LevelOne

VOI-7000 VoIP Phone

VOI-7100 PoE VoIP Phone

User Manual

Safety

FCC WARNING

This equipment may generate or use radio frequency energy. Changes or modifications to this equipment may cause harmful interference unless the modifications are expressly approved in the instruction manual. The user could lose the authority to operate this equipment if an unauthorized change or modification is made.

This 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 instructions, may cause harmful interference to radio communications. 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:

1) Reorient or relocate the receiving antenna.
2) Increase the separation between the equipment and receiver.
3) Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
4) Consult the dealer or an experienced radio/TV technician for help.

CE Declaration of conformity

This equipment complies with the requirements relating to electromagnetic compatibility, EN 55022 class B for ITE, the essential protection requirement of Council Directive 89/336/EEC on the approximation of the laws of the Member States relating to electromagnetic compatibility.

Table of Contents

1. Introduction

The VOI-7000 / VOI-7100 IP Phone is an LCD VoIP Phone with SIP Protocols for Voice over IP (VoIP) applications. This user's manual will explain the keypad instructions and web configurations for the VoIP Phone. IP Phone can make a VoIP call over the ADSL Internet connection, and it provides one RJ45 WAN port for ADSL Internet connections plus one RJ45 LAN port for Notebook PC connection. With the embedded NAT/DHCP server, IP Phone can be easily configured for different network diagrams by PC Web browser and telephone keypads. This is very suitable for ITSP (Internet Telephony Service Providers) and SOHO users to make VoIP calls.

Moreover, with PPTP VPN client supported, user can create secured tunnel between central office and IP Phone, make sure your communication is safe.

VOI-7100 IP phone adopts the latest Power Over Ethernet technology to not only save user investments, simplify network deployment, but also provide centralized power management.

1.1. Features

- SIP v1 (RFC2543), v2 (RFC3261) with MD5 authentication (RFC2069 and RFC 2617)
- RJ45 x 2 for Ethernet WAN and LAN ports
- ITU-T G.711, G.723, G.726, G.729A/B, VAD and CNG for Speech Codec
- ITU-T G.165/168 Echo Cancellation
- LCD Display for registered IP phone number
- Configurations by Web Browser and Telephone Keypads
- Embedded NAT/DHCP Server
- PPPoE/DHCP Client for Dynamic IP plus NAT, DNS, and DDNS Clients
- Support STUN server for NAT Traversal
- Speed Dial, Call Forward/Waiting/Transfer/Hold, and 3-Way Conference Call features
- Direct IP/URL Dial without SIP Proxy or Dial number via SIP server
- Phone book stores up to 140 records
- VPN PPTP Client embedded

VOI-7100 PoE

- Complies with 802.3af PoE Standard

1.2. Packing Contents

Open the shipping cartons of the Switch and carefully unpacks its contents.
The carton should contain the following items:

- SIP IP Telephone
- Power Adaptor
- Cat.5 Cable
- CD User Manual

If any item is found missing or damaged, please contact your local reseller for replacement

2. Hardware Description

2.1. LCD Display and Keypads

The LCD display and keypads of IP Phone are as the following.

Handset

LCD Display

VMS LED

Menu Key

Volume Up / Down

Handsfree

Number Keypads

2.2. Front Panel

VOI-7000

| Power | LAN RJ45 | WAN RJ45 |

VOI-7100

| Power | LAN RJ45 | WAN RJ45 (PoE) |
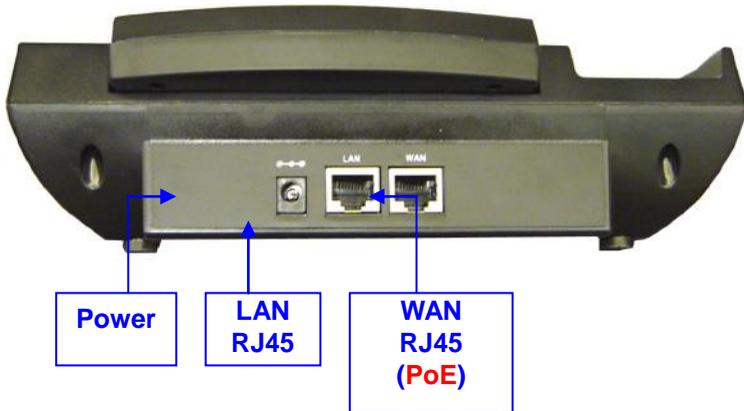
Memory Card

Use the memory card as a name index for speed dialler or extensions.

2.3. Connection Diagram

Without Broadband Router

Power

LAN

WAN

Internet

With Broadband Router

Internet

Broadband
Router

LAN
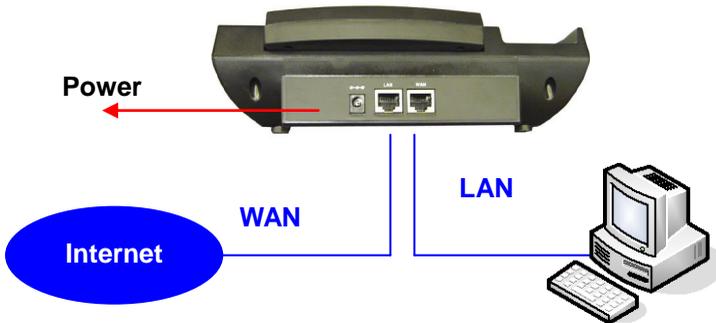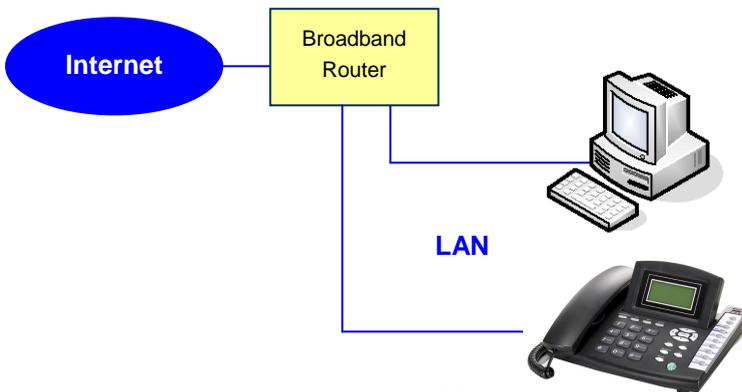
Note

Public Switched Telephone Network (PSTN), which refers to the international telephone system based on copper wires carrying analog voice data

Telephone service carried by the PSTN is often called **plain old telephone service** (POTS).

2.4. Installation

1. Connect IP Phone RJ45 WAN port to NAT Router using a Category 5 LAN cable.

2. Connect IP Phone RJ45 LAN port to Notebook PC using a Category 5 LAN cable.

3. Connect DC power adaptor, and the LCD panel will start showing Loading Program! and System Initialized.

4. The LCD panel will show Date, Time and No service without SIP registration, or <phone number> after successful SIP registration.

5. Pick up the phone, and the LCD panel will show IP Dialling.., and you should hear a dial tone. Please hang up. If not, please check if the RJ45 WAN port is connected.

6. Press MENU / 4.Network / 2.Status from the keypad to check the IP address for IP Phone. The MENU key is used for escape, and the ENTER key for selection. The default IP address is **192.168.1.100.** You need this IP address for Web configurations in Chapter 7.

7. Please refer to VoIP applications examples of SIP registrations, and register IP Phone into your SIP server.

8. The LCD panel will show Date, Time and registered <phone number> after successful SIP registration.

9. Press the Hand-Free key, and you should hear a dial tone. Press 123456# to call the party with the number 123456 registered in the SIP server. Note that # will dial out the number immediately. Dialling without # will not dial out until the auto dial timer (default=5 seconds) elapsed. In a moment, you should hear a ring back tone, and wait for answer.

2.5. Default Setting

IP Address : **192.168.1.100 (LAN)**

Login Name : **root**

Password : **root**

Note

WAN port IP address will depend on the device connect with. For example, Broadband Router provides DHCP server and assign IP address to the IP Phone when it connected.

2.6. Reset to Default

Press **MENU** / **7.Administrator** / **2.Default setting** / **1.Load default** by using Menu and arrow keys to reset back to factory defaults, and the LCD panel will start showing **Loading Program** and **System Initialized**. Please use the **MENU** key for escape, and the **ENTER** key for selection.

Press **MENU** / **7.Administrator** / **6.Restart** to reboot IP Phone.

.

3. Web Configuration

You may enter the IP address from PC Web browser to configure IP Phone. For example, enter **http://192.168.1.100** from Web browser to display login page as follows.

Login VoIP Gateway

Username root

Password ••••

Login   Clear

☐ Remember last login
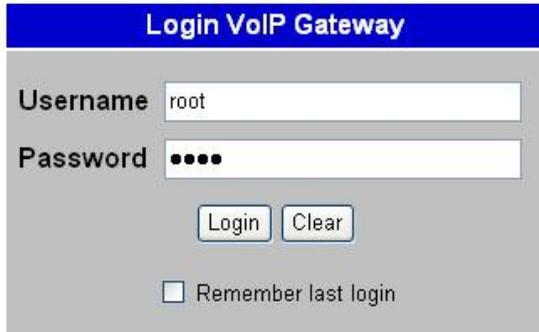
Enter the username and password into the blank field. The default settings are:

Username: **root**

Password: **root**

Click the "**Login**" button will enter the management information page for system setup.

Note

Whenever you change the setting in each Web page, please remember to click the "**Submit**" button in the page, and click the "**Save**" button to save into the non-volatile memory and click the "**Reboot**" button to activate the new settings.

System Information

After login, you will see the system information like firmware version, Codec, etc in this page. You may click the button list at the left hand side to configure the IP Phone.

Caution

VOI-7000 and VOI-7100 use different firmware format, check it carefully before upgrade

3.1. Phone Book

The Phone Book specifies pre-record phone list and speed dialling function, it allows up to 140 records on the phone book.

Phone Book

You could add/delete items in current phone book.

Phone Book Page: page 1 ▼

Phone	Name	URL	Select
0			☐
1			☐
2			☐
3			☐
4			☐
5			☐
6			☐
7			☐
8			☐
9			☐

[Delete Selected] [Delete All] [Reset]

Add New Phone

Position: [   ] (0~139)

Name: [   ]

URL: [   ]

[Add Phone] [Reset]

Input the Position (0~139), Name and URL, then click the "Add Phone" button to enter.

Note

URL can be either complete strings or numbers only, it depends on your service provider.

Example

Phone	Name	URL	Select
1	David	221	□
2	Bill	221090@sipcall.org	□
3	Jone	221080@192.168.12.234	□
4			

Speed Dial Setting

For Speed Dial function you can add/delete Speed Dial number up to maximum 10 entries in Speed Dial Phone List.

Speed Dial Phone List

You could set the speed dial phones in this page.

Phone	Name	URL	Select
0			☐
1			■
2			☐
3			■
4			☐
5			■
6			☐
7			■
8			☐
9			■

[Delete Selected]   [Delete All]   [Reset]

Add New Phone

Position: [　　　] (0~9)

Name: [　　　　　　]

URL: [　　　　　　　　　]

[Add Phone]   [Reset]

If you need to add a phone number into the Speed Dial list, you need to enter the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the "**Add Phone**" button.

If you want to delete a phone number, please select the phone number you want to delete then click "**Delete Selected**" button.
If you want to delete all phone numbers, please click "**Delete All**" button.

Example
Press **[2] [#]** on telephone to Speed Dial the phone number 2 immediately.

3.2. Phone Setting

The sub pages are as follows; Call Forward, SNTP, Volume, Melody (Ringer), DND, Auto Answer, Dial Plan, Flash Time, Call Waiting, Soft-key, Hotline and Alarm functions.

Phone Setting

Call Forward
SNTP
Volume
Melody
Block Settings
Auto Dial
Call Waiting

16
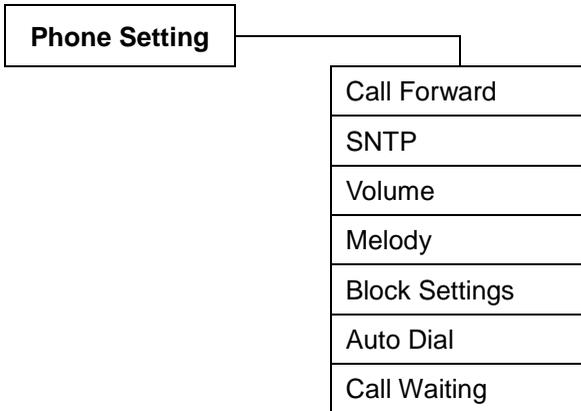
Call Forward

You can have your incoming calls forwarded to a specified destination. You can select the forward mode and enter the forward URL.

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	**URL**
All Fwd No.:		
Busy Fwd No.:		
No Answer Fwd No.:		

No Answer Fwd Time Out:	3	(2~8 Ring)

Submit   Reset

All Forward	All incoming calls are forwarded to the URL you choose.
Busy Forward	The incoming calls are forwarded to the URL when your line is busy.
No Answer Forward	All incoming calls are forwarded when you do not answer the call within specified time period

17

All Fwd No	Specify All Forward number
Busy Fwd No	Specify Busy Forward number
No Answer Fwd No	Specify No Answer Forward number
No Answer Fwd Time Out	Specify the time period before forward calls

Note

You have to set the Time Out Timer to start to forward the calls. It requires "Submit", "Save" and "Reboot" to activate new settings

SNTP

You can setup the primary and second SNTP Server IP Address, to get the date/time information. You may also set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the "Submit" button.

SNTP Settings

You could set the SNTP servers in this page.

SNTP:	⊙ On ○ Off
Primary Server:	time.window.com
Secondary Server:	208.184.49.9
Time Zone:	GMT [+ ▾] [08 ▾]:[00 ▾] (hh:mm)
Sync. Time:	[1] :[0] :[0] (dd:hh:mm)

[Submit] [Reset]

SNTP (Simple Network Time Protocol)

SNTP is an acronym that stands for Simple Network Time Protocol. SNTP enables IP Phone to synchronizing the clocks over Internet Time Servers which it is very precise timekeeping

19

Volume

Raise or lower the sound level by using the Volume Control. For example, if it is difficult to hear the other party's voice;

raise the Handset Volume, or If the other party has difficulty hearing you; raise the Handset Gain level.

Volume Setting

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

Handset Volume:	10	(0~15)
Speaker Volume:	10	(0~15)
Ringer Volume:	6	(0~10)
Handset Gain:	10	(0~15)
Speaker Gain:	9	(0~15)

Submit   Reset

Handset Vol.	Set the volume to hear from the handset
Speaker Vol.	Set the volume to hear from the Speaker
Ringer Vol.	Set the volume of ringer
Handset Gain	Set the volume send out to the other side's handset
Speaker Gain	Set the volume send out to the other side's speaker

Melody

You may set ON the ringer and select different ringer type for Melody settings.

Ringer Settings

You could set your favorite ringer in this page.

Ringer: ○ On   ◉ Off

Ringer Type: [ringer 1 ▼]

[Submit] [Reset]

Note

Because the default ringer is **ringer 1**, it means the setting will remain as **off** if you switch **On** and select **ringer 1**

Block Setting

You can setup the Block Setting to keep the phone silence. You may set this feature when you are in a meeting or busy.

Block Setting

You could set the block period of your phone in this page.

Always Block: ○ On   ⊙ Off

Block Period: ○ On   ⊙ Off

From: [00] : [00]   (hh:mm)

To: [00] : [00]   (hh:mm)

[Submit]   [Reset]

Always Block	All incoming call will be blocked when enabled
Block Period	Set a time period and the phone will be blocked during the time period When the time in "**From**" is greater than "**To**", the Block time will be from Day 1 to Day 2.

Auto Answer

You may enable the Auto Answer function to answer the incoming call by FXO port. When the ring count exceeds the number set in Auto Answer Counter, the FXO port will auto answer and allow for extension calls from PSTN to VoIP and vice versa.

For the incoming call from the Internet, the FXO port will answer with a PSTN dial tone and allow caller to redial to PSTN phone number. For the incoming call from PSTN, the FXO port will answer with a short beep tone and allow caller to redial to VoIP number. PIN Code is used to prevent from call piracy. The caller needs to enter the right PIN code followed by "**#**" to get the PSTN dial tone. Incorrect PIN Code will result in call disconnect. The Auto Answer is disabled at default.

Auto Answer

You could enable/disable the auto answer in this page.

Auto Answer:	○ On ⊙ Off
Auto Answer Counter:	3 (0~8)
PIN Code Enabled:	○ On ⊙ Off
PIN Code:	

[Submit] [Reset]

Auto Answer	Enable this function to answer the incoming calls from PSTN line automatically. It allows user to place call to Internet again.
Auto Answer Counter	Set time period before phone pick up the calls automatically
PIN Code Enabled	Enable the call restriction from PSTN line to VoIP or vice versa.
PIN Code	Set the PIN code. User requires to enter correct code which correspond with before get second dial tone.

Note

This function is only available on **VOI-7100**

Auto Dial

Auto dial timer settings can be set in this page. The auto dial timer specifies the elapse time between the dialling digits.

Auto Dial Setting

You could the time slice to auto dial in this page.

Auto Dial Time: 5 (3~9 sec)

Submit   Reset

Auto dial Time   The inter-digit timer. Default is 5 seconds

Call Waiting

You can enable the call waiting function in this page.   It allows answering another coming call by pressing flash key while holding the current call. You may switch back to previous call by pressing flash key again.

Call Waiting Setting

You could enable/disable the call waiting setting in this page.

Call Waiting: ⊙ On   ○ Off

[Submit]   [Reset]

Note

Flash key means On-hook and Off-hook in short period without hanging up the call.

3.3.   Network

You can check the Network status, and configure the WAN, LAN, DDNS, VLAN, DMZ, Virtual Server and PPTP settings in this section.

Network

Status
WAN
LAN
DDNS
VLAN
DMZ
Virtual Server
PPTP
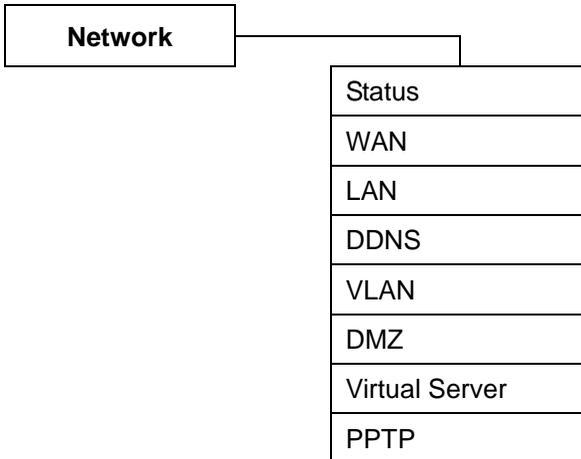
Network Status

You can check and show the current Network settings in this page. Interface 0 shows WAN port status, and Interface 1 shows LAN port status.

Network Status

This page shows current status of network interfaces of the system.

Interface 0	
Type:	DHCP Client
IP:	192.168.50.170
Mask:	255.255.255.0
Gateway:	192.168.50.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

Interface 1	
Type:	DHCP Server
IP:	192.168.123.1
Mask:	255.255.255.0
Gateway:	192.168.123.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

Bridge

The Bridge setting is used to configure the Ethernet port connects to the ADSL Modem/Router, or Ethernet switch.

Bridge Settings

You could configure your bridge settings in this page.

TCP/IP Configuration

IP Type:	⊙ Fixed IP ○ DHCP Client ○ PPPoE
IP:	192.168.50.136
Mask:	255.255.255.0
Gateway:	192.168.50.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1
MAC:	00116bc0144c

PPPoE Configuration

User Name:	
Password:	

Bridge

Bridge:	⊙ On ○ Off

Submit   Reset

Bridge	When setting to Bridge Mode, the WAN and the LAN ports will be bridged.
IP Type	There are three selections for Bridge: **Fixed IP**, **DHCP Client**, and **PPPoE** modes.
	For Fix IP Mode, please make sure the IP address. Net Mask, Gateway, and DNS settings are suitable in your current network environment.
	For PPPoE Mode, you have to enter correct username and password to get the IP address from your Internet Service Provider.

NAT Settings

This embedded NAT is useful for ADSL users without NAT router, and it separates the WAN port from the LAN port to perform router IP address translation. Connect your PC to the LAN port, set your PC as DHCP Client mode, and then the PC will get an IP address from the IP Phone automatically.

NAT Settings

You could configure your NAT settings in this page.

LAN Setting

IP:	192.168.50.136
Mask:	255.255.255.0
MAC:	00116bc0144c
DHCP Server:	○ On ◉ Off
Start IP:	101
End IP:	200
Lease Time:	1 : 0 (dd:hh)

WAN Setting

IP Type:	◉ Fixed IP ○ DHCP Client ○ PPPoE
IP:	0.0.0.0
Mask:	0.0.0.0
Gateway:	0.0.0.0
DNS Server1:	0.0.0.0
DNS Server2:	0.0.0.0
MAC:	00116bc0144c
PPPoE	
User Name:	
Password:	

Submit   Reset

DDNS Setting

DDNS (Dynamic DNS) A service that lets anyone on the Internet gain access to resources on your local network when the Internet address of that network is constantly changing. When it detects that the IP address of the cable or DSL modem has changed, it notifies the DDNS service provider of the new address.

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:	User Input
Type:	dyndns
Wild Card:	on

BACKMX:	On Off
Off Line:	○ On ⦿ Off

[Submit] [Reset]

You need to have a DDNS account before configuring the DDNS setting. Usually, most of the VoIP applications are working with a SIP Proxy Server. Nonetheless, you may have a DDNS account with a public IP address, and others can call you via the DDNS account.

Example

In this example, the other user can place VoIP calls to your IP Phone directly by your domain address.

3.4. SIP Settings

You can setup the Service Domain, Port Settings, Codec Settings, RTP Setting, RPort Setting and Other Settings for SIP Proxy Server registrations in this page.

SIP Setting	
	Service Domain
	Codec
	Codec ID
	Other
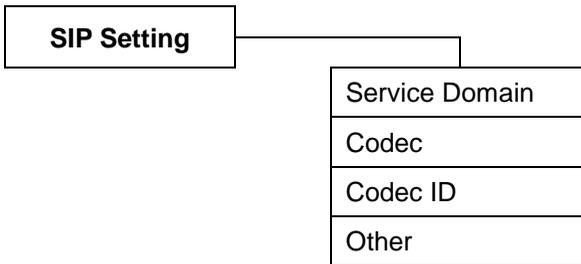
Understanding the SIP

SIP, the Session Initiation Protocol, is a signalling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. SIP was developed within the IETF MMUSIC (Multiparty Multimedia Session Control) working group, with work proceeding since September 1999 in the IETF SIP working group.

SIP enabled PBXes and/or SIP User Agents utilize the Session Initiation Protocol(SIP) to interconnect and to establish voice sessions between each other over an IP Network. SIP Telephony has emerged as a viable alternative to legacy (TDM) and fixed-line circuits for the establishment and transmission of voice communications.

Service Domain

You may register up to three SIP accounts in the IP Phone. You can call your friends via firstly enabled SIP account and receive the phone calls from all the three SIP accounts. It supports 3 services, allow user register on different service providers. Click "Active" ON to enable the Service Domain, then enter the following items:

Service Domain Settings

You could set information of service domains in this page.

Realm 1	
Active:	○ On ⊙ Off
Display Name:	
User Name:	
Register Name:	
Register Password:	
Domain Server:	
Proxy Server:	
Outbound Proxy:	
Status:	Not Registered

Realm 2	
Active:	○ On ⊙ Off
Display Name:	
User Name:	
Register Name:	

Realm (1 ~ 3)

Active	Enable the SIP account
Display Name	Enter the name you want to display
User Name	Enter the User Name given by your ITSP
Register Name	Enter the Register Name given by your ITSP
Register Password	Enter the Register Password given by your ITSP
Domain Server	Enter the Domain Server given by your ITSP
Proxy Server	Enter the Proxy Server given by your ITSP
Outbound Proxy	Enter the Outbound Proxy of ITSP. If not provided, you may skip this
Status	Shows register status

When it shows "Registered" in the Register Status, it indicates a successful registration to the ITSP, and the "**REG**" LED will On. The IP Phone is then ready for VoIP call. If you have more than one SIP account, please follow the steps to register to other ITSPs.

Note

After you finished the setting, please click the "**Submit**" button and click **Save Change**

DTMF Setting

☉ 2833

◯ Inband DTMF

◯ Send DTMF SIP Info

Port Setting

SIP Port:	5060	(1024~65535)
RTP Port:	60000	(1024~65535)

STUN Setting

STUN:	◯ On ☉ Off
STUN Server:	stun. xten. com
STUN Port:	3478

Submit   Reset

DTMF Setting   You can setup the options for DTMF function in this page. The options include RFC2833 (Outband DTMF), Inband DTMF, and Send DTMF SIP info. The default is set at Inband DTMF. If you are making two-stage callings for extension to PSTN, you may need to select Outband DTMF option.

Port Setting   The SIP Port and RTP Port numbers are default at 5060 and 60000, respectively. The RTP port number must be even number. If you have more than one VoIP phones under the same NAT router, it is recommended that different RTP port numbers be assigned to each of IP Phones.

STUN Setting   The STUN function must be enabled to
                    work properly behind NAT when
                    registered in SIP server. You may enter
                    the STUN server IP address and the
                    STUN port number. Please check your
                    ITSP for STUN information.

Codec

You can setup the Codec priority, RTP packet length, and VAD function in this page. Codecs basically convert analog signals to digital form and vice versa.

Codec Settings

You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.711 u-law
Codec Priority 2:	G.711 a-law
Codec Priority 3:	G.723
Codec Priority 4:	G.729
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40
Codec Priority 9:	GSM

RTP Packet Length	
G.711 & G.729:	20 ms
G.723:	30 ms

G.723 5.3K	
G.723 5.3K:	○ On ⊙ Off

Voice VAD	
Voice VAD:	○ On ⊙ Off

[Submit] [Reset]

Codec Priority	Adjust Codec priority to meet your requirement, lower number shows higher priority.
RTP Packet Length	Adjust Codec g711, g729 and g723 packet length
G.723 5.3K	Enables 5.3K bit/s rate when use g723
Voice VAD	VAD (Voice Activity Detection) is used to reduce the transmission rate during inactive speech periods. VAD classifies the input signal into active speech, inactive speech or background noise. Based on the VAD decisions.

One of the most important factors is how much bandwidth is used for each VoIP call. The higher the CODEC bandwidth is, the higher the cost of each call across the network will be.

Following is a list of CODECs and their associated bandwidth:

G.711 The G.711 pulse code modulation (PCM) coding scheme uses the most bandwidth. G.711 takes samples 8000 times per second, each of which is 8 bits in length, for a total bandwidth of 64,000 bps.

G.726 The G.726 adaptive differential pulse code modulation (ADPCM) coding schemes use somewhat less bandwidth. While each coding scheme takes samples 8000 times per second like PCM, G.726 ADPCM uses 4, 3, or 2 bits for each sample, thereby resulting in total required bandwidths of 32,000, 24,000, or 16,000 bps.

G.729 The G.729 and G.729A conjugate structure algebraic code excited linear prediction (CS-ACELP) coding scheme also compresses PCM using advanced codebook technology. It uses 8000 bps of total bandwidth.

G.723 The G.723 and G.723A multipulse maximum likelihood quantization (MPMLQ) coding schemes use a look-ahead algorithm. These compression schemes result in a required bandwidth of 6300 or 5300 bps.

GSM GSM (Global System for Mobile communications) is a cellular phone system standard popular outside the USA. The speech signal is divided into blocks of 20 ms. These blocks are then passed to the speech codec, which has a rate of 13 kbps, in order to obtain blocks of 260 bits.

Note

The network administrator should balance the need for voice quality against the cost of bandwidth in the network when choosing CODECs.

Codec ID

You can setup the Codec ID in this page. You need to follow the ITSP suggestion to setup these items.

Codec ID Setting

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

Codec Type	ID		Default Value
G726-16 ID:	23	(95~255)	☑ 23
G726-24 ID:	22	(95~255)	☑ 22
G726-32 ID:	2	(95~255)	☑ 2
G726-40 ID:	21	(95~255)	☑ 21
RFC 2833 ID:	101	(95~255)	☑ 101

Submit   Reset

Note

Two VoIP devices with different Codec ID will cause the interoperability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use, then you can change your Codec ID.

Other Settings

You can setup the Hold by RFC and QoS in this page. To change these settings please follows your ITSP information. The QoS is used to set the voice packet priority. Higher value other than zero will get higher priority for the voice packets in Internet. However, the QoS function still needs to cooperate with the other Internet devices. SIP Expire Time depends on your ITSP required.

Other Settings

You could set other settings in this page.

Hold by RFC:	○ On ● Off
Voice QoS (Diff-Serv):	40 (0~63)
SIP QoS (Diff-Serv):	40 (0~63)
SIP Expire Time:	60 (15~86400 sec)
Use DNS SRV:	○ On ● Off

Submit   Reset

Note

For more information about these advanced features, please ask your network administrator or service provider help desk.

3.5. Auto Config

Auto Configuration function can be used to download the original configurations stored in the TFTP or FTP server.

```
┌─────────────────┐
│  Auto Config    │────────────────┐
└─────────────────┘         ┌───────────────────┐
                            │ Auto Config       │
                            ├───────────────────┤
                            │ ICMP Settings     │
                            └───────────────────┘
```
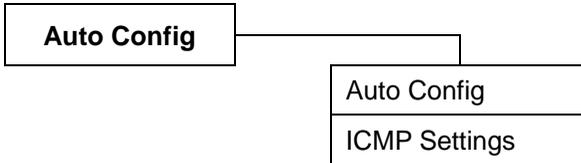
Auto Config

This feature allows service provider to provision their customer's IP Phone, end-to-end. By employing a TFTP / FTP server, the provisioning server writes the configuration files needed to automatically configure the IP Phone. Before enabling this auto configuration, you must select Bridge ON and Fixed IP type in Network settings.

Auto Configuration Setting

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

Auto Configuration: ⦿ Off   ○ By TFTP   ○ By FTP

TFTP Server:	0.0.0.0
FTP Server:	0.0.0.0
FTP Username:	
FTP Password:	

[Submit] [Reset]

Note

Auto Config is idea for ITSP or large network group to deploy VoIP devices easily

ICMP Setting

The ICMP function is to echo when someone ping this device. This can prevent from hacker attacking the device by not echoing.

ICMP Setting

You could enable/disable the ICMP setting in this page.

ICMP Not Echo: ○ On ⊙ Off

Submit Reset

ICMP Not Echo   ICMP is used to acknowledge and echo for the Ping request.   IP Phone will echo for the IP Ping request at default. Selecting ON for ICMP Not Echo will ignore the IP Ping request and keep silent. This is sometime useful for network security.

3.6. User Password

You may create the login name and password in this page.

User Password

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

New username:

New password:

Confirmed password:

Submit Reset

3.7. Save Change

You must save the changes you have made, and click the Save button. After clicking the "Save" button, the IP Phone will save the new settings into ROM and reboot it automatically
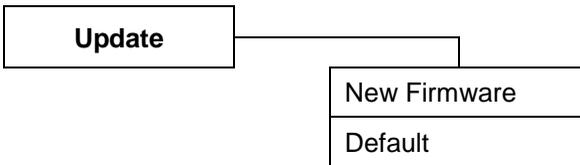
Save Changes

You have to save changes to effect them.

Save Changes: Save

3.8. Update

User can update the IP Phone firmware when new firmware is available. Make sure no power off during the firmware upgrade.

Update		
	New Firmware	
	Default	

Caution

VOI-7000 and VOI-7100 use different firmware format, check it carefully before upgrade

Update Firmware

Update Firmware

You could update the newest firmware.

Method: ⦿ Local PC   ◯ TFTP

Local PC

Code Type:	Risc ⌄
File Location:	[ ] [Browse...]

TFTP

TFTP Server:	192.168.1.250

[Update] [Reset]

The IP Phone provides two methods, HTTP or TFTP, to update new firmware
as the following steps:

1. Select the firmware code type, Risc or DSP code. (mostly for Risc code)
2. Click the "Browse" button to choose the updated file location for HTTP
   download, or
3. Select TFTP and enter the IP address of TFTP server for firmware
   download, then click the "Update" button.

Caution

VOI-7000 and VOI-7100 use different firmware format, check it carefully before upgrade

Do Not power off during the upgrade processing, it may damage the IP Phone

For update firmware by TFTP, the TFTP server is required. Contact your network administrator for more information

Default Setting

You can restore the IP Phone to factory default in this page. By clicking the "Restore" button, the IP Phone will restore to default and automatically restart again.

Restore Default Settings

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

Restore default settings: [Restore]

3.9. Reboot

You may click the Reboot button to restart, then IP Phone will automatically reboot with the stored configurations.

Reboot System

You could press the reboot button to restart the system.

Reboot system: [Reboot]

4. LCD Display and Keypad

You can use keypad to configure and to check the status of IP Phone. Make sure that the WAN port is connected to ADSL Ethernet, or you may hear a busy tone from the telephone.

4.1. Keypad Descriptions

Key	Descriptions
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 dialling process.

Key Name	Descriptions
PHONEBOOK	To show the phone book list.
CID	To show Incoming calls history.
OUT	To show Outgoing calls history.
ENTER	The "Enter" is for setting selections.
MENU	The "Menu" key is to set the IP Phone
▲ ▼ ◄ ►	Use navigate keys to select menu items.
VOL+/-	Set the volume High/Low
TRANSFER	"Transfer" to the other phone number.
DEL.	To delete parameters while change the setting
SPK	This is for Speaker Phone.
M1~M6	These are for 6 speed dial numbers.
HOLD	"HOLD" the current call.
CONF.	For 3-Way Conference Calls.
FWD	This is for forward function.
DND	To enable/disable DND for call reject.
ENTER	The "Enter" is for setting selections.

4.2. LCD Menu

1. Phone Book

1.Search	Search Phone Book
2.Add entry	Add new phone number to phone book
3.Speed dial	Add speed dial phone number
4.Erase all	Erase all phone number

2. Call History

1.Incoming calls	Show all incoming call.
2.Dialed numbers	Show all dialled call.
3.Erase record	Delete call history.

    1 All: Delete all call history.

    2 Incoming: Delete all incoming call.

    3 Dialled: Delete all dialled out call.

3. Call setting

1 Call forward

    1.All Forward:

        Activation: To Enabled/Disabled this function.
        Number: Forward to a registered or URL Number.

    2.Busy Forward.

        Activation: To Enabled/Disabled this function.
        Number: Forward to a registered or URL Number.

    3.No Answer Forward.

        Activation: To Enabled/Disabled this function.
        Number: Forward to a registered or URL Number.

4.Ring Timeout:	Set the Ring times to start the Forward function (2 ~ 8 Rings)	

2 Do not Disturb

1.Allways:	Block all calls
2.By Period:	Block calls by the period time
3.Period Time:	Set the start time and end time to Block calls.

3 Alarm Setting

1.Activation:	Enable/Disable alarm
2.Alarm Time:	Set the alarm time

4 Date/Time setting

1.Date & Time: Set the IP Phone Date and Time.

2.SNTP setting:

SNTP:	Enabled / Disable SNTP.
Primary SNTP:	Set Primary SNTP server IP address or URL.
Secondary SNTP:	Set Secondary SNTP server IP address or URL.
Time zone:	Set Time zone.
Adjustment Time:	Set adjustment time period.

5 Volume and Gain

1.Handset volume:	Set Handset volume from 0~15 (max.) for you to hear.
2.Speaker volume:	Set Speaker phone volume from 0~15 (max.) for you to hear.
3.Handset Gain:	Set Handset Gain from 0~15 (max.) for remote site to hear.
4.Speaker Gain:	Set Speakerphone Gain from 0~15 (max.) for remote site to hear.

6 Ringer

1.Ringer volume:	Ringer volume selection from 0~15 (max.).

2.Ringer type: Ringer tone selection from
    1~4.

7 Auto Dial   Auto Dial time selection from 3~9

seconds.

4. Network

1 General

   1 IP Type:
>   Fixed IP client
>   DHCP client:
>   PPPoE client:

   2 Fixed IP setting:
>   Host IP
>   Subnet mask
>   Gateway IP

   3 PPPoE setting:
>   User name
>   Password

   4 DNS Server
>   Primary DNS
>   Secondary DNS

   5 VLAN
>   VLAN for Phone
>   VLAN for NAT

2 Status: Show IP addresses and MAC address.

5. SIP Settings

1 Service Domain

    1 First realm

        Activation:

        User name:

        Display name:

        Register name:

        Register password:

        Proxy server: Proxy Server IP Address

        Domain server: Domain Server IP Address

        Outbound proxy: Outbound Proxy IP Address

    2 Second realm

    3 Third realm

2 Codec

    1 Codec type

G.711 uLaw:	G.711 uLaw
G.711 aLaw:	G.711 aLaw
G.723:	G.723.1
G.729:	G.729A
G.726-16:	G.726 16Kbps
G.726-24:	G.726 24Kbps
G.726-32:	G.726 32Kbps
G.726-40:	G.726 40Kbps

    2 VAD   Voice Activity Detection Enable/Disable.

3 RTP Setting

    1 Outband DTMF: Outband DTMF Enabled/Disabled
    2 Duplicate RTP

No duplicate:	No resend voice packets.
One duplicate:	Resend voice packets once.
Two duplicate:	Resend voice packets twice.

4 RPort Setting   RPort Enabled/Disabled.

5 Hold by RFC Hold by RFC3261 Enabled/Disabled.

6 Status   Use Up/Down keys to show the SIP
Proxy register status.

6. NAT Transversal

1 STUN setting

        1.STUN: STUN Enabled/Disabled

        2.STUN server: Server IP Address

7. Administrator

1 Auto Config

        1 Config Mode: Select Disable/TFTP/FTP/HTTP for auto config function with server.

        2 TFTP server: Set the TFTP server IP address.

        3 FTP server: Set the FTP server IP address.

        4 FTP Login Name: Set the login name to the FTP server.

        5 FTP Password: Set the Password to the FTP server.

2 Upgrade System   You can restore to the default setting

        1 Upgrade Now: Select Yes/No to upgrade with the upgrade Server.

        2 Upgrade via: Select Disable/TFTP/FTP/HTTP to do upgrade.

        3 Status:

        4 Reset Time: Set Yes/No to reset time.

3 Default setting   To load/abort the default setting

4 System Authority   Must enter the password first for SIP setting.   Default is "**root**".

5 Version   This shows the firmware version.
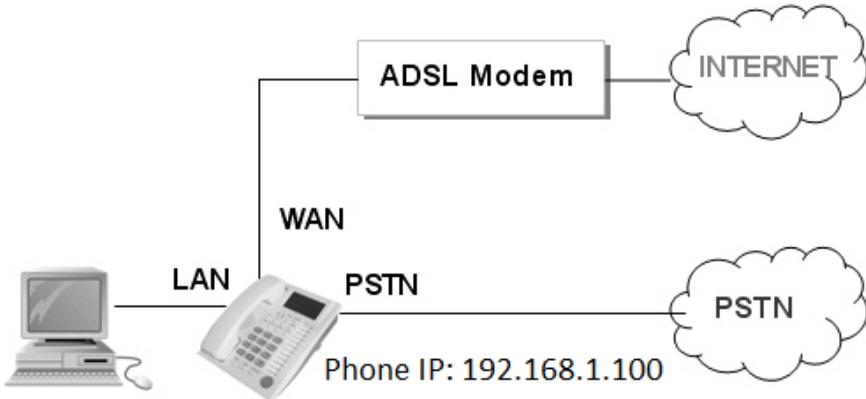
6 Watch Dog   This enables Watch Dog function for debugging.

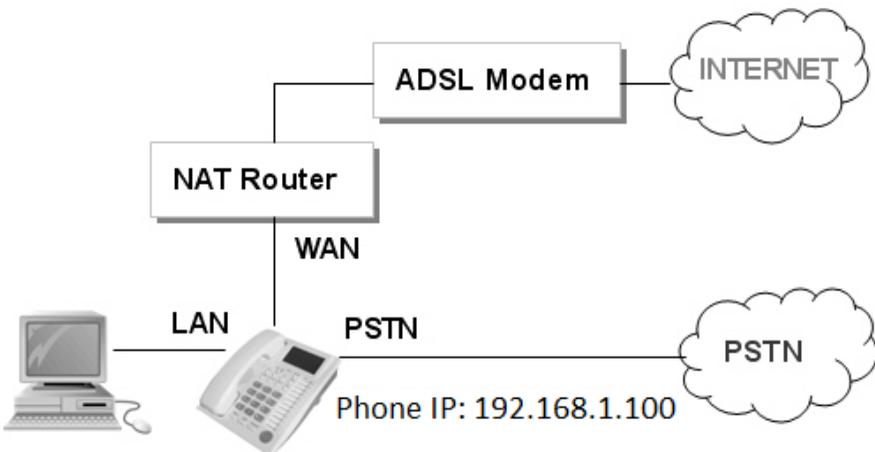### 7 Restart   This function will restart your IP Phone

5. Application Example

You can use PC Web browser to configure IP Phone.   For example, enter http://192.168.1.100 from PC web browser.

A. ADSL Connections with NAT enabled in IP Phone

B. ADSL Connections with external NAT Router

5.1. PSTN Calling

Applications:

VOI-7100 is default at the VoIP mode.　For PSTN calls, you may just pick up the phone, press **0*** key or **PSTN** function key, and dial directly to the PSTN number like a normal telephone.

Configurations:

The "**Auto Answer**" is OFF at default, and the function of extension call from SIP to PSTN is disabled.　The FXO port is for PSTN only and no configuration is needed.

Calling/Answering:

1. Pick up the phone and press **PSTN** function key, and you should hear a dial tone.
2. Press, e.g. **7654321**, to call the PSTN party with **7654321**. In a moment, you should hear a ring back tone, and wait for the called PSTN party to answer.
3. For receiving PSTN incoming calls, you just pick up the phone to answer when ringing.

5.2. SIP-to-SIP Calling

Applications:

The SIP-to-SIP calling works when both calling and answering parties are registered to SIP server with given registered phone numbers. The ADSL connections can be as in either Diagrams A or B.   Both parties are registered to SIP server under NAT router.   For Diagram A without NAT router, you may select NAT mode to enable the embedded NAT router.   For Diagram B with external NAT router, you may select Bridge mode to disable the embedded NAT.

Configurations:

1.   Select either "NAT" or "Bridge" in accord with your network in "WAN settings" page,
2.   Select "DHCP Client" to automatically get an IP address from NAT router.
3.   Remember to click the "Submit" button,
4.   Select Active "ON" in the "SIP settings / Service Domain" page,
5.   Enter the Register Name, Register Password, Proxy Server, and Outbound Proxy,
6.   Select "ON" in the "STUN setting", if Outbound Proxy is NOT available.
7.   Upon successful SIP registration, the **REG** LED indicator will be **ON** and the LCD will show registered <phone number>.

Callings:

8.   Pick up the phone, and you should hear a dial tone for VoIP mode.
9.   Press 1688# or 1688 to call the party with the registered SIP phone number 1688.   Note that # key will dial out the number immediately.

Dialling without # will not dial out until the auto dial timer (default=5 seconds) elapsed.

5.3. SIP-to-PSTN Calling

Applications:

The SIP-to-PSTN calling works when both calling and answering parties are registered to SIP server with given registered phone numbers. The ADSL can be as in both Diagrams A and B.   Both parties are registered to SIP server with either fixed real IP or private IP under NAT router.

Configurations:

1. Same as in Example 2.
2. Select "ON" in the "SIP settings / STUN setting" page, if Outbound Proxy is NOT available.
3. Select "ON" for the "Auto Answer" and "PIN Code" in Call settings. Set the Auto Answer Ring Counter, e.g. 3, and the PIN code, e.g. 1234
4. Upon successful SIP registration, the **REG** LED indicator will be **ON**.

Callings:

5. Pick up the phone for VoIP mode, and press 1688# or 1688 to call another IP Phone with registered SIP phone number 1688.
6. After 3 rings for Auto Answer, the FXO port will auto answer with a "dodo" tone (not dial tone). Press 1234# for PIN code and then you will hear a PSTN dial tone. Note you must add the postfix "#".   PIN Code is used to prevent from call piracy.   Incorrect PIN Code will result in call disconnect. If PIN code is OFF, the caller may press PSTN number directly.
7. Press 7654321 to call the PSTN party number of 7654321.

5.4. PSTN-to-SIP Calling

Applications:

The applications can be for ADSL connections as in both Diagrams A and B. Both parties are registered to SIP server with either fixed real IP or private IP under NAT router.

Configurations:

1. Same as in Example 2.
2. Select "ON" in the "SIP settings / STUN setting" page, if Outbound Proxy is NOT available.
3. Select "ON" for the "Auto Answer" and "PIN Code" in Call settings. Set the Auto Answer Ring Counter, e.g. 3, and the PIN code, e.g. 1234.
4. Make sure the REG LED is ON for a successful SIP registration.

Callings:

5. Call from PSTN line to the IP PHONE FXO number, e.g. 7654321.   In a moment, you should hear a ring back tone, and wait for the IP PHONE to answer. After 3 rings, the VoIP mode will auto answer with a "dodo" tone (not dial tone). Press 1234# for PIN code and then you will hear a dial tone for VoIP mode. Incorrect PIN Code will result in call disconnect.   If PIN code is OFF, there will be not "dodo" tone and the caller may press SIP number directly.
6. Press 1688# or 1688 to call the party with the registered SIP phone number 1688. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

5.5. 3-Way Conference Calling

Applications:

The Call Transfer and 3-Way Conference Call applications are for calls among Parties A, B, and C.　Three parties are registered to SIP server with either fixed real IP or private IP.　There are two kinds of call transfer; Blind Transfer and Attendant Transfer.

Blind Transfer:

1.　Party A calls Party B.
2.　While in conversation, Party B may press Transfer key, and should hear a dial tone.
3.　Party B press [Party C number] # and hang up to transfer to Party C.

Attendant Transfer:

1.　Party A calls Party B.
2.　While in conversation, Party B may press Transfer key, and should hear a dial tone.
3.　Party B press [Party C number] # and talk to Party C.
4.　Hang up from Party B, and then Party A will transfer and connect to Party C.

3-Way Conference Call:

1. Party A calls Party B.
2. While in conversation, Party B may press Hold key to hold the call, and should hear a dial tone.
3. Party B calls Party C.
4. While in conversation, Party may press Conf. key to join in Party A for three-way conference.

Call Waiting Application:

When a new call is coming while you are talking, you will hear an interrupt "dodo" tone and you can press Hold key to answer the new incoming call. You may press Hold key to switch back to the previous call.

Call Hold Application:

You may press Hold key to hold the current call for a while, then press Hold key again to resume conversations.

5.6. Direct IP to Direct IP Calling

Applications:

The applications are for ADSL connection without NAT router as in Diagram A. Both parties are with fixed real IP.   The Direct IP calling works when both calling and answering parties are with known fixed IP.   SIP server registrations are not required in this application.

Configurations:

1.   Select "Fixed IP" in the "Network / WAN settings" page,
2.   Enter the items of IP, Subnet Mask, Gateway IP,
3.   Click the "Submit" button.

Callings:

4.   Pick up the phone, and you should hear a dial tone.
5.   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. Dialling without # will not dial out until the auto dial timer (default=5 seconds) elapsed. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

5.7. FreeWorld Dialup (FWD)

Applications:

This shows how to use FWD as an example for free ITSP provider.   The applications are for both parties registered to FWD SIP server. Visit FWD web site and sign up for a new registered account number.   Follow the instructions for registration. After finished, you will receive a mail sent by the FWD mail system, and you will get one FWD phone number and password in the mail. For example, the register name/phone number is **636346** with password **xxxx**. Login to the Web configuration page.

FWD Web Site:

<http://www.freeworlddialup.com/>

Username: **FWD Number**

Password: **FWD Password**

Domain: **fwd.pulver.com**

SIP Proxy: **fwd.pulver.com:5060**

Outbound Proxy: **fwdnat.pulver.com:5082**

OR

STUN server: **stun.fwdnet.net:3478** (Phone must be STUN enabled)

Listen RTP Port: **8000**

Listen SIP Port: **5060**

SIP Settings

Realm 1	
Active:	⦿ On ○ Off
Display Name:	636346
User Name:	636346
Register Name:	636346
Register Password:	●●●●●●
Domain Server:	fwd.pulver.com
Proxy Server:	fwd.pulver.com:5060
Outbound Proxy:	fwdnat.pulver.com:5082
Status:	Not Registered

You have to enter the Display Name, User Name, Registered Name, Registered Password, Domain Server, Proxy Server, Outbound Proxy. After finished the setting, click the Submit button and the Save Change button.   The IP Phone will reboot automatically.   After boot up, the SIP setting page will show "**Registered**", and the LCD will show registered <phone number>; it will shows "**No service**" otherwise.

Codec Setting

Codec Settings

You could set the codec settings in this page.

Codec Priority

Codec Priority 1:	G.729
Codec Priority 2:	G.723
Codec Priority 3:	G.711 u-law
Codec Priority 4:	G.711 a-law
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40

RTP Packet Length

G.711 & G.729:	20 ms
G.723:	30 ms

G.723 5.3K

G.723 5.3K:	○ On ⊙ Off

Voice VAD

Voice VAD:	○ On ⊙ Off

Callings:

1. Pick up the phone, and the LCD will show FWD phone number <636346>.
2. Press 12345 to call the party with registered FWD phone number 12345. In a moment, you should hear the ring back tone, and wait for the called party to answer.

6. Specification

Model No.	VOI-7000	VOI-7100
Connector	1 x WAN 1 x LAN 1 x Headset Plug	1 x WAN 1 x LAN 1 x Headset Plug 1 x RJ11 FXO
LCD Size	16 x 2	
Network Protocol	SIP v1 (RFC2543), v2(RFC3261) IP/TCP/UDP/RTP/RTCP IP/ICMP/ARP/RARP/SNTP TFTP Client/DHCP Client/ PPPoE Client Telnet/HTTP Server DNS Client VLAN Setting DMZ Setting Virtual Server MAC Clone Setting	
Call Function	Call Hold Call Waiting Call Forward Caller ID 3-way conference	

|	
Codec	G.711: 64k bit/s (PCM) G.723.1: 6.3k / 5.3k bit/s G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) G.729A: 8k bit/s (CS-ACELP) G.729B: adds VAD & CNG to G.729
Voice	VAD: Voice activity detection CNG: Comfortable noise generator LEC: Line echo canceller Packet Loss Compensation Adaptive Jitter Buffer
DTMF Function	In-Band DTMF Out-of Band DTMF SIP Info
QoS	ToS field
NAT Traversal	STUN
Configuration	Web Browser Keypad
Power	DC12V, 1.0A
Operational Temperature	0 to 40°C

7. Trouble Shooting

7.1. Do not hear dial tone?

When you pick up the phone and hear a busy tone, it indicates the WAN port is NOT connected. The LCD will show Ethernet Error! Make sure the ADSL Ethernet cable is connected to the WAN port of IP Phone and Power Reset again.

7.2. Can not access web page?

IE Web Browser is a useful tool to configure IP PHONE.   When you have difficulties in accessing the default IP address http://192.168.1.100 of IP PHONE as in the following figure, the most possibility is that the PC might have different subnet IP settings from 192.168.1.xxx. In this case, you must change IP PHONE IP address to the same subnet as PC and NAT router.
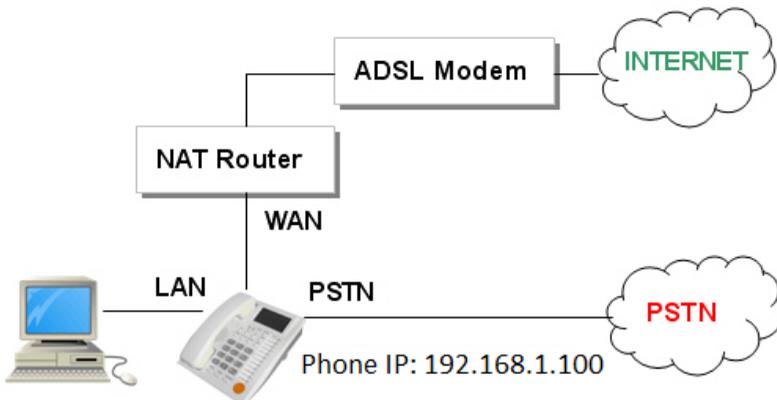
Example: To change IP PHONE IP address to the same subnet as PC and NAT router

1. Press the menu to enable DHCP Client mode.   IP PHONE will reboot, and LED will start flashing to get an IP address from NAT DHCP server.
2. Press Menu_4_5 to read IP Addresses for WAN and LAN Ports, for example, 192.168.62.51.
3. Enter from IE web browser http://192.168.62.51 to login IP PHONE web page for configurations.