



VOIP Phone
Model: TL-IP
User Manual



Version 1.2



(QR code)

Smartphone Scan to download PDF version

1	Introduction	3
1.1	Hardware Overview	3
1.2	Software Overview	5
2	Setup the IP Phone system by using Web Browser	5
2.1	Login	5
2.2	System Information	6
2.3	Phone Settings	7
2.3.1	Call Forward	7
2.3.2	SNTP Settings	8
2.3.3	Volume Settings	8
2.3.4	Melody Settings	9
2.3.5	DND Settings	9
2.3.6	Dial Plan	10
2.3.7	Call Waiting Settings	11
2.3.8	Alarm Settings	12
2.4	Network	12
2.4.1	WAN Settings	12
2.4.2	VLAN Settings	14
2.4.3	PPTP Settings	15
2.5	SIP Settings	15
2.5.1	Service Domain	15
2.5.2	Port Settings	17
2.5.3	Codec Settings	17
2.5.4	Codec ID Settings	19
2.5.5	DTMF Settings	19
2.5.6	STUN Settings	19
2.5.7	Other Settings	20
2.6	Others	21
2.6.1	Auto Config	21
2.6.2	Advanced Settings	23
2.7	Update	23
2.7.1	Auto Update	23
2.7.2	Update System	24
2.7.3	Default Settings	25
2.8	System Auth.	26
2.9	Save Change	26
3	Copyright and Trademarks	
27		

1 Introduction

Before using the SIPPhone, some configurations are required to make the IP Phone work properly. The manual will illustrate how to configure the ip phone via web page

1.1 Hardware Overview

One RJ-45 Networking interfaces for WAN port which supports 10/100Mbps Fast Ethernet. The default setting of WAN port is a DHCP client.

One LED that combine functions of ringer and message-waiting and one in-use LED.

The power is only supplied by switch hubs with POE (Power over Ethernet). There is no any jack for external power adaptor.





1.2 Software Overview

Network Protocol	Codec
<ul style="list-style-type: none"> • SIP 2 (RFC3261) • IP/TCP/UDP/RTP/SNTP • DHCPClient/PPPoE Client/PPTP Client • HTTPServer • DNSClient 	<ul style="list-style-type: none"> • G711 aLaw • G711 uLaw • G722 • G723 • G729 • iLBC
Voice Quality	Tone
<ul style="list-style-type: none"> • Comfortable noise generator • Voice Activity Detector • Voice QoS • SIP QoS • Jitter Buffer 	<ul style="list-style-type: none"> • Ring Tone • 350Hz/440Hz Dial Tone • 400Hz Dial Tone • Busy Tone
Phone Function	Call Function
<ul style="list-style-type: none"> • Speedial key • Volume Adjustment • 8 types of ringing melody • Do Not Disturb • Schedule Alarm 	<ul style="list-style-type: none"> • Call Hold • Call Mute • Call Waiting • Call Forward
IP Assignment	DTMF
<ul style="list-style-type: none"> • Static IP • DHCP • PPPoE 	<ul style="list-style-type: none"> • RFC2833 • Inband • SIPInfo
NAT Traversal	Firmware Upgrade
<ul style="list-style-type: none"> • STUN 	<ul style="list-style-type: none"> • TFTP • FTP • HTTP • Local Computer
SIP Server	Configuration
<ul style="list-style-type: none"> • Up to six SIPaccount • Outbound Proxy 	<ul style="list-style-type: none"> • Web Browser

2 Setup the IP Phone system by using Web Browser

Before configure the IP Phone, Firstly user should press the keys “123*#” keys of IP phone in order to obtain the IP address which is assigned from a DHCP server.

2.1 Login

Please input the default username and password into the blank fields. The default username of administrator is **superuser**, the default password is **123456**. Then click the Login button to login the SIP Phone.

Login VoIP

Enter your password to login VOIP server


Username

Password

2.2 System Information

After login the web page, user can see the system information such as model name and firmware version.

In addition, there is a function list in the left hand side. User can use mouse to click the function to setup and configure the IP phone.



+ Phone Settings
+ Network
+ Sip Settings
+ Others
+ Update
System Auth.
Save Change

System Information

This page illustrate the system related information

Item	Description
Model Name	TL-IP
Firmware Version	TL-IP-201204182

2.3 Phone Settings

Phone Settings contains Call Forward, SNTP Settings, Volume Settings, Melody Settings, DND Settings, Dial Plan, Call Waiting Settings and Alarm Settings.

2.3.1 Call Forward

2.3.1.1 All Forward

All incoming call will be forward to the number that is filled. Please input the name in the name field and the phone number or IP address in the URL field.

2.3.1.2 Busy Forward

New incoming call will be forwarded to the number that user chose while user is online.

2.3.1.3 No Answer Forward

User can have incoming calls answered by another phone whenever the IP phone is unanswered after several seconds of ring. How long the call will be forwarded is determined by the No Answer Fwd Time Out. Parameter range is from 5 seconds to 30 seconds.

If user wants to disable previous forward settings, choose “Disable” from the drop-down combo box. After finishing the setting, please click “Apply” button.

Forward Settings

You could set the forward number of your phone in this page.

Forward Type	<div> <div>All Forward</div> <div> <div>Disable</div> <div>All Forward</div> <div>Busy Forward</div> <div>No Answer Forward</div> </div> </div>
All Fwd No.	<input type="text"/>
Busy Fwd No.	<input type="text"/>
No Answer Fwd No.	<input type="text"/>
No Answer Fwd Time Out	<input type="text" value="5"/> (5~30 sec)
<div> <div>Apply</div> <div>Reset</div> </div>	

2.3.2 SNTP Settings

User can enable SNTP Setting to synchronize the time from an outside Time Server to the IP Phone. Since all time sources over Internet supply GMT time only, user should also set suitable Time Zone from the drop-down combo box. Also, user can disable SNTP and input time to the fields of Local Time manually. Please click the “Apply” button after finishing.

SNTP Settings

You could set the SNTP servers in this page.

SNTP

☐ Disable ☒ Enable

SNTP Server

time.windows.com

Time Zone

(GMT +08:00) Beijing, Chongqing, Hong Kong, Manila, Perth, Singapore, Taipei, Urumq

Local Time

2011

:

01

:

01

:

00

:

00

(Year:Month:Day Hour:Min)

Apply

Reset

2.3.3 Volume Settings

User can setup Handset Volume, Speaker Volume and Ringer Volume here. The higher number is set, the louder output user get. Please click the “Apply” Button after finishing.

Volume Settings

You could set the volume of your phone in this page.

Handset Output

3

(1~7)

Speaker Output

1

(1~7)

Ringer Volume

2

(1~7)

Apply

Reset

2.3.4 Melody Settings

User can select one of melodies from Ringer Type for ringing tone of incoming call. Please click the “Apply” button after finishing.

Ringer Settings

You could set your favorite ringer in this page.

Ringer	<input type="radio"/> Off <input checked="" type="radio"/> On
Ringer Type	<div>ringer 3 ▼</div>
<div>Apply Reset</div>	

2.3.5 DND Settings

User can setup Do Not Disturb either from a period of time or always on. Callers will hear busy ring tone while one of DND Settings is enabled. Please click the “Apply” button after finishing.

DND Settings

You could set the do not disturb period of your phone in this page.

DND Always	<input checked="" type="radio"/> Off <input type="radio"/> On
DND Period	<input checked="" type="radio"/> Off <input type="radio"/> On
From	<div>00 : 00 (Hour:mm)</div>
To	<div>00 : 00 (Hour:mm)</div>
<div>Apply Reset</div>	

2.3.6 Dial Plan

2.3.6.1 Replace Rule

The Dial Plan function provides basic dial number replacement or drop rule. Maximum 4 rules user can apply at a same time. The rules will only be effective when matching digits are located at the beginning of dialed numbers.

Example for operation of “replace number by”:

Digits for matching	Operation	Digits for operation
852	replace number by	1234

When user presses 85291234567 on the keypad, the IP Phone will send out 123491234567

Example for operation of “drop number”:

Digits for matching	Operation	Digits for operation
0050	Drop number	

When user presses 005091234567 on the keypad, the IP Phone will send out 91234567.

2.3.6.2 Dial Now

If user wants to dial some digits at once without waiting for timeout, please input the digits into the field of “Dial Now”.

User can set more than one rule in the field by adding “+”, e.g. *xx+#xx+11x+xxxxxxx. If the number dialed matches the rule “*xx”, e.g. “*11”, “*1123”. “*11” will be automatically dial out at once no matter there are more digits followed by “*11”.

2.3.6.3 Auto Dial Time

The Auto Dial Time instructs IP phone to treat input is completed and send out a call after how many seconds without pressing keypad.

2.3.6.4 Use Pound Key (#) As Send Key

User can enable the pound key (#) as an end signal. It instructs the IP Phone dial out the numbers at once by pressing pound key. For example, 91234567#.

2.3.6.5 Use Asterisk Key (*) For IP Dialing

User can enable the asterisk key (*) as a dot-decimal notation of IP address. After user enabled it, user can direct input IP address by keypad.

2.3.6.6 Dial Tone

For the Dial Tone option, default is dual tone 350Hz/440Hz, it is a standard of North American. The "General" one is 400Hz, it is suitable in Japan and China.

Please click the "Apply" button after finishing.

Dial Plan

You could set the dial plan in this page.

Name	Digits for matching	Operation	Digits for operation
Replace rule 1	<input type="text"/>	disable	<input type="text"/>
Replace rule 2	<input type="text"/>	drop number	<input type="text"/>
Replace rule 3	<input type="text"/>	replace number by	<input type="text"/>
Replace rule 4	<input type="text"/>	disable	<input type="text"/>

Dial Now

Auto Dial Time (3~9 sec)

Use # as send key ☐ No ☒ Yes

Use * for IP dialing ☒ No ☐ Yes

Dial Tone ☒ USA ☐ General

2.3.7 Call Waiting Settings

User can enable or disable the call waiting function. Please click the "Apply" button after finishing.

Call Waiting

You could enable/disable the call waiting settings in this page.

Call Waiting ☐ Disable ☒ Enable

2.3.8 Alarm Settings

User can let the IP phone ring as an alarm at dedicated time schedule. Please click the "Apply" button after finishing.

Alarm Settings

You could set the alarm time in this page.

Alarm ☐ Off ☒ On

Alarm Time : (Hour:mm)

Current Time: 2012-2-29 14:01

Apply

Reset

2.4 Network

In Network page, user can configure all the network settings and check the network status of IP phone.

2.4.1 WAN Settings

Let user configure all parameters for WAN port. You can set a fixed IP address for the WAN port or configure it to obtain the IP address through either DHCP client or PPPoE. You must choose one of IP Mode which is suitable to your current network environment.

2.4.1.1 Fixed IP Settings

User should input the IP address, the net mask and default gateway which are suitable to current network into the fields.

2.4.1.2 DHCP Settings

When DHCP is set, IP Phone acts as a DHCP client and obtains all TCP/IP parameters from DHCP server.

2.4.1.3 PPPoE Settings

Simply input the username of PPPoE account into the field of ID and the password into the field of Password. Both of them are provided by service provider of user.

2.4.1.4 DNS

User can manually input the IP address of Primary DNS server and Secondary DNS server, or set automatically obtain them from DHCP server. In general practice, IP address of DNS servers will be automatically assigned in both DHCP and PPPoE mode.

2.4.1.5 Vendor

In some cases, the vendor parameter must be submitted to PPPoE service provider during login. Should user enable it or not please refer to the user manual that provided by service provider.

WAN Settings

You could configure the WAN settings in this page.

IP Mode ☒ Fixed ☐ DHCP ☐ PPPoE

Fixed IP Settings

IP Address	192.168.204.66
Net Mask	255.255.255.0
Default GW	192.168.204.10

PPPoE Settings

ID	
Password	

DNS

Auto DNS Enable	<input checked="" type="radio"/> Off <input type="radio"/> On
Primary DNS	168.95.1.1
Second DNS	168.95.1.2

Vendor

Vendor Enable	<input checked="" type="radio"/> Off <input type="radio"/> On
Vendor	

2.4.1.6 Current Status

It shows the current status of connection, and the current information such as IP address, Netmask, Gateway, MAC address, IP address of Primary DNS and Secondary DNS.

Please click the "Apply" button after finishing.

Current Status: Fixed

IP: 192.168.204.066
Mask: 255.255.255.000
Gateway: 192.168.204.010
MAC Address: 2e:2e:d3:9d:31:73
DNS1: 168.095.001.001
DNS2: 168.095.001.002

Apply

Reset

2.4.2 VLAN Settings

User can create independent logical networks within a physical network by deploying VLAN environment.

2.4.2.1 VID (802.1Q/TAG)

If user enable VLAN Packet, VLAN ID/VLAN TAG should be given for inserting into packet header in order to classify the packets belong to Virtual Local Area Network..

2.4.2.2 User Priority (802.1P)

User can set the frame priority level for different classes of network traffic. Values are from 0 (best effort) to 7 (the highest). The smaller number is set, the lower priority is set.

VLAN Settings

You could set the VLAN settings in this page.

VLAN Packets

☒ Off ☐ On

VID (802.1Q/TAG)

(0 ~ 4094)

User Priority (802.1P)

(0 ~ 7)

Apply

Reset

2.4.3 PPTP Settings

To connect remote VPNserver by point-to-point tunneling protocol. After connected, the subnet of IP phone is equal to the network of remote VPNserver. The IP phone becomes a member of remote network. All data traffic between remote server and IP phone will be encrypted. To enable the PPTPconnection, user should input the user name and password of VPNaccount of remote server. Pleaseclick the “Apply” button after finishing. If the WAN is connected by using PPPoE,user can only obtain the IP address that assigned by PPTPserver by pressing“**47#” while the IP Phoneis idle.

PPTP Settings

You could set PPTP settings in this page.

PPTP	<input type="radio"/> Off <input checked="" type="radio"/> On
PPTP Server	<input type="text" value="vpn.example.com"/>
PPTP Username	<input type="text" value="tom_buehl"/>
PPTP Password	<input type="password" value="••••••••"/>
<input type="button" value="Apply"/> <input type="button" value="Reset"/>	

2.5 SIP Settings

In order to let SIPphone work properly, user should setup SIPService Domain, SIP Port, Codec, Codec ID, DTMF, STUNserver and others. Some information of them user should obtain from SIPserviceprovider.

2.5.1 Service Domain

Usercan setup total six SIPaccounts for receiving inbound calls and use the first realm for outbound call.

2.5.1.1 Use Service

To enable or disable the realm.

2.5.1.2 User Number

Extension number or telephone number of SIPaccount.

2.5.1.3 Authorized Name

Username of SIPaccount

2.5.1.4 Password

Password of SIP account

2.5.1.5 Proxy IP

IP address of Proxy Server that enables SIP connection to SIP domain. If there is no real proxy server between IP phone and domain, please input the IP address of domain into this field.

2.5.1.6 Domain

IP address of domain who are the service provider for SIP service or PBX server that SIP phone will connect to.

2.5.1.7 Outbound Proxy

IP address of Proxy Server that enables outbound call. If there is no additional server for outbound call, user can leave it blank or fill the IP address of domain.

2.5.1.8 SIP Expire Time

How long SIP phone is expired and should renew the registration status.

2.5.1.9 Status

Show registration status of this realm.

Please click the "Apply" button after finishing.

Service Domain Settings

You could set information of service domains in this page.

First Realm

Use Service	Enable ▾
User Number	6002
Authorized Name	6002
Password	●●●●
Proxy IP	192.168.204.55
Domain	192.168.204.55
Outbound Proxy	192.168.204.55
SIP Expire Time	300 (20~65535)
Status	Register

2.5.2 Port Settings

To change the port for SIP and RTP connection.

2.5.2.1 SIP Port

Default is 5060, user can change to any port number from 100 to 65535

2.5.2.2 RTP Port

Usually from 10000 to 20000, default is 20000, user can change to any port number from 100 to 65535.

Please click the "Apply" button after finishing.

Port Settings

You could set the port number in this page.

SIP Port	<input type="text" value="5060"/>	(100 ~ 65535)
RTP Port	<input type="text" value="20000"/>	(100 ~ 65535)

2.5.3 Codec Settings

2.5.3.1 Codec Priority

Setting for priority of preferred codecs. If first one is unsupported by domain, the second one will be automatically used and so on.

2.5.3.2 RTP Packet Length

To set the millisecond of RTP Packet Length for both codec G711 and G729.

2.5.3.3 iLBC 15K2

To enable or disable iLBC deploys 15k2 rate.

2.5.3.4 G723 5.3K

To enable or disable G723 deploys 5.3K rate.

2.5.3.5 Voice VAD

To enable or disable Voice Activation Detection.

2.5.3.6 Voice CNG

To enable or disable Comfort Noise Generator.

Codec Settings

You could set the codec settings in this page.

Codec Priority

Codec Priority 1	G.729 ▾
Codec Priority 2	G.711a ▾
Codec Priority 3	G.711u ▾
Codec Priority 4	G.723 ▾
Codec Priority 5	iLBC ▾
Codec Priority 6	G.722 ▾

RTP Packet Length

G.711 & G.729 20ms ▾

iLBC 15K2

iLBC 15K2 ☒ Off ☐ On

G.723 5.3K

G.723 5.3K ☒ Off ☐ On

Voice VAD

Voice VAD ☒ Off ☐ On

Voice CNG

CNG ☒ Off ☐ On

Apply

Reset

2.5.4 Codec ID Settings

To change RFC2833 event ID. When SIP phone communicate with other SIP device, if the codec ID that other SIP device deployed is non-standard, problem will occur. User can adjust the codec ID of SIP phone from default to a matching one. Please click the “Apply” button after finishing.

Codec ID Settings

You could set the value of Codec ID in this page.

Codec Type	ID	Default Value
RFC 2833 ID	<input type="text" value="101"/>	(95~127) <input checked="" type="checkbox"/> 101

2.5.5 DTMF Settings

There are three common standards of DTMF, RFC2833, Inband and SIP Info. User should consult to service provider which one is correct and suitable to setting. Please click the “Apply” button after finishing.

DTMF Settings

You could set the DTMF settings in this page.

DTMF ☒ RFC_2833 ☐ Inband_DTMF ☐ Send_DTMF_SIP_Info

Delay Time (60 ~ 300 msec)

2.5.6 STUN Settings

If SIP phone is behind NAT, user should use a STUN server which is outside current local network, to translate the IP address from the private to public. Setup of STUN server allows SIP phone connect to SIP service provider outside current local network. There are some STUN servers which are free of charge over internet. The famous one is stun.xten.com. Please click the “Apply” button after finishing.

STUN Settings

You could set the IP of STUN server in this page.

STUN ☒ Off ☐ On

STUN Server

STUN Port (100 ~ 65535)

2.5.7 Other Settings

2.5.7.1 Voice QoS (Diff-Serv)

Voice Quality of Service allows user set the priority of voice packet passthrough the router or firewall which connects to Internet. The higher value is set, the higher priority it gets.

2.5.7.2 SIP QoS (Diff-Serv)

SIP Quality of Service allows user set the priority of SIP packet passthrough the router or firewall which connects to Internet. The higher value is set, the higher priority it gets.

2.5.7.3 Send Keep Alive Packet

Keeping send UDP packets from SIP phone to domain or any other device that SIP phone connected to, in order to keep the transmission between two devices alive.

2.5.7.4 Keep Alive Period

Time interval that SIP phone sends packet to domain.

2.5.7.5 Jitter Buffer Max

Maximum millisecond for Jitter Buffer to discard delay packets.

2.5.7.6 Anonymous Call Rejection

Reject any inbound call which does not submit a caller ID.

2.5.7.7 Auto Answer

Automatically answer a call by handsfree.

2.5.7.8 Auto Answer Time Out

How long a call ring until it is automatically answered.

2.5.7.9 Subscribe for MWI

SubscribeMessageWaiting Indicator function from server in according with standard RFC3842.If there is a voice message for registered extension, WMI LED will keep on lighting until message is listened.

Other Settings

You could set other settings in this page.

Voice QoS (Diff-Serv)	<input type="text" value="40"/>	(0 ~ 63)
SIP QoS (Diff-Serv)	<input type="text" value="40"/>	(0 ~ 63)
Send Keep Alives Packet	<input checked="" type="radio"/> Off <input type="radio"/> On	
Keep Alives Period	<input type="text" value="60"/>	(15 ~ 250 sec)
Jitter Buffer Max	<input type="text" value="150"/>	(70~250 ms)
Anonymous Call Rejection	<input checked="" type="radio"/> Off <input type="radio"/> On	
Auto Answer	<input checked="" type="radio"/> Off <input type="radio"/> On	
Auto Answer Time Out	<input type="text" value="2"/>	(0~10 sec)
Subscribe for MWI	<input type="radio"/> Off <input checked="" type="radio"/> On	

2.6 Others

User can configure the IP Phone by config file in XML format and store system log of IP phone to a remote server.

2.6.1 Auto Config

To configure the IP phone by config file via local PC, tftp, ftp or http server. User should input IP address of Config Server, Path, Username and Password. To update the config file from server to IP phone, please click "Update" button. To upload the config file from IP phone to server, please click "Upload" button. If update the IP phone from config file in PC, just browse the location of config file in PC and click "Apply" button. The default name of config file that can be recognized is "config.xml".

2.6.1.1 Auto Configuration

To choose one of config method.

2.6.1.2 Config Sever

The IP address of either HTTP or FTP/TFTP server.

2.6.1.3 HTTP File Path

The exact path of config file in HTTP server.

2.6.1.4 FTP Username

Username of FTP account.

2.6.1.5 FTP Password

Password of FTP account.

2.6.1.6 FTP File Path

The exact path of config file in FTP or TFTP server.

2.6.1.7 Download XML

After input all required information, user can download the config file from server by right click the button of "Download XML". For Firefox user, choose "Save Link As". For Internet Explorer user, choose "Save Target As".

2.6.1.8 Name of config file with MAC address

IP Phone can be configured by different config file. It lets administrator more easily to manage a group of IP Phones with individual setting. If the name of config file in HTTP, FTP or TFTP server is exactly the MAC address of IP Phone, for example, MAC address of IP Phone is 2e:2e:d3:9d:31:72, name of config file is 2e2ed39d3172.xml, user can update the configuration by only clicking "Update" button. IP Phone will follow previous saved setting in the Auto Config page to seek for the config file which is totally matched to its MAC address and then update itself.

Auto Configuration Settings

You could enable/disable the auto configuration settings in this page.

Auto Configuration	<input checked="" type="radio"/> off <input type="radio"/> tftp <input type="radio"/> ftp <input type="radio"/> http	
Config Server	<input type="text"/>	Exp. 60.35.187.30
HTTP File Path	<input type="text"/>	Exp. /download/
FTP Username	<input type="text" value="anonymous"/>	
FTP Password	<input type="text"/>	
FTP File Path	<input type="text"/>	Exp. /file/load

Local PC

<input type="text"/>	<input type="button" value="Browse..."/>
<input type="button" value="Apply"/>	

Download XML (The right mouse button click on the download button, select 'save target as')

2.6.2 Advanced Settings

User can write system log of IP phone to a remote Log Server with syslogd running. Please input the IP address of Log Server, then click "Apply" button.

Advanced Settings

You could change advanced settings in this page.

System Log Server	<input type="text"/>
System Log Type	None ▼
<input type="button" value="Apply"/> <input type="button" value="Reset"/>	

2.7 Update

2.7.1 Auto Update

The IP phone will automatically update its firmware by checking the firmware version via TFTP, FTP or HTTP after rebooting. If the number of version file is larger than current version, IP phone will download the update file and upgrade itself. Otherwise, it will do nothing. The update procedure will take about several minutes, please make sure power supply to IP phone won't be interrupted during update.

2.7.1.1 Upgrade Mode

To choose one of upgrade method.

2.7.1.2 Upgrade Addr

The IP address of either HTTP or FTP/TFTP server.

2.7.1.3 FTP User

Username of FTP account.

2.7.1.4 FTP Password

Password of FTP account.

2.7.1.5 Update File

The name of firmware file in remote server.

2.7.1.6 FTP File Path

The exact path of firmware file in TFTP or FTP server.

2.7.1.7 HTTP File Path

The exact path of firmware file in HTTP server.

Auto Update

You must set parameter for Auto Update in this page.

Upgrade Mode	<input checked="" type="radio"/> off <input type="radio"/> ftp <input type="radio"/> tftp <input type="radio"/> http	
Upgrade Addr	<input type="text" value="192.168.18.200"/>	
Ftp User	<input type="text" value="anonymous"/>	
Ftp Pwd	<input type="text"/>	
Update File	<input type="text" value="JX840-0906090.tar.bz2"/>	
FTP File Path	<input type="text"/>	Exp. /download/
HTTP File Path	<input type="text"/>	

2.7.2 Update System

To manually update the firmware of IP phone via TFTP, FTP, HTTP or local PC. If update the IP phone from firmware file in PC, just browse the location of firmware file in PC and click "Apply" button.

2.7.2.1 Upgrade Mode

To choose one of upgrade method.

2.7.2.2 Upgrade Addr

The IP address of either HTTP or FTP/TFTP server.

2.7.2.3 FTP User

Username of FTP account.

2.7.2.4 FTP Pwd

Password of FTP account.

2.7.2.5 Update File

The name of firmware file in remote server.

2.7.2.6 FTP File Path

The exact path of firmware file in TFTP or FTP server.

2.7.2.7 HTTP File Path

The exact path of firmware file in HTTP server

Update System

You can update system in this page.

Upgrade Mode	<input type="radio"/> ftp <input checked="" type="radio"/> tftp <input type="radio"/> http	
Upgrade Addr	<input type="text" value="192.168.18.200"/>	
Ftp User	<input type="text" value="anonymouS"/>	
Ftp Pwd	<input type="text"/>	
Update File	<input type="text" value="JX840-0906090.tar.bz2"/>	
FTP File Path	<input type="text"/>	Exp. /download/
HTTP File Path	<input type="text"/>	

Local PC

<input type="text"/>	<input type="button" value="Browse..."/>
<input type="button" value="Apply"/>	

2.7.3 Default Settings

To restore the default setting. Please click "Apply" button after finishing.

Restore Default Settings

You could click the restore button to restore the factory settings.

Restore default settings:

2.8 System Auth

To change the user name and password of Super User.

System Authority

You could change the login password in this page.

Super User PWD

Confirm Password

2.9 Save Change

Some config items should be saved and the IP phone should be rebooted until the setting takes effect. Once user click the link of "Save Change", a page with "Reboot" button will appear to remind user reboot the IP phone.

Save Change

You have to save changes to effect them.

Copyright and Trademarks

Specifications are subject to change without notice.

Inn-Phone is a registered trademark or trademark of Med-Pat, Inc. and/or its affiliates in the U.S.A. and certain other countries.

Copyright © 2012 Med-Pat, Inc. All rights reserved.

Under the copyright laws, this manual may not be copied, in whole or in parts, without the written consent of Med-Pat

.

Every effort has been made to ensure that the information in this manual is accurate. Med-Pat is not responsible for printing or clerical errors

Other brands and product names are trademark or registered trademarks of their respective holders.

Contact Information

Need to contact Inn-Phone?

31 Riordan Place, Shrewsbury, NJ 07702, USA

Visit us online for information on our latest products and updates to your existing products at:

<http://www.inn-phone.com>

Can't find information about a product you want to buy on the web? Do you want to know more about networking with Inn-Phone products? Give our advice line a call at:

(877) 467-7864

Or fax your request in to:

(888) 962-3728

If you experience problems with any Inn-Phone products, you can call us toll-free at:

(877) 467-7864

Don't wish to call? You can e-mail us at:

info@inn-phone.com

