



G100 (P100) VoIP Phone

User Manual



Index

Introduction	2
Suitable Uses of G100 IP Phone	3
Appearance	4
Performance and Features	5
Main technical index	6
Standard and Protocol	7
Electric requirements	7
Operating requirements	7
Installation	8
Configuration	9
1. Phone keypad set	9
2. Configured by WEB	15
Usage of the Phone	30
Upgrade G100 IP Phone	32
View Records	33

Introduction

IP is the abbreviation of Internet Protocol. An IP phone is a telephone that transports voice data through the internet by using the IP protocol. It can be used for audio communication through LAN, Cable Modem or XDSL and wherever else the internet is available.

IP telephony generally comes with a cost advantage. The G100 IP phone features furthermore excellent sound quality.

Appearance

1. Front Panel

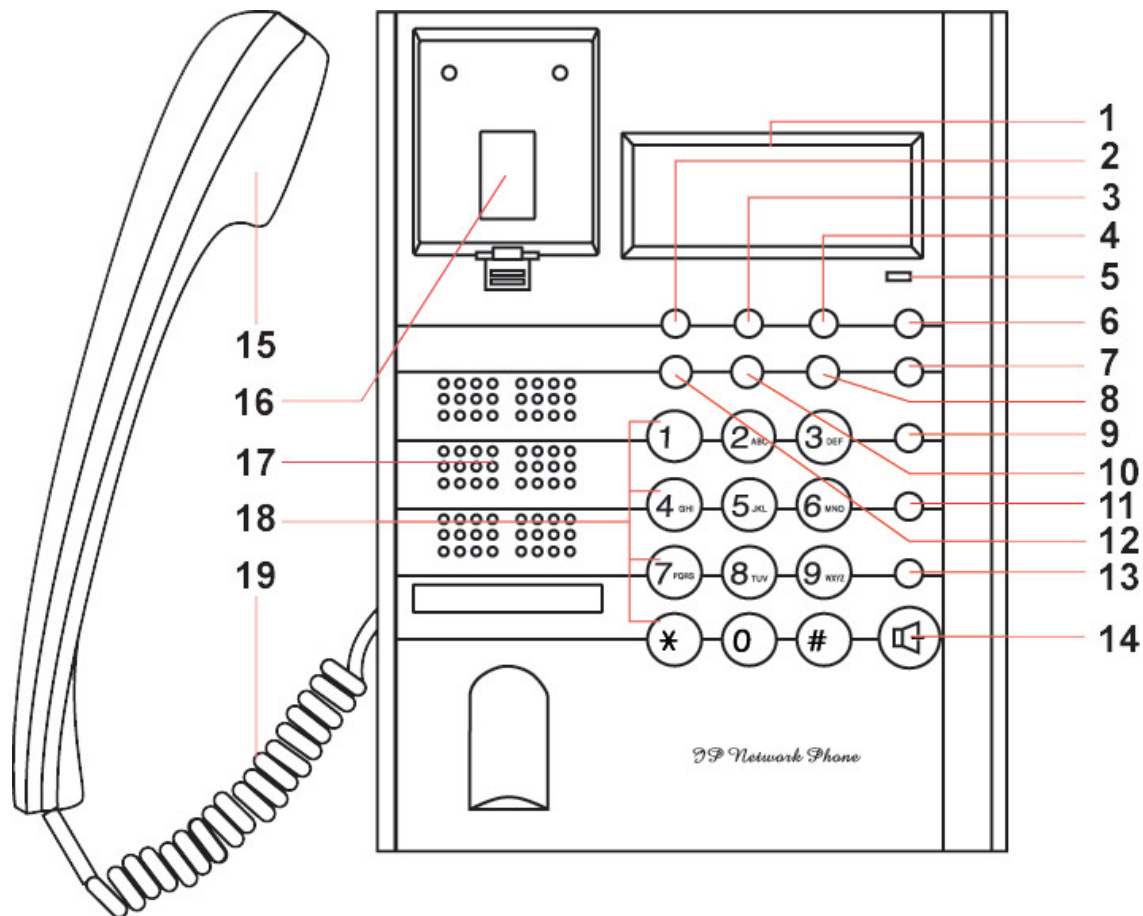


Figure 1 Front Panel

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

2. Rear view

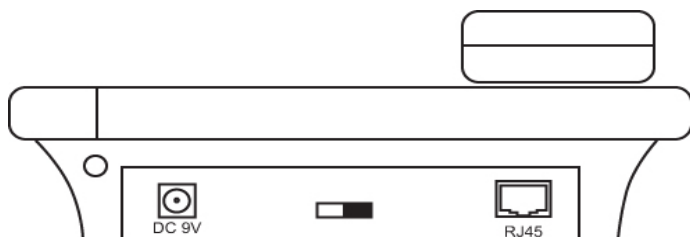


Figure 2 Backside view

- | | | |
|-----------------------|-----------------|---------------|
| 1. Power adapter port | 2. Power switch | 3. RJ-45 port |
|-----------------------|-----------------|---------------|

3. Function Keys

Key	Function
Missed	With handset on hook, press this key to review missed number
Local Num	With handset on hook, press this key to check your phone number
Local IP	With handset on hook, press this key to check phone ip address
Dialed	With handset on hook, press this key to review dialed number
Answered	With handset on hook, press this key to review answered number
Redial	IP phone dials the last dialed number when button is pressed
Speak	Press this key to place a call without taking handset from hook
Volume+	Increase the volumes of handset or speaker
Volume-	Decrease the volumes of handset or speaker
Keypad	With handset picked up or after pressing the “Spk/hand free” button, pressing these keys dials number

Performance and Features

- Supports H.323 v4, compatible with most H.323 v1-v4 system and devices;
- Built in H.323 proxy support to pass NAT;
- Supports MGCP RFC2705
- Supports 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;
- Provides 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;
- Automatic gatekeeper discovery in LAN;
- Call PSTN by ITSP's prepaid card (eTalk, italk, ringtec etc);
- DHCP support for automatic IP address assignment;
- PPPoE support for ADSL or Cable modems;
- Setup of all parameters through standard web browser (such as IE6.0), phone keypad or telnet;

- Supports G.723.1 5.3k/6.3k , G.729 , G.711 A-Law μ -Law audio codec algorithms;
- Dynamic voice detection; Echo cancellation; Comfort noise generation;
- Dynamic voice jitter buffer for high audio quality;
- Tone generation and Local DTMF generation/detection according with ITU-T;
- E.164 dial plan and customized dial rules;
- 80 entries each for missed call, answered call and dialed call;
- 100 phonebook entries;

- 2×16 LCD with backlight;
- Status LED;
- Speed dialing;
- Adjustable volume for both handset and speaker;

Technical parameters

- VoIP chip: Signal processor, 50Mhz with built-in coprocessor
- Data storage: 2MB SDRAM;
- Program memory: 1MB Flash memory;
- Network interface: 10Base-T/100Base-T, RJ-45 port;
- Echo cancellation: G165 16ms;
- Power consumption: 2.7W(max);
- Power adapter: input AC 110V or 220/230V, output DC 9V 500mA;
- Operating conditions:
 - Ambient temperature: 0-40° C (32-104°F)
 - Humidity: 10-90% non condensing
- Overall size: 215×190×70 mm (L×W×H).

Storage conditions

- Atmospheric pressure: 86-106Kpa;
- Storage temperature: -30° to 65° C (-22° to 149° F)
- Humidity: 10 to 90% non condensing

Installation

1. How to connect Handset and Phone

Insert Handset cord into handset cord jack of the base.

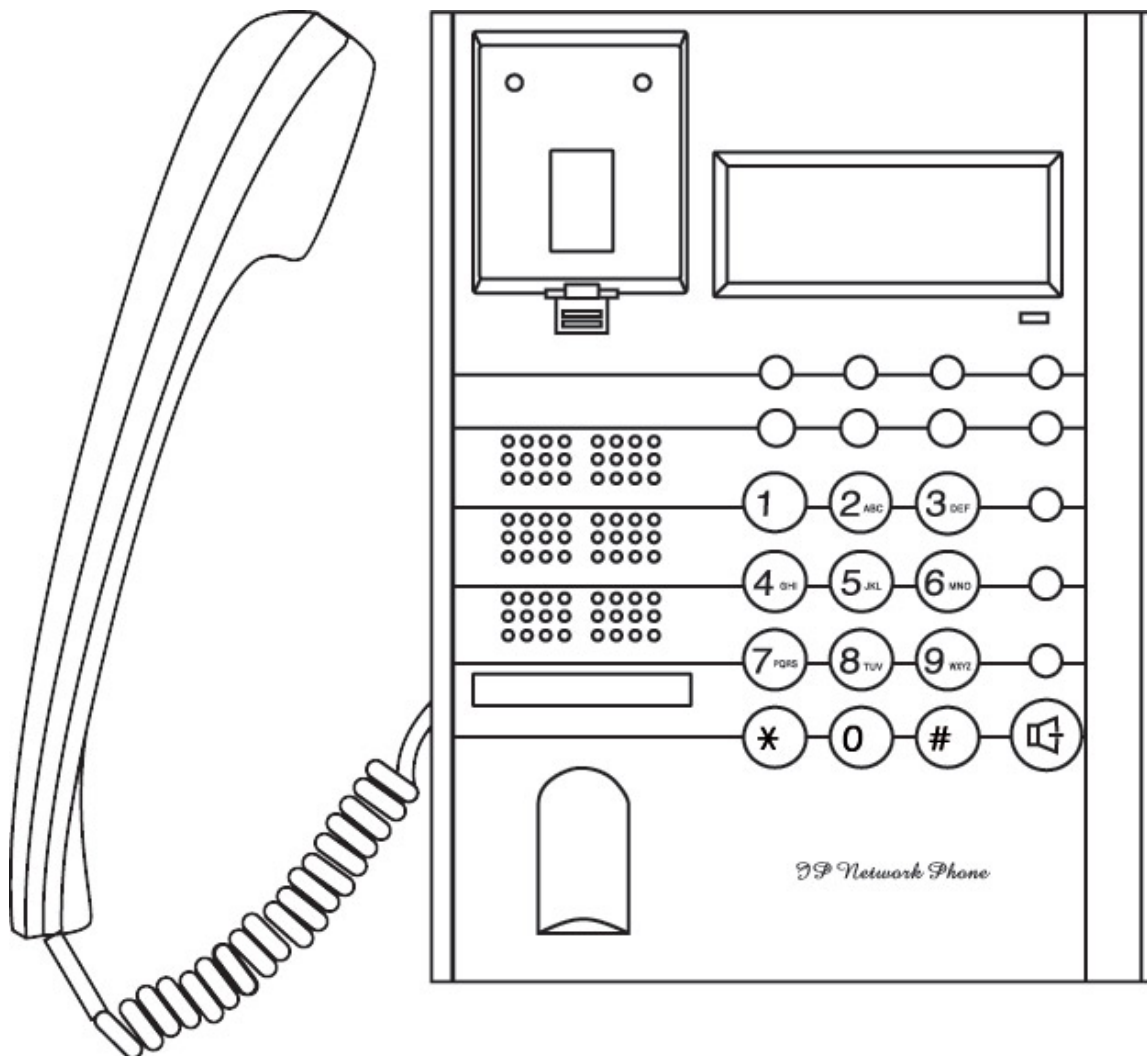


Figure 3 Handset and base of Phone connection

2. Connect Phone and Power

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. Please refer to Figure 4.

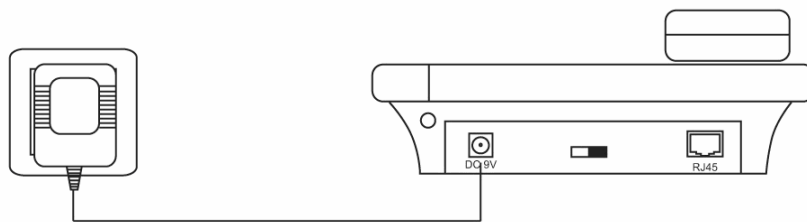


Figure 4 Connect Power

3. Connect the phone to the net

LAN users: Plug one end of the direct-connecting cable into RJ-45 jack which is located on the back of phone, then connect the other end of cable to a network hub or a network switch . Please see figure5.

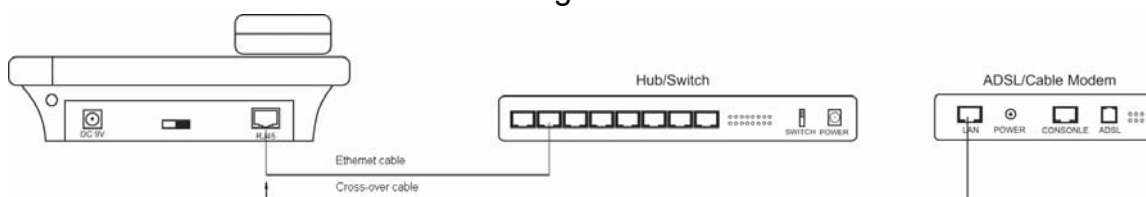


Figure 5 LAN Phone connected to net

ADSL/Cable Modem users: Plug the RJ-45 Ethernet cross-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 setting the switch on the back of the IP phone to “ON”. The LC-Display shows the status of the phone. After displaying “Booting...”, it shows that the phone is searching an IP telephony server. If no server can be found, please check your network connection and set up network and IP provider settings.

Configuration

There are three different ways to configure the G100 IP Phone: By using the phone keypad, a standard web browser or through Telnet.

1. Setup through phone keypad

Note: Default password for G100 phones is 1234.

(1) Entering into setting mode

Use the keypad to enter the password of the phone (when debug is not disabled (0[disable]), the default password is 1234; when debug is set to 0[disable], please use the super password 19750407, and then press “#” until this

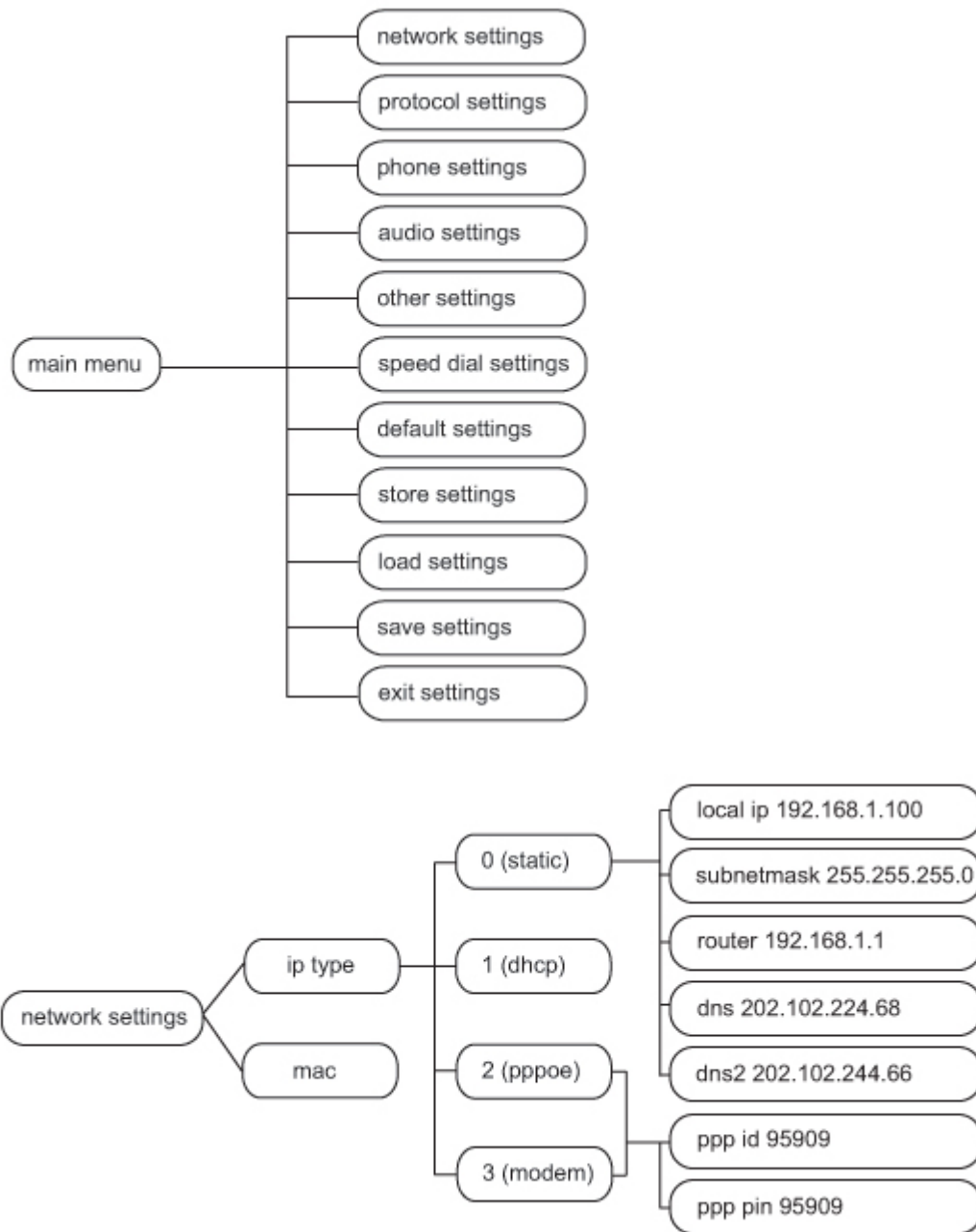
prompt is displayed: Password: .

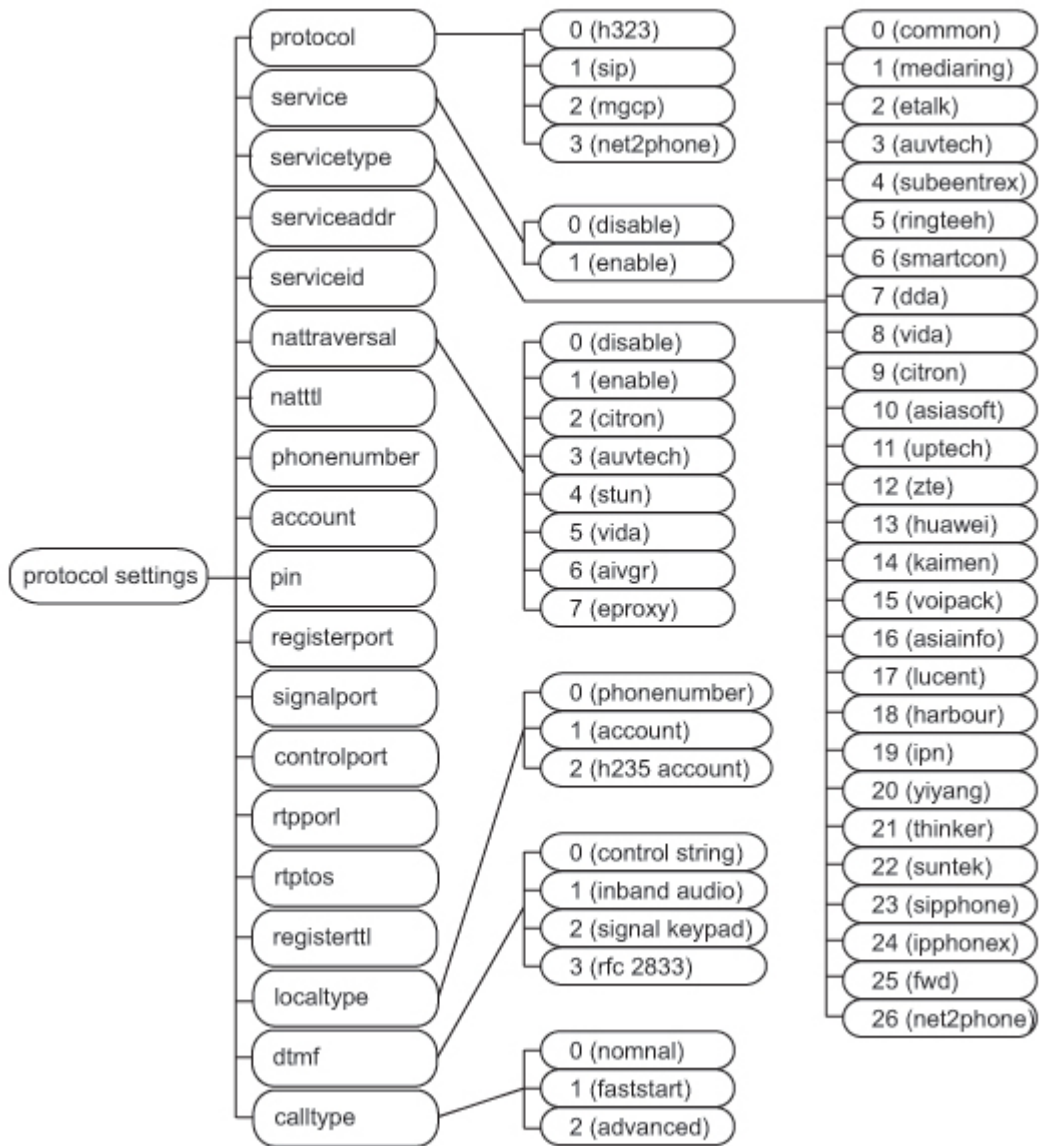
Then enter the super password again and press the red “Spk” button to let the phone enter setting mode.

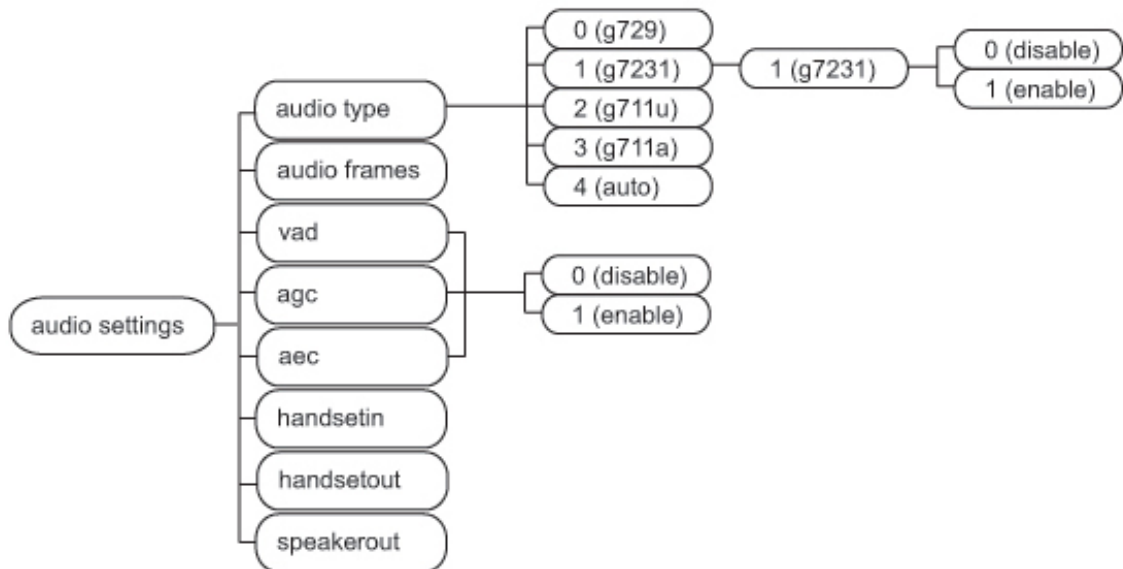
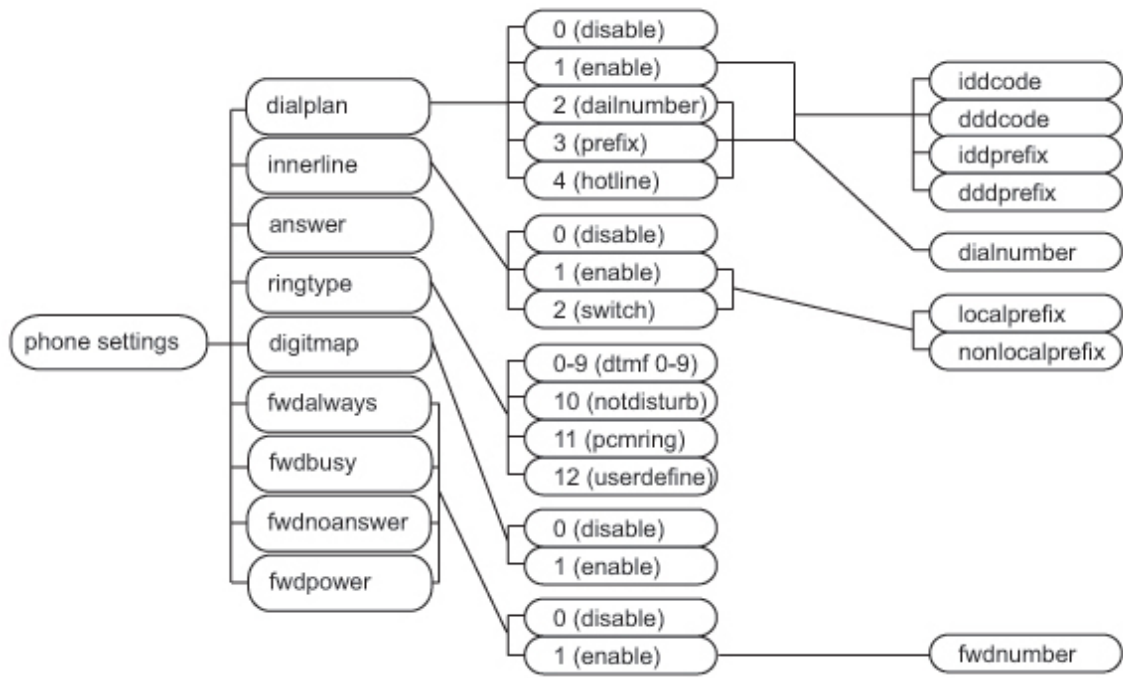
(2) Introduce of the function of keypad in the keypad setting mode

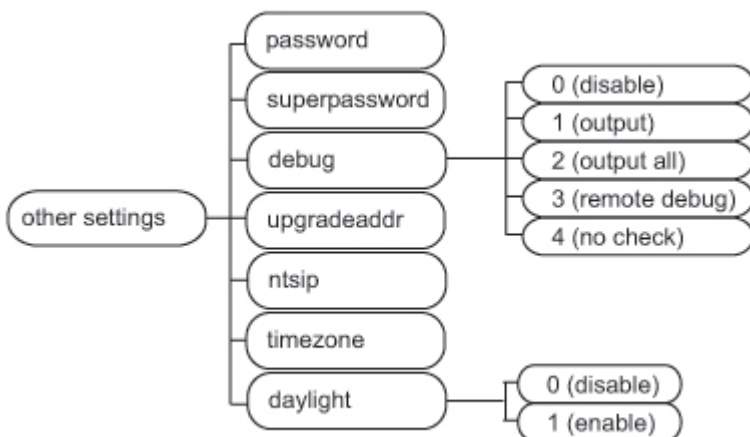
Press key	Function
Spk /Hand free	Enter into submenu of the current menu ;Acknowledge modification
Volume/+	Scroll menu forward
Volume/-	Scroll menu backward
Local Number	Press to change a setting
Speed Dial	Cancel current setting; return to parent menu
Back Space	Backspace during entering of setting
Number keypad	Input new setting. Please see appendix for character represented by each key

(3) Menu Structure









(4) Setting example

We take enabling eTalk prepaid card and setting the ID and pin as an example for how to set up the G100:

Enter setting mode

Use the keypad to input 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 make the phone enter “setup mode”.

Enable eTalk

Once the phone enters setup mode, **network settings** will be displayed, then press “Vol/+” to browse the menu until **protocol settings** is displayed. Press “Spk” to enter submenu and then continually press “Vol/-” to browse through the submenu until **Servicetype 0[normal]** is displayed on the screen. Press “Local Number” to enter change mode, then you will see a cursor blinking at the start of the second line. It indicates that you can change the value. Then press “2” to enable eTalk and then press “Spk” to confirm. **Servicetype 2[etalk]** will be displayed.

Set eTalk account (0117ETNS)

Press “Vol/-” to proceed in the menu, once **account** is displayed on the

screen, please press “Local IP” to enter change mode, then you will see a cursor blinking at the start of the second line. It indicates that you can change the value. Then press 0117 keys once, then for the alphanumeric characters the “3” key thrice, “8” key twice, “6” key thrice and “7” for times. Then press “Spk” to confirm.

After above operation, **account**
0117ETNS will be displayed on the LCD.

Set eTalk pin (409466193597)

Press “Vol/+” to proceed in the menu, once **pin** is displayed on the screen, please press “Local IP” to enter change mode then you will see a cursor blinking at the start of the second line. It indicates that you can change the value. Then press 409466193597 keys to input the password, and then press “Spk” to confirm. After above operation, **pin**
409466193597 will be displayed on the LCD.

Save setting and exit setting

Continually press “Redial” & “Speed Dial” to reach save setting menu **save settings**. Then please press the “Spk” key to confirm the change. When the menu **save Are you sure?** is displayed on the LCD, press “Spk” again to confirm. Once the IP phone restart successfully, the new setting will take effect.

Referring to the above operation; it is possible reach any menu item and to modify any value.

(5)Appendix

The following chart shows how often a key must be pressed to enter an alphanumeric character in setup mode.

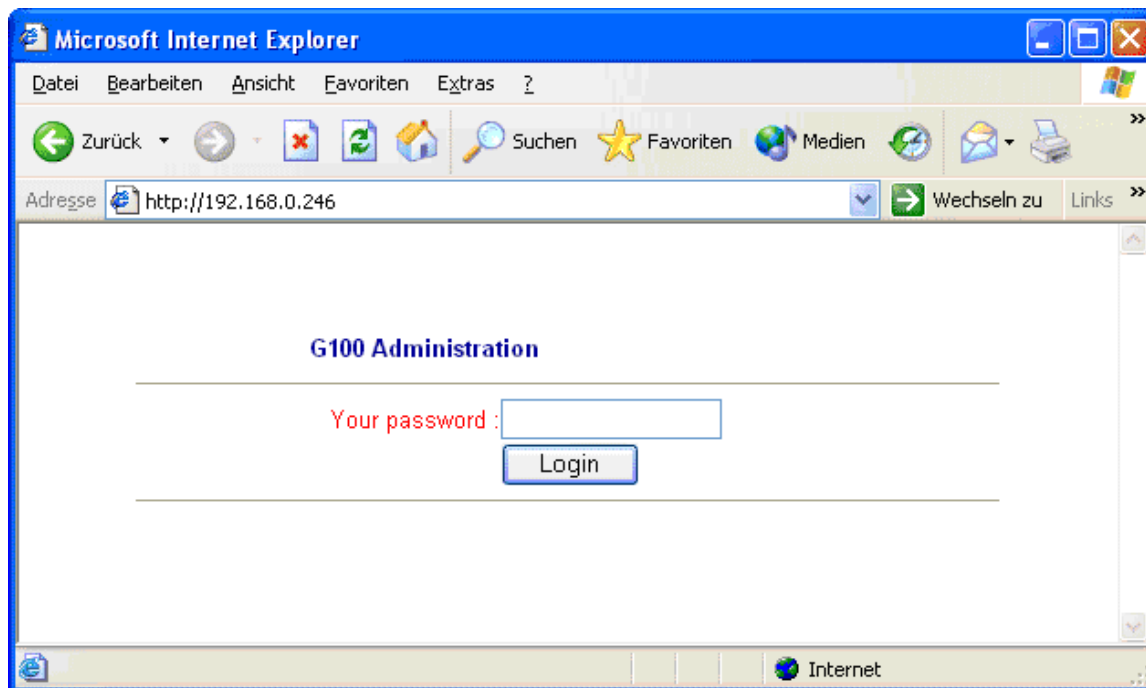
Press key	Press once	Press twice	Press three times	Press four times	Press five times
1	1	.	,	?	!
2	2	A/a	B/b	C/c	
3	3	D/d	E/e	F/f	
4	4	G/g	H/h	I/i	*
5	5	J/j	K/k	L/l	
6	6	M/m	N/n	O/o	#
7	7	P/p	Q/q	R/r	S/s

8	8	T/t	U/u	V/v	
9	9	W/w	X/x	Y/y	Z/z
0	0	Space	:/@	;/-	'/&
*					
#	Switch between upper case and lower case				

2. Configuration using a web browser

In order to use this functionality, a computer must be connected to the same network which the G100 IP phone is connected to. Open a web browser (e.g. Microsoft Internet Explorer). Input the IP address of the phone into address line () – the web page that is shown below must pop up. You can check the IP address of the phone by pressing the "Local IP" key on the G100. After pressing this button, the phone also registers with the protocol server of your service provider.

Please enter the password (defaults: 1234 or 19750407 for the super password) of the phone into the password edit line. Then click at the button.



(1) Network Settings

Network	Service	NAT	Telephone	Audio	Time	Password
Your type of internet connection	dhcp <input type="button" value="v"/>					
DHCP / statische IP / Modem / DSL						
Only for static IP address						
IP Adresse of this telephone	192.168.0.246					
DNS 1 IP Address	192.168.0.2					
DNS 2 IP Address	62.104.191.241					
Subnet Mask	255.255.255.0					
Router IP Address	192.168.0.2					
For direct ADSL / DSL connection						
User name / ID	<input type="text"/>					
Password / PIN	<input type="text"/>					
MAC address of this telephone	00-09-45-40-27-2a					
<small>Please change this only after a system crash when MAC address is reset</small>						
<input type="button" value="Save/Reboot"/>						

Your type of internet connection

- **static ip:** Select this item to set IP address and subnet mask manually.
- **dhcp:** Select this item to enable DHCP mode. If DHCP is selected, your router automatically assigns all the required network parameters to devices that are connected to it. The G100 IP phone is shipped with DHCP as factory default
- **pppoe:** ADSL and Cable Modem users can select this in order to let the IP phone establish directly the internet connection
- **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 pppin fields.
-

Only for static IP address

- **local ip:** With static ip selected in iptype drop down list, please enter IP address of IP phone here.
- **DNS 1 IP Address:** With static ip selected in iptype drop down list, please enter IP address of DNS (domain name) server here.
- **DNS 2 IP Address:** With static ip selected in iptype drop down list, please enter IP address of backup DNS server 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.

For direct ADSL / DSL connection:

- **User name / ID:** 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.

- **MAC address of this telephone:** MAC address is the physical address supplied by the Ethernet NIC that is built into the G100 IP phone. Each G100 phone is shipped with a unique MAC address that is also printed on the back of the base. Accidental changes of this parameter can cause the DHCP server to malfunction!

(2) Protocol settings

Network	Service	NAT	Telephone	Audio	Time	Password
Protocol		sip				
Service Type		common				
Protocol settings						
User name / account		Your user name				
Password / pin		Your password				
Phone number						
Type of account		auto				
Call type		advanced				
DTMF signal transmission		control string				
Protocol server / Ports						
Domain name / Realm						
Server to log in						
Register port		5060				
Signal port		5060				
Control port		0				
RTP Port / Voice		8000				
RTP Type of service		0				
Register ttl		60				
Save/Reboot						

- **Protocol:** Select an item from the dropdown list:
 - H.323: The phone will use the H323 protocol.
 - SIP: The phone will use the SIP protocol.
 - MGCP: The phone will use the MGCP protocol.
 - N2P: The phone will use the Net2phone private protocol.


Note With Net2phone selected here, please also enter the IP address or domain name of designated server into “Server to log in”; then set service port to

6801; fill account and password of Net2phone card into User name / account and Password /pin fields.


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.

- Common: no special requirements
 - Mediating: use Mediating prepaid
 - etalk: Use eTalk prepaid card by selecting this item. With eTalk selected here, please select dialnum in dial plan drop down list and then fill 00 into dial number field.
-

 **Note** When eTalk card is used, besides above configuration, please set other parameters as follows: Domain name / Realm: 202.83.204.213; Server to log in: ETNSGK; account: 0117ETNS.If IP phone cannot log into 202.83.204.213,please try to use 202.83.204.213. If the IP address of the phone is private, please do port mapping on NAT device.

- auvtech: Use Auvtech's h323 system by selecting this item.
-

 **Note** When connected with Auvtech's system, please select "auvtech" from the "nat traversal" list box, select "account" from the "local type" list box.

- subcentrex: Use subcentrex service by selecting this item.
- ringtec: Use Ringtec service by selecting this item. With this service used, please select dialnum in use dial plan drop down list and then fill account of Ringtec into dial number field.
- smartcon: Use smartcon service by selecting this item.
- dda: Use dda service by selecting this item.
- vida: Use Vida service by selecting this item.
- citron: Use citron service by selecting this item.
- asiasoft: Use asiasoft service by selecting this item
- Uptech: Use Uptech's MGCP system
- Zte: Use Zte's MGCP system
- Huawei: Use Huawei's MGCP or H.323 system
- Kaimen: Use Kaimen's MGCP system
- Voipack: Use Voipack's MGCP system

- Lucent: Use Lucent's H.323 system
- Harbor: Use Harbor's MGCP system
- IPN: Use IPN's H.323 system
- Yiyang: Use Yiyang's H.323 system
- Thinker: Use Thinker's H.323 system
- Suntek: Use Suntek's H.323 system
- Sipphone: Free SIP service on internet, please visit www.sipphone.com for more information.
- Inphonex: Free SIP service on internet, please visit www.inphonex.com for more information.
- Fwd: Free SIP service on internet, please visit www.freeworldialup.com for more information.
- Net2phone: Use Net2phone private protocol system.

Protocol settings

- Domain name / Realm and Server to log in: Please enter the URI (domain name/IP address : service port) of your service provider "Domain name / Realm". If default service ports are used, which is usually the case, ": service port" can be omitted. "Server to log in" is filled according to the protocol used. For the SIP protocol, this is generally the IP address of the SIP server.
- H.323: Please enter the URI of your gatekeeper into "Domain name / Realm". The default service port is 1719. If the gatekeeper has a GKID, please put it into "Server to log in".

To make IP-to-IP call, please clear the "Domain name / Realm" field. The default service port is 1720.

- MGCP: Please put the URI of the Call Agent into the "Domain name / Realm" field. The default service port is 2727. Leave "Server to log in" empty.
- SIP: Enter the URI of the SIP proxy server into "Domain name / Realm". Enter the domain name of the SIP proxy server into "Server to log in" or leave "Server to log in" empty. If the system has an Outbound Proxy, please put the URI of the Outbound proxy into "Domain name / Realm" and put the domain name of SIP proxy server into "Server to log in". The default service port is 5060.
- Net2phone: Please put the URI of the Net2phone server into "Domain name / Realm". The default service port is 6801. Leave "Server to log in" empty.
Net2phone usually use the following servers:
call1.net2phone.com, call2.net2phone.com, kip1.net2phone.com,
skip2.net2phone.com, skip1.net2phone.net, skip2.net2phone.net.

- **User name/ account:** If the H.323 protocol used with a calling card, please enter the account information on the calling card into this field..

If “Type of account” is set to “md5 account”, enter the ID in “User name/ account”.

If “Type of account” is set to “account”, enter the H.323 ID here.

If “Use dialplan” in the telephone settings (on next page) is set to “prefix”, enter country code and card number, followed by a # character here. Your service provider can tell you the correct settings.

If the SIP protocol is used, please enter your username into this field.

For the MGCP protocol, please enter local endpoint id (eg., aaln/0) here.


For Net2phone, please enter account of Net2phone card here.

- **pin:** H.323 protocol: please enter password of your calling card into this field; for md5 accounts, enter password of your account here. If “Use dialplan” in the telephone settings (on next page) is set to “prefix”, enter password, followed by a # character here. Your service provider can tell you the correct settings.

For the SIP protocol, please enter the password of your account in this field.

For the MGCP protocol, please enter domain name here.

For Net2phone, please enter password of Net2phone card here.

 **Note** When MGCP protocol is used, some servers require adding “[]” outside the domain name.

- **Phone number:** The local phone number or username of this phone. The phone number is part of the account information from your service provider.

- **RTP Type of service:** Enter data packet priority for IP packet headers here. Values range between 0 and 255 with 255 indicating highest priority. This parameter must be supported by the network in order to have effect.

- **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:** For the H.323 protocol, signal port is Q.931 port using TCP protocol, can be any number between 1024 and 65535.

- **Control port:** For the H.323 protocol, this port is H.245 port using TCP protocol, can be any number between 1024 and 65535.

- **Register ttl:** For H.323 or SIP protocol, the IP phone will send a keep-alive registration message to the H.323 gatekeeper or SIP proxy server every “Register ttl” seconds. The minimum value is 10, maximum value is 255. Default is 60.

- **RTP Port / Voice:** The RTP port is the port that is used to transmit and receive voice packets using UDP protocol. This is an even number between 1024 and 65535. It *cannot* be the same port as the **Register port**.

- **Type of account:** If the H.323 protocol is used, this parameter determines how the G100 IP Phone authenticates itself at the gatekeeper. SIP settings are also affected. For SIP, the “automatic” setting should be chosen:
 - 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 “User name / account” field as H323 ID to login GK.
 - Automatic: Best setting for SIP services
 - MD5 account: Use H235 encrypted username and password to log on at the gatekeeper
 - CAT account: Use Cisco access token
 - SHA-1: some service providers might use this feature
 - CNC: some service providers might use this feature

- **Call type:** The following choices are available from the 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. This is the recommended setting for the H.323 protocol.

- **DTMF signal transmission:** Sets dial tone signals for sending control strings: Select either “control string”, “inband audio”, “signal keypad” or “RFC2833” from dropdown list. Your service provider can advise for the best setting.

(3) NAT settings

Network	Service	NAT	Telephone	Audio	Time	Password
NAT: auto <input type="button" value="v"/> NAT address: <input type="text"/> NAT ttl: 30 <input type="text"/> <input type="button" value="Save/Reboot"/>						

- **NAT:** If the G100 IP Phone is part of a network that uses address translation (this is the case for every modern router) and if it needs to communicate with another IP phone which is part of a different LAN (e.g. another IP phone that is connected to the internet, please select an item from dropdown list to set the proxy used by the phone. “Auto” is the best setting for the SIP protocol.

- **auto:** Select this option if logon server and IP phone are in the same LAN, or the logon system supports IP phones working within a LAN (typical for routers).

- **enable:** When the system does not support IP phones working behind LAN, please select this item to search public IP address of the NAT device. With this item selected, the “NAT Address” field will be used. Besides, port mapping (port forwarding) needs to be properly set up on the NAT device.

- **Citron:** If the Citron private protocol is used, select this item to fit into the GnuGK system for transfer of voice and signal data through the router.

- **auvtech:** Select this item if Auvtech private system is used.

- **STUN:** Select this item if the SIP protocol used and STUN is required. Please enter the URI of a valid STUN server into the “NAT Address” field.

- **Vida:** Select this item if Vida private system is used.

- **Aivgr:** Select this item if aivgr private system is used.

- **NAT Address:** If “Nat traversal” is set to “enable”, please enter the domain name of your NAT server (such web servers help to determine the public IP address of the G100 IP phone) into “nat addr”, such as www.whatismyip.com.

If “Nat traversal” is set to “STUN”, please enter the URI of the STUN server for your service provider into “NAT Address”, in the format “domain name: service port or IP address:service port”. The default service port for stun is 3478.

- **NAT ttl:** If the IP phone is behind a NAT device, it will send packets to the 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.

(4) Telephone settings

Network	Service	NAT	Telephone	Audio	Time	Password	
Use dialplan							
			disable				
Dial number							
City code (ddd code)							
			10				
Country code							
			86				
International exit code							
			00				
National exit code							
			0				
Local prefix							
			0				
Nonlocal prefix							
			0				
Dialplan within the network							
			disable				
Time to answer							
			30				
Ring type							
			dtmf0				
Use Digitmap							
			<input type="checkbox"/>				
Forwarding of calls							
Forward to phone number							
If powered off							
			<input type="checkbox"/>				
On no answer							
			<input type="checkbox"/>				
Always							
			<input type="checkbox"/>				
If busy							
			<input type="checkbox"/>				
Save/Reboot							

● **Use dialplan:** Set whether a dial plan is used or if the phone performs standard number dialling. The following options are available:

- **disable:** Do not use the dial plan feature by selecting this option.
- **enable:** Use dial plan by selecting this option.
- **dialnum:** Use dial number by selecting this option. If this option is selected, please enter the dial prefix into the “Dial number” field.
- **prefix:** Use 179XX service by selecting this option.
- **Hotline:** Use Hotline function by selecting this option. IF this option is selected, please enter the hotline number that is provided by your service provider (if this feature is supported by the service) into dial number field.

Note If the 179xx service is used, please set as follows: fill call prefix into “Dial number” field, such as 17930; type country code, card number and # into “User name / account” field and fill password and # into “Password” field.

● **Dial number:** With dialnum selected in “Use dialplan” drop down list, please enter the dial prefix into this field according to the requirement of logon server. For


example, with eTalk card used, enter 00 here.

- **City code (ddd code):** With “enable” or “dialnum” selected in “Use dialplan” drop down list, set area code according to E.164 dial rule. For example, New York City: 917.

- **Country code:** With “enable” or “dialnum” selected in “Use dialplan” drop down list, set country code according to E.164 dial rule. For example, U.S.A: 1, Germany: 49 or China: 86.


- **International exit code:** With “enable” or “dialnum” selected in “Use dialplan” drop down list, set international call prefix according to E.164 dial rule: Usually the correct setting is 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. Usually the correct setting is 00.

 **Note** With “enable” or “dialnum” selected in “Use dialplan” drop down list, you can also set Country and exit codes according to requirements of the IP telephony service that you’re using.

- **Dialplan within the network:** Generally used for switching between dialplans, but this feature even allows for multiple services to be used with the same IP telephone! Enable/disable multi-settings by selecting corresponding option from dropdown list. The G100 IP phone allows to save up to 5 settings in total.

- **disable:** Disable multi-settings by selecting this option: The G100 IP phone will operate using the current setting.
- **enable:** Use designated service to place calls by selecting this item. By entering the Local prefix, the phone switches to backup setting 1.
- **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.

- **Ring type:** Set ring type by selecting corresponding item from drop down list.
 - dtmf 0-9: Standard ringtones
 - do not disturb: Set the phone to MUTE
 - pcmring: Set ringtone to music that is setup by the factory
 - user define : Set ringtone to music (loaded into the phone by user)

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

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

Forwarding of calls

- **Forward to telephone number:** Enter the IP phone number to which incoming calls are to be forwarded.

- **If powered off:** Forward calls if the G100 IP phone is powered off by checking this box. Please enter the IP phone number to which incoming calls are to be forwarded in the field “Forward to telephone number”.

- **On no answer:** Forward calls after some time if the call is not answered. Please enter the IP phone number to which incoming calls are to be forwarded in the field “Forward to telephone number”.

- **Always:** Forward all calls by checking this box. Please enter the IP phone number to which incoming calls are to be forwarded in the field “Forward to telephone number”.

- **If busy:** Forward calls if busy by checking this box. Please enter the IP phone number to which incoming calls are to be forwarded in the field “Forward to telephone number”. Depending on the service, this feature only provides users with a short tone if a new call is coming in.

(5) Audio settings

Network	Service	NAT	Telephone	Audio	Time	Password
Type of codec						
<input type="text" value="auto"/>						
Number of audio frames						
<input type="text" value="2"/>						
Audio settings						
Voice activity detection (VAD) <input type="checkbox"/>						
Automatic gain control (AGC) <input type="checkbox"/>						
Echo cancellation (AEC) <input checked="" type="checkbox"/>						
g.723.1 high data rate <input checked="" type="checkbox"/>						
Volume settings						
Microphone <input type="text" value="12"/> (max: 15)						
Handset out <input type="text" value="23"/> (max: 31)						
Speaker out <input type="text" value="24"/> (max: 31)						
<input type="button" value="Save/Reboot"/>						

- Type of codec:** Set coder/decoder (codec) type of the phone by selecting an option from the drop down list. The options are g.729, g.723.1, g.711u, g.711a and auto. "auto" is suggested ! If "auto" is selected, the IP phone will negotiate with the logon server which algorithm is best for incoming and outgoing calls. As the correct selection is determined for each call, optimum compatibility with any kind of hardware or software IP phones is guaranteed with the "auto" option.

- Number of audio frames:** Set number of audio frames in RTP package. If G.723 audio codec used, set this value to 1; while for the G.729 audio codec, 2 is the optimum value. Minimum is 1 and maximum is 8.

Audio settings

- Voice activity detection (VAD):** Enable/disable VAD (voice activity detection) by checking/unchecking this box. Bandwidth is lower if this checkbox is in checked state, but compatibility is better if not checked.

- Automatic gain control (AGC):** Enable/disable AGC of microphone gain by checking/unchecking this box.

- Echo cancellation (AEC):** Enable/disable AEC by checking/unchecking this box.

- G.723.1 high data rate:** If the G.723.1 codec is selected, enable/disable G.723.1 high rate by checking/unchecking this option.

Volume settings

- Microphone:** The value entered here represents the microphone gain (0:

minimum, ... 15: maximum)

- **Handset out:** The value entered here represents the output volume of the handset (0: minimum, ... 31: maximum)
- **Speaker out:** The value entered here represents the speaker volume (0: minimum, ... 31: maximum)

(6) Time settings

Network	Service	NAT	Telephone	Audio	Time	Password
Time server		210.59.157.10				
Timezone		(GMT)Dublin,Edinburgh,London,Lisbon				
automatic daylight savings		<input checked="" type="checkbox"/>				
Save/Reboot						

- **Time server IP:** Enter IP address of time server here.
- **Timezone:** Select correct time zone in dropdown list.
- **automatic daylight savings:** Enable/disable automatic daylight savings time switching by checking/unchecking this box.

(7) Password settings

Network	Service	NAT	Telephone	Audio	Time	Password
Password		●●●●				
Password repetition						
Super Password		●●●●●●●●				
Repetition of super password						(for TELNET and DEBUG Mode)
DEBUG Mode		no check				
Save/Reboot						

- **Password:** Set the password of the phone. (Default password is 1234).
- **Password repetition:** The same value as in “Password” must be entered in this field in order to make sure that no typos are in the new setting.


- **Super Password:** The super password of the phone. (Default password is 19750407).

- **Repetition of super password:** The same value as in “Super Password” must be entered in this field in order to make sure that no typos are in the new setting.

- **DEBUG Mode:** Sets the debug level of the phone. This parameter is used for finding errors in the firmware of the G100 IP phone or for checking compatibility issues for service providers.

- disable:** Disable output of bug messages by selecting this item.
- output:** Output operation informations.
- output all:** Output all informations.
- remote debug:** Save debug informations in SDRAM of IP phone by selecting this option.
- no check:** Disable checks.

- **Save/Reboot:** Click this button to save the configuration and restart the phone. Once the phone restarts successfully, the new configuration is effective.

 **Note** After loading settings page from the G100 IP Phone, the **Save/Reboot** button must be pressed within 5 minutes. Otherwise the validity of new settings expires. When clicking at the **Save/Reboot** button, the index page asking for your password will pop up again.

Phone book

No.	Name	Phone Number	No.	Name	Phone Number
001	Test	123456789	002	IP Test	192*168*62*34

Save/Back

• **Phone Book:** Click this button to access speed dial settings. On this page you can set and save speed dial numbers by typing names into **Name** fields and corresponding phone numbers in **Phone Number** fields. For example, input “Jack” in Name field following 001, and then input 5989426454 into Phone number field. Then Jack’s number 5989426454 is saved in phone book. Please click the **Save/Back** button subsequently. In normal state, you can use speed dial to call numbers saved in phone book.

🔧**Note** If H.323 or SIP protocol is used, you can save the IP address of another phone in the phone number field. Please use “*” instead of “.”. In order to save the IP address 192.168.1.221, please input 192*168*1*221.

Firmware / Ring File update

Firmware File Name:

Ring File Name:

Please download the required file to update from the internet. Next choose the downloaded file and click Update.

• **Upgrade Firmware:** Click this button to update the program of IP phone. Before updating, please select a valid *.bin file containing a new firmware in **Firmware File Name**, and then click at the “Update Firmware” button. Then the phone will read in the content of the bin file from your computer and subsequently it will store this data permanently in the phone.

🔧**Note** Please do not stop a firmware upgrade once it is in progress. The G100 IP phone could e.g. be damaged by power cuts.

- **Update Ring:** Click this button to update the user-defined ring tone. The functionality is analogous to updating the firmware.

Usage of the phone

1. Receiving calls

G100 IP phone can receive incoming calls from other IP phones and devices. It works just like an ordinary phone for incoming calls. When it rings, you can receive the call by the following methods:

(1) Use handset

Lift the handset and start speaking. When the call is over, put the handset back on the hook.

(2) Handset to handsfree


While receiving a call with the handset, press „Handsfree” on the keypad and then put down the handset. When the call is over, press “Handsfree” again.

(3) Use Handsfree

Press “Handsfree” to speak to the other party. When the call is over, press „Hand free” again.

(4) Hand free to handset

While receiving the call with the “Hand free” pressed, pick up the handset to continue the call. When the call is over, put the handset back on the hook.

 **Note** When you communicate with the other party without lifting the handset, please do not exceed a distance of about 1 to 2 feet from the G100 IP phone.

2. Place a call

(1) Call another IP phone (your G100 IP Phone must be logged on protocol server)

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 on the hook. The dialed number has been saved in the buffer.

Handsfree: 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 in 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 logging on protocol server

If the G100 IP phone does not log on a protocol server , 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".

(3)Place a call through Gateway

If G100 IP phone does not login the Gatekeeper, you can place a call through a Gateway directly by lifting the handset or pressing "Hand free" and then entering the IP address of the other party (e.g. by entering 192*168*0*6), and then pressing "#" or "Call" .

Note When you place a call without a Gatekeeper or with Gateway, please log off the Gatekeeper.

View Records

View missed calls

Hit the “Missed” (calls) button on the keypad. You will then be able to read records of missed calls. Hit Vol/+ or Vol/- on keypad to browse through the log.

View answered calls

Hit the “Answered” (calls) button on the keypad. You will then be able to read records of answered calls. Hit Vol/+ or Vol/- on keypad to browse through the log.

View dialed numbers

Hit the “Dialed” (calls) button on the keypad. Click dialed call. You will then be able to read records of dialed calls. Hit Vol/+ or Vol/- on keypad to browse through the log.

When you read the number you want to dial, please press “#” or “call” key to place a call directly.

Note The G100 IP phone supports saving over 100 entries of unanswered call, dialed calls and received calls. When the entries exceed 127, the latest record will overwrite the oldest entry. Records will be lost when the phone restarts or is turned off.

Copyright © 2005,
Global IP Telecommunications, Inc • 92 Derrick Road • Bradford, PA 16701 • U.S.A.
All rights reserved.