IPPBX FAQ

For Firmware Version: V2.0/V3.0

2013-12-11

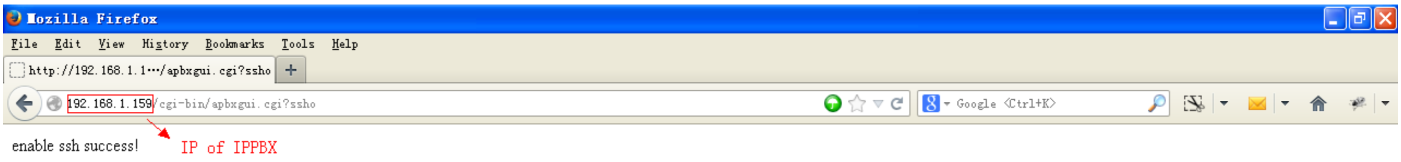
Contents

1. IPPBX Access

1.1 How to access IPPBX via SSH?

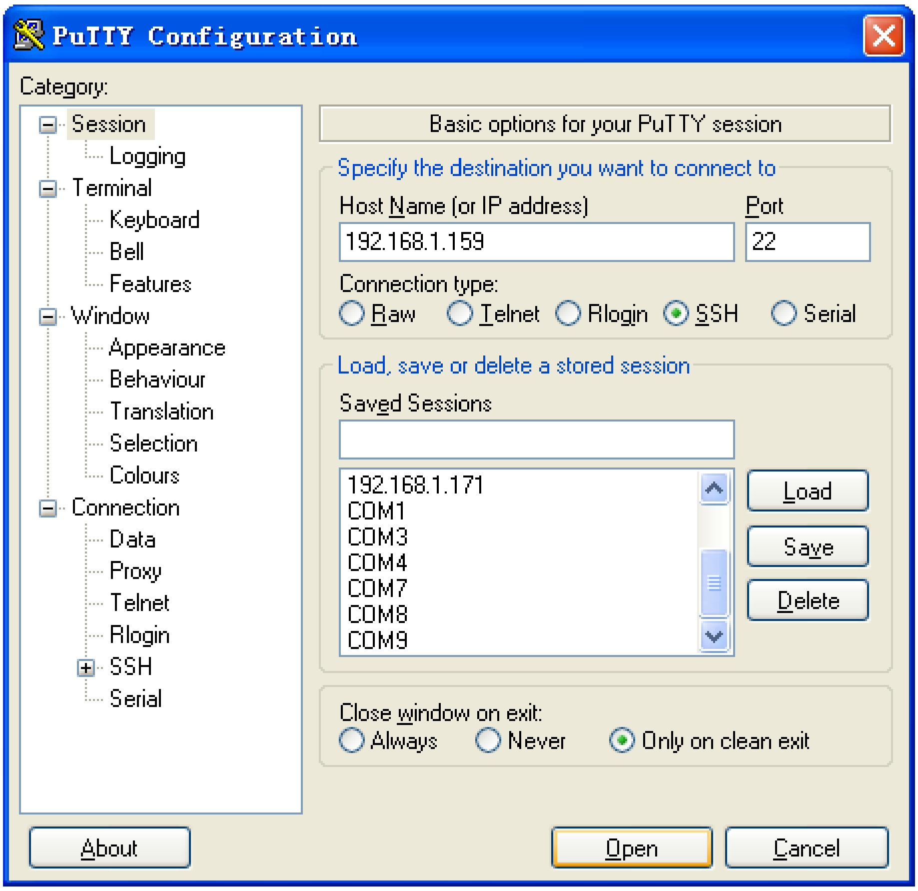
The SSH function is disabled by default, please enabled it through WEB GUI first.

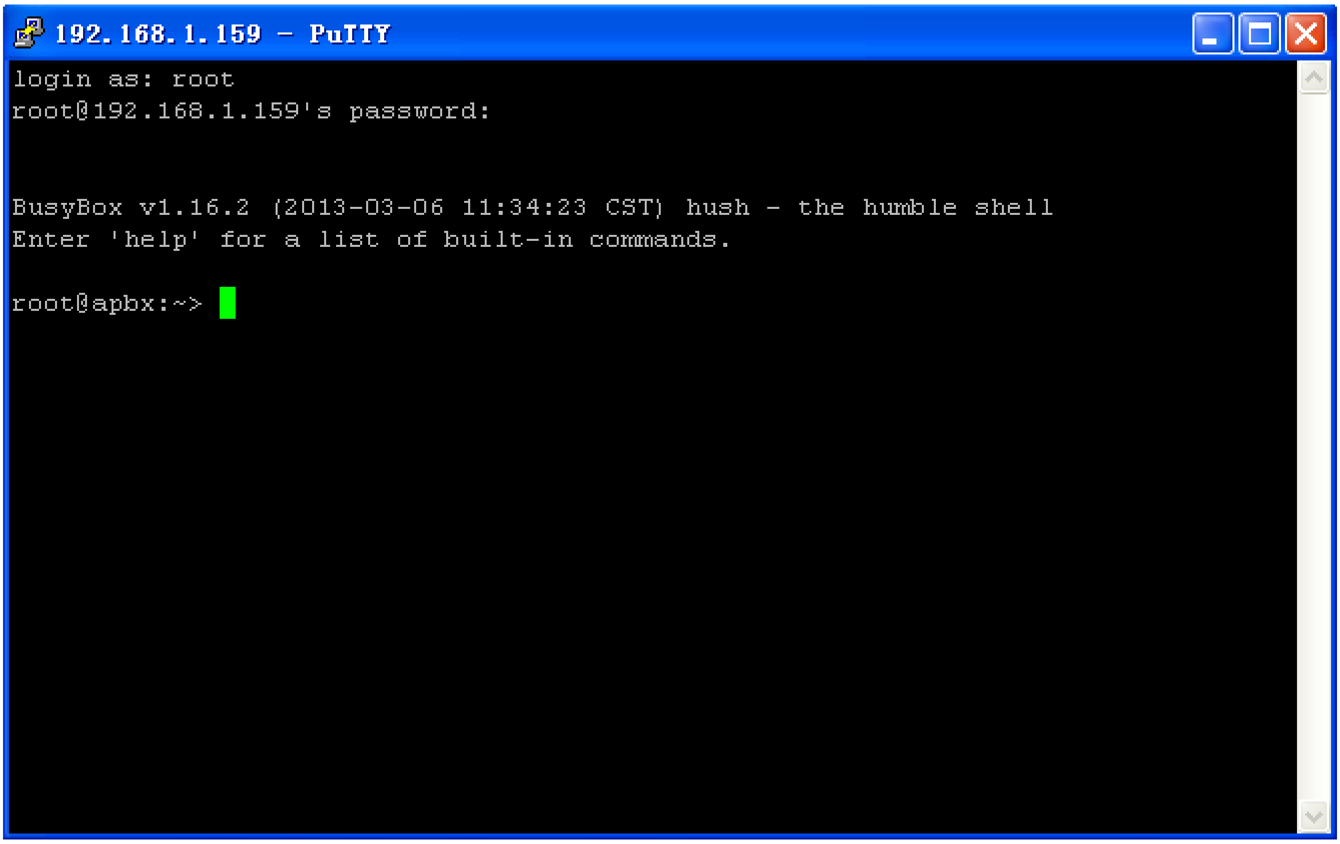
1) Enter xx.xx.xx.xx/cgi-bin/apbxgui.cgi?ssho in WEB Browser to enable it xx.xx.xx.xx/cgi-bin/apbxgui.cgi?sshf will disable it.



It needs to be re-enabled once PBX reboot.

2) Login IPPBX via SSH with username/password: root/atcombox





1.2 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.

1.3 How to retrieve WEB password via SSH?

1) Login PBX via SSH

2) Check the database, the password for WEB login is stored in database of IPPBX, several database operation statements will be used.

root@apbx:~> **cd /persistent/var/lib/** ; go to the directory of database
root@apbx:/persistent/var/lib> **sqlite3 apbx.db** ;enter the database
SQLite version 3.6.21
Enter ".help" for instructions
Enter SQL statements terminated with a ";"
sqlite> **.tables** ;show all the tables defined in database
sqlite> **select * from admin;** ;check the content of table admin, "**;**" is needed.
sqlite> **.quit** ;quit the database

```
root@apbx:~> cd /persistent/var/lib/
root@apbx:/persistent/var/lib> sqlite3 apbx.db
SQLite version 3.6.21
Enter ".help" for instructions
Enter SQL statements terminated with a ";"
sqlite> .tables
admin               exstorage           moh                 sipsettings
analogtrunk         extensions          mplog               slog
apbxman             feature             netoptions          speeddials
autobackup          firewallsettings    options             storagequota
autoprov            firmware            outbounds           trunks
autoprovcfg         fwauto              phonebook           version
bakres              fwcommon            phonebookcfg        vlansettings
blacklist           fwsip               playrule            vmsettings
callback            iaxsettings         portmappings        voiptrunk
conferences         inbounds            product             vpn
context             intercom            prompts             wansettings
datetime            ivr                 queues              whitelist
ddnssettings        kernellog           ringgroups
dhcpserversettings  language            ringtone
disa                lansettings         serviceprovider
sqlite> select * from admin;
admin|atcom|-1|0|||                password for WEB login
sqlite> .quit
```

Tips: if you enter some special mode unintentional in the database, and can't quit with command **.quit**, please use Ctrl + D.

1.4 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: **uImage_IP01.crc**

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

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

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

Kernel for IP2G4A/IP4G:**uImage_IP2G4A.crc**

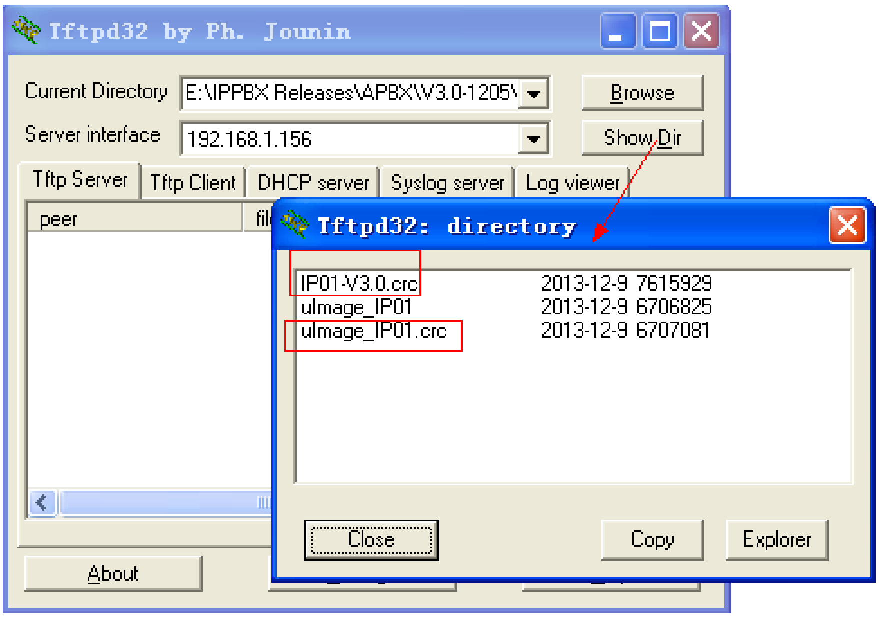
http://www.atcom.cn/cn/download/pbx/ip2g4a/uImage_IP2G4A.crc

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

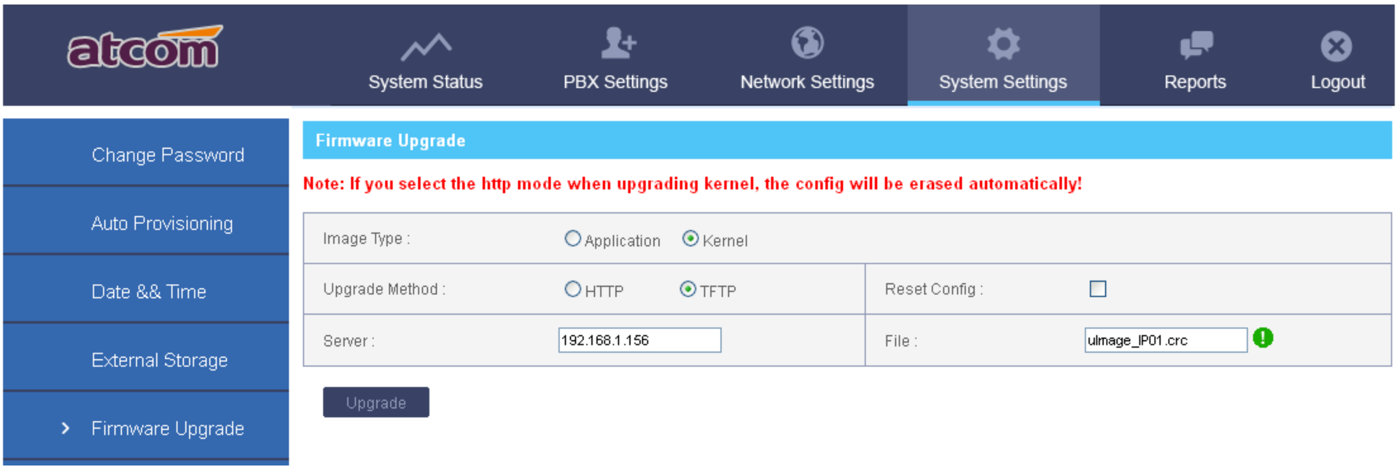
IP01/IP02: http://www.atcom.cn/cn/download/pbx/ip01/IP01-V3.0.crc

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

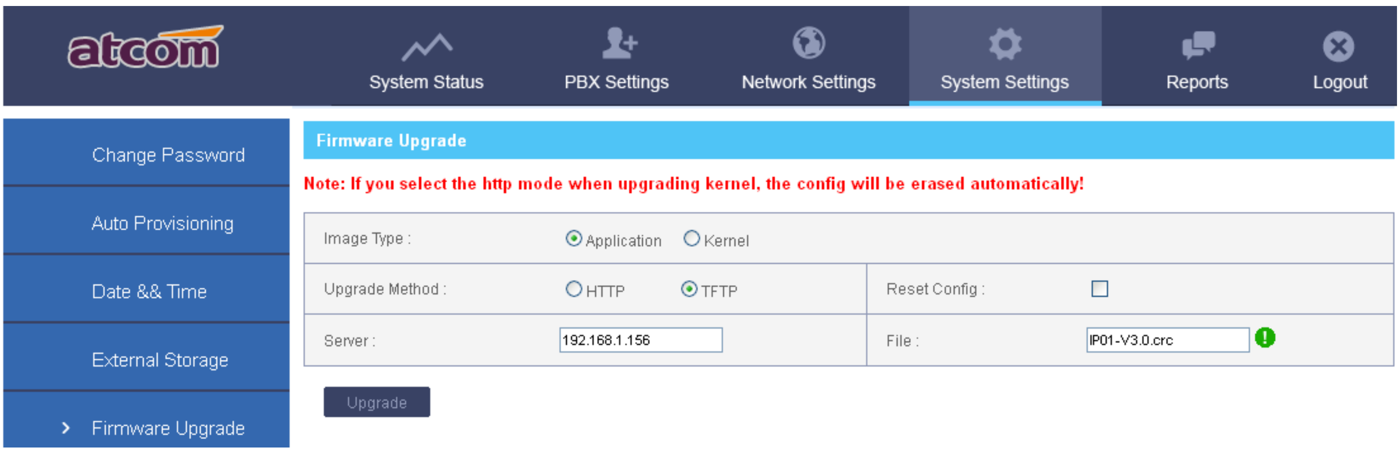
2) Set TFTP server: Choose firmware uploading directory as tftp server base directory.



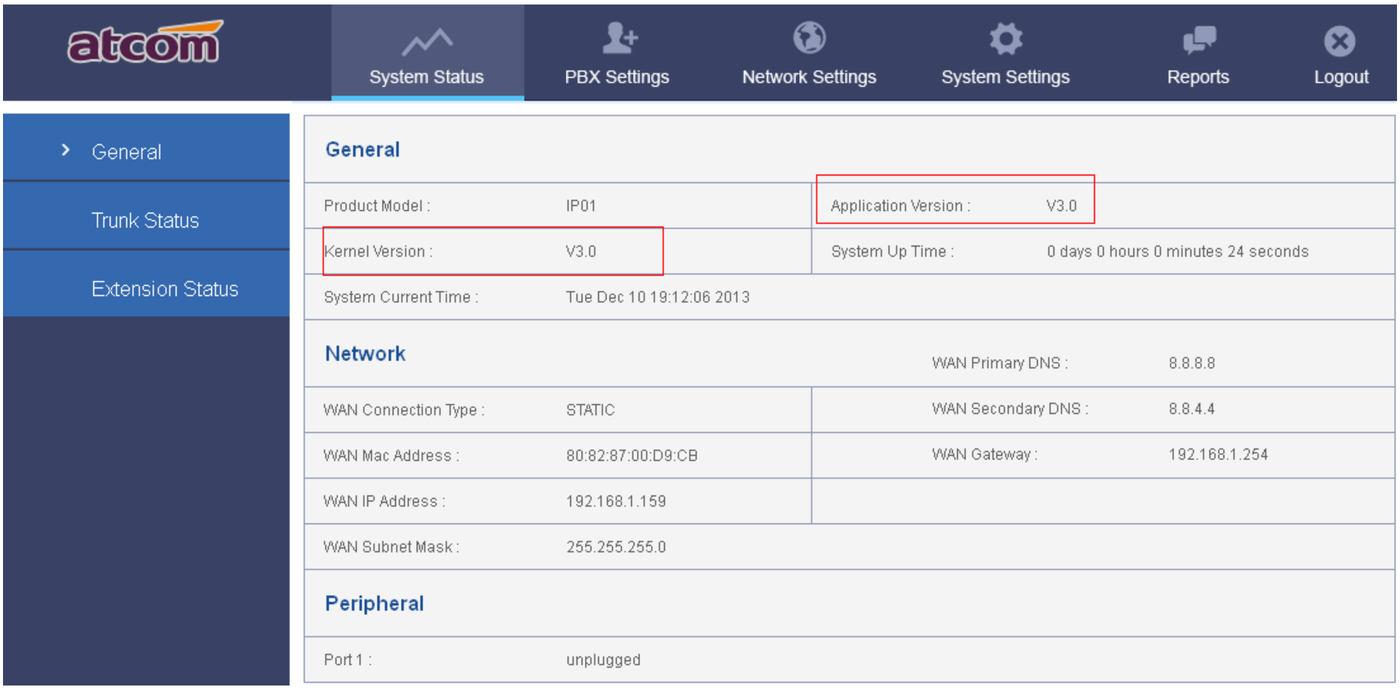
3) Upgrade Kernel without Reset Config chose



4) Upgrade Application without Reset Config chose



5) Check firmware version.



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.

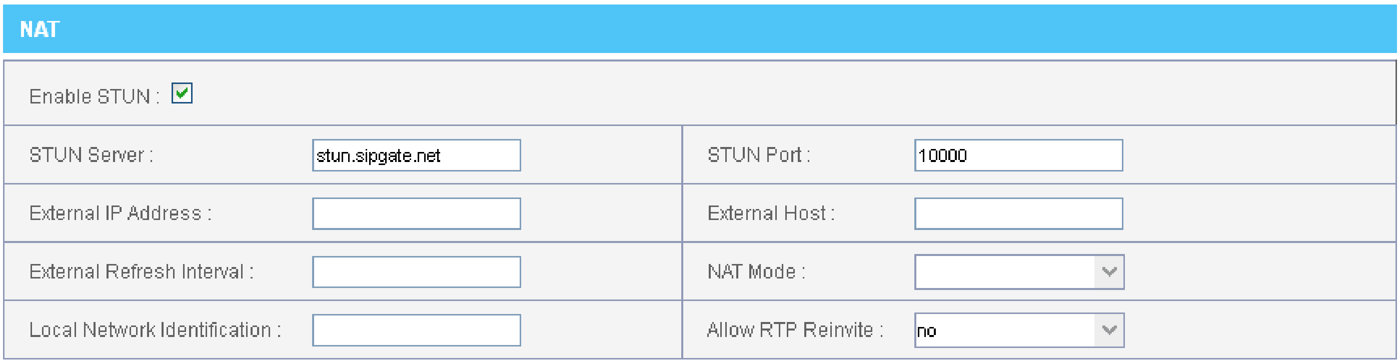
1) Please port forward below ports on your router first:

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

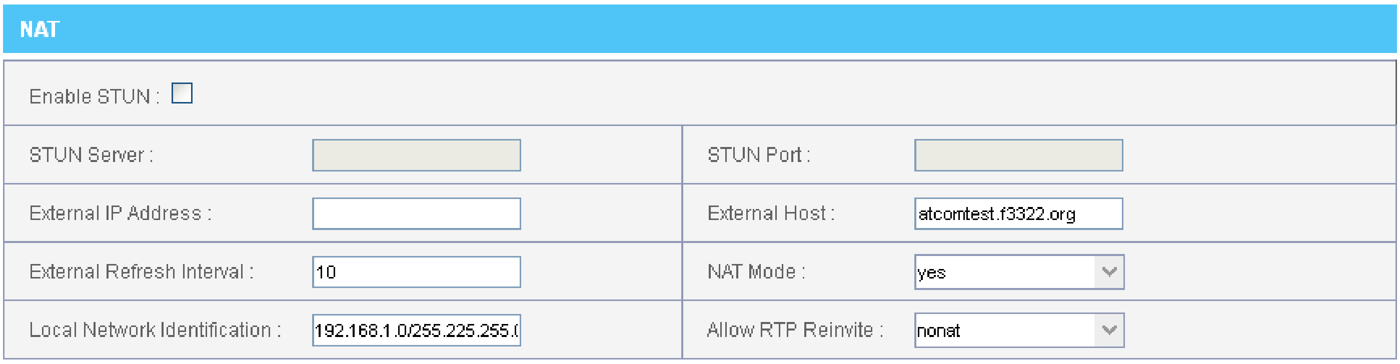
   RTP:10000~20000 (UDP)

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



b. Set NAT



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

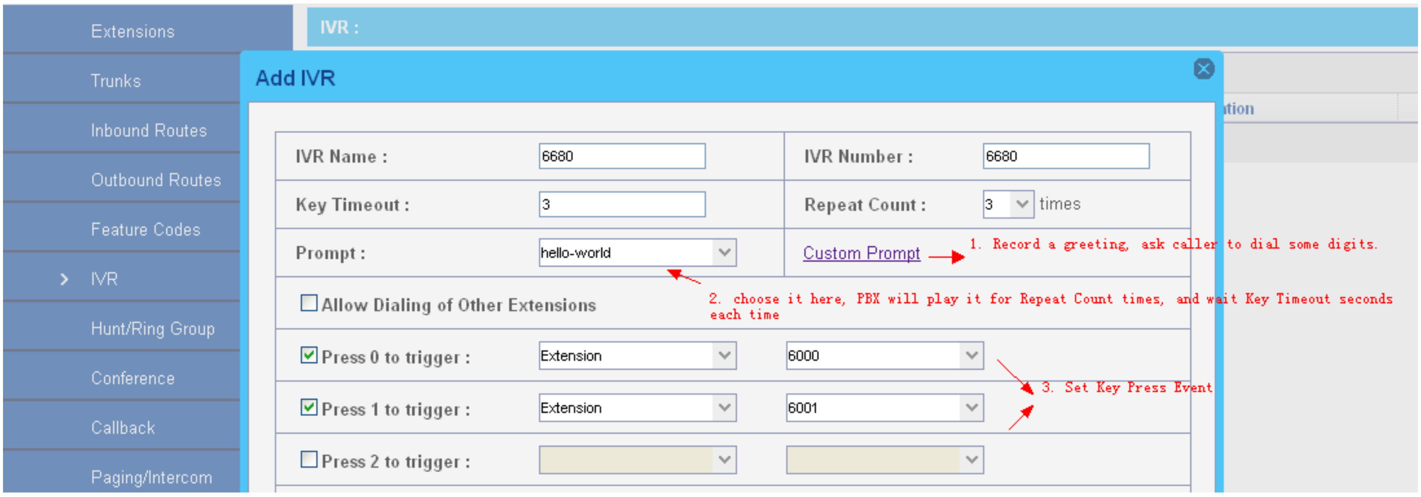
Your local network, format: sub-network/netmask.

Allow RTP Reinvite

nonat.

3.2 How to set IVR?

1) Record Custom Prompts
2) Set IVR like below



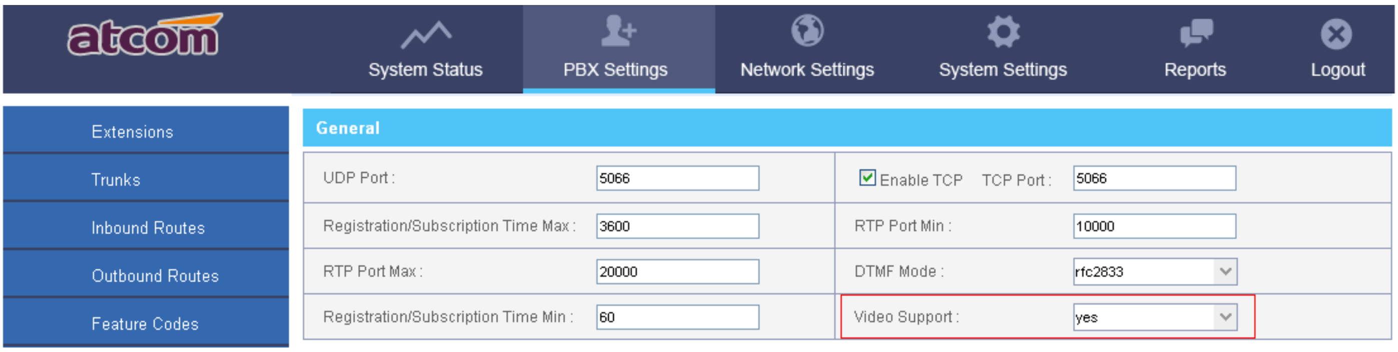
3.3 How to record your custom unavailable/busy/temp greetings for voicemail?

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

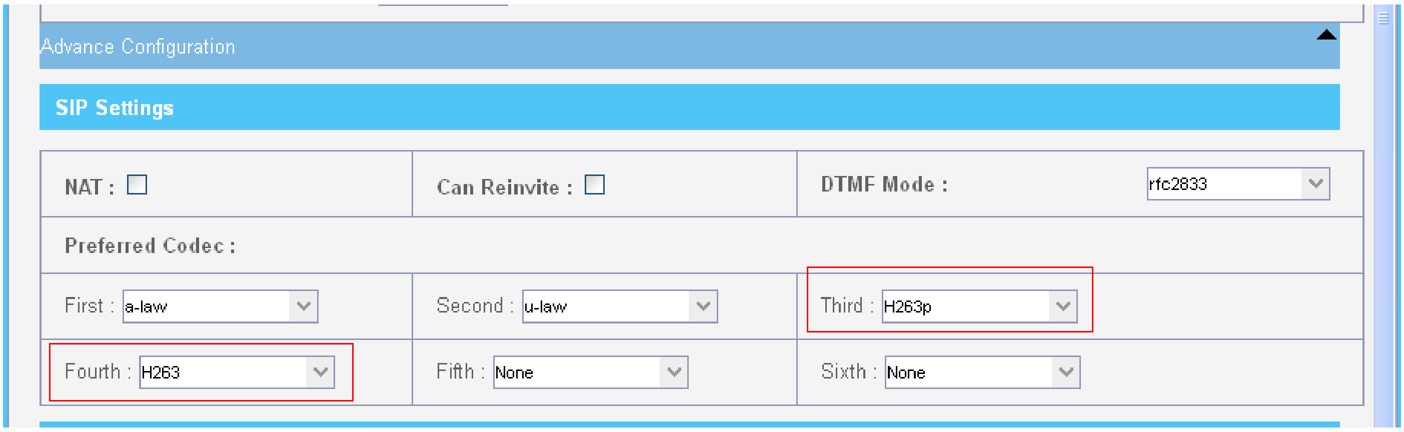
Dial *2(default setting) ->0 Mailbox options ->1 Record your unavailable message
->2 Record your busy message
->3 Record your name
->4 Manage your temporary recording

3.4 How to make video calls?

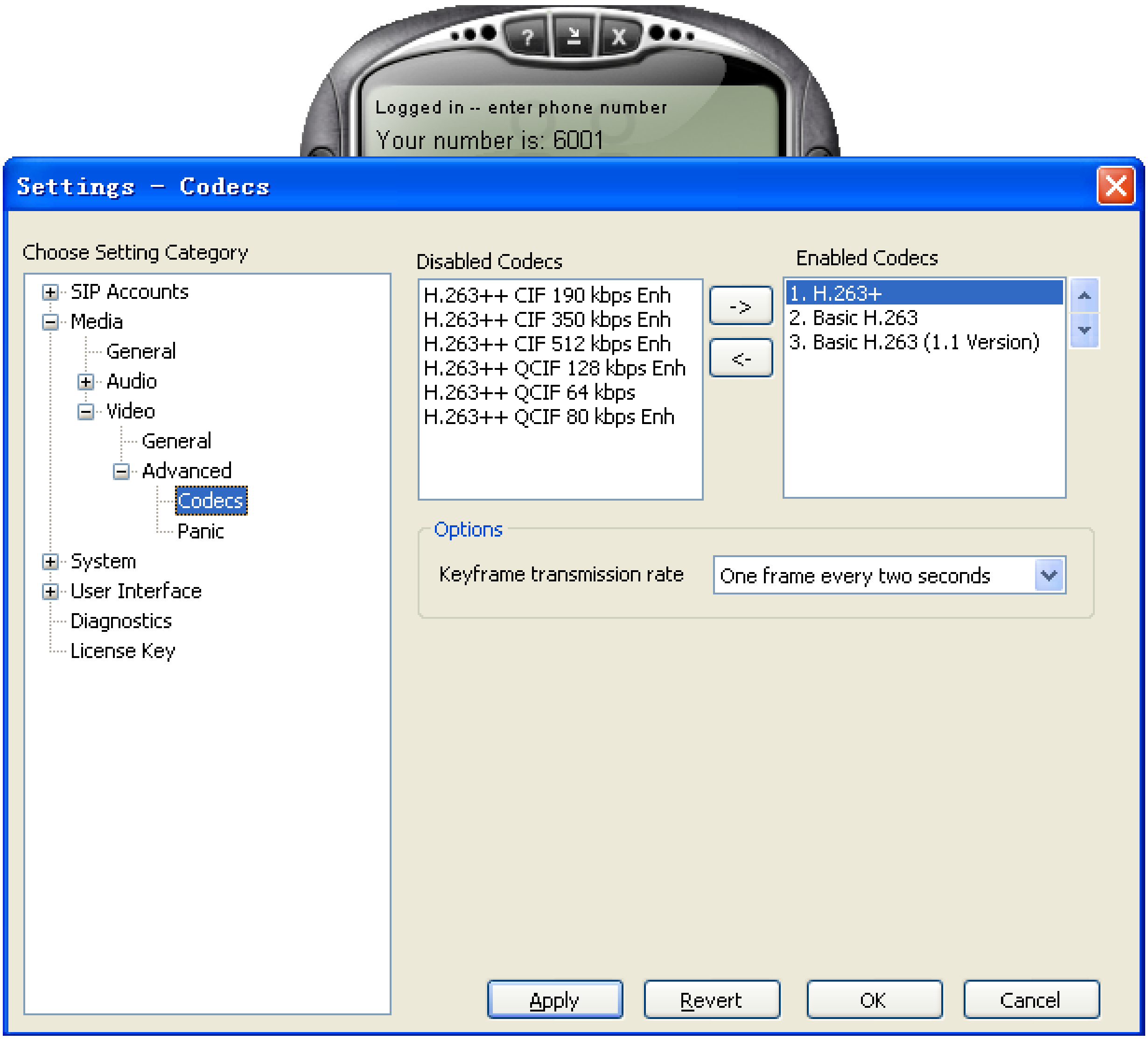
1) Make sure the Video Support on PBX Settings -> SIP Settings is set to yes



2) Select video code for the extension



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

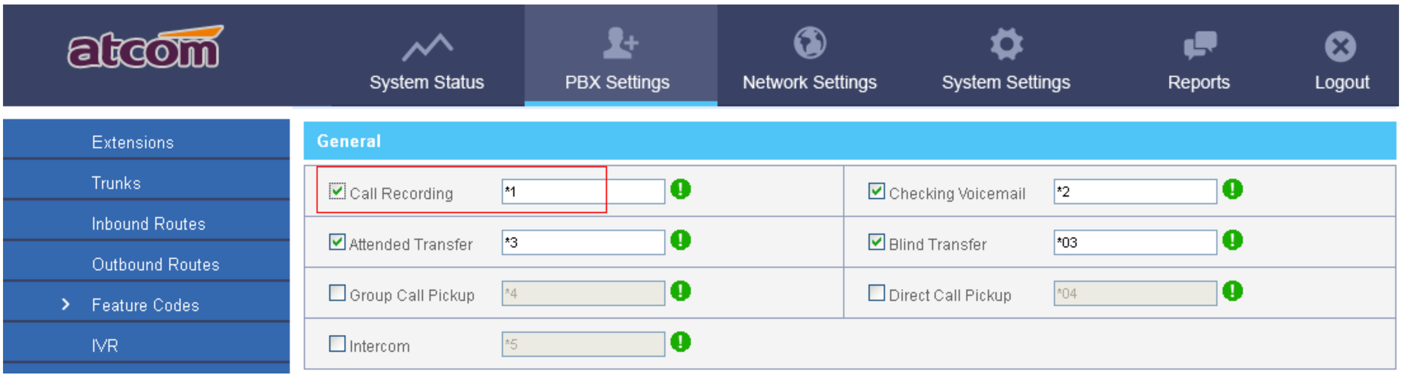
Make sure the modules are installed and detected before you use them, detailed WEB configuration can be found in user manual of each product.

3.6 How to check call recordings?

1. There are two ways to record calls.

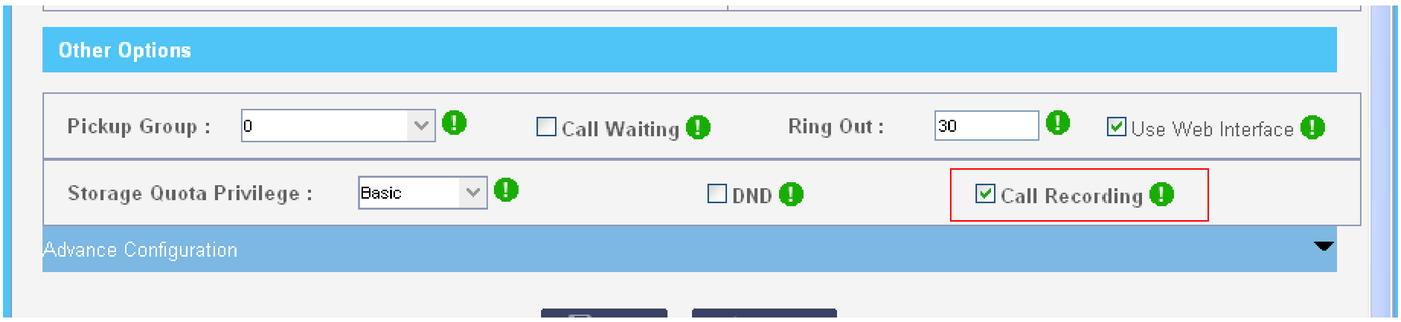
1) One touch record

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.

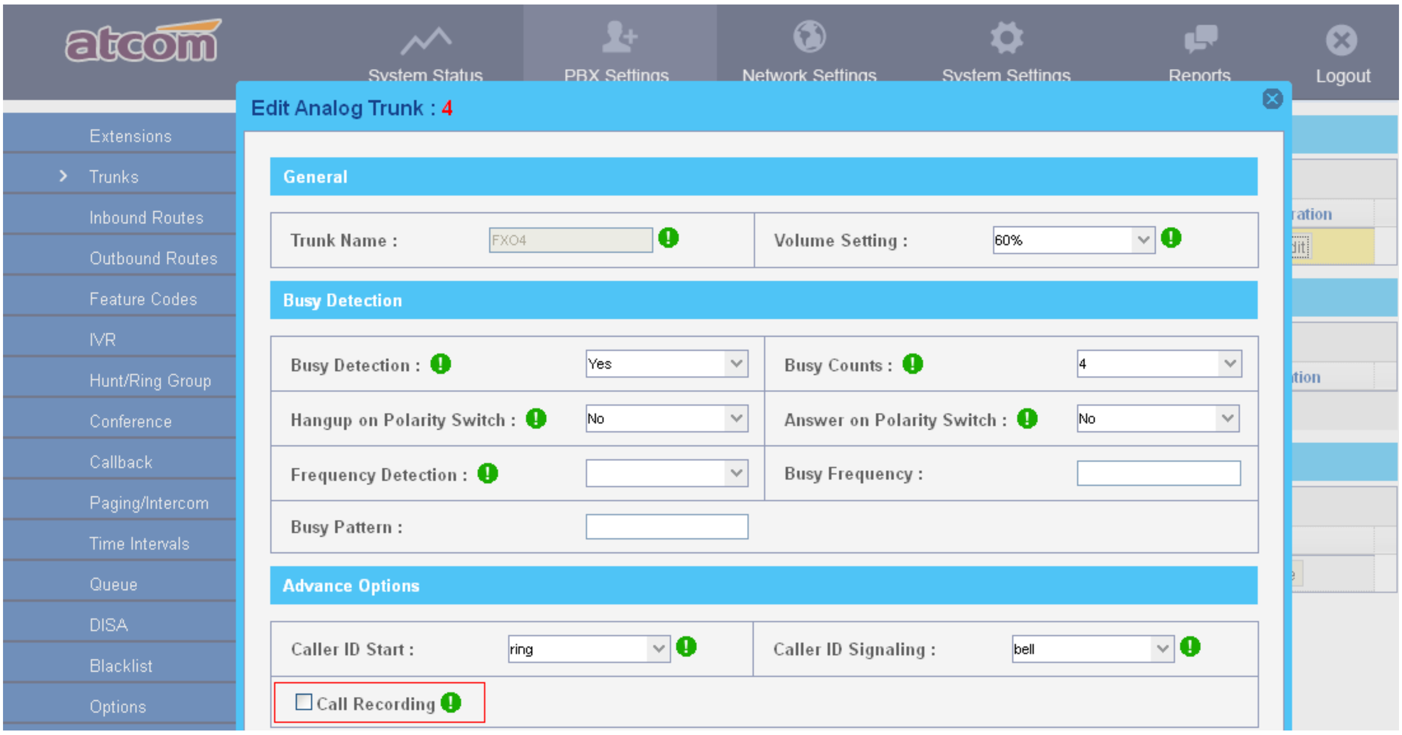


2) Automatic Call Record

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

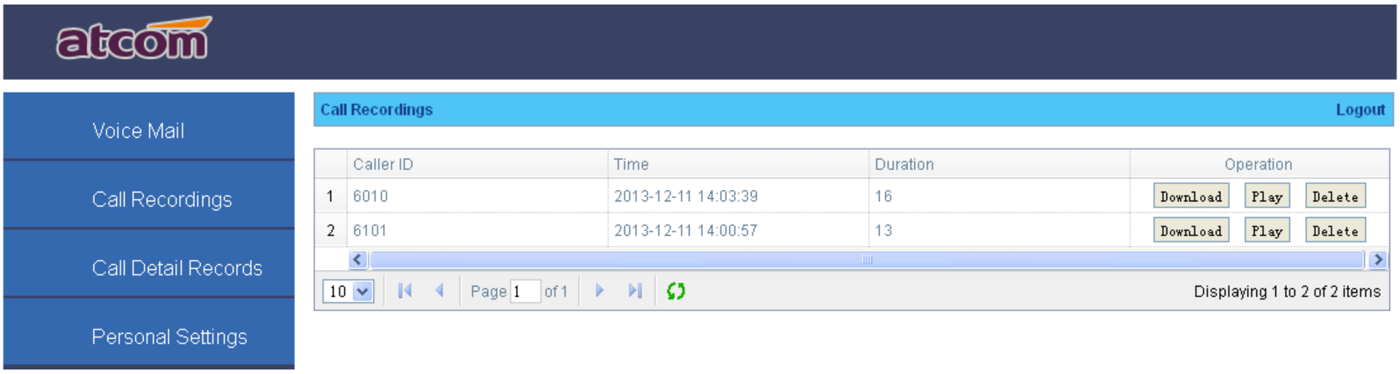


Enable call recording in PBX settings -> Trunks for a trunk, all incoming & outgoing calls through this trunk will be recorded automatically.

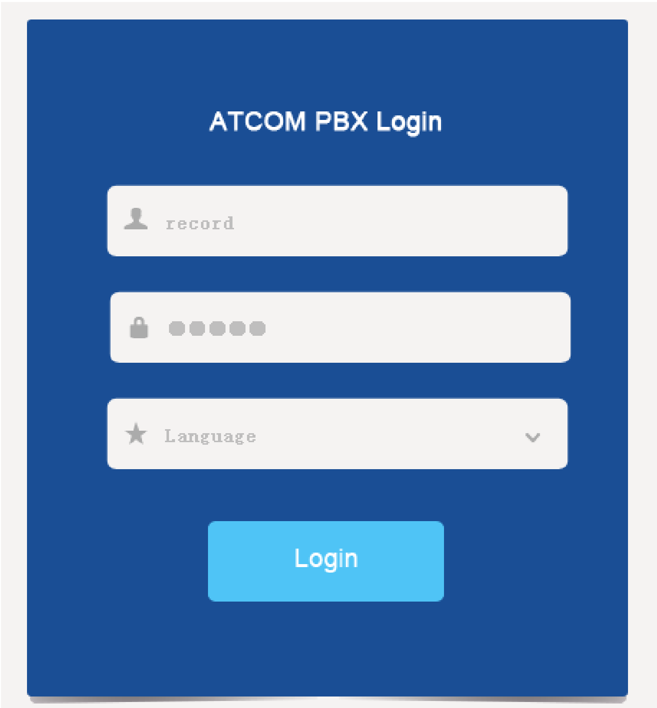


2. There are two ways to check call recordings.

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



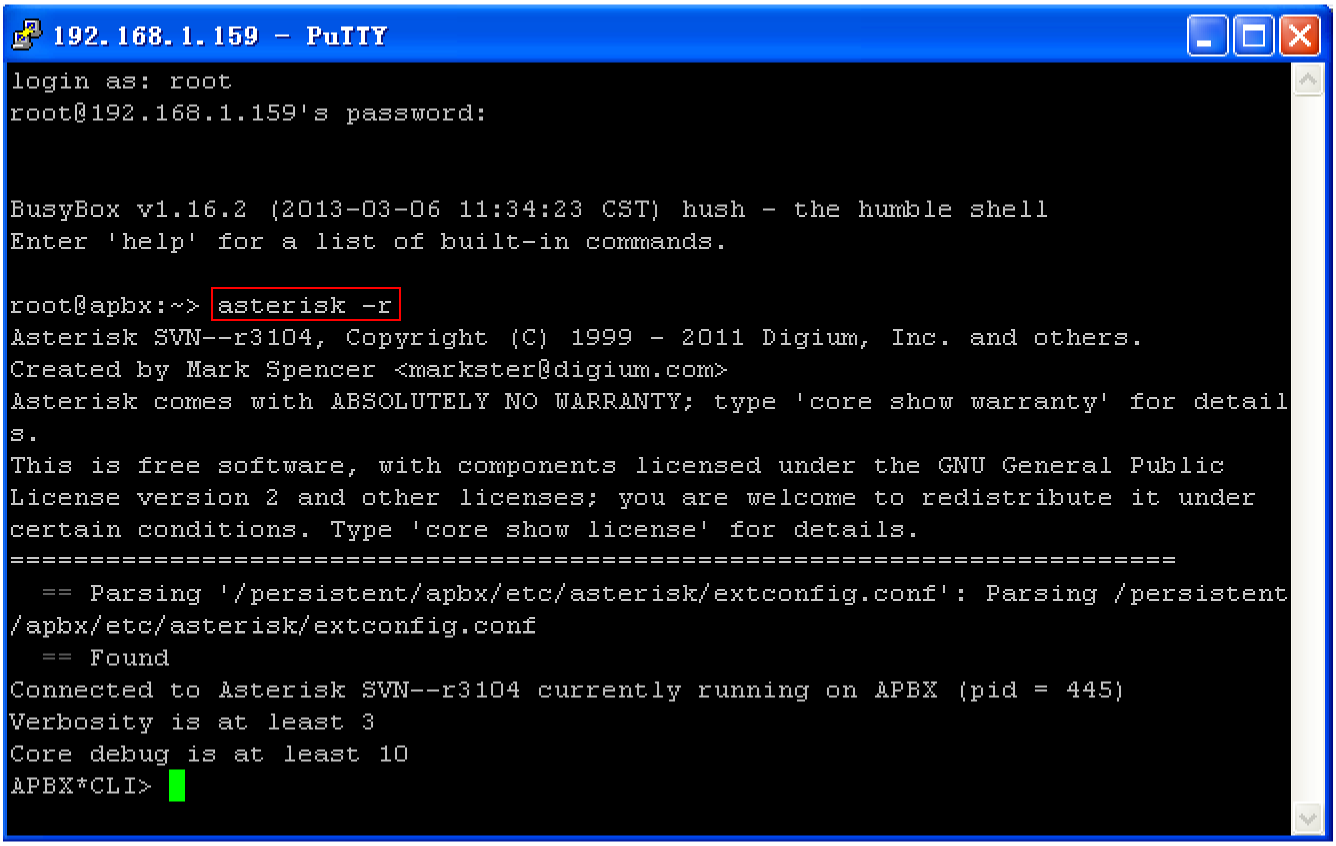
2) **record** user can login IPPBX with username/password: record/*the_password_of_admin* to check all recordings.



4. How to do Asterisk debug when there is call issue?

1) Login IPPBX via SSH as Chapter 1.1 do

2) Go to Asterisk CLI by running command: **asterisk –r** on Linux interface.



3) Make a call, all call processing information will be printed in the screen

4) Exit Asterisk CLI

APBX*CLI> **exit**
Executing last minute cleanups
Asterisk ending (0).
root@apbx:~>

4.1 Why I can't call out through trunk?

4.2.1 There is no right outbound route for the extension

I want to dial 83018618 with extension 6000, but PBX prompts extension can't be found:

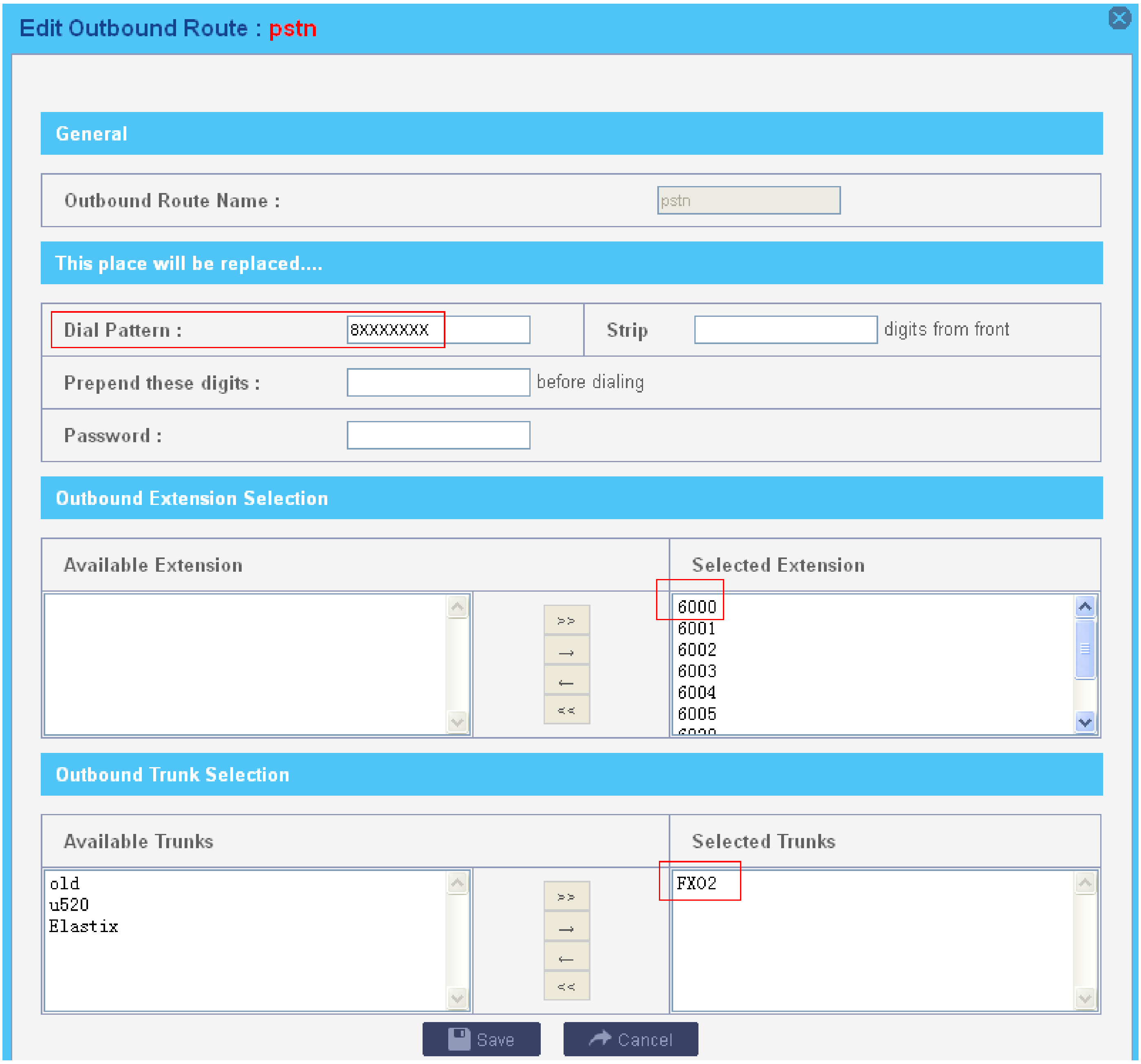
```
[2013-09-12 11:09:44] NOTICE[506]: chan_sip.c:22191 handle_request_invite: Call from '6000' (192.168.1.198:50
60) to extension '83018618' rejected because extension not found in context 'DIALPLAN_6000'.
APBX*CLI>
```

Solution:

1) Please check your outbound route, make sure the number you dialed can match the Dial Pattern which acts as a filter. 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)
. Wildcard, matches anything remaining (digits and/or *#)
! 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.

2) Make sure the extension you use in selected into Selected Extension



4.2.2 Use phone's send key wrong

I want to dial 1234567#, but IPPBX dials 1234567 only.

```
    -- Executing [1234567@DIALPLAN_6000:1] Macro("SIP/6000-00000000", "trunkdial-failover-0.1,1,1234567,TRUNK-FXO2,") in new stac
k
    -- Executing [s@macro-trunkdial-failover-0.1:1] Wait("SIP/6000-00000000", "1") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:2] GotoIf("SIP/6000-00000000", "0?3:4") in new stack
    -- Goto (macro-trunkdial-failover-0.1,s,4)
    -- Executing [s@macro-trunkdial-failover-0.1:4] Set("SIP/6000-00000000", "TCOUNT=3") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:5] Set("SIP/6000-00000000", "OldCallerID=6000") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:6] Set("SIP/6000-00000000", "RINGTIME=30") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:7] Set("SIP/6000-00000000", "CDR(userfield)=outbound") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:8] GotoIf("SIP/6000-00000000", "0?hangup") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:9] GotoIf("SIP/6000-00000000", "0?setdod") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:10] Set("SIP/6000-00000000", "CALLERID(name)=Tina") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:11] Set("SIP/6000-00000000", "CALLERID(num)=334455") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:12] Set("SIP/6000-00000000", "_SIPSRTP=") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:13] SetMusicOnHold("SIP/6000-00000000", "default") in new stack
    -- Executing [s@macro-trunkdial-failover-0.1:14] Dial("SIP/6000-00000000", "DAHDI/2/1234567,30,tTXxWwkKg") in new stack
    -- Called DAHDI/2/1234567
    -- DAHDI/2-1 answered SIP/6000-00000000
    -- Hanging up on 'DAHDI/2-1'
    -- Hungup 'DAHDI/2-1'
  == Spawn extension (macro-trunkdial-failover-0.1, s, 14) exited non-zero on 'SIP/6000-00000000' in macro 'trunkdial-failover-0.
```

Solution:

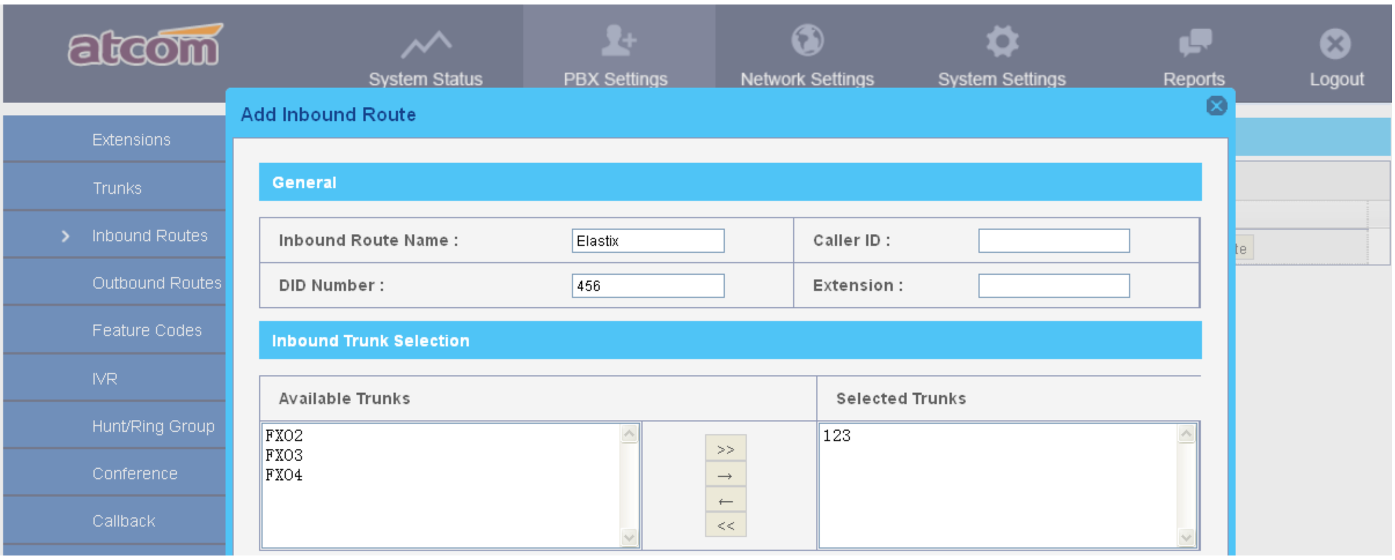
For some phones, # acts as a send key, a sign implying entering is finished and the call should be sent immediately. It will not be regarded as a part of the number. You need to cancel this function of # on your phones.

4.2 Why I can't receive incoming call from trunk?

4.2.1 Does DID number set right in inbound route?

For Analog Trunk, the DID number must be blank.

For SIP trunk, leaving DID number blank means set DID to as same as the account name set in selected SIP trunk. so if the number others dial to reach your account is different from the register account, please set the DID number to the one others dial to reach you.



5. How to secure your IPPBX?

5.1 Put IPPBX behind your firewall

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

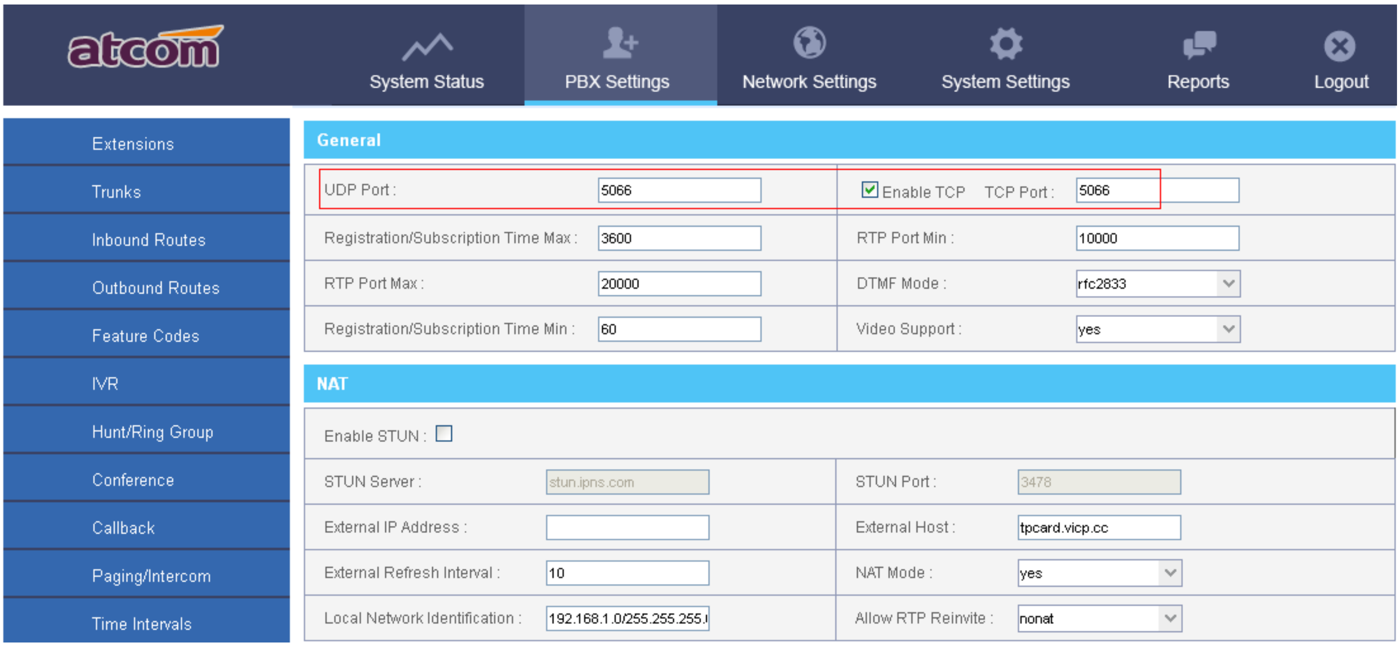
It is dangerous to open accessible port to the outside world, for example SSH (TCP :22), WEB (TCP: 80/443)

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.



5.2 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.

5.3 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.

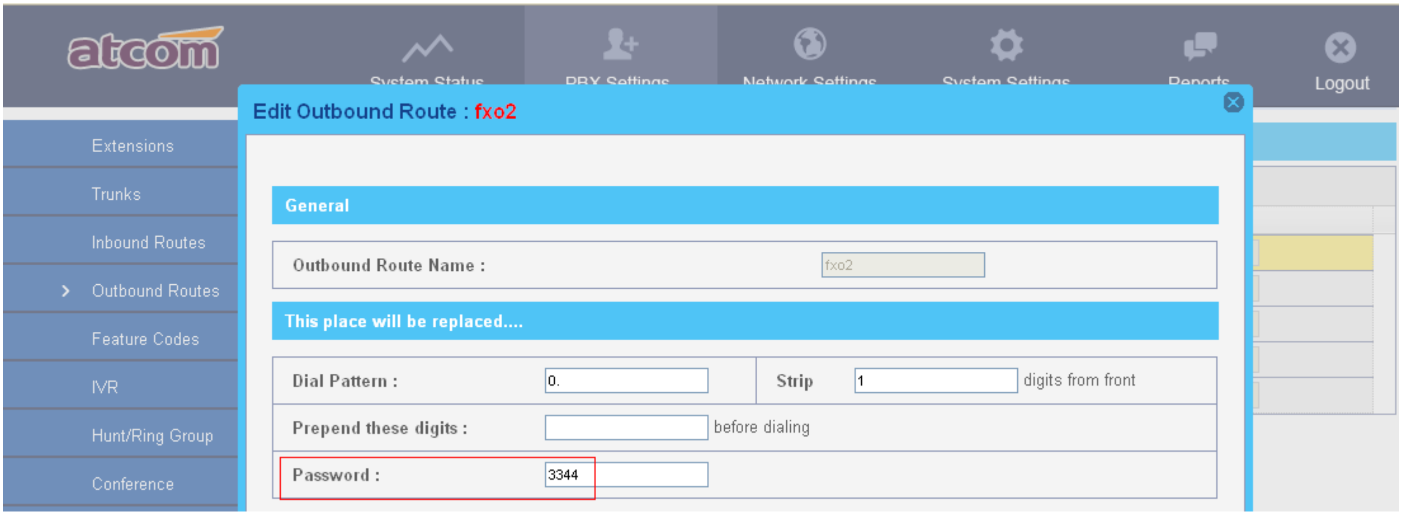
Go to PBX settings -> Extensions -> Edit an extension -> Advanced Configuration

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

5.4 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.



5.5 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	Edit Delete
2	ACCEPT	SIPprovider	UDP	216.207.245.47/255.255.255.255		5060:5063	Edit Delete
3	DROP	dropothers	UDP			5060:5063	Edit Delete

Auto Defense

New Rule

	Port	Protocol	Rate	Operation
1	80	TCP	50	Edit Delete

SIP Defense

New Rule

	SIP Packets	Time in Seconds	Operation
1	200	1	Edit Delete

Other Options

☐ Disable Ping ☐ Drop All

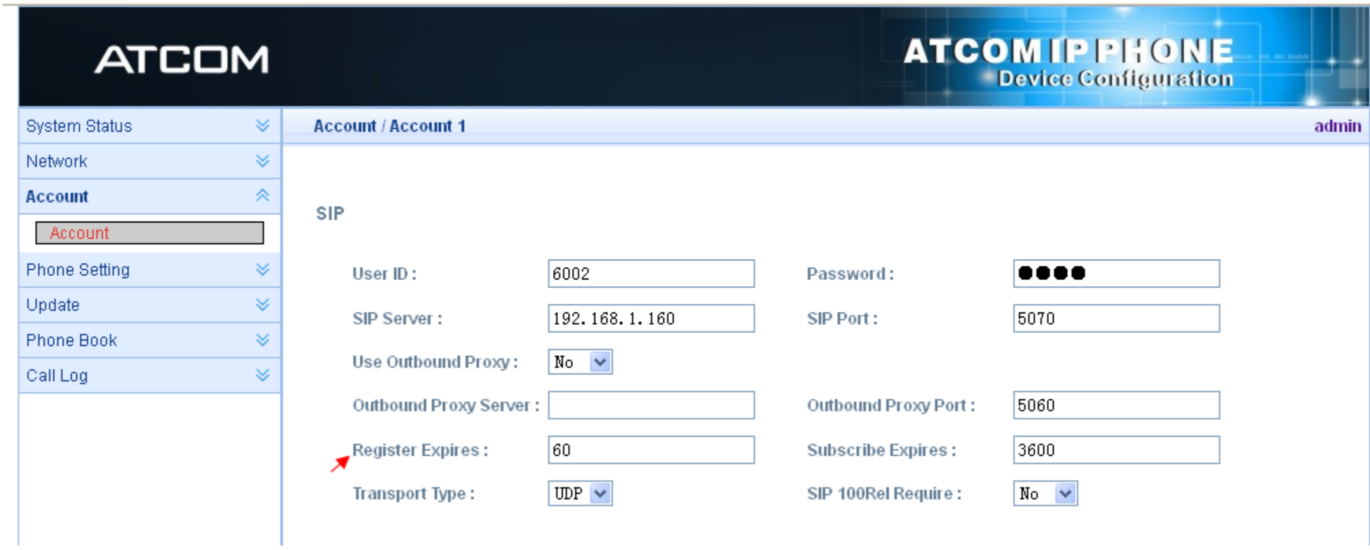
6. Others

6.1 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.

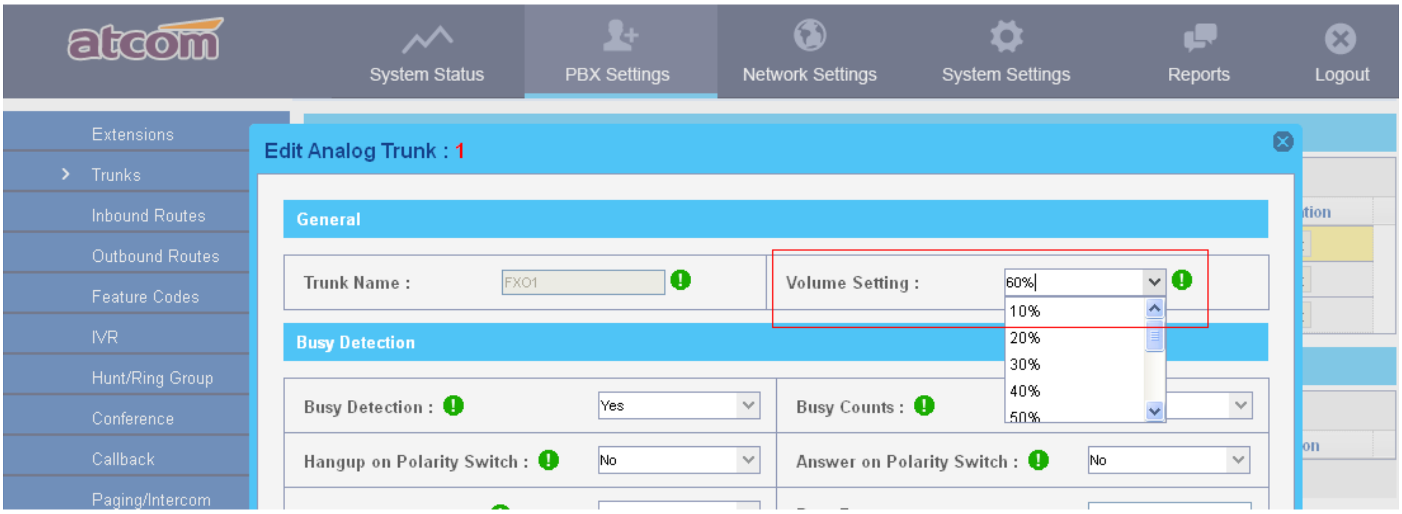
After PBX reboot, all the registry information is lost, PBX will think of that IP phone is unavailable until IP phone register again when the register expiration is reach.

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



6.2 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.



6.3 How to capture CID information?

1. Login IPPBX via SSH as chapter 1.1 do.
2. Execute command **dahdi_monitor n –r stream.raw**, n is channel number, for example, if you want to monitor channel 1's caller id signaling, the command is dahdi_monitor 1 –r stream.raw;

```
root@apbx:~> dahdi_monitor
Usage: dahdi_monitor <channel num> [-v[v]] [-m] [-o] [-l limit] [-f FILE | -s FILE | -r FILE1 -t FILE2] [-F FILE | -S FILE | -R FILE1 -T FILE2]
Options:
        -v: Visual mode.  Implies -m.
        -vv: Visual/Verbose mode.  Implies -m.
        -l LIMIT: Stop after reading LIMIT bytes
        -m: Separate rx/tx streams.
        -o: Output audio via OSS.  Note: Only 'normal' combined rx/tx streams are output via OSS.
        -f FILE: Save combined rx/tx stream to mono FILE. Cannot be used with -m.
        -r FILE: Save rx stream to FILE. Implies -m.
        -t FILE: Save tx stream to FILE. Implies -m.
        -s FILE: Save stereo rx/tx stream to FILE. Implies -m.
        -F FILE: Save combined pre-echocanceled rx/tx stream to FILE. Cannot be used with -m.
        -R FILE: Save pre-echocanceled rx stream to FILE. Implies -m.
        -T FILE: Save pre-echocanceled tx stream to FILE. Implies -m.
        -S FILE: Save pre-echocanceled stereo rx/tx stream to FILE. Implies -m.
Examples:
Save a stream to a file
        dahdi_monitor 1 -f stream.raw
Visualize an rx/tx stream and save them to separate files.
        dahdi_monitor 1 -v -r streamrx.raw -t streamtx.raw
Play a combined rx/tx stream via OSS and save it to a file
        dahdi_monitor 1 -o -f stream.raw
Save a combined normal rx/tx stream and a combined 'preecho' rx/tx stream to files
        dahdi_monitor 1 -f stream.raw -F streampreecho.raw
Save a normal rx/tx stream and a 'preecho' rx/tx stream to separate files
        dahdi_monitor 1 -m -r streamrx.raw -t streamtx.raw -R streampreechorx.raw -T streampreechotx.raw
root@apbx:~> dahdi_monitor 1 -f stream.raw
Writing combined stream to stream.raw
```

3. Make call to PBX via port 1, wait for 5 seconds, hang up the call, call into the same PSTN line again, wait for 5 seconds, then type Ctrl + c to stop the command;
4. Deploy TFTP server in your local network, and put the file to the TFTP server with below command

tftp –pr *file_name TFTP_server_address*

```
root@apbx:~> ls
stream.raw
root@apbx:~> tftp -pr stream.raw 192.168.1.156
```

5. Email that stream.raw file to us