



IPPBX FAQ

For Firmware Version: V2.0/V3.0

2014-05-28

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1. IPPBX Access

1.1 How to access IPPBX if I forget the IP of WAN?

- 1) If your IPPBX model is IP02/IP08/IP2G4A/IP4G, you can try to login IPPBX via LAN, then check the IP of WAN.

The default IP of LAN is **192.168.10.1/255.255.255.0**, and WAN is **192.168.1.100/255.255.255.0**.

- 2) You can login IPPBX via WAN with its failover IP

The failover IP of WAN is **172.31.0.254/255.255.0.0**, please set the IP of your PC to 172.31.xx.xx/255.255.0.0, then connect IPPBX via WAN.

1.2 How to do hardware reset?

If your IPPBX comes with firmware V1.4.0 or higher version, then it supports hardware reset.

There is a little bottom besides power slot in IPPBX, long press it until the SYS LED light, then let it go. The box will be reseted to factory default settings.

2. Upgrade

2.1 How to upgrade IPPBX from V1.4.0/V2.0 to latest firmware V3.0?

Please use **Firefox** and choose **TFTP** way.

- 1) Download Kernel and Application for IPPBX

Kernel for IP01: **ulmage_IP01.crc**

http://www.atcom.cn/cn/download/pbx/ip01/uImage_IP01.crc

Kernel for IP02/IP04/IP08: **ulmage.crc**

<http://www.atcom.cn/cn/download/pbx/ip02/uImage.crc>

Kernel for IP2G4A/IP4G: **ulmage_IP2G4A.crc**

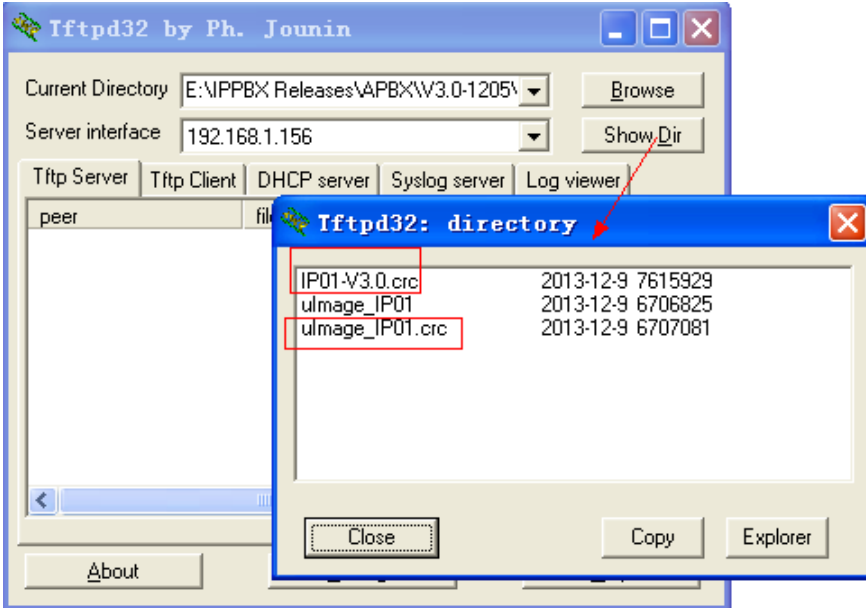
http://www.atcom.cn/cn/download/pbx/ip2g4a/uImage_IP2G4A.crc

IP01/IP02 and IP04/IP08/IP2G4A/IP4G use different applications:

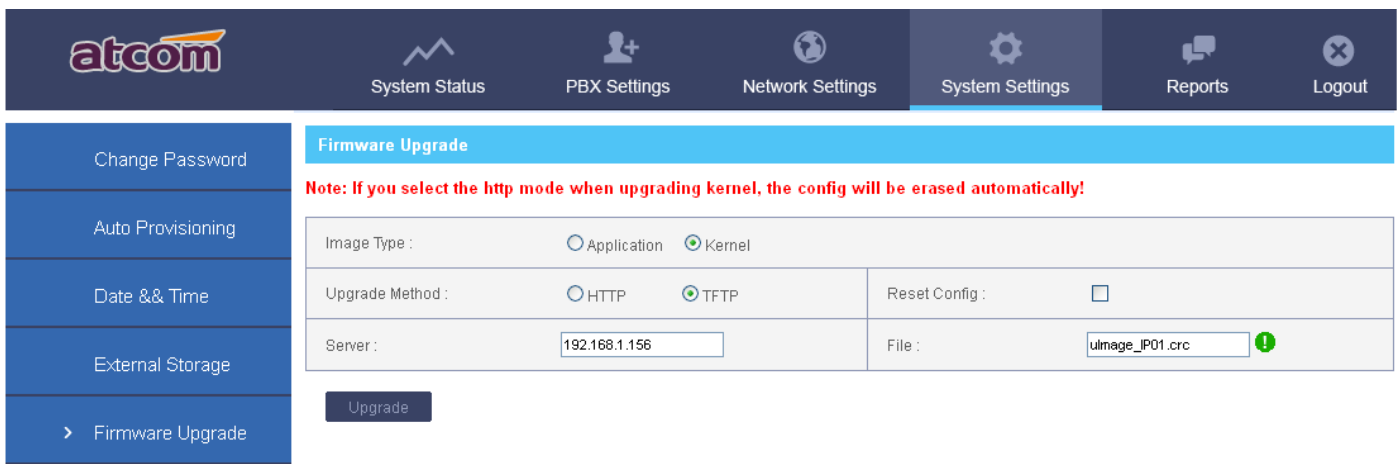
IP01/IP02: <http://www.atcom.cn/cn/download/pbx/ip01/IP01-V3.0.crc>

IP04/IP08/IP2G4A/IP4G: <http://www.atcom.cn/cn/download/pbx/ip04/IP04-V3.0.crc>

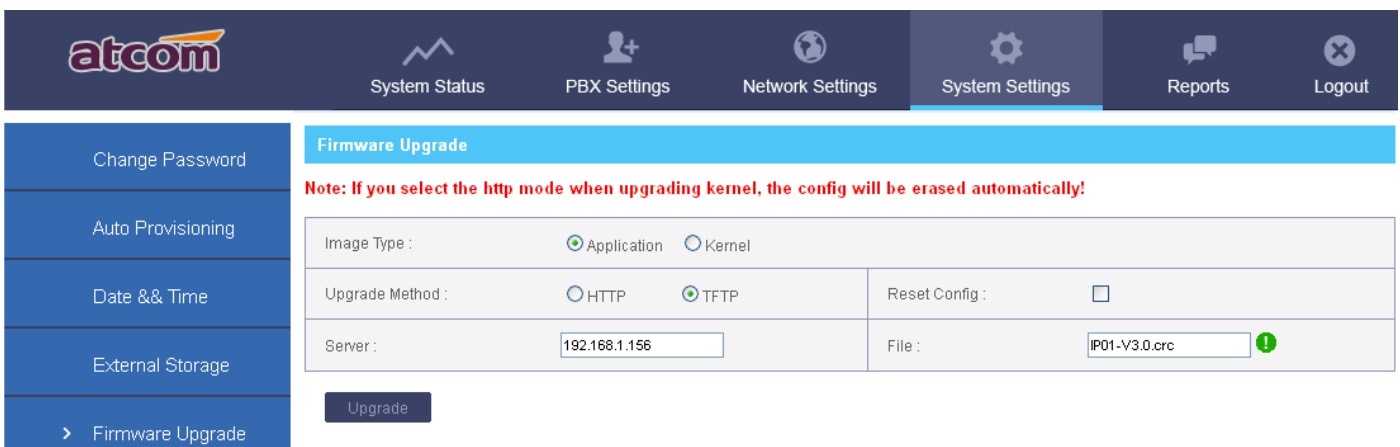
- 2) Set TFTP server: Choose firmware uploading directory as tftp server base directory.



3) Upgrade Kernel without Reset Config chose



4) Upgrade Application without Reset Config chose



5) Check firmware version.

atcom		System Status	PBX Settings	Network Settings	System Settings	Reports	Logout
General	General						
Trunk Status	Product Model :	IP01	Application Version :	V3.0			
Extension Status	Kernel Version :	V3.0	System Up Time :	0 days 0 hours 0 minutes 24 seconds			
	System Current Time :	Tue Dec 10 19:12:06 2013					
	Network			WAN Primary DNS :	8.8.8.8		
	WAN Connection Type :	STATIC	WAN Secondary DNS :	8.8.4.4			
	WAN Mac Address :	80:82:87:00:D9:CB	WAN Gateway :	192.168.1.254			
	WAN IP Address :	192.168.1.159					
	WAN Subnet Mask :	255.255.255.0					
	Peripheral						
	Port 1 :	unplugged					

NOTE: New firmware has problem in compatible with old firmware at the outbound route part, please delete outbound routes and recreate them.

3. Call Management

3.1 When a user is in two or three different Outbound Route groups, in what order the rules are applied?

There shouldn't be order since all the outbound routes shouldn't be conflict. Pattern in outbound route is like number filter, when you make a call, there should be one exact outbound route matches what you dial.

For example:

Outbound route 1:

Pattern: **00.**

Outbound route 2:

Pattern: **0Z.**

* Z means any digit in 1-9, . (dot) means any digits or characters (*,#)

When you dial 001234567, then PBX will forward the call via trunk selected in outbound 1, and that in outbound 2 if 011234567 is dialed. So you need to distinguish the outbound routes manually in order that PBX can distinguish them.

If you set pattern to **0X.** in outbound 2 (X means any digit in 0-9), when 001234567 is dialed, the call should be forwarded out also through trunks in outbound route 1, since 00. matches the dialed number precisely, if there are two outbound routes with more ambiguous patterns, PBX may choose one randomly.

3.2 How to resolve one-way audio issue?

If your IPPBX is behind router, there is always one-way audio issue.

1) Please port forward below ports on your router first:

SIP: 5060 (UDP), if you use others port/transport for SIP, port forward that port.

RTP:10000~20000 (UDP)

2) Then choose either below NAT solution.

a. Set STUN, just setting STUN server / port is OK.

There are many public STUN servers 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=""/>
Local Network Identification :	<input type="text"/>	Allow RTP Reinvite :	<input type="text" value="no"/>

b. Set 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="192.168.1.0/255.225.255.0"/>	Allow RTP Reinvite :	<input type="text" value="nonat"/>

External IP address

If you have fixed public IP for your router, fill it in here.

External Host

Otherwise, you need to apply for a DDNS, and fill it in here.

External Refresh Interval

How often to refresh External Host if used.

NAT Mode

Yes

Local Network identification

Your local network, format: sub-network/netmask.

Allow RTP Reinvite

nonat.

3.2 How to set IVR?

- 1) Record Custom Prompts
- 2) Set IVR like below

3.3 How to record your custom unavailable/busy/temp greetings for voicemail?

Please dial into your mailbox and operate according to the IVR prompts, the call flow should be:

- Dial *2(default setting) ->0 Mailbox options ->1 Record your unavailable message
 ->2 Record your busy message
 ->3 Record your name
 ->4 Manage your temporary recording

3.4 How to make video calls?

- 1) Make sure the Video Support on PBX Settings -> SIP Settings is set to yes

- 2) Select video code for the extension

Advance Configuration

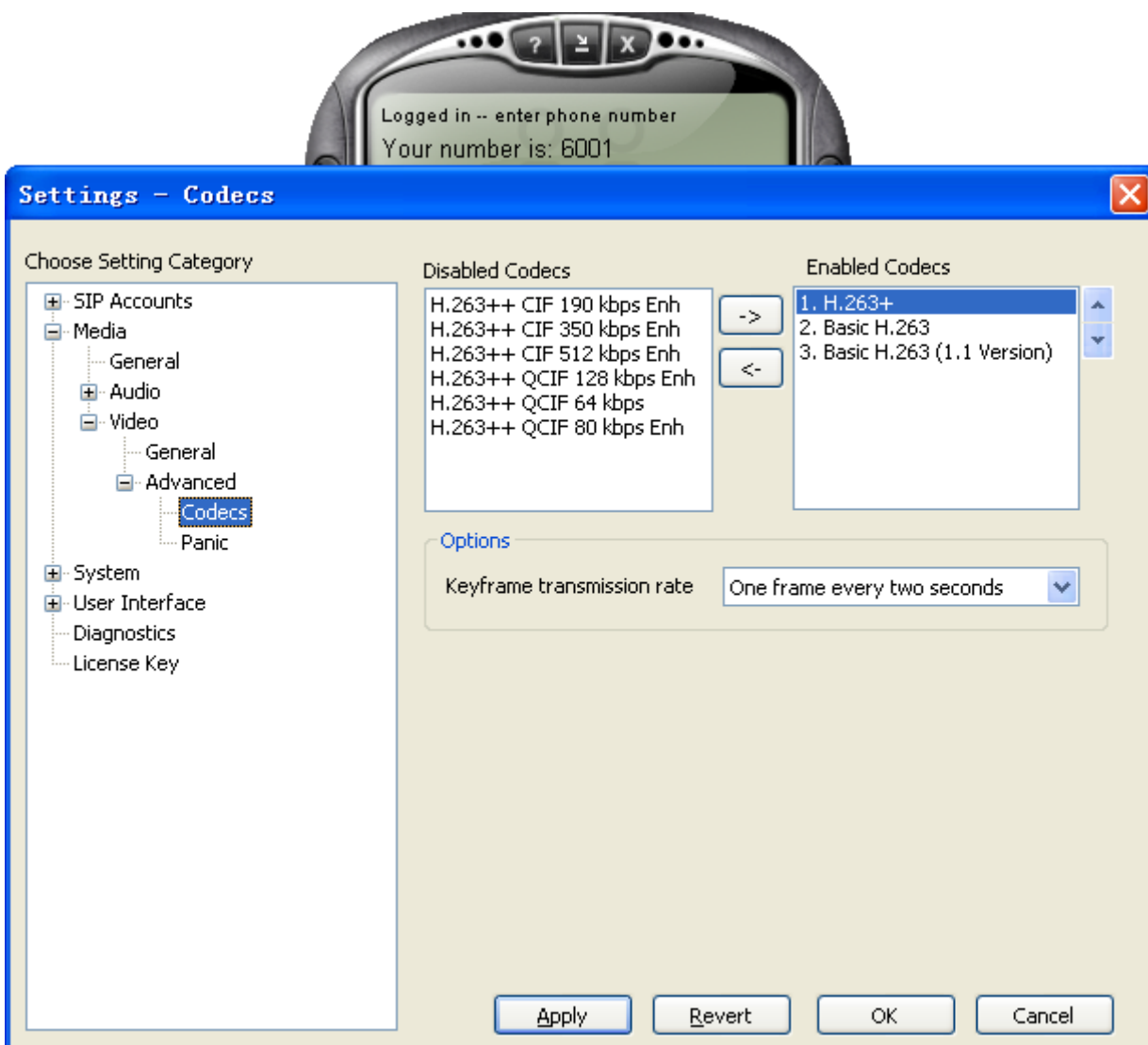
SIP Settings

NAT : Can Reinvite : DTMF Mode : rfc2833

Preferred Codec :

First : a-law	Second : u-law	Third : H263p
Fourth : H263	Fifth : None	Sixth : None

3) Select the same video code on the IP phone



3.5 What analog module is used for?

There are 3 kinds of modules:

FXO: is used to connect PSTN

FXS: is used to connect Analog phone

GSM: is used to connect GSM network, a special kind of FXO, can be used in IP2G4A/IP4G

LED for corresponding Ports:

If FXO module is detected: light red

If FXS module is detected: light yellow

If GSM module is detected: light red

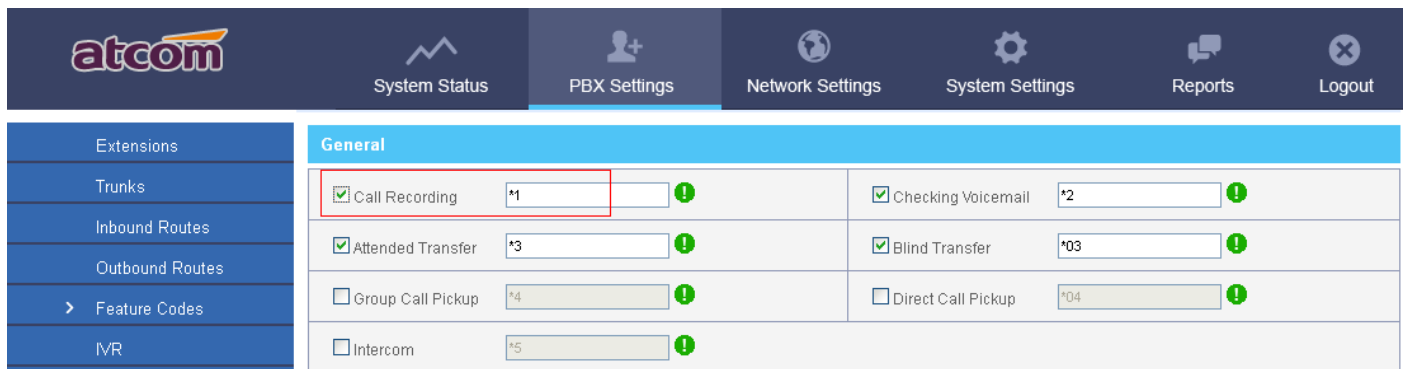
Make sure the modules are installed and detected before you use them, detailed WEB configuration can be found in user manual of each product.

3.6 How to check call recordings?

1. There are two ways to record calls.

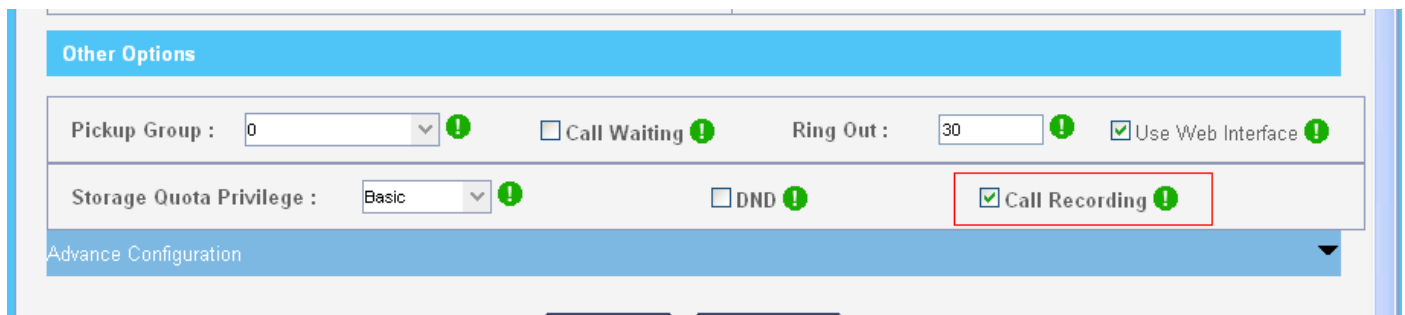
1) One touch record

You need to enable Call Recording in PBX Settings -> Feature Codes first and then dial Call Recording code to begin recording after a beep sound while in a call, and dial the same code again to stop recording.

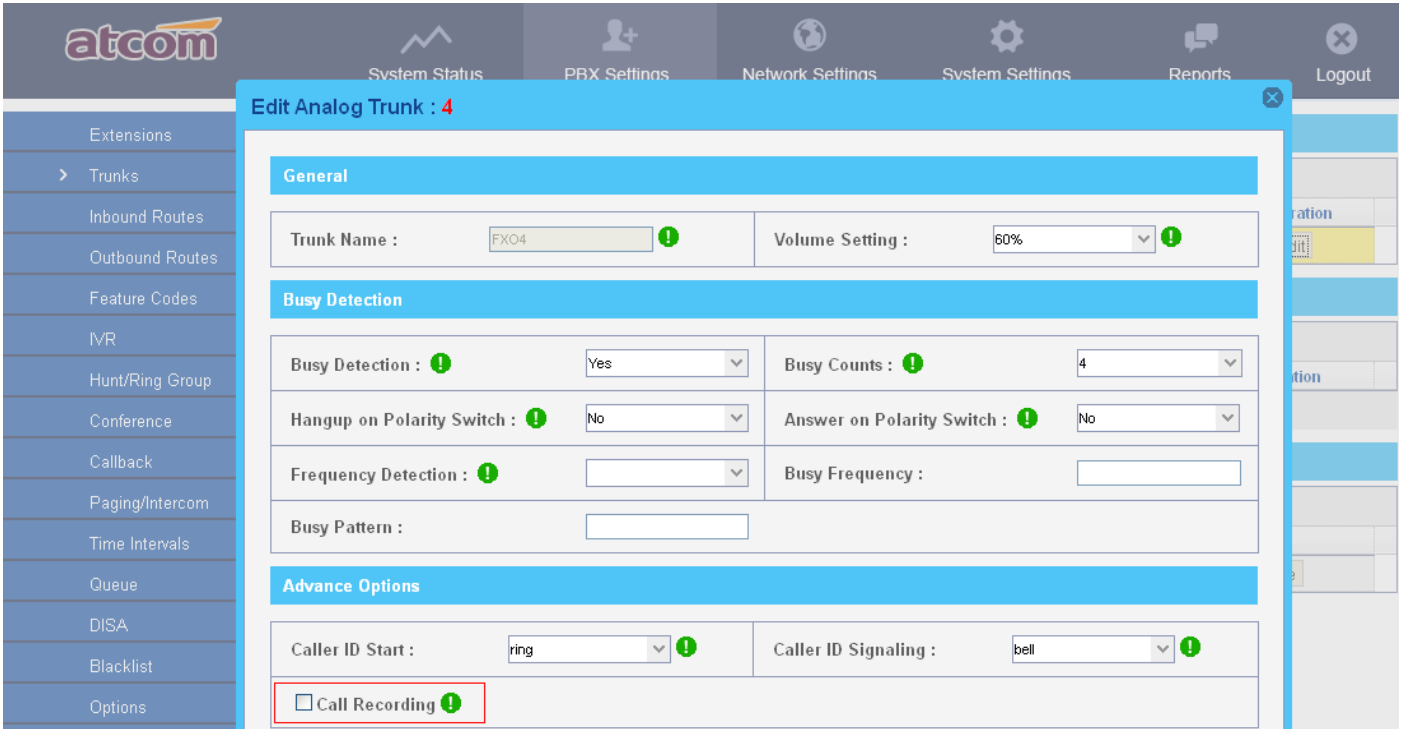


2) Automatic Call Record

Enable call recording in PBX Settings -> Extensions for an extension, all the calls made by this extension will be recorded automatically.

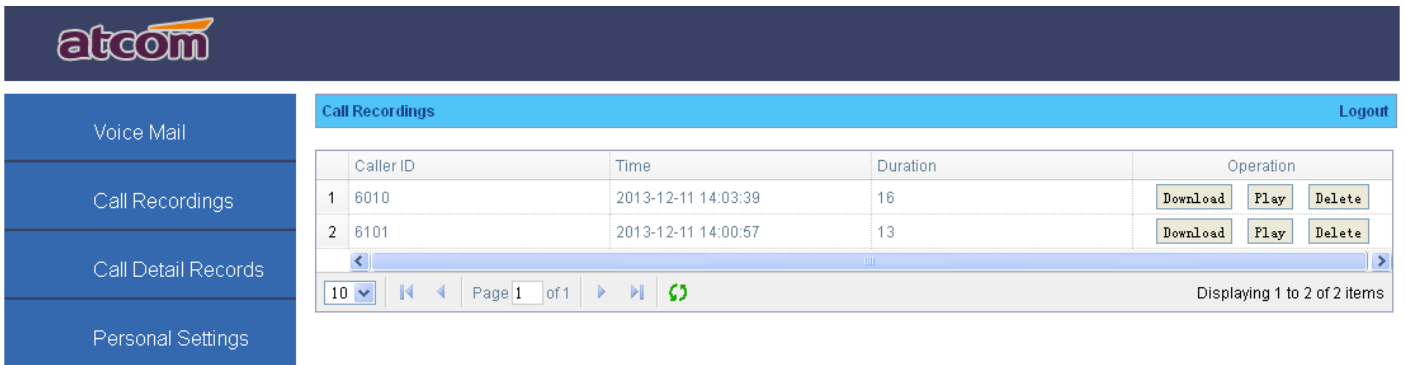


Enable call recording in PBX settings -> Trunks for a trunk, all incoming & outgoing calls through this trunk will be recorded automatically.

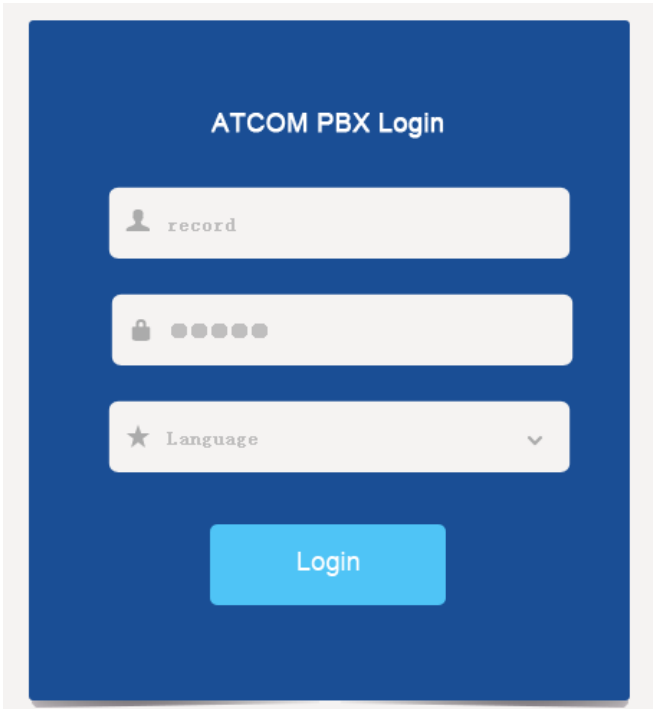


2. There are two ways to check call recordings.

1) Extension user can login its personal web portal to check his call recordings



2) **record** user can login IPPBX with username/password: `record/the_password_of_admin` to check all recordings.



3.7 How to set voicemail to email?

1) The voicemail2email needs a real email account as the sender, please configure it in **PBX Settings -->SMTP Settings** on the Web, make sure the SMTP server and Port is right. Some SMTP servers may require SSL/TLS support, I.E. Gmail.

SMTP Settings			
Email Address :	fax@atcom.com.cn	Password :
SMTP Server :	smtp.ym.163.com	Port :	25
Min Messages Time :	2	Max Messages Time :	120
Max Messages :	100	Say CID :	No
Say Duration :	No	Envelope :	No
Review :	Yes	Delete Message After Notification:	No
<input type="checkbox"/> Ask Caller to Dial 5		<input type="checkbox"/> Use SSL/TLS to send secure message to server	

2) Enable the voicemail for user and specify the Email Address as receiver

Edit Voip Extension : 8006

Type : SIP/AAX | Name : PengDi | Caller ID : 8006

Extension : 8006 | Password : 8006

Sip Settings

NAT : | Can Reinvite : | DTMF Mode : rfc2833

Transport : UDP | SIP Qualify

Voice Mail

Enable Voice Mail | Voice Mail Access PIN Code : 8006

Mail Setting

Enable Sending Voice Mail | Email Address : bty@atcomemail.com

3.8 How to set FAX to email?

1. Please set SMTP settings and Email Address first as previous chapter do.

2. Inbound Route Settings.

1) Forward to FAX always.

Set Destination Type to Faxes, Destination to an Extension Number.

If the extension is a VoIP extension, the FAX will be send to the Email Address configured for that extension.

If it's a FXS extension, PBX will dial the FXS extension directly, usually, the FXS port will be connected with an FAX machine.

Path

Destination Type : Faxes | Destination : 6000(SIP/AAX)

Fax Detection

Destination : No Detect

2) Fax Detection.

Path

Destination Type : Ring Group | Destination : 6600

Fax Detection

Destination : Custom Email | bty@atcomemail.com

When Destination Type is other functions, IVR, Ringgroup, etc, you can enable Fax Detection to detect the FAX, If a incoming call is a FAX call, PBX will send it to custom email box or email box according to the extensions or FAX machine directly.

4. How to secure your IPPBX?

4.1 Put IPPBX behind your firewall

5.1.1 Open SIP port & RTP port only to the outside world

It is dangerous to open accessible port to the outside world, for example SSH (TCP :22), WEB (TCP: 80/443)

It is recommend to open SIP port (UDP: 5060) and RTP port (UDP: 10000-20000) only for SIP communication with the outside world.

5.1.2 Change SIP port

Since it's impossible to access your IPPBX, hacker may try to register your IPPBX by guessing your extension number and password. Due to 5060 is a well known ports, it's more possible to be attacked, you can change it to 5061,5062, 5500,80 and so on.

Please go to PBX Settings -> SIP Settings option on the Web to change it.

General	
UDP Port :	5066
<input checked="" type="checkbox"/> Enable TCP	TCP Port : 5066
Registration/Subscription Time Max :	3600
RTP Port Min :	10000
RTP Port Max :	20000
DTMF Mode :	rfc2833
Registration/Subscription Time Min :	60
Video Support :	yes
NAT	
Enable STUN :	<input type="checkbox"/>
STUN Server :	stun.jpns.com
STUN Port :	3478
External IP Address :	
External Host :	tpcard.vicp.cc
External Refresh Interval :	10
NAT Mode :	yes
Local Network Identification :	192.168.1.0/255.255.255.
Allow RTP Reinvite :	nonat

4.2 Set strong password for Extension

It is possible for intruders to send to your system over 40 authentication requests per second by using a rolling number generator and a database of common words. A strong password should be a mixture of upper and lower characters and numbers.

4.3 Set 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.

Go to PBX settings -> Extensions -> Edit an extension -> Advanced Configuration

Set trusted IP: xx.xx.xx.xx/255.255.255.255, for example: 192.168.1.156/255.255.255.255

Set trusted network: xx.xx.xx.xx/subnet mask, for example: 192.168.1.156/255.255.255.0

4.4 Set password for Trunk

The hacker usually use your analog/VoIP trunk to make expensive international calls after register to your IPPBX successfully. It's an effective way to secure your account balance to set password for outbound routes.

4.5 Enable Firewall on IPPBX

Common Rule: Accept/Drop the connections from remote hosts.

Auto Defense: Limit connections from remote hosts.

SIP Defense: Limit connections to SIP port from remote hosts.

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

5. Others

5.1 Why IP phone do not register unless a cold reset after a reboot of the IPPBX?

Every SIP client has its own register expiration, normally, it's 3600s which means it will register to SIP server once an hour.

After PBX reboot, all the registry information is lost, PBX will think of that IP phone is unavailable until IP phone register again when the register expiration is reach.

You can set register expiration to a shorter time, like 60s. (register every minute, however this will occupy some network bandwidth)

ATCOM
ATCOM IP PHONE
Device Configuration

<ul style="list-style-type: none"> System Status Network Account <ul style="list-style-type: none"> <li style="background-color: #e0e0e0;">Account Phone Setting Update Phone Book Call Log 	<div style="background-color: #e0e0e0; padding: 2px; border: 1px solid #ccc;">Account / Account 1 admin</div> <p>SIP</p> <table style="width: 100%;"> <tr> <td>User ID :</td> <td><input type="text" value="6002"/></td> <td>Password :</td> <td><input type="password" value="●●●●"/></td> </tr> <tr> <td>SIP Server :</td> <td><input type="text" value="192.168.1.160"/></td> <td>SIP Port :</td> <td><input type="text" value="5070"/></td> </tr> <tr> <td>Use Outbound Proxy :</td> <td><input type="text" value="No"/></td> <td>Outbound Proxy Server :</td> <td><input type="text"/></td> </tr> <tr> <td>Outbound Proxy Port :</td> <td><input type="text"/></td> <td>Outbound Proxy Port :</td> <td><input type="text" value="5060"/></td> </tr> <tr> <td>Register Expires :</td> <td><input type="text" value="60"/></td> <td>Subscribe Expires :</td> <td><input type="text" value="3600"/></td> </tr> <tr> <td>Transport Type :</td> <td><input type="text" value="UDP"/></td> <td>SIP 100Rel Require :</td> <td><input type="text" value="No"/></td> </tr> </table>	User ID :	<input type="text" value="6002"/>	Password :	<input type="password" value="●●●●"/>	SIP Server :	<input type="text" value="192.168.1.160"/>	SIP Port :	<input type="text" value="5070"/>	Use Outbound Proxy :	<input type="text" value="No"/>	Outbound Proxy Server :	<input type="text"/>	Outbound Proxy Port :	<input type="text"/>	Outbound Proxy Port :	<input type="text" value="5060"/>	Register Expires :	<input type="text" value="60"/>	Subscribe Expires :	<input type="text" value="3600"/>	Transport Type :	<input type="text" value="UDP"/>	SIP 100Rel Require :	<input type="text" value="No"/>
User ID :	<input type="text" value="6002"/>	Password :	<input type="password" value="●●●●"/>																						
SIP Server :	<input type="text" value="192.168.1.160"/>	SIP Port :	<input type="text" value="5070"/>																						
Use Outbound Proxy :	<input type="text" value="No"/>	Outbound Proxy Server :	<input type="text"/>																						
Outbound Proxy Port :	<input type="text"/>	Outbound Proxy Port :	<input type="text" value="5060"/>																						
Register Expires :	<input type="text" value="60"/>	Subscribe Expires :	<input type="text" value="3600"/>																						
Transport Type :	<input type="text" value="UDP"/>	SIP 100Rel Require :	<input type="text" value="No"/>																						

5.2 How to remove the echo on analog trunk?

Decreasing the FXO gain and reboot the box can reduce the echo, generally, the high of the volume, the more of the echo.

System Status
PBX Settings
Network Settings
System Settings
Reports
Logout

- Extensions
- Trunks
- Inbound Routes
- Outbound Routes
- Feature Codes
- IVR
- Hunt/Ring Group
- Conference
- Callback
- Paging/Intercom

Edit Analog Trunk : 1

General

Trunk Name : <input type="text" value="FXO1"/>	Volume Setting : <input type="text" value="60%"/>
--	---

Busy Detection

Busy Detection : <input type="text" value="Yes"/>	Busy Counts : <input type="text" value="50%"/>
Hangup on Polarity Switch : <input type="text" value="No"/>	Answer on Polarity Switch : <input type="text" value="No"/>