

Ceiling Public Broadcasting IP Speaker
User Manual
UTT-760P



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About this manual

Thank you for choosing Public Broadcasting IP Phone UTT-760P/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 UTT-760P/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 UTT-760P/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.ultrative.com>.

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1. Getting Started

1.1 Outline

UTT-760P Ceiling Public Broadcasting Wired IP Phone is a new generation of IP speakers. The Multicast functions settings of the device are 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;Auto answer, 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: Static IP/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.711μ/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,Height 65mm,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 UTT-760P phone to the corporate IP telephony network. If not, please refer to the following instructions.

Open the UTT-760P telephone boxes, carefully contrast the packing list, check if the related accessories of UTT-760P phone are complete, the packing list as follows:

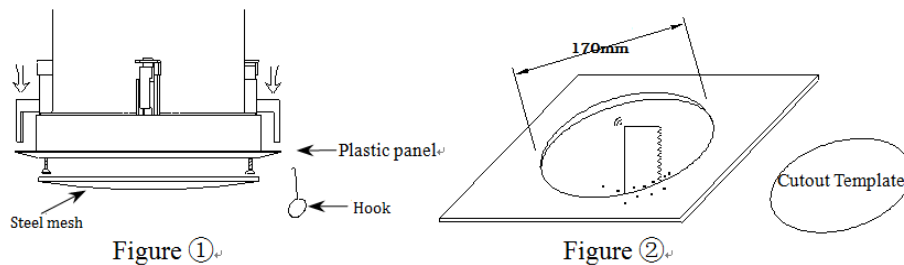
- 1 set UTT-760P Phone
- 1 steel mesh iron cover

- 12 screws
- 1 small pull hook
- 1 Quick Guide

In compliance with the following procedure, UTT-760P phone can be installed into the ceiling.

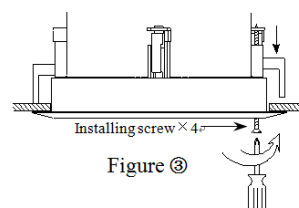
2.1 Ceiling entrance specification

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, when installation completed, load the steel mesh back to speaker, then electrify and test it.

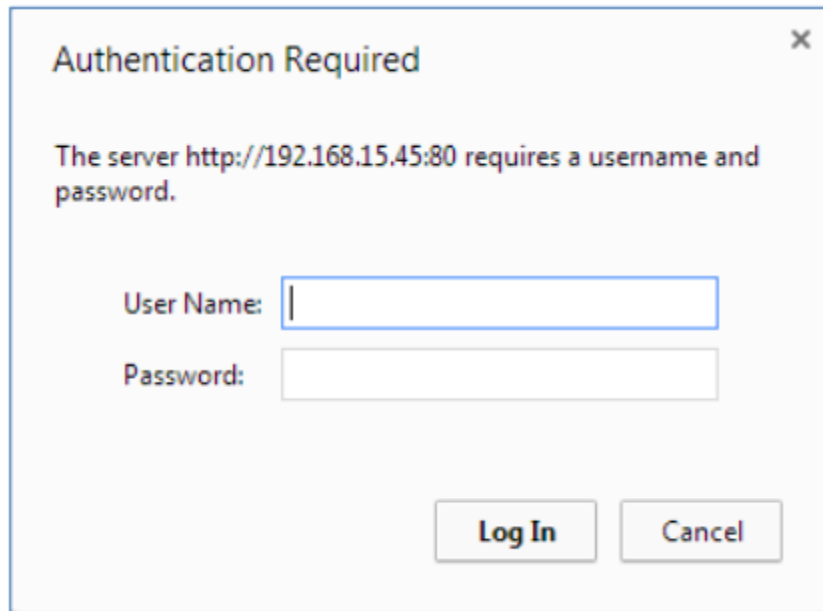
3. Configuration of speaker

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.

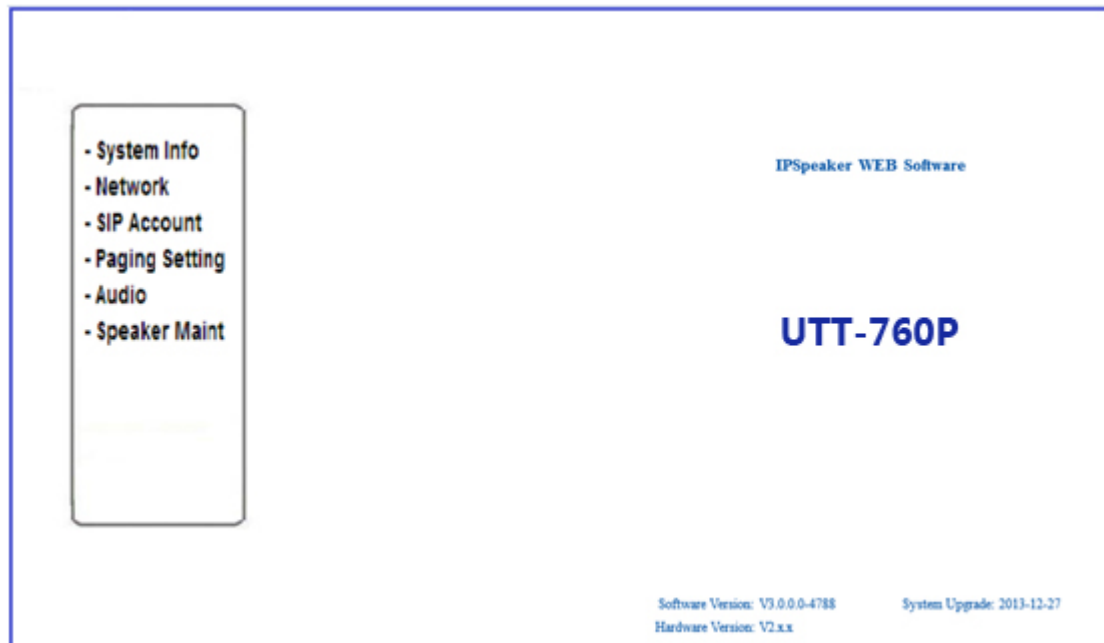
3.1 Login web management of speaker

Once you input the IP address on the web browser and tap 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 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 containing a vertical cursor, and "Password:" followed by a text box. At the bottom 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 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:

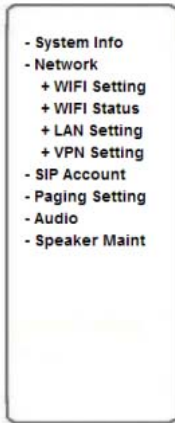
Username:

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:portnumner).

3.2.2 VPN Settings

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



A screenshot of the 'VPN Setting' configuration page. It features a checkbox for 'Enable VPN', a dropdown menu for 'VPN Type' set to 'L2TP', and a section labeled 'L2TP' containing three input fields: 'VPN Server Addr', 'VPN User Name', and 'VPN Password'. A 'Submit' button is located at the bottom of the form.

3.3 SIP Accounts

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

The screenshot shows a configuration window titled "Account1" with three main sections: SIP Settings, Call, and Security. The SIP Settings section includes fields for Account Mode (VOIP), Display Name (601), Username (601), Authenticate Name (601), Password (masked with asterisks), Label (601), SIP Server (192.168.15.101), Secondary server (Account1), OutboundProxy Server, and Secondary OutboundProxy Server. It also has a 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

The 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.

The screenshot displays a web interface titled "Paging Setting". It contains 20 rows, each representing a paging group from "Paging 1" to "Paging 20". Each row includes a radio button to toggle the paging status between "off" and "on", a text input field for the "Group IP", and a "Port" field with the value "10000". At the bottom of the form is a "Submit" button.

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 levels. For avoid power shortages and lead it restart, you need to use power adapter or support 802.3at's POE power adapter.

The screenshot shows the 'Audio' configuration window with the following settings:

- Tone:** Select Country: China
- Output Volume (1-9):** SpeakerPhone Volume: 2
- Input Volume (0-7):** SpeakerPhone Mic Volume: 0
- Voice Codec:** Payload Length: 20 ms, High Rate of G723.1:
- Jitter Buffer:** Type: Adaptive (selected), Fixed; Min Delay: 60, Normal Delay: 120, Max Delay: 150
- Other:** VAD: , SideTone: , Echo Suppression Mode:
- Ring:** Ring Type: Ring1, Delete button
- Uploading Ring Tone:** Browse... button, Upload and Cancel buttons. Note: (Please upload a ring tone with G711A audio coding, Maximum 10 rings and the total sizes must less than 150k.)
- Audio Codecs:** enableCode list: G722, G711A, G711U, G729A, G723. Buttons: Up, Down, <<, >>, disableCode
- Submit** button

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, and 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:

SIP

LCD

Log send to server: off on

Log Server Address: :

Capture Packet:

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: (0-63)

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:

The screenshot shows a configuration page for an IP phone. At the top, there are radio buttons for 'If No Answer' (off/on) and a 'Number' field. Below is 'Ring Frequency' set to 15. The 'Set Time Mode' section has radio buttons for SNTP, SIP Server (selected), PSTN, and Manual. The SNTP Server is set to 'sparky.services.adelaide.edu.au' with a dropdown menu. The SNTP Secondary server is set to 'www.time.ac.cn' with a dropdown menu. 'Update Interval(seconds)' is 600. 'Daylight Savings Time Mode' has radio buttons for 'always off', 'always on', and 'Auto' (selected). 'Time Format' has radio buttons for '24 Hour' (selected) and '12 Hour'. 'Date Format' is 'DD MM WWW' and 'Time Zone-GMT' is 'GMT+08:00 Beijing'. The 'Manual Setting' section includes fields for Year (2000), Month (1), Day (0), Hour (0), and Minute (0). The 'Other' section includes 'GoS' (40), 'Check When Upgrade Software' (Check), 'BLF Light' (On), 'Headset Mode' (Normal), 'Ring Type On Seat Mode' (Headset), and 'Network Packet Mirroring' (Off). A 'Submit' button is at the bottom.

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

The screenshot shows the 'VLAN Setting' configuration page. It has a section for 'VLAN' with an 'Enable Vlan' checkbox. Below are two columns: 'LAN Port' and 'PC Port'. Each column has a 'VID' field (0, range 0-4094) and a 'Priority' dropdown menu (0, range 0-7). A 'Submit' button is at the bottom.

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

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.

