SIP Phone 302

USER MANUAL

SIP-Phone 302 Administration Guide

SIP-Phone 302 Administration Guide

Chapter 1 Overview of the SIP-Phone 302

SIP-Phone 302 is a full-featured IP-based telephone set for home telephony via Ethernet base communication. It supports SIP RFC3261 protocol. Two 10/100BaseT embedded switch/hub RJ-45 ports allow connect to office LAN and PC on your table. It is easily interface with ADSL/Cable Modem that is provided by ITSP, ISP or Carrier company to provide VoIP services to residential and SOHO application.

SIP-Phone 302 provides two 10/100BaseT switch/hub RJ-45 ports. The internal two-port Ethernet switch allows for a direct connection to a 10/100BaseT Ethernet network via an RJ-45 interface with single LAN connectivity for both the phone and a co-located PC.

Note:
Marked with (**): May be not available yet, please contact with your distributor.

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1. Hardware Overview

1.Front View and Keypad function

- ♦ **System Indication LED**: When having not checked voice mail , system indication LED will be flashing until user checked voice mail.
- ♦ **MENU**: Press to enter LCD Menu when in standby mode; if already in LCD Menu, press this button can return to standby mode
- ♦ **MUTE**:
    1. Mute the voice of Microphone and let others can't hear from user in communication.
    2. Change input mode to be digit or character mode: When configuration in LCD menu

can change input mode to be input digit only or input character.

- ◆ **▲▼**: Move to up/down; left/right; increase/decrease value.
- ◆ **OK**: Press OK to confirm the modification.
- ◆ **Flash**:
  1. Transfer a call. User A can press FLASH button when in communication with user B, then input phone number can make call to User C, after talk with C, A can hang up, User B and User C can communicate.
  2. Back to upper level of menu: when in LCD Menu, press FLASH button can jump to upper level of menu.
- ◆ **REDIAL** / **HOLD**:
  1. Redial the last outgoing call or hold one call in communication.
  2. Upper-case/Lower-case character: change input character mode to be upper case or lower-case.
- ◆ **SPEAKER**: Speaking without picking up handset.
- ◆ **5 Graphic Memory key**: User presses these keys to do speed dial according to phone book data 1-5 (please refer to LCD configuration-Phone Book, Configuring the SIP-Phone 302 through Telnet command lines - [pbook] command, or Web Configuration-Phone Book chapter).
- ◆ **Number 1 –10, * and #**: The function is as the same as the general phone set. **Corresponding list of keypad and symbol:**

1	"1"
2	"a" ; " b" ; "c" ; "2"
3	"d" ; "e" ; "f" ; "3"
4	"g" ; "h" ; "i" ; "4"
5	"j" ; "k" ; "l" ; "5"
6	"m" ; "n" ; "o" ; "6"
7	"p" ; "q" ; "r" ; "s" ; "7"
8	"t" ; "u" ; "v" ; "8"
9	"w" ; "x" ; "y" ; "z" ; "9"
*	"." ; "@" ; "_" ; "-" ; "," ; "!" ; "?" ; "" ; "+" ; "$" ; "*"
0	"Space" ; "0"
#	"#"

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2.Back View

- ♦ **DC 5V**: DC 5V power input outlet
- ♦ **LAN**: 10/100 Base-T RJ-45 connector, connected directly to the **Hub** through the **straight** CAT-5 cable.
- ♦ **PC**: 10/100 Base-T RJ-45 connector, connected directly to the **PC** through the **straight** CAT-5 cable

3.Specification

1. Ethernet Port：

   Ethernet port is for connecting SIP-Phone to network, transmit rate supports 10/100 Base-T.
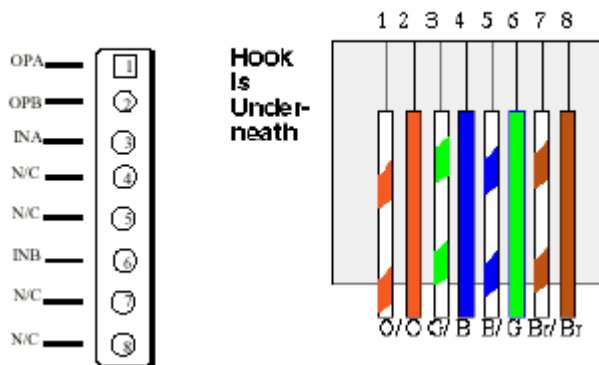
Ethernet connector（LAN）

2. Dimension: 200mm(W) x 79mm(H) x 210mm(D)
3. Weight: 830g

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2. Software Features and Specification

Application:
- ISP/ITSP (Internet Telephony Service Provider)
- IP-PBX with office telephony services
- Multi-nation enterprise communication
- SOHO Telephony

Calling Features
- Call Hold
- Call Transfer
- Blind Transfer
- Call Waiting
- Call Forward
- 5 configurable speed dials

Network Supported
- Fixed IP
- Dynamic Host Configuration Protocol (DHCP)
- PPPoE connection (When PPPoE disconnect, SIP-Phone 302 can automatically re-connect)
- Behind NAT IP Sharing Device
- Support QOS by setting DSCP (Differentiated Service Code Point) parameters of VoIP packet

Audio Features
- G.711 a/μ-Law, G.723.1, G.729, G.729a
- VAD, CNG
- Programmable Dynamic Jitter Buffer
- Bad Frame Interpolation
- Gain/Attenuation Settings

Provisioning and Configuration
- SIP (RFC3261) compliance
- LCD configuration password protection
- Provide Proxy Mode or Peer-to-Peer Mode (Non Proxy Server needed) selection
- Ring tone, Speaker and Handset volume adjustable
- Support DNS server inquiry

Management Features:

- Software Upgrade: TFTP/FTP download
- Three easy ways for system configuration
  - LCD Front Panel
  - Web Browser
  - TELNET

Environmental

- Operating and storage Humidity: 10 to 95 % (Non-condensing)
- Operational Temperature: 0 to +40 °C
- Storage Temperature: -10 to 60℃

Certification

- CE

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Chapter 2 Configuring the SIP-Phone 302 through LCD Phone menu

> **Note:**
> 1. After any configuration has changed for the SIP-Phone 302, user has to do **Reboot** in the selection 8 "Reboot".
> 2. **We suggest user to set IP address via LCD menu 6→2.3.4.5 first, then go to chapter 3 to do other configurations via web browser. If user need to do more detail or advanced configurations, please refer to chapter 4 and use Telnet command lines.**
> 3. User can also try to enter web configuration via default IP address of SIP-Phone 302: **10.1.1.3**

1. Initialize SIP-Phone 302

1. After SIP-Phone 302 finish initializing, it will get into standby mode:
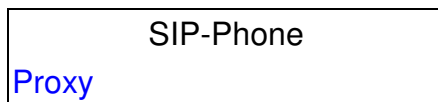
> SIP-Phone
> Proxy

The main LCD screen would be shown as similar as above. "Proxy" means the SIP-Phone 302 is in Proxy Mode.

> **Note:**
> If SIP-Phone didn't register to Proxy server, when SIP-Phone 302 been off-hook, user will hear busy tone instead of dial tone.

2. When SIP-Phone 302 is under peer-to-peer mode, on LCD will show "P2P" instead of "Proxy".

> SIP-Phone
> P2P
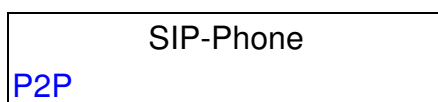
3. Press **MENU** to enter configuration mode then press **OK** to enter sub menus; press

FLASH can jump out current menu to previous level.

1. Voice Mail
2. Call List
3. Forward
4. Phone Book
5. Ringer
6. Network
7. Advanced Set (can be protected by password)
8. Reboot

2. LCD Menu Configuration
User can set the following configurations via LCD keypad.

> *Note:*
> 1. Press **REDIAL/HOLD** before input data can switch characters to be capital or lowercase.
> 2. Press **MUTE** before input data can switch input mode to be character mode or IP mode; for example, user wants to enter IP address, after pressing **MUTE** can enter digits directly.
> 3. When user is inputting data, press ▲ will clear previous input data.

1. **Voice Mail**

   User can check all voice mail in this Voice Mail Box, but the Server must support this function, too.

   (1) Information: to see all voice mail message.

   (2) MailBox No. : set Voice MailBox Number.

   (3) MailBox Key.: set Voice MailBox key.

   (4) Dail MailBox: choose this item then input OK button, it will call to Voice MailBox.

2. **Call List**

   User can check all call records in this call list menu.

   (1) Missed Calls：to see all missed calls in message box.

   (2) Received：to see all received calls in message box.

   (3) Dialed No.: to see all dialed numbers in message box.

   (4) Exit: return to upper level of LCD Menu

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3. Forward

There are 3 selections in Forward type, user must select under which condition to forward calls.

(1) Busy

When SIP-Phone 302 is in busy status, the incoming call will be forwarded to the assigned phone number.

A. Activate

Enter a forwarded phone number to activate busy forward function.

B. Deactivate

Deactivate Busy Forward function.

C. Exit

Return to upper level of LCD Menu

(2) No Answer

When SIP-Phone 302 has incoming call and hasn't been picked up for around 10 seconds, the incoming call will be forwarded to the assigned phone number.

A. Activate

Enter a forwarded phone number to activate No Answer Forward function.

B. Deactivate

Deactivate No Answer Forward function.

C. Exit

Return to upper level of LCD Menu

(3) Uncondition (Unconditional Forward)

It is included the above two types. Whether the SIP-Phone is in which status, calls will be automatically forwarded to the assigned phone number.

A. Activate

Enter a forwarded phone number to activate Unconditional Forward function.

B. Deactivate

Deactivate Unconditional Forward function.

C. Exit

Return to upper level of LCD Menu

(4) Delete All: Delete all forward activated data.

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(5) Exit: Return to upper level of LCD Menu.

4. Phone Book
1. List
List all records of name, telephone number, and IP address in the phone address book.

2. Edit/Del
Edit or delete a record of name, telephone number, and IP address of the phone address book.

3. New
Add a new record of name, telephone number, and IP address of the phone address book.

4. Exit
Return to upper level of LCD Menu

5. Ringer
1. Volume
User can adjust ring volume by press ▼ or ▲ on the keypad to decrease or increase ringer volume. (Level: 0~12)

2. Style
There are three tone styles for SIP-Phone 302. Move the ">" symbol by press ▼ or ▲ on the keypad to select the tone style preferred, then press OK to confirm it.

3. Exit
Return to upper level of LCD Menu

6. Network
1. Information
All information of network will slide through LCD screen slowly, if user wants to see each item only can press ▼ or ▲ to check detail information.

(1) Mode: Display current network connection mode of SIP-Phone to be Static (Fixed IP), DHCP, or PPPoE.

(2) IP, Mask, Gateway: display current IP information.

2. **Get IP Mode**
   Set network mode of SIP-Phone to be Fix (Fixed IP), DHCP, or PoE (PPPoE).
3. **IP address**
   Set IP address of SIP-Phone 302.
4. **Subnet Mask:**
   Set subnet mask address of SIP-Phone 302.
5. **Default GW**
   Set default gateway address of SIP-Phone 302.
6. **DNS (Domain Name Server)**
   Set IP address of Domain Name Server. Once SIP-Phone can connect to DNS server, user can set URL address for Proxy server or Phone book instead of IP address.
   (1) Primary
      Set Primary DNS server IP address
   (2) Secondary
      Set Secondary DNS server IP address
   (3) Exit
      Return to upper level of LCD Menu
7. **PoE Config (PPPoE Configuration)**
   (1) User Name
      Set PPPoE connection authentication user name.
   (2) Password
      Set PPPoE connection authentication password.
   (3) Reconnect
      Select ON or OFF to enable or disable this function. If user enables this function, after PPPoE disconnected, SIP-Phone will automatically reboot to re-connect, and after reboot, if SIP-Phone still can't connect with server, SIP-Phone will keep trying to connect. On the other hand, if user disables this function, SIP-Phone won't reboot and keep trying to connect.
   (4) Exit
      Return to upper level of LCD Menu
8. **IP-Sharing**
   (1) If SIP-Phone is behind IP sharing or NAT device, and SIP-Phone is under Peer-to-Peer mode or Proxy mode (doesn't support endpoint behind NAT function), on IP sharing must enable "**DMZ**" function or set "**Virtual Server**" to open ports (UDP port: 5060 and 16384, 16385). SIP-Phone must enable this IP sharing function.
   (2) User must enter public IP address of IP sharing.

9. **Exit:** Return to upper level of LCD Menu

7. **Advanced Set (can be protected by password)**
   ● **Password:**
   User must key in password to enter this menu if password has been set, items under this command are all important ones, which can only be configured by advanced users. If password hasn't been set, user can enter this menu directly without entering password.

Note:
   1. If user forget password, please contact with your distributor, we will generate a specific password according to your MAC address of SIP-Phone.
   2. User can also try to configure SIP-Phone via Telnet or Web browser with default IP address: 10.1.1.3. (If IP haven't been changed)

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1. **SIP**
   (1) Mode
   Select SIP connection mode to be peer-to-peer mode or Proxy mode.
   (2) Proxy
   A. **Proxy**
   Set Proxy IP address or Domain Name.
   B. **Outbound**
   Set Outbound Proxy IP address or Domain Name.
   C. **Px port**
   Set Proxy port for SIP-Phone to send messages.
   D. **Expire (in seconds)**
   Set expire time of registration, in the duration of 2/3 expire time, SIP-Phone will re-register to Proxy Server again.
   E. **Exit**
   Return to upper level of LCD Menu

   (3) User Info
   A. **User Name (Mandatory)**
   Set User Name of SIP-Phone to register to Proxy Server. If Proxy server doesn't request specific User name, please enter Line number here.
   B. **Line No.**
   Set Line Number of SIP-Phone to register to Proxy Server.

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C. Password

Set User Password of SIP-Phone to register to Proxy Server. This configuration is not necessary, if Proxy server doesn't request client to set password, user only has to set User Name the same as Line Number.

D. Exit

Return to upper level of LCD Menu

(4) Exit

Return to upper level of LCD Menu

2. SW Update

(1) Method

There are two methods to download new version file, please move the ">" symbol by press ▲or ▼ on the keypad to select TFTP or FTP method, then press OK to confirm it.

(2) Sever

User has to offer one TFTP/FTP server IP Address and set this IP Address via keypad. The IP Address is necessary for upgrading SIP-Phone new application rom file.

(3) Account

User has to input user name for FTP server login .It is necessary for upgrading SIP-Phone new application rom file via FTP method.

(4) Password

User has to input user password for FTP server login .It is necessary for upgrading SIP-Phone new application rom file via FTP method.

(5) File Name

User has to press the file name of new application rom file prepared for upgrading

(6) Version

Show versions of all software and hardware. (**)

(7) Upgrade

Select YES or NO to start upgrade.

(8) Exit

Return to upper level of LCD Menu

Note:

Download via LCD command can only upgrade new *application* rom file.

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3. **Menu Password**

Set entry password of phone LCD menu.

4. **Exit**

Return to upper level of LCD Menu

8. Reboot

Reboot machine. It is necessary and important for user to reboot SIP-Phone 302 after any configurations has been made. SIP-Phone will ask user again before reboot.

Chapter 3 Configuring the SIP-Phone 302 through Web Pages

The HTTPD web management interface provides user an easier way to configure rather than command line method through TELNET.

The configuration function and steps are similar with the way through command line. Please refer to the chapter 4-Configuring the SIP-Phone 302 through Telnet command lines for more detail information. Below is a guide for user to configure via web interface.

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Step 1. Browse the IP Address predefined via Keypad

Please enter IP address (user have to set via LCD menu first) of SIP-Phone in web browser. If user failed to set IP address via LCD menu, the **default IP address of SIP-Phone is 10.1.1.3**, user can also try to connect to SIP-Phone with this default IP via web interface.

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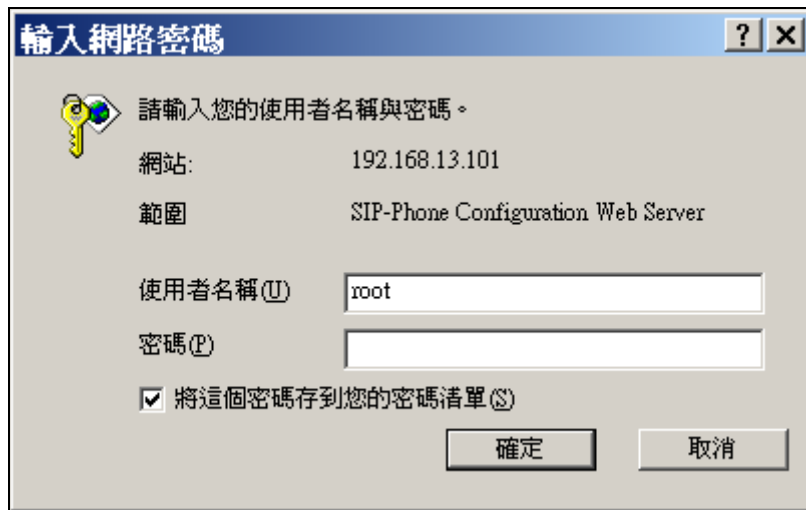
Step 2. Input the login name and password

- Login name: **root** or **administrator**
  User who requests authorization to execute **all** configuration commands needs to login with "root". If a user login with "administrator", commands below are not functional:
  1. **Password:** set password of login: "root" users.
  2. **Flash clean**: clean all current configurations
  3. **Rom configuration:** upgrade boot sector
  4. **Rom configuration:** upgrade whole 2m software file

- Password (The same with TELNET): Null (just press confirm, no need to key in password in default value)

Note: User can set password later in **PASSWORD** via web interface.

Step 3. Enter the web interface main screen

After enter login name and password, user can see web interface main screen as below.
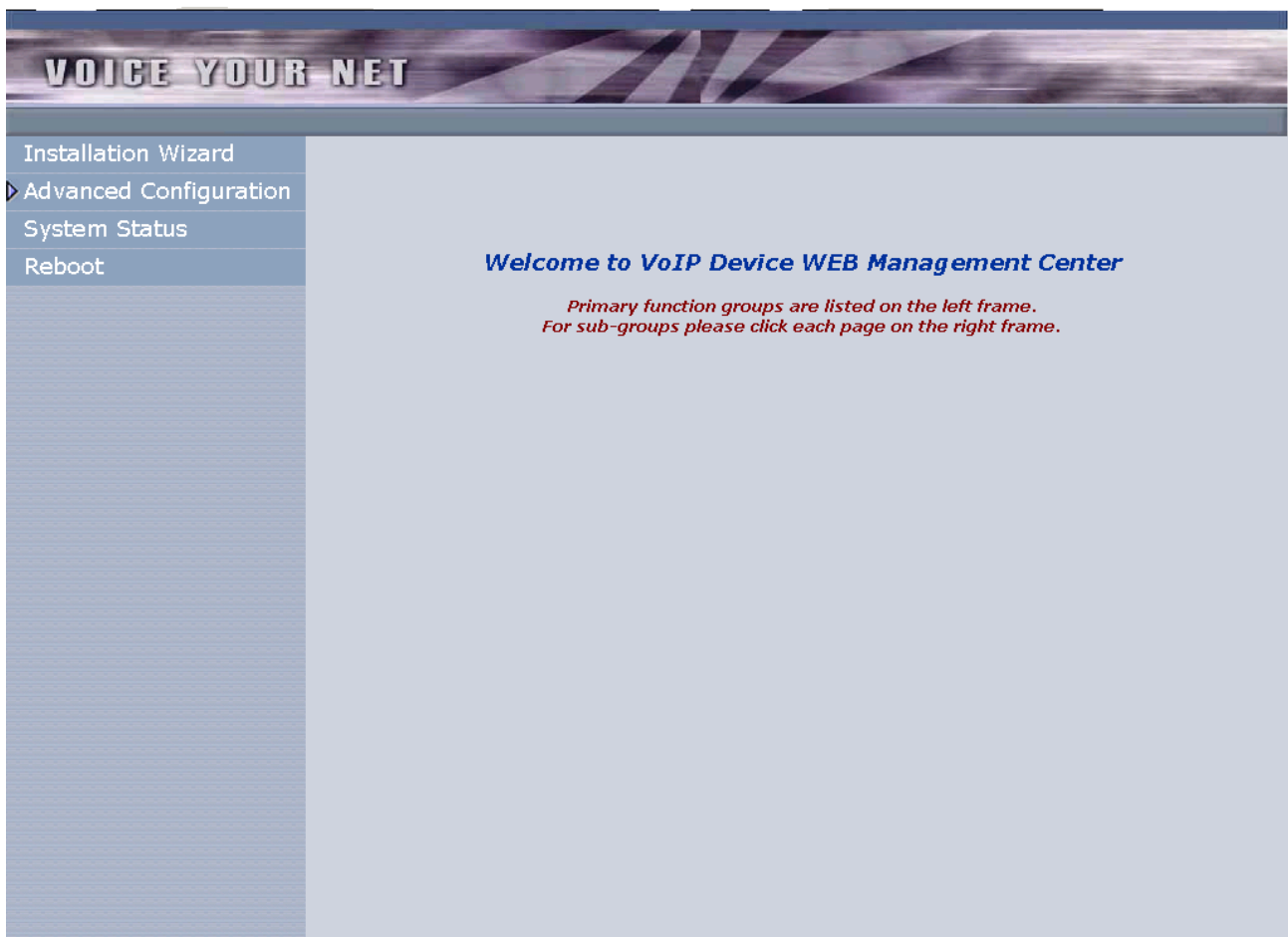
Step 4. Start to configure

After enter web management interface, user can see 4 main items.

1. **Installation Wizard:** User can follow steps in wizard to make first-time initial configuration.
2. **Advanced Configuration:** This menu includes other advanced configuration items. Please press triangle figure to list all items below Advanced Configuration.
3. **System Status:** User can check SIP-Phone current status here.
4. **Reboot:** After make any change, user has to reboot SIP-Phone to apply change.

Button Definition:

1. OK : After change or input any parameter, press this button will save data into SIP-Phone.

2. CANCEL : Press this button will clean data input by user and restore to original data.

    (A) ADD : Add a new data.

    (B) DELETE : Delete a specific data according to index number.

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Most important items are Network Interface, SIP Information, and Phone Book (in Peer-to-Peer mode). Please remember to configure these commands before start to work with SIP-Phone.

> Note:
> After change any settings, please remember to **reboot** (in Reboot System) SIP-Phone so that changes can take effect.

1. Installation Wizard

Installation Wizard includes 3 steps:

SIP-Phone 302 Administration Guide
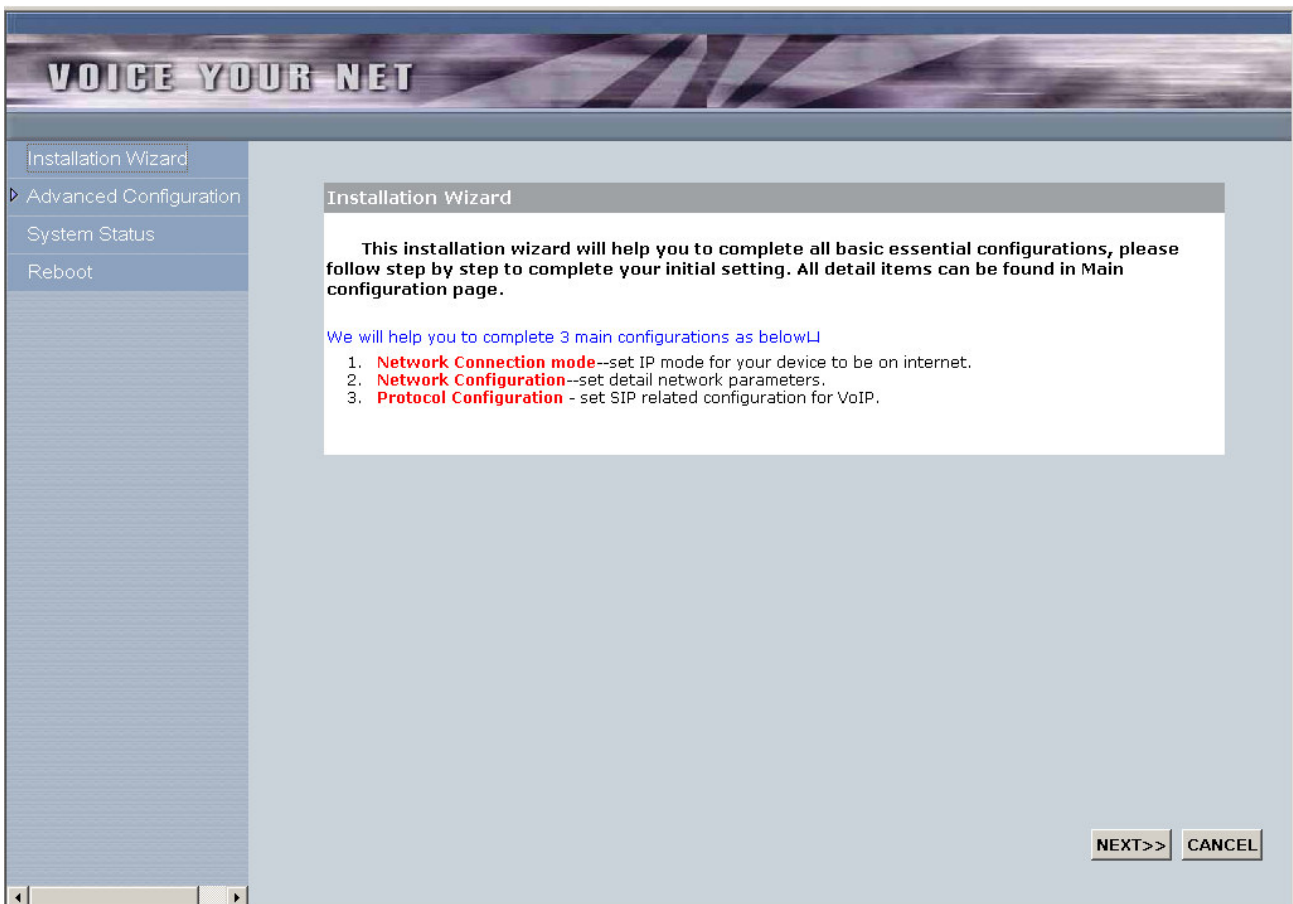
(A) Network Connection Mode: User has to select SIP-Phone network mode as Static IP,
    DHCP or PPPoE.

SIP-Phone 302 Administration Guide
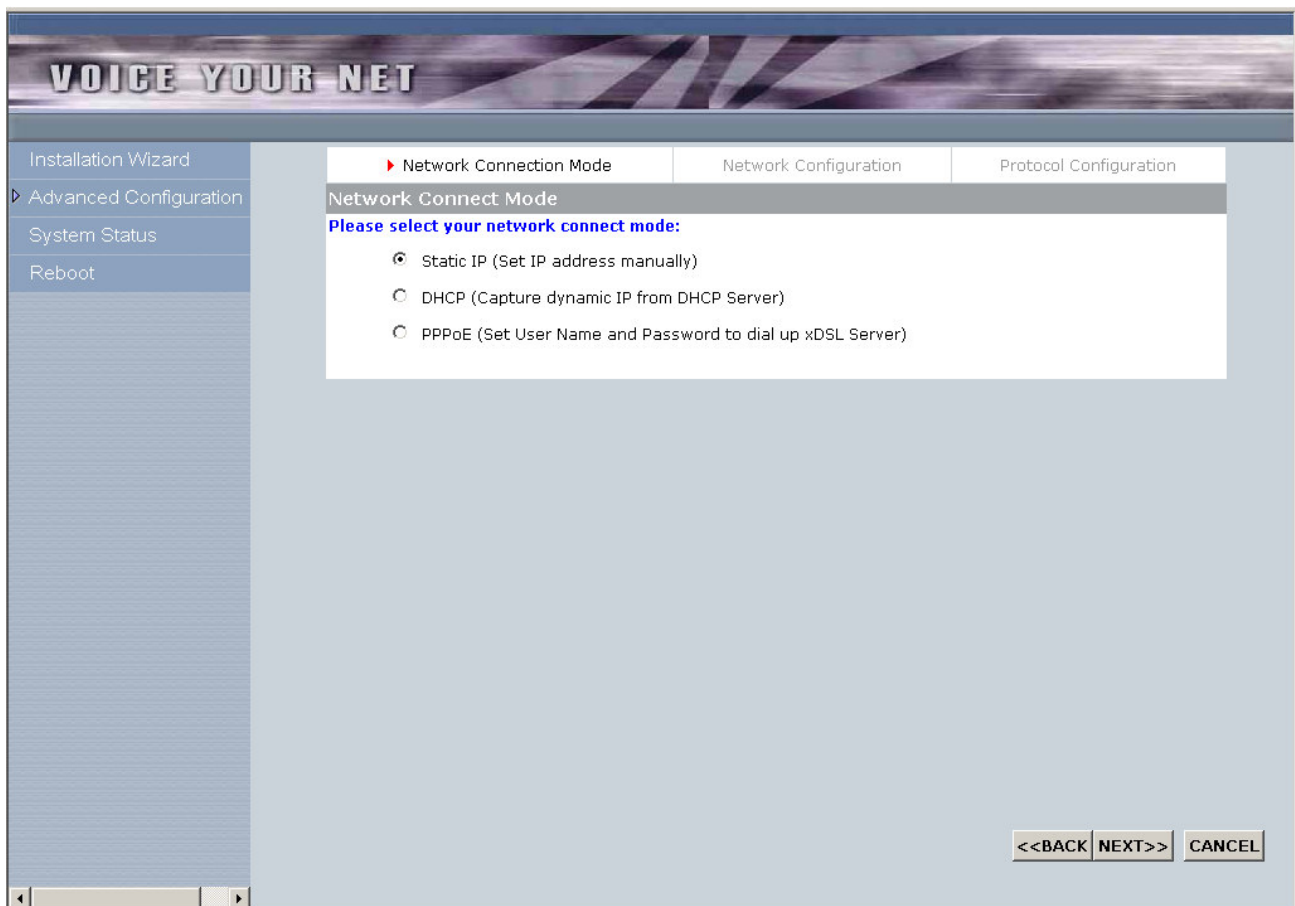
(B) Network Configuration: After selecting network connection mode, user has to input
related network parameters.

(1) Static IP: User has to input IP, subnet mask, default gateway, and DNS server
address.

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(2) PPPoE: User has to input PPPoE connection user name and password.

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(C) Protocol Configuration: After setting network, user has to set SIP related parameters .

- Operation Mode: User has to select SIP-Phone to work as Proxy mode or Peer-to-Peer mode.
- Primary Proxy Address and port: If user select Proxy mode in item A, please input Primary Proxy address and signaling port of Proxy.
- Secondary Proxy Address and port: User can also input secondary Proxy server and port for backup.
- Outbound Proxy Address and port: User can input outbound Proxy and port if necessary.
- Phone Number: Registering Phone number of SIP-Phone.
- Registering Account Name and password: If Proxy server need registration authentication please input user name and password here.

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2. Advanced Configuration

(A) Network Configuration

(1) Network Configuration

Network Connection Mode: User has select network configuration mode first, then configuration page will display related items.

(a) **Static IP**: If user selects Static IP mode, following items will be displayed.

- IP Address: Set IP Address of SIP-Phone.
- Subnet Mask: Set the Subnet Mask of SIP-Phone.
- Default routing gateway: Set Default gateway of SIP-Phone.
- Primary DNS Server: Set Primary Domain Server IP address.
- Secondary DNS Server: Set Secondary Domain Name Server IP address.
- HTTP Port for WEB Management: Set port number for user to configure SIP-Phone via WEB management. Default value is 80.

(b) DHCP: If user selects DHCP mode, following items will be displayed.

- DNS Server Obtained Mode: When SIP-Phone is in DHCP or PPPoE mode, user can determine DNS address is obtained from server or set manually.
- Primary DNS Server: If user determines to set DNS address manually, please set Primary Domain Name Server IP address here.
- Secondary DNS Server: If user determines to set DNS address manually, please set Secondary Domain Name Server IP address here.
- HTTP Port for WEB Management: Set port number for user to configure SIP-Phone via WEB management. Default value is 80.

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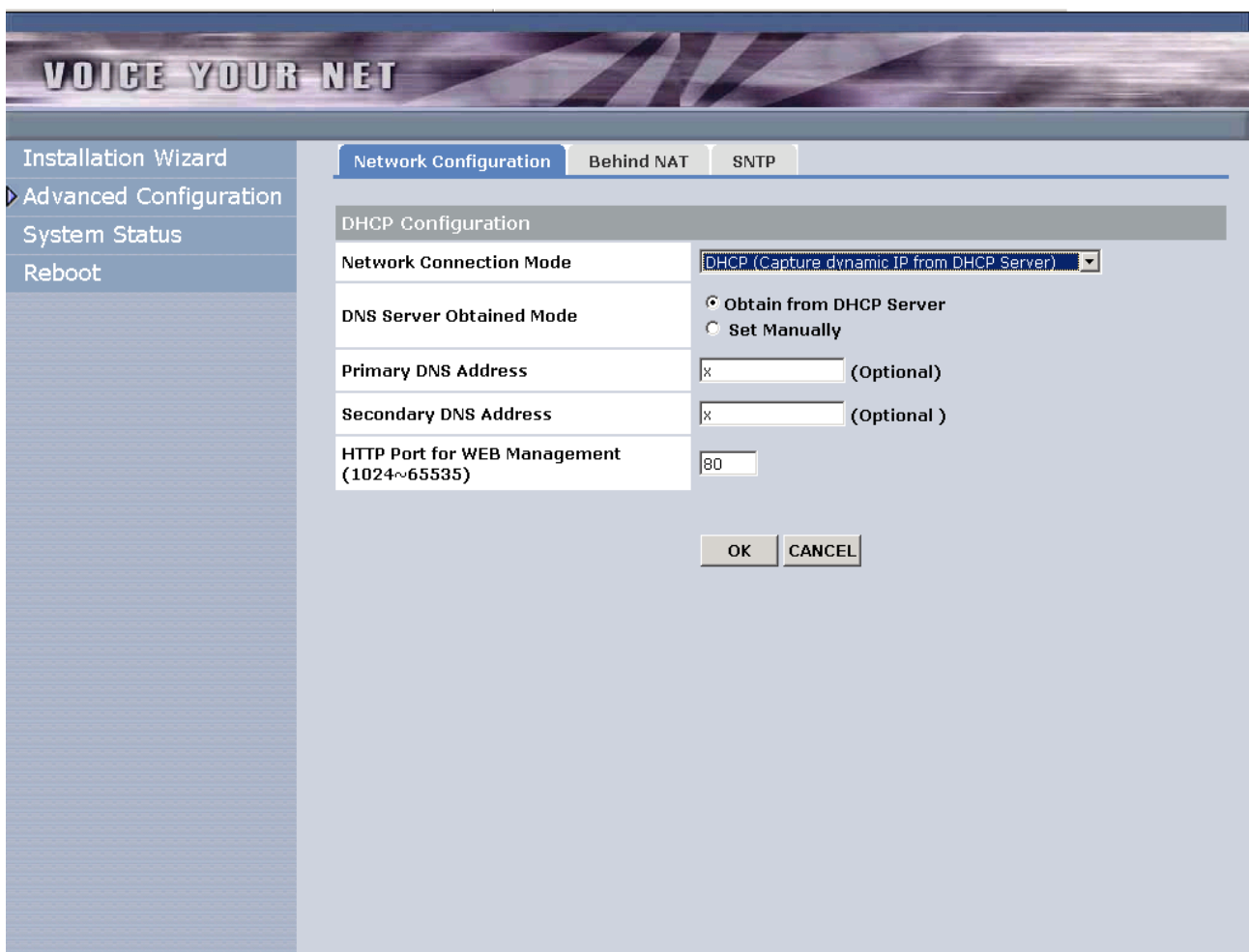
(c) PPPoE: If user selects PPPoE mode, following items will be displayed.

- DNS Server Obtained Mode: When SIP-Phone is in DHCP or PPPoE mode, user can determine DNS address is obtained from server or set manually.

- Primary DNS Server: If user determines to set DNS address manually, please set Primary Domain Name Server IP address here.

- Secondary DNS Server: If user determines to set DNS address manually, please set Secondary Domain Name Server IP address here.

- PPPoE User ID: Set PPPoE authentication User Name.

- PPPoE User Password: Set PPPoE authentication password.

- PPPoE Retry: Enable/Disable auto-retry function after PPPoE disconnection. If user enables this function, after PPPoE being disconnected, SIP-Phone will automatically reboot to re-connect, and after reboot, if SIP-Phone still can't get contact with server, SIP-Phone will keep trying to connect. After re-connected, SIP-Phone will also restart system. On the other hand, if user disables this function, SIP-Phone won't reboot and keep trying to connect.

- Send PPPoE Echo Request: Enable or Disable PPPoE Echo function. If user enabled this feature, SIP Phone will send out echo request to check PPPoE connection status. Please notice that if user disables this function, SIP Phone cannot detect if PPPoE connection is still alive or not.

- HTTP Port for WEB Management: Set port number for user to configure SIP-Phone via WEB management. Default value is 80.

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VOICE YOUR NET

Installation Wizard
Advanced Configuration
System Status
Reboot

| Network Configuration | Behind NAT | SNTP |

PPPoE Configuration

Network Connection Mode	PPPoE (Set User Name and Password to dial up) ▼
DNS Server Obtained Mode	⦿ Obtain from PPPoE Server ○ Set Manually
Primary DNS Address	0.0.0.0 (Optional)
Secondary DNS Address	x (Optional)
PPPoE User ID	84460791@hinet.net
PPPoE User Password	********
PPPoE Retry	⦿ Enable ○ Disable
Send PPPoE Echo Request	⦿ Enable ○ Disable
HTTP Port for WEB Management (1024~65535)	80

OK   CANCEL

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(2) Behind NAT

- Behind IP Sharing: Select if enable SIP-Phone behind IP Sharing router function.

- IP Sharing Public IP Address: Set Public IP Address of IP Sharing router for SIP-Phone to work behind IP sharing.

- STUN Server Address: Set STUN server address.

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(3) SNTP
- SNTP Mode: Enable / Disable the Simple Network Time Protocol function
- SNTP Server Address: Set SNTP Server Address.
  When SNTP server is available, enable SIP-Phone SNTP function to point to SNTP server IP address so that SIP-Phone can get correct current time.
- Time Zone-GMT: Set time zone for SNTP Server time.
  User can set different time zone according to the location of SIP-Phone. For example, in Taiwan the time zone should be set as 8,which means GMT+8.

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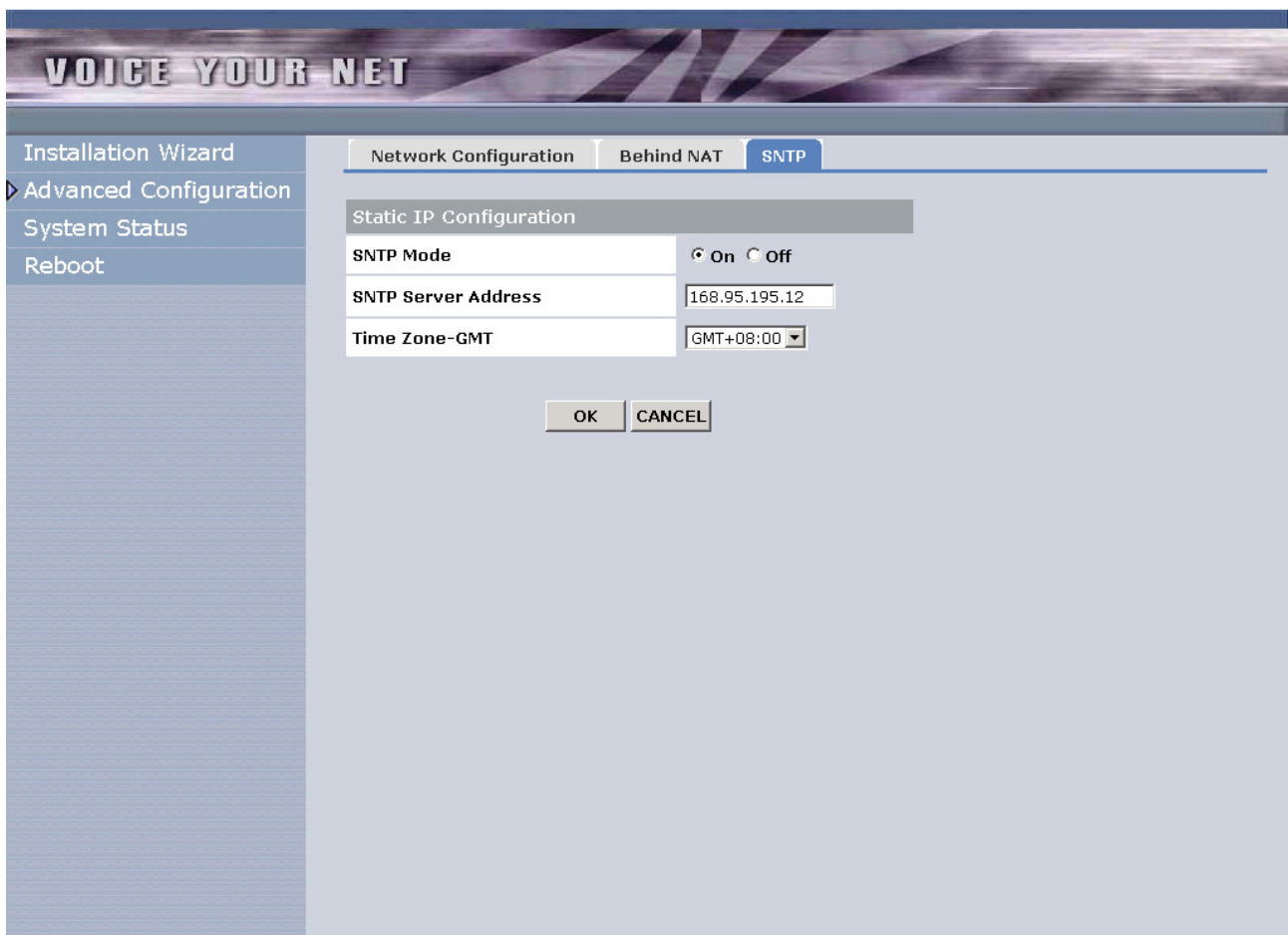
(B) SIP Configuration

(1) SIP Main Configuration-Settings.

- Operation Mode: User has to select SIP-Phone to work as Proxy mode or Peer-to-Peer mode.
- Primary Proxy Address and port: If user select Proxy mode in item A, please input Primary Proxy address and signaling port of Proxy.
- Secondary Proxy Address and port: User can also input secondary Proxy server and port for backup.
- Outbound Proxy Address and port: User can input outbound Proxy and port if necessary.
- Phone Number: Registering Phone number of SIP-Phone.
- Registering Account Name and password: If Proxy server need registration authentication please input user name and password here. Registering Account Name and password: If Proxy server need registration authentication please input user name and password here.

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(2) SIP Advanced Configuration
- Prefix String: set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, SIP-Phone will sent out messages as Account Name @"IP address of Proxy", if user set prefix as abc, SIP-Phone will set out as abc123@"IP address of Proxy". This function is for special proxy server.
- Expire Time: set expire time of registration. SIP-Phone will keep re-registering to proxy server before expire timed out
- Phone Book Search: enable/disable phone book search function. If user enabled this function, SIP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, SIP-Phone will make call to related IP address.
- Display Name: set SIP-Phone display name for caller ID information.
- Local SIP Port: set SIP UDP port.
- RTP Port: set RTP port for sending voice data.

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(C) System Configuration
(1) Feature Configuration

- **Inter Digit Time:** Set the DTMF inter digit time (second) To set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, SIP-Phone will dial out all number pressed.

- **Keypad DTMF Type:** set DTMF type. User can select DTMF type SIP-Phone transmits.

- **End of Dial Key:** select end of dialing key, e.g. set end of dial key as * button, after finished pressing dialing number then press * will dial out.

- **Blind Transfer key:** Set Blind Transfer key (default: *65), when you want to use the function, you can input *65 ahead other callee Number C then your LP302 will on-hook automatic and transfer to C.

- **User phone:** Enable the function. It will send parameter user=phone to server.

- **Call Waiting:** When you are communicating then have incoming call, LP302 will bring Insert Tone. (default : on)

- **Call Transfer:** You can enable or disable the function. (default: on)

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(2) Tos Configuration

Set Signal or RTP Packets DSCP value: The recommended DiffServ Codepoint is defined in RFC2597 to classify the traffic into different service classes.

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(D) Number Configuration

(1) Phone Book

- Add Data: User can specify 50 sets of phone book via web interface. Please input index, Name, E.164 number, IP Address, port of the destination device, drop prefix, and insert number.
    1. name: Specify name for one pbook data
    2. IP : Set IP address of callee.
    3. e.164: set phone number of callee.
    4. port: user can set callee listening port for SIP signaling, default value is 5060.
- Delete Date: User can delete any configured phone book data by index.

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(2) Hotline

If user set SIP-Phone as hotline mode, once SIP-phone is off-hook, it will automatically dials out IP address (Peer-to-Peer Mode) or phone number (Proxy Mode) set in hotline table.

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(3) Digit Manipulation

- add: Add a rule to drop or insert prefix digits of incoming call.
- prefix: Set which prefix number to implement digit manipulation rule.
- drop: Enable or disable drop function. If this function is enabled, Sip-Phone will drop prefix number on incoming call.
- insert: Set which digit to insert.
- delete: Delete a digit manipulation rule by index.

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(E) Media Configuration

        (1) Codec

- Codec Priority: set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need. For example, user can only set first priority as G.723.1, and set the others as x, that means only G.723.1 is available.

- G.723/G.729B Silence Suppression: Enable / Disable sound compression and comfort noise generation. It is only for codec G.723.1 and G.729B.

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(2) Voice

- Ring Volume: Adjust the volume of Ringer.
- Receive Volume: Adjust the volume of Receive (Local side hearing).
- Transmit Volume: Adjust the volume of transmit (remote side hearing).
- DTMF Volume: Adjust the volume of DTMF.
- Echo Cancelor: Enable / Disable (it is suggested to always Enable this function).

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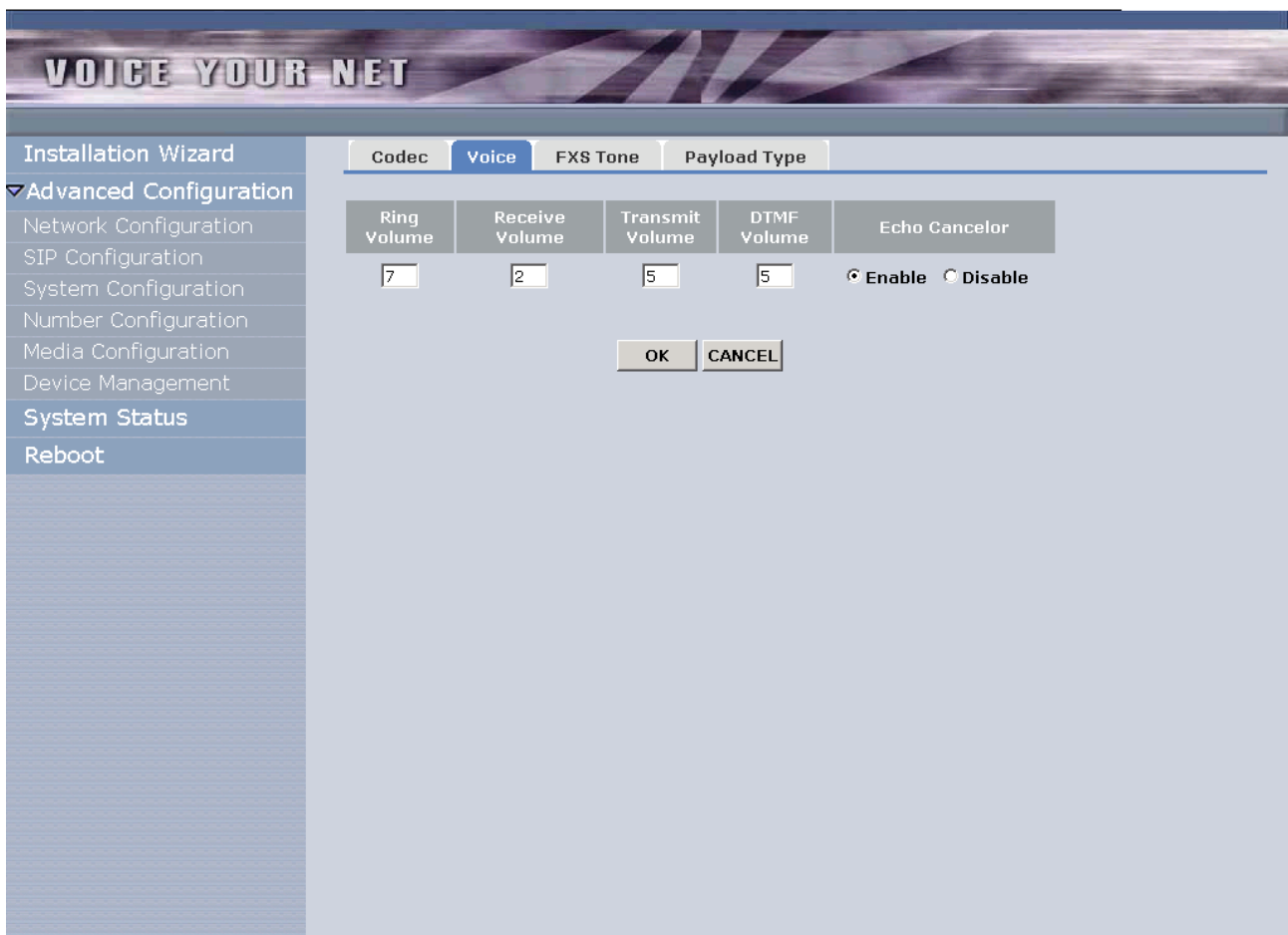
(3) FXS Tone

- Ring Back Tone: set ring back tone parameters.
- Busy Tone: Set busy tone parameters.
- Dial Tone: Set dial tone parameters.
- 2nd Dial Tone: Set 2nd Dial tone Parameters.
- Ring Gadence: Set ring Gadence parameters.
- Flash Time: Set transfer parameters.

	Frequency		Level		First Cycle of Time		Second Cycle of time	
	Low	High	Low	High	On	Off	On	Off
Ring Back Tone	440	480	155	155	2000	4000	2000	4000
Busy Tone	480	620	155	155	500	500	500	500
Dial Tone	350	440	155	155	2000	0	2000	0
2nd Dial Tone	350	440	8	8	25	25	1023	1023

Ring Cadence   Frequency 20   Level 94   On Time 2000   Off Time 4000

Flash Time   On Time 400   Off Time 800

[OK] [CANCEL]

Installation Wizard
Advanced Configuration
Network Configuration
SIP Configuration
System Configuration
Number Configuration
Media Configuration
Device Management
System Status
Reboot

Codec   Voice   FXS Tone   Payload Type

FXS Tone Configuration

(4) Payload Type

RFC2833 Payload Type: change RFC2833 Payload type. This is for special request from the other site, if RFC2833 payload types of 2 sites are different, it may cause some problem of connection.

SIP-Phone 302 Administration Guide
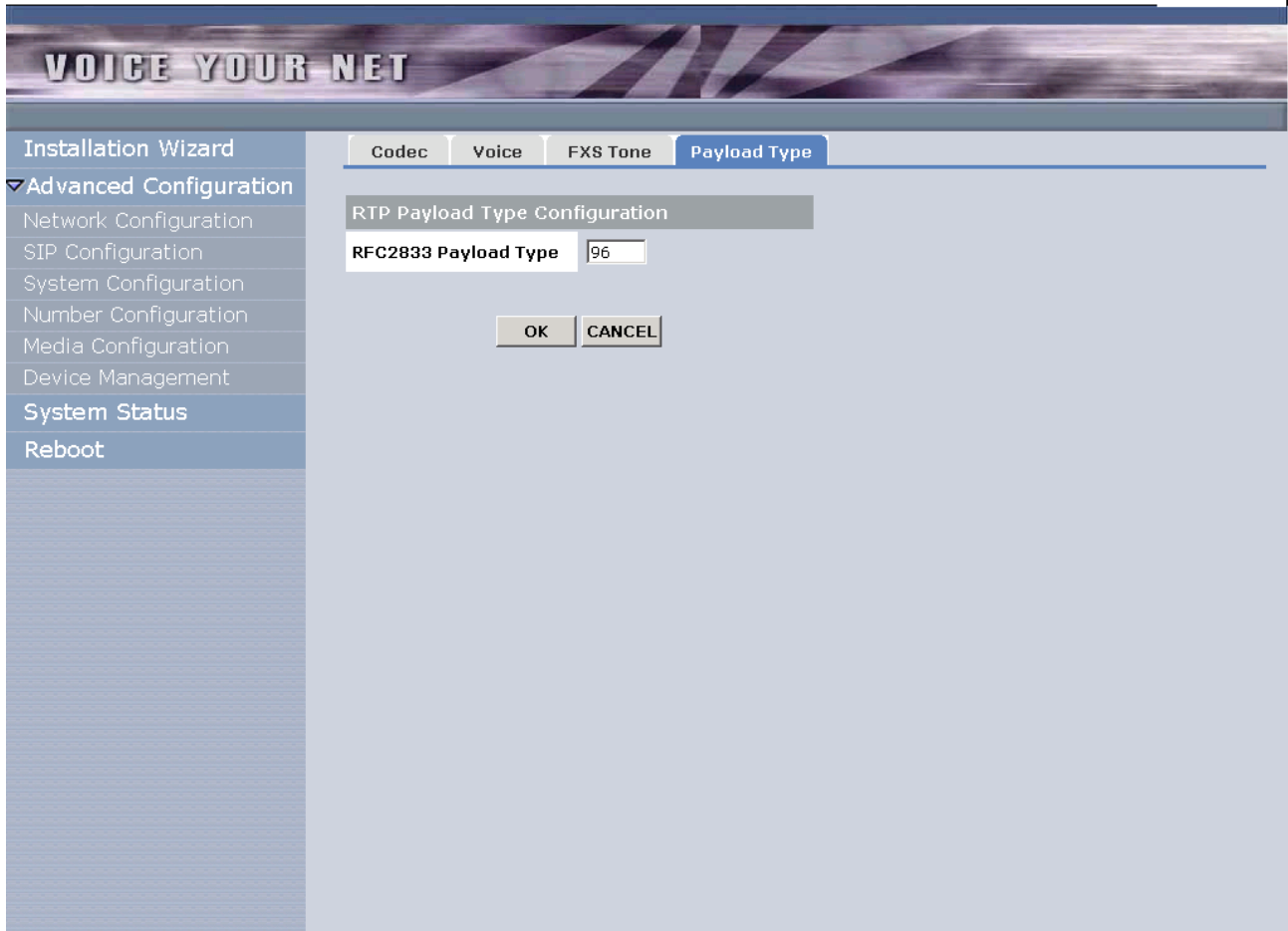
(F) Device Management
(1) Login Password

Change password configuration: First select login name as root or user, then enter current password, new password and confirm new password again to set new password.

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(2) Software Upgrade

- Download Mode: Select download method as TFTP or FTP
- TFTP/FTP Server IP Address: Set TFTP server IP address
- FTP Login: Set FTP login name and password
- Target File name: Set file name prepared to upgrade
- Target File Type: Select which sector of SIP-Phone to upgrade

Note:
After upgrade Application, please remember to execute Flash Clean, which will clean all configurations become factory values except Network settings..

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(3) Provision Server

- Provision Server Address: set Provision server IP address.
- Provision Server Login User Name: set Provision Server login user ID.
- Provision Server Login User Password: set Provision Server login user password.
- Provision Server Cycle Time: set Provision Server update cycle time.

(4) Flash Clean

Press CLEAN will clean all configurations except Network Configuration of
SIP-Phone and reset to factory default value.

| Note: |
| User must re-configure all commands all over again (except Network Configuration). |

3. System Status

(1) A. Network Status: Display all current network status of SIP-Phone.

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(2) Version Information: Display software version.

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4. Reboot

Press reboot will reset SIP-Phone.

Note: o execute reboot via web browser, SIP-Phone will automatically save all data before reboot. To execute reboot via TELNET command, please remember to do **Commit Data** before **Reboot System**.

SIP-Phone 302 Administration Guide
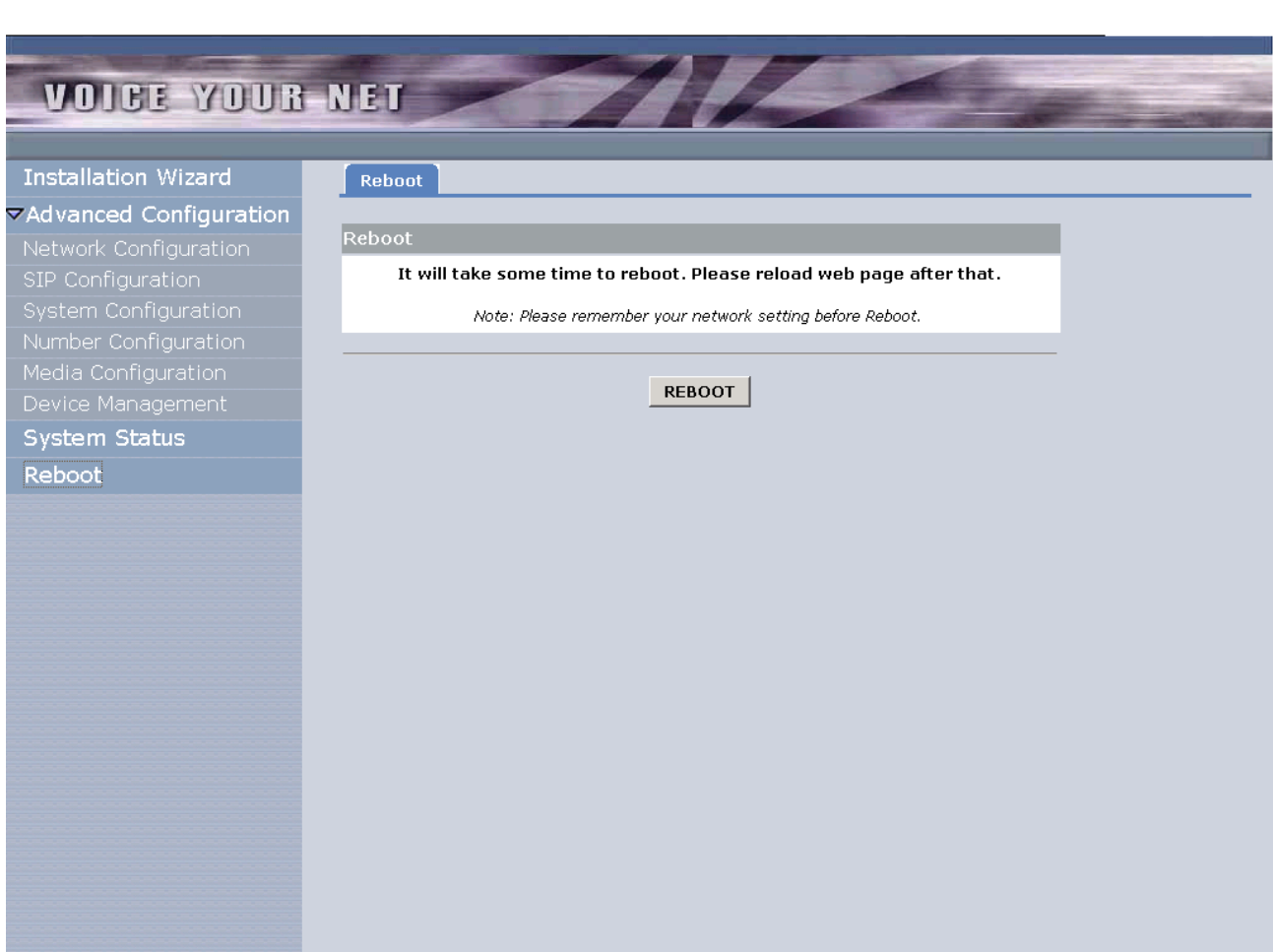
Chapter 4 Configuring the SIP-Phone 302 through Telnet command lines

After setting the IP Address of SIP-Phone 302 and reboot, (please refer to LCD Menu: 6-3.4.5), user can enter into Telnet command lines.

Note:
1. After user enter SIP-Phone configuration via telnet, please use login: "root", password: null, press enter to enter command lines.
2. If user forget login password, please contact with your distributor, we will generate one new password according to LAN Phone's MAC address. Please login with "mac" and this new password.
3. User must input lower-case command, but contents of configurations such as SIP alias or user name etc, user can set as capital case.
4. After any change of configuration, please remember to do **commit** command to save changes and then **reboot** command to reboot system.

---

1. [help] command

Type **help** or **man** or **?** to display all the command lists. The following figure is shown all commands of SIP-Phone 302.

```
usr/config$ ?

help            help/man/? [command]
quit            quit/exit/close
debug           show debug message
reboot          reboot local machine
pbook           Phonebook information and configuration
commit          commit flash rom data
ping            test that a remote host is reachable
time            show current time
ifaddr          internet address manipulation
pro             Provision Server manipulation
pppoe           PPPoE stack manipulation
flash           clean configuration from flash rom
sysconf         System information manipulation
sip             SIP information manipulation
security        Security information manipulation
line            Line information manipulation
voice           Voice information manipulation
tos             IP Packet ToS (Type of Service)values
bureau          Bureau line information manipulation
prefix          Prefix drop/insert information manipulation
auth            Set configuration items for "administrator" user.
rom             ROM file update
passwd          Password setting information and configuration

usage: help [command]

usr/config$
```
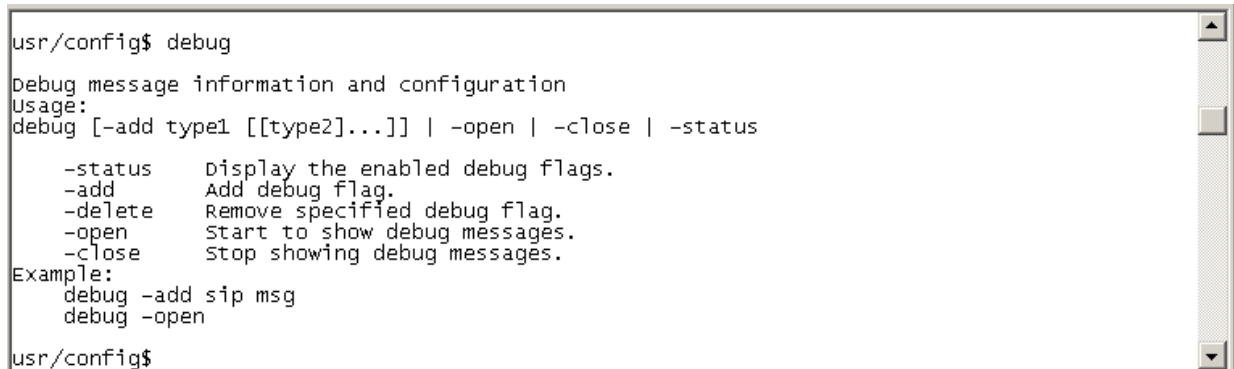
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2. [quit] command

Type **quit/exit/close** will logout SIP-Phone 302 and Telnet Program.

3. [debug] command

This command is for engineers to debug system of SIP-Phone 302. User can add debug flag via command **debug –add "debug flags"**, and then start debug function via command **debug –open**. When SIP-Phone 302 is working on screen will display related debug messages. Most frequently used debug flag are "sip", "fsm", "msg"…etc.

```
usr/config$ debug

Debug message information and configuration
Usage:
debug [-add type1 [[type2]...]] | -open | -close | -status

    -status     Display the enabled debug flags.
    -add        Add debug flag.
    -delete     Remove specified debug flag.
    -open       Start to show debug messages.
    -close      Stop showing debug messages.
Example:
    debug -add sip msg
    debug -open

usr/config$
```

4. [reboot] command

After typing **commit** command, type **reboot** to restart the SIP-Phone 302. Sometimes after user type reboot, on terminal screen will display: "Data modified, commit to flash rom?" which means SIP-Phone will record call history or not. (Ex. REDIAL, outgoing and incoming call data)

5. [pbook] command

This command is functional both in Proxy mode and Peer-to-Peer mode. In proxy mode, use speed dial or 10 DL button will dial out e.164 number in phone book. In the other hand, in peer-to-peer mode, SIP-Phone will dial out IP address.

1. **-print:** display phone book data. User can print all data in phone book by command (**pbook –print**). Furthermore, user can also print only a section of data by indicate parameter "start index" and "end index" (**pbook –print "start index" "end index"**). If parameter "end index" is omitted, only record "start index" will be displayed. (**pbook –print "start prefix"**).
2. **-add**: add a new record in phone book table by giving name, IP address, e.164 number of callee and port of allee endpoint.
   (**pbook –add name "X" ip "xxx.xxx.xxx.xxx" e164 "X" port "port number of callee"**)
3. **-delete**: delete a record of certain listed index in phone book table. (**pbook –delete**

"index number")

4. **-modify**: modify record of a certain index in phone book . Please notice that the name, IP address and e164 number must be modified together; user cannot just modify one parameter only.

   (pbook –modify "index" name "X" ip "xxx.xxx.xxx.xxx" e164 "X")

```
usr/config$ pbook

Phonebook information and configuration
Usage:
pbook [-print [start_record] [end_record]]
pbook [-add [ip ipaddress] [name Alias] [e164 phonenumber] [port portaddress]]
pbook [-delete index]
pbook [-modify [index] [ip ipaddress] [name Alias] [e164 phonenumber] [port portaddress]
]

    -print      Display phonebook data.
    -add        Add an record to phonebook.
    -delete     Delete an record from phonebook.
    -modify     Modify an exist record.
Example:
    pbook -print
    pbook -add name Test ip 210.59.163.202 e164 1001
    pbook -delete 3
    pbook -modify 3 name Test ip 210.59.163.202 e164 1001 port 5060

usr/config$ ▮
```

6. [commit] command

Save any changes after configuring the SIP-Phone 302.

```
usr/config$ commit

This may take a few seconds, please wait....

Commit to flash memory ok!
usr/config$
```

7. [ping] command

Command **ping** can test which the IP address is reachable or not.

Usage: **ping "IP address"**

The message will display packets transmitting condition or no answer from the IP address.

```
usr/config$ ping
usr/config$ ping 192.168.2.107
PING 192.168.2.107: 56 data bytes
64 bytes from 192.168.2.107: icmp_seq=0. time=5. ms
64 bytes from 192.168.2.107: icmp_seq=1. time=0. ms
64 bytes from 192.168.2.107: icmp_seq=2. time=0. ms
64 bytes from 192.168.2.107: icmp_seq=3. time=0. ms
----192.168.2.107 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)  min/avg/max = 0/1/5
usr/config$ ping 192.168.99.99
PING 192.168.99.99: 56 data bytes
no answer from 192.168.99.99
usr/config$ ▮
```

8. [time] command

When SIP-Phone 302 enable SNTP function and be able to connect with SNTP server, type **time** command will show the current time retrieved from SNTP server.

```
usr/config$ time
Current time is TUE FEB 03 14:50:08 2004
usr/config$
```

9. [ifaddr] command

Configure and display the SIP-Phone 302 IP information.

1. **–print:** print out all current configurations of ifaddr command.
2. **-ip, -mask, -gate:** Set SIP-Phone 302 IP Address, subnet mask and default gateway respectively.
1. **-ipmode:** Set SIP-Phone network mode to be Fixed IP, DHCP or PPPoE.
   When User set IP mode to be fixed IP, please set IP, subnet Mask, default gateway as mentioned in item 2.
   If User set IP mode to be DHCP, SIP-Phone will search for DHCP server to capture IP address after reboot.
   If user set IP mode to be PPPoE, please remember to set related parameters under **[pppoe] command**.
2. **-sntp:** When SNTP server is available, enable SIP-Phone 302 SNTP function and assign SNTP server IP address so that SIP-Phone can capture current time from SNTP server. **(ifaddr –sntp 1 "xxx.xxx.xxx.xxx")**
3. **-dns:** User can set primary and secondary Domain Name Server IP address. Once SIP-Phone can connect with DNS server, user can specify URL address instead of IP address for Proxy Server and phone book IP address...etc. **(ifaddr –dns 1 "primary DNS server address" –dns 2 "secondary DNS server address")**
4. **–timezone:** User can set different time zone according to the location SIP-Phone is. For example, in Taiwan the time zone should be set as 8,which means GMT+8. (GMT+8: **ifaddr –timezone 8**)
5. **–timeformat**: Set time display format as 12 or 24 hours. (**ifaddr –timeformat 0/1**, 0 as 24 hours, 1 as 12 houts)
6. **-ipsharing:** If SIP-Phone 302 is behind a IP-sharing , user must enable IP sharing function and specify public IP address.(**ifaddr –ipsharing 0/1 "public IP address of IP sharing" ,** 0 for disable and 1 for enable)
7. **–stun:** Enable/disable STUN function and set STUN server address. After user has set this function, SIP-Phone can work behind NAT device. (**ifaddr –stun "STUN server address"**)

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8. **–stunport:** Set STUN server port for inter-working with STUN Server.
9. **–http:** Set specify HTTP port for Web Management. (1024~65535)

> *Note:*
> Some Proxy servers support endpoint behind NAT function, in this case SIP-Phone doesn't have to enable IP sharing function, please contact with your Proxy Server vendor for detail information.

```
usr/config$ ifaddr    Paste

LAN information and configuration
Usage:
ifaddr [-print]|[-ipmode option]|[-sntp mode [server]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-dns index [dns server address]]
ifaddr [-autodns used]

    -print      Display LAN information and configuration.
    -ip         Specify WAN ip address.
    -mask       Set Internet subnet mask.
    -gate       Specify default gateway ip address
    -ipmode     Set get IP mode(0:Fixed IP/1:DHCP/2:PPPoE)
    -sntp       Set SNTP server mode and specify IP address.
    -autodns    Specify the way to obtain DNS Server (0:Manual/1:Auto).
    -dns        specify IP address of DNS Server.
    -timezone   Set local timezone.
    -timeformat Set time format(1:12/0:24)
    -ipsharing  Specify usage of an IP sharing device and specify IP address.
    -stun       Specify STUN Server address(x:disable STUN).
    -stunport   Specify STUN Server port.
    -http       Specify HTTP port for Web Management.(1024~65535)
Note:
    Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
    SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).
Example:
    ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
    ifaddr -ipmode 1
    ifaddr -sntp 1 210.59.163.254
    ifaddr -ipsharing 1 210.59.163.254
    ifaddr -dns 1 168.95.192.1
    ifaddr -autodns 1

usr/config$
```

10. [pro] command
Configure and display the Provision Server information.
1. **–pro:** set Provision server IP address.
2. **–proid:** set Provision server login user ID.
3. **–propwd:** set Provision server login user password.
4. **–protimer:** set Provision server recycle time.

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```
usr/config$ pro

Provision Server information and configuration
Usage:
pro [-proip ipaddress] [-proid loginid] [-propwd password] [-protimer cycletime]

    -proip      Provision Server IP address(x:disable)
    -proid      Provision Server login ID
    -propwd     Provision Server login password
    -protimer   Provision Server update cycle time (0~65535 minutes)

usr/config$
```

11. [pppoe] command

1. **-print:** display all current configurations and information.
2. **–id:** to set PPPoE authentication user name.
3. **–pwd:** to set PPPoE authentication password.
4. **–reboot:** Select enable or disable this function. If user enables this function, after PPPoE disconnected, SIP-Phone will automatically reboot to re-connect, and after reboot, if SIP-Phone still can't connect with server, SIP-Phone will keep trying to connect. On the other hand, if user disables this function, SIP-Phone won't reboot and keep trying to connect. (**pppoe –reboot 0/1**)
5. **–echo**: to set PPPoE send echo request function or not. Under some ISP sending echo request will cause abnormal behavior for LAN Phone, however, if user disable echo function, when ISP disconnect, LAN Phone will not try to reconnect. Suggest for most ISPs this function need to be enabled. Please refer to **pppoe –reboot** function.

```
usr/config$ pppoe

PPPoE device information and configuration
Usage:
pppoe [-print]
pppoe [-id username][-pwd password]

    -print      Display PPPoE device information.
    -id         Connection user name.
    -pwd        Connection password.
    -reboot     Reboot after remote host disconnection.
    -echo

usr/config$
```

12. [flash] command
This command will clean the configuration stored in the flash rom to default value and reboot the SIP-Phone 302.

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> Note:
> 1. After user upgrade new software version, suggested to execute this command to make sure new software work well on SIP-Phone 302.
> 2. To execute the command **flash –clean,** all configuration of SIP-Phone 302 stored in flash will be cleaned. It is authorized for the user whose login name is "root" only.

```
usr/config$ flash

Flash memory information and configuration
Usage:
flash -clean

Note:
    This command will clean the configuration stored in
    the flash and reboot it.
usr/config$
```

13. [sysconf] command

1. **-print:** display all current configurations.
2. **-idtime:** set the duration(in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, SIP-Phone will dial out all number pressed.
3. **-keypad:** set DTMF type .User can select DTMF type SIP-Phone receive and transmit.(**sysconf –keypad 0/1 ,** 0 for in band ,1 for RFC2833.)
4. **-2833type:** change RFC2833 Payload type.
5. **-eod**: select end of dialing key, e.g. set end of dial key as "＊" button , after finished pressing dialing number then press "＊" will dial out. (**sysconf –eod 0/1/2** , 0 for no end of dial key , 1 for "＊" button, 2 for "**#**" button)
6. **–privacy**: this function can only work in Japan and also user's service platform supports Japan standard telecom CLIR specification. When this function is set as Japan mode, other users can hide their caller ID by press special code before dial out phone number.
7. **–btstart:** Set blind transfer start key(default: *65), when you want to use the function, you only input *65 ahead other callee number C.
8. **–phone:** Add the parameter user=phone(0: off; 1: on)
9. **–Call Waiting:** When you are communicating then have incoming call, LP302 will bring Insert Tone. (default : on)
10. **–Call Transfer:** You can enable or disable the function. (default: on)
11. **–mwi:** Voice Mail function. You can enable the function.(0: off; 1: Client; 2: proxy)
12. **–mwiexpires:** MWI Subscription Keep Alive for Client(default: 1800 sec)
13. **–mwinumber:** Voice Mailbox Number

14. **–mwikey:** Voice Mailbox Hot Key(default: *69)

```
usr/config$ sysconf

System information and configuration
Usage:
 sysconf [-idtime digit] [-keypad dtmf]
         [-2833type type] [-eod digit]
 sysconf -print

 -print        Display system overall information and configuration.
 -idtime       Inter-Digits time.(1~10 sec)
 -service      Specify gateway service type. (0: Normal service,
               1: HotLine service.)
 -keypad       Select DTMF type: 0=In-band,1=RFC2833,2=INFO.
 -2833type     RFC2833 Payload Type (range:96~128 inter-used:100,102~105)
 -eod          End of Dial Digit setting(0: NONE, 1: *, 2: #)
 -privacy      Privacy Number Type(0: NONE, 1: Japan)
 -btstart      Blind Transfer Start Key
 -phone        Add the parameter user=phone(0: OFF, 1: ON)
 -waiting      Enable the Call Waiting function(0: OFF, 1: ON)
 -transfer     Enable the Call Transfer function(0: OFF, 1: ON)
 -mwi          Enable the MWI subscription function
               (0: OFF, 1: Client(Subscribe),2: Proxy(NOTIFY))
 -mwiexpires   MWI Subscription Keep Alive for Client(default:1800 sec)
 -mwinumber    Voice MailBox Number
 -mwikey       Voice MailBox Hot Key(default:*69)
Example:
  sysconf -keypad 0 -eod 2
  sysconf -btstart *65

usr/config$ █
```

14. [sip] command

4. **–print:** display all current configurations.
5. **–mode**: configure SIP-Phone 302 as Proxy or Peer-to-Peer Mode.
   Usage: **sip –mode 0/1**(1 for Proxy mode, 0 for Peer-to-Peer mode)
6. **–px**: set primary proxy server IP address or URL address (**sip –px "IP address or URL of Proxy server"**).
7. **–pxport:** set listening port of Primary Proxy server.
8. **–px2:** set secondary proxy server IP address or URL address (**sip –px2 "IP address or URL of Proxy server"**).
9. **–px2port:** set listening port of Secondary Proxy server.
10. **–outpx:** set IP address of outbound proxy server. After user set outbound proxy, all packets form SIP-Phone will be sent to outbound proxy server.
11. **–outpxport:** set listening port of outbound Proxy server.
12. **–prefix:** set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, SIP-Phone will sent out messages as Account Name @"IP address of Proxy", if user set prefix as abc, SIP-Phone will set out as abc123@"IP address of Proxy". This function is for special proxy server.
13. **–line**: identify one number for the SIP-Phone 302 to register to the Proxy   **(SIP –line**

"line number").

> **Note:** In proxy mode please remember to set user account information under security command.

14. **–pbsearch:** enable/disable phone book search function under Proxy Mode. If user enabled this function, SIP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, SIP-Phone will make call to related IP address.

15. **–expire:** set expire time of registration. SIP-Phone will keep re-registering to proxy server before expire timed out.

16. **–port:** set listening UDP port or SIP-Phone.

17. **–rtp:** set RTP port number. SIP-Phone will use this port to send and receive voice.

```
usr/config$ sip

SIP stack information and configuration
Usage:
sip [-mode  pxmode]
sip [-px address] [-prefix prefixstring]
    [-pxport ProxyPort][-outpx address][-line number]
    [-expire t1] [-port udpPort] [-rtp rtpPort]
sip -print

    -print      Display SIP stack information and configuration.
    -mode       Configure as Proxy mode or Peer-to-Peer mode.
    -px         Proxy server address. (Proxy IPv4 address or Proxy dns name)
    -px2        Secondary Proxy server address. (IPv4 address or dns name)
    -pxport     Proxy server port.    (the port of proxy)
    -px2port    Secondary Proxy server port.    (the port of Secondary proxy)
    -outpx      OutBound Proxy server address. (IPv4 address or Proxy dns)
    -outpxport  OutBound Proxy server port. (the port of OutBound proxy)
    -prefix     Specify prefix string, use it when UserID contains alphabets
                (if UserID uses numerals, specify as null)
    -line       TEL Phone number.
    -pbsearch   Search phone book    0:off/1:on.
    -expire     The relative time after which the message expires(0 ~ (2^31-1))
    -port       SIP local UDP port number (5060~5070). Default: 5060
    -rtp        RTP receive port number (2326~65534). Default: 16384
Example:
    sip -mode 1
    sip -px 210.59.163.171 -line 70
usr/config$
```

15. [security] command

1. **–print**: display all current configurations.
2. **-name**: set user ID of SIP-Phone 302 for registering. User can set user name and password for registering. If password is no need, please set user name the same as line number or SIP-Phone won't register successfully.
3. **–pwd**: set account password for registering.

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```
usr/config$ security

Secuirty information and configuration
Usage:
 security [-name username] [-password password]
 security [-print]

 -print        Display system account information and configuration.
 -name         Specify user name.
 -pwd          Specify password.
Example:
  security -name 1001 -pwd 1001

usr/config$
```

16. [line] command

The line command is Forward setting information and configuration.

1. –print: display Forward information.

2. –unconditional: It is included the above two types. Whether the SIP-Phone is in which status, calls will be automatically forwarded to the assigned phone number.

3. –busy: set busy forward number, when SIP-Phone 302 is in busy status, the incoming call will be forwarded to the assigned phone number.

4. –no answer: set no answer forward number, when SIP-Phone 302 has incoming call and hasn't been picked up for around 10 seconds, the incoming call will be forwarded to the assigned phone number.

```
usr/config$ line

Gateway line information and configuration
Usage:
line -print        Gateway line information.
line -fwd          Forward type and number.
    1:unconditional
    2:busy
    3:no answer
Example:
    line -print
    line -fwd 1 1001
    line -fwd 2 x

usr/config$
```

17. [voice] command

The voice command is associated with the voice codec setting information.

1. **-print**: display voice codec information and configuration.

2. **-send**: three voice packet size can be configured as 20 ms, 40 ms or 60 ms.(only 30 and 60 ms for G..723.1)

3. **-priority**: set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need, for example, **voice –priority g723** or **voice –priority g723 711a g711u g729 g729a** means SIP-Phone can support only one codec or four codecs.

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4. **-volume**: There are two types can be adjustable, voice volume and ring volume.
    Voice volume means the volume user can hear.

> **Note:**
> If value of volume set as 0 means –32db, 1 means –31db…etc.

5. **-nscng**: enable or disable sound compression and comfort noise generation. It is
    only for codec G.723.1. (0 for off, 1 for on)

> Note: It is for advanced administrator use only. Please ask your distributor
> before changing any settings of this command.

```
usr/config$ voice

Voice codec setting information and configuration
Usage:
      [-volume [voice level] [ring level]
      [-nscng [G711U used1] [G711A used2] [G723 used3]]
voice -print
voice -priority [G729] [G723] [G711U] [G711A] [G729B]

    -print      Display voice codec information and configuration.
    -priority   Priority preference of installed codecs.
                G.729
                G.723
                G.711U
                G.711A
                G.729B
    -volume     Specify the following levels:
                voice volume (0~9, default: 7).
                ring volume (0~12, default: 7).
    -nscng      No sound compression and CNG. (G.723.1 only, On=1, Off=0).
Example:
    voice -volume voice 7 ring 7
    voice -priority g729 g723 g711u g711a1
usr/config$ ▮
```

18. [tos] command

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Code
Point (DSCP) of the DS field in the IP packet header, and map each Code Point to a
corresponding egress traffic priority. As per the definition in RFC2474, the DS field is
Type-of-Service (TOS) octet in IPv4. The recommended DiffServ Code Point is defined
in RFC2597 to classify the traffic into different service classes. The mapping of Code
Point value of DS-field to egress traffic priorities is shown as follows.

1. High priority with DS-field.

Expected Forwarding (EF)	101110	====>	46 (Decimal System)
Assured Forwarding (AF)	001010	====>	10 (Decimal System)
	010010	====>	18 (Decimal System)

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	011010	====>	26	(Decimal System)
	100010	====>	34	(Decimal System)

2. Low Priority with DS-field:

Assured Forwarding (AF)	001100	====>	12	(Decimal System)
	010100	====>	20	(Decimal System)
	011100	====>	28	(Decimal System)
	100100	====>	36	(Decimal System)
	001110	====>	14	(Decimal System)
	010110	====>	22	(Decimal System)
	011110	====>	30	(Decimal System)
	100110	====>	38	(Decimal System)
	000000	====>	0	(Decimal System)

1. **-print** : display all current configurations.
2. **–rtptype:** set DSCP value of signaling packets from 0 to 63
3. **–siptype:** set DSCP value of RTP packets from 0 to 63

Note:

This command won't be functional until whole network environment support
DSCP function, e.g. all routers or switches in your network have enabled DSCP
feature.

```
usr/config$ tos
IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
tos -print
     [-rtpreliab mode]
tos -print

Example:
    tos -rtptype 10 -sigtype 0

usr/config$ 
```

19. [bureau] command
   1. **-print**: display all current configurations.
   2. **–hotline**: set hotline IP and remote phone number. If user has enable Hotline
      function, once SIP-Phone 302 been off-hook, it will automatically dial out to
      assigned IP and phone number.
      (**bureau –hotline "IP of destination" "Phone number of destination"**)

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```
usr/config$ bureau

Bureau line setting information and configuration
Usage:
bureau  [-hotline [Port DestIP TELnum]]
bureau -print

    -print     Display Bureau line information and configuration.
    -hotline   Set Hot line information.
Note:
    Hotline feature should be used together with:
        $sysconf -service 1 (HotLine service)
        $sip     -mode    0 (peer-to-peer mode)
Example:
    bureau -hotline 192.168.4.69 628

usr/config$
```

20. [prefix] command

1. **–print:** Display current configuration of digit manipulation.
2. **-add**: Add a rule of digit manipulation.
3. **-prefix:** Set which prefix number to implement digit manipulation rule.
4. **-drop:** Enable or disable drop function. If this function is enabled, Gateway will drop prefix number on incoming call. (0 as disable, 1 as enable)
5. **-insert:** Set which digit to insert on incoming call. (**digit -add prefix "prefix number" drop 0/1 insert "insert number"**)
6. **-modify:** Modify a rule to drop or insert prefix digits of incoming call. (**digit –modify "index" drop 0/1 insert "insert number"**)
7. **-delete:** Delete a rule to drop or insert prefix digits of incoming call. (**digit –delete "index"**)

```
usr/config$ prefix

Prefix drop/insert information and configuration
Usage:
prefix -add [prefix number][drop number][insert digits]
prefix -delete index
prefix -modify index [prefix number][drop number][insert number]
prefix -print    Prefix drop/insert information.
       prefix    The prefix of dialed number.
       drop      Drop prefix(Enable:1/Disable:0).
       insert    Insert digits.
Example:
    prefix -add prefix 100 drop 1 insert 2000
    prefix -add prefix 100 drop 1
    prefix -add prefix 100 drop 0 insert 200
    prefix -delete 1
    prefix -modify 1 prefix 100 drop 0 insert 300

usr/config$ █
```

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21. [auth] command

You can control what command administrator can use.

-print: display auth switch configuration.

   Use item name to do configuration name (0: Disable; 1: Enable)

```
usr/config$ auth

Root control what command administrator can use.
Usage:
auth -print Display auth switch configuration.
           Use item name to do config name (0=Disable, 1=Enabled).
Example: auth -ifaddr 1

usr/config$ █
```

22. [rom] command

1. **-print**: show all current configurations and version information.
2. **-app**,**-boot**, **-dsptest**, **-dspcore**, **-dspapp**,: upgrade main boot code, main application code, DSP testing code, DSP kernel code, DSP application code, Ring Back Tone PCM file and Hold Tone .

> **Note:**
> After upgrade Application, please remember to execute **flash –clean** command, which will clean all configurations become factory values except IP address.

3. **-boot2m**: to upgrade 2mb rom file, which includes all firmware file mentioned in item 2.
4. **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading firmware rom file.
5. **-f**: the file name prepared for upgrading is necessary as well.
6. **–method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
7. **–ftp**: specify user name and password for FTP download method.
   For example: User prepares to upgrade the latest app rom file – lpsip.100, the TFTP server is 192.168.1.1, User has to input command as below:
   rom –app –s 192.168.1.1 –f lpsip.100

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```
usr/config$ rom

ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp]
    -s TFTP/FTP server ip -f filename
rom -print
    -print    show versions of rom files. (optional)
    -app      update main application code(optional)
    -boot     update main boot code(optional)
    -boot2m   update 2M code(optional)
    -dsptest  update DSP testing code(optional)
    -dspcore  update DSP kernel code(optional)
    -dspapp   update DSP application code(optional)
    -s        IP address of TFTP/FTP server (mandatory)
    -f        file name(mandatory)
    -method   download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
    -ftp      specify username and password for FTP
Note:
    This command can run select one option in 'app', 'boot',
    , 'dsptest', 'dspcore', and 'dspapp'.
Example:
    rom -method 1
    rom -ftp vwusr vwusr
    rom -app -s 192.168.4.101 -f app.bin

usr/config$
```

Command **rom –print** can show current version installed in SIP-Phone 302. (**)

```
usr/config$ rom -print

Download Method  :  TFTP

      Boot Rom   :  lp302
Application Rom  :  lp302sip1113.bin
usr/config$
```

23. [passwd] command

For security protection, user has to input the password before entering **application user/config mode**. Two configurations of login name/password are supported by the system.

1. **–set**: set password of "root" users or "administrator" users.
   (**passwd –set root/administrator "password"**)

2. **–clean**: clean up password restored before, and user can login :"root/administrator", password: "press enter".

   User who requests authorization to execute **all** configuration commands needs to login with "root". If a user login with "administrator", commands below are not functional:

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```
usr/config$ passwd

Password setting information and configuration
 Usage:
   passwd -set Loginname Password
 Note:
   Loginname can be only 'administrator'
 Example:
   passwd -set administrator Your_Passwd_Setting

usr/config$
```

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Chapter 5 Upgrading Software on the SIP-Phone 302

SIP-Phone 302 supports three methods to upgrade the new version. All methods are necessary to prepare the **TFTP** or **FTP** program on the host PC as **TFTP/FTP server**. After installing **TFTP/FTP** program on one PC and connecting to network, SIP-Phone 302 is ready to be upgraded.

1. LCD Panel Control
2. Remote Control: Telnet
3. Web Management

1.From LCD Phone Menu

Please select the 7→2 selection-**SW Upgrade.** Press **OK** to enter into the sub-selection as below.

1. **SW Update**

    (1) Method

    There are two methods to download new version file, please move the ">" symbol by press ▲or ▼ on the keypad to select TFTP or FTP method, then press **OK** to confirm it.

    (2) Sever

    User has to offer one TFTP/FTP server IP Address and set this IP Address via keypad. The IP Address is necessary for upgrading SIP-Phone new application rom file.

    (3) Account

    User has to input user name for FTP server login .It is necessary for upgrading SIP-Phone new application rom file via FTP method.

    (4) Password

    User has to input user password for FTP server login .It is necessary for upgrading SIP-Phone new application rom file via FTP method.

    (5) File Name

    User has to press the file name of new application rom file prepared for upgrading

    (6) Version

    Show versions of all software and hardware. (**)

    (7) Upgrade

    Select YES or NO to start upgrade.

    (8) Exit

    Return to upper level of LCD Menu

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> **Note:**
> Download via LCD command can only upgrade new *application* rom file.

2. Web Management

Please refer to chapter **4.18 [rom] command**

- FTP/TFTP Server IP Address: Set TFTP server IP address
- Target File name: Set file name prepared to upgrade
- Method: Select download method as TFTP or FTP
- FTP Login: Set FTP login name and password
- Target File Type: Select which sector of SIP-Phone to upgrade

> Note:
> 1. After 2mb file download is finished, all configurations might change to default values, user has to configure again.
> 2. After upgrade Application, please remember to execute Flash Clean, which will clean all configurations become factory values except IP address.

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3. Telnet command lines

Please refer to chapter **4.22 [rom] command**

1. **-print**: show all current configurations and version information.
2. **-app**,**-boot**, **-dsptest**, **-dspcore**, **-dspapp**,: upgrade main boot code, main application code, DSP testing code, DSP kernel code, DSP application code, Ring Back Tone PCM file and Hold Tone .

> ***Note:***
> After upgrade Application, please remember to execute flash –clean command, which will clean all configurations become factory values except IP address.

3. **-boot2m**: to upgrade 2mb rom file, which includes all firmware file mentioned in item 2.
4. **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading firmware rom file.
5. **-f**: the file name prepared for upgrading is necessary as well.
6. **–method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
7. **–ftp**: specify user name and password for FTP download method.
   For example: User prepares to upgrade the latest app rom file – lpsip.100, the TFTP server is 192.168.1.1, User has to input command as below:
   rom –app –s 192.168.1.1 –f lpsip.100

```
usr/config$ rom

ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp]
    -s TFTP/FTP server ip -f filename
rom -print
    -print      show versions of rom files. (optional)
    -app        update main application code(optional)
    -boot       update main boot code(optional)
    -boot2m     update 2M code(optional)
    -dsptest    update DSP testing code(optional)
    -dspcore    update DSP kernel code(optional)
    -dspapp     update DSP application code(optional)
    -s          IP address of TFTP/FTP server (mandatory)
    -f          file name(mandatory)
    -method     download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
    -ftp        specify username and password for FTP
Note:
    This command can run select one option in 'app', 'boot',
    , 'dsptest', 'dspcore', and 'dspapp'.
Example:
    rom -method 1
    rom -ftp vwusr vwusr
    rom -app -s 192.168.4.101 -f app.bin

usr/config$
```

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