



# XT-07P

## User Manual



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## 1 Safety Instruction

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Please read the following safety notices before installing or using this unit. They are crucial for the safe and reliable operation of the device.

- Please use the external power supply that is included in the package. Other power supply may cause damage to the phone and affect the behavior or induce noise.
- Before using the external power supply in the package, please check the home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it because it may cause fire or electric shock.
- Do not drop, knock or shake the phone. Rough handling can break internal circuit boards.
- This phone is design for indoor use. Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure the phone to high temperature or below 0°C or high humidity.
- Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

## 2 Overview

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### 2.1 Overview

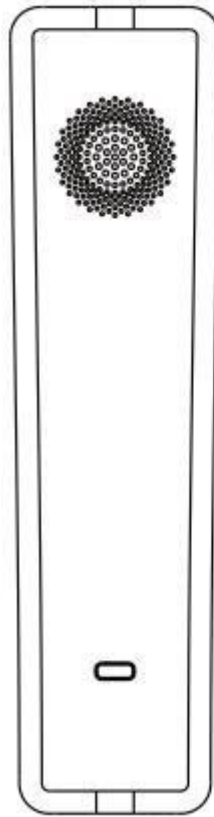
XT-07P is a network telephone specially designed for hotels. The simple design of the device brings excellent user experience for users. The equipment is not only a telephone, but also a masterpiece placed in the living room or office.

XT-07P is the latest generation of network telephone designed for the hotel, which still continues the excellent performance and specifications of traditional equipment; such as high-definition voice, high-performance echo cancellation, 100Mbps Ethernet, QoS, encrypted transmission, automatic configuration, etc.; new system, smooth operation, flat interface setting and many other advantages.

For enterprise users, the equipment is a cost-effective office equipment, while realizing environmental protection, it also provides convenient operation; For home users, the device is a highly efficient communication device. Users can flexibly configure and define the functions of one DSS keys, saving space and cost. It will be an ideal choice for enterprise users and home users who pursue high quality and high efficiency.

In order to help some interested users better understand the details of the product, this user manual can be used as a reference guide for the use XonTel XT-07P.

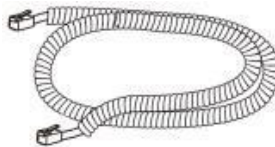
## 2.2 Packing Contents



IP Phone



Handset



Handset Cord

## 3 Desktop Installation

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### 3.1 PoE and the use of external power adapters

The device supports two power supply modes, power supply from external power adapter or over Ethernet (PoE) complied switch.

PoE power supply saves the space and cost of providing the device additional power outlet. With a PoE switch, the device can be powered through a single Ethernet cable which is also used for data transmission. By attaching UPS system to PoE switch, the device can keep working at power outage just like traditional PSTN telephone which is powered by the telephone line.

For users who do not have PoE equipment, the traditional power adaptor should be used. If the device is connected to a PoE switch and power adapter at the same time, the power adapter will be used in priority and will switch to PoE power supply once it fails.

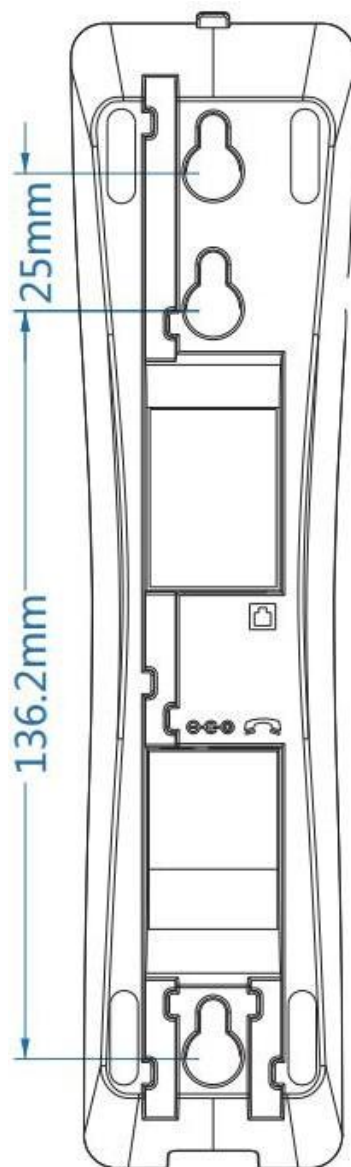
Please use the power adapter supplied by XonTel and the PoE switch met the specifications to ensure the device work properly.

### 3.2 Wall mounted installation method

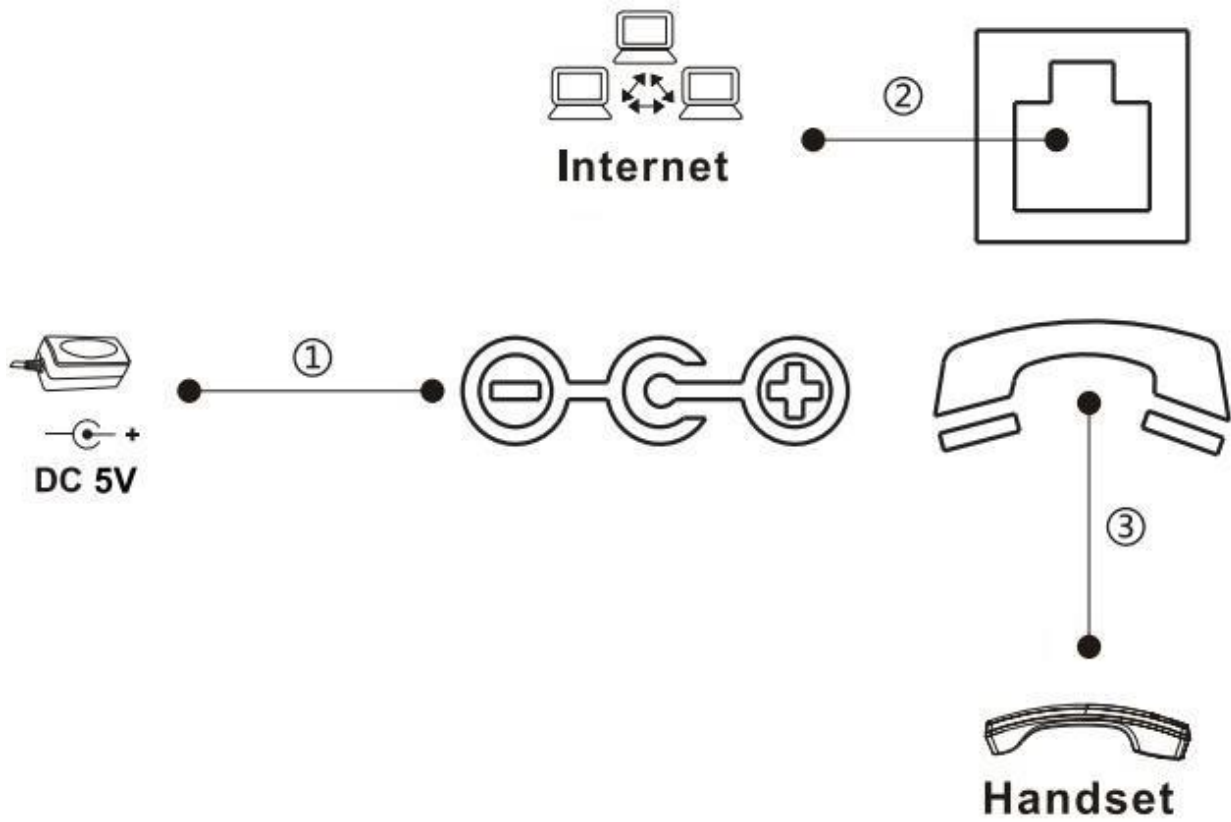
The device supports wall mounted.

Please follow the instructions in the picture below to install the phone :

- 1) Drill two holes in the wall with a vertical distance of 136.2 or 161.2mm.
- 2) Insert two rubber plugs and screws in turn. Note that 5mm is reserved between the nut and the wall, which is convenient for hanging the phone base.
- 3) Connect the cable, handle cable and power supply.
- 4) Align the wall hole on the base with the screws in step 2 and slide down to complete the installation.



Connect the power adapter, network, PC, phone to the appropriate port as shown in the picture below.





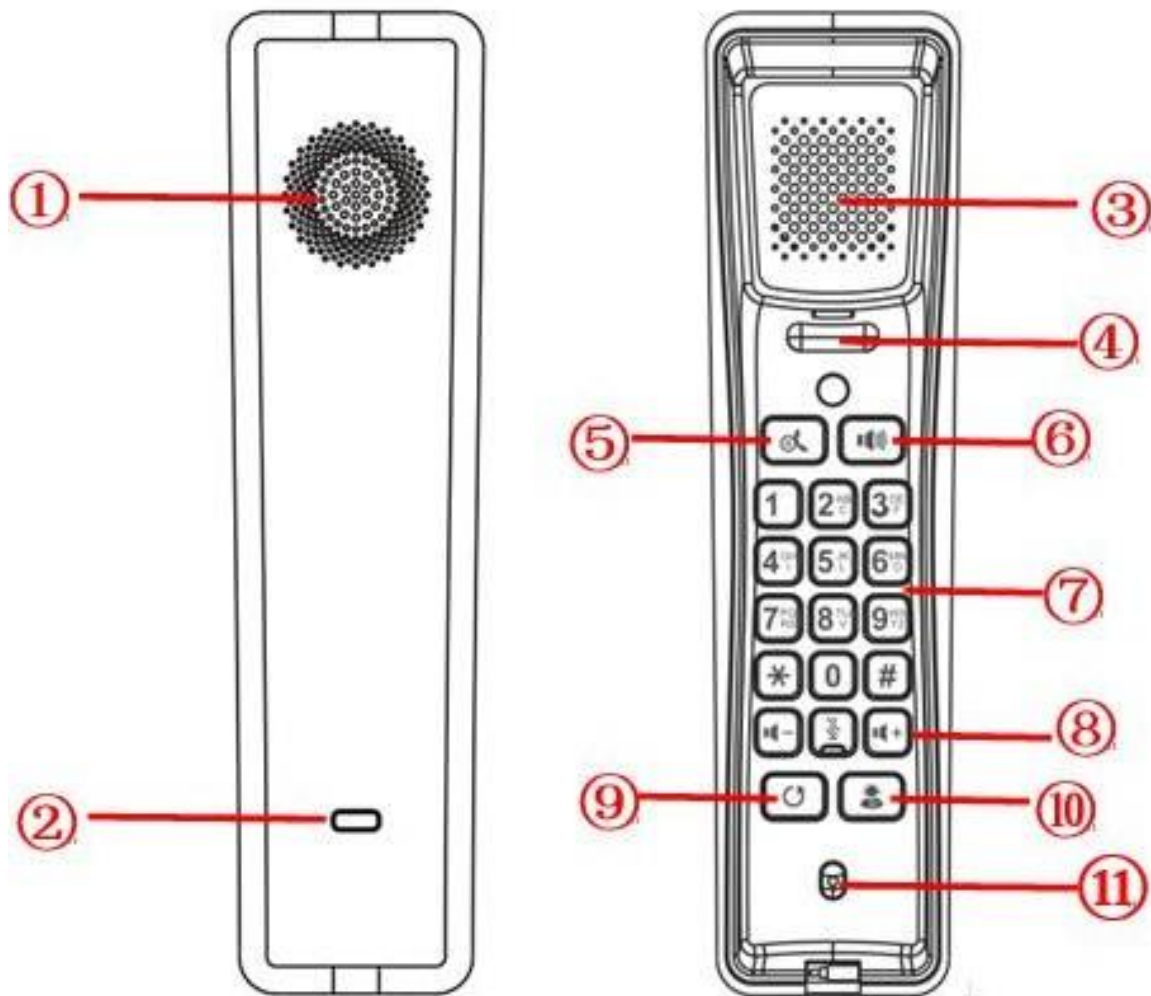
## 4 Appendix Table

### 4.1 Appendix I –LED Definition

Type	LED Light	State
default standby	standby	Green On
	mute	Green slow flash
	Line error (Registration failure)/Network disconnection	Red slow flash
call	calling/Pick up the handle	Red On
	mute	Green slow flash
	hold/held	Green slow flash
	Ringing	Red flash

## 5 Introduction to the User

### 5.1 Instruction of Keypad



The picture above shows the keypad layout of the phone. Each button provides its own specific function. Users can refer to the instructions for the keys in the illustration in this section to operate the phone.

Number	The keypad names	Instruction
①	Hands-free Speaker	The hands-free channel plays sound
②	Status indicator lamp	Power indication/line status indication
③	Handle the horn	The handle channel plays sound
④	Hook	Hang up the handle and hang up the phone
⑤	Hold Key	Press the "Hold" key during the call, the user can hold the call, and press it again to cancel the holding and restore the normal call state.
⑥	Hands-free Key	The user can press this key to open the audio channel of the speakerphone
⑦	Standard Press the "Hold" key	The 12 standard telephone keys provide the same function as standard telephones, but further to the standard function, some keys also provide special function by long-pressing the key, Key # - Long-pressed to broadcast IP (Default English) .
⑧	Volumes Key	The volume to add and subtract-In the standby state, ring and ring configuration interface, press this button to increase/reduce the ring volume; Press this button to increase/lower the volume on the call Mute Key-During a call, the user can press this key to mute the microphone.
⑨	Redial	Press the Redial key to redial the last number dialed
⑩	Function Key	User-defined functionality
⑪	Microphone	Listen when the receiver is answering (do not listen when the phone is hands-free)

## 5.2 Using Handset / Hands-free Speaker

### ■ Using Handset

About the use of the handle, the user can pick up the handle to dial the number, press the "#" button after pressing the number, the number will be dialed. Users can switch audio channels of the phone by pressing the hands-free button.

### ■ Using Hands-free Speaker

For the use of the speakerphone, the user can dial the number by pressing the speakerphone button, or by dialing the number and then pressing the speakerphone button. When the voice channel of the handle is opened, the user can switch the audio channel of the phone by pressing the button of the hands-free speaker.

## 6 Basic Function

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### 6.1 Making Phone Calls

#### ■ Default Line

The device provides two line services (1 main line and 1 standby line). if both lines are configured successfully, the user uses line 1 to make or receive calls by default.

#### ■ Dialing Methods

Users can dial a number in the following ways:

- The Device end
  - Dial directly, pick up the handle and input the number, then press "#" to call out
  - Redialing the last dialed number Redial)
- The Web end
  - Selecting a phone number from call logs

#### ■ Cancel Call

When calling a number, the user can cancel the call by putting back the handle/pressing down the spring.

### 6.2 Answering Calls

Users can answer the call by picking up the handle or pressing the speakerphone button to open the hands-free channel.

The telephone does not support multiple calls. When there is an established call, the user needs to hang up the current call before answering the second call.

### 6.3 End of the Call

When the call is over, the user can put the handle back on the phone and press the speakerphone button to end the call.

## 6.4 Redial

- Redial the last outgoing number:  
When the phone is in standby mode, press the redial button and the phone will call out the last number dialed.
- Call out any number with the redial key:  
Enter the number, press the redial key, and the phone will call out the number on the dial.
- Redial record clearing  
After the phone is used, redial will default to the last used number; therefore, it is necessary to clear the records used by the last customer without affecting the use of other customers.


## 6.5 Mute


You can turn on mute mode during a call and turn off the microphone so that the local voice is not heard. Normally, mute mode is automatically turned off at the end of a call.

You can also turn on mute on any screen (such as the free screen) and mute the ringtone automatically when there is an incoming call.


Mute mode can be turned on in all call modes (handles or hands-free).



### 6.5.1 Mute the Call

During the conversation, press the mute button on the phone:  the mute lamp is red and the power lamp is orange.

Cancel mute: press  cancel mute on the phone again. When the mute lamp goes out, the power lamp returns to its original state

### 6.5.2 Ringing Mute

- Mute: press the mute button when the phone is in standby mode:   
mute light red always bright, power lamp green flashing; There is no ringer for incoming calls.

Cancel ring tone mute: On the standby or incoming call screen, press the mute button again  or volume up cancel ring tone  mute

## 6.6 Call Hold/Resume

User can press [**Hold**] button to maintain the current call, and this button will become [**Resume**] button, and the user can press the "resume" button to restore the call.

## 7 Web Configuration

### 7.1 Getting an IP address

DHCP is the default network setting, and telephone will get the IP address from DHCP server (Router) after the cable connected.

Pick up the handset and long press the “#” key, then you can hear the IP address voice information.

### 7.2 Enter XT-07P web interface

1. Connect the telephone and PC in the same LAN.
2. Open the web browser in the PC, and input XT-07P IP address in the address bar.
3. Enter XT-07P username and password, default username is **admin** and password is **xontel**.
4. Click “**Logon**” button to enter into the web setting interface.



User:

Password:

Language:  ☐

## 7.3 Network Settings

Enter into the web setting interface and select **Network->Basic**, there are three mode of network:

### 1. Static IP

Set IP address manually.

The screenshot shows the 'Basic' tab of the Network settings. The 'Network Mode' is set to 'IPv4 Only'. Under 'IPv4 Network Status', the IP is 192.168.1.39, Subnet mask is 255.255.255.0, Default gateway is 192.168.1.1, and MAC is 0c:38:3e:45:fc:90. In the 'IPv4 Settings' section, 'Static IP' is selected with radio buttons. The fields for IP, Subnet mask, Default gateway, Primary DNS Server (8.8.8.8), and Secondary DNS Server (192.168.1.1) are filled. There is an 'Apply' button at the bottom.

### 2. DHCP

Get IP address from DHCP server automatically.

The screenshot shows the 'Basic' tab of the Network settings. The 'Network Mode' is set to 'IPv4 Only'. Under 'IPv4 Network Status', the IP is 192.168.1.39, Subnet mask is 255.255.255.0, Default gateway is 192.168.1.1, and MAC is 0c:38:3e:45:fc:90. In the 'IPv4 Settings' section, 'DHCP' is selected with radio buttons. The 'Enable Vendor Identifier' is set to 'Disabled'. The 'Vendor Identifier' is 'Xontel'. The 'DNS Server Configured by' is set to 'DHCP'. The 'Primary DNS Server' is 8.8.8.8 and the 'Secondary DNS Server' is 192.168.1.1. There is an 'Apply' button at the bottom.



### 3. PPPoE

PPPoE server, Username, Password

You should choose correct network mode according to your network environment.

Basic Service Port VPN Advanced

System  
Network  
Line  
Phone settings  
Phonebook  
Call logs  
Function Key  
Security  
Device Log

**Network Mode** ?

Network Mode: IPv4 Only

**IPv4 Network Status**

IP:	192.168.1.39
Subnet mask:	255.255.255.0
Default gateway:	192.168.1.1
MAC:	0c:38:3e:45:fc:90

**IPv4 Settings**

Static IP ☐ DHCP ☐ PPPoE ☒

PPPoE Username: user123

PPPoE Password: .....

Primary DNS Server: 8.8.8.8

Secondary DNS Server : 192.168.1.1

DNS Domain:

Apply

## 7.4 SIP Account Settings

Select **Line->SIP** and fill in the items that shown in the figure below.

1. Enable “ **Active** ” check box.
2. **Username**: Fill in the SIP line number or PBX extension number.
3. **Authentication User**: Fill in the SIP line number or PBX extension number.
4. **Authentication Password**: Fill in SIP line or PBX extension registration password.
5. **Server Address**: Fill in server IP address or PBX IP address.
6. **Server Port**: Fill in server register port number.
7. **Transport Protocol**: Select transport protocol (UDP, TCP, TLS) of the phone.

Click the “**Apply**” button to save the configuration, you can dial out after the register status is “**Registered**” with red color.

## 7.5 Auto answering calls

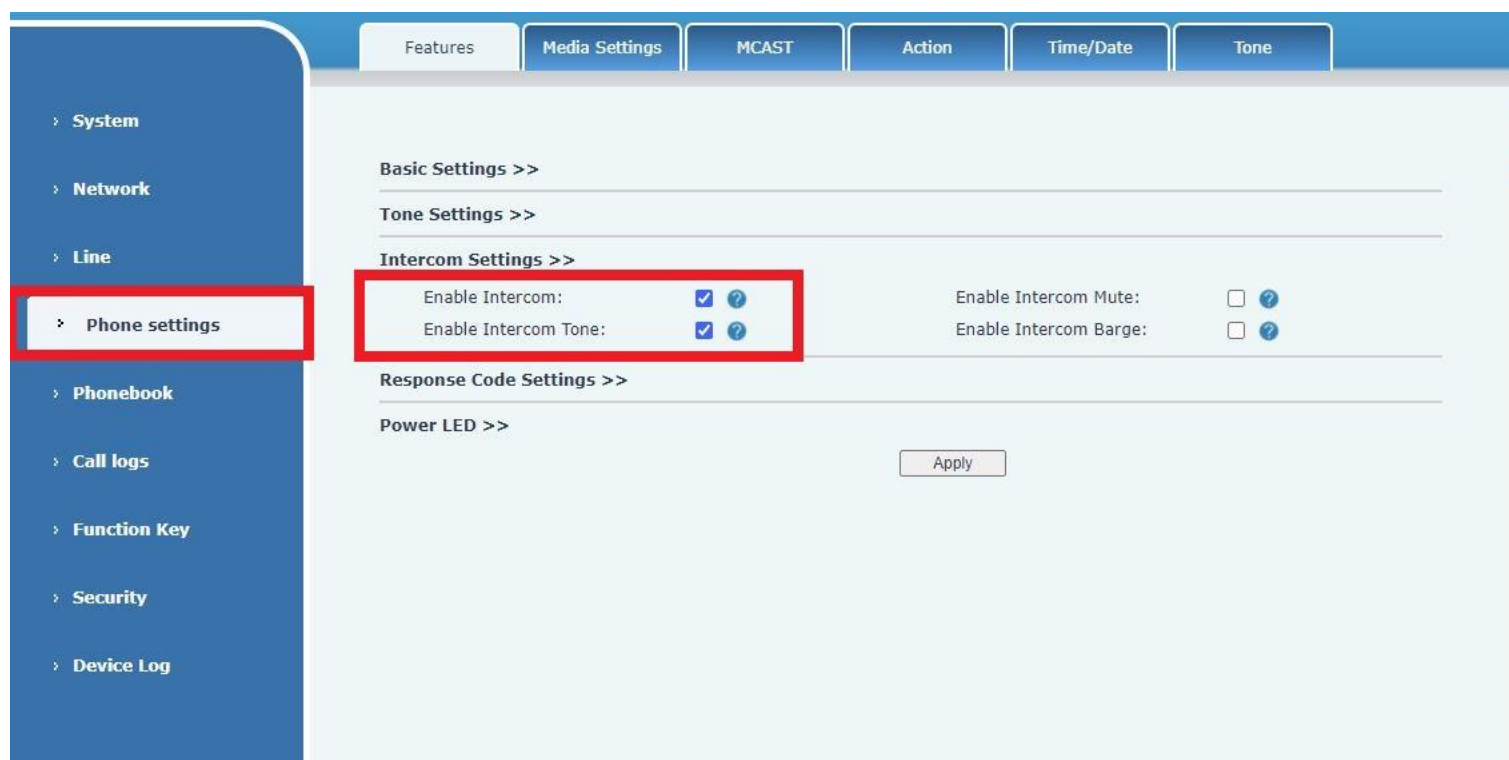
User may enable auto-answering feature on the device and any incoming call will be automatically answered. The auto-answering can be enabled on line basis.

Log in the phone page, enter [Line] >> [SIP], select [SIP] >> [Basic settings], **Enable Auto Answering** and click apply after setting the automatic answering time.



### Important note:

If user want to enable auto answer in the phone if the phone is a member in an paging group, enable intercom and intercom tone as shown below



## 7.6 Add contacts

User can add contacts on the device and specify special ringtone for each contact.

Log in the phone page, enter [Phonebook] >> [Contacts] >> **Add new contact**

The screenshot displays the XonTel XT-07P web interface. On the left is a navigation menu with options: System, Network, Line, Phone settings, Phonebook (selected), Call logs, Function Key, Security, and Device Log. The main area has tabs for Contacts, Cloud phonebook, Blacklist, and Advanced. The 'Contacts' tab is active, showing a 'Contact List' table with columns: Index, Name, Phone, and Phone1. There is one contact with Index 1, Name 607, and Phone 607. A modal dialog titled 'Add new contact' is open on the right. It contains the following fields: Name (basel), Phone (611), Phone1, Phone2, Line (Auto), and Ring (Type 5). The 'Name', 'Phone', and 'Ring' fields are highlighted with red boxes. At the bottom of the dialog are 'OK' and 'Cancel' buttons.

## 7.7 Dial Plan

### Basic Settings

- ☒ Press # to invoke dialing
- ☐ Dial Fixed Length  to Send
- ☒ Send after  second(s) (3~30)
- ☐ Press # to Do Blind Transfer
- ☐ Blind Transfer on Onhook
- ☐ Attended Transfer on Onhook
- ☐ Attended Transfer on Conference Onhook
- ☐ Enable E.164

Apply



Parameters	Description
Press # to invoke dialing	The user dials the other party's number and then adds the # number to dial out;
Dial Fixed Length	The number entered by the user is automatically dialed out when it reaches a fixed length
Timeout dial	The system dials automatically after timeout
Press # to Do Blind Transfer	The user enters the number to be transferred and then presses the "#" key to transfer the current call to a third party
Blind Transfer on Onhook	After the user enters the number, hang up the handle or turn off the hands-free function to transfer the current call to a third party.
Attended Transfer on Onhook	Hang up the handle or press the hands-free button to realize the function of attention -transfer, which can transfer the current call to a third party.
Attended Transfer on Conference Onhook	During a three-way call, hang up the handle and the remaining two parties remain on the call.
Enable E.164	Please refer to E.164 standard specification

## Add dialing rules:

### Dial Plan Add

Digit Map:

Apply to Call: Outgoing Call

Line: SIP DIALPEER

Alias(Optional): No Alias

Suffix:

Match to Send: No

Destination:

Phone Number:

Port:

Length:

Add

### Dial Plan Option

Delete

Modify

### User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type: Number(length)	Suffix
-------	-----------	------	---------------	------	----------------------------	--------

Parameters	Description
Digit Map	<p>There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules.</p> <p>In prefix matching, only part of the number is entered followed by T. The mapping will then take place whenever these digits are dialed.</p> <p>Prefix mode supports a maximum of 30 digits.</p>
<p>Note: Two different special characters are used.</p> <ul style="list-style-type: none"> <li>x -- Matches any single digit that is dialed.</li> <li>[ ] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.</li> </ul>	
Destination	Set Destination address. This is for IP direct.
Port	Set the Signal port, and the default is 5060 for SIP.
Alias	Set the Alias. This is the text to be added, replaced or deleted. It is an optional item.
<p>Note: There are four types of aliases.</p> <ul style="list-style-type: none"> <li>all: xxx — xxx will replace the phone number.</li> <li>add: xxx — xxx will be dialed before any phone number.</li> <li>del — The characters will be deleted from the phone number.</li> <li>rep: xxx — xxx will be substituted for the specified characters.</li> </ul>	
Suffix	Characters to be added at the end of the phone number. It is an optional item.

Length	Set the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. It is an optional item.
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This feature allows the user to create rules to make dialing easier. There are several different options for dial rules. The examples below will show how this can be used.

**Example 1:** All Substitution -- Assume that it is desired to place a direct IP call to IP address 172.168.2.208. Using this feature, 123 can be substituted for 172.168.2.208.

#### User-defined Dial Plan Table ?

Index	Digit Map	Call	Match to Send	Line	Alias Type: Number(length)	Suffix
1	"123"	Out	No	SIP DIALPEER(172.168.2.208:5060)		

**Example 2:** Partial Substitution -- To dial a long distance call to Beijing requires dialing area code 010 before the local phone number. Using this feature 1 can be substituted for 010. For example, to call 62213123 would only require dialing 162213123 instead of 01062213123.

#### User-defined Dial Plan Table ?

Index	Digit Map	Call	Match to Send	Line	Alias Type: Number(length)	Suffix
1	"1T"	Out	No	636@SIP1	rep:010(1)	



## 7.8 Function Key

The device has a total of 11 configurable custom function keys; One direct call foreground key and 10 custom digital speed dial keys.

Device direct call key, default configuration as a fixed number;(customizable replacement)

0~9 numeric keys can be used as customized shortcut keys, users can customize the configuration of 0~9 numeric keys in the web page, users can quickly dial the corresponding number by long press each shortcut key.

Webpage interface: **[Function key] >> [Function key]**.

Key	Type	Name	Value	Subtype	Line	PickUp Number
DSS Key 1	Memory Key	basel	611	Speed Dial	636@SIP1	

The DSS Key could be configured as followings,

- ◆ Memory Key
  - Speed Dial/Intercom/Call Park/Call Forward (to someone)
- ◆ Event
  - Call Transfer (to someone)
- ◆ DTMF
- ◆ MCAST Paging
- ◆ Action URL



Parameters	Description
Memory Key	<p><b>Speed Dial:</b> You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed.</p> <p><b>Intercom:</b> This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments.</p> <p><b>Call Park:</b> You can put a call on hold and continue the conversation from any other telephone set.</p> <p><b>Call Forward:</b> You can divert the call to the number you want.</p>
Event	<b>Transfer:</b> You can transfer the call to another number.
DTMF	It allows user to dial or edit dial number easily.
Multicast	Configure the multicast address and audio codec. User presses the key to initiate the multicast.
Action URL	The user can use a specific URL to make basic calls to the phone.

## 7.9 Speed Dial List

The user can configure the number button "0~9" to be the speed dial key. After the configuration is completed according to the figure below, the user can long press the corresponding number key to achieve speed dial.

webpage interface: [Function key] >> [Speed Dial List].

English

Function Key

Speed Dial List

System
Network
Line
Phone settings
Phonebook
Call logs
Function Key

Key	Type	Name	Value	Subtype	Line	PickUp Number
DSS Key 1	Memory Key	basel	611	Speed Dial	636@SIP1	
DSS Key 2	Memory Key	emma	613	Speed Dial	636@SIP1	
DSS Key 3	Memory Key	fuad	603	Speed Dial	636@SIP1	
DSS Key 4	None			None	AUTO	
DSS Key 5	None			None	AUTO	
DSS Key 6	None			None	AUTO	
DSS Key 7	None			None	AUTO	
DSS Key 8	None			None	AUTO	
DSS Key 9	None			None	AUTO	
DSS Key 0	None			None	AUTO	

Apply

## 7.10 Call Log

The phone can store up to 600 call records, the user can browse the complete call record in this page. The call record can be sorted by time, call number, contact name or line, and the call record can be screened by call record type (incoming call, outgoing call, missed call, forward call).

The user can also save the number in the call record to his/her phone book or add it to the blacklist/whitelist.

Users can also dial the web page by clicking on the number in the call log.

webpage interface: [Call logs]

**Call Information**

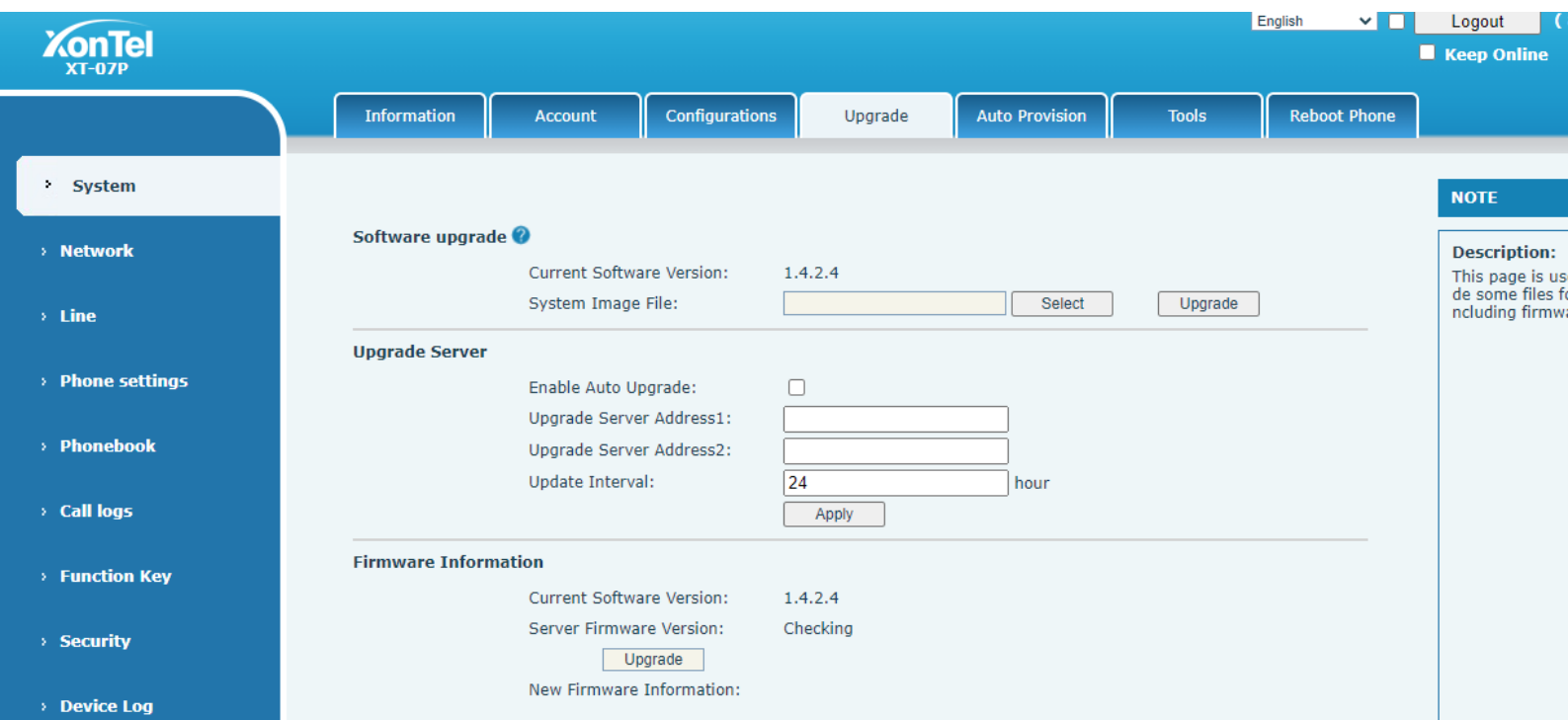
Call Type: All Previous Page: 1 Next

<input type="checkbox"/>	Index	Time	Call Type	Caller Number	Contact Name	Duration	Line	Add to phonebook
<input type="checkbox"/>	1	2020/12/31 16:14:15	Missed Calls	<a href="#">611</a>	Basel	00:00:00	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	2	2020/12/31 16:12:35	Missed Calls	<a href="#">611</a>	Basel	00:00:00	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	3	2020/12/31 16:12:17	Missed Calls	<a href="#">611</a>	Basel	00:00:00	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	4	2020/12/31 16:12:11	Missed Calls	<a href="#">611</a>	Basel	00:00:00	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	5	2020/12/31 16:12:1	Outgoing Calls	<a href="#">611</a>	611	00:00:00	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	6	2020/12/31 15:55:36	Missed Calls	<a href="#">611</a>	Basel	00:00:00	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	7	2020/12/31 15:45:10	Outgoing Calls	<a href="#">611</a>	611	00:02:01	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	8	2020/12/31 15:44:44	Missed Calls	<a href="#">611</a>	Basel	00:00:00	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	9	2020/12/31 15:43:32	Outgoing Calls	<a href="#">611</a>	611	00:01:05	636@SIP1	<input type="button" value="Add"/>
<input type="checkbox"/>	10	2020/12/31 15:43:12	Outgoing Calls	<a href="#">113</a>	113	00:00:07	636@SIP1	<input type="button" value="Add"/>

10 Entries per page

## 7.11 Upgrade

web interface: log into the phone web page and enter the **[System] >> [Upgrade]** page.



Parameter	Description
<b>Upgrade server</b>	
Enable Auto Upgrade	Enable automatic upgrade, if there is a new version txt and new software firmware on the server, phone will show a prompt upgrade message after Update Interval.
Upgrade Server Address1	Set available upgrade server address.
Upgrade Server Address2	Set available upgrade server address.
Update Interval	Set Update Interval.
<b>Firmware Information</b>	
Current Software Version	It will show Current Software Version.
Server Firmware Version	It will show Server Firmware Version.
[Upgrade] button	If there is a new version txt and new software firmware on the server, the page will display version information and upgrade button will become available; Click [Upgrade] button to upgrade the new firmware.
New version description information	When there is a corresponding TXT file and version on the server side, the TXT and version information will be displayed under the new version description information.