APBX IP04 User Manual

For Firmware Version: V1.4.0

2013-07-11

Contents

1. Introduction

Overview of the APBX

ATCOM IPPBX is a SIP-based IP voice switch with a small embedded OS and rich GUI (Graphical User Interface), providing a powerful networking and corporate communication function. With it, users can quickly deploy an internal communication system for enterprise, as well as configure conveniently applications and value-added services on IP PBX via its GUI, to fit enterprise's own various demands.

Targeting for SOHO user and SMB market with an easy to use graphical interface, IP04 provides a cost-saving solution on their telecommunication data needs. With IP04, company with branch offices in different countries can be easily combined together to work like a virtual single office through internet.

Hardware Specifications

CPU	400MHz Blackfin 532 Chip
NAND Flash	256 M
SDRAM	128 M
Analog Port	4
Network Interface	WAN

Measurement and Weight

Inner box	225 * 120 * 30 mm
G.W./unit	0.765KG
Carton MEAS	456 * 442 * 362 mm
Units per Carton	21 units/ CTN
G.W./CTN	18 KG/CTN

Function Features

Voicemail	**Authentication before call outbound**
Voicemail to Email	**User WEB portal**
Blind/Attended Transfer	**Blacklist**
Call Forward	**Call Detail Records(CDR)**
Call Parking	**Conference**
Do not Disturb (DND)	**Ring Group**
Group / Directed Pickup	**Call Queue**
Call Recording	**IVR**
Call Waiting	**Intercom/Paging**
Call Routing	**Firewalls**

Caller ID	**IP Restriction**
BLF Support	**DDNS**
Music on Hold	**VLAN**
Storage Quota Privilege	

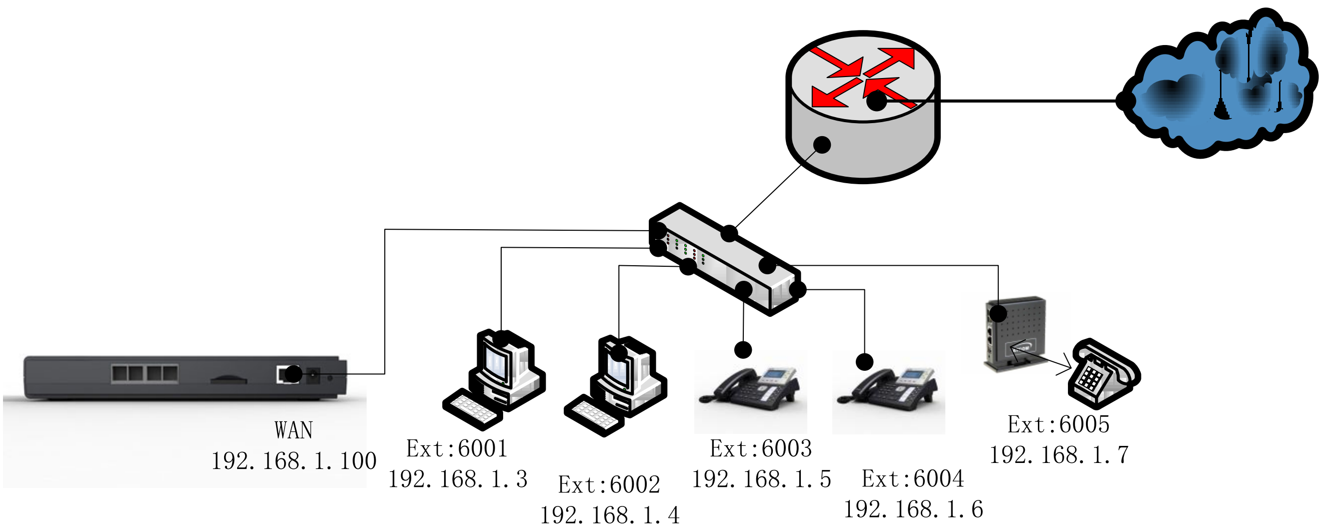
2. Connection and Change IP Address of APBX

2.1 Connection

The default IP address of APBX is:

WAN: 192.168.1.100/255.255.255.0

The network scenario should be like below:



1) Connect APBX to your PC directly or through switch.

   Make sure IP address of your PC is in network 192.168.1.0/255.255.255.0, if not, you need to appoint an IP address for your PC, for example, 192.168.1.3

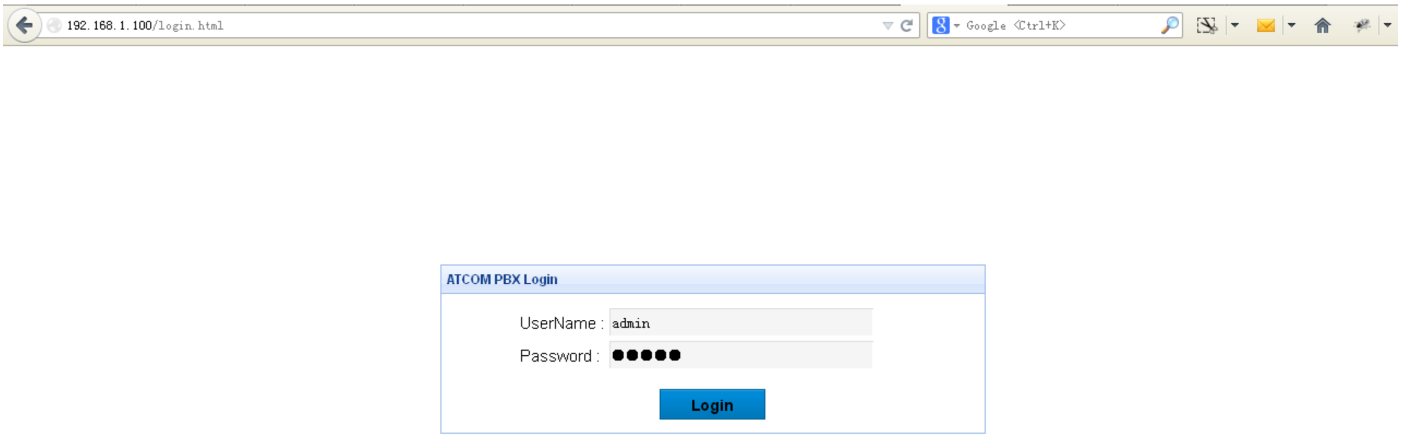
2) Login APBX as administrator via WEB GUI

User: **admin**

Password: **atcom**

3) Go to **Network Settings** ->**WAN**.

Re-set IP of WAN port.



2.2 WAN Settings

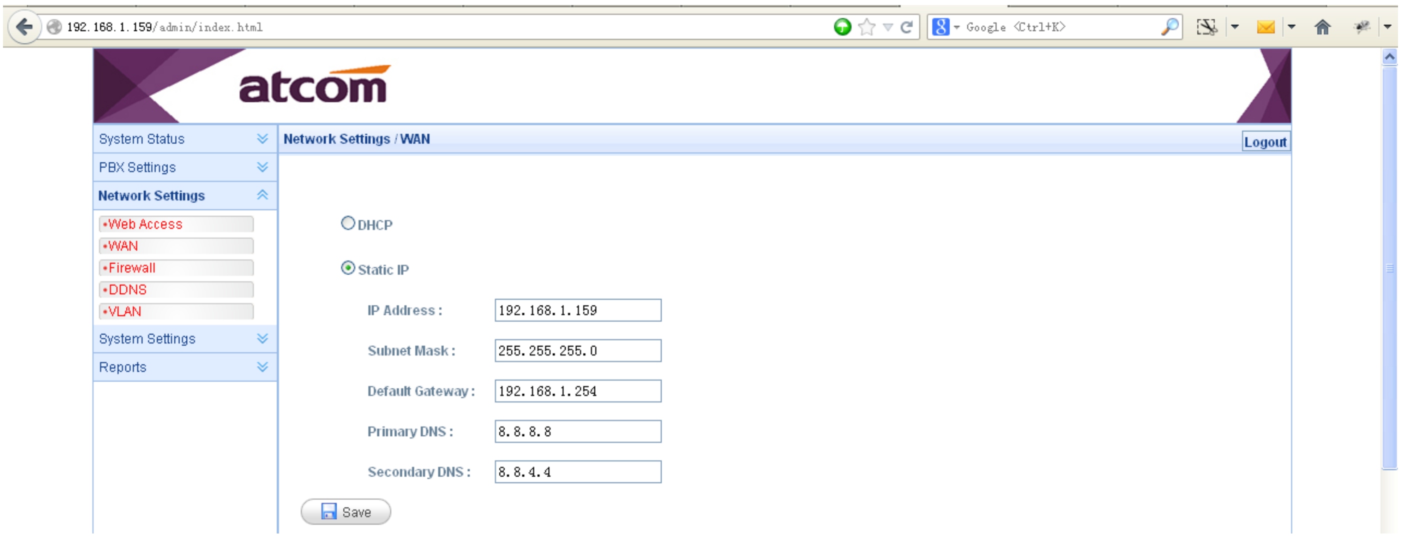
There are two ways to set an IP address to WAN port: DHCP, Static IP.

DHCP

APBX will obtain an IP address automatically from DHCP server when rebooting. It’s not recommended to choose this option unless there is a reserved IP for APBX in DHCP server so that APBX can keep the same IP all the time.

Static IP

Set an IP address manually according to the real network environment. If APBX is behind a router, the gateway is usually set to the IP of the router.



2.3 System Status

1) General

Product Model

Show the model of this APBX

Firmware Version

Show the firmware version

System Uptime

Show the time how long the system has been running

System Current Time

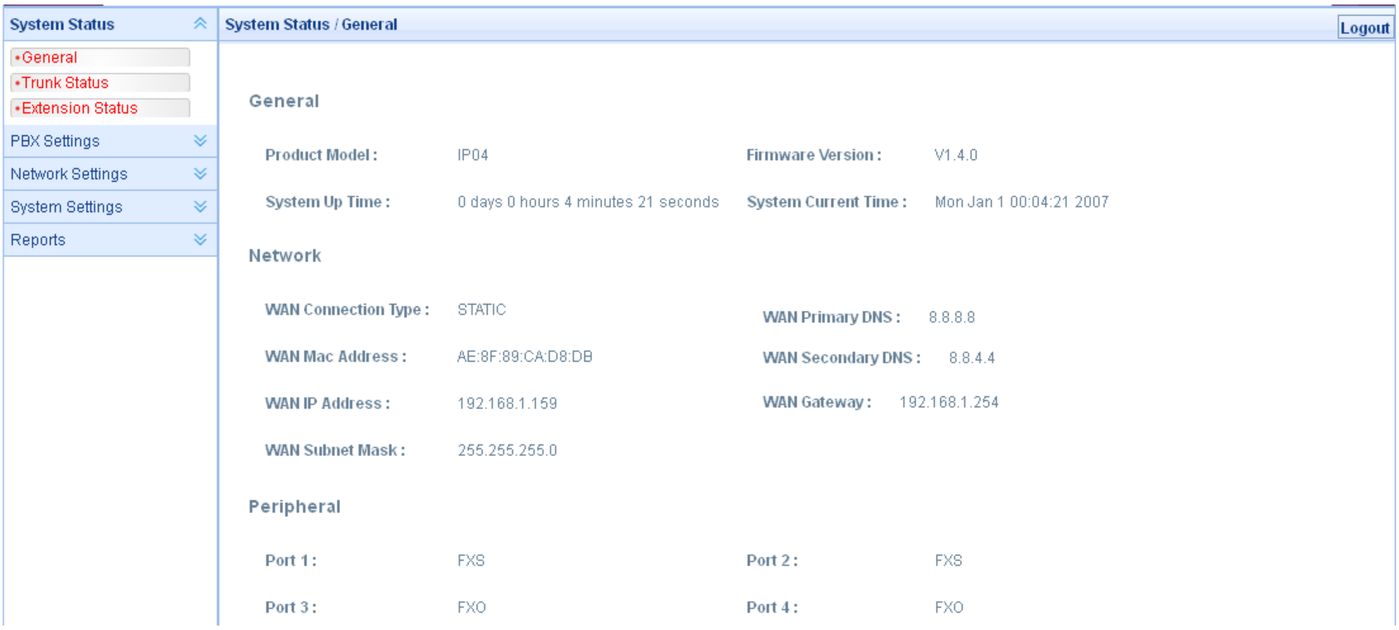
Show the current time

2) Network

Show the network setting of APBX

3) Peripheral

Show what kinds of / how many modules are detected.



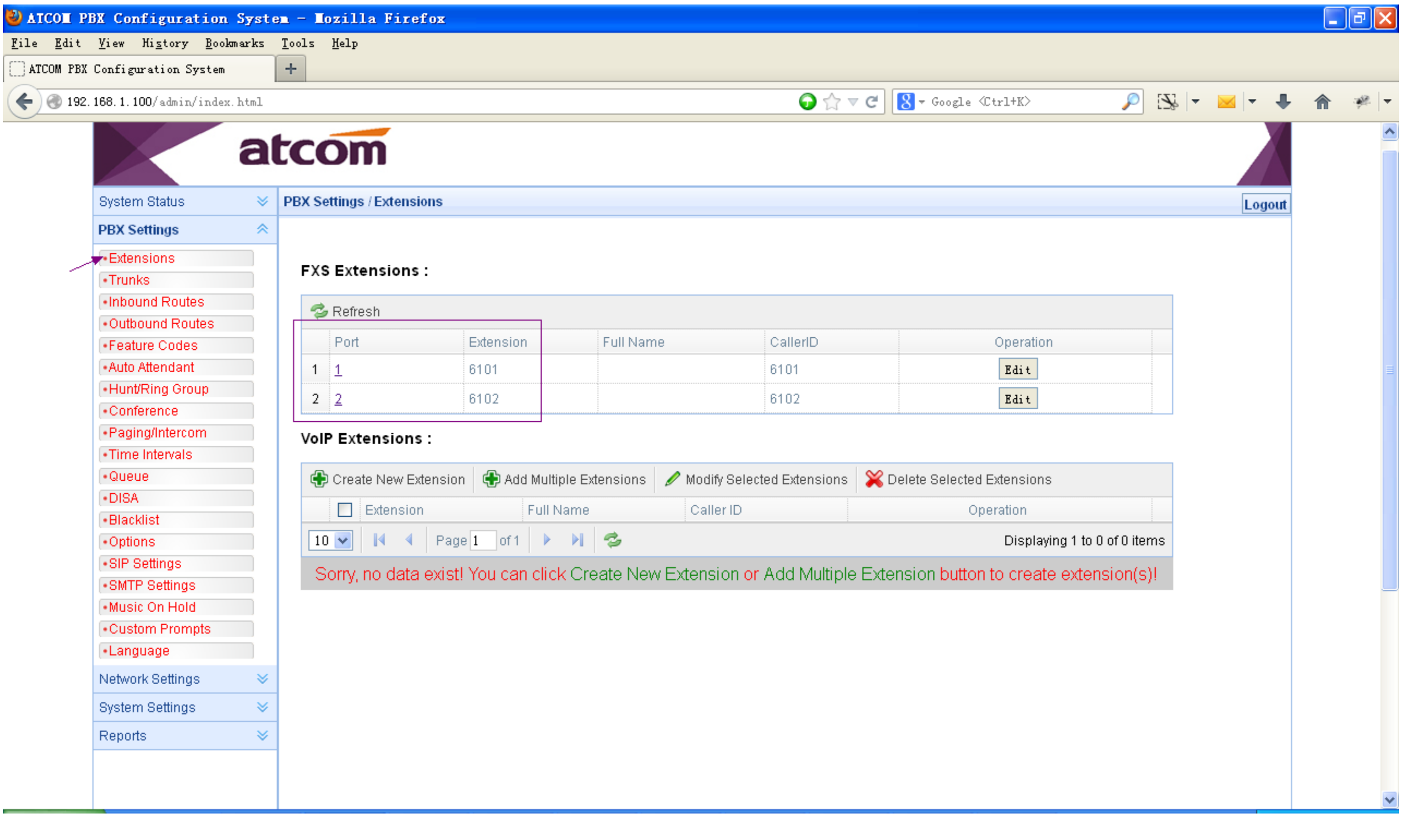
3. Create local extensions and make interior calls

There are two kinds of extensions in APBX: FXS extensions and SIP extensions.

3.1 FXS extensions

It needs support of FXS module, the module installed in IP04 can be: **AX110S**

Analog phone is available to make calls once connected to the corresponding FXS port, APBX configures FXS extension automatically when FXS module is detected. FXS extension can’t be deleted. The extension number is defined in **PBX Settings** -> **Options** -> **Extension Preference**, changing it can change the FXS extension number.



1) General

Extension

Extension number, i.e. 6101, it is associated with this particular user / phone.

Port

The analog port bound with extension.

Name

A character-based name for this extension, i.e. ‘Bob Jones’.

Caller ID

CID showed in the other’s phone during a call, default is Extension.

2) Voice Mail

Enable Voice Mail

Check this option to enable voice mail account for the extension. Enabled by default.

Voice Mail Access PIN Code

Password for accessing this voice mail account, default is 123456. It’s also the password for extension to login his administration web page.

3) Mail Setting

Enable Sending Voice Mail

Check this option to enable PBX send new voicemail to Email address below as an attachment.

Email Address

The Email address that new voicemail will be send to when Enable Sending Voice Mail is enabled and PBX settings -> SMTP settings is right set.

4) Flash

Hook Flash Detection Time

Sets the amount of time, in milliseconds, that the hook-flash must remain depressed in order for asterisk to consider such an event a valid flash event. The default value of it is 1250 ms and it can be configured in 1 ms increments.

Sequential Hook Flash Interval

Sets the amount of time, in milliseconds, that must have passed since the last hook-flash event received by asterisk before it will recognize a second event. If a second event occurs in less time than defined in here, then asterisk will ignore the event. The default value is 750 ms, and it can be configured in 1 ms increments.

5) Follow Me

Follow me is a feature to let an incoming call to a called party to be redirected to a third party, the third party can be a voicemail box, ring group, mobile telephone and so on.

When callee is No Answer / Busy / Unreachable, incoming calls will go to voicemail by default, if voicemail is disabled, call will be hung up.

6) Other Options

Pickup Group

Allows extension to answer someone else’s telephone call by dialing the group call pickup code (defined in **PBX Settings**->**Feature Codes**->**General**), the two extensions must be in a same pickup group.

APBX supports 10 pickup groups: 0-10, **None** means the extension belongs to none pickup group, extensions in group None can’t pick up others’ ring call and also can’t be picked up by others.

Call Waiting

Check this option to enable the Call Waiting capability for this extension. Then the extension can answer a new call when it is already on the line. If this Option is checked, the follow me option “When busy” will be unavailable.

Ring Out

Set the ring timeout for this extension. APBX will stop ringing the extension if the time is up and there is still no answer.

Use Web Interface

When checked, user can login the administration web page of this extension with extension number and voice mail pin code as username and password.

Storage Quota Privilege

Set capacity of disk space for this extension to store voicemail and call recording.

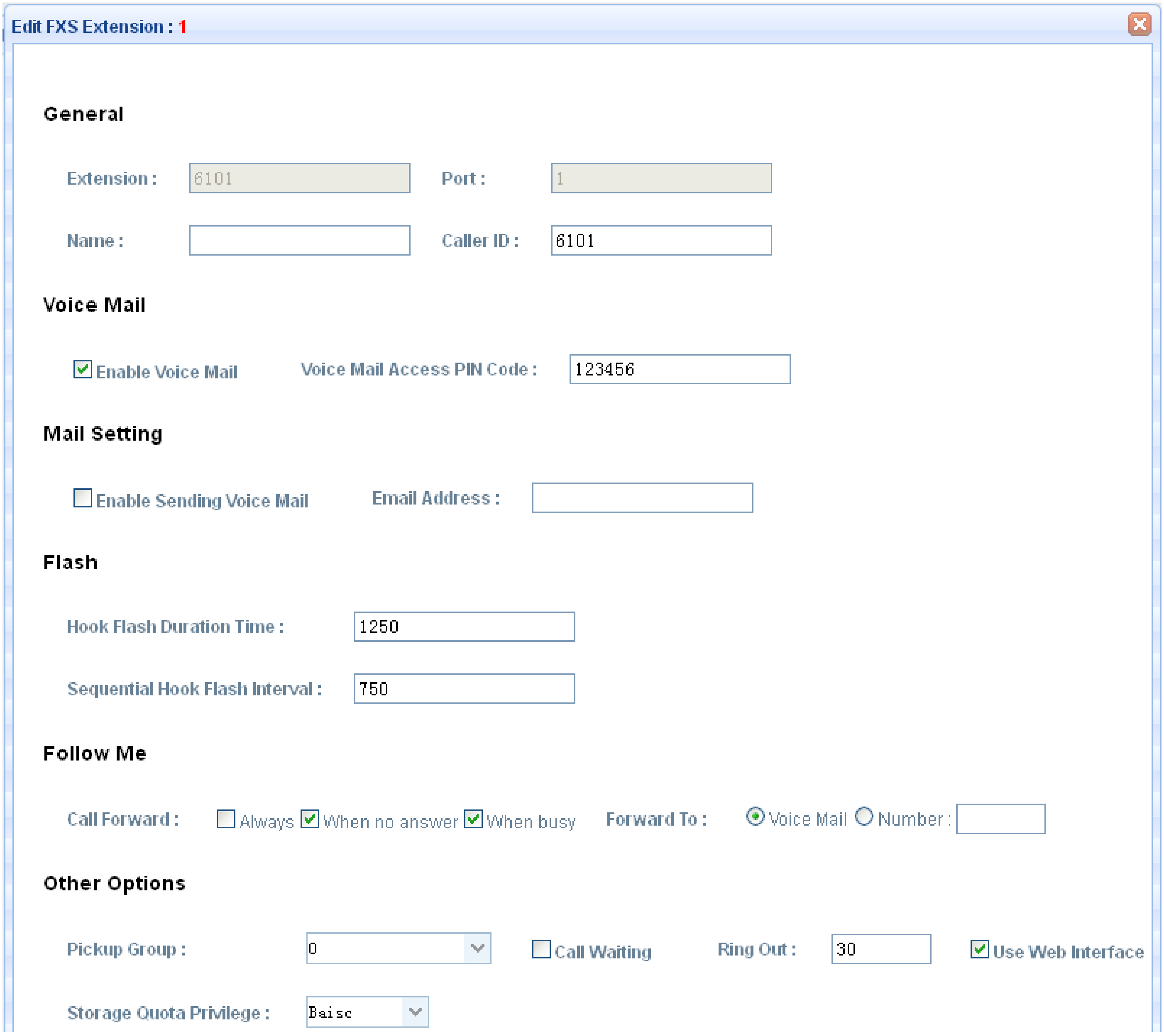
Restricted: 1 M

Basic: 2 M

Regular: 3 M

Privileged: 4 M

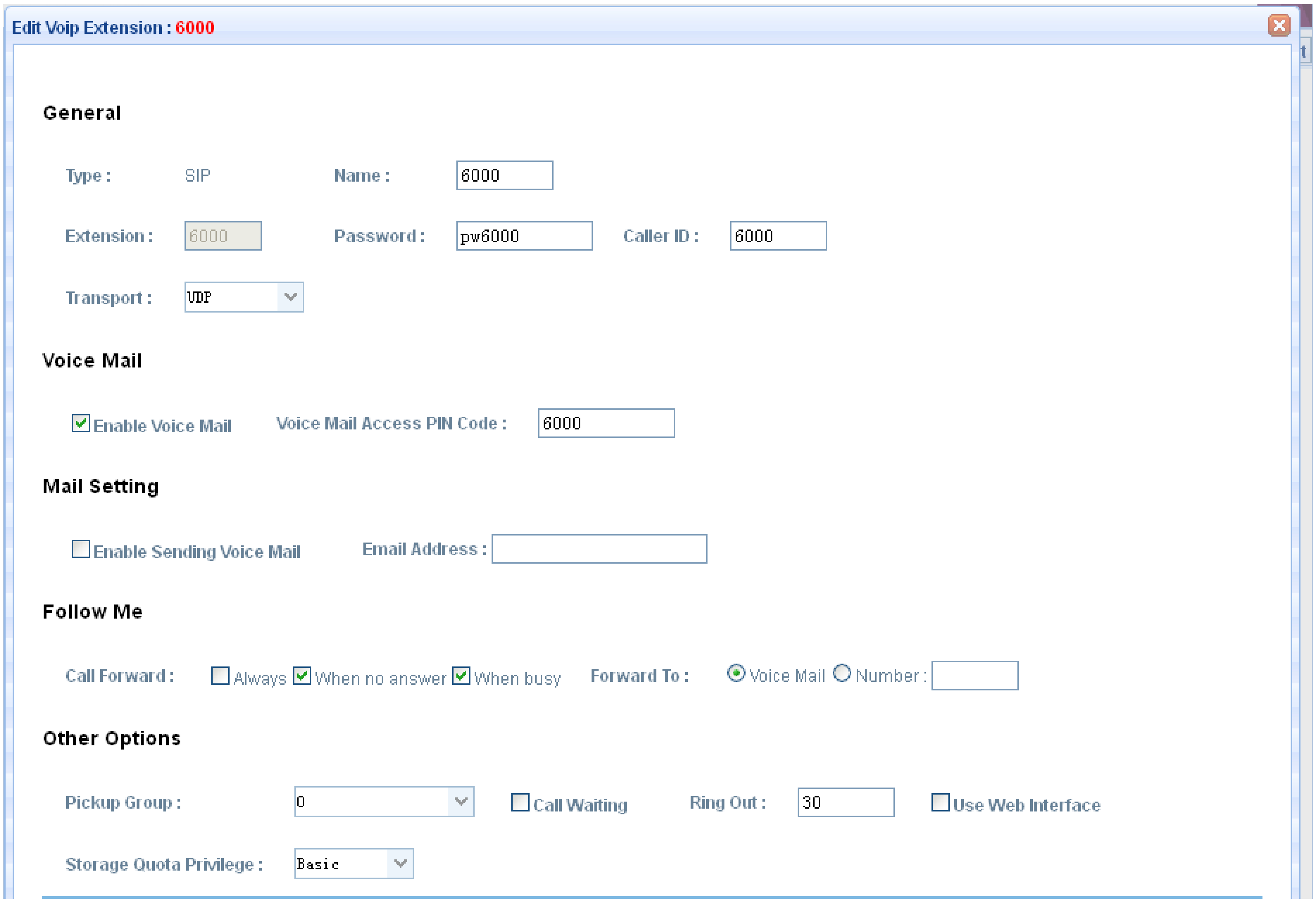
Super: 5 M



3.2 SIP extensions

SIP extension is an SIP account that allows IP phone or softphone to register to. It can be created / modified / deleted one by one or in batch.

Click [Create New Extension] to add an extension or [Add Multiple Extensions] to add multiple extensions. The extension number range is defined in **PBX Settings** -> **Options** -> **Extension Preference**, changing it can create extensions in others number range.



1) General

Name

A character-based name for this extension, i.e. 'Bob Jones'

Extension

Extension number, i.e. 6000, it is associated with this particular user / phone.

Password

Authentication for SIP phone to register and make calls.

Caller ID

CID showed in the other's phone during a call, default is Extension.

Transport

The transplant protocol type for VoIP data package, default is UDP. Please make sure TCP is enabled in PBX Settings -> SIP Settings before using TCP.

2) Voice Mail

Enable Voice Mail

Check this option to enable voice mail account for the extension. Enabled by default.

Voice Mail Access PIN Code

Password for accessing this voice mail account, default is the extension number. It's also the password for extension to login his administration web page.

3) Mail Setting

Enable Sending Voice Mail

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Email Address

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5) Other Options

Pickup Group

Allows extension to answer someone else's telephone call by dialing the group call pickup code (defined in **PBX Settings**->**Feature Codes**->**General**), the two extensions must be in a same pickup group.

APBX supports 10 pickup groups: 0-10, **None** means the extension belongs to none pickup group, extensions in group None can't pick up others' ring call and also can't be picked up by others.

Call Waiting

Check this option to enable the Call Waiting capability for this extension. Then the extension can answer a new call when it is already on the line. It also needs the call waiting support of IP phone. If this Option is checked, the follow me option "When busy" will be unavailable.

Ring Out

Set the ring timeout for this extension. APBX will stop ringing the extension if the time is up and there is still no answer.

Use Web Interface

When checked, user can login the administration web page of this extension with extension number and voice mail pin code as username and password.

Storage Quota Privilege

Set capacity of disk space for this extension to store voicemail and call recording.

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Regular: 3 M

Privileged: 4 M

Super: 5 M

6) Advanced Configuration

SIP Settings

NAT

Try this setting when APBX 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.

Can Reinvite

By default, Asterisk will route the media streams 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.

DTMF Mode

Select DTMF sending mode, there are three modes: **rfc2833**, **inband**, **info**. The DTMF setting in here should be as same as that in SIP phone, otherwise APBX will not detect the users' input correctly during a call.

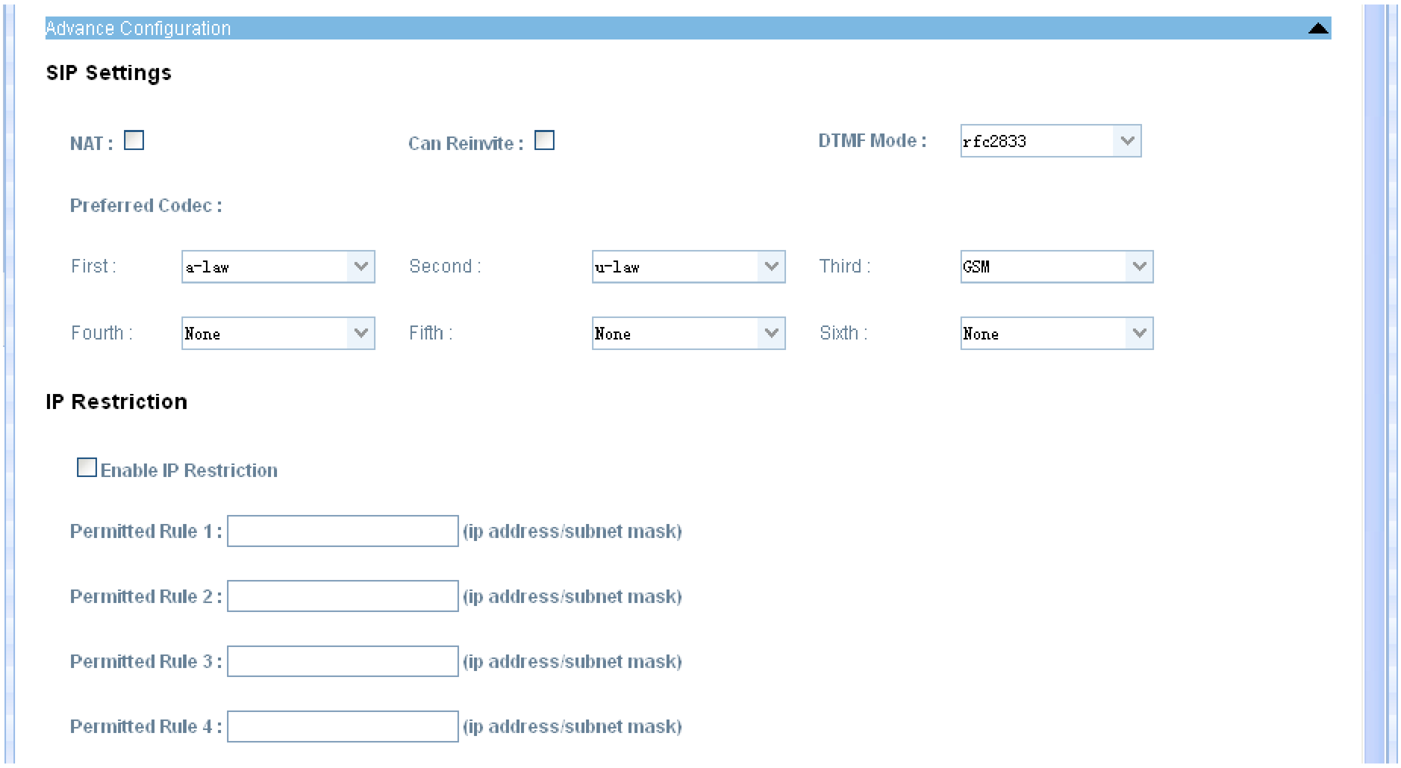
Auto means IPPBX will match anyone of them according to the setting of SIP phone.

Preferred Codec

Set the allowed codec and priority for SIP phone. The options are below:

A-law, **U-law**, **GSM**, **SPEEX**, **G726**, **G722**, **ADPCM**, **G729**

NOTE: There must be at least one same codec chose in APBX extension settings and SIP phone codec settings, otherwise, It's impossible to make calls between APBX and SIP phone.



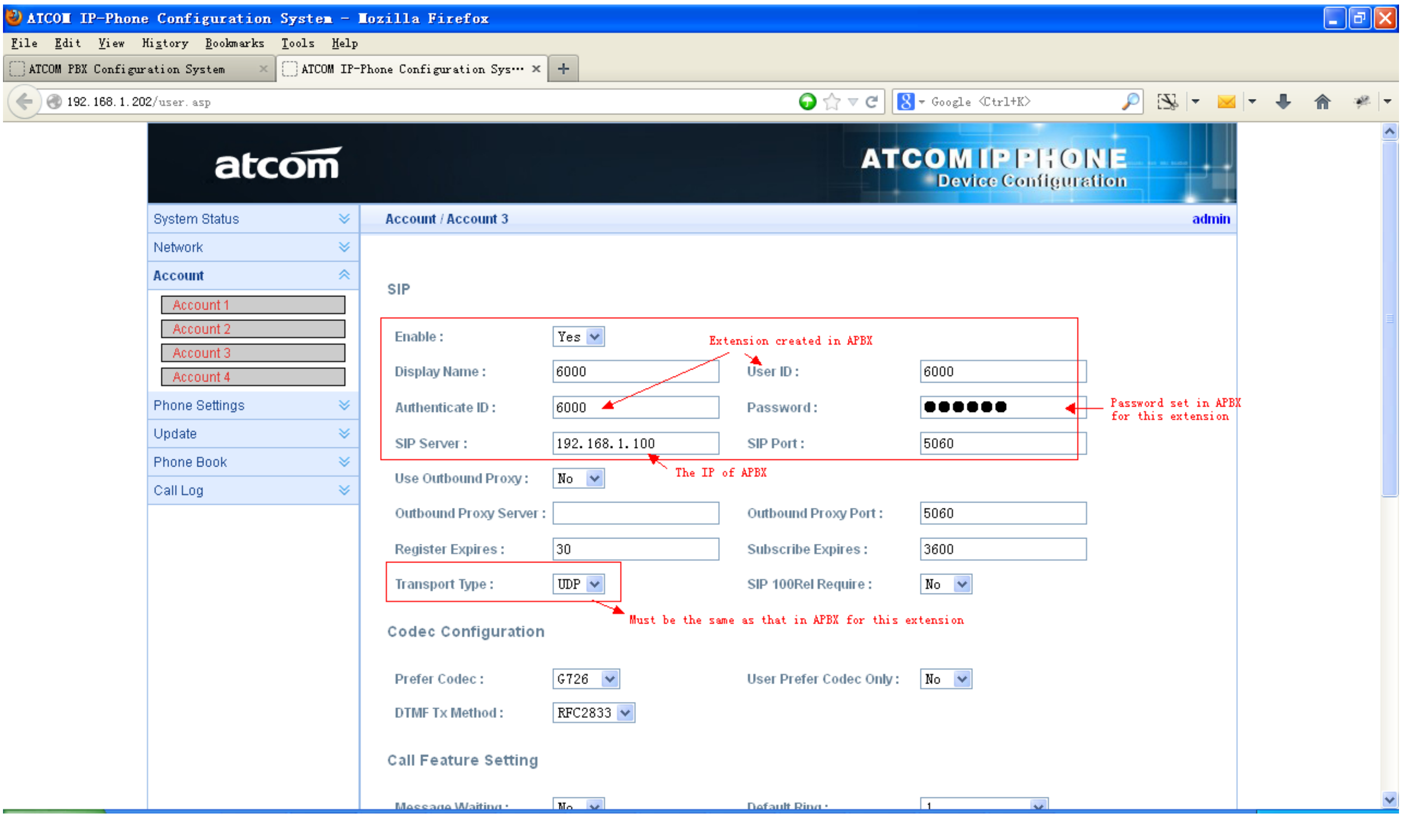
IP Restriction

Enable it to permit trusted IP/network register to this extension number. This is an useful way to improve the security of APBX.

Set trusted IP: xx.xx.xx.xx/255.255.255.255, for example: 192.168.1.160/255.255.255.255

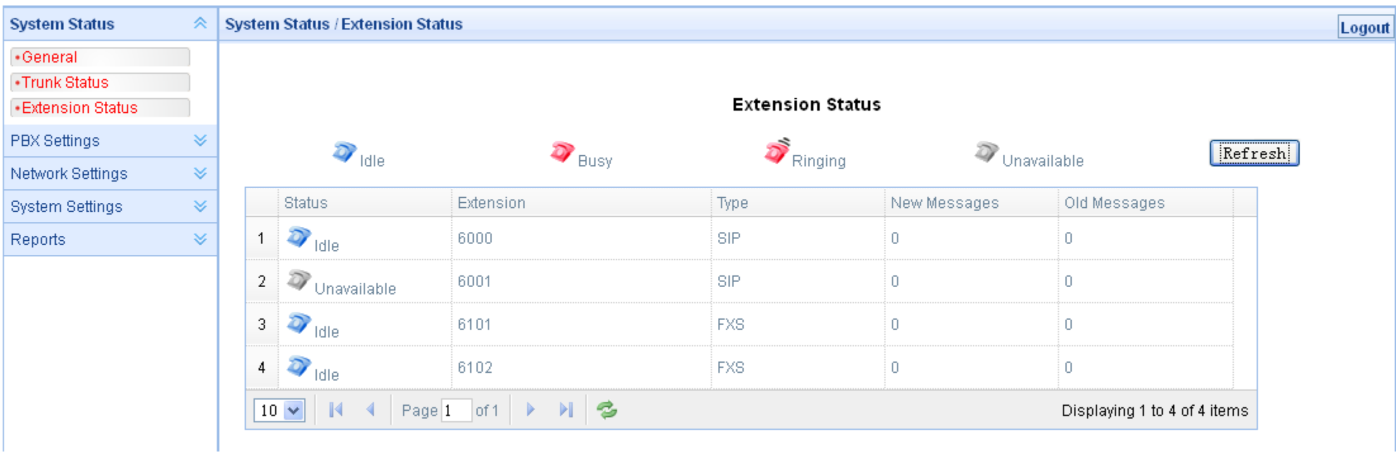
Set trusted network: xx.xx.xx.xx/subnet mask, for example: 192.168.1.0/255.255.255.0

3.3 Register onto APBX with your IP phone



After successfully register with 6000 and 6001, you can make interior calls among 6000, 6001, 6101(FXS), 6102(FXS) now.

3.4 Extensions Status



This page is used to check the extensions status.

Idle: The extension is registered and idle.

Busy: The extension is on the phone.

Ringing: The extension is ringing.

Unavailable: The extension is not registered and unreachable.

If this page response slowly, please be patient to wait the output before check other pages. Otherwise, other pages cannot be displayed correctly since APBX is accessing database while status checking, and database is locked for other pages' request.

3.5 Feature Codes

1) General

Call Recording

Record a call while in the call.

Dial Call Recording Code to begin recording and dial it again to stop recording during a call.

Checking Voicemail

Users can check their Voicemail by dialing this code on their phone.

Attended Transfer

Routed a call to a third party only if the third party answers the call. The call flow should be like below:

1. Phone A call B, B answers the call.
2. B presses feature code(*3) and C's number to transfer the call to C
3. If C answers B's call, B can talk to C and A is on hold
4. If Phone B hangs up, A will talk to C, transfer is successful.

4' If Phone C hangs up, B connects back to A, transfer is failed

Blind Transfer

Blind transfer is when a call is routed to a third party, the original call is ended, and no check is made to determine whether the transferred call is answered or if the number is busy.

The call flow should be like below:

1. Phone A call B, B answers the call.
2. B presses feature code(*03) and C's number to transfer the call to C

Group Call Pickup

Pick up a ring call for other extensions in the same pickup group.

The call flow should be like below:

1. C calls A, phone A is ring, but A is not at his/her seat.
2. Extension A and B are in the same pick up group, B can dial Group Call Pickup code to pick up the ring call, and talk to C.

Direct Call Pickup

Pick up a ring call for an appointed extension.

The call flow should be like below:

1. C calls A, phone A is ring, but A is not at his/her seat.
2. B can dial Direct Call Pickup code + A's extension number to pick up the ring call, and talk to C.

Intercom

Connect directly to a specified phone.

The call flow should be like below:

1. A dial Intercom code + B's extension nubmer.
2. If Phone B supports page/intercom, it will answer the call automatically.

2) Call Park

It allows a person to park a call on IPPBX and continue the conversation from any other telephone set.

The call flow should be like below:

1. A and B are on the conversation.
2. A dial call park code (e.g. *6), PBX will tell A a park extension (e.g. 701) and then hang up the call. B is parked on PBX.
3. C dial park extension: 701, PBX will bridge C and B.

3) Call Forward

Users can configure their follow me settings via their phones.

Reset to Defaults

Reset follow me settings by dialing *70 (default code, can be changed). After dialing in, PBX will prompt a "beep", then the setting is completed and the call will be hung up.

NOTE: Default Follow Me settings are as below:
Always: Disabled ☐
When no answer: Enabled ☑
When busy: Enabled ☑
Forward to: Voice Mail

Enable/Cancel Unconditional Call Forward

Enable/Disable call forward Always function.

Enable/Cancel Call Forward On Busy

Enable/Disable call forward When Busy function.

Enable/Cancel Call Forward On No-Answer

Enable/Disable call forward When No Answer function.

Call Forward to Number

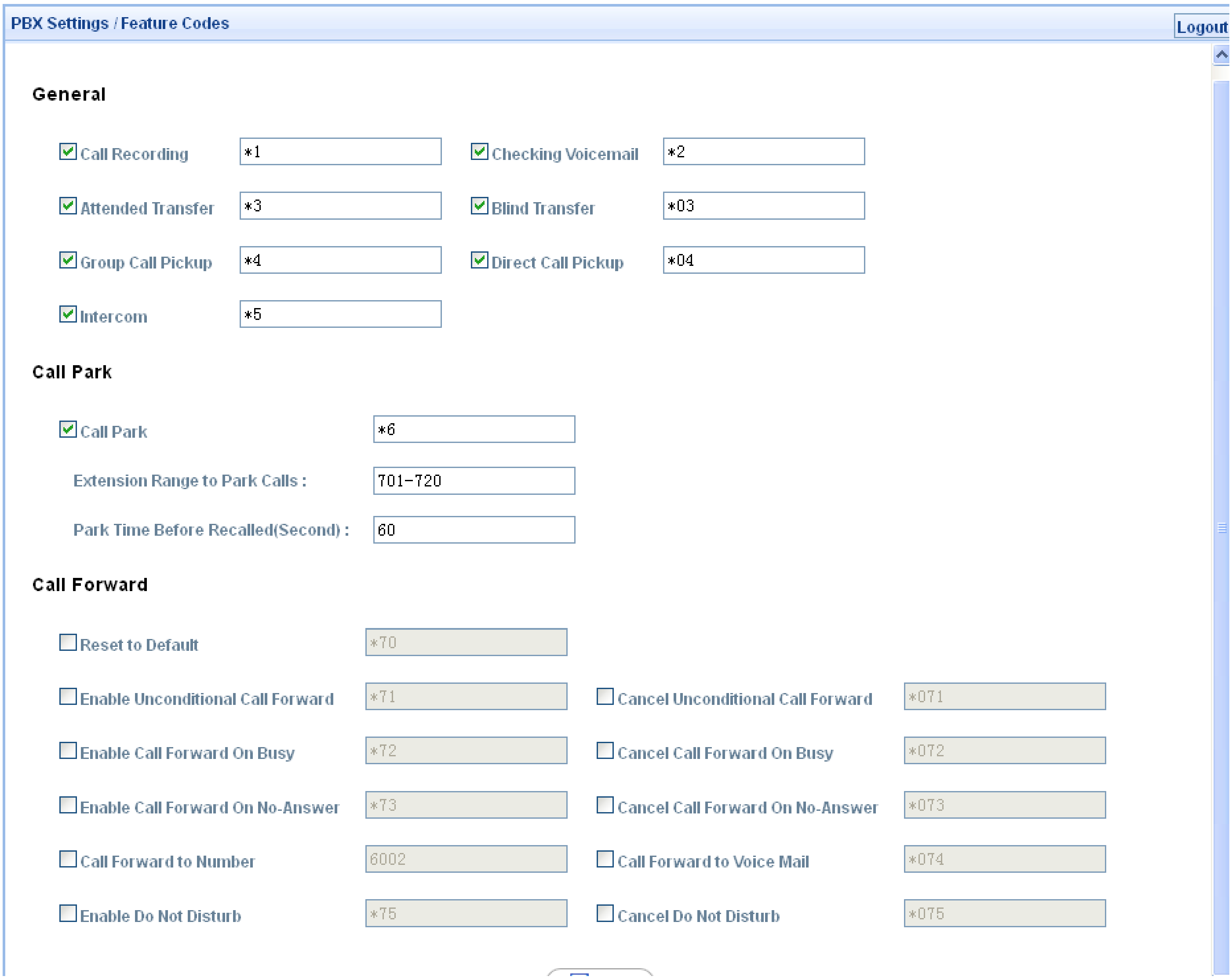
Set the destination for call forward to number by dialing *74 (default code, can be changed), if the number is not set yet, dial *74+number to set it.

Call Forward to Voice Mail

Set the destination for call forward to voicemail.

Enable/Cancel Do Not Disturb

Enable/Disable Do Not Disturb function



3.6 SMTP Settings

1) Voice Mail to Email Setting

Email Address

The Sender Email Address APBX used to send voicemail.

Password

The password for above Email Address/Account.

SMTP Server

SMTP server that above email address/account is located in.

Port

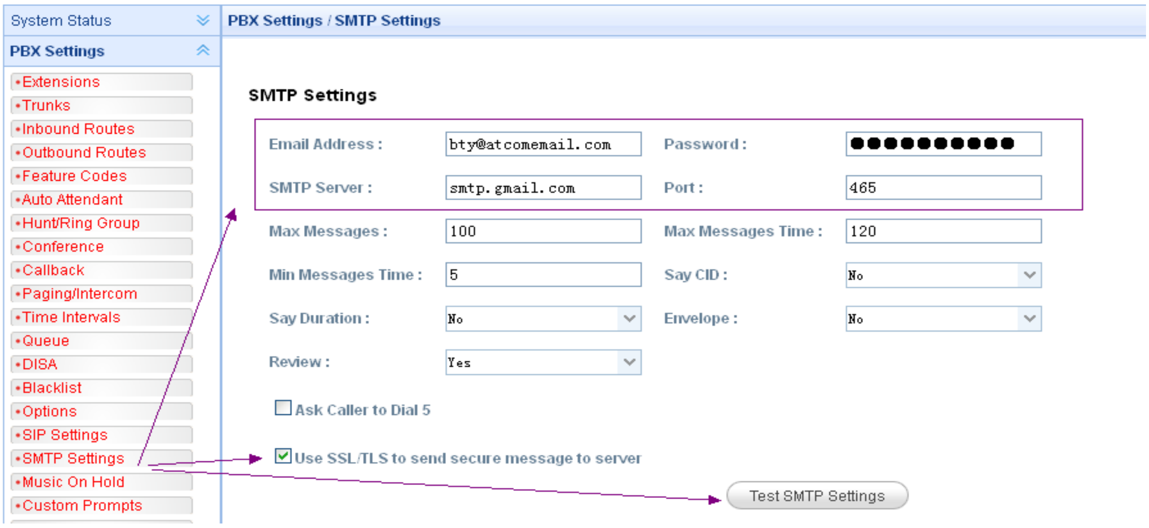
Port for SMTP server, for example: Gmail server use port 465 to send / receive email.

Use SSL/TLS to send secure message to server

Some servers need to authenticate sender before sending email, then the box should be checked.

Test SMTP Settings

Check whether the SMTP setup is OK. PBX will send an email to the test email address using above SMTP setting information. If the test failed, please check that information and network connection.



NOTE: After SMTP setting, please set Email address for each extension to achieve Voicemail to Email function.

2) Voice Mail Setting

Max Messages

This limits the number of messages in a voicemail folder. The maximum value is 9999 (hard coded) and the default 100. When a mailbox has more than this number of messages in it, new messages can not be recorded and “voice mail box is full” is played to the caller.

Max Messages Time

This defines the maximum amount of time in seconds of an incoming message. Use this when there are many users and disk space is limited. The default value is 120 (2 minutes), 0 means there will be no maximum time limit enforced.

Min Messages Time

This setting can be used to eliminate messages which are shorter than a given amount of time in seconds. The default value for this setting is 5.

Say CID/Duration

Read back caller’s telephone number / message duration prior to playing the incoming message when checking it.

Envelope

Envelope controls whether or not Asterisk will play the message envelope (date/time) before playing the voicemail message.

Review

Let a caller review their message before committing it to a mailbox.

Ask Caller to Dial 5

If this option is set, the caller will be prompted to press 5 before leaving a message.

3.7 Conference

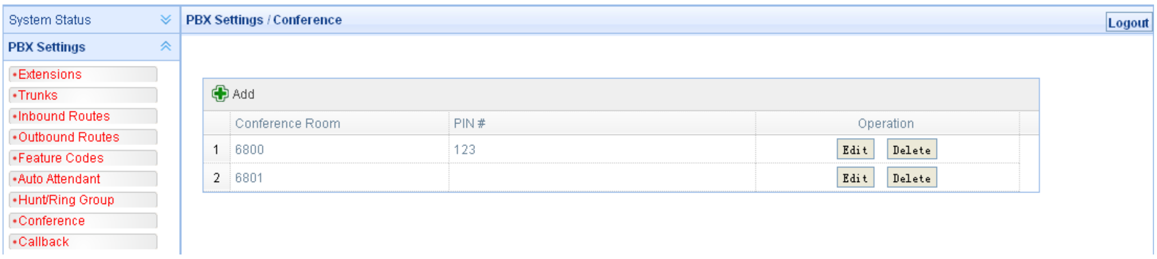
Allows participants dial into a virtual meeting room from their own phone, support up to 20 participants.

Conference Room

Extension number of conference room, participant dial it to get into the room.

PIN#

Used for authentication before participants dial into the room, APBX will playback MoH for the first participant.



3.8 Paging / Intercom

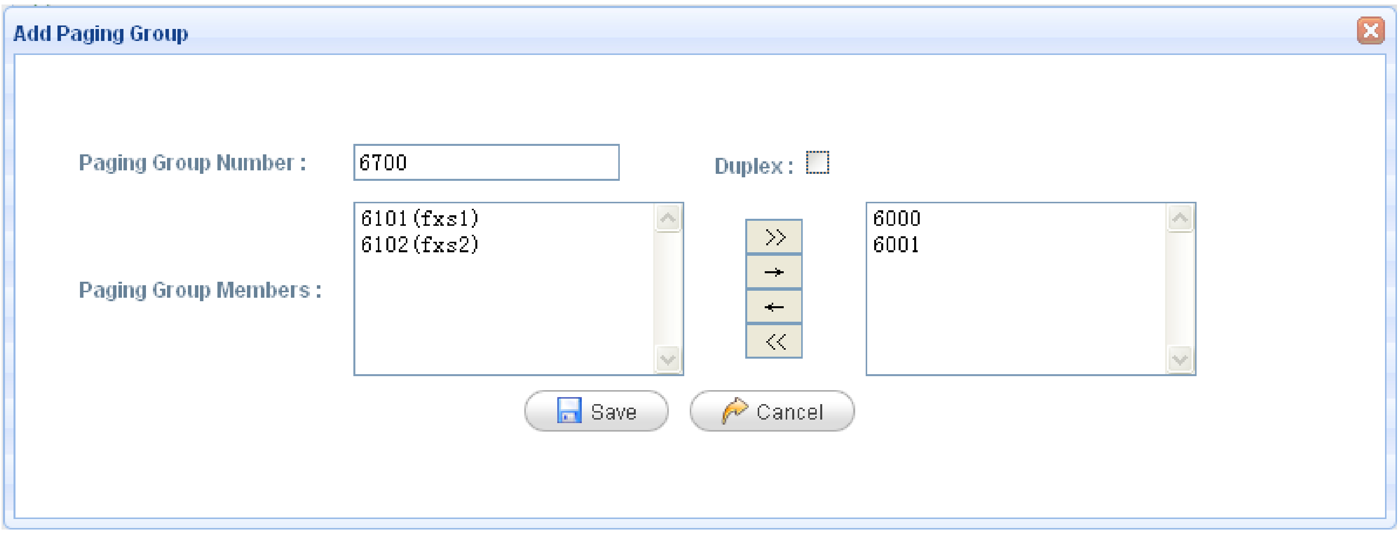
Dial a code and connect directly to a built-in two-way announcement and talkback function on one or more phones, support up to 20 participants.

Paging Group Number

Extension number of paging group, dial it to reach this group.

Duplex

If checked, caller and callees all can speak and hear. Otherwise, only caller can speak, and callees can hear.



3.9 Options

1) General Preference

Ring Timeout

Default Ring Timeout for an extension if Ring Out for it is not set.

Max Call Duration

This defines the maximum amount of time in seconds for a interior call, 0 means no limit, default is 6000s.

Music On Hold

This define which Music on hold is used when transfer/call park/on hold/Conference etc.

Tone Region

This defines how the default dial tone, busy tone, and ring tone look like, please select your country or nearest neighboring country here.

2) Extension Preference.

Defines the range for SIP / FXS / Ring Group / Voice Menu / Paging Group / Conference / Queue Extensions. The extension length must be between 3 and 9 digits. The maximum quantity can be supported for each are as below:

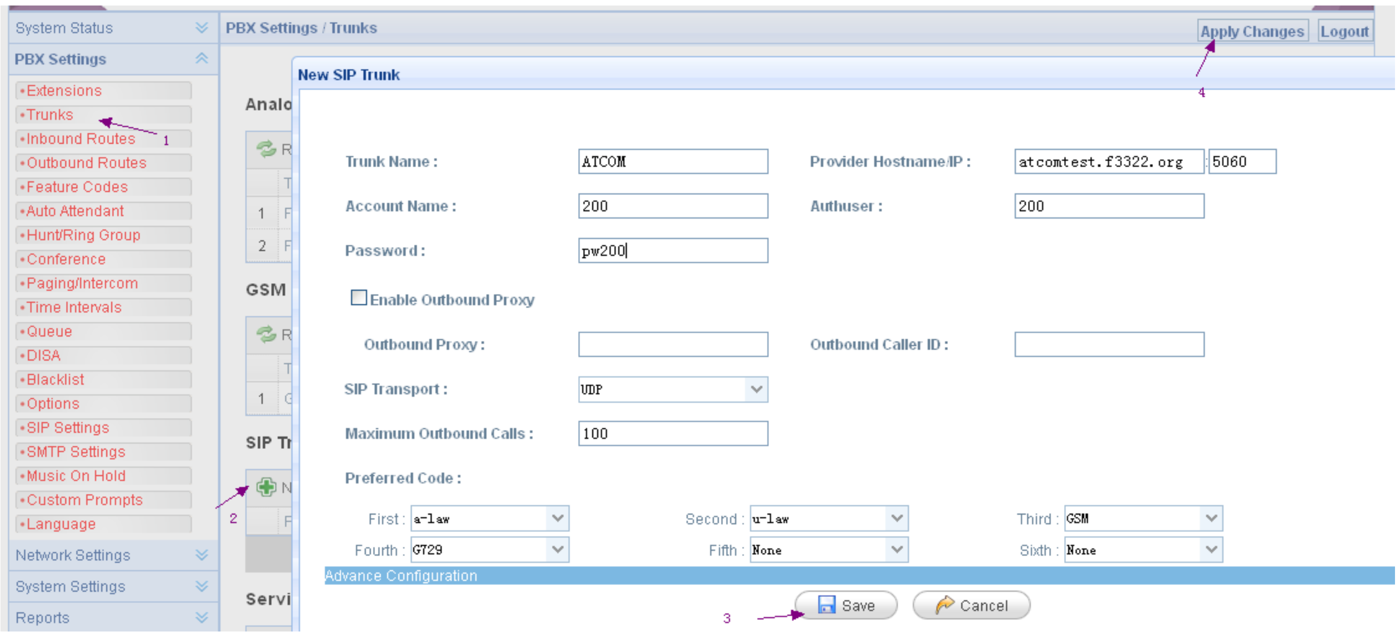
SIP extension	100
FXS extension	8
Ring Group	9
Voice menu/IVR	16
Paging Group	9
Conference	9
Queue	9

4. Create SIP trunk and make outbound / inbound calls to / from VoIP provider

4.1 Create SIP trunks

Go to **PBX Settings** -> **Trunks**, Click 

to add a new SIP trunk.



Trunk Name

A unique label to help you identify the trunk.

Provider Hostname/IP

Hostname or IP of your VoIP provider, default port is 5060

Account Name

The username that your service provider configured

Authuser

The username that your service provider configured for authentication, generally, it’s same as Account Name.

Password

The password configured for the user in your service provider side.

Enable Outbound Proxy

Outbound Proxy is a SIP proxy server, it acts, like any proxy server, as a middleman between two communicating agents, serving as a transit point for all SIP traffic. It can be used to solve the SIP one-way-audio issue.

Outbound Caller ID

The Caller ID used when using outbound proxy.

SIP Transport

The transplant protocol type for VoIP data package, default is UDP. Please make sure TCP is enabled in PBX Settings -> SIP Settings before using TCP.

Maximum Outbound Calls

Define the maximum quantity of outbound connections (simultaneous calls) that can be used on this trunk. Inbound calls are not counted in. 0 means no connection limit.

Preferred Code

Set the allowed codec and priority for this trunk.

Advance Configuration

DOD(Direct Outward Dialing Number) Setting

Set the Outbound number for different extensions.

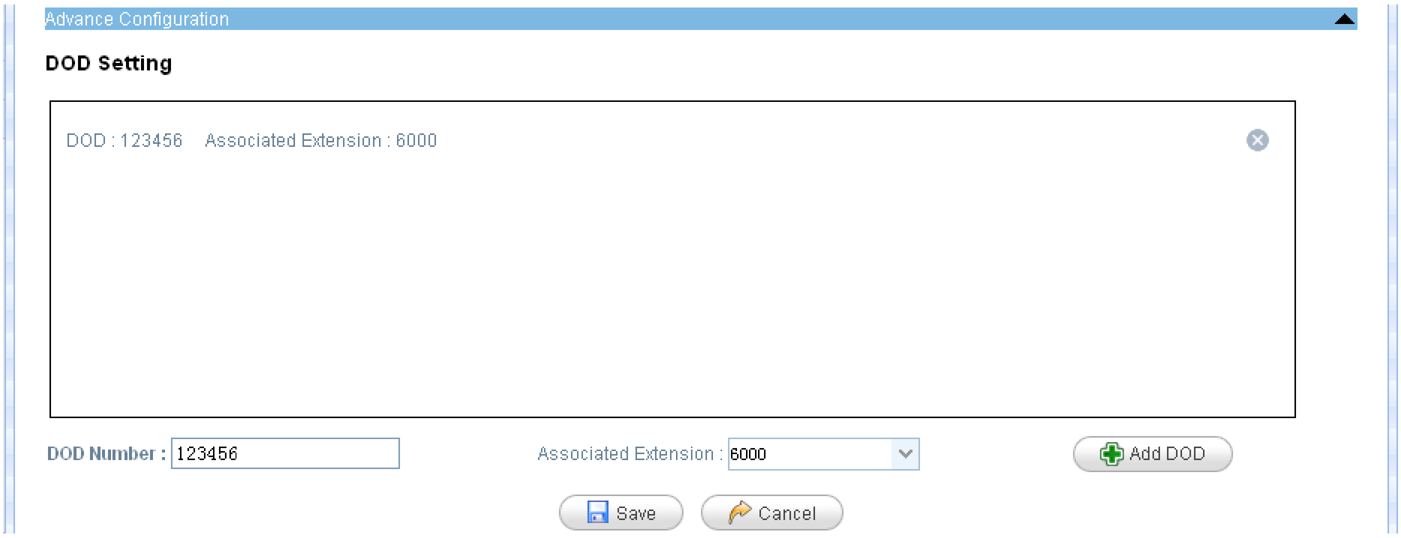
For example:

< === Do not set DOD,

The other end of the trunk will show original (interior) extension number.

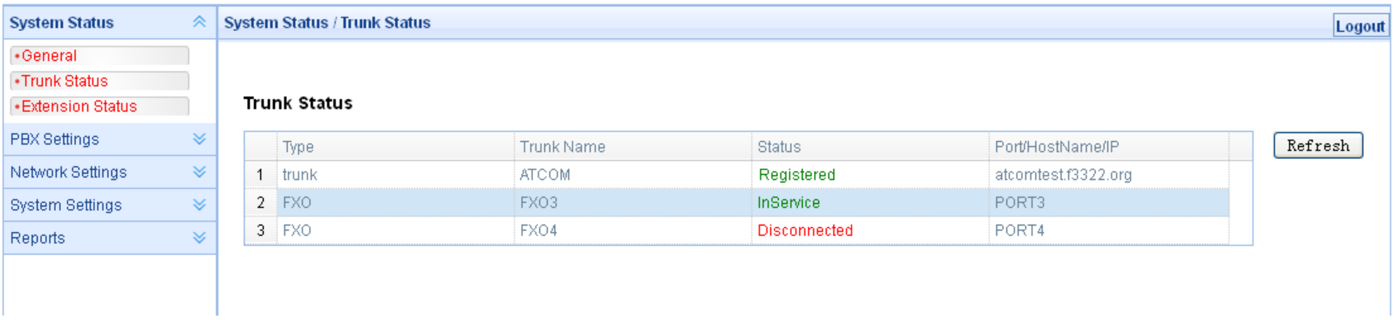
DOD is set ==== >

The other end of the trunk will show DOD number.



4.2 Check SIP Trunk Status

After creating trunk, go to **System Status** -> **Trunk Status** to check the SIP trunk Status, make sure it's registered.



4.3 Make outbound calls

Go to **PBX Settings** -> **Outbound Routes**, click  to add an outbound route.

Outbound Route Name

A unique label to help you identify the outbound route.

Dial Pattern

A filter for marching numbers you dial, the call will be forwarded out via Selected Trunks only when it matches the dial pattern here. In patterns, some characters have special meanings.

X means Any Digits from 0-9

Z means Any Digits from 1-9

N means Any Digits from 2-9

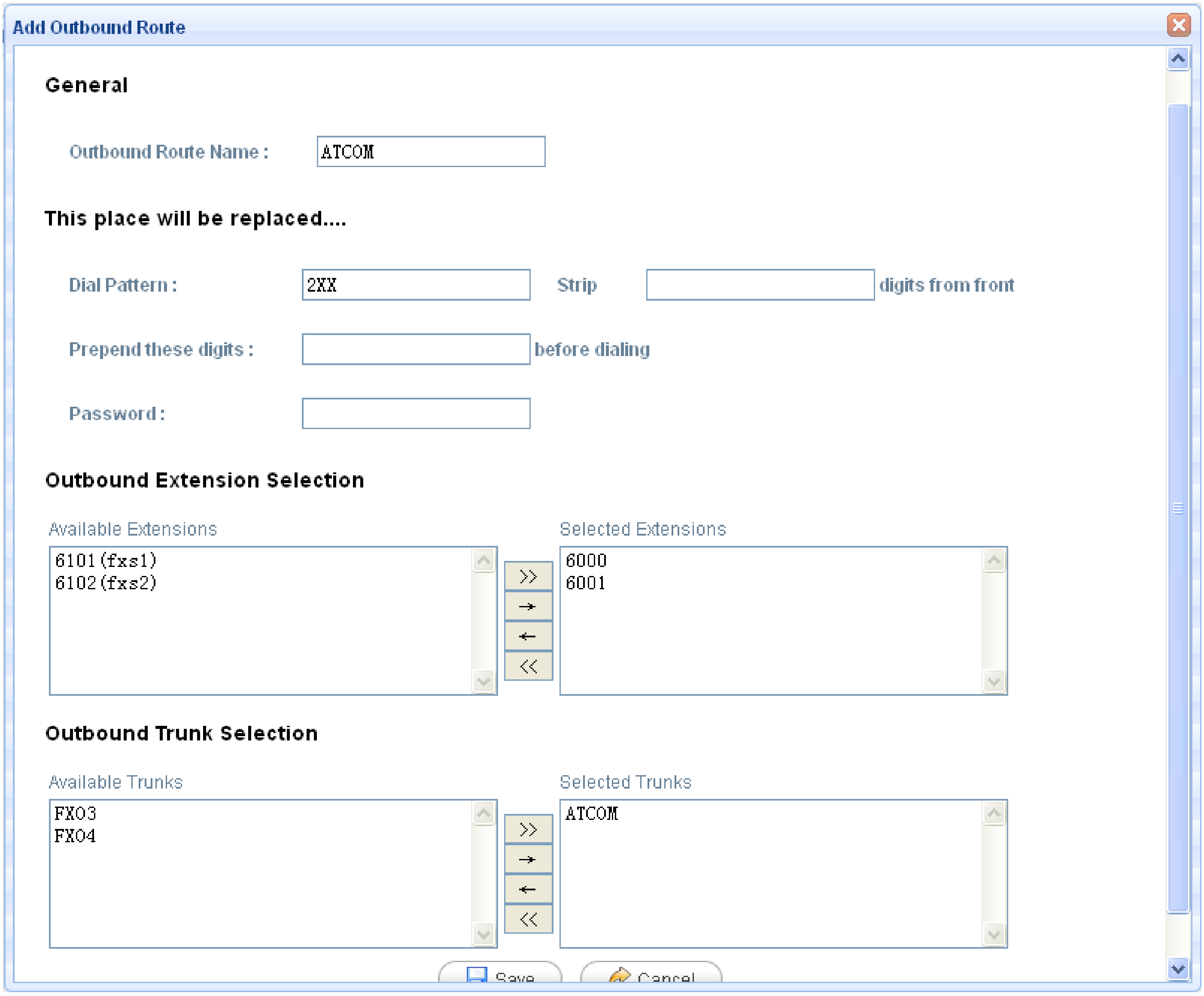
[1234-9] means Any Digits in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9)

. means one or more digits

! will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

For example: Once set Dial Pattern: 2XX Strip: 0, that means any calls to 200-299 will be forwarded out. Please do not simply set it to X., otherwise all telephone numbers with 2 or 2+ digits will be matched, this outbound route probably affect your interior calls, unless your local extensions is just a single figure.

Strip: The number of digits that will be stripped from the front of the dialing string before the call is placed via Selected Trunks. See example in Chapter 5.
Prepend these digits: Allows the user to specify digits that are prepended before the call is placed via the trunk. See example in Part 4.
Password: Authentication for Selected Extensions before dialing out.
Outbound Extension Selection: Select extensions which can dial out with this outbound route. In my case, only 6000 and 6001 can dial out with this trunk.
Outbound Trunk Selection: Select trunks which calls are forwarded out through.



4.4 Make inbound calls

Go to **PBX Settings** -> **Inbound Routes**, click 

to add an inbound route.

Caller ID

Define the Caller ID number to be matched on incoming calls. Leave this field blank to match any or no CID info. Special characters described in chapter 4.3 can be used here as same.

DID number

Define the expected DID number if your trunk passes DID on incoming calls. Leave this blank to match calls with any no DID info. Special characters described in chapter 4.3 can be used here as same.

Extension

Define the extension for DID number. This field is only valid when you use SIP trunk for this inbound router. You can only input number and '-' in this field, and the format can be XXX or XXX-XXX. The count of the number must be only one or equal the count of the DID number. Up to 100 DID numbers can be set.

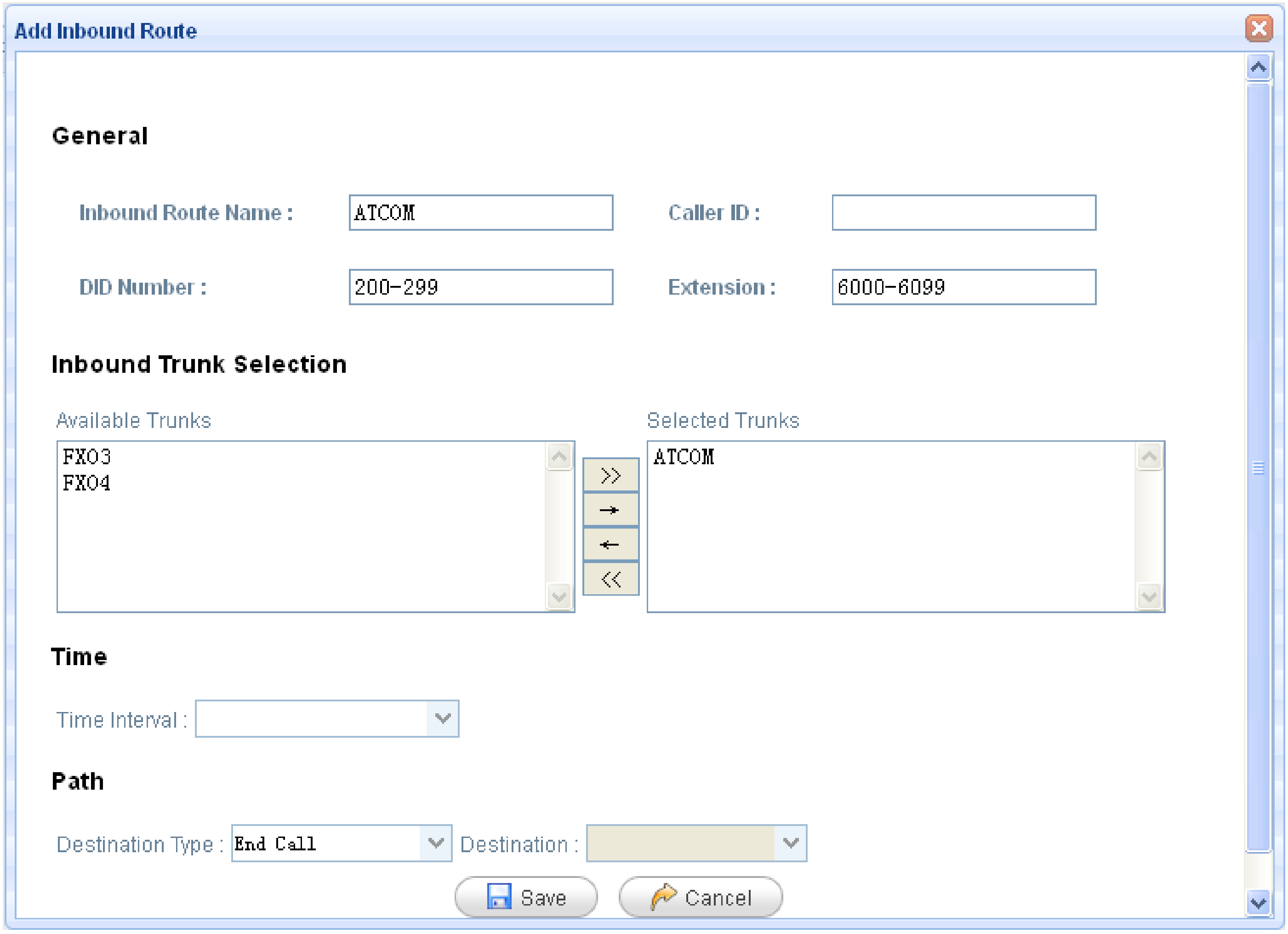
For example: Set DID number: 6000-6010，Extension: 6000，All inbound calls to 6000-6010 will be forwarded to extension 6000. Set DID number: 6000-6010, Extension: 6000-6010, inbound calls to 6000-6010 will be forwarded to corresponding extension.

Inbound Trunk Selection

Select the trunks for which this inbound route apply.

Time

Select appropriate time intervals for when this inbound route apply.



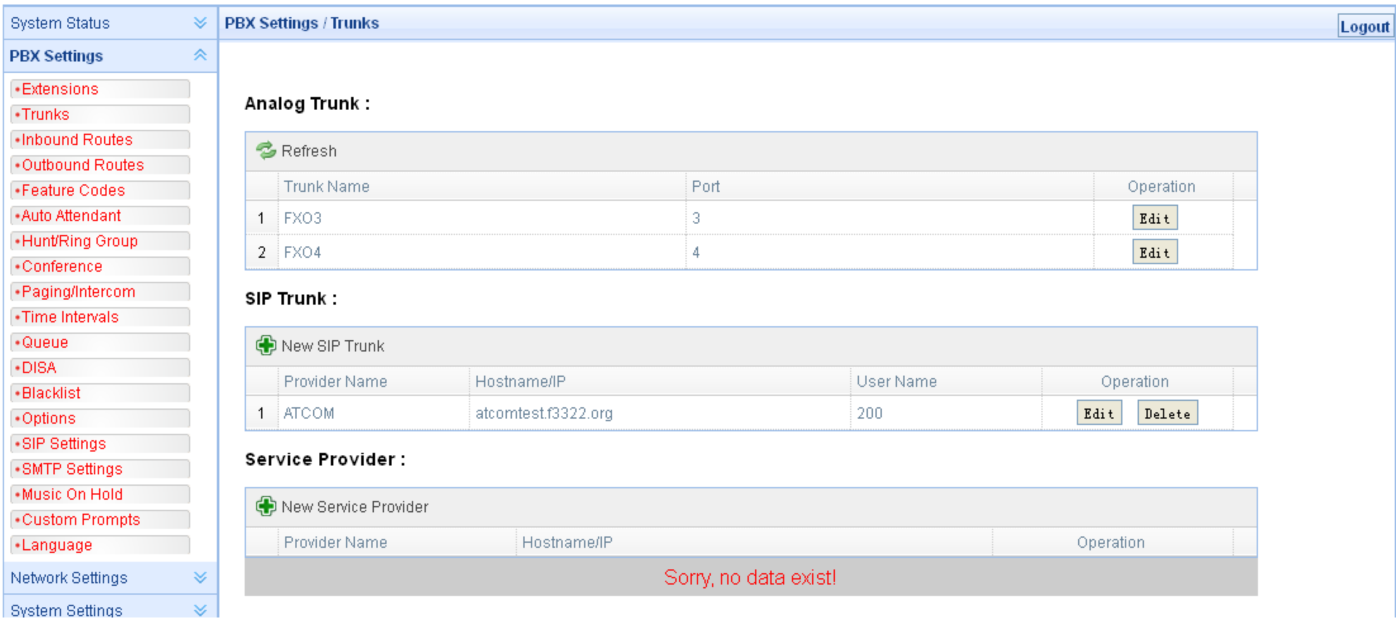
Path

Set the destination for incoming calls. If Extension is set, this option will not take effect.

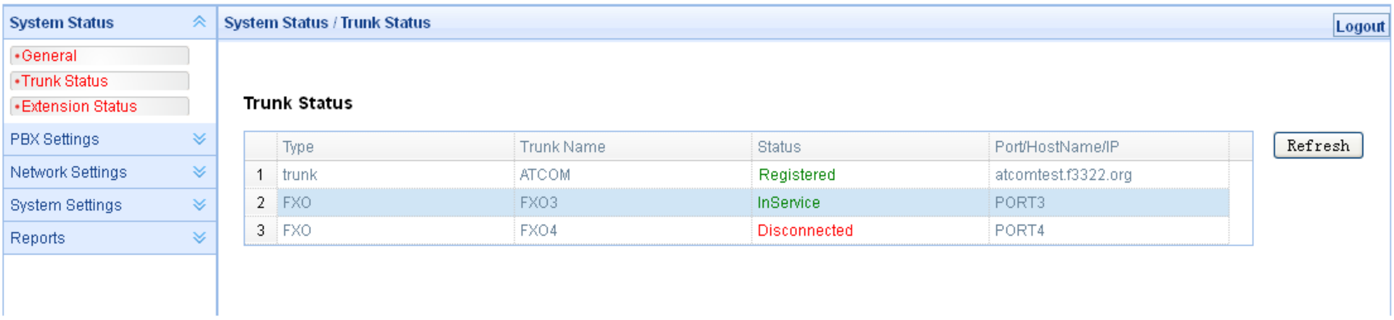
5. Make outbound / inbound calls to / from PSTN network

5.1 Make sure FXO modules are installed

If there are FXO modules installed in your APBX, APBX configures analog trunk automatically when they are detected. The module installed in IP04 can be: **AX110X**



NOTE: Before using them, please make sure FXO port is connected with PSTN line (InService). The connection status can be checked in **System Status -> Trunk Status**



5.2 Make outbound calls

Go to **PBX Settings** -> **Outbound Routes**, click 

to add an outbound route.

Outbound Route Name

A unique label to help you identify the outbound route.

Dial Pattern

A filter for marching numbers you dial, the call will be forwarded out via Selected Trunks only when it matches the dial pattern here. In patterns, some characters have special meanings.

X means Any Digits from 0-9

Z means Any Digits from 1-9

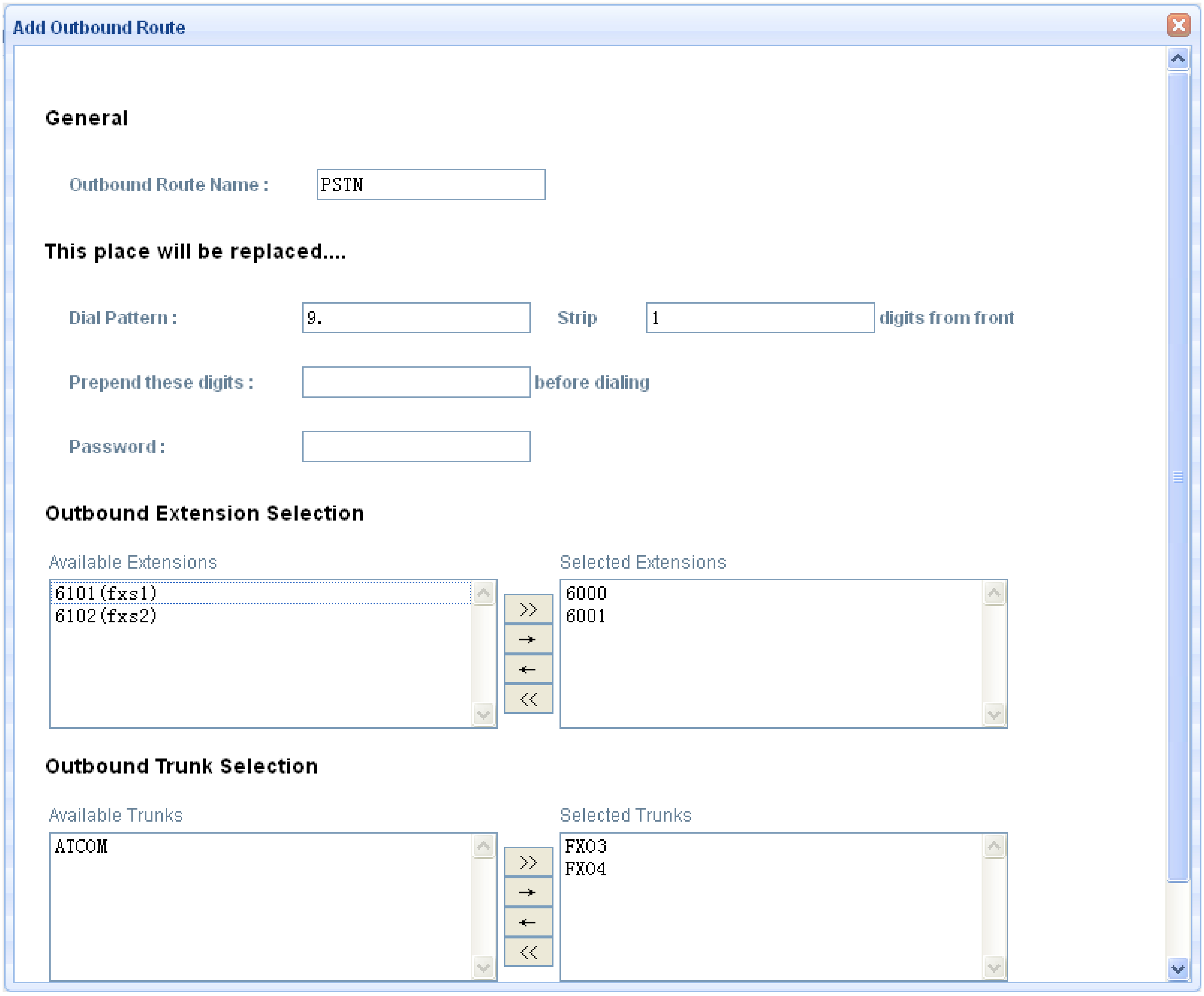
N means Any Digits from 2-9

[1234-9] means Any Digits in the brackets (in this example, 1, 2, 3, 4, 5, 6, 7, 8, 9)

. means one or more digits

Strip: The number of digits that will be stripped from the front of the dialing string before the call is placed via Selected Trunks.

For example: If set Dial Pattern: 9, Strip: 1, Prepend 123, user need to dial 94567 to dial PSTN number 1234567



Prepend these digits

Allows the user to specify digits that are prepended before the call is placed via the trunk.

Password

Authentication for Selected Extensions before dialing out.

Outbound Extension Selection

Select extensions which can dial out with this outbound route. In my case, only 6000 and 6001 can prefix 9 to dial out.

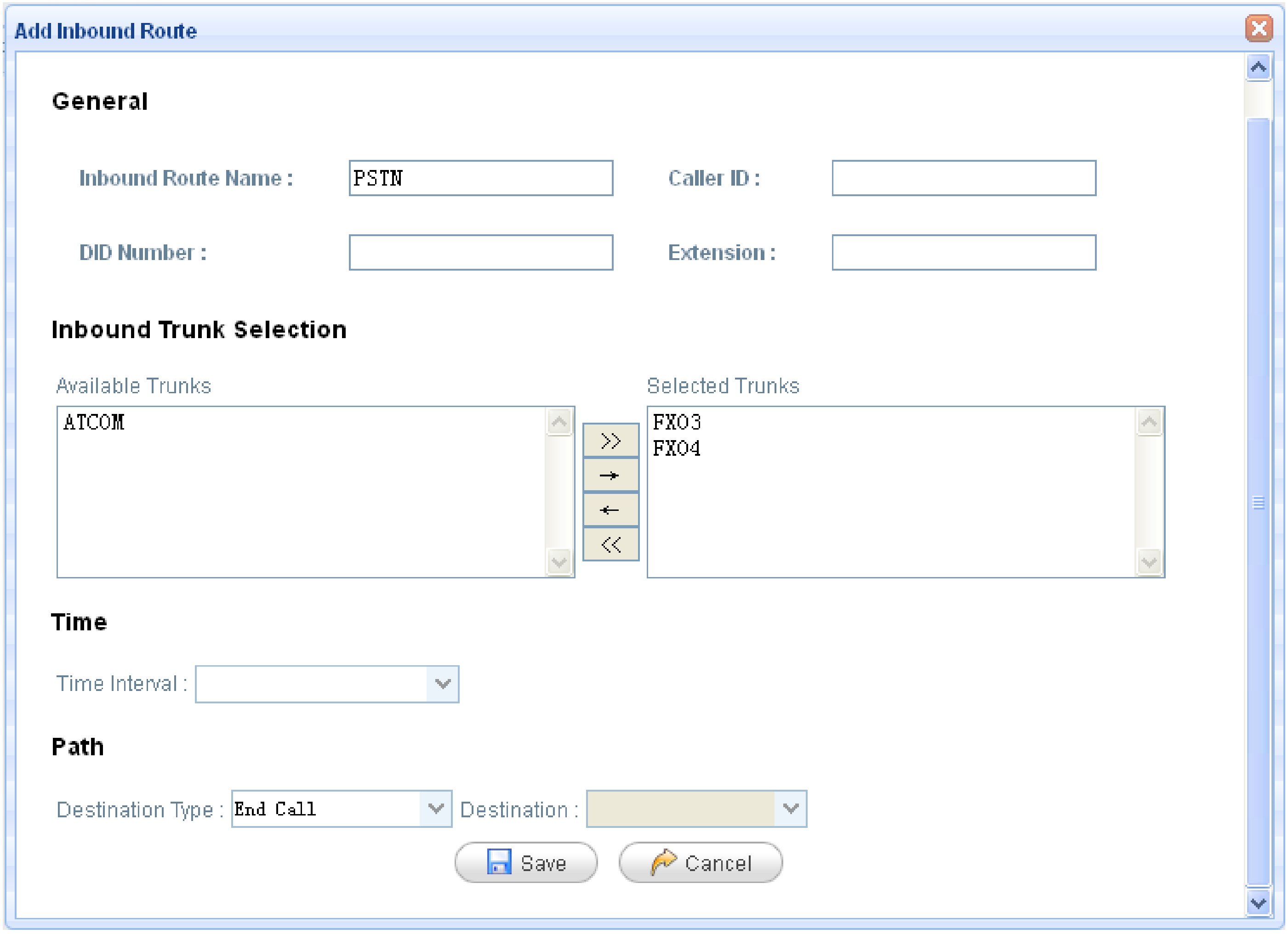
Outbound Trunk Selection

Select trunks which calls are forwarded out through. In my case, the call will be forwarded out via analog trunk FXO3 first, if failed, will try FXO4.

5.3 Make inbound calls

Go to **PBX Settings** -> **Inbound Routes**, click 

to add an inbound route. Just setting Selected Trunks and Path is OK.



Caller ID

Define the Caller ID number to be matched on incoming calls. Leave this field blank to match any or no CID info. Special characters described in chapter 5.2 can be used here as same.

DID number

Just leave it blank.

Extension

It’s unavailable for Analog trunk, leave it blank.

Inbound Trunk Selection

Select the trunks for which this inbound route apply.

Time

Select appropriate time intervals for when this inbound route apply.

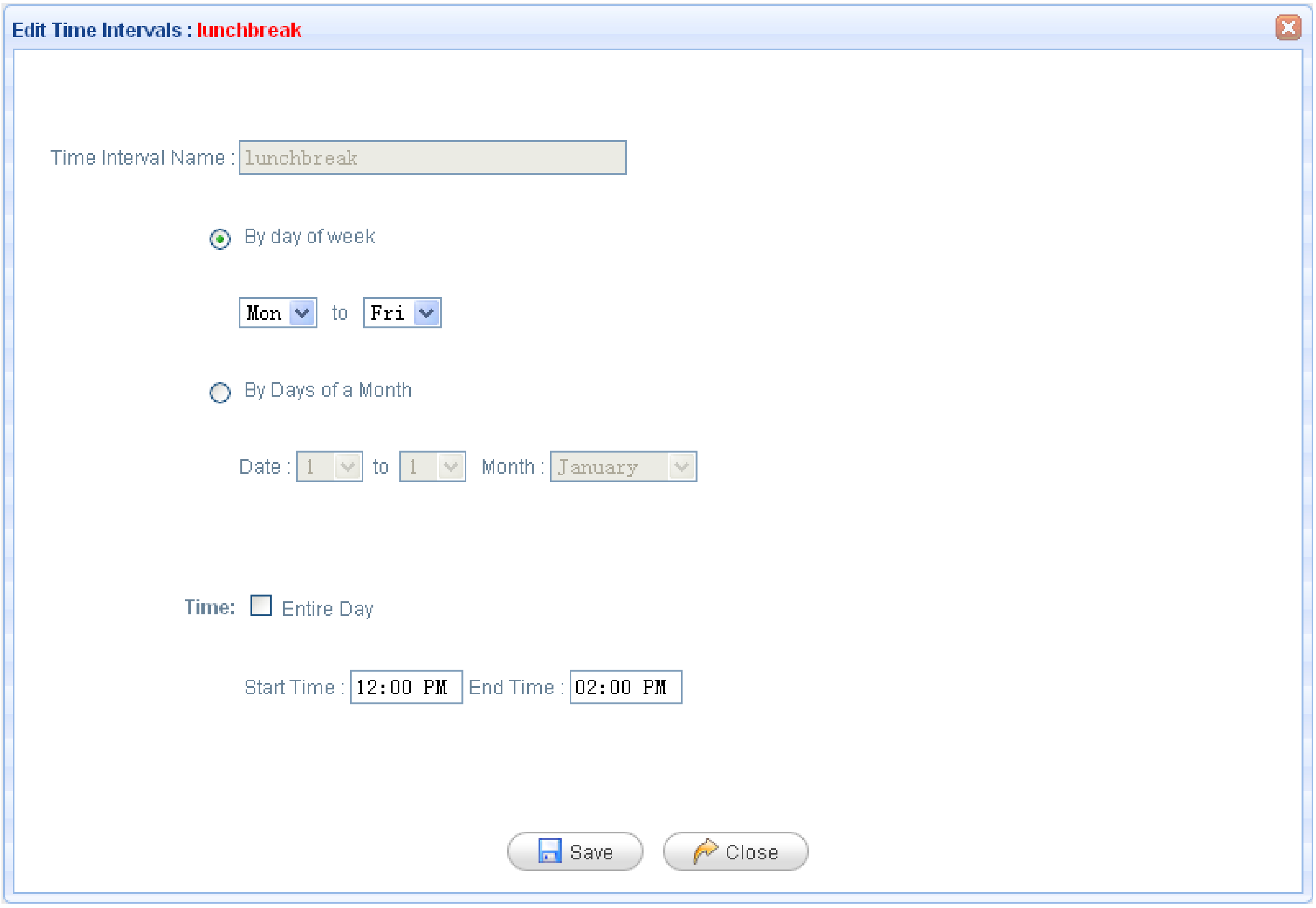
Path

Set the destination for incoming calls.

6. Inbound Call Control

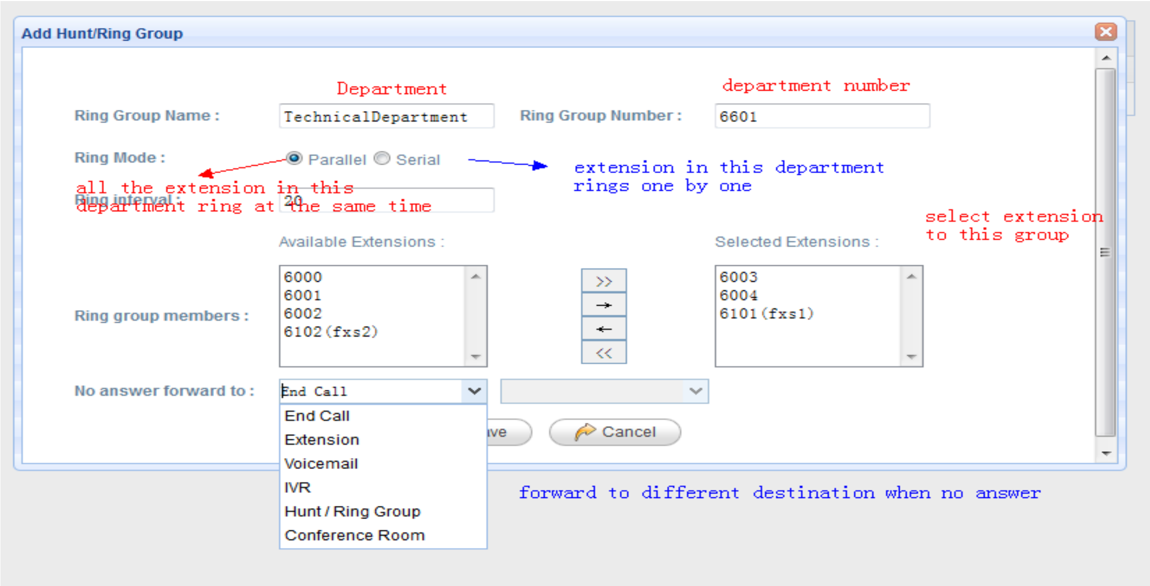
6.1 Time Interval

Set the Time Interval for inbound route.



6.2 Hunt / Ring Group

This defines a 'virtual' extension that rings a group of phones simultaneously / one by one, stopping until any one of them is picked up.



6.3 Queue

Usually used in Call Centre to queue customers for the next available operator.

1) General

Queue Name

Name of the queue

Queue Number

Extension number of the queue, dial it to get into the queue

Queue Password

Used as authentication for users before being dynamic agent.

Queue Agent Timeout

Ring timeout in seconds when calling an agent

Queue Max Wait Time

The maximum time in seconds for a caller can wait in the queue before being pulled out. (0 means unlimited)

Queue Ringing Strategy

Strategy for APBX ring the agents.

RingAll: Ring all available agents simultaneously until one answers.

LeastRecent: Ring agents which was least recently called.

FewestCalls: Ring agents with the fewest completed calls.

Random: Ring agents in a random way.

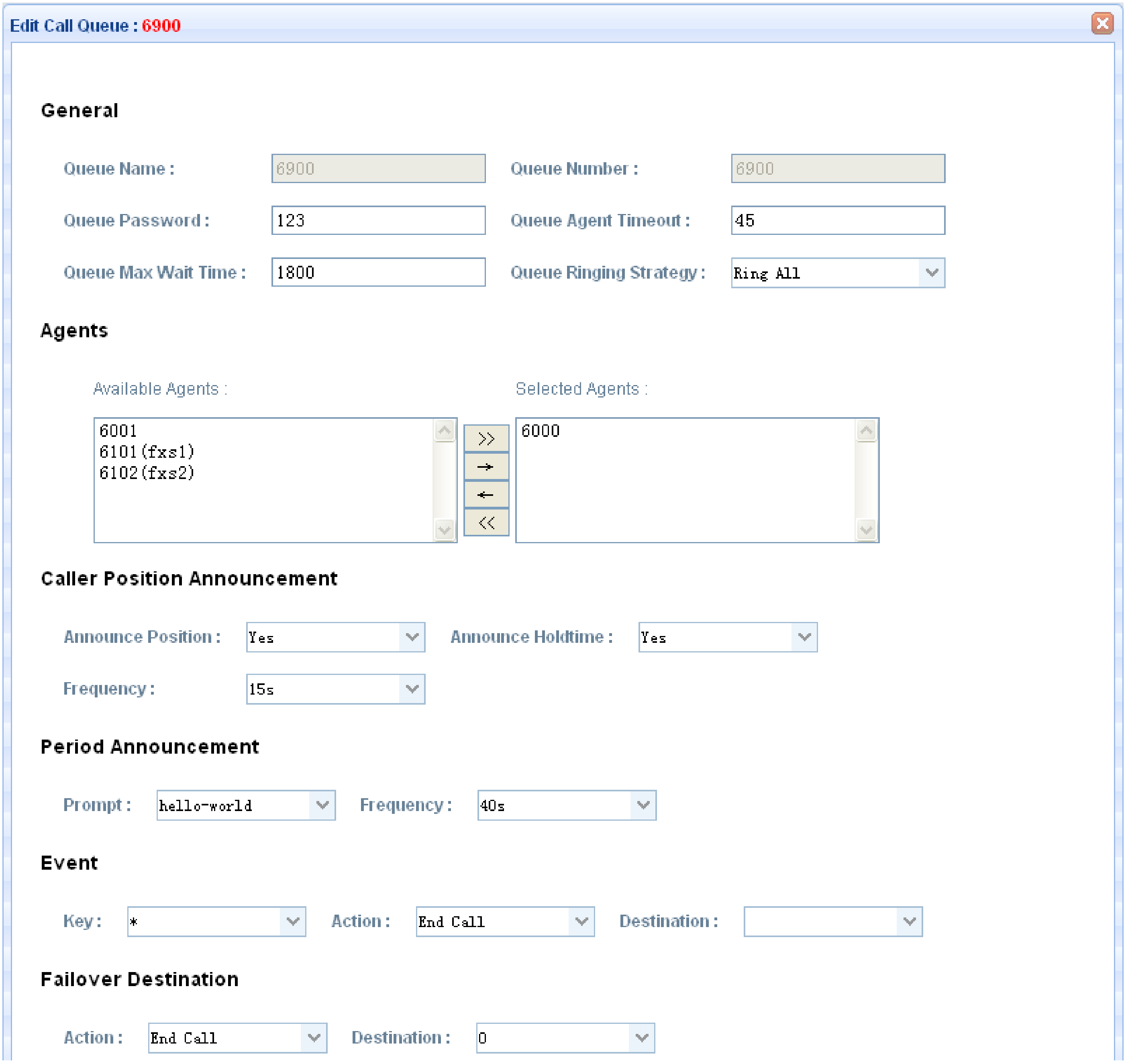
RRmemory: Round robin with memory, remembers where it left off in the last ring pass.

2) Agents

Select Static Agent here. there are two kinds of agents:

Static Agent: chose here

Dynamic Agent: users can dial ‘Queue number + *’ to log in as dynamic agent, and ‘Queue number + **’ to log out. In this case, users can dial 6900* to being a dynamic agent (need to enter password 123), and 6900** to log out.



3) Caller Position Announcement

Announce queue position and / or estimated hold time to caller

4) Period Announcement

This allows a message like "Thank you for holding, your call is important to us." to be played at regular intervals while a caller is in the queue

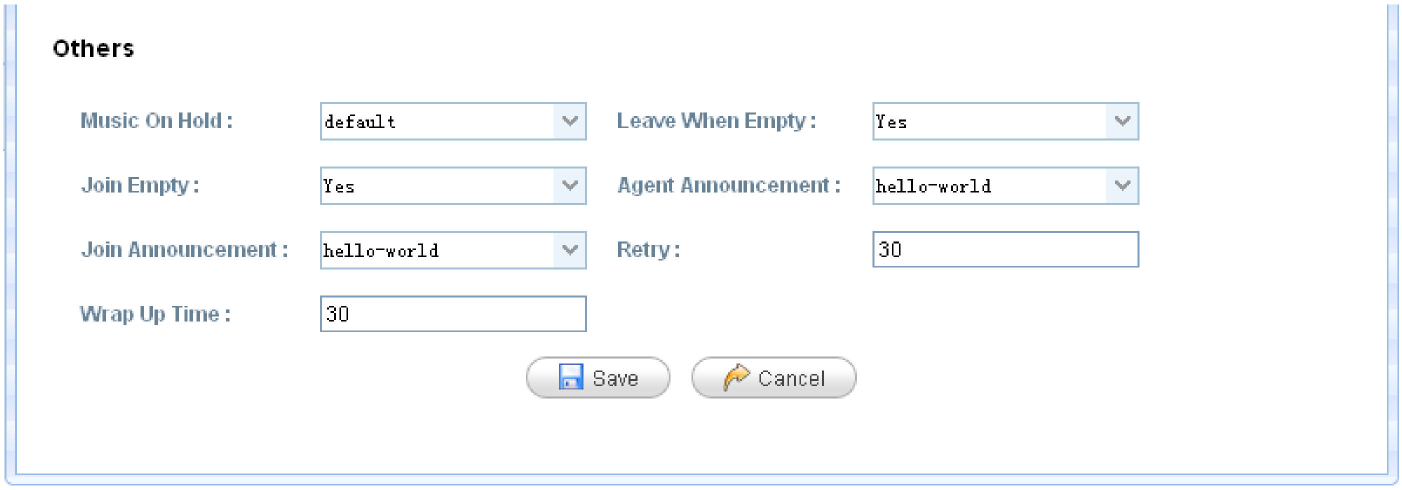
NOTE: The key point with announcements is that they are only played within the timeout/retry period set on the queue. For the most part this works OK as when all queue members are busy/unavailable, the timeout/retry period is effectively ignored (i.e. you can consider the queue to always be in this state) and announcements will be played as per your setting of the announce-frequency and periodic-announce-frequency parameters. When a handset is available and the queue is ringing it, the timeout/retry timeouts become critical. For example, if you want announcements every 20 seconds, but the timeout is set to 60 seconds, when a queue member is ringing, you will only ever get announcements every 60 seconds.

5) Event

This allows callers waiting in the queue to dial a key to go to other destination.

6) Failover Destination

This define the failover destination for callers when the max wait time is up.



7) Others

Music On Hold

Select Music On Hold Class for this Queue

Leave When Empty

This option controls whether calls already on hold are forced out of a queue that has no agents. There are two options:

Yes: Callers are forced out of a queue when no agents logged in, or if all logged in agents are unavailable.

NO: Callers will remain in a queue with no agents.

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 or if all agents are unavailable.

Agent Announcement

Announcement played to the agent prior to bridging in the caller.

Join Announcement

Announcement played to callers once prior to joining the queue.

Retry

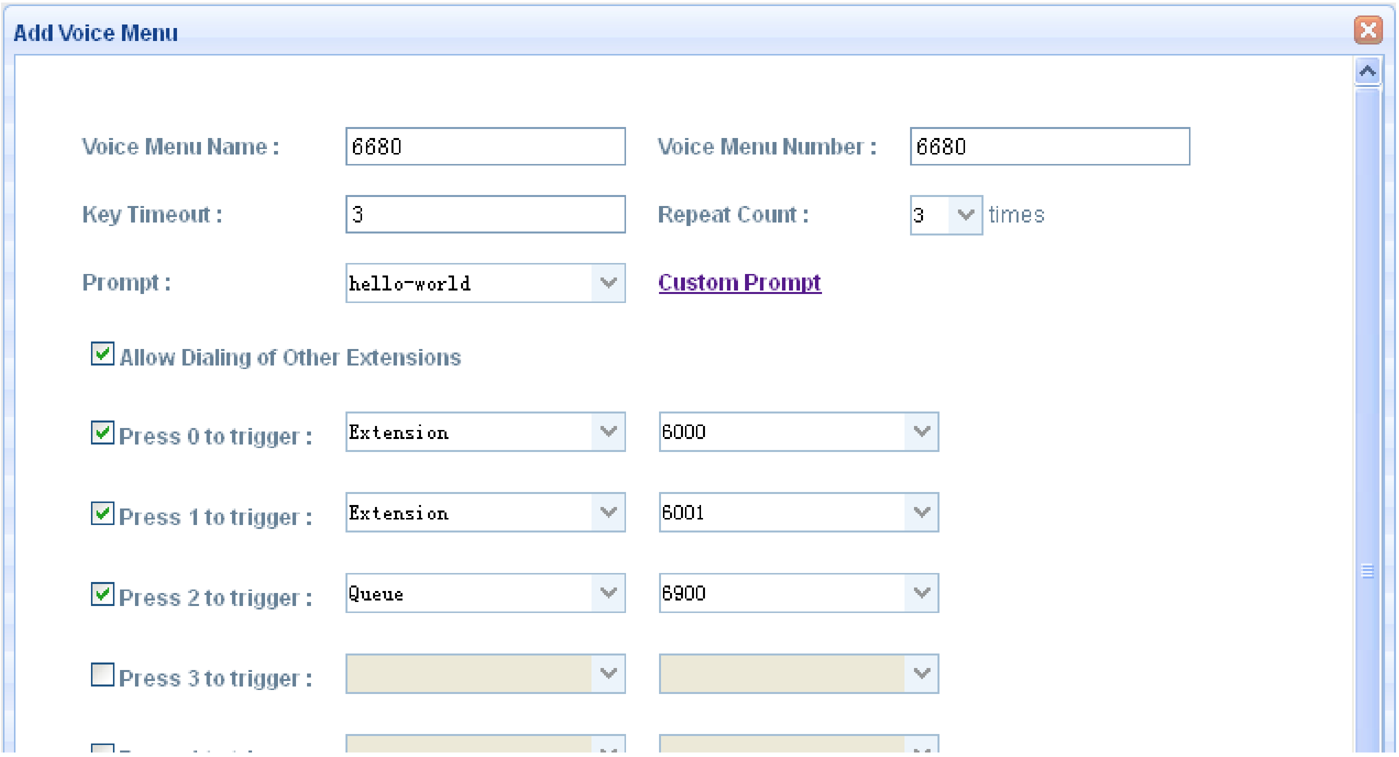
How long does APBX wait before trying all the members again.

Wrap Up Time

After a successful call, how long to wait before sending a potentially free member another call.

6.4 Auto Attendant / Voice Menu / IVR

Callers are presented with a recorded menu and respond by selecting a digit or, in some cases, by entering an extension number. The automated attendant eliminates the need for a live operator to handle the call.



Voice Menu Name

Name of the Voice Menu

Voice Menu Number

Extension number of the voice menu, dial it to get into the voice menu

Key Timeout

How long for APBX to wait user’s input

Repeat Count

How many times to play prompt

Allow Dialing of Other Extensions

Allow dialing local extensions

Key Press Event

Dial digit to trigger corresponding event

No Entry Forward to

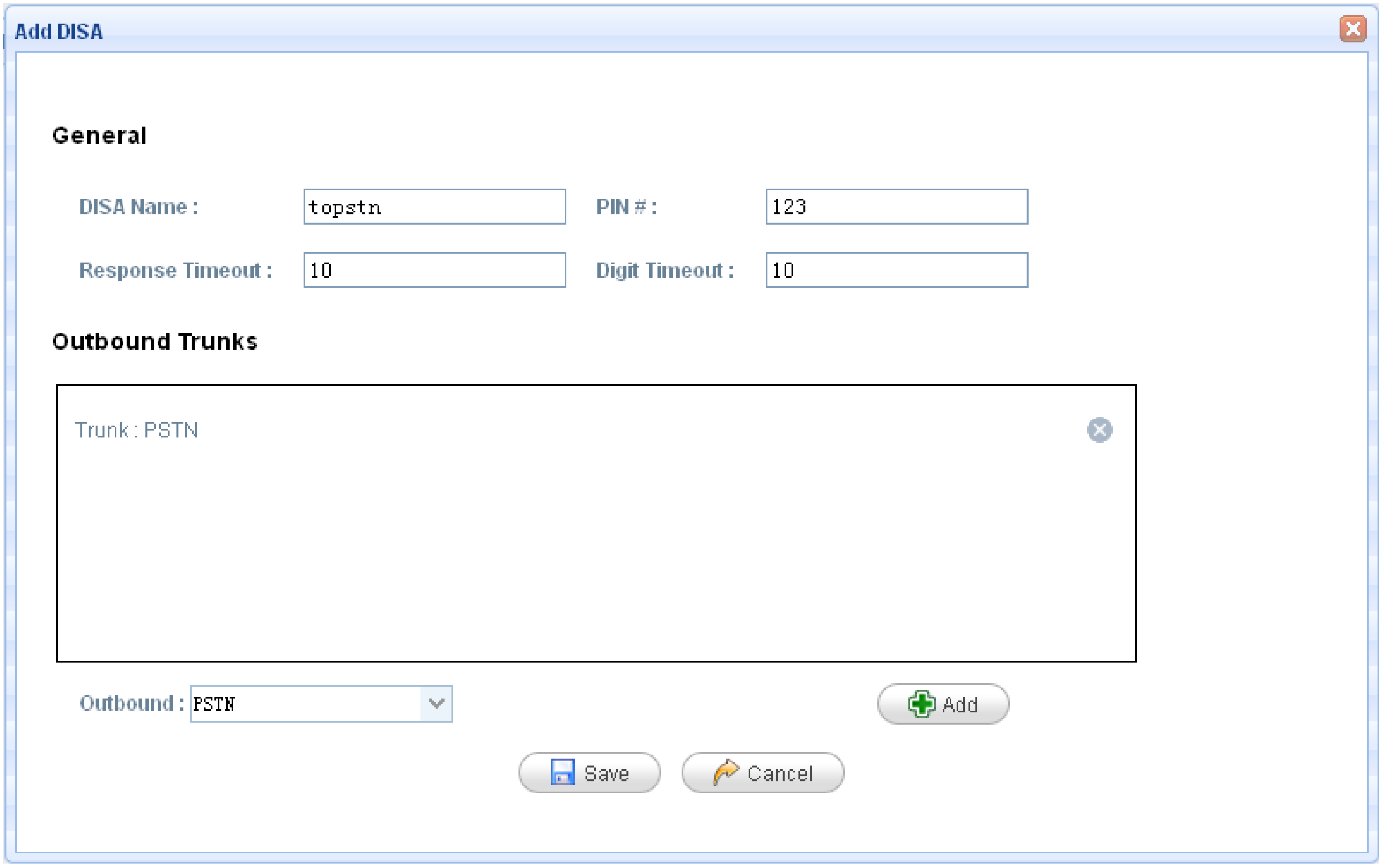
The destination for incoming call if there is none input

Invalid Forward to

The destination for incoming call if there is invalid input

6.5 DISA

DISA (Direct Inward System Access) allows someone calling in from outside to obtain an "internal" system dialtone and dial out as if a local extension.



1) General

DISA Name

A name for the DISA

PIN #

When caller get into the DISA, this password is needed to put before making calls.

Response Timeout

The maximum time in seconds APBX will wait for input from a user.

Digit Timeout

The maximum time allowed between entry of digits. If exceeded, user input is deemed to have finished.

2) Outbound Trunks

Choose the outbound route that callers can use to dial out.

For example:

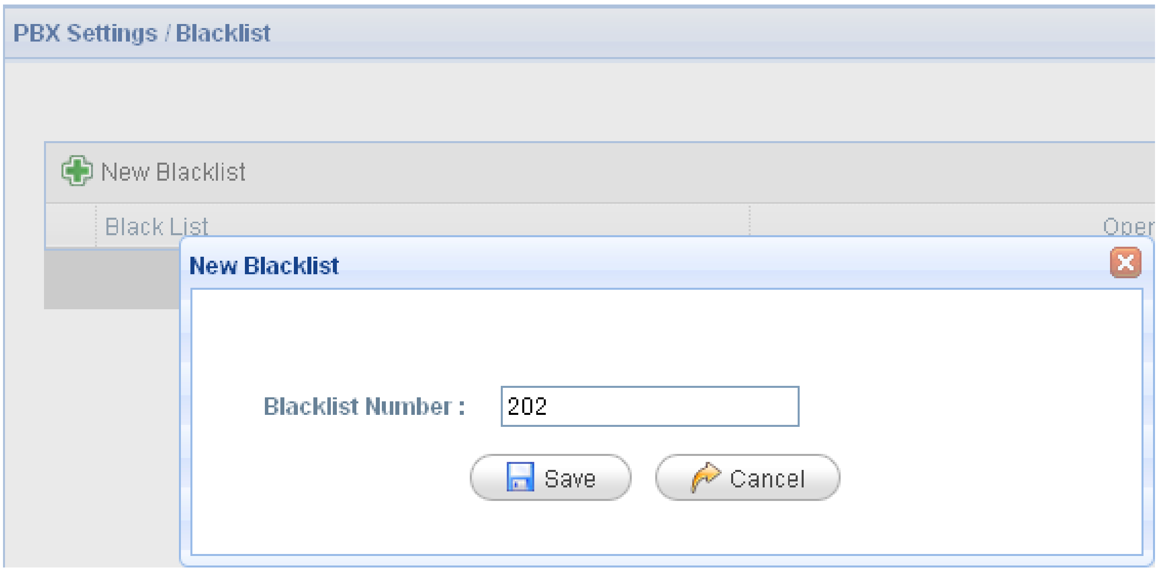
Both City A and B have a APBX, APBX-A and APBX-B, they are connected with SIP trunk, and APBX-A has FXO trunk to connect local PSTN and outbound route for that, DISA can be used as below:

1. Create a DISA in APBX-A including the FXO trunk.
2. Set it as the destination of inbound route for SIP trunk.

After users of APBX-B dial into DISA application in APBX-A, The DISA application in turn requires the user to enter his passcode, followed by the pound sign (#). If the passcode is correct, the user will hear dialtone on which a outbound call may be placed, so there is no long distance call fees.

6.6 Blacklist

Block incoming calls from specified numbers



If a number in blacklist dial into APBX, caller will hear following prompt: "Then number you have dialed is not in service. Please check the number and try again." Then system will then disconnect the call.

6.7 SIP Settings

1) General

UDP Port

Set the SIP port (UDP) which APBX is listening to.

Enable TCP

Enable TCP protocol for SIP.

TCP Port

Set the SIP port (TCP) which APBX is listening to.

Registration / Subscription Time Max

Maximum duration in seconds of a SIP registration / subscription.

Registration / Subscription Time Min

Minimum duration in seconds of a SIP registration / subscription.

RTP Port Min / Max

Set the RTP port range.

DTMF Mode

Set the default DTMF mode

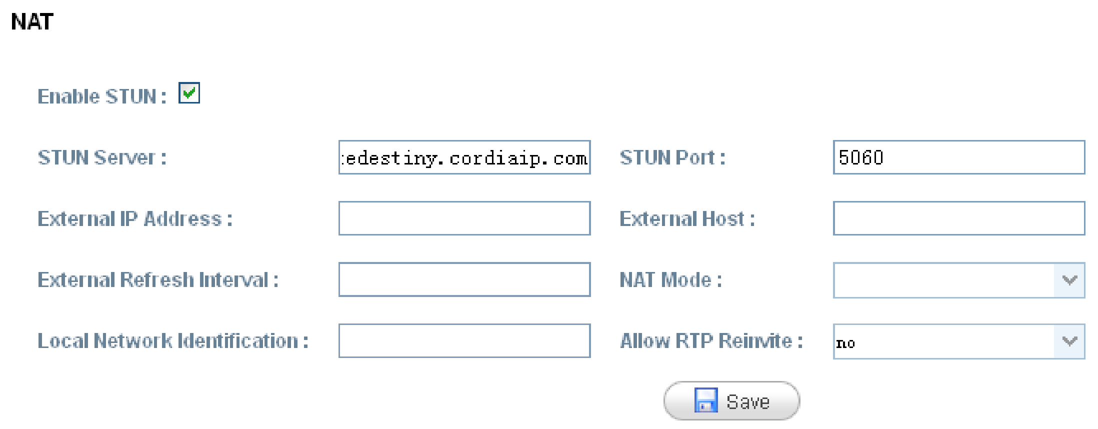
2) NAT

Here provide other two solutions for SIP one-way-audio issue besides outbound proxy. Using any one is OK.

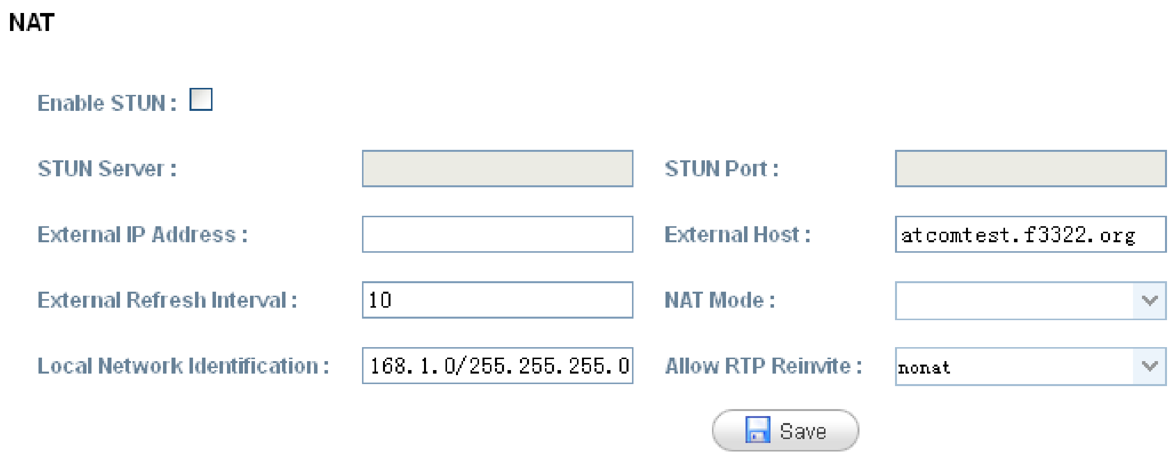
a. STUN

Just setting STUN server / port is OK.

There are many public STUN server on Internet: http://www.voip-info.org/wiki/view/STUN



b. NAT



The External IP, External Host and Local Network Identification settings are used if you use APBX behind a NAT device to communicate with services on the outside.

External IP address

Address that we're going to put in outbound SIP messages if we're behind a NAT. The externip and localnet is used when registering and communicating with other proxies that we're registered with.

External Host

Alternatively you can specify an external host, and APBX will perform DNS queries periodically. Not recommended for production environments! Use External IP instead.

External Refresh Interval

How often to refresh External Host if used.

NAT Mode

Global NAT settings (Affects all peers and users), is used 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.

Local Network identification

You may add multiple local networks. A reasonable set of defaults are set here.

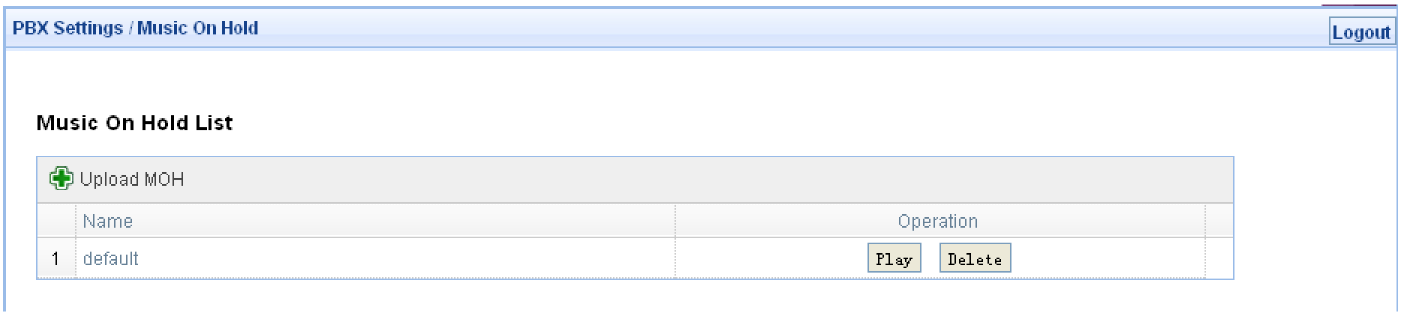
Allow RTP Reinvite

By default, Asterisk tries to re-invite the audio to an optimal path. If there's no reason for APBX to stay in the media path, the media will be redirected. This does not really work with in the case where APBX is outside and have clients on the inside of a NAT. In that case, you want to set this option to nonat.

7. Audios

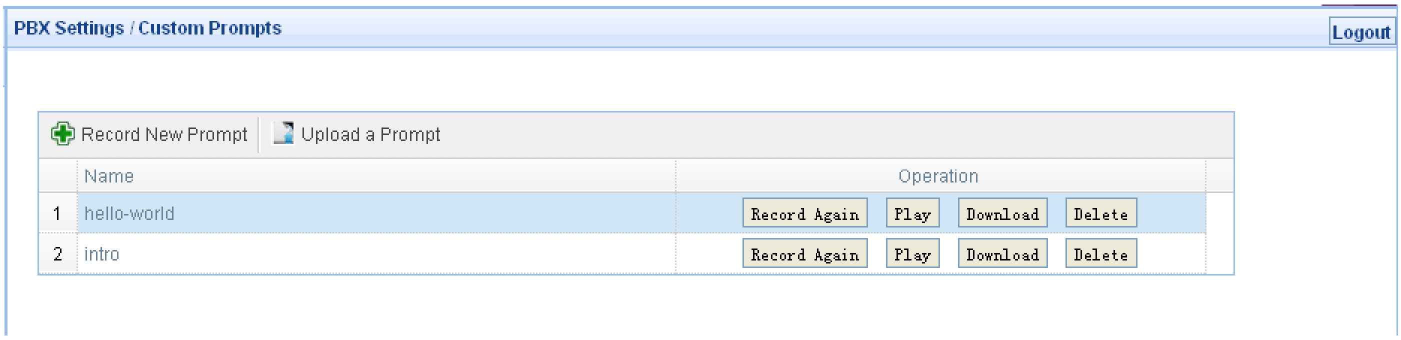
7.1 Music On Hold

Manage audio files for Music On Hold, the format should be .WAV and .GSM, the size should less than 4 MB.



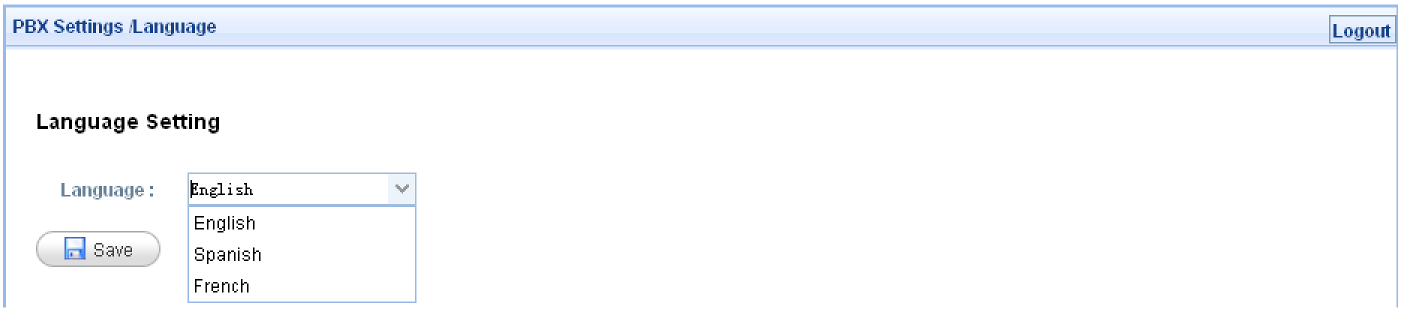
7.2 Custom Prompts

Manage prompts used for Voice Menu. It can be recorded by extensions or uploaded from local PC.



7.3 Language Setting

Set the language of default system prompt audio, English is supported by default. French and Spanish need to be download from Internet when chose at the first time. Make sure gateway is right set so that APBX can access Internet.

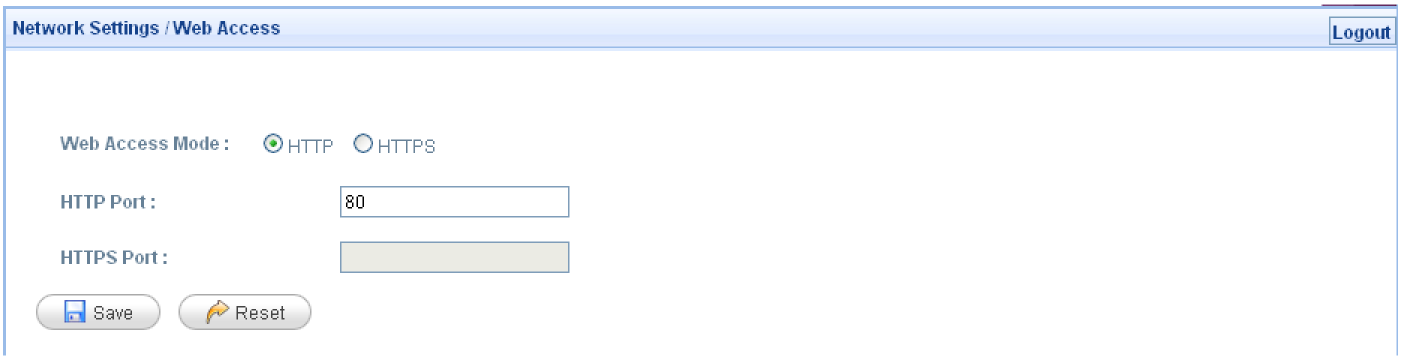


8. Network Settings

Description of LAN, WAN and DHCP server settings can be found in Chapter 2. All network settings will take effect after APBX reboot.

8.1 Web Access

Choose the web access protocol and port for web server here. HTTP and HTTPs are both supported, default port is 80 and 443 respective.



8.2 Firewall

Firewall is used to prevent unauthorized connections.

1) Enable Firewall

Check it to enable firewall.

2) Common Rule

Accept/Drop the connections from remote hosts.

Name

A name for the rule.

Description

Simple description for the rule.

Protocol

Set the protocol type for connection.

Port

Set the destination port range for connection. The main protocols and default ports APBX uses for each application are list below:

HTTP	TCP:80
HTTPS	TCP:443
SIP	UDP:5060
SIP	TCP:5060
RTP	UDP:50000-60000

IP

Set source IP of connection.

Format of IP: IP/mask

For example:

192.168.1.156/255.255.255.255 for IP 192.168.1.156

216.207.245.47/255.255.255.255 for IP 216.207.245.47

192.168.1.156/255.255.255.0 for network 192.168.1.0/24

Mac Address

Set source Mac of connection. Either IP or Mac Address must be set.

Action

Accept: Accept the access from remote hosts.

Drop: Drop the access from remote hosts.

3) Auto Defense

Limit connections from remote hosts.

Port

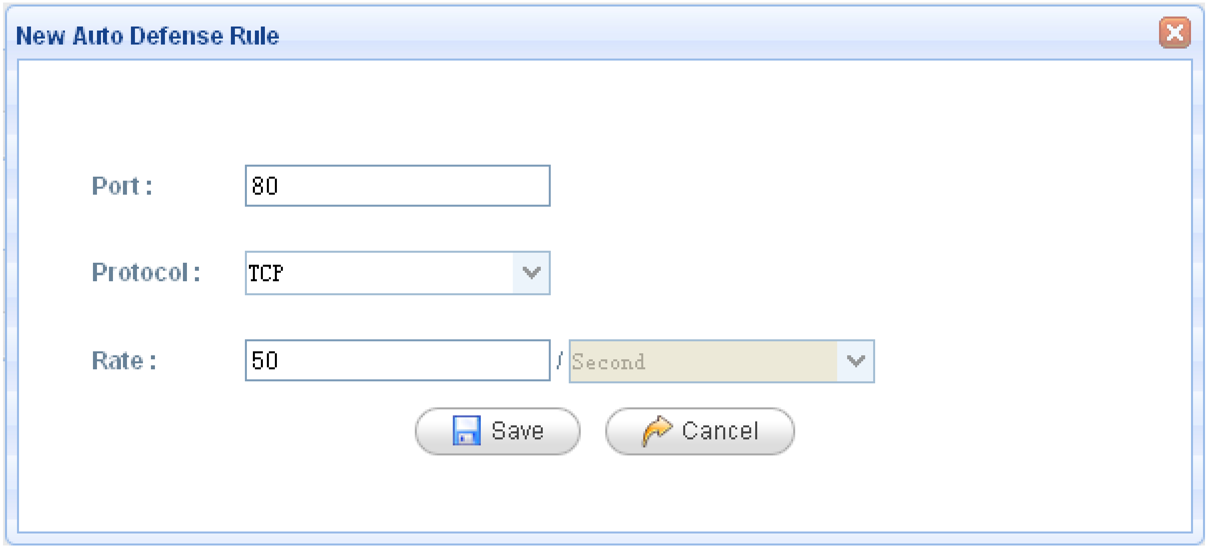
Set the destination port range for connection.

Protocol

Set the protocol type for connection.

Rate

The maximum packets or connections can be handled per second



4) SIP Defense:

Limit connections to SIP port from remote hosts.

Port

Set the destination port range for connection.

Protocol

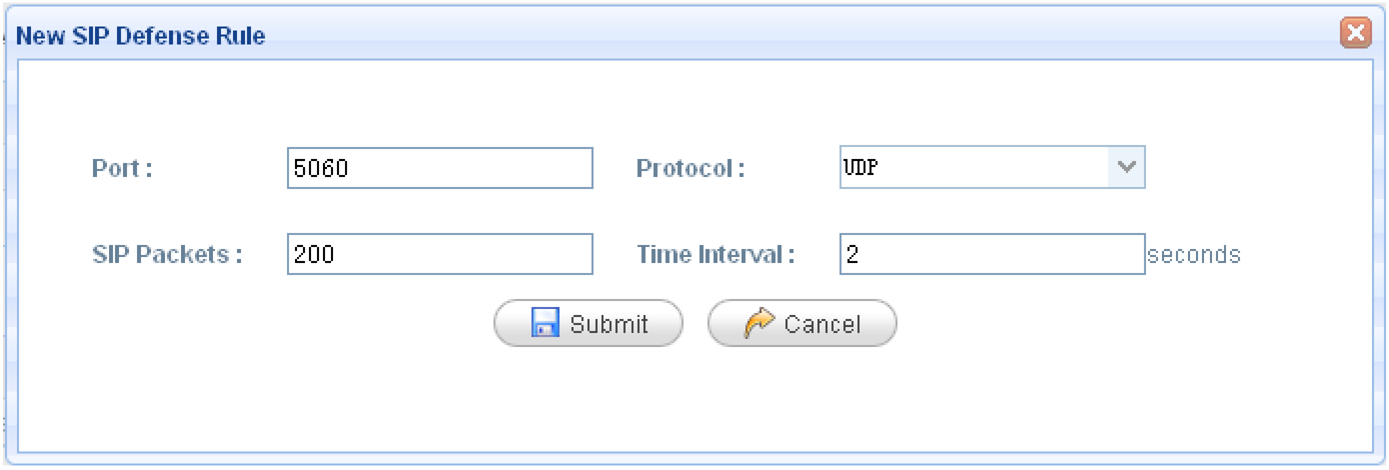
Set the protocol type for connection.

SIP Packets

The maximum packets can be handled per time interval.

Time Interval

Time unit which IPPBX uses to manage IP packets received.



5) Other Options

Disable Ping

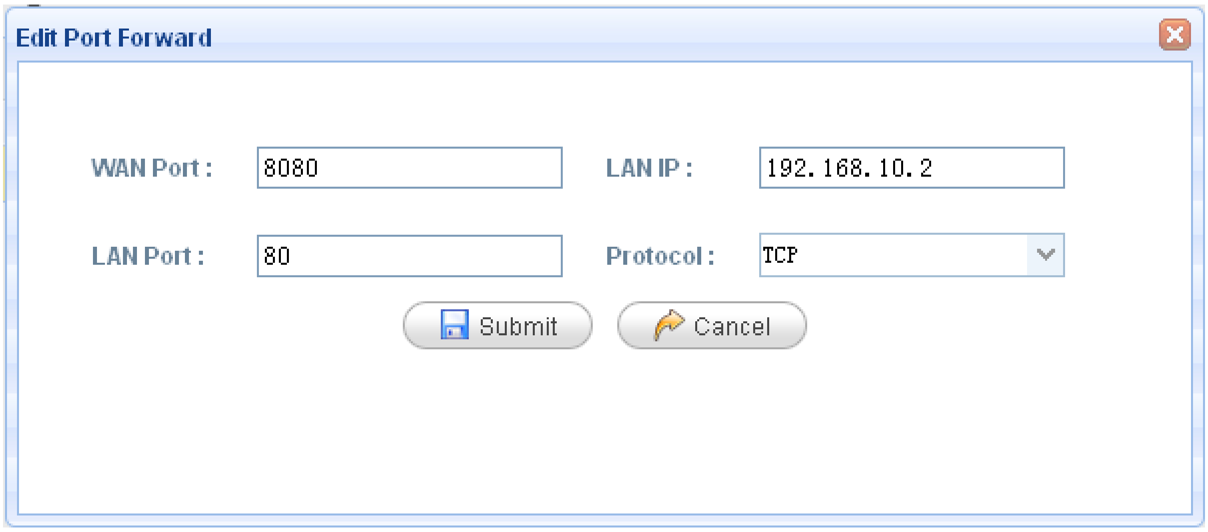
Check this to drop ping packets from remote hosts.

Drop All

Check this to drop all packets or connection from other hosts if there are no other rules defined.

8.3 Port Forwarding

When APBX works as a router, it can forward connections to WAN to a device connected to LAN network.



For example: user can access 192.168.10.2:80 (connected to LAN) by accessing xx.xx.xx.xx:8080 (xx.xx.xx.xx is the IP of WAN)

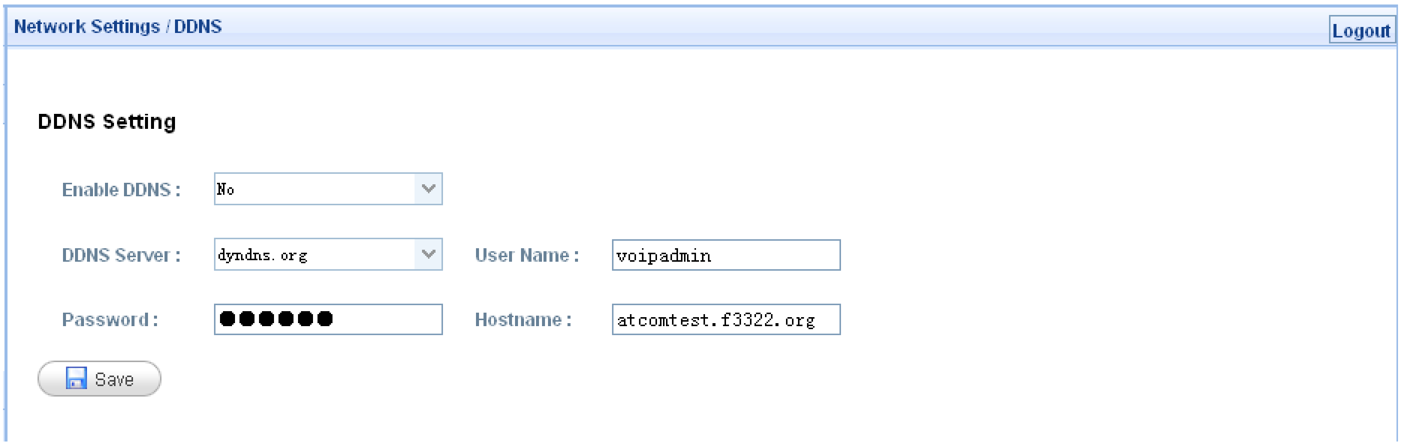
8.4 DDNS

Dynamic Domain Name Service (DDNS) is a service used to map a domain name to the dynamic IP address of a network device. IPPBX support 3 DDNS servers below, please go to the website of below servers and apply a domain name then fill related information here.

dyndns.org

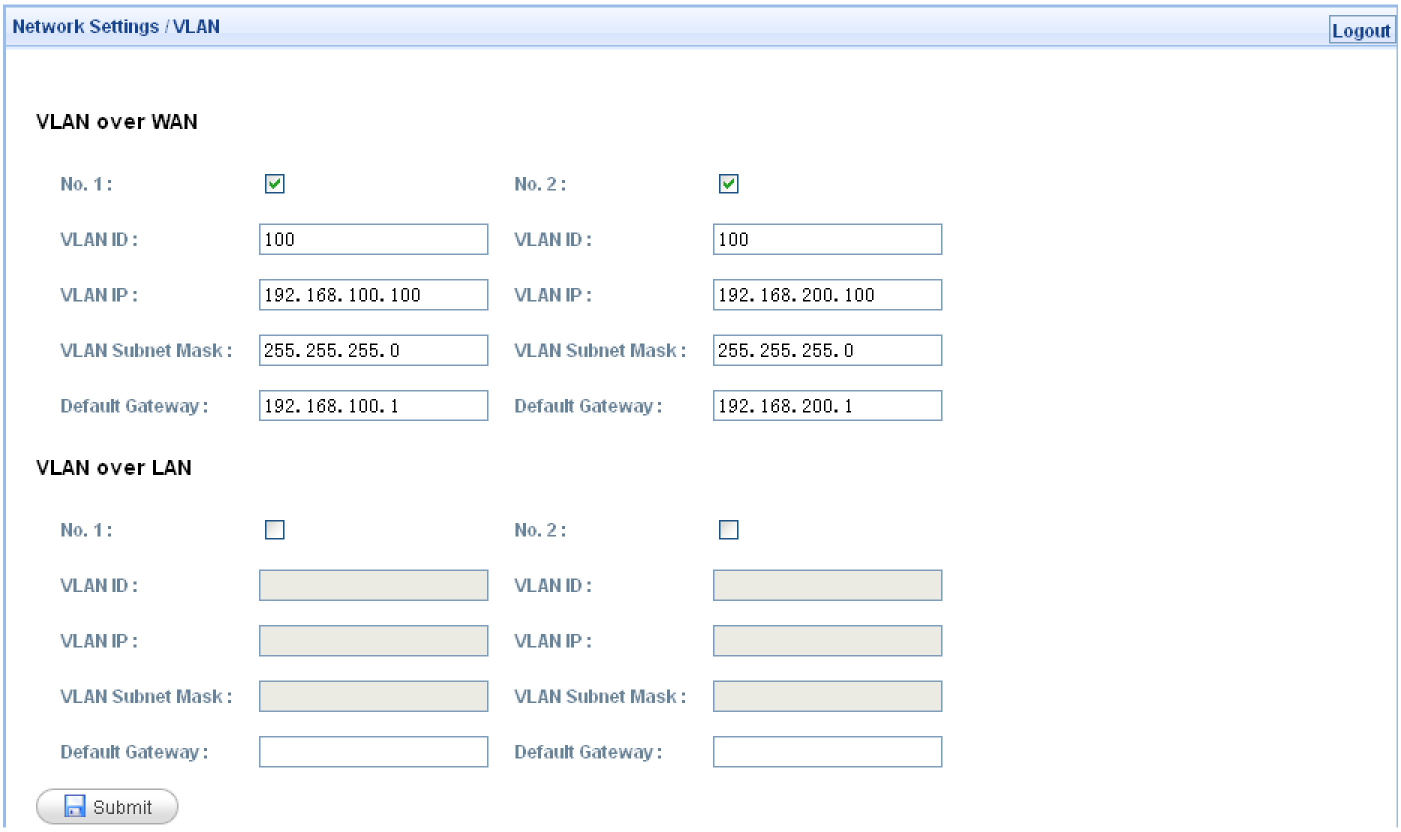
qdns

www.zoneedit.com



8.5 VLAN

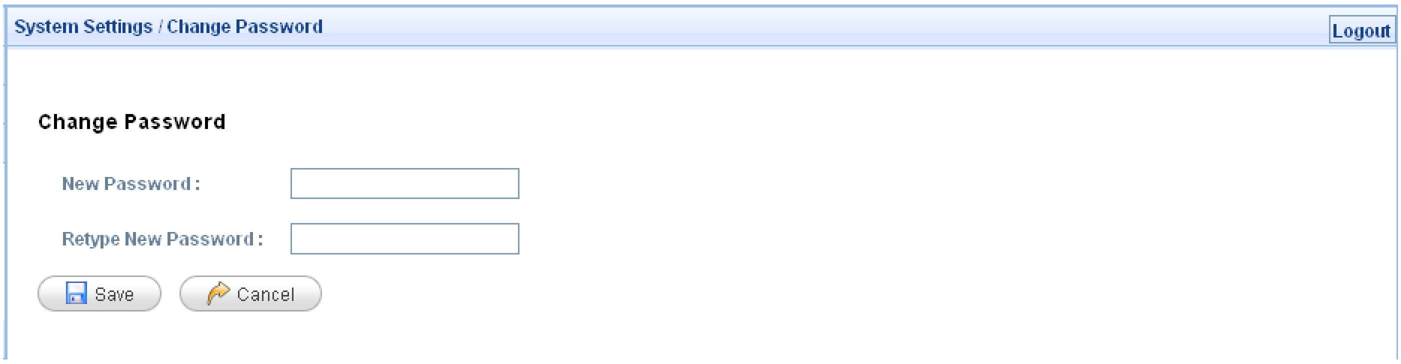
A VLAN (Virtual LAN) is a logical local area network (LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations. Both WAN and LAN support 2 VLANs.



9. System Settings

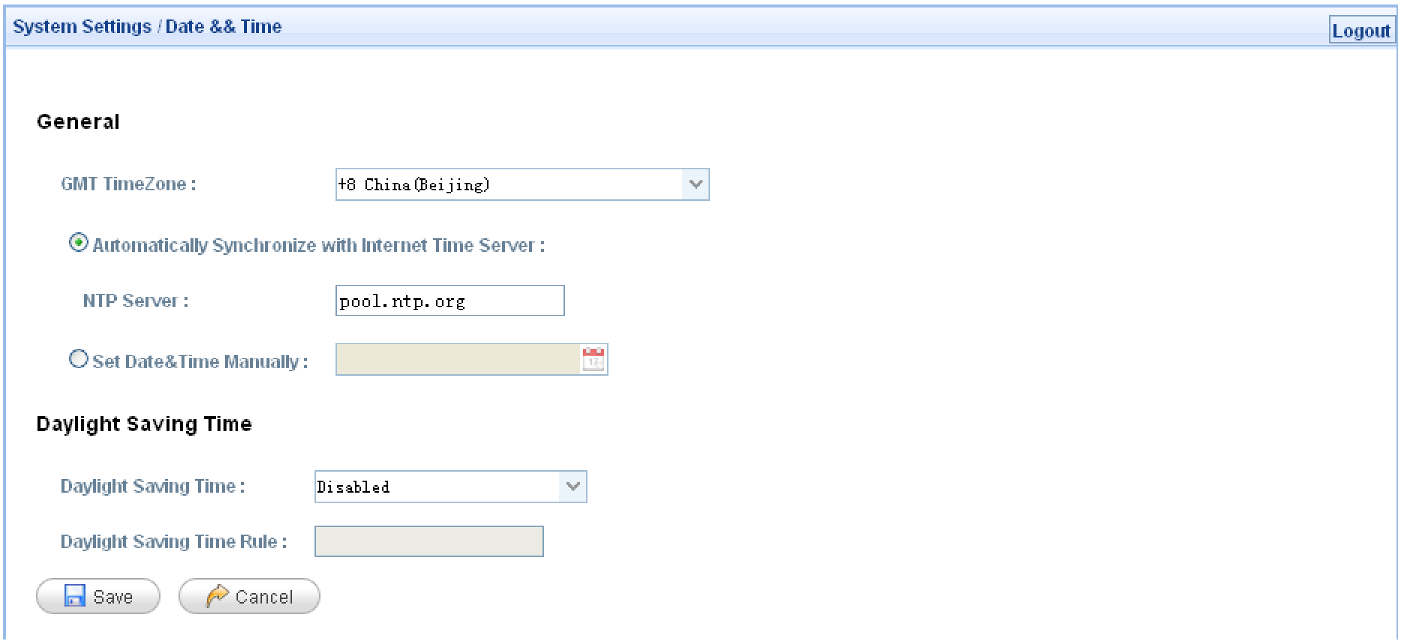
9.1 Change Password

Change the password for admin login, it will take effect immediately.



9.2 Date && Time

Set the date and time for APBX. The settings will take effect immediately.



1) General

There are two ways to set Date/Time for APBX:

a. NTP server

Make sure the connection between APBX and NTP server is OK, if the NTP server is located on Internet, the gateway of WAN should be right set so that APBX can access Internet.

GMT TimeZone is also an important arguments for time setting in this way. Please choose the right Time Zone.

b. Manually

2) Daylight Saving Time

There are two ways to set DST:

a. Automatic

Just making sure GMT Time Zone is right set is OK. There have already DST setting in each Time Zone.

b. Manually

However, the DST in some countries is changing every year. If the DST setting in Time Zone is not exact. Please set it manually, the format should be: **start=*start_time*;end=*end_time*;save=*offset***

The rule for start / end time is: **month/mday/wday/hour:min:second**

1<= month <=12 , 0< mday <=31 , 0<= wday <7

month/mday/wday means the first wday coming after month/mday

for example:

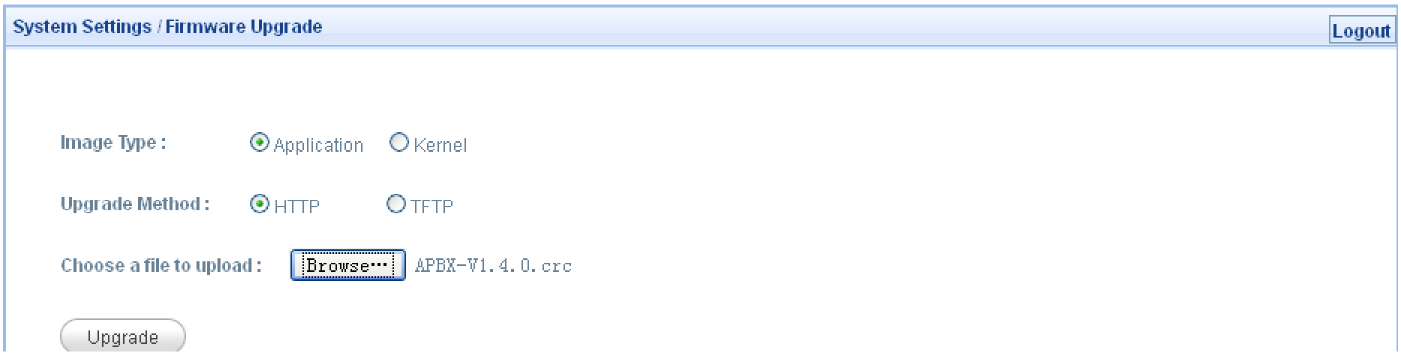
start=4/1/7/0:0:0;end=10/31/7/0:0:0;save=1 means APBX time from the first Sunday coming after April 1th to the first Sunday coming after October 31th will be one hour early.

9.3 Firmware Upgrade

There are two ways to upload Kernel / Application for APBX:

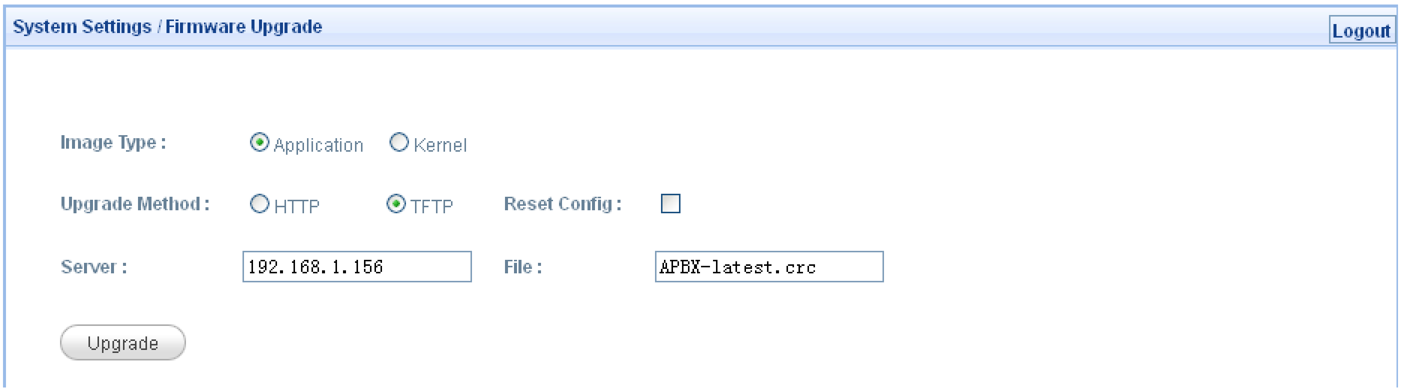
1) HTTP

Upload them from local PC.



2) TFTP

Upload them from TFTP server, the Kernel / Application must be located in base directory of TFTP server.

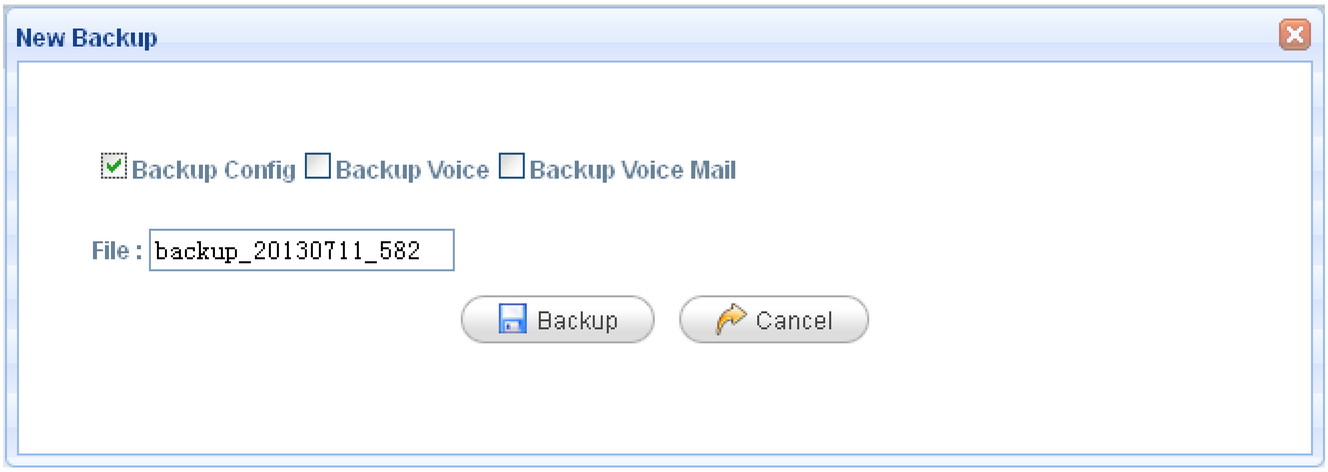


Choose Reset Config will reset the configuration.

9.4 Backup and Restore

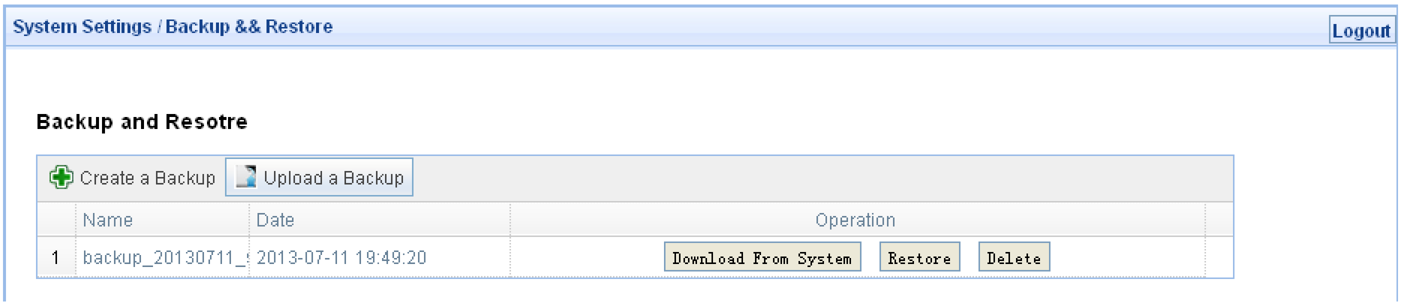
1) Backup

Create Backup for configuration / System audio prompt / Voice Mail. The backup can be downloaded to local PC.



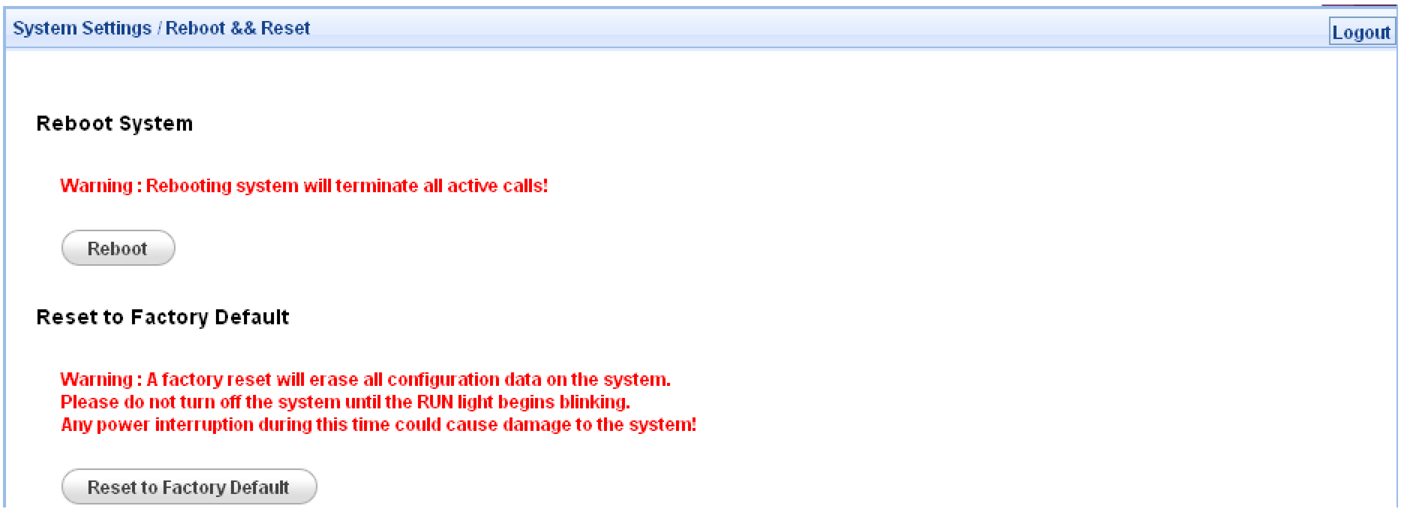
2) Restore

Click Restore to restore corresponding backup, backup file can be uploaded from local PC. Backup will be used after APBX reboot. It can't be used for different product models.



9.5 Reboot && Reset

Reboot or Reset APBX.



10.Reports

10.1 Call Detail Records

Display the Call Detail Records, the operation for it can be search, delete and download.

1) Search

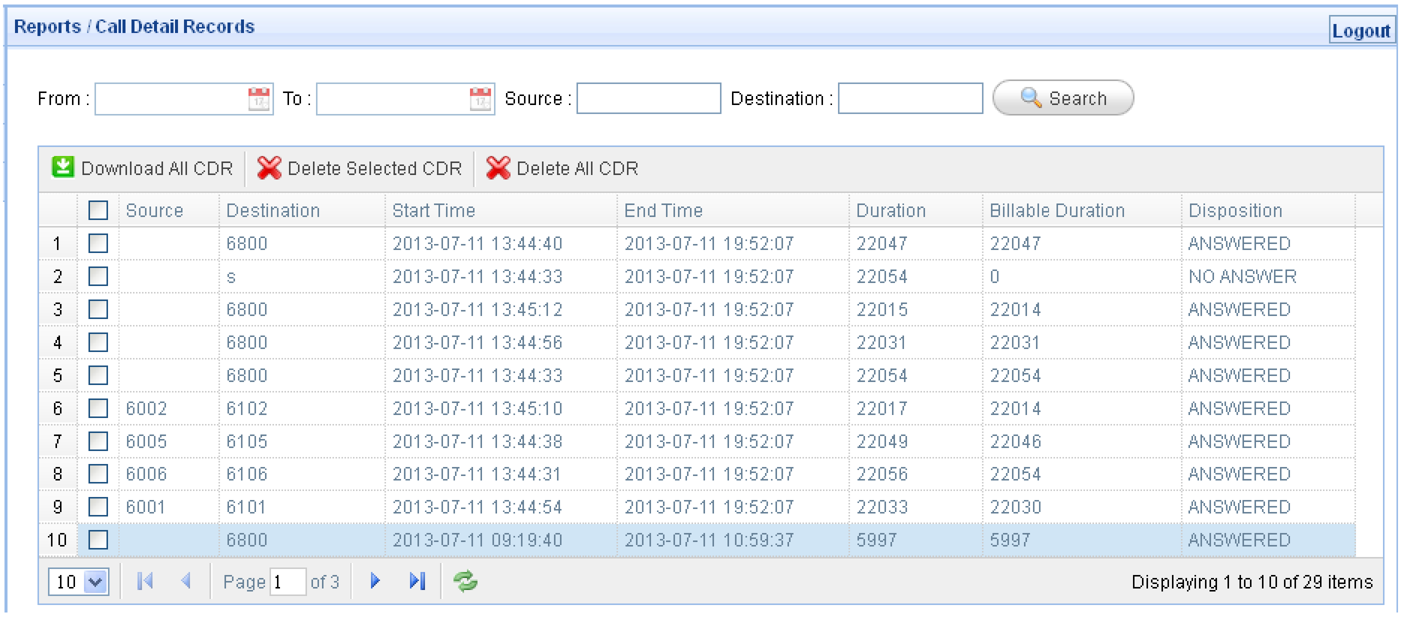
Users can search the records they needs according to Source, Destination, and / or Time.

2) Delete

APBX supports two delete operation: delete selected CDR and delete all CDR.

3) Download

It can be download to local PC

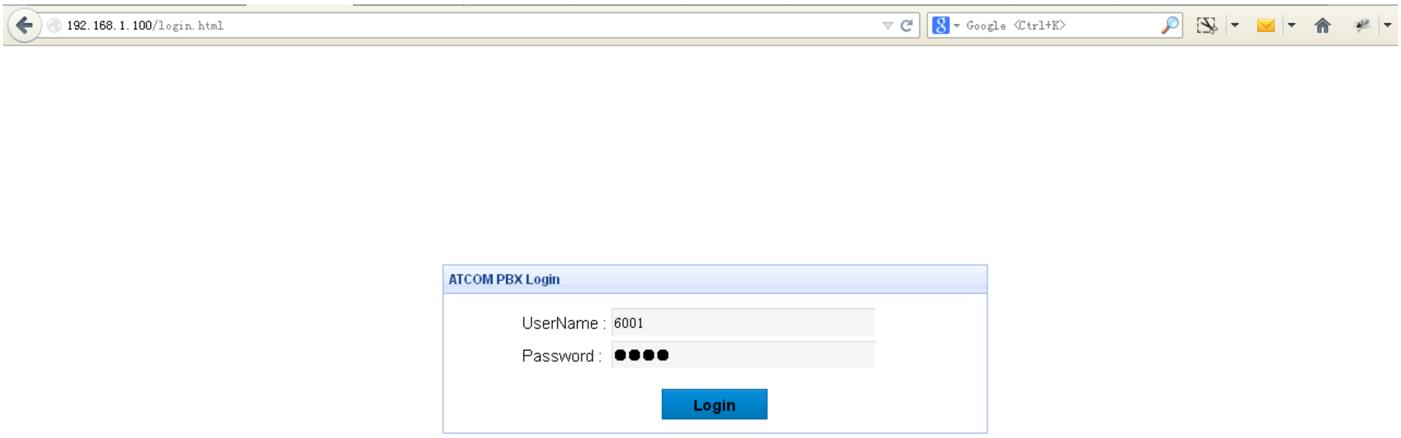


	Source	Destination	Start Time	End Time	Duration	Billable Duration	Disposition
1		6800	2013-07-11 13:44:40	2013-07-11 19:52:07	22047	22047	ANSWERED
2		s	2013-07-11 13:44:33	2013-07-11 19:52:07	22054	0	NO ANSWER
3		6800	2013-07-11 13:45:12	2013-07-11 19:52:07	22015	22014	ANSWERED
4		6800	2013-07-11 13:44:56	2013-07-11 19:52:07	22031	22031	ANSWERED
5		6800	2013-07-11 13:44:33	2013-07-11 19:52:07	22054	22054	ANSWERED
6	6002	6102	2013-07-11 13:45:10	2013-07-11 19:52:07	22017	22014	ANSWERED
7	6005	6105	2013-07-11 13:44:38	2013-07-11 19:52:07	22049	22046	ANSWERED
8	6006	6106	2013-07-11 13:44:31	2013-07-11 19:52:07	22056	22054	ANSWERED
9	6001	6101	2013-07-11 13:44:54	2013-07-11 19:52:07	22033	22030	ANSWERED
10		6800	2013-07-11 09:19:40	2013-07-11 10:59:37	5997	5997	ANSWERED

11. Web Interface for extension

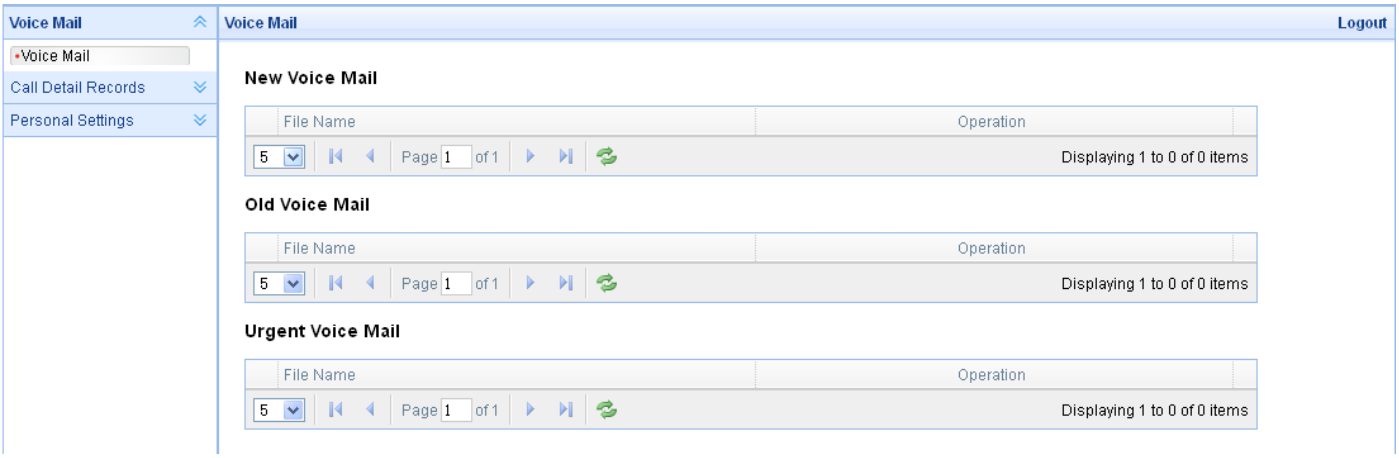
PBX allows users to check their voicemail / CDR, and set personal settings.

1. Check **Use Web Interface** option in **PBX Settings** -> **Extensions** management settings to allow this extension to login its own web interface.
2. Enter the IP of PBX in the browser.
3. Login with extension number / Voice Mail Access PIN Code as username / password



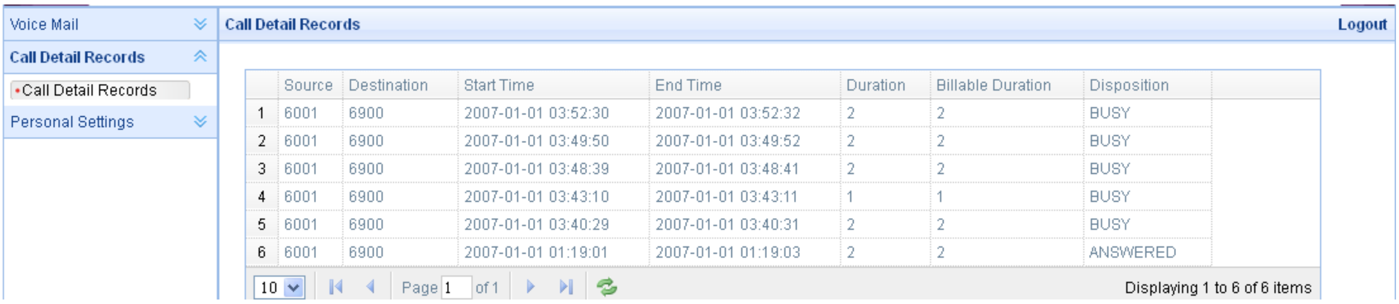
1) Voice Mail Checking

Users can listen / download / delete / move voice mail here.



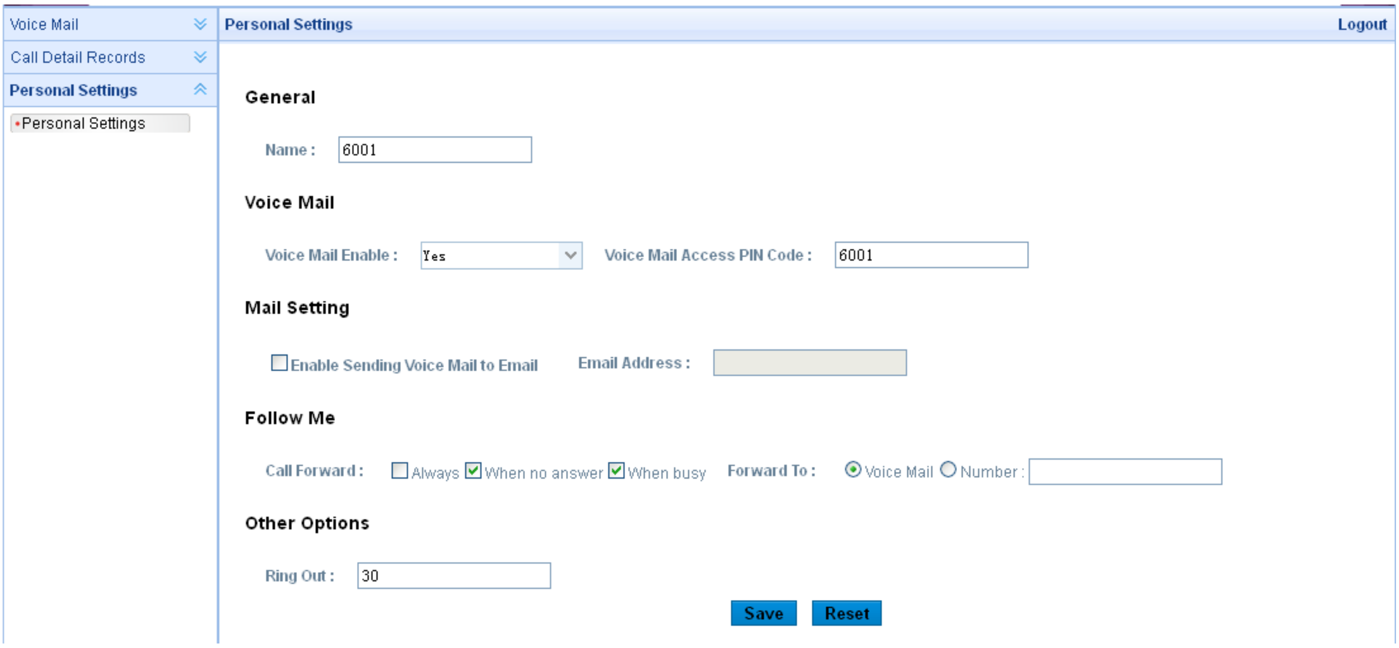
2) CDR Checking

Users can check their CDR here.



3) Personal Settings

Users can set voice mail / voice mail to email / follow me / ring timeout here.



--Finish--