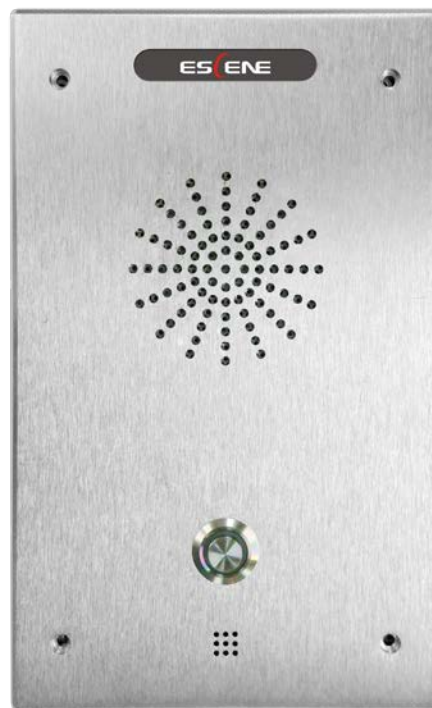




Smart Security IP Intercom IS710 User Manual



IS710-01

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About this manual

Thank you for choosing Smart Security IP Intercom IS710. 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 IS710. 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 IP Intercom IS710, 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>.

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1. Getting Started

1.1 Outline

ESCENE IP Intercom Phone IS710 is the newest 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.





IS710 is dust proof, water proof and dismantle prevention, having a fashion appearance and high protection. IS710 is a bond of door opening by long-distance DTMF. Its powerful performance, stability and reasonable price make it a perfect choice of industry user.

NOTE: IS710-01, support POE, IS is intercom series, 01 is 1 DSS key.

1.2 Product Features

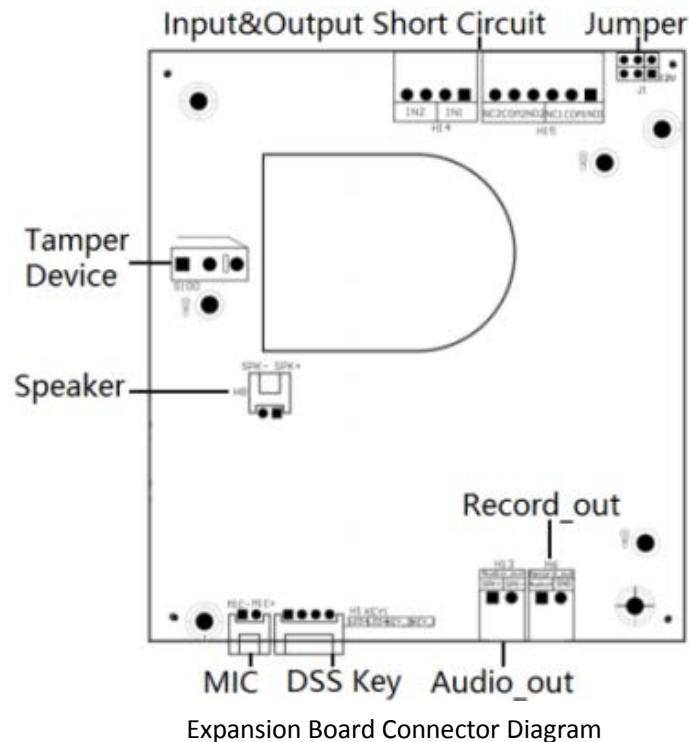
- Support embedded or wall-mounted installation.
- High-fidelity sound quality, HD codec, Full duplex hands-free calls;
- 2*RJ45 standard Ethernet Ports, integrated PoE;
- Built-in speaker and high sensitive microphone, support hand-free calls and receive emergency broadcast
- User-defined DSS key, which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

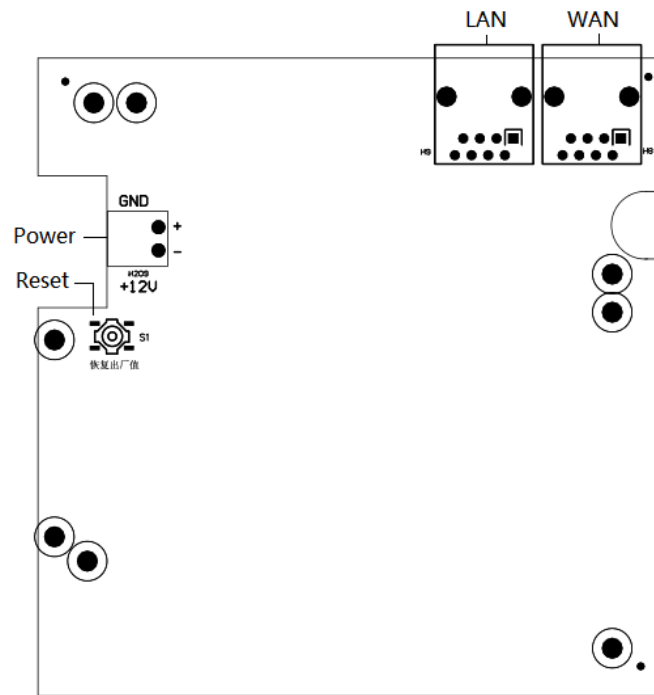
Intercom features
WEB support Multi-Language ;1 SIP account; Hotline; Passive support Call hold, Call waiting, Call forward, Call transfer (blind/busy/ask), Mute, DND, Auto-answer, 3-way conferencing; 1 DSS programmable key(Speed dial, Intercom etc.);Volume control; Direct IP call without SIP proxy; 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;
Physical properties
1 DSS programmable key(Speed dial, Intercom etc.) 1 LED light, 1 light touch button(Remote factory value) 2 RJ45 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: 190x115x41mm
Product Certification
   
Platform Compatibility Test (non-certificate)
ZTE/Alcatel-Lucent/Asterisk/Broadsoft/Metaswitch/Yeastar/Avaya/3CX/Elastix/HUAWEI etc.

1.3 Technical Information

INTERFACE SPECIFICATION:





Mother Board Connector Diagram

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

2. Intercom Installation

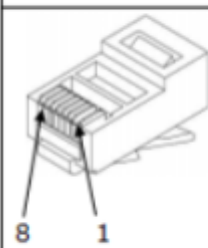








Generally system administrator will connect your new IS710 IP Intercom to company LAN network. If not, please refer to below illustration.

Open IS710 packing box, according to the packing list, check the related attachment to make sure to no omitting. Packing list as follows.

- IS710 Intercom
- Quick operating guide
- 8*Screws
- 2*Rubbers

IS710 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

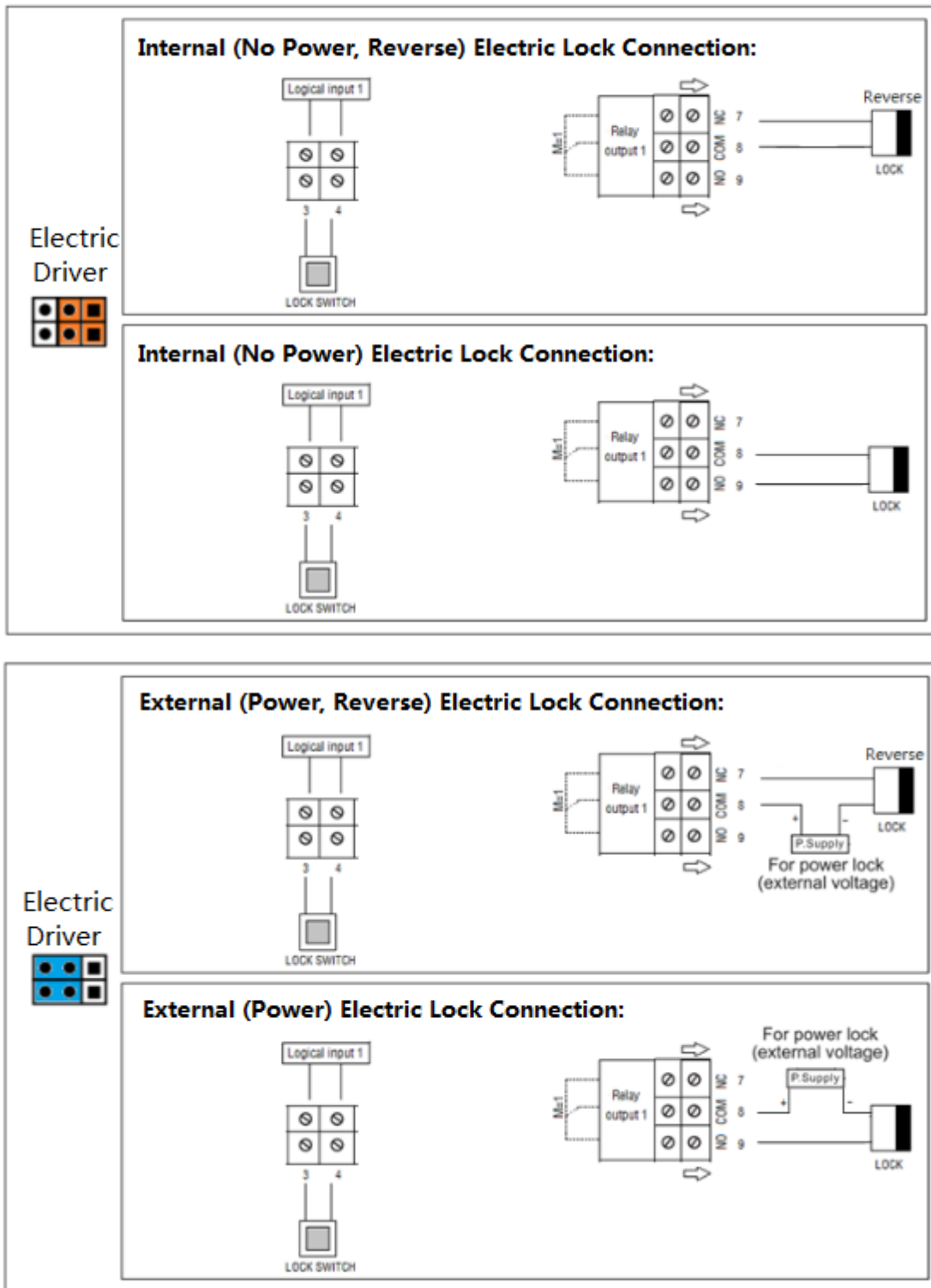
RJ-45				LAN	PC
	Pin No.	Marking	Colour		
	1	Tx+		TX1+	RX2-
	2	Tx -		TX1-	RX2+
	3	Rx+		RX1+	TX2-
	4	PoE+		POE+	Null
	5	PoE+			
	6	Rx -		RX1-	TX2+
	7	PoE-		POE-	Null
	8	PoE-			

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

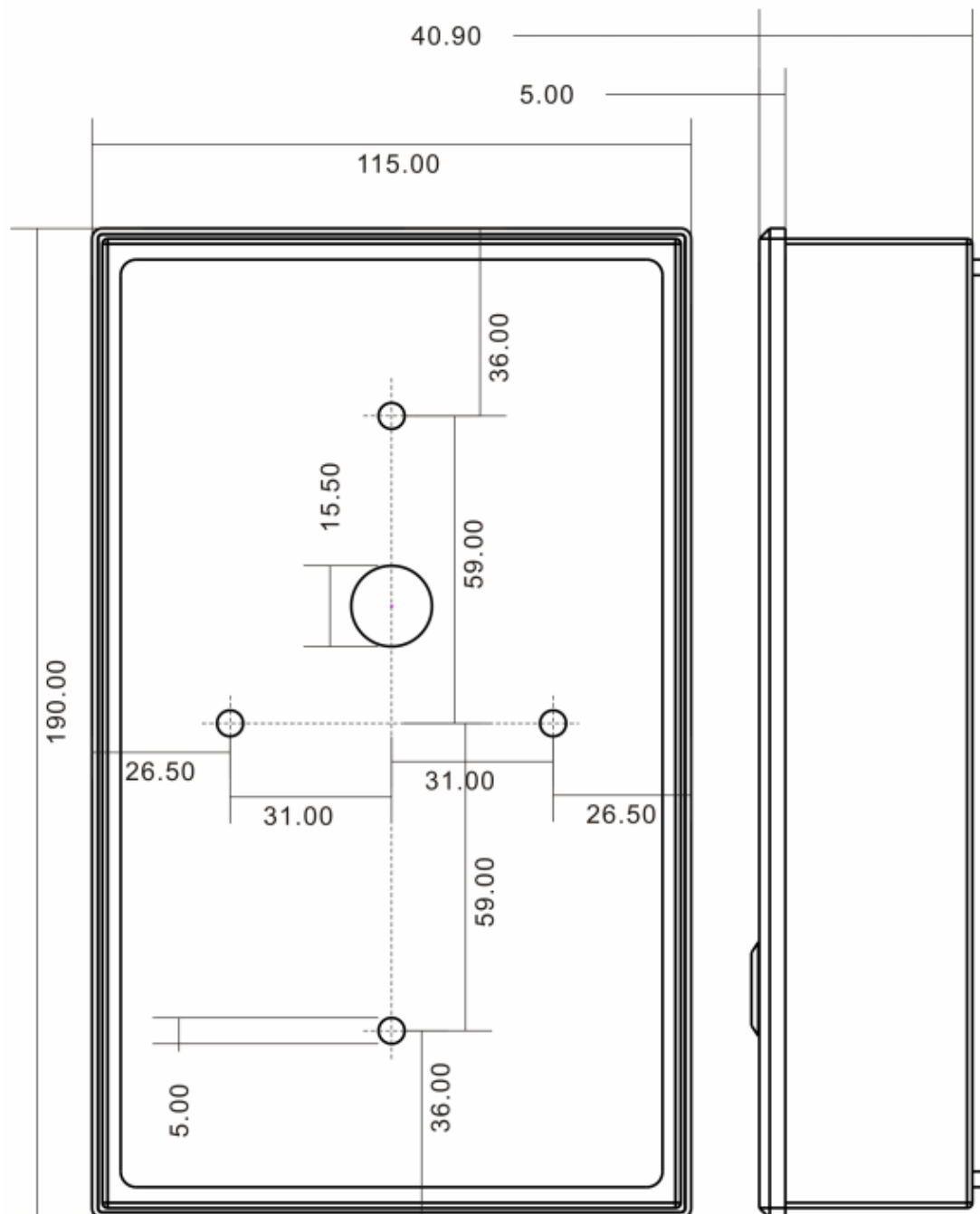


Internal/External Electric Lock Connection

2.1 Embedded

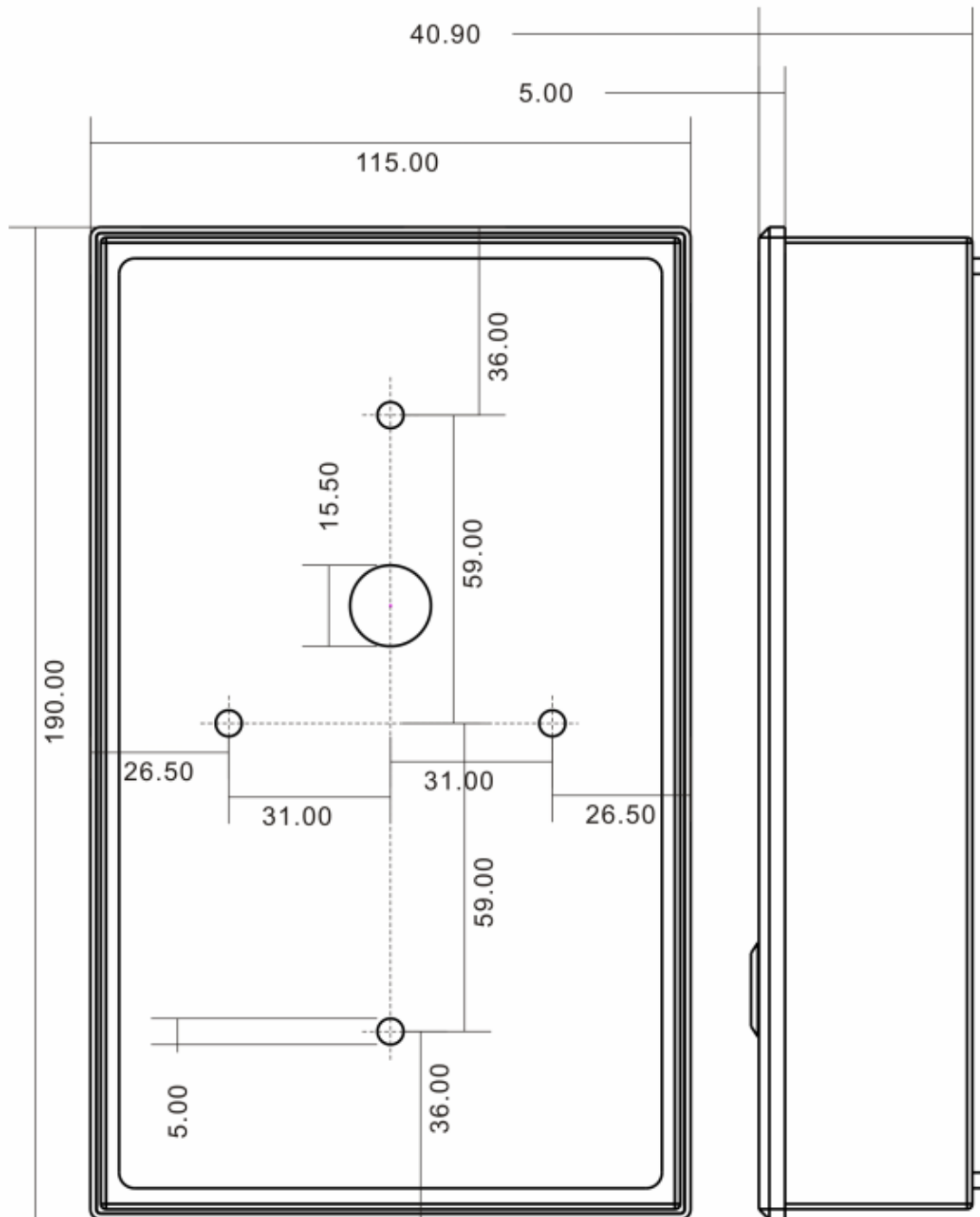
If the product is used for embedding, then the cutting of embedded need to be a little

bigger than the installing hole of standard dimension 190*115*41mm .(41mm means the extra aluminum housing thickness.),as follows. Notice: the embedded value is up to the actual situation.



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.)

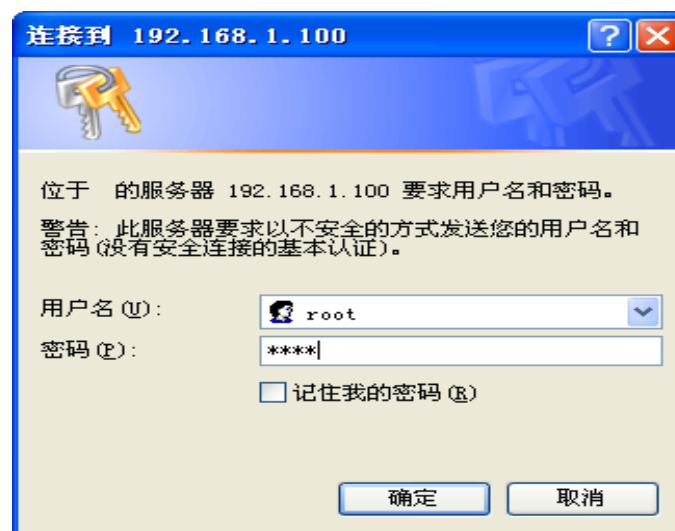
3.1 Remote WEB Management

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. As below is for example.

Intercom Settings

Phone Status

System Run Time 0 Days18 Hours40 Minutes21 Seconds

Register status ?

Account1	None
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/IV740-04
Software Version	V0.1.15.0118_Alpha(1305)
Hardware version	V2.x.x
Kernel Version	v1.0.0

Network

DHCP ?


Hostname(Option 12)	<input type="text"/>
Manufacturer(Option 60)	<input type="text"/>
User Class Information (Option 77)	<input type="text"/>

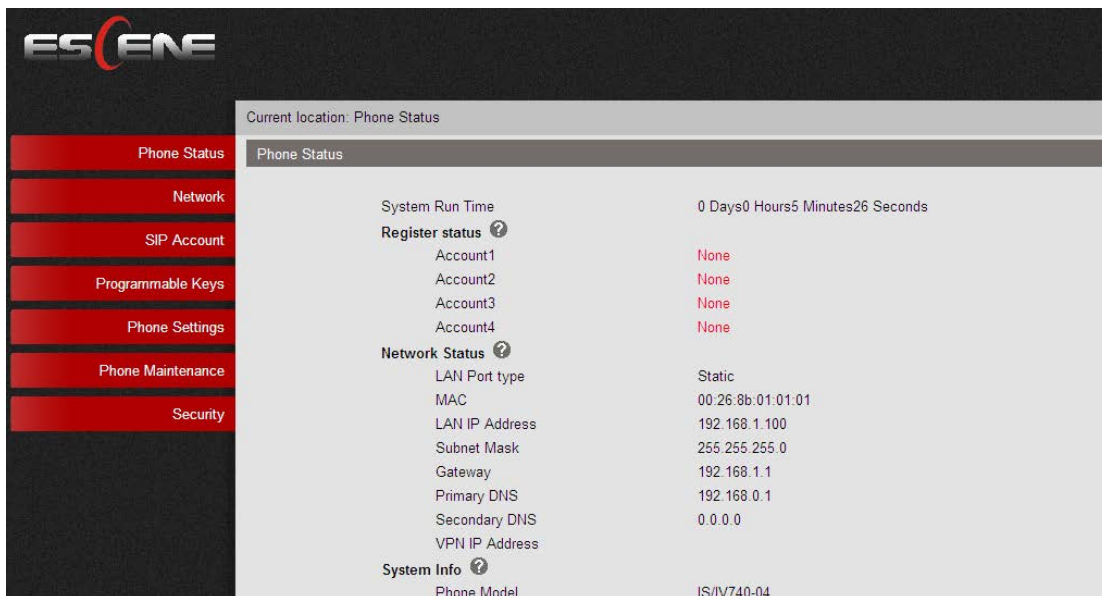
Static IP ?

IP Address	<input type="text" value="192.168.1.100"/>
Subnet Mask	<input type="text" value="255.255.255.0"/>
Gateway	<input type="text" value="192.168.1.1"/>
Static DNS	<input checked="" type="radio"/> on <input type="radio"/> off
Primary DNS	<input type="text" value="192.168.0.1"/>
Secondary DNS	<input type="text" value="0.0.0.0"/>
HTTP Port	<input type="text" value="80"/> (1-65535)

SIP Account	
Enable	<input type="checkbox"/> ?
Display Name	<input type="text"/> ?
Username	<input type="text"/> * ?
Password	<input type="text"/> ?
SIP Server	<input type="text"/> * ?
Polling interval time of registration	<input type="text" value="32"/> s Default value: 32s, range: 20s~60s
Register Expiration Time	<input type="text" value="3600"/> Default: 3600s, Min: 40s ?
Phone Settings	
Door Monitor Server URL	<input type="text"/>
OutPut1	<input checked="" type="checkbox"/> Press Key <input checked="" type="checkbox"/> InPut1 <input checked="" type="checkbox"/> InPut2 <input checked="" type="checkbox"/> Server Control <input checked="" type="checkbox"/> DTMF Number: # <input type="text"/> <input checked="" type="checkbox"/> DoorCard <input checked="" type="checkbox"/> Touch key Open Door Number: <input type="text"/> Short Circuit Time: <input type="text" value="3"/> s (1-3600)
OutPut2	<input type="checkbox"/> Press Key <input type="checkbox"/> InPut1 <input type="checkbox"/> InPut2 <input type="checkbox"/> Server Control <input type="checkbox"/> DTMF Number: # <input type="text"/> <input type="checkbox"/> DoorCard <input type="checkbox"/> Touch key Open Door Number: <input type="text"/> Short Circuit Time: <input type="text" value="3"/> s (1-3600)
Speakerphone volume (1~9)	<input type="text" value="6"/>
Speakerphone mic volume(1~7)	<input type="text" value="5"/>
Hot Number	<input type="text"/> ?
Auto Answer	<input type="radio"/> off <input checked="" type="radio"/> on
Phone Maintenance	
Select a File	<input type="text"/> 浏览...
Software Upgrade	<input type="button" value="Upgrade"/>
Configuration	<input type="button" value="Upload"/> <input type="button" value="Download"/>
Default Settings	<input type="button" value="Reset to Factory Settings"/>
Reboot	<input type="button" value="Reboot"/>
<input type="button" value="Submit"/>	

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

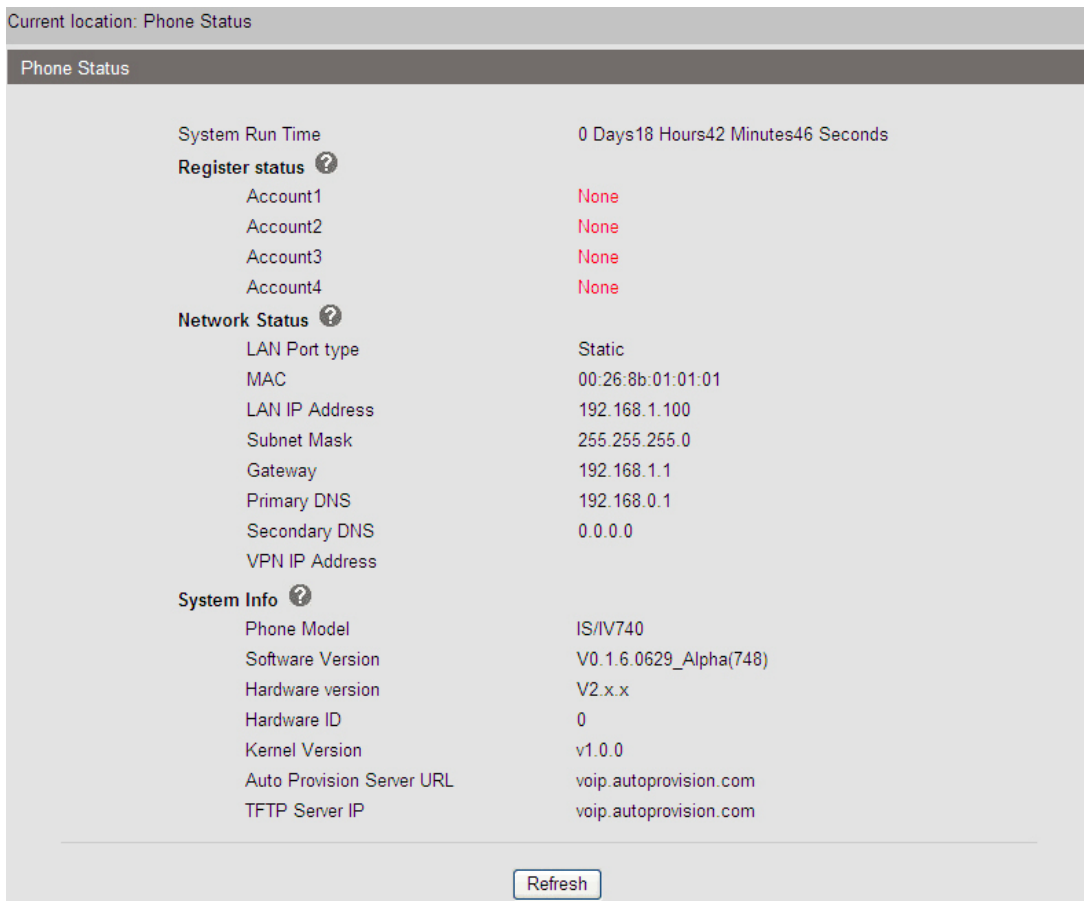
 <http://192.168.1.100/home.asp>



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

DHCP ?
 Hostname(Option 12)
 Manufacturer(Option 60)
 User Class Information(Option 77)

Static IP ?
 IP Address
 Subnet Mask
 Gateway
 Static DNS on off
 Primary DNS
 Secondary DNS

PPPoE ?
 Username
 Password
 MTU Default: 1500

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

VPN Setting >>

Enable VPN

VPN Type L2TP
OPEN VPN

L2TP

VPN Server Addr

VPN User Name

VPN Password

OPEN VPN

Attention: The trusted certs dir is /mnt/sip/vpn/

Upload VPN Config 浏览...

upload

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

VLAN Settings >>

<p>LAN Port</p> <p>Enable VLAN: <input type="checkbox"/></p> <p>VID: <input type="text" value="0"/> (0~4094)</p> <p>Priority: <input type="text" value="0"/> (0~7)</p>	<p>PC Port</p> <p>Enable VLAN: <input type="checkbox"/></p> <p>VID: <input type="text" value="0"/> (0~4094)</p> <p>Priority: <input type="text" value="0"/> (0~7)</p>
---	--

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

Port Management Settings >>

HTTP Port	<input type="text" value="80"/> (1-65535)
Telnet	<input type="radio"/> off <input checked="" type="radio"/> on
Telnet Port	<input type="text" value="23"/> (1-65535)
Local SIP port	<input type="text" value="5060"/> (Default: 5060)
RTP port range	<input type="text" value="10000"/> -- <input type="text" value="10128"/>

Please Note: After changing the default HTTP port 80, please restart the machine to take effect. Using the new HTTP port to access the Web user interface "http://ipaddr:port".

3.3. 5 QoS

Qos >> ?

SIP Qos	<input type="text" value="26"/> (0-63)
Voice Qos	<input type="text" value="46"/> (0-63)

3.3. 6 Network Packet Mirroring

Network Packet Mirroring >>

Network Packet Mirroring ▾

3.3. 7 LLDP

LLDP >>

LLDP	<input type="radio"/> off <input checked="" type="radio"/> on
LLDP Packet Interval	<input type="text" value="60"/> s(1-3600)

3.3. 8 Paging Settings

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

Paging Setting >>

Paging1 off on
 Group IP Port:

Paging2 off on
 Group IP Port:

Paging3 off on
 Group IP Port:

Paging4 off on
 Group IP Port:

Paging5 off on
 Group IP Port:

3.3.9 Socket5 Proxy Server

Socket5 Proxy Server >>

Socket5 Proxy Server off on
 Server IP *
 Port *
 Anonymous Login
 Username
 Password

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 username.

3.4 SIP Accounts

Current location: SIP Account >Account1

Enable	<input checked="" type="checkbox"/> ?
Server type	Default ▾
Display Name	<input type="text"/> ?
Username	3003 * ?
Authenticate Name	<input type="text"/> ?
Password	•••• ?
Label	<input type="text"/> ?
SIP Server	192.168.0.7 * ?
Secondary server	<input type="text"/> ?
Outbound Proxy Server	<input type="text"/> ?
Secondary Outbound Proxy Server	<input type="text"/> ?
Polling interval time of registration	32 s Default value: 32s, range: 20s~60s
NAT Traversal	Disabled ▾ ?
STUN Server	<input type="text"/> ?
Register Expiration Time	3600 Default: 3600s, Min: 40s ?
Auto Answer	<input type="radio"/> off <input checked="" type="radio"/> on
SIP Transport	<input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS ?
Ring type	None ▾ ?

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.

Current location: SIP Account >Account1

Advanced >>

RPort off on ?

Do not Disturb off on

Anonymous call off on ?

Anonymous Call Rejection off on ?

Use Session Timer off on ?

Session Timer (min: 30s) ?

Refresher ?

Call Method SIP TEL

DNS-SRV off on

Allow-events off on

Registered NAT off on

Keep-alive Type ?

Keep-alive Interval (15-60s)

Use user=phone off on ?

BLA off on ?

BLA Number

Subscribe Period Default: 1800s, Min: 120s ?

SIP Encryption off on ?

Encryption algorithm

Encryption key

Voice encryption (SRTP) ?

EP+ Outcode Switch off on

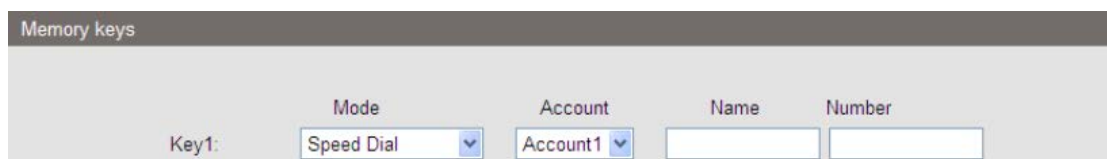
OutCode

OutCode Length

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
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

OutPut >>

Door Monitor Server URL

OutPut1 Press Key InPut1 InPut2 Server Control

DTMF Number #

Short Circuit Time: s (1-3600)

OutPut2 Press Key InPut1 InPut2 Server Control

DTMF Number: #

Short Circuit Time: s (1-3600)

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

Time Settings >>

Set time mode SIP Server ▾

Time zone-GMT GMT+08:00 China(Beijing) ▾ ?

Daylight Savings Time mode always off always on Auto ?

Update Interval (seconds) 600 Seconds ?

ITEM	DESCRIPTION
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 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.

Ring >> ?

Ring type Ring1 ▾ Delete

Ring type2 Ring1 ▾

Upload ring tone
 浏览...

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

3.6.4 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.

Volume Settings >>

Tone ?

Select country ▾

Ring volume(0~9)

The nightBegin time: -

Ring Volume in Night(0~9):

Output volume(1~9)

Speakerphone volume

Input volume(1~7)

Speakerphone mic volume

ITEM	DESCRIPTION
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

VoIP Call Forwarding >>

Always off To voicemail To this number: ?

If Busy off To voicemail To this number: ?

If No Answer off To voicemail To this number: ?

Ring Frequency Seconds (Default: 15s, Max: 15s)

ITEM	DESCRIPTION
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

Auto Redial >>

Auto Redial off on

Auto Redial Interval(1~300s)

Auto Redial Times(1-300)

3.7.3 Pickup function

Pickup function >>

Pickup function off on

Pickup code

3.7.4 Hotline function

Hot Line Function >>

Hot Line Function off Delay s (0-30)

Hot Number ?

IP Dail

3.7.5 Auto Answer

Default value is on, Values can be changed accordingly.

Auto Answer >>

Auto Answer off on Turn on Auto Answer Group:

3.7.6 Remote Control

A Third party is permitted to control this device

Remote Control >> ?

Action URI allow IP List ?

3.7.7 Action URL

The device will send orders to action URL initiative.

Action URL >> ?	
Off Hook	<input type="text"/> ?
On Hook	<input type="text"/> ?
Incoming Call	<input type="text"/> ?
Outgoing call	<input type="text"/> ?
Established	<input type="text"/> ?
Terminated	<input type="text"/> ?

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	<input checked="" type="radio"/> Automatic <input type="radio"/> Manual
EP+	<input type="radio"/> off <input checked="" type="radio"/> on
Password	<input type="text" value="7394"/>
OutCode	<input type="text"/>
OutCode Length	<input type="text" value="0"/>
Door bell Code	<input type="text"/>
Open Door Password	<input type="text"/>
Roaming Server Address	<input type="text"/> : <input type="text" value="0"/>

3.7.9 Other features settings

For other features such as call waiting, DTMF etc.

Other Features Settings >>

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

3.8 Advanced

3.8.1 Audio

For Audio Codecs setting and Jitter Buffer setting.

Audio >>

Audio Codecs ?	<input type="button" value="Up"/> <table border="1" style="display: inline-table; vertical-align: middle;"> <tr><td>G711A</td></tr> <tr><td>G711U</td></tr> <tr><td>G729</td></tr> <tr><td>G722</td></tr> <tr><td>G723</td></tr> </table> <input type="button" value="Down"/> <table border="1" style="display: inline-table; vertical-align: middle; margin-left: 10px;"> <tr><td><<</td></tr> <tr><td>iLBC</td></tr> <tr><td>G726_32</td></tr> <tr><td>>></td></tr> </table> disabled Codecs	G711A	G711U	G729	G722	G723	<<	iLBC	G726_32	>>
G711A										
G711U										
G729										
G722										
G723										
<<										
iLBC										
G726_32										
>>										
Jitter Buffer ?	<input checked="" type="radio"/> Adaptive <input type="radio"/> Fixed Type Min Delay <input type="text" value="60"/> Max Delay <input type="text" value="500"/>									
Other	Payload length <input type="text" value="20"/> ms High rate of G723.1 <input checked="" type="checkbox"/> VAD <input checked="" type="checkbox"/> ? Echo suppression mode <input type="checkbox"/> Side Tone <input type="checkbox"/>									

3.8.2 Dial Plan

Dial Plan >>

Send key * #

Dial length (1~32)

No Dial timeout (1~14s)

ID	Operation	Prefix	IP Address	Account	Description
----	-----------	--------	------------	---------	-------------

3.9 Phone call info

To call or hang up the phone via web

Current location: Phonebook > Phone Call Info

Phone Call Info

Dial a Number

Outgoing Account ▼

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. files by HTTP.

Current location: Phone Maintenance > Basic

Basic (Attention: Do NOT power off when upgrading!)

HTTP Upgrade >>

Select a File

Software Upgrade

Kernel Upgrade

Configuration

Log

All Config Files

3.10.2 FTP Upgrade

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

FTP Upgrade >>

Server IP

File name

Username

Password

Software Upgrade

Kernel Upgrade

Note: It's not necessary to input a file name for backup.

Configuration

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

File name

Software Upgrade

Kernel Upgrade

Note: It's not necessary to input a file name for backup.

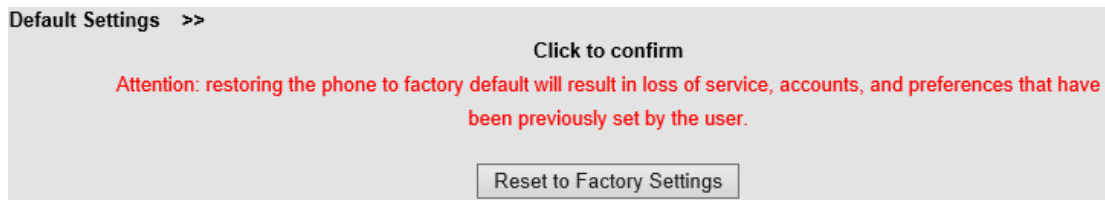
Configuration

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

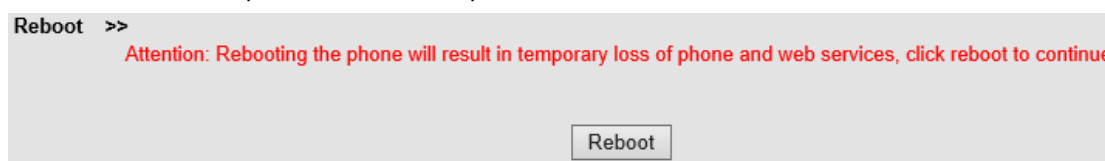
You can load the phone to the factory default setting in default setting option.



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.

Current location: Phone Maintenance > Advanced

Advanced

Log >>

No record

Call

SIP

LCD

Error Level

Log is sent to server off on

Log Server Address :

Capture packet

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](http://www.escene.cn/en)

Auto Provisioning >>

PNP active on off ?

PNP Interval (minutes)

Auto Provision on off

Option: (Default :66, Min:1, Max:254)

Protocol ▼

Software Server URL

Username

Password

Auto Download Software

Auto Download Kernel

Auto Download Config File

Bootling Checked

Zero Active off on ?

Wait Time(1~100s) ?

Disable the phone while booting off on

Auto Provision Frequency Hours (Default :7 days, Max:30 days)

Auto Provision Time ▼

Next Auto Provisioning

AES Enabled off on

AES Key

Download file name ▼

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

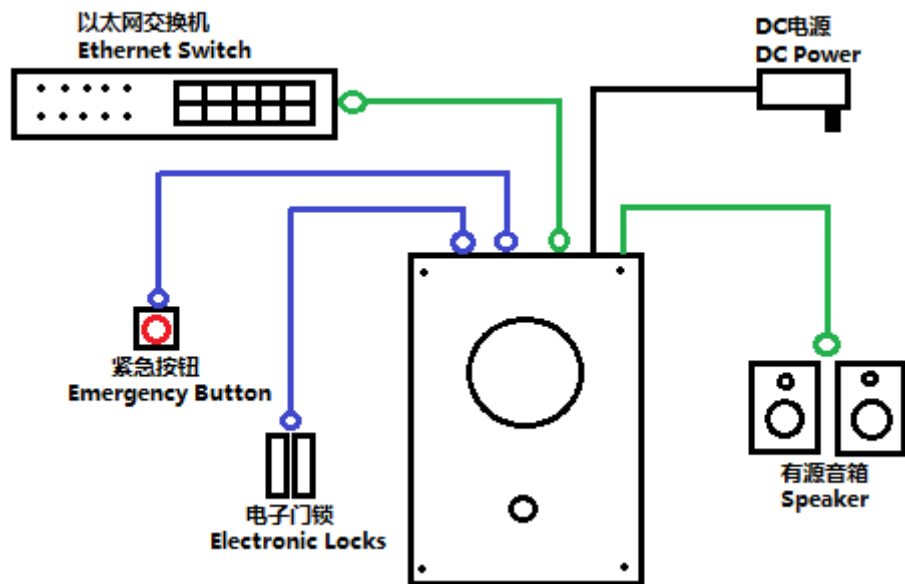
Auto Provision	
Auto Provision	You can enable/disable auto provision by select on/off
Protocol	Used for auto provision, it includes TFTP/HTTP/FTP
Software Server URL	The server address of the auto provision
Username	Providing by provision server
Password	Providing by provision server
Auto Download Software	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
Bootling 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

