



*SayHi*TM

Ceiling Public Broadcasting IP Speaker

User Manual

PS760-P



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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>.

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1. Getting Started

1.1 Outline

PS760-P Ceiling Public Broadcasting Wired 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 PBX 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/at) which cannot coexist with WIFI series.

NOTE: If speaker voice is up to 5 level. For avoid power shortages and lead it restart, you need to use power adapter or support 802.3at's POE power adapter.

1.3 Technical Information

Phone features
WEB Multi-language; Support 3 PBX accounts; 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)
Network parameters

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

Security

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

Audio features

Wideband encoding: G.722 ; Narrow band encoding:G.711 μ /A, G723.1, G726, G.729AB , iLBC; Support VAD,CNG,AEC,AGC etc. audio processing; Full duplex speaker, with automatic echo elimination (AEC).

Physical properties

One RJ-45 10/100M Ethernet interfaces(LAN), LAN port is for the use of ordinary model ;One power adapter; Speaker Output:13Watts;Power:DC 12V/1A;

ABC keys function definitions:

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.

ABC-LEDs keys function definitions:

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

Carton packaging

Size:8.5"x6";The mask width 215mm,Height65mm,The cover width x160mm ,Heightx135mm, Total height:155mm;Net weight:1.1kg Gross weight:2.0kg;

Product Certification



ISO 9001

Platform Compatibility Test (non-certificate)

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 phone to the corporate IP telephony network. If not, please refer to the following instructions.

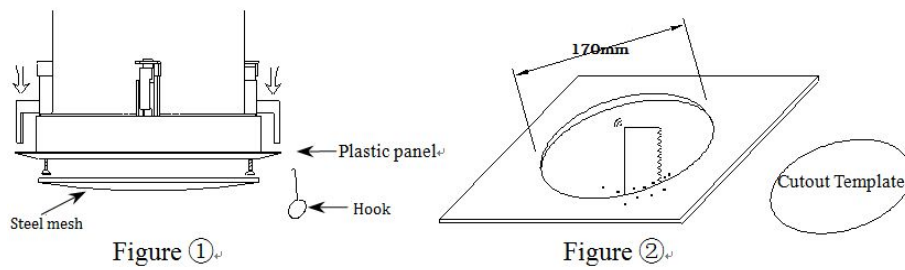
Open the PS760-P telephone boxes, carefully contrast the packing list, check if the related accessories of PS760-P phone are complete, the packing list as follows:

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

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

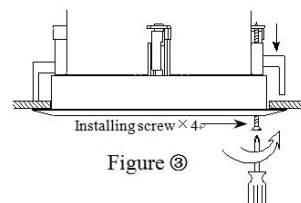
2.1 Ceiling entrance specification

Cut out a installation hole of $\varnothing 170\text{mm} \sim \varnothing 180\text{mm}$ 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 mash 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 mash back to speaker, then electrify and test it.

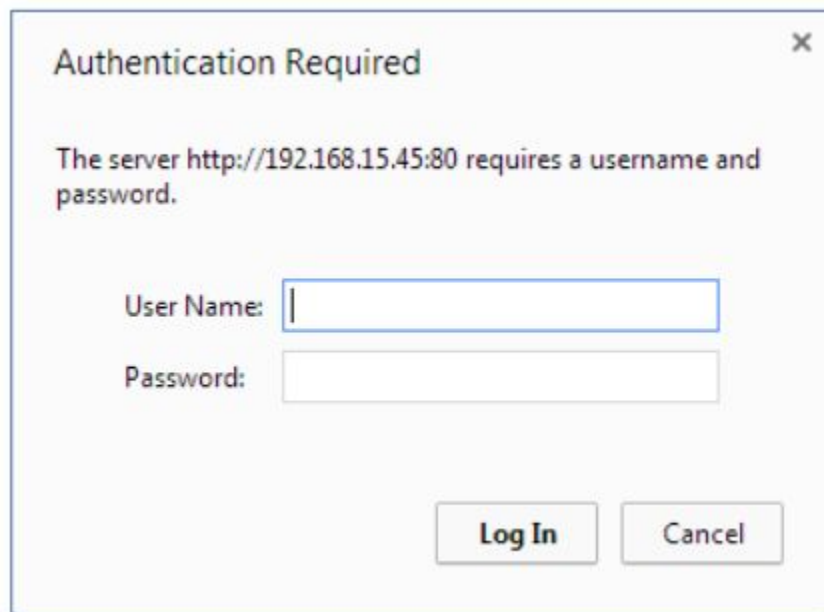
3. Configuration of speaker

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.

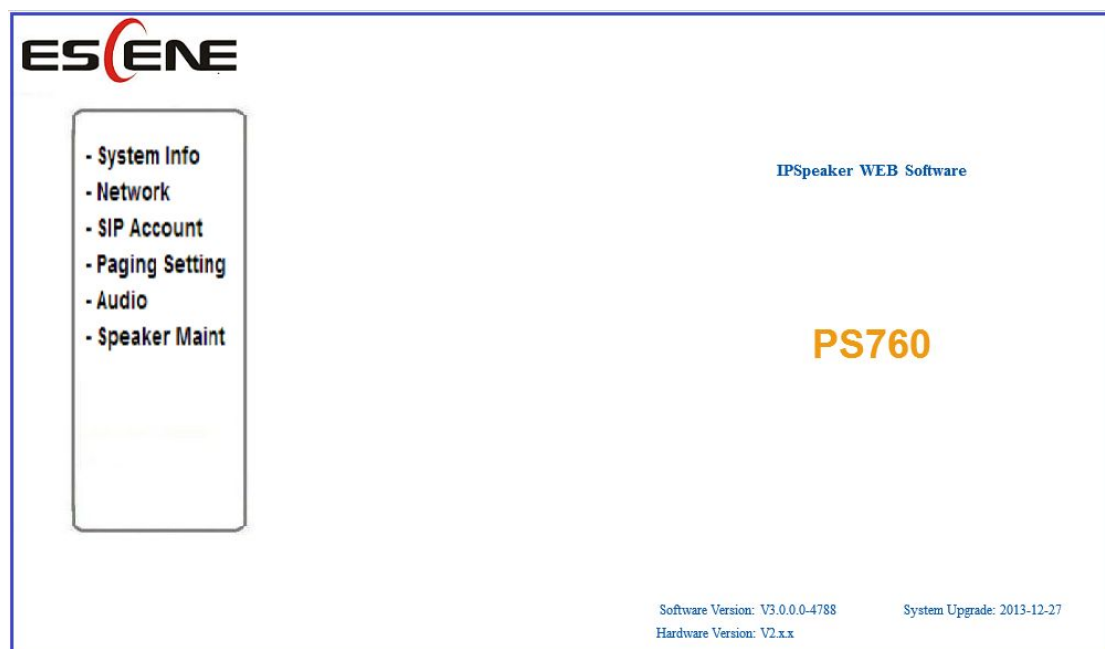
3.1 Login web management of speaker

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 web browser dialog box titled "Authentication Required" with a close button (X) in the top right corner. The text inside the dialog states: "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 containing a vertical cursor, and "Password:" followed by an empty text box. At the bottom right of the dialog 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 "ES CENE" IP Speaker WEB Software interface. The top left features the "ES CENE" logo. On the left side, there is a vertical menu with the following options: "- System Info", "- Network", "- SIP Account", "- Paging Setting", "- Audio", and "- Speaker Maint". The main area of the interface is white and contains the text "IPSpeaker WEB Software" in blue. Below this, the model number "PS760" is displayed in large orange letters. At the bottom of the interface, there is a status bar with the following information: "Software Version: V3.0.0.0-4788", "Hardware Version: V2.x.x", 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.

3.2.1 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: ☒

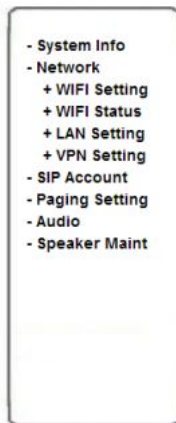
Username:

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).

3.2.2 VPN Settings

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



VPN Setting

Enable VPN: ☐

VPN Type: L2TP

L2TP

VPN Server Addr:

VPN User Name:

VPN Password:

Submit

3.3 SIP Accounts

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

The screenshot shows a configuration window titled "Account1". It is divided into three main sections: "SIP Settings", "Call", and "Security".

SIP Settings:

- Enable: ☒
- Account Mode: **VOIP** (dropdown)
- Display Name: **501** (text box)
- Username: **501** (text box) *
- Authenticate Name: **501** (text box)
- Password: ********* (text box) *
- Label: **501** (text box)
- SIP Server: **192.168.15.101** (text box)
- Secondary server: **Account1** (text box)
- OutboundProxy Server: (text box)
- Secondary OutboundProxy Server: (text box)
- Polling Interval Time Of Registration: **32** s Default Value: 32s. Range: 20s~~60s (text box)
- NAT Traversal: **Disable** (dropdown)
- STUN Server: (text box)
- BLA: ☒ off ☐ on
- BLA Number: (text box)
- Call Method: ☒ SIP ☐ TEL
- Subscribe Period: **1800** Default: 1800s, Min: 120s (text box)
- Register Expire Time: **3600** Default: 3600s, Min: 40s (text box)
- DNS-SRV: ☒ off ☐ on
- SIP Transport: ☒ UDP ☐ TCP ☐ TLS

Call:

- Amount Of Line Account Used: **2** (Default: 2) (text box)
- Do Not Disturb: ☒ off ☐ on
- Anonymous Call: ☒ off ☐ on
- Anonymous Call Rejection: ☒ off ☐ on
- Use Session Timer: ☒ off ☐ on
- Session Timer: **300** (min:150s) (text box)
- Allow-events: ☒ off ☐ on
- Registered NAT: ☐ off ☒ on
- Ring Type: **None** (dropdown)
- UDP Keep-alive Message: ☒ off ☐ on
- UDP Keep-alive Interval: **30** (15-60s) (text box)

Security:

- SIP Encryption: ☒ off ☐ on
- RTP Encryption: ☒ off ☐ on
- Encryption Algorithm: **RC4** (dropdown)
- Encryption Key: (text box)

At the bottom left is a "Submit" button.

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

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

NOTE: If speaker voice is up to 5 level. For avoid power shortages and lead it restart, you need to use power adapter or support 802.3at's POE power adapter.

Audio

Tone
Select Country:

Output Volume (1~9)
SpeakerPhone Volume:

Input Volume (0~7)
SpeakerPhone Mic Volume:

Voice Codec
Payload Length: ms
High Rate of G723.1: ☒

Jitter Buffer
Type: ☒ Adaptive ☐ Fixed
Min Delay:
Normal Delay:
Max Delay:

Other
VAD: ☐
SideTone: ☐
Echo Suppression Mode: ☐

Ring
Ring Type:

Uploading Ring Tone

(Please upload a ring tone with G711A audio coding. Maximum 10 rings and the total sizes must less than 150k.)

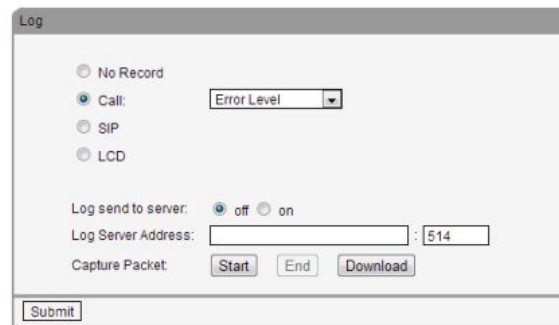
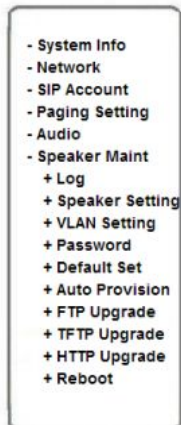
Audio Codecs :
enableCode
G722
G711A
G711U
G729A
G723
disableCode

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.



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: ☒ 70 s (Min:20, Max:1800)

Caller No AnswerTime: ☒ 180 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: (0-53)

Voice Qos: (0-53)

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

Set Time Mode: ☐ SNTP ☒ SIP Server ☐ PSTN ☐ Manual

SNTP Server: ☒ List Manual

SNTPSecondary server: ☒ List Manual

Update Interval(seconds):

Daylight Savings Time Mode: ☐ always off ☐ always on ☒ Auto

Time Format: ☒ 24 Hour ☐ 12 Hour

Date Format:

Time Zone-GMT:

Manual Setting

Year Month Day Hour Minute Second

Other

QoS: Diff-Serv or Precedence

Check When Upgrade Software: BLF Light:

Headset Mode: ☒ Normal ☐ Seat Mode

Ring Type On Seat Mode: ☒ Headset ☐ Speaker

Network Packet Mirroring:

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.

VLAN Setting

VLAN

Enable Vlan: ☐

LAN Port

VID: (0~4094)

Priority: (0~7)

PC Port

VID: (0~4094)

Priority: (0~7)

3.6.4 Password

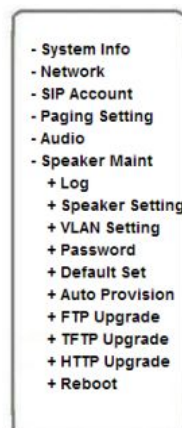
If you need to change the speaker's default password, you need to configure the password information. Please note that the password change should be ensured safety and not distribute to anyone, unless they are allowed to interact with the speaker.



A web form titled "Password" with the following fields: Username (with "root" entered), Old Password, New Password, and Confirm Password. Below the fields are two radio buttons: "Administrator" (selected) and "User". A "Submit" button is at the bottom.

3.6.5 Factory Defaults Setting

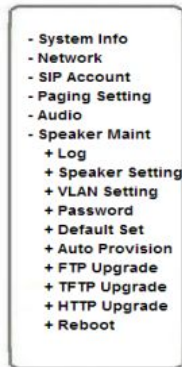
The following will direct you how to set the factory defaults. Please note that factory default setting will erase all the original configuration info, so please be cautious to use this function.



A web form titled "Default Setting" with the following text: "When click this button this equipment will restore to the default status" and "Pay Attention: it will take effect on next reboot." Below the text is a button labeled "Reset to Factory Setting".

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.



Auto Provision

Auto Provision: ☒ on ☐ off

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

Protocol:

Software Server URL:

Username:

Password:

☒ Auto Download Software

☒ Auto Download Enterprise Phonebook

☒ Auto Download Personal Phonebook

☒ Booting Checked

Disable the Speaker while booting checking: ☒ off ☐ on

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

Auto Provision Time:

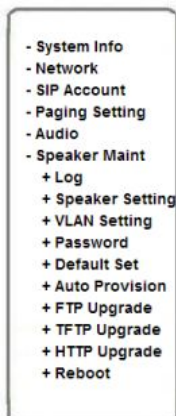
Auto Provision Next Time: Wed Dec 4 16:38:37 2013

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



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

Reboot

Attention: When click this button this equipment will be reboot, web service will be interred, please connect again.

3.7 Language

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

