VoIP/PSTN
IP-Phone

Quick Install Guide

Version: 1.3*(draft)*

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Chapter 1 Introducing IP-Phone

1.1 Introduction

This IP Phone (Internet Protocol Telephone) is a desktop network telephone that can act like a traditional telephone and deliver good quality voice. Basically, it will be connected to the LAN and PSTN network or enterprise as an office Dual VoIP (Voice over Internet Protocol) / PSTN phone; it also can be connected to broadband access devices (like XDSL/Cable Modem) for SOHO and residential users.

H.323 and SIP are the major type of VoIP protocol in the world. The IP Phone support both of H.323 and SIP communications simultaneously.

1.2 Package Contests

At first, we are thankful very much to you for using our IP Phone product. Upon opening the package, you may find the following items. Be sure that no damage is found on these items and the plug specification of Power adaptor meets the standard of your country. If you found any problem with them, please contact the supplier or distributor for assistance.

1. The main unit
2. Handset
3. Power adapter
4. Quick guide
5. CD
6. RJ-45 Ethernet cable
7. RJ-11 Telephone cable

1.3 Surface
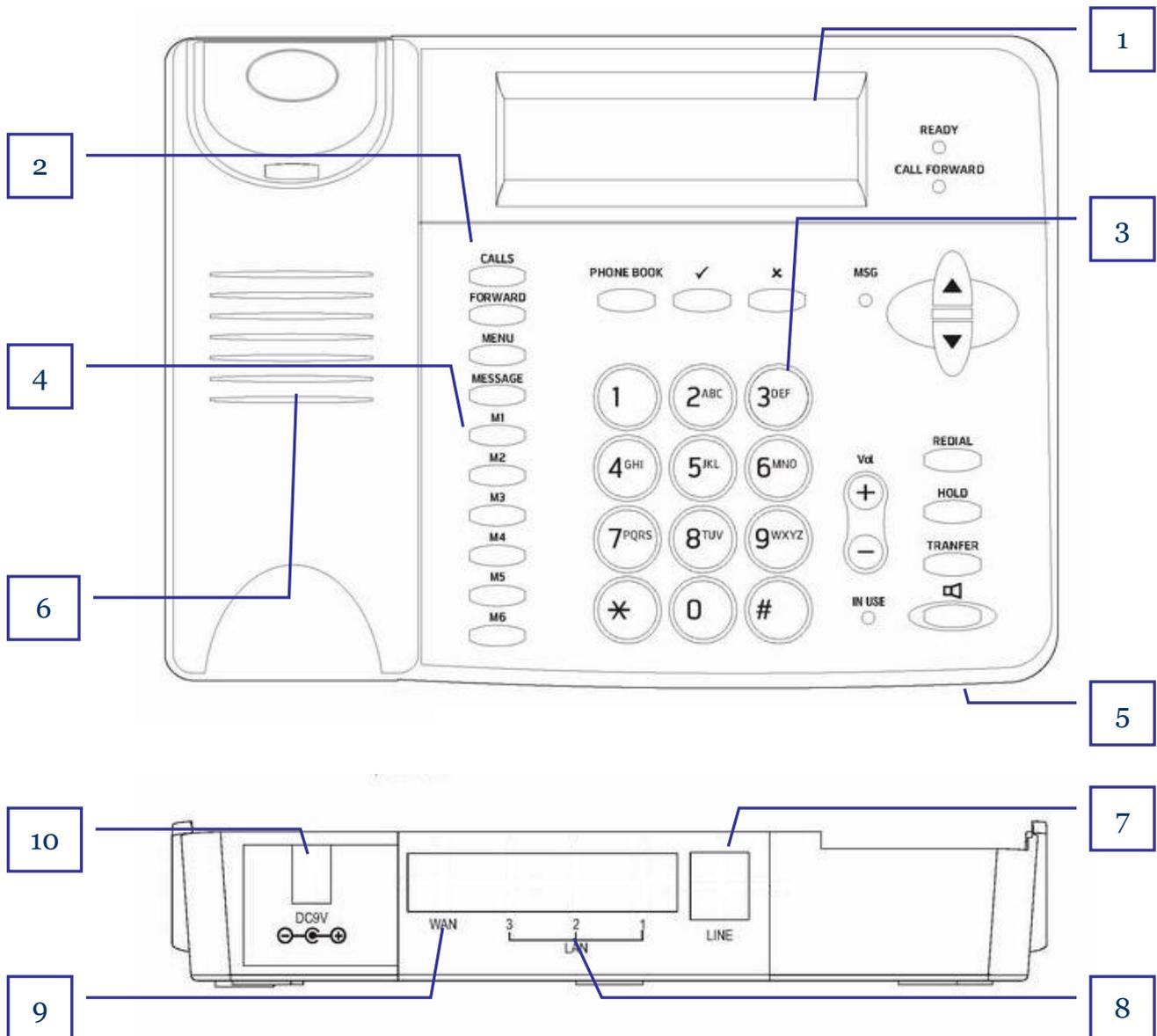
1. LCD Screen	6. Speaker
2. Function keys	7. PSTN Jack
3. Dialing Pad	8. LAN portx3
4. Speed Dial	9. WAN Port
5. MIC	10 Power Jack

READY LED:

>Flash: Failed on server registration

>Light: Successes on server registration

CALL FORWARD: when enable forward setting

MSG: when the new message or miss call in real time

IN USE: when use the hand-free speaker

1.4 Key Features

IP-Phone

1. VoIP/PSTN support
2. SIP/H.323 protocol simultaneously
3. Local 3-Way conference
4. PPTP and IPsec pass through
5. UPnP / Port Mapping support
6. SNTP Support
7. Static-IP/PPPoE/DHCP Support
8. Multiple Service Domain (up to 4)
9. STUN / Outbound Proxy Support
10. Customizable Ring melody for VoIP/PSTN/On-Hold
11. Message for Value-add
12. Forward (Always/Busy/No-Answer)
13. Call Rejection
14. Call Hold/Mute
15. Call timer display
16. IP Direct Call
17. Auto-Provision (up to 4)
18. Management by WEB/Keypad
19. Support IP TOS (Type of Service) for VoIP

1.5 Specifications

- 1x RJ-11 PSTN port
- 1 x RJ-45 10/100Mbps WAN Ethernet Port (Auto MDI-II/MDI-X)
- 3 x RJ-45 10/100Mbps LAN Ethernet Ports (Auto MDI-II/MDI-X)
- Display: 20(characters)*2(lines)
- Physical: 150 X 170 X 43 mm
- Power: AC100V-240V, DC 9V/1.3A (Max)
- Dimension: 215 x 160 x 55 mm (W/D/H)
- Weight: 615g
- LED Indicators : Ready, Call Forward, MSG, INUSE
- 33 feature keys
- Supported Protocol: UDP, TCP, Standard H.323 & SIP v2.0
- Audio CODEC:
  -G.711 A-law / u-law, (64kbps)
  -G.723.1, (5.3 or 6.3kbps)
  -G.729/G.729a, (8kbps)
- DTMF / Call progress tone detection and generation
- G.168 echo cancellation
- User friendly Web configure interface
- Configuration/Upgrade by Web or Provision Server
- Build-in watching dog for auto recovery

Chapter 2 The Basic Application Concept

The IP Phone supports 3 ways of calling method as described below
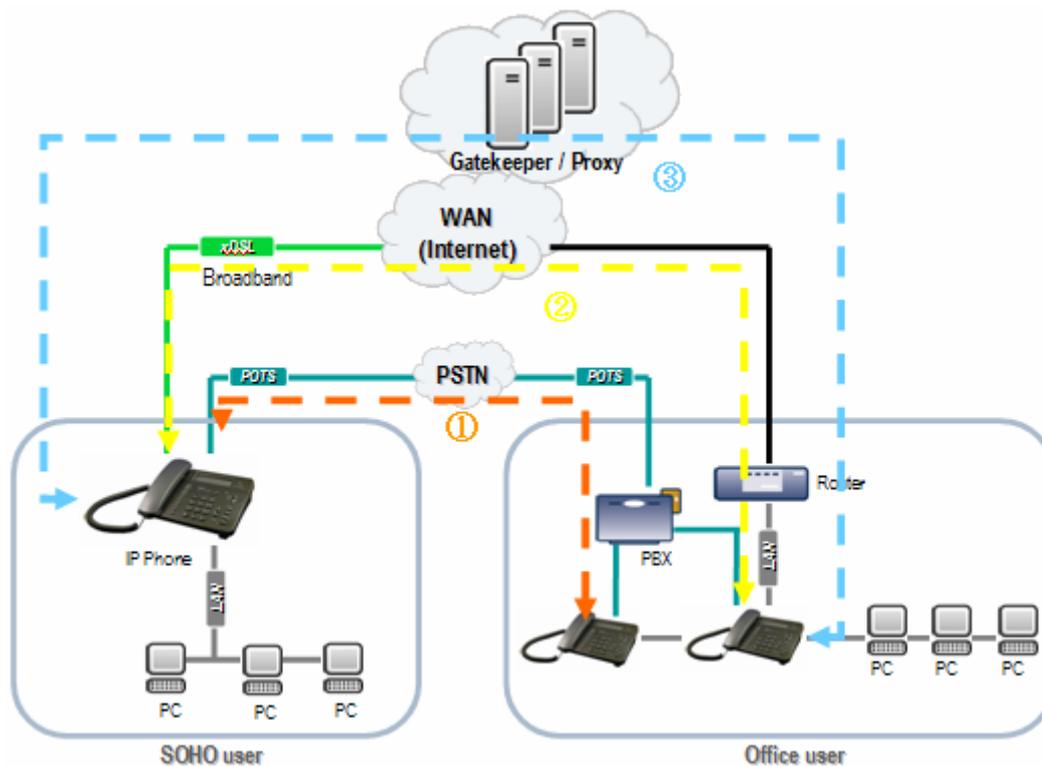
(Figure2-1)

2.1 Phone to Phone Direct Dial via PSTN network

The IP Phone support PSTN connectivity, you can easily to place a PSTN call.

Example: press "*" to switch PSTN mode, then start to make a PSTN call

(Please refer to Fig 2.1 for this call way ①)

2.1 Phone to Phone Direct Dial via Internet

This is peer-to-peer dialing method. You can please a VoIP call to a remote IP Phone with dialing its IP address directly via keypad. But, be sure that the calling and the called remote IP Phones must have fixed real IP address.

Example: press keypad to enter "Call by URL" mode, then enter the IP dial number step by step.

(Please refer to Fig 2.1 for this call way ②)

2.3 Phone through Gatekeeper/Proxy Dial

This is the most common dialing method. You can place a VoIP call to any IP Phone(s). In this case, the IP Phone doesn't need a fixed real IP address.

(Please refer to Fig 2.1 for this call way ③)

Chapter 3 Configuration

The default setting of DHCP Server inside IP-Phone is turn ON, So please set up your PC TCP/IP network as "Get IP Automatically" from DHCP to get internal IP from gateway. By default, the IP-Phone will become the network gateway and default IP is 192.168.22.1 and will assign your PC IP as 192.168.22.x.

Please go to "Control Panel"→"Network". In the "Configure" page, choose the TCP/IP of LAN card, and press "Properties" please choose "Obtain IP Address Automatically"

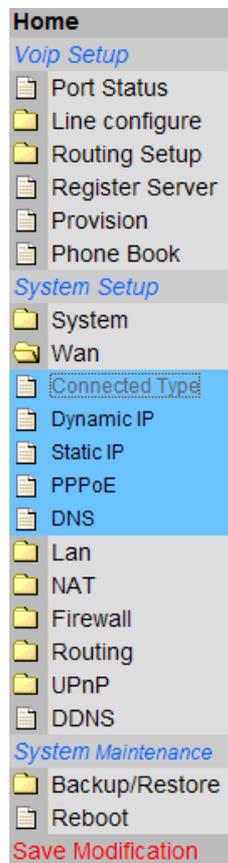Launch your browser and open the IP-Phone WEB-UI page as http://192.168.22.1
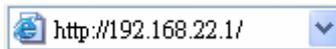
(Figure3-1)

  The default <u>User name</u> is **voip**
  The default <u>Password</u> is **1234**

3.1 Connection

Click "WAN" option below "System Setup" item:

(Figure3-2)

Please select the type of Internet connection you have and set up the router to use the IP Address Dynamic IP Address, Static IP Address or PPPoE connection.

If your ISP has not given you an IP address, please select Dynamic IP Address (default). If you have been given a specific IP address, select Specify an IP Address.
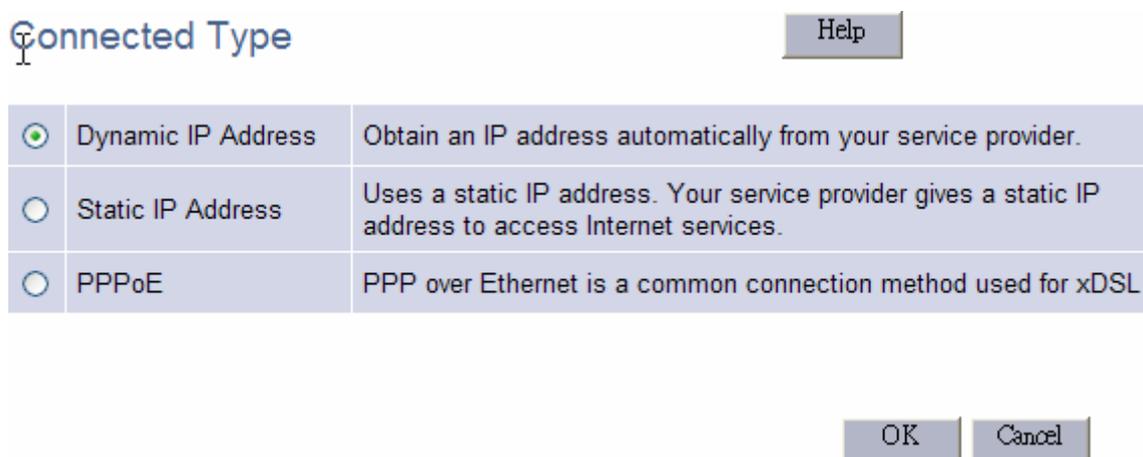
(Figure3-3)

Dynamic IP

Obtain an IP address automatically from your service provider.

MAC Address Clone

MAC address of WAN. If you use Cable Modem you must input it.

You can use the "Clone MAC Address" button to copy the MAC address of the Ethernet Card installed by your ISP and replace the WAN MAC address with this MAC address.

(Figure3-4)

Static IP

To use Static-IP connection, please select "Static IP Address" OK and enter WAN IP settings, supplied by your ISP.

(Figure3-5)

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PPPoE

 To use PPPoE ADSL connection, please select Yes in use **"PPPoE"** ADSL service and enter the "**username**" and "**password**" in the PPPoE setup section. Most of ADSL ISPs assign dynamic-IP settings to the router when using PPPoE. The IP-Phone automatically obtains these settings from your ISP when the PPPoE connection is successfully established.

(Figure3-6)

DNS

Domain Name Servers are used to map an IP address to the equivalent domain name (e.g.www.china.com). Your ISP should provide the IP address for one or more domain name servers.

(Figure3-7)

3.2 System status

System Status

You can use the Status screen to see the connection status of WAN/LAN interfaces, firmware and hardware version numbers, and the number of connected clients to your network.

The following items are included in this screen:

INTERNET - Displays WAN connection type and status.

GATEWAY - Displays system IP settings, as well as DHCP, NAT and Firewall status.

INFORMATION - Displays the number of connected clients, as well as the Router's hardware and firmware version numbers.

Please click to System Setup→System→System Status to see the follow information

System Status   Help

INTERNET   Refresh

Cable/DSL	Connected
WAN IP	172.16.7.136
Subnet Mask	255.255.0.0
Gateway	172.16.7.1
DNS	168.95.192.1
Secondary DNS	168.95.1.1
Domain Name	
Connection Type	Dynamic IP
Connection Time	00:57:00

Release   Renew

GATEWAY

IP Address	192.168.22.1
Subnet Mask	255.255.255.0
DHCP Server	Enabled
NAT	Enabled
Firewall	Enabled

INFORMATION

System Up Time	02:39:59
System Date	5/29/2006 11:46:45
Connected Clients	0
Runtime Code Version	V5.1.2.60
Boot Code Version	V0.1.5.9
LAN MAC Address	00:02:46:59:03:6D
WAN MAC Address	00:02:46:59:03:6C
Hardware Version	V0.1.2.3
Serial Number	12345678

(Figure3-8)

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3.3 VoIP Call Out

User key in the phone number through phone set dial pad, then IP-Phone will translate the phone number by the routing table setting here to destination IP & dial out number then Call out via network protocol

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
	MaxDigits: 20	FirstDigitTime(Sec): 30	OtherDigitTime(Sec): 5						
1	international	00	3	20	gk	3	00	A	Delete
2	taiwan	0	3	10	gk	1	00886	A	Delete
3	tw_mobile	09	3	10	gk	1	00886	A	Delete
4	OnNet	8	5	6	gk	2	63998512	A	Delete
5	taipei	2	3	10	gk	0	00886	A	Delete

Modify   Reset

Insert to: 6   Area Code: ___   IPAddress: ___
Add   Reset

(Figure3-9)

Remark: For remark this rule. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.

Area Code: Define the Prefix number fit this rule, any phone number prefix digits matched with the rule will call out by this rule define. Please Notify there is a compare order rule on this routing table. That mean the Gateway will check the rule list from top to bottom one by one, any rule item matched with the prefix digits that user key in will go to call out directly no regard to the rest rules below.   For Example, if a rule item for area code 8862 is on Index 5, another rule item for area code 886 on Index 6 below that will be ignored.

Min Digits: The length of the dialed number should not less than this digits. For example, if the field is entered into '3', the length of the dial number should be 3 digits at least.

Max Digits: The length of the dialed number should not more than this digits. For example, if the field is entered into '10', the length of the dial number should be 3 digits at most.

IP Address: Define the destination IP for call out number fit this rule, user can input below format:

- ■ IP address, for example: 168.56.9.22
- ■ URL, route via URL. For example: oneingate.allwin.com .This gateway can setup to register to DDNS service to let user call out to another gateway with dynamic IP by URL.
- ■ gkn , route via gatekeeper, it will get the destination IP by gatekeeper setting in advance. For example: gk1 for gatekeeper 1. gk2 for gatekeeper 2. gk for all the

gatekeepers available (search sequence: gk1 > gk2 > gk3 > gk4). Gk3_2_1 will try gk3 first, then gk2, then gk1.

- **rsn**, route via server mode, it will get the destination IP by register server setting in advance. For example: rs1 for register server 1. rs2 for register server 2. rs for all the register server available (search sequence: rs1 > rs2 > rs3 > rs4). rs3_2_1 will try rs3 first, then rs2, then rs1.

   All the setting above can be added by port number, for examples: 168.56.9.22:8495 will call to 8495 port.

Strip: the number of digits will be ignored by user input. For example, if user key in the number is 886285123390 and the STRIPE field is setting to 4, the first 4 digits 8862 will be truncated and actually call out number will be 85123390。

Prefix: The numbers will be added on the prefix of user key in number. For examples, if user key in the number is 85123390 and the PREFIX field is setting to 0028862, the actually call out number will be 002886285123390. Another example, if user key in the number is 90, STRIP field is setting to 2, and the PREFIX field is setting to 0,85123390, the actually call out number will be 0,85123390 (, mean wait 1 second). This example is especially for speed dial function.

Profile: Define the optional special call out parameters on this destination. Please input the name you defined on the profile list.   。

☞Note1:

VoIP Call rule of the IP-Phone, the delegation that preserve used for pstn call out, this line please keep to establish, don't do any change or delete either!

Index	Remark	Area Code	Min Digits	Max Digits	IP Address	Strip	Prefix	Profile	Delete
1		*	1	1	pstn				Delete

(Figure3-10)

IP address, for example: 168.56.9.22

All the setting above can be added by port number, for examples:

168.56.9.22:8495 will call to 8495 port.

Strip: the number of digits will be ignored by user input. For example, if user key in the number is 886212345678 and the STRIPE field is setting to 4, the first 4 digits 8862 will be truncated and actually call out number will be 12345678.

Prefix: The numbers will be added on the prefix of user key in number. For examples, if user key in the number is 12345678 and the PREFIX field is setting to 0028862, the actually call out number will be 002886212345678. Another example, if user key in the number is 90, STRIP field is setting to 2, and the PREFIX field is setting to 0,12345678, the actually call out number will be 0,12345678 (, mean wait 1 second). This example is especially for speed dial function.

To add new rule item on routing table, please assign the item number you want to insert before, input AREA CODE and IP address then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the modify button to take it effect. For store back to flash memory, please press "***Save Modification***" (***/System Maintenance/Save Modification/***).

3.4 Routing Profile

(Figure3-11)

This page defines the optional special VoIP parameters when making/received a VoIP call. For define some special parameters for different VoIP equipment or authorize purpose, please add a profile here and put the name of profile on the VoIP Call Out table.

Name: Specify a profile name. Please use UNDERLINE to replace the SPACE due to HTTP protocol limitation.

VAD:

    ON: turn on the VAD(Voice Active Detection) function.

    OFF: turn off the VAD function, please select ON for save the bandwidth.

CODEC: Select different voice CODEC for VoIP communication. The bit rate of G.723.1 is 5.3k/6.3k, G.729 is 8k, uLaw and aLaw is 64k per second. The G.723.1 is default CODEC.

H.245 tunneling:

    ON for enable.

    OFF for disable.

DTMF Relay:

  When select **Inband** to transfer the DTMF during VoIP, the user pressed DTMF tone will be treat as general voice and been compressed then transmit to remote side to decompress play back, it maybe cause some problem on duplicate or missing DTMF receive.

    When select **Outband** to transfer the DTMF during VoIP, the user pressed DTMF tone will be decode by local gateway then transmit as signal, after received on received remote gateway, it will be regenerate by remote Gateway. The default value is Outband.

T.38 FAX Relay: (This function is inactive on the IP Phone product)

Package Frame: Select the voice payload frame on each UDP package VoIP transmit. More frames into one package is save more bandwidth. The default frames on each package is 3.

Q.931 Fast Start:

    ON: Enable Fast Start capability during Q.931 handshaking.

    OFF: Disable Fast Start capability during Q.931 handshaking.

ID1: User defines ID #1 during this VoIP call.

As:

H.323 ID: Parameters on ID1 field is the H.323 ID during this VoIP call

E.164: Parameters on ID1 field is the E.164 during this VoIP call

Calling: Parameters on ID1 field is DID number during this VoIP call. If this optional is setting, it will override the LINE NUMBER on line Setting menu.

ID2,ID3,ID4: there are 4 fields for user define the ID parameters, please reference the ID1 setting above.

Delete: Delete this rule item on routing table.

To add new profile item on routing table, please assign the number you want to insert before, input profile NAME then press ADD button to add it on the list. Then modify the necessary information on the routing table list.

Please remember to press the modify button to take it effect. For store back to flash memory, please press Save Modification

3.5 Forwarding

Setup Help VoIP Call Out Routing Profile Forwarding

Forward

Always ☐   6582

OnBusy ☐   637602

No Answer ☐   59888   No Answer Sec 3000 s

Modify   Reset

Remark: The forward setting is only for on-net call , that means all the numbers are register on the same server !

Always: click and enter the number then that will enable always forward function. When the voip call has incoming, the IP-Phone will auto forward to the assign number.

OnBusy: click and enter the number then that will enable onbusy forward function. When the voip call incoming and the IP-Phone is on busy, it will auto forward to the assign number.

No Answer: click and enter the number, assign the X timer to enable no answer during the call, after that will enable no answer forward function. When the voip call incoming and the IP-Phone is no answer for X seconds, it will auto forward to the assign number.

Please remember to press the **modify** button to take it effect. For store back to flash memory, please press Save Modification

3.5 Register Server setting

Register Server Status

RS1 : ○ Disable   RS2 : ○ Disable   RS3 : ○ Disable   RS4 : ○ Disable   [Reload]

Register Server Config
Mac: 00024659036c

RS1 [Disable ▾]   **Remark:** [_____]   **Proxy:** ☐
IP Address: [sip:172.16.52.120]   Prefix: [_____]   *1: [_____]
ID1: [sip_test]   [UserName ▾] ID2: [●●●●]   [Password ▾]
ID3: [_____]   [H323ID ▾] ID4: [_____]   [H323ID ▾]

RS2 [Disable ▾]   **Remark:** [_____]   **Proxy:** ☐
IP Address: [h323:172.16.52.120]   Prefix: [_____]   *1: [_____]
ID1: [h323_test]   [H323ID ▾] ID2: [_____]   [E164 ▾]
ID3: [_____]   [H323ID ▾] ID4: [_____]   [H323ID ▾]

(Figure3-13)

Enable/Disable: Enable or disable this setting.

IP Address: Define the destination IP for call out number fit this rule, user can input below format:

■　URL, route via URL. For example: oneingate.allwin.com .This gateway can setup to register to DDNS service to let user call out to another gateway with dynamic IP by URL.

■　IP address, for example: 168.56.9.22

◆　For sip, please add "sip:" before the ip address, example: sip: 172.16.52.120.

◆　For h.323, please add "h323:" before the ip address, example: h323:172.16.52.120.

*1: SIP Outbound Proxy (IP address or Domain)

UserName: this SIP field is request form your ITSP

Password: this SIP field is request form your ITSP

H323ID: this H323 field is request form your ITSP

E164: this H323 field is request form your ITSP

Please remember to press the modify button to take it effect. For store back to flash memory, please press Save Modification

3.6 Line Configure

This page will setup the phone line and midi information

3.61 Line Configure

(Figure3-14)

Port: display the port number, e.g. 1

Interface: display the Phone interface type.

Name: Line name for this port. This will send and display on the remote side due VoIP call

Line number: Telephone number assigned to this line.

HandSet Tx: This will adjust the handset volume of local phone set. The adjust range is from 1 to 5. Higher value will cause louder sound come from local phone set.

HandSet Rx: This will adjust the handset volume of local phone set. The adjust range is from 1 to 10. Higher value will cause amplifier the sound get from local phone set.

SpeakPhone: This will adjust the speaker volume of local phone set. The adjust range is from 1 to 10. Higher value will cause amplifier the sound get from local phone set.

Mic: This will adjust the mic volume of local phone set. The adjust range is from 1 to 10. Higher value will cause amplifier the sound get from local phone set.

VoIP Ring Melody: This VoIP Ring Melody can be adjustable.

Line Ring Melody: This Line Ring Melody can be adjustable.

Hold Melody: This Hold Melody can be adjustable.

   Please remember to press the modify button to take it effect. For store back to flash memory, please press Save Modification

3.62 MIDI Setting

(Figure3-14)

This page is show how to setup and restore MIDI file.

Index: display the number

Remark: For remark this midi files.

File Name: Display the MIDI file name.

Listen: Click the Listen button, this midi file can be saved or opened on the computer.

Size: display the file size.

Delete: Delete this midi file.

Used: This value will display how many sizes are used.

Available: This value will display how many sizes are Available.

How to add midi file: Select the Index number, and browse the midi file, then restore it!

   Please remember to press the modify button to take it effect. For store back to flash memory, please press Save Modification

3.7 Phone Book

(Figure3-15)

This Phone book can be added on this page or form the keypad.
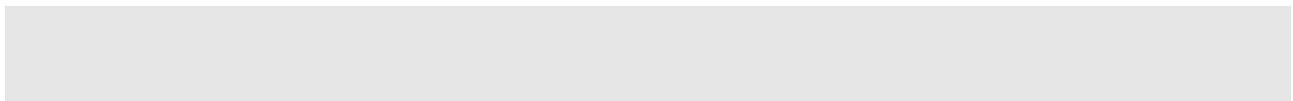
Index: Display the number

Name: Display the name.

Number: This number can be defined voip or pstn number.

Speed Dial: It can be defined up to 6 speed dial.

MIDI: Display the name.

Delete: Delete this item.

Chapter 4 The Setting with LCD Menus

4.1 Front Panel configure with LCD

a. **Calls**

- ➢ **Missed Calls** : display the missed calls list
- ➢ Dialed Calls : display the Dialed calls list
- ➢ Received Calls : display the Received calls list
- ➢ Forwarded Calls : display the forwarded calls list
- ➢ Delete Calls
  - ν Delete Missed : Delete the missed call records
  - ν Delete Dialed : Delete the dialed call records
  - ν Delete Received : Delete the received call records
  - ν Delete Forwarded : Delete the forwarded call records
  - ν Delete All : Delete all records

b. Forward

- ➢ Always Forward
- ➢ Forward On Busy
- ➢ No Answer

c. Menu

- ➢ View
  - ν Network Link
    1. Network Mode
    2. IP Address
    3. SubNet Mask
    4. Default Router
    5. Primary DNS
    6. Secondary DNS
  - ν Ping Test : testing the internet connection status.

  Example enter 168*95*192*1 , and press ok , then the LCD will display response time
  - ν Device Status
    1. Model
    2. Service Provider
    3. Firmware
    4. Provision Link
    5. SIP Server Link
    6. Last Provision
  - ν Call By URL : make a IP direct call
    1. Called IP: enter the IP address that you want to call
    2. Called IP Port: enter the port number (default is 5060 for SIP)

3. Telephone Number: optional
  ➢ Configure: administrator setting mode (default password is 1234)
    ν Network : select which internet mode that you used
      1. DHCP
      2. Fixed IP
      3. PPPoE
    ν Time Zone
    ν Provision: this is setting for Provision Server , please contact your ITSP!
      1. Provision Set
      2. Provision IP
      3. Provision Now
    ν SIP Set
      1. Server IP : setting the sip proxy address
      2. Server Port
      3. Outbound Proxy: please get this information form your ITSP
      4. Phone Number
      5. User Name
      6. Password
      7. Save Modify? : Please remember to save modify !
    ν Password Modify
  ➢ Setting
    ν VoIP Ring Melody (form 1 to 10)
    ν Line Ring Melody (form 1 to 10)
    ν Ring Volume (form 1 to 10)
d. Massage :real time message display
e. M1 ~M6 Speed Dial
f. PHONE BOOK
  ➢ List Phone Book
  ➢ Add Phone Book
  ➢ Speed Dial
g. ∨ for OK
h. ㄨ for Cannel
i. ▲ for Page up
j. ▼ for Page down
k. REDIAL
l. HOLD
m. TRANSFER
n. SPEAKER
o. Vol ＋
p. Vol －

4.2LED Display

a. READY LED: Power and register status
  - ➢ Light：Power on and register on server already
  - ➢ Flash：Power on but not register on the server or internet connection issue
  - ➢ Dark：Power off

b. CALL FORWARD
  - ➢ Light：Enable
  - ➢ Dark：Disable

c. MSG： Real time message
  - ➢ Flash：New message or miss call notify
  - ➢ Dark：No any message

d. INUSE：display the hand-free speaker
  - ➢ Light：It means the hand-free speaker is in use
  - ➢ Dark：It means the hand-free speaker is not use