



Two Handset VOIP Phone

Model: D2200-SIP

User Manual



Version 2.0



(QR code)

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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

Two RJ-45Networking interfaces support 10/100Mps FastEthernet. User can connect WAN port to ADSLor Soft PBX,the LANport to PCcomputer. The default setting of WAN port is a DHCPclient. The IP address of LANport is 192.168.2.1.

One 3.5mm Headsetjack is used to connect only Apple Type Headphone with Mic.

(User can only control the headset volume by the Volume-Control keys on the top of the phone)

One mute LED, one on-hold LED, one hand-free LED and one LED that combine functions of ringer and message-waiting.

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



1.2 Software Overview

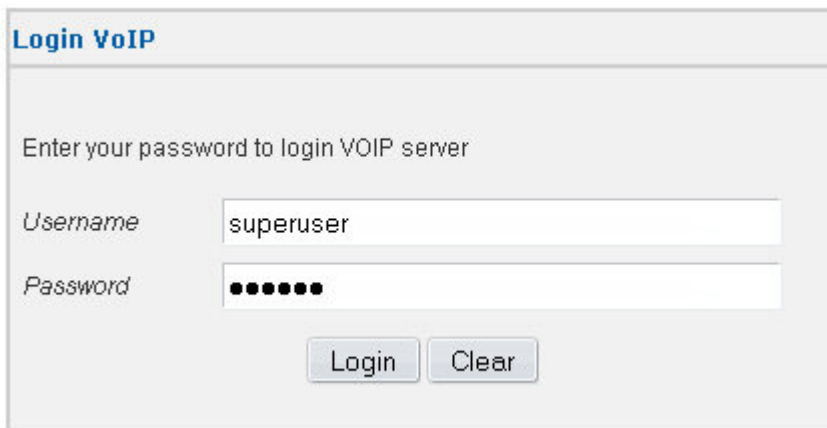
Network Protocol	Codec
z SIP 2 (RFC3261)	z G711 aLaw
z IP/TCP/UDP/RTP/SNTP	z G711 uLaw
z DHCPClient/PPPoE Client/PPTP Client	z G722
z LLDP-MED	z G723
z HTTP Server	z G729
z DNSClient	z iLBC
Voice Quality	Tone
z Comfortable noise generator	z Ring Tone
z Voice Activity Detector	z 350Hz/440Hz Dial Tone
z Voice QoS	z 400Hz Dial Tone
z SIP QoS	z Busy Tone
z Jitter Buffer	
Phone Function	Call Function
z Speedial key	z Call Hold
z Volume Adjustment	z Call Mute
z 8 types of ringing melody	z Call Waiting
z Do Not Disturb	z Call Forward
z Schedule Alarm	
IP Assignment	DTMF
z Static IP	z RFC2833
z DHCP	z Inband
z PPPoE	z SIPInfo
NAT Traversal	Firmware Upgrade
z STUN	z TFTP
	z FTP
	z HTTP
	z Local Computer
SIP Server	Configuration
z Up to six SIPaccount	z Web Browser
z Outbound Proxy	

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 DHCPserver. If there is no DHCPservice in local network, please connect computer to the LAN port by using Ethernet cable. Setup the IP address of computer as the same subnet of the LAN port. Inputting <http://192.168.2.1> (the default LAN IP address of IP Phone) in URL of your web browser, now user can reach the page of Login.

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 SIPPhone.



Login VoIP

Enter your password to login VOIP server

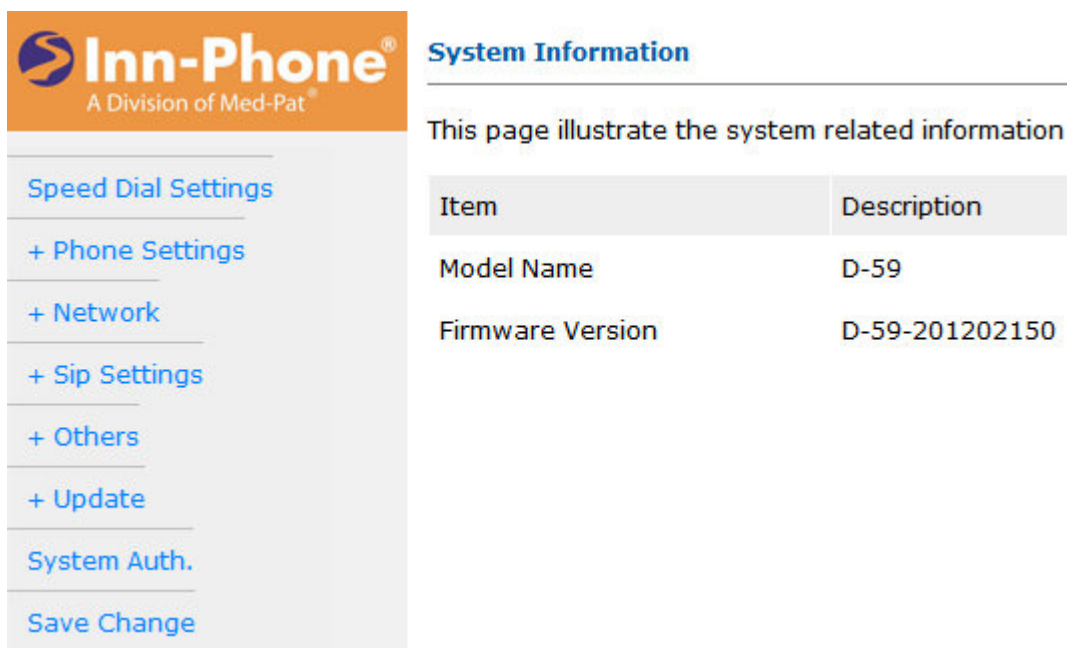
Username:

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.



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System Information

This page illustrate the system related information

Item	Description
Model Name	D-59
Firmware Version	D-59-201202150

Speed Dial Settings

+ Phone Settings

+ Network

+ Sip Settings

+ Others

+ Update

System Auth.

Save Change

2.3 Speed Dial Settings

2.3.1 Setting Speed Dial via web page

In SpeedDial Phone List, user can add or delete SpeedDial number. Maximum 10 entries can be setup in SpeedDial Phone List.

To add a phone number into the SpeedDial Phone List, user needs to input position, name, and URL. URL can be phone number or IP address. After finishing, click “Add Phone” button.

To delete a group of preset phone number, first user should select the number by clicking the select button, and then click “Delete Selected” button to delete selected number.

To delete all numbers, simply click “Delete All” button, a dialogue window will pop-up to let you confirm. Click “OK” button to delete all numbers.

Speed Dial Phone List

You could set the speed dial phones in this page.

Position	Name	URL	Select
0	Tom Buehl	192.168.204.55	<input type="checkbox"/>
1			<input checked="" type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Add New Phone

Position: (0~9)
 Name:
 URL:

2.3.2 Setting Speed Dial by keypad

The pattern for setting SpeedDial is ****73*** + position + ***** + phone number that user wants to add to speed dial list + **#** key to end the setup.

For example, dialing ****73*0*18004813293#** can add 18004813293 to position 0. Dialing ****73*1*12345678#** can add 12345678 to position1. Setting Speed Dial will only be complete after busytone is heard

2.4 Phone Settings

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

2.4.1 Call Forward

2.4.1.1 All Forward

All incoming call will be forward to the number that is filled. Pleaseinput the name in the name field and the phone number or IP addressin the URLfield.

2.4.1.2 Busy Forward

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

2.4.1.3 No Answer Forward

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

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

Forward Settings

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

Forward Type	<input type="text" value="All Forward"/>
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)
<input type="button" value="Apply"/> <input type="button" value="Reset"/>	

2.4.2 SNTPSettings

User can enable SNTPSetting 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	<input type="radio"/> Disable <input checked="" type="radio"/> Enable
SNTP Server	<input type="text" value="time.windows.com"/>
Time Zone	(GMT +08:00) Beijing, Chongqing, Hong Kong, Manila, Perth, Singapore, Taipei, Urumq <input type="text"/>
Local Time	<input type="text" value="2011"/> : <input type="text" value="01"/> : <input type="text" value="01"/> <input type="text" value="00"/> : <input type="text" value="00"/> (Year:Month:Day Hour:Min)
<input type="button" value="Apply"/> <input type="button" value="Reset"/>	

2.4.3 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	<input type="text" value="3"/>	(1~7)
Speaker Output	<input type="text" value="1"/>	(1~7)
Ringer Volume	<input type="text" value="2"/>	(1~7)

2.4.4 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 Off On

Ringer Type

2.4.5 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 Off On

DND Period Off On

From : (Hour:mm)

To : (Hour:mm)

2.4.6 Dial Plan

2.4.6.1 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.

2.4.6.2 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”.

2.4.6.3 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.

2.4.6.4 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#.

2.4.6.5 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.

2.4.6.6 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

2.4.7 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

2.4.8 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

2.5 Network

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

2.5.1 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.

2.5.1.1 Fixed IP Settings

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

2.5.1.2 DHCP Settings

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

2.5.1.3 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.

2.5.14 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.

2.5.15 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	

2.5.16 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

2.5.2 LAN Settings

Set network parameters for LAN port. user should refer to current network environment to configure the IP phone properly.

2.5.2.1 LAN Mode

Bridge Mode means the WAN port and LAN port are transparent. The IP address of the device which connects to LAN port, will be treated as same to WAN port over Internet. NAT Mode means the IP phone will act as a router, the IP address of LAN port will be translated to the same of WAN port. If user deploy NAT mode, parameters for NAT should be inputted. The LAN port can be disabled by selecting Disable option in LAN Mode.

2.5.2.2 NAT & DHCP Server

The subnet of LAN port and the device that connects to LAN port must be the same and must be different to WAN port. There is a feature of DHCP server in LAN port, user can enable it for automatically assigning IP address to the device that connected to LAN port.

LAN Settings

You could configure the LAN settings in this page.

LAN Mode Nat Bridge Disable

Nat

IP Address
 Net Mask
 DHCP Server
 IP Pool Start
 IP Pool End
 MAX Leases (1~250)

Apply

Reset

2.5.3 VLAN Settings

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

2.5.3.1 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..

2.5.3.2 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	<input checked="" type="radio"/> Off <input type="radio"/> On
VID (802.1Q/TAG)	<input type="text" value="136"/> (0 ~ 4094)
User Priority (802.1P)	<input type="text" value="0"/> (0 ~ 7)

2.5.4 DMZ Settings

To enable Demilitarized Zone by inputting the DMZ Host IP and Port. After enabled it, the IP phone will act as a DMZ host, external traffic will go through from the WAN port to the host which connected to the LAN Port. For example, IP address of WAN Port is 123.123.0.1, IP address of LAN Port is 192.168.1.1, IP address of the device (DMZ Host) which connected to LAN port is 192.168.1.2, DMZ Port is 90. Internet users can access 192.168.1.2 (DMZ Host) by connecting to 123.123.0.1 port 90.

DMZ Settings

You could configure your demilitarized zone settings in this page.

DMZ Off On

DMZ Host IP

DMZ Port (1 ~ 65535)

2.5.5 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 IPPhoneis idle.

PPTP Settings

You could set PPTP settings in this page.

PPTP Off On

PPTP Server

PPTP Username

PPTP Password

2.5.6 LLDPSettings

To enable/disable Link Layer Discovery Protocol. User can also set the interval for sendinginformation by IPPhoneafter enabling it.

LLDP Settings

You could set the LLDP in this page.

LLDP	<input type="text" value="Enable"/>
Packet Interval	<input type="text" value="120"/> (1 ~ 3600 sec)
<input type="button" value="Apply"/> <input type="button" value="Reset"/>	

2.5.7 Manually set Network Mode of WAN

User can manually set the two kind of network mode of SIPPhone by inputting some digits of keypad.

2.5.7.1 Fixed IP Mode

To press 20120912*# during SIPPhone is idle. SIPPhone will go to reboot and use the fixed IP address that was saved in configuration before.

2.5.7.2 DHCP IP Mode

To press 20120913*# during SIPPhone is idle. SIPPhone will go to reboot and use the DHCP mode.

2.5.8 DHCP Option 66

When user deploys a large number of SIPPhones on network, user can use DHCP Option 66 to automatically instruct the SIPPhone with the provisioning URL.

DHCP Server with Option 66 will instruct individual SIP Phone the URL path of individual configuration file that was stored in TFTP Server during SIPPhone acquires the IP address from it. The configuration file is named by its MAC address. The format of file name is something like 2E2ED39BD26D.xml.

2.6 SIP Settings

In order to let SIP phone work properly, user should setup SIP Service Domain, SIP Port, Codec, Codec ID, DTMF, STUN server and others. Some information of them user should obtain from SIP service provider.

2.6.1 Service Domain

User can setup total six SIP accounts for receiving inbound calls and use the first realm for outbound call.

2.6.1.1 Use Service

To enable or disable the realm.

2.6.1.2 User Number

Extension number or telephone number of SIP account.

2.6.1.3 Authorized Name

Username of SIP account

2.6.1.4 Password

Password of SIP account

2.6.1.5 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.

2.6.1.6 Domain

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

2.6.1.7 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.

2.6.1.8 SIP Expire Time

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

2.6.1.9 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

2.6.2 Port Settings

To change the port for SIP and RTP connection.

2.6.2.1 SIP Port

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

2.6.2.2 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	5060	(100 ~ 65535)
RTP Port	20000	(100 ~ 65535)

2.6.3 CodecSettings

2.6.3.1 CodecPriority

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

2.6.3.2 RTPPacket Length

To set the millisecond of RTPPacketLength for both codec G711 and G729.

2.6.3.3 iLBC 15K2

To enable or disable ILBC deploys 15k2 rate.

2.6.3.4 G723 5.3K

To enable or disable G723 deploys 5.3K rate.

2.6.3.5 Voice VAD

To enable or disable Voice Activation Detection.

2.6.3.6 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

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

2.6.4 Codec ID Settings

To change RFC2833event ID. When SIPphone communicate with other SIPdevice, if the codec ID that other SIPdevice deployed is non-standard, problem will occur. User can adjust the codec ID of SIPphone 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

2.6.5 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)

2.6.6 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)

2.6.7 Other Settings

2.6.7.1 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.

2.6.7.2 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.

2.6.7.3 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.

2.6.7.4 Keep Alive Period

Time interval that SIP phone sends packet to domain.

2.6.7.5 Jitter Buffer Max

Maximum millisecond for Jitter Buffer to discard delay packets.

2.6.7.6 Anonymous Call Rejection

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

2.6.7.7 Auto Answer

Automatically answer a call by handsfree.

2.6.7.8 Auto Answer Time Out

How long a call ring until it is automatically answered.

2.6.7.9 Clear redial in 10 min

To clear memory of a stored phone number for redialing after 10 minutes.

Please click the "Apply" button after finishing.

2.6.7.10 Subscribe for MWI

Subscribe Message Waiting 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.

2.6.7.11 Session Switch

To enable / disable Session Switch Feature which is based on Generalized Multiprotocol Label Switch (RFC4208)

2.6.7.12 Session Time

To set minimum Session Time for Session Switch.

2.6.7.13 Support 100rel

To enable / disable the module that provides management of Reliability of Provisional Responses.

2.6.7.14 Support Update Method

To enable / disable the update request for session.

2.6.7.15 Rport

To enable / disable support for response-port parameter in header (RFC3581).

2.6.7.16 Use TelURL

To enable / disable URL for telephone call (RFC2806)

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="5"/>	(0~10 sec)
Clear redial in 10min	<input type="radio"/> Off <input checked="" type="radio"/> On	
Subscribe for MWI	<input type="radio"/> Off <input checked="" type="radio"/> On	
Session Switch	<input type="text" value="Disable"/>	
Session Time (Min=90s)	<input type="text" value="1800"/>	
Support 100rel	<input type="text" value="Disable"/>	
Support Update Method	<input type="text" value="Disable"/>	
Rport	<input type="text" value="Enable"/>	
Use Tel URI	<input type="text" value="Disable"/>	

2.7 Others

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

2.7.1 ConfigSpeed Dial

To configure SpeedDial by config file via local PC, tftp, ftp or http server. The file name must be "speedbook.xml", any other name will be ignored. Format is as follows:

```
<?xml version="1.0" encoding="UTF-8"?>
<phonebookrecord xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
xsi:noNamespaceSchemaLocation="speedphonebook.xsd">
  <phonebookinfo>
    <phonebookindex>0</phonebookindex>
    <phonebookname>TomBuehl</phonebookname>
    <phonenumber>8001</phonenumber>
  </phonebookinfo>
  <phonebookinfo>
    <phonebookindex>1</phonebookindex>
    <phonebookname>Doug Zagha</phonebookname>
    <phonenumber>8002</phonenumber>
  </phonebookinfo>
</phonebookrecord>
```

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.

2.7.1.1 ConfigSpeed Dial

To choose one of config method.

2.7.1.2 ConfigSever

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

2.7.1.3 HTTPFile Path

The exact path of config file in HTTP server.

2.7.1.4 FTPUsername

Username of FTP account.

2.7.1.5 FTPPassword

Password of FTPaccount.

2.7.1.6 FTPFile Path

The exact path of config file in FTPor TFTPserver.

Speed Dial Configuration

You could enable/disable the config speed dial in this page.

Config Speed Dial	<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"/>	

2.7.2 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".

2.7.2.1 Auto Configuration

To choose one of config method.

2.7.2.2 ConfigSever

The IP address of either HTTPor FTP/TFTPserver.

2.7.2.3 HTTPFile Path

The exact path of config file in HTTPserver.

2.7.2.4 FTPUsername

Username of FTPaccount.

2.7.2.5 FTPPassword

Password of FTPaccount.

2.7.2.6 FTPFile Path

The exact path of config file in FTPor TFTPserver.

2.7.2.7 DownloadXML

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".

2.7.2.8 Name of configfile 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 exact 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')

2.7.3 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"/>	

2.8 Update

2.8.1 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.

2.8.1.1 Upgrade Mode

To choose one of upgrade method.

2.8.1.2 Upgrade Addr

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

2.8.1.3 FTP User

Username of FTP account.

2.8.1.4 FTP Password

Password of FTP account.

2.8.1.5 Update File

The name of firmware file in remote server.

2.8.1.6 FTPFile Path

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

2.8.1.7 HTTPFile 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"/>

2.8.2 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.

2.8.2.1 Upgrade Mode

To choose one of upgrade method.

2.8.2.2 Upgrade Addr

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

2.8.2.3 FTP User

Username of FTP account.

2.8.2.4 FTP Pwd

Password of FTP account.

2.8.2.5 Update File

The name of firmware file in remote server.

2.8.2.6 FTPFile Path

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

2.8.2.7 HTTPFile 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"/>	

2.8.3 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:

2.9 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	<input type="password"/>
Confirm Password	<input type="password"/>

2.10 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.

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