

*Public Broadcasting IP Phone*



*SayHi*<sup>TM</sup>

**Public Broadcasting IP Phone User Manual  
PS760-P(W)**



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Escene Communication Co.,Ltd

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

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Escene Communication

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## About this manual

Thank you for choosing Public Broadcasting IP Phone PS760-P(W).This IP Phone is specially designed for the user under the public environment with fashionable appearance and complete functions. This manual aims to help you quickly use IP Phone PS760-P(W).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 phone. If this is your first time to use IP Phone PS760-P(W),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>.

## Tables of Contents

Copyright and disclaimer.....	2
1.Getting Started.....	4
1.1 Outline.....	4
1.2 Product Features.....	4
1.3 Technical Information.....	4
2.Telephone installation.....	5
2.1 Ceiling entrance specification.....	6
2.2 Installation of speaker equipment.....	6
3. Configuration of speaker.....	7
3.1 Login web management of speaker.....	7
3.2 Network.....	8
3.2.1 WIFI Settings.....	8
3.2.2 WIFI Status.....	9
3.2.3 LAN Settings.....	9
3.2.4 VPN Settings.....	10
3.3 SIP Accounts.....	10
3.4 Multicast settings.....	11
3.5 Voice.....	12
3.6 Maintains for the speaker.....	13
3.6.1 Log.....	13
3.6.2 Speaker setup.....	14
3.6.3 VLAN setup.....	16
3.6.4 Password.....	16
3.6.5 Factory Defaults Setting.....	17
3.6.6 Auto provision.....	17
3.6.7 FTP Upgrade.....	18
3.6.8 TFTP Upgrade.....	18
3.6.9 HTTP Upgrade.....	19
3.6.10 Reboot.....	19
3.7 Language.....	20

# 1. Getting Started

## 1.1 Outline

PS760-P (W) Public Broadcasting Wired and Wireless-based IP Phone, a new generation of IP speakers. The Multicast functions settings of the device is more user-friendly than previous devices, the incidental tools and functions of this product will make Multicast easier to manage, notify, expand, and monitor.




The device adopting well-designed and elegant color (white), makes it look more beautiful, more suitable for installation on most of the ceiling without any influence. This device is more suitable for large and small public broadcasting projects e.g. schools, hospitals, stadiums, hotels, shopping malls, commercial buildings, venues, theaters, government buildings.

## 1.2 Product Features

- 3 VoIP (SIP) accounts.
- Support up to 20 groups multicast functions.
- LEDs and function keys embedded design, LEDs reflects the device work status, function keys can be used to adjust the volume, broadcast IP and a key to restore the factory values.
- Built-in microphone and external 3.5mm microphone interface can be switched freely.
- Support remote web management and maintenance, such as account maintenance, software upgrades, the volume control.
- POE power supply (802.3af) which cannot coexist with WIFI series
- WIFI Series is easy to be installed with strong expansion ,its RJ45 port having a dual-use nature can be used as LAN port which is the same with the use of ordinary series when people manually turn off WIFI.

## 1.3 Technical Information

<b>Phone features</b>
WEB Multi-language; Support 3 accounts line Active / passive support; Support calls holding, calls waiting and calls transfer; Call transfer (busy turn, blind turn, consulting turn), mute, don't disturb, Auto answer, three-way conference, the volume control and so on; support IP direct dial without account, Support up to 20 groups multicast functions simultaneously (the priority decreases from 1 to 20)
<b>Network parameters</b>
Support SIP V1 (RFC2543), V2 (RFC3261); Support DNS SRV (RFC3263); Support STUN network penetration; Support 3 DTMF model: In-band, RFC2833, SIP INFO; Support Network model: StaticIP/DHCP/PPPoE; built-in DNS/TFTP/FTP terminal; Support NAT/DHCP service; Support SIP and RTP Qos
<b>Security</b>
Support VLAN (802.1pq), LLDP, VPN (L2TP/OPEN_VPN); Support TLS (Transport Layer Security) protocol; Support information authentication mechanism MD5; Support AES encryption protocol; Support phone locks, support the Root/User level management mode
<b>Audio features</b>
Wideband encoding: G.722; Narrow band encoding: G.711 $\mu$ /A, G723.1, G726, G.729AB, iLBC; Support VAD, CNG, AEC, AGC etc. audio processing; Full duplex speaker, with automatic echo elimination (AEC).
<b>Physical properties</b>
One RJ-45 10/100M Ethernet interfaces(LAN), LAN port is for the use of ordinary model; LAN/PC port is for WS series, when WIFI function opening, it is PC port, when WIFI function closing, it is LAN port; One power adapter; Speaker Output: 13Watts; Power: DC 12V/1A; <b>ABC keys function definitions:</b> A key: volume increase; B key: quickly restore the factory value by pressing the button 20 seconds(refer to soft recovery, not hard recovery), press 1 time means broadcasting the current IP address; C key: volume decrease. <b>ABC-LEDs keys function definitions:</b> A (network) Green slow flash - network connection is fail; The light off - the network connection is normal; B (multicast) The light off - multicast is idle; red flash - multicast is busy; C (SIP Account) The light off - the registration is successful / idle; blue flash - Account busy; blue flashing slowly - registration is fail; Flash is 1s / times, slow flash is 2s / times
<b>Carton packaging</b>
Size: 8.5"x6"; The mask width 215mm, Height 65mm, The cover width x160mm, Height x135mm, Total height: 155mm; Net weight: 1.1kg Gross weight: 2.0kg;
<b>Product Certification</b>

   ISO 9001
<b>Platform Compatibility Test (non-certificate)</b>
ZTE/Alcatel-Lucent/Asterisk/Broadsoft/Metaswitch/Yeastar/Avaya/3CX/Elastix/HUAWEI etc.

## 2.Telephone installation

Generally ,the system administrator will connect your new PS760-P (W) phone to the corporate IP telephony network. If not, please refer to the following instructions.

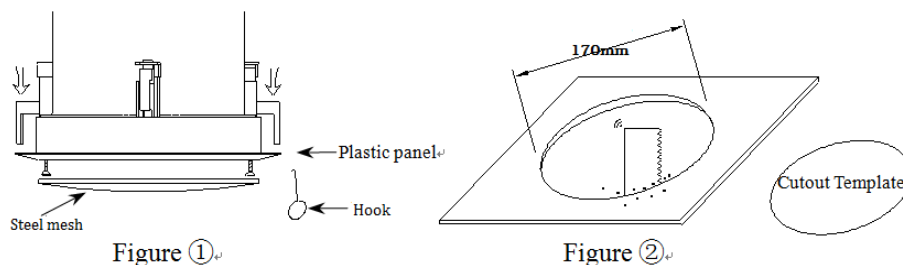
Open the PS760-P (W) telephone boxes, carefully contrast the packing list, check if the related accessories of PS760-P (W) phone are complete, the packing list as follows:

- 1 set PS760-P (W) Phone
- 1 steel mesh iron cover
- 12 screws
- 1 small pull hook
- 1 Quick Guide

In compliance with the following procedure,PS760-P (W) phone can be installed into the ceiling.

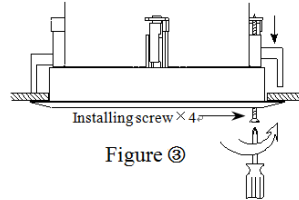
### 2.1 Ceiling entrance specification

Cut out a installation hole of  $\text{Ø}170\text{mm} \sim \text{Ø}180\text{mm}$  on the ceiling (Figure ②),Notice: The figure is just for your reference, as below



## **2.2 Installation of speaker equipment**

- 1、 open the cover(use a small hook to make steel mesh up)(Figure ①);
- 2、 Through the cut-down hole, build the speaker in, after the built-in, use screw driver to remount the four binding post.(Figure ③);



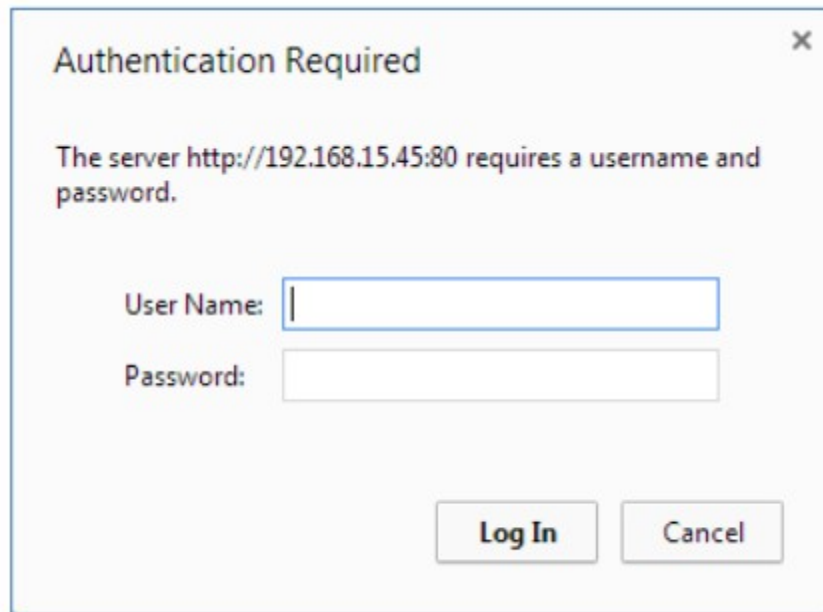
- 3、 3、When installation completed,load the steel mesh back to speaker, then electrify and test it.

## **3. Configuration of speaker**

Before you adjust the speaker, you need to know the IP address of the speaker. there a way you could learn how to get the IP address below. IP address will be gained by DHCP, you could press the second button from the left side. And it will let you know the speaker IP.

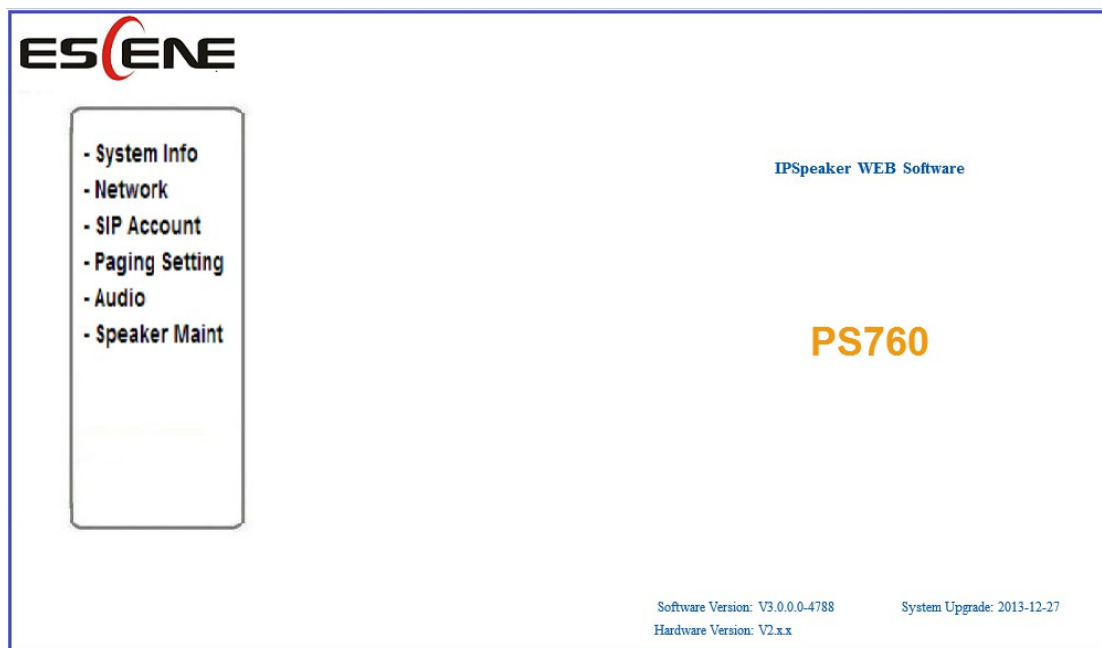
### **3.1 Login web management of speaker**

Once you input the IP address on the web browser and tape enter key on the key board, Then a login screen will pop up from the speaker equipment. You need to input user name and password. Both the tolerant user name and password of system is case letters "root"



The image shows a dialog box titled "Authentication Required" with a close button (X) in the top right corner. The text inside the dialog box reads: "The server http://192.168.15.45:80 requires a username and password." Below this text are two input fields: "User Name:" followed by a text box, and "Password:" followed by a text box. At the bottom of the dialog box are two buttons: "Log In" and "Cancel".

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



The image displays the administrative web interface for the ESENE IPSpeaker. The top left corner features the ESENE logo. On the right side, it says "IPSpeaker WEB Software". In the center, the model number "PS760" is displayed in large orange text. On the left side, there is a vertical menu with the following options: "- System Info", "- Network", "- SIP Account", "- Paging Setting", "- Audio", and "- Speaker Maint". At the bottom of the page, there is technical information: "Software Version: V3.0.0.0-4788", "Hardware Version: V2.xx", and "System Upgrade: 2013-12-27".

The setting and parameter of each option will be describe as follow.

### **3.2 Network**

This option is used for the network setting, please notice hat if you have WIFI speaker, then you could set the LAN and VPN.



### 3.2.1 WIFI Settings

Pls note that the option is just support item PS760. The below operation will guide you how to connect the WIFI speaker to WIFI network, if you know WIFI hot spot SSID, you could input in the text box are, you could also search "SITE SURVEY" according to the hot spot.

- System info
- Network
  - + WIFI Setting
  - + WIFI Status
  - + LAN Setting
  - + VPN Setting
- SIP Account
- Paging Setting
- Audio
- Speaker Maint

WiFi Setting

You can enter the Wireless Network Name of AP.

WiFi:  off  on

WPA/WPA2:  AES  TKIP

WiFi IP Bind:

Wireless Network Name (SSID):

SSID	BSSID	Channel	Type	Encrypt	Signal	Select
ALJAN	00a0f8d975f5	9	b	WEP	38	<input checked="" type="radio"/>
HAJDYAH TECH	442b038c09bb	6	b/g	WPA-PSK	32	<input type="radio"/>
AIC	00090fe6a0b9	3	b/g/n	WPA-PSK/WPA2-PSK	26	<input type="radio"/>
khalid-khobar	589835eaf901	6	b/g/n	WPA2-PSK	24	<input type="radio"/>
FORSANA	c8d719c9ca72	1	b/g/n	WPA2-PSK	22	<input type="radio"/>
ALJANLINK2	00a0f8df6a1	2	b	WEP	20	<input type="radio"/>
Auj_WIFI	22090fe6a0b9	3	b/g/n	WPA/WPA2	18	<input type="radio"/>
Auj_WIFI	22090fe6b649	3	b/g/n	WPA/WPA2	16	<input type="radio"/>
DELTA Company	4ceddee2dc5c	11	b/g/n	WPA-PSK	14	<input type="radio"/>
ATCO	00a0f8de7920	12	b	WEP	14	<input type="radio"/>

### 3.2.2 WIFI Status

Pls note that the option is just support item PS760-W. Once the device connect to the WIFI hot spot, you could use the below operation to check the WIFI connecting status.

- System Info
- Network
  - + WIFI Setting
  - + WIFI Status
  - + LAN Setting
  - + VPN Setting
- SIP Account
- Paging Setting
- Audio
- Speaker Maint

WiFi Status

Wireless Configuration

System Version: V2.0

WiFi MAC: BC:F6:85:FE:9E:E8

Mode: AP Client

SSID: HAJDYAH TECH

Encryption: WPA-PSK

BSSID: 44:2b:03:8c:09:bb

State: Connected

Signal: 35

### 3.2.3 LAN Settings

This option is used for setting speaker IP, as shown below.

LAN Port

**IP settings**

DHCP  
Hostname(Option 12):   
Manufacturer(Option 60):

Static IP  
IP Address:   
Netmask:   
Gateway:

PPPoE  
Username:   
Password:   
MTU:  Default: 1500

**DNS Settings**

Automatic  
 Manual DNS  
Primary DNS:   
Secondary DNS:

**MAC Address**  
MAC Address: 00:26:8b:01:7b:3c

**Port Management Settings**  
HTTP Port:   
Telnet Port:

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

Please Note: Changing the default HTTP Port (80) will require using the new port number to access the IP Speaker web interface. Please note that changes require a reboot. Use the following format when not using the default HTTP (<http://ip address:portnumber>).

### 3.2.4 VPN Settings

If your speaker install into VPN network, then you need to set the VPN as below.

- System Info
- Network
  - + WiFi Setting
  - + WiFi Status
  - + LAN Setting
  - + VPN Setting
- SIP Account
- Paging Setting
- Audio
- Speaker Maint

VPN Setting

Enable VPN:

VPN Type:

L2TP

VPN Server Addr:

VPN User Name:

VPN Password:

### 3.3 SIP Accounts

ESCENE IP speaker could be set three SIP accounts, user name, password and server address is necessary. if he complete information are correct and available, after submitting, there will be a register to the server.

The screenshot shows the configuration interface for 'Account1'. It is divided into three main sections: SIP Settings, Call, and Security. The SIP Settings section includes fields for Enable (checked), Account Mode (VOIP), Display Name (601), Username (601), Authenticate Name (601), Password (masked), Label (601), SIP Server (192.168.15.101), Secondary server (Account1), OutboundProxy Server, Secondary OutboundProxy Server, Polling Interval Time Of Registration (32s), NAT Traversal (Disable), STUN Server, BLA (off), BLA Number, Call Method (SIP), Subscribe Period (1800s), Register Expire Time (3600s), DNS-SRV (off), and SIP Transport (UDP). The Call section includes Amount Of Line Account Used (2), Do Not Disturb (off), Anonymous Call (off), Anonymous Call Rejection (off), Use Session Timer (off), Session Timer (300s), Allow-events (off), Registered NAT (on), Ring Type (None), UDP Keep-alive Message (off), and UDP Keep-alive Interval (30s). The Security section includes SIP Encryption (off), RTP Encryption (off), Encryption Algorithm (RC4), and Encryption Key. A Submit button is located at the bottom left.

### 3.4 Multicast settings

ESCENE IP speaker could be set 20 groups multicast. You could add them and input several IP address and ports according to the below diagram. Notice: The priority of multicast is from 1-20. The highest degree is 1, the lowest degree is 20.

**Paging Setting**

<b>Paging 1:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 2:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 3:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 4:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 5:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 6:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 7:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 8:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 9:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 10:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 11:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 12:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 13:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 14:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 15:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 16:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 17:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 18:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 19:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>
<b>Paging 20:</b>	<input checked="" type="radio"/> off <input type="radio"/> on	Group IP: <input type="text"/>	Port: <input type="text" value="10000"/>

### 3.5 Voice

You can manage the speaker volume ranks and microphones which showed in the following table. Please note that in most cases, if you open the microphone and switch to level 7, then don't use speaker volume more than level 4, unless you can ensure that your coverage is small or noise reduction level is high.

The screenshot shows the 'Audio' configuration window. It is divided into several sections:

- Tone:** Select Country: China (dropdown)
- Output Volume (1~9):** SpeakerPhone Volume: 2 (input field)
- Input Volume (0~7):** SpeakerPhone Mic Volume: 0 (input field)
- Voice Codec:** Payload Length: 20 ms (dropdown), High Rate of G723.1:
- Jitter Buffer:** Type:  Adaptive  Fixed; Min Delay: 60 (input field); Normal Delay: 120 (input field); Max Delay: 150 (input field)
- Other:** VAD: ; SideTone: ; Echo Suppression Mode:
- Ring:** Ring Type: Ring1 (dropdown); Delete (button)
- Uploading Ring Tone:** File input field; Browse... (button); Upload (button); Cancel (button)

Below the 'Uploading Ring Tone' section, there is a note: "(Please upload a ring tone with G711A audio coding, Maximum 10 rings and the total sizes must less than 150k.)"

At the bottom, there is an 'Audio Codecs' section with a list: G722, G711A, G711U, G729A, G723. Navigation buttons include Up, Down, <<, >>, and a vertical scrollbar. A 'disableCode' button is also present. A 'Submit' button is at the bottom left.

### 3.6 Maintains for the speaker

Sometimes, you need to check the speaker's status and usage, or configure some extra options. According to the below, you can reconfigure the speaker or view maintenance logs for more information.

#### 3.6.1 Log

For most well-known system, log file will provide the speaker behavior change and some causes of records, information contained in the log files you need to investigate the problem, if you need to get information from it, then it can upload the log file to the specified server.

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot

Log

No Record

Call: Error Level

SIP

LCD

Log send to server:  off  on

Log Server Address: : 514

Capture Packet: Start End Download

Submit

## 3.6.2 Speaker setup

Most of the following list will serve as a backup, you can set a time and date only and other options can be ignored directly.

**Speaker Setting**

**Basic**

Called No AnswerTime:   s (Min:20, Max:1800)

Caller No AnswerTime:   s (Min:90, Max:1800)

DTMF:  RFC 2833  Inband  SIP Info  Auto

Pound Send Method:  #  %23

RFC 2833 PayLoad:

BackLight:  off  Always On  timer  s (Min:1, Max:255)

Keyboard Lock:

**PSTN Setting**

PSTN Ring Type:  PSTN Ring  VOIP Ring

PSTN Prefix Code:

VOIP Prefix Code:

Hook:  off  on

Hook Frequency:  (Default:500 ms;min:100 ms;max:1600 ms)

**Qos**

SIP Qos:  (0-63)

Voice Qos:  (0-63)

**Call**

BLF Transfer In Taking  off  on

BLF Transfer Mode  Blind Transfer  Attended Transfer

Hot Line Function:  off  Immediately Hot Line  Delay  s (5-30)

Hot Number:

Call Waiting:  off  on

Call Waiting Tone:  off  Play on currently active device Frequency:  s (5-60)

Auto Answer:  off  on  Turn On But Filter This Group:

Auto Answer Mode:  Hands Free  Handle  Headset

Pickup Function:  off  on

Pickup Code:

Message:

Fuzzy Search:  off  on

Booking Voicemail:

Play Voicemail Tone:  off  on

Miss Call Display:  off  on

Call List Save:  off  on

DND Softkey:  off  on

Play Hangup Tone:  off  on

Transfer Code:  off  on Number:

Conference Exit Result:  Disconnect All  Others Remain Connected

Return code when refuse:

Return code when DND:

Flash hook time(≈800ms):

**VOIP Call Forward**

Always:  off  on Number:

If Busy:  off  on Number:

If No Answer:  off  on Number:

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

Set Time Mode:  SNTP  SIP Server  PSTN  Manual

SNTP Server: sparky.services.adelaide.edu.au  
 sparky.services.adelaide.edu.au List  
 sparky.services.adelaide.edu.au Manual

SNTPSecondary server: www.time.ac.cn  
 www.time.ac.cn List  
 www.time.ac.cn Manual

Update Interval(seconds): 600

Daylight Savings Time Mode:  always off  always on  Auto

Time Format:  24 Hour  12 Hour

Date Format: DD MM WWW

Time Zone-GMT: GMT+08:00 Beijing

**Manual Setting**  
2000 Year 1 Month 1 Day 0 Hour 0 Minute 0 Second

**Other**  
QoS: 40 Diff-Serv or Precedence

Check When Upgrade Software: Check BLF Light: On

Headset Mode:  Normal  Seat Mode

Ring Type On Seat Mode:  Headset  Speaker

Network Packet Mirroring: Off

Submit

### 3.6.3 VLAN setup

If you need your speaker work under the situation of VLAN, you need to configure the VLAN information, otherwise you will not be able to operate the speaker or play any announcements.

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot

VLAN Setting

VLAN  
Enable Vlan:

LAN Port  
VID: 0 (0~4094)  
Priority: 0 (0~7)

PC Port  
VID: 0 (0~4094)  
Priority: 0 (0~7)

Submit

### 3.6.4 Password

If you need to change the speaker's default password, you need to configure the password information. Please note that the password change should be ensured safety



and not distribute to anyone, unless they are allowed to interact with the speaker.

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot

Username:

Old Password:

New Password:

Confirm Password:

Administrator  User

### 3.6.5 Factory Defaults Setting

The following will direct you how to set the factory defaults. Please note that factory default setting will erase all the original configuration info, so please be cautious to use this function.

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot

When click this button this equipment will restore to the default status

Pay Attention: It will take effect on next reboot.

### 3.6.6 Auto provision

The auto provision function makes the speaker read the related configuration file by itself. The operation below will show the related operation, such as, name or address of Sever, verify by user name or password (optional), testing period, reboot testing, etc.

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot

Auto Provision

Auto Provision:  on  off

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

Protocol: TFTP

Software Server URL: TFTP://192.168.15.100

Username:

Password:

Auto Download Software

Auto Download Enterprise Phonebook

Auto Download Personal Phonebook

Bootling Checked

Disable the Speaker while bootling checking:  off  on

Auto Provision Frequency: 168 Hour (Default: 7 days, Max: 30 days )

Auto Provision Time: None

Auto Provision Next Time: Wed Dec 4 16:38:37 2013

AES Enable:  off  on

AES Key :

### 3.6.7 FTP Upgrade

You can use FTP/TFTP/HTTP to upgrade new firmware and the software of speaker. The following will direct you if use FTP to upgrade

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot

FTP Upgrade (Attention: Do not cut off the electricity when Upgrade!!)

Server IP:

Filename:

Username:

Password:

Software Upgrade:

Kernel Upgrade:

Note: It's no necessary to input filename when backup.

Configuration:

Phone Book:

EXT Module:

### 3.6.8 TFTP Upgrade

The following will direct you if use TFTP to upgrade.

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot

**TFTP Upgrade** (Attention: Do not cut off the electricity when Upgrade!!)

Server IP:

Filename:

Software Upgrade:

Kernel Upgrade:

Note: It's no necessary to input filename when backup.

Configuration:

Phone Book:

EXT Module:

### 3.6.9 HTTP Upgrade

The following will direct you if use HTTP to upgrade.

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot

**HTTP Upgrade** (Attention: Do not cut off the electricity when Upgrade!!)

HTTP Upgrade:

Select a File:

Software Upgrade:

Kernel Upgrade:

Configuration:

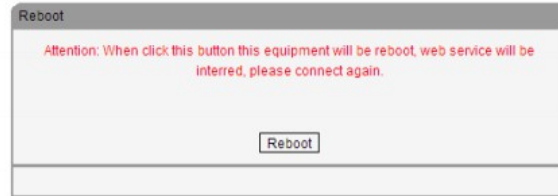
Log:

All Config File:

### 3.6.10 Reboot

The following will direct you if rebooting speaker. Please note that do not rebooting speaker unless hooking or no reaction situation which is rare occurrence.

- System Info
- Network
- SIP Account
- Paging Setting
- Audio
- Speaker Maint
  - + Log
  - + Speaker Setting
  - + VLAN Setting
  - + Password
  - + Default Set
  - + Auto Provision
  - + FTP Upgrade
  - + TFTP Upgrade
  - + HTTP Upgrade
  - + Reboot



### 3.7 Language

The following will direct you how to change language with the speaker.

