



IPPBX IP08 User Manual



For Firmware Version: V4.0

2015-11-12

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1. Introduction

Overview of the IPPBX

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, IP08 provides a cost-saving solution on their telecommunication/data needs. With IP08, 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	8
Network Interface	WAN,LAN

Measurement and Weight

Inner box	225 * 120 * 30mm
G.W./unit	0.79KG
Carton MEAS	456 * 442 * 362 mm
Units per Carton	21 units/ CTN
G.W./CTN	18 KG/CTN

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	Call Back
Call Waiting	IVR
Call Routing	Intercom/Paging

Caller ID	IP Restriction
BLF Support	Firewalls
Music on Hold	DDNS
Video Call	DHCP Server
PPPoE	VLAN
Port Forward	VPN client
External Storage	Auto Provisioning
Storage Quota Privilege	Custom Dialplan
Auto call recording	Integrated built-in packet capture tools

2. Connection and Change IP Address of IPPBX

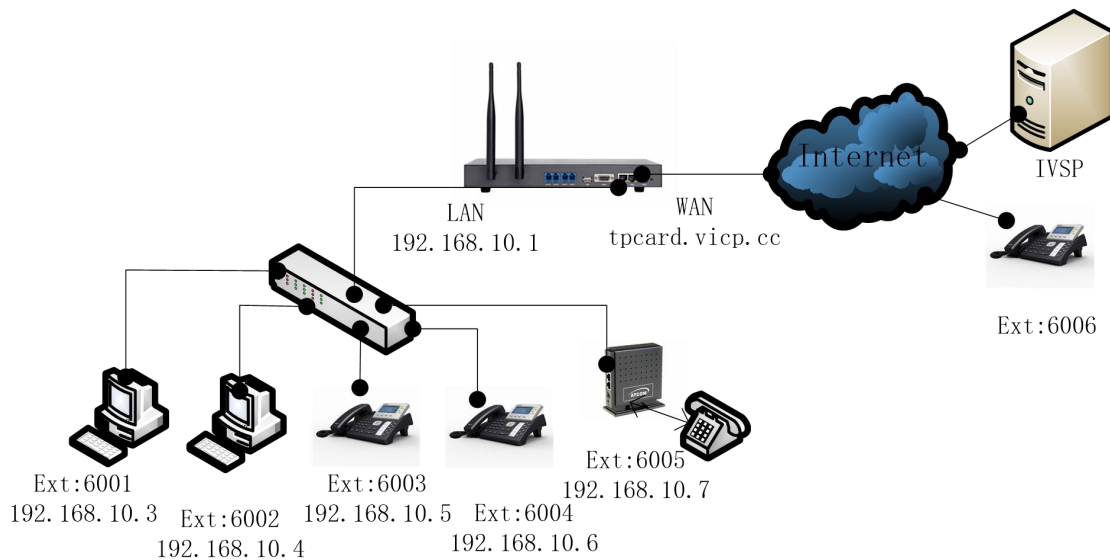
2.1 Connection

The default IP address of IPPBX is:

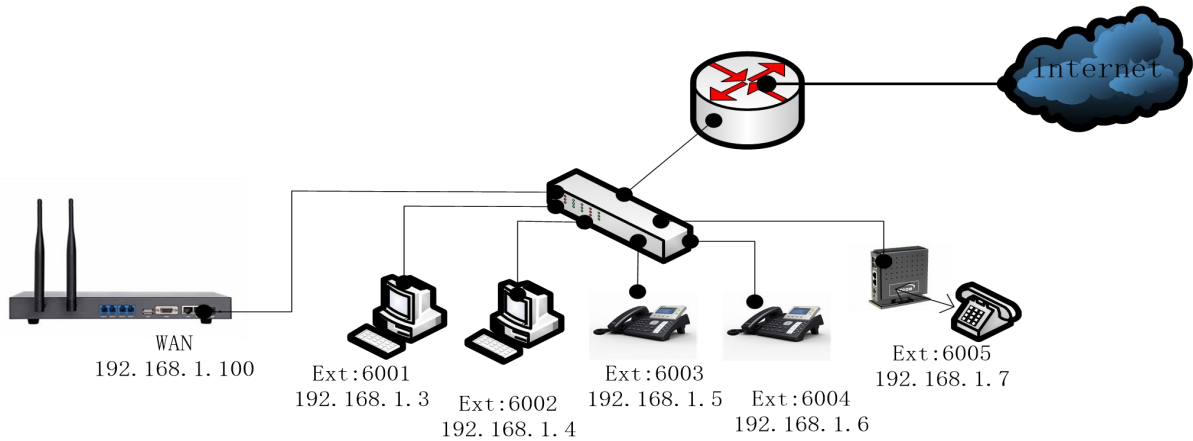
WAN: 192.168.1.100/255.255.255.0

LAN: 192.168.10.1/255.255.255.0

There are two network interfaces (WAN and LAN) in IP08, it can be used as a router (WAN port connects Internet and LAN port connects local network):



If you do not want IPPBX to act as a router and put it behind a router, please connect WAN port to your local network:



NOTE: It's recommended to re-set IP address of IPPBX via WAN at the first time, Since DHCP server on LAN port is enabled by default, Other DHCP client enabled devices in the network may obtain IP from IPPBX rather than real DHCP server. And then access IPPBX via LAN after configuring LAN settings.

- 1) Connect IPPBX 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 IPPBX as administrator via WEB GUI

User: admin

Password: atcom

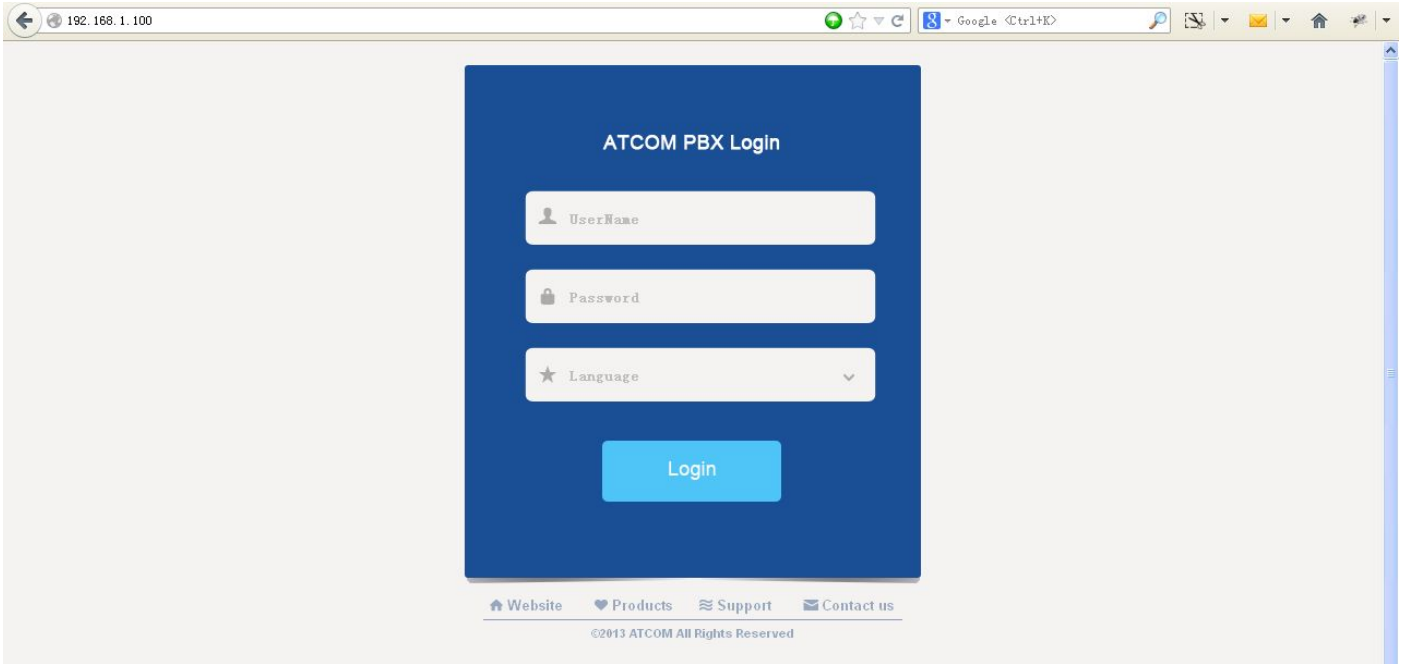
NOTE: Language option will be accomplished in future versions, currently, only English is supported. So it's not selectable now.

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

Re-set IP of WAN port.

- 4) Go to **Network Settings -> LAN.**

Re-set IP of LAN port.



2.2 WAN Settings

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

DHCP

IPPBX 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 IPPBX in DHCP server so that IPPBX can keep the same IP all the time.

Static IP

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

PPPoE

IPPBX will obtain an IP address automatically after establishing a connection to PPPoE server with username and password.

		System Status	PBX Settings	Network Settings	System Settings	Reports	Logout
Web Access > WAN LAN DHCP Server Firewall Port Forward DDNS VLAN VPN	<h3>WAN Setting</h3> Use WAN <input checked="" type="checkbox"/>						
<input type="radio"/> DHCP							
<input checked="" type="radio"/> Static IP							
IP Address : <input type="text" value="192.168.1.100"/>							
Subnet Mask : <input type="text" value="255.255.255.0"/>							
Default Gateway : <input type="text" value="192.168.1.1"/>							
Primary DNS : <input type="text" value="8.8.8.8"/>							
Secondary DNS : <input type="text" value="8.8.4.4"/>							
<input type="radio"/> PPPoE							
User Name : <input type="text"/>							
Password : <input type="password"/> <input type="checkbox"/> Show password							
WAN side HTTP(S) access : <input checked="" type="radio"/> Yes <input type="radio"/> No							
<input type="button" value="Save"/>							

2.3 LAN Settings

DHCP

Yes: IPPBX will get an IP address from DHCP server, Not recommended.

No: Configure IP address manually for LAN. Default is NO.

NAT

Yes: IPPBX will act as a router, all the devices connected to LAN can access the network connected to WAN.

Default is Yes.

No: Networks connected to LAN and WAN are independent.

NOTE: IP addresses of LAN and WAN can't be in the same network.

For example: IP of WAN is 192.168.1.100/255.255.255.0, IP of LAN can't be 192.168.1.xx/255.255.255.0

Hostname

Set the host name for IPPBX.

IP Address / Subnet Mask

Set the IP address / subnet mask for LAN network interface.

Gateway


Set the Gateway for LAN network interface.

Primary DNS / Secondary DNS

Set the DNS server for LAN network interface.

IP Address 2 / Subnet Mask 2

Set the failover IP address for LAN network interface.



The screenshot shows the 'Network Settings' page in the atcom web interface. The 'LAN Setting' section is active, displaying various configuration fields:

- DHCP:** No (dropdown menu)
- NAT:** Yes (dropdown menu)
- Host Name:** APBX (text input)
- IP Address:** 192.168.10.1 (text input)
- Subnet Mask:** 255.255.255.0 (text input)
- Default Gateway:** 192.168.10.1 (text input)
- Primary DNS:** 8.8.8.8 (text input)
- Secondary DNS:** 8.8.4.4 (text input)
- IP Address 2:** (empty text input)
- Subnet Mask 2:** (empty text input)

A 'Save' button is located at the bottom left of the configuration area.

2.4 DHCP Server

The Dynamic Host Configuration Protocol (DHCP) is a network protocol used to configure devices that are connected to a network (known as hosts) so they can communicate on that network using the Internet Protocol (IP). It involves clients and a server operating in a client-server model.

Enable DHCP server

Check to enable DHCP server.

Default Gateway

Gateway information that will be used in DHCP client.

Subnet Mask

Subnet Mask that will be used in DHCP client.

Primary DNS / Secondary DNS

DNS information that will be used in DHCP client.

Start IP / End IP

IP address pool, IP in which will be assigned to DHCP client.

Lease Time

The amount of time the address is leased for DHCP client, When the lease is up, it either renews or gets a new address.

		System Status	PBX Settings	Network Settings	System Settings	Reports	Logout
--	--	---------------	--------------	------------------	-----------------	---------	--------

Web Access	DHCP Server Setting					
WAN	Enable DHCP Server : <input checked="" type="checkbox"/>					
LAN	Default Gateway : <input type="text" value="192.168.10.1"/>		Subnet Mask : <input type="text" value="255.255.255.0"/>			
> DHCP Server	Primary DNS : <input type="text" value="8.8.8.8"/>		Secondary DNS : <input type="text" value="8.8.4.4"/>			
Firewall	Start IP : <input type="text" value="192.168.10.100"/>		End IP : <input type="text" value="192.168.10.254"/>			
	Lease Time : <input type="text" value="86400"/>					
	<input type="button" value="Save"/> <input type="button" value="Reset"/>					

NOTE: All the network settings will take effect immediately when you save the change.

3. System Status

		System Status	PBX Settings	Network Settings	System Settings	Reports	Logout
--	--	---------------	--------------	------------------	-----------------	---------	--------

> General	General					
Trunk Status	Product Model : IP08		Application Version : V4.0			
Extension Status	Kernel Version : V4.0		System Up Time : 0 days 0 hours 1 minutes 59 seconds			
	System Current Time : Mon Nov 16 09:52:31 2015					
	Network					
	WAN Connection Type : STATIC		PPPoE Status : DISABLED			
	WAN Mac Address : 80:82:87:01:89:4A		LAN Mac Address : 80:82:87:01:89:4B			
	WAN IP Address : 192.168.1.78		LAN IP Address : 192.168.10.1			
	WAN Subnet Mask : 255.255.255.0		LAN Subnet Mask : 255.255.255.0			
	WAN Gateway : 192.168.1.1		LAN NAT : yes			
	WAN Primary DNS : 8.8.8.8		LAN Primary DNS : 8.8.8.8			
	WAN Secondary DNS : 8.8.4.4		LAN Secondary DNS : 8.8.4.4			
	Peripheral					
	Port 1 : FXS		Port 2 : FXS			
	Port 3 : unplugged		Port 4 : unplugged			
	Port 5 : FXO		Port 6 : FXO			
	Port 7 : unplugged		Port 8 : unplugged			

3.1 General

Product Model

Show the model of this IPPBX

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

1) Network

Show the network setting of IPPBX

2) Peripheral

Show what kinds of / how many modules are detected.

3.2 Trunk Status

Type	Trunk Name	Status	Port/HostName/IP
1 FXO	FX05	InService	PORT5
2 FXO	FX06	Disconnected	PORT6

Type

It could show FXO,SIP,IAX,all the types of Trunks

Trunk Name

Show the name for all the trunks

Status

Show that trunk if it connects successfully.

When FXO port connects to the PSTN line or SIP/IAX Trunk connects to the server successfully ,the Status will show 'InService'.

When FXO connects failed, it will show 'Disconnected'

Also if the SIP/IAX Trunk connects failed, it will show 'Failed'

Ports/HostName/IP

Show which port does FXO connects

Show the HostName or IP address for SIP/IAX Trunk

3.3 Extension 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.

Hold: The extension is on hold

Unavailable: The extension is not registered and unreachable.

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

Status	Extension	Type	New Messages	Old Messages
Unavailable	6000	SIP	0	0
Idle	6001	IAX	0	0
Unavailable	6002	SIP/IAX	0	0
Idle	6003	SIP	0	0
Ringing	6006	SIP	0	0
Idle	6007	SIP	0	0
Unavailable	6008	IAX	0	0
Unavailable	6009	SIP/IAX	0	0
Ringing	6101	FXS	0	0
Idle	6102	FXS	0	0

4. PBX Settings

4.1 Extensions

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

4.1.1 FXS extensions



The screenshot shows the atcom web interface. The top navigation bar includes System Status, PBX Settings (selected), Network Settings, System Settings, Reports, and Logout. The left sidebar has a menu with Extensions (selected), Trunks, and Inbound Routes. The main content area is titled 'FXS Extensions' and contains a 'Refresh' button and a table with the following data:

Port	Extension	Full Name	CallerID	Operation
1 1	6101		6101	Edit

It needs support of FXS module, the module installed in IP08 can be:

AX210S: dual FXS port

AX210XS: one FXO port and one FXS port

Analog phone is available to make calls once connected to the corresponding FXS port, IPPBX configures FXS extension automatically when FXS module is detected and they can't be deleted.

NOTE: FXS extension number range 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) Volume Settings

Rxgain

Adjust the volume sent to FXS extension. There are 3 options: 20%,60%,100%.

Txgain

Adjust the volume sent out by FXS extension. There are 3 options: 20%,60%,100%.

5) Flash**Hook Flash Detection Time**

Sets the amount of time, in milliseconds, that the hook-flash must remain depressed in order for IPPBX 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 IPPBX before it will recognize a second event. If a second event occurs in less time than defined in here, then IPPBX will ignore the event. The default value is 750 ms, and it can be configured in 1 ms increments.

6) 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.

Edit FXS Extension : 1
✕

General

Extension : <input style="width: 90%;" type="text" value="6101"/>	Port : <input style="width: 90%;" type="text" value="1"/>
Name : <input style="width: 90%;" type="text"/>	Caller ID : <input style="width: 90%;" type="text" value="6101"/>

Voice Mail

<input checked="" type="checkbox"/> Enable Voice Mail	Voice Mail Access PIN Code : <input type="text" value="123456"/>
---	--

Mail Setting

<input type="checkbox"/> Enable Sending Voice Mail	Email Address : <input style="width: 90%;" type="text"/>
--	--

Volume Settings

Rxgain : <input type="text" value="60%"/>	Txgain : <input type="text" value="60%"/>
---	---

Flash

Hook Flash Duration Time : <input type="text" value="1250"/>
Sequential Hook Flash Interval : <input type="text" value="750"/>

Follow Me

Call Forward : <input type="checkbox"/> Always <input checked="" type="checkbox"/> When no answer <input checked="" type="checkbox"/> When busy	Forward To : <input checked="" type="radio"/> Voice Mail <input type="radio"/> Number : <input style="width: 80%;" type="text"/>
---	--

Other Options

Pickup Group : <input type="text" value="0"/>	<input type="checkbox"/> Call Waiting	Ring Out : <input type="text" value="30"/>	<input checked="" type="checkbox"/> Use Web Interface
Storage Quota Privilege : <input type="text" value="Basic"/>	<input type="checkbox"/> DND	<input type="checkbox"/> Call Recording	

7) 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.

IPPBX 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. IPPBX 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 in IPPBX.

Restricted: 1 M

Basic: 2 M

Regular: 3 M

Privileged: 4 M

Super: 5 M

NOTE: If you set **System Settings -> External Storage** and set **Move Files Created Before 0 days ago**. This storage limitation will be of no use.

DND

Do not disturb, once checked, the extension will be unavailable.

Call Recording

Check this option to record all interior calls made by this extension, user can check the recordings on the personal web portal and with record account (password is the same with admin) too.

4.1 SIP extensions

SIP extension is a SIP Account that allows an IP Phone or an IP SoftPhone client to register on IPPBX. It can be created / modified / deleted one by one or in batch.

4.2.1 Add new extensions

a. Add single extension

Click  to add an extension

NOTE: SIP extension number range is defined in **PBX Settings -> Options -> Extension Preference**, changing it can create extensions in others number range.

1) General

Type

SIP

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.

Extension

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

Password

Authentication for SIP phone to register and make calls.

2) SIP Setting

NAT

Try this setting when IPPBX 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, IPPBX will route the media streams from SIP endpoints through itself. Enabling this option causes IPPBX to attempt to negotiate the endpoints to route the media stream directly, bypassing IPPBX. It is not always possible for IPPBX 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 IPPBX 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.

Transport

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

SIP Quality

Check this option to force keep alive.

3) 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.

4) 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.

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.

IPPBX 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. IPPBX 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.

Edit Voip Extension : 6000

General

Type : **Name :** **Caller ID :**
Extension : **Password :**

Sip Settings

NAT : **Can Reinvite :** **DTMF Mode :**
Transport : **SIP Qualify**

Voice Mail

Enable Voice Mail **Voice Mail Access PIN Code :**

Mail Setting

Enable Sending Voice Mail **Email Address :**

Follow Me

Call Forward : **When no answer** **When busy** **Forward To :** **Voice Mail** **Number :**

Other Options

Pickup Group : **Call Waiting** **Ring Out :** **Use Web Interface**
Storage Quota Privilege : **DND** **Call Recording**

Storage Quota Privilege

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

Restricted: 1 M

Basic: 2 M

Regular: 3 M

Privileged: 4 M

Super: 5 M

NOTE: If you have set **System Settings** -> **External Storage** and set Move Files Created Before 0 days ago. This storage limitation will be of no use.

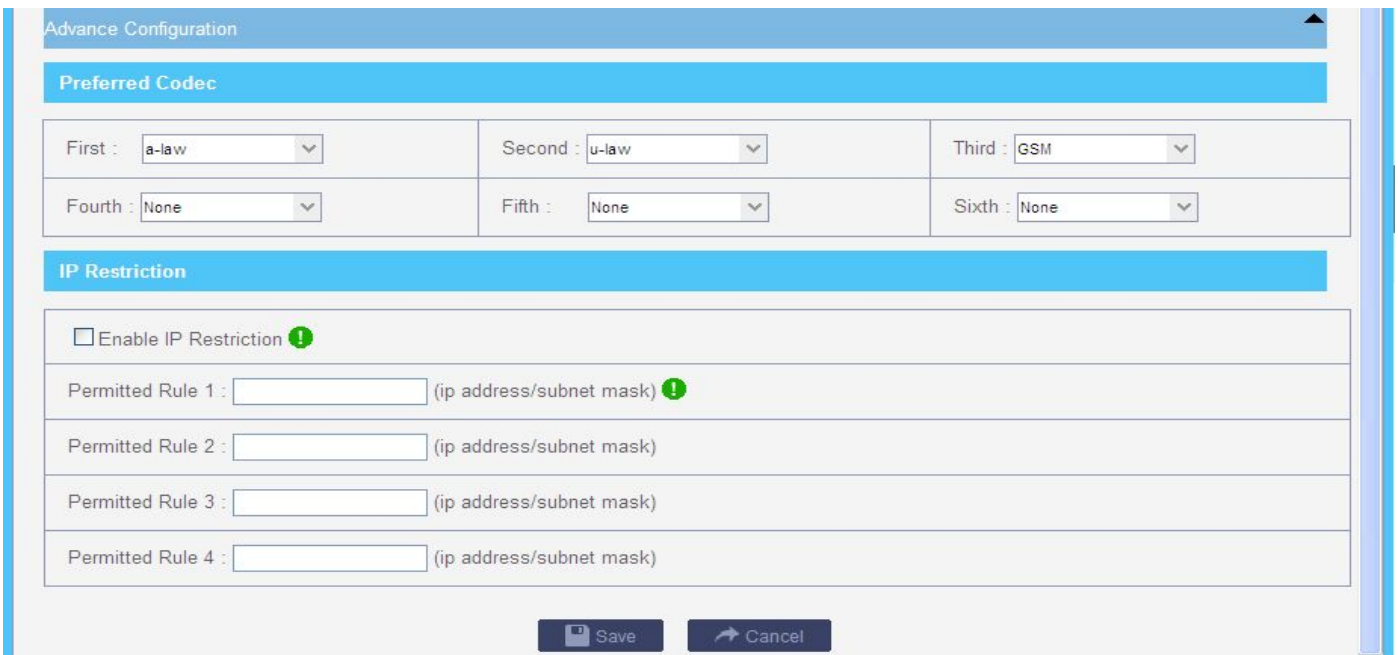
DND

Do not disturb, once checked, the extension will be unavailable.

Call Recording

Check this option to record all interior calls made by this extension, user can check the recordings on the personal web portal and with record account(password is the same with admin) too.

Advanced Configuration



The screenshot shows the 'Advance Configuration' page with two main sections:

- Preferred Codec:** A table with six columns for codec selection.

First : a-law	Second : u-law	Third : GSM
Fourth : None	Fifth : None	Sixth : None
- IP Restriction:** A section with a checkbox 'Enable IP Restriction' (unchecked) and four input fields for 'Permitted Rule 1' through '4', each with a '(ip address/subnet mask)' label and a warning icon.

At the bottom of the form are 'Save' and 'Cancel' buttons.

Preferred Codec

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

Audio: A-law, U-law, GSM, SPEEX, G726, G722, ADPCM, G729

Video: H261, H263, H263P, H264

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

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

- b. Click to add multiple extensions.

The dialog box titled "Add Multiple Extensions" contains the following fields and buttons:

- Create:** A dropdown menu with the value "10".
- Type:** A dropdown menu with the value "SIP/IAX".
- Extensions starting from:** An empty text input field.
- Buttons:** A "Create" button with a green checkmark icon and a "Cancel" button with a grey arrow icon.

4.2.2 Modify extensions

- a. Click behind an extension to edit the extension.
- b. Choose one or more extensions, click to modify them simultaneously.

3.1 Register onto IPPBX with your IP phone

The screenshot shows the "Account" configuration page for "Account 1". The page has a teal header with the "atcom" logo and navigation tabs: Basic, Account, Network, DSS Key, Phone, and Contacts. On the left, there is a sidebar with "Basic" selected, and sub-tabs for "Codecs" and "Advanced".

The main configuration area includes the following fields:

- Account: Account 1 (dropdown)
- Register Status: Registered
- Account Active: Enable (dropdown)
- Label: (empty text field)
- Display Name: (empty text field)
- Register Name: (empty text field)
- * User Name: 6000 (text field)
- Password: (masked with dots)
- * SIP Server: 192.168.1.78 (text field) | Port: 5060 (text field)
- Enable Outbound Proxy Server: Disable (dropdown)
- Outbound Proxy Server: (empty text field) | Port: 5060 (text field)
- Enable Peer to Peer: Disable (dropdown)
- Transport Type: UDP (dropdown)
- NAT Traversal: STUN (dropdown)

Red annotations with arrows point to specific fields:

- "Extension created in APBX" points to the "User Name" field.
- "The IP of APBX" points to the "SIP Server" field.
- "Must be the same as that in APBX for this extension" points to the "Transport Type" field.

Login the web administration page of IP phone and set the account information.

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

atcom

System Status
PBX Settings
Network Settings
System Settings
Reports
Logout

General
Trunk Status
> Extension Status

	Idle	Busy	Ringing	Hold	Unavailable
Status	Extension	Type	New Messages	Old Messages	
1	Unavailable	6000	SIP	0	0
2	Idle	6001	IAX	0	0
3	Unavailable	6002	SIP/IAX	0	0
4	Idle	6003	SIP	0	0
5	Ringing	6006	SIP	0	0
6	Idle	6007	SIP	0	0
7	Unavailable	6008	IAX	0	0
8	Unavailable	6009	SIP/IAX	0	0
9	Ringing	6101	FXS	0	0
10	Idle	6102	FXS	0	0

10 Page 1 of 1 Displaying 1 to 10 of 10 items

Refresh

3.2 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 number.
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

atcom		System Status	PBX Settings	Network Settings	System Settings	Reports	Logout	
<ul style="list-style-type: none"> Extensions Trunks Inbound Routes Outbound Routes > Feature Codes IVR Hunt/Ring Group Conference Callback Paging/Intercom Time Intervals Queue DISA Blacklist Options SIP Settings IAX Settings SMTP Settings Music On Hold Custom Prompts Custom Dialplan 	General							
	<input type="checkbox"/> Call Recording	*1	<input type="checkbox"/> Checking Voicemail	*2	<input checked="" type="checkbox"/> Attended Transfer	*3	<input checked="" type="checkbox"/> Blind Transfer	*03
	<input type="checkbox"/> Group Call Pickup	*4	<input type="checkbox"/> Direct Call Pickup	*04	<input type="checkbox"/> Intercom	*5		
	Call Park							
	<input type="checkbox"/> Call Park	*6	Extension Range to Park Calls :	701-720	Park Time Before Recalled(Second) :	60		
	Call Forward							
	<input type="checkbox"/> Reset to Default	*70	<input type="checkbox"/> Enable Unconditional Call Forward	*71	<input type="checkbox"/> Cancel Unconditional Call Forward	*071		
	<input type="checkbox"/> Enable Call Forward On Busy	*72	<input type="checkbox"/> Cancel Call Forward On Busy	*072	<input type="checkbox"/> Enable Call Forward On No-Answer	*73	<input type="checkbox"/> Cancel Call Forward On No-Answer	*073
	<input type="checkbox"/> Call Forward to Number	*74	<input type="checkbox"/> Call Forward to Voice Mail	*074	<input type="checkbox"/> Enable Do Not Disturb	*75	<input type="checkbox"/> Cancel Do Not Disturb	*075
	Save							

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

3.3 SMTP Settings

1) Voice Mail to Email Setting

Email Address

The Sender Email Address IPPBX 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 IPPBX 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.

Delete Messages After Notification

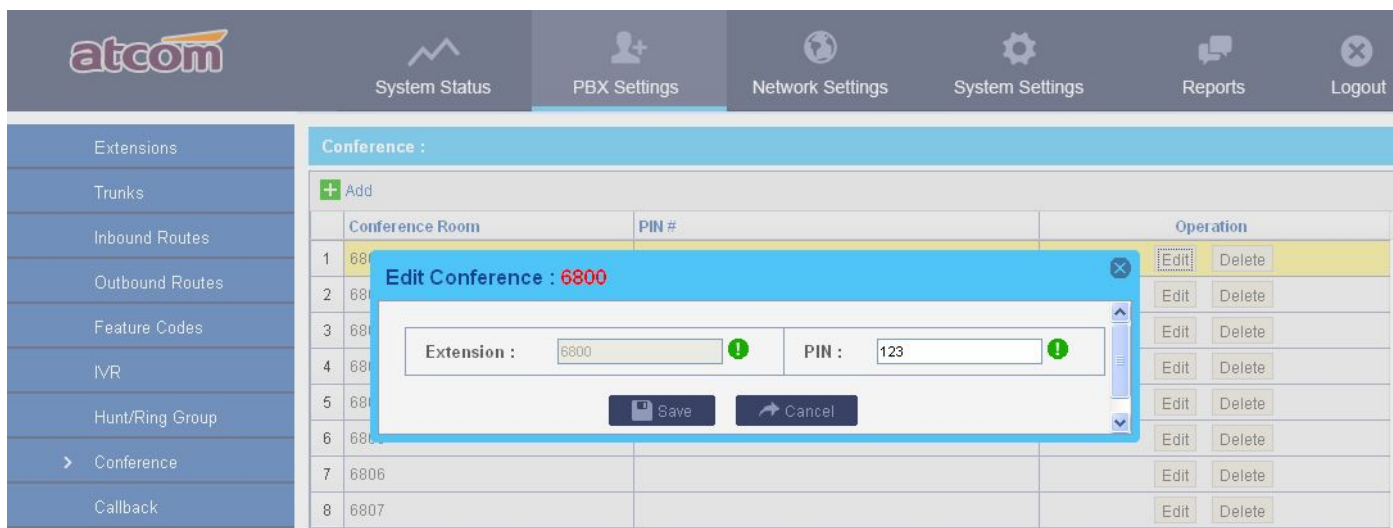
When voicemail to email is set, after voicemail is sent out via email successfully, the voicemail will be deleted from IPPBX. This is intended for use with users who wish to receive their voicemail ONLY by email.

Ask Caller to Dial 5

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

3.4 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, IPPBX will playback MoH for the first participant.

3.5 Paging / Intercom

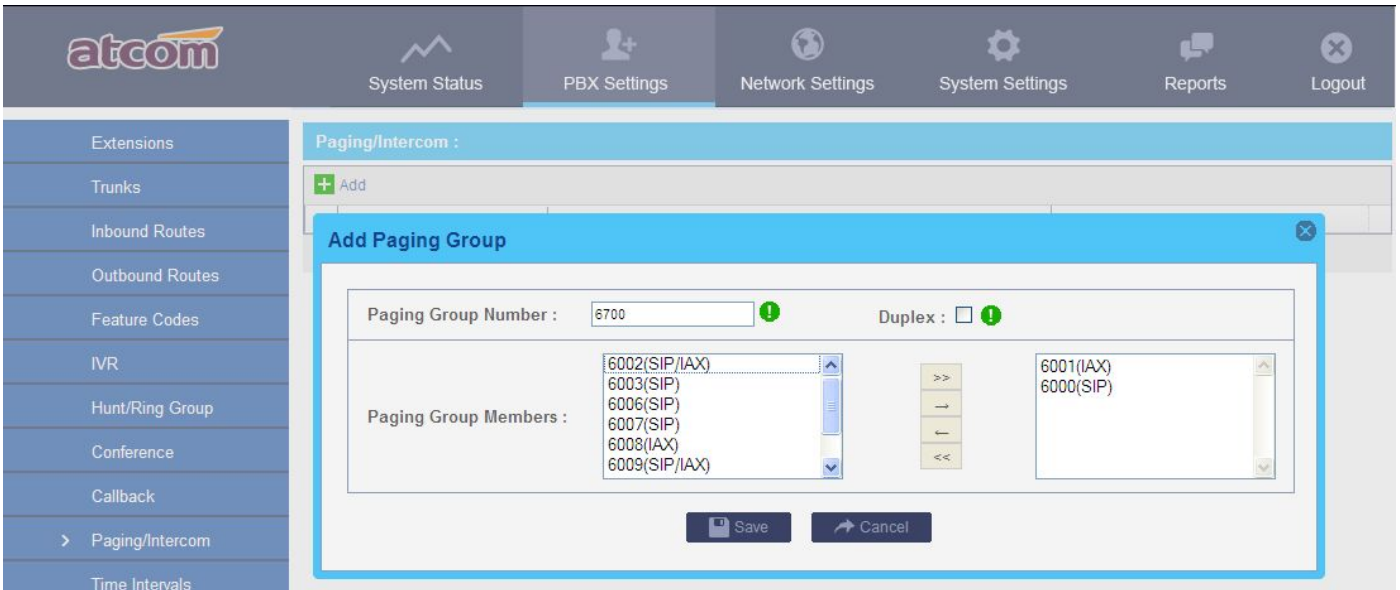
Dial a code and connect directly to a built-in two-way announcement and talkback function on one or more phones, support up to 20 participants.

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.



3.6 Options

1) General Preference

Ring Timeout

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

Max Call Duration

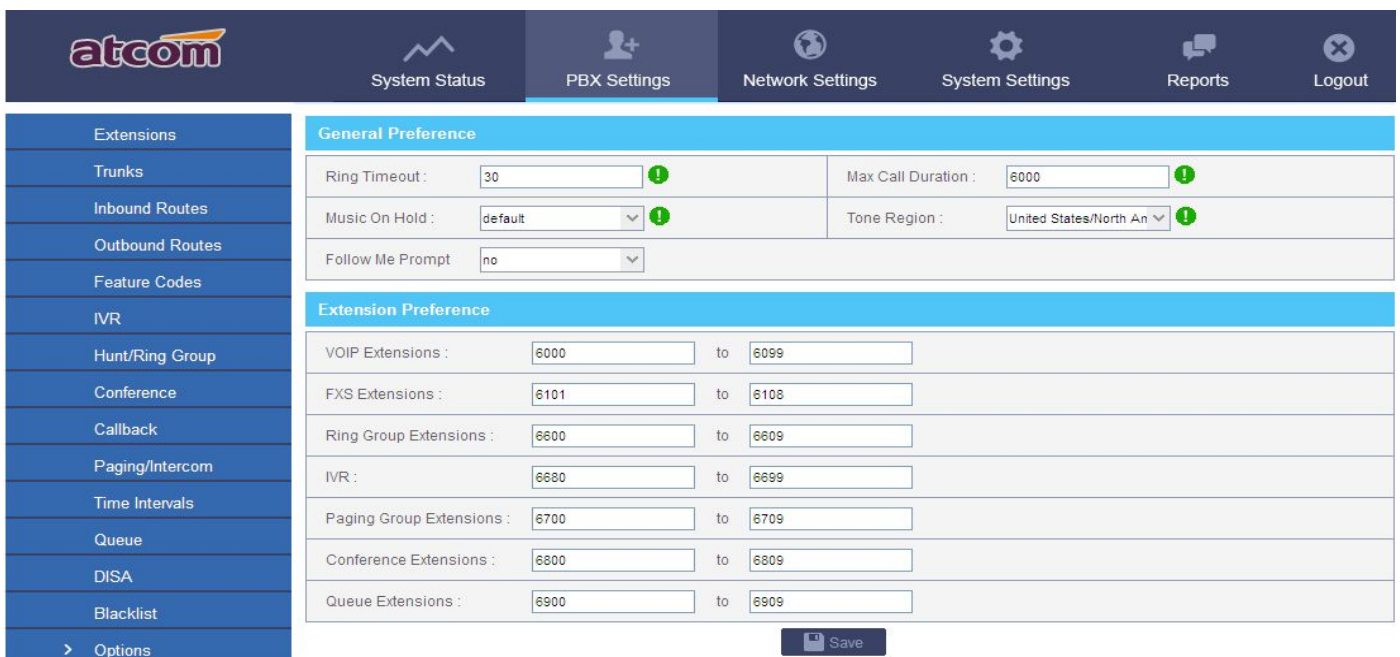
This defines the maximum amount of time in seconds for a interior call, 0 means no limit, default is 6000s.

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 VOIP / 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:

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

4. Create SIP trunk to communicate with VoIP provider

In this page, we can also configure the 'service provider' trunk, which doesn't need the use name and password for authorization, when you have bought a trunk from provide with IP address only, please choose 'service provider' trunk .

4.1 Create SIP trunks

Go to **PBX Settings** -> **Trunks**, Click  to add a new SIP trunk.

Type

SIP

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.

Call Recording

Check this option to record all exterior calls made by this trunk. User can check the recordings with record account(password is the same with admin) .

The screenshot shows a configuration window titled "New VoIP Trunk". The fields are as follows:

- Type: SIP
- Trunk Name: elastix
- Provider Hostname/IP: 192.168.1.173
- Account Name: 200
- Authuser: 200
- Password: 200
- Domain: (empty)
- From User: (empty)
- Call Recording:
- Enable Outbound Proxy:
- Outbound Proxy: (empty)
- Outbound Caller ID: (empty)
- SIP Transport: UDP
- Maximum Outbound Calls: 100
- Preferred Code:

First: a-law	Second: u-law	Third: GSM
Fourth: G729	Fifth: None	Sixth: None

Buttons: Save, Cancel

Advance Configuration

DOD(Direct Outward Dialing Number) Setting

Set the Outbound number for different extensions.

For example:

Advance Configuration

DOD Setting

DOD :123456 Associated Extension : 6000

DOD Number : Associated Extension :



< === 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.

atcom

System Status PBX Settings Network Settings System Settings Reports Logout

Type	Trunk Name	Status	Port/HostName/IP
1 SIP	Elastix	Registered	192.168.1.158
2 SIP-SIP	IP08	Registered	192.168.1.157
3 FXO	FXO2	Disconnected	PORT2

Refresh

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 6.

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 dial out with this trunk.

Outbound Trunk Selection: Select trunks which calls are forwarded out through.

Add Outbound Route
✕

General

Outbound Route Name : !

This place will be replaced....

Dial Pattern : ! Strip ! digits from front

Prepend these digits : before dialing !

Password : !

Outbound Extension Selection

Available Extensions		Selected Extensions
<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> 6002(SIP/IAX) 6003(SIP) 6006(SIP) 6007(SIP) 6008(IAX) 6009(SIP/IAX) </div>	>> → ← <<	<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> 6000(SIP) 6001(IAX) 6102(fxs2) </div>

Outbound Trunk Selection

Available Trunks		Selected Trunks
<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> FX03 FX04 GSM5 GSM7 sv4 u520 </div>	>> → ← <<	<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> elastix </div>

4.4 Make inbound calls

Go to **PBX Settings** -> **Inbound Routes**, click **Add** to add an inbound route.

1) General

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. The DID for SIP trunk is usually as same as the account (others can only dial the account number to dial into PBX), then just leave it blank is OK .

Extension

Define the extension for DID number. This field is usually unavailable for most VoIP trunk, if you VoIP provider support DID, you can set it. 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.

Add Inbound Route ✕

General

Inbound Route Name : <input style="width: 90%;" type="text" value="fromElastix"/> !	Caller ID : <input style="width: 90%;" type="text" value="2XX"/> !
DID Number : <input style="width: 90%;" type="text" value="200"/> !	Extension : <input style="width: 90%;" type="text"/> !

Inbound Trunk Selection

Available Trunks		Selected Trunks
<div style="border: 1px solid #ccc; padding: 2px;"> FX03 FX04 GSM5 GSM7 sv4 u520 </div>	>> → ← <<	<div style="border: 1px solid #ccc; padding: 2px;"> elastix </div>

Time

Time Interval : ▼

Path

Destination Type : <input style="width: 90%;" type="text" value="Extension"/> ▼	Destination : <input style="width: 90%;" type="text" value="6000(SIP)"/> ▼
--	---

Fax Detection

Destination : <input style="width: 90%;" type="text" value="Faxes"/> ▼	<input style="width: 90%;" type="text" value="6001(IAX)"/> ▼
---	---

Save
Cancel

2) Inbound Trunk Selection

Select the trunks for which this inbound route apply.

3) Time

Select appropriate time intervals for when this inbound route apply.

4) Path

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

5. Create Service Provider to connect two IP PBXes

Service Provider is used to interconnect two IP PBXes, it must be created in both IP PBXes, after setting service provider and outbound route, extensions of each IP PBX can call the other's directly according to the outbound route.

5.1 Create Service Provider

Go to **PBX Settings -> Trunks**, Click  to add a new service provider.

Type

SIP

Provider Hostname/IP

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

Provider Name

A unique label to help you identify the service provider.

Call Recording

Check this option to record all calls made by this trunk. User can check the recordings with record account(password is the same with admin) .

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.

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.

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. Detailed example can be found in Chapter 4.1.

5.2 Check Service Provider Status

After setting service provider in the other PBX, the status will become to Registered.

Type	Trunk Name	Status	Port/HostName/IP
1 SIP	Elastix	Registered	192.168.1.158
2 SIP-SIP	IP08	Registered	192.168.1.157
3 FXO	FXO2	Disconnected	PORT2

[Refresh](#)

5.3 Make outbound calls

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

The setting here is as same as that for SIP trunk, please refer to Chapter 4.3 to set it.

Add Outbound Route

General

Outbound Route Name : !

This place will be replaced....

Dial Pattern : ! Strip ! digits from front

Prepend these digits : before dialing !

Password : !

Outbound Extension Selection

Available Extensions	Selected Extensions
6002(SIP/IAX) 6003(SIP) 6006(SIP) 6007(SIP) 6008(IAX) 6009(SIP/IAX)	6000(SIP) 6001(IAX) 6101(fxs1)

Outbound Trunk Selection

Available Trunks	Selected Trunks
FXO3 FXO4 GSM5 GSM7 elastix sv4	IP08

Then 6000/6001 can dial 900-999 to corresponding extension (900-999) in other end directly.

5.4 Make inbound calls

Go to **PBX Settings -> Inbound Routes**, click  to add an inbound route.

Even there is no inbound route set, two IPPBXes can communicate with each other, however we can make it more functional with setting DID to those differ from local extensions.

Example 1: DID + Extension--Make calls more easier.

DID: 00-03 Extension: 6000-60003

Extensions from other end can dial 6000-6003 extension directly to reach these extensions, besides, they can dial 00-03 to reach them if their outbound route allow. In this case, Path is of no use.

Example 2: DID + Path--Forward calls to other applications.

DID: 6600 Path: Ring Group

Extensions from other end can dial local extensions directly, besides, they can dial 6600 to reach the destination set in Path. In this case, they can dial 6600 to reach the ring group.

Add Inbound Route ✕

General

Inbound Route Name : <input style="width: 90%;" type="text" value="IP08"/>	Caller ID : <input style="width: 90%;" type="text"/>
DID Number : <input style="width: 90%;" type="text" value="6600"/>	Extension : <input style="width: 90%;" type="text"/>

Inbound Trunk Selection

Available Trunks		Selected Trunks
FX04 GSM5 GSM7 elastix sv4 u520	>> ← ← <<	IP08

Time

Time Interval :

Path

Destination Type : <input type="text" value="Ring Group"/>	Destination : <input type="text" value="6600"/>
--	---

Fax Detection

Destination : <input type="text" value="Faxes"/>	<input type="text" value="6001(IAX)"/>
--	--

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

6.1 Make sure FXO modules are installed

If there are FXO modules installed in your IPPBX, IPPBX configures analog trunk automatically when they are detected.

System Status
PBX Settings
Network Settings
System Settings
Reports
Logout

Extensions > Trunks Inbound Routes	Analog Trunk :						
	Refresh <input type="button" value="Refresh"/>						
	<table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="width: 5%;">Trunk Name</th> <th style="width: 45%;">Port</th> <th style="width: 50%;">Operation</th> </tr> </thead> <tbody> <tr> <td style="text-align: center;">1</td> <td style="text-align: center;">FX02</td> <td style="text-align: center;">2 <input type="button" value="Edit"/></td> </tr> </tbody> </table>	Trunk Name	Port	Operation	1	FX02	2 <input type="button" value="Edit"/>
Trunk Name	Port	Operation					
1	FX02	2 <input type="button" value="Edit"/>					

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**

The screenshot shows the atcom web interface with a navigation menu at the top containing System Status, PBX Settings, Network Settings, System Settings, Reports, and Logout. On the left, there is a sidebar with options for General, Trunk Status (selected), and Extension Status. The main content area displays a table of trunk status information.

Type	Trunk Name	Status	Port/HostName/IP
1 SIP	Elastix	Failed	192.168.1.158
2 SIP-SIP	IP08	Registered	192.168.1.157
3 FXO	FXO2	InService	PORT2

A Refresh button is located at the bottom right of the table.

6.2 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 6000,6001 can prefix 9 to dial out.

Outbound Trunk Selection

Select trunks which calls are forwarded out through. If there are more than one FXO trunk chose, PBX will try to dial out through next trunk if the previous one failed.

Add Outbound Route ✕

General

Outbound Route Name : !

This place will be replaced....

Dial Pattern : ! Strip ! digits from front

Prepend these digits : before dialing !

Password : !

Outbound Extension Selection

Available Extensions		Selected Extensions
<div style="border: 1px solid #ccc; padding: 2px;"> 6003(SIP) 6006(SIP) 6007(SIP) 6008(IAX) 6009(SIP/IAX) 6102(fxs2) </div>	>> → ← <<	<div style="border: 1px solid #ccc; padding: 2px;"> 6000(SIP) 6001(IAX) 6101(fxs1) </div>

Outbound Trunk Selection

Available Trunks		Selected Trunks
<div style="border: 1px solid #ccc; padding: 2px;"> FXO4 GSM5 GSM7 IP08 elastix sv4 </div>	>> → ← <<	<div style="border: 1px solid #ccc; padding: 2px;"> FXO3 </div>

6.3 Make inbound calls

Go to **PBX Settings -> Inbound Routes**, click to add an inbound route. Just setting Selected Trunks and Path is OK.

1) General

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.

2) Inbound Trunk Selection

Select the trunks for which this inbound route apply.

3) Time

Select appropriate time intervals for when this inbound route apply.

4) Path

Set the destination for incoming calls.

Add Inbound Route ✕

General

Inbound Route Name : <input style="width: 90%;" type="text" value="fromPSTN"/> !	Caller ID : <input style="width: 90%;" type="text"/> !
DID Number : <input style="width: 90%;" type="text"/> ! Leave them blank	Extension : <input style="width: 90%;" type="text"/> !

Inbound Trunk Selection

Available Trunks		Selected Trunks
<div style="border: 1px solid #ccc; padding: 2px;"> GSM5 GSM7 IP08 elastix sv4 u520 </div>	>> → ← <<	<div style="border: 1px solid #ccc; padding: 2px;"> FX03 </div>

Time

Time Interval : v

Path

Destination Type : <input style="width: 90%;" type="text" value="Extension"/> v	Destination : <input style="width: 90%;" type="text" value="6002(SIP/IAX)"/> v
--	---

Fax Detection

Destination : <input style="width: 90%;" type="text" value="No Detect"/> v	<input style="width: 90%;" type="text"/> v
---	---

Save
Cancel

7. Inbound Call Control

7.1 Time Interval

Set the Time Interval for inbound route.

Add Time Intervals
✕

Time Interval Name :

By day of week

to

By Days of a Month

Date : to Month :

Time: Entire Day

Start Time : End Time :

7.2 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.

Edit Hunt/Ring Group : 6600
✕

Ring Group Name : ! Ring Group Number : !

Ignore Call Forward ! Enable MOH ! Music On Hold :

Ring Mode : Parallel ! Serial !

Ring interval : !

Ring group members : !

Available Extensions :

- 6002(SIP/IAX)
- 6003(SIP)
- 6006(SIP)
- 6007(SIP)
- 6008(IAX)
- 6009(SIP/IAX)

>>

→

←

<<

Selected Extensions :

- 6000(SIP)
- 6001(IAX)

No answer forward to : !

Ignore Call Forward

By default, if there is call forward enabled in IP phone, IPPBX will call that forwarding number. when this option is checked, IPPBX will ignore that call forward setting.

Enable MoH

When MoH is choose, caller will hear MoH other than ring tone.

Parallel

Ring all the members at the same time.

Serial

Ring all the members one by one.

No answer forward to

Set the failover destination if there is no answer.

7.3 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 IPPBX 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 IPPBX 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.

Edit Call Queue : 6900
✕

General

Queue Name : <input style="width: 90%;" type="text" value="6900"/>	Queue Number : <input style="width: 90%;" type="text" value="6900"/>
Queue Password : <input style="width: 90%;" type="text" value="1234"/>	Queue Agent Timeout : <input style="width: 90%;" type="text" value="45"/>
Queue Max Wait Time : <input style="width: 90%;" type="text" value="1800"/>	Queue Ringing Strategy : <input style="width: 90%;" type="text" value="Ring All"/>

Agents

Available Agents :

6002(SIP/IAX)
 6003(SIP)
 6006(SIP)
 6007(SIP)
 6008(IAX)
 6009(SIP/IAX)

>>
→
←
<<

Selected Agents :

6000(SIP)
 6001(IAX)
 6101(fxs1)

Caller Position Announcement

Announce Position : <input style="width: 90%;" type="text" value="Yes"/>	Announce Holdtime : <input style="width: 90%;" type="text" value="Yes"/>
Frequency : <input style="width: 90%;" type="text" value="15s"/>	

Period Announcement

Prompt : <input style="width: 90%;" type="text" value="hello-world"/>	Frequency : <input style="width: 90%;" type="text" value="40s"/>
---	--

Event

Key : <input style="width: 90%;" type="text" value="*"/>	Action : <input style="width: 90%;" type="text" value="End Call"/>	Destination : <input style="width: 90%;" type="text"/>
--	--	--

Failover Destination

Action : <input style="width: 90%;" type="text" value="End Call"/>	Destination : <input style="width: 90%;" type="text" value="0"/>
--	--

Others

Music On Hold : <input style="width: 90%;" type="text" value="default"/>	Leave When Empty : <input style="width: 90%;" type="text" value="No"/>
Join Empty : <input style="width: 90%;" type="text" value="Yes"/>	Agent Announcement : <input style="width: 90%;" type="text" value="hello-world"/>
Join Announcement : <input style="width: 90%;" type="text" value="hello-world"/>	Retry : <input style="width: 90%;" type="text" value="30"/>
Wrap Up Time : <input style="width: 90%;" type="text" value="30"/>	

7.4 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.

Edit IVR : 6680
✕

IVR Name : <input style="width: 90%;" type="text" value="6680"/>	IVR Number : <input style="width: 90%;" type="text" value="6680"/>
Key Timeout : <input style="width: 90%;" type="text" value="3"/>	Repeat Count : <input style="width: 20%;" type="text" value="3"/> times
Prompt : <input style="width: 90%;" type="text" value="hello-world"/>	Custom Prompt
<input type="checkbox"/> Allow Dialing of Other Extensions	
<input checked="" type="checkbox"/> Press 0 to trigger : <input style="width: 150%;" type="text" value="Extension"/>	<input style="width: 150%;" type="text" value="6001(IAX)"/>
<input checked="" type="checkbox"/> Press 1 to trigger : <input style="width: 150%;" type="text" value="Voice Mail"/>	<input style="width: 150%;" type="text" value="6000(SIP)"/>
<input type="checkbox"/> Press 2 to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press 3 to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press 4 to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press 5 to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press 6 to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press 7 to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press 8 to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press 9 to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press * to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
<input type="checkbox"/> Press # to trigger : <input style="width: 150%;" type="text"/>	<input style="width: 150%;" type="text"/>
No Entry Forward to : <input style="width: 150%;" type="text" value="No Action"/>	<input style="width: 150%;" type="text"/>
Invalid Forward to : <input style="width: 150%;" type="text" value="No Action"/>	<input style="width: 150%;" type="text"/>

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 IPPBX 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

7.5 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 IPPBX 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.

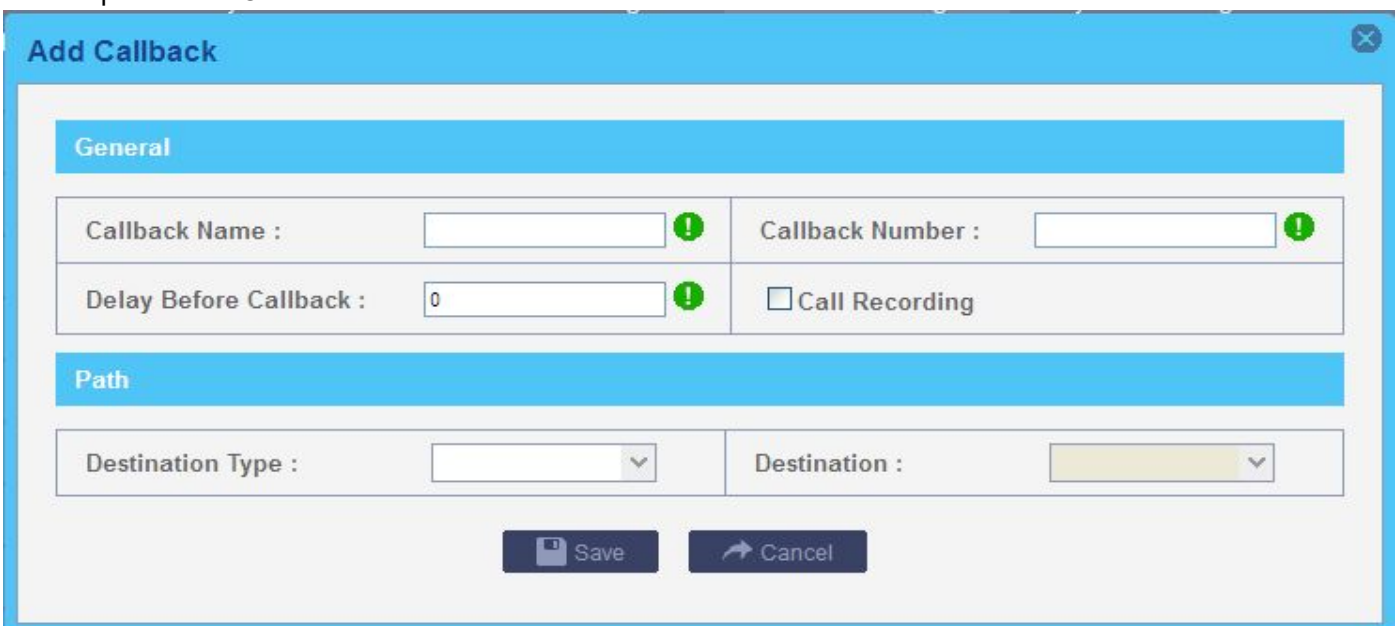
For example: Both City A and B have a IPPBX, IPPBX-A and IPPBX-B, they are connected with SIP trunk, and IPPBX-A has FXO trunk to connect local PSTN and outbound route for that, DISA can be used as below:

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

After users of IPPBX-B dial into DISA application in IPPBX-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.

7.6 Call Back

A callback will hang up on the caller and then call them back, directing them to the selected destination. This is useful for reducing mobile phone charges as well as other applications. Outbound calls will proceed according to the dial patterns in Outbound Routes



Callback Name

A name for the Callback

Callback Number

Enter the number to dial for the callback. Leave this blank to just dial the incoming Caller ID Number.

Delay before Callback

Enter the number of seconds the system should wait before calling back.

Call Recording

Check this option to record all calls made by this Call back. User can check the recording with record account(password is the same with admin).

PATH

Choose the destination which IPPBX will bridge to caller.

For Example:

1. Set Callback:

Edit Callback : pstn ✕

General

Callback Name : <input style="width: 90%;" type="text" value="pstn"/> !	Callback Number : <input style="width: 90%;" type="text"/> !
Delay Before Callback : <input style="width: 90%;" type="text" value="3"/> !	<input type="checkbox"/> Call Recording

Path

Destination Type : <input style="width: 90%;" type="text" value="Extension"/> ▼	Destination : <input style="width: 90%;" type="text" value="6002(SIP/IAX)"/> ▼
---	--

2. Set inbound route for FXO trunk:

Add Inbound Route ✕

General

Inbound Route Name : <input style="width: 90%;" type="text" value="fromPSTN"/> !	Caller ID : <input style="width: 90%;" type="text"/> !
DID Number : <input style="width: 90%;" type="text"/> !	Extension : <input style="width: 90%;" type="text"/> !

Inbound Trunk Selection

Available Trunks	Selected Trunks
<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> FXO4 GSM5 GSM7 IP08 elastix sv4 </div>	<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> FXO3 </div>

Time

Time Interval : <input style="width: 90%;" type="text"/>	▼
--	---

Path

Destination Type : <input style="width: 90%;" type="text" value="Callback"/> ▼	Destination : <input style="width: 90%;" type="text" value="pstn"/> ▼
--	---

Fax Detection

Destination : <input style="width: 90%;" type="text" value="No Detect"/> ▼	▼
--	---

3. Dial into IPPBX from PSTN number, and hang up immediately once the call is connected. Then IPPBX dial extension 6002, after 6002 answer the call, IPPBX dial caller and bridge them.

7.7 Inbound to Outbound

Outbound Routes can act as destination in Inbound route settings, it let two ends connecting to IPPBX with trunk communicate directly.

7.7.1 FXO trunk to SIP trunk

1. Set outbound route for SIP trunk.
2. Set Inbound route for FXO trunk.

Destination

Choose the proper SIP trunk

Number

Set a number on the other end of SIP trunk. it must be set.

3. Dial into IPPBX from PSTN, then IPPBX will dial 202 through SIP trunk.

Edit Inbound Route : fPSTN
✕

General

Inbound Route Name : <input type="text" value="fPSTN"/> !	Caller ID : <input type="text"/>
DID Number : <input type="text"/>	Extension : <input type="text"/>

Inbound Trunk Selection

Available Trunks	Selected Trunks
	FXO4

Time

Time Interval :

Path

Destination Type :
 Destination :
 Number :

Fax Detection

Destination :

7.7.2 SIP trunk to FXO trunk

1. Set outbound route for FXO trunk.
2. Set Inbound route for SIP trunk/Service Provider. Since service provider is more flexible, here we use Service Provider as an example

Destination

Choose the proper FXO trunk

Number

Set a number on the other end of FXO trunk. it can be blank, then it means the number caller dialed.

3. Dial into IPPBX from SIP trunk, e.g. 83018618, then IPPBX will dial that number through FXO trunk selected in PSTN outbound route.

Edit Inbound Route : fElastix
✕

General

Inbound Route Name : <input style="width: 90%;" type="text" value="fElastix"/> !	Caller ID : <input style="width: 90%;" type="text"/> !
DID Number : <input style="width: 90%;" type="text"/> !	Extension : <input style="width: 90%;" type="text"/> !

Inbound Trunk Selection

Available Trunks		Selected Trunks
FXO4	>> → ← <<	Elastix

Time

Time Interval :

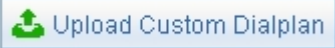
Path

Destination Type :
 Destination :
 Number :

Fax Detection

Destination :

7.8 Custom Dialplan

Click  to upload Custom Dialplan. There are three types file: user, dialplan and dahdi.

user

This file aimed at defining a "user". It can define a user with an optional SIP phone, Dahdi phone and/or just about any other type of phone.

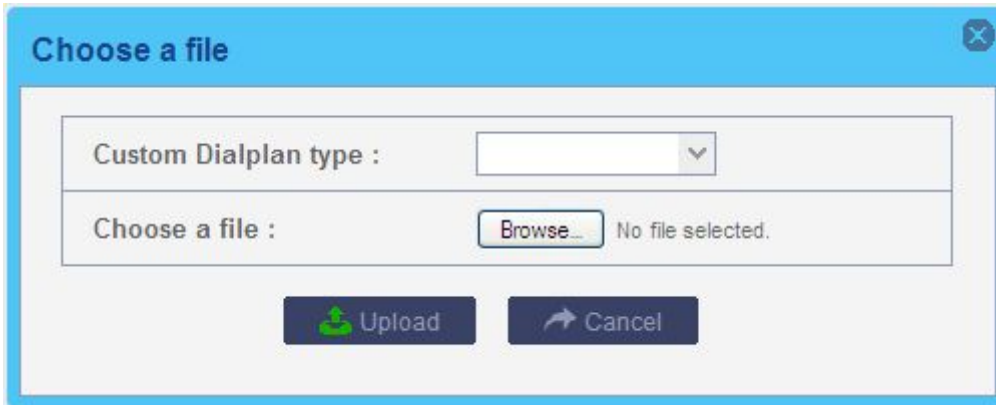
dialplan

This file contains the "dial plan" of Asterisk, the master plan of control or execution flow for all of its operations. It controls how incoming and outgoing calls are handled and routed. This is where you configure the behavior of all connections through IPPBX.

dahdi

Asterisk cmd DAHDILookup configuration.

NOTE: You must prefix `custom_` to your custom file and context to avoid conflict with the system file, `users.conf`, `extensions.conf` and `chan_dahdi.conf`.




Choose a file

Custom Dialplan type :

Choose a file : No file selected.

7.9 Blacklist

Block incoming calls from specified numbers



New Blacklist

Blacklist Number :

If a number in blacklist dial into IPPBX, 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.

7.10 SIP Settings

1) General

UDP Port

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

Enable TCP

Enable TCP protocol for SIP.

TCP Port

Set the SIP port (TCP) which IPPBX 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

General			
UDP Port :	<input type="text" value="5060"/>	<input type="checkbox"/> Enable TCP	TCP Port : <input type="text" value="5060"/>
Registration/Subscription Time Max :	<input type="text" value="3600"/>	Registration/Subscription Time Min :	<input type="text" value="60"/>
RTP Port Max :	<input type="text" value="20000"/>	RTP Port Min :	<input type="text" value="10000"/>
DTMF Mode :	<input type="text" value="rfc2833"/>	Video Support :	<input type="text" value="yes"/>

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>

NAT	
Enable STUN : <input checked="" type="checkbox"/>	
STUN Server : <input type="text" value="stun.sipgate.net"/>	STUN Port : <input type="text" value="10000"/>
External IP Address : <input type="text"/>	External Host : <input type="text"/>
External Refresh Interval : <input type="text"/>	NAT Mode : <input type="text" value="no"/>
Local Network Identification : <input type="text"/>	Allow RTP Reinvite : <input type="text" value="no"/>

b. NAT

NAT	
Enable STUN : <input type="checkbox"/>	
STUN Server : <input type="text"/>	STUN Port : <input type="text"/>
External IP Address : <input type="text"/>	External Host : <input type="text" value="atcomtest.f3322.org"/>
External Refresh Interval : <input type="text" value="10"/>	NAT Mode : <input type="text" value="yes"/>
Local Network Identification : <input type="text" value="92.168.1.0/255.255.255.0"/>	Allow RTP Reinvite : <input type="text" value="nonat"/>

The External IP, External Host and Local Network Identification settings are used if you use IPPBX 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 IPPBX 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 IPPBX to stay in the media path, the media will be redirected. This does not really work with in the case where IPPBX is outside and have clients on the inside of a NAT. In that case, you want to set this option to no nat.

8. Audios

8.1 Music On Hold

Manage audio files for Music On Hold, the format should be .WAV (16 bit, mono 8000 Hz) and .GSM, the size should less than 4 MB.

Music On Hold List	
 Upload MOH	
Name	Operation
1 default	<input type="button" value="Play"/> <input type="button" value="Delete"/>

8.2 Custom Prompts

Manage prompts used for Voice Menu. It can be recorded by extensions or uploaded from local PC, the format should be .WAV (16 bit, mono 8000 Hz) and .GSM, the size should less than 4 MB

Custom Prompts :	
<input type="button" value="+ Record New Prompt"/> <input type="button" value="Upload a Prompt"/>	
Name	Operation
1 hello-world	<input type="button" value="Record Again"/> <input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>
2 intro	<input type="button" value="Record Again"/> <input type="button" value="Play"/> <input type="button" value="Download"/> <input type="button" value="Delete"/>

8.3 Language Setting

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

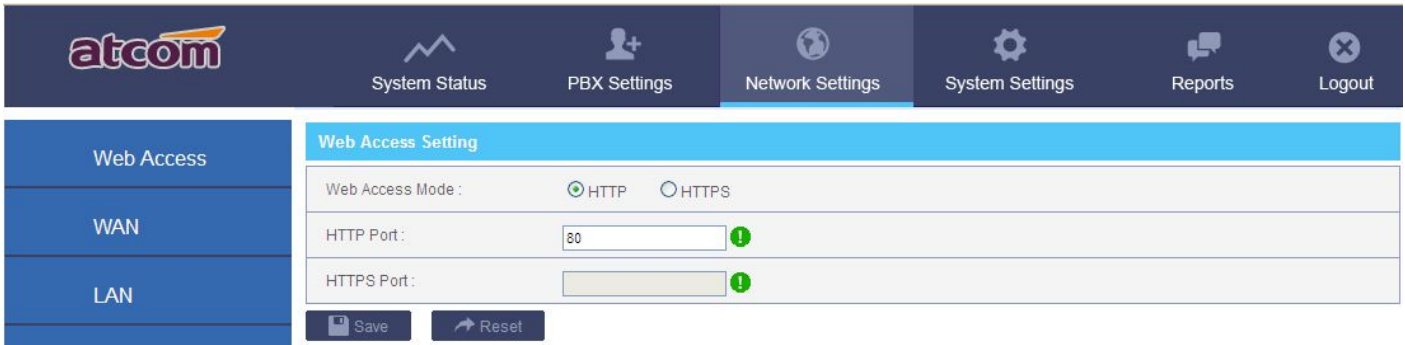
Extensions Trunks Inbound Routes Outbound Routes	Language Setting
Language : <input type="text" value="English"/> 	
<input type="button" value="Save"/>	

9. Network Settings

Description of LAN, WAN and DHCP server settings can be found in Chapter 2. All network settings will take effect immediately when you save the change.

9.1 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.



9.2 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 IPPBX uses for each application are list below:

HTTP	TCP:80
HTTPS	TCP:443
SIP	UDP:5060
SIP	TCP:5060
RTP	UDP:10000-20000

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.

New Firewall Common Rule

Name : SIPlocal !

Description : accept sip package from local network !

Protocol : UDP !

Port : 5060 : 5063 !

IP : 192.168.1.0 / 255.255.255.0 !

Mac Address : !

Action : ACCEPT !

Save Cancel

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

New Auto Defense Rule

Port : 80 !

Protocol : TCP !

Rate : 50 / Second !

Save Cancel

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.

Port :	<input type="text" value="5060"/>	Protocol :	<input type="text" value="UDP"/>
SIP Packets :	<input type="text" value="200"/>	Time Interval :	<input type="text" value="1"/> seconds

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.

6) Firewall Setting Example :

Firewall setting

Enable Firewall

Common Rule

+ New Rule

	Action	Name	Protocol	IP	Mac Address	Port	Operation
1	ACCEPT	SIPlocal	UDP	192.168.1.0/255.255.255.0		5060:5063	<input type="button" value="Edit"/> <input type="button" value="Delete"/>
2	ACCEPT	SIPprovider	UDP	216.207.245.47/255.255.255.255		5060:5063	<input type="button" value="Edit"/> <input type="button" value="Delete"/>
3	DROP	dropothers	UDP			5060:5063	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

Auto Defense

+ New Rule

	Port	Protocol	Rate	Operation
1	80	TCP	50	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

SIP Defense

+ New Rule

	SIP Packets	Time in Seconds	Operation
1	200	1	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

Other Options

Disable Ping
 Drop All

9.3 Port Forwarding

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

New Port Forward ✕

WAN Port : <input style="width: 80%;" type="text" value="8088"/>	LAN IP : <input style="width: 80%;" type="text" value="192.168.10.2"/>
LAN Port : <input style="width: 80%;" type="text" value="80"/>	Protocol : <input style="width: 80%;" type="text" value="TCP"/>

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)

9.4 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 4 DDNS servers below, please go to the website of below servers and apply a domain name then fill related information here.

dyndns.org

dyndns.com

qdns

www.zoneedit.com

DDNS Setting	
Enable DDNS :	<input type="text" value="Yes"/>
DDNS Server :	<input type="text" value="dyndns.org"/>
User Name :	<input type="text" value="voipadmin"/>
Password :	<input type="password" value="*****"/>
Hostname :	<input type="text" value="atcomtest.f3322.org"/>

9.5 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.

VLAN over WAN	
No. 1 :	<input checked="" type="checkbox"/>
No. 2 :	<input checked="" type="checkbox"/>
VLAN ID :	<input type="text" value="100"/>
VLAN ID :	<input type="text" value="200"/>
VLAN IP :	<input type="text" value="192.168.100.100"/>
VLAN IP :	<input type="text" value="192.168.200.100"/>
VLAN Subnet Mask :	<input type="text" value="255.255.255.0"/>
VLAN Subnet Mask :	<input type="text" value="255.255.255.0"/>
Default Gateway :	<input type="text" value="192.168.100.1"/>
Default Gateway :	<input type="text" value="192.168.200.1"/>

VLAN over LAN	
No. 1 :	<input type="checkbox"/>
No. 2 :	<input type="checkbox"/>
VLAN ID :	<input type="text"/>
VLAN ID :	<input type="text"/>
VLAN IP :	<input type="text"/>
VLAN IP :	<input type="text"/>
VLAN Subnet Mask :	<input type="text"/>
VLAN Subnet Mask :	<input type="text"/>
Default Gateway :	<input type="text"/>
Default Gateway :	<input type="text"/>

9.6 VPN

A virtual private network (VPN) extends a private network across a public network, such as the Internet. It enables a computer to send and receive data across shared or public networks as if it were directly connected to the


private network, while benefitting from the functionality, security and management policies of the private network. This is done by establishing a virtual point-to-point connection through the use of dedicated connections, encryption, or a combination of the two. Currently, only PPTP is supported.

VPN Setting	
<input checked="" type="checkbox"/> Enable VPN	
VPN Type :	<input type="text" value="PPTP"/>
VPN Server :	<input type="text" value="bijou.tenacy-free.com"/>
User Name :	<input type="text" value="tenacy"/>
Password :	<input type="text" value="91706"/>
<input type="button" value="Save"/>	

10. System Settings

10.1 Change Password

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

 System Status PBX Settings Network Settings System Settings Reports Logout									
<ul style="list-style-type: none"> > Change Password Auto Provisioning Date & Time 	<table border="1"> <thead> <tr> <th colspan="2">Change Password</th> </tr> </thead> <tbody> <tr> <td>New Password :</td> <td><input type="text"/></td> </tr> <tr> <td>Retype New Password :</td> <td><input type="text"/></td> </tr> <tr> <td colspan="2"><input type="button" value="Save"/></td> </tr> </tbody> </table>	Change Password		New Password :	<input type="text"/>	Retype New Password :	<input type="text"/>	<input type="button" value="Save"/>	
Change Password									
New Password :	<input type="text"/>								
Retype New Password :	<input type="text"/>								
<input type="button" value="Save"/>									

10.2 Auto Provisioning

IPPBX can only configure ATCOM AT8XX serials IP phone via DHCP + TFTP way and SIP Subscribe + Notify way. Make sure your AT8XX IP phone is using the latest firmware.

NOTE: DHCP + TFTP way can only be used when IPPBX is act as router and DHCP. User should reboot the phone manually after the configuration on web page.

10.2.1 Auto Provisioning

1) Phonebook

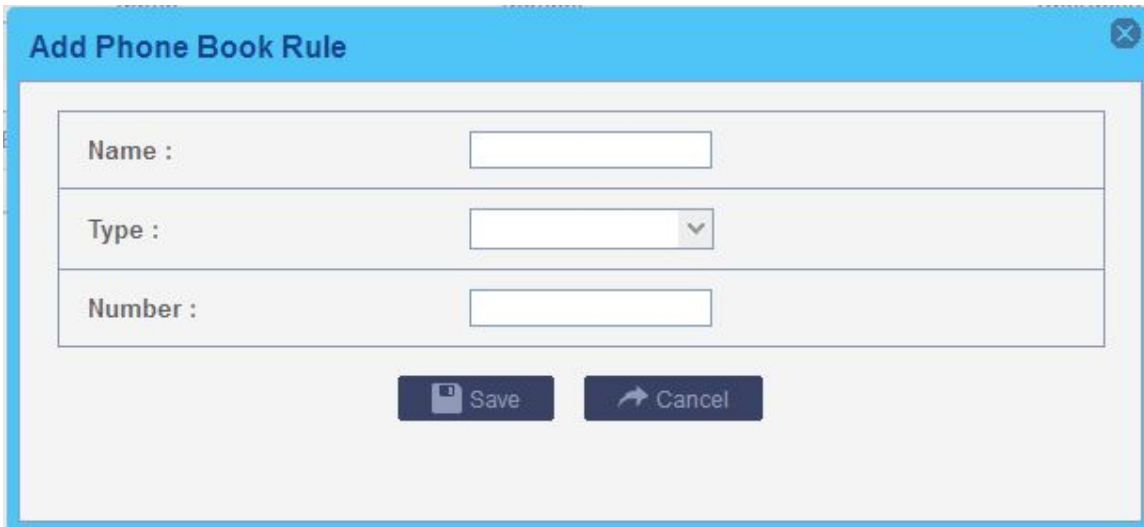
You can add your contacts here and when you use phone provisioning, IP phone will download the phone book.

Add

Name: Input the name of this contact.

Type: These are two types: Directory and Blacklist.

Number: Input the number here.



The screenshot shows a dialog box titled "Add Phone Book Rule". It contains three input fields: "Name" (text), "Type" (dropdown), and "Number" (text). Below the fields are "Save" and "Cancel" buttons.

Upload phonebook

You can upload a phonebook before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones. The format of phonebook should be *.xml.

NOTE: All the existing phonebooks of the IP phone will be replaced automatically if the phonebooks are configured in this way.

2) Ring Tone

You can upload a Ring Tone before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones.

NOTE: Ring tone must be wav file, 8k sampling rate, 8 Bit u-law compression. File size should < 200Kbytes.

3) Firmware

You can upload a firmware before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones.

NOTE: the file format must be .fw

10.2.2 Configured Phones

Create New Phone

Enable

Enable auto provisioning for this phone

Phone Type

Choose the Phone Type

MAC Address

Input the MAC address for the phone

New Version

Make sure your phone is with the newest version. If yes, IPPBX will generate the configuration file for IP phone.

Manufacturer

Choose the manufacturer.

DND

Enable or Disable DND.

Call Waiting

Enable or Disable Call Waiting.

Auto Answer

Enable or Disable Auto Answer.

Phone book

Enable or Disable Phone book update .If this option is checked, you must add or upload phone book first.

New Configured Phone ✕

General

Enable :	<input type="text" value="Yes"/>	Phone Type :	<input type="text" value="AT820"/>
MAC Address :	<input type="text" value="80:82:87:01:04:D2"/>	New Version :	<input type="text" value="Yes"/>
Manufacturer :	<input type="text" value="ATCOM"/>	DND :	<input type="text" value="Disabled"/>
Call Waiting :	<input type="text" value="Enabled"/>	Auto Answer :	<input type="text" value="Disabled"/>
Phone Book :	<input type="text" value="Enabled"/>	Ring Tone :	<input type="text" value="Enabled"/>
Firmware :	<input type="text" value="Disabled"/>	Firmware Version :	<input type="text"/>

Line

	line	Extension :	DisplayName :	Line Active :
<input type="checkbox"/>	line1	<input type="text"/>	<input type="text"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/>	line2	<input type="text" value="6007"/>	<input type="text" value="Robin"/>	<input checked="" type="checkbox"/>

Ring Tone

Enable or Disable Ring Tone. If this option is checked, you must upload phone book first.

Firmware

Enable or Disable Firmware update. If this option is checked, you must upload firmware first.

Firmware version

If Firmware option is enable, you must input the correct firmware version here.

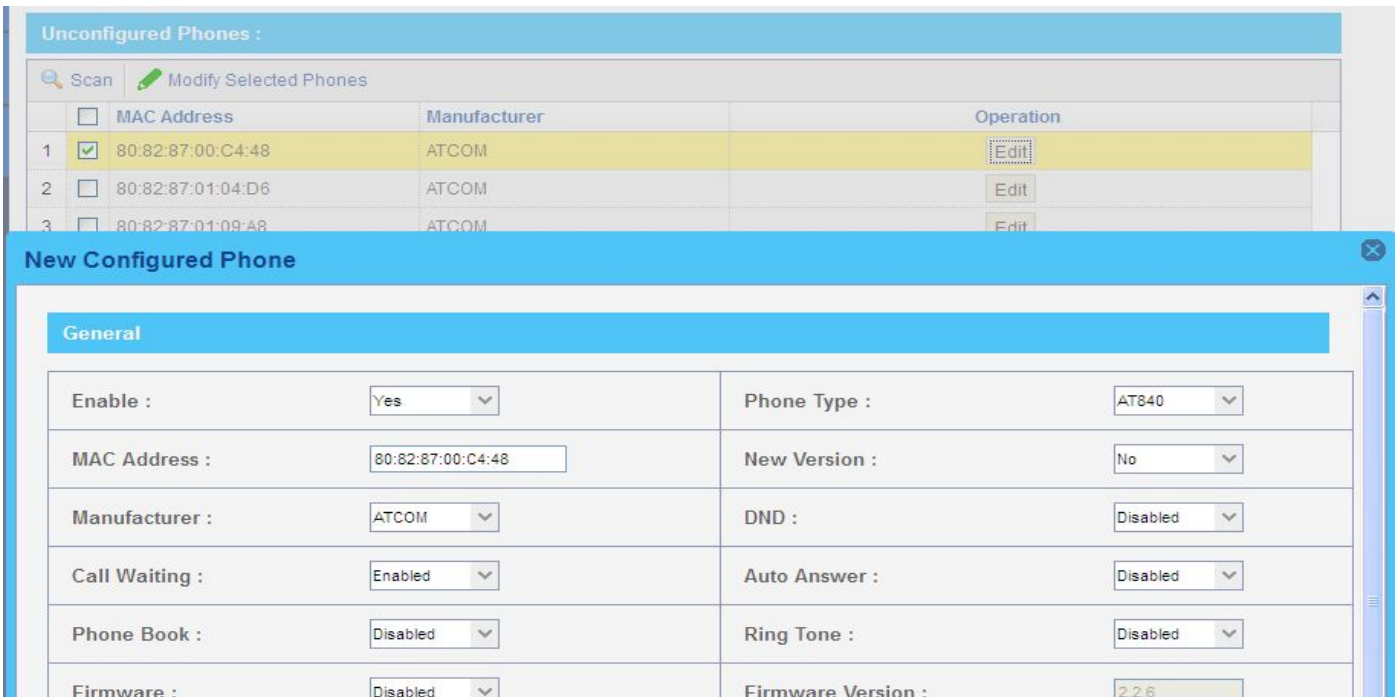
NOTE: If the firmware version is different from the one you're going to load in IP phone, it will cause problems in IP phone.

Line

You can configure the line key settings for this IP phone.

10.2.3 Unconfigured Phones

In this section, IPPBX will scan all the supported IP phones and display here, you can click the 'Edit' button of IP phone and input the corresponding information in the pop-up window.



10.2.4 Upload Phone Configure

Click "Upload a file" and choose the configure file of IP phone in the popup window.

NOTE: the file format must be .xml.

10.3 Date & Time

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

General	
GMT TimeZone :	<input type="text" value="+8 China(Beijing)"/>
<input checked="" type="radio"/> Automatically Synchronize with Internet Time Server :	
NTP Server :	<input type="text" value="pool.ntp.org"/>
<input type="radio"/> Set Date&Time Manually :	<input type="text"/>
Daylight Saving Time	
Daylight Saving Time :	<input type="text" value="Disabled"/>
Daylight Saving Time Rule :	<input type="text"/>
<input type="button" value="Save"/>	<input type="button" value="Cancel"/>

1) General

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

a. NTP server

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

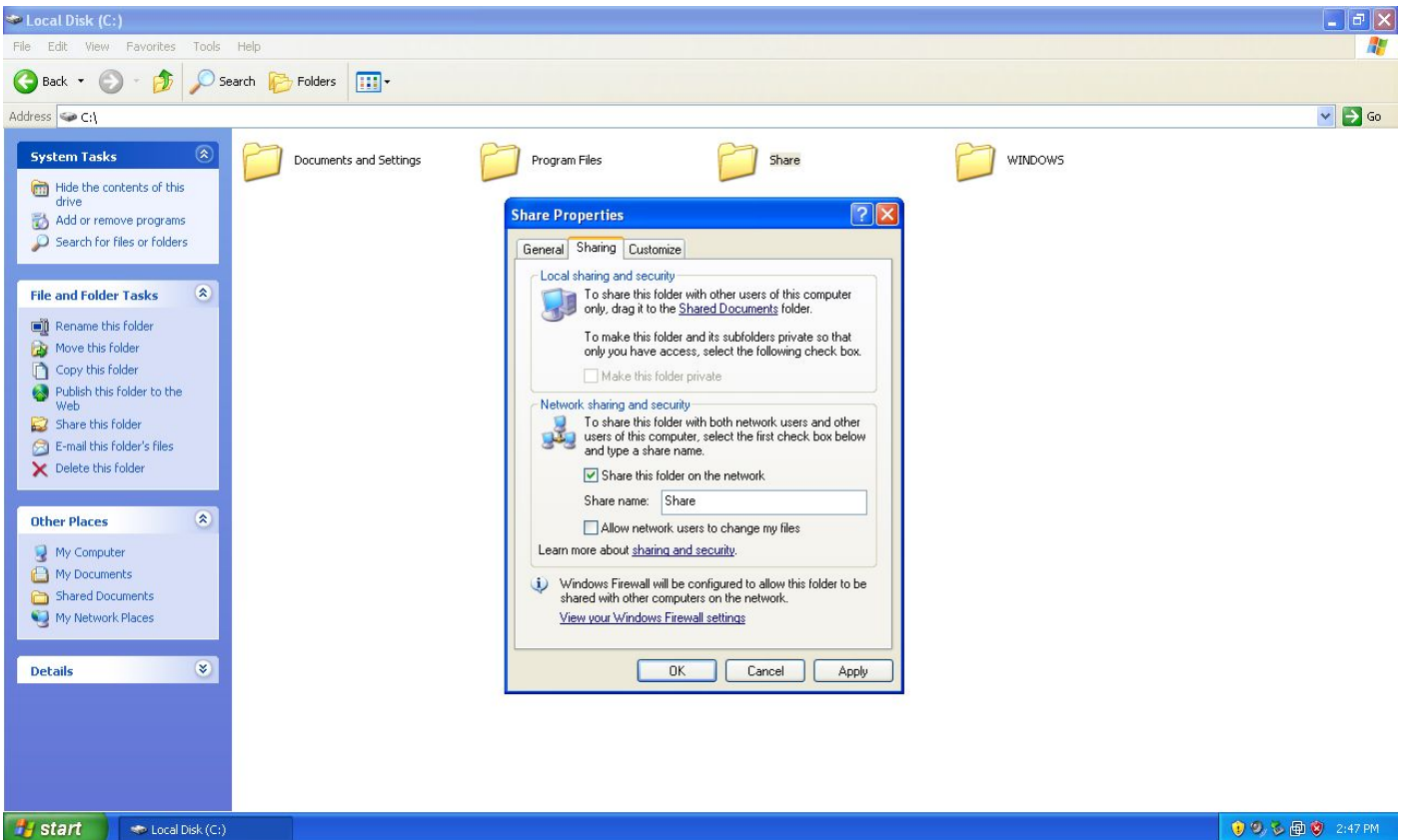
start=4/1/7/0:0:0;end=10/31/7/0:0:0;save=1 means IPPBX time from the first Sunday coming after April 1th to the first Sunday coming after October 31th will be one hour early.

10.4 External Storage

The External Storage feature is used to extend storage space. Once configured, the files (voicemail/Call recording (>60M) / CDR(>5M)) created before the configured days will be moved to the Net-Disk.

NOTE: The shared folder must be based on Windows operation system.

1. Choose a window-based computer that is always in service
2. Create a folder
3. Share this folder



4. Set External Storage

Net-Disk Host/IP

IP of the PC

Net-Disk Share Name

The name of the share folder

Net-Disk Access User Name

Account in that PC

Net-Disk Access Password

Password for the account, if there is no password required, leave it blank.

Backup Period

How often PBX move its voicemail/call recording/CDR data to PC

Move Files Created Before days ago

Choose what files should be move to PC according to their creation time. 0 means immediately. Note that even 0 is set, the size of call recording must be more than 60 M, that of CDR must be 10M. Then IPPBX will move them to PC.

Change Password	Net Disk Settings
Auto Provisioning	Net-Disk HostIp : <input type="text" value="192.168.1.156"/>
Date && Time	Net-Disk Share Name : <input type="text" value="Share"/>
> External Storage	Net-Disk Access User Name : <input type="text" value="user"/>
Firmware Upgrade	Net-Disk Access Password : <input type="password" value="*****"/>
	Backup Period : <input type="text" value="30"/> minutes
	Move Files Created Before : <input type="text" value="30"/> days ago
	<input type="button" value="Save"/>

10.5 Firmware Upgrade

Firmware for IPPBX There are two ways to upload Kernel / Application for IPPBX:

1) HTTP

Upload them from local PC. It will reset IPPBX to factory default settings by default.

2) TFTP

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

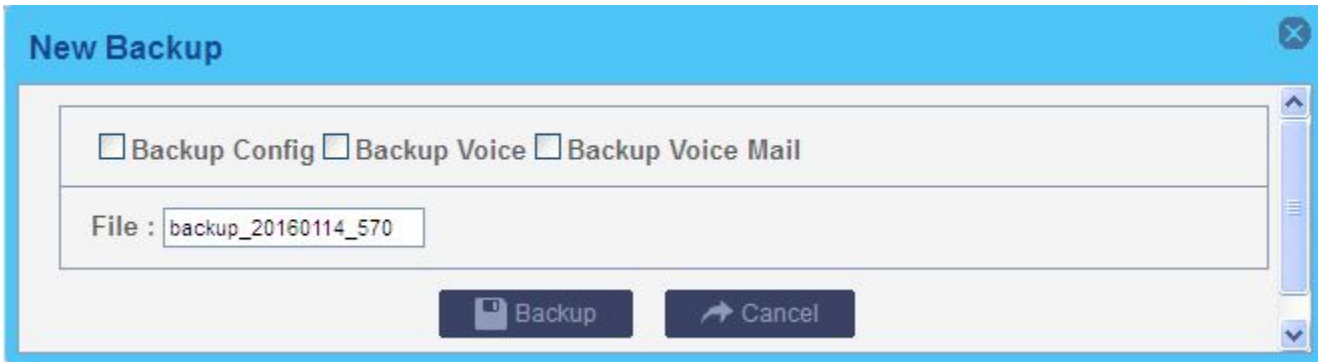
Firmware Upgrade			
Note: If you select the http mode when upgrading kernel, the config will be erased automatically!			
Image Type :	<input checked="" type="radio"/> Application	<input type="radio"/> Kernel	
Upgrade Method :	<input type="radio"/> HTTP	<input checked="" type="radio"/> TFTP	Reset Config : <input type="checkbox"/>
Server :	<input type="text"/>	File :	<input type="text"/>
<input type="button" value="Upgrade"/>			

Choose Reset Config will reset the configuration.

10.6 Backup and Restore

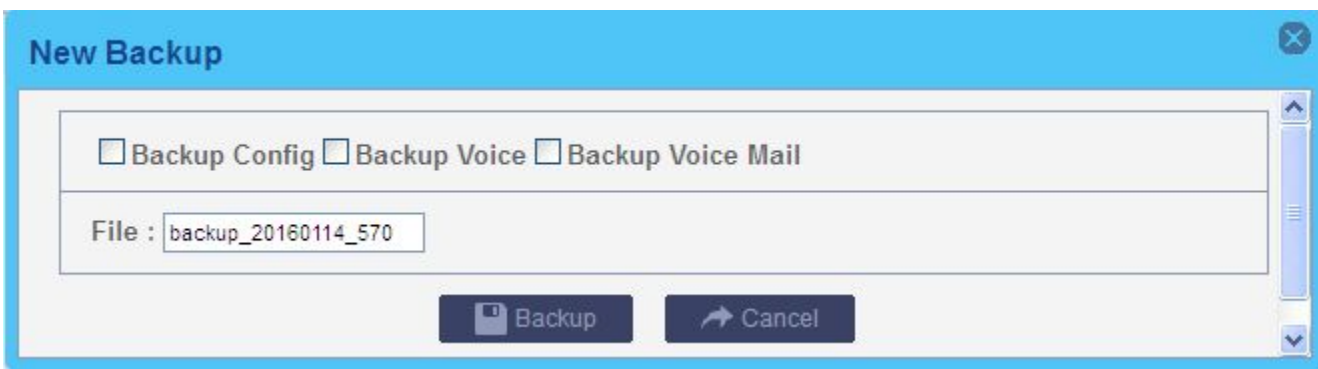
1) Backup

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 IPPBX reboot. It can't be used for different product models.



10.7 Reboot && Reset

Reboot or Reset IPPBX.

Reboot System

Warning : Rebooting system will terminate all active calls!

Reboot

Reset to Factory Default

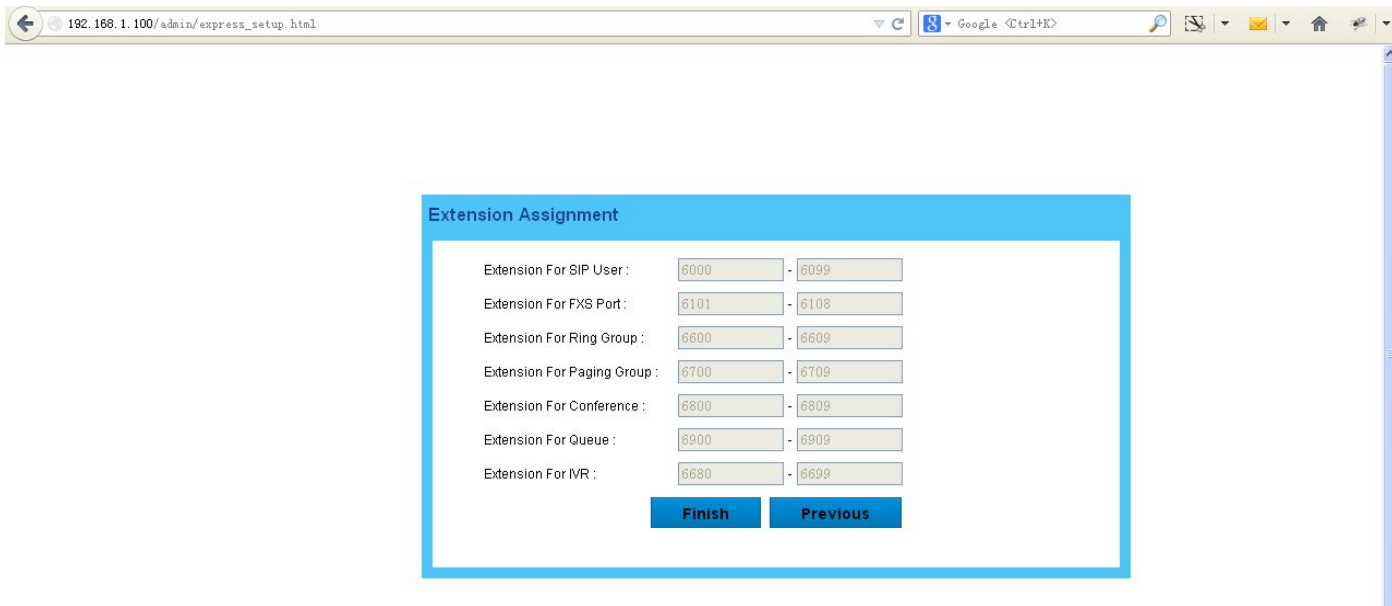
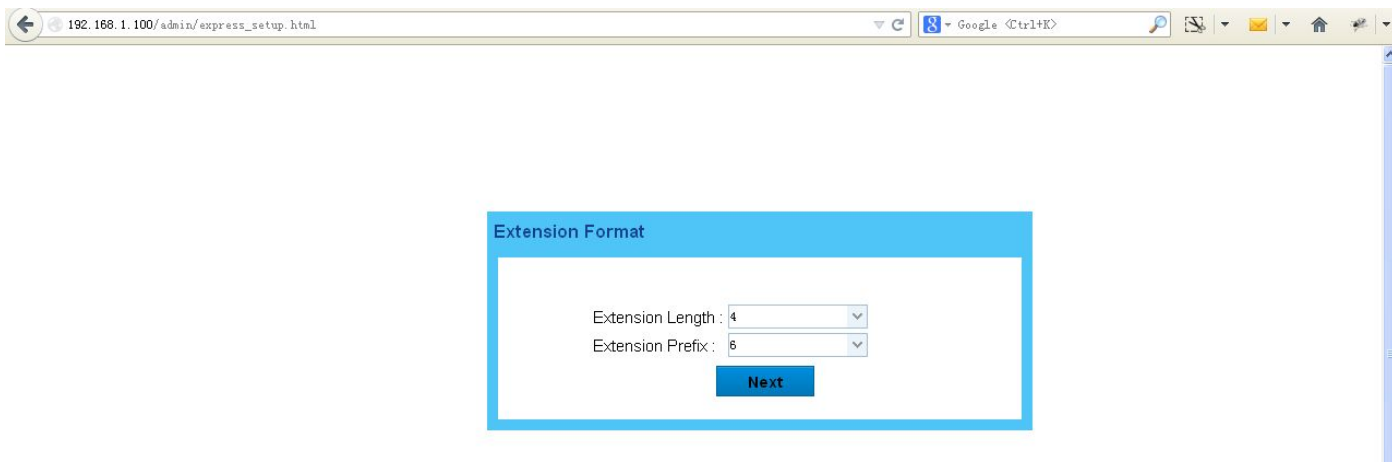
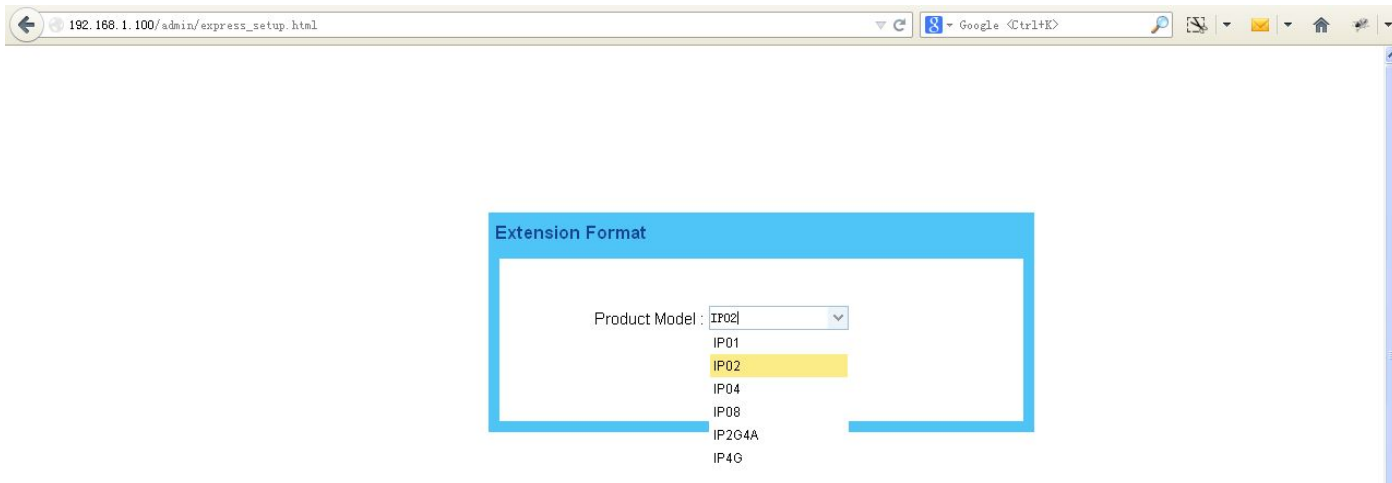
Warning : A factory reset will erase all configuration data on the system.

Please do not turn off the system until the RUN light begins blinking.

Any power interruption during this time could cause damage to the system!

Reset to Factory Default

You need to choose the product model and extension length/prefix the first time to login IPPBX after it is reset to factory default settings.



11. Reports

11.1 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

IPPBX supports two delete operations: delete selected CDR and delete all CDR.

3) Download

It can be downloaded to local PC

	Source	Destination	Start Time	End Time	Duration	Billable Duration	Disposition
11		s	2013-08-21 08:18:41	2013-08-21 08:18:47	6	0	ANSWERED
12	02196802000	s	2013-08-21 08:18:36	2013-08-21 08:18:38	2	0	ANSWERED
13		s	2013-08-20 19:25:15	2013-08-20 19:25:21	6	0	ANSWERED
14		s	2013-08-20 19:25:06	2013-08-20 19:25:12	6	0	ANSWERED
15		s	2013-08-20 19:24:56	2013-08-20 19:25:02	6	0	ANSWERED
16		s	2013-08-20 19:24:51	2013-08-20 19:24:52	1	0	ANSWERED
17	900	6003	2013-08-20 18:48:14	2013-08-20 19:00:25	731	729	ANSWERED
18	900	6003	2013-08-20 18:48:11	2013-08-20 18:48:12	1	1	ANSWERED
19	900	6003	2013-08-20 18:48:03	2013-08-20 18:48:09	6	6	ANSWERED
20	900	6003	2013-08-20 18:47:10	2013-08-20 18:47:17	7	5	ANSWERED

11.2 Syslog

1) Syslog

Set the Syslog level and download it.

2) Packet Capture Tool

This feature always used to capture packets for technician. Users could specify the destination IP address and port to get the packets.

IP

Specify the destination IP address to get the packets.

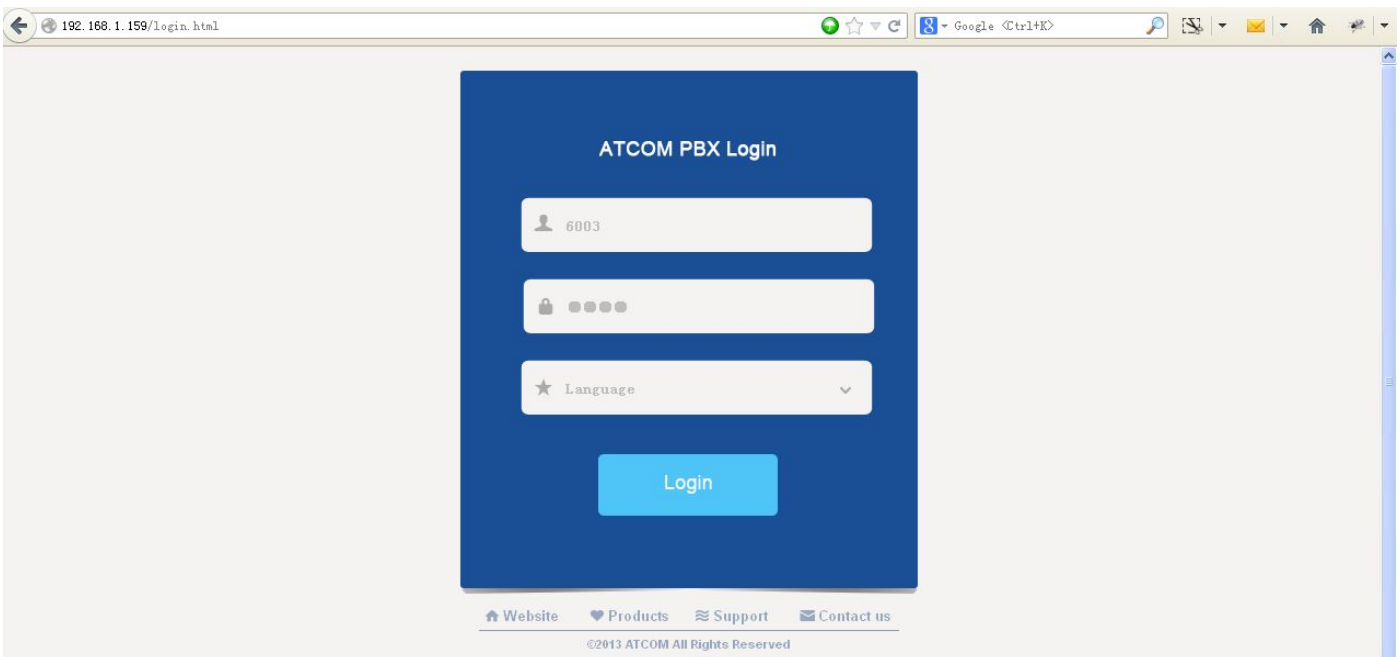
Port

Specify the destination port to get the packets.

Packet Capture Tool	
IP :	<input type="text"/>
Port :	<input type="text"/>
<input type="button" value="Start"/> <input type="button" value="Stop"/> <input type="button" value="Download"/>	

12. Web Interface for extension

PBX allows users to check their voicemail /Call Recording/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) Call Recording Checking

3) CDR Checking

Users can check their CDR here.

4) Personal Settings

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

--Finish--