G300

Voice over IP Telefone

www.globaliptel.com

Inhaltsverzeichnis

Preface

Thanks for buying G300 from Global IP Telecommunications.
We would like to give you a quick start guide with the following pages.

Unpacking and installation

Remove all packing material and ensure that box contains:

1 x G300 VoIP Phone
1 x Power adapter with power cable
1 x Network cable

Place the phone close to power socket and network port on a stable, flat surface that is clean and free of dust.

Connect handset with phone, power adapter with power socket and power cable with phone.

Link phone to network. Phone can be used with DSL-Modem by using a Cross-Over-Wire. By using a patch cable you can connect the phone to a Hub or a Switch.

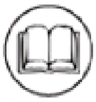
Now please switch on the phone by sliding the on/off switch at the back of phone towards the middle of backside.

The phone and its functions

Keys and functions

P1: Programmable key (local IP as default)

P2: Programmable key (enter microphone volume setting mode while talking to someone)

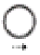
 **:** Access phone book

+: Increase volume (navigate the menu)

-: Decrease volume

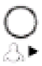
 : Cancel / Backspace

 **:** Redial

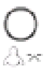
 **:** Flash

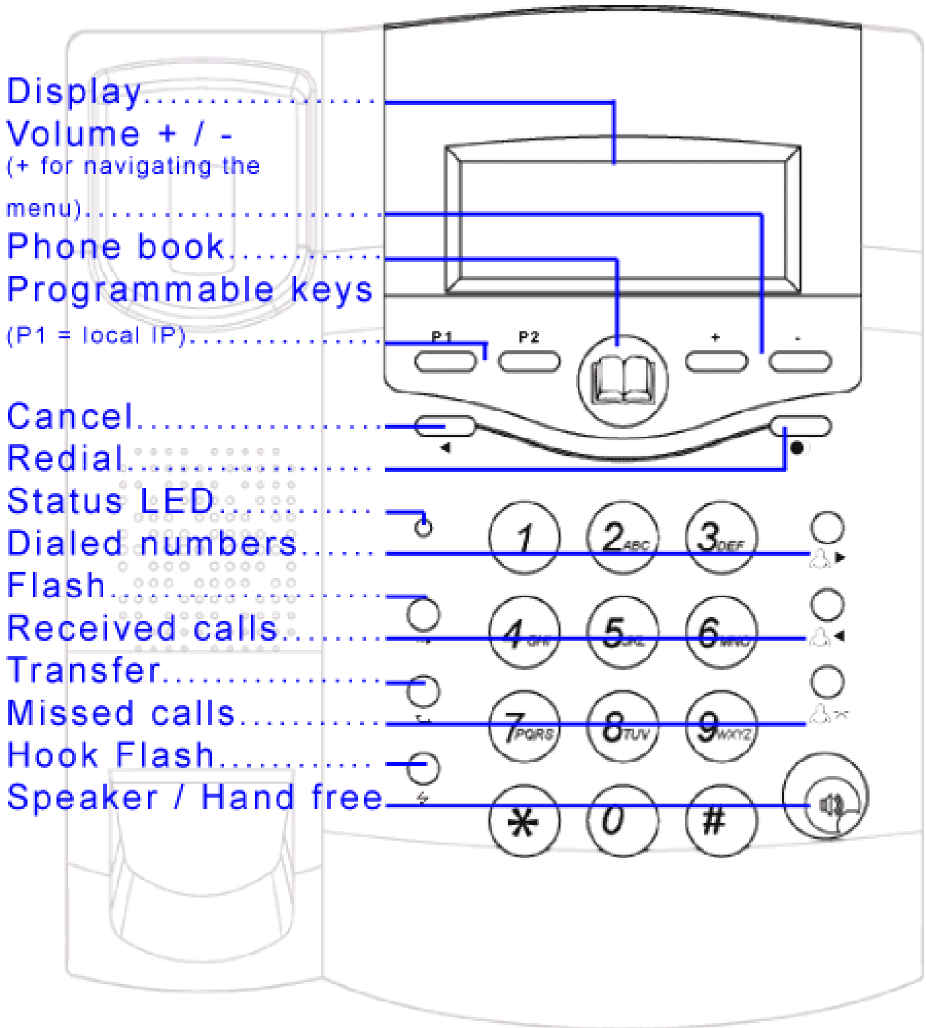
 **:** Transfer

 **:** Hook Flash

 **:** Access dialed numbers list

 **:** Access received numbers list

 **:** Access missed calls list



1. On/Off switch
2. Power
3. RJ-45 port for connecting phoen with network
4. Additional RJ-45 port for connecting computer to phone and thus to network

Konfiguration of phone by phone menu or WEB

interface

You can configure phone by integrated menu or by using a browser.
It is recommended to use Microsoft Internet Explorer if you want to configure phone by WEB interface.
MAC users can download Microsoft Internet Explorer from http://www.microsoft.com/mac/products/ie/.

Setup with phone menu

Phones' factory default password is 1234. Default super password is 19750407. Please ask your dealer if he has changed the password while entering account data for you.
Enter password - or super password if debug mode is enabled - and press "#" until "password:" is displayed. Enter password again and confirm with "Speaker/Hand free". You can go through the menu with the "+" key.

Keyfunctions during menu enabled

Speaker/Hand free:
Select sub menu and confirm changes

+:
Go through the menu.

Numerical keys:
Enter new values.

-:
Backspace.

Redial:
Go back without saving.

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
        - jittersize
```



```
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]
        - call waiting
        - fwdpoweroff
        - dualmode

Audio settings:
        - codec 1 ---------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]
        - upgradetype        |1[output]
        - upgradeaddr        |2[output all]
        - sntpip             |3[remote debug]
        - timezone           |4[no check]
        - daylight
Exit:

Safe:

Auto test settings:

Load settings:

Store settings:

Default settings:

Speeddial settings:
         - enter numers and names
```


Setup by WEB interface

You can use Microsoft Internet Explorer for setting up the phone if a computer is part of same network segment.

Open a browser window and enter phone's IP address (e.g. 192.168.0.99) at the address line. IP address will be displayed if you press "P1 (Local IP)" with handset down. A login dialog appears. Please enter either **password (1234)** or **super password (19750407)** and click "Login".

Network

DHCP/IP/Modem/DSL:	Please choose the way you want to feed your phone with network parameter.
Static IP:	Enter IP address, subnet maske and router IP manually.
DHCP:	Source network parameter by DHCP.
PPPoE:	For direct use with ADSL and Cable Modem. Source network parameter direct from Internet Service Provider (ISP).
Modem:	For sourcing the data from ISP. You will have to enter your ID and PIN at ppp id and ppp pin.
PPPoE ID:	Enter ISP Username if PPPoE or Modem selected.
PPPoE PIN:	Enter ISP Password if PPPoE or Modem selected.
Local IP:	Please enter phone's IP if Static IP selected.
Subnet Mask:	Please enter subnet mask if Static IP selected.
Router ip:	Please enter router IP if Static IP selected.
DNS:	Please enter DNS Server IP if Static IP selected.
DNS 2:	Please enter alternative DNS Server IP.
MAC:	Phones MAC is printed on a lable at the underside.

Service provider

Service type:	Leave as standard if you are not using one of the specified services.
User name / account:	Enter user name / account.
Phone number:	Enter phone number. Mostly it is user name again.
Type of account:	Leave as "auto" if you can not select the type used by your VoIP provider.
Call type:	Choose "advanced" or leave as is.
DTMF signal trans.:	Choose special method if required or leave as is.
DTMF payload:	Choose special value if required or leave as is.
Domain name / Realm:	Enter service address (e.g. subdomain.yourprovider.com)
Server to log in:	Enter registration server's address (e.g. subdomain.yourprovider.com)
Register port:	With SIP it is 5600
Signal port:	With SIP you can set to 0.
Controlport:	With SIP you can set to 0.
RTP port (voice):	With SIP it is 8000.
RTP type of service:	With SIP you can set to 0.
Register ttl:	With SIP you can set to 60.
g.729 jitter buffer size:	With SIP you can set to 0.

NAT

NAT:	Select the way you want to handle NAT. Mostly "auto" or "STUN" is used.
NAT address:	Enter STUN-Server address and port if "STUN" selected.
NAT ttl:	Select interval for sending keep alive packet to NAT device.

Phone

Use dial plan:	Enable "dial plan" if you want to use it or select "dialnum".
Dial number:	Please enter prefix for using "dial number".
City code (ddd code)	Enter city code. E.g. 69 for Frankfurt/Germany or a three digits city code like 212 for calling New York/USA.
Country code:	Enter country code. E.g. 49 for Germany or 1 for USA.
International exit code:	Enter the exit code for international calling. E.g. 00 for Germany.
National exit code:	Enter national exit code. E.g. 0 in Germany.
Local prefix:	Enable, disable or omit local prefix
Non local prefix:	Enter your non local prefix.
Use dial plan:	Check "use dial plan" if you want to enable the service.
Phone number to forward to:	Enter the phone number you want incomming calls to be diverted to.
Ring type:	Select ring tone or mute speaker for incomming call indication.
Time to answer:	Enter a number from 0 to 60 to set the entries of seconds before phone answers or forwards the call. If you want to disable auto answer or forwarding, please set this parameter to 0.
If power off:	Divert calls if phone is switched off.
On no answer:	Divert calls if no answer.
Always:	Divert directly.
If bussy:	Divert calls if bussy.

Audio

Codec 1...5:	Select codecs and set priority by sorting from preferred to 4th alternative..
Number of audio frames:	With G.723 selected, please enter 2. If using G.729, please enter 1.
VAD	Enable/disable voice activity detection by checking/unchecking this option.
AGC	Enable/disable automatic gain control by checking/unchecking this option.
AEC	Enable/disable acoustic echo cancellation
G.723.1 high data rate	Enable if G.723.1 is enbaled.
Microphone	Set microphone volume. (max. 15)
Handset speaker:	Set handset speaker volume. (max. 31)
Speaker:	Set speaker volume. (max. 31)

PWD/Time

Password:	AGC ein- und ausschalten.
Password repetition:	VEC ein- und ausschalten.
Supper password:	Mirkrofonlautstärke am Hörer einrichten.
Supper password repetition:	Lautstärke am Hörer einrichten.
DEBUG mode:	Einrichten der Lautstärke für die Freisprechfunktion.

Time server:	Enter time server's IP.
Time zone:	Select your time zone.
Daylight saving:	By checking "daylight saving" local time becommes lowered by one hour.

Phone book

Name:	Enter contact's name.
Number:	Enter contact's number.

Update Firmware

Firmware file:	Search for Firmware file you have saved to your computer and update phone by clicking "Update Firmware".
Ring tone update:	Search for Ring tone you have saved to your computer and update ring tone by clicking "Update ring tone".

Place and receive calls and look at call lists

Placing and receiving a call is as easy as working with a ordinary phone.

Receiving calls:
Phone can receive calls from other SIP Phones and any other phone which is calling by using a gateway.

All you will have to do if phone rings is to lift up the handset or to press "Speaker/Hands free".

For ending the call simply ring off or press "Speaker/Hands free" again.

Please note that space to handsfree's microphone should not be more than 40 cm if you are using handsfree mode.

Placing a call:
Lift up the handset and dial the phone number.

Please note that you will have to dial the phone number with country and city code with some providers. Please ask your VoIP provider for his format.

If handset is down, you also can enable speaker mode first. Then dial the number and press "#".

Missed calls:
Please press "Missed" if you want to list missed calls numbers. Click "+" to turn the number orderly or "-" to sort reservedly.

Answered calls:
Please press "Answered" if you want to list answered calls numbers. Here you can sort the numbers with "+" and "-", too.

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

You can call a listed number by pressing "#" if VoIP provider supports calling with SIP-format.

Main technical data

Technical data	
Chip:	50Mhz
Data storage:	2MB SDRAM
Program memory:	1MB Flash Memory
Network:	10BaseT/100BaseTX. 2 x RJ 45, one for LAN/WAN and one for connecting a computer with the phone, which will act as Hub.
Echo cancellation:	G165 16ms
Power loss:	2,7W (max)
Adapter:	Eingang AC 110-230V 1A 400MA, Ausgang DC 9V 500mA
Overall size:	215×190×70mm(L×W×H)
Voltage:	9V DC
Power:	0.5W (max.)
Certified:	BACL CE (RSZ04052413-1&2).
	EMC EN55022: 1998+A2:2003 und EN55024: 1998+A1:2001
Operating requirements	
Operating temperature:	0° - 40° C (32° - 104° F)
Storage temperature:	-30° - 65° C (-22° to 149° F)
Relative humidity:	10 - 90% (no dew!)
Atmosphere pressure:	86 - 106Kpa

Note

Further information are available at www.globaliptel.com.

Please contact your VoIP provider if you do have questions according to how to configure your account.

We can not support you with generell network problems.

Preface

Thanks for buying G300 from Global IP Telecommunications.
We would like to give you a user manual with the following pages.

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 the broad band IP Network environment which accord with TCP/IP protocol, such as in the LAN or WAN of Enterprises and Institutions, Telecom IP phone services provider's network and broad band Internet user, who log on internet through LAN, Cable Modem or XDSL and so on.
The mostly significant features of IP phone is transporting voice message over data communication network at an extremely low price with excellent sound quality. Using IP phone, you can save dramatically on international calls and long distance calls.

G300 IP Phone has 2 x 16 characters LCD with backlit and supports being set by either keypad or Internet Explorer.

Suitable use of G300 VoIP 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.

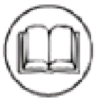
Important note: DO NOT INSTALL A FIRMWARE WHICH IS NOT DESIGNATED FOR THE DEVICE. WITH A WRONG FIRMWARE YOU CAN MAKE THE PHONE UNUSABLE OR YOU CAN DESTROY IT. WE ARE EXEMPT FROM LIABILITY FOR FIRMWARE UPDATES WHO ARE MADE BY CUSTOMERS. FIRMWARE UPDATES WILL BE DONE BY YOUR OWN RISK.

The phone and its functions

Keys and functions

P1: Programmable key (local IP as default)

P2: Programmable key (enter microphone volume setting mode while talking to someone)

 **:** Access phone book

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 : Cancel / Backspace

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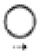
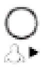
 **:** Flash

 **:** Transfer

 **:** Hook Flash

 **:** Access dialed numbers list

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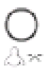
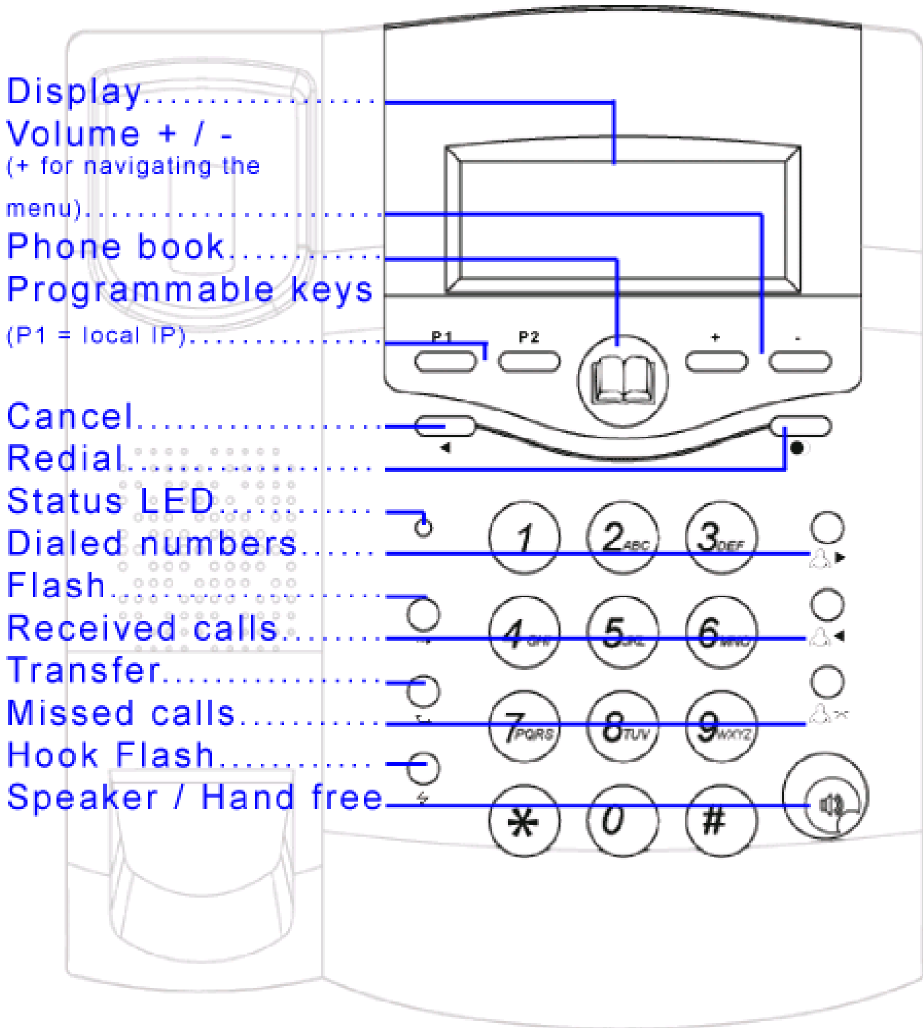
 **:** Access missed calls list



1. On/Off switch
2. Power
3. RJ-45 port for connecting phoen with network
4. Additional RJ-45 port for connecting computer to phone and thus to network

Performance and features

Supports SIP and H.323 with special firmware installed

Supports H.323 v4, compatible with most H.323 v1-v4 systems and devices

Supports MGCP RFC2705 and SIP RFC3261

Supports 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

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 Gatekeeper;

Automatically gatekeeper discovery in LAN

Call PSTN by ITSP’s gateway

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

Firmware update/upgrade

Support G.723.1 5.3k/6.3k, G.729, G.711 A-Law, µ-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

40 entries each for missed call, answered call and dialed call

112 entries for speed calling

Flüssigkristalldisplay mit 2×16 Zeichen

LCD display for showing dial data, caller name, caller number and so on

Working status indicating Lamp and LCD jacklight

Volume of handset's mikrophone and speaker and hand free speaker adjustable

14 function keys for operating and setting phone besides standard keys 0-9, # and *.

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:
40 records

Power loss:
2.7W (max)

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×W×H).

Standard and protocol

H.323 V4 calling signal protocol if special firmware installed

MGCP RFC2705 and SIP RFC3261

Net2phone 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;

Speed 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 and G729 VAD/CNG economical bandwidth

G.165 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

HTTP: Hyper Text transfer Protocol

Build in H.323 proxy support (with special H.323 firmware only)

Automatische Proxysuche.

Electric requirements

Voltage:
9V DC

Power:
0.5W (max.)

Power adapter:
Input 230V, 500mA

Network interface:
2 RJ-45 Ethernet ports

Operating requirements

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

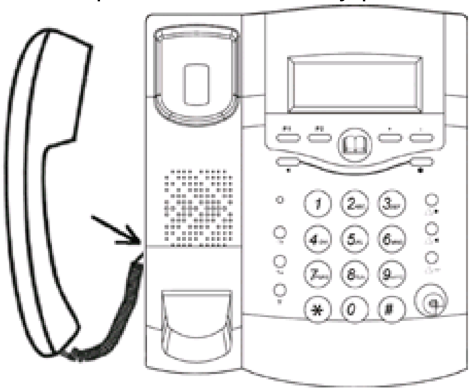
Storage temperature:
-30° to 65° C (-22° to 149° F)

Humidity:
10 to 90% (no dew)

Installation

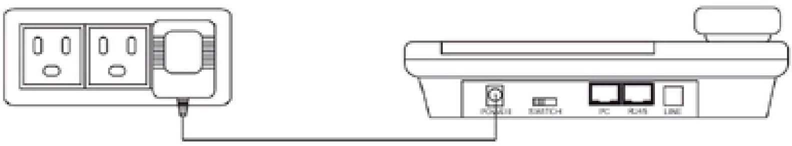
1. Connect phone and handset

Connect phone and handset by provided cord.



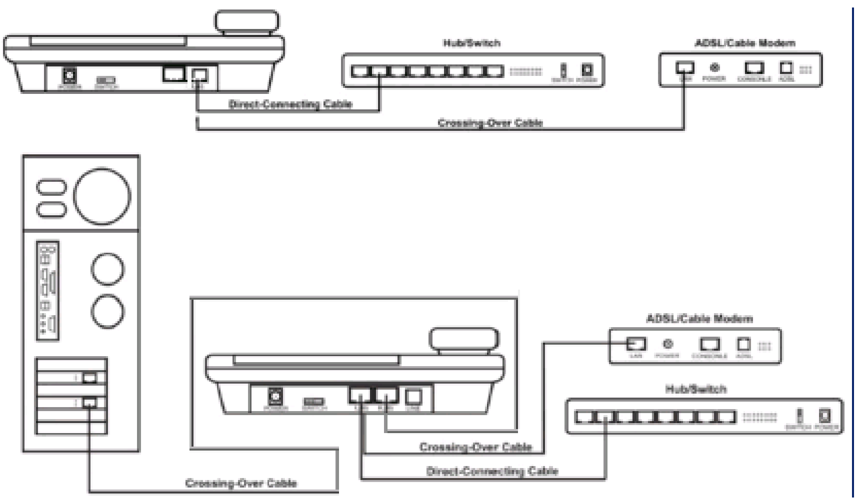
2. Connect to power supply

Place the phone nearby of power 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. Establish connection to the network

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



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

4. Switch on the phone

Switch on the phone by sliding the switch which is located at the back side of the phone to the left. Ensure that LED and display indicate activity.

Setup by phone

Default password is 1234.

1) Entering into setting mode

Note:
if "debug" is not set to 0[disable], please use default password 1234. Please use super password 19750407 if "debug" is set to 0[disabled]. Please note that you can not enter alphanumerical passwords with phone keys. If you want to enter setting with alphanumerical digits, please refer to page 15. You can navigate the menu with "+" and enter setting mode with "missed calls". If you want to confirm changes, pleas press "missed calls" and cancel with "+".

If phone is switched off:
Press "#" and switch on the phone. Enter your password and confirm with "Missed calls".

If phone is allready running:
Enter password and press "#" until „Password:" is displayed. Enter password again and access configuration menu by pressing "Missed calls".

2) Key functions in the keypad setting mode

Key	**Function**
+	**Navigate menu // cancel changes**
Missed calls	**Enter menu // confirm changes**
Speaker	**Backspace while entering new values**
Numerical pad	**Enter values and changes**

3) 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
 - jittersize
```



```
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]
        - call waiting
        - fwdpoweroff
        - dualmode

Audio settings:
        - codec 1 ---------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]
        - upgradetype        |1[output]
        - upgradeaddr        |2[output all]
        - sntpip             |3[remote debug]
        - timezone           |4[no check]
        - daylight
Exit:

Safe:

Auto test settings:

Load settings:

Store settings:

Default settings:

Speeddial settings:
         - enter numers and names
```

4) Appendix

Here you can see the characters you will receive by pressing a key for one to five times.

Press key	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	:/@	;/-	'/&
*					
#	Switch between uppercase and lowercase				

We recommend to restart system after having changed settings.

Configuration

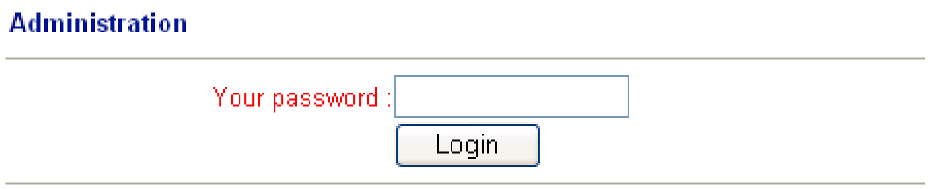
There are three ways to configure this IP Phone:

Phone keypad set, web browser and Telnet commands.

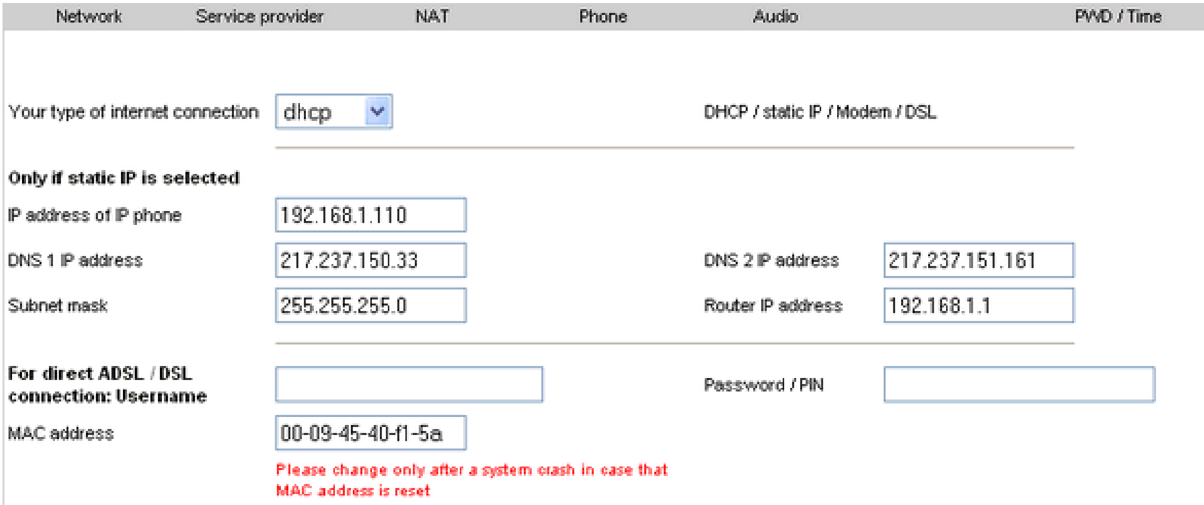
Configuration with WEB browser

If phone and computer are connected to the same network, you can configure the phone by WEB browser. It is recommended to use Microsoft Internet Explorer 6.0 or higher. Open new browser window, enter phones IP (e.g. http://192.168.0.110) at address line and press "Enter" at your keyboard.

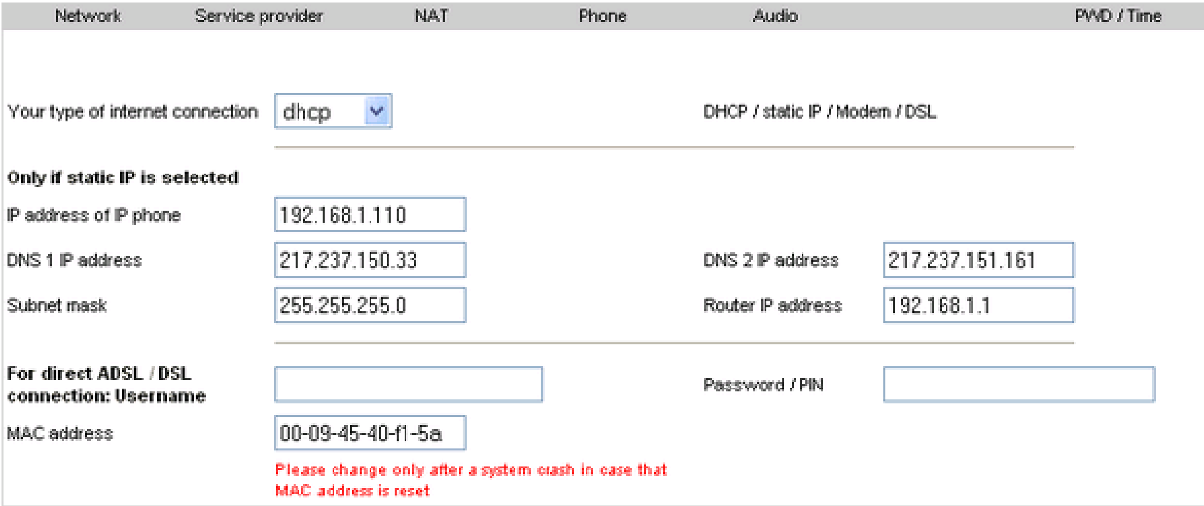
Phones IP will be displayed by pressing "P1". Please note that IP can not be displayed if phone is running DHCP-mode and DHCP-Server is unavailable.
You can either use default password 1234 or super password 19750407 for login. Super password is required if debug mode is set to 0[disable].
Please note, that you will have to enter your individual password if you have updated the password with you individual one.



The following window will appear if you have entered a valid password and clicked "Login":



Network settings



How are you connected with the Internet:
Choose a setting for either sourcing network parameter or entering manually.

Static IP: Choose Static IP if you want to enter IP-Addresse Subnet Mask and Router IP-address manually.
DHCP: Choose DHCP for sourcing necessary network parameter from DHCP Server or Router, if connected with same network. Your IP Network Phone comes with DHCP as default. By this your phone tries to source network parameter when connecting phone to LAN with power on.
PPPoE: If you want to use your device directly at XDSL-modem, please select PPPoE. Your ISP transfers required network parameters automatically if phone is connected to modem and switched on.
Modem: If you want to use phone with modem, please select modem from drop down menu and enter ID/Username and PIN/Password.

Username / ID:
Please enter user name if PPPoE or Modem selected.

Password / PIN:
Please enter password for Internet connection if PPPoE or Modem selected.

IP address of IP phone:
Please enter an IP if you want to use device with static IP.

Subnet mask:
Please enter a Subnet mask if you want to use device with static IP.

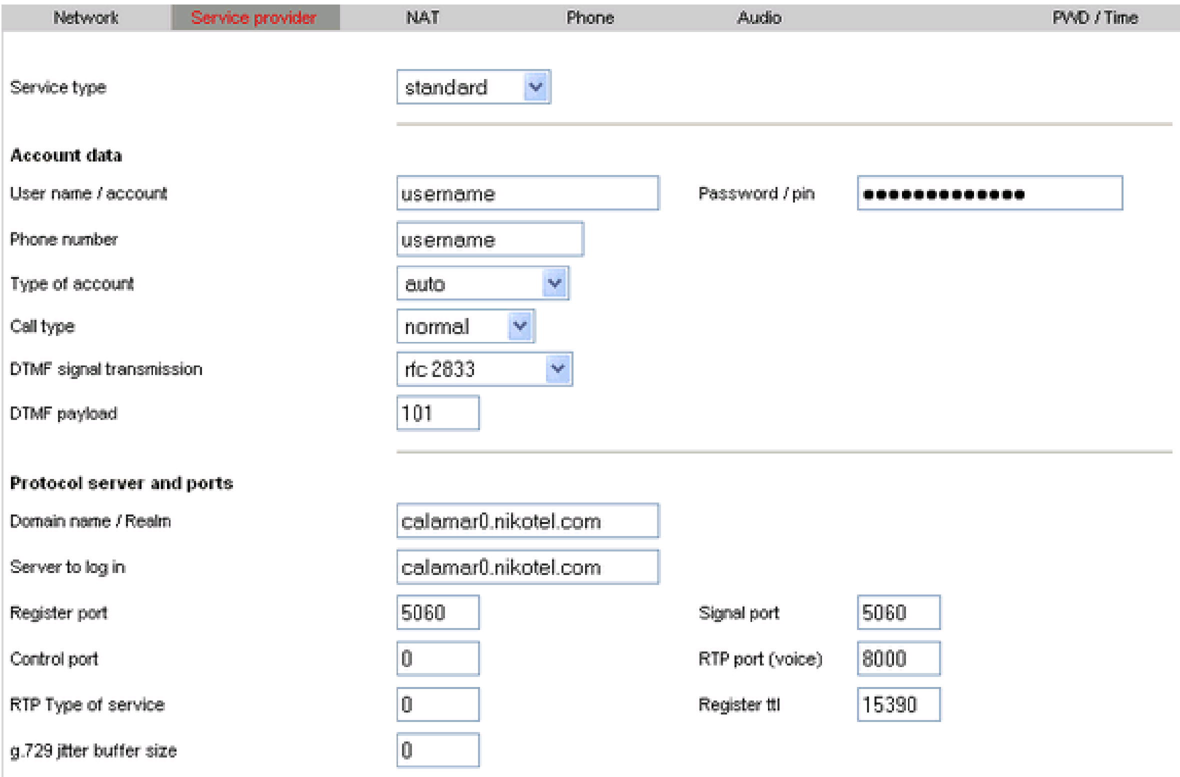
Router IP Address:
Please enter Router's IP if you want to use device with static IP.

DNS 1 IP address:
Please enter an DNS-Server's IP if you want to use device with static IP.

DNS 2 IP address:
Please enter a backup DNS-Server's IP if you want to use device with static IP.

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

Service provider



Service provider

Service type:
Leave as "Standard" or select your service from drop down list if necessary.

You can decide either to use h323 or sip or IAX2 by installing adequate firmware.

h323: H.323.
SIP: SIP Protokoll.
IAX2: Asterisk.

H.323:
Please ask your provider for the settings and enter URI of Gatekeeper at “Domainname/Realm”. Standard port of service is 1719. Does your Gatekeeper have an own ID, so please enter at "Server to log in".

For IP-to-IP-calling please please delete "Server to log in" data. Standard port is 1720 in both cases.

SIP:
Please enter SIP Proxy server's URI at „Server to log in“ ein. Enter Domain name of SIP Proxy server or URI at “Domainname/Realm” or leave empty. Please ask your provider for the settings. Please enter Outbound Proxy's URI if specified by your provider. Standard port is 5060.

IAX2:
Open source linux PBX protocoll.

Phone number or GUI:
Enter your phone number or display name here.

User name / account:
Please enter user name or user ID as requirded by your service provider.

Password / pin:
If your are using H323 with calling card please enter calling card's password. If you are using a md5 acount please enter its password. If dial plan selected, please enter an aditional # (e.g. 6234#). Please enter your password or pin if you are using a SIP account. MGCP-protocol users please enter Domain name. Net2phone needs password of Net2phone card.

Some MGCP system do need additional characters. The best would be if you do enter the domain in brakets (e.g. "voiptest.com").

Register port:
Please enter 2427 if using MGCP or 5060 (additional clients do need the port increased by 1 or 2 - depends on service) if using SIP. If using H.323 or Net2phone you can use any port between 1024 and 65535.

Signal port:
5060 if using SIP or any TCP port between 1024 and 65535 if using H323.

Control port:
With H323 protocol used, this port is H.245 port using TCP protocol. Please enter a number between 1024 and 65535. .

Register ttl:
Timer for sending keep alive signal to either Gatekeeper or SIP Proxy. Default is 60. Value is specified in seconds.

RTP type of service:
Set RTP packet priority. As higher the value as higher the priority.

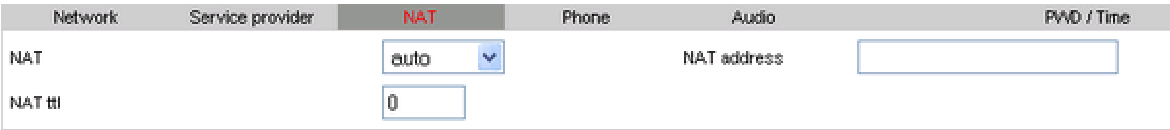
RTP port / Voice:
RTP port is the port transferring and receiving voice flow using UDP protocol. Please fill an even number between 1024 and 65535 into this field. Can not be the same like signal port.

Type of account:
Select an other value then "Auto" if using H323 and as specified by your provider.
SIP users will use "user name".

Call type:
Select either "normal" or "advanced".

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

NAT



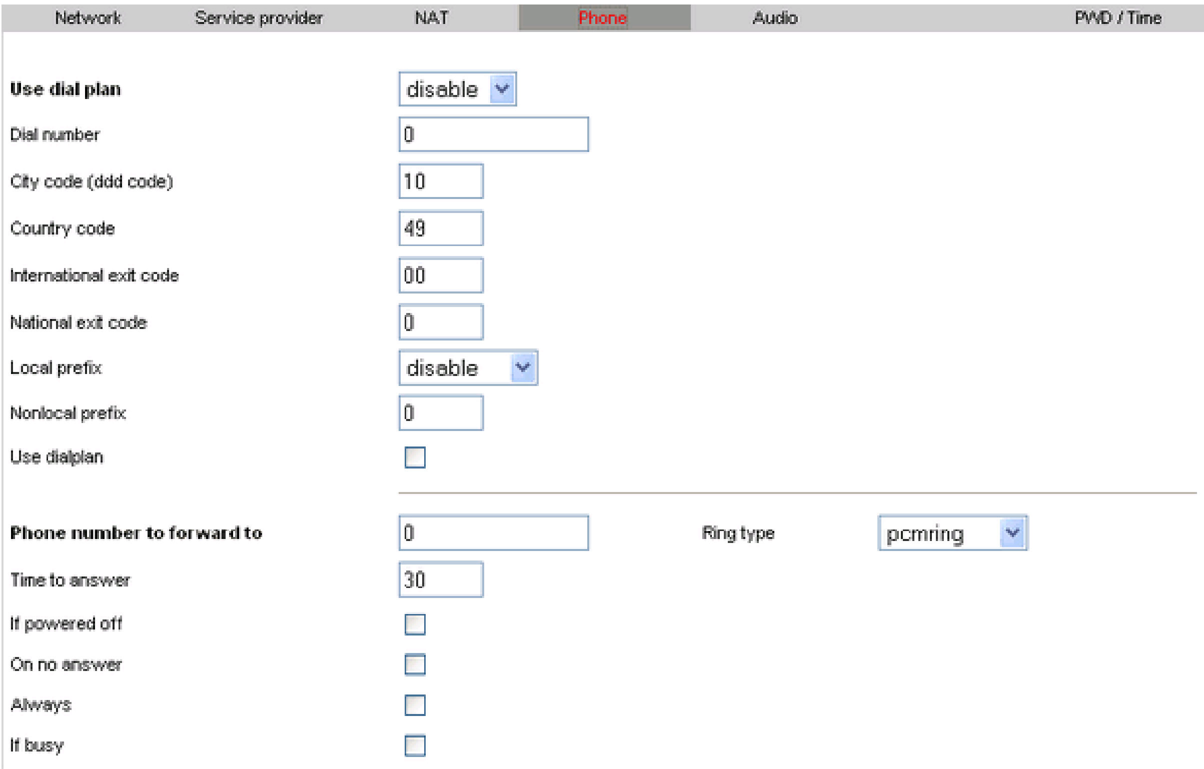
- **NAT:**
  You can choose NAT if necessary for registering and calling.

  Select "auto" or "STUN" or a value as told by service provider or network administrator.

  NAT Address:
  Enter your Router's IP if "enable" selected from NAT drop down menu.
  Enter STUN-Server's URI (e.g. IP:Port or Domain:Port) if "STUN" selected. Default STUN port is 3478.

  NAT ttl:
  Send keep alive for port mapping. You can choose a value between 10 and 60 seconds. Default is 20.

Phone



Use dial plan: 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: Iprefix: Use service by selecting this item.

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 (ddd code): With enable or dialnum selected in use dialplan drop down list, set area code according to E.164 dial rule.

country code: With enable or dialnum selected in use dialplan drop down list, set country code according to E.164 dial rule.

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

national exit code: 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.

Use dial plan: Check this box if you want to use the dial plan.

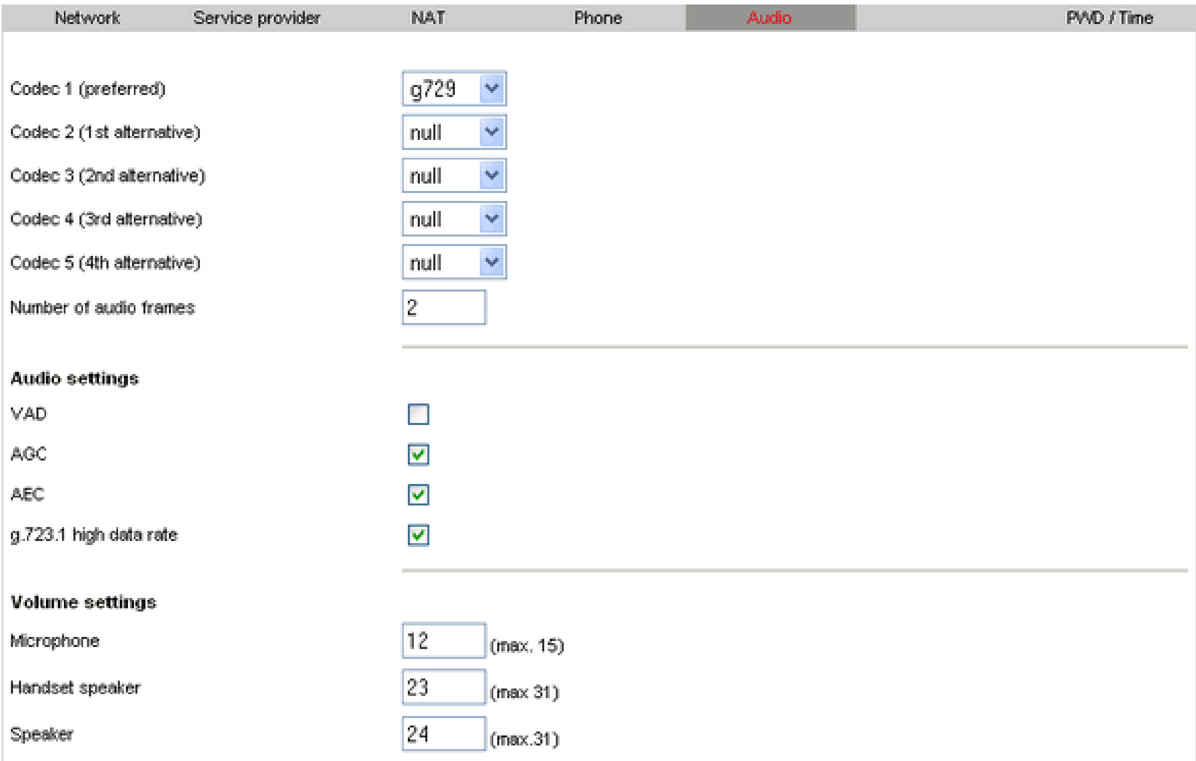
Phone number to forward to: Enter phone number you want to foward to.

- **Ring type:** Select ring tone. Additional ring tones are available at www.globaliptel.com. Please refer to chapter "Update firmware".

- **Time 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.

- Now you can make the phone to divert calls "**if power off**", "**on no answer**", "**allways**" and "**if busy**" by checking the boxes.

Audio



Codec x: Select the codecs you want to use.

Number of audio frames: Set audio frames in RTP package. With G723 audio codec used, set it as 1; with G729 audio codes used, set it as 2.

VAD: Enable voice activity detection by checking this box.

AGC: Enable auto gain control by checking this box.

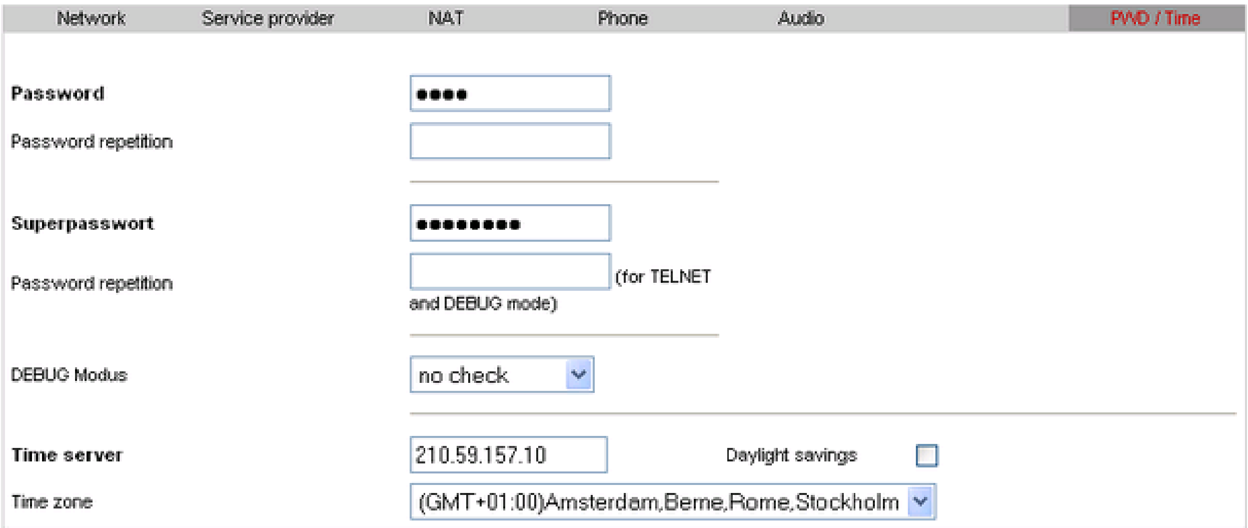
AEC: Enable acustic echo cancelation by checking this box. .

g.723.1 high date rate: With g.723.1 selected in audio type dropdown list, enable/disable g.723.1 high rate by checking / unchecking this option.

Volume settings:

Microphone, Hand set and **Speaker:** Adjust the volume to a comfortable level.

Password and time



Password / Time

Password: Update password for more security. Default password is 1234. Please note that you can enter numerical passwords by phone keypad only if you want to configure phone by phone menu.

Super password: Update super password for more security and do not forget it. Phone leaves factory with super password 19750407. Super password is necessary for accesing phone if debug mode is disabled.

DEBUG Modus: .

disable: Disable output of error messages.
output: Showes status message.
output all: Showes error messages at a special support tool.
remote debug: Saves error to SDRAM.
no check: No output.

Time server: Here you can enter a timeserver's address if you do not want to use default time server.

Daylight saving: You can enable/disable daylight saving by checking the box.

Time zone: Select time zone from drop down menu.

Save/Reboot: Klick it if you want to save settings and restart system. System restart is necessary to apply new settings.

Phone book

No.	Name	Phone Number	No.	Name	Phone Number
001	test1	8704087	002	test2	192*168*1*136
003			004		

You can access the phone book by the hyperlink at the right of the web settings dialog page.
Here you can enter names and number.

If you want to place direct IP-to-IP calls, please enter a * instead of an . while entering an IP.

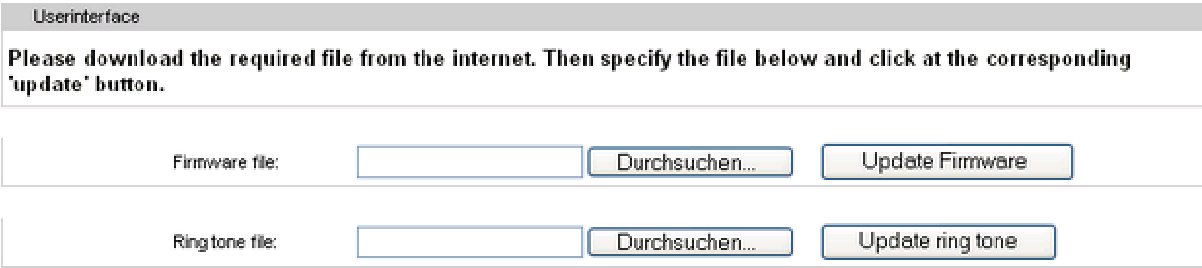
Update firmware and ring tone

Download firmware from www.globaliptel.com

Third party firmware is not supported and can damage your phone. You are only allowed to use Global IP Telecommunications firmware which is approved for your device.

Install new firmware or new ring tone

Open update page and "search" for firmware file or ring tone saved by you. Update firmware or ring tone by clicking the corresponding "update" button.



Do not modify file name and do not use firmware not approved by Global IP Telecommunications.

Call records

Missed calls

Click missed call keypad. Record of missed call will be displayed. Click "+" keypad to turn the numbers orderly.

Answered calls

Click answered call keypad. Record of answered call will be displayed. Click "+" keypad to turn the numbers orderly.

Dialed numbers

Click dialed numbers keypad. Dialed number will be displayed. Click "+" keypad to turn the numbers orderly.

When you see a number you want to dial, please press "#" to place a call directly.

Phone saves up to 120 records. When the entries arrives 120, the latest record will cover the first one. The record will be lost when the phone restarts or turned on.

Receive a call

Your IP Network Phone can receive calls from other IP Network phones, phones which are using same protocoll or which are calling you by using a media gateway (e.g. PSTN calls SIP phone). If there is an incomminc call, IP Network Phone acts as an ordinary PSTN phone. You can answer the call by either

Picking up the hand set

or

activating speaker / hand free mode.

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

Placing a call

(1) Call member of same service or PSTN by service providers gateway (payment required)

Call with hand set:
Pick up the handset and listen for the Internet dial tone. Then dial the phone number you wish to call and press "#". 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.

Call with speaker mode:
Press "Speaker/Hand free" and listen for the Internet dial tone. Then input the phone number you want to call and press "#". 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 start talking. Press "Speaker / Hand free" again when the call is over. The dialed number has been saved into the buffer.

(2) IP-to-IP dialing (with h323 only)

You can either call with hand set lifted or with speaker mode activated as described before. The difference while giving a IP-to-IP call is that you will have to enter the other party's IP if you are part of the same network. Please note that you will have to enter a * instead of a . while entering the IP.

Legal note

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