Smart Security IP Intercom

IS/IV 750 User Manual

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Declaration

Information regarding this guide is subject to change without any notice.

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About this manual

Thank you for choosing Smart Security IP Intercom IS/IV 750.This IP Intercom is specially designed for the user under the public environment with fashionable appearance and complete functions. This manual aims to help you quickly use Smart Security IP Intercom IS/IV 750.Before use ,please read the packing list and safety notes section of this manual ,communicate with the system administrator to confirm if the current network environment can meet the requirements of configuring the Intercom. If this is your first time to use Smart Security IP Intercom IS/IV 750, we recommend that you should read the quick installation guide and product technical manual. The document can be downloaded from the following website: http://www.escene.cn/en/en.

Summary

1.Getting Started

1.1 Outline

ESCENE SN-SG IP Door Phone IS/IV 750 is the newest indoor VoIP intercom which professional designed as the requirements from industry users. The device has the characteristic of well compatibility with different platform, offering users a convenient service.

IS/IV 750 is dust proof, water proof and dismantle prevention, having a fashion appearance and high protection. IS/IV 750 is a bond of door opening by long-distance DTMF, Password and RFID. Its powerful performance, stability and reasonable price make it a perfect choice of industry user.

NOTE: IS/IV 750-PRT, P is POE, R is Reader(optional), T is Touchpad(optional).

1.2 Product Features

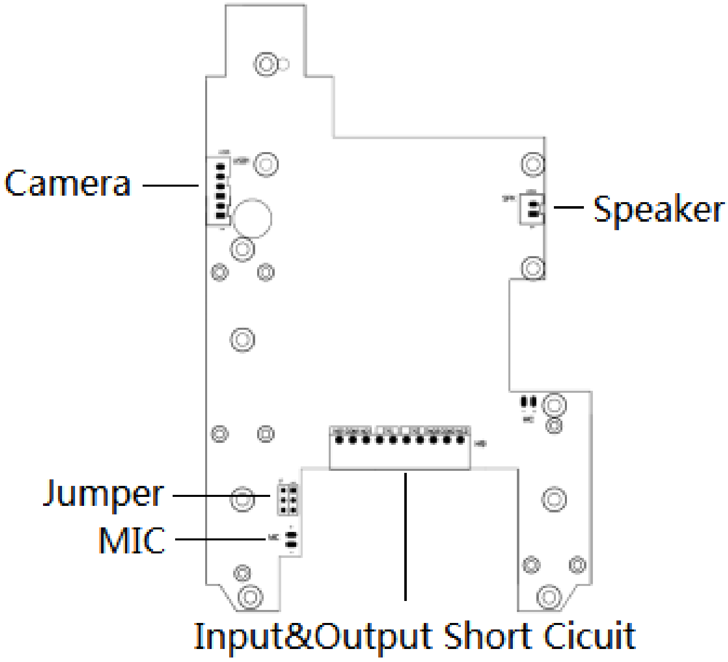
- Support embedded or wall-mounted installation.
- High-fidelity sound quality, HD codec, Full duplex hands-free calls;
- 2*RJ45 standard Ethernet Ports, integrated PoE;
- User-defined DSS key, which can be set up to Speedy dial, intercom,etc.
- Support Plug-and-Play, auto-provision, remote maintenance and management; TR069.

Intercom features
WEB support Multi-Language ;4 SIP account; Hotline; Call hold, Call waiting, Call forward; Call transfer (blind/busy/ask);Mute, DND; Auto-answer, 3-way conferencing;1 DSS programmable key(Speed dial, Intercom etc.);Volume control; Direct IP call without SIP proxy; Default Ring tone 1 selection/import/delete; Custom Ring tone 2 selection/import/delete; Time setting(SNTP/SIP Server/Manual);Support SIP main/standby server;
LED Status
Available--OFF；Busy--Steady；registration failed--Flashing
Network parameters
SIP v1 (RFC2543), v2 (RFC3261);DNS SRV (RFC3263);NAT Traversal: STUN mode; DTMF: In-Band, RFC2833, SIP Info, Auto; HTTP/HTTPS Web Management; IP Assignment: Static/DHCP/PPPoE; Network support Bridge mode; TFTP/DHCP/PPPoE client; DNS client, NAT/DHCP server;

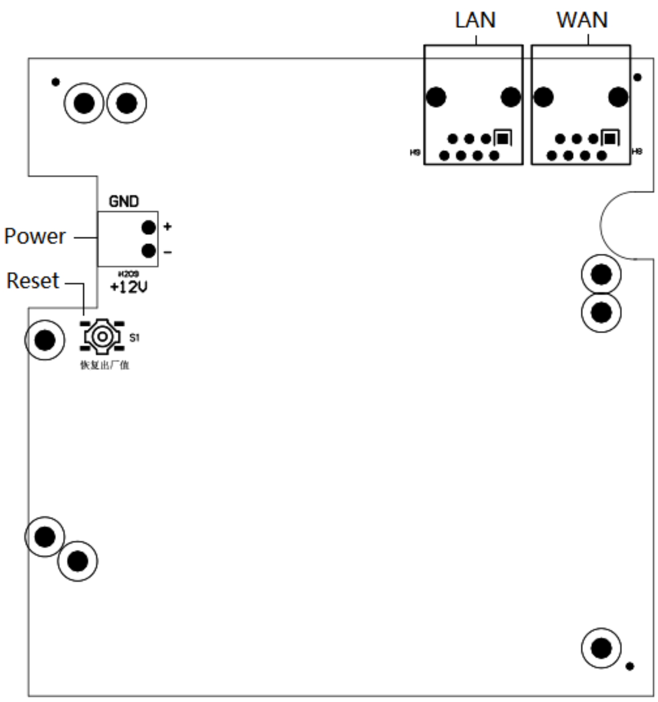
Security
LLDP, VLAN QoS (802.1pq),VPN(L2TP);Transport Layer Security (TLS);Digest authentication using MD5/MD5-sess;Secure configuration file via AES encryption; Admin/User 2-level configuration mode;
Voice features
Wideband Codec: G.722;Narrowband codec: G.711μ/A, G.723.1;G.726, G.729a/b, iLBC; VAD, CNG, AEC, AGC; Full-duplex;
Video features
Video codec: H.264; Image codec: JPEG/PNG/BMP/GIF; Video format: MP4/3GP/FLV; Video call resolution: QCIF / CIF / VGA / 4CIF (1280x720P); Bandwidth selection: 64kbps~4Mbps; Frame rate selection: 10~30fps.
Physical properties
1 DSS programmable key with led light 2 RJ-45 10/100m Ethernet ports, Power adapter: DC 12V/1A; Power over Ethernet ,IEEE 802.3af,class 0; Each motherboard port, check the picture illustration below "Mother Broad Interface".
Carton packaging
The whole Size：162*112*40mm
Product Certification
RoHS Compliant CE FC ISO 9001
Platform Compatibility Test (non-certificate)
ZTE/Alcatel-Lucent/Asterisk/Broadsoft/Metaswitch/Yeastar/Avaya/3CX/Elastix/HUAWEI etc.

1.3 Technical Information

INTERFACE SPECIFICATION:



Expansion Board Connector Diagram



Mother Board Connector Diagram

*This data is for information purposes only and is subject to change without notice.

2. Intercom Installation

Generally system administrator will connect your new IS/IV 750 IP Intercom to company LAN network. If not, please refer to below illustration.

Open IS/IV 750 packing box, according to the packing list, check the related attachment to make sure to no omitting. Packing list as follows.

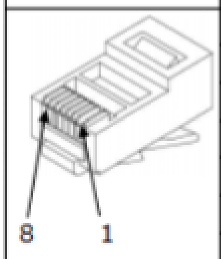
- IS/IV 750 Intercom
- Quick operating guide
- 8*Screws
- 2*Rubbers

IS/IV 750 could be installed to internet according to the below steps.

A) Connect Network

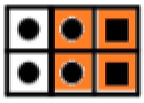
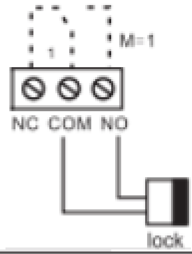
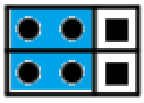
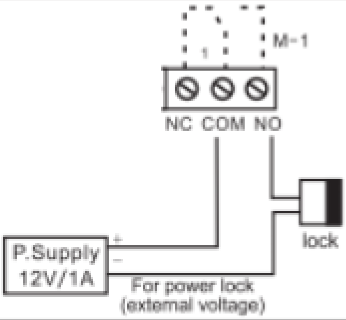
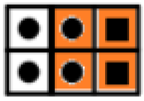
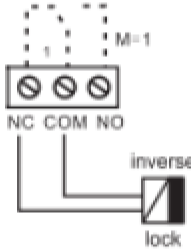
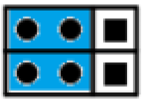
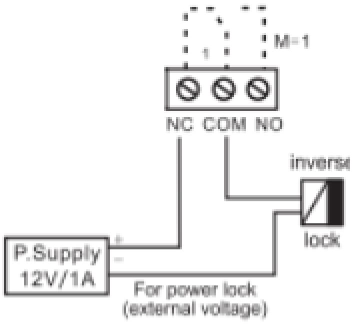
B) Internal/External Electric Lock Connection Driver Option

C) Internal/External Electric Lock Connection

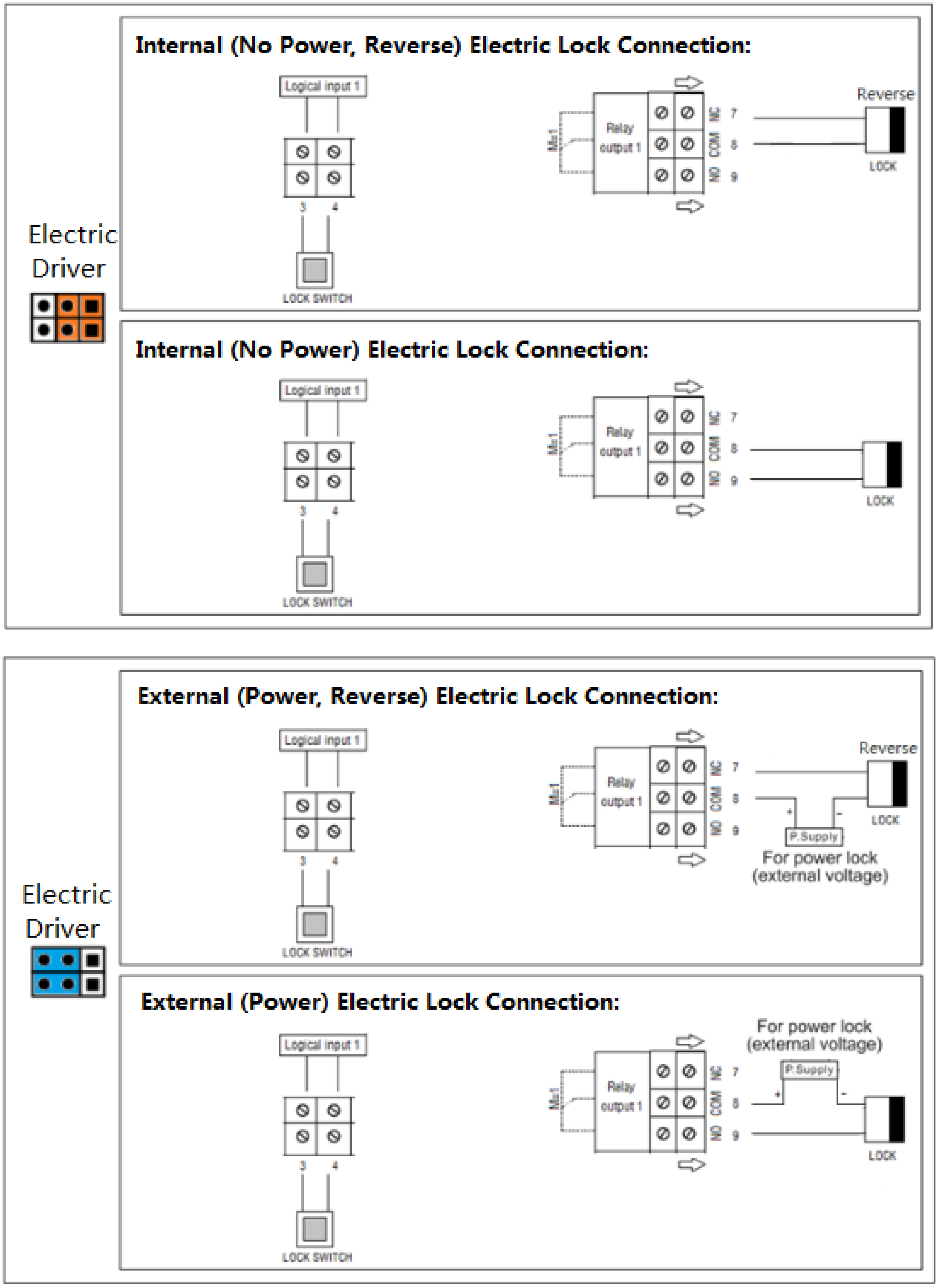
RJ-45				IS720	
	Pin No.	Marking	Colour	LAN	PC
	1	Tx+		TX1+	RX2-
	2	Tx -		TX1-	RX2+
	3	Rx+		RX1+	TX2-
	4	PoE+		POE+	Null
	5	PoE+			
	6	Rx -		RX1-	TX2+
	7	PoE -		POE-	Null
	8	PoE -			

Connect network

NOTE: When the initial electric current of the lock is less than 500mA/12V, you can access to the internal driven mode and use the POE of the Voice Access System or 12V DC to control the switch of the electric lock; When the initial electric current of the lock is more than 500mA/12V, you need to access to the external driven mode(Use specialized DC power to control the electric lock).

Electric lock power supply mode		Electric lock		Jumper	Mode of connection
Internal	External	NO	NC		
√		√			
	√	√			
√			√		
	√		√		

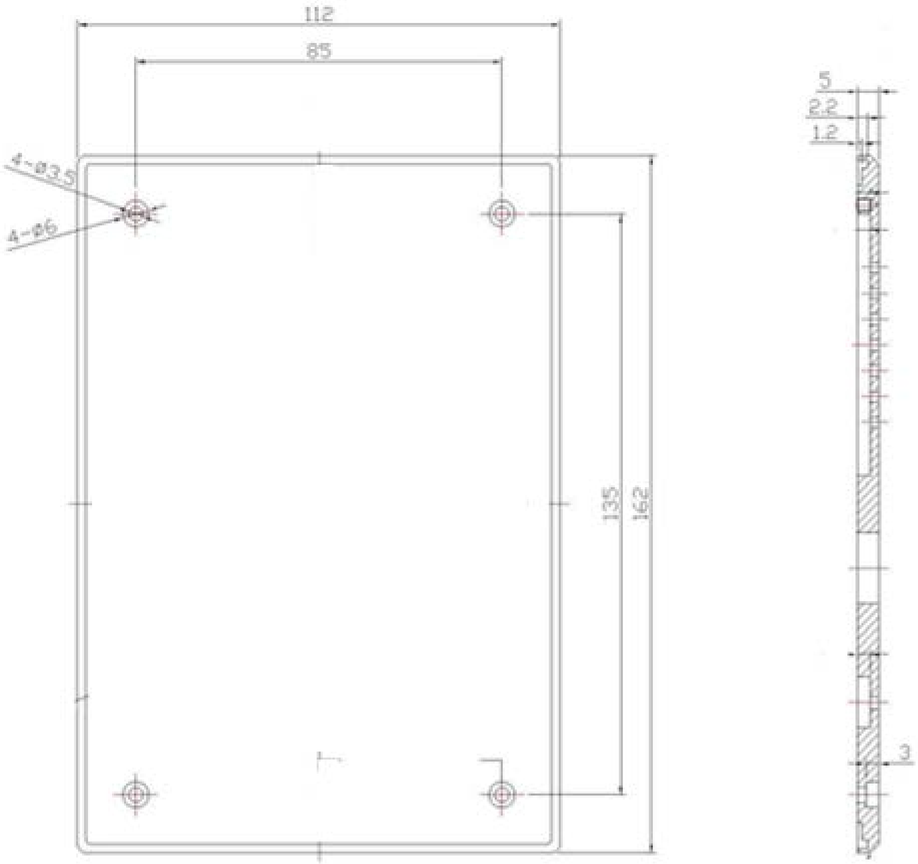
Internal/External Electric Lock Connection Driver Option



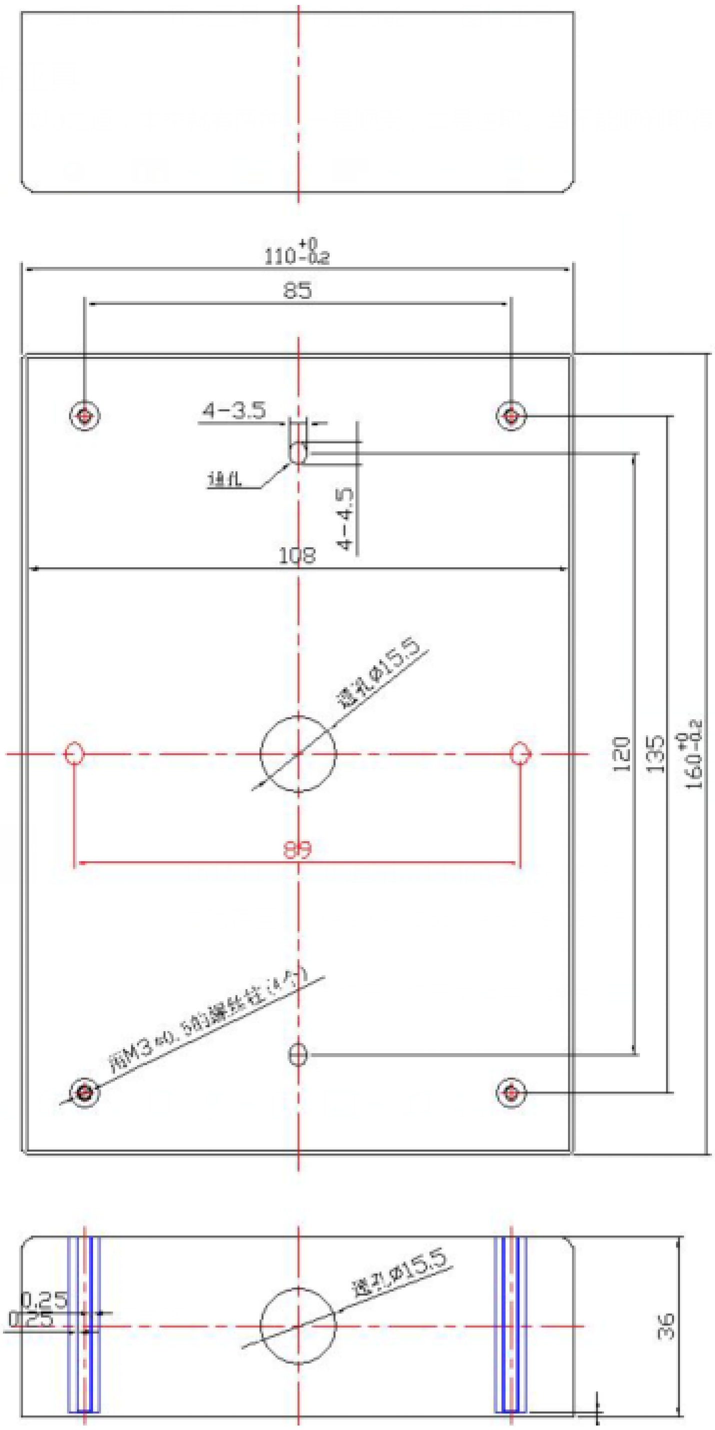
Internal/External Electric Lock Connection

2.1 Embedded

If the product is used for embedding, then the cutting of embedded need to be a little bigger than the installing hole of standard dimension 168*116*42mm .(42mm means the extra aluminum housing thickness.),as follows. Notice: the embedded value is up to the actual situation.



Front and side structure

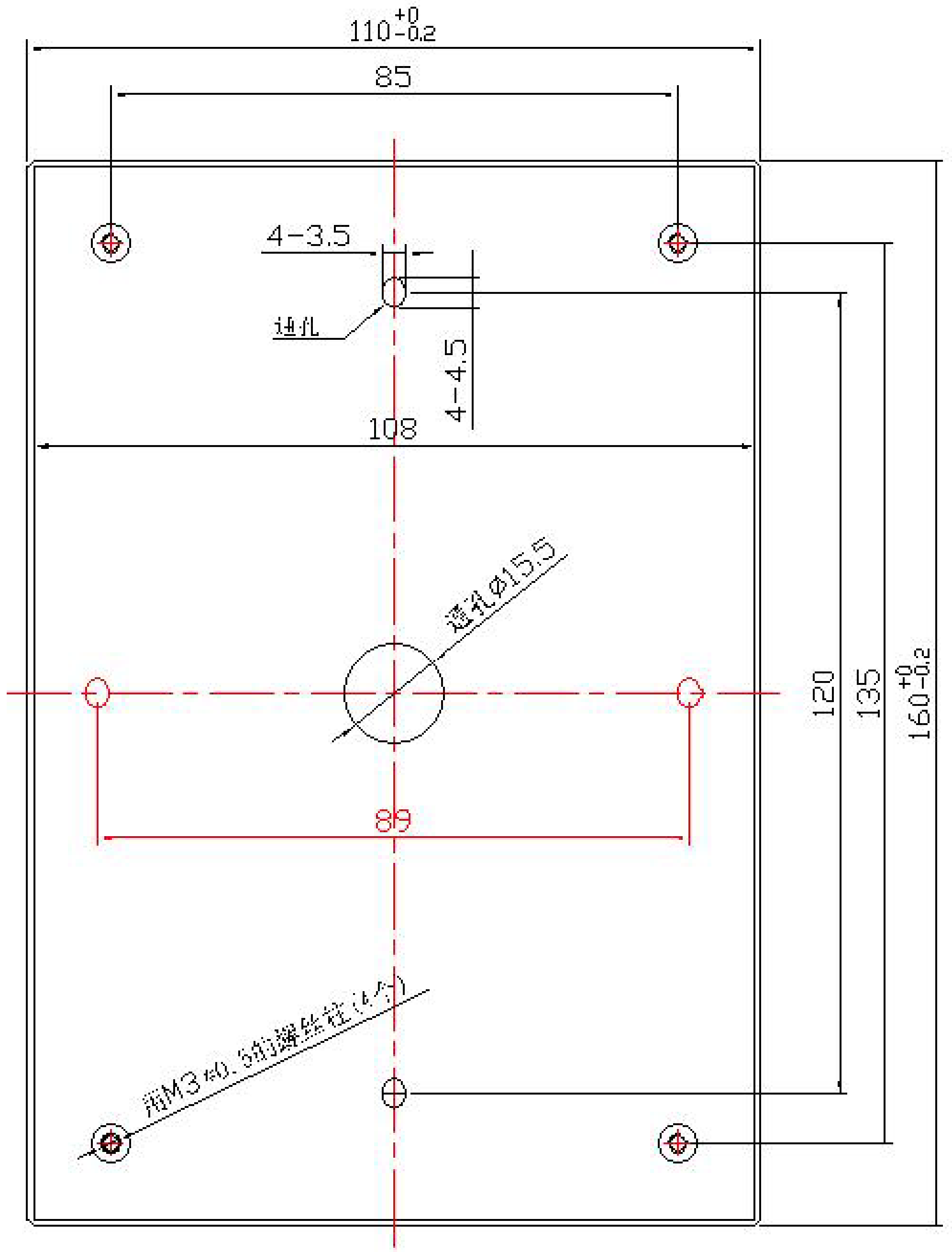


Back structure

2.2 Equipment installation

1、Open the aluminum housing.

2、Put the IP intercom into it according to the cutting embedded specification. After that, fix four M3*12 screws on the wall with the screw driver.



3、After installing inter housing, well-set the related wiring and replace the aluminum housing. Power on and start testing.

3.Configuration of IP intercom

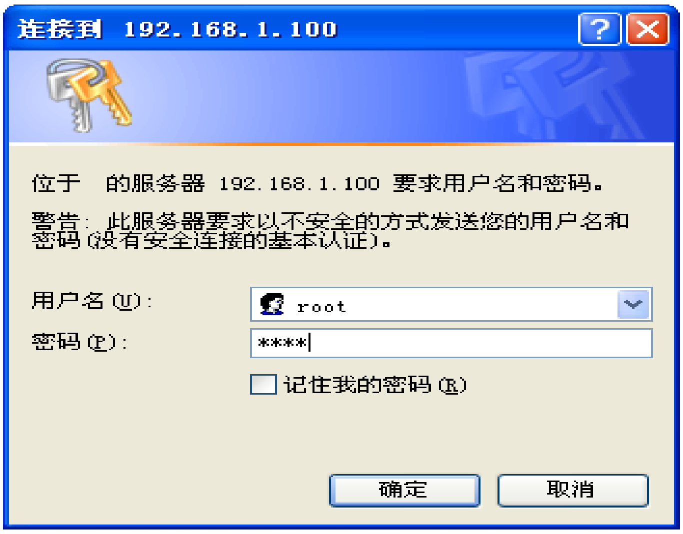
You need to know the IP address of IP Intercom before starting setting. You could learn how to get IP address below. Default IP address is 192.168.1.153.in static status. If getting from DHCP, you need to search the related IP address by the third software scanning equipment MAC.(Getting IP by DHCP is not suggested.)

3.1 Remote WEB Management

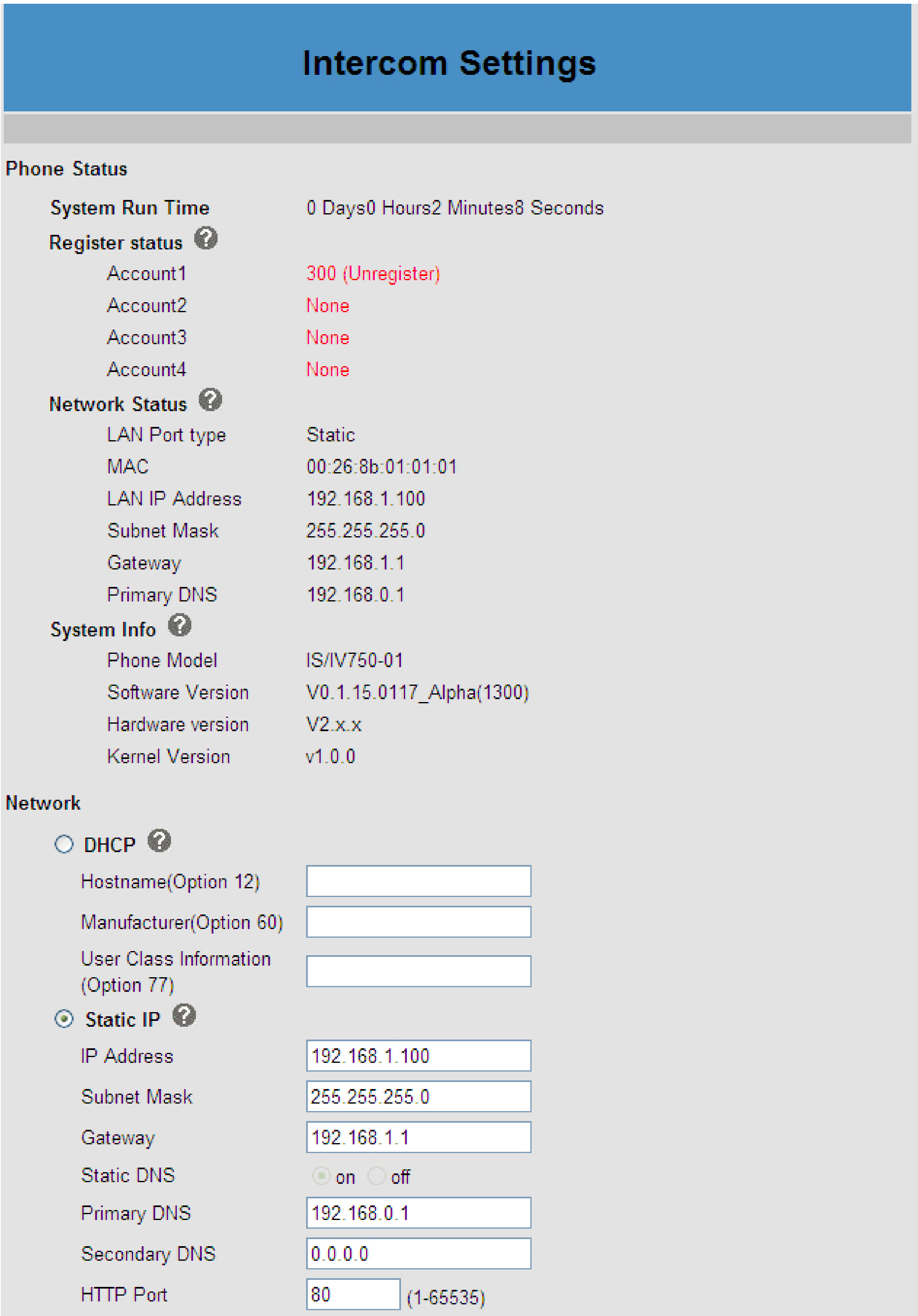
This equipment's factory IP address is using static IP(ip:192.168.1.100, Gateway:192.168.1.1).

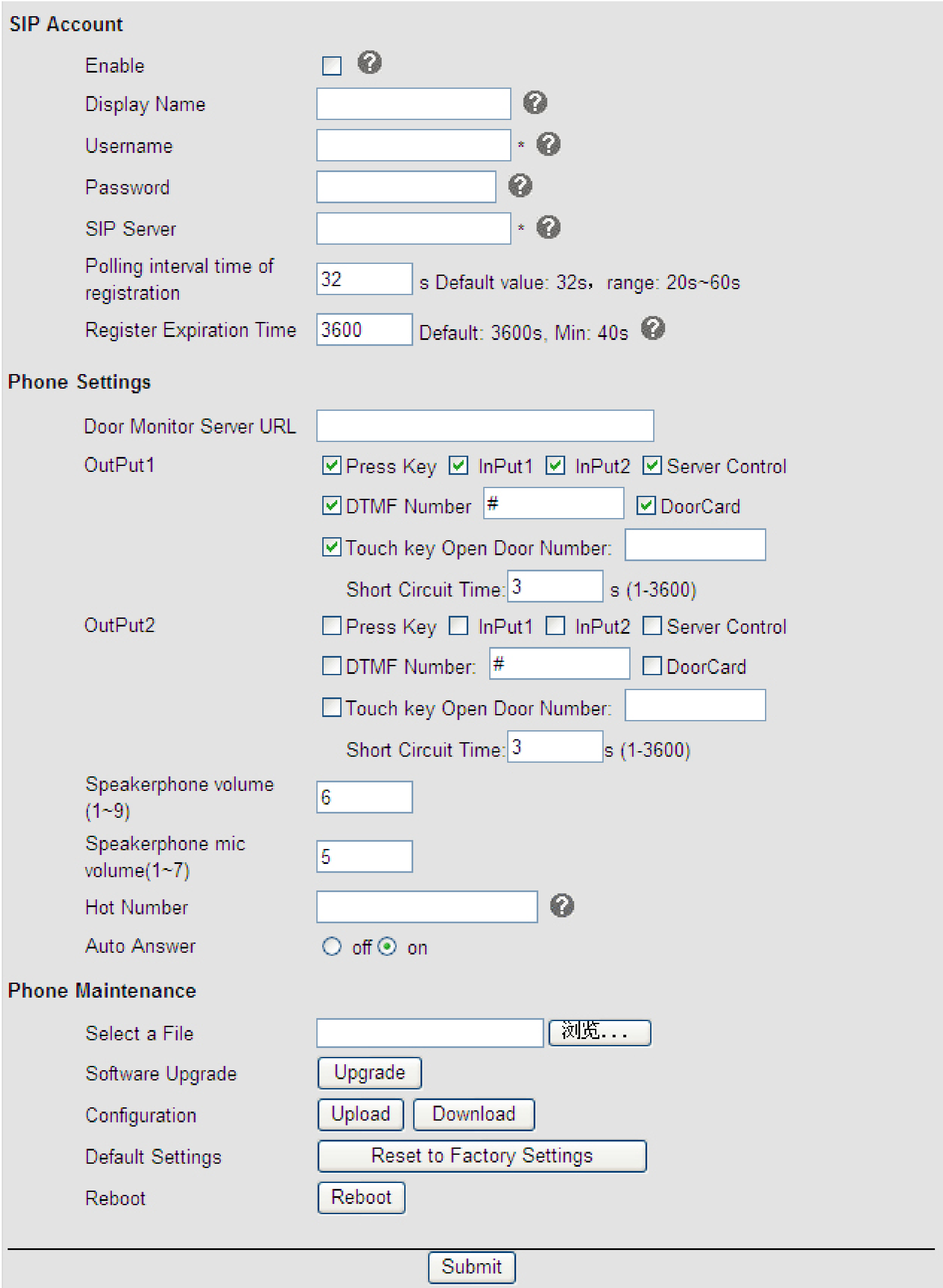
http://192.168.1.100/user.asp is easy web management.
http://192.168.1.100/home.asp is all-round web management.

Once input the IP address of intercom on the web browser and tap the “enter” on the keyboard. Then a login screen will pop up from the intercom equipment. You need to input user name and password. Both default user name and password of system is case letters “root”



After the log-in，the easy administrate web page of the intercom will pop up.


Intercom Settings
Phone Status
System Run Time
0 Days0 Hours2 Minutes8 Seconds
Register status
Account1
300 (Unregister)
Account2
None
Account3
None
Account4
None
Network Status
LAN Port type
Static
MAC
00:26:8b:01:01:01
LAN IP Address
192.168.1.100
Subnet Mask
255.255.255.0
Gateway
192.168.1.1
Primary DNS
192.168.0.1
System Info
Phone Model
IS/IV750-01
Software Version
V0.1.15.0117_Alpha(1300)
Hardware version
V2.x.x
Kernel Version
v1.0.0
Network
DHCP
Hostname(Option 12)
Manufacturer(Option 60)
User Class Information (Option 77)
Static IP
IP Address
192.168.1.100
Subnet Mask
255.255.255.0
Gateway
192.168.1.1
Static DNS
on
off
Primary DNS
192.168.0.1
Secondary DNS
0.0.0.0
HTTP Port
80
(1-65535)



SIP Account

Enable

Display Name

Username *

Password

SIP Server *

Polling interval time of registration: 32 s Default value: 32s，range: 20s~60s

Register Expiration Time: 3600 Default: 3600s, Min: 40s

Phone Settings

Door Monitor Server URL

OutPut1: Press Key, InPut1, InPut2, Server Control, DTMF Number #, DoorCard, Touch key Open Door Number:, Short Circuit Time: 3 s (1-3600)

OutPut2: Press Key, InPut1, InPut2, Server Control, DTMF Number: #, DoorCard, Touch key Open Door Number:, Short Circuit Time: 3 s (1-3600)

Speakerphone volume (1~9): 6

Speakerphone mic volume(1~7): 5

Hot Number

Auto Answer: off on

Phone Maintenance

Select a File: 浏览...

Software Upgrade: Upgrade

Configuration: Upload Download

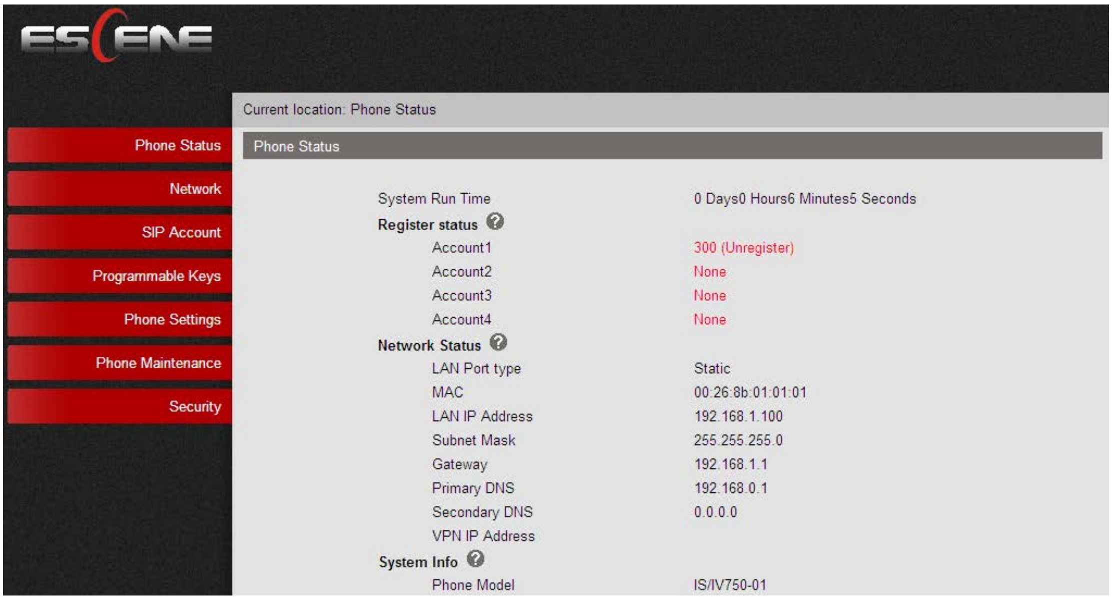
Default Settings: Reset to Factory Settings

Reboot: Reboot

Submit

If you want to open the all-round web management, you can enter an URL as follow:

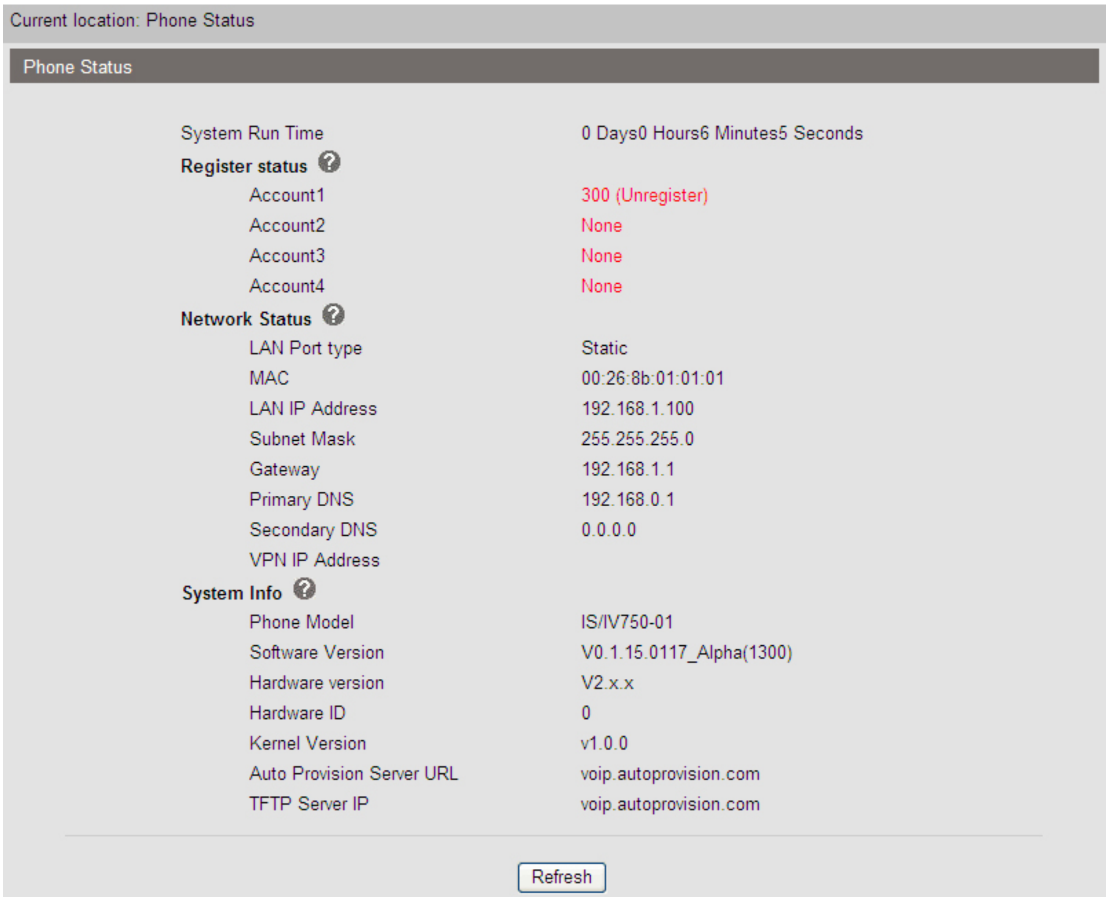
http://192.168.1.100/home.asp



Here you can see as below information: System Run Time, Register Status, Network Status, System Information,

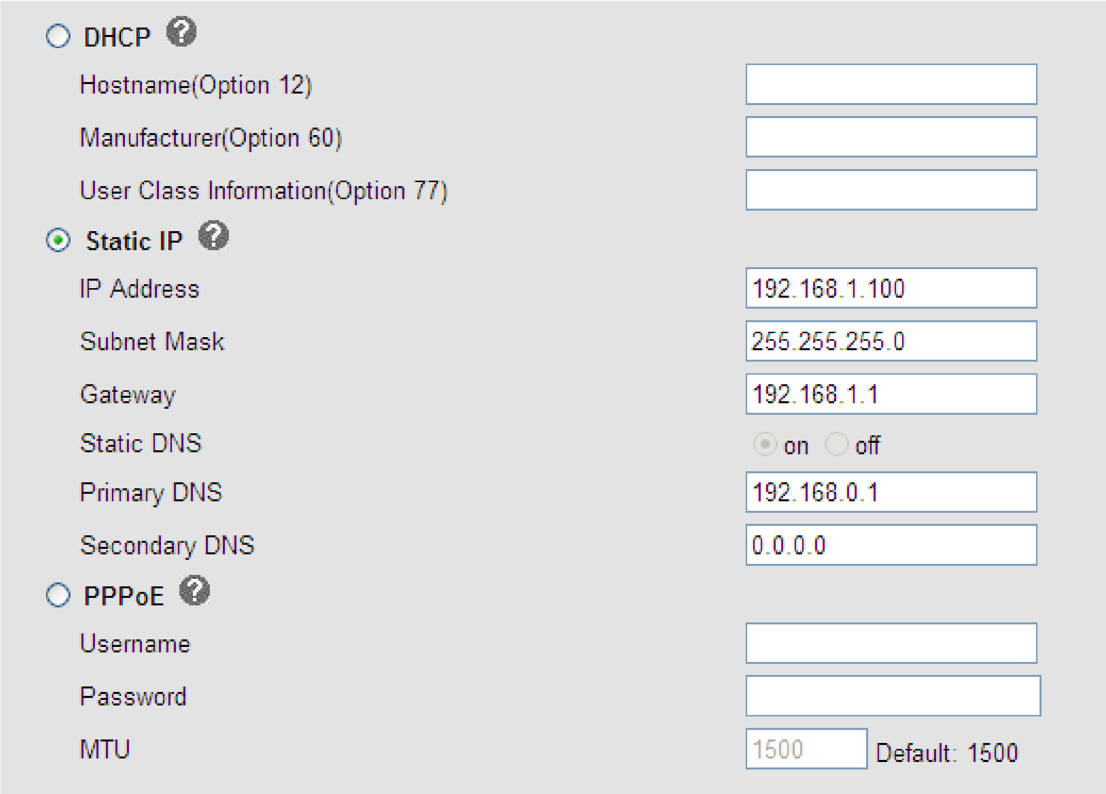
3.2 Phone Status

Here you can see as below information: System Run Time, Register Status, Network Status, System Information,



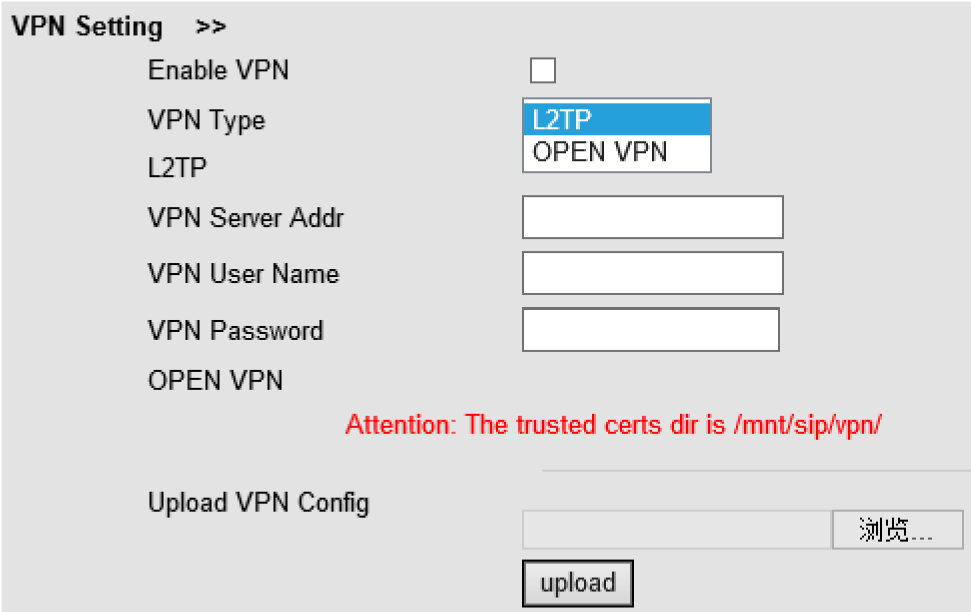
3.3 Network

3.3.1 LAN Port Configuration



ITEM	DESCRIPTION
Network Connection Mode	Network Connection Mode has DHCP, Static IP, PPPoE.
DNS SETTINGS	Select the DNS mode that you want.

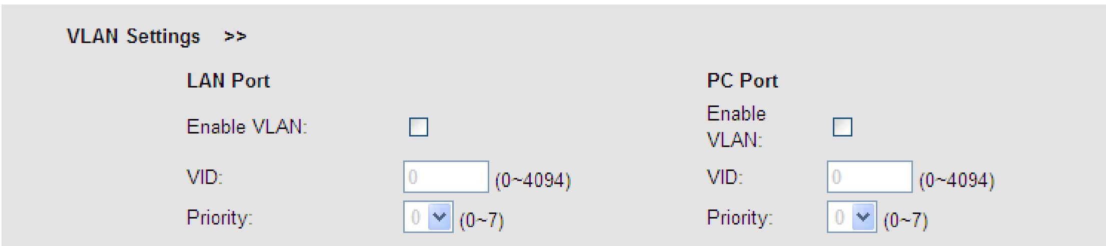
3.3. 2 VPN Settings



When using VPN Setting option, you can set several parameters as follow:

VPN Setting	
Enable VPN	You can enable/disable VPN for phone and pc.
VPN Type:	Choose the appropriate type of VPN.
VPN Server Addr	VPN server's IP.
VPN User Name	VPN user's name
VPN Password	A password be used for authentication
OPEN VPN	Upload the *.ovpn file to the phone

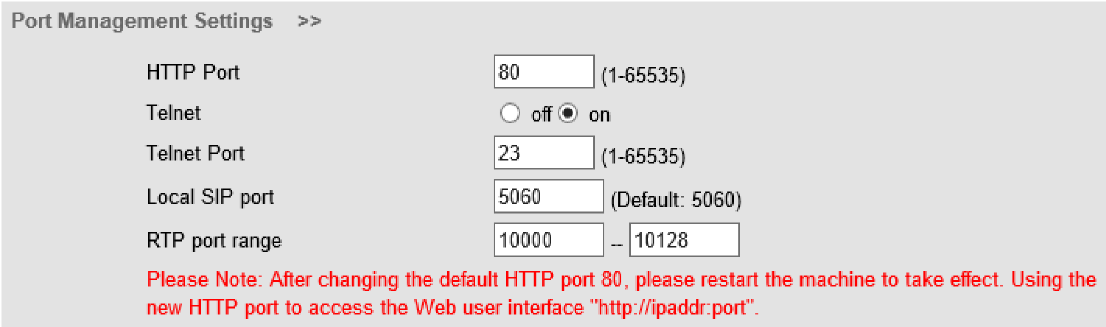
3.3. 3 VLAN Settings



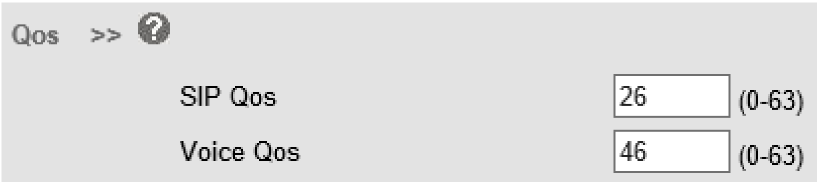
When using VLAN Setting option, you can set several parameters as follow:

VLAN Setting	
Enable VLAN	You can enable/disable VLAN for phone and pc
VID [LAN/PC Port]	The VLAN ID you want the phone or pc to join

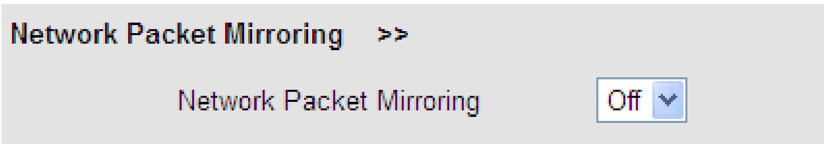
3.3. 4 Port management Settings



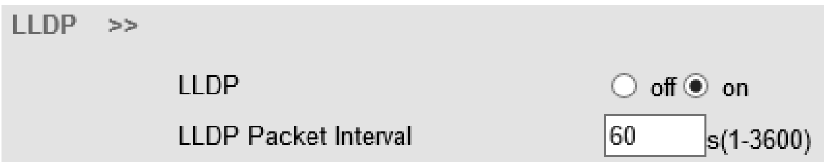
3.3. 5 QoS



3.3. 6 Network Packet Mirroring



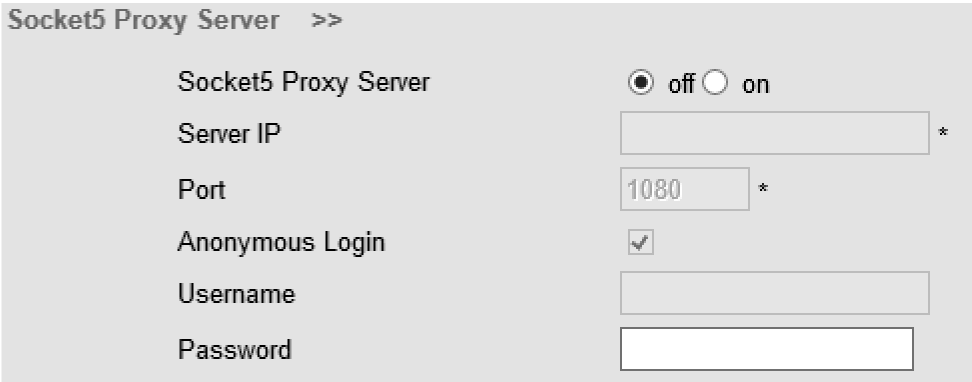
3.3. 7 LLDP



3.3. 8 Paging Settings

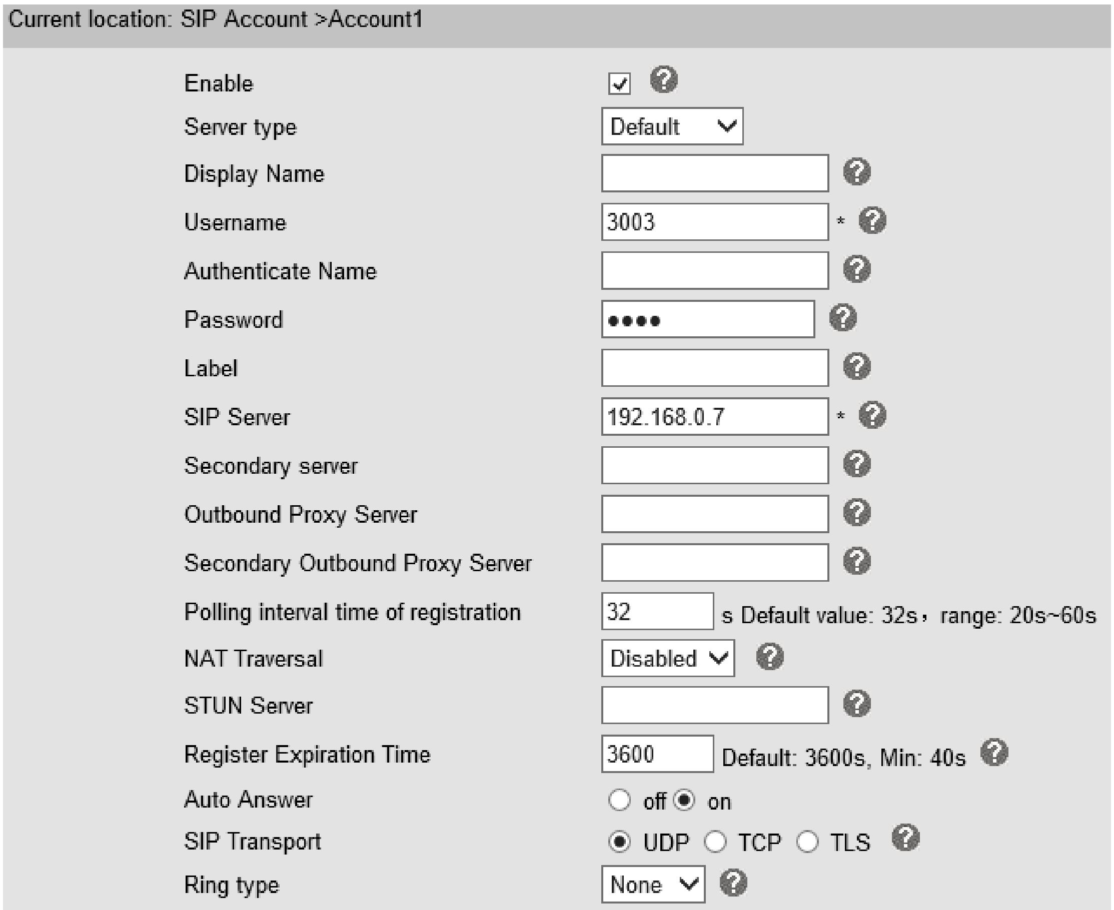
Paging Settings (NOTE: This feature priority is followed the serial number, In other words, "paging1" is the highest priority)

3.3.9 Socket5 Proxy Server



Socket5 Proxy Server	
Socket5 Proxy Server	Enable/Disable Socket5 Proxy Server.
Server IP	Socket5 Proxy Server IP address.
Port	Socket5 Proxy Server port, default is 1080.
Anonymous Login	Enable/Disable Socket5 Proxy Server login userame.

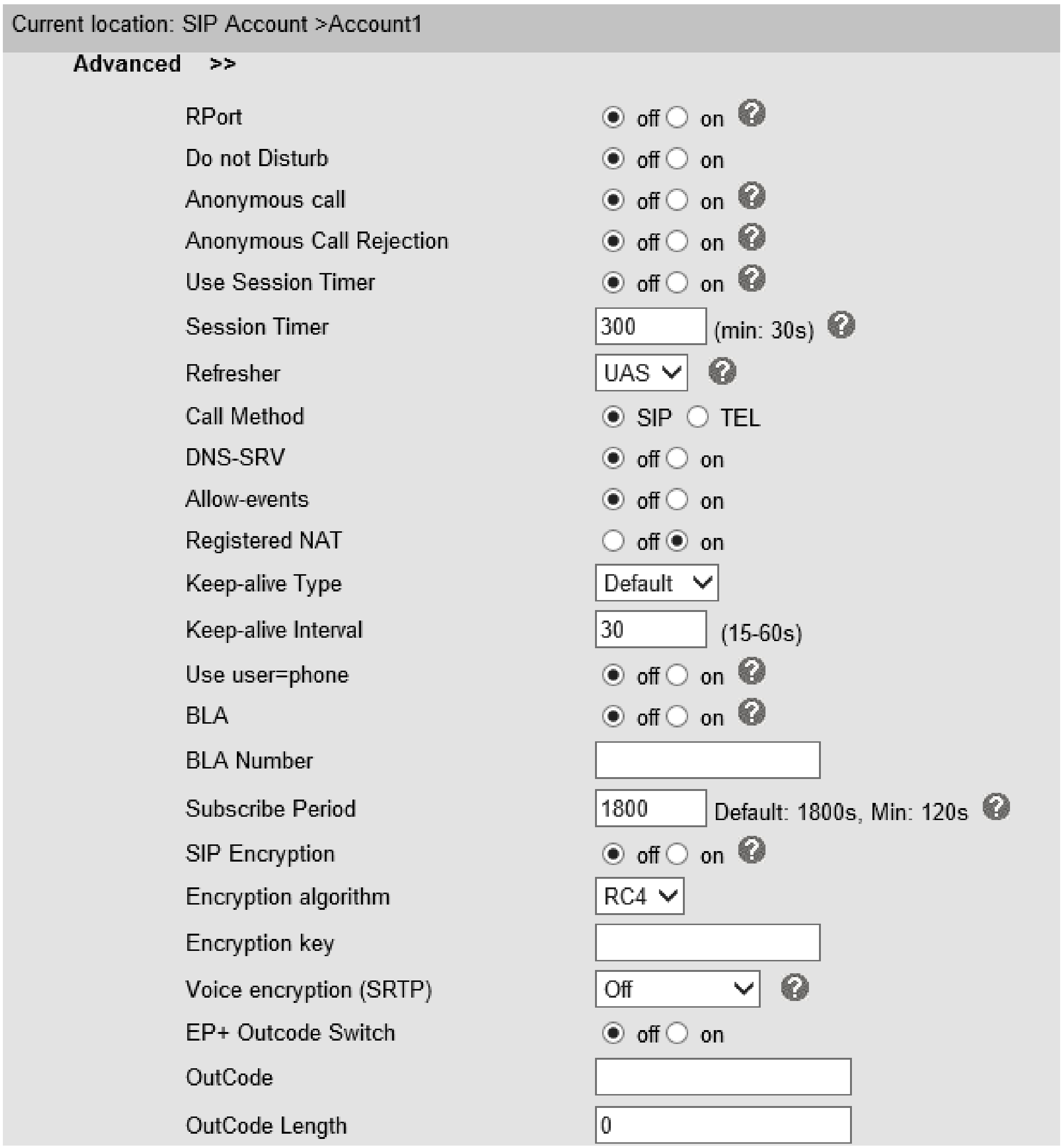
3.4 SIP Accounts



Choose one Account, you will find the following parameters:

ITEM	DECSRIPTION
Enable	You can choose on/off to enable/disable the line.
Account Mode	You can choose VOIP/PSTN, but this model nonsupport PSTN, If you need, Pls contact us to buy another model that can supports PSTN.
Display Name	It is showed as Caller ID when making a phone call
Username	It is a username provided by SIP Server
Authenticate Name	It is authenticated ID for authentication
Password	It is a password provided by SIP Server
Label	Label with this account.
SIP Server	Server for registration, provided by administrator
Secondary server	When the main server can't work, it also can register in this secondary server.
Outbound Proxy Server	Put into the address with the outbound proxy server.
Secondary Outbound Proxy Server	When the main out bound server can't work, it also can use this secondary server.

Poling Interval Time Of Registration	Poling Interval Time Of Registration, default is 32 s.
NAT Traversal	Defines the STUN server will be active or not
STUN Server	Session traversal utilities for NAT.
BLA	Share with the line.
BLA Number	BLA Number
Subscribe Period	Subscribe expire time.
Register Expire Time	IP phone automatically registered every time
SIP Transport	There are UDP/TCP/TLS three options
Ring Type	Select this account ringing type.

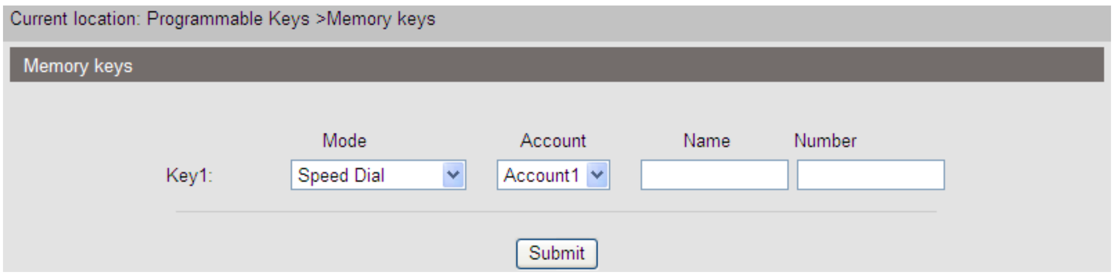


ITEM	DECSRIPTION

Call	
Do Not Disturb	Enable/Disable Do Not Disturb
Anonymous Call	Enable/Disable anonymous call.
Anonymous Call Rejection	Enable/Disable anonymous call rejection.
Use Session Timer	Enable/Disable refresh session function. The device will send an Invite packet to refresh the session during a call if it enable.
Session Timer	The refresh session time interval.
Call Method	This method include SIP and TEL.
DNS-SRV	Enable/Disable DNS-SRV.
Allow-events	Enable/Disable Allow-events.
Registered NAT	Enable/Disable Registered to NAT
UDP Keep-alive Message	The phone periodically sends a UDP packet to keep the port active and to avoid the server to shut down the port
UDP Keep-alive Interval	Default is 30 second.

ITEM	DECSRIPTION
Security	
SIP Encryption	Enable/Disable SIP encryption.
RTP Encryption	Enable/Disable RTP encryption.
Encryption Algorithm	The encryption algorithm at this time we only have RC4.
Encryption Key	The key with encryption.

3.5 Programmable Keys



ITEMS	DESCRIBES
Line	The default value.
Speed Dial	You can use this key feature to speed up dialing the numbers often used or hard to remember.
Speed Dial Prefix	You can use this key feature to speed up dial a call with a specified prefix number.
DTMF	You can use this key feature to send the specification of arbitrary key sequences via DTMF.

BLF	You can use the BLF feature to monitor a specific user for status changes on the phone.
Paging	You can use multicast paging to quickly and easily forward time sensitive announcements out to people within the multicast group.
Call Park	You can use call park feature to place a call on hold, and then retrieve the call from another phone in the system (for example, a phone in another office or conference room).
Intercom	You can press the configured intercom key to automatically connect with a remote extension for outgoing intercom calls, and the remote extension will automatically answer the incoming intercom calls
BLA	This feature such as the BLF.

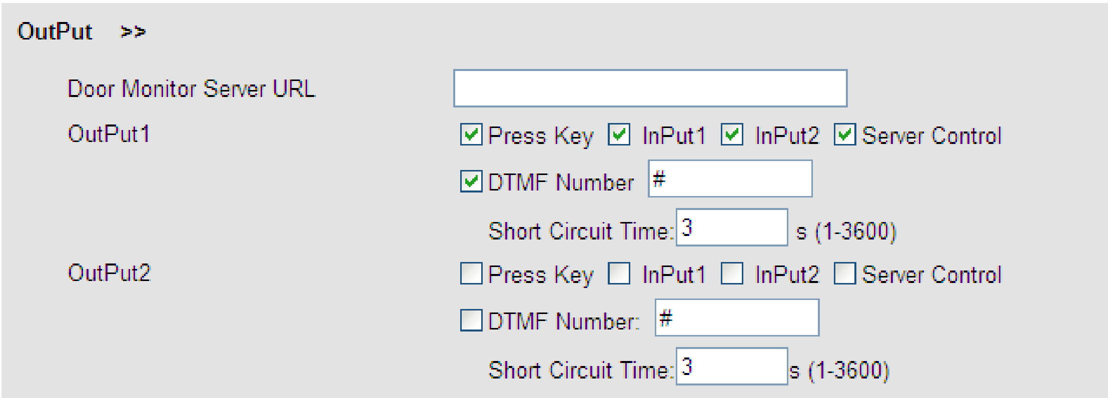
NOTE: ONLY WHEN YOU CHOOSE "SPEED DIAL", THE RIGHT OF "NAME","NUMBER" WILL TAKE EFFECT.

3.6 Phone Settings

3.6.1 Output

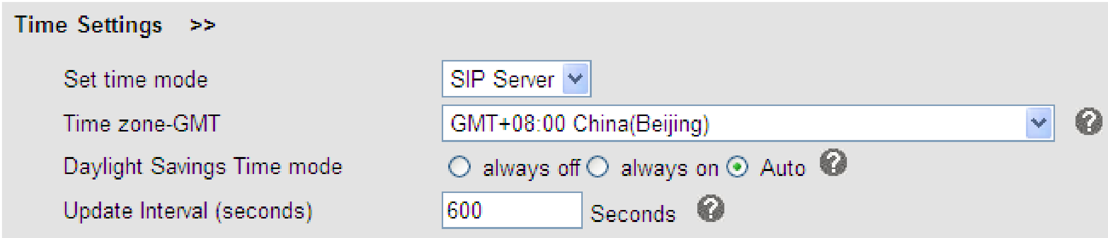
Signal output 1&2 is controlled by several variables. Among them, Server control is a custom variable for some specified platforms. Once custom option is selected, Output variable will be activating accordingly.

Note: Both 'signal input' and 'signal output' are on/off switch



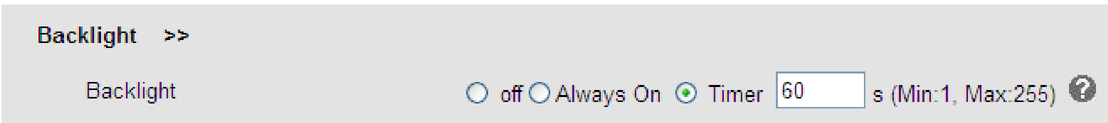
Output1	Variables for output1
output2	Variables for output2
Press key	Press the dial button to trigger the relay.
Input1/2	Shortcut the input1 logic to trigger the relay. See the diagram
Server Control	Use API command to trigger the relay. Ask us for dev manual please.
DTMF Number	Pressing DTMF key to trigger the relay when the phone talking.
Short Circuit time	The relay circuit timer.

3.6.2 Time Settings



ITEM	DECSRIPTION
Time Settings	
Set Time Mode	Include SNTP/SIP Server/PSTN/Manual
SNTP Server	You can select in the list or input owner server address.
Update Interval	The update interval with SNTP.
Day Light Saving Time	Enable/disable the DST for the phone
Time Format	You can use 24 hour time format or 12 hour time format
Date Format	You can choose the appropriate time format.
Time Zone-GMT	You can select different time zone for the phone
Manual Setting	Setting time manually.

3.6.3 Backlight



3.6.4 Ring tone

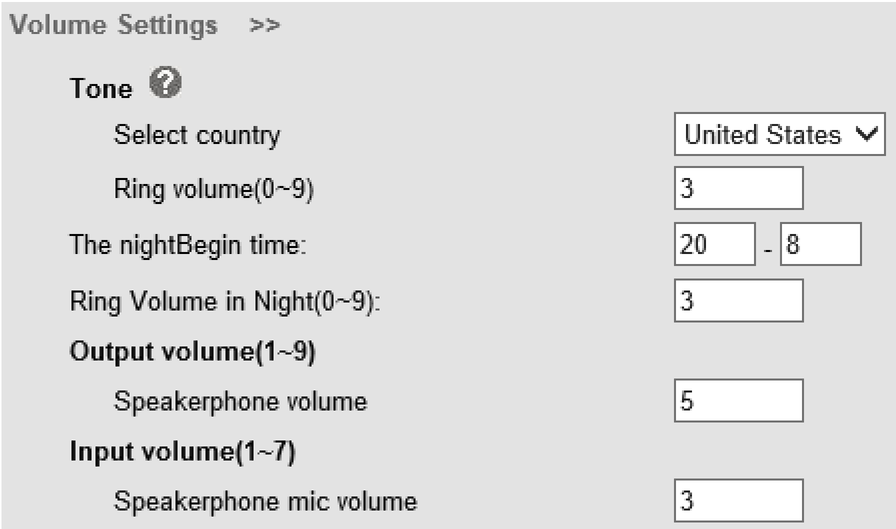
Ring1 is for the speaker on the panel, Ring2 is for external speaker

Note： Only Support a ring tone with G711A (*.wav) audio coding, maximum is 10 rings and the total size must be less than 150kB.

3.6.5 Volume Setting

You can manage the volume level and mic level as below form.

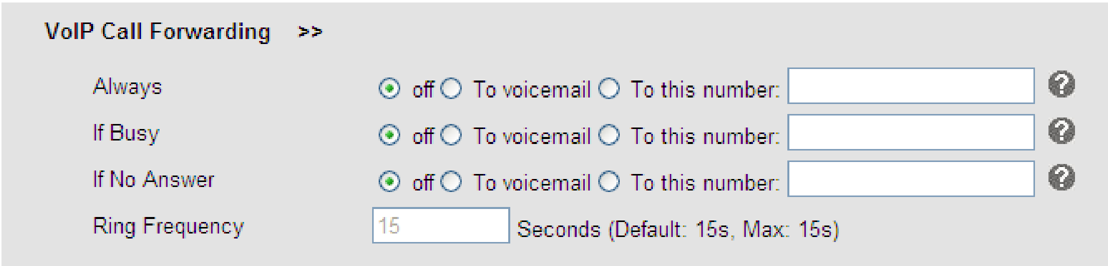
Note: Normally if the mic is on level 7. Please keep the volume below level 4.Unless the using area is small or you have good ability of noise reduction.



ITEM	DECSRIPTION
Basic	
Select Country	Select the country dial tone. Default is United States.
Ring Volume	The ring volume default is Lv3, the range is 0~9.
Handset Volume	The handset volume default is Lv5, the range is 1~9.
Speaker Phone Volume	The speaker volume default is Lv5, the range is 1~9.
Headset Volume	The headset volume default is Lv3, the range is 1~9.
Handset MIC Volume	The handset MIC volume default is Lv3, the range is 1~7.
Speaker Phone MIC Volume	The speaker MIC volume default is Lv3, the range is 1~7
Headset MIC Volume	The headset MIC volume default is Lv3, the range is 1~7

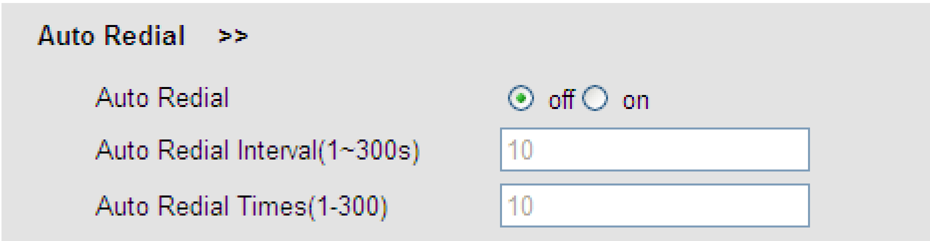
3.7 Features

3.7.1 VoIP Call Forward

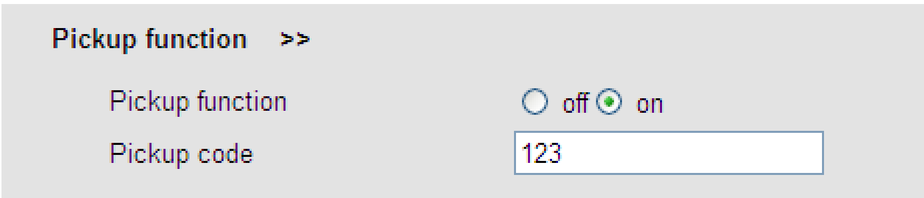


ITEM	DECSRIPTION
Always	All ways transfer the call to others.
If Busy	If the phone was busy working, the call will be transfer to others.
If No Answer	If the phone was no answer, the call will be transfer to others.
Ring Frequency	The ring frequency with the VOIP Call Forward.

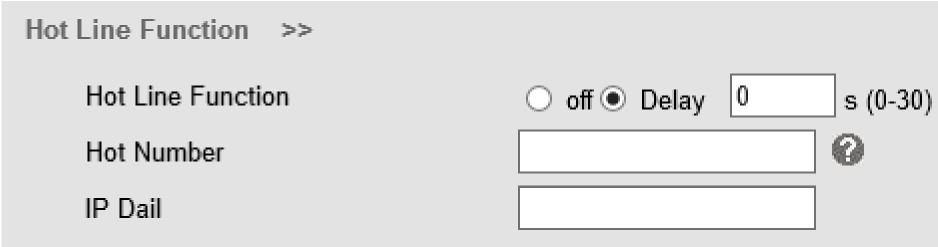
3.7.2 Auto Redial



3.7.3 Pickup function

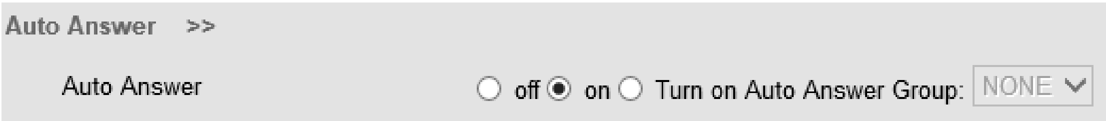


3.7.4 Hotline function



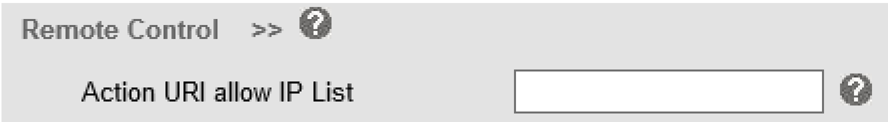
3.7.5 Auto Answer

Default value is on, Values can be changed accordingly。



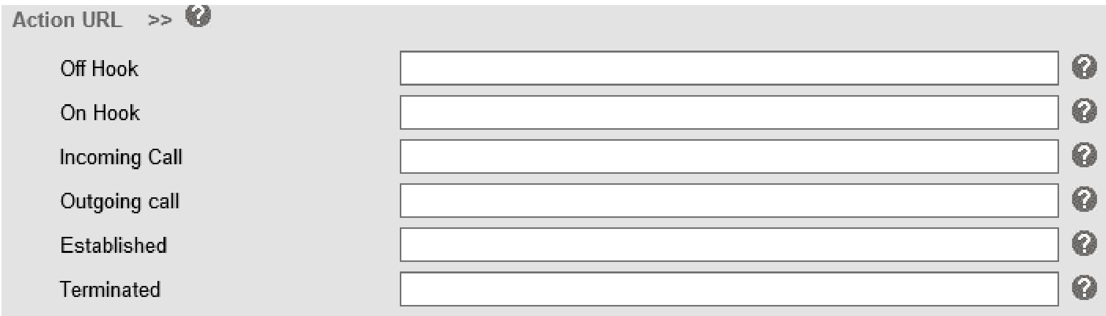
3.7.6 Remote Control

A Third party is permitted to control this device



3.7.7 Action URL

The device will send orders to action URL initiative。



3.7.8 EP+

EP+ options are for the users who download the EP+ application on mobile phone. After Completing below settings, EP+ will be activated. For more details, please refer to www.escene.cn/en/en

EP+ >>

Configure Mode	◉ Automatic ○ Manual
EP+	○ off ◉ on
Password	7394
OutCode	
OutCode Length	0
Door bell Code	
Open Door Password	
Roaming Server Address	: 0

3.7.9 Other features settings

For other features such as Call waiting, DTMF etc.

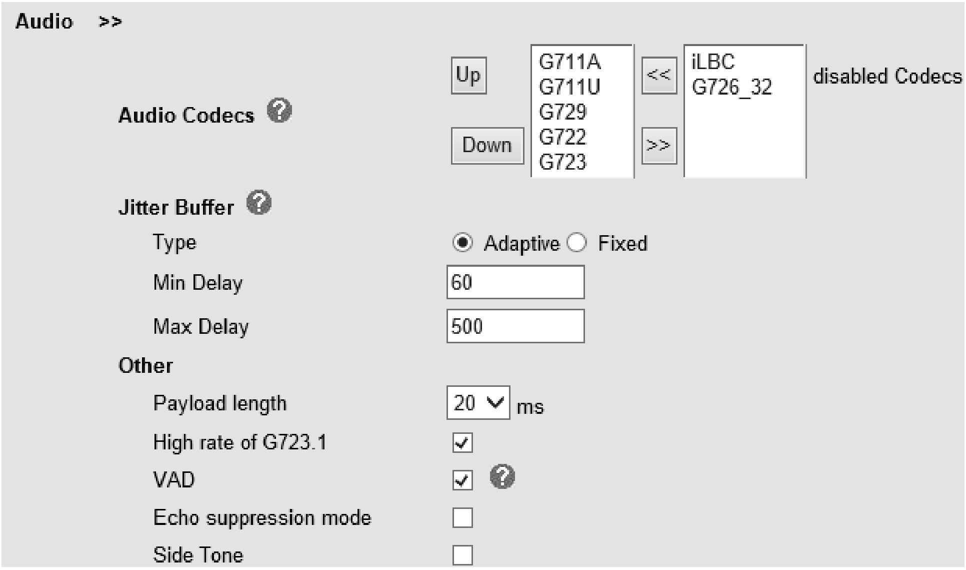
Other Features Settings >>

Call Waiting	○ off ◉ on
Call Waiting Tone	○ off ◉ Play on currently active device Frequency: 10 s (5-60)
Play Hold Tone	○ off ◉ Play on currently active device Frequency: 30 s (5-60)
DTMF	◉ RFC 2833 ○ Inband ○ SIP Info ○ Auto
Suppress DTMF Display	◉ off ○ on
100 Reliable retransmission	○ off ◉ on
Play Hangup Tone	○ off ◉ on
Conference Code	◉ off ○ on Number:
Hold Code	◉ off ○ on Number:
Conference exit result	◉ Disconnect all ○ Others remain connected
Return code when refused	603(Decline)
Return code when DnD	603(Decline)
Called No Answer Time	☑ 70 s (Min:20, Max:1800)
Caller No AnswerTime:	☑ 180 s (Min: 90s, Max: 1800s)
RFC 2833 PayLoad	101
Caller ID source	FROM
SIP Session Timer(seconds) T1	0.5
SIP Session Timer(seconds) T2	4
SIP Session Timer(seconds) T4	5
Affiliated Port	○ off ◉ on

3.8 Advanced

3.8.1 Audio

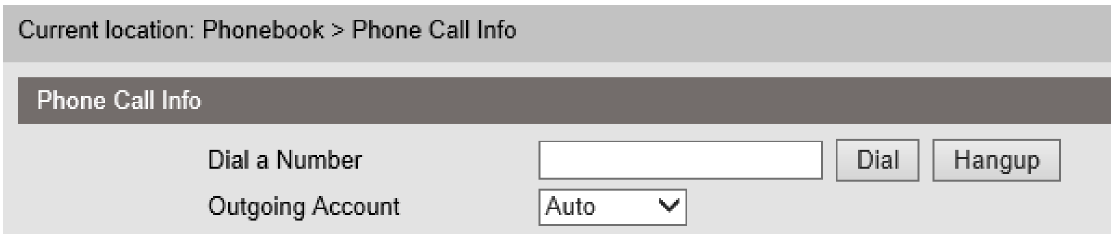
For Audio Codecs setting and Jitter Buffer setting.



3.8.2 Dial Plan

3.9 Phone call info

To call or hang up the phone via web

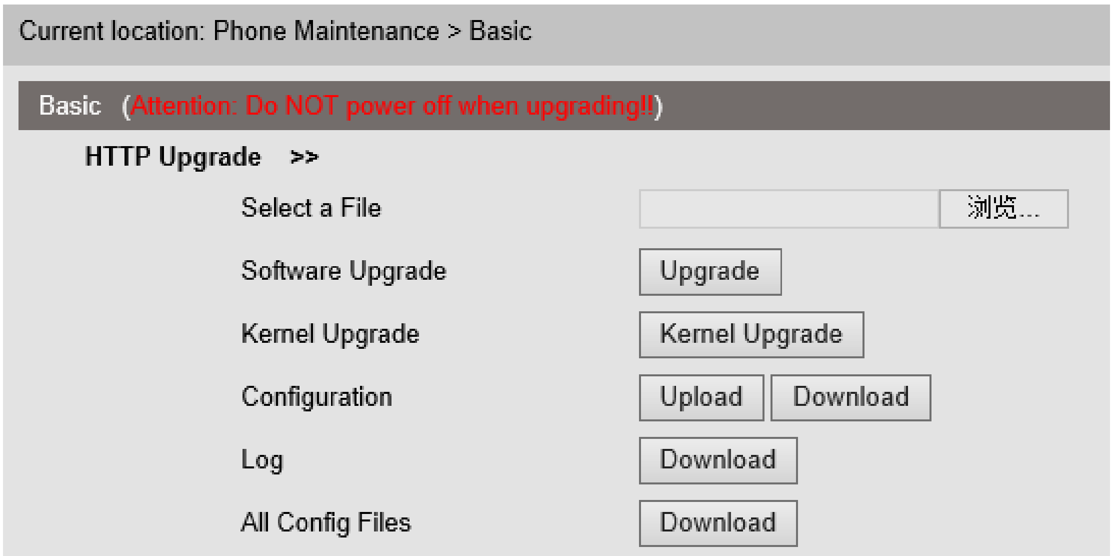


3.10 Maintenance

This part mainly introduces some maintenance method. According to the below, you can reconfigure Intercom IP Phone or view Intercom IP Phone log to gain more information about maintenance.

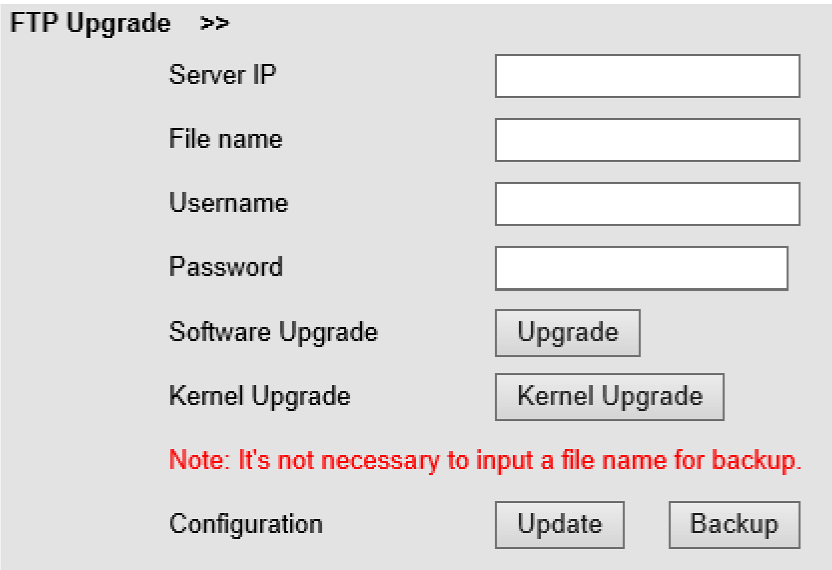
3.10.1 HTTP Upgrade

You can upgrade the software, kernel and configuration etc. filesby HTTP.



3.10.2 FTP Upgrade

You can upgrade the software, kernel and configure filesbyFTP.

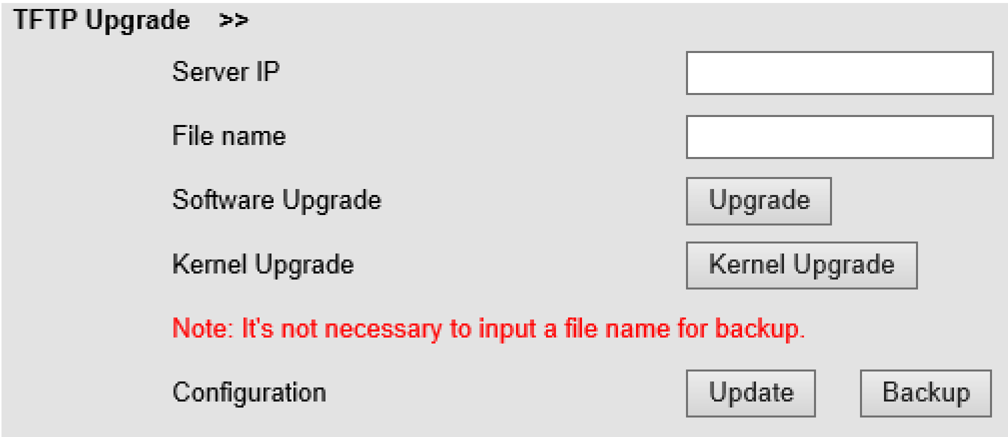


When using FTP upgrade, you can set several parameters as follow:

FTP Upgrade	
Server IP	The IP address of the FTP server
Filename	Downloading from FTP server
Username	Providing by FTP server
Password	Providing by FTP server
Software Upgrade	Used for upgrading the software of the phone
Kernel Upgrade	Used for upgrading the kernel of the phone
Configuration	Used for updating/backup to update/backup the configure file of the phone
Phone Book	Used for updating/backup to update/backup the phonebook of the phone
EXT Module	Used for updating/backup the expansion of the phone **[NOTES：The mode doesn't support this feature]**

3.10.3 TFTP Upgrade

You can upgrade the software, kernel and configure files by TFTP.

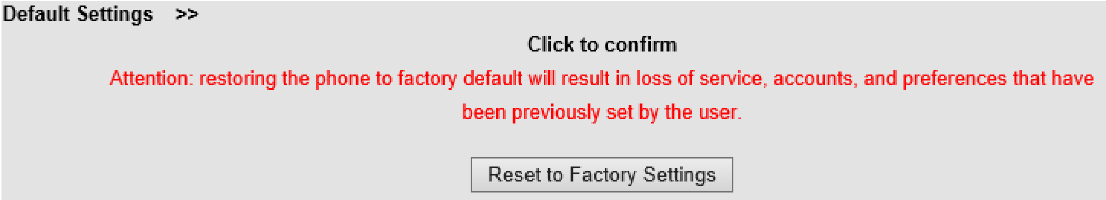


TFTP Upgrade	
Server IP	The IP address of the TFTP server
Filename	Downloading from FTP server
Software Upgrade	Used for upgrading the software of the phone
Kernel Upgrade	Used for upgrading the kernel of the phone
Configuration	Used for updating/backup the configure file of the phone
Phone Book	Used for updating/backup the phonebook of the phone
EXT Module	Used for updating/backup the expansion of the phone **[NOTES：The mode doesn't support this feature]**

NOTES: It's not necessary to input filename when doing backup Configuration, Phone Book, EXT Module.

3.10.4 Factory reset

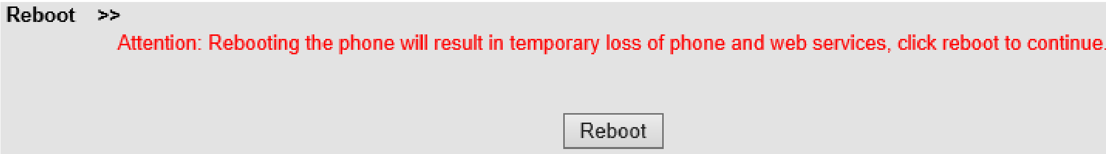
You can load the phone to the factory default setting in default setting option.



Press the 'Reset to Factory Setting' option, the phone will load to factory default setting on next reboot.

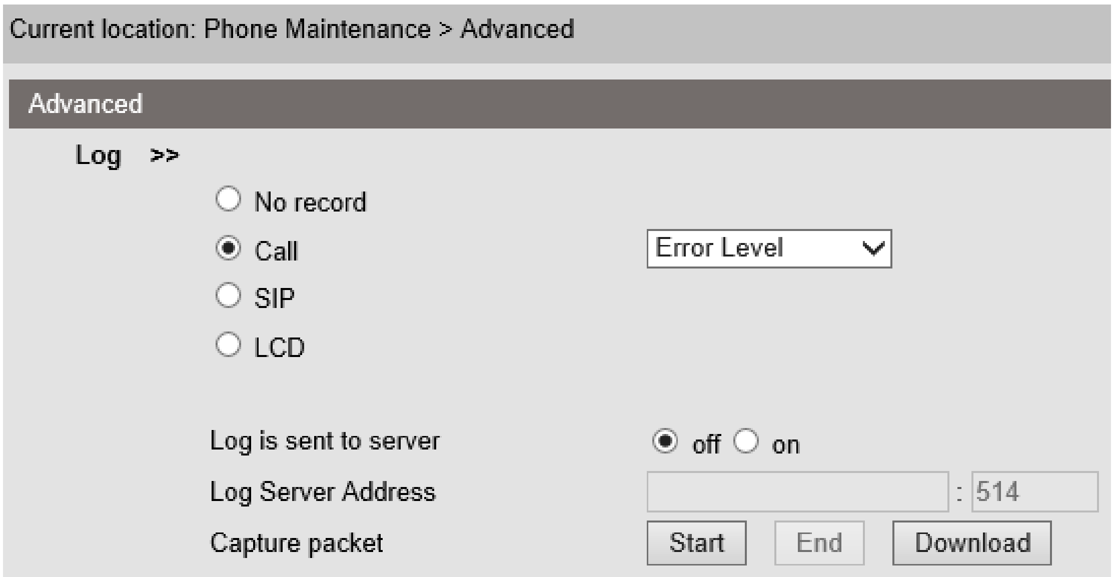
3.10.5 Reboot

You can use reboot option to reboot the phone.



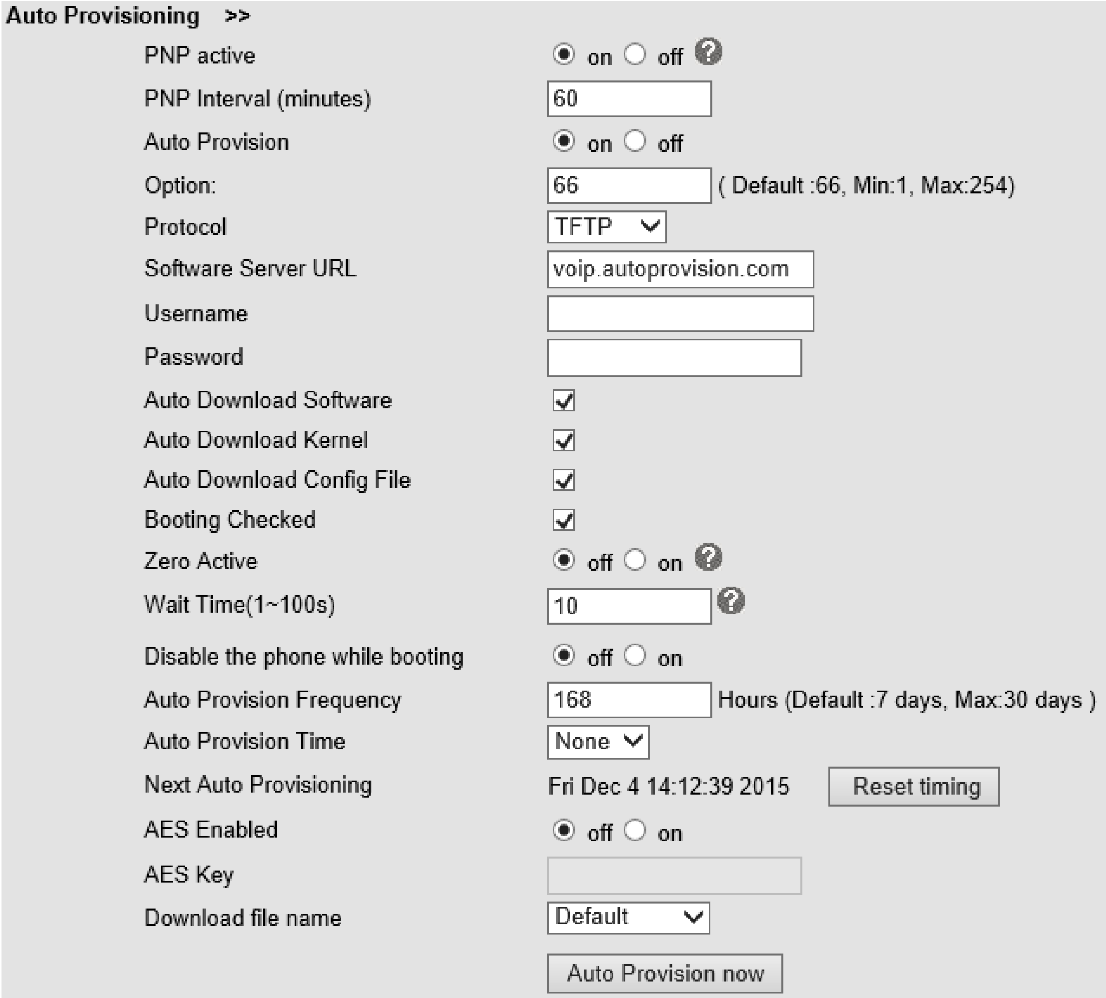
3.10.6 Log

This feature is use for the administrator to managing the equipment, like debugging, SIP etc,. If you need to catch a debugging Level, you need to setup on this interface.



3.10.7 Auto Provision

When you open this auto provision feature, the phone will do auto provision after it detect a different software or kernel (Higher or Lower) which are putted on the TFTP,HTTP,HTTPS,FTP, server. For the detailed information about auto provision, you can find it in the official website: HTTP://www.escene.cn/en



When using auto provision, you can set several parameters as follow:

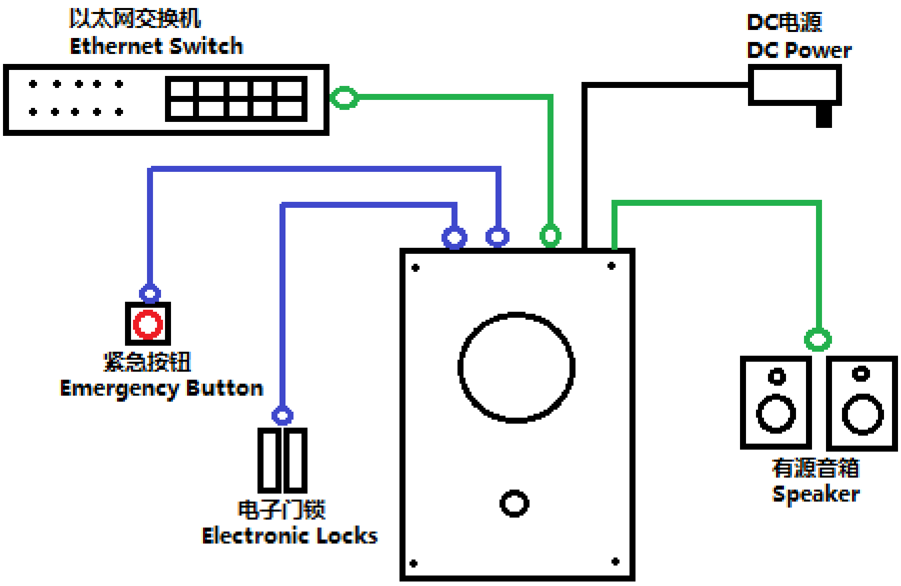
Auto Provision	
Auto Provision	You can enable/disable auto provision by select on/off
Protocol	Used for auto provision, it includes TFTP/HTTP/FTP
Software Server URL	The server address of the auto provision
Username	Providing by provision server
Password	Providing by provision server
Auto Download Software	Used for auto download software from server
Auto Download Kernel	Used for auto download kernel from server
Auto Download Config File	Used for auto download config file from server
Auto Download Expansion	NOTES: The model doesn't support this feature.
Auto Download Enterprise Phonebook	Used for auto download Enterprise Phonebook from server
Auto Download Personal Phonebook	Used for auto download personal phonebook from server

Booting Checked	Used for checking the auto provision when phone booting
Disable the phone while booting checking	Enable/Disable the booting checking feature.
Auto Provision Frequency	Used for setting the time interval for auto provision
Auto Provision Time	Used for the specific time for auto provision
Auto Provision Next Time	Reset the Auto Provision Next Upgrading time.
AES Enable	You can enable/disable AES encrypt for auto provision
AES Key	The key of the AES
Auto Provision Now	Used for doing auto provision immediately

4. Brief pictures for application environment

The following pictures introduce the practical application of IP intercom.Take door security and fire protection for example. More compatibility application is subject to actual test.

4.1 Door security system application



4.2 Fire protection system application