



Xcelerator IP



**IP2007 User Guide**

Issue	Release Date	Changes	Page
1.0	3-07	Initial release	--

**“WARNING:** Handling the cord on this product will expose you to lead, a chemical known to the State of California to cause [cancer, and] birth defects or other reproductive harm. ***Wash hands after handling.***”

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set/2007

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# 1

## Introduction

Read the *Introduction* chapter to:

- review the contents of the IP2007 phone package
- identify the phone buttons and the hardware



## IP2007 Product Description

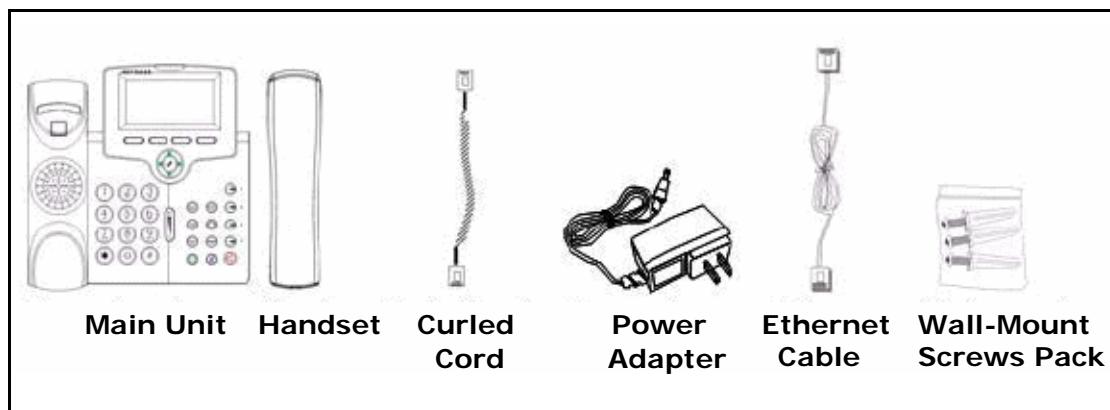
The IP2007 IP Phone is a fully featured IP phone integrating voice and data connectivity to the Xcelerator IP gateway. The IP2007 is feature rich SIP endpoint that provides a full suite of telephony features and a dual port 10/100baseT LAN switch to allow for a single wire to the desktop solution where the PC is connected through the second port on the IP2007 IP phone.

The IP2007 provides both aural and visual queues to support the various options and activities supported. Please refer to the section "Identifying Buttons and Hardware" for further descriptions of the features, functions, and operations associated with the IP2007.

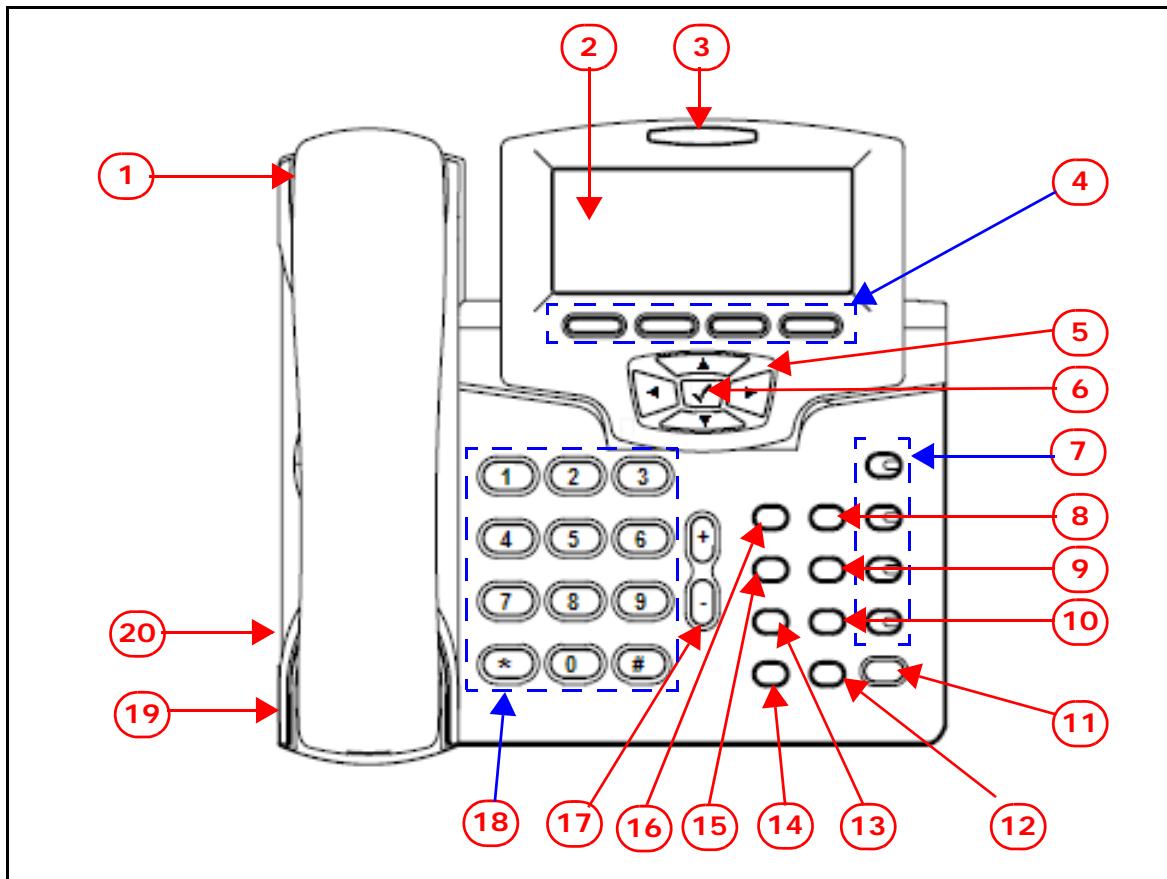
### Package Contents

Upon opening the package, ensure that the following items are included and that there is no damage. If you find any problem with them, contact the reseller or supplier for assistance.

- IP phone (with stand kit assembled on the back of phone)
- Handset
- Curled phone cord
- Power adapter (Different appearance for different country area)
- Ethernet cable
- Wall-mounting screw pack
- User Guide
- Warranty card



## Identifying Buttons and Hardware



### IP Phone Function Keys

No.	Part Name	Description of Function
1	Handset	Phone Handset
2	LCD	The LCD shows phone features such as the time, date, your phone number, caller ID, line/call status, and soft key tabs.
3	Message LED	This LED illuminates red to indicate a message waiting status.
4	Soft Keys	These four keys are used for item selection or control on the LCD screen. Each key function depends on its corresponding content displayed on the LCD at that time. (Example: Press soft key 1 to enter the configuration menu).
5	Navigator Control Keys	Four arrows on the Navigator Control are used to scroll through items on the LCD screen.
6	Navigator [✓] OK Key	This key is normally used to confirm a configuration setting or phone number dial.

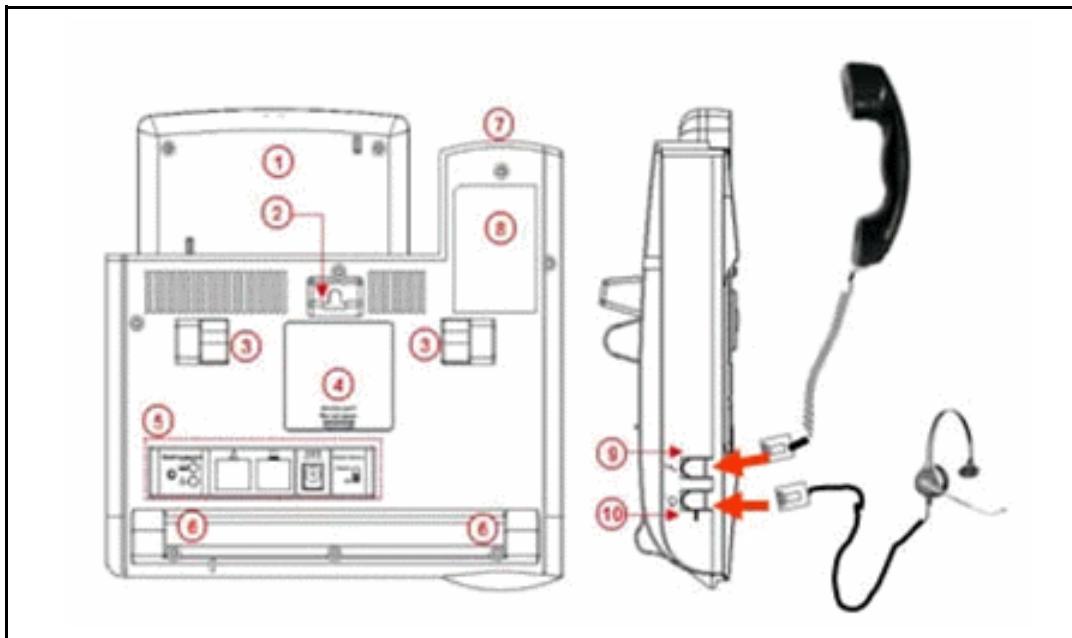
No.	Part Name	Description of Function
7	Number Keys (1,2,3,4) 	These four keys can be used for line selection or programmable features. A green LED is associated with each key to indicate its line/call status.
8	Transfer Key 	This key is used to transfer a call to another IP phone.
9	Redial Key 	This key is used to redial the last dialed number automatically.
10	Hold Key 	The Hold key is used to place the current call on hold and allows you to then answer another new call. Pressing the key again will release the hold function.
11	Speaker Key 	This key is used to activate/de-activate the Handsfree dial or talk. A green LED is associated to indicate its status.
12	Mute Key 	This MUTE key is used to activate/de-activate the voice transmit from this IP phone. A red LED is associated to indicate its status.
13	Phone Book Key 	This key is used to enter the Phone Book to place a call or to edit. The LCD screen will provide Phone Book prompts once this key is pressed.
14	Headset Key 	This key is used to activate/de-activate the headset. A green LED is associated to indicate its status.
15	Conference Key 	This key is used to drop the current multiple-connected phones into a conference. It is a phone-bridged function.
16	Message Key 	This key is used to get access to the voice mail system for message retrieval.
17	Volume Control Key 	This key is used for volume control: When the phone is idle, the ringer volume is adjusted. When talking on an active call, the Handset/Speaker/Headset output volume is adjusted.
18	Numeric Keypad (0-9, *, #)	This is the numeric keypad for dialing numbers.
19	Headset Wire Port	RJ-9 jack by the left bottom side of IP phone
20	Headset Cord Port	RJ-9 jack by the left bottom side of IP phone

## LED Status Indicators

No.	Key/Indicator	Color	Static OFF	Static ON	Blinking ON/OFF
8	Message indicator	Red	No new messages	N/A	Slow blinking for new voice message indication
13	Number keys	Green	Default	Mode Headset: During call Mode Handset: During call only if hands-free is active.	Headset/Handset Mode: Ringing cadence on incoming call; <b>Group listening in active cadence</b>
14	Speaker key	Green	Default; Handset or Headset modes.	When SPKR is on for on-hook dialing or hands-free talking.  Headset & Handset are off.	When an incoming call is ringing (in Handset mode)
15	Mute key	Red	Mute is off.	Mute is on.	N/A
16	Headset key	Green	Default; Handset	When Headset is on for on-hook dialing or hands-free talking.  Handset & Speaker are off.	When an incoming call is ringing (in Headset mode).  [The rings take place in speaker and headset]
8, 13, 14, 15, & 16	All LEDs for system status	Green	Ready	Boot: During system booting	Boot: When system booting failure or fault occurs.

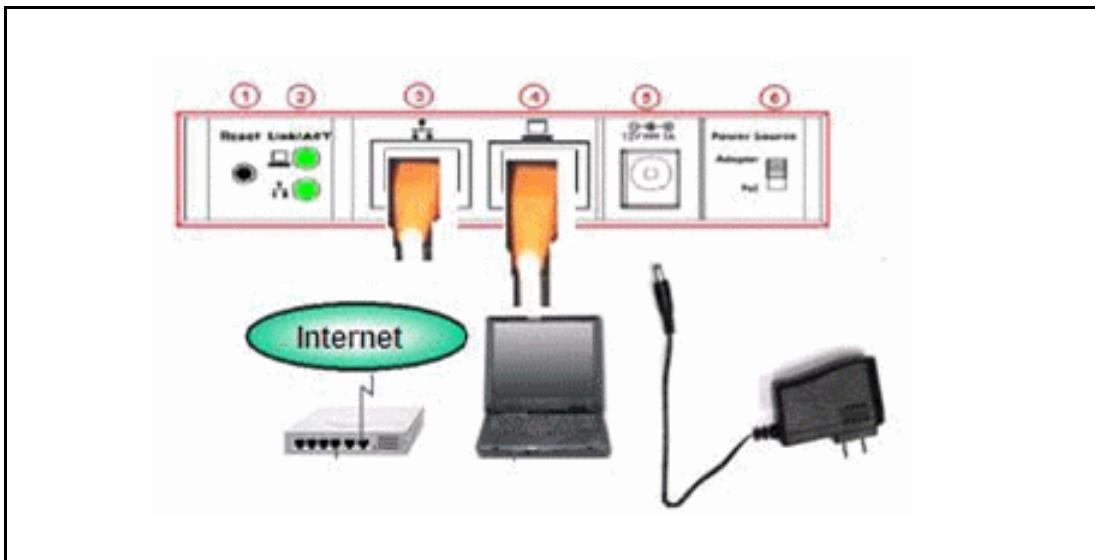
## IP Phone Rear & Side Views

The following illustrates the rear and side views of the IP phone. Refer to the callout numbers, and associated simple description of the part in the following table.



No.	Part Name	Description of Function
1	LCD screen cover	The back of the LCD screen
2	Wall-Mount Hole	For mounting the phone on the wall
3	Hinge of Stand	A leg is installed here to support the IP phone at different angles. For wall-mounting, remove the leg.
4	Service door	The Service Door is only for engineering use. Inside, there is a console port. Users should not open this door. [Note: To use this port, you need a special converter cable (RJ-45 to DB-9)]
5	Input/Output ports	For installation, connect the cables here (refer to the next figure and associated table for the details).
6	Hinge of Chassis	The small frame is fixed here. You do not need to disassemble this frame for wall-mounting.
7	The back of Cradle	This is the back cover of the handset cradle.
8	Product label	This shows product production information, including product model, serial number, and MAC address.
9	Handset cord port	RJ-9 jack on the side of IP phone.
10	Handset wire port	RJ-9 jack on the side of IP phone.

## IP Phone In/Out Ports



No.	Part Name	Description of Function
1	Reset Button	The Reset Button is for system engineering use (The configuration will be reset to factory settings when pressed).
2	Link/Activity LEDs	The LEDs indicate the link status of the WAN and LAN ports.
3	WAN Port	RJ-45 Jack 100/10Mbps Ethernet port for connecting to IP network
4	LAN Port	RJ-45 Jack 100/10Mbps Ethernet port for connecting to PC or Notebook
5	Power Jack	If a power source from adaptor is required, use the standard power adaptor supplied in the package (12V/1000mA).
6	Power Source	There are two available sources for power: from Power Adaptor or PoE (Power over Ethernet). The default is Adaptor. Please set to the appropriate option before installation. Default setting is Power Adaptor. Only the Ethernet WAN port supports PoE.

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# 2

## Install & Connect the Phone

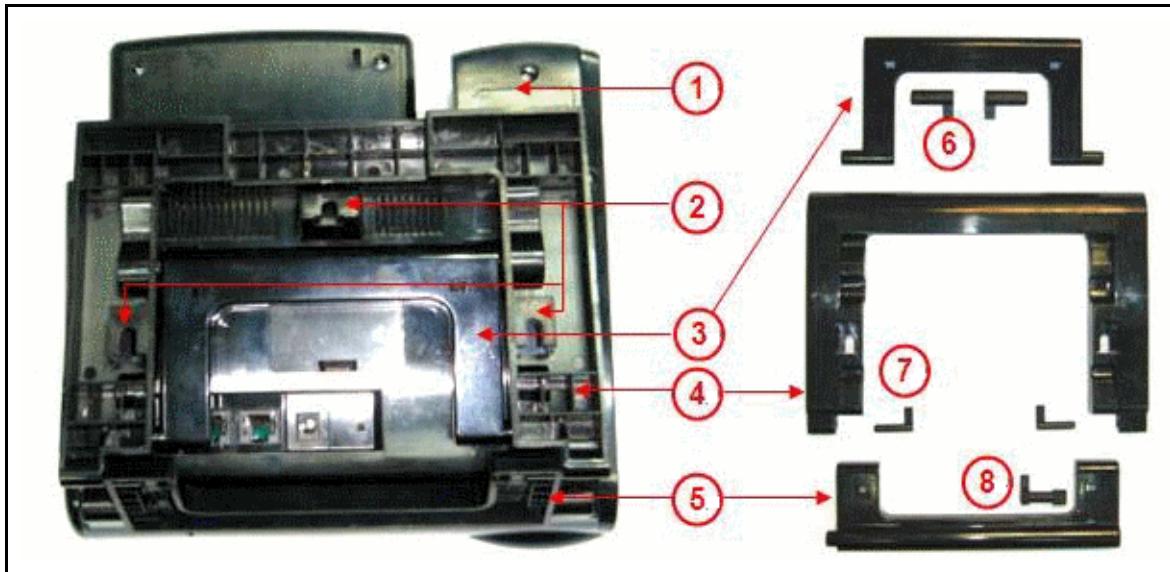
Read the *Install & Connect the Phone* chapter to perform the following functions:

- set up the IP2007 Phone, the phone stand, and the power source switch
- connect the handset and the cables
- mount the IP2007 Phone on the wall



## Assembling the IP Phone

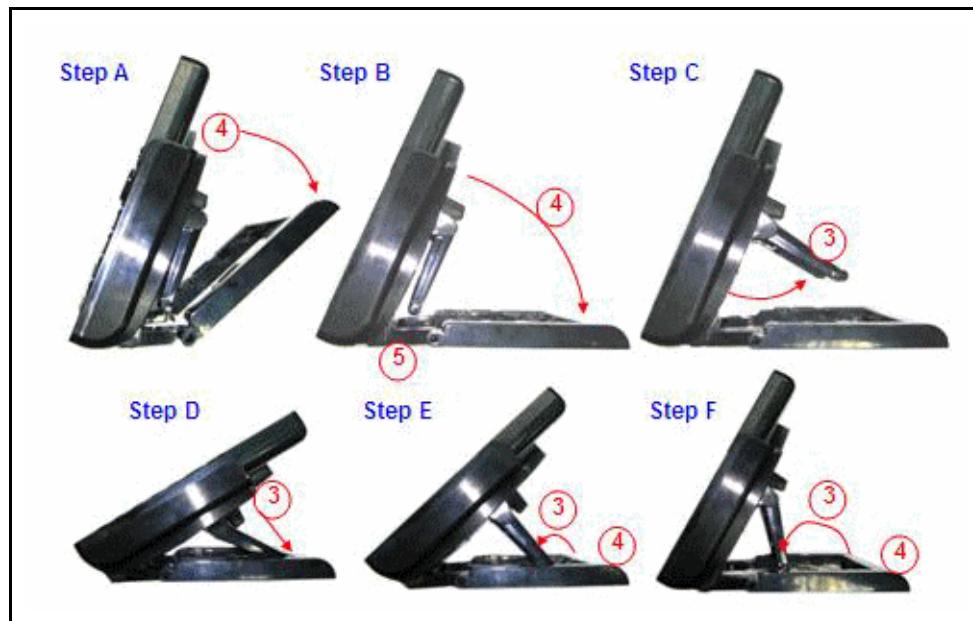
After removing the IP phone from the package, you will see the back of IP phone as shown in the following illustration. The phone (1) is already assembled with three frames. The U-type leg (3) is used with the big U-type frame (4) that has three step cavities for three angle positions. Another small U-type frame (5) is fastened to the big U-type frame (4) and the main body of phone (1). You may find these frames are locked together with different latches (6), (7), and (8) respectively. Do not touch latches (7) and (8). The latch (6) will be removed if you are do a wall-mount.



## Setting Up the Stand

Refer to the following steps and the associated illustration to setup the stand of the IP phone.

- Step A Move the big U-type frame (4) downward.
- Step B Move the big U-type frame (4) downward to the flat level. The small U-type frame (5) will get flat concurrently.
- Step C Move the U-type leg (3) upward.
- Step D Seat the U-type leg (3) onto the big U-type frame (4) at about a 30° angle position.
  - or-
- Step E Pull-in the U-type leg (3) and seat it at about a 45° angle position in the big U-type frame (4).
  - or-
- Step F Pull-in the U-type leg (3) and seat it at about 60° angle position in the big U-type frame (4).



### Connecting the Handset

Use the curled cord (1) to connect to the phone jack (3) at the bottom of the handset (2), as shown in the following illustration. Plug the other end of curled cord to the Handset jack (5) on the side of IP phone. The other RJ-9 jack (4) is for a headset. This completes the assembly of the IP phone.



### Setting the PoE or PA Power Source Switch

Before connecting the cables, set the power source switch to the correct position. It is located to the right of the I/O ports on the back, as shown in the following illustration.

Use a pen to set it to one of the following:

**Adaptor position** - Use this position if you are using an external standard power adaptor.

**PoE position** - Use this position if you can offer Power over Ethernet (PoE, IEEE802.3af compliant) to the phone.

The Power source switch default setting is "Adaptor".



## Connecting the Cables

To avoid any problems caused by power-on, the power source will be applied to the phone last. Do not plug the adaptor head to the wall socket if you are using Power adaptor. Do not plug the Ethernet cable to the uplink switch if you are using PoE.

Prepare the Ethernet cable(s), attached in the package, and power adaptor (no need for PoE user). Plug the cable(s) to their corresponding ports on the back of the IP phone. It is recommended that you set the stand angle at a 60° angle so that room is sufficient for easy access to connect the cables. Arrange the cables to go under the big U-type frame as shown in the illustration.



## Full Connection Example

The following illustration is an example of a full connection.

(1) To Handset thru curled cord, (2) to IP Network thru Ethernet cable, (3) to PC/Notebook thru Ethernet cable, (4) to wall socket thru Power adaptor, and (to use Headset, plug the headset with R-J9 plug to the Headset jack on the left bottom of the IP phone) (5) to headset thru headset cable.



## Mounting the Phone On the Wall

The IP phone can be used on the desktop or mounted on a wall.

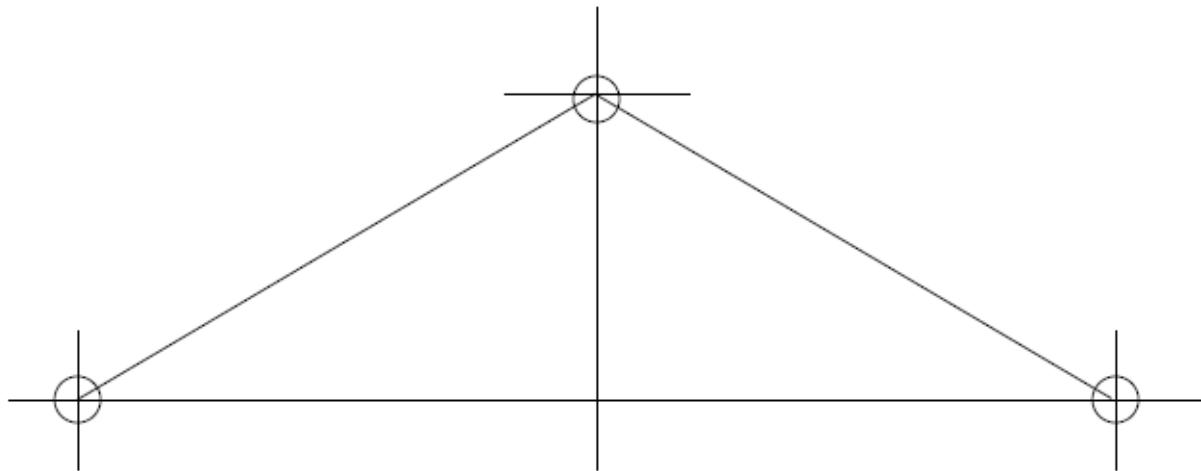
Print the screw alignment template (on the next page) and tape it on the wall to position the 3 screws provided at the right place for the mount-holes on the phone.

Before mounting the IP phone to the wall, detach the handset and curled cord from the IP phone, then perform the following steps:

- Step A On the back of the phone, find the latch (3) on the U-type leg and remove it.
- Step B Remove the other latch (4).
- Step C Remove the U-type leg (1). Retain the leg (1) and latches (2) for possible reuse in a desktop mode.
- Step D Seat the big U-type frame onto the back of IP phone.
- Step E Re-check that the power source switch is in the correct position. Connect all cables to the jacks on the phone, then mount the phone's holes (5 top, 6 left and 7 right) to the screws on the wall.
- Step F Under the hook switch, find and press the cradle latch (8) to let it down. Then hang up the handset and re-attach the curled cord.



## Screw Alignment Template



### *Resetting the Phone to Factory Defaults*

If you want to reset the configuration to the factory default settings, locate the Reset switch on the back of the phone and complete the following steps:

- Step A Remove the power source (adaptor or Ethernet cable) at the remote end.
- Step B Use a pen to press and hold the Reset switch.
- Step C To turn on the power again press and hold the Reset switch for over 3 seconds.
- Step D Release the Reset switch. The phone will automatically reset the configuration and the old settings will be gone.

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# 3

## IP2007 Phone Features

Read the *IP2007 Phone Features* chapter to discover the many features of the IP2007 telephone, and to operate its basic user functions.



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## Answer a Call

There are various ways to answer an incoming call:

- Lift the handset to answer the call.
- Press  to answer the call in handsfree mode.
- Press  to answer the call in headset mode.

---

## Auto Answer

You can use the Auto Answer function to automatically answer all incoming calls when you are busy. You can enable or disable this function through the LCD menus or via the Web. The default value is disabled.

### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key >  arrow to 8. *Auto Answer* >  for options
- » *Web Menu* = *Login* > *Phone tab* > *Phone Parameters*  *Enable Auto Answer*

---

## Basic Call

*To make an intercom call ... dial a Station number (IP Terminal, SLT) or a Voice Mail number.*

*To make an outgoing call ... dial a phone number. The system chooses a PSTN line or IP trunk via the Call Routing Table to dial out. If it includes \* in the phone number, and the call is dialed through PSTN Trunk, the \* will be interpreted as a one second pause.*

*-or-*

*Dial a PSTN, an IP Trunk or a Trunk Group number first. After hearing a dial tone, dial the phone number.*

## Call Forward

You can use Call Forward to redirect your incoming calls from your IP Phone to another number. You can set three different types of call forward as described in the following sections.

### *Always Forward*

If you enable Always Forward, all your incoming calls will redirect to another destination.

### *Busy Forward*

If you enable Busy Forward, your incoming calls will redirect to another number when your phone is connected to a call and you do not wish to receive a second call simultaneously.

### *No Answer Forward*

If you enable No Answer Forward, your incoming call will redirect to another number after this call is not answered by you for a specific amount of seconds.

#### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key >  arrow to 7. Call Forward ...
- » *Web Menu* = Login > Personal tab > Call Forward Settings

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## Call Log

The IP2007 phone can store a call log for your reference. To access your call log, use the LCD menus. There are three types of Call Logs - Missed Calls, Received Calls, or Dialed Calls. To dial from a listing, press the soft keys corresponding to the LCD menus display.

## Call Waiting

If Call Waiting is enabled for a specific IP2007 station, an alert (muted ring) will be played on the called party IP2007 when a second call is received and the IP2007 is in use.

*To enable Call Waiting ... dial \*99*

If Call Waiting is disabled for a specific IP2007 station, the IP2007 will return a busy tone to any calling party while the IP2007 is in use.

*To disable Call Waiting ... dial \*\*99*

---

## Caller Blocking

You can block up to 10 phone numbers from reaching your phone when a caller attempts to call you from one of these numbers.

### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key >  arrow to 15. Blocking List
- » *Web Menu* = Login + Personal tab + phone number + Save Settings

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## Caller ID

The *Xcelerator-IP*, by default, accepts Calling Party Name and Calling Party Number ID. Calling Party Name is only displayed if the Calling Party Name and Calling Party Number are entered in the IP2007 Phone Book. The Calling Party Number will be displayed if delivered from the serving CO.

### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key >  arrow to 12. Phone Book
- » *Web Menu* = Login > Phone Book tab > enter a party's name and number in order for caller ID to display on incoming calls

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## Conference

Conference calling allows three parties to simultaneously participate in a call.

### ***To establish a Conference call:***

1. Place an internal or external call.
2. Press the **HOLD** key  to place the 1st party on hold.
3. Press the **NAV** key down arrow  to place a second call on this line.
4. Dial the phone number of the 2nd party of this conference.
5. Press  to start the conference.

---

## Distinctive Ringing

Distinctive ring cadences can be selected allowing adjacent users to discern which extension is ringing. It also provides different ring tones for intercom and trunk calls. You can enable or disable this function through the LCD menus or via the Web.

### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key +  arrow to 2. Ring
- » *Web Menu* = Login > Phone tab > Tones Used

## Do Not Disturb

You can enable the Do Not Disturb (DND) function, if you do not want any incoming calls to interrupt your work. All incoming calls will receive a busy tone when they call your phone number. You can enable or disable this function through the LCD menus or via the Web.

### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key >  arrow to 6. DND
- » *Web Menu* = Login > Phone tab > Phone Parameters  Enable DND

## Drop a Call

There are various ways to drop an existing call:

- Place the handset on hook if talking in handset.
- Press  to drop a handsfree call.
- Press  to drop a headset call.

## Feature Key Programming

Feature Keys can be programmed by phone users. A feature key can be programmed for line appearance.

*To program a Feature Key ... dial \*70 + (Feature Key number: 1 - 4) + (PSTN, IP Trunk or Trunk Group number)*

## Hold/Resume/Navigation in Calls

Only one call can be active at any given time; another call must be placed on hold.

### **Hold a Call**

After you have a connected call, you can use  to hold a call.

### **Resume a Held Call**

You can resume a held call by using the soft key under the "Resume" label on the LCD. The call will return to the connected status.

### **Navigation in Calls**

You can jump between two calls by using . Press the  or  arrow of the navigator control key to toggle between the two calls.

---

## Lines and Calls

### *Lines*

The term "Line" in this manual represents how many phone numbers are supported on one phone. A phone can have more than one phone number so that the user can subscribe to different VOIP service providers.

### *Calls*

The term "Call" in this manual represents how many simultaneous connections can be made to a single phone number. Each line has two calls capability, so that you can hold one call and talk to another person in another call.

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## Mute

To mute the microphone during a call, press the  button. The button will illuminate to indicate that the microphone has been muted.

Press the MUTE button again to re-enable the microphone.

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## Page Allow/Deny

You can block one-way pages (internal, group, and all page) over the IP phone speaker by dialing the Page Deny code.

#### Feature Code Access

*To enable Paging ... dial \*99*

*To disable Paging ... dial \*\*99*

---

## Phone Lock/Unlock

The IP2007 Lock feature is used to prevent unauthorized trunk calls from being made from a specific extension. A locked extension will continue to receive incoming trunk calls, and a user can continue to place and receive intercom calls; however, outgoing trunk calls are blocked. You can access this feature via the Phone or the Web.

#### Feature Code Access

*To lock the phone ... dial \*97 + (VM password)*

*To unlock the phone ... dial \*\*97 + (VM password)*

#### Phone & Web Access

» *Phone Menu = Left soft key to CONFIG >  key > ▼ arrow to 9. Phone Lock*

» *Web Menu = Login > Phone tab > Phone Parameters  Phone Lock*

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## Phone Book

The *Xcelerator-IP* provides users with a Phone Book, with each entry containing a user programmed Phone Number and User Name. The phone number can be an extension number, phone number, or IP address. Up to 400 entries per station user is supported.

### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key >  arrow to 12. Phone Book
- » *Web Menu* = Login > Phone Book tab > enter a caller's name and number for caller ID to display on incoming calls

---

## Place a Call

There are various ways to place a call:

- Lift the handset and dial the number pad as a regular phone.
- Press  to perform handsfree dialing.
- Press  to perform headset dialing.
- Press  to pickup a specific line.
- Dial directly, the phone will automatically pickup in handsfree mode and perform handsfree dialing.



*You can use the pound [#] key or the [OK] key to complete the number. This will speed up the calling process.*

---

## Redial

The Redial feature automatically dials the last number dialed from the phone.

Press  to dial the last number automatically. If the phone is on hook, it will pickup in handsfree mode automatically and perform handsfree dialing.

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## Register to a Server

The phone should be configured before it can perform some basic functions. Although the phone can make a peer to peer VOIP call (the user must remember the IP address of the called party), it is desirable to have a centralized server to provide the directory service. This server can be a soft-switch, an IP-PBX, or a simple proxy.

The server has two basic functions:

- 1) The first is to track active phones and their IP address.
- 2) The second is passing signaling messages between communication parties.

To make your phone reachable in this VOIP network, your phone has to register to the server so that when someone dials your phone number the server knows where you are and informs your IP phone that someone is calling.

### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key >  arrow to 16. Admin > SIP
- » *Web Menu* = Login + SIP tab (refer to "[SIP Page" on page 5-10](#))

## *Speakerphone/Headset/Handset*

There are 3 major input/output devices on this phone. To switch between these devices, follow the audio path transitions described below:

**Handset to Handsfree** -- You are using the handset and want to switch to handsfree talking. Press  and place the handset on hook. *NOTE:* The voice will not switch to speakerphone until you place the handset on hook.

**Handset to Headset** -- You are using the handset and want to switch to the headset. Press  and place the handset on hook. *NOTE:* The voice will not switch to headset until you place the handset on hook.

**Handsfree to Handset** -- You are using handsfree talking and want to switch to the handset. Lift the handset off hook and the voice will switch to handset immediately.

**Handsfree to Headset** -- You are using handsfree talking and want to switch to the headset. Press  and the voice will switch to headset immediately.

**Headset to Handset** -- You are using the headset and want to switch to handset. Lift the handset off hook and the voice will switch to handset immediately.

**Headset to Handsfree** -- You are using the headset and want to switch to handsfree talking. Press  and the voice will switch to speakerphone immediately.

## Speed Dial

You can preset 10 speed dial numbers for fast dialing. Use the following steps to use speed dial:

1. Lift the handset.
2. You will see "SPD" on the LCD menus. Press the soft key corresponding to the LCD menus display.
3. You will see "SpeedDial:\_" on the LCD display. Enter a valid speed dial bin number (0-9) and the IP phone will dial out with the number that is preset in the IP phone.

### Phone & Web Access

- » *Phone Menu* = Left soft key to *CONFIG* >  key >  arrow to 13. Speed Dial
- » *Web Menu* = Login > Personal tab > Speed Dial Entry Settings

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## Transfer

Transfer redirects a connected call. You can use a Blind or Supervised transfer method to transfer the call to an extension or an outside phone number.

### *Blind Transfer*

#### ***While on a call:***

1. Press the TRANSFER key  to transfer the call.
2. Dial the desired phone number and the call will transfer automatically.
3. Hang up to release the line.

### *Supervised Transfer*

#### ***While on a call:***

1. Press the HOLD key  to hold this call.
2. Press the down arrow  of the navigation control key to start a 2nd call on this line.
3. Dial the desired phone number. You will hear a ring back tone.
4. When the called party answers, advise them of the call being transferred.
5. Press the TRANSFER key  and hang up. The Supervised Transfer is complete.

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## Voice Mail Access

Ask your system administrator for the Voice Mail access number in order to access your voice mailbox. You will find the "MSG number" field blank; enter an appropriate number.

If you have voice mail in your voice mailbox, you will see a voice mail icon  on the left top corner of the LCD. Press the message icon  to access your voice mail.



*The Total Recording Message Time for one extension depends on how many extensions are connected to the Xcelerator IP. (See current members in the Phone Extension Table.) The current Xcelerator IP configuration allows 240 minutes of recording time.*

*Recording Time Per Extension = (240 minutes) divided by (Total Members)*

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## Volume Adjustment

Use the VOLUME CONTROL button  to adjust the volume.

- When using the handset ... use this button to adjust the handset output volume.*
- When using handsfree talking ... use this button to adjust speaker output.*
- When using a headset ... use this button to adjust the headset output volume.*
- When the phone is ringing ... use this button to adjust the ringing volume.*



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# 4

# Configuration Via Menus

Read the *Configuration Via Menus* chapter to learn how to navigate in the menus of your IP2007 telephone.



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## Menu

### *Startup*

You can use LCD menus to configure most settings of your IP phone.

Press the LEFT soft key to start menu selection process.

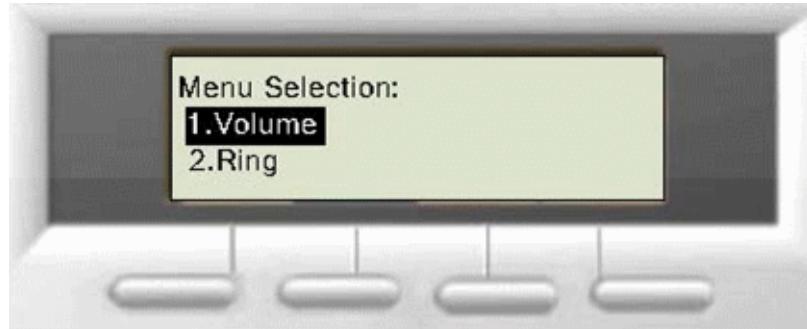


## Menu Tree

- Menu Selection
- **1. Volume**
  - + Ring Volume
  - + Handset Speaker
  - + Handset Mic
  - + Handsfree Speaker
  - + Handsfree Mic
  - + Headset Speaker
  - + Headset Mic
- + **2. Ring**
- + **3. Tone**
- + **4. Time**
- + **5. Info**
- + **6. DND**
- **7. Call Forward**
  - Always Forward
    - + Toggle
    - + Forward Number
  - On Busy Forward
    - + Toggle
    - + Forward Number
  - No Answer Forward
    - + Toggle
    - + Forward Number
    - + No Answer Time
- + **8. Auto Answer**
- + **9. Phone Lock**
- **10. Call Record**
  - + Dialed
  - + Missed
  - + Received
- + **11. Ping**
- + **12. Phone Book**
- + **13. Speed Dial**
- **14. Network**
  - Network Type
    - + Static IP
    - + DHCP
    - + PPPoE
    - + Netmask
  - + Default Gateway
  - DNS
    - + Primary DNS
    - + Secondary DNS
    - + Third DNS
  - + SNTP
- + **15. Blocking List**
- **16. Admin**
  - + Phone Number
  - Account
    - Admin
      - + Name
      - + Password
    - User
      - + Name
      - + Password
  - NAT
    - + NAT Type
    - STUN
      - + STUN Server IP
      - + STUN Port
    - + SIP PING
    - Port Mapping
      - + Extern Router IP
      - + Signal Port
      - + Extern RTP Port 1
      - + Extern RTP Port 2
  - SIP
    - SIP Server
      - + SIP Proxy
      - + Outbound Proxy
      - + Registrar Server
      - + Registrar Outbound
      - + Port
    - Authentication
      - + Authorized ID
      - + Authorized Password
    - + DTMF
    - Codec
      - + G.711 u-law
      - + G.711 a-law
      - + G.723.1
      - + G.729
      - + Packet Time
      - + Caller ID
      - + User Name
    - + VLAN
      - Download
        - + TFTP Server
        - + Firmware Upgrade
        - + Profile Download
      - + Reset
      - + Boot
  - + **17. Quit**

The menu tree begins at the top of the left column. After "Blocking List" at the bottom of that column, the menu tree continues at the top of the right column.

This is the first display shown when you press the left soft key.



### *Navigate in the Menu*

Use the Navigator Control key  to scroll to a desired item.



Press  to enter the sub-tree for the selected item.



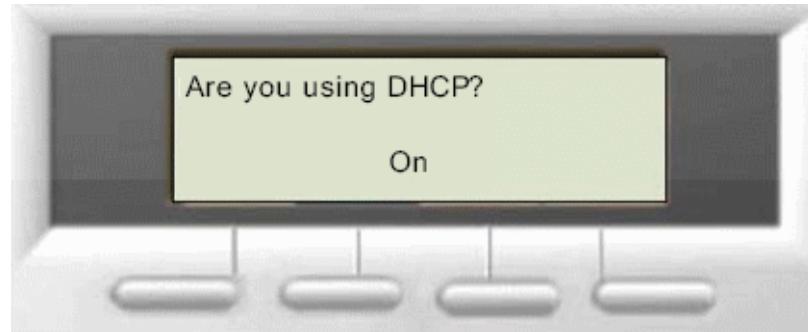
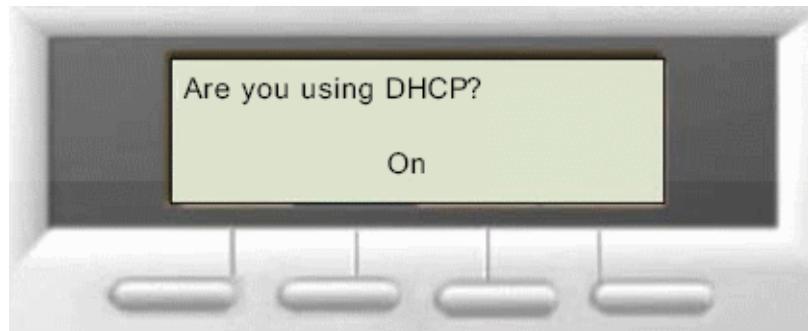
Press  to confirm and save the setting of a specific item. Thereafter, the menu will roll back to the upper level of the menu tree. You can also press the left arrow of the Navigator Control key to roll back to the upper level of the menu tree without causing the system to resave.

## Enable or Disable Settings

You can enable or disable certain item settings. Use the up or down arrow to toggle an item option. Example:



to



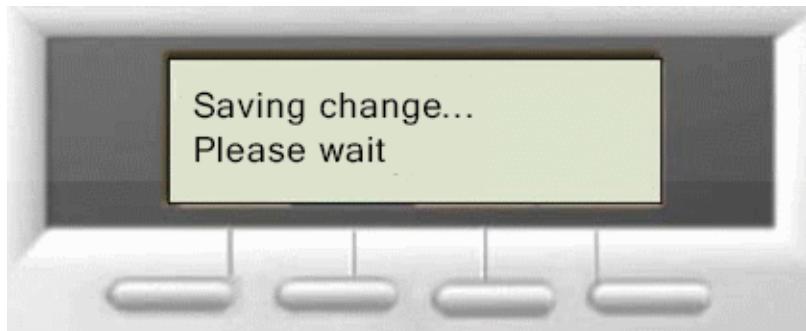
## Numeric and Alpha Characters

You can input numeric or alphabetic characters in certain menus, such as in the Phone Book. When you are on a menu that accepts characters, you will see "ABC" above the left soft key. Press the left soft key to cycle through the options of ABC, 123, and abc. These options represent characters in upper case, numeric digits, and characters in lower case. You can switch between the three options during one entry to use upper and lower case characters as well as numbers.

To enter alphabetic characters, select ABC or abc with the left soft key, then repeatedly press the dial pad button that has the character you want until the desired character appears. The display will cycle through all of the characters associated with the button.

## Saving the Configuration

Some changes are saved immediately as you change settings. All settings will be saved when you leave the Menus. You will see "Saving change ... Please wait" on the LCD.





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# 5

# Configuration Via Web

Read the *Configuration Via Web* chapter to learn how to use your IP2007 telephone by using the web interface.



## Web

### Access

To access the phone configuration, enter the IP address in the Address field of your web browser, <http://192.168.1.10>.

*To confirm IP address --* If you are unsure about the IP address, you can verify the current IP address on your IP phone by pressing the Left soft key to access the CONFIG menu and using the down arrow to select the 5. Info menu.

### Login

The following dialog box will pop up and prompt you to provide the user name and password in order to prevent unauthorized users from accessing your phone.

The default names and passwords for the Web are as follows:

- User Name = user
- User Password = 1111
- (Administrator) User Name = admin
- (Administrator) Password = 1234



### Web Page Menus

When accessing the web pages for the IP2007 Phone, you can make configuration changes in each category. Click on a menu to access a specific group of settings.

Information | Network | Phone | SwUpgrade | Personal | SIP | System | Phone Book



### Web Page Buttons

Button	Web Page	Description
Cancel	All	Discard all changes entered on this page
Download Settings	SW Upgrade	Download files to upgrade phone firmware
Logout	All	Exit and close browser window
Next/Previous	Phone Book	View next or previous page of Phone Book entries
Reboot	Information	Update phone to capture changes made to s/w
Save Settings	All	Save changes entered on this page to the IP Phone
Set All to Default	System	Change ALL parameters to factory default settings
Update	SW Upgrade	Locate new file provided (s/w, profile, logo, font)

## Information Page

The Information Page displays when you log into the phone's web page. This "view only" page shows general information regarding your IP2007 Phone.

### Network Information

IP Type | Current IP | Subnet Mask | Default Gateway | Primary DNS | Secondary DNS

### Product Information

Product Name | Software Version | MAC Address | Hardware Version

### Profile Status

Phone Number | Registration State | SIP Proxy Server

Information	Desktop SIP Phone							IP-Phone PhoneBook
	Information	Network	Phone	SwUpgrade	Personal	SIP	System	
<b>Network Information</b>								
	IP Type:	DHCP		Current IP:	192.168.1.3			
	Subnet Mask:	255.255.255.0		Default Gateway:	192.168.1.1			
	Primary DNS:	192.168.1.1		Secondary DNS:				
<b>Product Information</b>								
	Product Name:	IP-Phone		Software Version:	V3.0.3 - IG			
	MAC Address:	00-16-38-88-37-78		Hardware Version:	R1			
<b>Profile Status</b>								
	Phone Number:	100		Registration State:	Registrar OK			
	SIP Proxy Server:	192.168.1.1						
	<a href="#">Reboot</a> <a href="#">Logout</a>							

## Network Settings

The Network Settings page allows you to access the following network categories:

Internet Connection Items | Optional Network Items | QoS Items

Network Settings	Desktop SIP Phone							IP-Phone PhoneBook
	Information	Network	Phone	SwUpgrade	Personal	SIP	System	
<b>Internet connection items</b>								
	IP Type:	<input type="button" value="DHCP"/>		IP Address:	<input type="text" value="0.0.0"/>			
	Subnet Mask:	<input type="text" value="255.255.255.0"/>		Default Gateway:	<input type="text" value="192.168.1.1"/>			
<b>Optional network items</b>								
	Primary DNS:	<input type="text" value="168.95.1.1"/>		Secondary DNS:	<input type="text"/>			
	Third DNS:	<input type="text"/>		Time Server:	<input type="text" value="140.112.2.189"/>			
<b>QoS items</b>								
	DIFFSERV for RTP:	<input type="button" value="Best Effort"/>						
	DIFFSERV for Signal:	<input type="button" value="Best Effort"/>						
	VLAN Mode:	<input type="button" value="None"/>						
	Voice VLAN ID:	<input type="text" value="2"/>		VLAN Option ID:	<input type="text" value="176"/>			
	<a href="#">Save Settings</a> <a href="#">Cancel</a> <a href="#">Logout</a>							

## Network Parameters

Internet Connection Items	
IP Type	Select how this phone receives IP address - DHCP or Static IP; Default = DHCP
IP Address	Static IP address; Default = 0's
Subnet Mask	Static Subnet Mask
Default Gateway	Static Default Gateway
Optional Network Items	
Primary DNS	Static Primary DNS
Secondary DNS	Static Secondary DNS
Third DNS	Static Third DNS
Time Server	Static Time Server
QoS Items	
DIFFSERV for RTP	<p><i>Options:</i></p> <p>Best Effort  AF Class 1 (DROP: Low/Medium/High)  AF Class 2 (DROP: Low/Medium/High)    AF Class 3 (DROP: Low/Medium/High)  AF Class 4 (DROP: Low/Medium/High)  Expedited Forwarding</p> <p>Default = Best Effort</p>
DIFFSERV for Signal	<p><i>Options:</i></p> <p>Best Effort  AF Class 1 (DROP: Low/Medium/High)  AF Class 2 (DROP: Low/Medium/High)  AF Class 3 (DROP: Low/Medium/High)  AF Class 4 (DROP: Low/Medium/High)  Expedited Forwarding</p> <p>Default = Best Effort</p>
VLAN Mode	<i>Options:</i> None   Automatic   Static; Default = None
Voice VLAN ID	Voice VLAN ID; Default = 2
VLAN Option ID	VLAN option ID; Default = 176

## Phone Settings

The Phone Settings page allows you to access the following phone categories:

- Volume Control
- Tones Used
- Phone Parameters
- Prefix Entry Settings
- Dial Plan

**Phone Settings**      **Desktop SIP Phone**      IP-Phone

Information    Network    Phone    SwUpgrade    **Personal**    SIP    System    PhoneBook

**Volume control**

Handset Mic	6	Handset Speaker	6
Speaker Mic	6	Speakerphone	6
Headset Mic	6	Headset Speaker	6
Ring Tone Volume	6	Side Tone	5

**Tones Used**

Tone Type	United States	Ring Type	United States
DTMF Relay	RFC2833	RFC2833 PayLoad:	101
RFC2833 DTMF Tone Duration	800 ms		

**Phone Params**

Hold Remind Time	60	Enable DND	<input type="checkbox"/>
Enable Auto Answer	<input type="checkbox"/>	Enable VAD+CNG	<input type="checkbox"/>
Enable Echo Canceller	<input checked="" type="checkbox"/>		
Enable Phone Lock	<input type="checkbox"/>		

**Prefix Entry Settings**

Prefix Entry 0 Type:	Disabled	Prefix Entry 0 Replace:	
Prefix Entry 0 Pattern:			
Prefix Entry 1 Type:	Disabled	Prefix Entry 1 Replace:	
Prefix Entry 1 Pattern:			
Prefix Entry 2 Type:	Disabled	Prefix Entry 2 Replace:	
Prefix Entry 2 Pattern:			
Prefix Entry 9 Type:	Disabled	Prefix Entry 9 Replace:	
Prefix Entry 9 Pattern:			

**Dial Plan**

Dial Plan:	
------------	--

**Save settings**    **Cancel**    **Logout**

## IP Phone Parameters

Volume Control	
Handset Mic	Set input level of handset microphone: 1-10. Default = 6
Handset Speaker	Set output level of handset speaker: 1-10. Default = 6
Speaker Mic	Set input level of hand-free microphone: 1-10. Default = 6
Speakerphone	Set output level of hand-free speaker: 1-10. Default = 6
Headset Mic	Set input level of headset microphone: 1-10. Default = 6
Headset Speaker	Set output level of headset speaker: 1-10. Default = 6
Ring Tone Volume	Set output level of ring tone: 1-10. Default = 6
Side Tone	Set sound level between mouthpiece and earpiece: 1-10. Default = 5
Tones Used	
Tone Type	Select tone type for your country; Default = U.S.
Ring Type	Select ring type for your country; Default = U.S.
DTMF Relay	Select way to send DTMF through in-band or out band mechanism: None   Tone (In-Band)   RFC2833   SIP-INFO. Default = RFC2833
RFC2833 PayLoad	SIP DTMF detection; Default = 101
RFC2833 DTMF Tone Duration	<i>Options:</i> 560ms   640ms   800ms   960ms   1120ms   1280ms; Default = 800
Phone Parameters	
Hold Remind Time	Call on hold reminder tone (10-120 seconds, increments of 10 secs); Default = 60
Enable Auto Answer	Turn on auto answer function; Default = Disabled
Enable DND	Turn on DND function; Default = Disabled
Enable Echo Canceller	Remove echo to improve voice quality; Default = Enabled
Enable VAD+CNG	Voice Activity Detection + Comfort Noise Generator; Default = Disabled
Enable Phone Lock	Lock user's phone; Default = Disabled
Prefix Entry Settings	
Prefix Entry Type (0-9)	<i>Options:</i> Disabled   Add   Replace; Default = Disabled
Prefix Entry Pattern (0-9)	Digit to dial out
Prefix Entry Replace	Replaces digit to dial out
Dial Plan	Determines how to route calls

## Software Upgrade

The Software Upgrade page allows you to access the following s/w upgrade categories:

- Server Items
- Firmware Files

Phone Software Update		IP2007 SIP Phone						IP-Phone	
		Information	Network	Phone	SW Upgrade	Personal	SIP	System	Phone Book
Server Items									
		Server Type:	<input style="border: 1px solid black; padding: 2px; width: 100px; height: 25px; border-radius: 5px;" type="button" value="TFTP"/>		File Directory				
		Server IP Address	<input style="width: 150px; height: 25px; border: 1px solid black; border-radius: 5px;" type="text" value="172.19.5.37"/>		User Password				
		Server User Name	<input style="width: 150px; height: 25px; border: 1px solid black; border-radius: 5px;" type="text" value="root"/>						
Firmware Files									
		Software:	<input style="width: 150px; height: 25px; border: 1px solid black; border-radius: 5px;" type="text" value="app.bin.gz"/>		<input style="border: 1px solid black; padding: 2px; width: 100px; height: 25px; border-radius: 5px;" type="button" value="Update"/>				
		Profile:	<input style="width: 150px; height: 25px; border: 1px solid black; border-radius: 5px;" type="text" value="IP2007.cfg"/>		<input style="border: 1px solid black; padding: 2px; width: 100px; height: 25px; border-radius: 5px;" type="button" value="Update"/>				
		Logo:	<input style="width: 150px; height: 25px; border: 1px solid black; border-radius: 5px;" type="text" value="logo tcbmp"/>		<input style="border: 1px solid black; padding: 2px; width: 100px; height: 25px; border-radius: 5px;" type="button" value="Update"/>				
		Font:	<input style="width: 150px; height: 25px; border: 1px solid black; border-radius: 5px;" type="text" value="local.font"/>		<input style="border: 1px solid black; padding: 2px; width: 100px; height: 25px; border-radius: 5px;" type="button" value="Update"/>				
		WEB Logo:	<input style="width: 150px; height: 25px; border: 1px solid black; border-radius: 5px;" type="text" value="WebLogo.jpg"/>		<input style="border: 1px solid black; padding: 2px; width: 100px; height: 25px; border-radius: 5px;" type="button" value="Update"/>				
		<input style="border: 1px solid black; padding: 2px; width: 150px; height: 25px; border-radius: 5px;" type="button" value="Save Settings"/> <input style="border: 1px solid black; padding: 2px; width: 100px; height: 25px; border-radius: 5px;" type="button" value="Cancel"/> <input style="border: 1px solid black; padding: 2px; width: 100px; height: 25px; border-radius: 5px;" type="button" value="Logout"/> <input style="border: 1px solid black; padding: 2px; width: 150px; height: 25px; border-radius: 5px;" type="button" value="DownLoad Settings"/>							

## S/W Upgrade Parameters

Server Items	
Server Type	TFTP   FTP; Default = TFTP
Server IP Address	FTP server address; Default = 192.168.1.100
Files Directory	FTP directory where the firmware and profile is located
Server User Name	User name of the account on the FTP server; Default = root
User Password	Password of the account on the FTP server
Firmware Files	
Software	app.bin.gz
Profile	IP2007.cfg
Logo	logo tcbmp
Font	local.font
WEB Logo	WebLogo.jpg

## Personal Settings

The Personal Settings page allows you to access the following call categories:  
 Call Forward Settings | Caller Blocking Settings | Speed Dial Entry Settings

**IP2007 SIP Phone**

Personal Settings		Information	Network	Phone	SW Upgrade	Personal	SIP	System	IP-Phone	Phone Book	
<b>Call Forward Settings</b>		Enable Always Forward	<input type="checkbox"/>	Forward to user[@host]							
		Enable On Busy Forward	<input type="checkbox"/>	Forward to user[@host]							
		Enable No Answer Forward	<input checked="" type="checkbox"/>	Forward to user[@host]	voicemail						
		No Answer Timeout (sec)	20								
<b>Caller Blocking Settings</b>		Enable Block Anonymous call	<input type="checkbox"/>								
		Caller Blocking Entry 0									
		Caller Blocking Entry 1									
		Caller Blocking Entry 2									
		Caller Blocking Entry 9									
<b>Speed Dial Entry Settings</b>		Speed Dial Entry 0									
		Speed Dial Entry 1									
		Speed Dial Entry 2									
		Speed Dial Entry 9									
		<div style="text-align: center;"> <input type="button" value="Save Settings"/> <input type="button" value="Cancel"/> <input type="button" value="Logout"/> </div>									

## Personal Parameters

Call Forward Settings	
Enable Always Forward & Forward to user@[host]	Turns on unconditional forward, all calls will automatically forward to the number entered; Default = disabled (unchecked)
Enable On Busy Forward & Forward to user@[host]	Turns on the On Busy Forward function. When extension is busy, calls will forward to the number entered; Default = disabled (unchecked)
Enable No Answer Forward & Forward to user@[host]	Turns on the No Answer Forward function; Default = enabled (checked) - sends calls to Voice Mail
No Answer Timeout (sec)	Sets the timeout for the No Answer Forward function. Default=20
Caller Blocking Settings	
Enable Block Anonymous call	Default = disabled (unchecked)
Caller Blocking Entry (0-9)	10 caller blocking settings. You can set up 10 numbers to prevent them from ring you if a caller calls you from one of these numbers.
Speed Dial Entry Settings	
Speed Dial Entry (0-9)	22 speed dial entries 1-22 (equals speed bins 0-21)

## SIP Page

**ADMIN Function** -- In order to access the SIP parameters, you must log in as a System Administrator. (Logging in as a station user will prevent this page from loading.)

The SIP page allows an administrator to set up the following SIP parameters:

- SIP Proxy Server
- SIP Registrar Server
- Subscriber Information
- Optional SIP Header
- RTP Parameters
- Voice Mail Items
- Codec Settings

Line1 Settings		Desktop SIP Phone						IP-Phone PhoneBook		
		Information	Network	Phone	SwUpgrade	Personal	SIP			System
SIP Proxy Server		SIP Proxy Server	192.168.1.1	Outbound Proxy Server						
		Server Port	5060	SIP Surviving Proxy Server						
SIP Registrar Server		Registrar Server	192.168.1.1	Registrar Outbound Server						
		Registrar Server Port	5060	Registrar Expire Time (sec)	180					
Subscriber Information		Phone Number	100	User Name	100					
		Authorized ID	100	Authorized Password	*****					
		Enable Caller ID	<input type="checkbox"/>	Display Name						
		Locating SIP Server:	<input type="checkbox"/>	SIP Domain:						
		End dial on #:	<input type="checkbox"/>	Dial Timeout (sec):	6					
Optional SIP Header		Optional Header 1			Optional Header 2					
RTP Parameters		RTP Port 1	10002	RTP Port 2	10004					
		Enable Statistic:	<input type="checkbox"/>	Statistic Server:	10000					
Voice Mail Items		Subscribe MWI:	<input type="checkbox"/>	MSG Number	100					
		Voice Mail Server:								
Codec Settings		Codec G.711 u-law	First	G.711u Packet Time	30(ms)					
		Codec G.711 a-law	Second	G.711a Packet Time	30(ms)					
		Codec G.729	Fourth	G.729 Packet Time	30(ms)					
		Codec G.723.1	Third	G.723 Packet Time	30(ms)					
		G.723.1 Bit Rate	5.3kb/s							
NAT items		NAT Type:	None	SIP PING Interval Time (ms):	6					
		STUN Server IP:			STUN Server Port:	3478				
		Extern Router IP:			Extern Signal Port:	5060				
		Extern RTP Port 1:	10002	Extern RTP Port 2:	10004					
		<input type="button" value="Save Settings"/> <input type="button" value="Cancel"/> <input type="button" value="Logout"/>								

## SIP Parameters

SIP Proxy Server	
SIP Proxy Server	SIP proxy server address
Outbound Proxy Server	Outbound proxy server address
Server Port	Proxy server port; Default = 5060
SIP Secondary Proxy Server	Redundant proxy server
SIP Surviving Proxy Server	Creates a SIP proxy loop to find available SIP proxy server
SIP Registrar Server	
Registrar Server	Registrar server address
Registrar Outbound Server	Registrar outbound server address
Registrar Server Port	Registrar server port; Default = 5060
Registrar Expire Time (sec)	Expiration time for REGISTER request; Default = 180
Subscriber Information	
User Name	User name to appear in phone LCD display
Phone Number	Phone number assigned to phone
Authorized ID	Authentication user name
Authorized Password	Authentication password
Enable Caller ID	Enables sending Display Name
Display Name	Name to appear in phone LCD display to remote users
Locating SIP Server	Default = disabled (unchecked)
SIP Domain	Use SIP Domain name instead of IP address
End dial on #	Allows entering # sign for speed dialing after dialing number
Dial Timeout (sec)	Default = 2
Optional SIP Header	
Optional Header 1	Optional SIP header in message
Optional Header 2	Optional SIP header in message
RTP Parameters	
RTP Port 1	RTP Port for channel 1; Default = 10002
RTP Port 2	RTP Port for channel 2; Default = 10004
Enable Statistic	Default = disabled (unchecked)
Statistic Server	Default = 10004
Statistic Port	Default = 10000

Voice Mail Items	
Subscribe MWI	Subscribe to voice mail server
Voice Mail Server	Voice mail server address
MSG Number	Default = voicemail
Codec Settings	
Codec G.711u-law	Set G.711u-law protocol preference: None   First   Second   Third   Fourth; Default = First
G.711u Packet Time	<i>Options:</i> 10-80(ms) Default = 30(ms)
Codec G.711a-law	Set G.711a-law protocol preference: None   First   Second   Third   Fourth; Default = Second
G.711a Packet Time	<i>Options:</i> 10-80(ms) Default = 30(ms)
Codec G.729	Set G.729 protocol preference: None   First   Second   Third   Fourth; Default = Fourth
G.729 Packet Time	<i>Options:</i> 10-80(ms) Default = 30(ms)
Codec G.723.1	Set G.723.1 protocol preference: None   First   Second   Third   Fourth; Default = Third
G.723 Packet Time	<i>Options:</i> 10-80(ms) Default = 30(ms)
G.723.1 Bit Rate	Set the bit rate for G.723.1 protocol: 5.3kb/s   6.3kb/s Default = 5.3kb/s.
NAT Items	
NAT Type	<i>Options:</i> None   STUN   SIP PING   Port Mapping   UDP Heartbeat Default = None
SIP PING Interval Time (ms)	SIP PING frequency; Default = 6
STUN Server IP	STUN server IP address
STUN Server Port	STUN server port; Default = 3478
Extern Router IP	External router address for port mapping
Extern Signal Port	External SIP signaling port for port mapping; Default = 5060
Extern RTP Port 1	External RTP port for port mapping; Default = 10002
Extern RTP Port 2	External RTP port for port mapping; Default = 10004

## System Settings

The System Settings page allows you to access the following system categories:

- Administer Settings
- Time Settings
- Xcelerator IP Feature Items
- Feature Key Settings

System Settings

Desktop SIP Phone

Information Network Phone SwUpgrade Personal SIP System PhoneBook

IP-Phone PhoneBook

**Administer settings**

Administrator Name: admin      User Name: user      Administrator Password: \*\*\*\*\*

User Password: \*\*\*\*\*

Used log server:       Log level: Emergency

System Log Address: 0.0.0.0      System Log Port: 514

System Language: English

**Time settings items**

Auto DST:       Daylight save time: 0

Starts on: Month: JAN, Day: 1, Time: 00:30      Ends on: Month: JAN, Day: 1, Time: 00:30

Time Zone: GMT+12:00 (Auckland, Wellington, ...)

**Gw6000 Feature Items**

Gw6000 Flag: <input checked="" type="checkbox"/>	Feature Key 1 String: 700
GW6000 Key 1 Type: Line	Feature Key 2 String: 701
GW6000 Key 2 Type: Line	Feature Key 3 String: 702
GW6000 Key 3 Type: Line	Feature Key 4 String: 710
GW6000 Key 4 Type: Line	

**Feature Key settings**

Key Input:  SendKey

Set All to Default

Save settings      Cancel      Logout

## System Parameters

Administer Settings	
Administrator Name	The name of the administrator; Default = admin
Administrator Password	The password for the administrator; Default = 1234
User Name	The name of the station user; Default = user
User Password	The password for the user; Default = 1111
User System Log Server	Default = disabled (unchecked)
Log level	<i>Options:</i> Emergency   Alert   Critical   Error   Warning   Notice   Informational   Debug; Default = Emergency
System Log Address	The IP address of the System Server
System Log Port	System Log port number (standard)
System Language	System Language supported is English; Default = English
Time Settings Items	
Auto DST	Automatic daylight saving time flag
Daylight save time	<i>Options:</i> -1   0   +1; Default = 0
Starts on	<i>Options:</i> Month   Day   Time; Defaults = JAN   1   00:30 (respectively)
Ends on	<i>Options:</i> Month   Day   Time; Defaults = JAN   1   00:30 (respectively)
Time Format	TIME_24_HOUR   TIME_12_HOUR Default = TIME_24_HOUR
Time Zone	Default = GMT+12:00 (Auckland, Wellington, ...)
Xcelerator IP Feature Items	
Xcelerator IP flag	Using Xcelerator IP gateway flag; Default = enabled (checked)
Xcelerator IP Key 1 Type (1-4)	<i>Options:</i> None   Line   Feature Key; Default = Line
Feature Key 1 String (1-4)	<i>Options:</i> 700   701   702   710 (respectively)
Feature Key Settings	
Key Input	This field is for Engineering input only ( <i>do not use</i> )

## Phone Book

The *Xcelerator IP* allows each user to store up to 400 name/phone number entries in their individual Phone Book. Once completed, you will be able to place a call to the desired person by clicking the Phone Book entry number next to the person's name/number.

PhoneBook Entry (001-080)	Dial	User Name	Phone Number	Ring Type
001	<input type="button" value="Dial"/>	Mike John	4804436179	1
003	<input type="button" value="Dial"/>		0	0
005	<input type="button" value="Dial"/>		0	0
007	<input type="button" value="Dial"/>		0	0
009	<input type="button" value="Dial"/>		0	0
011	<input type="button" value="Dial"/>		0	0
013	<input type="button" value="Dial"/>		0	0
015	<input type="button" value="Dial"/>		0	0
017	<input type="button" value="Dial"/>		0	0
019	<input type="button" value="Dial"/>		0	0
021	<input type="button" value="Dial"/>		0	0
023	<input type="button" value="Dial"/>		0	0
025	<input type="button" value="Dial"/>		0	0
027	<input type="button" value="Dial"/>		0	0
029	<input type="button" value="Dial"/>		0	0
031	<input type="button" value="Dial"/>		0	0
033	<input type="button" value="Dial"/>		0	0
035	<input type="button" value="Dial"/>		0	0
037	<input type="button" value="Dial"/>		0	0
039	<input type="button" value="Dial"/>		0	0
041	<input type="button" value="Dial"/>		0	0
043	<input type="button" value="Dial"/>		0	0
045	<input type="button" value="Dial"/>		0	0
047	<input type="button" value="Dial"/>		0	0
049	<input type="button" value="Dial"/>		0	0
051	<input type="button" value="Dial"/>		0	0
053	<input type="button" value="Dial"/>		0	0
055	<input type="button" value="Dial"/>		0	0
057	<input type="button" value="Dial"/>		0	0
059	<input type="button" value="Dial"/>		0	0
061	<input type="button" value="Dial"/>		0	0
063	<input type="button" value="Dial"/>		0	0
065	<input type="button" value="Dial"/>		0	0
067	<input type="button" value="Dial"/>		0	0
069	<input type="button" value="Dial"/>		0	0
071	<input type="button" value="Dial"/>		0	0
073	<input type="button" value="Dial"/>		0	0
075	<input type="button" value="Dial"/>		0	0
077	<input type="button" value="Dial"/>		0	0
079	<input type="button" value="Dial"/>		0	0
081	<input type="button" value="Dial"/>		0	0
083	<input type="button" value="Dial"/>		0	0
085	<input type="button" value="Dial"/>		0	0
087	<input type="button" value="Dial"/>		0	0
089	<input type="button" value="Dial"/>		0	0
091	<input type="button" value="Dial"/>		0	0
093	<input type="button" value="Dial"/>		0	0
095	<input type="button" value="Dial"/>		0	0
097	<input type="button" value="Dial"/>		0	0
099	<input type="button" value="Dial"/>		0	0
400	<input type="button" value="Dial"/>		0	0

## Phone Book Parameters

Phone Book Entry (001-400)	
User Name	The user name entered is used to identify the entry while in the Phone Book. Name is used for caller ID when available.
Phone Number	The number entered is used to place a call to the person associated with that phone number. <i>NOTE -- Enter number only (no dashes or spaces).</i>
Ring Type	You can assign a unique ring type (1-10) to help distinguish between types of callers (ex: Executives, Sales Dept., family members, etc.).



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# 6

# Software Upgrade

Read the *Software Upgrade* chapter to learn how to update your IP 2007 telephone with the latest software configuration.



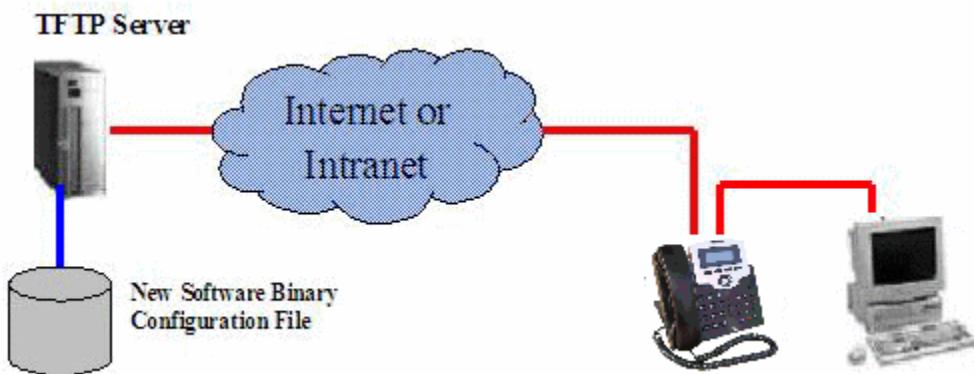
---

## Required Components

- A working IP network that is established and configured for IP phone.
- A TFTP server that is configured on your network (example: TFTP32 server)
- Latest software image or configuration file.

---

## Environment Setup



### *Prepare TFTP Server*

- Make sure the TFTP server has been properly installed in the server.
- Put the software file in the TFTP root directory.

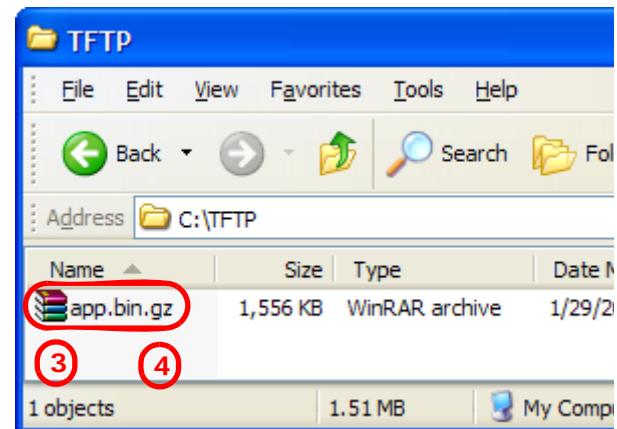
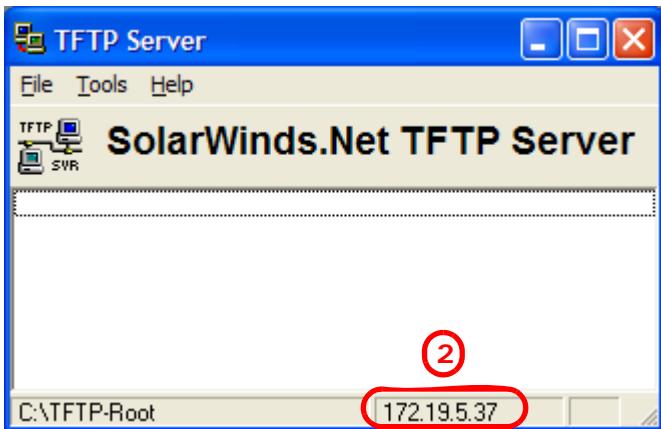
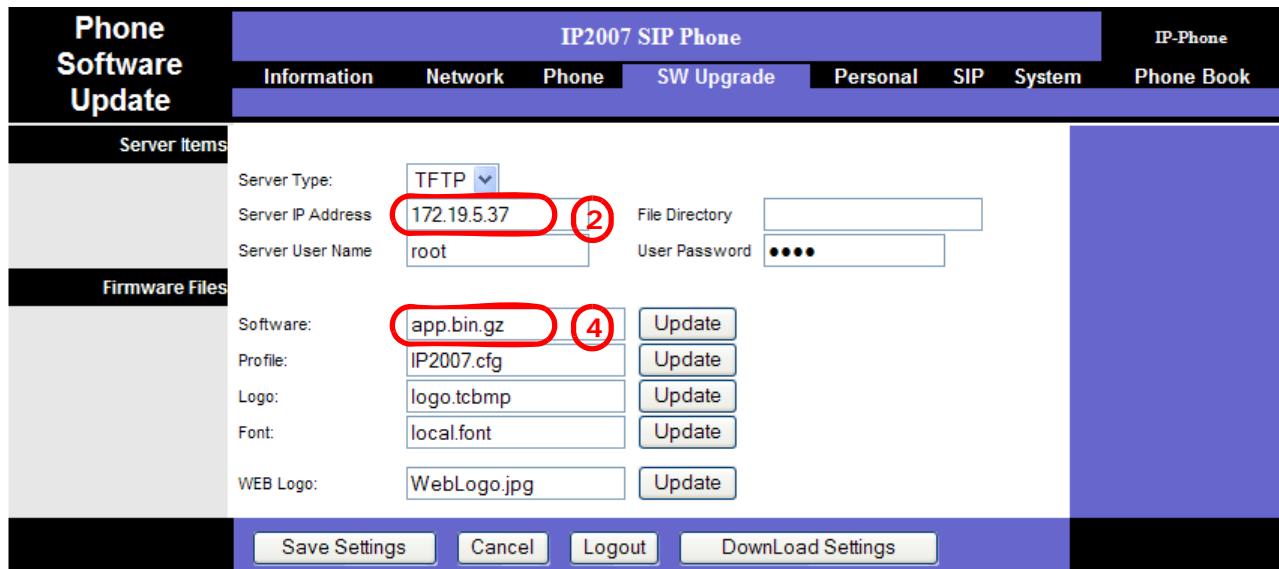
### *Configure Phone for TFTP Access*

- Configure the correct TFTP server address.
- Configure the file name to be downloaded.

## Software Upgrade Procedure

### To upgrade the phone software via the Web:

1. On the Server computer, create a TFTP or FTP folder on the c: drive "root" directory.
2. Make sure the same Server IP address appears in both areas:  
IP2007 SIP Phone web page  
-and-  
TFTP/FTP software program
3. Copy the updated file provided into the newly created TFTP/FTP directory.
4. If necessary, change the name of the software file on the web or in the TFTP/FTP program, so that the filename matches in both places.



5. Click the "Save Settings" button to capture the new settings.
  - During the upgrade process, the phone LCD will display the following messages: "Erasing Flash", "Writing Flash", and "Write Complete".
  - When the software download is complete, the phone will automatically reboot.
6. To verify a successful download, you can use the web interface or the IP2007 phone:  
*via Web* -- click on the "Information" tab to view the Software Version, OR  
*via Phone* -- press the Left soft key, go to "5. Info" and scroll to "Firmware Version".

## Upgrade Troubleshooting

*If the phone download image fails, it can be caused by any of the following reasons:*

- The TFTP server is not working. (You can Ping the server from your computer. You can also login to the server from your computer to verify this.)
- The account information is incorrect (Incorrect user name or password).
- The TFTP server address is incorrect.
- The network configuration of the phone is incorrect (You can issue a Ping from IP Phone to verify this).
- The file is not located where you expected it to be. You can use your TFTP client login to the TFTP server (using the same account) to verify this.
- The file is in position, but the folder setting on the phone is incorrect.
- The file is in position and the folder is correct, but the filename is incorrect.

## TFTP Software - Free Download

If you do not already have an TFTP program, you can go to the link provided and download an application for free ... <http://www.solarwinds.net/downloads>



*Vertical Communications does not provide support for this TFTP product.*

*For technical assistance, contact Solar Winds.*

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Notice : The changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

IMPORTANT NOTE: To comply with the FCC RF exposure compliance requirements, no change to the antenna or the device is permitted. Any change to the antenna or the device could result in the device exceeding the RF exposure requirements and void user's authority to operate the device. The antenna used for this transmitter must be installed to provide a separation distance of at least 20cm from all persons and must not be co-located or operating in conjunction with any other antenna or transmitter.

#### FCC INFORMATION

The Federal Communication Commission Radio Frequency Interference

Statement includes the following paragraph:

The equipment has been tested and found to comply with the limits for a Class B Digital Device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction, may cause harmful interference to radio communication. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

□

--Reorient or relocate the receiving antenna.

--Increase the separation between the equipment and receiver.

--Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

--Consult the dealer or an experienced radio/TV technician for help.

□

The user should not modify or change this equipment without written approval from Vertical Communications. Modification could void authority to use this equipment.

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