IP-01P User Manual

1 Introduce

This Article

This article is the user manual for the IP-01P Asterisk Appliance. It also includes the application notes for how to use ATCOM products to build a telephony system for small office. Through this article, we hope that users can build the IP telephony system via IP-01P Asterisk Appliance.

2 Configure the device via GUI

2.1 Access the GUI

The IP01P GUI is immigrated from Asterisk Now 2.0 version. The default IP for IP-01P is 192.168.1.100. Put the default ip in your web browser and it will redirect to the setting page of IP-01P, the default password for the web access is:

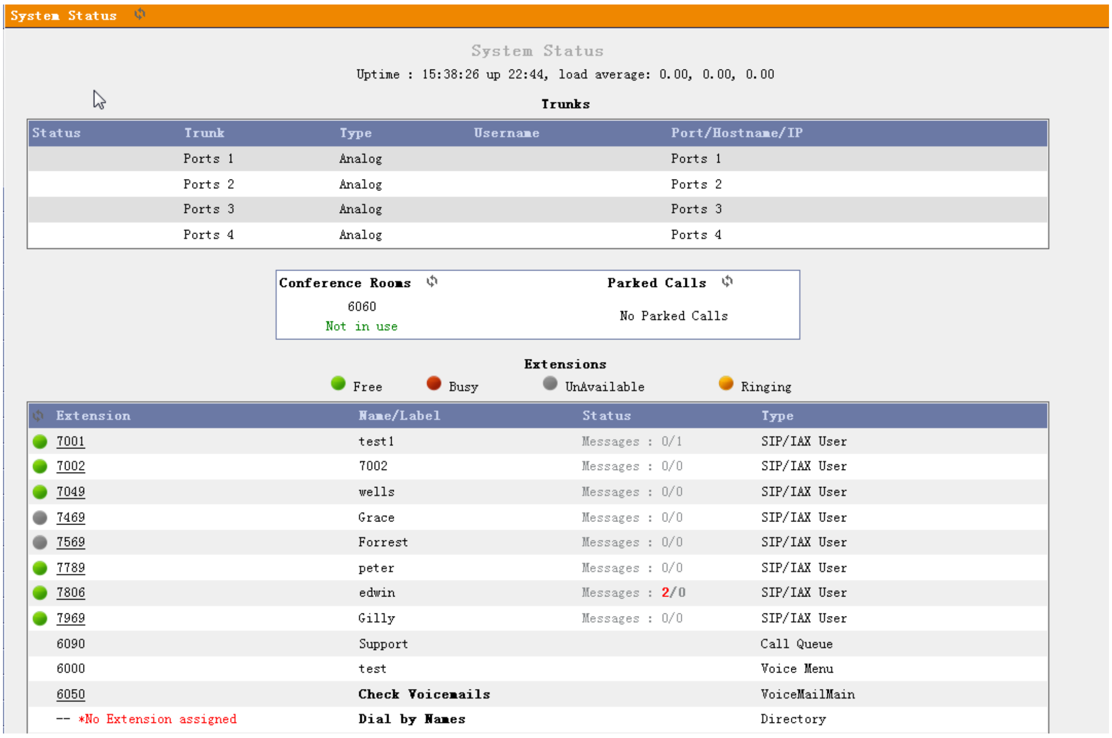
Username: admin
Password: mysecret

If you can't access the IP-01P, please check if you have connect the RJ45 cable to the WAN port and your computer is in the same network 192.168.1.xxx as the IP-01P.

Note: the recommend web browser of IP-01P is Firefox.

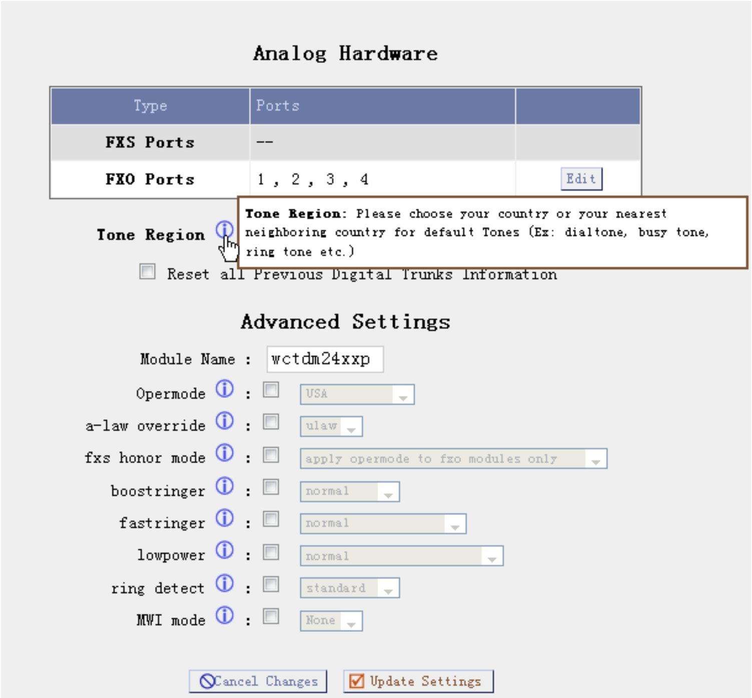
2.2 System Status

When you have entered the IP-01P setting page, the system status will be showed and you can see the system status as below:



2.3 Configure Hardware

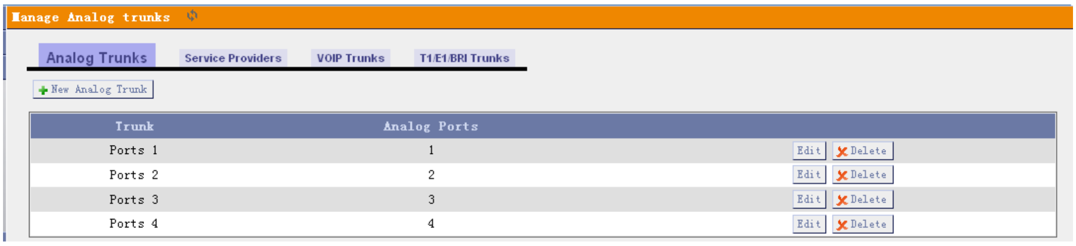
The Configure Hardware page lists the available telephony ports in your system. You can configure the hardware to comply with your local telephony environment.



Note: Hover on the (i) and you can see the comment of every settings.

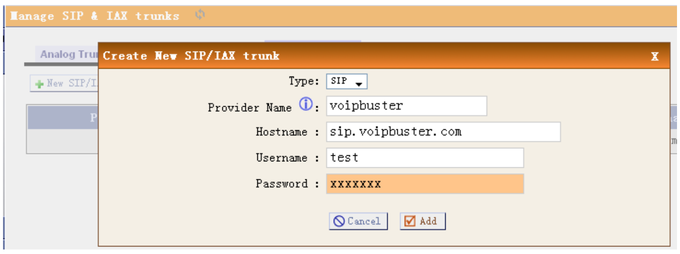
2.4 Trunks

Trunks are used to make outbound call to the real world. There are different trunks we can set here.



We have put the IP-01P with four FXO ports so there are four analog trunks in this setting page.

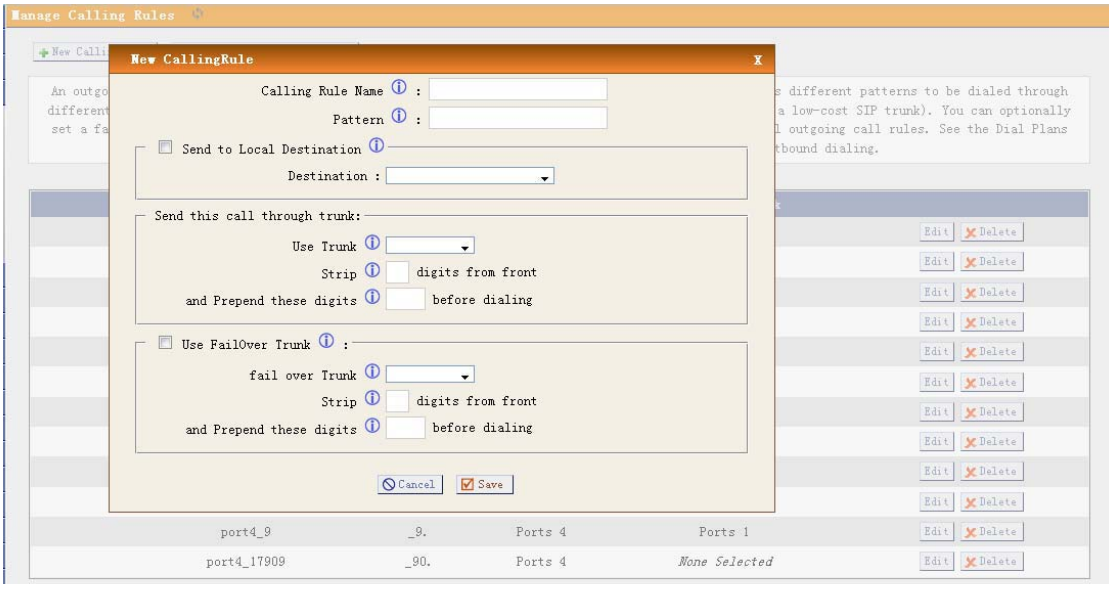
VoIP trunks (SIP&IAX2) are also available in the IPXX.



More info about how to set up the trunks, please refer the application notes.

2.5 Outgoing Calling Rules

Outgoing Calling Rules defines the calling permission sand the routing rules when making calls.



- <u>Calling Rule Name</u>: The name of your calling rule
- <u>Pattern</u>: Describe what numbers should use this rule:

  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 numbers, while _1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one.
- <u>Use/Trunk</u>: Describe which trunk should be used in this rule
- <u>Strip</u>: Define how many digits should be removed from the dialstring.

Two samples for calling rule setting:

1) Cut the first digit for all dialstring start with 9, and make outgoing call via port1:

   Calling Rule Name: Out_9

   Pattern: _9.

   Use/Trunk: Ports 1

   Strip: 1 digits from the front

   In this case, if you dial 983018806 in your extension, 83018806 will be sent via port1

2) Cut the first digit for all dialstring start with 0, and prepend 86 then dial via voipbuster

trunk.

Calling Rule Name: Out_voipbuster

Pattern: _0.

Use/Trunk: Voipbuster

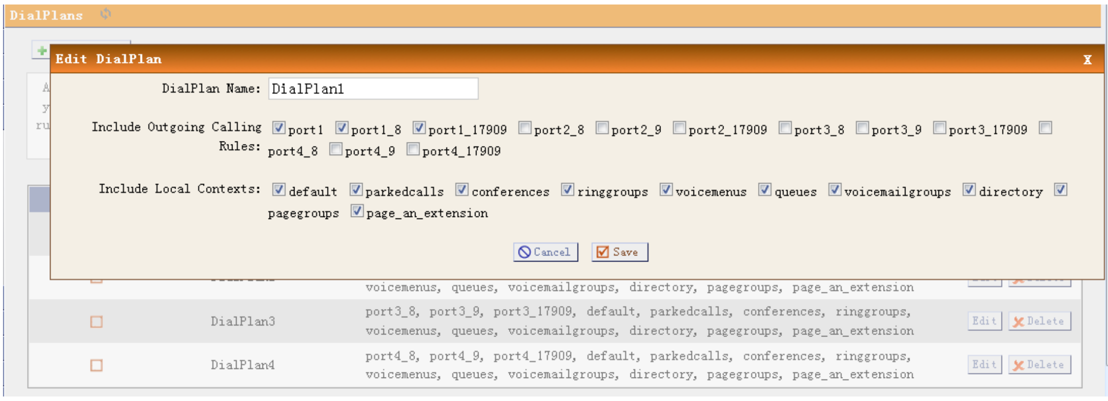
Strip: 1 digits from the front

And prepend these digits 86

In this case, if you dial 075583018806 in your extension, 8675583018806 will be sent via voipbuster trunk

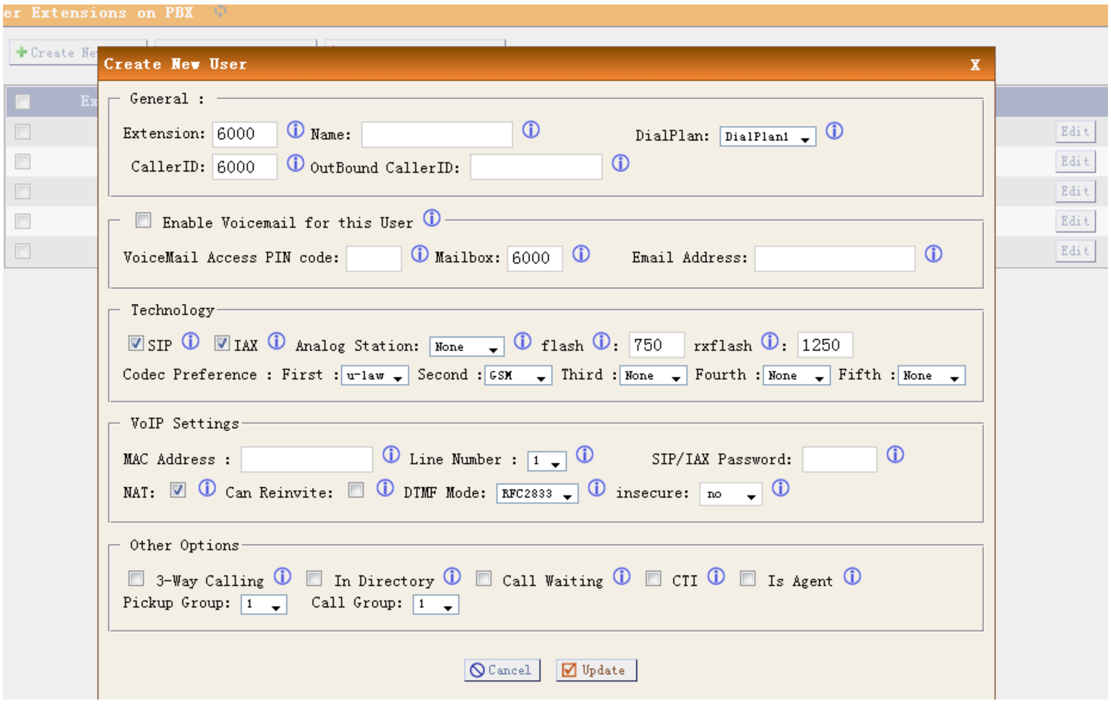
2.6 Dial Plans

A Dial Plan is a collection of Outgoing Call Rules. Dial Plans are assigned to Users to specify the dialing permissions they have. For example, you might have one Dial Plan for local calling that only permits users of that Dial Plan to dial local numbers, via the "local" outgoing calling rule. Another user may be permitted to dial long distance numbers, and so would have a Dial Plan that includes both the "local" and "longdistance" outgoing calling rules.



2.7 Users

This page allow administrator to create extensions for every user.



General:

Extension: The number you can dial to reach this user.

Name: CallerID name of the user.

CallerID: The CID string when you dial to other internal users.

Outbound CallerID: Specify the public CallerID for outbound calls, it is only available when your digital or voip provider support this feature.

VoiceMail:

Voicemail Access PIN code: The password of your voicemail box

Email address: The email address for the voicemail to email function.

Technology:

SIP: enable this option so the extension can be a SIP device.

IAX2: enable this option so the extension can be an IAX2 device.

Analog Station: If you have analog FXS ports in your IP-01P, you can select the port here for your extension.

flash/rxflash: flash parameter for the users.

Codec preference: specify the preference codec for the users.

Voip Setting:

MAC address: used for polycom phone provisioning.

Line Number: used for polycom phone provisioning.

Linekeys: used for polycom phone provisioning.

SIP/IAX2 password: user password for SIP/IAX2 registration.

NAT: enable this when you use the IP-01P in public network and the sip devices are in private network.

Can Reinvite: enable this and the IP-01P will try to negotiate the endpoints to route the media string directly (not through IP-01P). This can reduce the CPU load of the IP-01P and you will get better voice performance because the media string are sent directly from endpoint to endpoint.

DTMF mode: DTMF uses on conversation, the RFC2833 is the most common.

Insecure: method of authentication,

Other Options:

3-Way Calling: enable/disable 3-way calling

In directory: check this if the user is listed in the directory.

Call waiting: enable/disable Call waiting

CTI: Computer Telephony Integration, allows access to 3rd party applications over Asterisk Manager Interface.

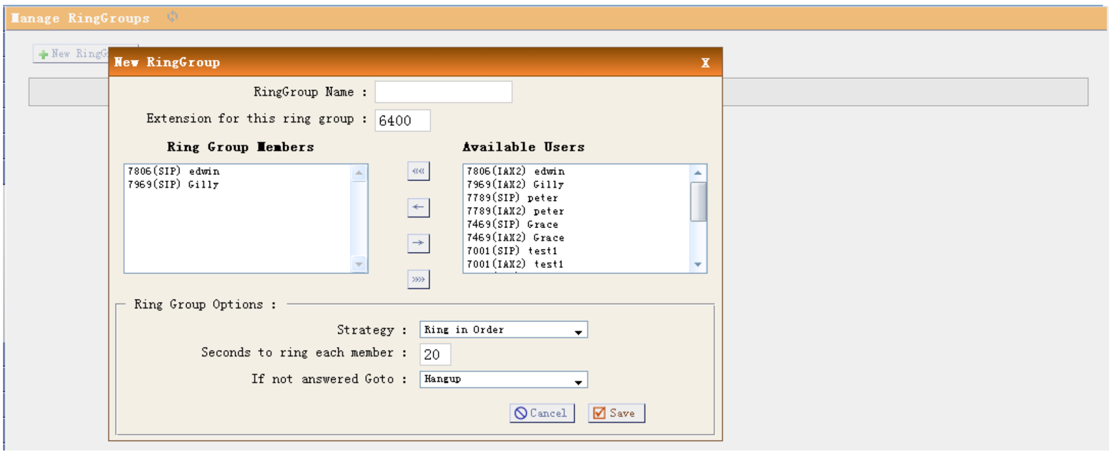
Is Agent: check this if the user is available in call queue.

Pick up Group: Specify the call pick up group.

Call Group: Specify the call group for the user.

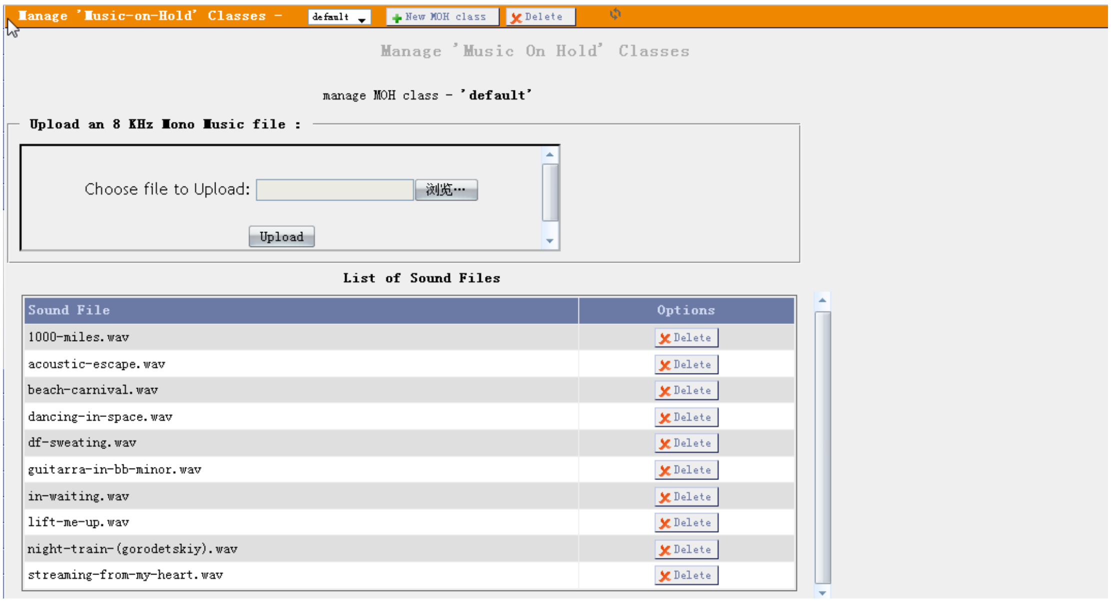
2.8 Ring Groups

Define the Ringroups to dial more than one extension simultaneously, or to ring more than one phone sequentially.



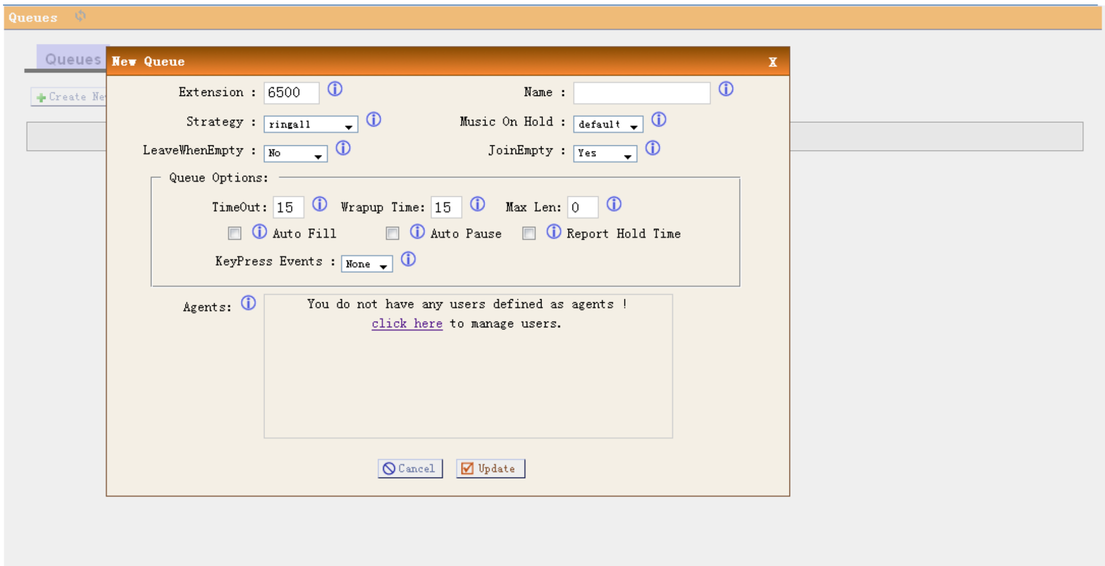
2.9 Music on Hold

Customize audio tracks for different queues, parked calls etc.



2.10 Call Queues

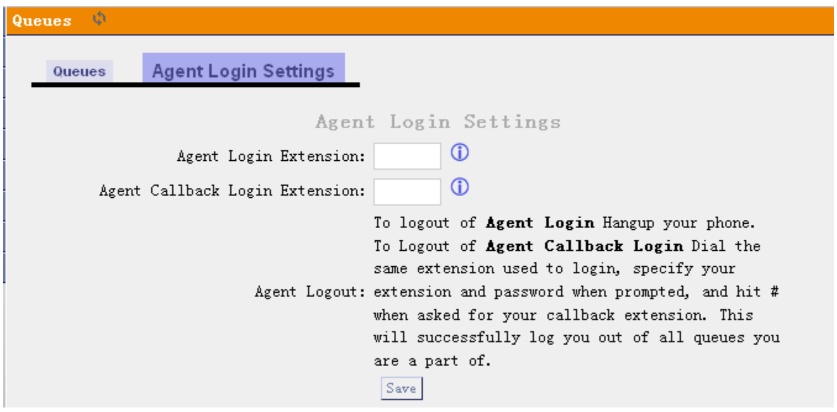
Call queues allow calls to be sequenced to one or more agents.



- Extension: This option defines the numbered extension that may be dialed to reach this Queue.
- Name: This option defines a name for this Queue, i.e. "Sales". 'Name' is a label to help you see this queue in the queue list.
- Strategy: This option sets the Ringing Strategy for this Queue. The options are
  RingAll: Ring All available Agents simultaneously until one answers.
  RoundRobin: Take turns ringing each available Agent.
  LeastRecent: Ring the Agent which was least recently called
  FewestCalls: Ring the Agent with the fewest completed calls.
  Random: Ring a Random Agent.
  RRmemory: RoundRobin with Memory, Remembers where it left off in the last ring pass.
- Agents: This selection shows all Users defined as Agents in their User conf. Checking a User here makes them a member of the current Queue.
- Music On Hold: Select the 'Music on Hold' Class for this Queue. 'Music on Hold' classes can be managed from the 'Music On Hold' panel.
- LeaveWhenEmpty: This option controls whether callers already on hold are forced out of a queue that has no agents. There are three options.
  Yes: Callers are forced out of a queue when no agents are logged in.
  No: Callers will remain in a queue with no agents.
  Strict: Callers are forced out of a queue with no agents logged in, or if all logged in agents are unavailable.
- Join Empty:This option controls whether callers can join a call queue that has no agents. There are three options,
  Yes: Callers can join a call queue with no agents or only unavailable agents
  No: Callers cannot join a queue with no agents
  Strict: Callers cannot join a queue with no agents or if all agents are unavailable.

Queue Options:

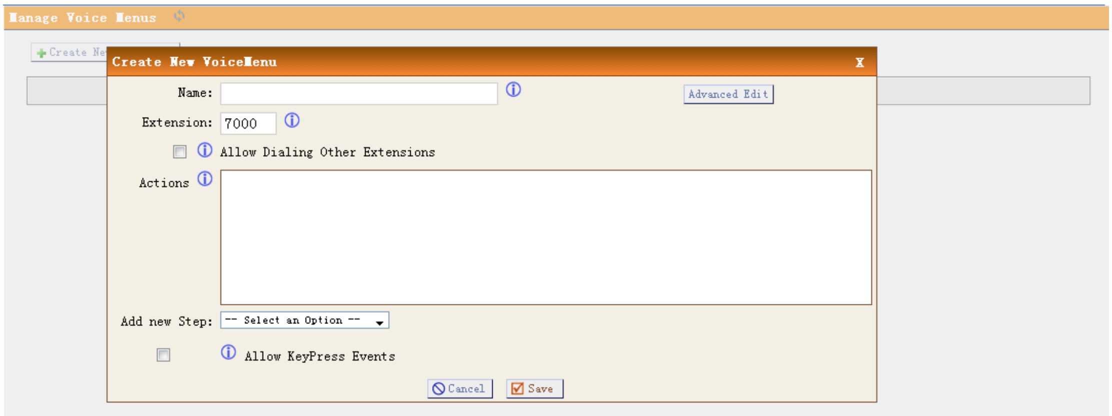
- ➢ Timeout: How many seconds an Agent's phone will ring before the Queue tries to ring the next Agent.
- ➢ Wrapup Time: How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 0, which is no delay.
- ➢ MaxLen: How many calls can 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.
- ➢ AutoFill: 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.
- ➢ AutoPause: 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.
- ➢ 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.
- ➢ KeyPress Events: If a caller presses a key while waiting in the queue, this setting selects which voice menu should process the key press.



- ➢ Agent Login Extension: Extension to be dialed for the Agents to Login to the Specific Queue. This is an extension that all the Agents can Call to Login to their specified Queues.
- ➢ Agent Callback Login Extension: Extension to be dialed for the Agents to Login to the Queues they are apart of. Same as Agent Login Extension, except you do not have to remain on the line.

2.11 Voice Menu (IVR)

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



- Name: Name of this Voice Menu
- Extension: If you want this Voicemenu to be accessible by dialing an extension, then enter that extension number.
- Dial other Extensions: Is the caller allowed to dial extensions other than the ones explicitly defined.
- 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.

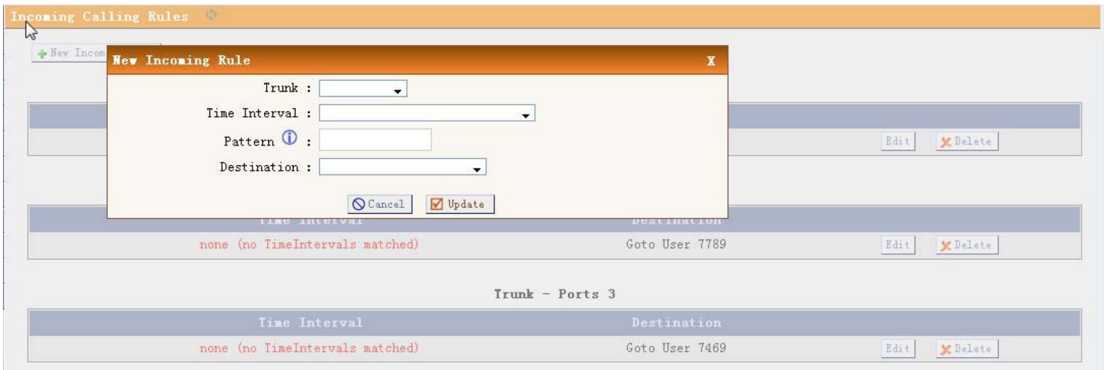
2.12 Time Intervals

Time Intervals are defined ranges of time that will be used by call routing features.

- Name: Name of the time Interval
- By day of week: Define the day range by week for time interval
- By days of a month: Define the day range by Month for time interval
- Time /Entire day: Define if the time interval is available for the whole day or only for the specified hours.

2.13 Incoming Calling Rules

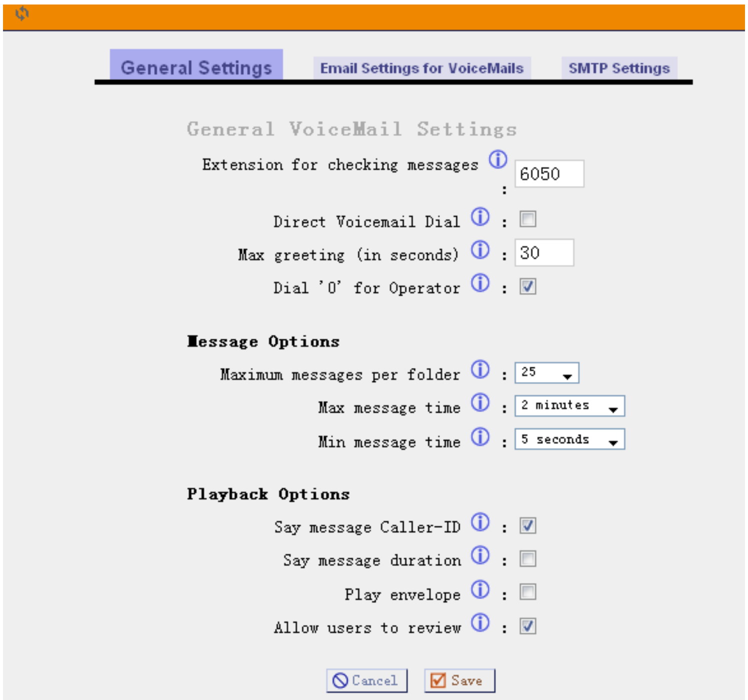
Create, modify, prioritize and delete incoming call rules for handling Incoming calls based on service provider and/or the number called based on Time Intervals.



- Trunk: Which service provide should use this trunk for its incoming calls.
- Time Interval: ranges of time that will be used in this rule.
- Pattern: Incoming call pattern.
- Destination: where should the incoming call is routed.

2.14 Voicemail

General settings for voicemail function.

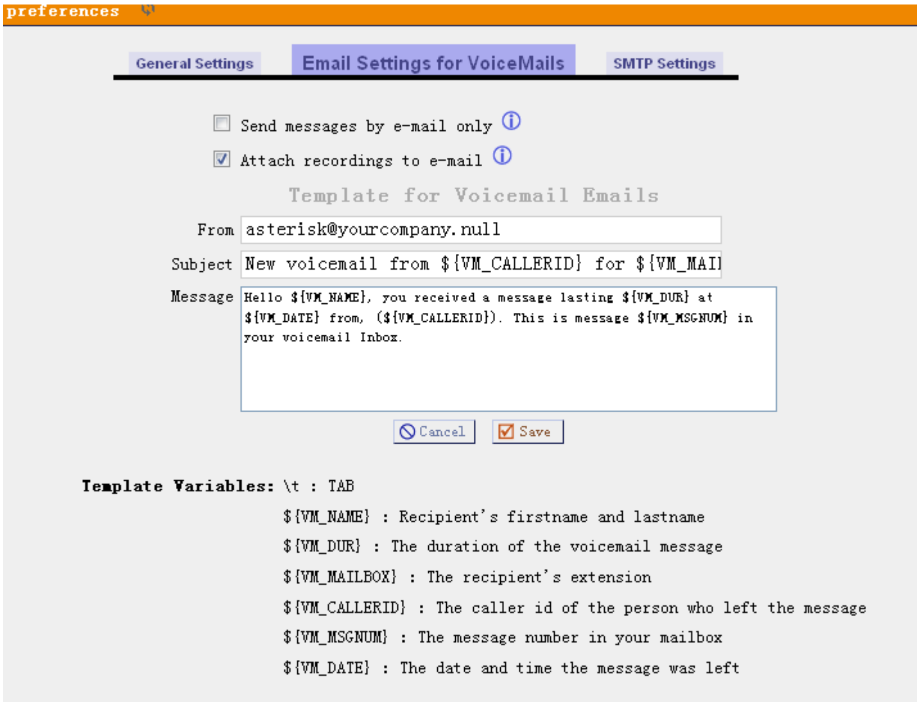


General Voicemail Settings:

- Extension for checking Message: This option, i.e. "2345" defines the extension that Users call in order to access their voicemail accounts.
- 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.
- Max Greeting:Set the maximum number of seconds for a User's voicemail greeting.
- Dail 'O' 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.
- Maximum messages per folder: This select box sets the maximum number of messages that a user may have in any of their folders.
- Maximum 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.
- Minimum message Time: This select box sets the minimum duration of a voicemail message in seconds. Messages below this threshold will be automatically deleted.
- 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.
- Say Message Duration (in minutes): If this option is set, the duration of the message in mintues will be played back before the voicemail message begins playing.
- Allow callers to Review: Checking this option allows the caller to review their message before it is submitted as a new voicemail message.
- Play Envelope: Turn on/off playing introductions about each message when accessing them from the voicemail application.

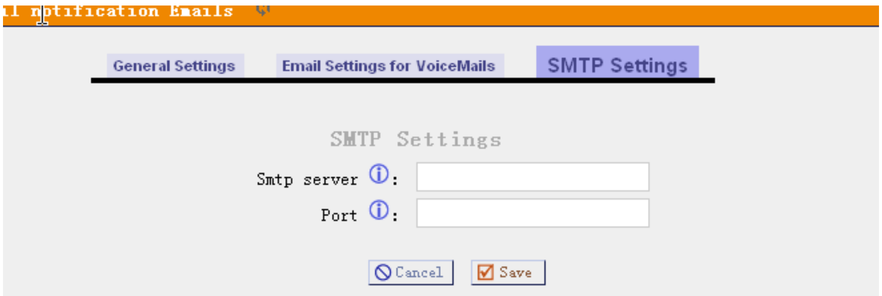
Voicemail to Email: with this function configured, when there is a new voicemail for users, the IP-01P will automatically send the voicemail to the user's email address set in the user's profile.

Voicemail to Email Preference:



- 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.
- Attach recording 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.

SMTP server setting

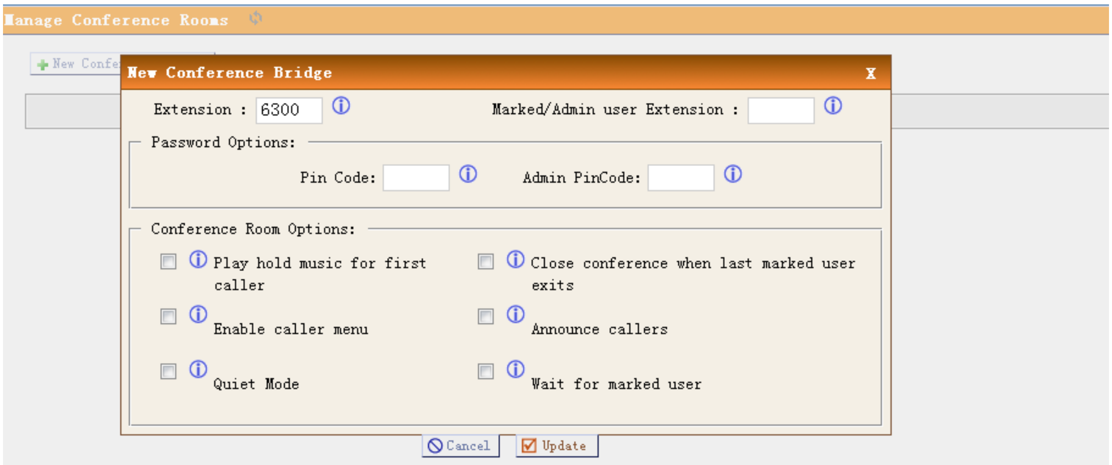


- SMTP Server: The IP address or hostname of an SMTP server that your Astfin box may connect to, without authentication, in order to send e-mail notifications of your voicemails; i.e. mail.yourcompany.com
- Port: The port number on which the SMTP server is running; generally port 25.

Note: for Setup example for the Voicemail to Email please refer the application note.

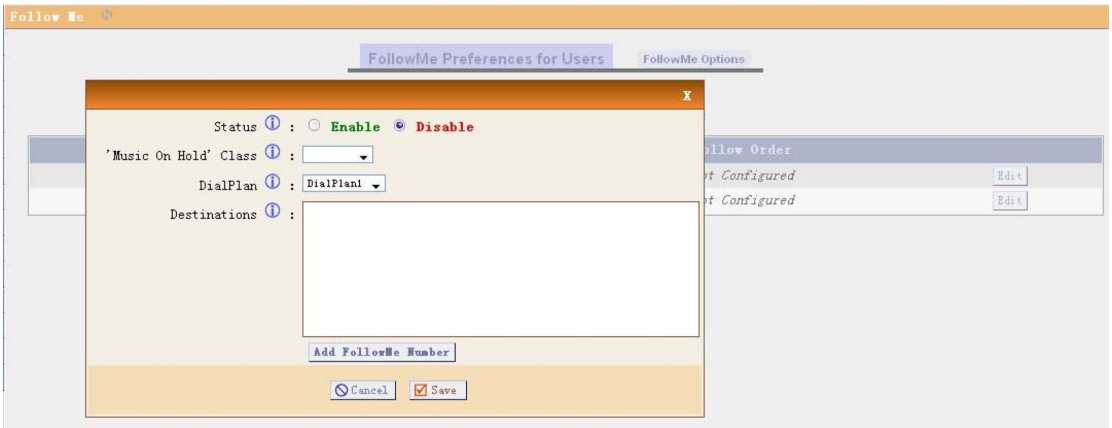
2.15 Conferencing

MeetMe conference bridging allows quick, ad-hoc conferences with or without security.



- Extension: This is the number dialed to reach this Conference Bridge.
- 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 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.
- Administrator PIN Code: Defining this option sets a PIN for Conference Administrators.
- Play Hold Music for 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.
- Enable Caller Menu: Checking this option allows a user to access the Conference Bridge menu by pressing the * "Asterisk" key on their dialpad.
- Announce Callers: Checking this option announces, to all Bridge participants, the joining of any other participants.
- Quiet Mode: Do not play enter/leave sounds
- Wait for Marked User: Prevent conference participants from hearing each other until the marked user has joined.

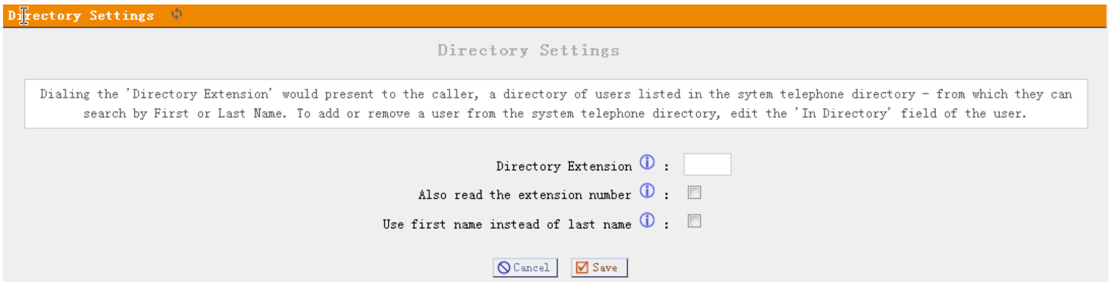
2.16 Follow Me



- ➢ Status: Enable/Disable FollowMe for this user.
- ➢ Music On Hold class: that the caller would hear while tracking the user.
- ➢ DialPlan: DialPlan that would be used for dialing the FollowMe numbers. By default this would be the same dialplan as that of the user.";
- ➢ Destinations: List of extensions/numbers that would be dialed to reach the user during FollowMe.

2.17 Directory

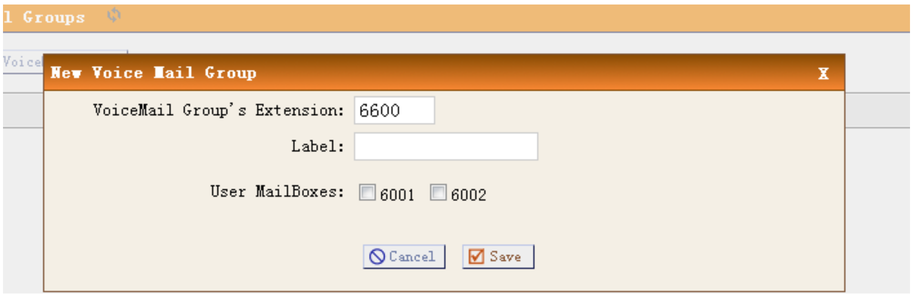
Preferences for 'Dialing by Name Directory'



- Directory Extension: Extension to dial for accessing the Name Directory
- Read Extension number: In addition to the name, also read the extension number to the caller before presenting dialing options
- 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

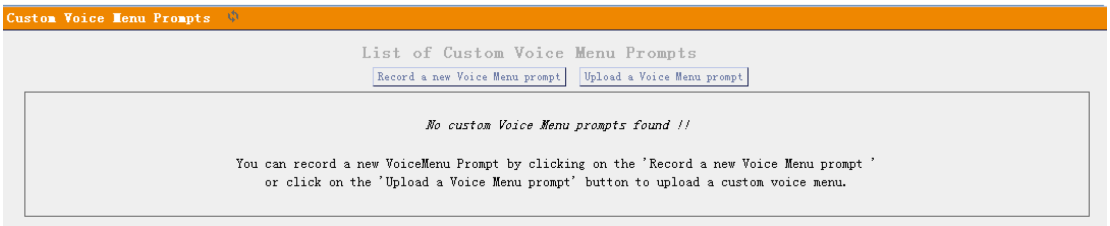
2.18 Voicemail Group

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



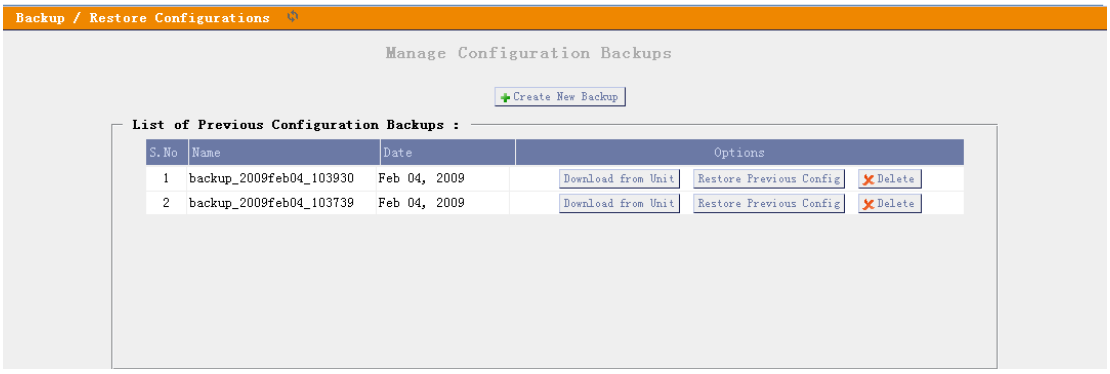
2.19 Voice Menu prompts

Record or Upload custom VoiceMenu prompts.



2.20 BackUp

Backup or Restore the configure files.



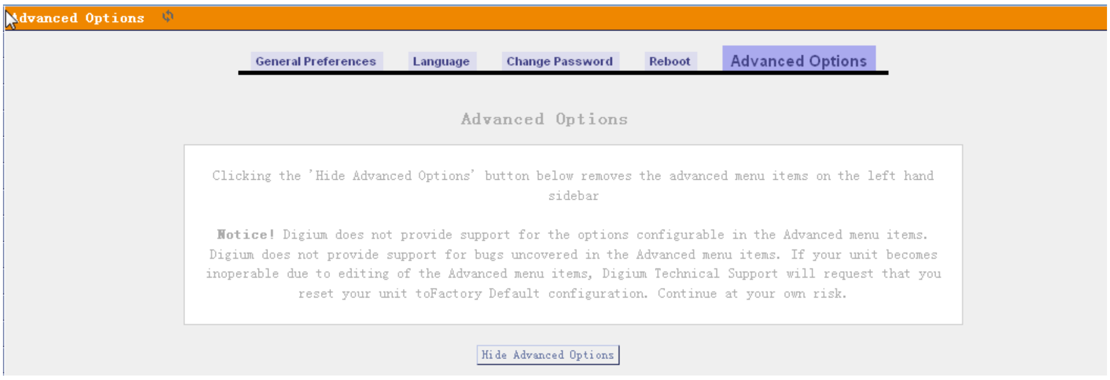
Note: Restored the backup won't take effect on the network setting. You need to modify the network setting in the GUI and save/reboot.

2.21 Active Channels

Displays current Active Channels on the PBX, with the options to Hangup or Transfer.

2.22 Advance Options

In the Options Panel, choose Advanced Options--> show Advanced Options then the advanced options will be showed in the left menu.

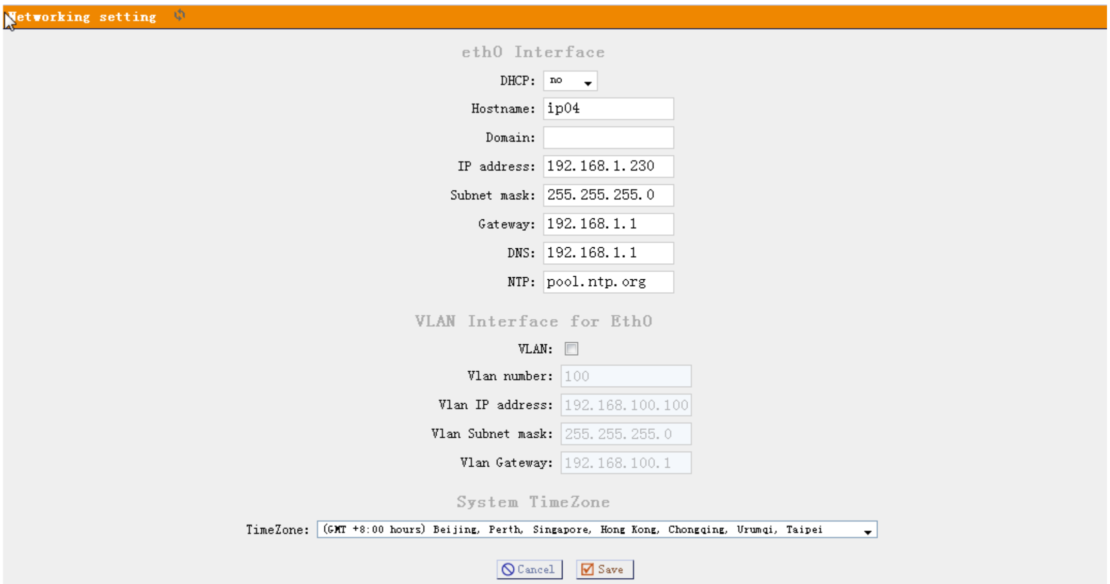


The advances options include:

- Call Detail Records
- Active Channels
- Bulk Add
- File Editor
- Asterisk CLI
- IAX Settings
- SIP Settings
- Network Settings
- Firmware update

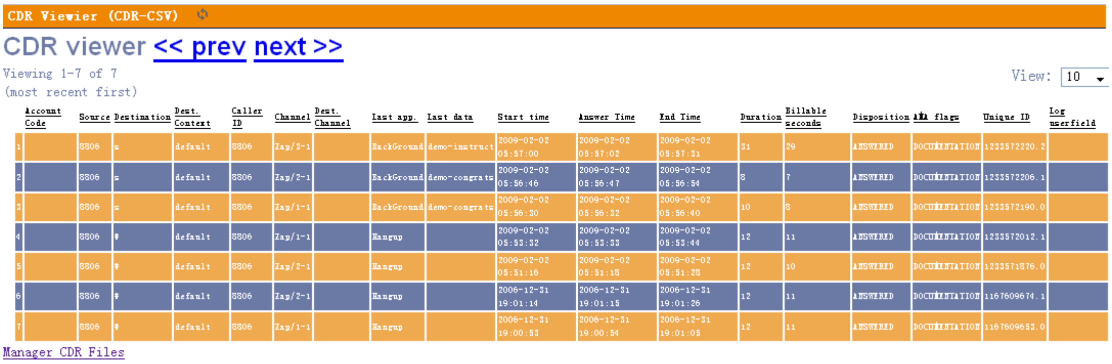
2.23 Advance Options—Network Settings

Network and time zone settings.



2.24 Advance Options—Call Detail Records

Shows the call details.



CDR Viewier (CDR-CSV)

CDR viewer << prev next >>

Viewing 1-7 of 7
(most recent first)

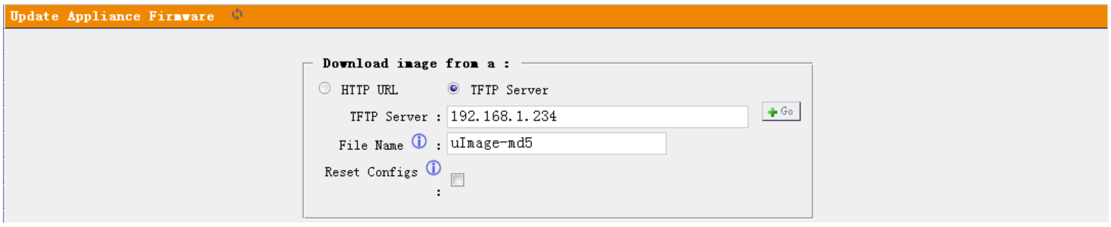
View: 10

	Account Code	Source	Destination	Dest. Context	Caller ID	Channel	Dest. Channel	Last app.	Last data	Start time	Answer Time	End Time	Duration	Billable seconds	Disposition	AMA flags	Unique ID	Log userfield
1		8806	s	default	8806	Zap/3-1		BackGround	demo-instruct	2009-02-02 05:57:00	2009-02-02 05:57:02	2009-02-02 05:57:31	31	29	ANSWERED	DOCUMENTATION	1233572220.2	
2		8806	s	default	8806	Zap/2-1		BackGround	demo-congrats	2009-02-02 05:56:46	2009-02-02 05:56:47	2009-02-02 05:56:54	8	7	ANSWERED	DOCUMENTATION	1233572206.1	
3		8806	s	default	8806	Zap/1-1		BackGround	demo-congrats	2009-02-02 05:56:30	2009-02-02 05:56:32	2009-02-02 05:56:40	10	8	ANSWERED	DOCUMENTATION	1233572190.0	
4		8806	#	default	8806	Zap/1-1		Hangup		2009-02-02 05:53:32	2009-02-02 05:53:33	2009-02-02 05:53:44	12	11	ANSWERED	DOCUMENTATION	1233572012.1	
5		8806	#	default	8806	Zap/2-1		Hangup		2009-02-02 05:51:16	2009-02-02 05:51:18	2009-02-02 05:51:28	12	10	ANSWERED	DOCUMENTATION	1233571876.0	
6		8806	#	default	8806	Zap/2-1		Hangup		2006-12-31 19:01:14	2006-12-31 19:01:15	2006-12-31 19:01:26	12	11	ANSWERED	DOCUMENTATION	1167609674.1	
7		8806	#	default	8806	Zap/1-1		Hangup		2006-12-31 19:00:53	2006-12-31 19:00:54	2006-12-31 19:01:05	12	11	ANSWERED	DOCUMENTATION	1167609653.0	

Manager CDR Files

2.25 Advance Options—Firmware Update

Update the firmware of your device

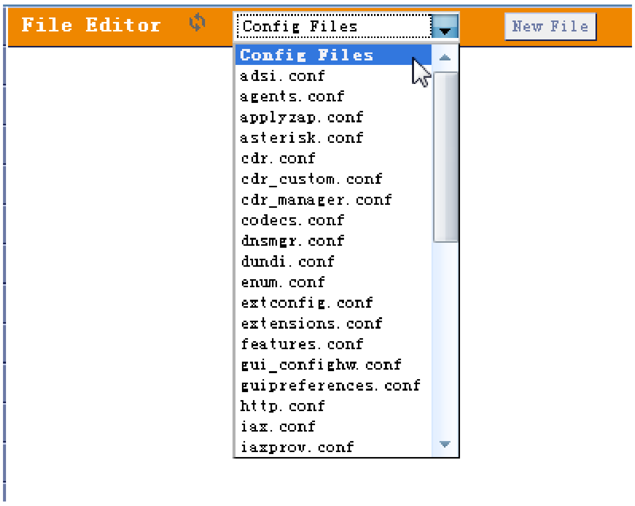


HTTP and TFTP update is available for the firmware update.

- TFTP server: TFTP server which include the update firmware
- File Name: name of the new firmware, please make sure that you are using a md5 firmware for the updating.
- Reset Configs: enable this if you want to reset the networking and asterisk configs after upgrade.

2.26 Advance Options—File Editor

Here you can modify the asterisk configure files directly.



Note: Please make sure you know what the meaning in the files before trying to modify these files.

3 Application notes

3.1 Different methods to access the IPxx

There are several ways to access the IPXX series products. Different ways has different usage. The web/SSH/telnet accesses are base on network connection, and the console port access is via the console cable which allows you to access the devices on a lower level.

- Web access

It is the most common way to access the IPxx. Most settings can be done through the web interface. Simply put the device's IP address in your web browser (better use Firefox) and enter the username and password to access the device. The web access username/password is **admin/mysecret**

- SSH access

This is the advance way to access the device, you can use the putty software to access the device. in the ssh access, you can access the Linux directly and do more advance linux setting and debug. The SSH user/password is **root/uClinux**

- Telnet access

The telnet access is similar with the SSH access but it is not suggested because it is not as convenient as SSH access.

- Console access

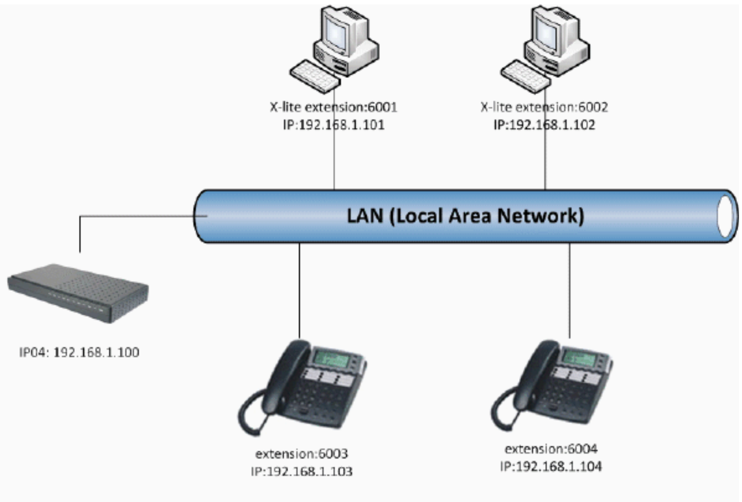
The console access is used mainly for develope purpose or in the case when network is down.
Below is the console port setting to access the IPXX
Running the Hyper Terminal or Minicom in your computer to connect the IP-01P, the setting of the console port should be:

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

3.2 Make free internal calls

Making internal calls are the base requirement for a telephony system. Below are the settings for this usage. It is base on IP-01P but setting is the same in other IPXX products.

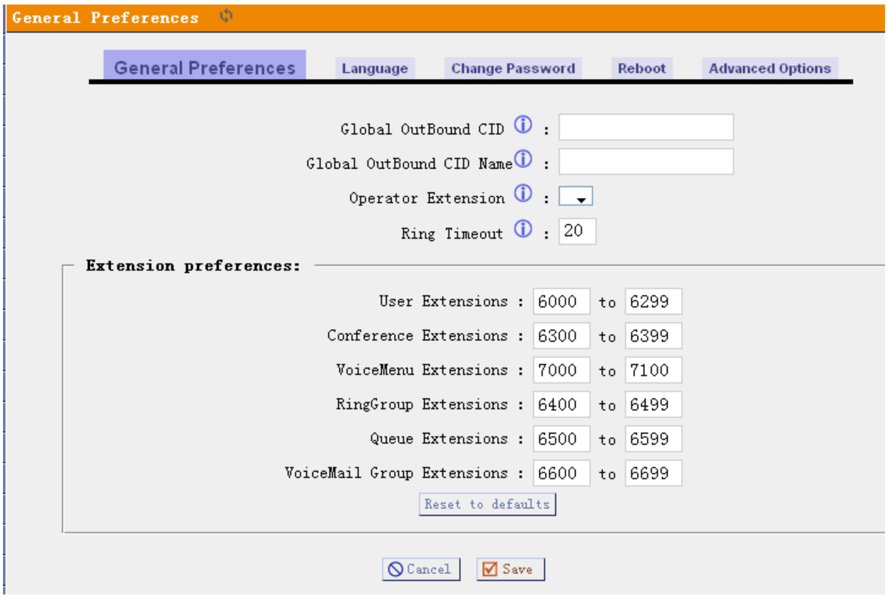
System Setup



Internal calls in the same office

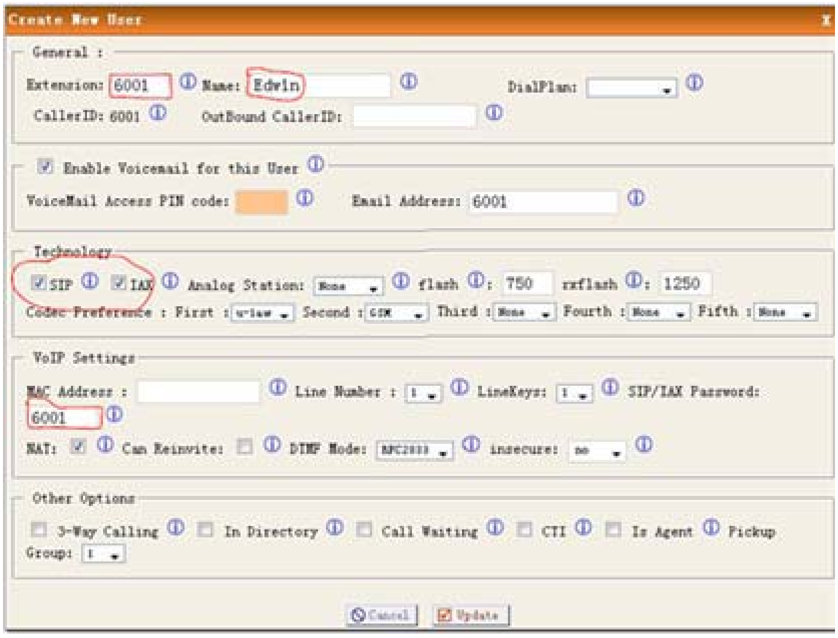
At the beginning, we need to add some extensions to make internal calls. Each extension acts as an internal number. There are three types of extensions we can add: SIP, IAX2 and ZAP.

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 6000~6299. The extension 6000 is used for auto-attendant so don't register on this extension.



Then go to page **Dial Plan-->Create New Dialplan** to create a default Dial Plan.

Then go to page **Users-->Create New User** to create the extensions: 6001



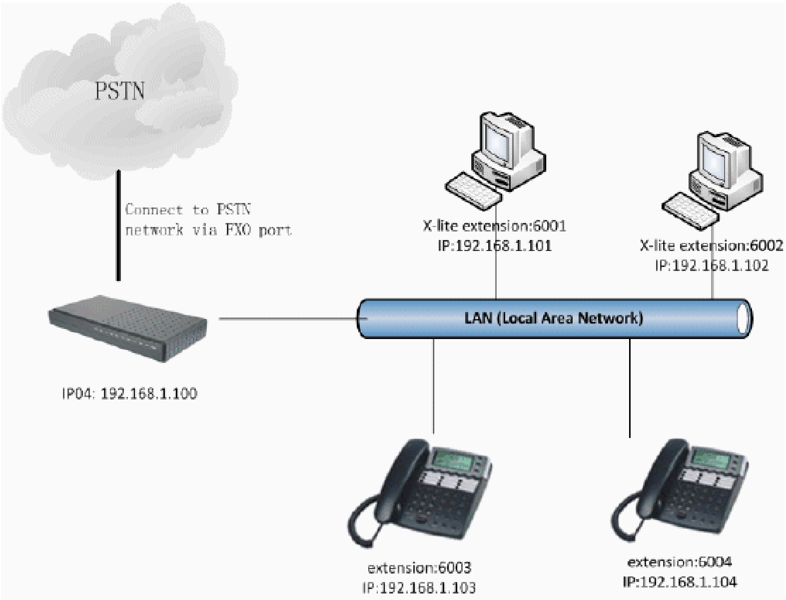
Use the same method to add the extensions: 6002, 6003 and 6004. In our system picture, we use softphone x-lite to register on 6001 and 6002. Use AT-530 IP phone to register on the extensions 6003 and 6004. Then these four extensions can communicate with each other use the numbers 6001, 6002, 6003 and 6004.

3.3 Make outbound calls to PSTN

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

3.3.1 Analog/FXO trunking

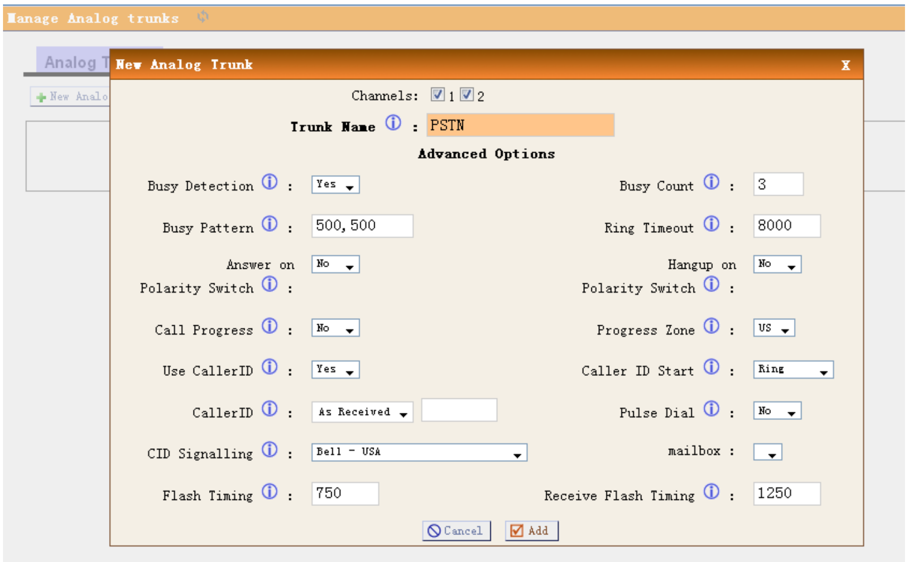
For the IP01/04/08, you can install FXO module and use the FXO trunking to make outgoing call via your local PSTN line. The set up is as per below:



Outgoing Call To PSTN Network

Step 1: Create FXO trunk

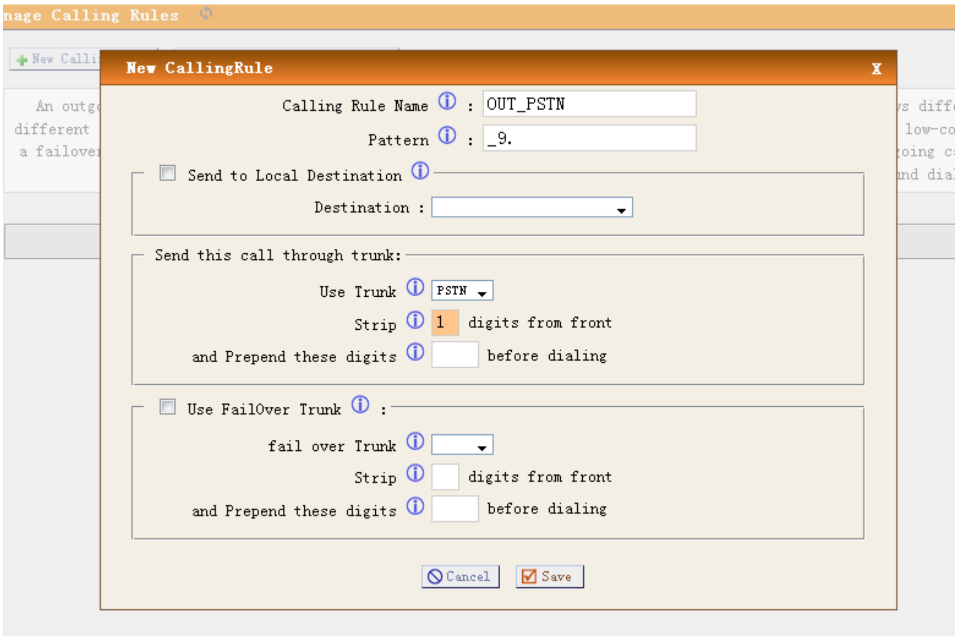
Go to page Trunks--> Add New Analog Trunk



Note: The port1 and port2 of IP-01P 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

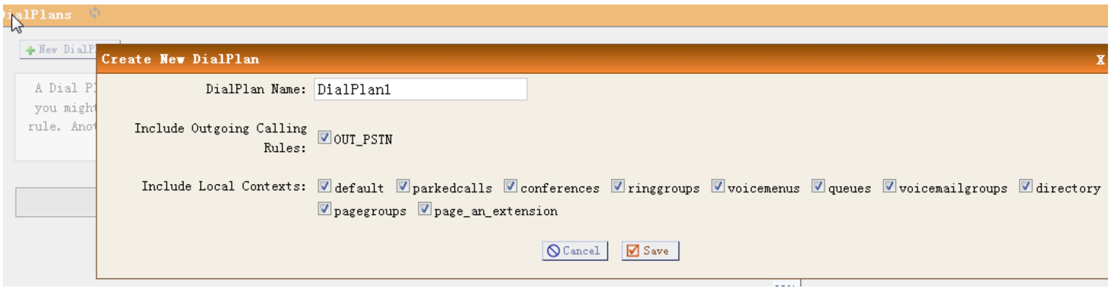
Go to page Outgoing Calling Rules.



The pattern _9. and strip 1 digits means all calls start with 9 will be cut the first digit and sent out via this rule. for example, if you dial 983018049, the IP-01P will send 83018049 to port1 or port2.

Step 3: Add New DialPlan

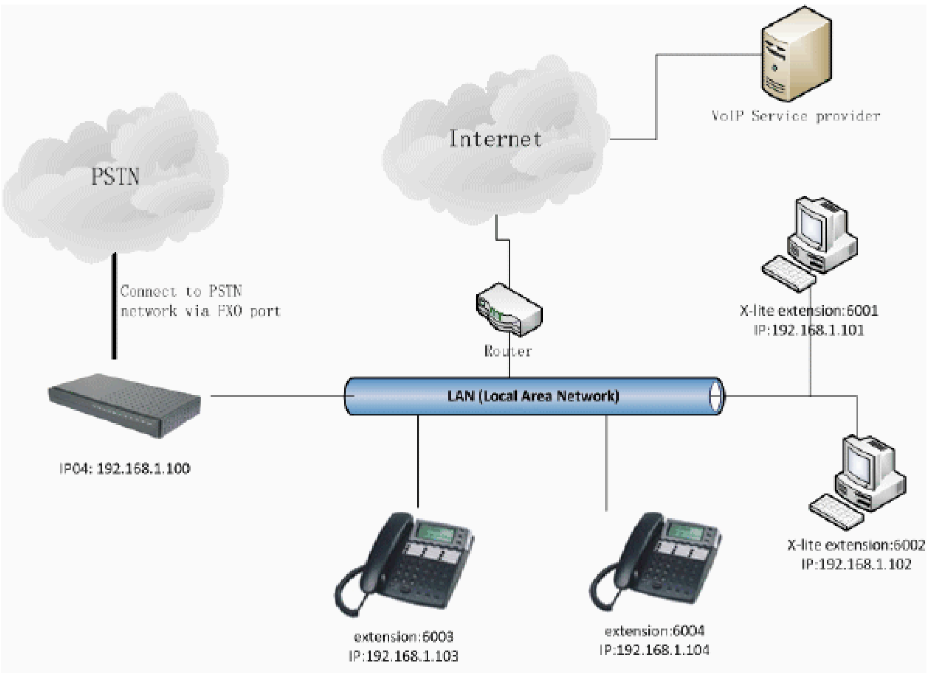
Go to page Dial Plans--> Crear New Dial Plans



Set the DialPlan1 to default dial plan so every new extension will use this dialplan in default. Then you can use your extensions to dial out via the port1 and port2.

3.3.2 VoIP trunking

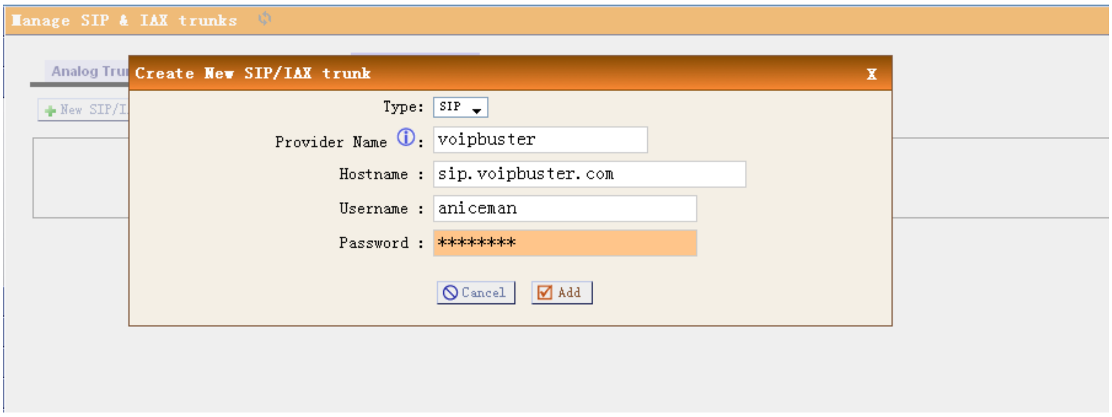
Via the voip trunking we can dial call via the voip service to reduce our cost when making international calls.



Outgoing Calls Network

Step 1: Add Voip trunks

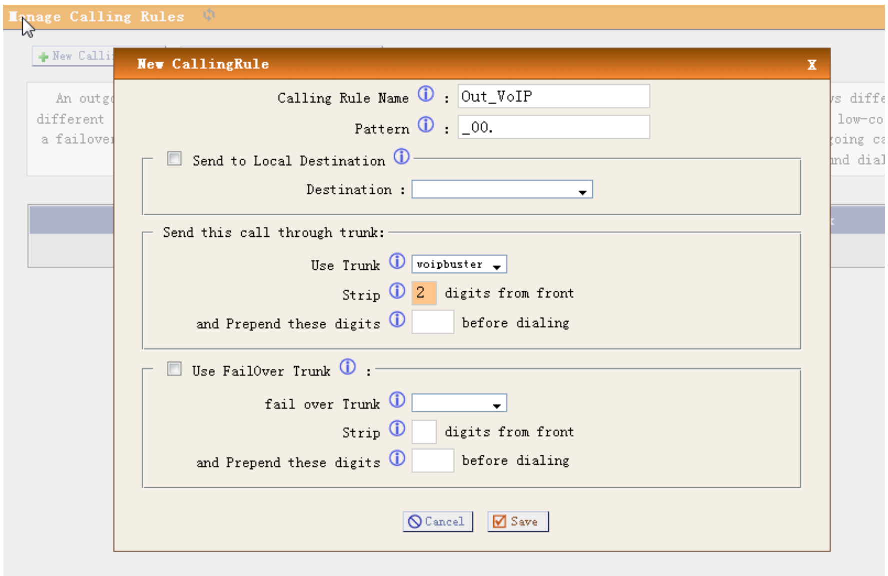
Go to page Trunks--> Voip Trunks--> Add New Sip trunks



We use the Voipbuster as our voip service provider here.

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.

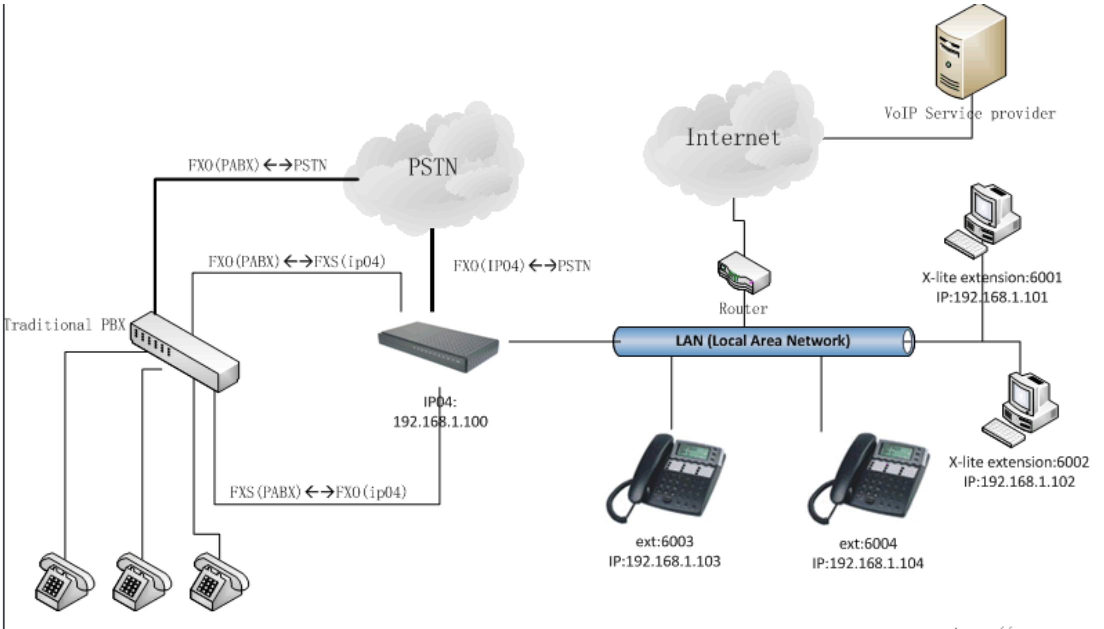
Step 3: Add this new calling rule to the dial plan1

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

3.4 Combine the IP-01P with exist traditional PBX

Introduce:

Assume that we already have a (3-8 3fxo 8extensions) traditional PABX in our office and we want to add more pstn lines/extensions or use voip solution in the exist solution. We can combine the IP-01P with the exist PABX solution use below structure.



Combine the IP04 with exist traditional PBX

1/ Connect the FXO port of the PABX to IP-01P's FXS port. so the PABX will be one of the IP-01P's FXS extensions and all the extensions under pabx can use all the fxs functions from IP-01P, the functions include:

make calls to the IP-01P's other extensions
make calls use the IP-01P voip trunk
make PSTN via IP-01P's PSTN trunk

2/ Connect the FXS port for the PABX to IP-01P's FXO port. So the IP-01P will be one of the PABX extensions and all the extensions under pabx can use all the fxs functions from pabx, the functions include:

make calls to the pabx's other extensions
make PSTN via pabx's PSTN trunk

3.5 Intercommunication between two IPXX

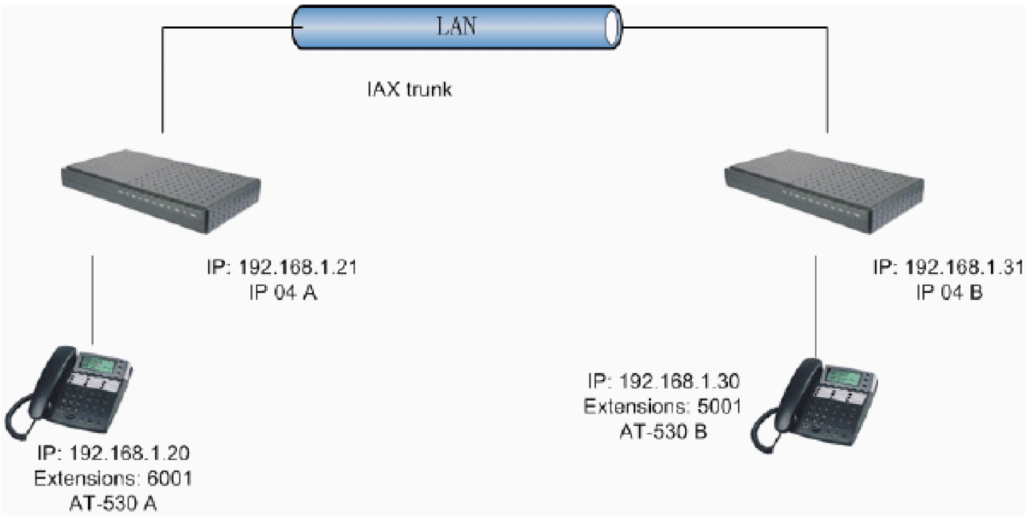
Introduce:

This application note shows how to link two IP-01P in different location. With this function, we can link branches together with IP-01P. Same method can be used when connect more than 2 IP-01P in different branches.

3.5.1 Link two IP-01Ps in the same network

The simplest case to link two IPxx together is in the same network. We start from this and then try to expand to different network. We use IP-01P here, same method for other IPXXs products.

Below is the structure of how to link two IP-01Ps in the same LAN:



The method of connecting two IP-01Ps in different location is: 1) register the IP-01PA as an extension in IP-01PB(via IAX2 trunk) so the extensions in IP-01PA can make calls to IP-01PB's extensions via this "special" trunk. 2) use the reverse method in IP-01PB to register to IP-01PA.

In above structure:

1) AT-530A registers to IP-01PA as an extension 6001.
2) AT-530B registers to IP-01PB as an extension 5001.
3) All the extensions under IP-01PA are in the format 6XXX.
4) All the extensions under IP-01PB are in the format 5XXX
5) Extensions under IP-01PA can make calls to extension under IP-01PB use format 5XXX.
6) Extensions under IP-01PB can make calls to extension under IP-01PA use format 6XXX.
7) The two IP-01P links each other via IAX2 trunk.

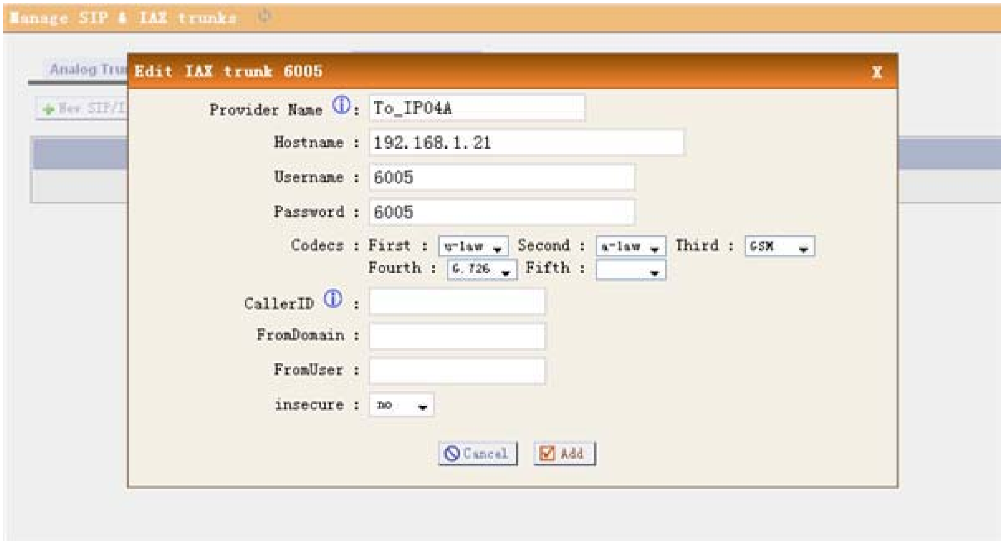
Step 1: Set up a extension 6005 in IP-01PA

Extension: 6005 ; Phone number of this extension
Name: User_IP-01PB ;

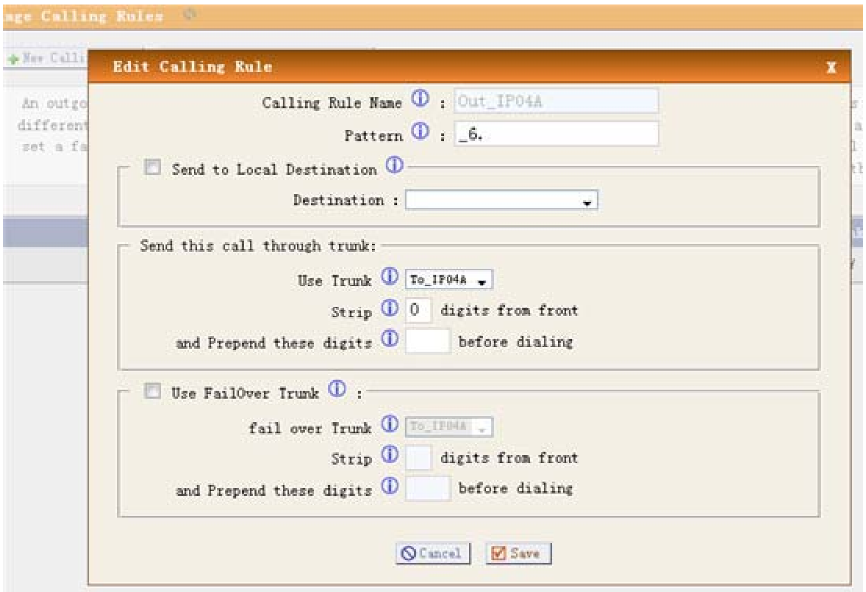
Password: 6005 ;IAX2 Log on password

Caller ID: 6005 ; Caller ID

Step 2: Set up an IAX trunk in IP-01PB to link to IP-01PA via this User_IP-01PB extension. In the page Trunks--> Add Voip Trunk



Step 3: Set Calling Rule in IP-01PB, all calls start with 6 will be sent to IP-01PA. In the page: Outgoing Calling Rules--> Add New Calling Rule



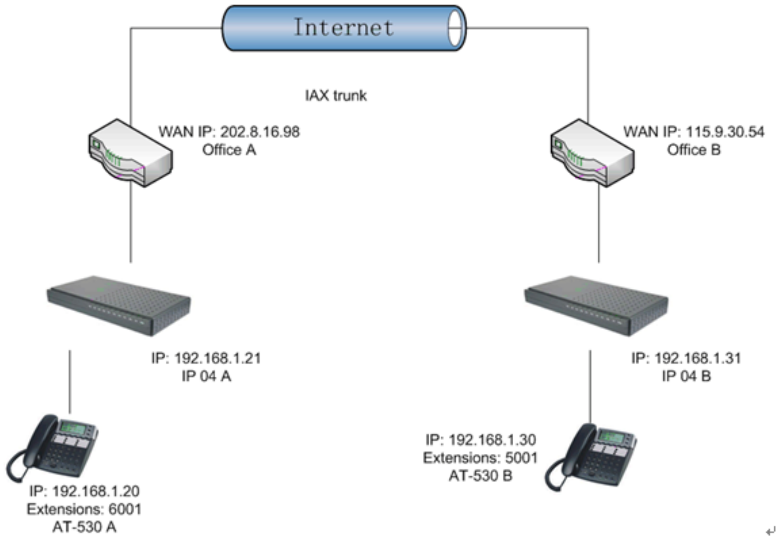
Step 4: Add this new calling rule "Out_IP-01PA" to the exist dial plan. In the page: DialPlan --> Edit DialPlan1

Active the change and apply the test:
1/Register an IP phone AT-530B to IP-01PB with 5001 extension.
2/Register an IP phone AT-530A to IP-01PA with 6001 extension.
3/Use 5001 to dial 6001. And you can see 6001 is ringing and you can pick up the calls.
Above is the way to router IP-01PB's call to IP-01PA, the method to link IP-01PA to IP-01PB is the same as above.

3.5.2 Link two IP-01Ps in different location

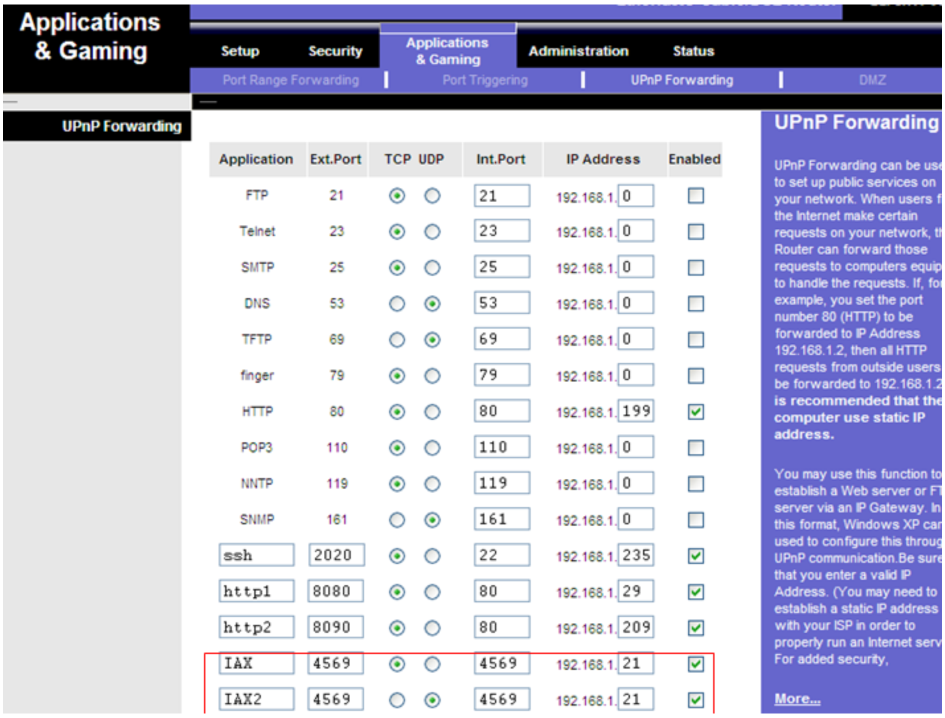
The generally environment for two IP-01P in different location is: two IP-01P are both behind router and using the private IP.

Since the IP-01P doesn't have the public IP, we need to do port forwarding in the router and make IP-01P is reachable to others.



Step 1: Set port forwarding in the router for IP-01PA

The IP-01PA is behind the router, to register to IP-01PA via the internet, you need to forward the IAX2 port in your router, so all the packets received on the router WAN port (202.8.16.98:4569) will be forwarded to the IP-01PA (192.168.1.21:4569). Below is the setting page in a linksys router:



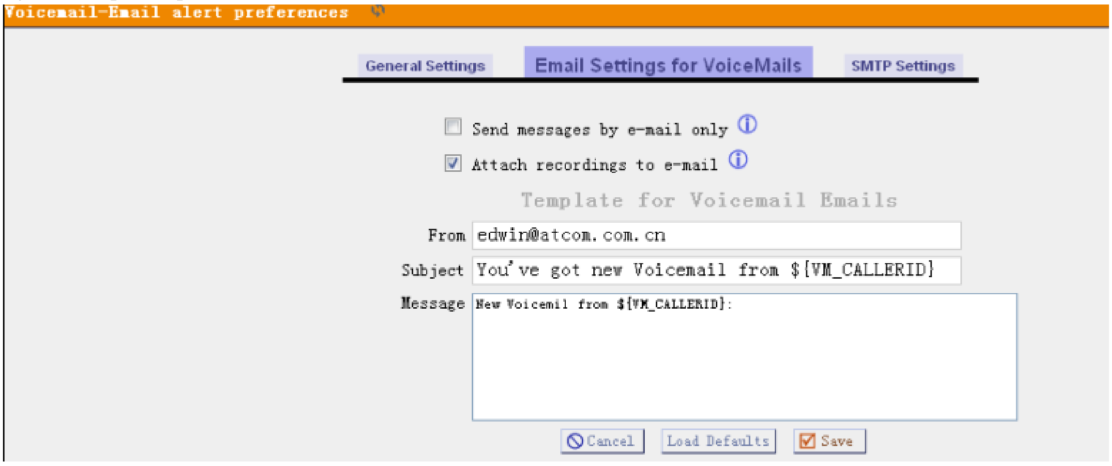
Step 2. Set up the service provider and calling rule in IP-01PB to make it register to IP-01PA. This method is almost the same as above, EXCEPT you need to use the 202.8.16.98 as the service provider instead of 192.168.1.21.

Step 3. Use the same method do port forwarding in routerB for IP-01PB. Your public address from network provider maybe a dynamic ip which will be changed periodically. To overcome the problem of dynamic ip, you may need to use the DDNS service , for more info please google the internet.

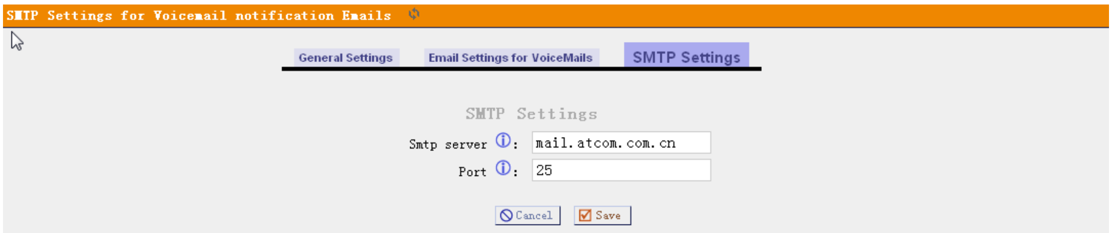
3.6 Voicemail to Email Configure example

The IP-01P will send a notify Email to your mail box when you have set up the Voicemail to Email function.

1) Set up the preference



2) Configure your SMTP server



If your SSMTP server needs Authentication, you need to put your username and password in the file ssmtp.conf (via SSH access) as below:

[/etc/ssmtp/ssmtp.conf]

root=edwin@atcom.com.cn //mailbox account

mailhub=mail.atcom.com.cn //smtp server

rewriteDomain=atcom.com.cn

hostname=edwin@atcom.com.cn

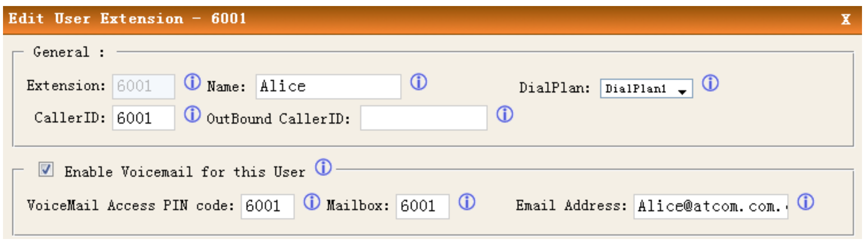
AuthUser=edwin@atcom.com.cn //mailbox account

AuthPass=xxxxxxx //mailbox password

AuthMethod=LOGIN

FromLineOverride=YES

3) Enable the voicemail for users and put the corresponding Email Address.



3.7 Call Features

3.7.1 Call Pick Up

The default feature code to pick up a call is *8. If there is an incoming call for a user in the call group 2, then the users in the pickup group 2 is able to pick up this call by dialing *8.
The pickup group and call group can be defined in the user setting page.

3.7.2 Call Park

The default call park extension is 700. The call park features code can be found in the file: /etc/asterisk/features.conf

Park a call on eye-beam:
--Press XFER button , then it will shows Enter Number + press XFER
--Enter the default park extension 700 and press the XFER button again. The call will be parked to the extension range 701~720
--dial 701~720 to get the parked call in another extension.

3.8 Cron

Cron is the name of program that enables Linux users to execute commands or scripts (groups of commands) automatically at a specified time/date. It is normally used for sys admin commands, like makewhatis, which builds a search database for the man -k command, or for running a backup script, but can be used for anything.

You can start the cron service for IP-01P by:
/etc/init.d/cron enable
/etc/init.d/cron start

The crontab file locats in /etc/config. More info for how to use cron in linux, please search in the internet.

4 Application notes

5 Known Issue

5.1 Call Detail Record

You need to use firefox to open the CDR, it doesn't work in the IE.

FCC WARNING

This device complies with Part 15 of the FCC Rules. Operation is
subject to the following two conditions:
(1) this device may not cause harmful interference, and
(2) this device must accept any interference received, including interference that may cause undesired operation.

NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

NOTE: The manufacturer is not responsible for and radio or TV interference caused by unauthorized modifications to this equipment. Such modifications could void the user’s authority to operate the equipment.