ACBEZ VoIP

PRO5301/5302
IP Phone
User Manual

Chapter 1 Introduction

The IP Phone PRO530X is a full-featured IP-based telephone system, which provides VoIP service on LAN or any IP based environment. By using IP environment for voice communication, company or individual can save lots of expenses and make data and voice network converged.

IP Phone PRO530X also supports PSTN analog line connection. Therefore, it can perform as the same as traditional POTS (Plain Old Telephone Service).

＊Product Description

1.Features and Specification

Basic Features:

- ITU-T H.323 v3 compliance
- DTMF detection/generation
- TFTP/FTP software download
- Remote configuration/reset via Telnet
- LED indication for system status：Speaker, Hold ,Mute , Message , PSTN
- Network Interface:
  - Switch Hub inside, providing 2 RJ-45 sockets for 10/100Base-T connection
  - 1 RJ-11 socket for PSTN connection
- Microsoft Net Meeting v3.0 compatible
- SNTP (Simple Network Time Protocol)
- Call HOLD/TRANSFER/FORWARD/MUTE
- PSTN/IP side access switch
- 10 Direct Line Button for speed dial
- Support Speaker Mode
- System Configuration from keypads and displayed on LCD
- Ring Tone selection
- Password setting for security
- Function Keys: Speaker, Redial, Mute, Hold, Transfer, Forward, Message, PSTN
- Ten sets last Phone Number redial
- Dial plan
- Provide TOS (Type Of Service) function
- Gatekeeper mode or Peer to Peer mode selection

Caller ID:

- IP side display H323-ID and E.164.
- Display the count of total call received.
- Display un-answered call name, number.
- Show caller's name, number, calling time.

Volume Adjustment:

- Speaker volume level adjustable.
- Handset Receiver volume level adjustable.

LCD:

- 2 lines, 24 character Dot Matrix display.
- Indicator messages of HOLD, MUTE, PSTN, Directed Line 1~10.
- Display current date and time.
- Display of call duration.

Audio features:

- Codec -- G.711 a/μ law, G.723.1 (6.3Kbps),G.729, G.729a, G.729b, G.729ab
- VAD (Voice Activity Detection), CNG (Comfort Noise Generate)
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Bad Frame Interpolation
- Provide H.245 Out-band DTMF message
- Call Transfer (H.450.2)
- Call Forward (H.450.3)
- Call Hold (H.450.4)
- Gain (Voice Volume) Settings
- Provide Call Progress Tone: Dial tone, busy tone, call-holding tone and ring-back tone

Firmware Upgrade:

- Firmware version displayed on LCD and Web.
- Three ways to upgrade firmware:
  - Download via KEYPAD configuration (can only upgrade application rom file) .
  - Download via Telnet.
  - Download via Web Browser

Management Features:

- Three easy ways for system configuration
  - IP Phone 530X KEYPAD
  - TELNET
  - Web Browser

2.Appearance

1.Front View and Keypad function

- **C**: Cancel and Clear
- **←→** : Move to left /previous and right/next.
- **OK**: Press OK to confirm the modification.
- **Direct Line (DL) Button 1 – 10**: User press DL button to do speed dial according to phone book data 1-10 (please refer to LCD configuration-Phone Address Book, Advanced Configurations via Telnet- [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	"Space" ; "," ; "." ; "!" ; "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	“0”
#	“_” ; “@” ; “#”

- **MUTE**: Mute the voice of MIC and let others can’t hear from user in communication.
- **PSTN**: Press PSTN to switch IP Phone 530X as PSTN or IP Phone Mode. In PSTN mode, “PSTN” characters will be displayed on LCD left bottom side, then users can dial out as if standard telephone set in PSTN; in IP Phone mode, “GK” characters will be displayed on LCD left bottom side.

Note:

1. ***When IP Phone is in PSTN mode, only*** **PSTN** ***and*** **SPEAKER** ***function key can work.***
2. ***On LCD will display ”…Incoming Call... ” to inform user when IP Phone has both IP and PSTN side incoming calls.***
3. ***If in communication with IP side, user can press*** **HOLD** ***to hold IP side, then press*** **PSTN** ***to pick up PSTN side, after that can press*** **HOLD** ***again to retrieve IP side.***
4. ***If in communication with PSTN side, user must hang up PSTN side before pick up IP side.***

- **HOLD**: To hold a call with H.450 function.
- **SPEED**: Press SPEED and number (Phone book index) to do speed dial according to phone book data (please refer to LCD configuration-Phone Address Book or Advanced Configurations via Telnet- [pbook] command).
- **FORWARD**: Forward an incoming call to another IP device by H.450 forward function. (Please refer to *LCD configuration-Indicate Forward Type*)

- **MESSAGE and its indicated LED light**: When missed incoming calls, the MESSAGE LED will be flashing. User can check the information of missed calls by pressing the MESSAGE button.
- **TRANSFER**:
  1. Transfer a call by H.450 transfer function. Press TRANSFER button in

communication and press phone number which user want to transferred to can transfer this call.

2. Change characters to be capital or lowercase: when pressing TRANSFER before press letters can switch type of letters.

- **REDIAL**: Redial the last outgoing call.
- **+** And **-**: Adjust the voice volume in communication.
- **SPEAKER**: Speaking without picking up handset.

Note: All function keys mentioned above (except Number 1 –10, SPEAKER* and #) are effective only in IP Phone mode.

2.Rear View

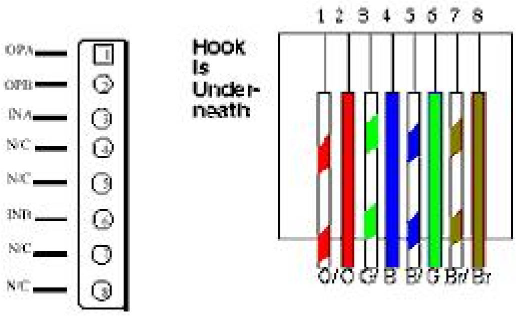
- **DC 9V**: DC 9V power input outlet
- **LAN**: RJ-45 connector, connected directly to the **Hub** through the **straight** CAT-5 cable.
- **PC**: RJ-45 connector, connected directly to the **PC** through the **straight** CAT-5 cable
- **Line**: RJ-11 connector, connected directly to the PSTN analog line.

Note: There are two LED indicated lights: LINK/ACT and 10/100 for LAN port and PC port. When network status is in normal, LED of LINK/ACT will be flashing; when transmit rate is in 10 mbps/100mbps, LED of 10/100 will light off/on.

3.Specification of connector

1、 Ethernet Port：

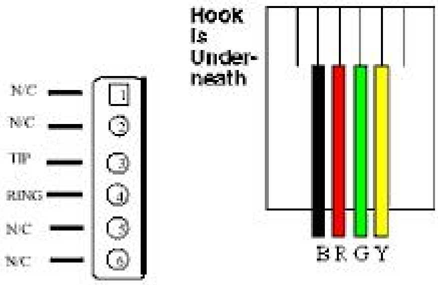
Ethernet port is for connecting IP Phone to network, transmit rate supports 10/100 Base-T.



Ethernet connector（LAN）

2、 RJ11connector：

RJ11 connector is for connecting IP Phone with PSTN.



RJ11connector

Chapter 2 Start up – from LCD Configuration

Note:

1. Any configurations that have made for the IP Phone 530X, user has to do **Reboot** in the selection 1→6 "Reboot".
2. **We suggest user to set IP address via LCD menu 1→1 first, then go to chapter 3 to do other detail configurations via web browser.**

1.Initialize IP Phone 530X

1. When power on the IP Phone 530X, the LCD screen shows as below.

IP-Phone Board Start Booting

2. Wait around one minute until the IP Phone 530X finishes boot program initialization.
3. User can see flashing greeting as below:

System Initializing…………..

4. Then IP Phone get into standby mode:

	IP-Phone
GK	10:10:10 AM

The main LCD screen would be shown as similar as above. The GK word means the IP Phone 530X is in Gatekeeper Mode, and when IP Phone is connected to SNTP server, on LCD will show current time from SNTP server.

5. After pressing the **PSTN** button, the GK word will be replaced by PSTN. Then IP Phone is in PSTN Mode. Please notice that in PSTN mode user must plug PSTN line in RJ-11 port.

	IP-Phone
PSTN	10:10:10 AM

Press 🡐 or 🡒 to enter configuration mode then press **OK** button to enter certain menu.

1. System Configuration (mandatory, protected by password)
2. User Line Number (mandatory)
3. Ring Configuration
4. Indicate Forward Type
5. Message Box
6. Phone Address Book

Note: ***LCD Panel of IP Phone 530X is operated manually by moving ← or → on the keypad. Press OK to enter separate configuration menu. Press C to go back to the main menu.***

2.LCD Configuration

User can set the following configurations manually by operating the commands displayed on LCD.

1. **System Configuration (mandatory, protected by password)**
   - **Please Enter Password:**
     User must key in password to enter this menu, selections under this command are all important ones, which can only be configured by administrators.

Note: 1.Password to enter System Configuration: The same as H.323 token password (please refer to item (11)), default value is **x**. **(Please press TRANSFER button to switch as lowercase characters first, then press 9 twice.)** 2.If user forget password, please contact with our company, we will generate a specific password according to your MAC address of IP Phone. 3.User can also try to configure IP Phone via Telnet or Web browser with default IP address: 10.1.1.3.

1. **Connect Configuration**

   It is necessary for user to set sub-configurations included in **Connect Configuration** in order to run the IP Phone 530X correctly.

   In addition, user has to prepare one valid IP Address to meet your network environment. (Global public IP Address or Virtual IP Address)

   (1) IP address

   In this configuration mode, user presses the prepared IP Address on the IP Phone keypad. Please input IP address as format: xxx.xxx.xxx.xxx.

   (2) Subnet Mask:

   User has to press the subnet mask IP Address on the IP Phone keypad**.**

   (3) Gateway

   User has to press the default gateway IP that meets IP Address on the IP Phone keypad.

   (4) Primary DNS

   User can set IP address of Domain Name Server, then for Gatekeeper and Phone book can enter URL address or IP address. Please refer to **4.9 [pbook]** and **4.13 [h323]** command.

(5) Secondary DNS

User can set IP address of secondary DNS, once primary DNS cannot work normally, IP Phone can refer to secondary DNS.

(6) Gatekeeper

User has to offer one available Gatekeeper server IP Address and set this IP Address on the IP Phone keypad.

(7) Second Gatekeeper

IP Phone provide alternative Gatekeeper feature, if IP Phone can't register to Main Gatekeeper for 10 times, it will try to register to the Second Gatekeeper. When main Gatekeeper can't work normally, IP Phone can still keep working with Second Gatekeeper

(8) SNTP Configuration

IP Phone 530X supports that user can assign one SNTP (Simple Network Time Protocol) Server in your country by setting in IP Phone. User has to offer one available SNTP server IP Address and set this IP Address on the IP Phone keypad.

(A) SNTP Mode: User can set different time can set SNTP function too be on/off/broadcast, which means IP Phone will capture current time from SNTP server or not, or broadcast to find a SNTP server and capture current time.

(B) SNTP Server: User can specify a SNTP server for IP Phone to capture current time.

(C) Time Zone: User can set time zone according to the location IP Phone is. For example, in Taiwan the time zone should be set as 8,which means GMT+8. (User can press "∗" as "-")

Note: If user didn't set SNTP server, on LCD won't display current time.

(9) Connection Mode

There are 2 types for IP Phone 530X to connect to the other devices. They are **GK**, **P2P.** The default mode is in GK mode. When user would like to connect via P2P mode, the IP Phone 530X must change as well. Move the "→" symbol by press 🡐 or 🡒 on the keypad to select one mode.

(10) DHCP Mode (ON/Off)

User can set IP Phone in DHCP mode, which means IP Phone will get a dynamic IP automatically.

(11) Token Password

(A) LCD menu password: User can enter LCD system configuration by key in this password

(B) H.235 security: To set RRQ/ARQ authentication token password. If IP Phone wants to register to a Gatekeeper, which implement H.235 security token feature, IP Phone has to set a RRQ/ARQ authentication token password, which is provided by Gatekeeper manager. IP Phone can't work normally with this Gatekeeper unless Token Password is set.

(12) GRQ Option

Set gatekeeper auto-discovery function to be OFF or ON. If this function is enabled and IP address of Gatekeeper is set as 255.255.255.255, IP Phone will multicast to search a Gatekeeper on network with configured Gatekeeper ID (please refer to **(12) Gatekeeper ID**); if IP address of Gatekeeper is set, before IP Phone register to the assigned Gatekeeper, it will send out GRQ (Gatekeeper Request) message with configured Gatekeeper name to Gatekeeper first.

(13) Gatekeeper ID

Set Gatekeeper name for Gatekeeper discovery. When IP Phone send out Gatekeeper discovery message will search Gatekeeper with this Gatekeeper name

2. User Line Name

User has to identify one ID name for the IP Phone 530X to register to the Gatekeeper.

3. Firmware Update

(1) Download method

There are two methods to download new rom file, please move the "→" symbol by press 🡸 or 🡺 on the keypad to select TFTP or FTP method, then press OK to confirm it.

(2) Set File Server IP

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

(3) Set FTP user account

User has to press user name and password for FTP server login .It is necessary for upgrading IP Phone new application rom file in FTP method.

(4) Indicate file name

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

(5) Start Download

Press OK to start download new application rom file. After download is finished, IP Phone will automatically reboot.

(6) Firmware Version

Show versions of all rom files and hardware.

Note: 1.Download via LCD command can only upgrade new ***application*** rom file. 2.If IP Phone fails to upgrade via LCD menu, IP Phone will automatically reboot.

4. **Hardware Test**

   IP Phone 530X provides self-test for all functions buttons. Once user press hardware test selection, IP Phone will start testing programs. Please follow the direction from LCD panel to operate and complete whole procedure.

5. **PPPoE Configuration**

   (1) PPPoE Mode

   Choose ON or OFF to enable or disable PPPoE function.

   (2) Username And Password

   Set PPPoE authentication user name and password.

   (3) Retry When Disconnect

   Choose ON or OFF to enable or disable this function. If user enables this function, after PPPoE being disconnected, IP Phone will automatically reboot to re-connect, and after reboot, if IP Phone still can’t get contact with server, IP Phone will keep trying to connect. On the other hand, if user disables this function, IP Phone won’t reboot and keep trying to connect.

6. **Reboot**

   It is necessary and important for user to reboot it after any configurations has been made to the IP Phone 530X.

2. **User Line Number**

User has to identify at least one number for the IP Phone 530X to register to the Gatekeeper. User can set up to 10 different numbers for one IP Phone.

(1) Line Number 1
(2) Line Number 2
(3) Line Number 3
(4) Line Number 4
(5) Line Number 5
(6) Line Number 6
(7) Line Number 7
(8) Line Number 8
(9) Line Number 9
(10) Line Number 10

Note: User can set six zero "000000" on LCD to disable this number and the number after this one. Ex. Line Number 1-5 is configured, if user set Line 3 as "000000", Line number 3-5 will be disabled and user will see x displayed on LCD.

3. **Ring Configuration**

(1) Ring Style Selection

There are three tone styles for IP Phone 530X. Move the "→" symbol by press 🡐 or 🡒 on the keypad to select the tone style preferred, then press OK to confirm it.

Choose Tone Style: Low Middle High

(2) Ring Volume Control

User can adjust ring volume by press 🡐 or 🡒 on the keypad to decrease or increase volume.

4. **Indicate Forward Type**

There are two selections to activate or deactivate forward function. After selection please press **OK**

(1) Activate: choose under which situation to forward call to another endpoint. After selection please press **OK**, then enter LINE NUMBER of another endpoint prepared to forward to.

A. Busy

When IP Phone 530X is in busy status, the incoming call will be directly forwarded to the assigned phone number.

B. No response

When IP Phone 530X is continuing ringing around 10 seconds, the incoming call will be directly forwarded to the assigned phone number.

C. Unconditional

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

(2) Deactivate: choose under which situation to deactivate forward function. After selection, please press **OK,** then user can see LINE NUMBER of the endpoint that is already configured to forward to, now press OK again.

Note: The number that user prepares to forward to is E.164 number which is registered on the Gatekeeper.

5. **Message Box**

If there is an unanswered IP call, it will be kept in message box. **MESSAGE** LED will be flashing until user press **MESSAGE** to check miss call and re-press MESSAGE to return to main screen.

1. **New Call**：to see all incoming missed call records in message box.
2. **History**：to see all outgoing records in message box.

6. **Phone Address Book**

1. **Display**

   Display all records of name, telephone number, and IP address in the phone address book.
2. **Add**

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

   Edit a record of name, telephone number, and IP address of the phone address book.
4. **Delete**

Delete one record in the phone address book.

7. **PPPoE Information**

All items below can only be displayed when PPPoE connection is established, user can check related information here.

1. **PPPoE Status**
2. **Note Information**
3. **PPPoE IP address**
4. **Destination Host**
5. **DNS Primary**
6. **Subnet Mask**
7. **Authentication**
8. **Protocol**
9. **Device**

Chapter 3 Web Configuration

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

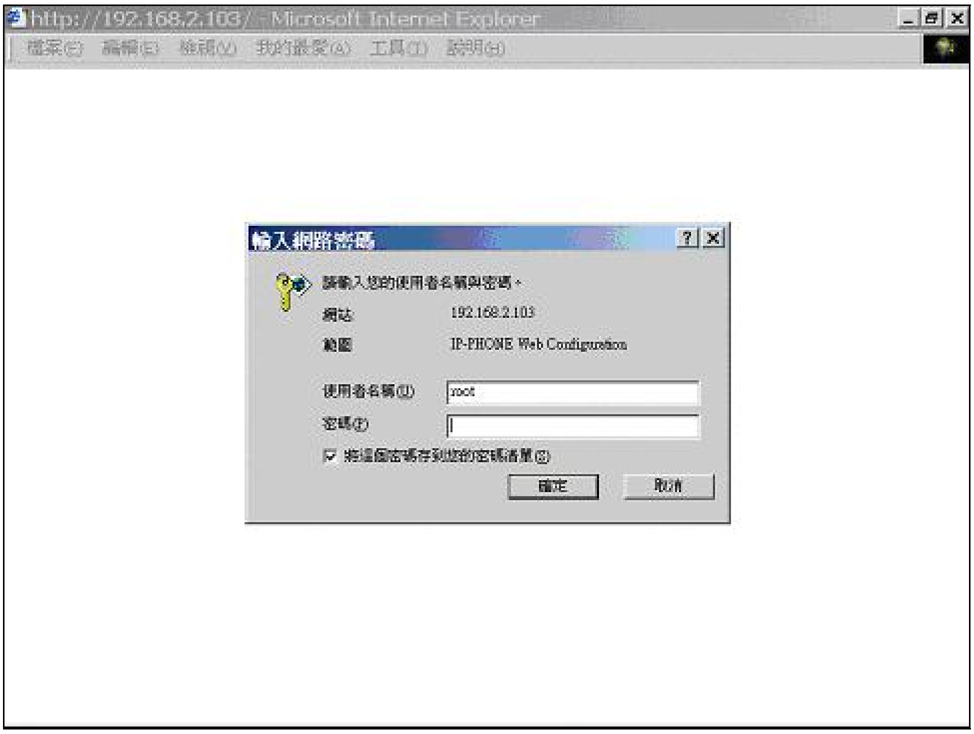
The configuration function and step is similar with the way through command line. Please refer to the chapter 4-Advanced Configurations via Telnet for more detail information. Below is a guide for user to configure via web interface.

Step 1. Browse the IP Address predefined via Keypad

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

Step 2. Input the login name and password

- Login name: root / administrator
- 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 **12.PASSWORD** via web interface.

Step 3. The web interface main screen

After enter login name and password, user can see web interface main screen as below.

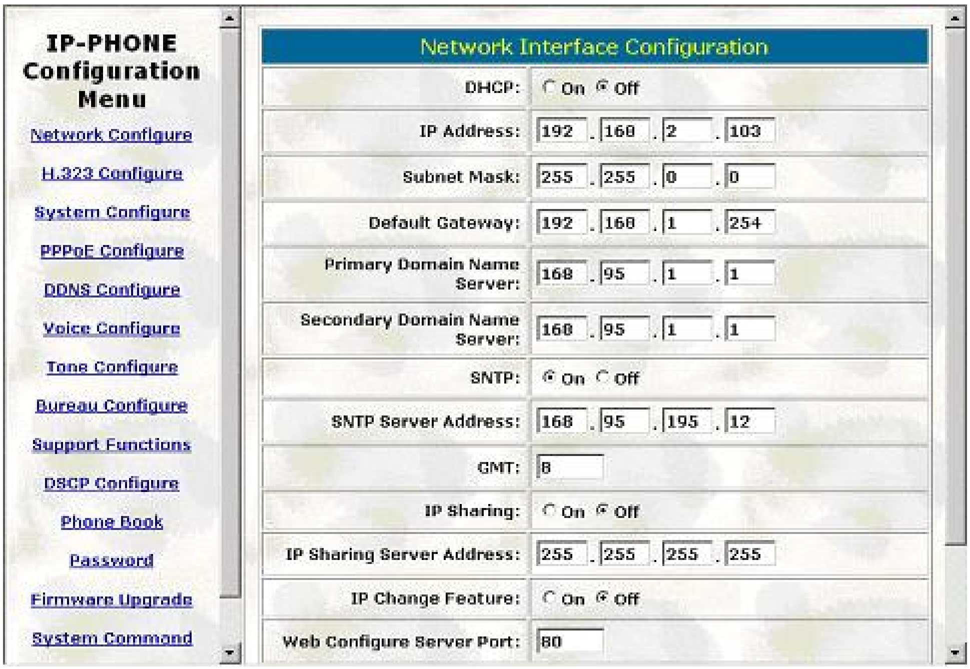
Step 4. Start configure

Most of all commands displayed in telnet are transfer to web interface. Most important items are Network Configure, H323 Configure, and System Command. Please remember to configure these commands before start to work with IP Phone.

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

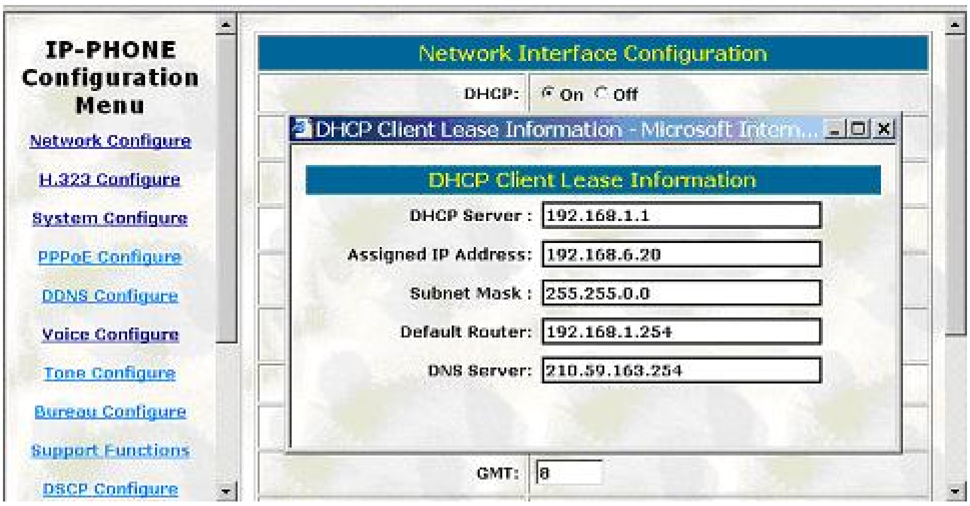
1. Network Configure

Please refer to chapter **4.6 [ifaddr] command.**



- DHCP: Enable / Disable to DHCP mode
  When DHCP function enables, IP Phone 530X will automatically search DHCP server after reboot.

Note: After IP Phone catches a dynamic IP address form DHCP server, user can see this IP address on LCD connect configuration. When user checks Network Configure via web interface, browser will pop up one small window for current DHCP network information.



- IP Address: Set IP Address of IP Phone
- Subnet Mask: Set the Subnet Mask of IP Phone
- Default gateway: Set Default routing gateway of IP Phone
- Domain Name Server: Set Domain Name Server IP address.
  User can set Domain Name Server IP address. Once IP Phone can connect with DNS server, user can specify URL address instead of IP address for Gatekeeper and phone book IP address. (Please refer to **4.13 [h323]** command and **4.9 [pbook] commands**)

- SNTP: Enable / Disable the Simple Network Time Protocol function
- SNTP Server Address: Set SNTP Server Address
  When SNTP server is available, enable IP Phone 530X SNTP function to point to SNTP server IP address so that IP Phone can get correct current time.

- GMT: Set time zone for SNTP Server time
  User can set different time zone according to the location of IP Phone. For example, in Taiwan the time zone should be set as 8,which means GMT+8.

- IP Sharing: Enable it if IP Phone is behind IP Sharing router.
- IP Sharing Server Address: Set Public IP Address of IP Sharing router if it is a fixed one.
- IP Change Feature: enable/disable IP change Function

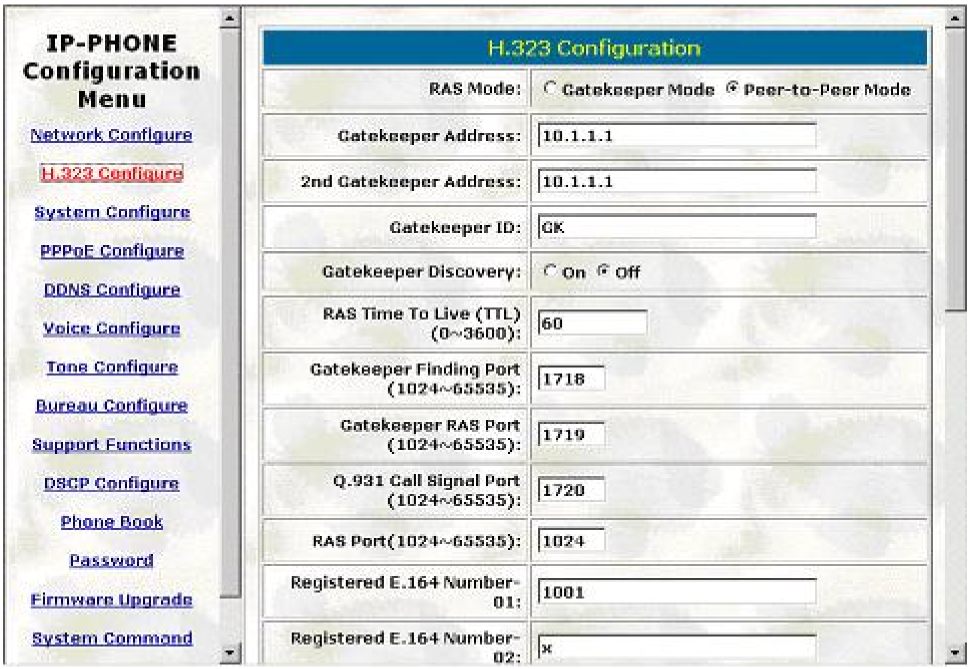
Note:

1. If user uses NAT device which supports multiple public IP address, and IP Phone enables IP change function, IP Phone will register to Gatekeeper with the public IP IP Phone uses but not the IP NAT uses.
2. If IP Phone 530X is behind an IP-sharing, user must enable IP sharing function.
3. If public IP address of NAT is fixed, please set the address in columns of IP Sharing Server Address, if it is dynamic, IP Phone must register to specific vendor of GK and no need to set IP Sharing Server Address. Please contact with your vendor for more information.

- Web Configure Server Port: set http port for configuration via web browser
  User can configure IP Phone via web browser, default http port is 80, if port 80 is not available or user has more than 1 IP Phone behind NAT, http port can be changed to another available port.

2. H323 Configure

Please refer to chapter **4.13 [h323] command**



- Mode: Select GK mode or Peer-to-Peer mode
- Gatekeeper IP Address: Set Gatekeeper IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Configure**). User can also set IP as 255.255.255.255 and let IP Phone auto discovery for Gatekeeper. Please notice that in this case user must enable Gatekeeper discovery function. Please refer to Gatekeeper ID and Gatekeeper Discovery.

- 2nd Gatekeeper IP: Set IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Configure**) of alternative Gatekeeper. If IP Phone tries to register to main Gatekeeper for 10 times but still failed, IP Phone will try to register to alternative Gatekeeper.

- Gatekeeper ID: Set Gatekeeper ID (Gatekeeper Name)
- Gatekeeper Discovery: Enable/Disable GRQ function
  Set Gatekeeper name for Gatekeeper discovery. When IP Phone send out Gatekeeper discovery message will search Gatekeeper with this Gatekeeper name.

- RAS Time To Live (TTL)(0~3600): set RAS TTL time (0-3600 second). IP Phone will keep re-registering to GK before TTL timed.

- Gatekeeper Finding Port (1024~65535): assign Gatekeeper finding port number (1024-65535)
- Gatekeeper RAS Port (1024~65535): assign Gatekeeper RAS port (1024-65535).
- Q.931 Call Signal Port (1024~65535): assign Q.931 port for call signaling.
- RAS Port (1024~65535): assign RAS port.
- Registered E.164 Number-01: Set 1st phone number
- Registered E.164 Number-02: Set 2nd phone number
- Registered E.164 Number-03: Set 3rd phone number
- Registered E.164 Number-04: Set 4th phone number
- Registered E.164 Number-05: Set 5th phone number
- Registered E.164 Number-06: Set 6th phone number
- Registered E.164 Number-07: Set 7th phone number
- Registered E.164 Number-08: Set 8th phone number
- Registered E.164 Number-09: Set 9th phone number
- Registered E.164 Number-10: Set 10th phone number
  Identify one number for the IP Phone 530X to register to the Gatekeeper. IP Phone can set up to 10 sets of phone numbers, if user wants IP Phone to register 10 different numbers for one set of IP Phone.

- Registered H.323 ID: Set Registered Alias as H323 ID.
  Identify ID for the IP Phone 530X to register to the Gatekeeper. The default alias is related to MAC address of IP Phone, so each IP Phone has different alias.

Note: Under GK mode, each IP Phone must has different H.323 ID and E.164 number to register to GK.

- Token Password: Set Token password for H.235 security or LCD menu use. User can clean this password via web configuration, so that will be no need to enter any password when entering LCD menu.

(A) LCD menu password: User can enter LCD system configuration by key in this password and default value is lowercase “x.” (Press TRANSFER to switch lowercase and uppercase, then press 9 twice).
(B) H.235 security: To set RRQ/ARQ authentication token password. If IP Phone wants to register to a Gatekeeper, which implement H.235 security token feature, IP Phone has to set a RRQ/ARQ authentication token password, which is provided by Gatekeeper manager. IP Phone can’t work normally with this Gatekeeper unless Token Password is set.

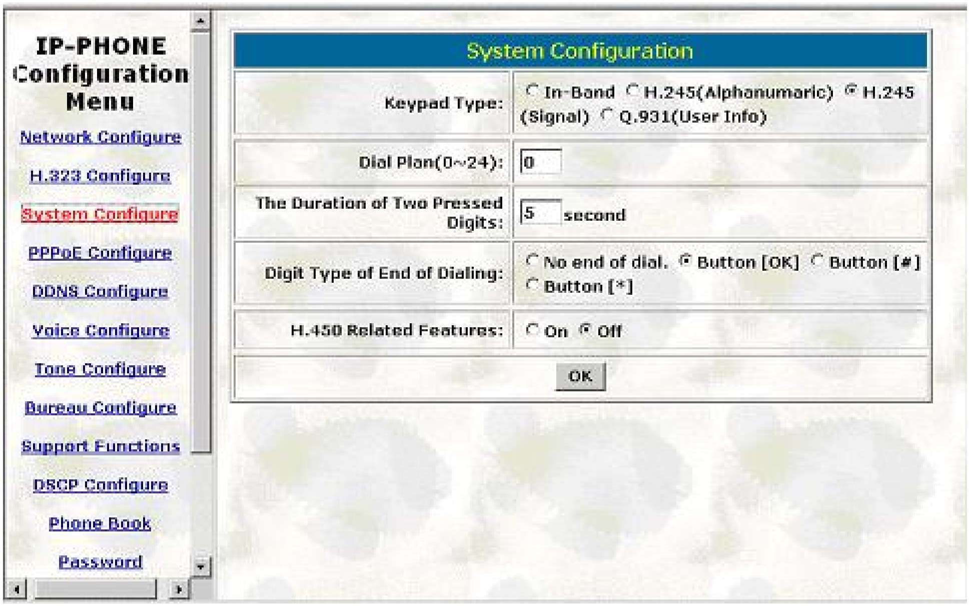
- RTP Port: Assign RTP port number (1024-65535). IP Phone need 2 RTP ports for voice communication, the port number assigned here is a base, for example, default value is 16384, that means IP Phone will use port 16384 and 16385 for RTP packets.

- Response Timeout: set max waiting time for first response to a new call. After dial phone number without getting response in max waiting time, user will hear busy tone. (1-200 seconds)

- Connection Timeout: set max waiting time for call establishment after receiving first response of a new call (1-20000 seconds).

3. System Configure

Please refer to chapter **4.12 [sysconf] command**



- Keypad Type: set DTMF type. User can select DTMF type IP Phone receive and transmit.

- Dial Plan: Set dialed DTMF digit limitation (0 is for any digits)
  It is for setting dial-numbering plan. While user will dial out e164 number only for three digits, the plan can be set as 3. Once input 3 digits, IP Phone will immediately dial out. The plan 0 is for any possible digits use, when after inter digit timed, IP Phone will dial out the number no matter how many digits user input.

Note: Before change to Peer-to-Peer mode from GK mode, please remember to set dial plan as 0, or it may works not normally in P2P mode.

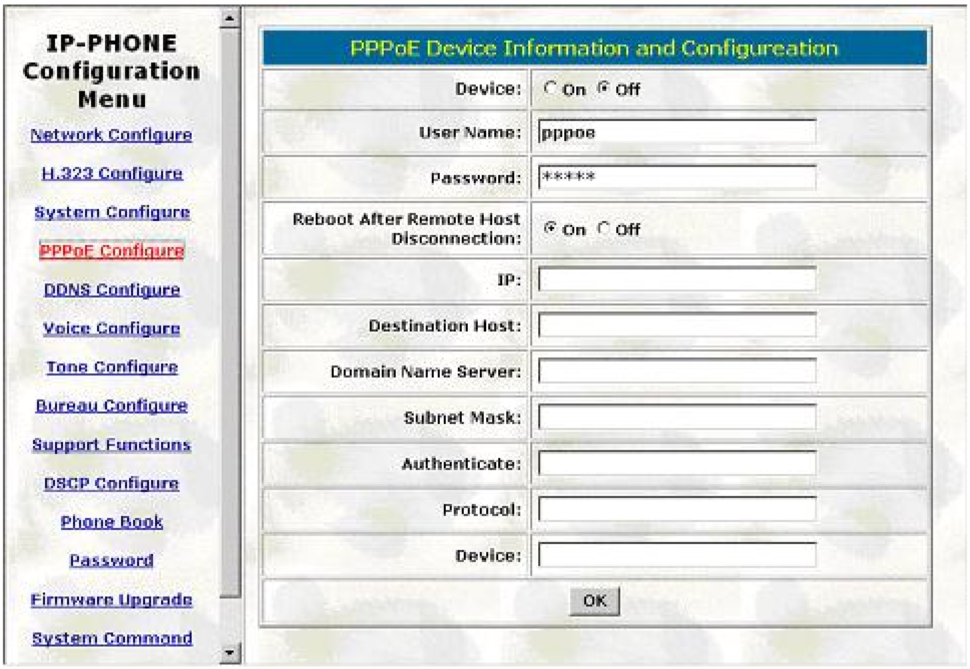
- The Duration of Two Pressed Digits: Set the DTMF inter digit time (second)
  To set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, IP Phone will dial out all number pressed.

- Digit Type of End of Dialing: select end of dialing key, e.g. set end of dial key as **OK** button, after finished pressing dialing number then press **OK** will dial out.
- H.450 Related Features: Enable/Disable H.450 Button functions, which include

transfer, hold and forward. Please Note that disable this function will cause **TRANSFER** button lost function of switching characters to be lower or upper case.

4. PPPoE Configure

Please refer to chapter **4.11[pppoe] command**

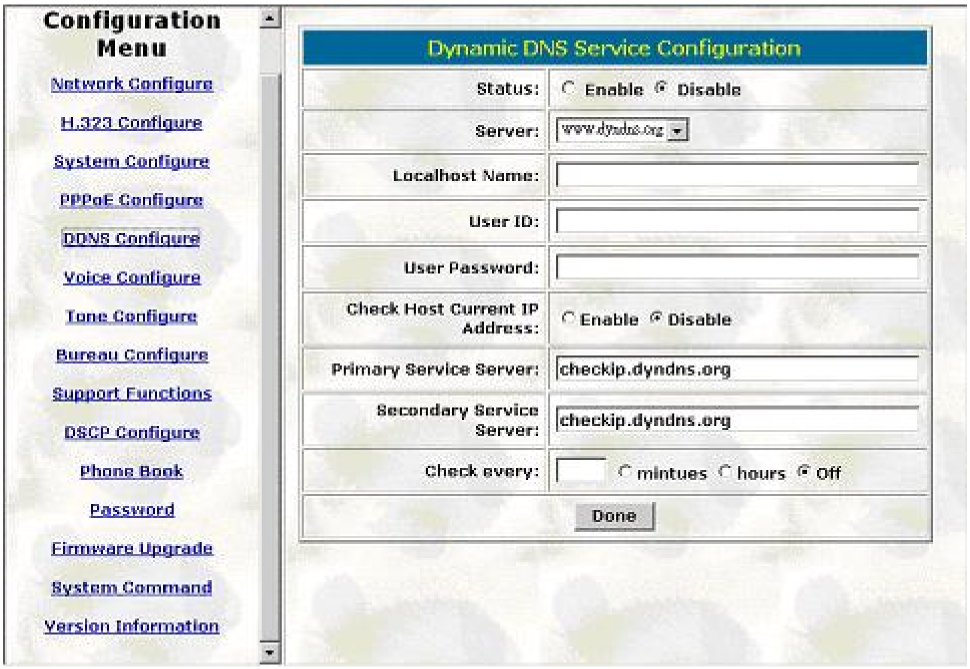


- Device: Enable/Disable PPPoE function
- User Name: Set PPPoE authentication User Name.
- Password: Set PPPoE authentication password.
- Reboot After Remote Host Disconnection: Enable/Disable auto reboot after PPPoE disconnection

  If user enables this function, after PPPoE being disconnected, IP Phone will automatically reboot to re-connect, and after reboot, if IP Phone still can't get contact with server, IP Phone will keep trying to connect. After re-connected, IP Phone will also restart system. On the other hand, if user disables this function, IP Phone won't reboot and keep trying to connect.
- Other items: After PPPoE connection established, related information will be displayed

5. DDNS Configure

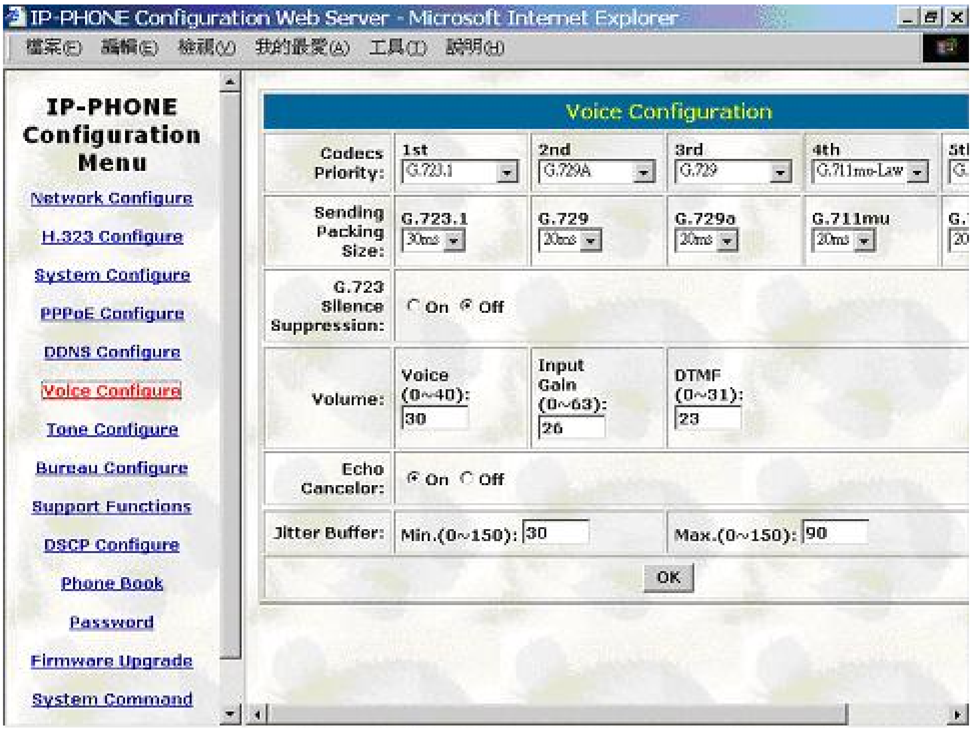
Please refer to chapter **4.10 [ddns] command**



- Status: to enable/disable DDNS function
- Server: to choose one DDNS server, on which user has already registered. (Now only one DDNS server is available---www.dyndns.org)
- Localhost Name: to set the registered Domain Name of IP Phone
- User ID: to set login ID of registered account to log in DDNS server
- User Password: set password of registered account to log in DDNS server
- Check Host Current IP Address: to enable/disable check IP function. If IP Phone is behind IP sharing, when this function is enabled, IP Phone will check it's public IP address by asking IP address check server and send to DDNS server to update DDNS data. If this function is disabled, when IP Phone is behind IP sharing, it will send it's private IP address to DDNS server
- Primary Service Server: to set IP address check server
- Secondary Service Server: to set secondary IP address check server
- Check every /minutes /hours off: to set the update interval time. IP Phone will re-update its IP address in this time.

6. Voice Configure

Please refer to chapter **4.14 [voice] command**



- Codec Priority: set codecs priority in order. Please notice that user can set from 1 to 7 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.
- Sending Packet Size: User can set different packet size for each codec. (For Advanced User)
- G.723 Silence Suppression: Enable / Disable sound compression and comfort noise generation. It is only for codec G.723.1 (For Advanced User)
- Volume: Adjust the volume in “Voice” (sending out); “Input” (receiving); “ DTMF” (DTMF sending out). Please Noted the value is limited.
- Echo Cancel: Enable / Disable (suggested always Enable this function).
- Jitter Buffer: Min. Delay and Max. Delay (For Advanced User)

7. Tone Configure

(For Advanced User)

Please refer to chapter **4.16 [tone] command**

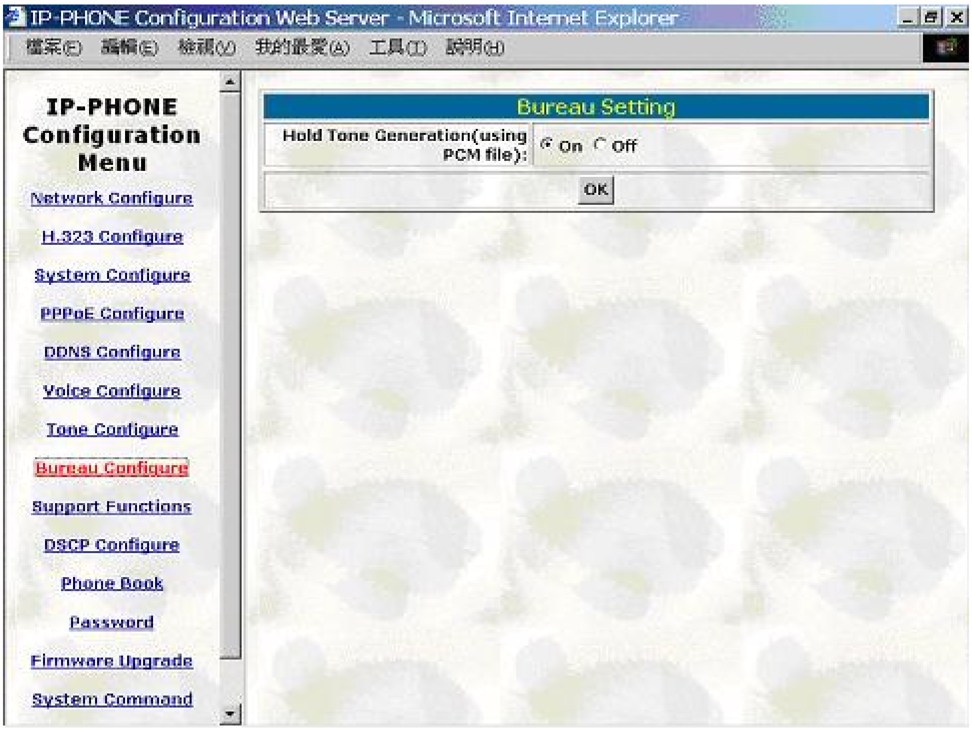
- Busy Tone
- Reorder Tone
- Ring Tone 1
- Ring Tone 2
- Dial Tone

IP Phone 530X is configurable of busy tone, reorder tone, ring tone and dial tone. However, only ring tone and dial tone is functional for now, busy tone and reorder tone are reserved for future feature.

User must key in 8 sets of number to finish this configuration. If it is single-frequency tone, please set high frequency and related items as 0. Furthermore, unit of on/off time is 1/100 second, and suggest keeping level as default value-8. User can also increase the value of level to increase the volume.

8. Bureau Configure

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

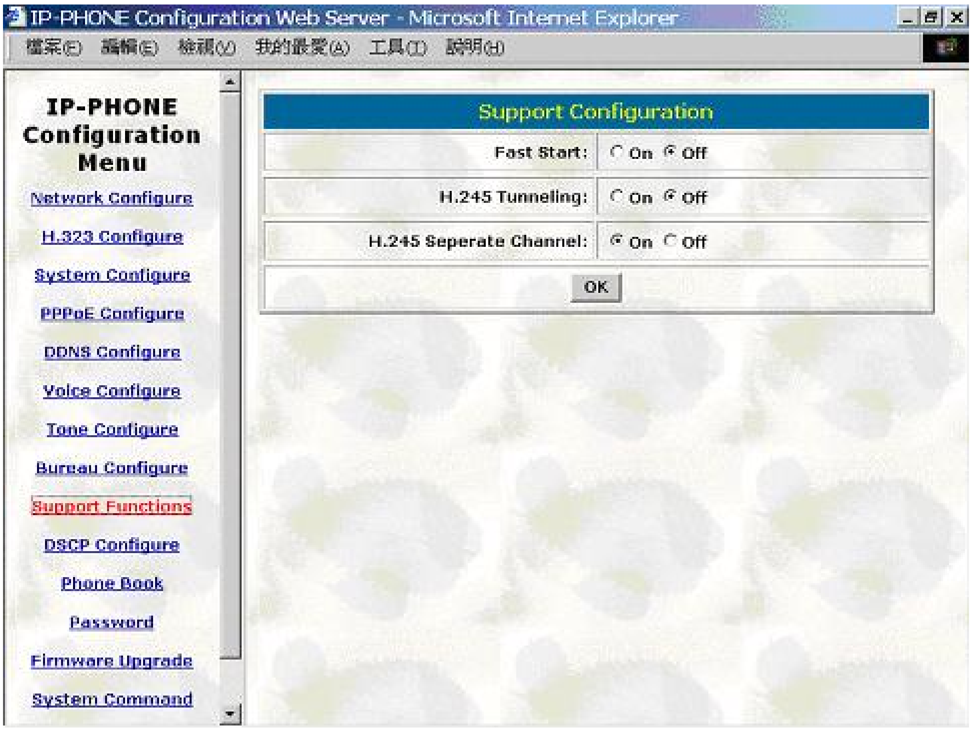


- Hold Tone Generation (using PCM file): Enable/Disable hold tone generation
  If other terminals support H.450 hold function, and execute hold function when connecting with IP Phone 530X, user will hear hold tone from IP Phone 530X.

9. Support Functions

(Function of this page will work only when 2 sides of communication both supports.)

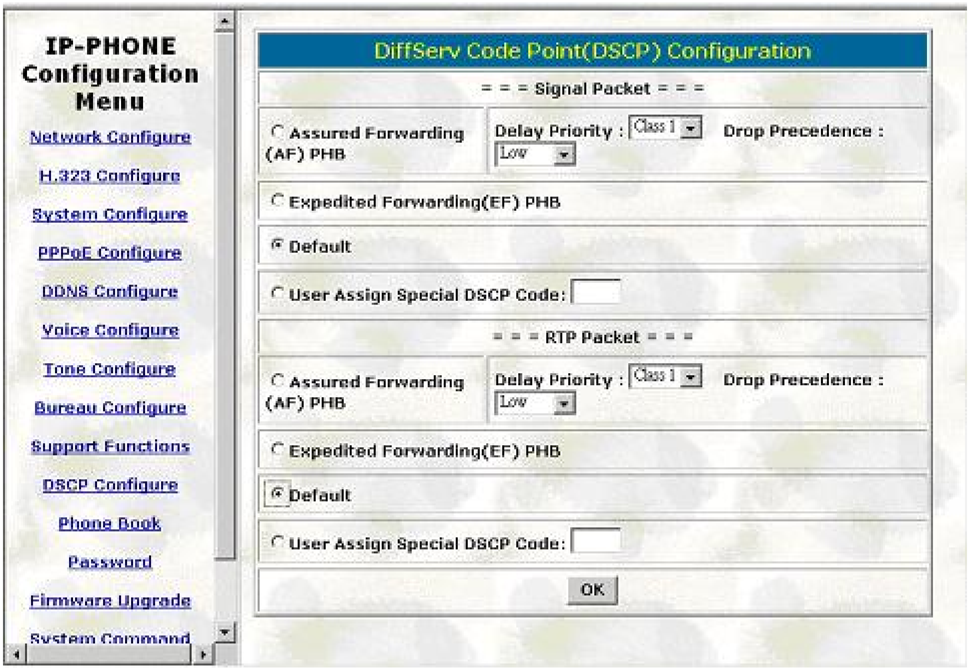
Please refer to chapter **4.17 [support] command**



- Fast Start: Enable/disable to do Fast Start function.
- H.245 Tunneling: Enable/disable H.245 Tunneling function.
- H.245 Separate Channel: Enable/Disable function of opening H.245 channel after fast start connection.

10. DSCP Configure

Please refer to chapter **4.15 [tos] command**



Set Signal or RTP Packet DSCP value:

- Assured Forwarding (AF) PHB: Select Delay priority and Drop Precedence
- Expedited Forwarding (EF) PHB: Select TOS value as EF
- Default: Select TOS value as 0
- User Assign Special DSCP Code: User can set other unspecified value here.

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Codepoint (DSCP) of the DS field in the IP packet header, and map each Codepoint 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 Codepoint is defined in RFC2597 to classify the traffic into different service classes. The mapping of Codepoint value of DS-field to egress traffic priorities is shown as follows.

DROP Precedence	**Class #1**	**Class #2**	**Class #3**	**Class #4**
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010

Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

Please refer to RFC standard documents for more information about what is DSCP.

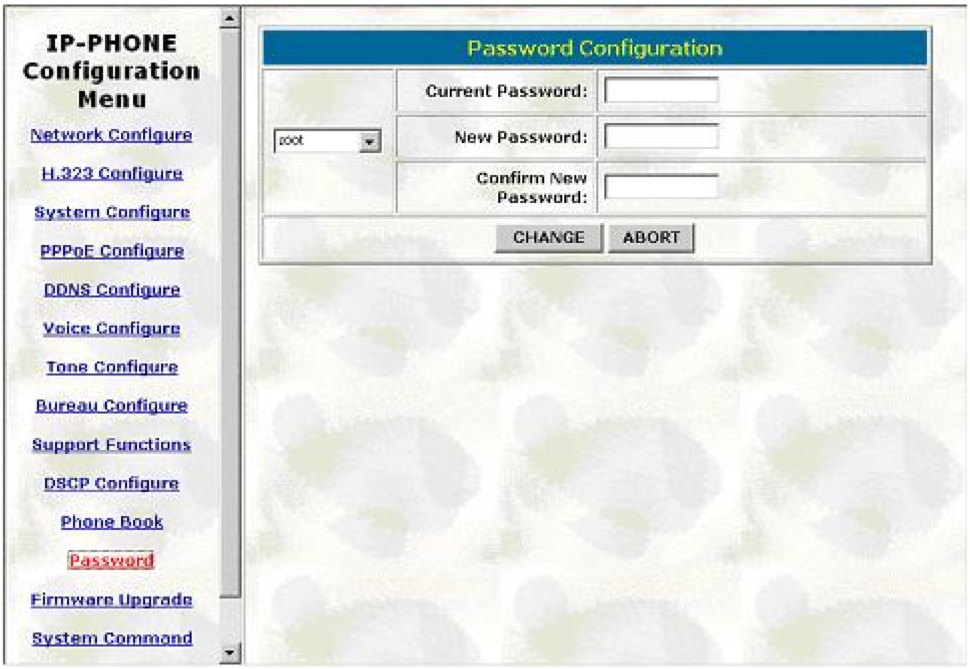
11. Phone Book

Please refer to chapter **4.9 [pbook] command**

- Add Data: User can specify 20 sets of phone book via web interface. Please input index, Name, IP Address and E.164 number of the destination device.
- Delete Date: User can delete any configured phone book data by index.

12. Password

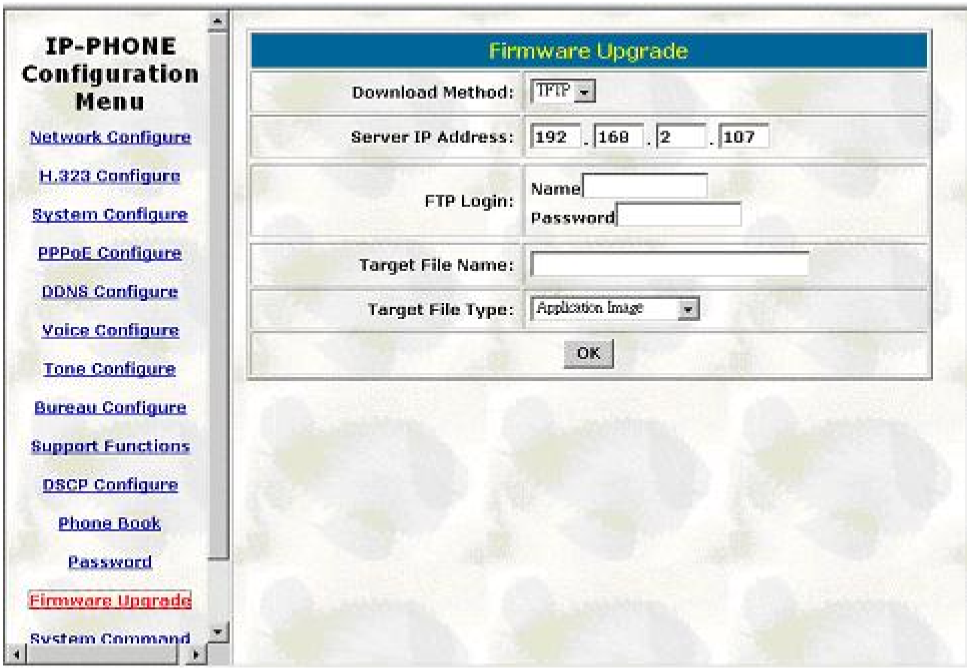
Please refer to chapter **4.20 [password] command**



- Change: First select login name as root or administrator, then enter current password, new password and confirm new password again to set new password.
- Abort: Press abort will clean all inputs.

13. Firmware Upgrade

Please refer to chapter **4.19 [rom] command**

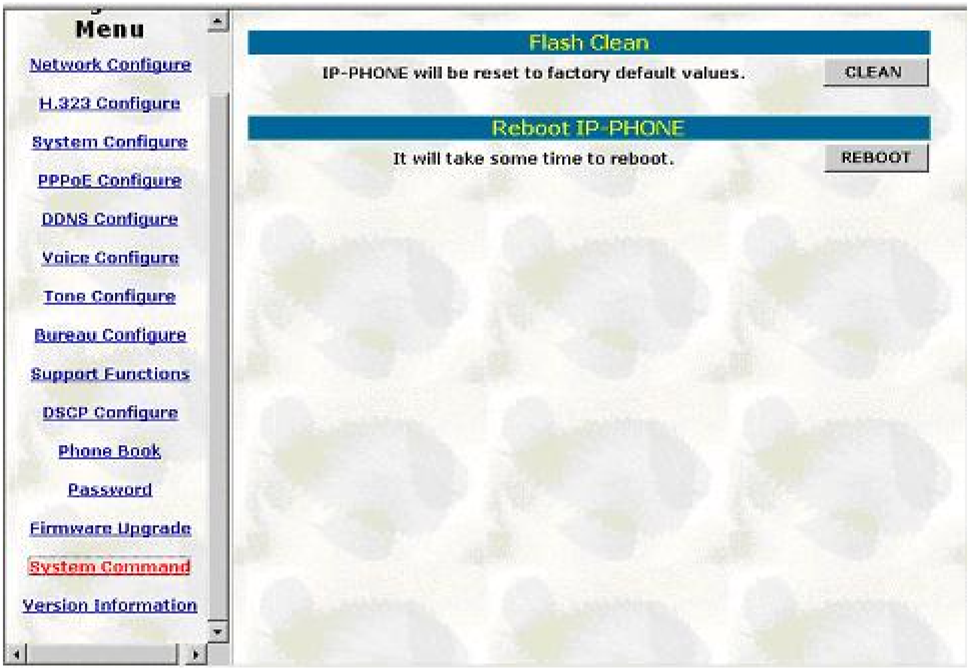


- Download Method: Select download method as TFTP or FTP
- Server IP Address: Set TFTP server IP address
- FTP Login: Set FTP login name and password
- Target File name: Set file name prepared to upgrade
- Target File Type: Select which sector of IP Phone to upgrade

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** (under **System Command**), which will clean all configurations become factory values except IP address.

14. System Command



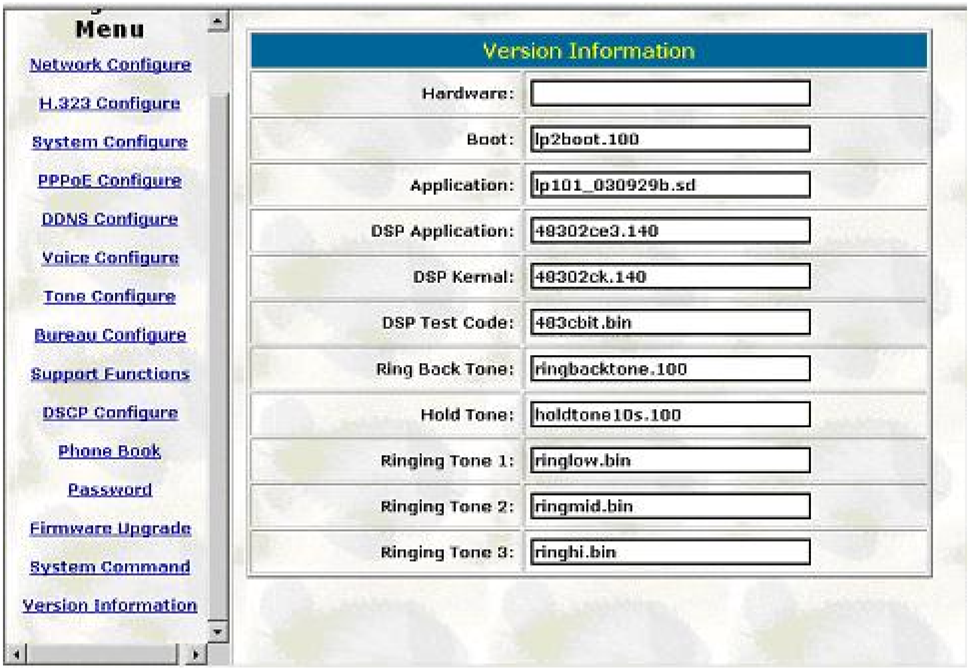
- Press CLEAN will clean all configurations of IP Phone and reset to factory default value.

Note: User must re-configure all commands all over again (except Network Configure) once execute this function,

- Press reboot will reset IP Phone.

Note: To execute reboot via web browser, IP Phone will automatically save all data before reboot. To execute reboot via TELNET command, please remember to do **commit** before **reboot**.

15. Version Information



Display all current version lists of hardware and software.

Chapter 4 Advanced Configurations via Telnet

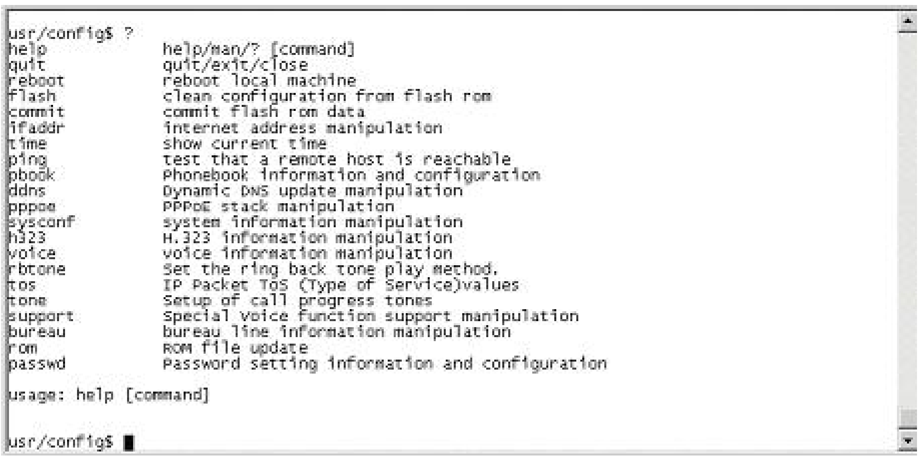
After initializing the IP Phone 530X IP Address setting (please refer to LCD Configuration: 1.Connect Configuration), user can enter into configuration mode via telnet.

Note:

1. After user enter IP Phone configuration via telnet, please use login: ”root”, password: null, press enter to enter command line.
2. Each command user must key-in with lower case, but contents of configurations such as h.323 alias or user name etc, user can set as capital case.
3. User who changes any configuration needs to do the **commit** command then **reboot** command.

1. [help] command

Type **help** or **man** or **?** to display all the command lists. The following figure is shown all commands of IP Phone 530X.



2. [quit] command

Type **quit/exit/close** will logout IP Phone 530X and Telnet Program.

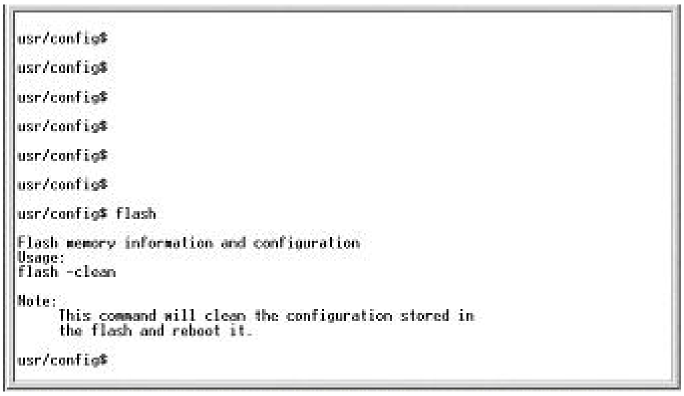
3. [reboot] command

After typing **commit** command, type **reboot** to restart the IP Phone 530X.
Sometimes after user type reboot, on terminal screen will display:"Data modified, commit to flash rom?", which means IP Phone will record call history or not.(ex. REDIAL, outgoing and incoming call data)

4. [flash] command

This command will clean the configuration stored in the flash rom to default value and reboot the IP Phone 530X.
Note: After user upgrade new software version, suggested to execute this command to make sure new software work well on IP Phone 530X.

Note: To execute the command **flash –clean,** all configuration of IP Phone 530X stored in flash will be cleaned. It is authorized for the user whose login name is “root” only.



```
usr/config$
usr/config$
usr/config$
usr/config$
usr/config$
usr/config$
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$
```

5. [commit] command

Save any changes after configuring the IP Phone 530X.

```
quit            quit/exit/close
debug           show debug message
reboot          reboot local machine
flash           clean configuration from flash rom
commit          commit flash rom data
ifaddr          internet address manipulation
time            show currently time
ping            test that a remote host is reachable
sysconf         System information manipulation
h323            H.323 information manipulation
voice           Voice information manipulation
tone            Setup of call progress tones
bureau          Bureau line information manipulation
rom             ROM file update

usage: help [command]

usr/config$ commit

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

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

6. [ifaddr] command

Configure and display the IP Phone 530X IP information.

1. **-ip, -mask, -gate**: Set IP Phone 530X IP Address, subnet mask and default gateway respectively.
2. -**dhcp**: When DHCP function enables (**ifaddr –dhcp 1**), IP Phone 530X will automatically search DHCP server after execute the **commit** and **reboot** command. Note: After IP Phone catches a dynamic IP address form DHCP server, user can see this IP address on LCD connect configuration.
3. -**sntp**: When sntp server is available, enable IP Phone 530X SNTP function and point to sntp server IP address. **(ifaddr –sntp 1 "xxx.xxx.xxx.xxx**")
4. -**dns**: User can set Domain Name Server IP address. Once IP Phone can connect with DNS server, user can specify URL address instead of IP address for Gatekeeper and phone book IP address. (Please refer to **4.13 [h323] command** and **4.9 [pbook] command**)
5. **–timezone**: User can set different time zone according to the location IP Phone is. For example, in Taiwan the time zone should be set as 8,which means GMT+8. (GMT-8: **ifaddr –timezone -8**)
6. **-ipsharing**: If IP Phone 530X is behind a IP-sharing , user can 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. **–ipchange**: If user uses NAT device which supports multiple public IP address, and IP Phone enables IP change function, IP Phone will register to Gatekeeper with the public IP IP Phone uses but not the IP NAT uses.

8. **–http**: set http port. User can configure IP Phone via web browser, default http port is 80, if port 80 is not available or user has more than 1 IP Phone behind NAT, http port can be changed to another available port.

```
usr/config$

usr/config$ ifaddr
LAN information and configuration
Usage:
ifaddr [-print]|[-dhcp used]|[-sntp mode [server]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway] [-cmcenter cmcenter]
ifaddr [-dns index [dns server address]]

    -print      Display LAN information and configuration.
    -ip         Specify ip phone ip address.
    -mask       Set Internet subnet mask.
    -gate       Specify default gateway ip address
    -dhcp       Set DHCP client service flag (on/off).
    -sntp       Set SNTP server mode and specify IP address.
    -dns        specify IP address of DNS Server.
    -timezone   Set local timezone.
    -ipsharing  Specify usage of an IP sharing device and specify IP address.
    -ipchange   Set the change IP address feature.
    -http       Specify web configure server port.
Note:
    Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
    DHCP client setting value (on=1, off=0). If DHCP set to 'on',
    Obtain a set of Internet configuration from DHCP server assgined.
    SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).
Example:
    ifaddr -ip 192.168.0.1 -mask 255.255.255.0 -gate 210.59.163.254
    ifaddr -dhcp 1
    ifaddr -sntp 1 213.91.2.137
    ifaddr -ipsharing 1 210.66.155.66
    ifaddr -dns 1 168.95.192.1
    ifaddr -timezone 8
    ifaddr -http 8080

usr/config$
```

7. [time] command

When SNTP server is established as well as the SNTP function of IP Phone 530X is enabled, type **time** command should show the current time what is retrieved from the assigned SNTP server.

Note: Please refer to the Chapter **4.6 [ifaddr] command** to configure SNTP server.

```
usr/config$ time
Current time is MON MAR 18 10:49:32 2002

usr/config$
```

8. [ping] command

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

Usage: ping “xxx.xxx.xxx.xxx(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$
```

9. [pbook] command

The command is for Peer-to-Peer Mode use only. Therefore, VoIP products such as IP Phone 530X that support Peer-to-Peer Mode are also available to be addressed on the IP Phone 530X phone book.

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_record' and 'end_record' (**pbook –print "start prefix" "end prefix"**). If parameter 'end_record' is omitted, only record 'start_record' will be display (**pbook –print "start prefix")**.

2.**-add**: add a new record in phone book table by give a name and e164 number for the Gateway / Terminal IP address .
(**pbook –add name "X" ip "xxx.xxx.xxx.xxx" e164 "X"**)
User can set IP or URL address(Domain Name Server must be configured. Please refer to **4.6 [ifaddr] command**)

3. **-search**: search any record such as ip address, name and e164 addressed on the phone book.

4. **-delete**: delete a record with index listed in phone book table. (**pbook –delete "index number"**)

5. **-insert**: insert an record in specified index of phone book.

6. **-modify**: modify any record that has addressed to index number. The name, IP address and e164 number should be modified together in one **modify** command.
(**pbook –modify "index" name "X" ip "xxx.xxx.xxx.xxx" e164 "X"**)

Note: Please dial "#" after dial e.164 of pbook.

10. [ddns] command

This function is for Dynamic Domain Name Server service. Once user register to one DDNS server, he can specify domain name for the IP Phone. When IP Phone reboot, it will automatically update it's IP address to DDNS server. In this way, even IP Phone is using dynamic IP address, other endpoint can locate this IP Phone by its domain name.

1. -print: display DDNS overall information and configuration.
2. –enable: to enable/disable DDNS function.(**ddns –enable 0/1**, 0 for disable and 1 for enable).
3. -server: to set IP address of DDNS login server. (Now only one DDNS server is available---www.dyndns.org)
4. -hostname: to set the registered Domain Name of IP Phone. (Ex. **ddns –lp001.ddns.org**)
5. –id: to set login ID of registered account to log in DDNS server.
6. –passwd: to set password of registered account to log in DDNS server.
7. –checkip: to enable/disable check IP function. If IP Phone is behind IP sharing, when this function is enabled, IP Phone will check it's public IP address by asking IP address check server and send to DDNS server to update DDNS data. If this function is disabled, when IP Phone is behind IP sharing, it will send it's private IP address to DDNS server.
8. -checkipsrv: to set IP address of IP address check server.
9. -delay: to set the update interval time. IP Phone will re-update its IP address in this time. (**ddns –delay 1-59m/1-24h** , m means minute, h means hour)
10. –force: to force to execute DDNS update. Once user enters this command, IP Phone will update DDNS data immediately. (**ddns –force "IP address of IP Phone"**)

Note:

1. For now we only support DDNS server as www.dyndns.org and www.3322.org.
2. User must register to DDNS server first, and specify user name and password in **ddns -id** and **ddns –passwd**.
3. The default IP address of DDNS login server is member.dyndns.org and members.3322.org.
4. User has to specify domain name applied for IP Phone in **ddns –hostname**.
5. The default IP address of check IP server in is checkip.dyndns.org.

11. [pppoe] command

1. -print: display system overall information and configuration. IF IP Phone has already connected to PPPoE server, user can see IP address and related information with this command.
2. -dev: to enable or disable PPPoE function.(**pppoe –dev 0/1**)
3. –open: to open PPPoE connection(If IP Phone is not in PPPoE connection, user can try to connect with **pppoe -open**)
4. –close: to close PPPoE connection(If IP Phone is in PPPoE connection, user can disconnect with **pppoe -close**)
5. –id: to set PPPoE authentication user name.
6. –pwd: to set PPPoE authentication password.
7. –reboot: If user enable this function, after PPPoE being disconnected, IP Phone will automatically reboot to re-connect, and after reboot, if IP Phone still can't get contact with server, IP Phone will keep trying to connect. After re-connected, IP Phone will also restart system. On the other hand, if user disables this function, IP Phone won't reboot and keep trying to connect. (**pppoe –reboot 0/1**)

```
usr/config$ pppoe
PPPOE device information and configuration
usage:
pppoe [-print]|[-open]|[-close]
pppoe [-dev on/off][-id username][-pwd password]

    -print      Display PPPOE device information.
    -dev        Enable(=1) or Disable(=0) device.
    -open       Open PPPoE connection.
    -close      Disconnect PPPoE connection.
    -id         Connection user name.
    -pwd        Connection password.
    -reboot     Reboot after remote host disconnection.

usr/config$
```

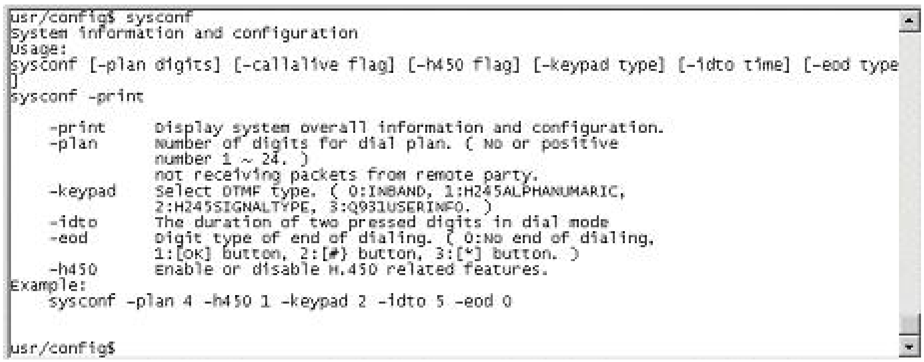
12. [sysconf] command

This command displays the system information and configuration.

1. **-print**: display system overall information and configuration.
2. **-plan**: It is for setting dial-numbering plan. While user will dial out e164 number only for three digits, the plan can be set as 3. Once input 3 digits, IP Phone will immediately dial out. The plan 0 is for any possible digits use, when after inter digit timed, IP Phone will dial out the number no matter how many digits user input.

Note: Before change to Peer-to-Peer mode from GK mode, please remember to set dial plan as 0, or it may works not normally in P2P mode.

3. **-keypad**: set DTMF type .User can select DTMF type IP Phone receive and transmit.(**sysconf –keypad 0/1/2/3 , 0 for in band ,1 for H.245 alphanumeric, 2 for H.245 signal type, 3 for Q.931 user info, 4 for RFC2833.**)
4. **-idto**: 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, IP Phone will dial out all number pressed.
5. **-eod**: select end of dialing key, e.g. set end of dial key as **OK** button , after finished pressing dialing number then press **OK** will dial out. (**sysconf –eod 0/1/2/3** , 0 for no end of dial key , 1 for "OK" button , 2 for "**#**" button , 3 for "＊" button)
6. **-h450**: Enable or disable H.450 related features, which include transfer, hold and forward. Please Note that disable this function will cause **TRANSFER** button lost function of switching characters to be lower or upper case.



```
usr/config$ sysconf
System information and configuration
Usage:
sysconf [-plan digits] [-callalive flag] [-h450 flag] [-keypad type] [-idto time] [-eod type
]
sysconf -print

    -print      Display system overall information and configuration.
    -plan       Number of digits for dial plan. ( No or positive
                number 1 ~ 24. )
                not receiving packets from remote party.
    -keypad     Select DTMF type. ( 0:INBAND, 1:H245ALPHANUMARIC,
                2:H245SIGNALTYPE, 3:Q931USERINFO. )
    -idto       The duration of two pressed digits in dial mode
    -eod        Digit type of end of dialing. ( 0:No end of dialing,
                1:[OK] button, 2:[#} button, 3:[*] button. )
    -h450       Enable or disable H.450 related features.
Example:
    sysconf -plan 4 -h450 1 -keypad 2 -idto 5 -eod 0

usr/config$
```

13. [h323] command

1. **–print**: display H.323 stack information and configuration.
2. **–mode**: configure IP Phone 530X as Gatekeeper or Peer-to-Peer Mode.
   Usage: **h323 –mode 0/1**(0 for Gatekeeper mode, 1 for Peer-to-Peer mode)
3. **–gk**: set Main Gatekeeper IP address**(h323 –gk "xxx.xxx.xxx.xxx")** or URL address(Domain Name Server must be configured. Please refer to **4.6 [ifaddr] command**)**.** User can set IP as 255.255.255.255 and let IP Phone auto discovery for Gatekeeper. Please notice in this case that user must enable Gatekeeper discovery function. Please refer to **6. -gkname** and **11.-gkdis**.
4. **–dfgw**: to set IP address of default gateway, this function is the same as Microsoft NetMeeting.
   A. To implement this feature both calling and called endpoints must be under peer-to-peer mode.
   B. If the called party is FXO products, such as PRO 5373, which Have to set **sysconf –2nddial 0** to make one-stage dialing.
      - Dial remote PSTN number under default gateway, IP Phone will

automatically dial to default gateway, then default gateway will dial this number to PSTN side.

- For example, user wants to dial to ext.888 under PRO 5373, user only have to dial 888 from IP Phone.

C. If called party are FXS products such as PRO 5371 : user can dial line number of PRO 5371 from IP Phone.

- For example ,user wants to dial to PRO 5371，the configuration of 5371 is **h323 –line1 530X –line2 102**，user can press 530X or 102 dialing to line1 or line2 of PRO 5371.

5. **–algk**: set IP address or URL address(Domain Name Server must be configured. Please refer to **4.6 [ifaddr] command**) of alternative Gatekeeper. If IP Phone tries to register to main Gatekeeper for 10 times but still fail, IP Phone will try to register to alternative Gatekeeper.
6. **–gkname**: set Gatekeeper name for Gatekeeper discovery. When IP Phone send out Gatekeeper discovery message will search Gatekeeper with this Gatekeeper name.(please refer to **11.-gkdis**)
7. **–e164**: identify one number for the IP Phone 530X to register to the Gatekeeper **(h323 –e164 "X").**
8. **–e164-x**:user can assign other 10 telephone numbers .For example, 10 users share the same IP Phone, they can assign phone numbers as 100, 200, 300….(**h323 –e164 100 –e164-1 200 –e164-2 300….**) User can disable one number and the number after this one. Ex. from set 1-5 is configured, if user set the third number as "x", from third to fifth number will be disabled at the mean time. (ex. **h323 –e164-2 x**)
9. **–alias**: identify ID for the IP Phone 530X to register to the Gatekeeper **(h323 –alias "X").**.The default alias is related to MAC address of IP Phone, so each IP Phone has different alias.
10. **–tokenpwd**: To set RRQ/ARQ authentication token password.(**h323 –tokenpwd "password" ; h323 –tokenpwd x** to disable this function**)**

(A) LCD menu password: User can enter LCD system configuration by key in this password and default value is lowercase "x." (press TRANSFER to switch lowercase and uppercase).
(B) H.235 security: To set RRQ/ARQ authentication token password. If IP Phone wants to register to a Gatekeeper, which implement H.235 security token feature, IP Phone has to set a RRQ/ARQ authentication token password, which is provided by Gatekeeper manager. IP Phone can't work normally with this Gatekeeper unless Token Password is set.

11. **–gkdis**: set auto discovery function on or off. If this function is enabled and IP address of Gatekeeper is set as 255.255.255.255, IP Phone will multicast to search a Gatekeeper on network with configured Gatekeeper name (please refer to **6. -gkname**); if IP address of Gatekeeper is set, before IP Phone register to the assigned Gatekeeper, it will send out GRQ (Gatekeeper Request) message with configured Gatekeeper name to Gatekeeper first.

12. **–rtp**: assign RTP port number(1024-65535). IP Phone need 2 RTP ports for voice communication, the port number assigned here is a base, for example, default value is 16384, that means IP Phone will use port 16384 and 16385 for RTP packets.
- **–ttl**: set RAS TTL time (0-3600 second). IP Phone will keep re-registering to GK before TTL timed.
13. **–gkfind**: assign Gatekeeper finding port number(1024-65535)
14. **–ras**: assign Gatekeeper RAS port(1024-65535)
15. **–range**: assign dynamically allocated port range(1500-65535)
16. **–respto**: set max waiting time for first response to a new call. After dial phone number without getting response in max waiting time, user will hear busy tone.(1-200 seconds)
17. **–connto**: set max waiting time for call establishment after receiving first response of a new call (1-20000 seconds).
18. **–q931**: assign Q.931 port for call signaling.
19. **–ras**: assign RAS port.

Note: 1.Items from 9-20 are for advanced user only.
2.In Peer-to-Peer mode, **h323 –print** will only display e164, alias, mode, RTP port, and allocated port range.
3.In P2P mode, please dial "#" after press IP address (ex.10.1.1.1 please dial 10*1*1*1#) or e.164 of Phone book (Please refer to chapter **4.9 [pbook] command**).

```
usr/config$ h323
H.323 stack information and configuration
Usage:
h323 [-mode gkmode]
h323 [-gk ipaddress] [-algk ipaddress] [-gkdis used] [-e164 number] [-e164-x number]
     [-alias h323id] [-tokenpwd password]
     [-rtp port] [-ttl time] [-gkfind port] [h225 port]
     [-range [start num1] [end num2]] [-respto t1] [-connto t2] [-dfgw IP]
h323 -print

    -print      Display H.323 stack information and configuration.
    -mode       Configure as Gatekeeper mode or Non-Gatekeeper mode.
    -gk         Gatekeeper ip address. (0.0.0.0 ~ 255.255.255.255)
    -dfgw       Default Gateway IP Address. (0.0.0.0 - 255.255.255.255)
    -algk       Second Gatekeeper ip address. (0.0.0.0 - 255.255.255.255)
    -gkname     Gatekeeper ID
    -e164       IP side registered number.
    -e164-x     IP side registered number.(x:1~9)
    -alias      IP side registered H323 ID.
    -tokenpwd   RRQ/ARQ authentication token password.
    -gkdis      Gatekeeper auto discovery (multicast, On=1, Off=0).
    -rtp        RTP port number (1024-65532).
    -ttl        RAS TTL time (0~3600 second).
    -gkfind     Gatekeeper finding port (1024~65535).
    -h225       Gatekeeper RAS port (1024~65535).
    -range      Dynamically allocated port range (1024~65535).
    -respto     Max waiting time for 1st response to a new call (1~200).
    -connto     Max waiting time for call establishment after receiving 1st
    -q931       Q.931 call signal port
    -ras        RAS port
                response of a new call (1~20000).
Note:
Options -gk -e164 -alias -gkdis -ttl -gkfind -h225 are ignored when
    RAS mode is configured as Non-GK mode.
Example:
    h323 -gk 210.66.155.88 -e164 101 -alias IP-PHONE
    h323 -mode 1
```

14. [voice] command

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

1. **-print**: display voice codec information and configuration.
   There are five voice codecs included in IP Phone 530X:G.723.1, G.711u, G.711A, G.729a, G.729, G.729ab, G.729b.
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)
3. **-priority**: set codecs priority in order. Please notice that user can set from 1 to 7 codecs as their need, for example, **voice –priority g723** or **voice –priority g723 711a g711u g729a g729** means IP Phone can support only one codec or five codecs.
4. **-volume**: There are three types can be adjustable, voice volume, input gain and DTMF volume.
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)
6. **-echo**: echo canceller can be made to each specified port. The default value is on to 6 ports.
7. **-mindelay**:set minimum delay of jitter buffer(0~150)
8. **-maxdealy**:set maximum delay of jitter buffer(0~150)

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:
voice [-send [G723 ms] [G729 ms] [G729A ms] [G729B ms] [G729AB ms] [G711U ms] [G711A ms] ]
      [-volume [voice level] [input level] [dtmf level]]
      [-nscng G723 used] [-echo used] [-mindelay/maxdelay used]
voice -print
voice -priority [G723] [G729] [G729A] [G729B] [G729AB] [G711U] [G711A]

    -print      Display voice codec information and configuration.
    -send       Specify sending packet size.
                G.723   (30/60/90 ms)
                G.729   (20/40/60 ms)
                G.729A (20/40/60 ms)
                G.729B  (20/40/60 ms)
                G.729AB (20/40/60 ms)
                G.711U (20/40/60 ms)
                G.711A (20/40/60 ms)
    -priority   Priority preference of installed codecs.
                G.723
                G.729
                G.729A
                G.729B
                G.729AB
                G.711U
                G.711A
    -volume     specify the following levels:
                voice volume (0-40, default: 30),
                input gain (0-35, default: 26),
                dtmf  volume (0-31, default: 27),
    -nscng      Silence suppression and CNG. (G.723.1 only, On=1, Off=0).
    -echo       Setting of echo canceller. (On=1, Off=0, per port basis).
    -mindelay   Setting of jitter buffer min delay. (0-150, default: 100).
    -maxdelay   Setting of jitter buffer max delay. (0-150, default: 150).
Example:
    voice -send g723 60 g729 60 g729a 60 g729b 60 g729ab 60 g711u 60 g711a 60
    voice -volume voice 20 input 32 dtmf 27
    voice -echo 1 1
```

14. [rbtone] command

1. **-print**: display rbtone information and configuration.
2. **-mode**: set ring back tone generation mode. 0 means IP Phone will always wait remote site sending ring back tone, 1 means IP Phone will automatically detect if IP Phone needs to generate ring back tone, 2 means IP Phone will always play local ring back tone.

15. [tos] command

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Codepoint (DSCP) of the DS field in the IP packet header, and map each Codepoint 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 Codepoint is defined in RFC2597 to classify the traffic into different service classes. The mapping of Codepoint value of DS-field to egress traffic priorities is shown as follows.

1. High priority with DS-field.

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

For example, to configure TOS at gateway via Telnet command:

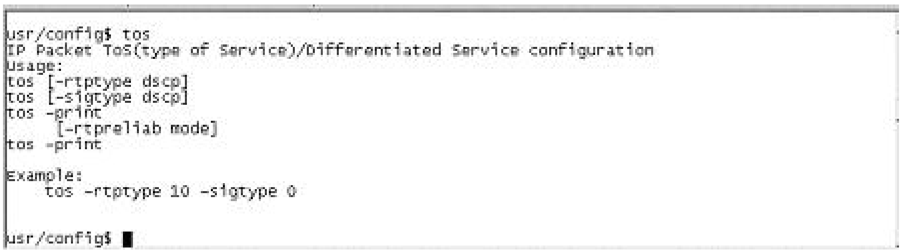
```
usr/config$ tos -rtptype 10
usr/config$ tos -sigtype 10

usr/config$ tos -print

IP Packet ToS information:
Signaling Packet:
      DSCP Code: 10        <====== Configure control signal DSCP code
Media Packet:
      DSCP Code: 10        <====== Configure RTP (voice) DSCP code
usr/config$
```

Or, the ToS function can be configured via Web Browser selection by entering the above DSCP Decimal Code.
This command is for setting IP packet TOS values to determine IP Packets priority on network.

1. **-print** : display current TOS values configurations.
2. **-sigtype**: configure DSCP value of signaling packets from 0 to 63
3. **-rtptype**: configure DSCP value of RTP packets from 0 to 63

Note:

1. This command won't be functional until network environment can be capable with TOS function.
2. tos -rtptype 14 -sigtype 10 is top priority of package.



16. [tone] command

IP Phone 530X is configurable of busy tone, reorder tone, ring tone and dial tone. However, only ring tone and dial tone is functional for now, busy tone and reorder tone are reserved for future feature.

Usage：tone –ringtone1/ringtione2/dialtone "low frequency" "high frequency" "low frequency level" "high frequency level" "low frequency on time" "low frequency off time" "high on time" "high frequency off time"; user must key in 8 sets of number to finish this configuration. If it is single-frequency tone, please set high frequency and related items as 0. Furthermore, unit of on/off time is 1/100 second, and suggest keeping level as default value (8). User can also increase the value of level to increase the volume.

17. [support] command

1. **-print** : display current SUPPORT values configurations.
2. **-fstart**: enable or disable fast start (**support –fstart 0/1** , 0 for disable and 1 for enable.)

Note: When fast start function is enabled, if user wants to send DTMF message after connection, IP Phone will send out Q.931 message. (Please refer to **4.11 sysconf –keypad** command, which can only set keypad as q.931 message at this time)

3. **-tunnel**: enable or disable H.245 tunnel function.(**support –tunnel 0/1** , 0 for disable and 1 for enable)
4. **-h245fs**:set if open H.245 separate channel after fast start or not. (**support –h245fs 0/1** , 0 for open and 1 for not.)

```
usr/config$ support
Special Voice function support manipulation
Usage:
support [-fstart enable] [-tunnel enable] [-h245fs enable]
support -print

    -fstart    Fast start enabled/disabled.
    -tunnel    H245 Tunneling enabled/disabled.
    -h245fs    H245 seperate channel after faststart.
Example:
    support  -fstart 1
    support  -tunnel 0
    support  -h245fs 1
usr/config$
```

18. [bureau] command

Type **bureau** can display commands below.

1. **-print**: display bureau line information and configuration.
2. **-hold**: set hold tone generation on or off. If other terminals support H.450 hold function, and execute hold function when connecting with IP Phone 530X, user will hear hold tone from IP Phone 530X. (0 for off,1 for on)

```
usr/config$ bureau
Bureau line setting information and configuration
Usage:
bureau [-hold used]
bureau -print

    -print     Display Bureau line information and configuration.
    -hold      Specify the hold tone generation (using PCM file). (On/Off)
               Setting value (On=1, Off=0).
Example:
    bureau -hold 1

usr/config$
```

19.[rom] command

1. **-print**: show versions of all rom files.
2. **-app**,**-boot**, **-dsptest**, **-dspcore**, **-dspapp**, **-rbpcm** and **-htpcm**: 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

Note:

1. After 2mb file download is finished, all configurations might change to default value, user has to configure again.
2. MAC address might change to default value also, please **MUST** use command ***setmac***:

   Usage: key in command ***setmac***

   key in MAC address with format: 0001a800xxxx

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. **–server**: specify TFTP/FTP server IP address. It is corresponding to LCD configuration -firmware upgrade-Set file Server IP.
7. **–method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
8. **–ftp**: specify user name and password for FTP download method

For example: User prepares to upgrade the latest app rom file – wtlp.103c, the TFTP server is 192.168.4.530X.

rom –app –s 192.168.1.1 –f lp.100(If –server is specified , can just type **rom –app –f lp.100**)

```
usr/config$ rom
ROM files updating commands
Usage:
rom [-print] [-boot] [-app] [-dsptest] [-dspcore] [-dspapp] [-rbpcm] [-htpcm]
     -s TFTP/FTP server ip -f filename
rom [-method mode] [-ftp username password] [-server serverIP]
rom -print
    -print      show versions of rom files. (optional)
    -boot       update main boot code(optional, only root user has authority.)
    -boot2m     update 2M code(optional, only root user has authority.)
    -app        update main application code(optional)
    -dsptest    update DSP testing code(optional)
    -dspcore    update DSP kernel code(optional)
    -dspapp     update DSP application code(optional)
    -rbpcm      update RingBack Tone PCM file(optional)
    -htpcm      update Hold Tone PCM file(optional)
    -s          IP address of TFTP/FTP server (mandatory)
    -f          file name(mandatory)
    -server     TFTP/FTP server IP address (store server IP in flash)
    -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', 'dsptest', 'dspcore',
 'dspapp', and 'rbpcm'.
Note:
    Once downloading server IP address is set via -server option,
    user can omit the -s option the next time when downloading.
    We keep -s option for backward compatibility.
Example:
    rom -method 1
    rom -ftp vvusr vvusr
    rom -server 192.168.4.101
    rom -app -f app.bin
```

Command **rom –print** can show current version installed in IP Phone 530X.

```
usr/config$ rom -print

Download Method  :  TFTP
 Server Address  :  192.168.2.107

   Hardware Ver. :  4.0
       Boot Rom  :  nblp-boot.102a
Application Rom  :  utlp.108f
        DSP App  :  48302ce3.127
     DSP Kernel  :  48302ck.127
  DSP Test Code  :  483cbit.bin
  Ringback Tone  :  wg-ringbacktone.100
      Hold Tone  :  wg-holdtone.101
  Ringing Tone1  :  ringlow.bin
  Ringing Tone2  :  ringmid.bin
  Ringing Tone3  :  ringhi.bin

usr/config$
```

20. [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. (**password –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”, two commands are not functional:

1. **password –set root**: set password of login : “root”.
2. **passwd –clean**: clean up password restored before, and user can login :”root/administrator”, password: ”press enter”.
3. **flash –clean:** only “root” users can clean all configurations stored in flash.
4. **rom –boot** :only “root” users can upgrade IP Phone boot rom version.
5. **rom –boot2m** : only “root” users can upgrade IP Phone 2mb firmware.

```
usr/config$ passwd
Password setting information and configuration
Usage:
passwd -set Loginname Password
passwd -clean
Note:
   1. Loginname can only be 'root' or 'administrator'.
   2. Only root user has authority to set root password.
   3. passwd -clean will clear all passwd stored in flash,
      please use it with care. (root user only)
Example:
   passwd -set root 1p101

usr/config$
```

Chapter 5 Upgrade the IP Phone 530X

IP Phone 530X 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, IP Phone 530X is ready to be upgraded.

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

∗Download Procedure

1.LCD Panel Control

1.Choose the 1→3 selection-**Firmware Upgrade.** Press **OK** to enter into the sub-selection as below.

2.Download method

There are two methods to download new rom file, please move the "→" symbol by press 🡐 or 🡒 on the keypad to select TFTP or FTP method, then press OK to confirm it.

3.Set File Server IP

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

4.Set FTP user account

User has to press user name and password for FTP server login. It is necessary for upgrading IP Phone new application rom file in FTP method.

5.Indicate file name

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

6.Start Download

Press OK to start download new application rom file.

7.Firmware Version

Show versions of all rom files.

Note:
1.Download via LCD command can only upgrade new ***application*** rom file.
2.If IP Phone fails to upgrade via LCD menu, IP Phone will automatically reboot.

2.Remote Control: Telnet

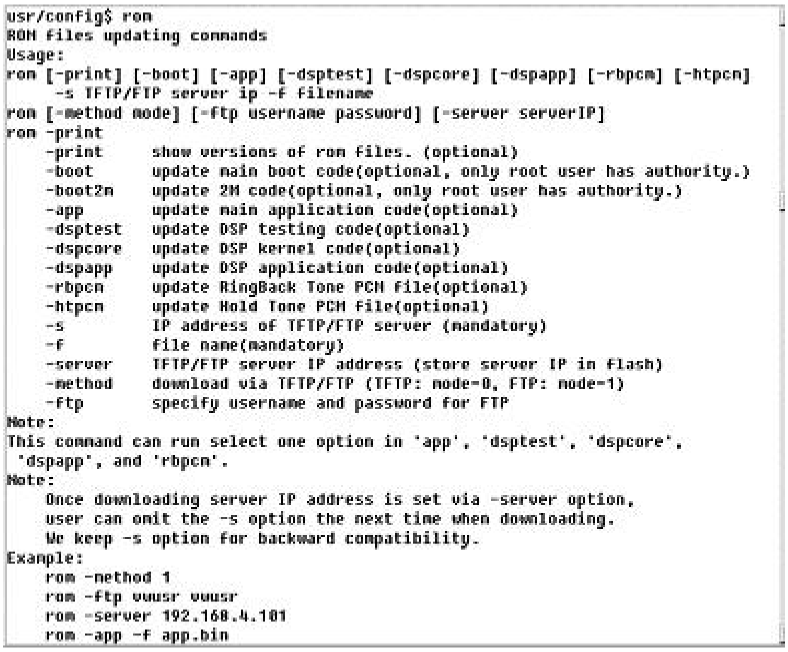
Associated with the Chapter **4.19 [rom] command**:

1. **-print**: show versions of all rom files.
2. **-app**,**-boot**, **-dsptest**, **-dspcore**, **-dspapp**, **-rbpcm** and **-htpcm**: upgrade main boot code, main application code, DSP testing code, DSP kernel code, DSP application code, Ring Back Tone PCM file and Hold Tone .
3. **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading rom file.
4. **-f**: the file name prepared for upgrading is necessary as well.

9. **–server**: specify TFTP/FTP server IP address. It is corresponding to LCD configuration -firmware upgrade-Set file Server IP.
10. **–method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
11. **–ftp**: specify user name and password for FTP download method

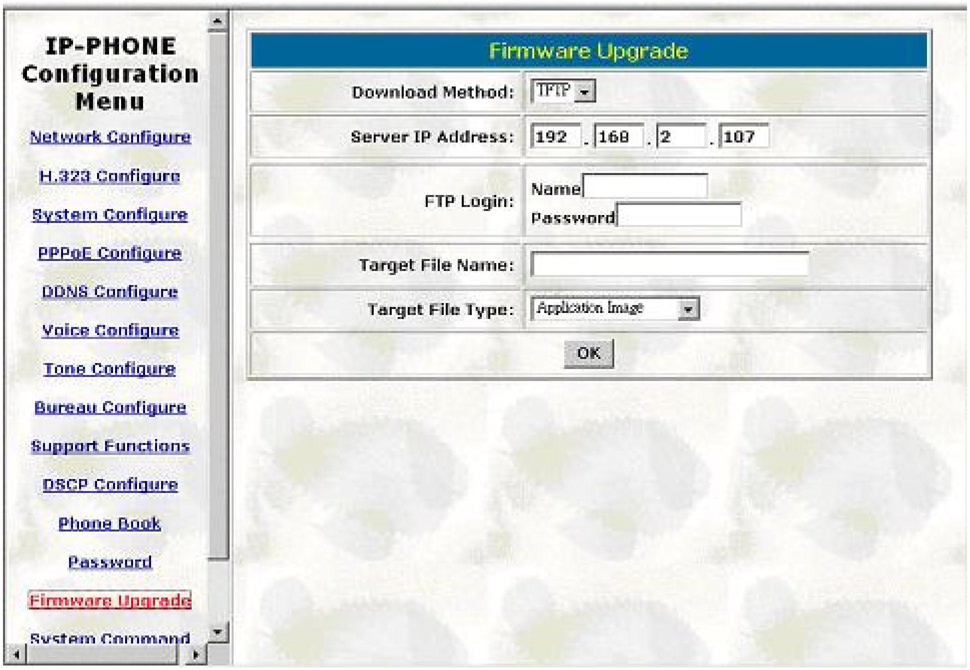
For example: User prepares to upgrade the latest app rom file – wtlp.103c, the TFTP server is 192.168.4.530X.

rom –app –s 192.168.1.1 –f lp.100 (If –server is specified , can just type **rom –app –f lp.100**)



```
usr/config$ rom
ROM files updating commands
Usage:
rom [-print] [-boot] [-app] [-dsptest] [-dspcore] [-dspapp] [-rbpcm] [-htpcm]
     -s TFTP/FTP server ip -f Filename
rom [-method mode] [-ftp username password] [-server serverIP]
rom -print
    -print      show versions of rom files. (optional)
    -boot       update main boot code(optional, only root user has authority.)
    -boot2m     update 2M code(optional, only root user has authority.)
    -app        update main application code(optional)
    -dsptest    update DSP testing code(optional)
    -dspcore    update DSP kernel code(optional)
    -dspapp     update DSP application code(optional)
    -rbpcm      update RingBack Tone PCM file(optional)
    -htpcm      update Hold Tone PCM file(optional)
    -s          IP address of TFTP/FTP server (mandatory)
    -f          file name(mandatory)
    -server     TFTP/FTP server IP address (store server IP in flash)
    -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', 'dsptest', 'dspcore',
 'dspapp', and 'rbpcm'.
Note:
    Once downloading server IP address is set via -server option,
    user can omit the -s option the next time when downloading.
    We keep -s option for backward compatibility.
Example:
    rom -method 1
    rom -ftp vvusr vvusr
    rom -server 192.168.4.101
    rom -app -f app.bin
```

3.Web Management



- Download Method: Select download method as TFTP or 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 IP 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** (under **System Command**), which will clean all configurations become factory values except IP address.