G102

Voice over IP Telefone

www.globaliptel.com

Table of Contents

Preface

Thanks for your interest in our product. At the following pages we would like to give you some information and hints for getting started with our phone quickly.

Unpacking and connecting

Remove whole packaging material and place the phone near to a power socket and a network port (Hub, Switch or DSL-Modem) at a dry place.

Insert handset cord into handset cord jack of the base. Plug the power cord adapter into the Power Jack. Then plug the other end of the power cord adapter into the appropriate power socket.

LAN users: Plug one end of the direct-connecting cable into RJ-45 jack which is located in the back of phone, and connect the other end of cable to hub.

ADSL/Cable Modem users: Plug the RJ-45 Ethernet crossing-over cable into the RJ-45 Ethernet Jack. Plug the other end of the cable into an ADSL/Cable modem router port.

Switch on the phone by using the switch located at the backside of the device.

The phone and its functions

Keys and functions

Missed:
Hang up and press to see missed calls

Answered:
Hang up and press to see answered calls

Local Num:
Hang up and press to see your number

Local IP:
Hang up and press to see your phone's IP

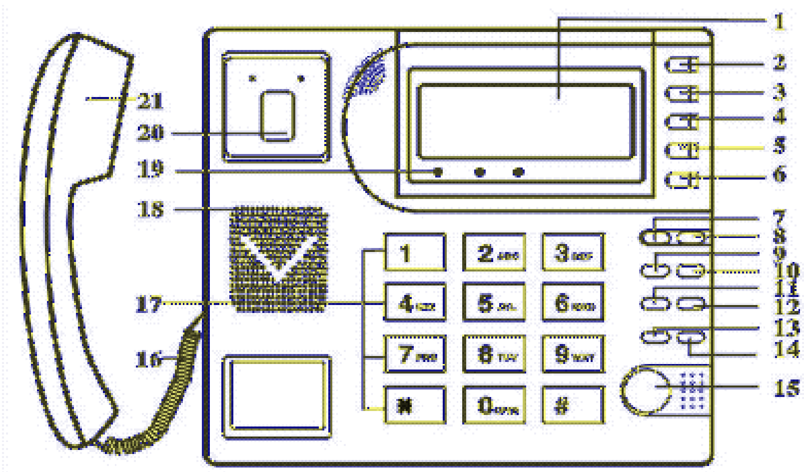
Dialed:
Hang up and press to see you dialed numbers

Redial:
Redial

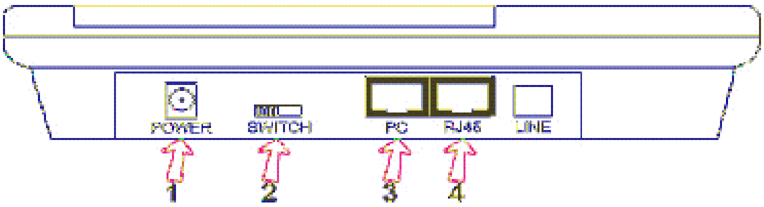
Speak:
Press to activate/deactivate hand free

Volume +/-:
Volume up and down

Number:
Dial a number or enter settings



1.	LCD	2.	Speed Dial	3.	Phone Book
4.	Back Space	5.	Volume +	6.	Volume -
7.	Service IP	8.	Local Number	9.	Missed
10.	Dialed	11.	Answered	12.	Call
13.	Local IP	14.	Redial	15.	Spk/Hand free
16.	Cord	17.	Number	18.	Speaker
19.	LED	20.	Handset		



1. Power adapter port
2. Power switch
3. PC(RJ-45) port
4. RJ-45 port

Configure phone by phone menu or WEB interface

You can configure your IP phone by phone menu or browser.

Configure the phone by phone menu

Use the keypad to enter the password of the phone (when debug is not set as 0[disable], default password is **1234**; when debug is set as 0[disable], please use super password **19750407**), and then press "#", till "Password:" is displayed. Then enter the password again and press "Spk" to let the phone enter setting mode.

Introduce of the function of keypad in the keypad setting mode

Press key	**Function**
Spk /Hand free	Enter into submenu of the current menu ;Acknowledge to modification
Volume/+	Scroll menu forward
Volume/-	Scroll menu backward
Local IP	Enter into modification status
Redial	Cancel current setting ; restore to its father catalogue
Back Space	Backspace during the setting
Number keypad	Input updating content according to require. Please see appendix for character represented by each key

Menu structure

```
Network Settings:
        - ip type -----------0[static]----------------- local ip

        - mac               |1[dhcp]                   | subnetmask
                            |2[pppoe]------|ppp id     | router
                            |3[modem]------|ppp pin    | dns

Protocol Settings:
        - protocol ----------0[h323]
        - service           |1 [sip]
        - servicetype       |2 [mgcp]
        - serviceaddr       |3 [net2phone]
        - serviceid
        - nattraversal
        - phonenumber
        - dtmf ------------------------------------0 [control string]
        - calltype ----------0 [normal]         |1 [inband audio]
        - account           |1 [faststart]      |2 [signal keypad]
        - pin               |2 [advanced]       |3 [rfc 2833]
        - registerport
        - signalport
        - controlport
        - rtpport
        - rtptos
        - registerttl
      - localtype
```



```
Phone Setting:
        - dialplan ---------------------------0[disable]     |- iddcode
        - innerline ---------0[disable]      |1[enable]------|- dddcode
        - answer            |1[enable]       |2[dialnum]     |-
iddprefix
        - ringtypel ---     |2[switch]       |3[prefix]      |-
dddprefix
        - digitmap    |-- 0-9[dtmf 0-9]
        - fwdalways   |10[not disturb]
        - fwdbusy     |11[pcmring]
        - fwdnoanswer |12[user define]
        - fwdpoweroff

Audio Settings:
        - audiotype ---------0[gt231]
        - audioframes       |1[g729]
        - vad               |2[]
        - agc               |3[g711u]
        - aec               |4[g711a]
        - handsetin         |5[auto]
        - handsetout
        - speakerout

Other Settings:
        - password
        - superpassword
        - debug --------------0[disable]
        - upgradeaddr        |1[output]
        - sntpip             |2[output all]
        - timezone           |3[remote debug]
        - daylight           |4[no check]
```


Setup by WEB interface

(If phone is connected to an same IP range as your computer or notebook.)

Enter your phone's IP into the address line of your web browser and click "Return".IP will be displayed if you hang up and key down "Local IP". A logon window appears where you can enter either password (**1234**) or super password (**19750407**). Then please click "Login".

Network

• Type of Connection:	Set how IP phone gets relevant network parameters by selecting corresponding item from drop down list..
- static:	Select this item to authorize users set IP address, subnet mask and router IP address of IP phone manually.
- dhcp:	Select this item to enable DHCP mode. With this system, your LAN or router automatically assigns all the required network parameters to any device connected to it when the device log on. The IP phone is shipped from the factory with DHCP on. So, if your LAN or router is configured to use DHCP addressing, the IP phone's LAN parameters will automatically be configured as soon as it is connected to the LAN or router and powered up.
- pppoe:	Those ADSL and Cable Modem users please select this item for it is a protocol especially designed for them. With this system, ADSL ISP automatically assigns all the required IP parameters to any device connected to it when the device log on.
- modem:	If the IP phone used with modem, please select this item to get relevant network parameters auto. Then please fill ID and pin into ppp id and ppppin fields.
• ADSL/DSL Username:	With pppoe or modem selected in iptype drop down list, please enter the user name here.
• Password / PIN:	With pppoe or modem selected in iptype drop down list, please enter the password here.
• IP address of IP Phone:	With static ip selected in iptype drop down list, please enter IP address of IP phone here.
• Subnet mask:	With static ip selected in iptype drop down list, please enter subnet mask of IP phone here.
• Router IP Adress:	With static ip selected in iptype drop down list, please enter router IP address of IP phone here.
• DNS 1:	With static ip selected in iptype drop down list, please enter IP address of DNS server here.
• DNS 2:	With static ip selected in iptype drop down list, please enter IP address of backup DNS server here.
• MAC:	MAC address is the physical address supplied by the Ethernet NIC. The phone is shipped from the factory with a unique algorism MAC address printed on the back of the base.

Protocol settings

• Protocol:	Select an item from dropdown list to set the protocol used by the phone (do not have all devices).
• Service type:	Enable/disable service by checking/unchecking this box (do not have all devices).
• Server to log in:	Enter service ID if required.
• Domain name / Realm:	Enter service's IP or Domain.
• RTP Tye of service:	Set the TOS field of the IP header of the RTP packets. The bigger this value is, the higher priority the packet has.

• RTP Port:	RTP port is the port transferring and receiving voice packets using UDP protocol. This is an even number between 1024 and 65535, can’t be the same as “register port”.
• Register Port:	Depends on used protocoll. So please enter according value:
- H323	Number between 1024 and 65535.
- MGCP	2427
- SIP	5060 (remember in putting up this value for each additional phone within your network)
- Net2phone	Any number of provider's choise.
• Signal port:	By using H323 it is a port between 1024 and 65535.
• Control port:	By using H323 it is a port between 1024 and 65535.
• Type of account:	
- phone number:	Use phone number as E.164 and H323 ID to login the GK.
- account:	Use phone number as E.164 and designated H323 ID filled in account field as H323 ID to login GK.
- H235 account:	Enable or disable H235 encryption.
• Call type:	
- normal:	Call out in normal way by selecting this item.
- faststart:	Call out in faststart way by selecting this item.
- advanced:	Call out in faststart and tunneling way by selecting this item. It is a recommended way with H323 protocol used.
• DTMF:	Set DTMF signal sending way by selecting control string, inband audio, signal keypad or rfc 2833 from dropdown list.
• Service type:	Leave as is if you do not use prepaid card system.
• Password / PIN:	password or pin.
• Service ttl:	This parameter is responsible for how long keep alive message will be sent (do not have all devices). - TTL min.10 seconds, max. 255. 60 seconds is default.

Phone Settings:

• Phone Number to fwd to :	Enter phone number or username if running SIP mode. Phone number you want to forward to.
• If powered off:	Check if incomming call should be forwarded if phone is switched off.
• Always:	Check if incomming call should allways be forwarded.
• If busy:	Check if incomming call should be forwarded if busy.
• On no answer:	Check if incomming call should be forwarded if no answer.
• Use dial plan:	Use dialplan or forward number.
• Dial Number:	With dialnum selected in use dialplan drop down list, please enter the dial prefix into this field according to requirement of log in server.
• City code:	With enable or dialnum selected in use dialplan drop down list, set area code according to E.164 dial rule.
• County code:	With enable or dialnum selected in use dialplan drop down list, set country code according to E.164 dial rule.
• International prefix:	With enable or dialnum selected in use dialplan drop down list, set international call prefix according to E.164 dial rule, such as 00.
• Local prefix:	Enable/disable multi-settings by selecting corresponding items from dropdown list. With enable or switch selected in innerline dropdown list, please fill the number switching to backup setting 1 here, such as 56.
• Nonlocal prefix:	With **enable** or **switch** selected in **innerline** dropdown list, please fill the number switching to backup setting 2 here, such as 57.

• Use Digitmap:	Enable/disable digitmap by checking/unchecking the box (do not have all devices).
• Ring type:	Set ring type by selecting corresponding item from drop down list.
• Rime to answer:	Enter a number from 0 through 60 to set the entries of the seconds before the phone answer the call auto or forward the calls. To disable auto answer function, please set this parameter as 0.
• predial time:	Set time limit from picking up the speaker to dialing the first number (do not have all devices).
• interdial time:	Set time limit between dialing two numbers (do not have all devices).
• postdial time:	Set time limit from dialing the last number to placing a call. If the next number is not dialed within the post dial time limit, then the phone will call the dialed number auto (do not have all devices).

Audio settings:

• audio type:	Set audio type of the phone by selecting item from drop down list.
• audio frame:	Please set "1" if using G723 and "2" if using G729.
• g.723.1 high rate:	With g.723.1 selected in audio type dropdown list, enable/disable g.723.1 high rate by checking/ unchecking this option.
• VAD:	Enable/disable VAD (voice activity detection) by checking/ unchecking this box.
• AGC:	Enable/disable AGC by checking/ unchecking this box.
• AEC:	Enable/disable AEC by checking/ unchecking this box.
• Microphone:	Drag the slider to adjust the volume of handset input. Drag it to the left to reduce the volume; while drag it to the right to increase the volume.
• Handset speaker:	Drag the slider to adjust the volume of handset output. Drag it to the left to reduce the volume; while drag it to the right to increase the volume.
• Speaker:	Drag the slider to adjust the volume of handfree output. Drag it to the left to reduce the volume; while drag it to the right to increase the volume.

Other settings:

• Password:	Set the password of the phone. (Default password is 1234).
• DEBUG Modus:	Set the debug level of the phone.
• Upgrade adress:	Enter upgrade address (do not have all devices).
• Time server:	Fill IP address of time server here.
• Daylight savings:	Enable/disable daylight by checking/unchecking this box.
• Time zone:	Select correct time zone in dropdown list.
• Update:	Save and reboot.
• Adress book:	Save names and numbers for phone book and speed dial.
• Update Firmware:	Install new firmware from local folder

Receive calls, give someone a call and have a look at the historie

Placing and receiving a call is as easy as doing with a PSTN phone.

Receiving calls
The IP phone can receive incoming calls from other devices that support the same protocol. It works just like an ordinary phone for incoming calls. When it rings, you can receive the call by following methods:

Lift the handset or press "Hand free" and begin speaking. When the call is over, put the handset back or press "Hand free" again if you have spoken with "Hand free".

Note
When you communicate with the other party without lifting the handset, please do not exceed 40 CM from speaker.

Place a call
Pick up the handset and listen for the Internet dial tone. Then dial the phone number you wish to call and press "#" or "Call" to end the dialing. Once the call connection has been established and the ring tone has sounded, wait for the other party to answer. When the other party answers, you can begin speaking. When the call is over, put back the handset. The dialed number has been saved into the buffer.

Please take note that it is mosty neccesarry to dial the country code as prefix if you are dialing a PSTN number.

Press "Hand free", input the phone number you wish to call and press "#" or "Call" to dial when you have not lifted the hend set.

Missed calls
Press "Missed" if you want to see missed calls. You can sort the listing by keydown "Vol/+". After having pressed "Vol/-" the numbers will be sorted by incoming time.

Answered calls
Press "Answered" if you want to see answered calls. You can sort the listing by keydown "Vol/+". After having pressed "Vol/-" the numbers will be sorted by incoming time.

Dialed numbers
Press "Dialed" if you want to see dialed numbers.

You can directly place a call to the visible number by pressing "#".

Technical data, protocols and electronic information

Functions and features

H.323 functions with special firmware only. Support H.323 v4,compatible with most H.323 v1-v4 system and devices; Built in H.323 proxy support to pass NAT; Support MGCP RFC2705, Support SIP RFC3261; Support Net2phone private protocol; Fast start and H.245 tunneling; Outband DTMF transmit by H.245 user input or Q.931 keypad; IEEE 802.3 /802.3 u 10 Base T / 100Base TX; Major G.7XX and gsm610 audio codec; Caller ID send and receive by Q.931; Provide direct IP address to IP address call mode, direct PSTN call by a voice gateway ,E.164 phone number call under the mode of GK; Automatically gatekeeper discovery in LAN; Call PSTN by ITSP's prepaid card (eTalk,italk,ringtec etc); DHCP support for automatically assign IP address and others relevant parameters; PPPoE support for ADSL or Cable modem; Setting IP Net Phone parameters by standard web browser (such as IE6.0), phone keypad or standard telnet; Upgrade program by FTP mode; Support G.723.1 5.3k/6.3k; G.729; G.711 A-Law & U-Law audio codec algorithm; Dynamic voice detection; Echo cancellation; Comfort noise generation; Dynamic voice jitter buffer which minimize effect to the voice caused by the audio delay and jitter and as a result the quality of voice is high; Tone generation and Local DTMF generation and detection according with ITU-T; E.164 dial plan and customized dial rules; 80 entries each for missed call, answered call and dialed call; 112 entries for quick dial; 16 entries for voice message; 2 x 16 digits LCD display dial data; caller name; caller number and so on ; Working status indicating Lamps (red, yellow and green) and keypad backlight; Independently digit adjust the volume of handset and hand free; 14 function keys for operating and setting phone besides standard keys 0-9,#; Speed dialing; Adjustable volume for both handset and speaker; 16 function keys, background LED and states indicating lights; Settings by HTTP web browser (IE6.0); Advanced settings by Telnet; Voice prompt; Upgrade by TFTP; Hotline

Main technical index
Main chip: 50Mhz;
Data storage:2MB SDRAM;
Program memory: 1MB Flash memory;
Application Network environment: 10Base-T/100Base-T;
Echo cancellation:G165 16ms;
Store quick dial number: 100;
Record phone number of missed call:80;
Power loss:2.7W(max);
Power adapter: input AC 220V,output DC 9V 500mA;
Employing condition: Ambience temperature 0-40° C / -32-104 F
Relative humidity 10-90%
Atmosphere pressure 86-106Kpa;
Overall size:215×190×70mm(L×W×H).

Standard and Protocol
H.323 V4 calling signal protocol;
MGCP RFC2705
SIP RFC3261
Net2phone,IAX2,WP private protocol
IEEE 802.3 10Base-T/100Base-TX RJ-45 port;
Major G.7XX and gsm610 audio codec
Audio codec algorithm include G.711A,G.711u,G.723.1(5.3k/6.3k) and G.729;
Quick dial setting and user-defined dial plan setting;
TCP/IP: Internet Transport and Control Protocol;
RTP: Real-time transport Protocol;
RTCP: Real-time Transport Control Protocol;
G.723.1• AG729 VAD/CNG economical bandwidth;
G165 16ms Echo cancellation;
DTMF: Tone Detection and Generation;
DHCP: Dynamic Host Configuration Protocol;
PPPoE: Point to Point Protocol Over Ethernet;
DNS: Domain Name Service;
NTS: Network Time service;
Telnet: Remote Host log on Protocol;
FTP: File Transfer Protocol;
HTTP: Hyper Text transfer Protocol;
Build in H.323 proxy support;

Auto search proxy

Electric requirements
Voltage: 9V DC
Power: 0.5W (max.)
Power adapter: AD/DC input 110-230V, 1A 400Ma
Network interface:1 or 2 RJ-45 Ethernet Connect

Operating requirements
Operation temperature: 0 to 50° C (-32° to 122° F)
Storage temperature: -30° to 65° C (-22° to 149° F)
Humidity: 10 to 90% no dew

Note

Further information available at our WebPages (please refer to Important Information).

If there are questions according to a special user account please talk to your VoIP provider's support.

Introduction

IP is abbreviation of Internet Protocol. An IP phone is a telephone transporting voice using grouping data package of IP protocol. It can be used widely for audio communication in a broad band IP Network environment which accord with TCP/IP protocol, such as in the LAN or WAN of Enterprises and Institutions, Telecom Providers, IP phone services provider's network and broad band INTERENT user, who log on internet through LAN, Cable Modem or XDSL and so on.

The most significant benefit of IP Telephony is cost savings achieved through transporting high quality voice messages over data communication networks. Using IP Telephony, you will be save dramatically on both domestic and international calls.

Our IP Telephone uses unique generalized out line and inner line modes. It functions much like an ordinary telephone switching between inner line and out line, so it supplies great conveniences to the users. When the IP Phone is in generalized inner line mode, it can call another IP device such as another IP phone worldwide for free. When the device is in generalized out line mode, it can place calls to ordinary telephones worldwide at a dramatically low price, because the phone supports using prepaid card supplied by ISP such as Net2phone or eTalk. Moreover, it possesses excellent sound quality just like an ordinary phone. The IP Phone has 2 x 16 English LCD with backlit and supports keypad setup.

Suitable use of the IP Phone

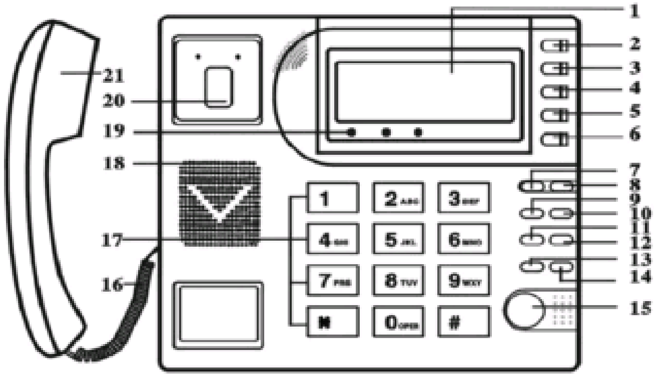
Our IP Phone is an inexpensive starting point for Voice over IP and is primarily designed for personal usage.

The phone is for indoor use only. Keep from direct and indirect contact with water. Do not send the box and hand it to by us autorised person for repair work. You will loose warranty claim if phone case has been opened. Setup phone in a safe, fixed location, ensuring that cables are placed in a manner that does not endanger yourself or others, including children. When cleaning phone, use dry cloth only, without any fluids or thinners. Prior to cleaning, it is necessary to switch the phone off and unplug power cable. You will also have to separate from power supply before maintenance work and if you will discover any electrical defect.

WICHTIG : DO NOT INSTALL A FIRMWARE WHICH IS NOT DESIGNATED FOR THE DEVICE. WITH A WRONG FIRMWARE YOU CAN MAKE THE PHONE UNUSABLE OR YOU CAN DESTROY IT. WE ARE EXEMPT FROM LIABILITY FOR FIRMWARE UPDATES WHO ARE MADE BY CUSTOMERS. FIRMWARE UPDATES WILL BE DONE BY YOUR OWN RISK.

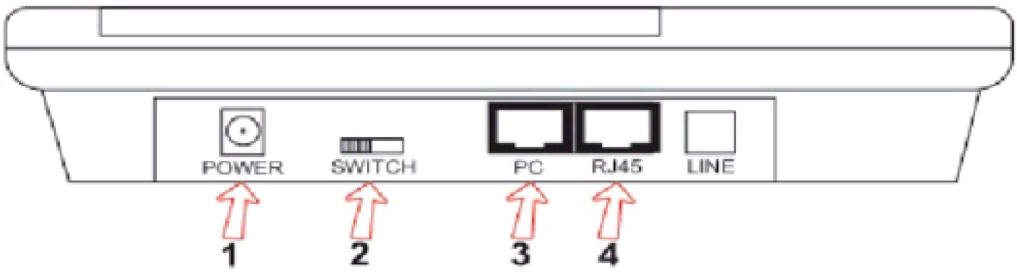
The phone

1. Front panel



Front panel

1.	LCD	2.	Speed Dial	3.	Phone book
4.	Backspace	5.	Volume +	6.	Volume -
7.	Service IP	8.	Local Number	9.	Missed
10.	Dialed number	11.	Answered	12.	Call
13.	Local IP	14.	Redial	15.	Spk / Hand free
16.	Cord	17.	Number	18.	Speaker
19.	LED	20.	Handset		



2. Backside view

1. Power adapter port   2.Switch   3. RJ-45 for PC   4. RJ-45

3. Function keys

Function Keys of IP Phone Introduction:

Missed:
With handset hung, press this key to review missed number

Answered:
With handset hung, press this key to review received number

Local Num:
With handset hung, press this key to get phone number

Dialed:
With handset hung, press this key to review dialed number

Redial:
While reviewing missed, received or dialed number, press this key to

dial current number

Speak:
Press this key to have a call without lifting the handset

Volume+:
Increase the volumes of handset or speaker; turn over the record backward

Volume-:
Decrease the volumes of handset or speaker; turn over the record forward

Keypad:
With handset picked or pressing speaker, press this key to dial number

Performance and Features

- Support H.323 v4,compatible with most H.323 v1-v4 system and devices;
- Built in H.323 proxy support to pass NAT;
- Support MGCP RFC2705
- Support SIP RFC3261
- Support Net2phone private protocol
- Fast start and H.245 tunneling;
- Outband DTMF transmit by H.245 user input or Q.931 keypad;
- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- Major G.7XX and gsm610 audio codec
- Caller ID send and receive by Q.931;
- Provide direct IP address to IP address call mode, direct PSTN call by a voice
- gateway ,E.164 phone number call under the mode of GK;
- Automatically gatekeeper discovery in LAN;
- Call PSTN by ITSP's prepaid card (eTalk,italk,ringtec etc);
- DHCP support for automatically assign IP address and others relevant parameters;
- PPPoE support for ADSL or Cable modem;
- Setting IP Net Phone parameters by standard web browser (such as IE6.0), phone
- keypad or standard telnet;
- Upgrade program by FTP mode;
- Support G.723.1 5.3k/6.3k & G.729 & G.711 A-Law & U-Law audio codec algorithm ;
- Dynamic voice detection; Echo cancellation; Comfort noise generation;
- Dynamic voice jitter buffer which minimize effect to the voice caused by the audio
- delay and jitter and as a result the quality of voice is high;
- Tone generation and Local DTMF generation and detection according with ITU-T;
- E.164 dial plan and customized dial rules;
- 80 entries each for missed call, answered call and dialed call;
- 112 entries for quick dial;

- 16 entries for voice message;
- 2 x 16 digits LCD display dial data; caller name; caller number and so on ;
- Working status indicating Lamps(red• Ayellow and green) and keypad jacklight;
- Independently digit adjust the volume of handset & hand free;
- 14 function keys for operating and setting phone besides standard keys 0-9,#;
- Speed dialing;
- Adjustable volume for both handset and speaker;
- 16 function keys, background LED and states indicating lights;
- Settings by HTTP web browser (IE6.0);
- Advanced settings by Telnet;
- Voice prompt;
- Upgrade by TFTP.
- Hotline

Main technical index

Main chip:
RISC CPU mit Koprozessor, 50MHz

Data storage:
2MB SDRAM

Program memory:
1MB Flash mamory

Application Network environment:
10Base-T/100Base-T

Echo cancellation:
G165 16ms

Speed dial numbers:
100

Record phone number of missed call:
80

Power loss:
2.7W (max)

Power adapter:
input AC 220V
Output DC 9V 500mA

Employing condition:
Ambience temperature 0-40° C (32°-104°F)
Relative humidity 10-95%
Atmosphere pressure 86-106Kpa;

Overall size:
215×190×70 mm (L×B×H).

Standard and protocols

- H.323 V4 calling signal protocol
- MGCP RFC2705
- SIP RFC3261
- Net2phone,IAX2,WP private protocol
- IEEE 802.3 10Base-T/100Base-TX RJ-45 port
- Major G.7XX and gsm610 audio codec
- Audio codec algorithm include G.711A,G.711u,G.723.1(5.3k/6.3k) and G.729
- Quick dial setting and user-defined dial plan setting
- TCP/IP: Internet Transport and Control Protocol
- RTP: Real-time transport Protocol
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- G165 16ms Echo cancellation
- DTMF: Tone Detection and Generation
- DHCP: Dynamic Host Configuration Protocol
- PPPoE: Point to Point Protocol Over Ethernet
- DNS: Domain Name Service
- NTS: Network Time Service
- Telnet: Remote Host log on Protocol
- FTP: File Transfer Protocol
- HTTP: Hyper Text transfer Protocol
- Build in H.323 proxy support
- Auto search proxy

Electric requirements

Voltage:
9V DC

Power:
0.5W (max.)

Power adapter:
AD/DC input 110-230V, 1A 400 mA

Network:
2 RJ-45 Ethernet ports

Operating requirements

Operation temperature:
0 to 50° C (32° to 122° F)

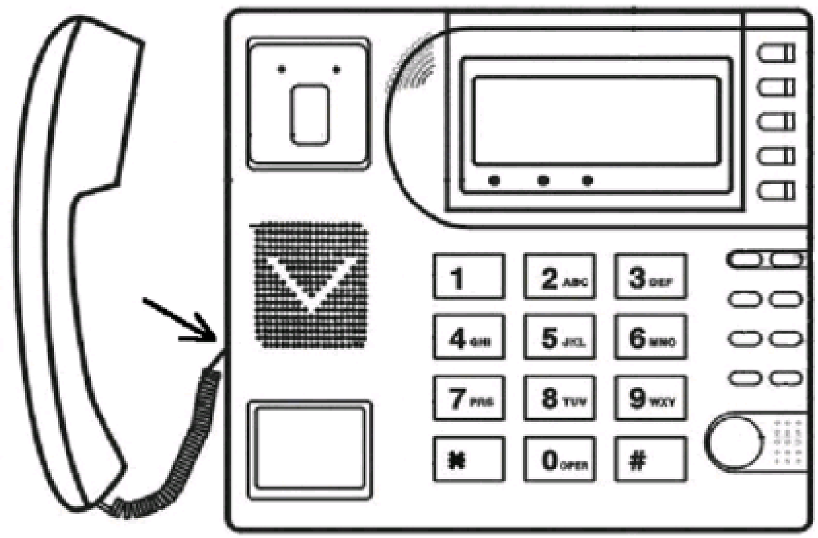
Storage temperature:
-30° to 65° C (-22° to 149° F)

Humidity:
10 to 90% no dew

Installation

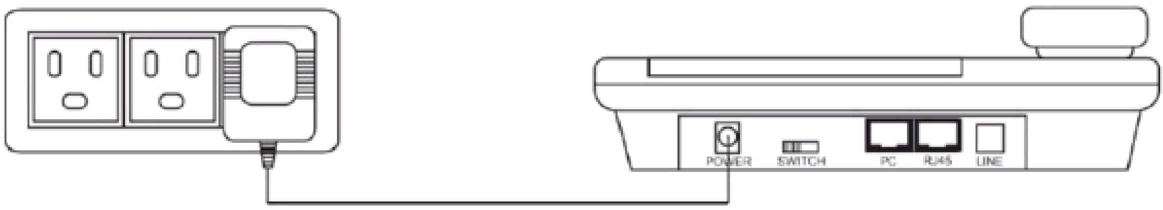
1. Connect handset and phone

Insert handset cord into handset cord jack of the base.



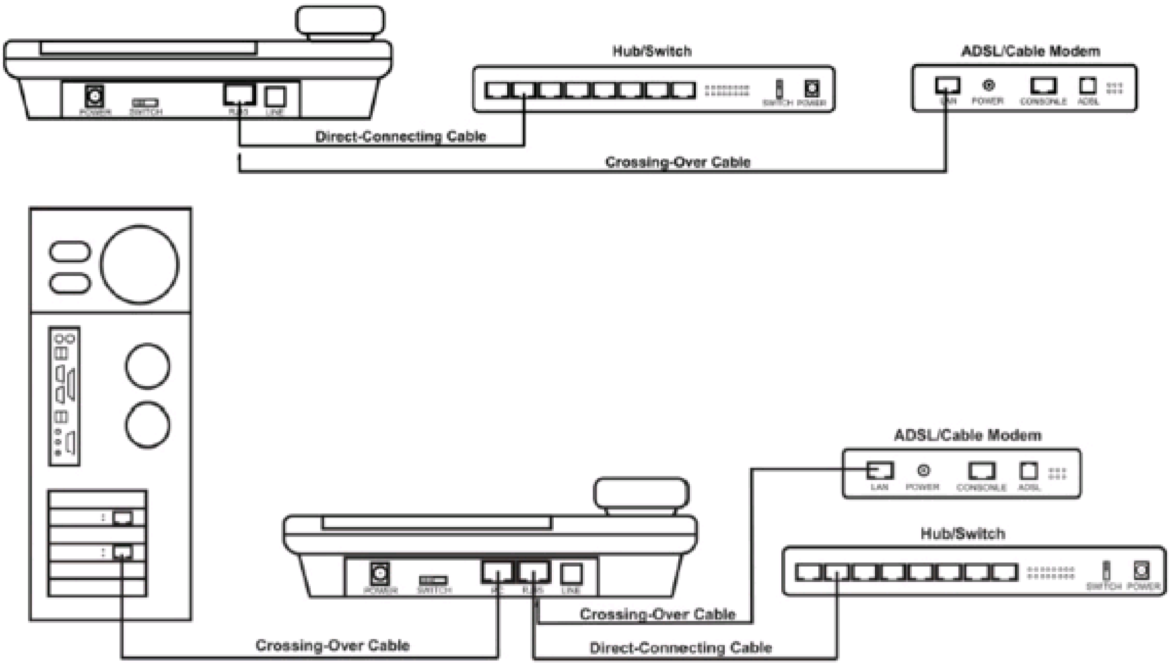
2. Connect phone and power

Palce the phone nearby of Ppwer socket.Plug the power cord adapter into the power jack. Then plug the other end of the power cord adapter into the appropriate power socket.



3. Connect the phone into the net

LAN users: Plug one end of the direct-connecting cable into RJ-45 jack which is located in the back of phone, and connect the other end of calbe to hub.



ADSL/Cable Modem users   Plug the RJ-45 Ethernet corssing-over cable into the RJ-45 Ethernet Jack. Plug the other end of the cable into an ADSL/Cable modem router port.

4. Start phone

Turn on the phone by pushing the switch to ON. Verify that yellow, green and red lights are on together, and then red light is off; green light blinks or is off ; yellow light blinks or is on, which behalf the success of starting phone and phone enter into normal standby.

Configuring by phone menu

Delivery condition default password of the device is 1234.

1) Entering into setting mode

Note:
Use the keypad to enter the password of the phone (when debug is not set as 0[disable], default password is 1234; when debug is set as 0[disable], please use super password 19750407), and then press "#" untill "Password:" is displayed on the phone. Then enter the password again and press "Spk" to let the phone enter setting mode. Alphanumerical digits could be entered by pressing a key multiple. Navigate the menu to "Save" or "Exit" by using the "Vol -" key if you want to save changes or to exit without saving.

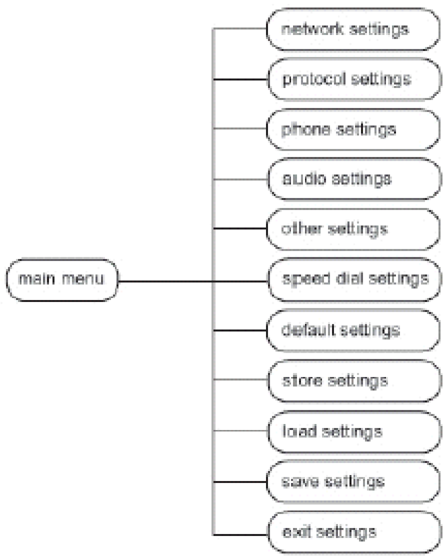
When phone is off:
Key down "#" and switch on the phone. Enter password and press "Hands free".

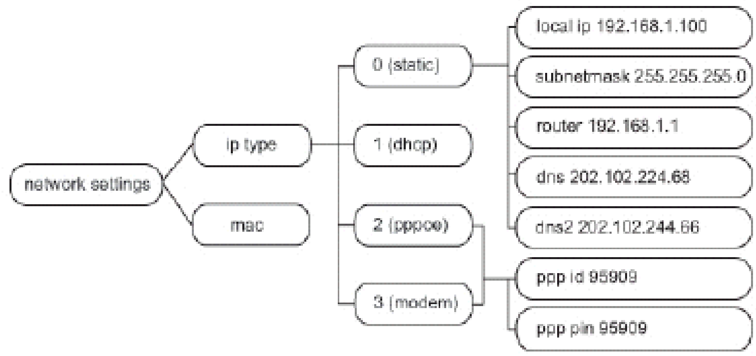
When phone is on:
Use dial keys for entering the password. Press "#" until "Password:" is displayed. Enter password again and press "Hands free" for entering the configuration menu.

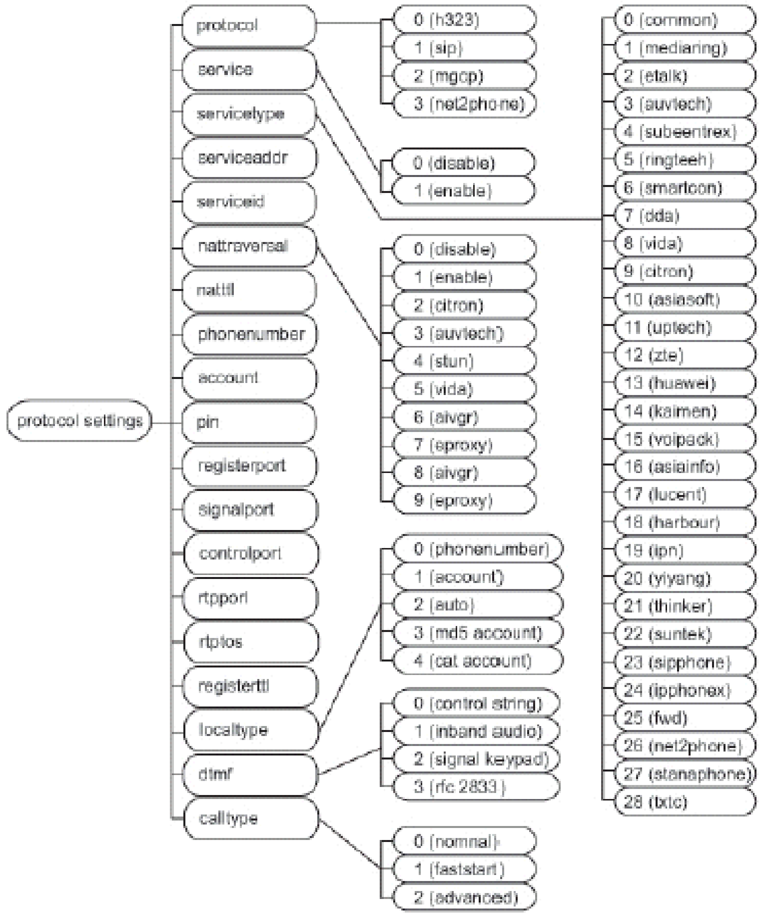
2) Introduction of keypad functions in keypad setting mode

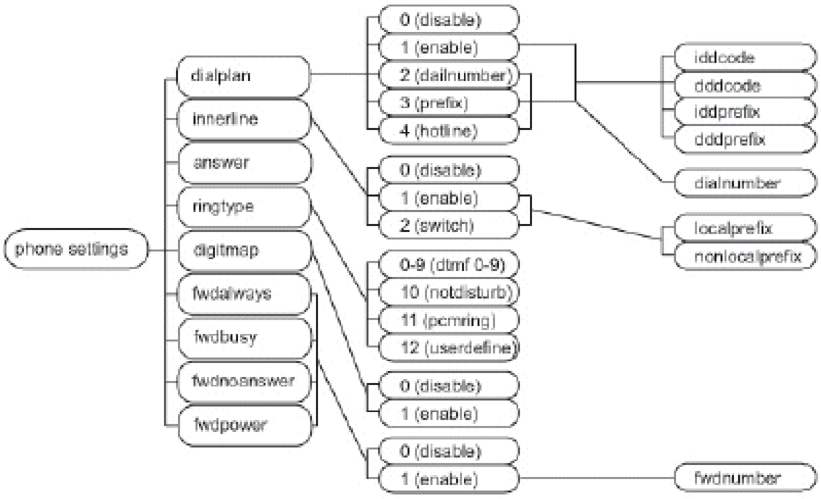
Press key	Function
Spk /Hand free	Enter into submenu of the current menu ;Acknowledge to modification
Volume/+	Scroll menu forward
Volume/-	Scroll menu backward
Local IP	Enter into modification status
Redial	Cancel current setting ; restore to its father catalogue
Back Space	Backspace during the setting
Number keypad	Input updating content according to require. Please see appendix for character represented by each key

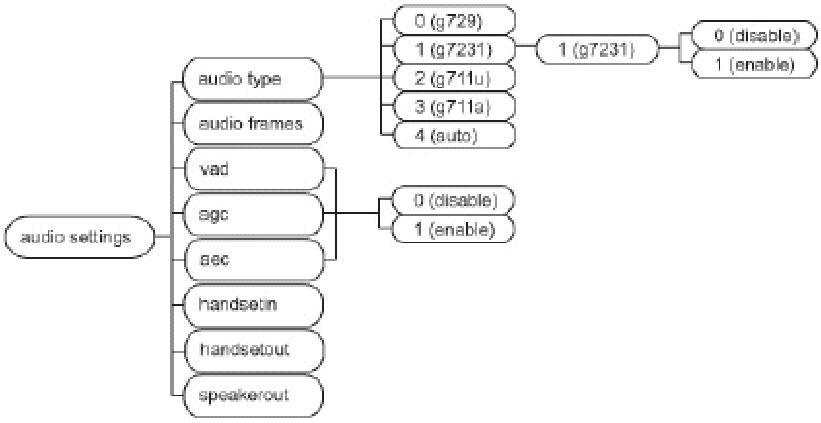
3) Menu structure




network settings
ip type
mac
0 (static)
1 (dhcp)
2 (pppoe)
3 (modem)
local ip 192.168.1.100
subnetmask 255.255.255.0
router 192.168.1.1
dns 202.102.224.68
dns2 202.102.244.66
ppp id 95909
ppp pin 95909


protocol settings
protocol
service
servicetype
serviceaddr
serviceid
nattraversal
nattl
phonenumber
account
pin
registerport
signalport
controlport
rtpport
rtptos
registerttl
localtype
dtmf
calltype
0 (h323)
1 (sip)
2 (mgcp)
3 (net2phone)
0 (disable)
1 (enable)
0 (disable)
1 (enable)
2 (citron)
3 (auvtech)
4 (stun)
5 (vida)
6 (aivgr)
7 (eproxy)
8 (aivgr)
9 (eproxy)
0 (phonenumber)
1 (account)
2 (auto)
3 (md5 account)
4 (cat account)
0 (control string)
1 (inband audio)
2 (signal keypad)
3 (rfc 2833)
0 (nomnal)
1 (faststart)
2 (advanced)
0 (common)
1 (mediaring)
2 (etalk)
3 (auvtech)
4 (subeentrex)
5 (ringteeh)
6 (smartcon)
7 (dda)
8 (vida)
9 (citron)
10 (asiasoft)
11 (uptech)
12 (zte)
13 (huawei)
14 (kaimen)
15 (voipack)
16 (asiainfo)
17 (lucent)
18 (harbour)
19 (ipn)
20 (yiyang)
21 (thinker)
22 (suntek)
23 (sipphone)
24 (ipphonex)
25 (fwd)
26 (net2phone)
27 (stanaphone)
28 (txtc)





4) Table

The following table shows the digits and number of presses you will need to receive a symbol.

Key pressed	1x	2x	3x	4x	5x
1	1	.	,	?	
2	2	A/a	B/b	C/c	
3	3	D/d	E/e	F/f	
4	4	G/g	H/h	I/I	*
5	5	J/j	K/k	L/l	
6	6	M/m	N/n	O/o	#
7	7	P/p	Q/q	R/r	S/s
8	8	T/t	U/u	V/v	
9	9	W/w	X/x	Y/y	Z/z
0	0	Space	:/@	;/-	'/&
*					
#	Change between upper and lower case				

Please restart the IP phone manually after you have changed the settings if you want to ensure that they will become effective.

Configuration

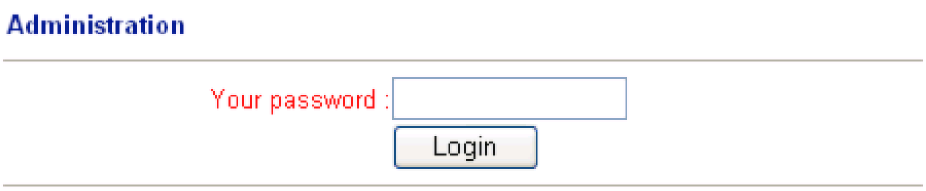
There are three different ways for configuring the device:

By phone keys, web browser or telnet commands.

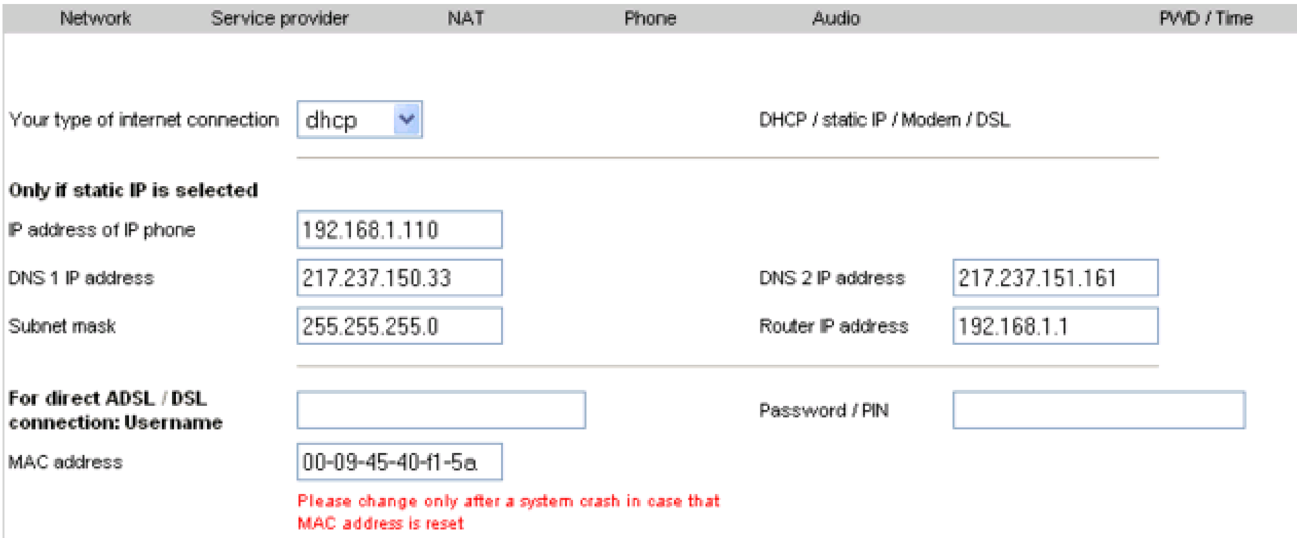
Configure by WEB

On a PC connected with the phone or at the same segment of the phone, open your Internet browser. Input the IP address of the phone into address bar and then input password of the phone into the following page.

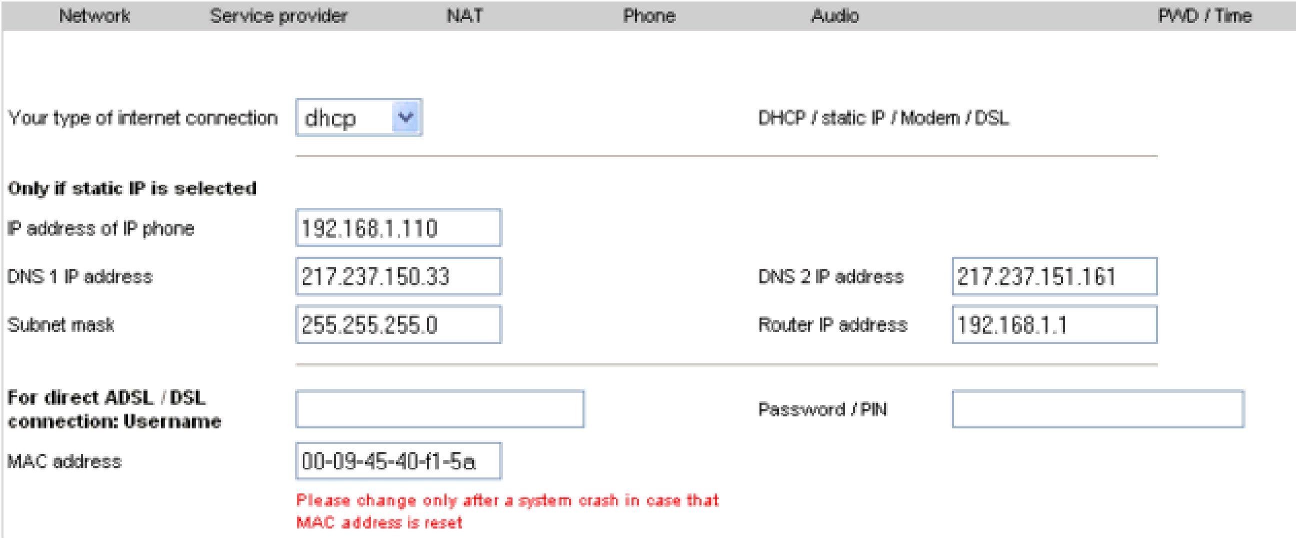
There are two passwords for the IP phone:
Ordinary password and super password. Default password 1234 is ordinary password and super password is 19750407. With Debug set 0[disable], please input super password; while Debug is not set as 0[disable], please input ordinary password . Then click [login] button. The following configured page will pop up:



1. If you have entered a valid password and pressed [login] the following window appears.



Network settings



(1) Network Setting

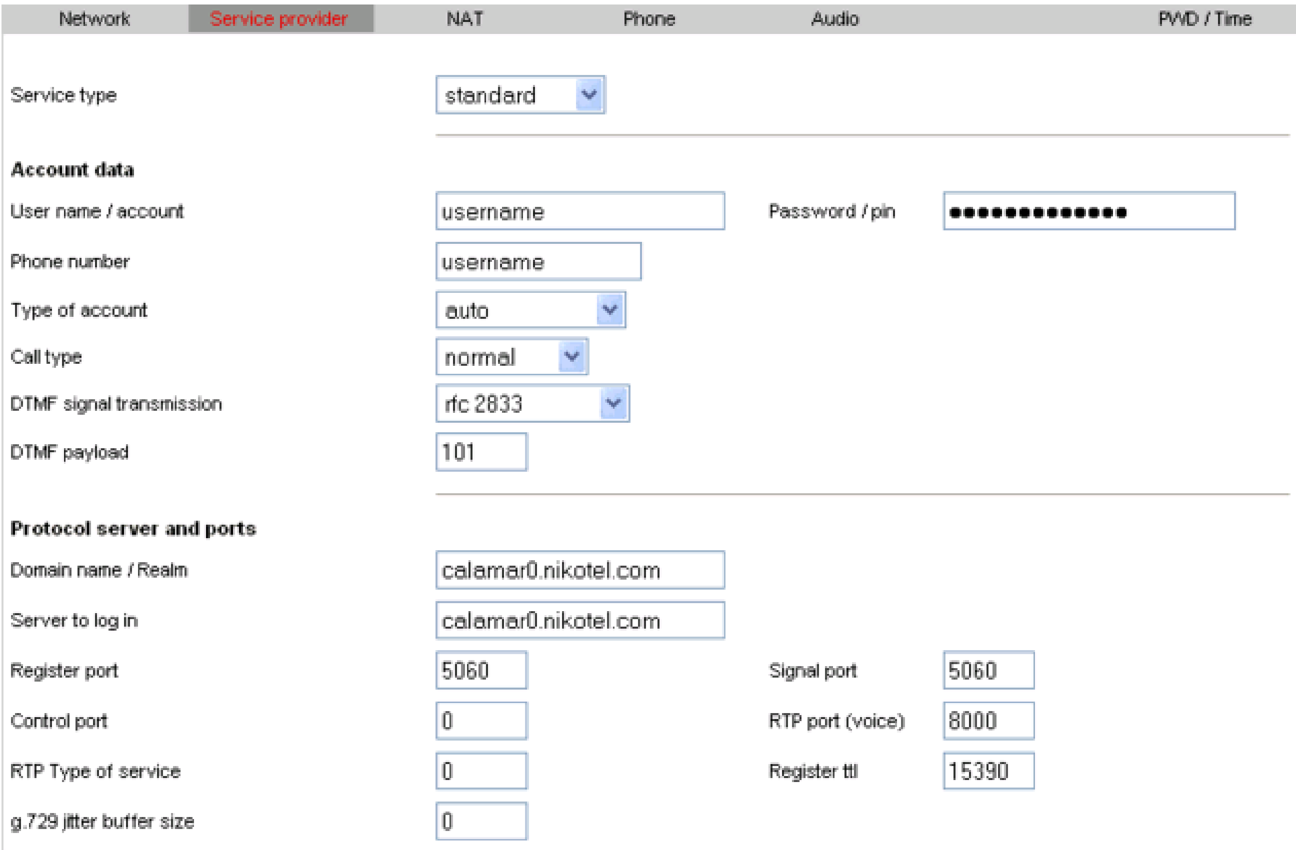
- **iptype**:
  Set how IP phone gets relevant network parameters by selecting corresponding item from drop down list.
  - **static ip:**
    Select this item to authorize users set IP address, subnet mask and router IP address of IP phone manually.
  - **dhcp:**
    Select this item to enable DHCP mode. With this system, your LAN or router automatically assigns all the required network parameters to any device connected to it when the device log on. The IP phone is shipped from the factory with DHCP on. So, if your LAN or router is configured to use DHCP addressing, the IP phone's LAN parameters will automatically be configured as soon as it is connected to the LAN or router and powered up.
  - **pppoe:**
    Those ADSL and Cable Modem users please select this item for it is a protocol especially designed for them. With this system, ADSL ISP automatically assigns all the required IP parameters to any device connected to it when the device log on.
  - **modem:**
    If the IP phone used with modem, please select this item to get relevant network parameters auto. Then please fill ID and pin into ppp id and ppppin fields.
- **ppp id:**
  With pppoe or modem selected in iptype drop down list, please enter the user name here.
- **ppp pin:** With pppoe or modem selected in iptype drop down list, please enter the password here.
- **local ip:** With static ip selected in iptype drop down list, please enter IP address of IP phone here.
- **subnet mask:** With static ip selected in iptype drop down list, please enter subnet mask of IP phone here.
- **router ip:** With static ip selected in iptype drop down list, please enter router IP address of IP phone here.

- **dns:** With static ip selected in iptype drop down list, please enter IP address of DNS server here.

- **dns 2:** With static ip selected in iptype drop down list, please enter IP address of backup DNS server here.

- **mac:** MAC address is the physical address supplied by the Ethernet NIC. The phone is shipped from the factory with a unique algorism MAC address printed on the back of the base.

Access and protocoll settings



2) Access and protocol settings

- **protocol:** Select an item from dropdown list to set the protocol used by the phone.

  - **sip:** Select this item to set the phone use SIP protocol.
  - **h323:** Select this item to set the phone use H323 protocol.
    mgcp: Select this item to set the phone use MGCP protocol.
  - **n2p:** Select this item to set the phone use Net2phone private system.
  - **iax2:** Select this item to set the phone use IAX2 private system.
  - **wp:** Select this item to set the phone use WP private system

Note:
With Net2phone selected here, please set other necessary parameters: check use service option, and then fill IP address or domain name of designated server into the "Domain name/Realm" field; then set service port as 6801; fill account and password of Net2phone card into account and pin fields.

- **use service:** Enable/disable service by checking/unchecking this box. Different service responses different protocol as follows:

  With H323 protocol used, the protocol service refers to the gatekeeper searching the address. To let IP phone call each other by E.164 number, please check this box and then fill the IP address or domain name of corresponding gatekeeper into the "Domain name/Realm" field. Without this check box being selected, the phone can call by gateway or just by dialing IP address of other IP phone or H323 device (such as Netmeeting) at the same network segment.

  With MGCP protocol used, the protocol service refers to Call Agent. Please check this box according to system, and then fill the "Domain name/Realm" field with Call Agent IP address or domain name.

  With SIP protocol used, the protocol service refers to SIP Proxy Server. Check this box according to system, and then fill the "Domain name/Realm" field with SIP Proxy Server IP address or domain name.

With Net2phone protocol used, the protocol service refers to designated server.

Please check this box, and then fill the designated IP address or domain name into the "Domain name/Realm" field.

Note:
Designated Net2phone server IP address are: 216.53.3.52; 4.43.114.39; 4.43.114.38 or 205.228.245.8. Domain names are: call1.net2phone.com; call2.net2phone.com; skip1.net2phone.com; skip2.net2phone.com; skip1.f8g9h0.net or skip2.f8g9h0.net.

- **Service type**: This option is used to accommodate the miscellaneous requirements of the system providers. When IP phone is connected to these systems, please select the corresponding service type. Please ask your service provider for all required settings.

- **nat traversal:** When the IP phone with private IP address need communicate with other IP phones in a different LAN or on Internet, please select an item from dropdown list to set the proxy used by the phone.
    - **disable:** Select this item when the log in server and IP phone in the same LAN, or the log in system supports the IP phone working behind the LAN.
    - **enable:** When the system does not support IP phone working behind the LAN, please select this item to search public IP address of the NAT device. With this item selected, “nat addr” field will be activated. Besides, port mapping (port forwarding) needs to be properly set up on NAT device.
    - **stun:** Select this item with SIP protocol used according to requirement of system. With this item selected, nat addr field is activated.

- **NAT:** When “nat traversal” is set to “enable”, please put the domain name of the servers (These web server helps to find out the public IP of the IP phone) into “nat addr”, such as www.whatismyip.com. When “nat traversal” is set to “stun”, please put the URI of the stun server into “nat addr”, in the format as “domain name/IP address : service port”. The default service port for stun is 3478.

- **nat ttl:** When IP phone sit behind a NAT device, it will send packets to server every “**nat ttl**” seconds to keep the port mapping on the NAT device alive. “**nat ttl**” is an integer between 10 and 60, default value is 20.

- **phone number:** The local phone number or username of this phone, usually is allocated by system.

- **account**: With H323 protocol used, while calling card is set, please type the account of chosen card into this field; while md5 account item selected in local type dropdown list, enter ID here; while account is selected in local type dropdown list, enter H323 ID here. While prefix item selected in use dialplan dropdown list, enter language indicating number, card number and # here, such as 14589653185”. With SIP system which requires authentication, please put the username/account into this field. With MGCP protocol used, please enter local endpoint id (eg., aaln/0) here. With Net2phone system used, enter account of Net2phone card here.

- **pin:** With H323 protocol used, while calling card is set, please type the password of chosen card into this field; while md5 account item selected in local type dropdown list, enter password here. While prefix item selected in use dialplan dropdown list, enter password and # here, such as 3185”. With SIP system which requires authentication, please put the password into this field. With MGCP protocol used, please enter domain name here. With Net2phone system used, enter password of Net2phone card here.

Note:
When MGCP protocol is used, some system requires adding “[]” outside the domain name. So please fill the domain name with “[]” into pin fields, such as "voiptest.com".

- **rtp tos**: Fill TOS segment of IP head package in RTP digital follow here.

- **register port:** The local UDP port registered with server to accept incoming handshaking messages. The default port number for MGCP protocol is 2427. The default port number for SIP protocol is 5060. For H.323 or Net2phone, any number between 1024 and 65535 is acceptable.

- **signal port:** With H323 protocol used, signal port is Q.931 port using TCP protocol, can be any number between 1024 and 65535.

- **control port:** With H323 protocol used, this port is H.245 port using TCP protocol, can be any number between 1024 and 65535.

- **register ttl :** With H323 or SIP protocol, IP phone will send a keep-alive registration message to H323 gatekeeper or SIP proxy server every "register ttl" seconds. The minimum value is 10, maximum value is 255. Default is 60.

- **Rtp tos:** Set the TOS field of the IP header of the RTP packets. The bigger this value is, the higher priority the packet is.

- **rtp port:** RTP port is the port transferring and receiving voice packets using UDP protocol. This is an even number between 1024 and 65535, can't be the same as "register port".

- **local type:** With H323 protocol used, this parameter refers to how IP phone authenticate itself to the gatekeeper. The meaning of each item is as follow:

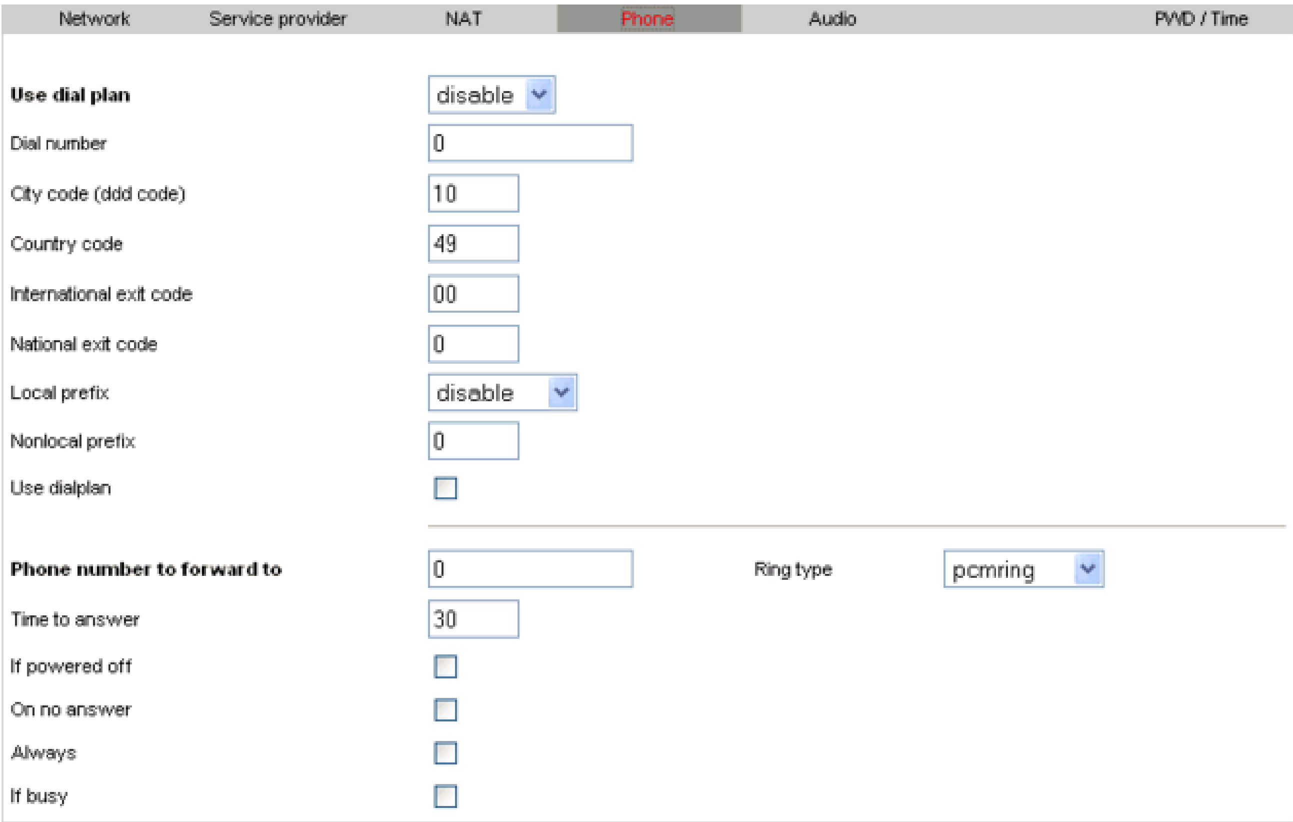
    - **phone number:** Use phone number as E.164 and H323 ID to login the GK.
    - **account:** Use phone number as E.164 and designated H323 ID filled in account field as H323 ID to login GK.
    - **auto:** Use MD5 or CAT encryption by the mode of auto negotation, on the condition of H.235 Encryption portocol.
    - **MD5 account:** Use H235 encrypted username and password to login the gatekeeper.
    - **CAT accout:** Use Cisco access token
    - **sha1:** Use SHA1(Secure Hash Algorithm v1) encryption mode

- **Call type**: Set call type by selecting the items in drop down list.

    - **normal:** Call out in normal way by selecting this item.
    - **faststart:** Call out in faststart way by selecting this item.
    - **advanced:** Call out in faststart and tunneling way by selecting this item. It is a recommended way with H323 protocol used.

- **dtmf:** Set DTMF signal sending way by selecting control string, inband audio, signal keypad or rfc 2833 from dropdown list.

Phone settings



3) Phone settings

- **fwd number:** Enter receiving forwarded calls phone number into this field; If the IP phone used with modem, with modem item selected in iptype list box, and then fill ISP number into this field.

- **fwd poweroff:** Forward calls if power off by checking this box. Please enter receiving forwarded calls phone number into fwd number field.

- **fwd always:** Forward all calls by checking this box. Please enter receiving forwarded calls phone number into fwd number field.

- **fwd busy:** Forward calls if busy by checking this box. Please enter receiving forwarded calls phone number into fwd number field.

- **fwd noanswer:** Forward calls without replying by checking this box. Please enter receiving forwarded calls phone number into fwd number field.

- **use dialplan:** Set whether use dial plan or use dial number by selecting the corresponding item in drop down list.

    - **disable:** Do not use dial plan or dial number by selecting this item.
    - **enable:** Use dial plan by selecting this item.
    - **dialnum:** Use dial number by selecting this item. With this item selected, please enter the dial prefix into dial number field.
    - **prefix:** Use 179XX service by selecting this item.
    - **Hotline:** Use Hotline function by selecting this item. With this item selected, please enter the hotline number into dial number field.

Note:

With 179xx service used, please set as follows: fill call prefix into dial number field, such as 17930; type ;language indicating number, card number and # into account field; fill password and # into pin field.

- **dial number:** With dialnum selected in use dialplan drop down list, please enter the dial prefix into this field according to requirement of log in server. For example, with eTalk card used, enter 00 here.

- **ddd code:** With enable or dialnum selected in use dialplan drop down list, set area code according to E.164 dial rule. For example, Beiing 10; Shanghai 21.

- **idd code:** With enable or dialnum selected in use dialplan drop down list, set country code according to E.164 dial rule. For example, China 86; U.S.A .1.

- **idd prefix:** With enable or dialnum selected in use dialplan drop down list, set international call prefix according to E.164 dial rule, such as 00.

- **ddd prefix:** With enable or dialnum selected in use dialplan drop down list, set long distance call prefix according to E.164 dial rule, such as 0.

Note With dialnum seletcted in use dialplan drop down list, you can also set dddcode, iddcode, iddprefix and dddprefix according to requirement of system.

- **innerline:** Enable/disable multi-settings by selecting corresponding items from dropdown list. The IP phone allows saving 5 settings totally.

    - **disable:** Disable multi-settings by selecting this item, then the phone will call out using current setting.
    - **enable:** Use designated system to place calls by selecting this item.
    - **switch:** Enable multi-settings by selecting this item. Then please fill the prefix switching to backup setting 1 and backup setting2 into local prefix and nonlocal prefix fields.

Note To modify the parameters of backup settings, please use Telnet commands.

- **local prefix:** With **enable** or **switch** selected in **innerline** dropdown list, please fill the number switching to backup setting 1 here, such as 56.

- **nonlocal prefix:** With **enable** or **switch** selected in **innerline** dropdown list, please fill the number switching to backup setting 2 here, such as 57.

- **use digitmap:** Enable/disable digitmap by checking/unchecking the box.

- **ring type:** Set ring type by selecting corresponding item from drop down list.

    - **dtmf 0-9:** Set ring as ordinary rings in different frequency
    - **not disturb:** Set the phone do not ring by selecting this item.
    - **pcmring**: Set ring as music shipped from factory by selecting this item.
    - **user define:** ring as music saved by user by selecting this item.

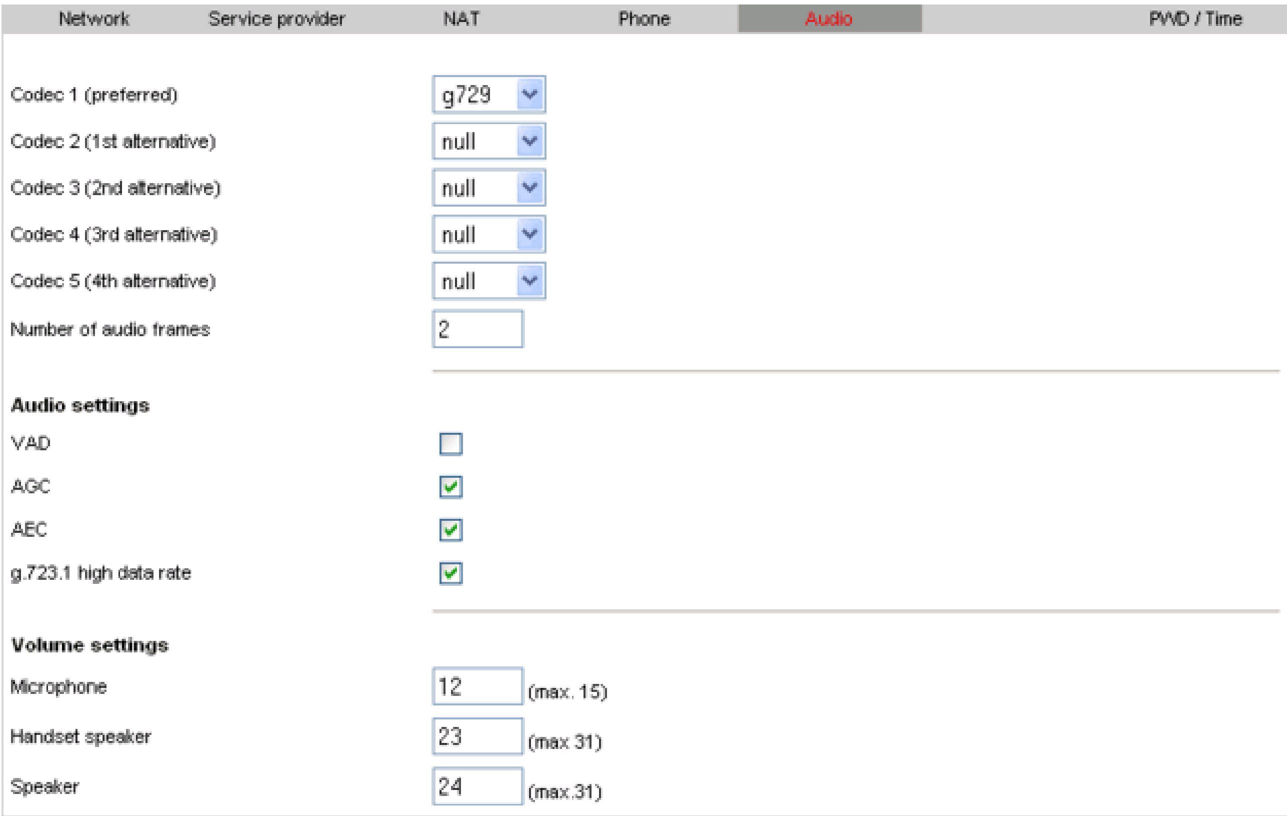
- **answer:** Enter a number from 0 through 60 to set the entries of the seconds before the phone answer the call auto or forward the calls. To disable auto answer function, please set this parameter as 0.

- **predial time:** Set time limit from picking up the speaker to dialing the first the number.

- **interdial time:** Set time limit between dialing two numbers.

- **postdial time:** Set time limit from dialing the last number to placing a call. If the next number is not dialed within the post dial time limit, then the phone will call the dialed number auto.

Audio settings

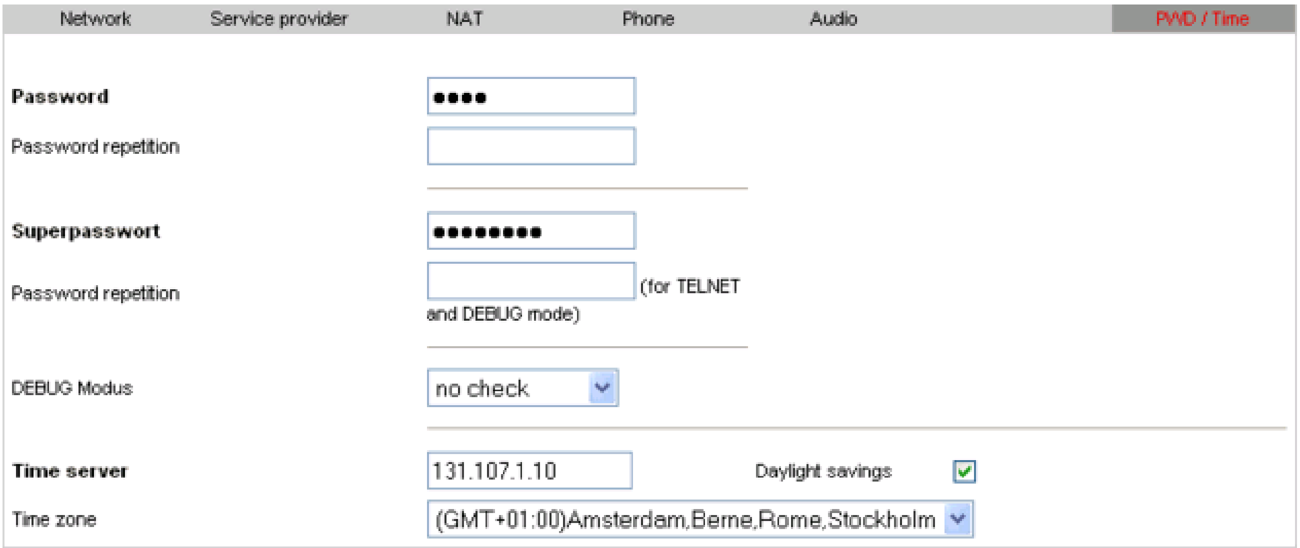


4) Audio settings

- **audio type:** Set audio type of the phone by selecting item from drop down list. The options are g729 • Cg7231• Cg711u• Cg711a and auto. “auto” is suggested, since when “auto” is selected, IP phone will negotiate with system about which algorithm to use and be able to communicate with more terminals and systems.
- **audio frame:** Set audio frames in RTP package. With G723 audio codec used, set it as 1; with G729 audio codes used, set it as 2. Minimum is 1 and maximum is 8.
- **g.723.1 high rate:** With g.723.1 selected in audio type dropdown list, enable/disable g.723.1 high rate by checking/ unchecking this option.
- **vad:** Enable/disable VAD (voice activity detection) by checking/ unchecking this box.
- **agc:** Enable/disable AGC by checking/unchecking this box.
- **aec:** Enable/disable VEC by checking/unchecking this box.
- **handset in:** Drag the slider to adjust the volume of handset input. Drag it to the left to reduce the volume; while drag it to the right to increase the volume.
- **handset out:** Drag the slider to adjust the volume of handset output. Drag it to the left to reduce the volume; while drag it to the right to increase the volume.
- **speaker out:** Drag the slider to adjust the volume of handfree output. Drag it to the left to reduce the

volume; while drag it to the right to increase the volume.

Other settings

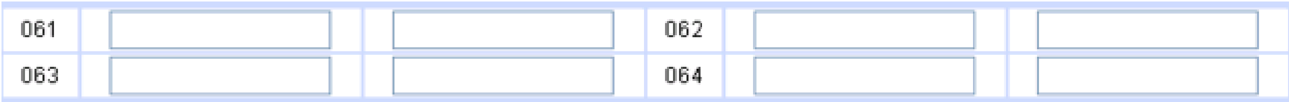


5) Other settings

- **password:** Set the password of the phone. (Default password is 1234).
- **super password:** Set the super password of the phone
- **debug:** Set the debug level of the phone.
    - **disable**: Disable output the bug message by selecting this item.
    - **output**: Output the operation information to the window, such as register, input by selecting this item.
    - **output all**: Output all bug information and data in test window by selecting this item.
    - **remote debug:** Save the bug information in SDRAM of IP phone by selecting this item.
    - **no check**: Disable checks the mark by selecting this item
- **nts ip:** Fill IP address of time server here.
- **use daylight:** Enable/disable daylight by checking/unchecking this box.
- **upgrade addr:** Enter IP address or domain name obtained by ISP of FTP server supplying updated program here.
- **timezone:** Select correct time zone in dropdown list.

Phone book

Click at the link "Phone book" at the right side of WEB interface if you want to change to phone book. There you can save numbers and names. After having clicked "Save" the entries are stored to phone and also available by speed dial.



Note:

If you want to place direct IP-to-IP calls, you can save IPs to phone book, too. Please use “*” instead of “.”. For example, save IP address 192.168.1.221, as 192*168*1*221.

Update firmware

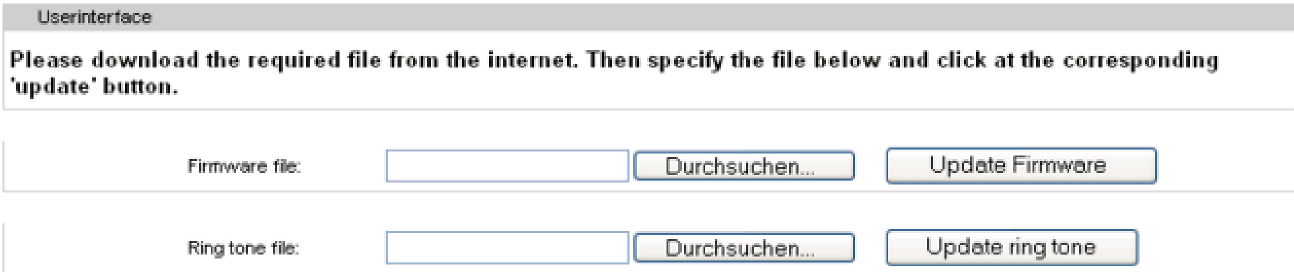
Download newest firmware

You can find newest firmware version for your phone on our webpage (please refer to Important Information).
Download it to your Computer or Notebook.

Install the firmware

Open the web configuration and follow the link "Update Firmware" at the right side below the logo.

Following page appears:



Click "Search" at line "Firmware" and select saved firmware from your system. Accept file and click „Update Firmware“.

Note:
Never change file name or its extension and never interrupt update process. Otherwise device could be damaged or become inoperable.

Looking at the call historie

Looking at the call history

Missed calls

Press "Missed" if you want to see missed calls. You can move through the listing with "Vol/+" and "Vol/-". If there are no numbers you have no missed call.

Answered calls
Press "Answered" if you want to see answered calls. You can move through the listing with "Vol/+" and "Vol/-". If there are no numbers you have no answered call.

Dialed numbers
Press "Dialed" if you want to see dialed numbers. You can move through the listing with "Vol/+" and "Vol/-". If there are no numbers you have not dialed a number.

You can directly place a call to the visible number by pressing "#".

Note:
The IP phone saves up to 127 entries. If there is no free storage oldest entry will be overwritten with by new entry. All data will be deleted if you do restart, switch off or reset phone.

Receiving calls

The phone can receive incoming calls from other devices that support the SIP or H.323* protocol. It works just like an ordinary phone for incoming calls. When it rings, you can receive the call by following methods:

1) Use handset

Lift the handset and begin speaking. When the call is over, put the handset back.

2) Use Hand free

Press "Hand free" to speak to the other party. When the call is over, press "Hand free" again.

Note:
When you communicate with the other party without lifting the handset,
please do not exceed 40 CM from speaker.

* Special Firmware required

Place a call

(1) Call another IP phone under the same Gatekeeper

Handset: Pick up the handset and listen for the Internet dial tone. Then dial the phone number you wish to call and press "#" or "Call" to end the dialing. Once the call connection has been established and the ring tone has sounded, wait for the other party to answer. When the other party answers, you can begin speaking. When the call is over, put back the handset. The dialed number has been saved into the buffer.

Hand free: Press "Hand free" and listen for the Internet dial tone. Then input the phone number you wish to call and press "#" or "Call" to end the dialing. Once the call connection has been established and the ring tone has sounded, wait for the other party to answer. When the other party answers, you can begin speaking. When the call is over, Press "Hand free" again. The dialed number has been saved into the buffer.

Blind dialing: Use the keypad to enter the phone number you wish to call and then press "#" or "Call" to make the call. Once the call connection has been established and the ring tone has sounded, wait for the other party to answer. When the other party
answers, you can begin speaking. When the call is over, Press "Hand free" again. The dialed number has been saved into the buffer.

(2) Place a call without login the Gatekeeper
If the IP phone does not login the Gatekeeper, you can place a call by lifting the handset or pressing "Hand free" and then inputting the IP address of the other party, and then pressing "#" or "Call".

Note
When you place a call without Gatekeeper or with Gateway, please log off Gatekeeper. To get the detailed operation please refer to Configuration chapter.

Rechtlicher Hinweis

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