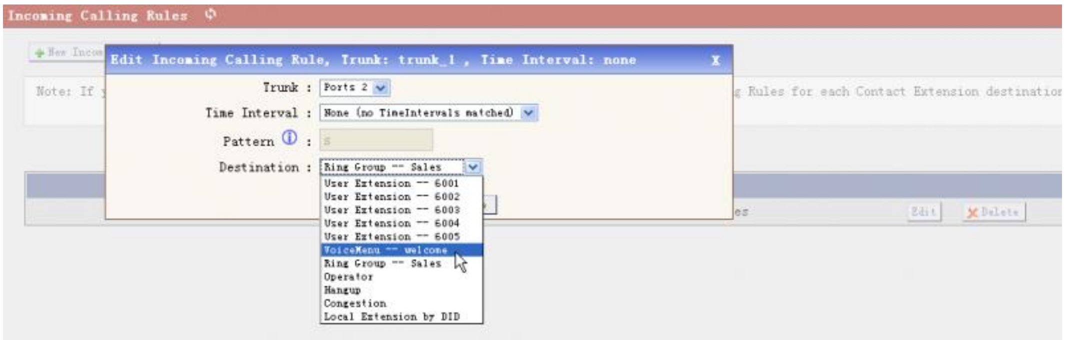
Go to the **Voice Menus** Page and create a new Voice Menu.

- **Name**: Name of this Voice Menu
- **Extension**: Extension of this Voice Menu, the other extension can reach this IVR by dialing this extension.
- **Allow Dialing Other Extensions**: Allow user to dial other extensions when listening this IVR.
- **Actions**: A sequence of actions performed when a call enters the menu.
- **Add a new step**: Add additional steps performed during the menu.
- **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. If you want to set up multiply levels IVR, you can choose the voice menu here for the key press.

Step3: Point your incoming calling route to the IVR

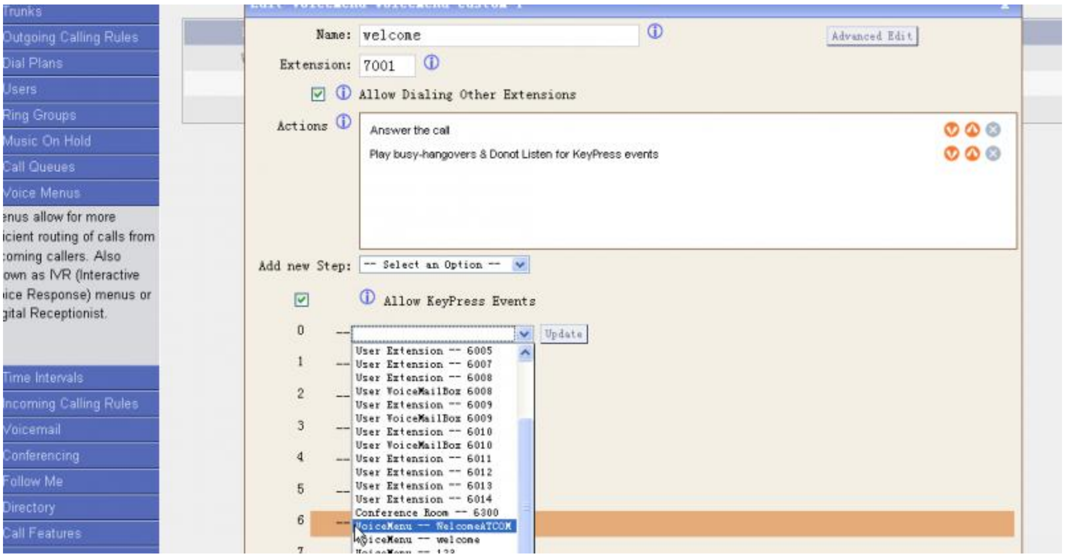
In the page ***Incoming Calling Rules*** you can define which Voice Menu the incoming calls should route to.



Advance option for IVR

Option1: More IVR Level

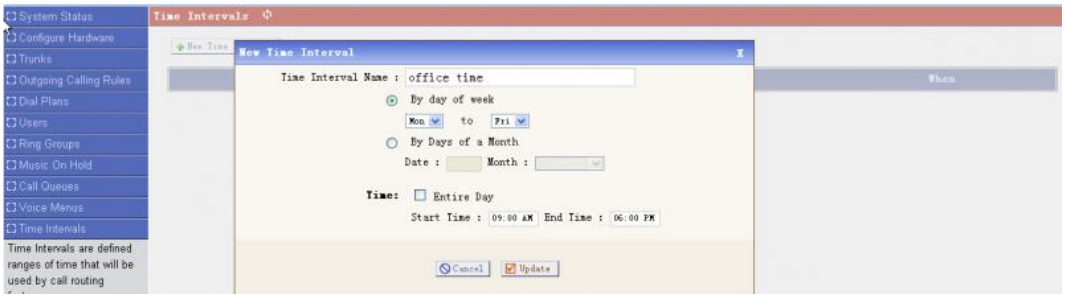
If you want to set up more than one IVR level, you can enable **allow key press event** as below, you can determine the key press event to reach another voice menu.



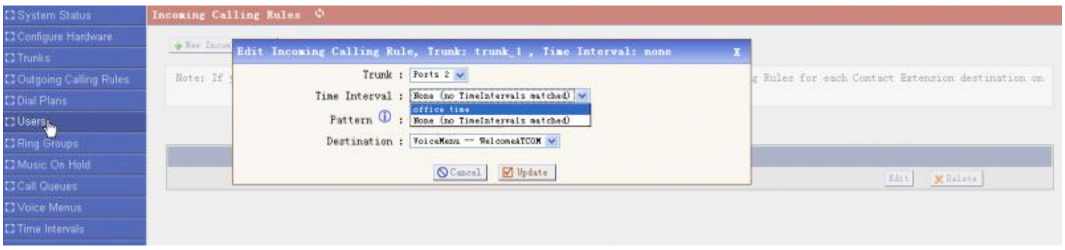
Option2: Set different IVR for different hours

You can special difference Voice menu for different hours, for example: a voice menu for working time and another menu for non-working time.

1/in the page **Time Intervals**, set up the time intervals for office time.



2/In the page **Incoming Calling Rule**, set up the rule with time Interval, then the incoming calls will be routed to the corresponding destination during these hours.



IVR Configure Example

Target: The IP-0X has four analog ports. Anyone who dials to these ports will hear a company welcome voice. The caller can dial the extension to reach the one he wants to call.

Step1: Record the welcome prompt

In the page **VoiceMenu prompts**:

- Click *Record a New Voice prompt*
- Enter the filename "welcome" and my extension "6001".
- Click *Record*
- Extension 6001 rings and answer the call then record the welcome prompt "Welcome to ATCOM, please dial the extensions".
- Hung up the call.

Step2: Create a new Voice Menu

In the page **Voice Menu**:

- *Create New VoiceMenu*
- Enter the VoiceMenu Name "WelcomeATCOM",Enter the extension "7001",Enable "Allow dialing other Extensions", Add a New step "Answer", Add a second New step "BackGround record/welcome",
- Save the Voice Menu.

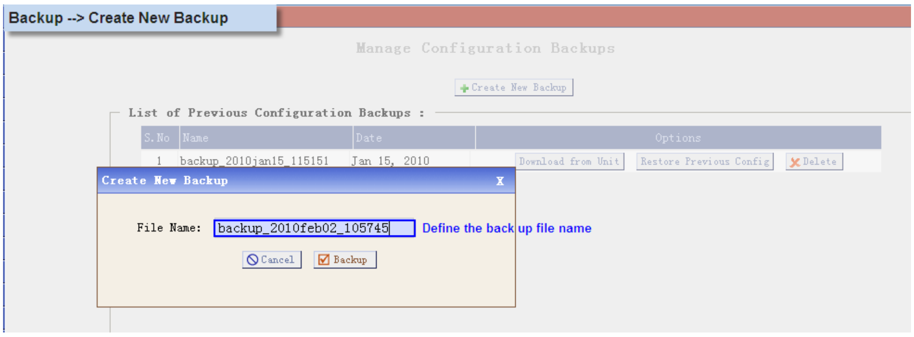
Step3: Point the incoming calling rule to voice menu

In the page **Incoming Calling Rule**:

- *Create A New Incoming Rule*
- Trunk select Port1,2,3,4. Time Interval choose "None". Pattern set to "S". and then set destination to "Voicemenu -- WelcomeATCOM"
- Save the Incoming Rule

Backup and restore configure

You can backup your configure by in the GUI→Backup



After the back up, you can see the back up in the back up file list. you can restore the configure in this page also. Note that the restore will only work after reboot.

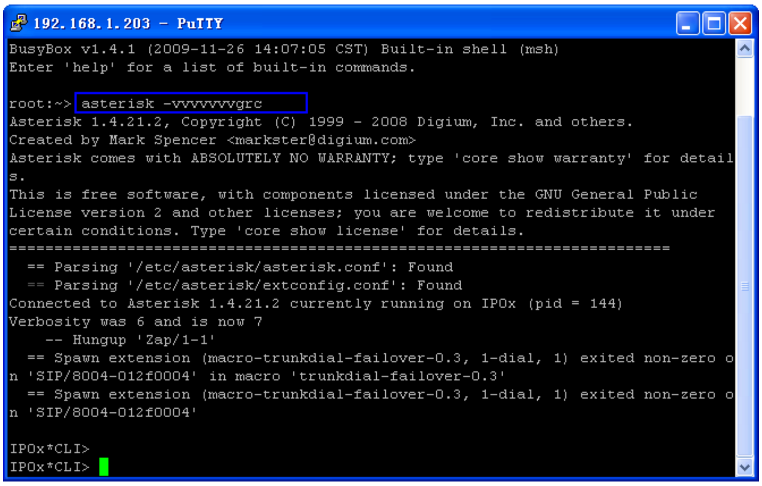
Debug the IP-PBX

If you have problem in making calls via the IP-PBX, you can debug it in the SSH and try to solve it yourself or give us the debug info to find the issue.

Debug the dial plan

Step1: Use the SSH access to connect to the IP-0X

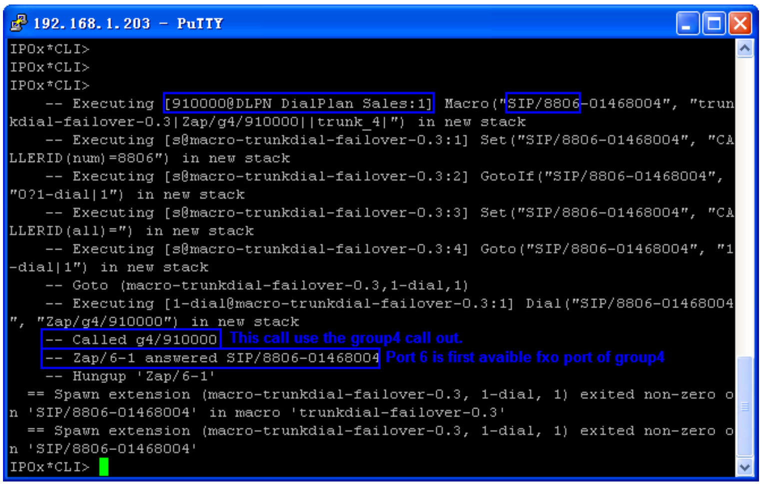
Step2: Run “asterisk –vvvvvvgrc” to connect to IP0x*CLI.



In the IP0x CLI, you can see how your calls process in the IP0x and try to sort out where the problem is.

For example:

Below is the call log when I use my extension 8806 to call a number 910000 via the FXO port.



If you have problem when making calls and can’t solve it yourself. Please copy above info to us and descript the problem you meet. We will help you to solve it in the soonest.