A-100 SIP Phone User Manual

A-100 IP phone Appearance Introduction:

- **A-100 IP Phone Front Illustration** (Refer to Fig 1.1):

Fig 1.1 A-100 IP Phone Illustration

➢ Function Keys of A-100 IP Phone Introduction (Normal State):

Keys	Function
LOCIP	With handset hung, press this key to get local IP address of the phone
MISCAL	With handset hung, press this key to review missed number
ANSCAL	With handset hung, press this key to review received number
NUMBER	With handset hung, press this key to get phone number
DIACAL	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

SPEAKER	Press this key to have a call without lifting the handset
VOL+	Increase the volumes of handset or speaker; turn over the record backward
VOL-	Decrease the volumes of handset or speaker; turn over the record forward

➢ Features

➢ Hardware

- Main chip—PA1688 50MHz
- Data Memory—2MB SDRAM
- Program Memory—1 MB Flash memory
- Ethernet Jack—1/2 10/100M jacks
- AC/DC adapter—Input AC100--- 230V，Output 9V DC, 1A

➢ Software

- DHCP support for LAN or Cable modem
- PPPoE support for ADSL or Cable modem
- Set phone by HTTP web browser (IE6.0) or Telnet
- Upgrade by FTP
- Support major G.7XX ;GSM610 audio codec
- VAD(Voice active detect)
- CNG (Comfort noise generation)
- Dynamic voice jitter buffer
- G.168/165 compliant 16ms echo cancellation
- Tone generation and Local DTMF re-generation according with ITU-T
- E.164 dial plan and customized dial rules
- 100 entries for quick dial
- 80 entries each for missed calls, answered calls and dialed calls
- Adjustable volume for both handset and speaker
- Voice prompt
- Hotline

➢ Standard and Protocol

A-100 IP Phone supports following standard and protocol:

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- Major G.7XX; GSM610 audio codec
- SIP(RFC 2543;RFC3261)
- TCP/IP: Internet transfer and control protocol
- RTP: Real-time Transport Protocol
- RTCP：Real-time Control Protocol
- VAD/CNG save bandwidth
- DHCP：Dynamic Host Configuration Protocol
- PPPoE：PPP Protocol over Ethernet
- DNS：Domain Name Server
- Telnet：Internet's remote login protocol
- FTP：File Transfer protocol
- HTTP：Hyper Text Transfer protocol

➢ **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

➢ **Electric requirements:**

- Voltage: 9V~24V
- Power adapter: output DC 12V/450mA
- Network interface:1/2X RJ-45 Ethernet Connectors

- **Size：**

200 x 195 x 87 mm (L x W x H)

- ## Installation:

1、 Connect handset to base: insert handset cord into handset cord jack at the left side of the base.
2、 Connect IP phone to Internet: plug the RJ-45 Ethernet cable into the Ethernet Jack. Plug the other end of the cable into HUB.

Power on IP phone: plug the power cord adapter into the Power Jack. Then plug the other end of the power cord adapter into the appropriate wall outlet.

- **Configuration**

Four different ways can be used to configure A-100 IP phone: phone keypad, web browser, Telnet commands and PalmTool configuration tool on computer.

- **Phone Keypad setup**

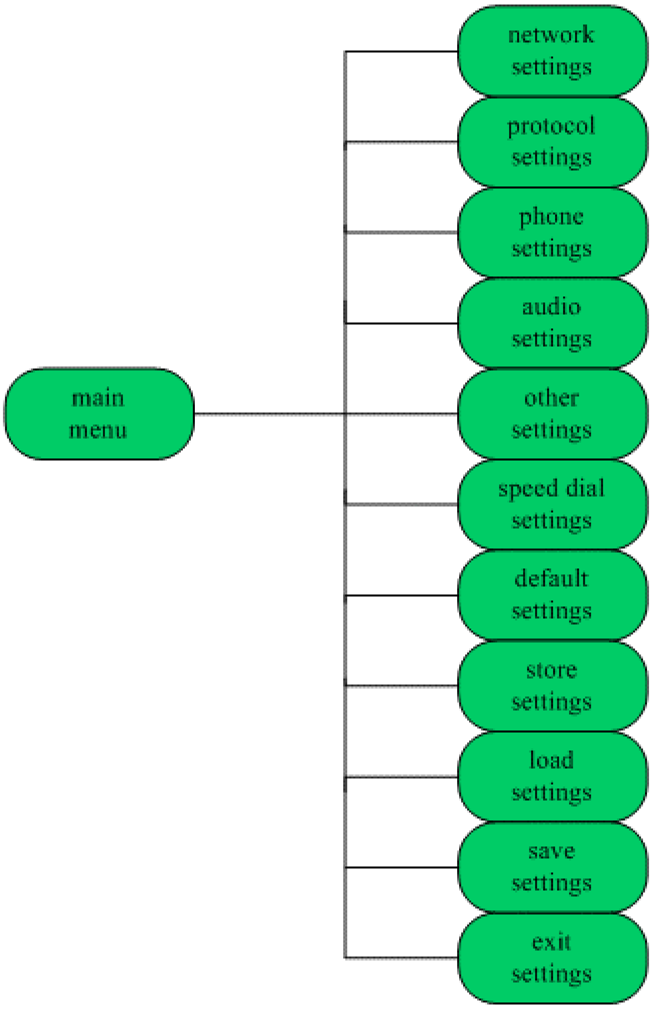
  - **Function Keys Introduction:** When using keypad and LCD to configure the settings of IP phone, following keys will be used:

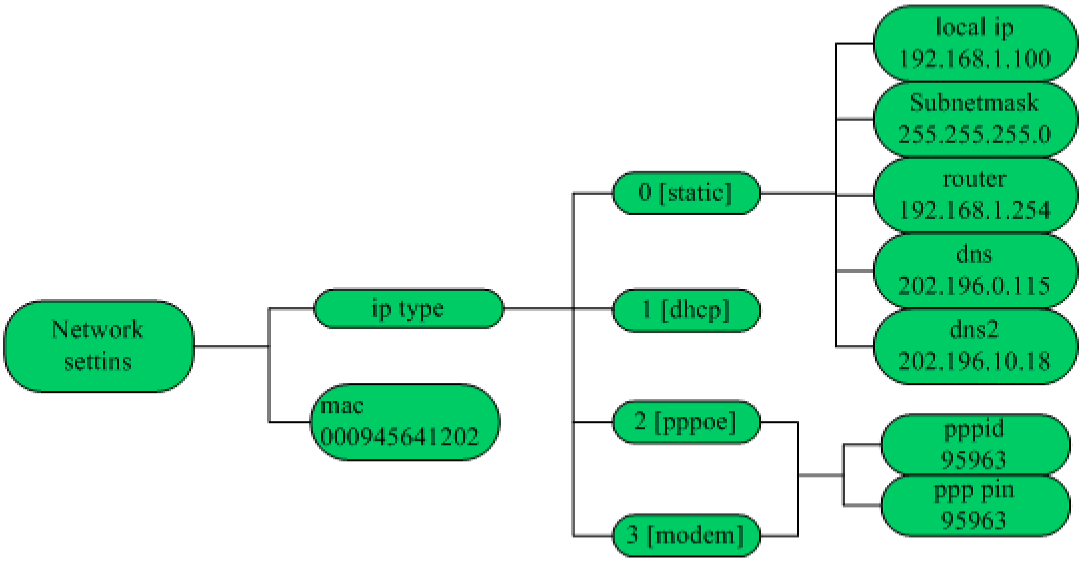
Keys	Function	Keys	Function
LOCIP	Enter setting mode	**SPEAKER/OK**	Enter submenu; confirm change
VOL+	Turn over menu backward	**VOL-**	Turn over menu forward; move cursor backward
LOCIP/SET	Modify values	**REDIAL/EXIT**	Exit current menu; exit setting mode

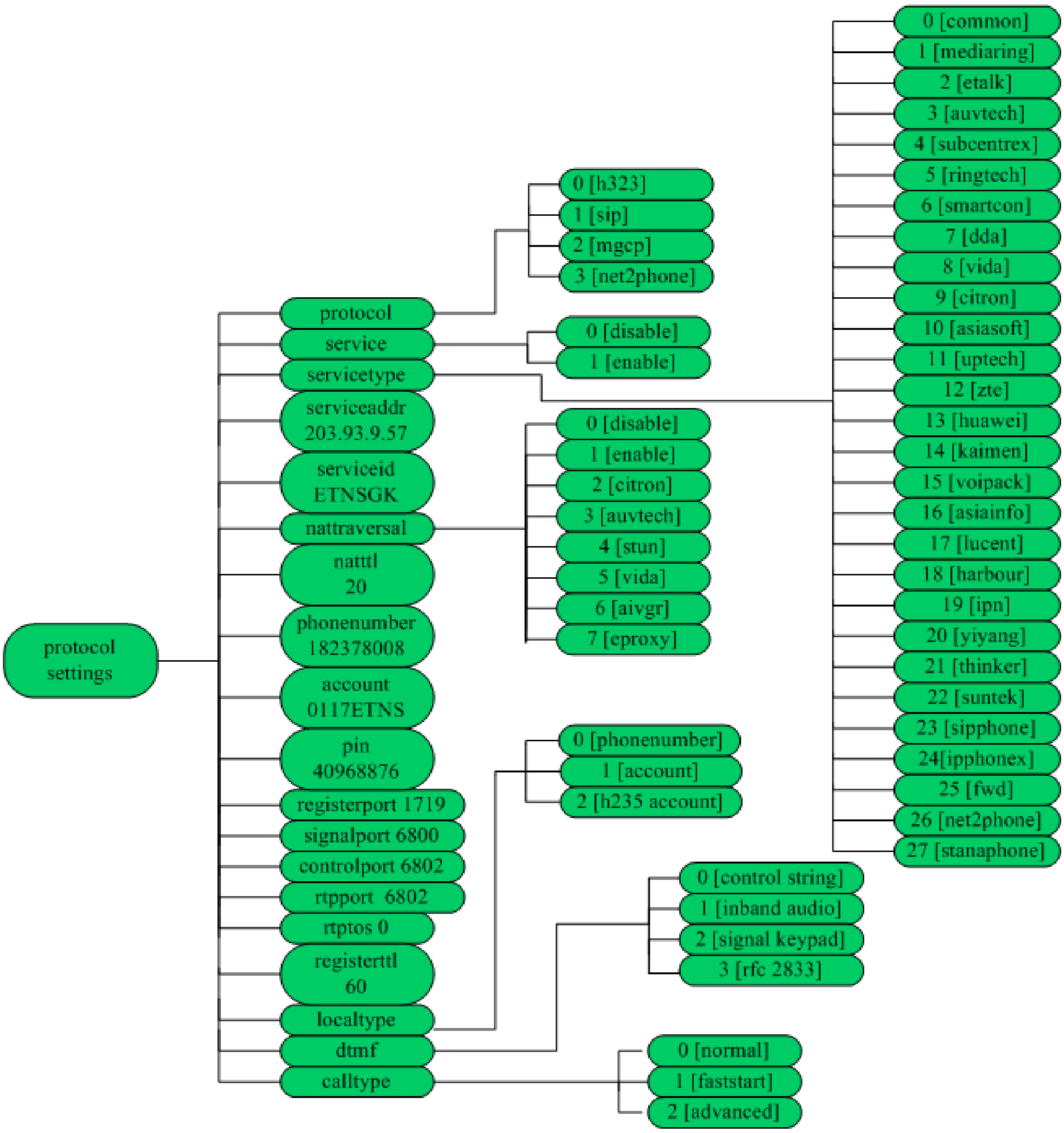
Refer to above operation; you can reach any menu to modify any value.

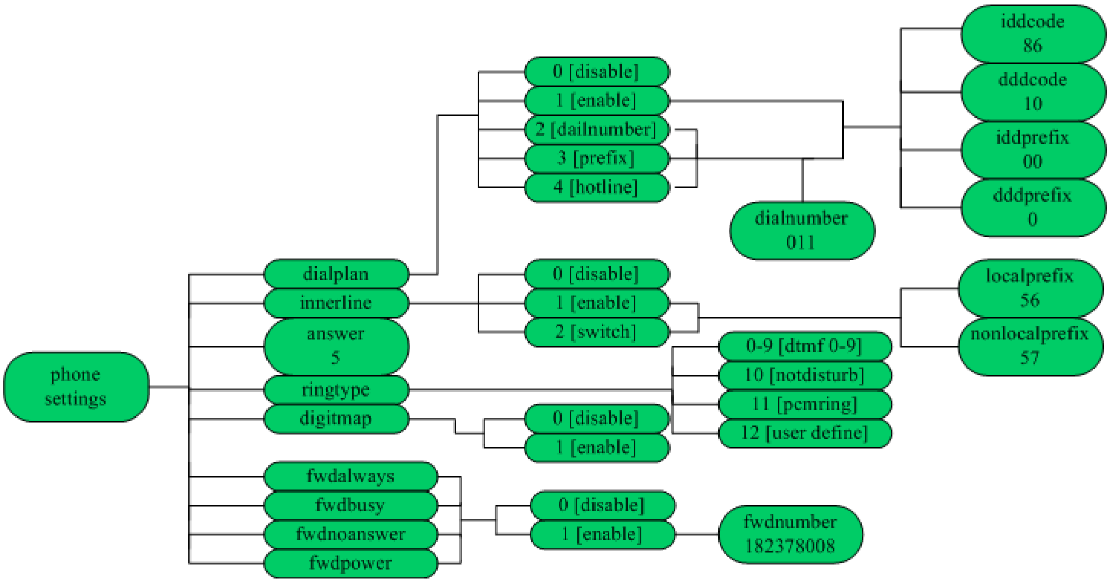
Please refer to following structure illustrations to learn the values of each menu item. As for the meaning of each item and value, please refer to ***Web Browser Setting*** chapter.

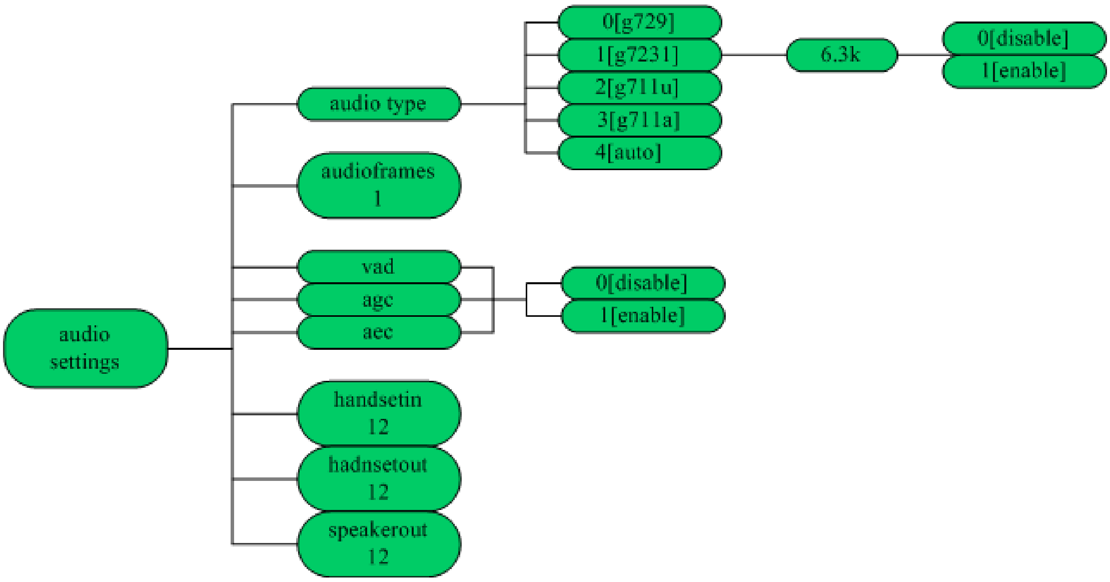
- **Menu Structure:**

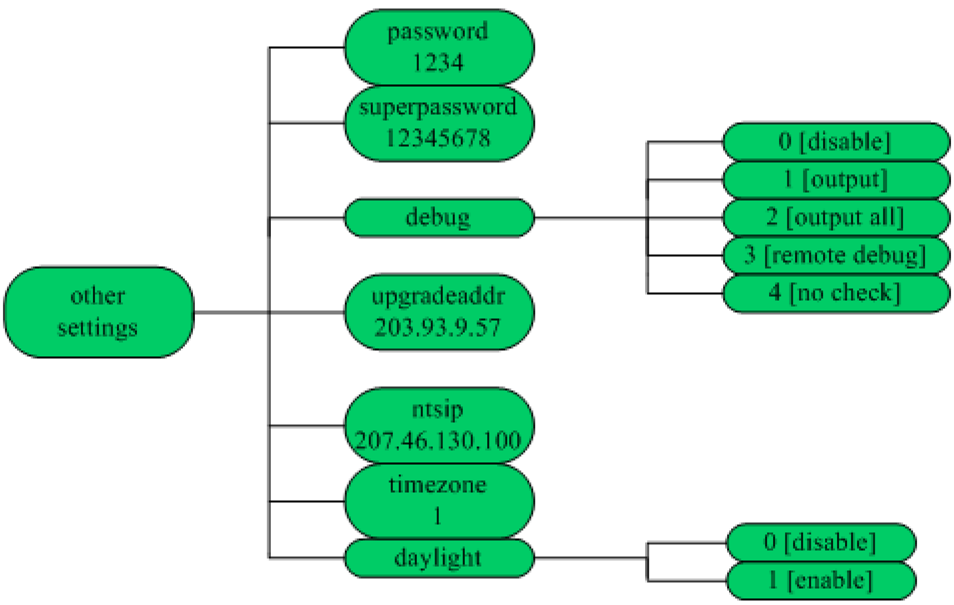



Network settins
ip type
mac 000945641202
0 [static]
1 [dhcp]
2 [pppoe]
3 [modem]
local ip 192.168.1.100
Subnetmask 255.255.255.0
router 192.168.1.254
dns 202.196.0.115
dns2 202.196.10.18
pppid 95963
ppp pin 95963


protocol settings
protocol
service
servicetype
serviceaddr 203.93.9.57
serviceid ETNSGK
nattraversal
natttl 20
phonenumber 182378008
account 0117ETNS
pin 40968876
registerport 1719
signalport 6800
controlport 6802
rtpport 6802
rtptos 0
registerttl 60
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]
0 [phonenumber]
1 [account]
2 [h235 account]
0 [control string]
1 [inband audio]
2 [signal keypad]
3 [rfc 2833]
0 [normal]
1 [faststart]
2 [advanced]
0 [common]
1 [mediaring]
2 [etalk]
3 [auvtech]
4 [subcentrex]
5 [ringtech]
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]


phone settings
dialplan
innerline
answer 5
ringtype
digitmap
fwdalways
fwdbusy
fwdnoanswer
fwdpower
0 [disable]
1 [enable]
2 [dailnumber]
3 [prefix]
4 [hotline]
iddcode 86
dddcode 10
iddprefix 00
dddprefix 0
dialnumber 011
0 [disable]
1 [enable]
2 [switch]
localprefix 56
nonlocalprefix 57
0-9 [dtmf 0-9]
10 [notdisturb]
11 [pcmring]
12 [user define]
0 [disable]
1 [enable]
0 [disable]
1 [enable]
fwdnumber 182378008


audio settings
audio type
audioframes 1
vad
agc
aec
handsetin 12
hadnsetout 12
speakerout 12
0[g729]
1[g7231]
2[g711u]
3[g711a]
4[auto]
6.3k
0[disable]
1[enable]
0[disable]
1[enable]


other settings
password 1234
superpassword 12345678
debug
upgradeaddr 203.93.9.57
ntsip 207.46.130.100
timezone 1
daylight
0 [disable]
1 [output]
2 [output all]
3 [remote debug]
4 [no check]
0 [disable]
1 [enable]

➢ Configured by WEB

Double click  icon to open the IE browser. Input the IP address of the phone into address bar (Address 192.168.1.100), and then input password of the phone into the following page. Default password 1234 is ordinary password and super password is 12345678. 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 wills popup. Refer to Fig 3.1 please.

network settings					
iptype	dhcp	ppp id		ppp pin	
local ip	192.168.1.212	subnet mask	255.255.255.0	router ip	192.168.1.254
dns	202.106.46.151	dns2	202.96.128.68	mac	00-09-45-60-16-63
protocol settings					
use service	☑	register ttl	15390	jitter size	0
service type	common	service addr	203.93.9.57	service id	
nat traversal	citron	nat addr		nat ttl	0
phone number	1067	account		pin	
register port	1720	signal port	1720	control port	1722
dtmf	control string	dtmf payload	101	rtp port	1722
local type	account	call type	advanced	rtp tos	0
phone settings					
use dialplan	enable	dial number		dddcode	10
iddcode	86	iddprefix	00	dddprefix	0
innerline	enable	local prefix	0	nonlocal prefix	0
ring type	dtmf0	use digitmap	☐	call waiting	☐
forward number	82378009	fwd poweroff	☐	fwd noanswer	☑
fwd always	☐	fwd busy	☐	answer	30
audio settings					
vad	☑	agc	☐	aec	☑
codec1	g7231	codec2	g729	codec3	g711u
codec4	gsm	codec5	null		
g.723.1 high rate	☑	audio frames	2		
handset in	7	handset out	20	speaker out	20
other settings					
password	1234	super password	12345678	debug	disable
sntp ip	210.59.157.10	use daylight	☐	upgrade addr	
timezone	(GMT+08:00)Beijing,Hong Kong,Urumqi				
Save Settings		Address Book		Upgrade Firmware	

Fig 3.1 Http Setting

Network Setting：

network settings					
iptype	static	ppp id	test1	ppp pin	test1
local ip	192.168.1.100	subnet mask	255.255.255.0	router ip	192.168.1.254
dns	202.106.196.115	dns2	202.96.128.68	mac	00-09-45-63-17-d8

Fig 3.2 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. A-100 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. A-100 phone is shipped from the factory with a unique algorism MAC address printed on the back of the base.

Protocol Setting：

protocol settings					
use service	☑	register ttl	15390	jitter size	0
service type	common	service addr	203.93.9.57	service id	
nat traversal	citron	nat addr		nat ttl	0
phone number	1067	account		pin	
register port	1720	signal port	1720	control port	1722
dtmf	control string	dtmf payload	101	rtp port	1722
local type	account	call type	advanced	rtp tos	0

Fig 3.3 Protocol Setting

- **use service:** Enable/disable service by checking/clearing this box. To make calls through SIP Proxy Server, please check this box; otherwise, phone can only make IP-to IP calls.

- **register ttl :** IP phone will send a keep-alive registration message to SIP proxy server every "**register ttl**" seconds. The minimum value is 10, maximum value is 65535. Default is 60.

- **jitter size :** Set buffer size of RTP package. The value range is 0-32.

- **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
  - **utstarcom:** Use UtstarCom's SIP system
  - **zte:** Use ZTE's SIP system
  - **ngtel:** Use NGTEL's SIP 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's SIP system
  - **stanaphone:** Use Stanaphone's SIP system
  - **italkbb:** Use ItalkBB's SIP system

- **service addr, service id**:

  If "**use service**" is checked, please put the URI of the SIP proxy server into "**service addr**". Put the domain name of the SIP proxy server into "**service id**" or leave "**service id**" empty. If the system has an Outbound Proxy，please put the URI of the Outbound proxy into "**service addr**" and put the domain name of SIP proxy server into "**service id**". The default service port is 5060. If "**use service**" is not checked, please clear "**service addr**" and "**service id**".

- **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 addr:** 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 65535, default value is 20.

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

- **account**: With SIP system which requires authentication, please put the username/account into this field.

- **pin:** With SIP system which requires authentication, please put the password into this field.

- **register port:** The local UDP port registered with server to accept incoming handshaking messages. The default port number is 5060.

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

- **dtmf payload :** When DTMF select **rfc 2833**.This parameter can be used indicating type of RTP payload type. The value can be use integer 96-101.

- **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**".

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

  - **normal:** When SIP system not support "outbound proxy", selecting this item.

- **advanced:** When SIP system support "outbound proxy", selecting this item. default setting is this item.

- **rtp tos:** Set the TOS field of the IP header of the RTP packets. The bigger this value is 0, the higher priority the packet is .

phone settings:

phone settings					
use dialplan	enable	dial number		dddcode	10
iddcode	86	iddprefix	00	dddprefix	0
innerline	enable	local prefix	0	nonlocal prefix	0
ring type	dtmf0	use digitmap	☐	call waiting	☐
forward number	82378009	fwd poweroff	☐	fwd noanswer	☑
fwd always	☐	fwd busy	☐	answer	30

Fig 3.4 Phone Setting

- **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 specially 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.
- **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. A-100 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.
- **local prefix:** With **enable** selected in **innerline** dropdown list, please fill the number switching to local call, such as 0.
- **nonlocal prefix:** With **enable** selected in **innerline** dropdown list, please fill the number switching to long-distance call, such as 9.
- **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**：Set ring as music saved by user by selecting this item.
- **Use digitmap:** Enable/disable digit map by checking/unchecking the box.

- **Call waiting:** Enable/disable call waiting by checking/unchecking the box.

- **forward 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 noanswer:** Forward calls without replying 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.

- **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.

Audio settings:

audio settings					
vad	☑	agc	☐	aec	☑
codec1	g7231	codec2	g729	codec3	g711u
codec4	gsm	codec5	null		
g.723.1 high rate	☑	audio frames	2		
handset in	7	handset out	20	speaker out	20

Fig 3.5 Audio Setting

- **vad:** Enable/disable VAD (voice activity detection).

- **agc:** Enable/disable AGC.

- **aec:** Enable/disable VEC.

- **codec1:** Set the priority 1of the audio compression algorithm. The options are **g729**，**g7231**，**g711u**，**g711a** and **gsm**.

- **codec2:** Set the priority 2of the audio compression algorithm. The options are **g729**，**g7231**，**g711u**，**g711a** and **gsm**.

- **codec3:** Set the priority 3of the audio compression algorithm. The options are **g729**，**g7231**，**g711u**，**g711a** and **gsm**.

- **codec4:** Set the priority 4of the audio compression algorithm. The options are **g729**，**g7231**，**g711u**，**g711a** and **gsm**.

- **codec5:** Set the priority 5of the audio compression algorithm. The options are **g729**，**g7231**，**g711u**，**g711a** and **gsm**.

- **g.723.1 high rate:** enable/disable g.723.1 high rate. G.723.1 high rate is 6.3kbps, low rate is 5.3kbps.

- **audio frame:** Set audio frames in RTP package. Minimum is 1 and

  maximum is 8.

- **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:

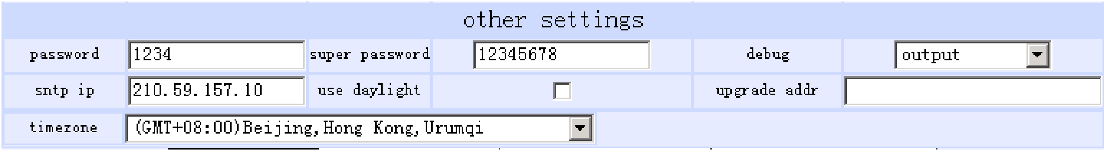


Fig 3.6 Other Setting Page

- **password:** Set the password of the phone. (Default password is 1234).
- **super password:** Set the super password of the phone.(Default super password is 12345678).
- **debug:** Set the debug level of the phone.
  - **disable:** Disable output the debug 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 debug information and data in test window by selecting this item.
  - **remote debug:** Save the debug information in SDRAM of IP phone by selecting this item.
  - **no check:** Disable checking firmware tags when upgrading. This is not suggested, because it will increase the risk of upgrading the wrong firmware into the phone.
- **upgrade addr:** Put IP address or domain name obtained by ISP of FTP server supplying upgrade program into this field.

- **nts ip:** Fill IP address of time server here.

- **use daylight:** Enable/disable daylight.

- **timezone:** Select correct time zone in list box.

When debug set as 0[disable], if input ordinary password (default one is 1234), then following page will pop up after clicking login. And only those parameters can be modified.

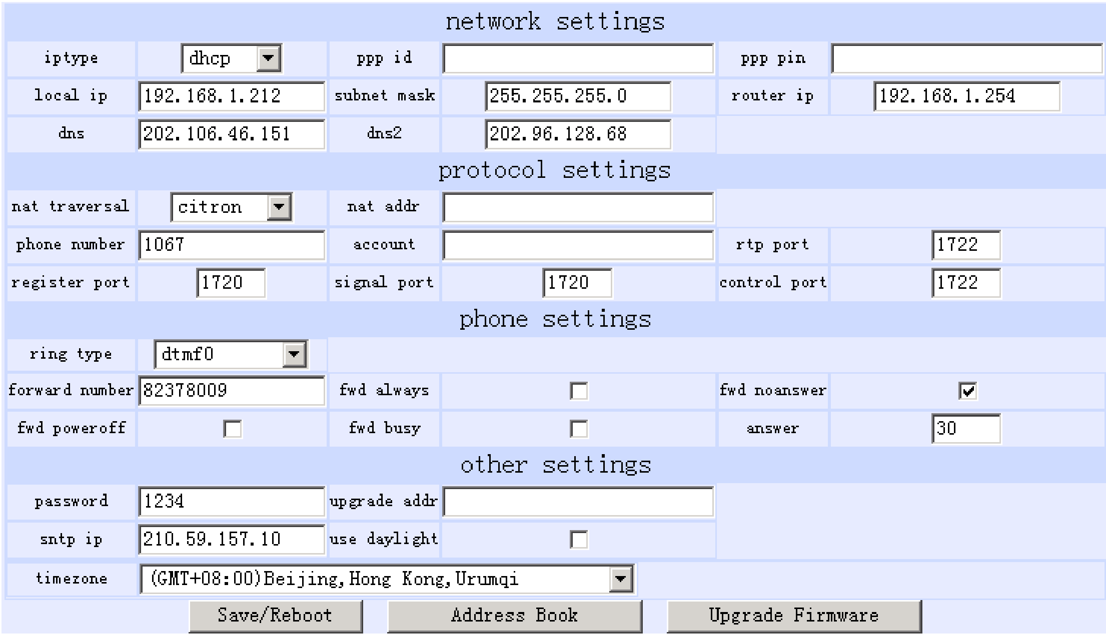


Fig 3.7 Setting Page using ordinary pin with Debug set as 0 [disable]

- **Update:** Click this button to save the configuration and the phone will reboot. Once the phone reboots successfully, the new configuration is effective.

✎**Note** After entering set page, if **Update** button is not clicked within 5 seconds, then when you click it again, the index page asking for pin will pop up again. Then please input the password again to enter the set page and then click **Update** button to confirm the modification.

Phone Book: Click this button to open the speed dial settings page. Please refer to Fig 3.7. In this page, you can set and save the speed dial number by typing the name into the **Name** field and then entering the corresponding number following the name. 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. Then please click **Save/Back** button. In normal state, you can use speed dial to call numbers saved in phone book.

Phone Book

No.	Name	Phone Number	No.	Name	Phone Number
001	Jack	5989426454	002	Allen	192.168.1.56
003			004		
005			006		
007			008		
009			010		

Fig 3.8 Phone Book Illustration

Upgrade Program: Click this button to update the program of IP phone. Before updating, please fill IP address of FTP server into **upgrade addr** field, and then click this button. Then the phone will read the corresponding bin files from the server and then load into the phone.

Update Digitmap: Click this button to update the digitmap of the phone. Before updating, please fill IP address of FTP server into **upgrade addr** field, and then click this button. Then the phone will read the corresponding map files from the server and then load into the phone.

Note Please refer to *PalmTool User Guide* to learn how to write digitmap or just download TXT file from our site. Then please save it as "phone type.map" file, such as A-100.map.

➢ Configured by PalmTool

PalmTool is a tool designed especially to configure and upgrade the PA168 IP phone. On a PC double click  icon to open the PalmTool. The index page of PalmTool will popup.

a) Input the IP address of the phone into **Local IP** field (such as 192.168.1.100), and then click "Phone Settings" button.

From Version1.24, use PalmTool to set the IP phone, please set debug as output or output all firstly, or PalmTool cannot connect IP phone. The parameters of PalmTool is same as the parameters in HTTP, so please refer to HTTP set chapter to learn how to set IP phone.

➢ Telnet Configuration

- On the PC choose **Start>Run**, and then type **telnet 192.168.1.100** into **Run** field in popuping Run dialog. Or input **telnet 192.168.1.100** in the DOS window. Then the following information will be displayed.

A-100 V1.39 settings
Password :

Then please type password. With debug is set as 0[disable], if type ordinary password (default one is 1234), after Retun, you will see：

Password : **
P:\>

If you type super password, then you will see:

Password : ******
P:\>

Above information indicates that IP phone is under setting mode, and then you can set the A-100 IP phone by using the telnet commands.

➢ A-100 Telnet Commands Explanation

A-100 Telnet Commands

Command	**Function**
?	Supply command name and parameters
get	Display basic parameters of the A-100 IP phone
set	Set parameters of the A-100 IP phone
store	Save current settings to designated position
load	Load designated settings to current position
exit	Exit from the setting mode without saving the configuration
write	Exit with saving all configurations and restart A-100
ping	Ping other net parameter
ftp	The phone connects to FTP server and then get the files

Detail description of A-100 Telnet commands

Command ?

Syntax description: No optional parameter

Usage: Type command name and parameters following P:\> . Be used as the keyword to supply keyword and parameters of the relevant commands.

Relevant usage: None

Detailed description:

? List help of all commands

For example:

```
P:\>?
set
get             list settings
store x         store current to xth settings
load x          load xth settings to current
exit
write           save settings
```

Command get

Syntax description: No optional parameter of keywords

Usage: Display basic parameters of the A-100 IP phone

Relevant usage: None

Detailed description:

get Display basic running parameters of the A-100 IP phone. Input ordinary password without debug being set as 0[disable], or input super password with debug set as 0[disable], then following parameters of IP phone will be displayed:

```
A-100 V1.39 settings
Password: ********
P:\>get
iptype 0[static]
ip 192.168.1.100            subnetmask 255.255.255.0   router 192.168.1.254
dns 202.106.196.152         dns2 202.106.196.115       mac 00-09-45-65-a3-e6
service 1[enable]
servicetype 0[common]       serviceaddr 203.93.9.57    serviceid [empty]
nattraversal 1[enable]      nataddr www.showmyip.com   natttl 30
phonenumber 182378009 account [empty]                  pin [empty]
registerport 6800           signalport 6800            controlport 6802
registerttl 60              rtptos 0                   rtpport 6802
jitter size 0
calltype 1[advanced]        localtype 0[phonenumber]   dtmf 0[control string]
dtmfpayload 101
dialplan 2[dialnum]         dialnumber 17911           dddcode [empty]
iddcode [empty]             iddprefix [empty]          dddprefix [empty]
innerline 1[enable]         localprefix 0              nonlocalprefix 0
answer 5                    ringtype 0[dtmf0]          digitmap 0[disable]
fwdnumber [empty]           fwdpoweroff 1[enable]
fwdalways 0[disable]        fwdbusy 0[disable]         fwdnoanswer 0[disable]
audiotype 0[g7231]          audioframes 1              6.3k 1[enable]
vad 1[enable]               agc 0[disable]             aec 1[enable]
handsetin 9                 handsetout 21              speakerout 21
codec1 1[g7231]             codec2 0[g729]             codec3 2[g711u]
codec4 4[gsm]               codec5  5[null]
password 1234               superpassword 19750407     debug 1[output]
upgradeaddr [empty]
sntpip 0.0.0.0              daylight 0[disable]
timezone 55[(GMT+08:00)Beijing,Hong Kong,Urumqi]
```

Input ordinary password with debug set as 0[disable], following information will be seen:

```
A-100 V1.39 settings
Password:****
P:\>get
```

```
iptype 0[static]
ip 192.168.1.100     subnetmask 255.255.255.0      router 192.168.1.254
dns 202.106.196.152        dns2 202.106.196.115
nattraversal 1[enable]    nataddr                        rtpport 5144
registerport 5142          signalport 5142             controlport 5144
account [empty]            pin [empty]
phonenumber 182378009 fwdpoweroff 0[disable]
fwdalways 0[disable]       fwdbusy 0[disable]          fwdnoanswer 0[disable]
ringtype 0[dtmf0]          answer 5
password 1234
upgradeaddr [empty]
sntpip 0.0.0.0              daylight 0[disable]
timezone 55[(GMT+08:00)Beijing,Hong Kong,Urumqi]
```

Command set

Syntax description: set keywords value

Usage: Used to configure password and other running parameters of A-100 IP phone.

Detailed description:

set iptype X

Set how IP phone gets relevant network parameters. X ranged from 0 through 3: 0: authorize users set IP address, subnet mask and router IP address of IP phone manually; 1: use 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. A-100 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; 2: use PPPoE mode. 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; 3: use modem mode. Those who use IP phone with modem, please set the value as 3.

set pppid XXX

With **iptype** set as **2**, use this command to set ADSL ID; with **iptype** set as **3**, use this command to set Modem ID.

set ppppin XXX

With **iptype** set as **2**, use this command to set ADSL pin; with **iptype** set as **3**, use this command to set Modem pin.

set ip XXX.XXX.XXX.XXX

With **iptype** set as **0**, use this command to set IP address of A-100 IP phone.

set subnetmask XXX.XXX.XXX.XXX

With **iptype** set as **0**, use this command to set subnet mask of A-100 IP phone.

set router XXX.XXX.XXX.XXX

With **iptype** set as **0**, use this command to set router IP of network with A-100 IP phone.

set dns XXX.XXX.XXX.XXX

With **iptype** set as **0**, use this command to set IP address of DNS server.

set dns2 XXX.XXX.XXX.XXX

With **iptype** set as **0**, use this command to set IP of backup DNS server.

set mac XX-XX-XX-XX-XX-XX

Set MAC address of the A-100 IP phone. Parameter xx-xx-xx-xx-xx-xx must be an HEX number.

set service X

Set register the SIP proxy server or not. X ranged from 0 through 1. 0: do not register; 1: register.

set service type X

Enable/disable the repaid and service system .choose the repaid server provider. Parameter x ranged from 0 through 29:

0:common: disable repaid card; 4: use Utstarcom system; 12: use ZTE system; 23: use sipphone service;24: use ipphonex service; 25: use fwd(freeworlddialup) service; 26: use net2phone SIP service; 27: use stanaphone service; 29: use iTalkBB system

set serviceaddr XXXX

Set IP address or domain name of SIP Proxy Server.If the system has an Outbound Proxy，set IP address or domain name of Outbound Proxy.

set serviceid XXXXX

If the system has an Outbound Proxy，Set the domain name of SIP proxy server such as xxxx.

set nattraversal X

X ranged from 0 through 7: 0: do not use NAT traversal. 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; 1: Use NAT traversal. When the login system does not support IP phone working behind the LAN, With this item selected, please make port mapping on NAT device; 4: stun.

set nataddr XXXXX

When "**nattraversal**" is set to "**1**";set IP address of NAT device wan port or URI of free assistant service (Such as www.showmyip.com etc.) in Internet.

When "**nattraversal**" is set to "**4**", set IP address or URI of the stun server, in the format as "domain name/IP address : service port". The default service port for stun is 3478.

✎**NOTE** The free service list of Internet：www.ip-calculator.com; www.ipchicken.com; www.ipchicken.com;www.showmyip.com;www.whatismyip.com; www.myipaddress.com; www.whatismyipaddress.com; ip.sbbs.net; www.whatismyipaddress.net;checkip.dyndns.org

set natttl XX

Set NAT TTL XX is an integer between 10 and 65535 sec. default value is 20 sec.

set phonenumber XXXXXXXX

Set a local ID of A-100 IP phone. Value xxxxx must be an Arabic numeral and no longer than 16 characters.

set account XXXXXX

Set the account; Value xxxxxx must be an Arabic numeral and no longer than 32 characters.

set pin XXXXXXXXXX

Set the account; Value xxxxxx must be an Arabic numeral and no longer than 32 characters.

set registerport XXXX

Set register port. Value XXXX default is 5060.

set registerttl X

Set register TTL. X is range from 10 through 65535 Sec. default value is 60 Sec.

set rtptos X

Set TOS segment of IP head package in RTP digital follow.

set rtpport XXXX

RTP port is the port transferring and receiving voice flow using UDP protocol. XXXX is range from 1024 through 65535.

set jittersize X

Set buffer size of RTP package. X is range from 0-32.

set calltype X

Set call type of the phone. X is ranged from 0 through 1: 0: when

SIP system not support “outbound proxy”; 1:when SIP system not support “outbound proxy”.

set dtmf X

Set DTMF relay type. X is ranged form 0 through 3: 0:control string；1:inband audio；3:rfc 2833.

set dtmf payload X

When **dtmf X** select 3(**rfc 2833).**This parameter can be used indicating type of RTP payload type. The value can be use integer 96-101.

set dialplan X

Enable/disable dial plan and dial number. Parameter X ranged from 0 through 3: 0: disable dial plan; 1: enable dial plan; 2: use dial number; 3: use special prefix service; 4: use hotline function.

set dialnumber XX

When **set dialplan** value set as 2, please use this command to set **dial number.** When **set dialplan** value set as 4, please use this command to set **hotline number.**

set dddcode XX

Set the area code when set **dialplan** value set as **1** or **2**. For example, the area code of Beijing is 10; the area code of Shanghai is 21, and the area code of Chengdu is 28, etc. Parameter xxx must be an Arabic numeral and no longer than 3 characters.

set iddcode XXX

Set the country code when set **dialplan** value set as **1** or **2**. For example, the country code of China is 86; the country code of USA is 1, etc. Parameter xxxx must be an Arabic numeral and no longer than 4 characters.

set iddprefix XX

Set IDD service prefix number when set **dialplan** value set as **1** or **2**. For example, IDD service prefix number of china is 00; IDD service prefix number of USA is 1, etc. Parameter xxx must be an Arabic numeral and no longer than 3 characters.

set dddprefix XX

Set DDD service prefix number when set **dialplan** value set as **1** or **2**. For example, DDD service prefix number of china is 0; DDD service prefix number of USA is 1, etc. Parameter xxx must be an Arabic numeral and no longer than 3 characters.

set innerline X

Set use innerline call or not. X ranged from 0 through 1: 0: disable; 1: enable innerline call.

set localpreifx X

With **innerline** set as **1[enable]** , please set the number switching to local call, such as 9.

set nonlocalprefix X

With **innerline** set as **1[enable]** , please set the number switching to long-distance call, such as 9.

set answer X

Set the ring seconds before the phone answers the call auto or forward the calls. X is ranged from 0 through 60.

set ringtype X

Set types of ring. X is ranged from 0 tohrough 12: 0-9: ring as ordinary rings in different frequency; 10: do not ring; 11: ring as music shipped from factory; 12: ring as music saved by user

set digitmap X

Set whether to use digitmap. X ranged from 0 to 1: 0: do not use digitmap; 1: use digitmap.

set fwdnumber XXXXXXX

Set receiving forwarded calls phone number. XXXX must be an Arabic numeral and no longer than 16 characters

set fwdpoweroff X

Enable/disable forward calls if power off. X is ranged from 0 through 1. 0: do not forward calls if power off; 1: forward call if power off.

set fwdalways X

Enable/disable forward all calls. X is ranged from 0 through 1. 0: do not forward all calls; 1: forward all calls.

set fwdbusy X

Enable/disable forward calls if busy. X is ranged from 0 through 1.

0: do not forward calls if busy; 1: forward call if busy.

set fwdnoanswer X

Enable/disable forward calls without replying. X is ranged from 0 through 1. 0: do not forward calls without replying; 1: forward call without replying.

set audioframes X

Set audio frames in RTP package. X is Arabic numerals between 0 and 7.

set 6.3k X

With G.7231, set A-100 IP phone to use 6.3K rate or not. X is ranged from 0 through 1: 0: use 6.3K rate; 1: use 5.3K rate.

set vad X

Enable/disable VAD. X is ranged from 0 through 1: 0: disable VAD; 1: enable VAD.

set agc X

Enable/disable AGC. X is ranged from 0 through 1: 0: disable AGC; 1: enable AGC.

set aec X

Enable/disable AEC. X is ranged from 0 through 1: 0: disable AEC; 1: disable AEC.

set handsetin X

Set initial volume of handset. X is ranged from 0 through 15.

set speakerin X

Set initial volume of microphone of the base. X is ranged from 0 through 15.

set handsetout X

Set initial volume of handout. X is ranged from 0 through 31.

set codec1 X

Set the priority 1of the audio compression algorithm. X is range from 0 through 4: 0: g729; 1:g7231; 2: g711u; 3: g711a; 4: gsm.

set codec2 X

Set the priority 2 of the audio compression algorithm. X is range from 0 through 4: 0: g729; 1:g7231; 2: g711u; 3: g711a; 4: gsm.

set codec3 X

Set the priority 3 of the audio compression algorithm. X is range from 0 through 4: 0: g729; 1:g7231; 2: g711u; 3: g711a; 4: gsm.

set codec4 X

Set the priority 4of the audio compression algorithm. X is range from 0 through 4: 0: g729; 1:g7231; 2: g711u; 3: g711a; 4: gsm.

set codec5 X

Set the priority 5 of the audio compression algorithm. X is range from 0 through 4: 0: g729; 1:g7231; 2: g711u; 3: g711a; 4: gsm.

set password XXXX

Set password of the A-100 IP phone. XXX must be ASCII characters .

set superpassword XXXX

Set super password of the A-100 IP phone. XXX must be ASCII characters.

set debug X

Set open debugging message output grade for special tool. X is ranged from 0 through 5: 0: close debugging output; 1: output the operation information to the window; 2: output all the bug information and data in test window; 3: save the bug information into SDRAM; 4: disable checks the mark.

set upgradeaddr XXX.XXX.XXX.XXX

Set IP address or domain name of FTP server supplying upgraded program of A-100 IP phone.

set ntsip XXX.XXX.XXX.XXX

Set IP address of time server.

set daylight X

Set use daylight or not. X ranged from 0 through 1: 0: do not use daylight; 1: use daylight.

set timezone XX

Set time zone.

Command store

Syntax description: no keyword. Parameter ranged from 0 through 4.

Usage: Save the current settings to the designated position.

Relevant Usage: store 1

Command load

Syntax description: no keyword. Parameter ranged from 0 through 4.

Usage: Load the designated settings to the current position.

Relevant Usage: load 1

Command exit

Syntax description: no keyword and parameter

Usage: Exit from Telnet command window without saving the configuration.

Relevant usage: None

Command write

Syntax description: No keyword and parameter

Usage: Save the configuration and restart the A-100 IP phone.

Command ping

Syntax description: **ping IP address**

Usage: ping IP address of other NAT device

Relevant usage: In telnet window, input ping xx.xxx.xx.xx（an IP address） and return, then the result will be displayed. If the address is effective, "ping OK" will be seen; if the address is ineffective, nothing will be seen. Fox example:

```
P:\>ping 203.93.9.57
P:\>
ping OK
P:\>ping 27.56.120.56
P:\>
```

Note Usually, the echo time of ping command is no more than 1 second. So if the result is not displayed in 5 seconds, ping command is fail.

ftp command

Syntax description: **ftp value**

Usage: the system connects to the FTP server auto to get the corresponding file and deal with it.

Relevant usage: ftp X

X ranged from 0 through 2:

X-0: Connect to FTP Server to get the file of updating program and save it to the SDRAM of the phone. Then the file can be read by PalmTool. This operation aims at testing.

X-1: Connect to FTP Server to get the file of updating program and update program Flash. This operation aims at updating program.

X-2: Connect to FTP Server to get the file of updating dial rules and update program Flash. This operation aims at updating dial rule.

✎**Note** When you use ftp 0 and ftp 1 commands, if the file get from FTP server is too large or the net speed is too slow, then the process will not be seen in telnet window. Please be patient. Using ftp command in telnet to get file spends almost same minutes as getting file using phone. So if nothing is displayed after too long time, it means that ftp is fail.

✎**Note** All the Telnet commands of A-100 IP phone should be written in low case and the password is case sensitive.

➢ Upgrade A-100 IP phone

➢ Set FTP server

FTP server can be supplied by the server provider as well as setup by the users in LAN. Please set the IP address of FTP server.

➢ Prepare Updated program

You can ask the server provider for the latest version of program

➢ Operation

If you have got the IP address of the FTP server from ISP, please do as follows:

a) Use keypad to enter setting mode
b) Use keypad to input the IP address of FTP server
c) Press 8 Local IP , then No19 light will blink twice a second. Once the ph ccessfully, the new program is effective.

✎ **Note** Please do not change the name of the upgraded program, or the operation will be fail.

➢ Usage of the phone

➢ Receiving calls:

A-100 IP phone can receive incoming calls from other A-100 IP phone and devices that support the SIP 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 Handset to hand free: While receiving call with handset, press Speake on the keypad and then put down the handset. When the call is over, press Speake gain.

③ Hand free: Press Speake o speak to the other party. When the call is over, press Speake ain.

④ Hand free to handset: While receiving the call with Speake pressed , pick up the handset to continue the call. When the call is over, put back the handset.

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

➢ Place a call

- **Call another PA168 IP phone under the same Gatekeeper:**

1. Handset: Pick up the handset and listen for the Internet dial tone. Then dial the phone number you wish to call and press Call or # 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.
2. Hand free: Press Speaker and listen for the Internet dial tone. Then input the phone number you wish to call and press Call or # 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 Speaker again. The dialed number has been saved into the buffer.
3. Blind dialing: Use the keypad to enter the phone number you wish to call and then press or 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

Speaker

begin speaking. When the call is over, Press again. The dialed number has been saved into the buffer.

- **Place a call without login the Gatekeeper**

If A-100 IP phone does not login the Gatekeeper, you can place a call by lifting the handset or pressing and Speaker tting the IP address of the other party, and then pressing or Call #

- **Place a call through Gateway**

If A-100 IP phone does not login the Gatekeeper, you can place a call through Gateway directly by lifting the handset or pressing and then inputting the IP address of the other party, and then pressing or Speaker .

✎**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.

➢ **View Records**

- View missed calls

Click Record , then you will hear the record of missed call. Click Vol+ to turn the numbers orderly; click Vol - to turn the numbers reservedly. If there is no record, you will hear nothing.

- View received call

Click Play , then you will hear the record of the received call. Click

Vol+ to turn the number orderly; click Vol- to turn the numbers reservedly. If there is no record, you will hear nothing.

- View dialed number

Click Play , then you will hear the record of the dialed call. Click Vol + to turn the number orderly; click Vol - to turn the numbers reservedly. If there is no record, you will hear nothing.

- When you hear the number you want to dial, please press # to place a call directly.

✎ **Note** A-100 IP phone supports saving 80 entries unanswered call, dialed call and received call ranged from 1-80 at best. When the entries arrives 80, the latest record will cover the first one. The record will be lost when the phone restarts or turned on.

Appendix Table： A-100 IP phone digital-character key map:

Keys	Press Once	Press Twice	Press Thrice	Press quartic	Press quintic
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	:/@	;/-	\ /&

#	Case change

Reserves the right to make changes in technical and product specification without prior notice.

A-100 SIP Phone User Manual (V1.39)