



WideBand IP Telephone

User's Manual

Models 6000/7000

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1 Introduction

This user's manual is for the WideBand IP Telephone, models 6000 and 7000. This user's manual will explain the IVR instruction and web configuration for the IP phone. Before using the IP phone, some setup processes are required to make the phone work properly. Please refer to the Setup Menu for further information.

1.1 Hardware Overview

The IP phone has the following interfaces for Networking, and one power connector.

Two RJ-45 networking interfaces

These two interfaces support 10/100Mps Fast Ethernet.

WAN: Connect to the ADSL modem or router.

LAN: Connect to your PC computer. You can then connect to the Internet through the Internet Sharing function of the phone adapter.

1.2 Software Overview

Network Protocol	Tone
<ul style="list-style-type: none"> • SIP v1 (RFC 2543), v2 (RFC 3261) • IP/TCP/UDP/RTP/RTCP • IP/ICMP/ARP/RARP/SNTP • TFTP Client/DHCP Client/PPPoE Client • Telnet/HTTP Server • DNS Client • NAT/DHCP Server 	<ul style="list-style-type: none"> • Ring Tone • Ring Back Tone • Dial Tone • Busy Tone • Programming Tone
Codec	Phone Function
<ul style="list-style-type: none"> • G.711:64kb/s (PCM) • G.723.1:6.3k / 5.3kb/s • G.726:16k / 24k / 32k / 40kb/s (ADCPM) • G.729A:8kb/s (CS-ACELP) • G.729B:adds VAD & CNG to G.729 	<ul style="list-style-type: none"> • Volume Adjustment • Speed Dial Key • Phone Book • Flash
Voice Quality	IP Assignment
<ul style="list-style-type: none"> • VAD: Voice activity detection • CNG: Comfortable noise generator • LEC: Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer 	<ul style="list-style-type: none"> • Static IP • DHCP • PPPoE
Call Function	Security
<ul style="list-style-type: none"> • Call Hold • Call Waiting • Call Forward • Caller ID • 3-way conference 	<ul style="list-style-type: none"> • HTTP 1.1 basic/digest authentication for Web setup • MD5 for SIP authentication (RFC 2069/RFC 2617)
DTMF Function	QoS
<ul style="list-style-type: none"> • In-Band DTMF • Out-Band DTMF • SIP Info 	<ul style="list-style-type: none"> • ToS Field
SIP Server	NAT Traversal
<ul style="list-style-type: none"> • Registrar Server (three SIP accounts) • Outbound Proxy 	<ul style="list-style-type: none"> • STUN
	Configuration
	<ul style="list-style-type: none"> • Web Browser • IVR/Keypad
	Firmware Upgrade
	<ul style="list-style-type: none"> • TFTP • HTTP

2 Keypad Interface

Key Name	Description
1	“, “!” “?” “1” “-“
2	“2”, “a”, “b”, “c”, “A”, “B”, “C”
3	“3”, “d”, “e”, “f”, “D”, “E”, “F”
4	“4”, “g”, “h”, “i”, “G”, “H”, “I”
5	“5”, “j”, “k”, “l”, “J”, “K”, “L”
6	“6”, “m”, “n”, “o”, “M”, “N”, “O”
7	“7”, “p”, “q”, “r”, “s”, “P”, “Q”, “R”, “S”
8	“8”, “t”, “u”, “v”, “T”, “U”, “V”
9	“9”, “w”, “x”, “y”, “z”, “W”, “X”, “Y”, “Z”
0	“0”, “space”
*	“*”, “.” “:”, “@”
#	Start dialing process
TRANSFER	Transfer to the other phone number
REDIAL	Redial the same number again
HOLD	Put the call on Hold
MUTE	Mute your side of the conversation
DND	Do Not Disturb functionality
OK	OK, accept setting
DEL	Delete a word or phone number
UP/DOWN	The up and down keys
LEFT/RIGHT	The left and right keys
MENU	Show phone configuration menu
SPK	Turn on/off Speaker Phone
Line1 - Line3	Available conversation lines
M1 - M5	The 5 speed dial numbers
Conf	Three-way conferencing
Call In	The list of incoming calls
Call Out	The list of outgoing calls
Volume +/-	Alter volume settings

3 Configure the IP Phone

The IP phone provides a built-in web server, so you can use a web browser to configure the phone. First, open a browser and type the IP address of the phone, specifying port 9999. Example: <http://192.168.123.1:9999>

The default LAN port IP is **192.168.123.1**, with a DHCP server enabled. The default WAN port has DHCP client enabled; the IP would be automatically assigned if it is connected to a network with a DHCP server.

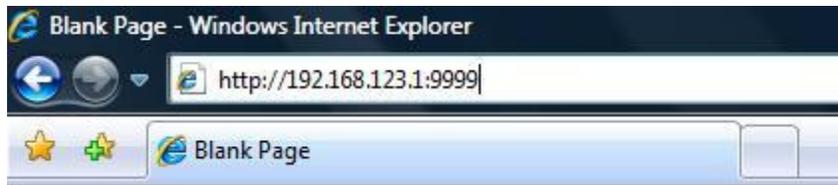
3.1 First Time Login

STEP 1 - Power Up and Connect the Phone

- Connect the WAN Port to a NAT Router or ADSL Modem using a Category 5 network cable. Then connect the LAN Port to a PC, also using a Category 5 network cable.
- Connect the 9V power adapter to the phone's power socket.
- The LCD Panel should then light up, showing the date, time, and "No Service" if a SIP account is not yet set up.
- Pick up the phone, and the LCD Panel will show "IP Dialing...". You should hear a ring back tone. If not, please make sure the RJ-45 WAN port is connected properly.

STEP 2 – Set Up the VoIP Account

- To find the phone's IP address from the phone's keypad, press the MENU key, then arrow down to the "4. Network" option and hit ENTER. Then, arrow down to the "5. Status" and hit ENTER. The MENU key is used for escape, and the ENTER key for selection. The default IP address is 192.168.123.1.
- Next, from your PC, open a browser (Internet Explorer), and enter "http://<IP address of IP Phone>:9999" (e.g. <http://192.168.123.1:9999>) as follows:



- The default login settings are a user named "root", with the password "test". There is another, limited account named "user", also with the password "test". If you use the root account to log in, you can configure all of the settings.

A screenshot of a web form titled "Login VoIP". The form has a light blue background and a white border. At the top, it says "Enter your username and password to login VoIP server". Below this, there are two input fields: "Username" and "Password". At the bottom of the form, there are two buttons: "Login" and "Clear".

- Click "SIP Settings"->"Account Settings" to set up a SIP account in your IP phone.

SIP Accounts Settings

You could set information of service domains in this page.

Realm 1 (Default)

Active: On Off

Display Name:

User Name:

Register Name:

Register Password:

Domain Server:

Proxy Server:

Outbound Proxy:

Subscribe for MWI: On Off

Status: Registered

Please refer to section 5.5 for details. Once you change the setting in the Web Management interface, please remember to click the “Submit” button on that page. Then, you will need to save the settings.

- The LCD panel will show the date, time, and **registered** <phone number> after successful SIP registration.

STEP 3 – Make your First Call

- Pick up the handset; you should now hear a dial tone.
- Enter the phone number you would like to call, optionally ending with a ‘#’ (e.g. 1003#). The ‘#’ tells the phone to call the number immediately. Dialing without # will not call until the auto dial timer (default=5 seconds) elapses.

3.2 Default Reset from Keypad

To restore the factory defaults using the phone’s keypad, press MENU, and arrow down to the “7. Administrator” option, and press ENTER. Then, arrow down to the “3. Default setting” option, and press ENTER. Finally, select “1. Load default”. The LCD panel will read **Loading Program**, and **System Initialized**. Please use the MENU key for escape, and the ENTER key for selection. Then, reboot the phone (option 7 in the Administrator menu).

3.3 Default Setting

LAN IP Address: 192.168.123.1

WAN IP Address: Dynamic Address/DHCP

Web Management Page

Login Name: root

Password: test

4 Application Example

4.1 SIP-to-SIP Calling/Answering

Applications:

Both parties are registered to a SIP server with either fixed real IP or private IP behind a NAT router. The SIP-to-SIP calling works when both calling and answering parties are registered to a SIP server with given registered phone numbers.

Configurations:

- Select “DHCP Client”, and bridge “ON” in the “Network / Network settings” page
- Remember to click the “Submit” button,
- Select Active “ON” in the “SIP settings / Service Domain” page
- Enter the Register Name, Register Password, Proxy Server, and Outbound Proxy,
- Select “ON” in “NAT settings / STUN setting” page, if Outbound Proxy is NOT available.
- Upon successful SIP registration, the LCD will show registered <phone number>.

Callings:

- Pick up the phone, and you should hear a dial tone.
- Press 1688# or 1688 to call the party with the registered SIP phone number 1688. Note that the # key will dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapses.

4.2 SIP to Direct IP Calling

Applications:

The application is for the calling party with an ADSL connection. The calling party is registered to a SIP server with either a fixed real or private IP, behind a NAT router. The answering party has a fixed real IP.

Configurations:

- Same as in Example 4.1
- Select “ON” in “NAT settings / STIN setting” page, if Outbound Proxy is NOT available.
- Upon successful SIP registration, the LCD will show registered <phone number>.

Calling:

- Press Hand-Free key for speakerphone, and you should hear a dial tone.
- Press 211*21*191*4# or 211*21*191*4 to call the party with the real IP address of 211.21.191.4. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

4.3 Direct IP to Direct IP Calling/Answering

Applications:

The applications are for ADSL connection without a NAT router. Both parties have fixed real IP addresses. SIP server registrations are not required in this application.

Configurations:

- Select “Fixed IP”, and bridge “ON” in the “Network / Network settings” page
- Enter the items of IP, Subnet Mask, and Gateway IP
- Click the “Submit” button

Callings:

- Pick up the phone, and you should hear a dial tone.
- Press 211*21*191*4# or 211*21*191*4 to call the party with the real IP address of 211.21.191.4. Note that # key will dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapses. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

4.4 Direct IP to Direct IP Calling within a NAT Router

Applications:

For the calling party with an ADSL connection behind a NAT router, this direct IP calling can work when the answering parties have fixed private IP addresses within the same VPN network, or with fixed real IP addresses:

Configurations:

- Select “Fixed IP”, and bridge “ON” in the “Network / Network settings” page
- Enter the items of IP, Subnet Mask, and Gateway IP
- Click the “Submit” button

Callings:

- Pick up the phone, and you should hear a dial tone.
- Press 192*168*1*51# or 192*168*1*51 to call the party with the private IP address of 192.168.1.51. Press 211*21*191*4 to call the party with the real IP address of 211.21.191.4. In a moment, you should hear a ring back tone, and wait for the called party to answer.

4.5 3-Way Conference Call, Call Waiting, Call Hold

4.5.1 3-Way Conference Calling Application

This is for 3-way conference call among Parties A, B, and C. Three parties are registered to a SIP server with either fixed real IP or private IP. The Flash/Transfer key is used to switch to the other phone line or HOLD, and is quite useful for the 3-way conference call and the call waiting function.

Calling:

- Make a phone call from Party A to the first phone number Party B.
- After the first call is established, press HOLD key (or Flash key) from Party A to hold the call, and Party A should hear a dial tone.
- Make another phone call from Party A to the second phone number Party C.
- After the second call is established, press CONF key from Party A to join in Party B for a three-way conference call.

4.5.2 Call Waiting Application

When a new call is coming while you are talking, you can push the HOLD / FLASH key to switch to the new call. You can push the HOLD / FLASH key to switch between the two calls.

4.5.3 Call Hold Application

You can push the HOLD key to hold the current call for a while, then push HOLD key again to resume talking.

4.5.4 Call Transfer

You can transfer the current call to another user using the Call Transfer function.

- While Party A is having a phone call with Party B, and Party A would like to transfer the call to Party C.
- Party A should press HOLD key, a dial tone will then be heard. Party A should then press TRANSFER key, followed by Party C's number and a “#” number.
- Then Party A's call will end, and Party C's phone will ring. Once Party C picks up the call, he can speak to Party B.

Alternative Way to Transfer a Call

- While Party A is having a phone call with Party B, and Party A would like to transfer the call to Party C.
- Party A will press the TRANSFER key, and a dial tone will be heard. Party A will then dial Party C's number. Once a Waiting tone is heard, Party A should hang up.
- Party C's phone will ring. Once Party C picks up the call, he can speak to Party B.

4.5.5 Call Forward

You can forward incoming calls to a third-party.

- Set up the Call Forward Number using the Web Configuration (Refer to 3.2.1), or using the LCD Menu (“Phone Setting” -> “Call Forward”)
- Press the FORWARD key to enable Call Forwarding

5 Phone Configuration

5.1 System Information

When you login the web page, you can see the phone's current system information, such as the firmware version, company name, etc. You will also see a menu of configuration categories on the left side.

VOIP

- Home
- Phone Book »
- Phone Settings »
- Network »
- SIP Settings »
- Advanced »
- Administration »
- Save Change
- Reboot

System Information

This page illustrate the system related information.

Model Name:	VoIP
Firmware Version:	2.2.488 (712280)
Codec Version:	Tue Aug 28 15:45:53 2007.
Call Status	
Protocol	SIP
Status	Registered
Phone In Use	0 Phone In Use
WAN Status	
MAC address	00:11:78:70:2a:1e
IP Address / Netmask	172.21.173.247/255.255.0.0[DHCP]
Gateway	172.21.1.2
DNS	172.21.1.2/0.0.0.0
LAN Status	
IP Address / Netmask	192.168.123.1/255.255.255.0[Static]

5.2 Phone Book

The phone book contains Speed Dial settings. If you want to use speed dial, just press the speed dial number (from 0~9), then press “#”.

Add New Phone

Position: (0~139)

Name:

Number or URL:

Ringer type: default ▼

Entry Name: Speed dial number
 Number or URL: Real address to be dialed

Check the examples for details.

Example:

Phone Book

You could add/delete items in current phone book.

Phone Book Page: page 1 ▾

Phone	Name	Number or URL	Ringer	Select
0	203	192.168.123.1		<input type="checkbox"/>
1	206	17476433364		<input type="checkbox"/>
2	301	192.168.1.202:5062		<input type="checkbox"/>
3				<input type="checkbox"/>
4				<input type="checkbox"/>
5				<input type="checkbox"/>
6				<input type="checkbox"/>
7				<input type="checkbox"/>
8				<input type="checkbox"/>
9				<input type="checkbox"/>

Example 1: Name 203; URL 192.168.123.1
If a user dialed "203#", the phone would dial 192.168.123.1.

Example 2: Name: 206; URL: 17476433364
If a user dialed "206#", the phone would dial out to (747) 643-3364.

Example 3: Name: 301; URL: 192.168.1.202:5062
If a user dialed "202#", the phone would dial out to 192.168.1.2:5064.

5.3 Phone Settings

Phone Settings contains Call Forwarding settings, along with SNTP, Volume, Do Not Disturb, Caller ID, and Call Waiting configurations.

5.3.1 Call Forwarding

You can set up the phone number you want to forward in this page. There are three types of forwarding modes: **All Forward**, **Busy Forward**, and **No Answer Forward**.

Forward Setting

You could set the forward number of your phone in this page.

All Forward: Off On

Busy Forward: Off On

No Answer Forward: Off On

	Name	Number or URL
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

No Answer Fwd Time Out: (2~8 Ring)

5.3.1.1 All Forward

All incoming calls will be forwarded to the number you choose. You can input the name and the phone number in the URL/Number field.

5.3.1.2 Busy Forward

If you are on the phone, incoming calls will be forwarded to the number you choose. You can input the name and the phone number in the URL field.

5.3.1.3 No Answer Forward

If you do not answer the phone, incoming calls will be forwarded to the number you choose. You can input the name and the phone number in the URL field. You will also need to specify the number of rings before the call is forwarded.

5.3.2 SNTP Setting

You can set up SNTP server IP addresses to automatically get date and time information. You can also set your time zone, and how often you would like to sync the time.

5.3.3 Volume Setting

There are several volume configuration options here. The range of valid values is given beside each field. For example, a range of “(0~14)” indicates that any value between 0 and 14 (inclusive) is valid.

Volume Setting

You could set the volume of your phone in this page.

Handset Volume:	<input type="text" value="8"/>	(0~14)
Speaker Volume:	<input type="text" value="3"/>	(0~14)
Ringer Volume:	<input type="text" value="6"/>	(0~10)
Handset Gain:	<input type="text" value="8"/>	(0~14)
Speaker Gain:	<input type="text" value="9"/>	(0~14)

5.3.4 DND Setting

This setting stands for “Do Not Disturb,” and blocks all incoming calls. You can configure this to always block calls, or to block calls just for a certain amount of time.

Always Block: All incoming calls will be blocked until this feature is disabled.

Block Period: Set an amount of time specified by “From” and “To”.

DND Setting

You could set the do not disturb period of your phone in this page.

DND Always:	<input type="radio"/> On	<input checked="" type="radio"/> Off
DND Period:	<input type="radio"/> On	<input checked="" type="radio"/> Off
From:	<input type="text" value="00"/> : <input type="text" value="22"/>	(hh:mm)
To:	<input type="text" value="00"/> : <input type="text" value="44"/>	(hh:mm)

5.3.6 Call Waiting Setting

You can enable or disable Call Waiting. When you are talking with someone, if there is an incoming call, and call waiting is enabled, you will hear the call waiting tone.

5.3.7 Dial Plan

Dial Plan

You could the set the dial plan in this page.

Drop prefix : Yes No

Replace rule 1: +

Drop prefix : Yes No

Replace rule 2: +

Drop prefix : Yes No

Replace rule 3: +

Drop prefix : Yes No

Replace rule 4: +

Dial now:

Auto Dial Time: (3~9 sec)

Use # as send key: Yes No

Use * for IP dialing: Yes No

Blocked Prefix:

Example 1: Drop prefix: No, Replace rule 1: 002, 8613+8662

1: If what you dialed begins with 8613, "002" is added to the beginning; so the actual number dialed would be [002+8613+xxx].

2: If what you dialed begins with 8662, "002" is added to the beginning; so the actual number dialed would be [002+8663+xxx].

Example 2: Drop prefix: Yes, Replace rule 2: 006, 002+003+004+005+007+009

1: If what you dialed begins with 002, the number "002" would be replaced with "006"; so the actual number dialed would be [006+xxx].

2: If what you dialed begins with 003, the number "003" would be replaced with "006"; so the actual number dialed would be [006+xxx].

Example 3: Drop prefix: No, Replace rule 3: 009, 12

1: If what you dialed begins with 12, the number "12" would be prefixed by number "009"; so the actual number dialed would be [009+12+xxx].

Example 4: Drop prefix: No, Replace rule 4: 007, 5xxx+35xx+21xx

1: If what you dialed begins with 5, followed by any 3 number, the number would be prefixed by "007"; so the actual number dialed would be [007+5xx].

2: If you dialed 534, the number 5 is only followed by two numbers, so it does not match the dialing rule, and the actual number dialed is still 534.

3: If what you dialed begins with 35, followed by any two numbers, it would be prefixed by "007"; so the actual number dialed would be [007+35xx].

4: If you dialed 358822, the number 5 is only followed by two numbers, so it does not match the dialing rule, and the actual number dialed is still [358822].

5.4 Network

In Network you can check the Network status, configure the Network and DDNS settings. On this screen you will find information about the current settings of both the LAN and WAN network interfaces.

5.4.1 WAN Settings

You can configure the IP Phone Network setting in this page.

WAN Settings

You could configure the WAN settings in this page.

LAN Mode: Bridge NAT

WAN Setting

IP Type: Fixed IP DHCP Client PPPoE

IP:

Mask:

Gateway:

DNS Server1:

DNS Server2:

MAC:

Host Name:

PPPoE Setting

User Name:

Password:

Service Name:

5.4.1.1 Mode

There are two modes for the WAN port: Bridge or NAT. Bridge mode: The two WAN and LAN Ethernet ports will be bridged and transparent. NAT mode: The embedded NAT will be enabled. The IP phone comes with a default static IP address of 192.168.1.100. You may select a proper IP type for your network requirements.

5.4.1.2 NAT Settings

To enable embedded NAT, you must set mode of WAN port to "NAT". This embedded NAT is useful for ADSL users without a NAT router, and it separates the WAN port from the LAN port to perform router IP address translation. For the WAN port, please select a proper IP type:

DHCP Client (default): Automatically get an assigned IP from the DHCP server.

PPPoE: This is a common way to connect to your ISP through ADSL. Enter the username and password your ISP provided.

Fixed IP: Assign a static IP address to that network interface.

MAC settings: This must be different between the LAN and WAN. The defaults should be acceptable.

5.4.2 LAN Settings

LAN Settings

You could configure the LAN settings in this page.

LAN Setting

IP:

Mask:

MAC:

DHCP Server

DHCP Server: On Off

Start IP:

End IP:

Lease Time: : (dd:hh)

5.4.2.1 LAN Setting

In this section you can configure the IP address and the subnet mask of the LAN adapter.

5.4.2.2 DHCP Server

There is a built-in DHCP server in the phone adapter. The default is enabled. The range of IP addresses to be assigned is specified by the Start IP and End IP fields. The Lease Time field specifies the amount of time before DHCP expires for a given client.

5.4.3 DDNS Setting

You can configure the DDNS setting on this page. You need to have the DDNS account and input the information properly. If you have a DDNS account with a public IP address, others can call you via the DDNS account. Now most VoIP applications work through a SIP Proxy Server.

DDNS Settings

You could set the configuration of DDNS in this page.

DDNS: On Off

Host Name:

User Name:

Password:

E-mail Address:

DDNS Server:

DDNS Server List:

Type:

Wild Card:

BACKMX: On Off

Off Line: On Off

5.4.4 VLAN Settings

VLANs can be used to separate VoIP traffic from other kinds of traffic, and give it a higher priority. This is one way to guarantee Quality-of-Service within a private network.

VLAN Settings

You could set the VLAN settings in this page.

VLAN Packets: On Off

VID (802.1Q/TAG): (2 ~ 4094)

User Priority (802.1P): (0 ~ 7)

CFI: (0 ~ 1)

5.4.5 PPTP Settings

Here you can connect the phone to a VPN (PPTP) Server. Using a VPN, you can connect to a SIP Server (SIP Proxy or IP PBX) inside a private network, or make a secure phone call.

PPTP Server: The VPN Server to connect to.

PPTP Username/Password: Enter your username and password for the VPN account.

PPTP Settings

You could set the PPTP server in this page.

PPTP: On Off

PPTP Server:

PPTP Username:

PPTP Password:

5.5 SIP Settings

This section will cover configurations such as the Service Domain, Port Settings, Codec Settings, RTP, and RPort Settings.

5.5.1 Service Domain

You may register up to three SIP accounts in the IP phone. You can make calls through the first SIP account, and receive calls from all three. The following figure shows a single example setup.

SIP Accounts Settings

You could set information of service domains in this page.

Realm 1 (Default)

Active: On Off

Display Name:

User Name:

Register Name:

Register Password:

Domain Server:

Proxy Server:

Outbound Proxy:

Subscribe for MWI: On Off

Status: **Registered**

Display Name: The name to be displayed on your phone.

User Name: Enter the username configured on the VoIP server.

Register Name: Enter the Register Name configured for your account on the VoIP server.

Register Password: Enter the VoIP server's account password corresponding to your username.

Domain Server: Enter the Domain server given by your VoIP server.

Proxy Server: Enter the proxy server; this is usually your VoIP server.

Outbound Proxy: This can be left blank, as it is usually the same as the proxy server.

Subscribe for MWI: Click On if your VoIP server supports Voice Mail notification.

Once the status is "Registered", you have successfully registered with the VoIP server.

5.5.2 Port Settings

Here you can set up the SIP and RTP port numbers. Each ISP provider may have different SIP/RTPport settings, so please refer to the ISP to set the port numbers correctly.

Port Settings

You could set the port number in this page.

SIP Port: (10~65533)

RTP Port: (10~65533)

5.5.3 Codec Settings

You can set up Codec priority, RTP packet length, and VAD on this page. Please follow the recommendations of your ISP to configure these settings.

Codec Settings

You could set the codec settings in this page.

Codec Priority

Codec Priority 1:

Codec Priority 2:

Codec Priority 3:

Codec Priority 4:

Codec Priority 5:

Codec Priority 6:

Codec Priority 7:

Codec Priority 8:

Codec Priority 9:

RTP Packet Length

G.711 & G.729:

G.723:

G.723 5.3K

G.723 5.3K: On Off

Voice VAD

Voice VAD: On Off

5.5.4 Codec ID Setting

Sometimes multiple VoIP devices with different codec IDs will cause interoperability issues. If you have trouble communicating with someone, make sure you are using the same codec IDs.

Codec ID Setting

You could set the value of Codec ID in this page.

Codec Type	ID	Default Value
G726-16 ID:	<input type="text" value="23"/> (95~255)	<input checked="" type="checkbox"/> 23
G726-24 ID:	<input type="text" value="22"/> (95~255)	<input checked="" type="checkbox"/> 22
G726-32 ID:	<input type="text" value="2"/> (95~255)	<input checked="" type="checkbox"/> 2
G726-40 ID:	<input type="text" value="21"/> (95~255)	<input checked="" type="checkbox"/> 21
RFC 2833 ID:	<input type="text" value="101"/> (95~255)	<input checked="" type="checkbox"/> 101

5.5.5 DTMF Setting

Here you can configure your DTMF settings. The options are RFC 2833, Inband DTMF, and Send DTMF SIP Info. Please configure this based on your VoIP Provider's recommendation.

5.5.6 RPort Setting

Here you can enable or disable RPort. Please configure this according to your ISP's recommendation.

5.5.7 Other Settings

You will find some miscellaneous SIP settings here. For the text fields, the range of valid values is shown on the right. Notable options here include the QoS settings. For these options, higher values specify a higher priority.

Other Settings

You could set other settings in this page.

Hold by RFC: On Off

Voice QoS (Diff-Serv): (0~63)

SIP QoS (Diff-Serv): (0~63)

SIP Expire Time: (15~86400 sec)

Use DNS SRV: On Off

Send Keep Alives Packet: On Off

Keep Alives Period: (15~250 sec)

Jitter Buffer: (0~250 packets)

5.6 NAT Trans

Here you can configure the STUN settings. This can help your IP phone work properly behind NAT.

STUN Setting

Here you can enable or disable STUN, and set up the STUN server IP address. Please configure these options based on the recommendations of your ISP.

STUN Setting

You could set the IP of STUN server in this page.

STUN: On Off

STUN Server:

STUN Port: (80~65535)

5.7 Others

5.7.1 Auto Configuration Setting

You can set up Auto Configuration to automatically configure your phone from a configuration file hosted on an HTTP, FTP, or TFTP server that you specify.

Auto Configuration Setting

You could enable/disable the auto configuration setting in this page.

Auto Configuration: Off TFTP FTP HTTP

TFTP Server:

HTTP Server: Exp. 60.35.187.30

HTTP File Path: Exp. /download/

FTP Server: Exp. 60.35.17.1

FTP Username:

FTP Password:

FTP File Path: Exp. /file/load

5.7.2 ICMP Setting

This specifies whether or not you want anyone to be able to ICMP ping your phone. This is often disabled on internet-facing computers for security reasons.

5.8 System Authority

Here you can add administrative accounts to your phone, or change the password of an existing account.

System Authority

You could change the login username/password in this page.

New username:

New password:

Confirmed password:

5.9 Save Changes

Whenever you make changes to the phone's configuration, you have to save the changes through this page in order for the changes to be applied. Saving the changes will reboot the phone.

Save Changes

You have to save changes to effect them.

Save Changes:

5.10 Update

5.10.1 Update Firmware

Here you can download a firmware update. To update the firmware:

- Select the firmware code type: CPU or DSP.
- Click the "Browse" button on the right side of the File Location, or type the correct path, including the filename, in the File Location field.
- Select the correct file you want to download to the IP phone, then click the Update button.

5.10.2 Default Settings

Here you can restore the factory defaults to the IP phone. Just click the Restore button to restore the defaults and reboot.

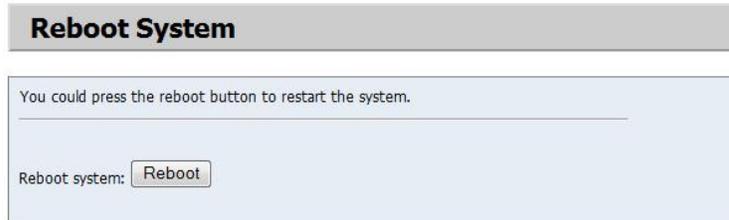
Restore Default Settings

You could click the restore button to restore the factory settings.

Restore default settings:

5.11 Reboot

Here you can reboot the IP phone. You can also reboot the phone using the reboot function on the phone's LCD menu.



The screenshot shows a web interface for rebooting the system. At the top, there is a grey header bar with the text "Reboot System". Below this, a light blue box contains the text "You could press the reboot button to restart the system." followed by a horizontal line. At the bottom of the box, the text "Reboot system:" is followed by a button labeled "Reboot".

6 Troubleshooting Configuration

6.1 The LCD is showing "Ethernet Error!"

Make sure the WAN port of the IP phone is connected properly to your ADSL Modem or Router. Reboot the phone and check again.

6.2 The LCD is showing "No Service"

"No Service" indicates that the IP Phone is not successfully registered to the VoIP server.

- Check if the Account information is set up correctly. Go to "SIP Settings"->"Account Settings" of the Web Management Interface. Make sure all the information is entered properly, especially the password. Press Submit, and reboot the phone.
- Make sure the IP phone can connect to the Internet. Check if your ADSL Modem is working properly.
- Call you VoIP Service Provider to check if there is a problem with their server.

6.3 DO NOT HEAR DIAL TONE?

No dial tone happens when there is a problem with the phone's connection to the internet, or when registration fails. Check the LCD display and refer to 6.1 and 6.2.