IPF-2600　User's Manual/使用手冊

V1.2

ArtDio Company Inc.
www.artdio.net

Safety Instructions

- Do not attempt to service the product by yourself. Any servicing of this product should be referred to qualified service personnel.
- To avoid electrical shock, do not put your finger, any pin, wire, or other metal objects into the vents and gaps.
- To avoid accidental fire or electrical shock, do not twist power cord or place it under heavy objects.
- The product should be connected to a power supply of the type described in the operating instructions or as marked on the product.
- To avoid hazard to children, dispose of the product's plastic packaging carefully.
- Please read all the instructions before using this product.

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Ⅰ. Product Features

The ArtDio IPF-2600 is the latest generation of Enterprise IP Phone that supports SIP (Session Initiation Protocol）, the innovative technology and excellencies in engineering deliver the best voice quality and multiple control functionality.

1.1 Software Features

- Support SIP v1.0 (RFC 2543)，SIP v2.0(RFC 3261)，TCP/UDP/IP, RTP/RTCP, HTTP，ICMP，ARP/RARP，DNS，DHCP（include DHCP client and DHCP server），SNTP，PPPoE，STUN，TFTP protocols.
- Support multiple audio codecs, including 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 and G.729B .
- Support VAD (Voice Activity Detection)，CNG (Comfort Noise Generation)，LEC （Line Echo Cancellation)，Packet Loss Compensation and Adaptive Jitter Buffer control.
- Support standard calling functions, such as Caller ID Display and Block，Call Waiting，Call Hold，Call Transfer (attended /blind)，Call Forwarding and 3-way Conference.
- Support in-band and out-of band DTMF (RFC2833) and SIP INFO.
- Support for Layer 3 QoS（Tos Field）settings.
- Support HTTP1.1 website settings，MD5 For SIP certification（RFC2069/RFC2617）, to ensure security level.
- Support up to 3 SIP accounts，supports Outbound Proxy and up to 11 media channels concurrently.
- Support configuration settings using phone keypad, through the website, Console control and Telnet settings, or provide automated provisioning by downloading encrypted configuration file via HTTP/TFTP for enterprise users.
- Built-in different ring tones allow for user-customization on ring tone settings.
- Support firmware upgrade via HTTP or TFTP/FTP.

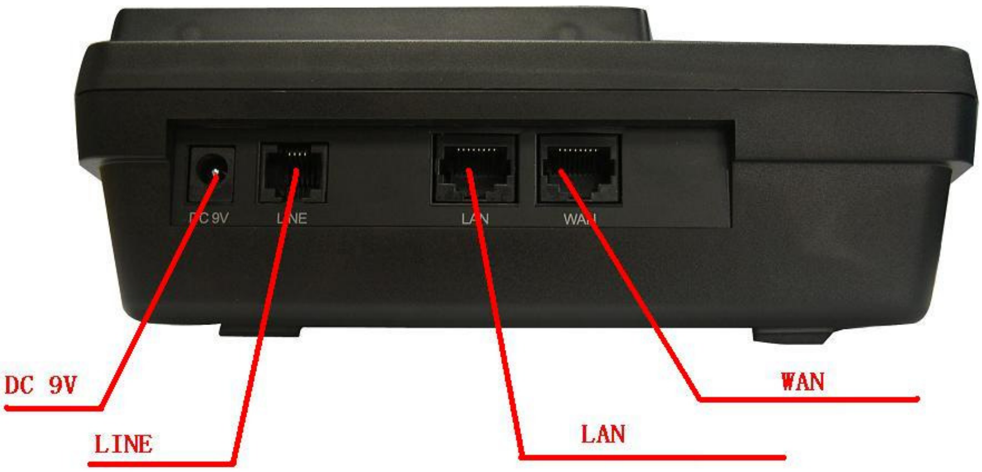
1.2 Hardware Features

- 2 x RJ-45 10/100M Base-T Ethernet ports, which enable to connect to WAN or LAN settings.
- 1 x RJ-11 port (FXO) connects the PSTN or extension line (FXO).
- MIC & Line-Out jack，support headset which will auto switch to headset when plugged in.
- Support functions of Speakerphone, Redial, Incoming/Dialed Call, Volume up and down, address book, Call Transfer, 3-way Conference, DND (Do-Not-Disturb), Microphone Mute, IP to PSTN exchange.(default IP)/ PSTN to IP exchange (default PSTN)
- With 8 LED phone status indicator.
- Phone keypad supports switching between 2 lines (L1－L2, Support 1 PSTN line and 1SIP line or 2 IP phone lines), and 4 speed dial keys (M1－M4）.

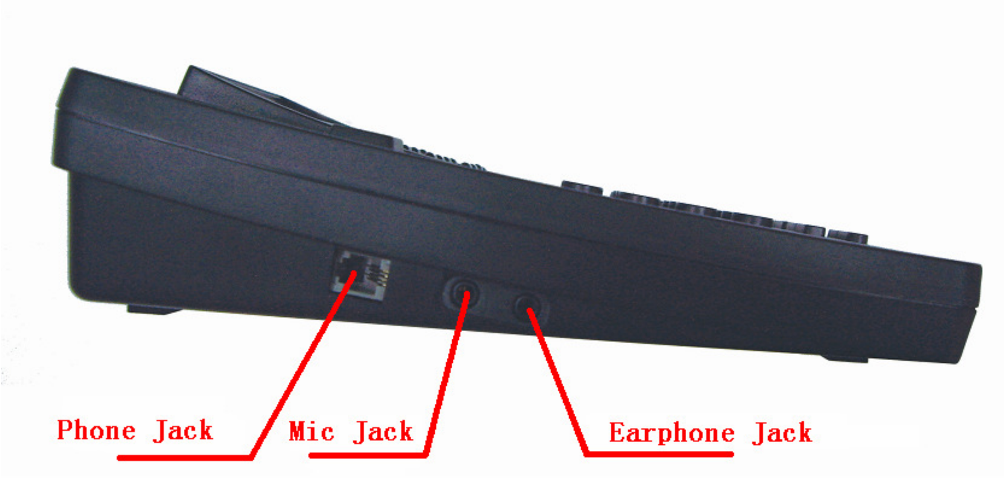
1.3 Product Overview

Front View

Back View



Side View



Ⅱ. INSTALLATION

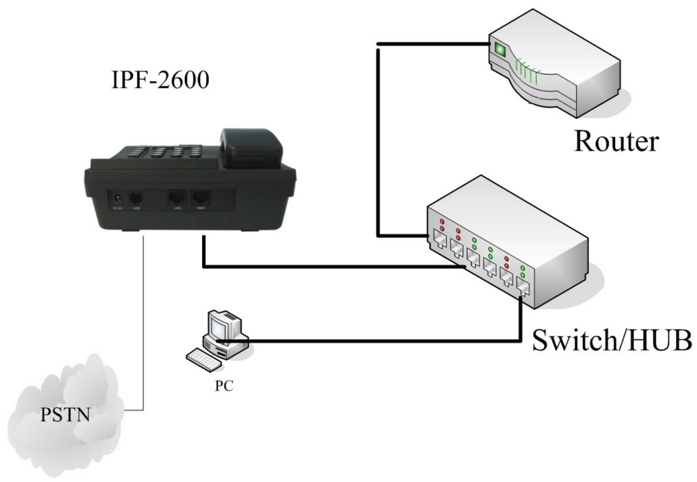
2.1 Package Content

- 1 x IPF-2600 IP Phone
- 1 x handset
- 1 x power adapter
- 1 x RJ-45 Cable
- User's Guide

2.2 Hardware Installation

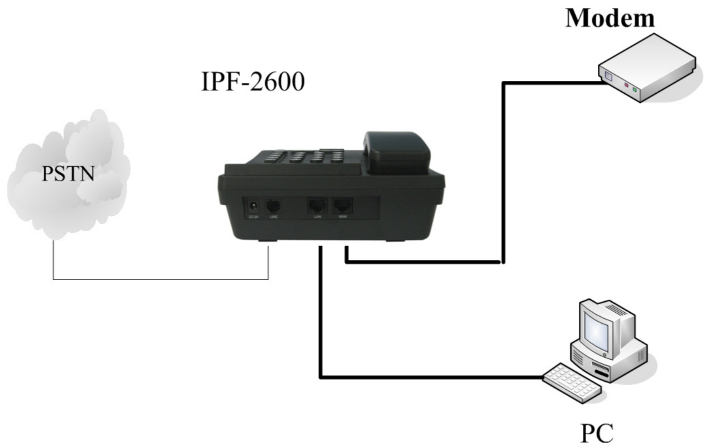
2.2.1 Connection with Router Existed in the Current Network

If the network environment has a Router existed, the IPF-2600 IP phone router function should be disabled. One end of the RJ-45 cable into the LAN port on the router (or the LAN port of the hub connected to the Router), then the other end connect to the WAN port on the IPF-2600 IP Phone. As the following image:



2.2.2 Connection without Router in the Current Network

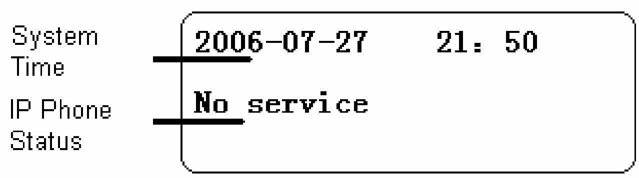
If the network environment does not have a router, the IPF-2600 router should been enabled. One end of the RJ-45 cable into the WAN port on IPF-2600, the other end into Internet port on the ADSL modem, and the LAN port of the IPF2600 could connect to a computer or a hub, as the following image:



2.3 LCD Status Display

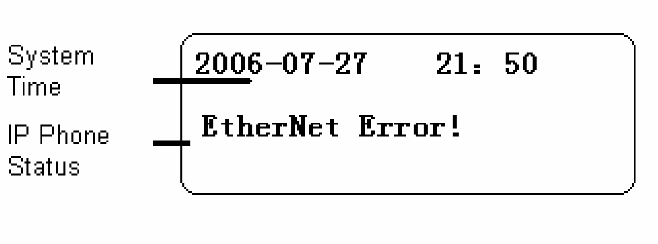
When the phone is installed, the LCD will display the connection status to let the users understand the connection status.

1.



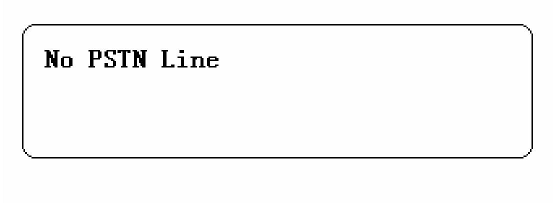
The status indicates the server is not found.

2.

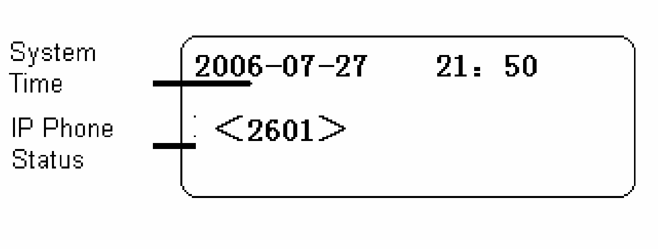


The status indicates the Ethernet link failure.

3． When the handset is picked up or when the speakerphone mode is turned on(default PSTN mode) , Default IP mode press" IPCALL" key exchange to PSTN mode ,if LCD display as follow, it indicates PSTN Link failure. Please check the PSTN line connection.

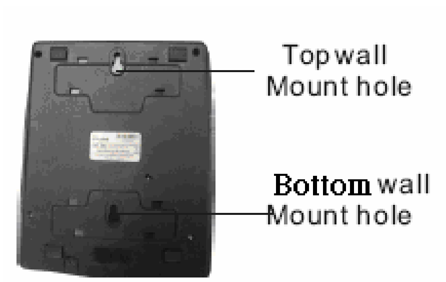


4． The status indicates normal connection state.

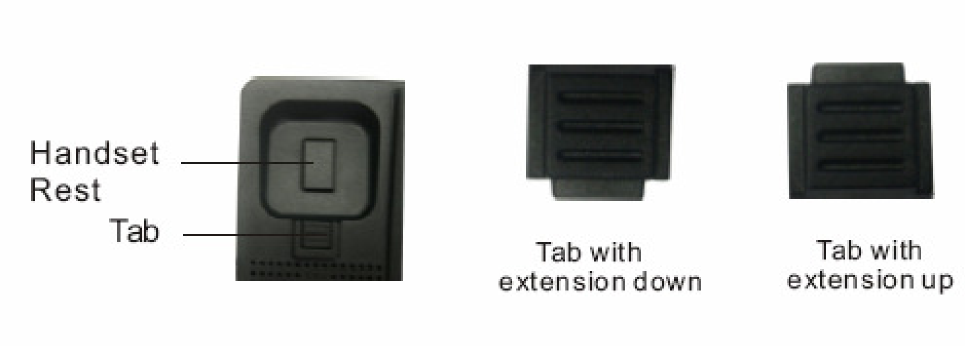


2.4 Wall mount

IPF2600 can be wall mounted. There are two wall mount holes on the bottom of the IPF2600 main body:



User can simply place the device against the wall with two holes placed to the fixed hanger position on the wall



After wall mounting the main body of IPF2600, user will need to pull out the tab (extension downward) from handset cradle on the top of the handset rest, and rotate the tab and plug it into the slot with the extension up for handset holding.

2.5 Safety Compliances

The IPF2600 phone is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. The phone should only be operated with the universal power adaptor provided with the package. Damages to the phone caused by using other unsupported power adaptors are not covered by the manufacturer's warranty.

2.6 Warranty

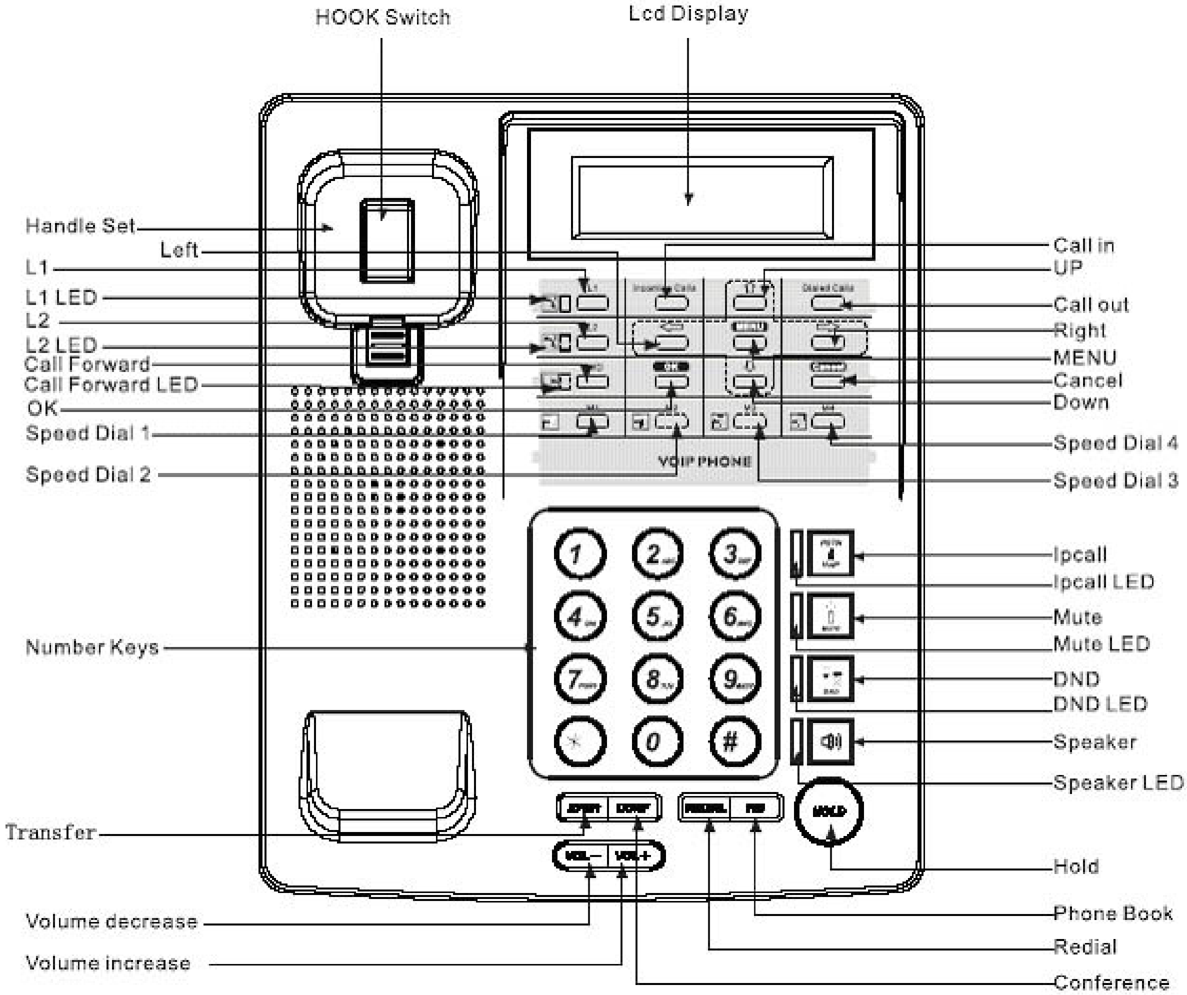
ArtDio has a reseller agreement with our reseller customer. End user should contact the company from whom you purchased the product for replacement, repair or refund. If you purchased the product directly from ArtDio, contact your ArtDio Sales and Service Representative for a RMA (Return Materials Authorization) number.

ArtDio reserves the right to remedy warranty policy without prior notification.

Ⅲ. Basic operation

3.1 Configuration using Phone Keypad

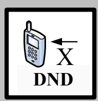
3.1.1 Keypad Display



3.1.2 Keypad Display Functions

Categories	Keypad	Description
MENU	MENU	Press to enter "Menu" mode to make changes on settings
	⇧ Up	Press to navigate up and scroll up, and to increase settings
	⇩ Down	Press to navigate down and scroll down, or decrease settings
	⇦ Left	Shift cursor to left, delete entered numbers or character

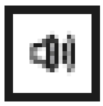
	Right	Shift cursor to right
	OK	To confirm the settings
	Cancel	Cancel settings, and exit from the settings.
Call History	Incoming Calls	Retrieve Incoming Calls history, press up/down key to navigate the selection. If the key pressed at idle Mode LCD will display 1:Received calls (Max 31 track record) 2:Missed Calls (Max 31track record) If the speakerphone is turned on or the handset is picked up, the list of incoming calls history will automatically display
	Dialed Calls	Retrieve Dialed Calls history, press up/down key to navigate the selection. If the speakerphone is turned on or the handset is picked up, the list of dialed calls history will automatically display
Call Forward	FWD	Turn on and off Call Forward function
Lines	L1－L2	For outgoing calls, press any of the L1- L2 Lines to use the Line to make a call. If the other Line is pressed, it will put the previous call on hold and dial the next call. For incoming calls, press any of these Lines to pick up the call. If there is a second incoming call, the LINE LED will flash, press on the Line which the LINE LED is flashing and pick up the second call and put the first call on hold. Note: The first incoming call must press “Speaker” key or pick up Handset.
	LINE LED (L1-L2)	Orange LED; will flash for incoming calls and lights up when the call is connected, it will light up when the call is on Hold.
Speed Dial	M1－M4	Press the Key to call a speed dial number. To enable the function, go to Phone book to enter a phone book entry, and add the number into the Speed dial settings. If a “-“ symbol is added before the numbers, the call will be dialed using the IP lines; if not, the call will be dialed using the PSTN line. After the call, it will reset to default mode. Note: Only at Default PSTN support M1-M4 PSTN Speed Dial.
Numbers （In the settings mode, continuously pressing the number key will display the following different characters in order. ）	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”
	*	“*”, “•”, “:”, “@”

	#	During IP line mode, press "#" key before dialing a call.
Function Keys	VoIP To PSTN 	If default "IP" mode, pressing the Key switch "IP" mode to "PSTN" mode. During "IP" mode, press to switch to "PSTN" mode. After entering the "PSTN" mode, the orange LED will light. The function is limited to switching from "IP" to "PSTN" mode only. If default "PSTN" mode, pressing the Key switch "PSTN" mode to "IP" mode only, During "PSTN" mode press to switch to "IP" mode, After entering the IP mode the orange LED will light .
	MUTE 	During the IP mode, press to mute an active call, the orange LED will light.
	DND 	Do-Not-Disturb, press to enter DND mode, all IP phone calls cannot call in, the orange LED will light up (IP mode only) PSTN calls could call in at DND mode
	SPEAKER 	Press to enter "Speaker" mode, the Orange LED will light, Pick up handset the phone will automatic switch to "handset" mode the Orange LED will turn off
	XFER	Press to transfer an active call to another number. When a call is connected, press the button and dial a number, the call will be transferred to the number.
	Conf	Press to bring calling/ called party into conference
	Redial	Press to redial the last dialed number
	P · B	Phonebook, press to retrieve phone book entries to select or edit contacts.
	Vol＋/-	VOL-: Reduce handset/speakerphone volume when phone is ACTIVE VOL+: Increase handset/speakerphone volume when phone is ACTIVE

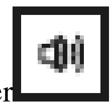
3.2 Making and Answering Phone Calls

3.2.1 Handset, Speakerphone and Headset Mode

Handset can be switched with either Speaker or Headset, however, whenever the Headset is plugged in, Speaker will be switched to Headset at Handset mode.

To Switch between Speaker and Handset /Headset, simply press the Speaker Key  or Hook.

Note:

During Handset mode press Speaker  switch to Speakerphone mode.
During Speakerphone mode press Hook or pick up Handset switch to Handset/Headset mode.

3.2.1 Multiple incoming calls

IPF2600 support 2 Line incoming calls , Orange LED; will flash for incoming calls and lights up when the call is connected, it will light up when the call is on Hold.

3.2.2 Making Calls

There are three ways to make phone calls:

1. Pick up Handset /press Speaker Key, or press the available LINE (L1/L2) key the corresponding LINE LED will light up in Orange. Enter the phone numbers and press the "#" key (default IP mode),if Default is PSTN mode could direct input target PSTN number.
2. Pick up Handset /press Speaker Key, or press the available LINE (L1/L2) key the corresponding LINE LED will light up in Orange Press the "Redial" button to redial the last Dialed number (default PSTN mode)
   Note:
   If the last number is PSTN number, could not call out at Default IP mode.
3. Pick up Handset /press Speaker Key, or press the available LINE (L1/L2) key the corresponding LINE LED will light up in Orange. Press the Speed Dial key to call the preset calling party number.

Note:

・ Once pressed, the dialed number is displayed on the LCD as the corresponding DTMF tone is played out.

・ If the "#" button is not pressed after the phone number, the phone will wait for "3-9" seconds before initiating the call.

3.2.3 Making Calls using IP Address

Direct IP calling allows two phones to talk to each other in an ad hoc fashion without a SIP proxy. VoIP calls can be made between two phones if

- Both phones have public IP addresses, or
- Both phones are on a same LAN using private or public IP addresses, or
- Both phones can be connected through a router using public or private IP addresses.

To make a direct IP calling, first pick up the phone or turn on the speakerphone, then input the 12-digit target IP address followed by "#"

Examples:

If the target IP address is 192.168.1.160, the dialing convention is **192*168*001*160#.**

3.2.4 Receiving Calls

There are two states when IPF2600 receives a call:

1. When receiving an initial call. Besides ringing with selected Ring Tone, the corresponding LINE will flash, taking Handset/SPEAKER/Headset off hook or press "Speaker" Key will enable user to hear the calling party in Handset/SPEAKER/Headset.
2. When receiving second incoming calls, besides playing stutter Call Waiting tone.
   The other Line LED will flash, pressing "Hold" Key or L1/L2 Key will enable user to hear the calling party in Handset/SPEAKER/Headset., the mean time the first incoming call will be hold.

3.2.5 Call Hold

While in conversation, pressing the "HOLD" button will put the other party on hold. User can resume the conversation by pressing the corresponding LINE. User will also automatically put the current line on "HOLD" by pressing another available LINE for making or receiving other phone calls.

3.2.6 Call Waiting and Switch between Calls

IPF2600 can support 2 Lines, user can switch to another line for making or answering calls and automatically put an ACTIVE call on Hold.

When receiving second incoming calls, besides playing a stutter Call Waiting tone,IPF2600 will pick up the next corresponding available LINE.

3.2.7 Call Transfer

IPF2600 supports both BLIND and ATTENDED Transfer:

1. Blind Transfer: When in conversation with an "ACTIVE" LINE If there is no LINE on HOLD, user press "Hold" button, hold current Line call, Then press "XFER" button Input target number, this will directly transfer the holden party in the corresponding LINE to the dialed number.
2. Attended Transfer: When a LINE is "ACTIVE", user will get a dial tone by pressing the "XFER" button, then dial the number and then press the "#" button, user could talk with the Line and then hang up, the Holden Line will connect with the current Line.

Note:

・ Transferring calls across SIP domains need to be supported by SIP services.

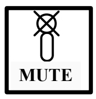
3.2.8 3-Way Conference

IPF2600 supports 3-way conference. With one LINE ACTIVE and another LINE on HOLD, press the CONF button then the LINE that is on HOLD will join the three parties together in a conference.

If after pressing the "CONF" button, a user (not Holder) decides not to conference anytime, user can cancel it and resume the conversation by pressing "Hold" or the original LINE button.

If the conference holder wishes to end a conference, simply Hook Off or press "Speaker" button, the conference Holder could not to hold /resume the conference, if the conference holder hang up, the other user could continuous talk with each other, which breaks the conference and places both parties, user can then talk to each individual party.

3.2.9 Mute

When in conversation with an ACTIVE LINE, pressing "MUTE" 

will mute the conversation, that is, you can hear the other party but the other party cannot hear you. Pressing the button again will resume the conversation.

3.2.10 Speed Dial

There are 10(M1-M4 and "0","5-9") speed dial buttons, each can be configured with a different account to dial.

Press M1-M4 will directly dial the preset calling party number at Speakerphone mode

Pick up handset or Press "Speaker" button, Press "0-9" followed by "#" to call the preset calling party number.

Note:

M1-M4 Support PSTN Speed Dial if No PSTN Active/Hold (Default PSTN).

"0-9" only Support IP/SIP Speed Dial.

Ⅳ. Configuration Settings

The IPF-2600 IP phone supports multiple configuration settings. In this section, it mainly explains configuring using the phone keypad, and through the website. When the phone is connected properly, press the "Menu" button to enter the Menu mode to make changes on settings. Press the "Menu" button again to exit from the Menu. If no action is made after entering the Menu mode for more than 10 seconds, it will exit automatically.

4.1 Phone Book

1．Search	Search the phone book entries, to make a call or edit the entries.
2．Add Entry	Add a new phone book entry. Note: If set IP code please add "-" Symbol before the numbers
3．Speed Dial	Set speed dial numbers Note: 1:M1-M4 cloud set PSTN speed dial number 2: the other only set IP speed dial number
4．Erase All	Delete all phone entries

4.2 Call History

1．Incoming calls	Enter to retrieve Incoming Calls history, press OK for options, select "Save" to save the changes or "Erase" to delete.
2．Dialed numbers	Enter to retrieve Dialed Calls history, press OK for options, select "Save" to save the changes or "Erase" to delete.
3．Erase record	Enter to erase record of call history. Press "OK" for options, select between "Incoming"、"Dialed"、"All" to erase records.

4.3 Phone Setting

4.3.1 Call Forward

1．All Forward	Forward all calls – go to Activation and select the options of "Enable" to turn on or "Disable" to turn off the function. Go to "Number" option to enter the phone number for call forwarding. Note: PSTN and IP (SIP) support but PSTN number must add "-" Symbol before the number.
2．Busy Forward	Busy calls forward - go to Activation and select the options of "Enabled" to turn on or "Disable" to turn off the function. Go to "Number" option to enter the phone number for busy forwarding. Note: Busy forward only support IP (SIP).
3．No Answer FWD	No answer call forward - go to Activation and select the options of "Enable" to turn on or "Disable" to turn off the function. Go to "Number" option to enter the phone number for no answer forwarding. Note: PSTN and IP (SIP) support but PSTN number must add "-" Symbol before the number.
4．Ring Time Out FWD	Ring time out call forwarding – enter the number of ring times before a call times out, then enter the phone number for call

	forwarding.

4.3.2 Do Not Disturb

1．Always	Block all calls
2．By Period	Black calls by certain time setting
3．Period Time	Block calls by certain time duration setting

4.3.3 Alarm Setting

1．Activation	On/Off clock Alarm
2．Alarm time	Set Alarm time, IPF2600 automatic Alarm when at the preset time.

4.3.4 Date/Time Setting

1．Date&Time	Date and time setting
2．SNTP Setting	SNTP：Turn on the SNTP to automatically update the time Primary SNTP：Primary SNTP address Secondary SNTP：secondary SNTP address Time Zone：Time Zone setting Adjustment time：Time adjustment setting

4.3.5 Volume & Gain

1．Handset Volume	Adjust handset volume, the maximum volume is 15.
2．Speaker Volume	Adjust speakerphone volume, the maximum volume is12.
3．Handset Gain	Adjust handset gain, the maximum gain is 15.
4．Speaker Gain	Adjust speakerphone gain, the maximum gain is 15.

4.3.6 Ringer Settings

1．Ringer Volume	Adjust ringer volume, the maximum is 10.
2．Ringer Type	Select between 4 different types of Ring tones

4.3.7 Auto Dial

1．Auto Dial	Auto dial setting – set the time interval in seconds for auto dial function. After the phone number is entered and without pressing the "#" sign, it will auto dial out based on the time set. Time interval setting is between 3 – 9 seconds.

4.4 Network Setting

4.4.1 WAN Setup

1、IP Type	IP mode type： Fixed IP client：set fixed IP address DHCP client：Obtain IP address automatically from DHCP server PPPoE client：PPPoE dial-up setting NAT setting: enable NAT function（setting is done through the website）
2、Fixed IP setting	Fixed IP address： Host IP：host IP address（"*" indicates the ".”in the IP address） Network mask: set submnet mask IP address Gateway IP：set gateway IP address MAC address：set MAC address
3、PPPoE Setting	PPPoE setting： User name：Enter PPPoE user ID Password：Enter PPPoE password

4.4.2 LAN Setup

Bridge	ON/OFF Bridge function
NAT	ON/OFF Router function

4.4.3 DNS

Primary DNS	Mostly DNS Setting
Secondary DNS	Secondary DNS Setting

4.4.4 VLAN

Activation	ON/OFF VLAN function
VID	Default is 136, Virtual LAN Server setting , VLAN Identifier (Virtual LAN ID. Name VLAN ID or VID).
Priority	Default is 0, user priority Setting , user priority(0-7)
CFI	Default is 1, Canonical Format Indicator (CFI). CFI=1: head of the packet contains spaces RIF , And the NCFI Flag which inside RIF decision information is carried by MAC address. "Standard format" (Canonical Format) or "non-standard format"

	(Non-Canonical Format). CFI=0: head of the packet Do Not contains spaces RIF, information is carried by MAC address is "Standard format" (Canonical Format).

4.4.5 Status

To check the IP phone for the IP address and MAC number.

4.5 SIP Setting

If the need for the Setting ,Please firstly Enter "System Authent" Sub of "Administrator", Input Password (pre-set value is the test) if the incorrect password, LCD display "incorrect", correct password LCD display "Correct." When the password was confirmed and then user could SIP set.

4.5.1 Service domain

First realm	Register the first SIP account: Activation: Enable Register the SIP account Disable Not Register the SIP account Display name: LCD Display Name User name: User name Register name: register the account name Register Password: register the account's Password Domain server: Net domain names Proxy Server: Voice Gateway for a landing on the heads of the IP address or Net domain names Outbound proxy: Speech Proxy
Second realm	Register Secondary SIP account, like as up.
Third realm	Register third SIP account, like as up.

4.5.2 Codec

Codec type	Set Speech Coding format
VAD	Enable :Voice Activity detection ON Disable: Voice Activity detection OFF

4.5.3 RTP Setting

Outband DTMF	ON/OFF Out band DTMF setting
Duplicate RTP	Retransmission number of voice packets

4.6 NAT Traversal

STUN setting	STUN：Confirm to use STUN setting
	STUN server：STUN server address

4.7 Administrator

1．Auto config	Auto configuration：Config mode：select configuration mode（Disable/TFTP/FTP） TFTP server：TFTP server address

	FTP server：FTP server address FTP login name：FTP server login name FTP password：FTP server password
2．Default Setting	Default setting：Load default：restore default setting Abort：abort the process
3．System authent	System authentication setting
4．Version	Display software version
5．Watch dog	IP phone main control
6．Restart	Restart the IP phone

Ⅴ. Configuration using Web Browser

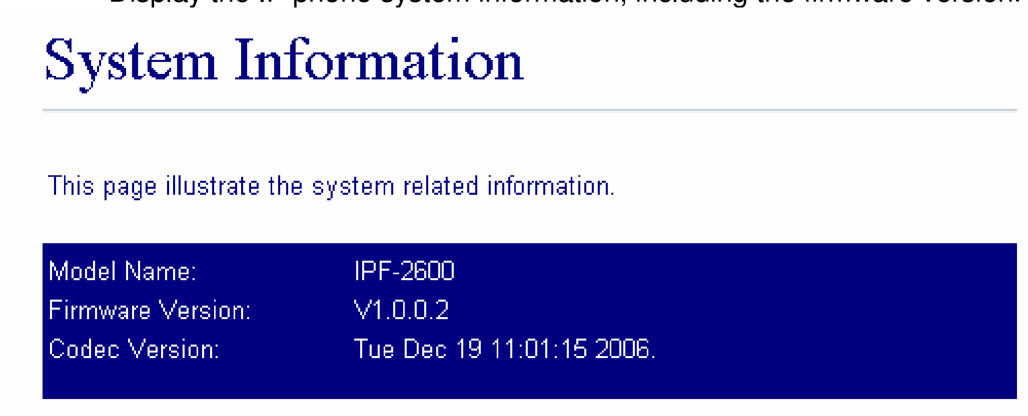
Enter the Menu on the IP phone, go to Network/status to retrieve the IP address, then check on the IP address on the connected computer system, Then, type the address http:// Phone-IP-address:9999 in the browser URL address, and login with the user name and password to enter the configuration setting page. The default username and password is "root" and "test".

To save the configuration settings through the website, press "submit" after configuring on each page. Then, press the "save changes" on the left-hand side column, click on "save". After the IP phone restart, it will save all the changes made through the website.

The following display shows the on-screen instructions for the configuration using the web browser

5.1 Information

Display the IP phone system information, including the firmware version.



Model Name: Product mode
Firmware Version: Firmware Version
Codec Version: Codec Version Date

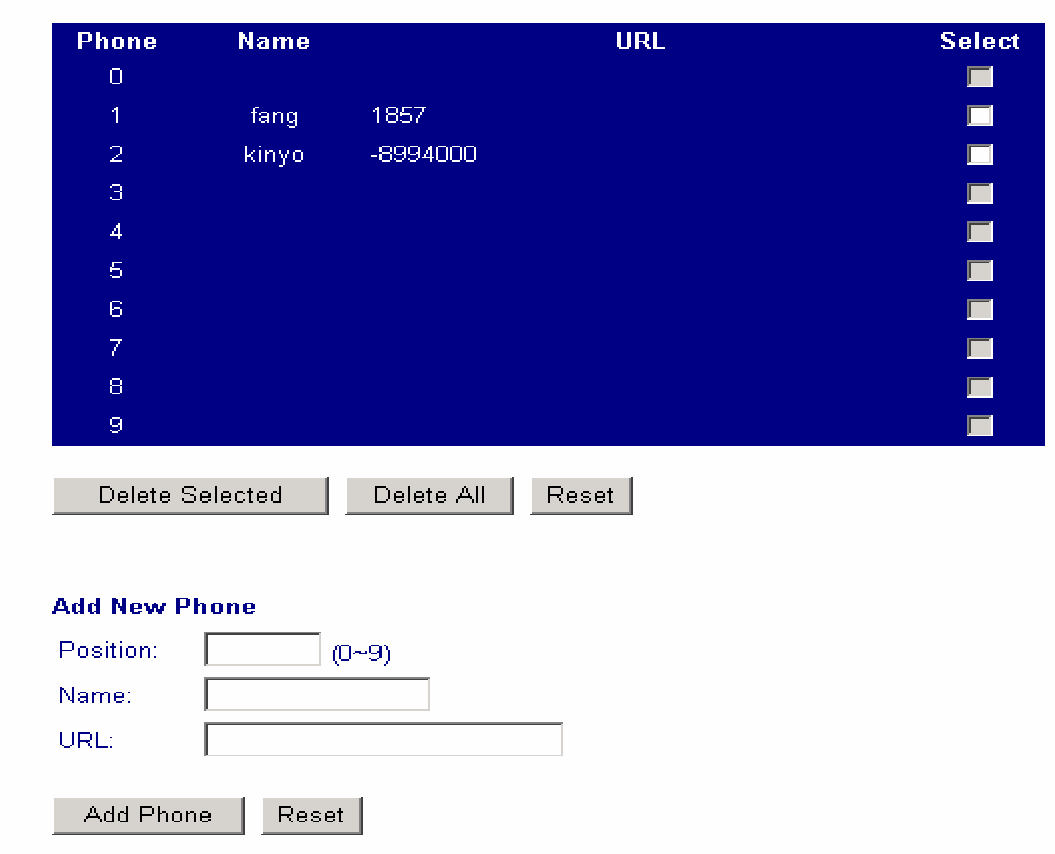
5.2 Phone Setting

5.2.1 Phone Book Setting

Phone Book	
Item	Description
Delete Selected	Delete the selected entries.
Delete All	Delete all entries.
Position	Select the position to add phone entries.
Name	Enter the name for the phone entries.
URL	Enter the phone number. Add "-" Symbol for an IP phone number to differentiate between the PSTN number.

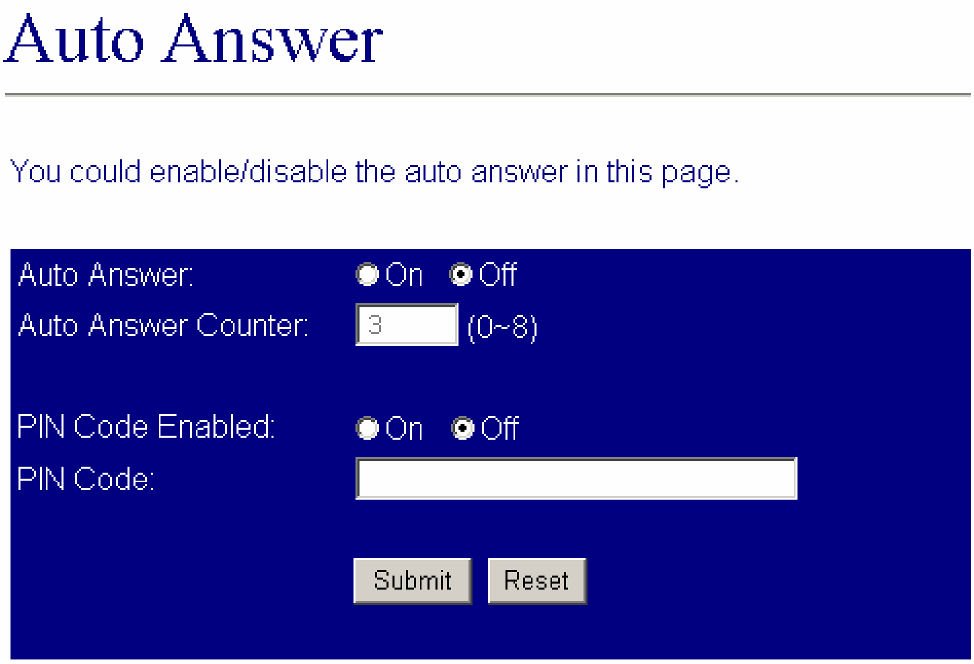
Add Phone	Press to confirm to add phone entries. Maximum phone book entries are 140.
Reset	Reset the phone book entries

5.2.2 Speed Dial



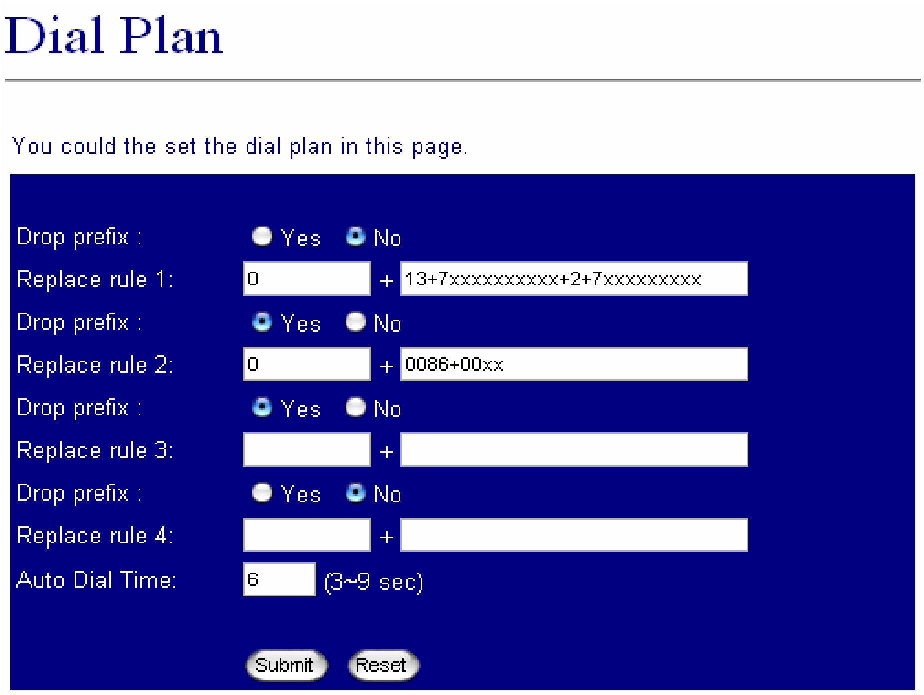
Speed Dial	
Item	Description
Delete Selected	Delete selected entries
Delete All	Delete all entries
Position	Select the position to add phone entries. The position for 1,2,3 & 4 is set-up for M1-M4 speed dial lines, Note: if Default IP mode only Support IP(SIP), if Default PSTN mode M1-M4 Support PSTN The other only support IP(SIP).
Name	Enter the name for the speed dial entries
URL	Enter the speed dial number. Only support IP(SIP)
Add Phone	Press to confirm to add phone entries. Maximum speed dial entries are 10.
Reset	Clear the data that set just now

5.2.3 Auto Answer



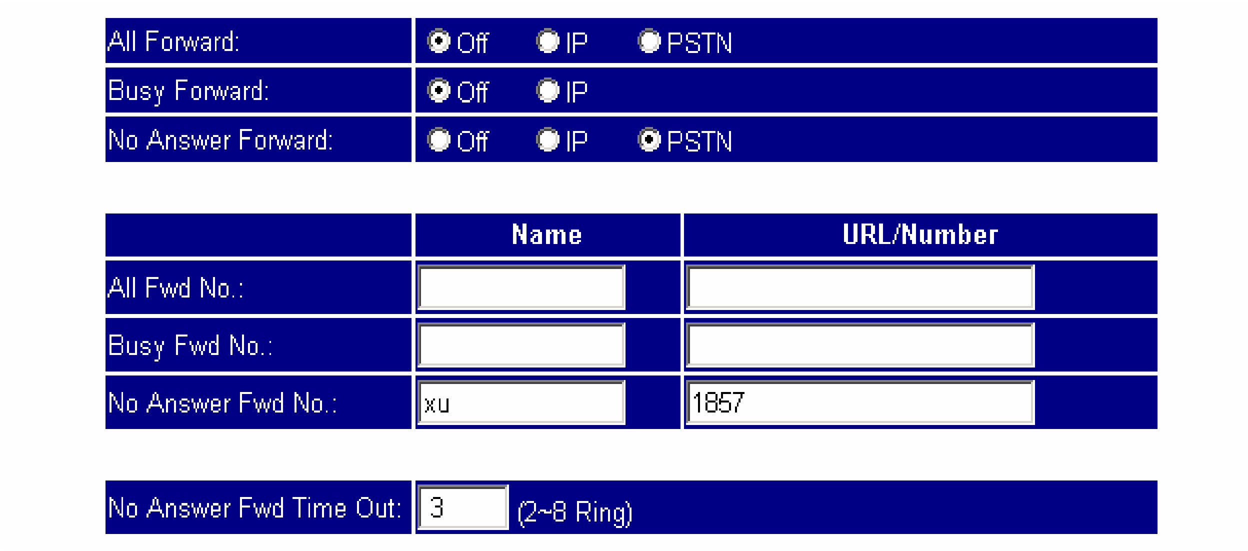
Auto Answer	
Item	Description
Auto Answer	Select "On" to turn on the auto answer function.
Auto Answer Counter	Set the incoming call's ring time before automatically answer the call function. Select between 0-8 ring times.
PIN Code Enabled	Select on to enable Pin code setting.
PIN Code	Enter a Pin code

5.2.4 Dial Plan Setting



Dial Plan Setting	
Item	Description
Drop Prefix	Default No(Add Prefix).When set Yes (subtract Prefix from dialed number), subtract Prefix from dialed number. Group 4 data set - No (add Prefix): When the rule matches ,directly add Prefix Before dialed number.Replace Item could be set seven number. - Yes (minus Prefix): When the rule matches, Then subtract compatible code top from Dialed number, Can import 39 number.
Replace rule1	Set dial rule: Add or Subtract xxx: Designated code length.
Auto Dial Time	Default Value is 5S.After long wait automatically dial .
Submit	Changes in the implementation of storage set.
Reset	Clear the data which have set.

5.2.5 Forward Setting

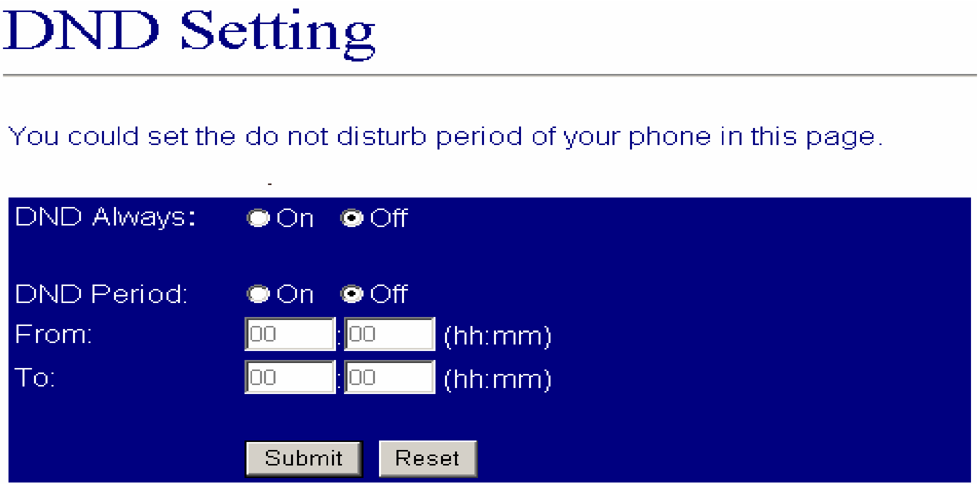


Forward Setting	
Item	Description
All Forward	Forward all calls to another phone number. Capable to forward all calls to IP or PSTN telephone lines If All forward ON, FWD LED light up ,LCD display "AF XXXXXX", "AF" Show ALL Forward ,"XXXXXX" Show destination number, user could select to forward to IP/PSTN Phone. Note: Support: IP－IP－IP,IP－IP－PSTN or PSTN－PSTN－IP For Example: IP(incoming calls IP(SIP) mode) －IP(receiving IP(SIP) mode) －IP(forward to IP(SIP))
Busy Forward	Forward busy calls to another phone number (this function is limited to IP incoming call) , If Busy forward ON, FWD LED light up, LCD Display "BF XXXXX" , Could not Forward to PSTN Phone at this item "BF" Show Busy Forward ,"XXXXX" Show destination number
No Answer Forward	When Ring Time out, Forward no answer calls to another number. If No Answer forward ON FWD LED light up, LCD Display "NF XXXXXX" , user could select to forward to PSTN/IP Phone. Note: Support: IP－IP－IP,IP－IP－PSTN or PSTN－PSTN－IP For Example: IP(incoming calls IP(SIP) mode) －IP(receiving IP(SIP) mode) －IP(forward to IP(SIP))
All Fwd No.	Set the name and phone number for All Forward calls.
Busy Fwd No	Set the name and phone number for Busy Forward calls.
No Answer Fwd No	Set the name and phone number for No Answer Forward calls.
Time Out	Set the ring time for call forwarding.
Reset	Reset call forwarding setting

5.2.6 Call Waiting

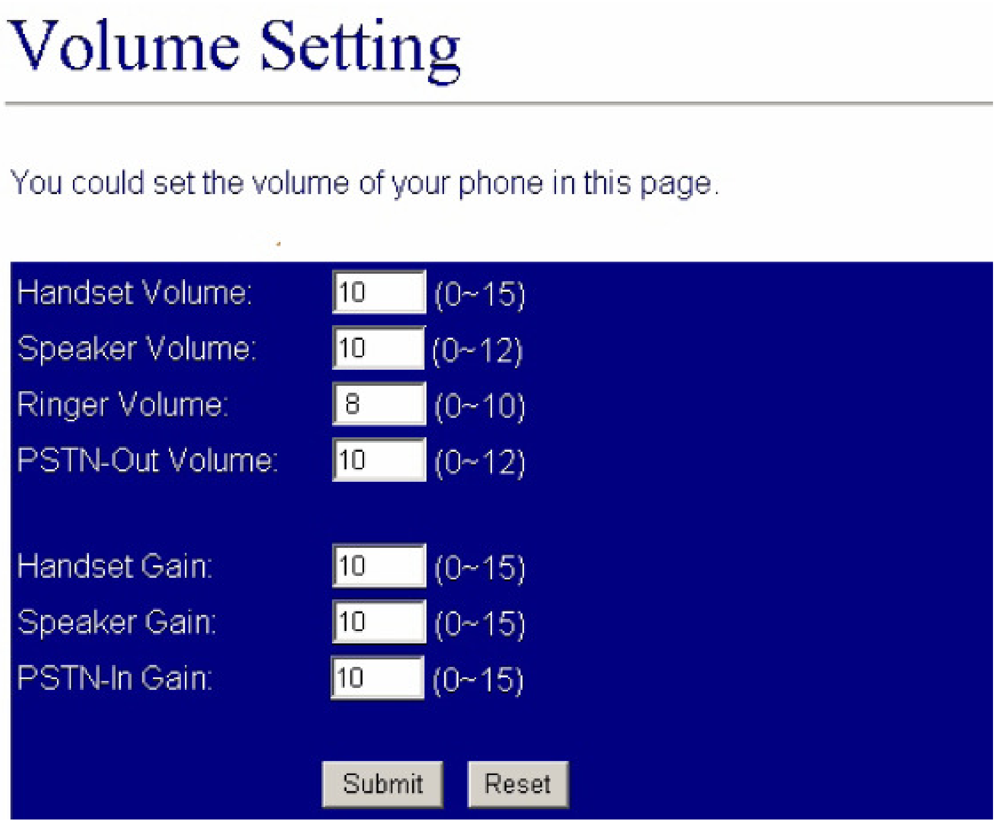
Call Waiting	
Item	Description
Call Waiting	Call Waiting – While talking to A party, B is calling in, at this time there will be beeping sound every 3 seconds to indicate a call waiting, press "hold" key or press L1/L2 button to put caller A on hold and switch to caller B. You can press "hold" key or press L1/L2button to put caller B on hold and switch back to caller A.

5.2.7 DND Setting



Call Block	
Item	Description
DND Always	Always block incoming IP calls – Select "On" to enable the function, all incoming IP calls will hear a busy tone. Select "Off" to disable the function. Note: this function does not affect PSTN calls
DND Period	Block period setting – set the time to block IP calls. Enter start time at the "From" column, and enter end time at the "To" column.

5.2.8 Volume Setting

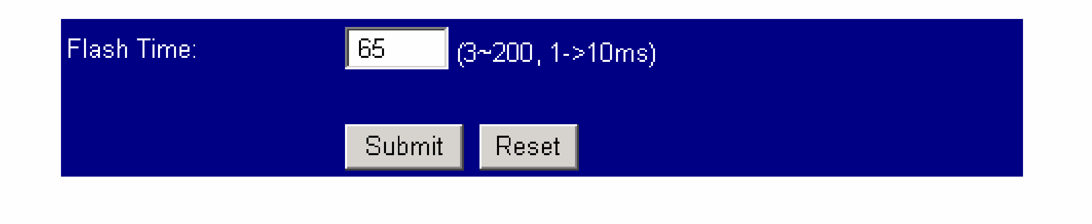


Volume Setting	
Item	Description
Handset Volume	Adjust handset volume.
Speaker Volume	Adjust Speaker volume.
Ringer Volume	Adjust Ringer volume
PSTN-Out Volume	Adjust volume for using PSTN lines
Handset Gain	Adjust handset gain (suggested not to exceed 13)
Speaker Gain	Adjust Speaker Gain.
PSTN-In Gain	Adjust the gain for using PSTN lines.

5.2.9 Melody Setting

Ringer Setting	
Item	Description
Ringer	Select “On” to enable ringer setting, select “off” to turn off.
Ringer Type	Select between the 4 different types of ringer

5.2.10 Flash Time Setting

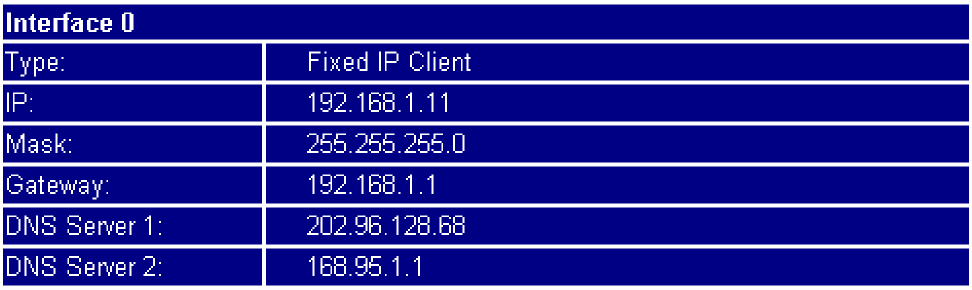


Flash Time Setting	
Item	Description
Flash Time	Flash time setting – the hook button act as a switching call function, set the time limit for the hook button when it is pressed till it is released. If the hook button is pressed longer than the time setting, it will be treated as hand up; if it is released before the time setting, it will be treated as a switching call function.

5. 3 Networking

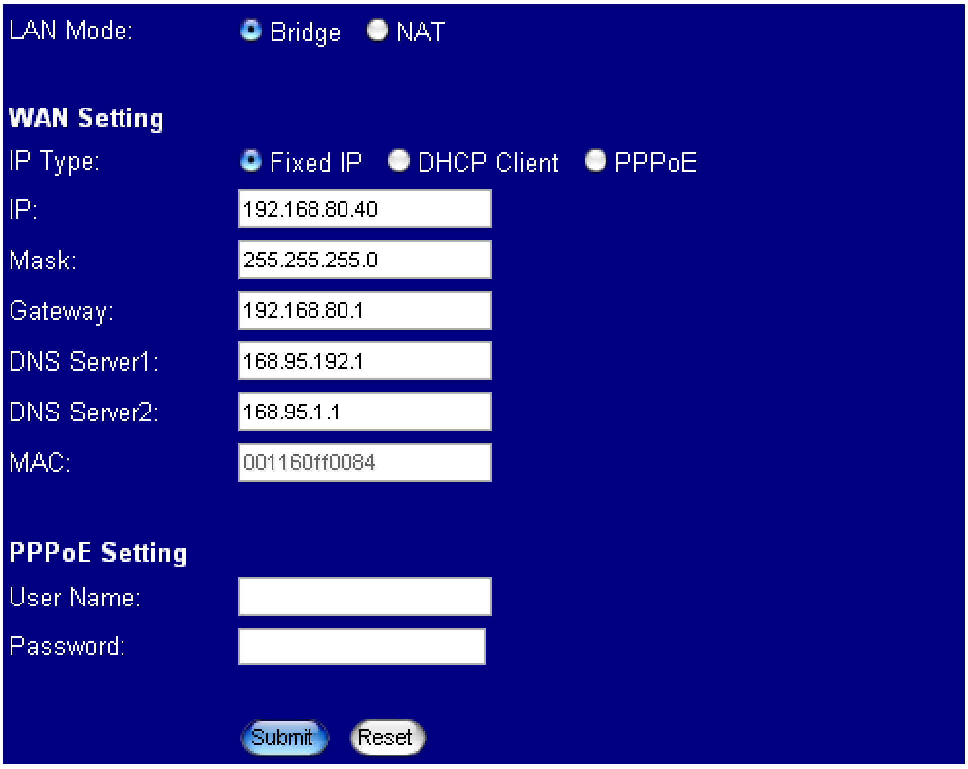
5.3.1 Status

IPF-2600 General Information Display



Status	
Item	Description
Type	IP address type show
IP	Show IP address
Mask	Show network shielding
Gateway	Show the Set Gateway's IP address
DNS Server 1	Show the Set Primary DNS address
DNS Server 2	Show the Set Secondary DNS address

5.3.2 Network Setting

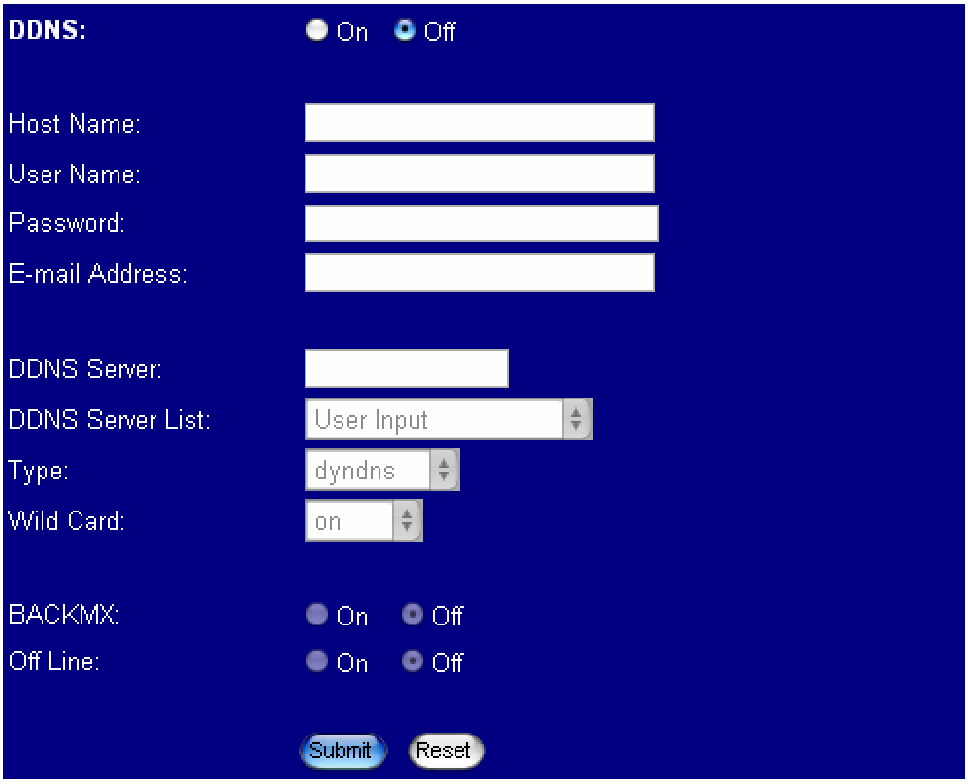


Network Setting	
Item	Description

LAN Mode	Setting LAN mode Bridge : Select whether turn on/turn off Bridge function NAT: Select whether turn on/turn off Router function
IP Type	IP Type. Select a proper connection for network setting: ● Input Local IP, Subnet Mask, Gateway IP, DNS after Fixed IP is selected. ● DHCP server received IP information after DHCP Client is selected. (Suitable for Cable Modem users)
IP	Set up IP Address
Mask	Set up Subnet Mask Address
Gateway	Set up Gateway Address
DNS Server 1	Set up Primary DNS IP Address
DNS Server 2	Set up Secondary DNS IP Address
MAC	Set up MAC Address
PPPoE Configuration	
PPPoE	ADSL User Options
User Name	When selecting PPPoE for IP Type, set up a user name for ADSL account.
Password	When selecting PPPoE, set up password for ADSL account.
Bridge	
Bridge	Option for bridging the WAN and LAN port.

5.3.3 DDNS Setting

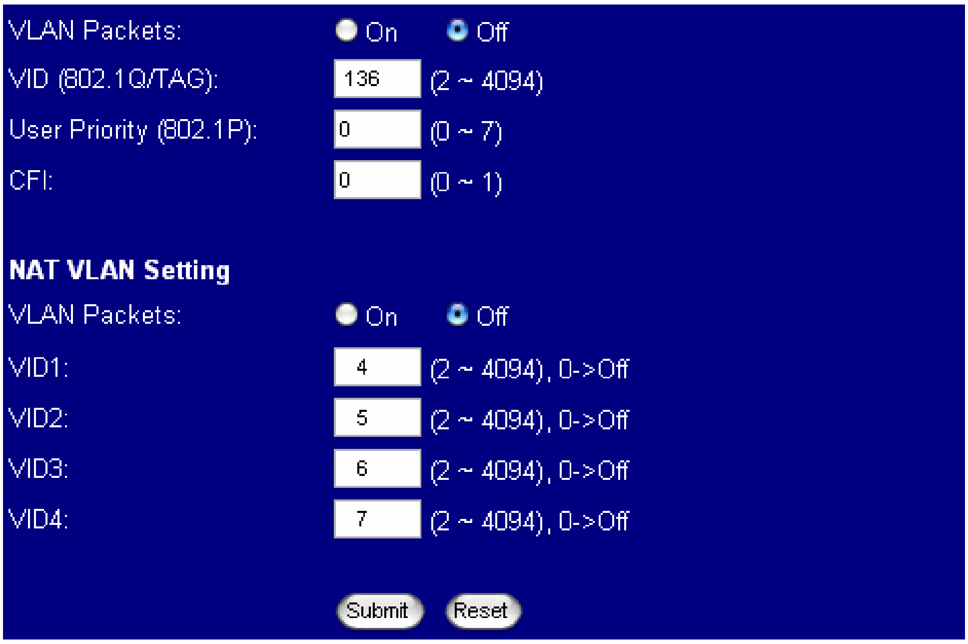
DDNS Setting



DDNS Setting	
Item	Description
DDNS	DDNS setting. Account and password for DDNS service. The status of Gateway is under SIP operation. If you need any help, please refer DDNS instruction for more detail, Set On/Off DDNS function
Host Name	DDNS Server Name
User Name	DDNS Server Registration Account
Password:	Account Password
E-mail Address	E-mail Address
DDNS Server	DDNS the IP address or domain name
DDNS Server List	DDNS service provider list.
Type	DDNS mode of choice.
Wild card	DDNS Wild card function ON/OFF
Back MX	DDNS Back MX function ON/OFF
Off Line	DDNS Off Line function ON/Off
Submit	Changes in the implementation of storage set.
Reset	Clear the data which have set just now.

5.3.4 VLAN Setting

VLAN Setting

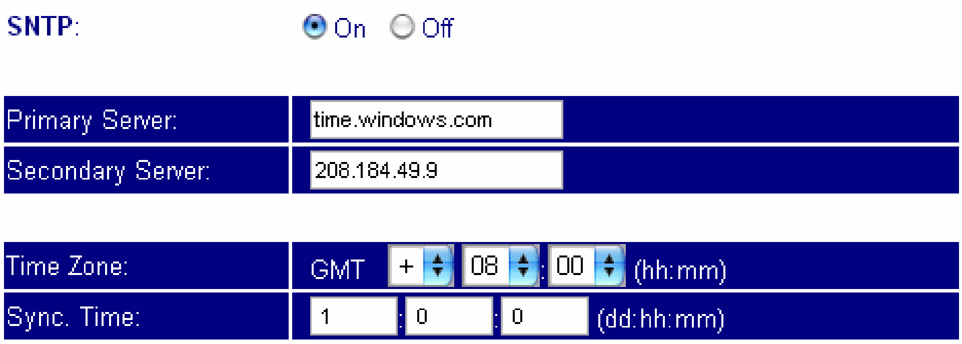


VLAN Setting	
Item	Description
VLAN Packets	Default Off.When set "On", turn on receive VALN Packets function
VID	Default is 136, Virtual LAN Server setting, VLAN Identifier (Virtual LAN ID. Name VLAN ID or VID).
User Priority	Default is 0, user priority Setting, user priority(0-7).

CFI	Default is 1, Canonical Format Indicator (CFI). CFI=1: head of the packet contains spaces RIF , And the NCFI Flag which inside RIF decision information is carried by MAC address. "Standard format" (Canonical Format) or "non-standard format" (Non-Canonical Format) CFI=0: head of the packet Do Not contains spaces RIF, information is carried by MAC address is "Standard format" (Canonical Format)
NAT VLAN Setting	
VLAN Packets	Default Off.When set "On", turn on receive "VALN Packets" function.
VID1 ~ 4	Default is 136, Virtual LAN Server setting, VLAN Identifier (Virtual LAN ID. Name VLAN ID or VID).
Submit	Changes in the implementation of storage set.
Reset	Clear the data which have set just now.

Please refer to VLAN instruction for setting configuration.

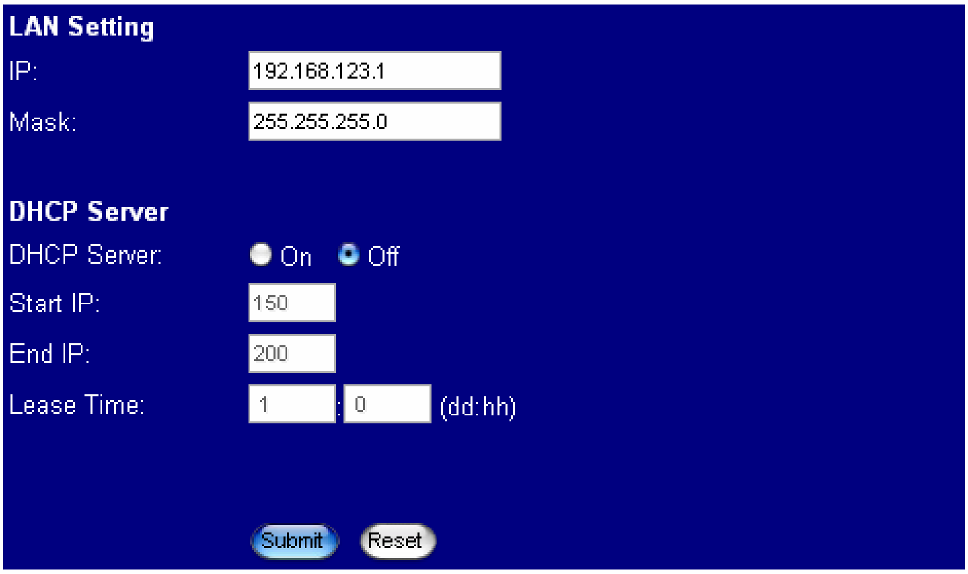
5.3.5 SNTP Setting



SNTP Setting	
Item	Description
SNTP	When setting the use time for a server, On is "in use", Off is "not in use".
Primary Server	Primary Server. Input the Primary Time Server Address.
Secondary Server:	Secondary Server. Input the Secondary Time Server Address.
Time Zone	Time Zone. Input the local time.
Sync. Time	Sync. Time. Input the Sync. Time.

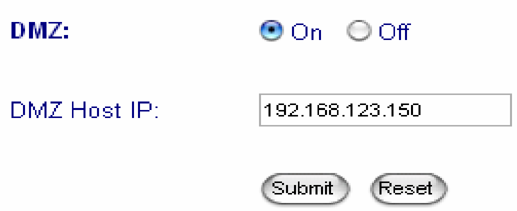
5.4 NAT Router

5.4.1 LAN Setting



LAN Setting	
Item	Description
IP	LAN port will become a virtual domain When the IPF2600 Router function turn on
Mask	Set up LAN Subnet Mask Address
DHCP Server	Set to Open and Close DHCP Server. To retrieve IP address automatically from Start IP and End IP server when Client End is open.
Start IP	Set Start IP of DHCP
End IP	Set End IP of DHCP
Lease Time	Lease Time is the available use time for designated IP assigned by server.
Submit	Submit
Reset	Reset Default Settings

5.4.2 DMZ Setting

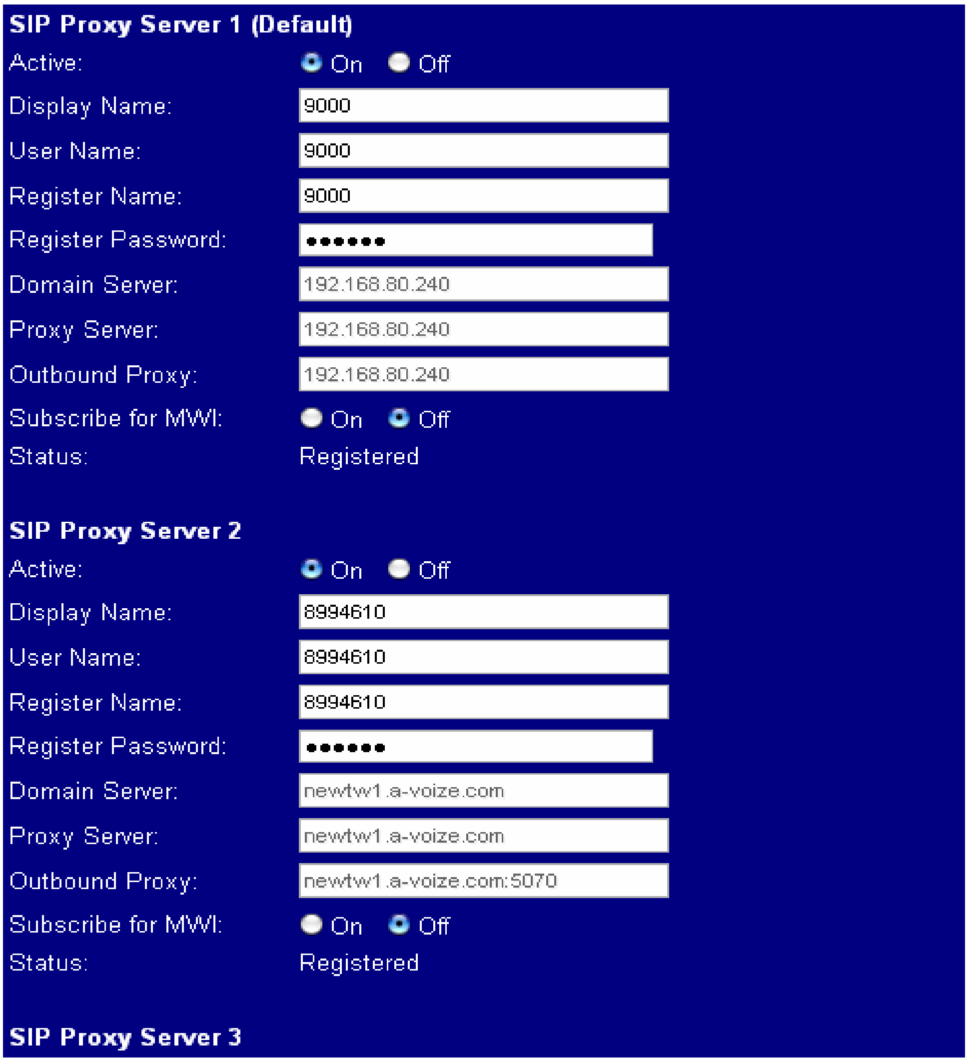


DMA Setting	
Item	Description
DMZ	DMZ. DMZ is to set a domain server outside of the firewall, it allows to designate

	a host to have mutual communication with external network without protections, so that it can function for other required applications with an external host.
On	Open DMZ Function
Off	Close DMZ Function
DMZ Host IP	DMZ Host IP

5.5 SIP Setting

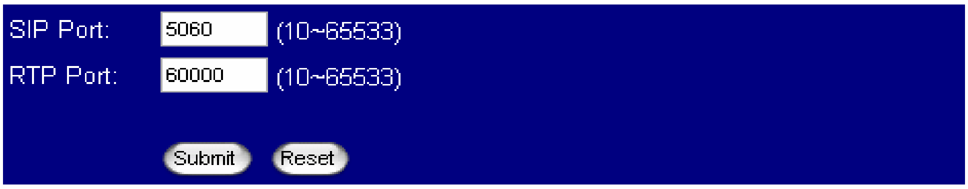
5.5.1 Service Domain



Service Domain	
Item	Description
Active	Account Options ● You can register 3 different accounts at the same time. When selecting "On" at the current field, input corresponding register number in the tab. ● Dial-out number is the first number registered (the first pick is the number with "On" option). IF the first accounts could not be registered automatically Switch to register on the next account. Any of 3 registered numbers can

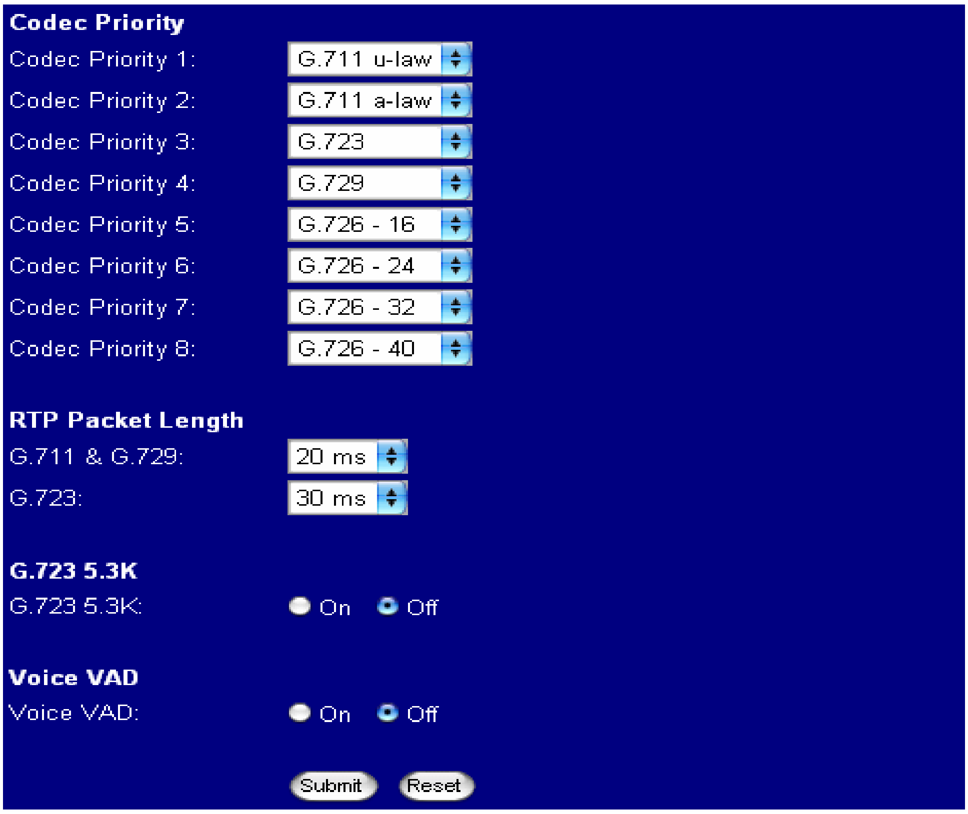
	receive incoming calls.
Display Name	Account Label
User Name	User Name. the number in use is the phone number of the device.
Register Name	Registered Name
Register Password	Registered Password
Domain Server	SIP server domain name or IP address
Proxy Server	● Please input registered Gateway IP address or domain name after "On" is selected. ● Please input registered Gateway IP address or domain name
Outbound Proxy	Outbound Proxy Server
Status	Registered Status. "Registered" indicates that register process is completed, "Not Registered" indicates that Registration process is failed.

5.5.2 Port Setting



Port Setting	
Item	Description
SIP Port	SIP Protocol port, the default setting is 5060.
RTP Port	RTP port number is the number for Vocal sending and receiving setting. It can be arranged from 1024 to 65533 but it must be even numbers.

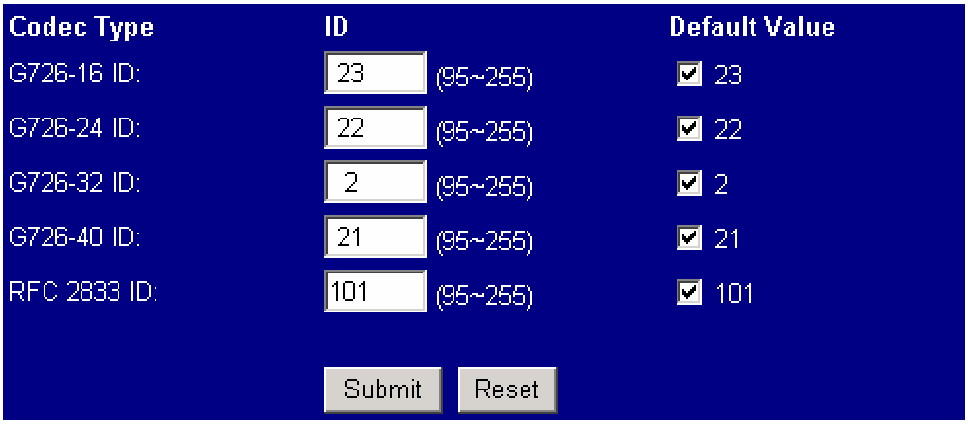
5.5.3 Codec Setting



Codec Setting	
Item	Description
Codec Priority	Codec Options. Select the Codec Priority(Codec Priority 1 as the first choice), and Voice Codec format
G.723 5.3K:	When selecting g.723 5.3K as Voice Codec, set up to "On" for Codec of 5.3K/S.
Voice VAD	Voice Activity detection ON/OFF
Submit	Submit
Reset	Reset Default Settings.

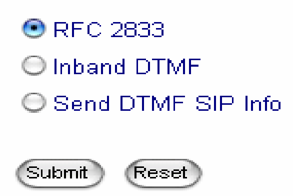
5.5.4 Codec ID

For the settings of G726 Codec and RFC 2833, we suggest setting them up with default value unless the Registration platform has other requirements.



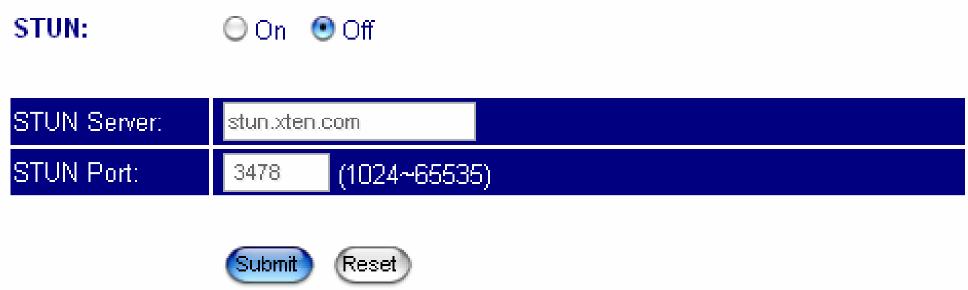
5.5.5 RTP Setting

The options of Outband DTMF and Send DTMF SIP Info are provided by ISP.



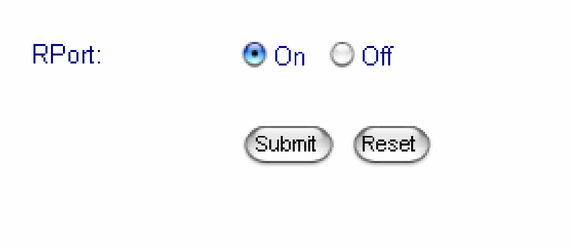
DTMF Setting	
Item	Description
RFC2833	transmission DTMF signals Using RFC2833
Inband DTMF	transmission DTMF signals Using Inband DTMF
Send DTMF SIP Info	transmission DTMF signals Using SIP Info
Submit	Changes in the implementation of storage set.
Reset	Reset Default Settings.

5.5.6 STUN Setting



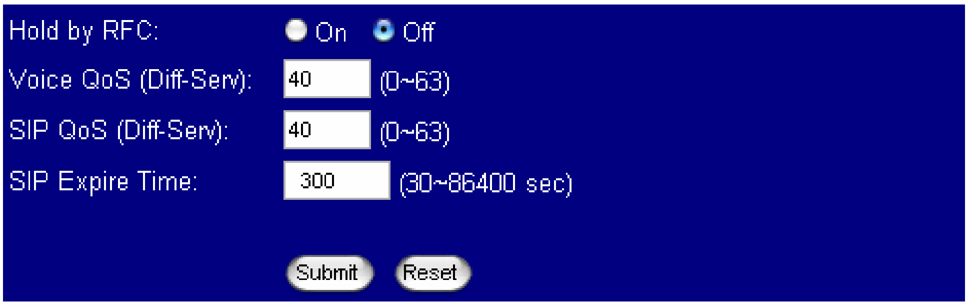
STUN Setting	
Item	Description
STUN	Check whether the setting uses STUN Protocol
STUN Server	STUN Server Address
STUN Port	STUN Port Number
Submit	Changes in the implementation of storage set.
Reset	Reset Default Settings .

5.5.7 RPort Setting



RPort Setting	
Item	Description
RPort	Set whether using RPort Protocol
Submit	Changes in the implementation of storage set.
Reset	Reset Default Settings .

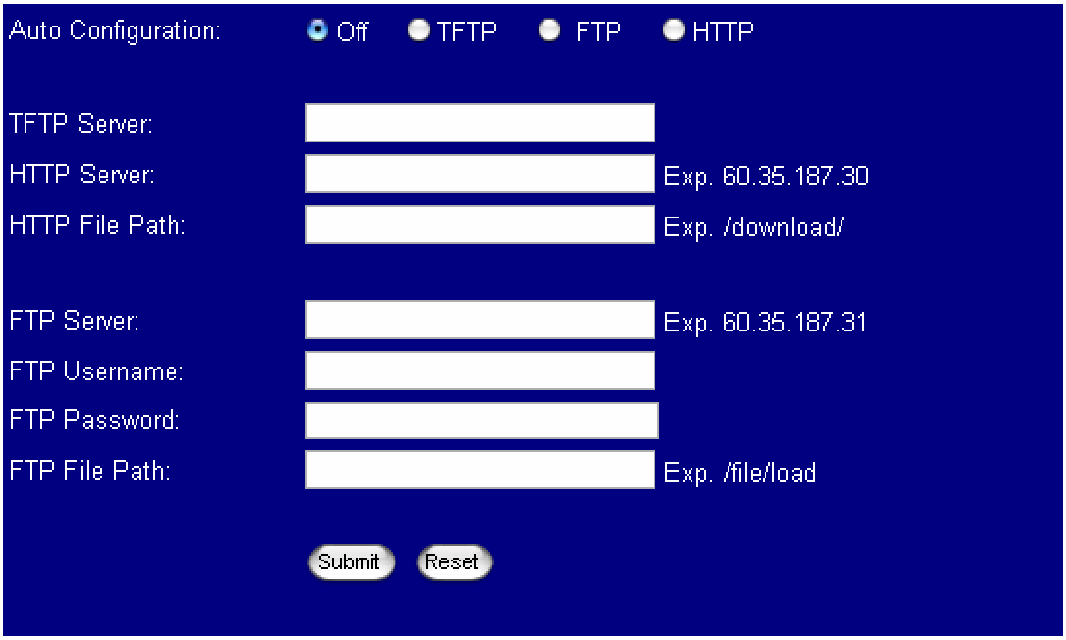
5.5.8 Other Setting



Other Setting	
Item	Description
Hold by RFC	The RFC option is provided by ISP.
Voice QoS	Set up the size of Voice QoS. Higher number is larger than lower number.
Sip QoS	Set up the size of SIP QoS.
SIP Expire Time	SIP Expire Time

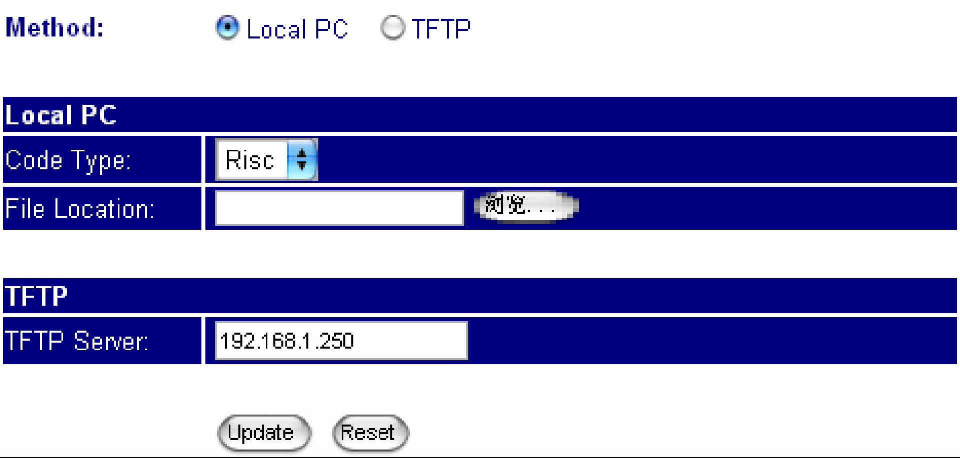
5.6 Other

5.6.1 Auto Config



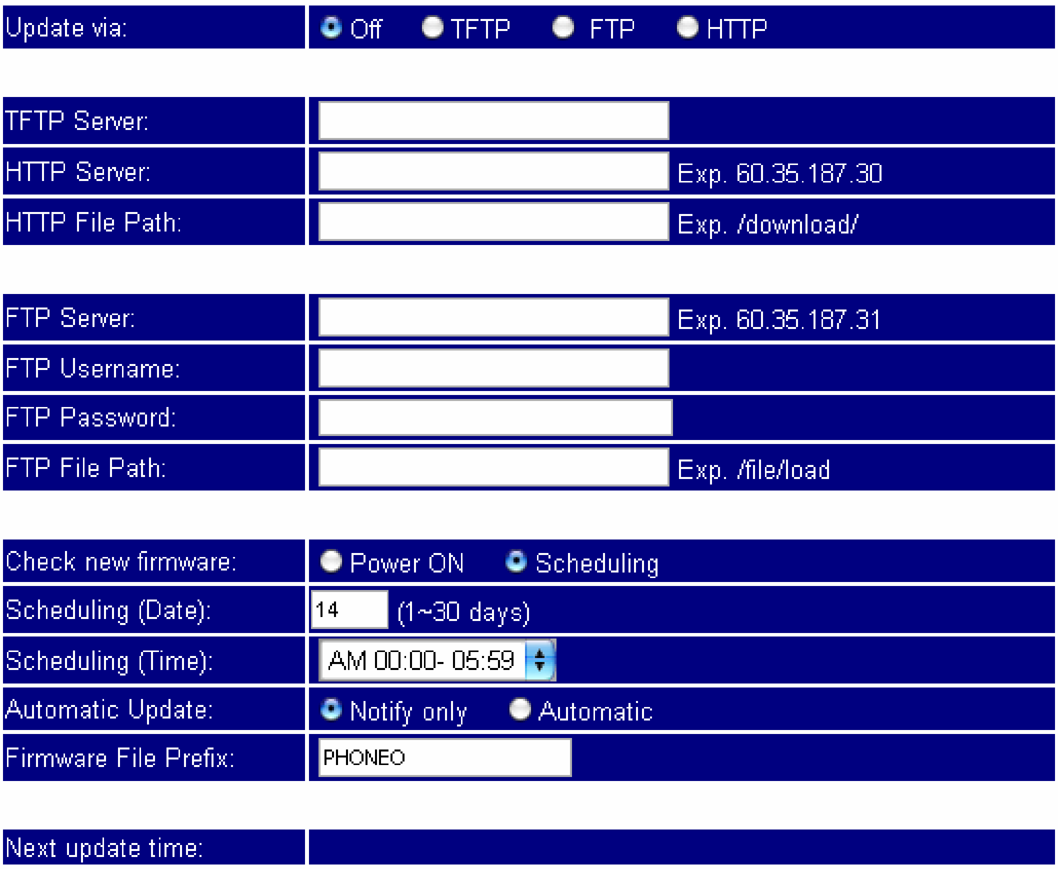
Auto Config	
Item	Description
Auto Configuration	Default Is Off, Automatically Configuration environmental setting, with TFTP, FTP and HTTP three ways.
TFTP Server	TFTP Server IP address or Domain Name.
HTTP Server	HTTP Server IP address or Domain Name.
HTTP Path	Set File Path in Http Server, for example: /123/.
FTP Server	Set FTP Server IP address or Domain Name.
FTP Username	FTP Username
FTP Password	FTP Password
Submit	Submit
Reset	Reset Default Settings

5.6.2 Firmware Upgrade



Other Setting	
Item	Description
Method	Select HTTP or TFTP to update software.
Code Type	Select Risc and DSP type to update Program Code Type
File Location	Select Upgrade File Location
TFTP Server	Set TFTP server address.

5.6.3 Auto Update

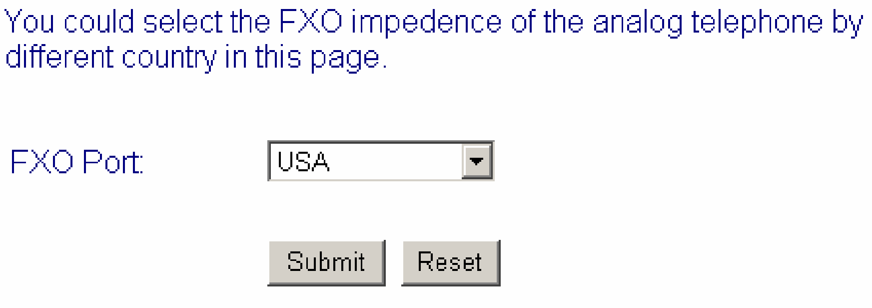


Auto Update	
Item	Description
Update via	Default is Off(Do not Auto Update). Automatically update environmental setting, with TFTP, FTP and HTTP three ways
Check new Firmware	Default Scheduling(In accordance with schedules).Ways of Checking whether have new Version for the IP Phone Power ON: Check whether have new Version for the IP Phone at Power ON Scheduling: Check whether have new Version for the IP Phone In accordance with schedules.
Scheduling (Date)	Default is 14 days, Every once in the days when the IP Phone check whether have new firmware Version for it Auto Update, the shortest one day, the Longest thirty days
Scheduling (Time)	Set Every time going to check the time span, Each section to check the time, a detailed time randomly generated. Section 4 provides for AM 00:00-05:59,AM 06:00-11:59, AM 12 : 00-17:59 AM 18:00 - 23:59. Default is AM00:00-05:59
Automatic Update	Updated automatically, providing Notify only (notification messages). Automatic (automatically updated) Default is Notify Only -Notify only: Does not automatically update the implementation of the action ,Phone : LCD Display the message "Found new Firmware", users can choose whether update to the new software -Automatic: If Scheduling has set ,IP Phone could automatically update New Firmware version .But POWER ON ,need user to choose whether update the new Firmware .
Firmware File Prefix	Default Product model Inspection Firmware version of the product model data
Next update time	Next checking or Updating date and time
Submit	Submit
Reset	Reset Default Settings

5.6.4 Default Setting

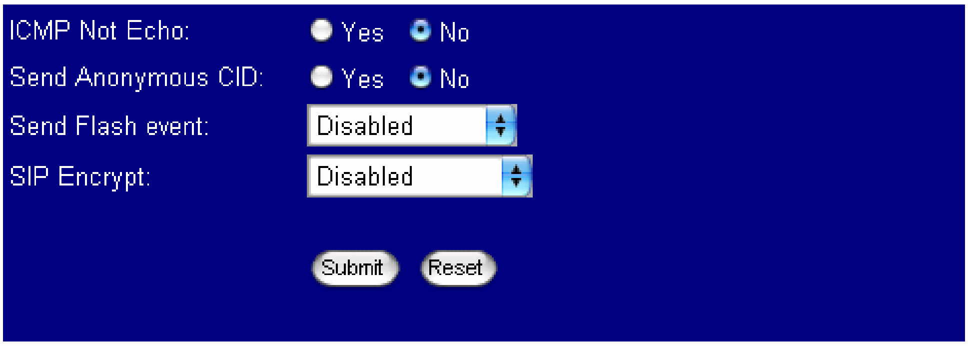
5.6.5 FXO Port Setting

Select appropriate parameters based on the features of local telecommunication equipments. It must be advised by distributors or engineers.



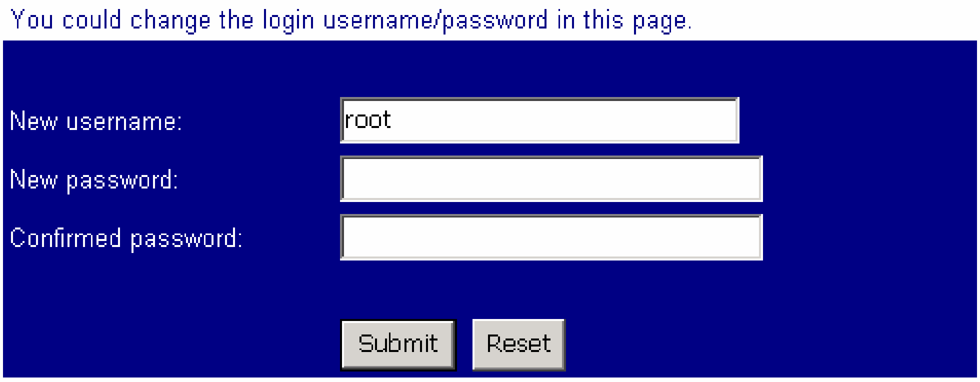
5.6.6 Advanced Setting

Check if the signals sent by the host are reporting back. We suggest using default values.



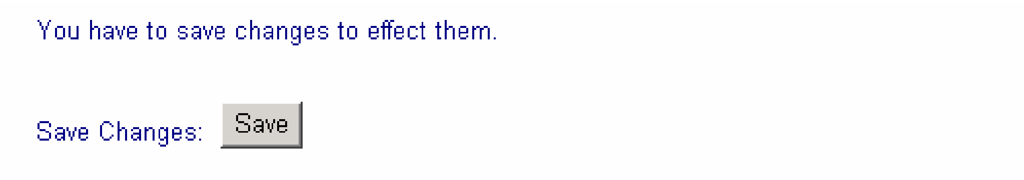
Advanced Setting	
Item	Description
ICMP Not Echo	Default No, responding to the message of ping, if select "Yes" do not responding to the message of ping.
Send Anonymous CID	Default No, Have no Call out security function, send information of the IP Phone, if Select "Yes" do not send information of the IP Phone
Send Flash event	Default Disable , User could Select DTMF Event or SIP Info to Send flash event.
SIP Encrypt	Default Disable , IPF2600 have INFINET, AVS, WALKERSUN1. WALKERSUN2 four format for SIP encryption. This service is provided by use environment
Submit	Submit
Reset	Reset Default Settings

5.6.7 System Auth

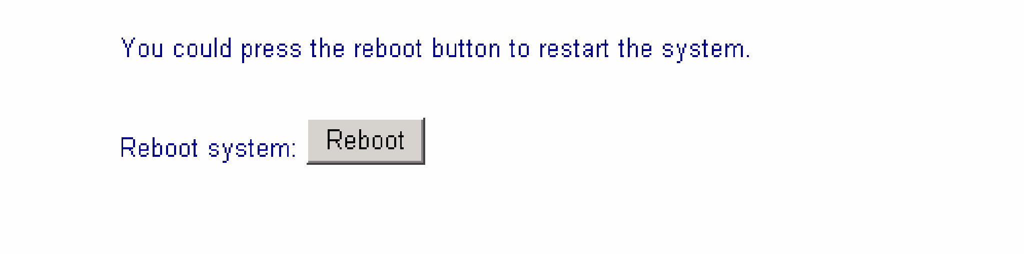


System Auth	
Item	Description
New username	Username. Set up a New Username for Registration.
New password	Set up a New Password
Confirmed password	Confirm Password

5.7 Save Changes



5.8 Reboot



Ⅵ. Product Specifications

Voice Interface	
Call Control Protocol	SIP v1 (RFC 2543), v2 (RFC 3261)
Voice Compression	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)
Delayed (Point to Point)	< 100ms
Echo Cancellation	Packet Loss Compensation, Adaptive Jitter Buffer, VAD (Voice activity detection), CNG (Comfortable noise generator), AEC (Acoustic echo canceller), G.165 (LEC, Line echo canceller), G.168 (EC, Digital network echo canceller)
Flow of the Average	5.3K(G.723.1) ~ 64K(G.711) bps
Other Support	In-Band DTMF, Out-of Band DTMF, SIP Info
LAN Interface	
Interface Spec	10/100Mbps Fast Ethernet
Interface Connector	RJ-45 Connector
Management	Web Browser
IP Address	Static IP / DHCP Client / PPPoE Client / TFTP Client / HTTP Server / DNS Client / Telnet / SNTP / RTP / RTCP / DDNS
Firmware Update	HTTP / TFTP / Telnet
Call Features	Call Hold / Call Waiting / Call Forward / Call Transfer / Caller ID / Call Block / 3-way Conference
Other Specs.	
Input AC/DC Range	100 – 240VAC, 50 – 60Hz, 12VDC
Power Consumption	6W
Operation Environment	Working Temporature : 0 ~ 40℃ (32° ~ 104℉)
	Storage Temporature : -30 ~ 65℃ (-22° ~ 149℉)
	Relative Humidity: 10 ~ 95% Non-Condensing
Weight	0.75g
Certificates	CE / FCC
Dimensions	219mm *180 mm *75 mm
Others	140 Phone Contacts
	IVR in English version
	NAT Pass-through (STUN / uPnP / R-Port)
	Networking Status, Firmware Update, No-Answer, Call Forward, Busy Call Forward, Call Forward, Missed Call Forward, Web Management, LED indicators

Ⅶ. Frequently Asked Questions (FAQ)

1. How do I know that IPS-2600 Registration is complete?

A: If IPF-2600 is registered successfully; the Registration number will be displayed on LCD. If "No Service" is displayed on LCD that means the Registration is not complete. System requires 2 minutes to complete Registration for IPF-2600.

2 . After connecting to power source and activating phone device, why do I see "ETHERNET ERROR" on my LCD?

A: Check if Ethernet Cable is correctly connected. If it is connected, please check your network connection.

3. Can I dial the phone if a blackout occurs?

A: Gateway will have no power supply if a blackout occurs. At this point of time, the IP phone will not be able to dial or receive phone calls. A blackout won't have any effects on the PSTN line, so the PSTN line can not still operate normally.

4. Does IPF-2600 IP phone supports Inbound Transit function?

A : Yes, it supports PSTN Inbound Call. Inbound Transit means that users use a PSTN line to make calls to IP telephone which connected a PSTN line, and this call will be forwarded to another IP phone through IVR answer function.

How to Dial An Inbound Transit Call:
Dial a call to your IP telephone through the PSTN line, you will hear a dial tone after the phone rings (3-9) times, and dial the SIP account number. If PIN code is set you need to input the PIN code before dialing the SIP number.

5. How to make a PSTN phone call?

A: Pick up the handset, after a dial tone, you can dial a PSTN number or push IP Call switch button to make an internet call.

6. How to use "Call Hold" function?

A: When you are on a call, press the "Hold" key or "Swap" key to put the caller on hold; to retrieve the call, press the "hold" key or swap key again.

7. How to use "Call Waiting" Function?

A: When you are on a call, if you hear a dial tone twice that means there is another incoming call for you. If you want to answer it, press "hold" key or "Swap" key to pick up the call.

8. How to make a "3-way Conference" call?

A: **"3-way Conference" call**
When you are on a phone with party A (L1) and you want to invite party B to join your conversation, press "hold " key to put party A (L1) on hold, and dial Party B's number (L2) after you hear a dial tone. When party B is connected, inform party B that there is a "3-way Conference" call you like to invite

him/her to participate.

a. If party B accepts your invitation, press "Conf" key to enter the 3-way conference call
b. If party B rejects your invitation or call forwarding is failed, press "hold" or "Swap" key to cancel call forwarding and retrieve the call back to party A.

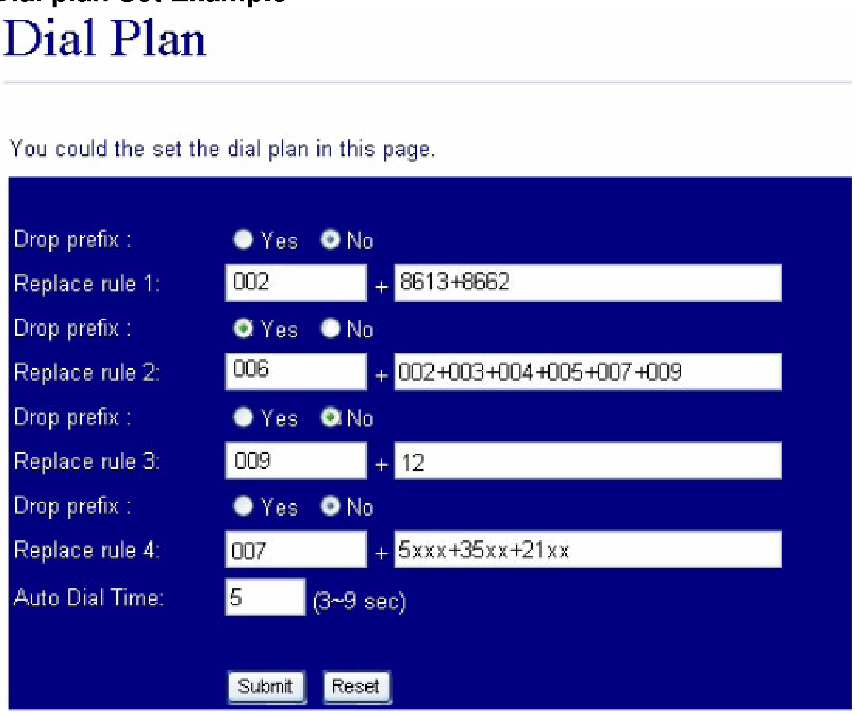
9. How to make "Call Transfer"?

Connect a call with party A, press "XFER" key and input party B's number. When the line is connected, "Call Transfer" is complete.

10. How to use Speed Dial?

A. Step1. Store the phone numbers in the speed dial list
Step2. Pick up the handset or press "Speak" key
Step3. Press "*" key to switch to IP line
Step4. Press speed dial number
Step5. Press "#" key to complete
(Note: the speed dial function only supports IP to IP calls.)

11. Dial plan Set Example



Example:

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

1: When input numbers include 8613, as long as the numbers begin with 8613, 002 will be automatically added in front of all the numbers, the actual number is [002+8613+xxx].
2: When input numbers include 8662, as long as the numbers begin with 8662, 002 will be automatically added in front of all the numbers, the actual number is [002+8662+xxx].

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

1: When input numbers include 002, as long as the numbers begin with 002, 006 will automatically instead of 002, the actual number is [006+xxx].
2: When input numbers include 003, as long as the numbers begin with 003, 006 will automatically instead of

003, the actual number is [006+xxx].
3: Drop prefix: No, Replace rule 3: 009, 12.
1: When input numbers include 12, as long as the numbers begin with12, 009 will be automatically added in front of all all the numbers, the actual number is [009+12+xxx].
4: Drop prefix: No, Replace rule 4: 007, 5xxx+35xx+21xx.
1: When input number include 5XXX, as long as the numbers begin with 5, followed by three numbers; 007 will be automatically added in front of all all the numbers, the actual number is [007+5xxx]
2: When input number include 534, the numbers begin with 5, follower by two numbers, it is unfit to add Prefix rule, the actual number is [534].
3: When input numbers include 358822, the numbers beginn with 35, followed by four numbers, it is unfit to add Prefix rule, the actual number is [358822].

12. How to switch with Multiple SIP Platform?

A : If IPF2600 has set 2-3 different SIP platform account, User want to switched to the platform which User wish to use ,Please refer to the following method :
For example, IPF2600 has set A, B, C three SIP platforms, A as preset platform; Switching from A to B platform
1: If the IP Phone at idle status, LCD would Display number at A Platform.
2: Pick up Handset or Press "Speaker" button (if default is PSTN, must switch to IP mode), Press "2" key and then press "*" key.
3: When the IPF2600 go to idle status, LCD will display number at B platform, user could make calls at B platform.
If user want to switch to C Platform
4: Pick up Handset or Press "Speaker" button (if default is PSTN, must switch to IP mode).
5: Press "3" key and then press "*" key.
6: When the IPF2600 go to idle status, LCD will display number at C platform, user could make calls at C platform.

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The company reserves the rights to continuously improve the product at any time and support will be based upon the latest release of the product.

Please visit our website at www.artdio.net to download the latest driver and user's manual.

一. 產品介紹

ArtDio IPF-2600 是基於 SIP（Session Initation Protocol）協議的新一代企業級 IP 電話解決方案,創新的技術工藝使其擁有絕佳的音質表現和豐富的功能。

1.1 軟體功能

- 支援 SIP v1 (RFC 2543),SIP v2(RFC 3261),TCP/UDP/IP, RTP/RTCP, HTTP,ICMP,ARP/RARP,DNS,DHCP Client,DHCP Server,SNTP,PPPoE,STUN,TFTP 等網路協定
- 支援多種編解碼協定,如：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, G.729B
- 支援 VAD 動態語音檢測（Voice Activity Detection）,CNG 舒適噪音生成(Comfortable Noise Generator),LEC 線性回音消除（Line Echo Canceller）,語音封包丟失補償（Packet Loss Compensation）, 語音抖動緩衝（Adaptive Jitter Buffer）
- 支援來電顯示及回撥功能（Caller ID Display or Redial）,來電等待(Call Waiting),通話保留(Call Hold),來電轉移(Call Transfer (attended /blind)),來電轉接(Call Forward),三方會議(3-way Conference)
- 支援 In-Band 和 Out-Of Band DTMF(RFC2833),Sip Info。
- 支援第三層 QoS（Tos Field）協定。
- 支援 HTTP1.1 網頁驗證,MD5 SIP 驗證（RFC2069/RFC2617）。
- 可以同時註冊三個不同 SIP 平臺,支援 Outbound Proxy。
- 提供電話按鍵設定、網頁設定、Console 控制臺和 Telnet 設定或提供 HTTP/TFTP/FTP 自動下載設定檔。支援 4 種鈴聲。。
- 支援 HTTP/TFTP/FTP 自動更新軟體。

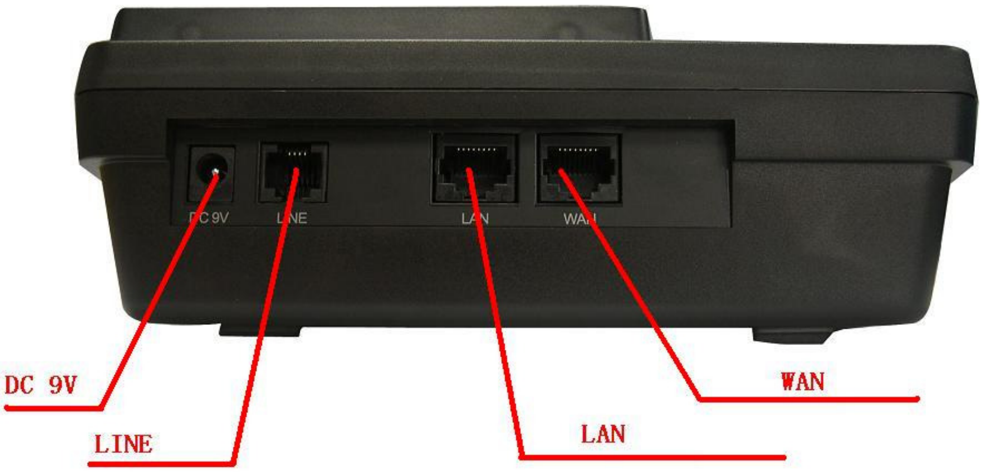
1.2 硬體功能

- RJ-45 *2　10/100Mps 乙太網路介面
- RJ-11*1　局端電話線介面,用於連接局端電話線或分機線（FXO）
- MIC & Line Out　耳麥介面,支援撥入時自動切換到耳麥。
- 全方位免持喇叭擴音器（Speaker）、重撥(Redial)、來電日誌(Incoming/Dialed Calls)、音量控制（VOL-/+）、電話簿（P.B）、來電轉接（XFER）、三方會議（Conf）、勿打擾（DND）、MIC 靜音（Mute）、IP to PSTN 切換。
- 擁有一個兩行背光 LCD 顯示器,八個 LED 指示燈話機狀態顯示。
- 按鍵支援兩路電話切換（L1－L2　支援一路 PSTN 和一路 IP 電話或兩路 IP 電話）,支援四個預設號碼快速撥號（M1－M4）

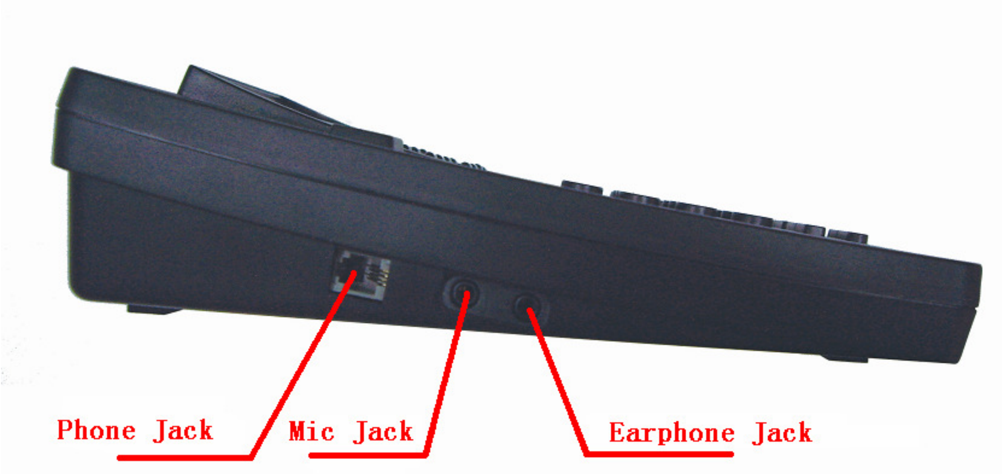
1.3 產品外觀

1. 話機正面

2・話機背面（話機後部圖及各連接埠的名稱）



3.話機側面



二. 產品安裝

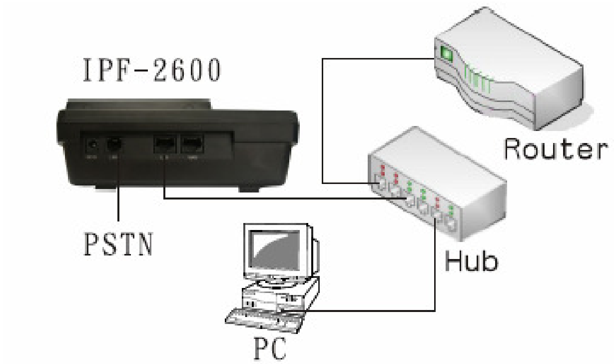
2.1 包裝內容

- IPF-2600 話機一台
- 電話聽筒一隻
- 電源供應器一個
- RJ-45 網路線一條
- 產品使用說明書一份

2.2 設備連接

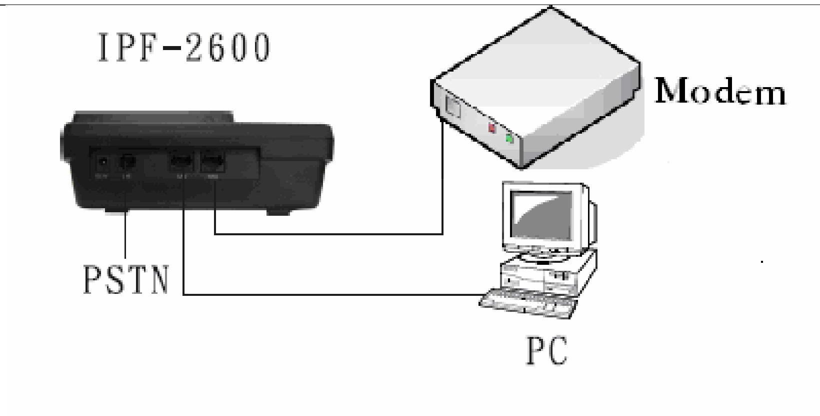
2.2.1. 有 IP 分享器（路由器）的接法：

當你的使用環境有 IP 分享器時,將 RJ-45 網路線一端接入 IP 分享器（或 IP 分享器下集線/交換器）的 LAN 介面,另一端接入 IPF-2600 的 WAN 介面,IPF-2600 接入 LAN 介面連接網路交換設備或 PC；也可以直接將 IPF-2600 的 LAN 介面接入 IP 分享器。如下圖所示：



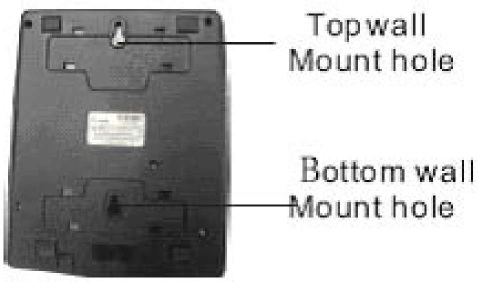
2.2.2. 沒有 IP 分享器的接法

當使用環境沒有 IP 分享器,可以將 IPF-2600 安裝在 ADSL Modem 和 PC（或集線/交換器之間）充當 ADSL 撥號設備。連接方法即 IPF-2600 的 WAN 介面接入 ADSL Modem 的網路介面,LAN 介面連接 PC 或網路交換設備。連接示意圖如下：

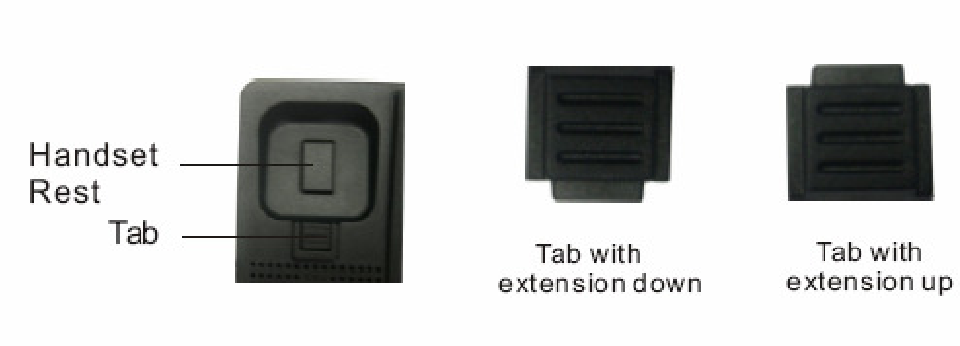


2.2.3. 牆面安裝

IPF2600 可以安裝在牆上,電話主機背部有兩個掛牆孔：



將電話的掛牆孔掛在牆壁的掛鈎上,使用者可以容易地將電話安裝在牆面上

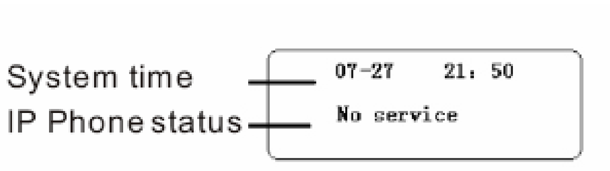


電話安裝好後,使用者需要把掛機孔下的掛牆鈕拔出來放聽筒

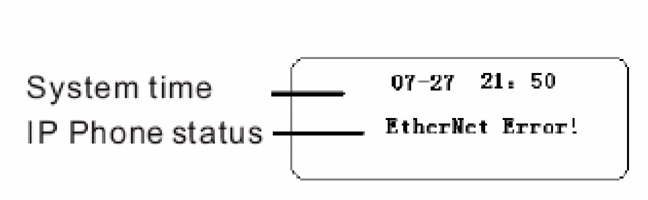
2.3 設備接入狀態顯示

當設備接連接後,會在 LCD 上顯示

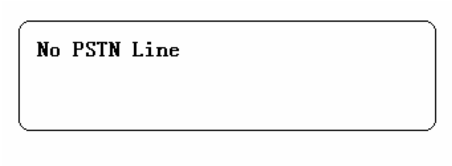
1．此狀態表示話機未成功註冊平臺。



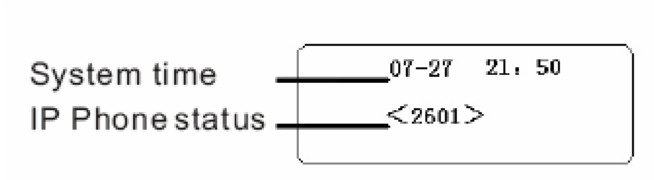
2．此狀態表示話機網路連接錯誤,请用户检查网络线路连接情况。



3．當出現拿起聽筒或按下免持鍵出現以下情況表示 PSTN 線路故障,請檢查電話線路。



4．此狀態表示話機狀態正常,能正常使用。

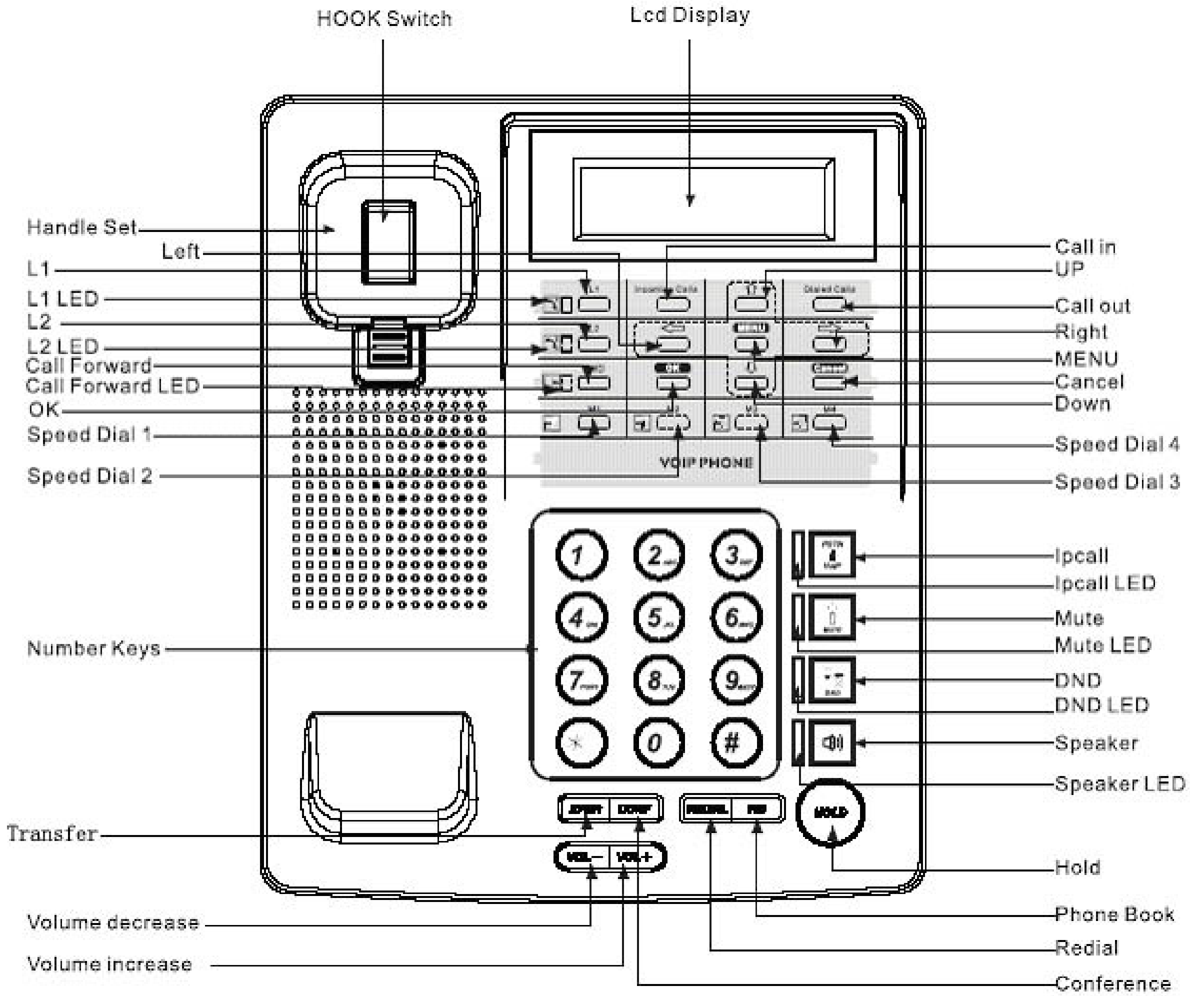


2.4 安全標準

IPF2600 遵循各種安全標準,包括 FCC/CE。該產品的電源供應器遵循 UL 標準,該電話只能使用包裝內提供的電源供應器,因使用其他電源供應器引起的產品損壞,不屬於製造商的保證範圍。

三. 基本操作

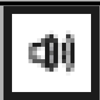
3. 1 話機按鍵示意圖



3. 2 話機按鍵功能說明

按鍵分類	按鍵名	功能/操作
單選鍵	MENU	進入“MENU”模式,按下後出現功能設定功能表,進行相關設定。
	↑	向上選擇,向上移動遊標。
	↓	向下選擇,向下移動游標。
	←	向左選擇,向左移動游標,刪除輸入的字元。
	→	向右選擇,向右移動游標。
	OK	確認設定值。
	Cancel	取消輸入的設定值,輸入號時刪除遊標前字元。

來電記錄查詢鍵	In coming Calls	查詢記錄,用【↑】/【↓】鍵上下選擇通話記錄,按下免持鍵或拿起話筒會自動撥出此筆記錄。
去電記錄查詢鍵	Dialed Calls	查詢記錄,用【↑】/【↓】鍵上下選擇通話記錄,按下免持鍵或 拿起話筒會自動撥出此筆記錄。
來電轉接功能鍵	FWD	開啓與關閉來電轉接功能
線路切換鍵	L1－L2	撥號時按下任意一個鍵,表示用此通路進行通話,如果在按下另一鍵會保留上一通電話,撥打下一通電話。如果已經接聽一路電話,此時再有來電時,指示燈會閃爍,再按下此鍵會保留上一通電話,接通此路電話。
	L1-L2 相關指示燈	顏色爲橙色,來電時閃爍,接通的一路恆亮,Hold 狀態下爲恆亮。
預撥號碼鍵	M1－M4	按下此鍵將撥出相應的預設速撥號碼,此功能要完成必須要先在 Speed Dial 中設定相應的速撥鍵,在 PSTN 優先模式下，在號碼前加“-”撥出時會走 IP 線路,不加則會走 PSTN 線路,IP 優先模式下只支援 IP 路線快速撥號。
數字鍵 （在進入設定模式後連續快速的按鍵會輸入相應字元）	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”
	*	“*”、“.”、“:”、“@”
	#	特殊撥號命令碼,在 IP 模式下,輸入完號碼按此鍵後開始撥號

功能鍵	PSTN VoIP	1．在 PSTN 優先模式下： 按下此鍵會進入 IP 模式,可以撥打 IP 電話。（當進入 IP 模式後橙色 LED 燈會長亮,此鍵只能單向切換即 PSTN to IP）。只要有一次轉換而且該路處於通話狀態,即使切換到另一路爲 PSTN 電話該功能指示燈會恆亮，如果在話機待機狀態切換到 IP 模式該指示燈亮。 2．在 IP 優先模式下： 按下此鍵會進入 PSTN 模式可以撥打 PSTN 電話。（當進入 PSTN 模式後橙色 LED 燈會恆亮,此鍵只能單向切換,即 IP to PSTN）。只要有一次轉換而且該路處於通話狀態,該功能指示燈會恆亮即使切換到另一路爲 IP 電話,如果在話機待機狀態切換到 PSTN 模式該指示燈亮。
	MUTE	MIC 靜音鍵,在通話時按下此鍵 MIC 會進入 MIC 靜音狀態同時橙色 LED 燈會亮起。
	X DND	來電拒接模式鍵,按下此鍵話機進入來電拒接模式,所有 IP 電話將無法撥入,同時橙色 LED 燈會亮起（但 PSTN 電話仍可撥入）。
		免持通話鍵,按下此鍵話機進入免持通話狀態。同時橙色 LED 燈會亮起。
	XFER	來電轉接鍵,按下此鍵後,輸入號碼,此通電話會被轉接至輸入的號碼上。支援 IP－IP－IP,PSTN－PSTN－PSTN
	CONF	三方会议功能键。,在接通兩路電話情況下按下此鍵後使用者進入三方會議模式。
	REDIAL	重撥鍵,按下此鍵會自動撥出最新一筆使用者撥出的電話號碼。
	P.B	電話薄功能鍵,按下此鍵可以對電話薄進行查詢功能。
	Vol+/-	音量調整鍵。
	Hold	通話中可以保留當前一通通話。

3. 3 撥打和接聽電話說明

3.3.1 免持、話筒和耳機模式

話筒模式可切換到免持模式，同樣免持模式可以切換到話筒。當耳機插入耳機插孔只能切換免持或者耳機模式。

3.3.2 多路來電

通常 IPF-2600 支援 2 路來電；當前正在通話的一路電話爲“通話中”，對應 LED 指示燈會恆亮。

3.3.3 建立通話

建立通話有三種方式：

1. 通過舉起話筒/按下免持按鍵,或者按下 L1 鍵/L2 鍵(按下後相應的 LED 指示燈會恆亮)並輸入電話號碼按

＃鍵即可。
2. 通過舉起話筒/按下免持按鍵或者按下 L1 鍵/L2 鍵(按下後相應的 LED 指示燈會恆亮)然後直接重撥鍵即可重撥上次去電的號碼。
3. 通過按下 M1~M4 任意一個快速撥號鍵,此時 L1 鍵和 Speaker 鍵相應的 LED 指示燈會恆亮)即可撥打事先設定好的號碼。
注意：
- 一旦按下重撥鍵上次撥的號碼就會顯示在 LCD 上,同時撥出該號碼。
- 如果撥號後不按＃鍵，根據設定的自動撥出時間（秒）後電話會自動撥出號碼。
  例：如果設定的自動撥出時間為 5 秒,那就在撥號後不按＃鍵,5 秒後電話會自動撥出號碼。

3.3.4 使用 IP 位址建立通話
不需使用 SIP 代理直接利用 IP 位址進行來電是一種特殊的來電方式需要滿足以下條件之一：
- 雙方的電話都需要有合法的 IP 位址
- 雙方的電話在同一個內部網域內(LAN)
- 雙方的電話利用合法或非法 IP,通過分享器連接起來

用 IP 直接撥號：
首先舉起話筒(或按免持鍵),然後輸入目的地的 IP 位址(例 192.168.1.1)，接著輸入＃號即可通話；或舉起話筒(或按免持)輸入目的地的 IP 位址(例 192.168.1.1)後，等待設定的自動撥出時間（秒）所需時間後電話會自動撥出號碼。

3.3.5 接聽來電
IPF-2600 呈現出兩種狀態：
1. 當接收到第一路來電時，電話除了播放出預先設定好的鈴聲外 L1 相應的 LED 會不停地閃爍。此時舉起話筒或按免持鍵就能聽到對方的聲音，L1 相應 LED 會亮。
2. 如果第一路電話通話中,當接收到第二路時 L2 相應 LED 會不停閃爍。這時 IPF-2600 可接通第二路電話,同時第一路將進入來電保留狀態。

3.3.6 來電保留
來電保留有兩種狀態：
1.通話時,按下 HOLD 鍵將會暫停和對方的通話，可通過按相應的"LINE"鍵來恢復通話。
2.使用者同樣可以直接按下另一路可用的 LINE 鍵暫停當前的通話來撥打或接聽其他來電。

3.3.7 來電等待與通話切換
IPF-2600 支援 2 路通話。因此使用者可以按下 L2 保留當前通話並自動切換到第二線路來撥打或接聽電話。或者按下 HOLD 鍵保留當前通話並接通第二路來電。

3.3.8 來電轉移
IPF-2600 支援兩種轉移方式：直接轉移(Blind Transfer)和間接轉移(Attend Transfer)
1. 直接轉移：當有一路通話時，使用者按下 "HOLD"鍵後會保留通話並自動切換到第二線路。再按下 "XFER" 鍵後輸入第三方號碼並按 "＃"鍵送出號碼。這樣 IPF-2600 退出通話而通話被無條件地轉接到第三方。
2. 間接轉移:；當有一路通話時，使用者按下 "XFER"鍵後會保留通話,並自動切換到第二線路。輸入第三方號碼再按 "＃"鍵送出號碼，使用者可以先與第三方通話暫時讓第二方處於等待狀態，當 IPF-2600 掛斷與第

三方的通話,此時第二方與第三方通話被無條件地接通。
3. 來電轉移僅在純 IP 模式下使用。

3.3.9 來電轉接（CALL FORWARD）
IPF2600 支援三種轉接方式：
- All Forward 無條件轉接（FWD 指示燈恆亮,LCD 顯示 AF）
- Busy Forward 遇忙轉接（FWD 指示燈恆亮,LCD 顯示 BF）
- No Answer Forward 無應答轉接（FWD 指示燈恆亮,LCD 顯示 NF）

注意：
Busy Forward 遇忙轉接只能在 IP 模式下有作用

3.3.10 靜音與刪除
在通話時按下"MUTE"鍵會使通話靜音。能聽到對方的聲音但對方聽不到你的聲音。再次按下"MUTE"鍵會恢復通話,雙方都能聽到對方的聲音。

3.3.11 快速撥號
IPF-2600 有 4 個"快速撥號"鍵（M1~M4），每個鍵可以設定一個號碼，按下 M1~M4 任何一個快速撥號鍵可以直接撥通預設號碼。

注意：
如果第一路已撥電話為 IP 電話，按快速撥號為 PSTN 電話將不能撥出。如果已撥的為 IP 電話只有在 IP 撥號模式下按已設定的號碼為 IP 的快速撥號號碼鍵才有效。

四. 話機按鍵設定

IPF-2600 支援多種設定模式。當話機啓動完成後按下"MENU"鍵進入"MENU"模式，出現設定功能表進行相關設定。如果再按一次"MENU"鍵會退出"MENU"模式，進入待機模式。如果進入"MENU"模式後,不進行操作 10 秒後也會自動退出。

4.1 Phone Book（電話簿）

1・Search	進行號碼查詢以便進行快速撥號和編輯。選擇此項目第一次按下「OK」進入快速查詢模式，用戶可以輸入 Name 或 Number 進行快速查詢；連續兩次按下「OK」鍵進入瀏覽查詢模式。 在此模式下選擇項目後按下「OK」鍵進入編輯模式,對 Name 和 Number 進入編輯。查詢模式中選中號碼後舉起聽筒或按下免持鍵會撥出此號碼。
2・Add Entry	增加 Phone Book 記錄。選擇此項目後,第一次按下「OK」在 Name 輸入要儲存號碼的使用者名稱或代號。按下「OK」儲存並進入 Number 項目輸入號碼然後按下「OK」鍵儲存並退出電話簿編輯。 ＊如果輸入的是 IP 線路號碼需要在號碼前加「-」號以區別 PSTN 號碼。
3・Speed Dial	設定快速撥號號碼。選擇此項目後,按下「OK」進入 Name 欄輸入速撥號碼的使用者名稱或代號按下「OK」鍵進入 Number 項目輸入速撥號碼後按下「OK」輸入速撥代碼完成該筆速撥號碼的輸入。話機支援九筆速撥號碼,代碼可以從 0－9 之間選擇，如果該代碼以前已儲存過號碼,將會被新號碼覆蓋。

	＊1－4 欄對應的是話機按鍵 M1－M4 快速撥號按鍵。如果輸入的是 IP 號碼剛需在號碼前加「-」以區別 PSTN 號碼。使用時直接按下待撥號碼相應的鍵即可撥出。IP 優先的不支援 PSTN 快速撥號。 ＊在其他欄位輸入的號碼則只能從 IP 線路撥出。無需添加「-」號。使用時,先將話機切換至 IP 模式,然後按下待撥號碼相應的序號以「#」與結束,即可撥出該號碼。
4．Erase All	清除所有 Phone Book 記錄,。再按下"OK"鍵會出現清除選項,可以選擇 "Yes", "NO"可選擇是否刪除 Phone Book 的所有記錄。

4.2 Call History（通話紀錄）

1．Incoming calls	來電記錄,進入功能表後,再次按下"OK"鍵會出現儲存或清除選項,選擇"save"或"erase"進行相關操作。
2．Dialed numbers	已撥記錄,進入功能表後,再次按下"OK"鍵會出現儲存或清除選項,選擇"save"或"erase"進行相關操作。
3．Erase record	清除記錄,進入功能表後,再次按下"OK"鍵會出現清除選項："Incoming"刪除來電記錄、"Dialed"刪除已撥電話記錄、"All"刪除來電和已撥的所有記錄。

4.3 Phone Setting（話機設定）

4.3.1 Call Forward

1．All Forward	無條件轉接。進入功能表後在"Activation"選項中選擇"Enabled"或"Disable"確認開啓與關閉該功能。在"Number"選項中輸入轉接號碼。 ＊ 可以在號碼前加"-"設定 PSTN 線路。
2．Busy Forward	遇忙轉接。進入功能表後,在"Activation"選項中選擇"Enabled"或"Disable"確認開啓與關閉該功能。如果選擇 Enable 須在"Number"選項中輸入轉接號碼 ＊ 只能設定轉接至 IP 線路的號碼,目前只支援轉接到 IP 線路。
3．No Answer FWD	無應答轉接。進入功能表後,在"Activation"選項中選擇"Enabled"或"Disable"確認開啓與關閉該功能。在"Number"選項中輸入轉接號碼。 ＊ 可以在號碼前加"-"設定 PSTN 線路。
4．Ring TimeOut FWD	震鈴逾時轉接,進入功能表後,要求先輸入震鈴次數,然後輸入轉接號碼

4.3.2 Do Not Disturb

1．Always	拒接所有來電。如果已設定來電拒絕話機將聽不到來電鈴聲,而撥號端則聽到忙線音。對 PSTN 撥入的電話無效。
2．By Period	部分時間拒接來電。如果已設定來電拒絕在該時間機將聽不到來電鈴聲而撥號端則聽到忙線音。
3．Period Time	設定拒接來電持續的起止時間。如果已設定話在該時間機將聽不到來電鈴聲而撥號端則聽到忙線音。

4.3.3 Alarm setting

1.Activation	設定是否啟用鬧鐘功能
2.Alarm time	設定鬧鐘時間,當該時間到時話機會自動震鈴。

4.3.4 Date/Time Setting

1．Date&Time	手動設定時間
2．SNTP Setting	SNTP：開啓時間伺服器,自動更新時間。 Primary SNTP：主要時間伺服器地址。 Secondary SNTP：備用時間伺服器地址。 Time zone：時區設定，設置負時區時可按"*" 號鍵。 Adjustment Time：自動對時週期設定。

4.3.5 Volume&Gain

1．Handset Volume	話筒輸出音量調整,最大値 15。
2．Speaker Volume	免持輸出音量調整,最大値 12。
3．Handset Gain	話筒輸入音量調整,最大値 15。
4．Speaker Gain	免持輸入音量調整,最大値 15。

4.3.6 Ringer

1．Ringer Volume	鈴聲音量調整,最大値 10。
2．Ringer Type	鈴聲模式設定,有 4 種模式。

4.3.7 Auto Dial

1．Auto Dial	此項設定在號碼輸入完成後,沒有按＃號確認撥出時,話機自動撥出的等待時間,設定範圍 3-9 秒。

4.4 Network（網路設定）

4.4.1 WAN Setup

1．IP Type	IP 模式選擇： ● Fixed IP client：指定 IP 地址。 ● DHCP client：DHCP 自動獲取 IP。 ● PPPoE client：PPPoE 撥號模式。
2．Fixed IP setting	指定 IP 地址： ● IP Address：主機 IP 位址。 ● Subnet mask：子網路遮罩。 ● Default Gateway：預設閘道地址。
3．PPPoE Setting	PPPoE 撥號設定： ● User name：輸入 PPPoE 帳號。, ● Password：PPPoE 密碼。

4.4.2 LAN Setup

1．Bridge	開啓橋接器功能。

2．NAT	開啓 IP 分享器功能。

4.4.3 DNS

1．Primary DNS	主要 DNS 設定。
2．Secondary DNS	備用 DNS 設定。

4.4.4 VLAN

1．Activation	設定 VLAN 是否開啓。
2．VID	預設爲 136。設定 VLAN Server 提供之虛擬區域網路識別碼 (Virtual LAN ID, 簡稱 VLAN ID 或 VID)。
3．Priority	預設爲 0。設定使用者優先權(user priority),優先權範圍爲 0~7。
4．CFI	預設爲 1。設定一個位元之「制式格式指示」Canonical Format Indicator (CFI)。 CFI = 1 表示標籤頭中包含 RIF 欄位,而且 RIF 中的 NCFI 旗標值決定訊框資料中所攜帶的 MAC 位址是「制式格式」(Canonical Format)或「非制式格式」(Non-Canonical Format)。 CFI=0 則表示此標籤標頭不含 RIF 欄位,而且訊框中所包裝的 MAC 位址是「制式格式」。

4.4.5 Status

可以查看話機的 IP 位址及 MAC 號碼。

4.5 SIP setting（SIP 設定）

如果需要對該項進行設定請先進 Administrator 子目錄 System authent 項輸入密碼（預設值爲 test）。
如果密碼不正確 LCD 顯示“incorrect”密碼正確 LCD 顯示“Correct”。當密碼被確認後再進入 SIP setting.。

4.5.1 Service domain

First realm	註冊第一個 SIP 帳號 Activation：選擇是否註冊本帳號. Display name：LCD 顯示名稱. User name：用戶名, 使用的號碼, 在此設定本話機之電話號碼. Register name：註冊帳號名. Register password：註冊密碼 Domain server：網域功能變數名稱. Proxy server：要登陸的語音閘道器的 IP 位元位址或者網域功能變數名稱. Outbound proxy：語音代理伺服器。
Second realm	注册第二个 SIP 帐号, 设置项功能同上。
Third realm	注册第三个 SIP 帐号, 设置项功能同上。

4.5.2 Codec

Codec type	设置语音编码格式。

VAD	設置是否開啓動態語音檢查功能。

4.5.3 RTP setting

Outband DTMF	Outband DTMF 啓動、停止設置。
Duplicate RTP	語音封包重傳次數設置。

4.6 NAT transversal（NAT 設定）

STUN setting	設定是否使用 STUN 協定。 STUN 服務器位址。

4.7 Adminisrtator（管理設定）

1. Auto config	話機自動控制。 ● Config mode：選擇控制模式（Disable/TFTP/FTP）。 ● TFTP server：TFTP 控制伺服器位址。 ● FTP server：FTP 控制伺服器位址。 ● FTP Login Name：FTP 服務器登錄名。 ● FTP Password：FTP 服務器登錄密碼 。
2. Upgrade System	話機軟體升級，允許話機從網路上下載軟體並自動更新更新模式需在 WEB 頁面設定。 ● Upgrade Now：當有新的軟體時可立即更新。 ● Status：查看話機更新排程時間表。 ● Reset Time：重置話機更新排程時間表。
3. Default Setting	話機預設值設定。 ● Load default：載入預設值, ● Abort：放棄操作。
4. System authent	供"root" 用戶輸入密碼登錄話機以具有權限進入 SIP Setting 功能表項。
4. Version	查看話機軟體版本。
5. Watch dog	話機監控
6. Restart	重新啟動話機

五. 網頁設定

進入話機 Network/Status 項取得話機 IP 位址。然後在要與之連接的電腦上設定相同網段的 IP 位址。設定 IPF-2600 的 IP 與電腦處於同一網段（**IP 位址前三段一樣,最後一段主機位址不同即可**）然後在電腦的網路瀏覽器的位址欄輸入"http：//IPF-2600 IP：9999 " 進入網頁設定登錄介面輸入 username 和 password。話機出廠預設的三個用戶是"root" 、"system"和"user",密碼均爲"test"（用戶可以自行更改）。輸入完密碼後,按 Login 便可以進入 IPF-2600 的網頁設定畫面。

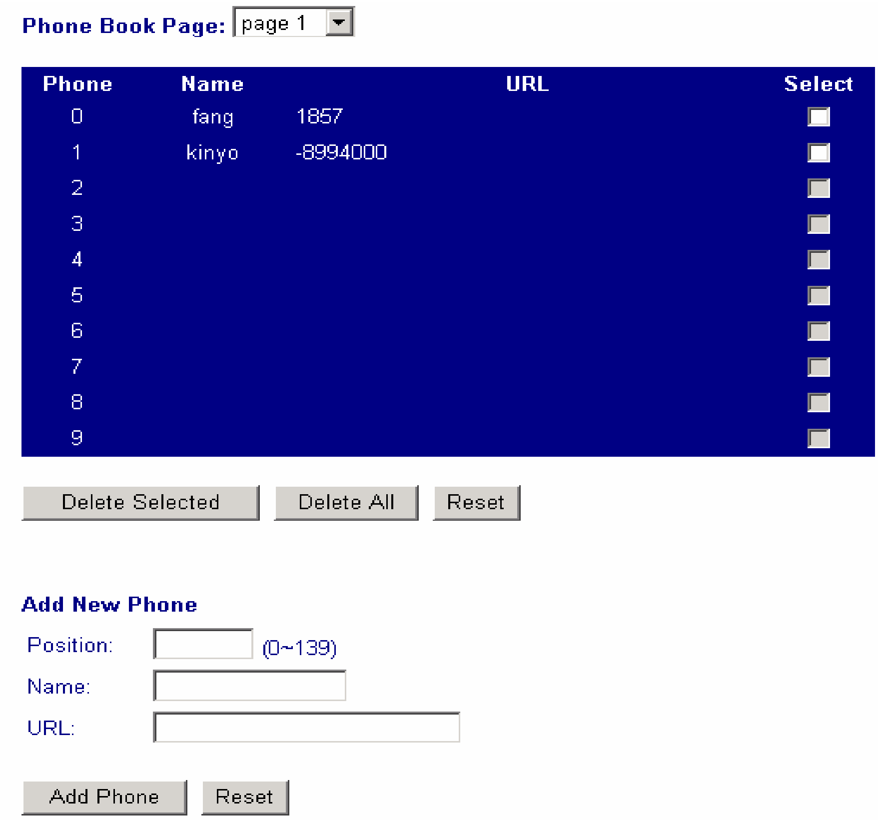
5.1 Information（話機資訊）

系統基本資訊,可以看到軟體的版本資訊。

Model Name　　產品型號
Firmware Version　　Firmware 軟體的版本
Codec Version　　Codec 軟體的編譯日期

5.2 Phone Setting（話機設定）

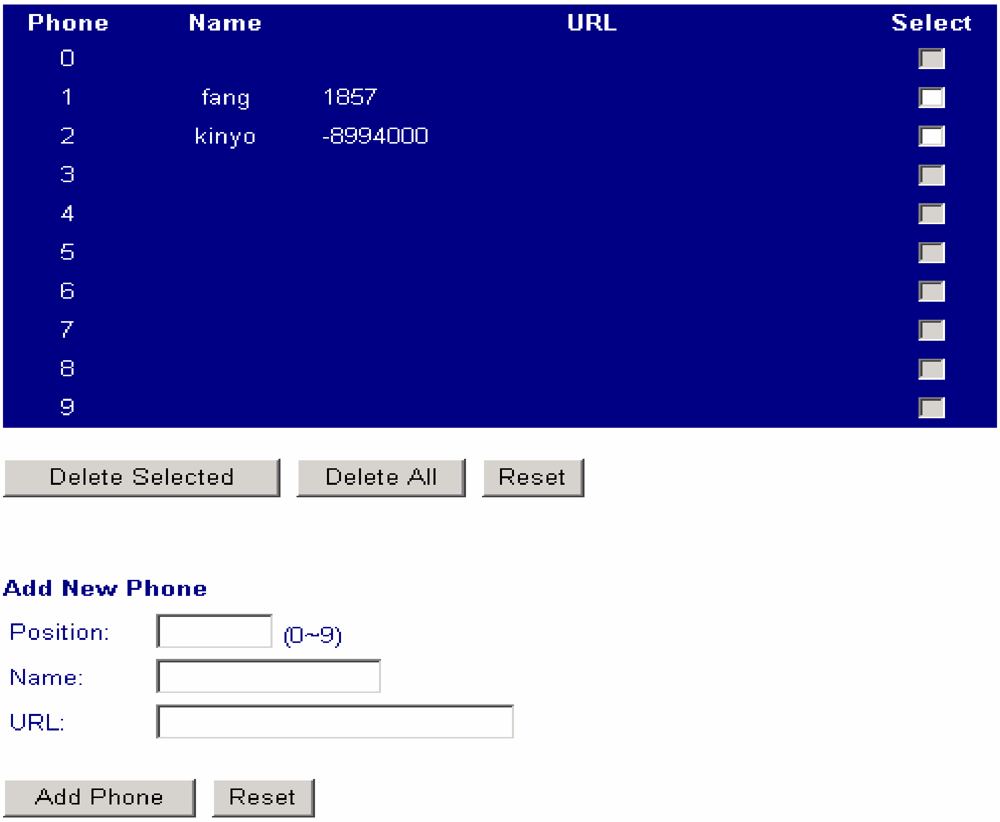
5.2.1 Phone Book（電話簿）



PHONE Book	
欄位名稱	說明
Delete Selected	刪除按鈕。刪除所選擇項內容。
Delete All	刪除按鈕。刪除所有內容。
Position	在填入欄裏填寫所對應的序號,隨後填入的號碼將寫到對應欄位裏。
Name	輸入撥號代碼。
URL	輸入外撥號碼或 IP 位置資料。如果輸入的是 IP 號碼剛需在號碼前加 “-”，

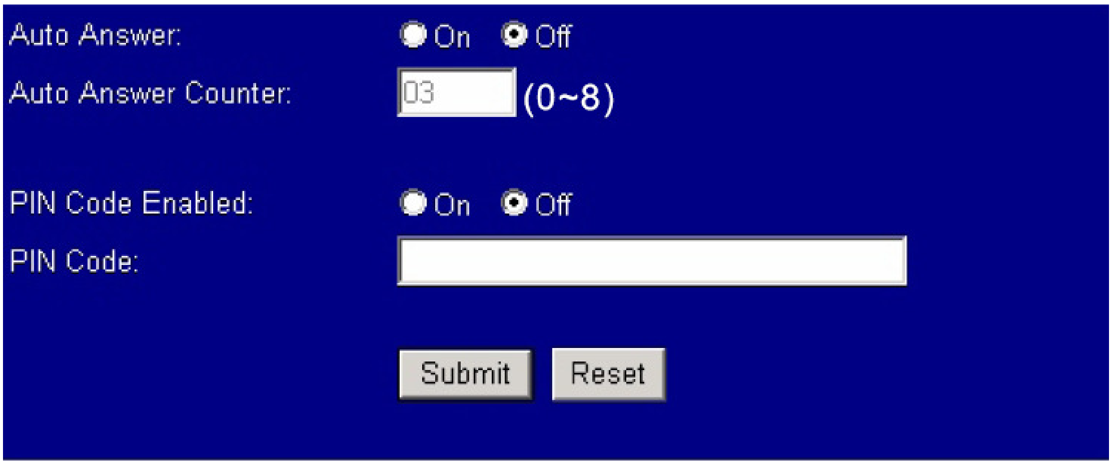
	以區別 PSTN 號碼。
Add Phone	新增此筆資料。(可填入 140 個電話號碼)
Reset	清除已設定之資料。

5.2.2 Speed Dial（速撥）



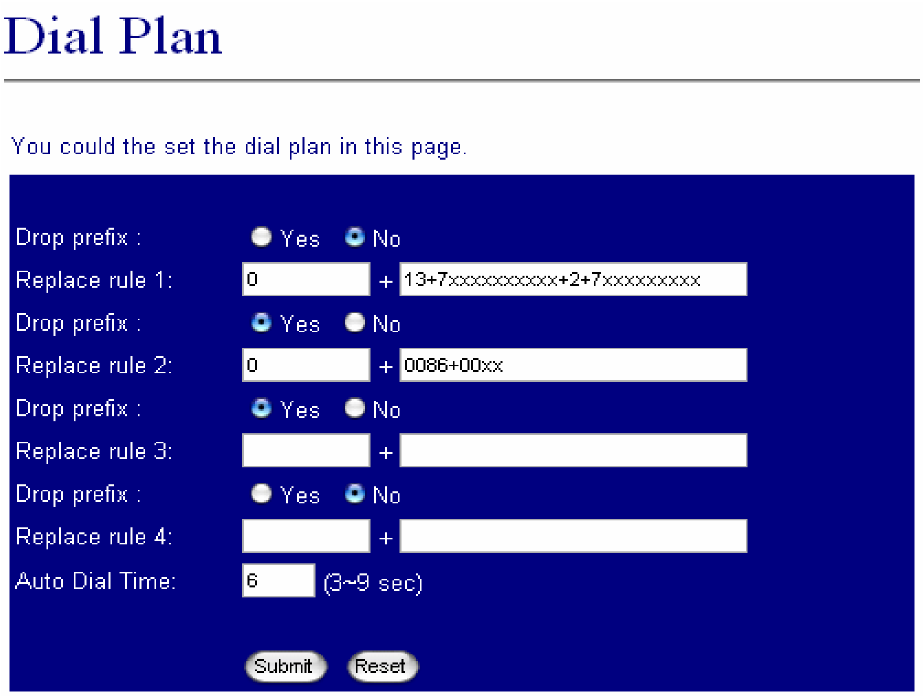
Speed Dial	
欄位名稱	說明
Delete Selected	刪除按鈕。刪除所選擇項內容。
Delete All	刪除按鈕。刪除所有內容。
Position	在填入欄裏填寫所對應的數字,隨後填入的號碼將寫到對應欄位裏,欄位編號依次從 0—9,對應於話機鍵盤上的 0—9 數字鍵,欄位元編號即爲該欄的 URL 號碼的快速撥號鍵。其中 1、2、3、4 欄位對應話機 M1－M4 預撥鍵。
Name	輸入撥號代碼。
URL	輸入外撥號碼或 IP 位置資料。
Add Phone	填寫完後按下此鍵,名字和號碼將填入到對應的欄位中。(可填入 10 個快速撥號號碼)
Reset	清除已設定之資料。

5.2.3 Auto Answer（自動應答）



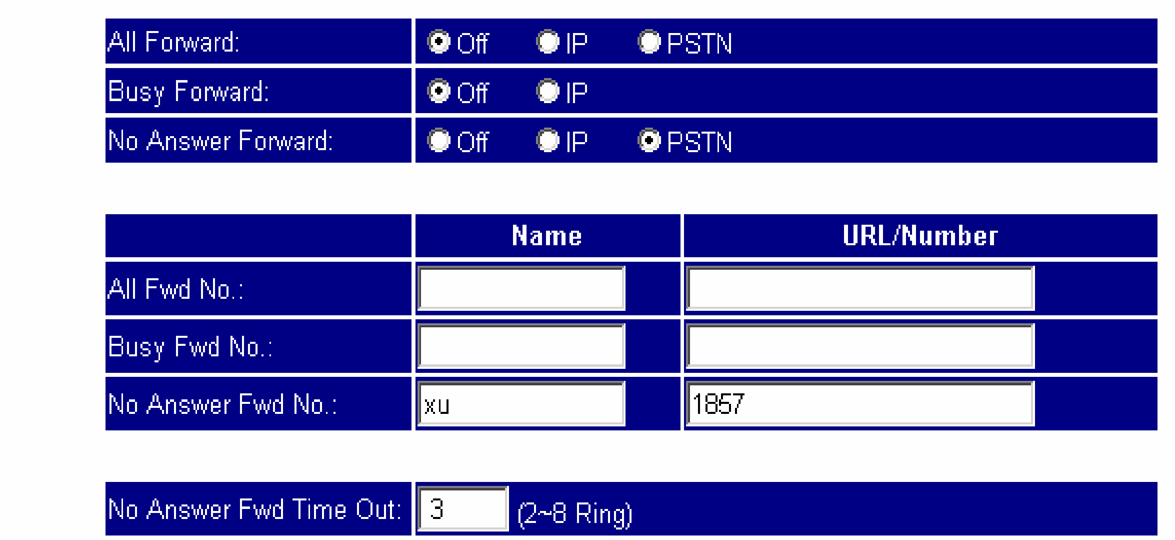
Auto Answer	
欄位名稱	說明
Auto Answer	選擇使用自動語音應答。選擇開啓自動應答功能後便使用上下車功能
Auto Answer Counter	自動應答時間。設定來電響鈴 X 聲後（X 爲 0～8）,沒人接聽則進入自動應答。若設爲 0 則話機不會先響鈴,直接聽到上下車的二次撥號音 ＊不能與 No Answer Fwd Time Out 設定相同,否則以 call forward 優先。
PIN Code Enabled	設定上、下車是否使用密碼認證
PIN Code	設定上、下車密碼。

5.2.4 Dial plan setting（撥號規則）



Dial Plan Setting	
欄位名稱	說明
Drop Prefix	預設爲 No (加碼)。當設定爲 Yes (減碼)時,則執行減碼的動作。 設定 4 組資料 - No (加碼): 當 rule 符合時,則直接加碼。可輸入 7 位數。 - Yo (減碼): 當 rule 符合時,則減掉符合之代碼再加碼。可輸入 39 位數。
Replace rule1	設定撥號規範。 +: 或。 xxx: 指定碼長。
Auto Dial Time	預設爲 5(秒)。等待多久後自動執行撥號。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.2.5 Forward Setting（轉接設定）



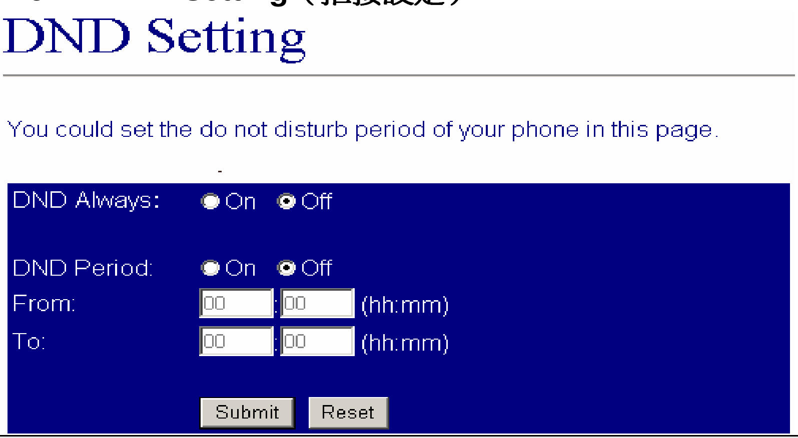
Forward Setting	
欄位名稱	說明
All Forward	全部轉接。打進來的所有電話都轉撥到指定號碼,使用者可以選擇轉接到 IP 或是 PSTN 線路。當 All Forward 功能啓動,FWD 按鍵指示燈亮起,LCD 上顯示 AF　XXXX。AF 表示 All Forward；XXXX 表示转接的目的地號碼。
Busy Forward	遇忙轉接。電話忙線時,啓用轉撥功能。即當電話打進來後,若電話占線,則將轉撥到指定號碼。 ＊無 PSTN 線路來電功能。
No Answer Forward	無應答轉接。電話無人應答時,啓用轉撥功能。即當電話打進來後,在震鈴逾時後,則將轉撥到指定號碼。使用者可以選擇轉接到 IP 或是 PSTN 線路。
All Fwd No.	全部轉接號碼。填入開啓全部轉接後對應的轉撥姓名和電話號碼。
Busy Fwd No	遇忙轉接號碼。填入開啓遇忙轉接後對應的轉撥姓名和電話號碼。
No Answer Fwd No	無應答轉接號碼。填入對應的無人接聽電話時轉撥的姓名和電話號碼
No Answer Fwd Time Out	設定無應答響鈴時長,超過此響鈴次數,話機會進入無應答轉接狀態。

	＊不能與 Auto Answer Counter 設定相同,否則以 call forward 優先。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.2.6 Call Waiting（來電等待）

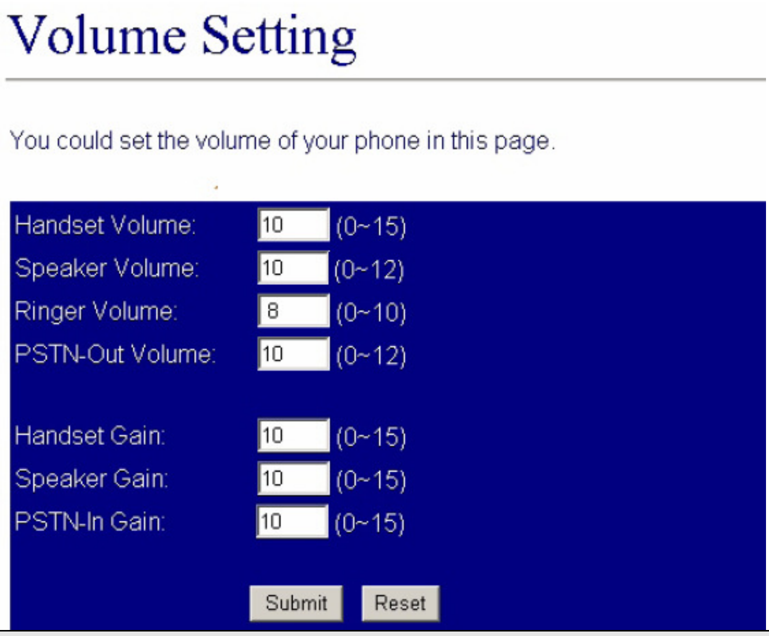
Call Waiting	
欄位名稱	說明
Call Waiting	來電等待。設定當前話機使用來電等待功能。本機和 A 通話時, B 撥入,本機會每隔 3 秒聽到 "嘟,嘟" 兩聲,此時按下 "HOLD"鍵來保持 A 並接聽 B 的電話,並且可以按 "hold "鍵在 A 和 B 之間切換通話。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.2.7 DND Setting（拒接設定）



DND Setting	
欄位名稱	說明
DND Always	拒接所有撥入電話,選擇 on 後所有打入的電話將聽到忙線音,off 為不使用。 ＊僅對 IP 線路撥入有效。
DND Period	某時段拒接設定。阻止某段時間打入的電話。From 中填入開始時間,To 爲截至時間。 ＊僅對 IP 線路撥入有效。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.2.8 Volume Setting（音量設定）



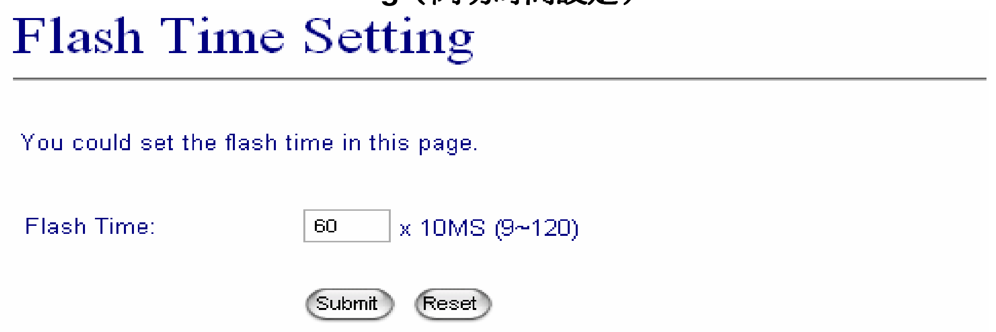
Volume Setting	
欄位名稱	說明
Handset Volume	設定聽筒輸出音量大小
Speaker Volume	設定免持輸出音量大小
Ringer Volume	設定震鈴音量大小
PSTN-Out Volume	設定 PSTN 線路輸出的音量大小
Handset Gain	設定免持輸入音量大小
Speaker Gain	設定免持喇叭輸入音量大小
PSTN-In Gain	設定 PSTN 線路輸入的音量大小。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.2.9 Melody Setting（鈴聲設定）

Ringer Setting	
欄位名稱	說明
Ringer	是否開啓用戶設定鈴聲　On 開啓,Off 爲關閉。選擇 off 情況下默認爲第一種

	鈴聲。
Ringer Type	選擇鈴聲模式,一共有四種模式可供選擇。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.2.10 Flash Time Setting（閃切時間設定）

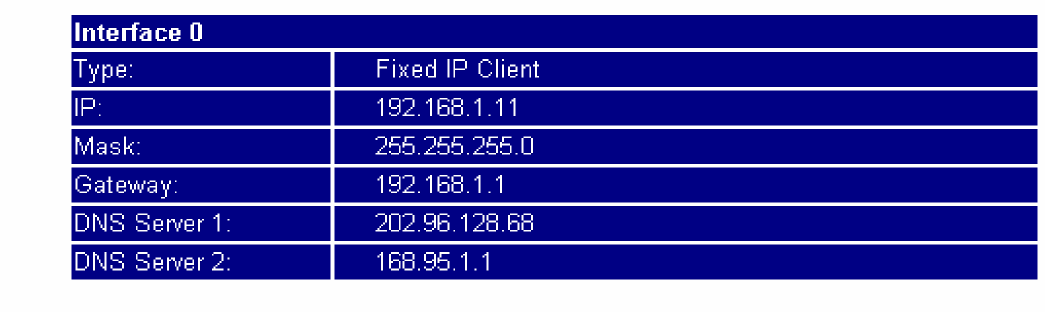


Flash Time Setting	
欄位名稱	說明
Flash Time	如使用插簧鍵或 Flash(閃切)來做切換鍵,設定插簧鍵按下到彈起的間隔時間。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.3 Networking（網路設定）

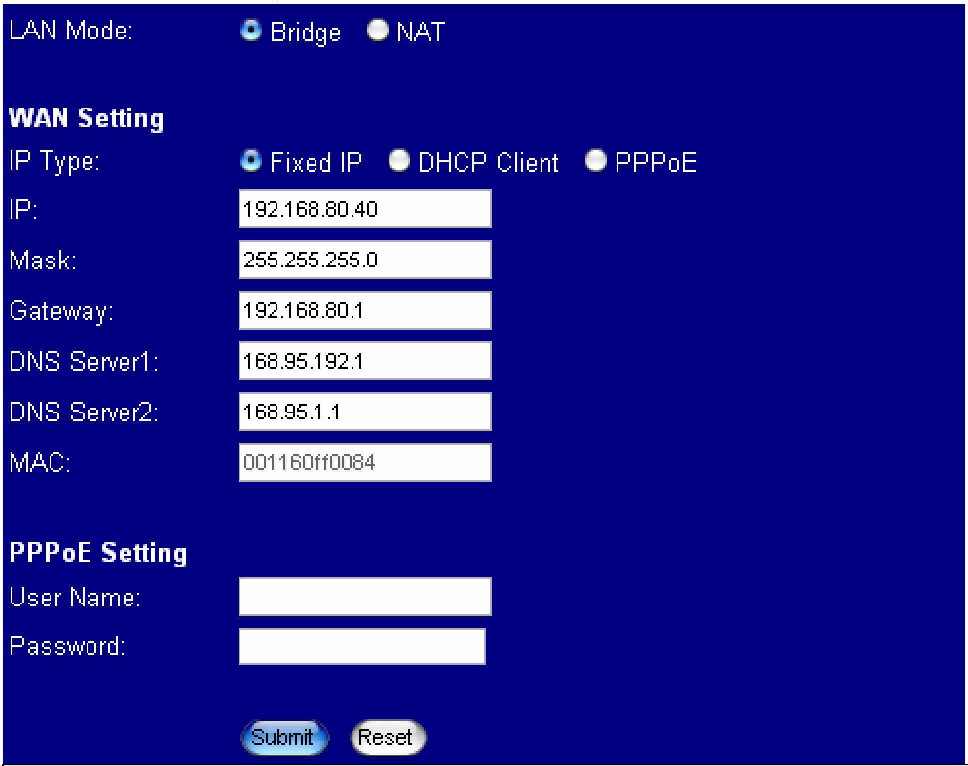
5.3.1 Status（狀態）

IPF-2600 設備基本資訊顯示



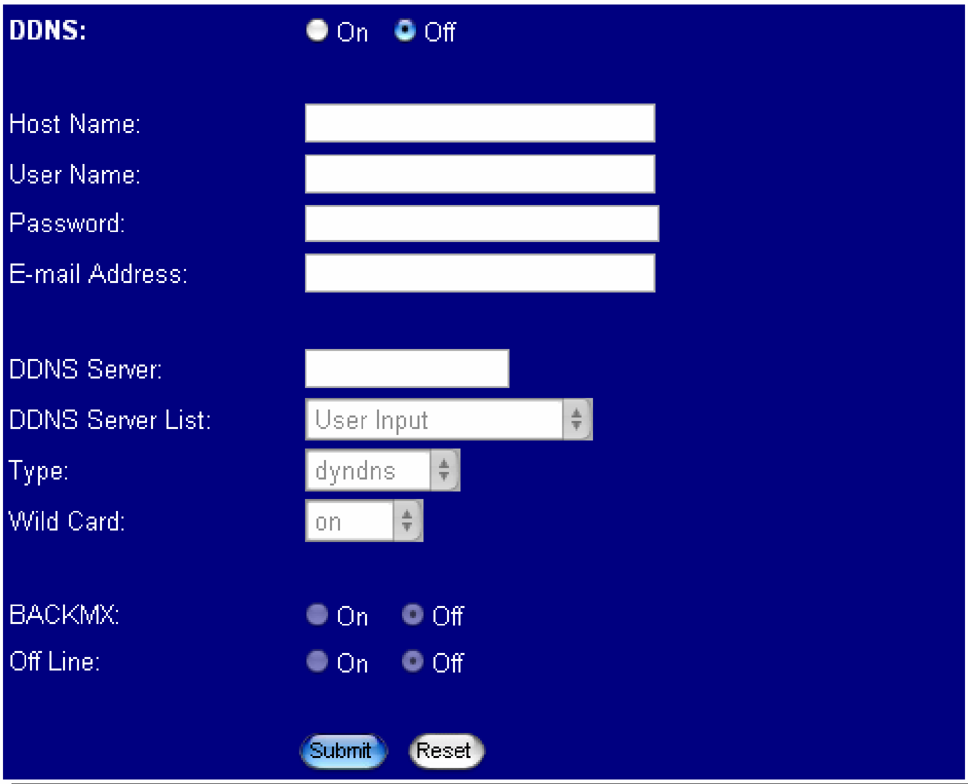
Status	
欄位名稱	說明
Type	顯示 IP 地址類型
IP	顯示 IP 地址
Mask	顯示子網路遮罩
Gateway	顯示已設定的預設閘道的 IP 地址
DNS Server 1	顯示已設定的主要的網路域名地址
DNS Server 2	顯示已設定的備用的網路域名地址

5.3.2 Network Setting（網路設定）



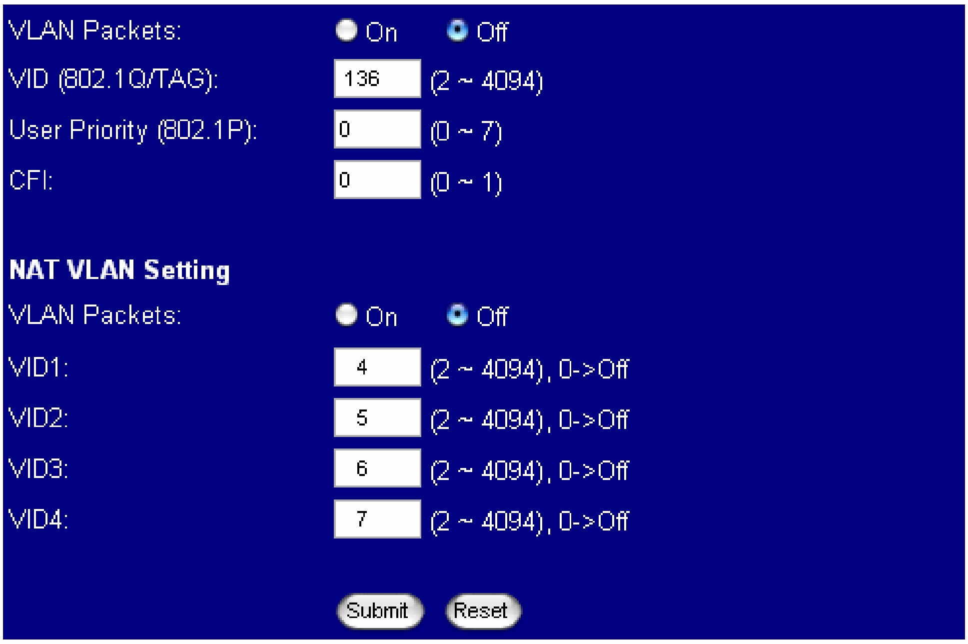
Network Setting	
欄位名稱	說明
LAN Mode	Bridge:開啓橋接器功能。 NAT:開啓路由器功能
Ip Type	IP 類型。依實際網路環境選擇適當的網路型態: ● Fixed IP：選擇後需將 Local IP、Subnet Mask、Gateway IP、DNS 等欄位填入相關資料。 ● DHCP Client：選取後會由 DHCP server 取得相關 IP 資料。 ● PPPoE：撥撥式 ADSL 用戶選擇。
IP	設定 WAN 的 IP 位址。
Mask	設定 WAN 的子網路遮罩位址
Gateway	設定 WAN 的預設閘道的 IP 地址
DNS Server 1	設定 WAN 的主要 DNS 的 IP 位址
DNS Server 2	設定 WAN 的備用 DNS 的 IP 位址
MAC	本機網路卡號
PPPoE Setting	
User Name	選取 pppoe 時,設定用戶的 ADSL 帳戶名。
Password	選取 pppoe 時,設定用戶的 ADSL 帳戶的密碼。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.3.3 DDNS Setting



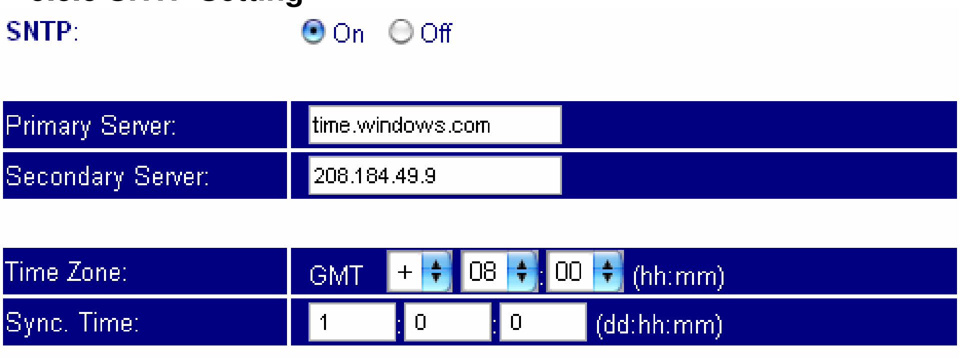
DDNS Setting	
欄位名稱	說明
DDNS	設定使用 DDNS。
Host Name	DDNS 伺服器主機名。
User Name	DDNS 伺服器註冊帳戶。
Password:	帳戶密碼。
E-mail Address	電子郵件位址。
DDNS Server	DDNS 的 IP 位址或域名。
DDNS Server List	DDNS 服務提供者列表。
Type	選擇 DDNS 工作模式。
Wild card	選擇是否啓用 Wild Card 功能
Back MX	選擇是否啓用 Back MX 功能
Off Line	選擇是否啓用 DDNS 的 Off Line 功能
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.3.4 VLAN Setting



VLAN Setting	
欄位名稱	說明
VLAN Packets	預設為 Off(不執行)。當設定為 On (執行)時,則啓動接收 VALN Packets 功能。
VID	預設為 136。設定 VLAN Server 提供之虛擬區域網路識別碼 (Virtual LAN ID, 簡稱 VLAN ID 或 VID)。
User Priority	預設為 0。設定使用者優先權(user priority),優先權範圍為 0~7。
CFI	預設為 1。設定一個位元之「制式格式指示」Canonical Format Indicator (CFI)。CFI = 1 表示標籤頭中包含 RIF 欄位,而且 RIF 中的 NCFI 旗標值決定訊框資料中所攜帶的 MAC 位址是「制式格式」(Canonical Format)或「非制式格式」(Non-Canonical Format)。 CFI=0 則表示此標籤標頭不含 RIF 欄位,而且訊框中所包裝的 MAC 位址是「制式格式」。
NAT VLAN Setting	
VLAN Packets	預設為 Off(不執行)。當設定為 On (執行)時,則啓動接收 VALN Packets 功能。
VID1 ~ 4	預設為 136。設定 VLAN Server 提供之虛擬區域網路識別碼 (Virtual LAN ID, 簡稱 VLAN ID 或 VID)。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

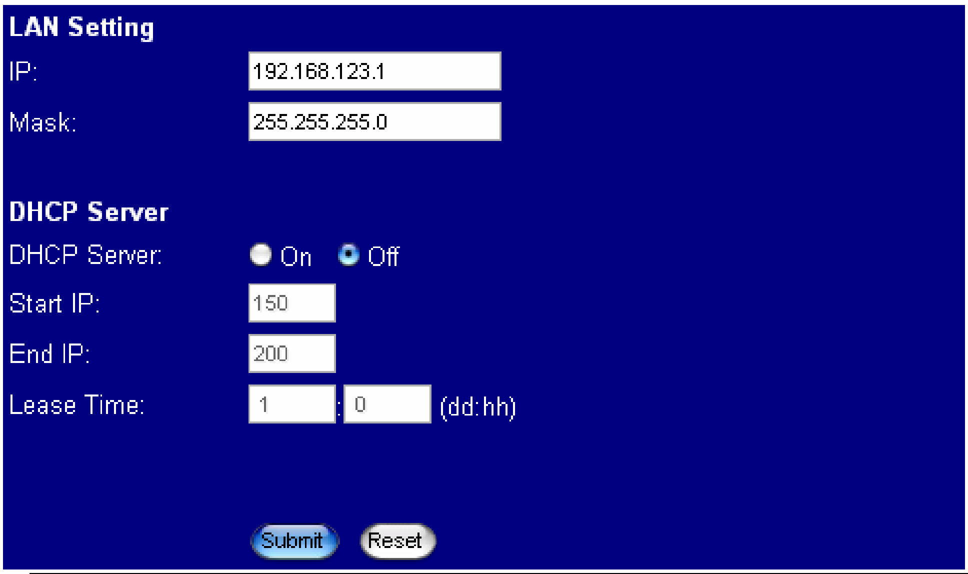
5.3.5 SNTP Setting



SNTP Setting	
欄位名稱	說明
SNTP	設定使用時間伺服器,on 爲使用,off 不使用。
Primary Server	主伺服器。填寫主時間伺服器地址。
Secondary Server:	次伺服器。填寫副時間伺服器地址。
Time Zone	時區。填入當地時區。
Sync. Time	自動對時週期設定。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5. 4 NAT Router（NAT 路由器）

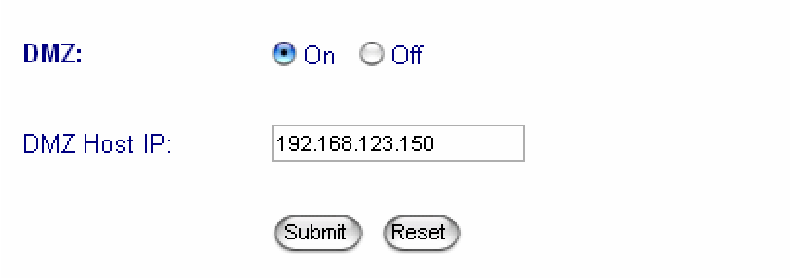
5.4.1 LAN Setting



LAN Setting	
欄位名稱	說明
IP	在開啓 NAT 功能後,LAN 埠將自己成爲另一個虛擬網域。
Mask	設定 LAN 端子網路遮罩。
DHCP Server	開啓 DHCP 伺服器服務。

Start IP	設定 DHCP 分配的起始 IP 其數據不能大於 255。
End IP	設定 DHCP 分配的終止 IP 其數據不能大於 255。
Lease Time	租約時間。由 DHCP 伺服器提供,指定 IP 位址可以使用的時間。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

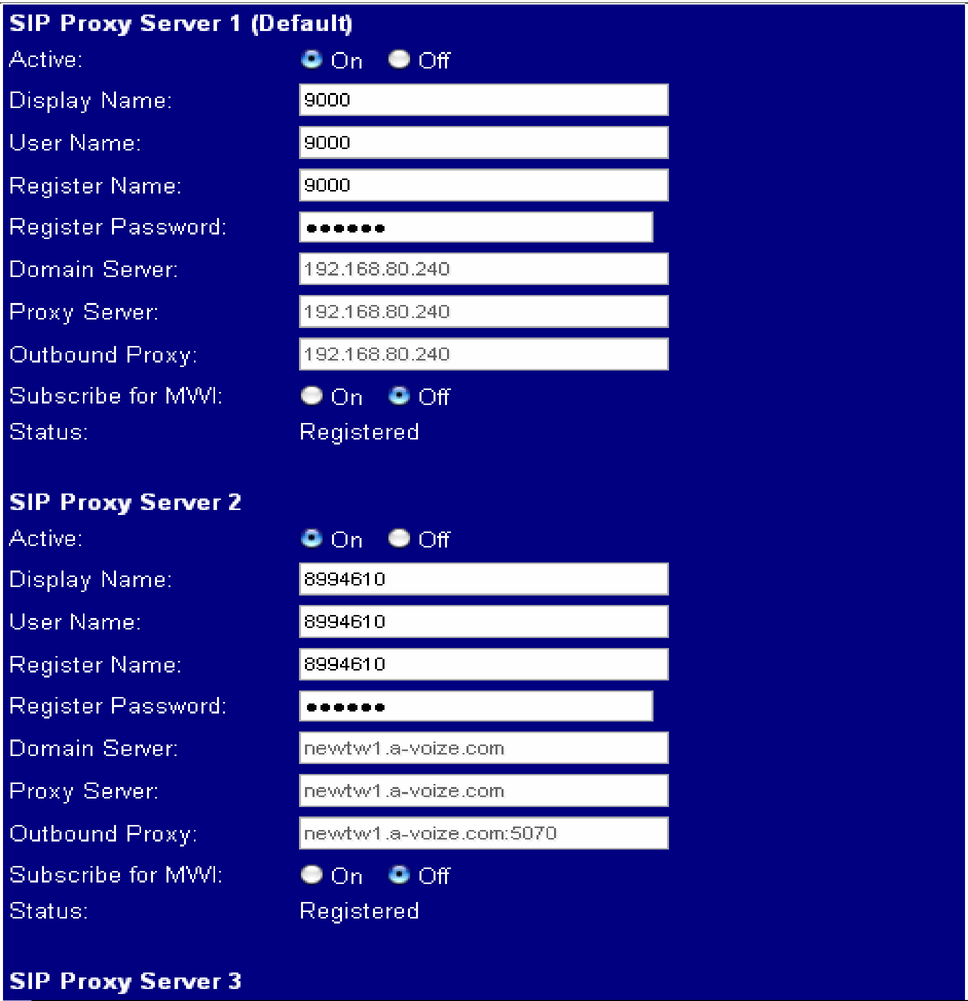
5.4.2 DMZ Setting



DMZ Setting	
欄位名稱	說明
DMZ	預設爲 Off(不執行)。當設定爲 On (執行)時,所有的封包(除了 SIP 相關)都會往該 IP 位置傳送。
DMZ Host	輸入一個指定的 IP Address。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.5 SIP setting（SIP 設定）

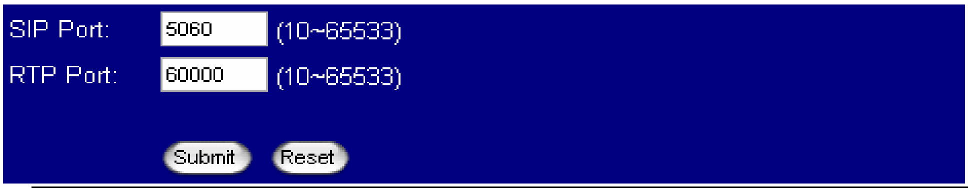
5.5.1 Service Domain（服務平台設定）



Service Domain	
欄位名稱	說明
Active	帳號選擇。 ● 此頁中可填入 3 個不同帳號,並且同時註冊上,當選擇 on 時爲使用當前欄,再在當前欄填入對應註冊帳號。 ● 撥出的號碼以註冊的第一個號碼爲主(以選取 on 選項的第一個選項欄爲首選). 第一個帳號註冊不上將自動跳轉到第二個註冊上的帳號,撥入到註冊上的任一號碼都可以接聽到電話。
Display Name	LCD 顯示名稱
User Name	用戶名,使用的號碼,在此設定本話機之電話號碼。
Register Name	註冊帳號名。
Register Password	註冊密碼。
Domain Server	網域功能變數名稱。
Proxy Server	要登錄的語音閘道器的 IP 位地址或者網域功能變數名稱。如果特殊埠號則加 :埠號。
Outbound Proxy	語音代理伺服器。如果特殊埠號則加 :埠號。
Subscribe of MWI	預設爲 Off(不執行)。當設定爲 On (執行)時,週期性的傳送”來話訊息留言偵

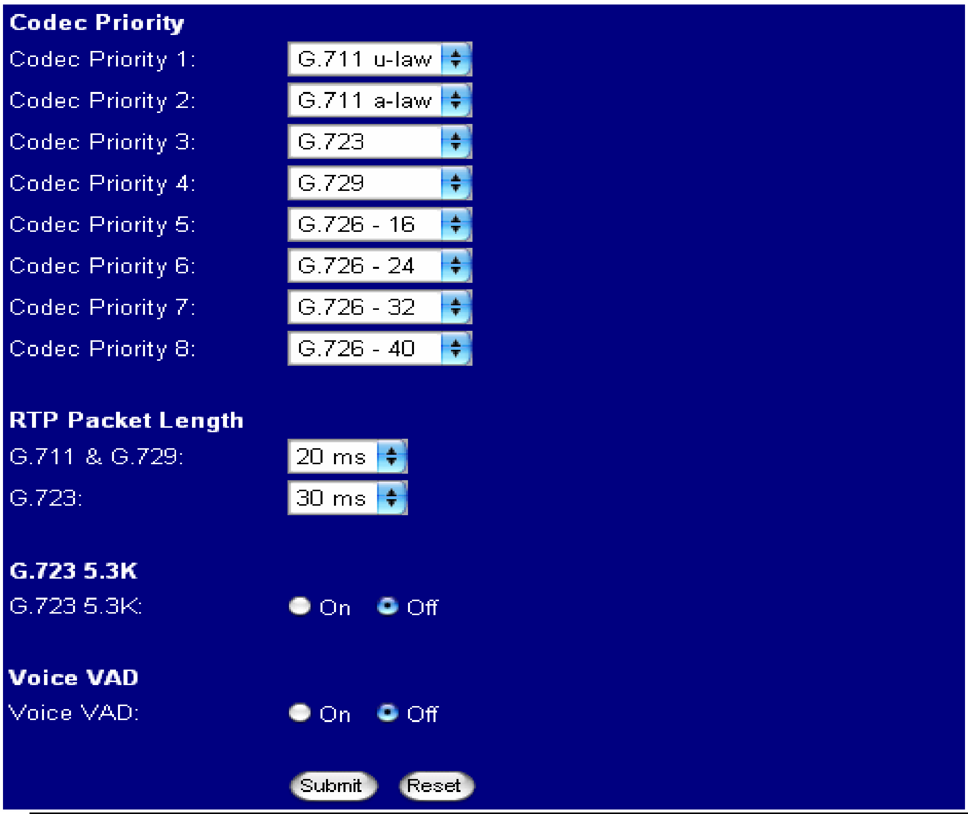
	測”的動作。
Status	註冊狀態,**Registered 為註冊成功,Not Registered 為註冊失敗。**
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.5.2 Port Setting（埠號設定）



Port Setting	
欄位名稱	說明
SIP Port	SIP 協定註冊埠,預設值為 5060。
RTP Port	RTP 埠號指語音傳送與接收的埠號。埠號可在 1024-65535 的範圍內選用,但一定為偶數。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.5.3 Codec Setting（語音編解碼設定）

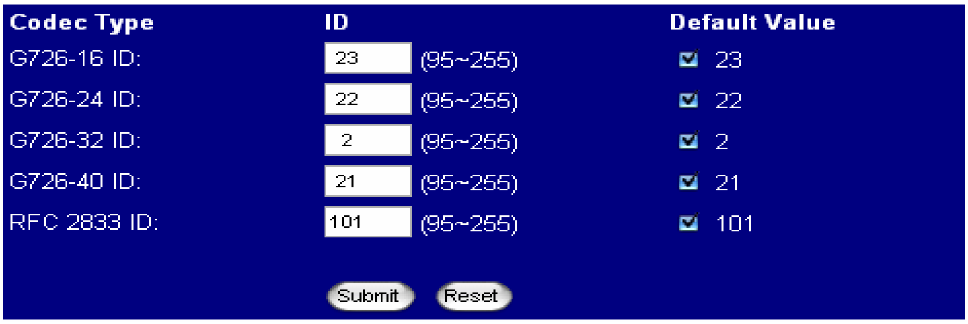


Codec Setting	
欄位名稱	說明

Codec Priority	編解碼選擇。選擇編解碼的優先權（以 Codec Priority1 爲首選）,及語音編解碼格式。
G.723 5.3K:	使用 g.723 語音編解碼時,設定 on 爲使用 5.3K/S 之 Codec。
Voice VAD	使用動態語音檢測。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.5. 4 Codec ID

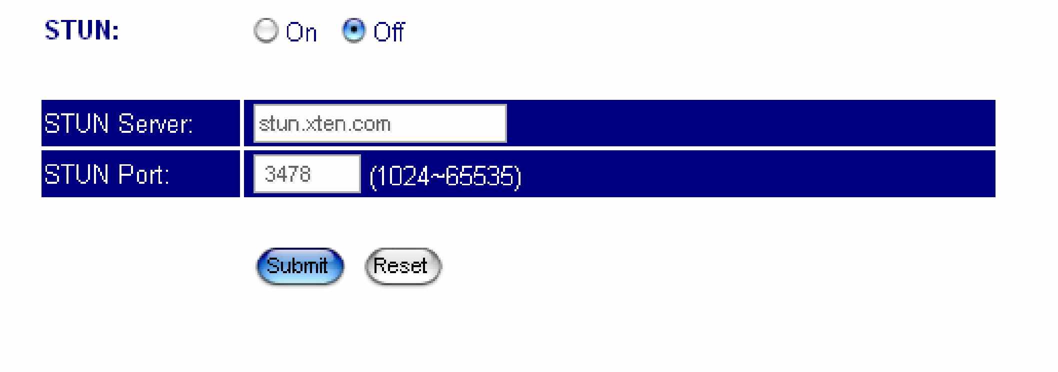
G726 各編解碼的設定及 RFC 2833 的設定,建議使用預設值,除非註冊平臺有其他要求。



5.5.5 DTMF Setting

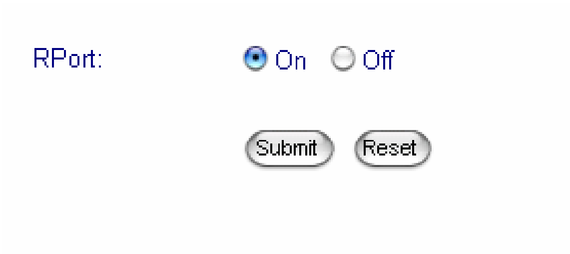
DTMF Setting	
欄位名稱	說明
2833	使用 rfc2833 方式傳送 DTMF 訊號。
Inband DTMF	使用 Inband DTMF 方式傳送 DTMF 訊號。
Send DTMF SIP Info	使用 SIP Info 方式。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.5.6 STUN Setting（STUN 設定）



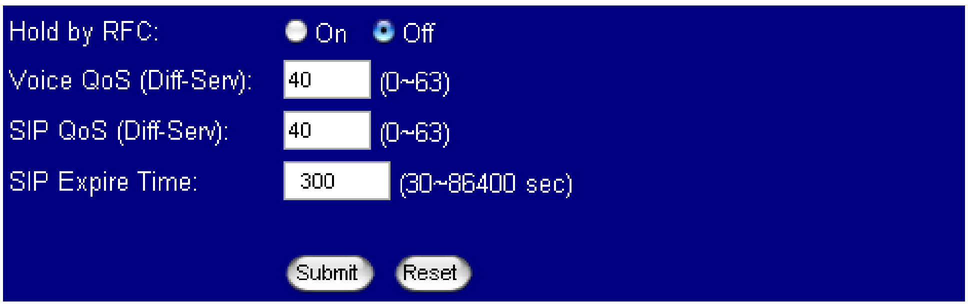
STUN Setting	
欄位名稱	說明
STUN	設定是否使用 STUN 協定。
STUN Server	STUN 服務器位址。
STUN Port	STUN 埠號。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.5.7 RPort Setting



RPort Setting	
欄位名稱	說明
On/Off	設定是否使用 RPort 協定。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.5.8 Other Setting（其他設定）

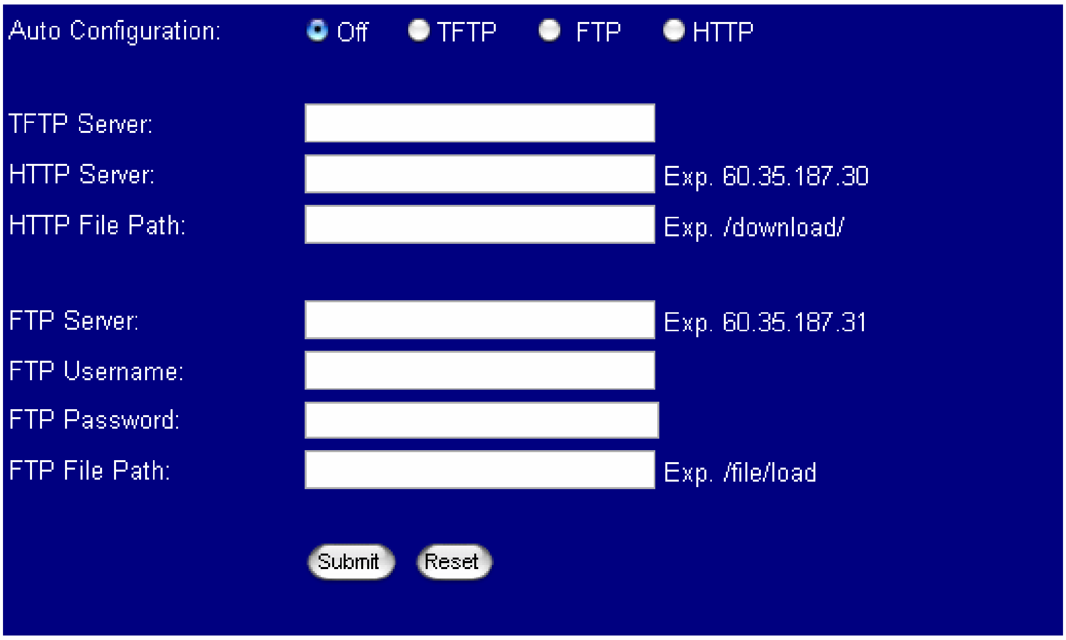


Ohter Setting	
欄位名稱	說明

Hold by RFC	RFC 選擇使用。此資訊由服務商提供。
Voice QoS	語音服務品質。
Sip QoS	SIP 服務品質。
SIP Expire Time	SIP 包傳送時間。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

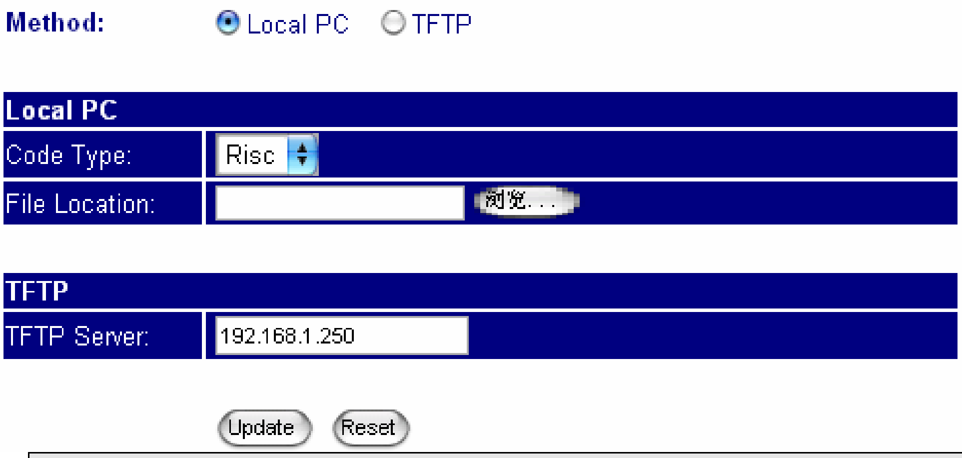
5.6 Others（其他）

5.6.1 Auto Config（自動設定）



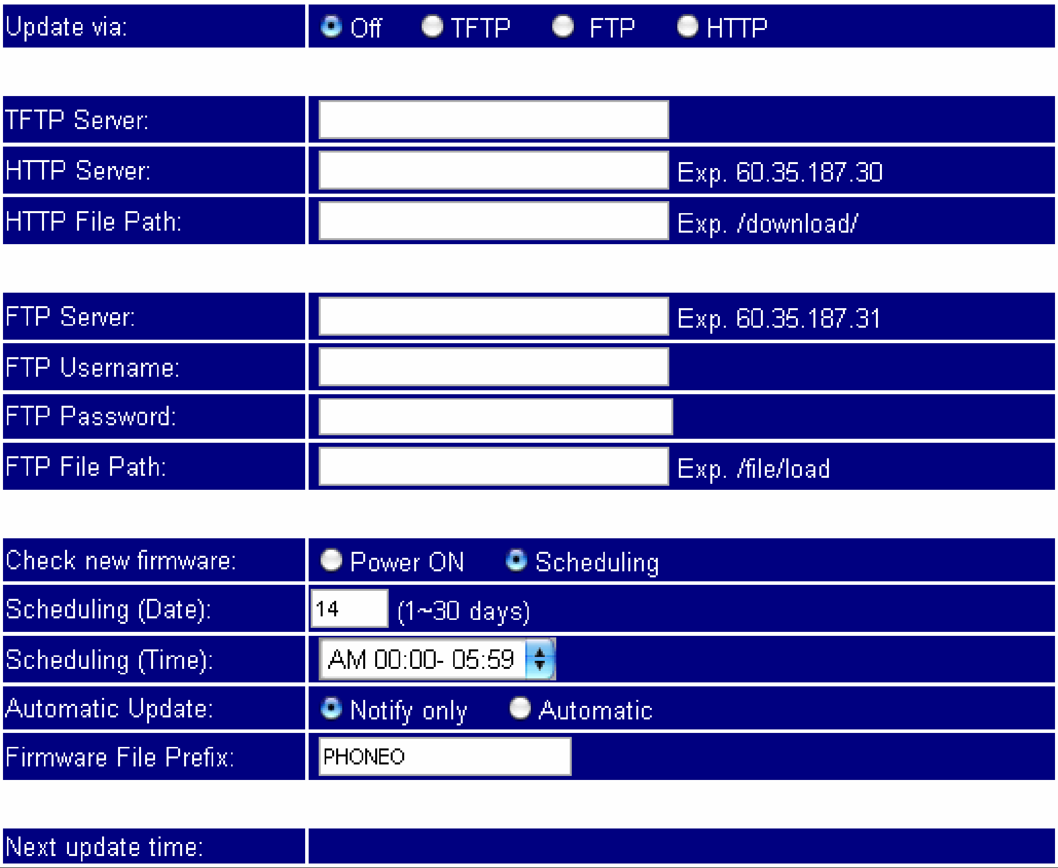
Auto Config	
欄位名稱	說明
Auto Configuration	預設為 Off(不執行)。自動更新環境設定方式,提供 TFTP,FTP 及 HTTP 等三種方式。
TFTP Server	設定 TFTP Server 位置,可以輸入 IP 或 Domain Name 資料。
HTTP Server	設定 HTTP Server 位置,可以輸入 IP 或 Domain Name 資料。
HTTP Path	設定路徑名稱,例如: /123/。
FTP Server	設定 FTP Server 位置,可以輸入 IP 或 Domain Name 資料。
FTP Username	登入 FTP Server 之使用者(User name)帳號資料。
FTP Password	登入 FTP Server 之密碼(Password)帳號資料。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.6.2 Firmware upgrade（韌體升級）



Ohter Setting	
欄位名稱	說明
Method	選擇使用 Local PC 方式或 TFTP 升級方式更新軟體。
Code Type	選擇更新程式編碼類型,可選擇 Risc 和 DSP 類型。
File Location	選擇升級程式位址。
TFTP Server	填寫升級的 tftp server 的 IP 位址。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.6.3 Auto Update（自動升級）



Auto Update	
欄位名稱	說明
Update via	預設爲 Off(不執行更新)。版本自動更新方式,提供 TFTP,FTP 及 HTTP 等三種方式。
Check new Firmware	預設爲 Scheduling(依照時間排程)。檢查是否有新的版本提供版本,提供 Power ON (每次開機)或 Scheduling(按照排程)。 - Power On(開機檢查): Power on + Scheduling,即每次開機時及依照時間排程檢查是否有新的版本可供更新。 - Scheduling: 依照時間排程檢查是否有新的版本可供更新。
Scheduling (Date)	預設爲 14 天。每隔幾天去檢查一次,提供最短爲 1 天,最長爲 30 天。
Scheduling (Time)	預設爲 AM 00:00 – 05:59。每次去檢查的時間區段,詳細的時間爲隨機產生。提供四個區段分別爲 AM 00:00 – 05:59, AM 06:00 – 11:59, AM 12:00 – 17:59, AM 18:00 – 23:59。
Automatic Update	預設爲 Notify only (發送訊息通知)。自動更新的方式,提供 Notify only(發送訊息通知),Automatic(自動執行更新)。 - Notify only: 發訊息通知有新的版本,但不執行自動更新的動作；Phone: LCD 會有提示訊息 “Find new S/W”,用戶可以選擇是否更新爲新的軟體。 - Automatic: 在 Scheduling 已設定的條件下自動執行版本更新的動作,在 POWER ON 時需用戶自己選擇是否更新爲新的軟體。
Firmware File Prefix	預設爲產品型號。檢查符合產品型號之版本資料。

Next update time	下次檢查或更新之日期與時間資料。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

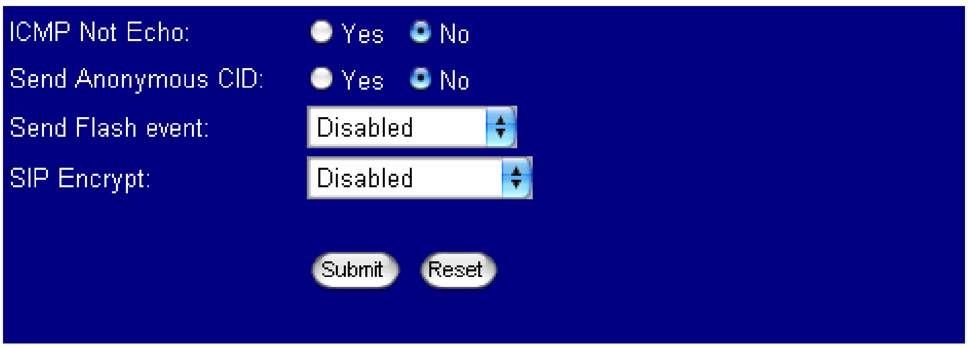
5.6.4 Default Setting（回復出廠設定）

使用此項功能後,所有設定值將會回復出廠預設值

5.6.5 FXO Port Setting（FXO 設定）

依地區電信設備之特性選擇適當之參數。

5.6.6 Advanced Setting（進階設定）

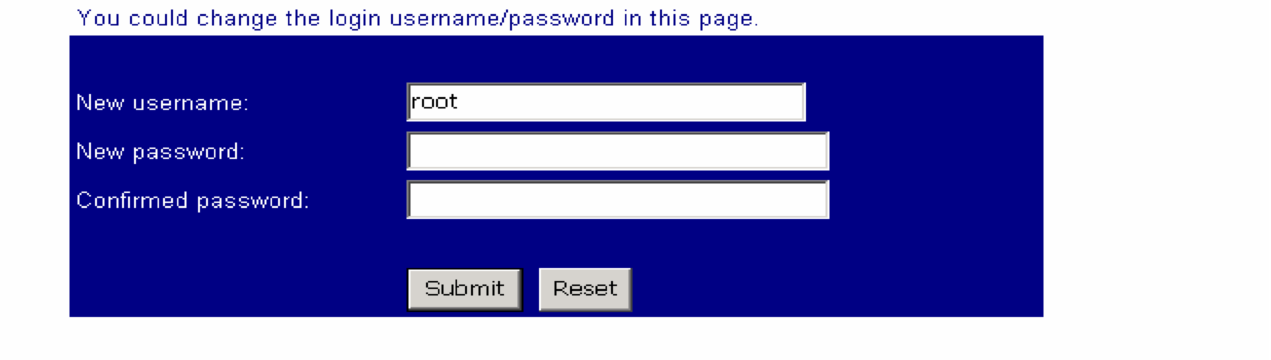


Advanced Setting	
欄位名稱	說明
ICMP Not Echo	啓動不回應 ping 的訊息。預設爲 No (不執行)。
Send Anonymous CID	啓動去電保密功能,不送本機之號碼資料。預設爲 No (不執行)。
Send Flash event	送 flash event 格式,提供 DTMF Event 及 SIP Info 二種格式。預設爲 Disable (不執行)。
SIP Encrypt	SIP 加密方式,提供 INFINET, AVS, WALKERSUN1, WALKERSUN2 四種格式。

	僅提供有此服務之環境使用。預設為 Disable (不執行)。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

5.6.7 System Auth（系統管理）

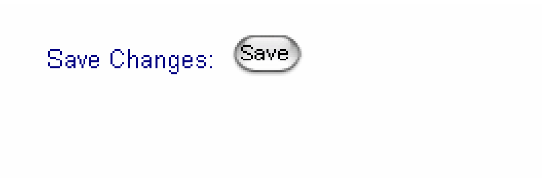
修改用戶密碼



System Auth	
欄位名稱	說明
New username	用戶名。
New password	設定新用戶密碼。
Confirmed password	確認密碼。
Submit	執行儲存變更設定。
Reset	清除已設定之資料。

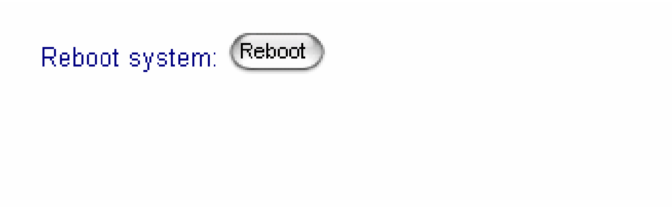
5.7 Save Changes（儲存設定）

儲存所有設定並重新啓動



5.8 Reboot（重新啓動）

不儲存更改的設定並重新啓動



六. 產品規格

語音介面	
來電控制協定	SIP v1 (RFC 2543), v2 (RFC 3261)
語音壓縮	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)
延遲 (點對點)	< 100ms
回音抵消	Packet Loss Compensation, Adaptive Jitter Buffer, VAD (Voice activity detection), CNG (Comfortable noise generator), AEC (Acoustic echo canceller), G.165 (LEC, Line echo canceller), G.168 (EC, Digital network echo canceller)
通話平均流量	5.3K(G.723.1) ~ 64K(G.711) bps
其他支援	In-Band DTMF, Out-of Band DTMF, SIP Info
區域網路介面	
介面規格	10/100Mbps Fast Ethernet
介面連接器	RJ-45 Connector
管理工具	網頁瀏覽器
支援網路協定	Static IP / DHCP Client / PPPoE Client / TFTP Client / HTTP Server / DNS Client / Telnet / SNTP / RTP / RTCP / DDNS
韌體更新	HTTP / TFTP / FTP
來電支援	Call Hold / Call Waiting / Call Forward / Call Transfer / Caller ID / Call Block / 3-way Conference
其他規格	
電源	100 – 240VAC, 50 – 60Hz, 9VDC
耗電量	6W
工作環境	工作溫度：0 ~ 40℃ (32°~ 104°F)
	儲存溫度：-30 ~ 65℃ (-22°~ 149°F)
	相對濕度：10 ~ 95% Non-Condensing
重量	0.75g
認證	CE / FCC
尺寸	219mm *180 mm *75 mm
其他	140 筆資料的電話薄
	NAT 穿透（STUN / uPnP / R-Port）
	網路狀況顯示、韌體更新、無應答轉接、忙線轉接、來電直接轉接、網頁式管理設定介面等等,LED 指示燈。

七.常見問題與排解（Q&A）

1. 怎樣確定 IPF-2600 註冊成功?

答：如果 IPF-2600 帳號註冊成功,IPF-2600 LCD 會正確顯示您註冊的帳號。如果 LCD 顯示幕顯示 No Service 則表示註冊不成功。一般 IPF-2600 在 2 分鐘內能註冊成功。

2 . 接上電源後,話機啟動後 LCD 顯示幕顯示 ETHERNET ERROR,這是為什麼?

答：這時請您檢查連接線路是否有鬆動,是否有連接網路線;如果連接正常,請檢查您的網路是否正常。

3. 如果停電了是否會影響正常打電話?

答：停電後,IPF-2600 將沒有電源供應,這是 IP 撥入或撥出的電話將無法接通。PSTN 路線的電話也無法正常撥打和接聽。

4. 是否支援上下車功能？

答：支援.

上車使用方法:

- 直接從 PSTN 外線撥入,聽到數聲震鈴聲(此時 IPF-2600 亦會跟著震鈴)後會出現第二次撥號音,再轉撥所要撥打的電話。
- 若有設定 PIN CODE 時,在撥通後會聽到數聲震鈴聲(此時 IPF-2600 亦會跟著震鈴).接著數聲急促的嘟嘟聲,此時按 Pincode#後就會聽到第二次撥號音,再轉撥所要撥打的電話。

下車使用方法:

- 撥打 IPF-2600 的 SIP 門號,聽到數聲震鈴聲(此時 IPF-2600 亦會跟著震鈴)後會出現第二次撥號音,再轉撥所要撥打下車的電話。
- 若有設定 PIN CODE 時,在撥打 IPF-2600 的 SIP 門號撥通後會聽到數聲震鈴聲(此時 IPF-2600 亦會跟著震鈴).接著數聲急促的嘟嘟聲,此時按 Pincode#後就會聽到第二次撥號音,再轉撥所要撥打下車的電話。

5.如何使用電話轉接？

答：**(1)三方會談**：

- 與使用者 A 建立通話。
- 通話中按下<HOLD>鍵將 A 通話保留,待聽到撥號音後撥打欲轉接之目標號碼（B）。
- 接通 B 後,告知有來電並詢問接聽意願。
- 如果 B 不願意接聽,或轉接（諮詢）B 失敗,請按下<hold>鍵取消轉接並取回與 A 之通話。
- 如果 B 願意接聽,按下<CONF>鍵,進入三方會談。

(2)來電轉移：

Blind Transfer (直接轉移)

B 來電 A, A 與 B 在通話的過程中,A 執行轉接給第三方,A 先按[Hold]鍵 Hold 住與 B 的通話,再按[Transfer/Flash]鍵接著輸入第三方的號碼,結束加"#"字鍵；即可將電話轉給第三方。

Attendant Transfer(間接轉移)

B 來電 A,A 與 B 在通話的過程中,A 執行轉接給第三方,A 按[Transfer/Flash]鍵接著輸入第三方的號碼,結束加"#"字鍵；則第三方開始振鈴,A 與 C 通話後,A 掛斷電話,則 B 與 C 可以互相通話。

6. 如何正確撥打電話?

答:拿起話筒,聽到撥號音後,即可撥打 PSTN 號碼或者或者按 IP Call 切換按鈕撥打網路電話。

7. 保留(Hold)如何使用?

答：當正在接聽電話時,通過按一下 Hold 或者插簧鍵可以保留對方,再按下一次將恢復通話。

8. 如何使用來電等待(Call Waiting)?

答：當正在通話時,從話機聽筒傳出“嘟嘟”兩聲,表示有插撥進來， 若想接聽插撥， 可以按 <HOLD> 鍵或者插簧鍵,就可以接聽插撥了。

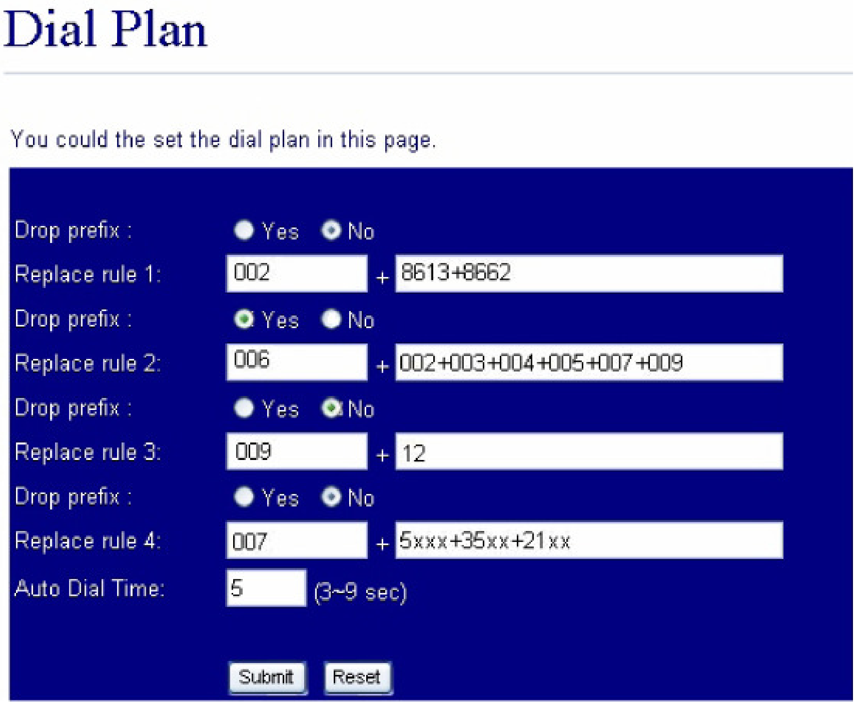
9. 速撥鍵(SPEED DIAL) 要怎麼用?

答：先在 Speed Dial 的網頁裏填入您需要快速撥號的號碼,然後拿起聽筒或者按下免持鍵後,按下對應的快速撥號號碼,接著按下＃號鍵後速撥完畢。(注：只能速撥 IP 線路的電話號碼。)

10. PSTN 來電為什麼不會顯示?

答：需當地電信局支持。

11. Dial plan 設定範例。



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

說明 1：當撥號時有輸入 8613 時,只要符合 8613 開頭的號碼,全部在前面自動加上 002；則實際號碼爲[002+8613+xxx]。

說明 2：當撥號時有輸入 8662 時,只要符合 8662 開頭的號碼,全部在前面自動加上 002；則實際號碼爲[002+8662+xxx]。

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

說明 1：當撥號時有輸入 002 時,只要符合 002 開頭的號碼,全部將 002 開頭的號碼,置換成 006；則實際號碼爲[006+xxx]。

說明 2：當撥號時有輸入 003 時,只要符合 003 開頭的號碼,全部將 003 開頭的號碼,置換成 006；則實際號碼爲[006+xxx]。

範例 3:Drop prefix: No, Replace rule 3: 009, 12。

說明 1：當撥號時有輸入 12 時,只要符合 12 開頭的號碼,全部在前面自動加上 009；則實際號碼爲[009+12+xxx]。

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

說明 1：當撥號時有輸入 5xxx 時,要符合 5 開頭,後面接著 3 碼的資料；全部在前面自動加上 007；則實際號碼爲[007+5xxx]。
說明 2：當撥號時有輸入 534 時,符合 5 開頭,後面接著 2 碼的資料；不符合加碼規則;實際號碼爲[534]。
說明 3：當撥號時有輸入 35xx 時,要符合 35 開頭,後面接著 2 碼的資料；全部在前面自動加上 007；則實際號碼爲[007+35xx]。
說明 4：當撥號時有輸入 358822 時,符合 35 開頭,後面接著 4 碼的資料；不符合加碼規則;實際號碼爲[358822]。

12. 如何實現多個 SIP 平台切換?

如果話機已同時設定好 2-3 個 SIP 平台帳號,若要切換到想使用的平台則請照下列方法操作
例如話機同時設定有 ABC 三個 SIP 平台,A 爲預設平台;要由 A 平台切換到 B 平台則
- 舉起話筒或按下免持鍵切換到 IP 撥號模式.此時話機 LCD 會顯示 A 平台的電話號碼
- 按 2*
- 掛上話筒或按免持鍵保持話機侍機狀態此時話機 LCD 會顯示 B 平台的電話號碼
- 現在即可以使用 B 平台撥打電話

如果要使用 C 平台則
- 舉起話筒或按下免持鍵切換到 IP 撥號模式
- 按 3*
- 掛上話筒或按免持鍵保持話機侍機狀態此時話機 LCD 會顯示 C 平台的電話號碼
- 現在即可以使用 C 平台撥打電話

如果又要使用 A 平台則
- 舉起話筒或按下免持鍵切換到 IP 撥號模式
- 按 1*
- 掛上話筒或按免持鍵保持話機侍機狀態此時話機 LCD 會顯示 A 平台的電話號碼
- 現在即可以使用 A 平台撥打電話

若話機重新啓動後則預設平台爲 A 平台

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