
Citel. The VoIP Migration Company™



Installation and User Guide

Table of Contents

Citel IP Phone C4110	4
Installation and User Guide.....	4
Citel IP Phone C4110 Installation.....	5
Connecting your C4110 Phone	6
1. Initialization	7
1.1 To set Network setting	7
1.2 To set SIP setting.....	11
1.3 Customizing your C4110 IP Phone	15
1.3.1 To set Ringer Type	15
1.3.2 To set screen setting sublist.....	15
1.3.3 To set Volume Control sublist	15
1.3.4 To change the display Language	16
1.3.5 To set Greeting Message	16
1.3.6 To set Memory key setting.....	17
1.3.7 To Model Information Sublist.....	17
1.3.8 To reboot system	17
2. Configuration with web browser	18
2.1 User verification.....	18
2.2 Basic Status.....	19
2.2.1 Running Status	19
2.2.2 Wizard	19
2.3 Network configuration	21
2.3.1 Wide area network (WAN).....	21
2.3.2 LAN Configuration	22
2.4 VOIP Configuration	24
2.4.1 SIP Line advanced configuration	25
2.4.2 IAX2 configuration	27
2.4.3 STUN Set	28
2.4.4 Dial peer configuration	29
2.5 DSP configuration.....	31
2.5.1 Call Service configuration.....	32
2.5.2 Digital Map configuration.....	33
2.5.3 Phone Book configuration	34
2.5.4 Function Key configuration	34
2.6 Phone Maintenance	36
2.6.1 Auto Update configuration.....	36
2.6.2 Syslog configuration	37
2.6.3 Save Config.....	38

2.6.4 Upload WEB page.....	38
2.6.5 FTP download	39
2.6.7 Account management.....	40
2.6.8 Reboot.....	41
2.7 Security	42
2.7.1 MMI Filter configuration	42
2.7.2 Firewall configuration.....	42
2.7.3 VPN Tunnel configuration	44
2.8 Logout.....	44
3. Citel IP Phone C4110 Basic Operation	45
3.1 Making a call.....	45
3.2 Answering a call.....	45
3.3 Using Hold and Mute.....	45
3.4 Volume adjustment.....	45
3.5 Redial	45
3.6 Call lists	46
3.7 Auto dial/Feature.....	46
3.8 Auto dial/Feature Access.....	46
3.9 Voice mail	46
3.10 N'pad(Note Pad).....	46
3.11 Call transfer (Blind and Consult)	47
3.12 Call waiting.....	47
3.13 Conference calling	47
3.14 Directory /Phone Book.....	47
3.15 Auto dial/Feature Access.....	48
4 Trouble shooting	49

Citel IP Phone C4110

Installation and User Guide

Citel IP Phone C4110 Introduction

Internet Protocol (IP) telephone is the next generation of telecommunication services that integrates voice and data capabilities to enhance everyday communications and increase productivity, it takes advantage of new high-bandwidth central office technologies to provide service, anywhere, at anytime instantly.

Today, many small, medium and large businesses in the world are deploying IP telephone systems to reduce the cost of operations and increase productivity of workers.

The design of the C4110 IP Phone includes one-button access to features like Speakerphone, Phonebook, Call history, Transfer, Conference, Auto dial, etc. And supports the unique requirements of various users.

Additional features and functionality of C4110 IP Phone includes a display screen, soft keys, Note Pad (N'pad), etc.

Citel IP Phone C4110 Features

- SIP & IAX2 Support
- Wide ranges of codec supported G.729, G.723, Wideband G.722 & G.711 a/ulaw.
- Three-line display screen with backlight and adjustable contrast
- Two lines supported with corresponding LED indicators
- Three screen-driven soft keys
- 4 programmable buttons for auto dial and specific function access
- High quality Full-duplex speaker phone for hands free calls
- Two port 10/100M Ethernet switch for shared PC access
- 8 button memory key expansion model (Option)
- RJ9 headset connection jack
- Hearing Aid Compatibility(HAC)
- In line power support (PoE/802.3af), which eliminates power adapters



Figure 1 Features and Keys

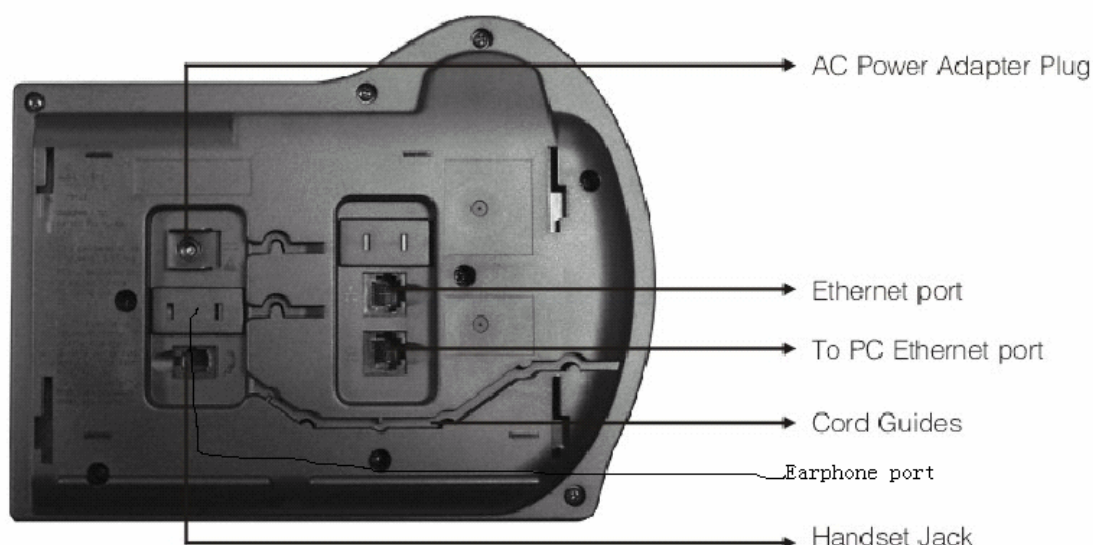
Citel IP Phone C4110 Installation

The C4110 IP Phone must be set up and configured prior to its first use. If your system administrator has not already set up your C4110 Phone, please refer to the C4110 User Guide for basic installation and physical set up configuration information. Please contact your service provider or regional distributor.

Caution:1. Use your IP Phone C4110 with the supplier approved AC adapter for this model of phone only.

Caution:2 . Severe damage to your IP Phone will occur if this phone is plugged into PSTN telephone line and an ISDN connection. Consult your system administrator to ensure that you are plugging your IP Phone into a 10/100 Base-T Ethernet Jack.

Connecting your C4110 Phone



Follow the following steps to connect the C4110 IP Phone

Step1: Pull upward the stand through the slots on the base, remove the stand.

Step2: Connect one end of the handset cord to the handset jack with symbol on the telephone base and connect the other end of the handset cord to the handset. Route the cord through the cord guide channel.
Slide the stand back until it clicks into the lock position.

Step3: Connect one end of the supplied Ethernet cable with RJ45 connector to the Ethernet port on the telephone base and connect the other end of the cable to an IP voice network jack through the catch on the telephone stand.

Step4: If you are connecting your PC Ethernet through the phone, connect one end of the PC Ethernet cable to your phone using the RJ45 connector marked with the symbol on the telephone base, and thread it through the catch on the stand. Connect the other end to the LAN port on your PC.

Step5: Plug AC power adapter into the base of the telephone, and plug the adapter into the nearest AC power outlet. If you are using PoE (Power over Ethernet), the AC power adapter is not needed.

Step6: To wall mount the C4110 IP Phone, you need to remove the stand and reverse it, then slide it back. Use the wall mounting holes on the stand to attach the IP Phone on the wall.

1. Initialization

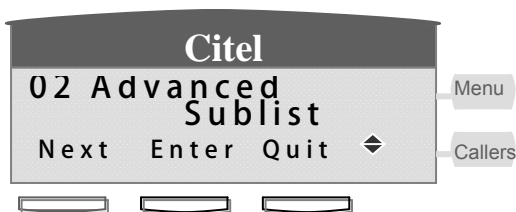
1.1 To set Network setting

C4110 can provide web setting mode and phone setting mode.

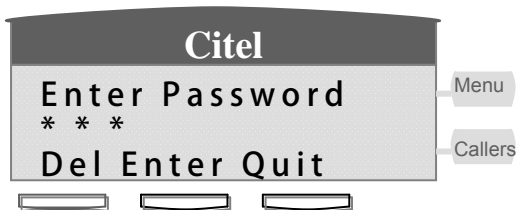
1. Setting through phone setting

Use the following procedure to configure the Network setting.

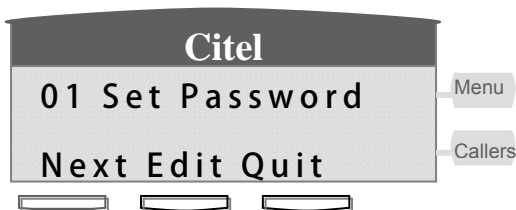
1. Press "Menu" into main menu, and use **Up** and **Down** navigation key or **Next** key to select user mode to "Advanced Sublist" Mode.



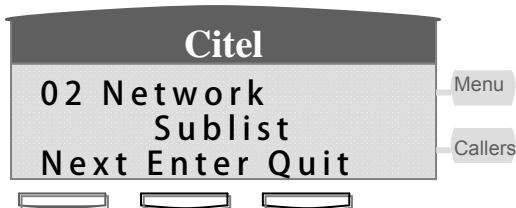
2. Press "Enter" into 'Advanced Sublist' menu.



3. Input the default password : 1 2 3

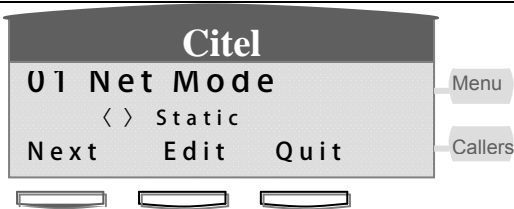


4. Press use **Up** and **Down** navigation key or **Next** key to select user mode to "Network Setting" Mode



To set the Net Mode

1. Press **Edit** use **Left** and **Right** key to select IP type as Static, DHCP or PPPOE.



2. Press **Save** to save your setting.
3. Press **Quit** key back to last menu and non-save.

Remark :

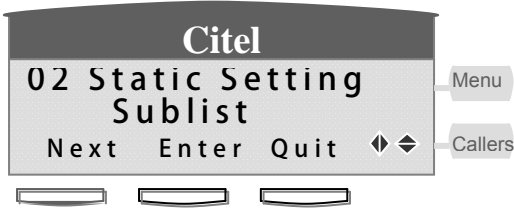
If “DHCP” mode is selected, Serve will use static distribution IP address, and it means the networking setting is done.

If “PPPoE” mode is selected, you need access “PPPoE Setting” mode and set account No#, Passwords and related information.

If “Static” mode is selected, you need access “Static Setting” mode and set related information.

To set the Static setting

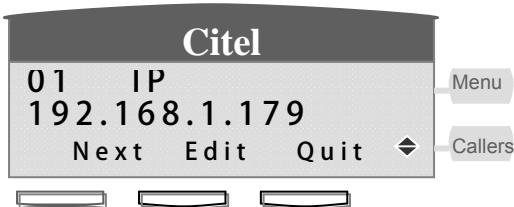
1. Press **Enter** into Static IP editing menu.



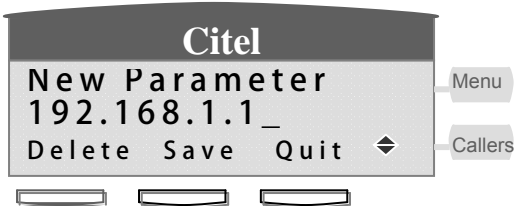
2. Press **Next** or **Up** and **Down** key to switch between “ Static IP ”、 “ S tatic Netmask ”、 “ S tatic gateway ” “ S tatic DNS ” menu.
3. Press “Quit” B ack to last menu.

Static IP editing

1. Press **Edit** into Static IP editing menu.



2. Input the Static IP address in the IP editing line, the ‘*’ means ‘.’.

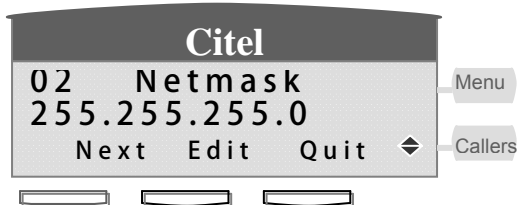


3. Press “Delete ” to delete character in the editing line.

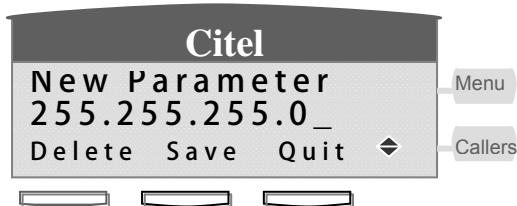
4. Press “Save” to save the IP address you input.
5. Press “Quit” back to last menu and non-save.

Static Netmask editing

1. Press **Edit** into StaticNetmask editing menu.



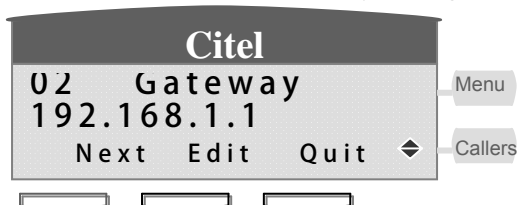
2. Input the Static Netmask in the editing line, the ‘*’ means ‘.’.



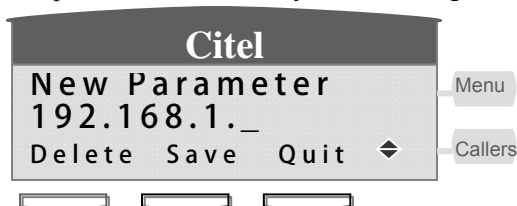
3. Press “Delete ” to delete character in the editing line.
4. Press “Save” to save the IP address you input.
5. Press “Quit” back to last menu and non-save.

Static Gateway editing

1. Press **Edit** into Static Gateway editing menu.



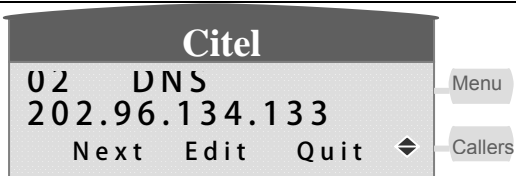
2. Input the Static Gateway in the editing line, the ‘*’ means ‘.’.



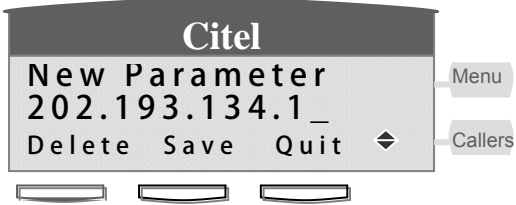
3. Press “Delete ” to delete character in the editing line.
4. Press “Save” to save the Gateway you input.
5. Press “Quit” back to last menu and non-save.

Static DNS editing

1. Press **Edit** into Static DNS editing menu.



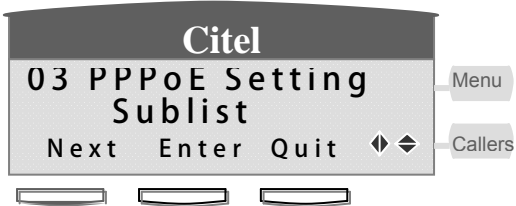
- Input the Static DNS the editing line, the '*' means '.'.



- Press "Delete" to delete character in the editing line.
- Press "Save" to save the DNS you input.
- Press "Quit" back to last menu and non-save.

To set the PPPoE setting

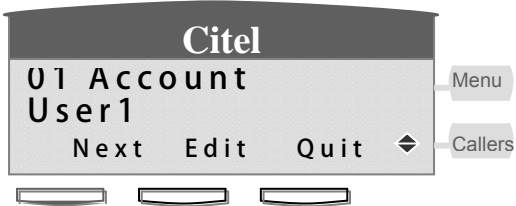
- Press **Enter** into PPPoE Setting editing menu.



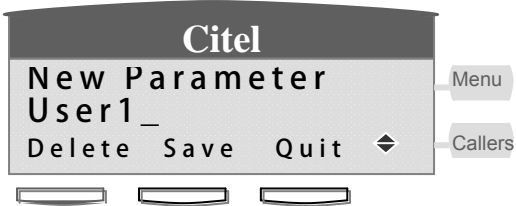
- Press **Next** or **Up** and **Down** key to switch between "Account" and "Password" menu.
- Press "Quit" Back to last menu.

Account editing

- Press **Edit** into Account editing menu.



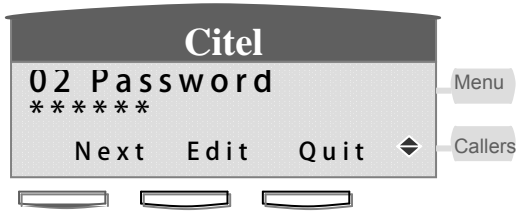
- Input the Account in the editing line.



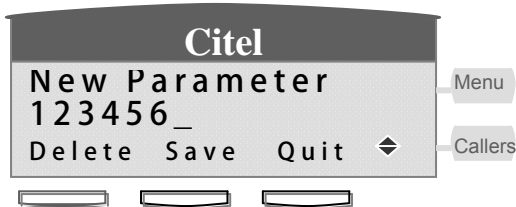
- Press "Delete" to delete character in the editing line.
- Press "Save" to save the Account you input.
- Press "Quit" back to last menu and non-save.

Password editing

1. Press **Edit** into Account editing menu.



2. Input the Password in the editing line.



3. Press "Delete" to delete character in the editing line.
4. Press "Save" to save the Password you input.
5. Press "Quit" back to last menu and non-save.

1.2 To set SIP setting

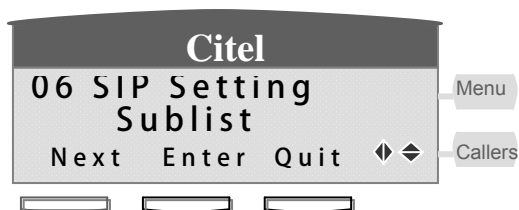
Use the following procedure to configure the SIP setting sublist.

For example :

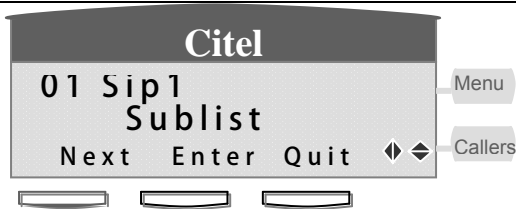
SIP server (Register Server Add): 10.10.10.1 (is your ip pbx address or provided by your local telecomm company)
SIP Number(Register Phone number): 622 (provided by your ip pbx or your local telecomm company)
SIP Account (Register Number) : 622 (provided by your ip pbx or your local telecomm company)
SIP Password (Register password): 261972 (provided by your ip pbx or your local telecomm company)

To enter SIP setting mode

1. Press "Menu" and using Up, Down or "Next" key into "SIP setting" mode.



2. Press "Next" and Up, Down to select "SIP1" or "SIP2" mode.
3. Press "Quit" back to last menu.
4. Press "Enter" into SIP1 menu



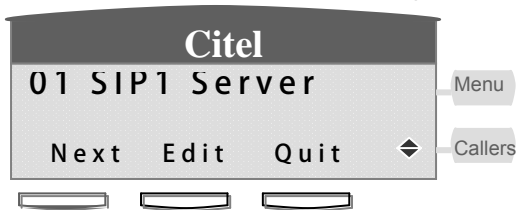
5. Press **Enter** into select “SIP1 Name” menu, and use Next or Up, Down button to switch between “SIP1 Name”, “ SIP1 Sever ” , “ SIP1 Number ” , “ SIP1 Account ” , “ SIP1 password” and “ SIP1 Register” menu

Enter SIP1 Server menu

To set the SIP1 Sever

Enter SIP1 Server menu

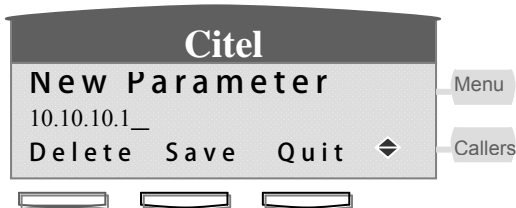
1. Press **Edit** into SIP1 Server editing menu.



2. Press **Quit** back to last menu.
3. Press **Next** into SIP Number menu.
4. Press “Delete ” to delete
5. Press “Save” to save as diag.3, Back to diag.2-1
6. Press “Quit” back to last menu and non-save.

SIP1 editing

1. Input SIP1 Server address 10.10.10.1 in the SIP Server editing line, the ‘*’ means ‘.’. The server address can be a web address or a IP address.

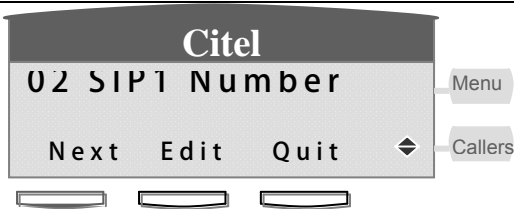


2. Press “Delete ” to delete character in the SIP Server editing line.
3. Press “Save” to save the SIP Name you input.
4. Press “Quit” back to last menu and non-save.

To set the SIP1 Number

Enter SIP1 Number menu

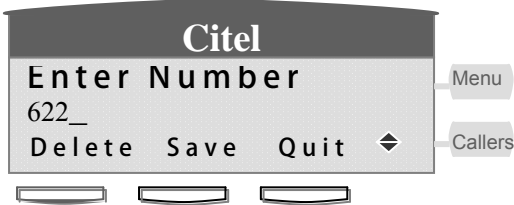
1. Press **Edit** into SIP1 Number editing menu



2. Press **Quit** back to last menu.
3. Press **Next** into SIP Account menu.

SIP1 Number editing

1. Input SIP1 Number 622 in the SIP Number editing line

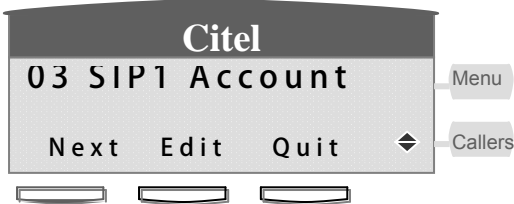


2. Press "Delete" to delete character in the SIP Number editing line.
3. Press "Save" to save the SIP Number you input.
4. Press "Quit" back to last menu and non-save.

To set the SIP1 Account

Enter SIP1 Account menu

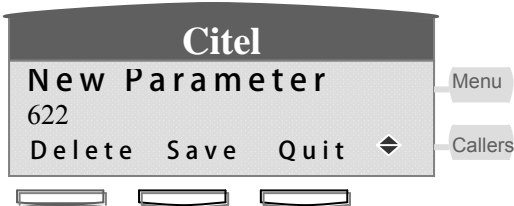
1. Press **Edit** into SIP1 Account editing menu



2. Press **Quit** back to last menu.
3. Press **Next** into SIP Account menu.

SIP1 Account editing

1. Input SIP1 Account 622 in the SIP Account editing line.

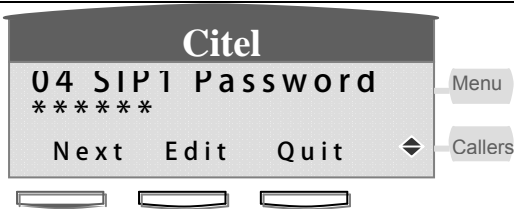


2. Press "Delete" to delete character in the SIP Account editing line.
3. Press "Save" to save the SIP Account you input.
4. Press "Quit" back to last menu and non-save.

To set the SIP1 Password

Enter SIP1 Password menu

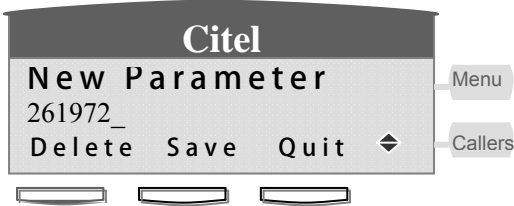
1. Press **Edit** into SIP1 Password editing menu.



2. Press **Quit** back to last menu.
3. Press **Next** into SIP1 Register menu.

SIP1 Password editing

1. Input SIP1 Password 261972 in the SIP Password editing line.

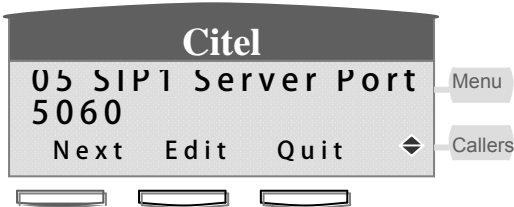


2. Press "Delete" to delete character in the SIP Account editing line.
3. Press "Save" to save the SIP Account you input.
4. Press "Quit" back to last menu and non-save.

To set the SIP1 Server Port

Enter SIP1 Server Port menu

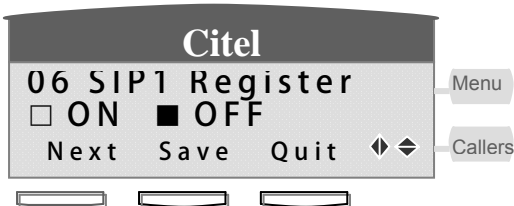
1. Press **Edit** into SIP1 Server Port editing menu. Default Port : 5060



2. Press **Quit** back to last menu.
3. Press **Next** into SIP1 Register menu.

To set the SIP1 Register

1. Press Left, Right key to ON/OFF the SIP1.





2. Press "Save" to save and quit.
3. Press "Quit" back to last menu and non-save.

1.3 Customizing your C4110 IP Phone

1.3.1 To set Ringer Type

You can select different ringer type to match your preferences and distinguish between calls on lines.

-To change the ringer type

1. Press  key and select “User preferences” item
2. If two lines are used, first select which line to change.
3. Press  key to select desired rings , then press **Save** soft key to save your settings.

1.3.2 To set screen setting sublist

1. Press  and go into “User Preferences Sublist” press “Enter”


2.

Either press  or Next softkey to scroll. “Screen Setting”


3.

Press **Enter** soft key and select Contrast or Brightness function.

To set Contrast


- Enter into Contrast menu.
- find desired setting using. 
- Press **Save** softkey to save your setting

To set Brightness

- Enter into Brightness menu.
- Select one of three kinds of brightness: Off, Low or High as you like using 
- Press **Save** softkey to save and quit or **Next** softkey directly to save and go to the next.


1.3.3 To set Volume Control sublist

1. Press  and go into “User Preferences Sublist” press “Enter”

2. Either press  or **Next** softkey to scroll. “Volume Control sublist ”

3. Press **Enter** soft key and select Ringer volume or Mic Volume function

To set Ringer volume:


- Enter the Ringer volume menu.
- Press  to adjust and reach the desired ringer volume.
- Press **Save** softkey to save your setting

You have 2 other ways to adjust your ringer volume:



To adjust ringer volume directly when a call is coming:

You can press left or right side of volume bar  directly to adjust ringer volume while a call is coming and the telephone is ring before you pick up the call.

To adjust ringer volume anytime




You can press left or right side of volume bar  directly to adjust ringer volume anytime if you like. But please don't press **Handsfree** or lift the handset.

To set Mic volume:

- Enter the Mic volume menu.
- Press   to adjust and reach the desired Mic volume.
- Press Save softkey to save your setting


1.3.4 To change the display Language

For changing business requirements, you may wish to change the language of your display messages and prompts.

1. Press **Menu** and go into “User Preferences Sublist” press “Enter”
2. Either press  or **Next** softkey to scroll. “Language Set”
3. “English is in use” is the default setting.
4. There are 2 kinds of language for your choice: English, Chinese. Use   to select your display language.
5. Press **Save** softkey to store your setting and quit, or press **Quit** softkey to give up the setting and go back up level.

1.3.5 To set Greeting Message

Greeting Message function lets your C4110 IP Phone display owner's phone number or name on the phone

1. Press **Menu** and go into “User Preferences Sublist” press “Enter”
2. Either press  or **Next** softkey to scrol “Greeting Message ”
3. Press **Edit** soft key to enter this setting, use dial /pad to input your desired information, then press **Save** soft key.



1.3.6 To set Memory key setting

You can save the numbers of your customers, business associates, or anyone you call frequently and phone systems' feature access code in C4110's four memory keys.



To save numbers in memory keys

1. select one of the four memory keys by one you want to store the number or feature code as the instructions on the display and enter the title ,number or feature code.
2. Press **Save** softkey to store your setting or press **Quit** softkey to give up the setting and return back up level.
3. You can delete the number stored in one memory key and save a new one in it or let it empty.

1.3.7 To Model Information Sublist

1. Press  **Menu**
2. Either press  or **Next** softkey to scroll "Model Information Sublist".
3. Press **Enter** soft key and select "Model Number , "Mac Address"or "Software Version"

1.3.8 To reboot system

Press  **Menu** and use  key or **Next** to select "reboot system" Press "*reboot*" button to reboot the device

2. Configuration with web browser

The IP Phone Web Configuration Menu can be accessed by the following URL: `http:// "Phone-IP-Address"`. e.g. `http://20.20.20.119`. The default LAN IP address is "192.168.1.179". If the web login port of the gateway is configured as non-80 standard port, the user need to input `http://xxx.xxx.xxx.xxx:PORT/`, otherwise the web will show that no server has been found, and it will be shown as follows:

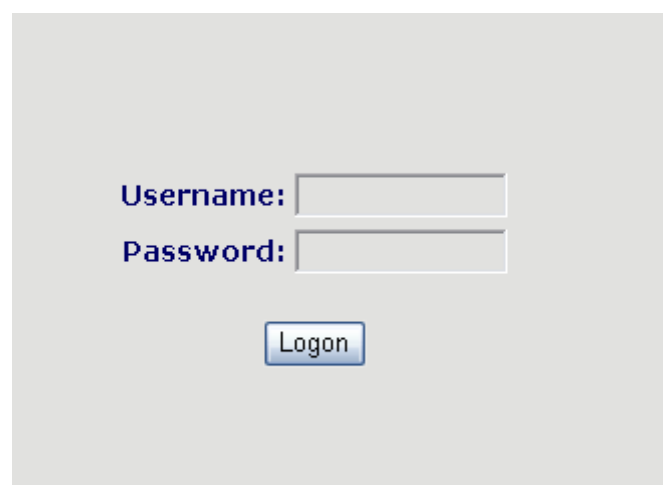


A screenshot of a web login interface. It features two text input fields: the first is labeled "Username:" and the second is labeled "Password:". Below these fields is a button labeled "Logon". The entire form is centered on a light gray background.

2.1 User verification

Users should login before configuring dialogue machine. Guest account: the default username and password are both "admin" can have a browse of system.

Note: after inputting username and password, user press carriage return directly to enter the page.



A screenshot of a web login interface, identical to the one above. It features two text input fields: the first is labeled "Username:" and the second is labeled "Password:". Below these fields is a button labeled "Logon". The entire form is centered on a light gray background.

2.2 Basic Status

2.2.1 Running Status

On this page user can gather information of each commonly-used parameter of the phone, it is shown as the following figure: the network section shows the current WAN, LAN configurations of the phone: including connect mode of WAN IP and IP(static state, DHCP, PPPoE), MAC address, WAN IP address of the gateway, LAN IP address of the gateway, the state of LAN DHCP server.

The screenshot displays the 'BASIC' configuration page. On the left is a blue sidebar with menu items: BASIC, Network, VOIP, PHONE, MAINTENANCE, SECURITY, and LOGOUT. The main content area has a title 'BASIC' and a navigation bar with 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. Below this is a 'Network' section with a table for WAN and LAN settings. The WAN table shows 'Connect Mode' as 'Static', 'IP Address' as '20.20.20.119', and 'Gateway' as '20.20.20.100'. The LAN table shows 'IP Address' as '192.168.10.1' and 'DHCP Server' as 'ON'. Below the network section is a 'Phone Number' section with a table for SIP lines. SIP LINE 1 is '616@210.21.235.238 :5060' and is 'Registered'. SIP LINE 2 is '617@210.21.235.238 :5060' and is 'Registered'. At the bottom, the version information is 'Version: VOIP PHONE V1.7.64.52 Jan 18 2008 18:51:25'.

Network			
WAN		LAN	
Connect Mode	Static	IP Address	192.168.10.1
MAC Address	00:0e:14:36:51:d2	DHCP Server	ON
IP Address	20.20.20.119		
Gateway	20.20.20.100		

Phone Number		
SIP LINE 1	616@210.21.235.238 :5060	Registered
SIP LINE 2	617@210.21.235.238 :5060	Registered

Version: VOIP PHONE V1.7.64.52 Jan 18 2008 18:51:25

The Phone Number section show the current registered phone number that work with sip server. The phone can register (two) number on the same server or different sip server.

The version number and date of issue have been shown at the end of the page.

2.2.2 Wizard

In order to make user easier set C4110's parameters, C4110 provides a wizard flow to set step by step.

First step is selecting network mode

[Wizard](#)

Network Mode Select

Static IP MODE

DHCP MODE

PPPoE MODE

Second step is setting network parameters if user uses static or PPPoE, else jump to step 3 directly

[Wizard](#)

Static IP Set

Static IP Address

Netmask

Gateway

DNS Domain

Primary DNS

Alter DNS

Third step is setting SIP

[Wizard](#)

SIMPLE SIP SET

Display Name

Server Address

Server Port

Account/User Name

Password

PhoneNumber

Enable Register

The last step is show the result of wizard. If user makes mistakes during the wizard,

he might click back to correct the error. Click Finish button, C4110 will save the setting and reboot. Then user might start to make/answer calls.

The screenshot shows a 'Wizard' configuration interface. It has two main sections: 'WAN' and 'SIP'. At the bottom, there are 'BACK' and 'Finish' buttons.

WAN	
Connect Mode	Static
Static IP Address	192.168.1.145
Gateway	192.168.1.1

SIP	
Register Server	192.168.1.4
Account/User Name	4141
PhoneNumber	4141
Register	ON

2.3 Network configuration

2.3.1 Wide area network (WAN)

User can view the current network IP linking mode of the system on this page.

The screenshot shows the 'NETWORK' configuration page. On the left is a navigation menu with options: BASIC, Network (selected), VOIP, PHONE, MAINTENANCE, SECURITY, and LOGOUT. The main content area is titled 'NETWORK' and has tabs for WAN, LAN, QOS, SERVICE PORT, DHCP SERVER, and SNTP. The 'WAN' tab is active, showing 'Wan Status' and 'WAN Setting' sections.

Wan Status	
Active IP	20.20.20.119
Current Netmask	255.255.255.0
MAC Address	00:0e:20:da:2f:5a
Current Gateway	20.20.20.100
Mac Authenticating Code	Valid MAC

WAN Setting		
Static <input type="radio"/>	DHCP <input checked="" type="radio"/>	PPPOE <input type="radio"/>
<input type="button" value="APPLY"/>		

User will be authorized to set the network IP, Gateway and DNS if the system adopts the static linking mode.

If the system selects DHCP service in the network which is using DHCP service, IP address will be gained dynamically.

If the system selects PPPOE service in the network which is using the PPPOE service, then the IP address will be gained by the set PPPOE ISP

Current gateway IP, subnet mask, MAC address and current gateway IP:

Active IP	20.20.20.119
Current Netmask	255.255.255.0
MAC Address	00:0e:20:da:2f:5a
Current Gateway	20.20.20.100

Configure Mac Authenticating code, the phone don't work if error code.

Mac Authenticating Code	<input type="text"/>	Valid MAC
--------------------------------	----------------------	------------------

Select acquisition option; Configure way of IP for WAN; This is single static IP parameter for WAN

Static <input type="radio"/>	DHCP <input checked="" type="radio"/>	PPPOE <input type="radio"/>
-------------------------------------	--	------------------------------------

Select "static". These are must be filling

Static IP Address	<input type="text" value="192.168.0.5"/>
Netmask	<input type="text" value="255.255.255.0"/>
Gateway	<input type="text" value="192.168.1.1"/>
DNS Domain	<input type="text"/>
Primary DNS	<input type="text" value="202.96.134.133"/>
Alter DNS	<input type="text" value="202.96.128.68"/>
APPLY	

Select "PPPoE". These are must be filling.

PPPOE Server	<input type="text" value="ANY"/>
Username	<input type="text" value="user123"/>
Password	<input type="password" value="••••••••"/>
APPLY	

PPPoE Server: the name of ISP PPPoE server, we suggest don't change if no especial request.

2.3.2 LAN Configuration

User can make local area network (LAN) configuration on this page, when bridge mode is selected, the local area network (LAN) configuration will no longer go into

effect.

The screenshot shows a web interface for network configuration. On the left is a blue sidebar with menu items: BASIC, Network, VOIP, PHONE, MAINTENANCE, SECURITY, and LOGOUT. The main content area is titled 'NETWORK' and has several tabs: WAN, LAN, QOS, SERVICE PORT, DHCP SERVER, and SNTP. The 'LAN' tab is active, showing a 'Lan Set' section with the following fields:

LAN IP	192.168.10.1
Netmask	255.255.255.0
DHCP Service	<input checked="" type="checkbox"/>
NAT	<input checked="" type="checkbox"/>
Bridge Mode	<input type="checkbox"/>

An 'APPLY' button is located at the bottom right of the form.

Configuration Item explanation:

Select “Bridge Mode”. LAN and WAN are in the same network.

A close-up of the 'Bridge Mode' configuration item, showing the label 'Bridge Mode' and an unchecked checkbox.

Static Address and Netmask. When the “DHCP service is” selected, the phone as a DHCP Server. When the “NAT” is selected, the phone as a router.

LAN IP	192.168.10.1
Netmask	255.255.255.0
DHCP Service	<input checked="" type="checkbox"/>
NAT	<input checked="" type="checkbox"/>

2.4 VOIP Configuration

User can configure specific parameters of SIP line on this page ;Click Basic to set the common parameters of SIP line selected.

The screenshot shows a web interface for VOIP configuration. On the left is a blue sidebar with navigation links: BASIC, Network, VOIP (highlighted), PHONE, MAINTENANCE, SECURITY, and LOGOUT. The main content area is titled 'VOIP' and contains tabs for SIP, IAX2, STUN, and DIAL PEER. Below the tabs is a 'SIP Line Select' section with a dropdown menu set to 'SIP 1' and a 'Load' button. The 'Basic Setting' section is a table with the following fields:

Regist Status	Registered	Display Name	
Server Address	20.20.20.200	Proxy Server Address	
Server Port	5060	Proxy Server Port	
Account Name	100	Proxy Username	
Password	•••	Proxy Password	
Phone Number	100	Domain Realm	
Enable Register	<input checked="" type="checkbox"/>		

Below the table are 'APPLY' and 'Advanced Set' buttons.

Definition of each parameter described as below

SIP Line Select	Select a line of SIP to configured, then click load button to get ready to set parameters
Registrar status	SIP register state ; if register successfully, show “Registered, otherwise show different results
Server address	Set SIP server IP address or Domain
Server Port	Set SIP server signal port
Account/User Username	Set SIP server account username (Usually it is the same with the config port number)
Password	Set password of SIP server account
Phone Number	Set assigned phone number
Display Name	Set assigned SIP display name as caller
Proxy Server Address	Set proxy server IP address if SIP

	account's server has proxy
Proxy Server Port	Set proxy server signal port
Proxy Username	Set proxy server account username (Usually it is the same with server account name)
Proxy Password	Set password of proxy server account
Domain Realm	Enter the sip domain if any, otherwise C4110 will use the proxy server address as sip domain. (Usually it is same with registered server and proxy server IP address).
Enable Register	After setting all the parameters above, enable register and click apply to make C4110 register to SIP server

Click Advanced set button to set the some parameters of SIP line selected which SIP server need to them.

2.4.1 SIP Line advanced configuration

VOIP
PHONE
MAINTENANCE
SECURITY
LOGOUT

Advanced SIP Setting

Register Expire Time	<input type="text" value="60"/> seconds	Forward Type	Off <input type="button" value="v"/>
Subscribe Expire Time	<input type="text" value="300"/> seconds	Forward Phone Number	<input type="text"/>
NAT Keep Alive Interval	<input type="text" value="60"/> seconds	Ring Type	Type 1 <input type="button" value="v"/>
User Agent	Voip Phone 1.0	Server Type	common <input type="button" value="v"/>
Signal Key	<input type="text"/>	DTMF Mode	DTMF_RFC2833 <input type="button" value="v"/>
Media Key	<input type="text"/>	RFC Protocol Edition	RFC3261 <input type="button" value="v"/>
Local Port	<input type="text" value="5060"/>	Transport Protocol	UDP <input type="button" value="v"/>
Enable Subscribe	<input type="checkbox"/>	Auto TCP	<input type="checkbox"/>
Enable Keep Authentication	<input type="checkbox"/>	Signal Encode	<input type="checkbox"/>
NAT Keep Alive	<input type="checkbox"/>	Rtp Encode	<input type="checkbox"/>
Enable Via rport	<input checked="" type="checkbox"/>	Enable Session Timer	<input type="checkbox"/>
Enable PRACK	<input type="checkbox"/>	Answer With Single Codec	<input type="checkbox"/>

Definition of each parameter described as below

Register Expire Time	Set expire time of SIP server register, the final actual time maybe decided by server
NAT Keep Alive Interval	Set NAT keep alive interval time to make C4110E could being contacted if server could not support

	keeping NAT alive
User Agent	Set the user agent name to show the information of C4110
Signal Key	Fill in the key for SIP encryption according to the Server type selected. Default is common
Media Key	Fill in the key for Media encryption according to the Server type selected. Default is common
Server Type	Set SIP server type if server is special or encrypted server
DTMF Mode	Set DTMF sending mode, support RFC2833, DTMF_RELAY and SIP info
RFC Protocol Edition	Enable the phone to use protocol edition. When the phone needs to communicate with phones using SIP1.0 such as CISCO5300 and so on, needs to modify into RFC2543. the default is to RFC3261
Forward Phone Number	Set the selected Line's call forward number
Enable PRACK	Enable / Disable SIP prack function. Default is unchecked. Prack is same with ACK that is used for temporary response. Prack is one of SIP request info
Enable Keep Authentication	Enable /disable support register with authentication, so that C4110 needn't carry authentication field, response each time. When server receive the register request with authentication, server will response 200 message, not 401
NAT Keep Alive	Enable/disable NAT keep alive function
Enable Session Timer	Enable/disable support RFC4028.
Signal Encode	Enable/disable signal Encryption according to the server type.
Rtp Encode	Enable/disable Voice encryption according to the server type

Enable Via rport	Enable/disable support RFC3581. When Phone is behind a NAT. "rport " allows request that the server send the response back to the source IP address and port from which the request originated.
------------------	---

Click Global Set button to set the parameters efficiently for all SIP lines.

2.4.2 IAX2 configuration

Definition of each parameter described as below

Register Status	Shows if the phone has been registered the IAX2 server or not.
IAX2 Server Addr	Input your IAX2 server address.
IAX2 Server Port	Set your IAX2 server port, the default is 4569.
Account Name	Input your IAX2 register account name.
Account Password	Input your IAX2 register password.
Phone Number	Input your assigned phone number (usually it is same you're your IAX2 account name).
Local Port	Set your local sport, the default is 4569.
Voice Mail Number	Specify the voice mail's number.
Voice Mail Text	Specify the voice mail's name.
Echo Test Number	Set echo test number. If IAX2 server supports echo test, and echo test number is non- numeric, system could set an echo test number to

	replace the echo test text. So user can dial the numeric number to test echo voice test. This function is provided with server to make endpoint to test whether endpoint could talk through server normally.
Echo Test Text	Specify echo test text's name.
Refresh Time	Set expire time of IAX2 server register, you can set it between 60 and 3600 seconds.
Enable Register	Start to register the IAX2 server or not by selecting it or not.
Enable G.729	Enable or disable code G.729 by selecting it or not
IAX2 (Default Protocol)	Select it to make all outgoing calls through the IAX2 server by default. If you also need make a call through SIP server, you can make prefix in dial peer setting to realize SIP calling. Note: any incoming call can be from both IAX2 and SIP. ;

2.4.3 STUN Set

The screenshot shows the VOIP configuration page with a sidebar on the left containing menu items: BASIC, Network, VOIP, PHONE, MAINTENANCE, SECURITY, and LOGOUT. The main content area is titled 'VOIP' and has tabs for SIP, IAX2, STUN, and DIAL PEER. The 'STUN Set' section includes the following fields:

STUN NAT Transverse	FALSE
STUN Server Addr	
STUN Server Port	3478
STUN Effect Time	50 Seconds
Local SIP Port	5060

Below these fields is an 'APPLY' button. The 'Set Sip Line Enable Stun' section includes a dropdown menu set to 'SIP 1', a 'Load' button, and a 'Use Stun' checkbox which is currently unchecked. An 'APPLY' button is located at the bottom of this section.

Definition of each parameter described as below

STUN NAT Transverse	Indication that current NAT is type that could be transverse or not
STUN Server Addr	Set the STUN server's IP address/domain
STUN Server Port	Set the STUN server's port
STUN Effect Time	Set the interval time of STUN detection
Local SIP Port	Set C4110 SIP signal port

After finished the aforesaid network and VoIP configurations on the phone and network communication has been implemented, the user can make VoIP calls by the

calling register and proxy server.

2.4.4 Dial peer configuration

Number IP table configuration

Function of number IP table is one way to implement the phone's calling online, and the calling of the phone will be more flexible by configuring the number IP table. For example, user knows the other party's number and IP and want to make direct call to the party by point-to-point mode : the other party's number is 1234, make a configuration of 1234 directly ,then the phone will send the called number-1234 to the corresponding IP address ; Or set numbers with prefix matching pattern. For example, user wants to make a call to a number in a certain region (010) , user can configure the corresponding number IP as 010T— protocol-IP, after that, whenever user dial numbers with 010 prefix(such as 010—62201234), the call will be made by this rule.

Base on this configuration, we can also make the phone use different accounts and run speed dialing without manual swap. When making deletion or modification, select the number first and click Modify button to change the rule.

The screenshot displays the 'VOIP' configuration interface. On the left is a vertical navigation menu with categories: BASIC, Network, VOIP, PHONE, MAINTENANCE, SECURITY, and LOGOUT. The main content area is titled 'VOIP' and contains several sections:

- Navigation Tabs:** SIP, IAX2, STUN, DIAL PEER (selected).
- Dial Peer Modify:** A table with columns: Number, Destination, Port, Mode, Alias, Suffix, Del Length.
- Add Dial Peer:** A form with fields for Phone Number, Destination (optional), Port (optional), Alias (optional), Call Mode (dropdown menu set to SIP), Suffix (optional), and Delete Length (optional). A 'Submit' button is located below the fields.
- Dial Peer Option:** A section with a dropdown menu and 'Delete' and 'Modify' buttons.

Click Add, add Dial Peer configuration will be shown at the upper part of the page.

Phone Number is to add outgoing call number, there are two kinds of outgoing call number setup : One is exactitude matching, after this configuration has been done, when the number is totally the same with the user's dialing number, the phone will make the call with this number's IP address image or configuration; Another is prefix matching (be equivalent to PSTN's district number prefix function, if the previous N digit of this number are the same with that of the user's dialing number(the prefix number length), then the phone will use this number's IP address image or configuration to make the call. When configuring the prefix matching, letter "T" should be added behind the prefix number to be distinguished from the exactitude matching.

Destination(optional) is configuration of destination address, if it is point-to-point call, then input the opposite terminal's IP address, it can also be set as domain name and resolved the specific IP address by DNS server of the phone. If no configuration has been made, then the IP will be considered as 0.0.0.0. This is an optional configuration item.

Port(optional) is configuration of the called party's protocol signal port, this is optional configuration item; if nothing is input, C4110 will set it as the default value 5060 of sip protocol.

Configure alias, this is optional configuration item : it is the number to be used when the other party's number has prefix; when no configuration has been made, C4110 will be shown as no alias.

- add:** xxx add xxx before number. in this way it can help user save the dialing length;
- all:** xxx the number is all replaced by xxx; speed dialing can be implemented, for example, user configure the dialing number as 1, with the configuration "all" the actual calling number will be replaced;
- del** delete n bit in the front part of the number, n can be decided by the replacing length. this configuration can decide the protocol for appointed number.
- rep:** xxx n bit in the front part of the number will be replaced. n is decided by the replacing length.

Configure suffix, this is optional configuration item; it is the additive dial-out number behind the number; when no configuration has been made, C4110 will be shown as no

2.5 DSP configuration

On this page, user can set speech coding, IO volume control, cue tone standard, caller ID standard and so on.

DSP Configuration			
First Codec	g711Ulaw64k	Second Codec	g723
Third Codec	g711Alaw64k	Fourth Codec	g729
Fifth Codec	g711Alaw64k	Ring Type	Type 1
Input Volume	3 (1-9)	Output Volume	2 (1-9)
Handfree Volume	1 (1-9)	Ring Volume	5 (1-9)
G729 Payload Length	20ms	Signal Standard	China
Handdown Time	200 ms	VAD	<input type="checkbox"/>
<input type="button" value="APPLY"/>			

1. Configure Coding Rule according to network bandwidth and VoIP server; support G.711a/u G.729 G723 and Wideband G.722 Codec for better speech quality.
2. Input Volume is to adjust mike volume. The value is from 1 to 9
3. Output Volume is to adjust handset out volume. The value is from 1 to 9.
4. Configure Signal Standard according to country's phone voice. It acts on the tone when you pick up phone, ring back tone and so on.
5. Normally, G729 Payload Length does not need to be changed into 10 ms; the voice 6, payload size can be represented in terms of the codec samples. For example, a G.729 voice payload size of 20 ms (two 10 ms codec samples) represents a voice payload of 20 bytes [(20 bytes * 8) / (20 ms) = 8 Kbps]
6. Adjust the hand down time .The time range is from 80ms to 800ms.
7. VAD Enable/disable VAD (Voice activity detection).

2.5.1 Call Service configuration

On this page, user can set value added services such as hot-line, blacklist, out-limit list and so on

The screenshot shows a web interface for configuring phone services. On the left is a vertical navigation menu with options: BASIC, Network, VOIP, PHONE, MAINTENANCE, SECURITY, and LOGOUT. The main content area is titled 'PHONE' and contains several tabs: DSP, CALL SERVICE (selected), DIGITAL MAP, PHONE BOOK, and FUNCTION KEY. Below the tabs are three main sections:

- Call Service Setting:** A table with settings for Hot Line, P2P IP Prefix, Do Not Disturb, Enable Call Transfer, Enable Three Way Call, No Answer Time (set to 20 seconds), Auto Answer, Ban Outgoing, Enable Call Waiting, and Accept Any Call. An 'APPLY' button is at the bottom.
- Black List:** A section with an 'Add' button, a dropdown menu, and a 'Delete' button.
- Limit List:** A section with an 'Add' button, a dropdown menu, and a 'Delete' button.

Configuration Items Explanation:

Hotline	Configure hot-line number of the port. With this number of the port, this hot-line number will be dialed automatically as soon as off hook and user can't dial any other number.
No Disturb	Do no disturb
Enable Call Transfer	Configuration enable/disable call transfer
Enable Three way call	Configuration enable/disable three way call
Enable call waiting	Configuration enable/disable call waiting
Answer any call	Configuration enable/disable answer any call
Enable voice recording	Configuration enable/disable voice recording
Incoming record playing	Configuration enable/disable playing record.
No Answer time	Configuration time how long is no answer, the default is 20 seconds
Black list	Phone in this list will refuse when it coming call

2.5.2 Digital Map configuration

Digit map is a set of rules to determine when the user has finished dialing. Digital Map is based on some rules to judge when user end their dialing and send the number to the server.

With digital map, users don't have to press '#' key or "call" key after finish dialing. If the number dialed matches some item in the digital map table, or it doesn't match with any item, this number will be sent out immediately. It is not like using dial peer; using digital map won't change the number dialed, the number sent is the same as the number dialed.

The screenshot shows the 'PHONE' configuration page with the 'DIGITAL MAP' tab selected. The 'Digital Map Set' section contains the following configuration:

<input checked="" type="checkbox"/>	End With "#"		
<input type="checkbox"/>	Fixed Length	11	
<input checked="" type="checkbox"/>	Time Out	5	(3--30)

Below the 'Digital Map Set' is an 'APPLY' button. The 'Digital Rule table' section is currently empty and includes 'Add' and 'Del' buttons.

End With "#": Checked- use # for the end of dialing. User dials 200300#. phone send 200300 to server immediately.

Fixed Length: when the length of the dialing is matched, the call will be sent.

Timeout: Specify the timeout of the last dial digit. The call will be sent after timeout.

End with "#", fixed length and Time out, these options are not repulsion. That means, these three ways can be use together or alone.

User could also use digital map table to make own rule.

X Represents any one number between 0 and 9.

Tn Represents the last digit timeout. Here [n] represents the time from 0~9 second, it is necessary. Tn must be the last two digit in the entry. If Tn is not included in the entry, we use T0 as default, it means system will sent the number immediately if the number matches the entry.

. (Dot) represents any number and no length limit.

[] Number location value range. It can be a number range(such as [1-4]), or number is separated by comma such as [1,3,5], or use a list such as [234]

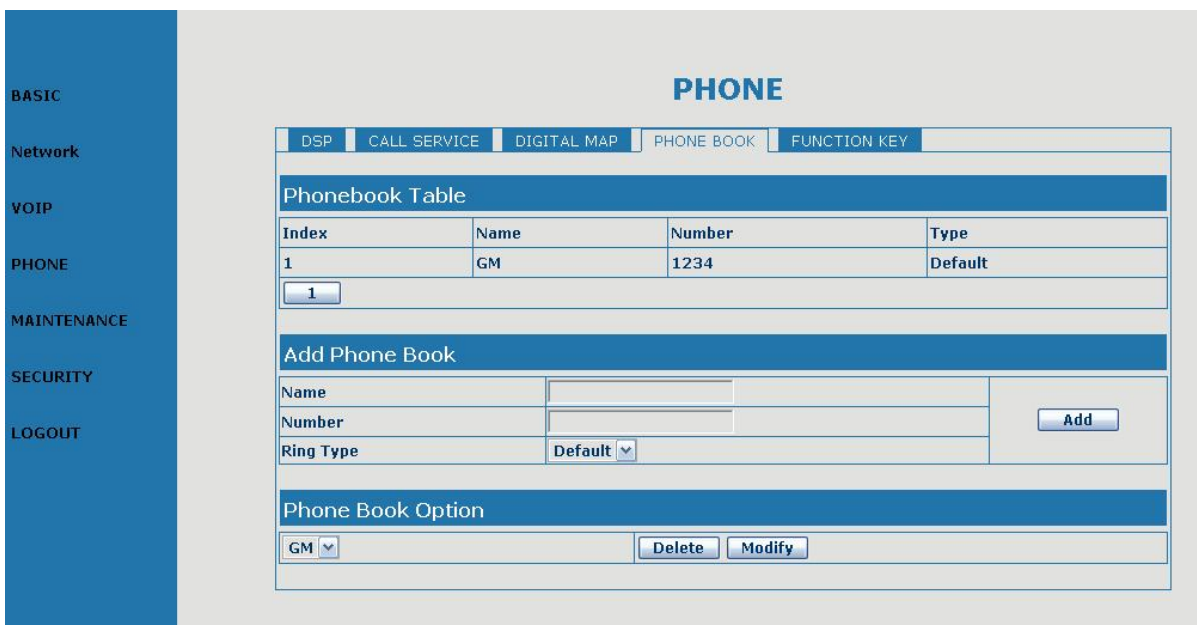
Example:

[1-8]xxx any 4 digits number between 1000 and 8999 sending out immediately.

9xxxxxxx any 8 digits number starting with 9 will be sent out immediately.
 911 after finishing dialing 911, it will be sent out immediately.
 911T4 after finishing dial 911, it will be sent out in 4 second.
 9911x.T4 any number more than 5 digits length starting with 9911, will be sent out in 4 seconds.

2.5.3 Phone Book configuration

Phone book configuration is used for add phone Book, Delete phone Book, Modify phone Book. Support up to 500 items.



2.5.4 Function Key configuration

There are 4 programmable memory keys on the C4110 phone and 8 programmable memory keys on the extended model . You can set the programmable keys with any of the following functions: Line, memory keys, key event.

PHONE

DSP | CALL SERVICE | DIGITAL MAP | PHONE BOOK | FUNCTION KEY

Interface Configuration

Contrast	5 (1-9)	Luminance	1 (0-2)
MWI Number			

Function Key Setting

F 1	Memory Key	
F 2	Memory Key	
F 3	Memory Key	
F 4	Memory Key	
F 5	Memory Key	
F 6	Memory Key	
F 7	Memory Key	
F 8	Memory Key	
F 9	Memory Key	
F 10	Memory Key	
F 11	Memory Key	
F 12	Memory Key	

Setting a Memory Key: The speed dial function can be set using the IP phone UI or the C4110 Web UI. You can save up to 12 speed dial numbers and names on the C4110. These can be programmed to quickly access features such as (*97), Voicemail.

Setting a Key Event: You can set DND, Pbook, Redial, on the programmable keys using the IP phone Web UI only.

PHONE

DSP | CALL SERVICE | DIGITAL MAP | PHONE BOOK | FUNCTION KEY

Interface Configuration

Contrast	5 (1-9)	Luminance	1 (0-2)
MWI Number			

Function Key Setting

F 1	Key Event	F_DND
F 2	Key Event	F_PBOOK
F 3	Key Event	F_REDIAL
F 4	Memory Key	
F 5	Memory Key	
F 6	Memory Key	
F 7	Memory Key	
F 8	Memory Key	
F 9	Memory Key	
F 10	Memory Key	
F 11	Memory Key	
F 12	Memory Key	

2.6 Phone Maintenance

System manage is used for firmware/config file update, config file backup, config Account save/clear config, and reboot function.



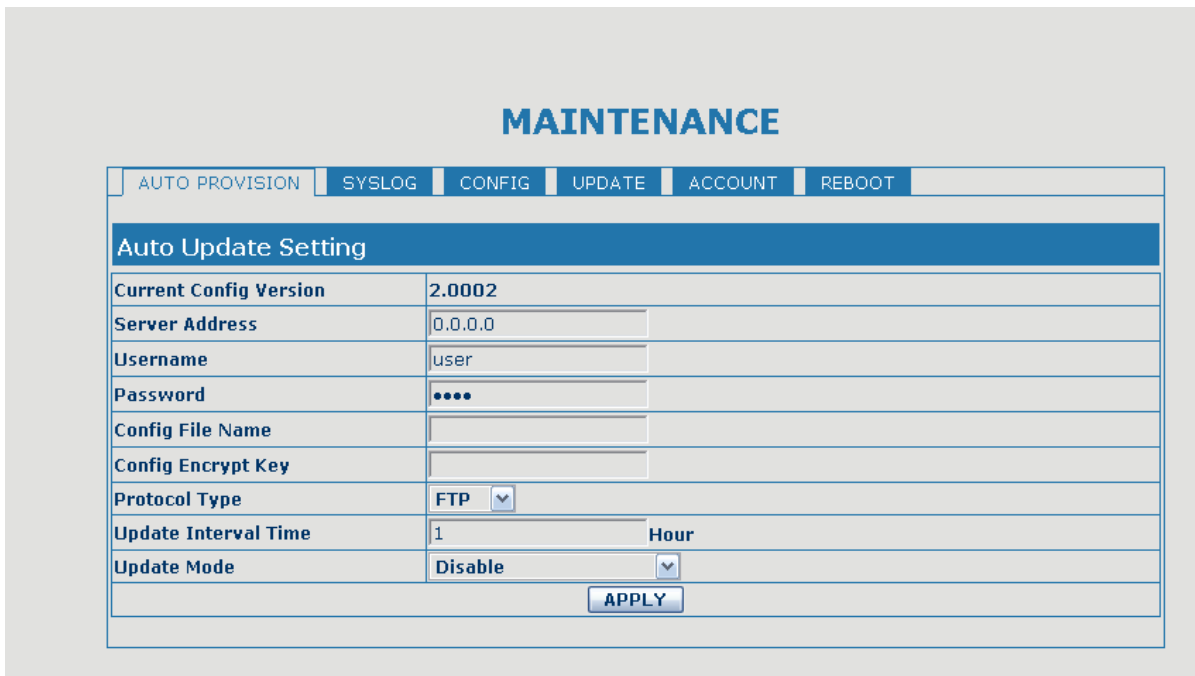
The screenshot shows the 'MAINTENANCE' configuration page. On the left is a vertical navigation menu with options: BASIC, Network, VOIP, PHONE, MAINTENANCE (highlighted), SECURITY, and LOGOUT. The main content area has a title 'MAINTENANCE' and a sub-menu with tabs: AUTO PROVISION, SYSLOG, CONFIG, UPDATE (selected), ACCOUNT, and REBOOT. Below the tabs is the 'Auto Update Setting' form. The form contains the following fields:

Current Config Version	2.0002
Server Address	0.0.0.0
Username	user
Password	••••
Config File Name	
Config Encrypt Key	
Protocol Type	FTP
Update Interval Time	1 Hour
Update Mode	Disable

An 'APPLY' button is located at the bottom right of the form.

2.6.1 Auto Update configuration

Definition) of each parameter described as below



This is a detailed view of the 'Auto Update Setting' form. It includes the same navigation and menu elements as the previous screenshot. The form fields are:

Current Config Version	2.0002
Server Address	0.0.0.0
Username	user
Password	••••
Config File Name	
Config Encrypt Key	
Protocol Type	FTP
Update Interval Time	1 Hour
Update Mode	Disable

An 'APPLY' button is located at the bottom right of the form.

Server Address: your FTP or TFTP server address.

Username: FTP server login user name. If it is used with TFTP, user need not fill anything.

Password: FTP server login user password. If it is used with TFTP, user need not fill anything.

Config File name: the name of config file in FTP or TFTP server. If “config file name” is unfilled, that means the phone will auto search the right config file from FTP /TFTP server by MAC address matching.

Config Encrypt Key: If the config file is encrypted, C4110 need to be filled in the encrypt key. config file supports AES (The Advanced Encryption Standard) 64. After download config file to phone, the phone will auto make decryption.

Protocol Type: update protocol selecting, Support FTP and TFTP protocol. It also can support optional HTTP.

Update interval time: the interval time of deciding to update the config file. It cooperate with update mode “update at a time interval”.

Update mode: there are three option for selection. Disable-disable auto update function; Update after reboot-when rebooting, it will auto connect server to update the config file; Update at a time interval-C4110 will check the config file of FTP server every interval time after reboot.

2.6.2 Syslog configuration

Users can star or close syslog function and configure syslog server IP address & port.

Syslog Set	
Server IP	0.0.0.0
Server Port	514
MGR Log Level	None
SIP Log Level	None
IAX2 Log Level	None
Enable Syslog	<input type="checkbox"/>
<input type="button" value="APPLY"/>	

Definition of each parameter described as below

Enable Syslog	Config enable/disable syslog function, choose it and then click “Apply” button to go into effect
----------------------	---

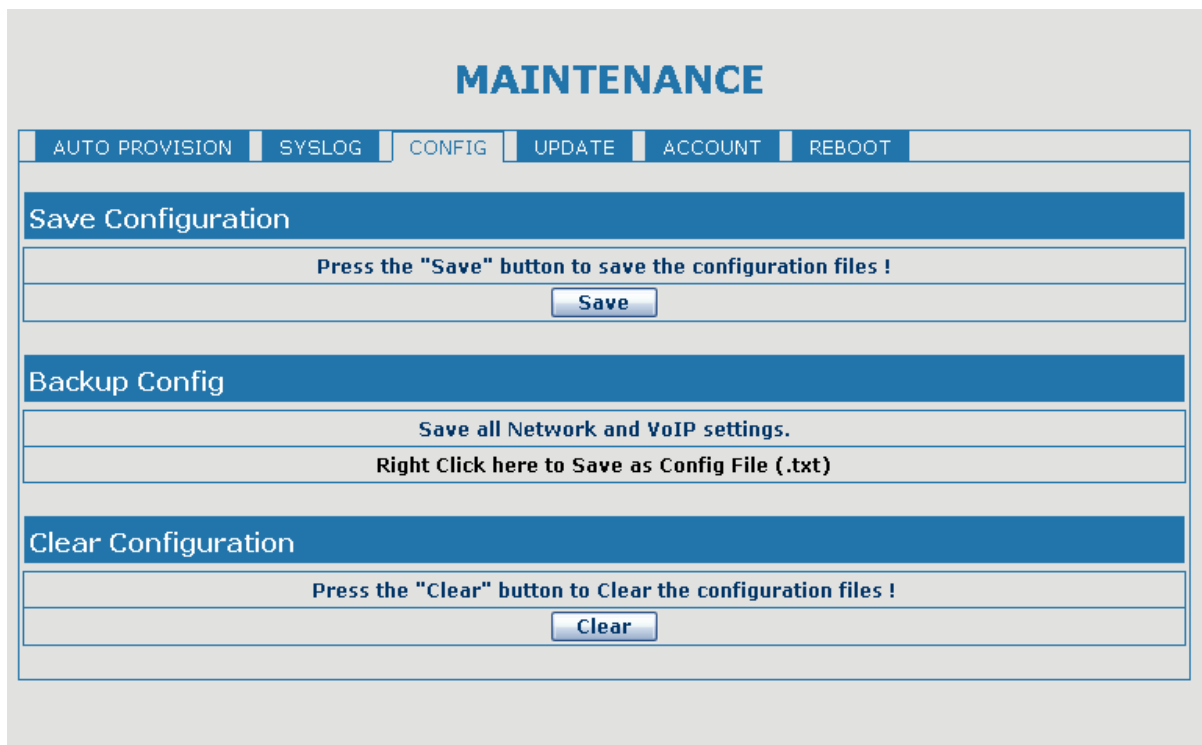
Server IP	Config syslog server IP address
Server Port	Config syslog server port, click “ Apply” button after inputting server IP & server port to take effect
MGR Log Level	Set system manager debug messages open with different level
SIP Log Level	Set SIP debug messages open with different level

2.6.3 Save Config

After setting parameters to C4110, user click save button to save the new config to C4110

If user want to make C4110 return to the factory default configuration, user could login as root level(such as admin), then click Clear button to use all default set. C4110 will clear config file in the system and reboot with default setting. If user use general level(such as guest), C4110 will not clear all set but part of them to default setting.

User could click the right key of mouse to save the config file of C4110 to the local PC when mouse move to “Right Click here to Save as Config File(.txt)”.



2.6.4 Upload WEB page

On this page, user can select the upgrade documents (firmware or config file) on hard disk of the computer directly to run the system upgrade. After the upgrade has been completed, restart the gateway and it will be usable at once.

MAINTENANCE

AUTO PROVISION
SYSLOG
CONFIG
UPDATE
ACCOUNT
REBOOT

Web Update

Select file 浏览... (*.z,*.txt,*.au) Update

FTP Update

Server	<input type="text"/>
Username	<input type="text"/>
Password	<input type="text"/>
File Name	<input type="text"/>
Type	Application update ▼
Protocol	FTP ▼

APPLY

2.6.5 FTP download

On this page, user can upgrade system and configure files by FTP or TFTP mode.

MAINTENANCE

AUTO PROVISION
SYSLOG
CONFIG
UPDATE
ACCOUNT
REBOOT

Web Update

Select file 浏览... (*.z,*.txt,*.au) Update

FTP Update

Server	<input type="text"/>
Username	<input type="text"/>
Password	<input type="text"/>
File Name	<input type="text"/>
Type	Application update ▼
Protocol	FTP ▼

APPLY

Configuration Item Explanation

Server	IP address of FTP/TFTP server
Username	Name of user
Password	Password
File name	Name of download/upload file name
Type	Configure the operation type. There are 3 types: 1. "application update": update the system through ftp or tftp 2. "config file export": Upload the gateway configuration file to

	the remote server through ftp or tftp protocols 3. "config file from gateway import": Download the configuration file from remote server and import to local
Protocol	Server type: FTP or TFTP

2.6.7 Account management

Users can edit users (add or delete) account and modify existing users' authority on this web page.

The screenshot shows a web interface titled "MAINTENANCE" with several navigation tabs: AUTO PROVISION, SYSLOG, CONFIG, UPDATE, ACCOUNT, and REBOOT. The "ACCOUNT" tab is selected. Below the tabs, there are three main sections:

- Set Keyboard Password:** A form with a "Keyboard Password" input field (masked with dots) and a "Set" button.
- User Set:** A table showing existing users:

User Name	User Level
admin	Root
guest	General
- Add User:** A form with fields for "User Name", "User Level" (a dropdown menu currently set to "Root"), "Password", and "Confirm", followed by a "Submit" button.

At the bottom, there is a partially visible "Account Option" section.

Definition of each parameter described as below

- User Name List existing phone user account name
- User Level Show existing user account level [two option]:Root and General: Root level users have the right to modify config; General level users have the right to read-only
- Add Add user account to IP phone
- Delete Delete increased user account
- Modify Modify increased user level and password

Operation Example:

Add one new account:

Click "Add" button →input User name (No-Modify) →Choosing User level from dropdown menu →set new user password →confirm password →submit the new account info by clicking "submit" button →show "submit success" on screen →return to account configuration interface by clicking "Return" button

Delete increased account:

Choosing the account need to del from dropdown menu→ Delete account by pressing “Delete button” →show “Submit Success” on screen

Modify account’s password:

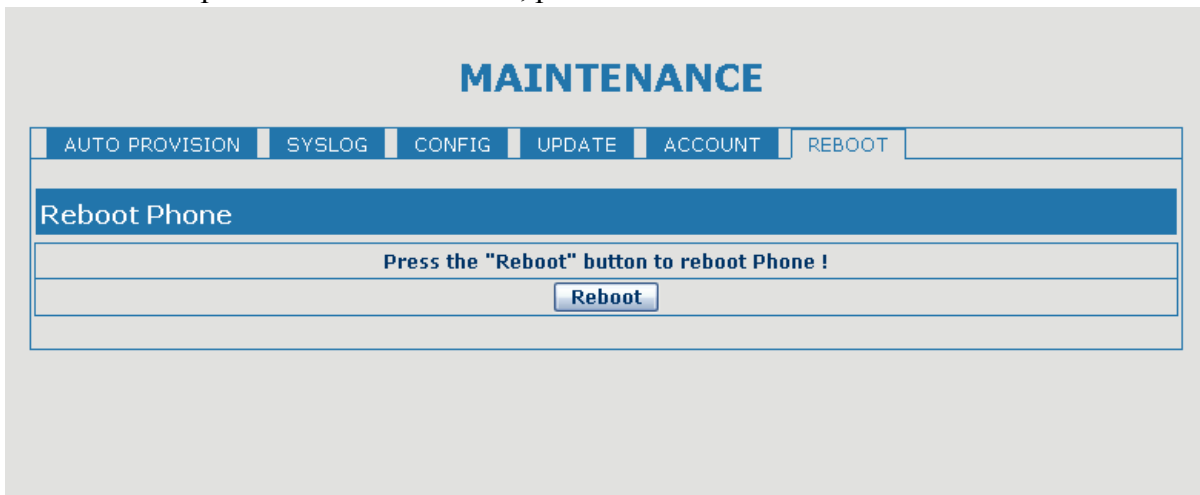
Choosing the modified account →enter below interface →modify user password →click “Submit” button to submit the modification

Owing to the phone's default account : accounts of the administrator level-admin account and the ordinary level—guest account are all weak account and weak password, the username and password will be easily to guess on public network, so the user had better modify the administrator and ordinary user.

Enter with manager level when making modification, create a administrator account and a browse account (you'd better not set the name as admin, administrator, guest, etc.), set password and then save configuration, entering with new manager account, delete default manager and browse account and save configuration, security will be enhanced!

2.6.8 Reboot

The phone will reboot when Administrator have modify some configuration. Configuration not effect until phone reboots. To reboot, press “Reboot” button.



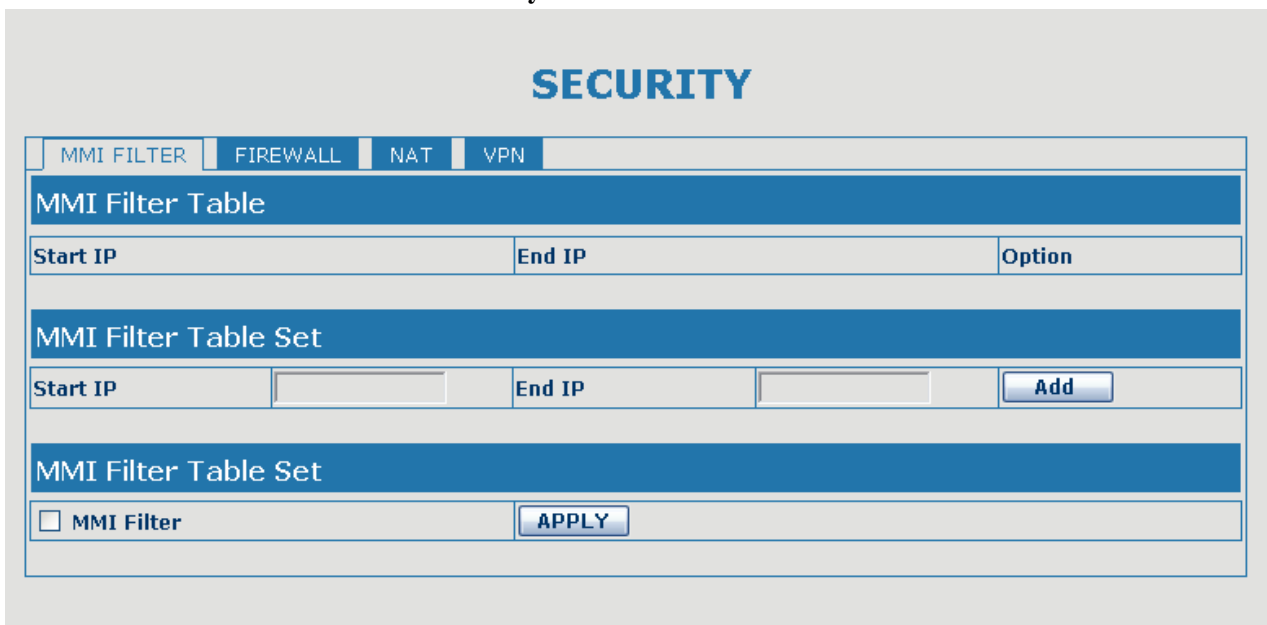
2.7 Security

2.7.1 MMI Filter configuration

MMI filter is used to make access limit to C4110.

When MMI filter is enable. Only IP address within the start IP and end IP can access C4110

. WARNING! If you try to Enable MMI filter check box without setting any item in filter table. You could not access gateway any more after you close this page. Therefore make sure that your IP address is included in the filter table before you want to enable MMI Filter.



The screenshot shows a web interface for configuring security settings. At the top, the word "SECURITY" is displayed in large blue letters. Below it, there are four tabs: "MMI FILTER", "FIREWALL", "NAT", and "VPN". The "MMI FILTER" tab is currently selected. The main content area is divided into three sections:

- MMI Filter Table:** A table with three columns: "Start IP", "End IP", and "Option".
- MMI Filter Table Set:** A form with two input fields for "Start IP" and "End IP", and an "Add" button.
- MMI Filter Table Set:** A section containing a checkbox labeled "MMI Filter" and an "APPLY" button.

2.7.2 Firewall configuration

User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the internet.

SECURITY

MMI FILTER
FIREWALL
NAT
VPN

Firewall Type

In_access Enable
 Out_access Enable

Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port
-------	-------------	----------	----------	----------	----------	----------	-------	------

Firewall Output Rule Table

Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port
-------	-------------	----------	----------	----------	----------	----------	-------	------

Firewall Set

Input/Output	Input	Src Addr				<input type="button" value="Add"/>
Deny/Permit	Deny	Des Addr				
Protocol Type	UDP	Src Mask				
Port Range	more than	Des Mask				

Rule Delete

Input/Output	Input	Index To Be Deleted		<input type="button" value="Delete"/>
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Access list support two type limits: input access or output access. Each type supports 10 items maximum.

C4110E firewall filter is base WAN port. So the source address or input destination address should be WAN port IP address.

Configuration:

in access enable Enable in access rule.

out access enable Enable out access rule.

Input/Output: Specify current adding rule is input rule or output rule.

Deny/Permit: Specify current adding rule is deny rule or permit rule.

Protocol Type: protocol using in this rule: TCP/IP/ICMP/UDP.

Port Range: port range if this rule.

Src Addr: Source address. can be single IP address or network address.

Dest Addr: Destination address. can be IP address or network address.

Src Mask: Source address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID.

Des Mask: Destination address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID.

2.7.3 VPN Tunnel configuration

SECURITY

MMI FILTERFIREWALLNATVPN

VPN IP

0.0.0.0

VPN Mode

UDP Tunnel L2TP Enable VPN

UDP Tunnel

VPN Server Addr	0.0.0.0	VPN Server Port	80
Server Group ID	VPN	Server Area Code	12345

L2TP

VPN Server Addr	VPN User Name
VPN Password	

2.8 Logout

After setting the C4110, user had better click Logout to end the management, keep C4110 in safe and authorized environment.

System Logout


Logout

Press the "Logout" button to Logout Phone !





3. Citel IP Phone C4110 Basic Operation


3.1 Making a call

-Using handset

Pick up the handset, and dial the desired number, or dial the desired number first and then pick the handset when your call is completed, put the handset on hook or press  key


-Using the speaker phone



With the handset on hook, press  key, dial the desired number on the dial pad, or dial the desired number and then press  or **Dial** soft key. When your call is completed, press  key or press  key again.

Note: During a call, you can alternate between handset and handsfree mode by picking up handset or press  key.



(Note: Please dial the number end with number”#” for quickly call set up. If you have two lines, press **SIP** soft key to select one of the lines.)



3.2 Answering a call

Answer a ringing call by picking up the handset or pressing the  key.


When your call is completed, put the handset on hook or press  key, or press  key again.

3.3 Using Hold and Mute

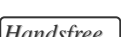
During a call, press  key will mute your handset or hands-free Microphone and you can hear the other party on the phone, the LED will light on while Mute is enabled. Press  again to speak with your caller.


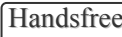
During a call, press the  key will place your caller on hold. To retrieve your call, press  key again. When you place your caller on hold, the caller may hear music on hold. You can not hear the speech of the caller.

3.4 Volume adjustment

4. Press  key to adjust handset and hands free speaker volume during a call,
5. Pressing the volume key in ringing or idle state adjusts the ringer volume.


3.5 Redial

Pick up handset or press  key, then press “Redial” softkey to redial the last dialed number.

Press “Redial” softkey first, then use  key to scroll 100 recently dialed numbers. When you get your desired number, pick up handset or press  key or press **Dial** soft key

to make a call.

3.6 Call lists

1. Press **Callers** key, then select “Callers” use  key or **Next** softkey to view missed, incoming, outgoing call list. Press **Enter** soft key to check detail caller information.
2. Press **Delete** soft key to clear the call records from the list
3. Press **Edit** soft key to save the number and name into Directory
4. press **Dial** soft key, then you can insert number or edit number before dialing and press **Call** soft key to return the call

3.7 Auto dial/Feature

There are four memory keys on the phone can be programmed as Auto dial or Feature Access key like, Ring again, Call forward or Pick up etc.

Press **Menu** key, follow the instruction on the screen to complete your Auto Dial keys programming. Pick up the handset, press the Auto dial key then the stored number will be dialed or press Auto dial key

first then followed by pressing **Handsfree** key or pressing **Dial** soft key or pick up handset.

3.8 Auto dial/Feature Access

(For programming Auto Dial key, please refer to the” Customizing your phone”)

3.9 Voice mail

Message waiting indicator (MWI) light will flash to indicate that messages are waiting at the voice mail or message center. Press voice mail access code,(like*97)to enter voice mail system and follow voice prompts to listen to the messages. When you heard messages, then MWI light will turn off.

Note:1. *Voice mail is an optional feature that must be configured on the call server. Particulars and menu options may vary.

2. When you have missed call or call records, the MWI light will light on as well.

3.10 N’pad(Note Pad)

N’pad gives the phone user a choice to stored a number on the screen and can be easily to dial out and save to the directory.

During a call, press **N’pad** soft key, dial the number you want to store. When you finish, press **Enter** soft key to save the number.

To retrieve the number in N’pad, press **N’pad** soft key, your stored number will display on the screen.

Use **Save** soft key to save into Directory, pick up handset or press **Handsfree** to dial this

number.

3.11 Call transfer (Blind and Consult)

-Blind Transfer

During a call, press this will make the caller on hold and not provide the dial tone, input transfer number with dial pad, press then hang up.

-Consult transfer

1. During a call, press key on the phone, this will make the caller on hold and provide dial tone, dial the transfer number with dial pad, press “#” or **Send** soft key
2. When the second party picks up, you may consult with them to transfer the call, then press to transfer the call.

3.12 Call waiting

When a second call is received while a user is engaged in a call, the user is informed via screen display and call waiting tone.

1. Press “Answer” softkey to answer the second call, this will place first call on hold
2. Press “resume” softkey, will resume the first call
3. To end the active call, press “End” or, press to end Talk calls.

3.13 Conference calling

-To create a three-way conference call:

1. Call the first party
2. press the “Conf” softkey on the phone, then call the second party. When the second party answers, press the **Conf.** soft key to add all callers to the conference call.

-To split a conference call

1. Press “Split” will make one of the callers on hold, keep to press this key will change call between two callers.
2. If you like to disconnect one of the callers, press “End” will end the active caller, then you can continue to talk to the other caller.


3.14 Directory /Phone Book

-To add or edit a contact in your phone directory.

1. Press the “Callers” key on the phone, select “PBook”, the screen will display “XX items in the list”
2. press a **Add** soft key to input your contact name and phone number with dial pad.

-

To search a contact number in the Directory

-
1. Press the “Callers” key on the phone,select “PBook, the screen will display “XX items in the list”
 2. Press **Enter** soft key, then use dial pad to input the contact name (letter) for searching your contact or use key to scroll.
 3. When find the desired contact, pick up handset, press Handsfree or press Dial softkey to call the contact.


3.15 Auto dial/Feature Access

User can save instant dialed number or feature access code on the four memory keys of the IP Phone.

1. choose the memory key then you will program , follow the instruction on screen to program features, like, Pick up, Ring Again, Message, Call Forward etc.or store the instant phone number on the memory keys.
2. When you have finished programming, remove the key cap, put a label in it to mark your function or stored numbers.

4 Trouble shooting

What's the Web GUI Default IP and Password?

Your phone system Default IP is **192.168.1.179** and username /password is **admin/admin**. If Phone is in DHCP Mode find the IP by pressing  Button.

The display is in an alternate language.

You can change the display language using the Set Options key. See **'To change display language'** of **'Setting options by using Menu'** for more information.


Why is the light not coming on with a new voice mail message?.

Your phone system or service provider must provide 'Visual' message waiting service for this function to work, check with your system administrator for more information.

How do I switching between two Number ?

Press the SIP soft key

How do I find the IP address of my phone ?

Press  and you can find it.


How do I make sure my phone was registered server?

Press   to check it

Why does my phone display 'Wan Disconnected' ?

Make sure your telephone was connected to the network.

How do I restart the IP phone ?

Enter into the Menu and press Next softkey or  to reach 'Reboot system', enter the **Reboot** softkey.

Where I can find the latest Firmware?

Latest Firmware can be downloaded from Citel website Firmware upgrade section at <http://www.citel.com/Products/ip-phones.asp>

Note: Your phone will be out of service temporarily during the restart and downloading process.