



User's Manual

High Definition PoE IP Phone (1-Line)

▶ VIP-1010PT



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CE Mark Warning

The is a class B device. In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.



Energy Saving Note of the Device

This power required device does not support standby mode operation. For energy saving, please remove the DC-plug or push the hardware Power Switch to OFF position to disconnect the device from the power circuit.

Without removing the DC-plug or switching off the device, the device will still consume power from the power circuit. In view of Saving the Energy and reducing the unnecessary power consumption, it is strongly suggested to switch off or remove the DC-plug from the device if this device is not intended to be active.

WEEE Warning



To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal

waste and have to collect such WEEE separately.

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Revision

User's Manual on PLANET SIP PoE IP Phone: Model: VIP-1010PT Rev: 1.0 (June, 2014) Part No. EM-VIP-1010PT_v1.0



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1 Introduction



Cost-effective, High-performance PoE Vol P Phone

To build high-performance VoIP communications at a low cost, PLANET has integrated high-definition voice into a cost-effective SIP phone. It complies with IEEE 802.3af PoE interface for flexible deployment. The VIP-1010PT makes it simple for the enterprise featuring voice and data system or expanding voice system to new locations. It helps the company to save money on long-distance calls; for example, the remote workers can dial in through a Unified VoIP Communication System just like an extension call but no long-distance call charge would occur. The VIP-1010PT also allows call to be transferred to anyone at any location within the voice system, which enables the enterprise to communicate more effectively and is helpful to streamline business processes.





High-quality HD VoIP Voice

The VIP-1010PT delivers HD voice (High-definition Voice) which is the next generation of voice quality for telephony audio, making the quality of voice better than that (toll quality) of the standard digital telephony and even close to that of a room conversation. HD voice is transmitted in the audio frequency range of 50 Hz to 7 kHz or higher over telephone lines, resulting in higher quality voice and clearer communication.



Standard Compliance

The VIP-1010PT supports Session Initiation Protocol 2.0 (RFC 3261) for easy integration with general voice over IP system. The VIP-1010PT is able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.

Compliant with standard SIP RFC 3261





Enhanced, Full-Featured Business IP Phone

The VIP-1010PT is a full-featured, enhanced business IP Phone that addresses the communication needs of the enterprises. It provides 1 voice line and dual 10/100Mbps Ethernet. Furthermore, the VIP-1010PT delivers user-friendly design containing a 132x64 graphic LCD with white backlight.

The VIP-1010PT supports all kinds of SIP-based phone features including LDAP, Call Waiting, Auto Answer, Music on Hold, Caller ID 3-way Conferencing, Call on Hold, Call Forwarding, Black List, DTMF Relay, In-Band, Out-of-Band (RFC 2833) and SIP Info, among others. Besides office use, the VIP-1010PT is also the ideal solution for VoIP service offered by Internet Telephony Service Provider (ITSP).



1.1 Features

Highlights

- Supports SIP 2.0 (RFC3261)
- Supports 1 SIP voice line
- IEEE 802.3af Power over Ethernet compliant
- Supports HD voice
- LDAP/ TR-069 / SNMP

Phone Features

- 1 line (supporting 1 SIP account)
- Supports call waiting, call forwarding, call transfer
- 3-way conferencing
- Call on hold, mute, auto-answer, redial
- Phonebook (500 groups), blacklist (100 groups), call logs (100 entries)
- 5 remote phone book URL supported
- Keypad Lock
- DND (Do Not Disturb)



- Volume adjustable, ring tones selectable
- Call Pickup/Group Call Pickup
- Speed Dial
- Intercom
- Daylight Saving
- Network Packet Capture
- Country Ringtone Signal
- Direct IP Call
- Auto Redial / Hot Desking
- Hotline / XML Browser / Action URL
- Multi-Languages: Default: English and Simple Chinese

IP PPX Features

- HD Voice
- Dial Plan
- SMS, Voicemail, MWI Message Notification
- Wideband Codec: G.722
- Narrowband Codec: PCMA, PCMU, G.729, G.722, G723_53, G23_63, G726_32
- VAD, CNG , Echo Canceller
- Full-Duplex Speakerphone

Security Features

- Supports HTTPS (SSL)
- Supports SRTP for Voice Data Encryption
- Supports Login for Administration
- SIP Over TLS

Network Features

- SIP V1(RFC2543), V2(RFC3261)
- Static IP/DHCP for IP configuration
- 3 DTMF modes: In-Band, RFC2833, SIP INFO
- HTTP/HTTPS Web Server for Management
- NTP for Auto Time Setting



Administration Features

- Auto provisioning using FTP/TFTP/HTTP/HTTPS/PnP
- Dial through IP PBX using Phone Number
- Dial through IP PBX using URL Address
- Configuration Managements with Web, Keypad on the phone and Auto Provisioning
- SNMP
- TR069

1.2 Application

Enterprise IP PBX Deployment of VIP-1010PT

The VIP-1010PT is much easier to install and configure than the traditional phone system. Its low cost and high-definition voice quality give you value for money. Base on standard SIP 2.0, it is compatible with all the standard SIP-based servers.







1.3 Product Specifications

Product	VIP-1010PT	
Hardware		
Lines (direct numbers)	1-line cost-effective IP phone	
Display	132 x 64 graphic LCD with blue backlight	
Footuro Kovo	4 Soft Keys	
reature keys	10 Programmable Keys	
Protocols and Standard		
	MAC Address (IEEE 802.3)	
	IPv4 (RFC 791)	
	Address Resolution Protocol (ARP)	
	DNS: A record (RFC 1706), SRV record (RFC 2782)	
	Dynamic Host Configuration Protocol (DHCP) client (RFC 2131)	
	TCP (RFC 793)	
Data Networking	User Datagram Protocol UDP (RFC 768)	
	Real-time Protocol RTP (RFC 1889, 1890)	
	Real-time Control Protocol (RTCP) (RFC 1889)	
	Simple Network Time Protocol (SNTP) (RFC 2030)	
	Backward compatible with RFC 2543	
	Session Timer (RFC 4028)	
	SDP (RFC 2327)	
	SIP version 2 (RFC 3261, 3262, 3263, 3264)	
	Message Waiting Indicator (RFC 3842)	
	Voice algorithms:	
	- PCMA	
	- PCMU	
	- G.729	
Voice Gateway	- G.722	
	- G723_53	
	- G23_63	
	- G726_32	
	Dual-tone Multi-frequency (DTMF), In-Band and Out-of-Band (RFC	
	2833) (SIP INFO)	
	Voice Activity Detection (VAD) with Silence Suppression	



Pestures Phone Features 1 line (supporting 1 SIP account) Supports call waiting, call forwarding, call transfer 3-way conferencing Call on hold, mute, auto-answer, redial Phonebook (SOO groups), blacklist (100 groups), call logs (100 entries) S Remote Phone Book URL supported LDAP DND (Do Not Disturb) Yolume adjustable, ring tones selectable Call Pickup/Group Call Pickup Speed Dial Intercorn Daylight Saving Network Packet Capture Country Ringtone Signal Direct IP Call Auto Redial Hotline XML Browser Hot Desking Keypad Lock Keypad Lock Action URL Multi-Languages: Default: English and Simple Chinese ID al Plan SMS, volcemail, MWI Message Notification Wideband Codec:: PCMA, PCMU, G.729, G.722, G723_53, G23_63, G726_32 YaD, CNG, Echo Canceller Full-Duplex Speakerphone Supports HITTPS (SSL)		Comfort Noise Generation	
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	Security Features	Supports HTTPS (SSL)	



	Supports SRTP for Voice Data Encryption
	Supports Login for Administration
	SIP Over TLS
	Dial without Register
	SIP V1(RFC2543), V2(RFC3261)
	Static IP/DHCP for IP configuration
Network Features	3 DTMF modes: In-Band, RFC2833, SIP INFO
	HTTP/HTTPS Web Server for Management
	NTP for Auto Time Setting
	Auto provisioning using FTP/TFTP/HTTP/HTTPS/PnP
	Dial through IP PBX using Phone Number
	Dial through IP PBX using URL Address
Administration Features	Configuration Managements with Web, Keypad on the phone and
	Auto Provisioning
	SNMP
	TR069
Environments	
Power Requirements	IEEE 802.3af
Operating Temperature	0 ~ 40 degrees C
Operating Humidity	10 ~ 65% (non-condensing)
Weight	651g (without box) / 920g (with box)
Dimensions (W x D x H)	193 x 190 x 35 mm
Emission	CE, FCC
	Two 10/100BASE-T RJ-45 Ethernet ports
	Handset: RJ-9 connector
Connectors	Headset: RJ-9 connector
	DC power jack



1.4 Physical Specifications and Packaging

Physical Specifications

Dimensions

Dimensions (L x W x H)	193 x 190 x 35 mm
Net Weight	651g (without box) / 920g (with box)

BASIC PACKAGING

- SIP IP Phone Unit
- Quick Installation Guide
- RJ-45 Cable x 1
- Stand x 1

1.5 Keypad

Keypad, LED, and Function Key Definitions





Keypad Description

Кеу	Key Name	Function Description
(15)	Navigation	Assists you in selecting an item that you want to process
< OK >		under the menu by pressing the Up, Down, Right or Left
		key. Press the OK key to save.
		Key combination includes functions such as
Destroy [List. Crick Trans]	Soft Keys	History/Favorites/Redial/CallReturn/HotDesking/DND/Menu/M
0000	1/2/3/4	SG/Status/Book/FWD/PickUp/Group PickUp/Intercom/Speed
		Dial/and so on.
	Home	Back to the Home page
E	Book	View Local Phone Book/Blacklist/Remote Phone Book
\bigcirc	Headset	Use the headset to call out or call in
FWD	Forward	Forward the call to the third party
	Redial	View the Missed Calls, Incoming Calls and Dialed Calls.
(m)	Muto	Press this key in calling mode and you can hear the other side,
	witte	but the other side cannot hear you.
(Volume /	Turn down or turn up the volume by pressing the "-" key or the
(4- 4+	volume -/+	"+" key.
U »	Hands-free	Make the phone into hands-free mode.
1 2.00 3.001	Divital	
4 _{ght} 5 _{1kt} 6 _{moo}	Digital	Input the phone number or DTMF.
	Keyboard	
	Indicator	
	light	Blinking light indicates there is an incoming call.



Rear View and Panel Descriptions



Keypad Description

Port	Port Name	Description
	Power Switch	Input: 5V DC, 1.2A
C	Internet	10/100M Connect it to Network
	PC	10/100M Connect it to PC
	Handset	Port type: RJ-9 connector
	Headset	Port type: RJ-9 connector



Account and Status Display

Port Name	Description	
Register success	(0	
Register failure	<u>6</u>	
Registering	g	
Deactivated account	Ŋ	
Auto answer	88	
No disturb	0	
Always forward	⊑ ,	
Network disconnection		
Ring off	u(×	
Headset mode	ប	
New voice message	80	
New text message	\boxtimes	
Missed calls	\$	



2 Initial Connection and Login

The package should contain the following items plus the VIP-1010PT. If any item is missing or damaged, please contact the seller immediately.



Step 1. Handset Connection

Plug one end of the handset cord into the handset and the other end into the handset jack



Step 2. Connecting Power System

The VIP-1010PT can be powered either by its external AC/DC adapter or by connecting to an IEEE 802.3af/at PSE device such as 802.3af injector / hub or 802.3af/at POE switch. Once the VIP-1010PT is powered, the LCD screen will prompt for POST.



Note 1: This unit does not include the 5V/1.2A power adapter. Note 2: Only WAN supports POE.



Step 3. Connecting Network



Step 4. Computer Network Setup

Set User computer's IP address to 192.168.0.x, where x is a number between 2 to 254 (except 1 where is being used for the phone by default). If User don't know how to do this, please ask User network administrator.

Connecting User PC to the VIP-1010PT PC port.



Step 5. Login Prompt

Use Web browser (Internet Explorer 6.0 or above) to connect to 192.168.0.1 (type this address in the address bar of Web browser).

User'll be prompted to input user name and password: admin and 123





3 LCD Basic Functions

3.1 Making a call

3.1.1 Call Device

User can make a phone call via the following methods :

- 1. Pick up the handset, Cicon will be shown on the idle screen.
- 2. Press the Hand-free key, icon will be shown on the idle screen.
- 3. Press the Headset key if the headset is connected to the Headset Port in advance.
- The **\mathbf{k}** icon will be shown on the idle screen.

User can also dial the number first, and then choose the method user will use to speak to the other party.

3.1.2 Call Methods

User can press an available line key if there is more than one account, then

1. Dial the number user wants to call.

2. Press History soft key. Use the navigation keys to highlight user choice (press Left/Right key to choose Missed Calls, Incoming Calls and Outgoing Calls).

3. Press the Redial key twice to call the last number called or press Redial key to enter All Calls interface to choose the number to dial out.

4. Press the programmable keys which are set as speed dial keys. Then press the Send key or Dial soft key to make the call if necessary.

3.2 Answering a call

1. If user is not on another phone call, lift the handset to use, or press the Speaker key/Answer soft key to answer using the speaker phone, or press the Headset key to answer the headset.

2. If user is on another phone call, press the answer soft key to answer new



incoming and hold the current talking. During the conversation, user can alternate between Headset, Handset and Hand-free by pressing the corresponding keys.

Note: The will flash during the Incoming interface

3.3 Mute

User can press the Mute key to make the user NOT be heard by the other party, but user can hear the other party. Icon will be shown on the LCD, and press the Mute key again to recover.

3.4 Call Hold / Resume Forward

1. Press the Hold button or Hold soft key to put user active call on hold.

2. If there is only one call on hold, press the hold soft key to retrieve the call.

3. If there is more than one call on hold, press the line button, and the Up/Down button to highlight the call, and then press the Resume button to retrieve the call.

3.5 DND (Do Not Disturbed)

If user enables the DND mode, the phone will reject to answer all calls automatically and play busy tone; the UI will present missed calls at the same time.

3.6 Call Waiting

To configure Call Waiting via Phone interface:

- 1. Press Menu -->Features-->Call Waiting-->Enter, ;
- 2. Use the Left or Right key to activate or deactivate call waiting.
- 3. Then press the Save key to save the changes.



3.7 Call Forward

User can set the static forward to transfer all the incoming calls to specified number; Also user can use dynamic forward to transfer all the incoming calls forward to the number inputted when the phone is ringing.

Forward: Enable call forward feature; Options are as follows :

- Always Forward: All the incoming calls will be transferred unconditionally to specified number.
- Busy Forward: The incoming calls will be transferred to specified number when the phone is busy.
- No Answer Forward: The incoming calls will be transferred to the specified number when the ring tone is time out without answer.

To configure Call Forward via Phone interface:

1. Press Menu -->Features-->Call Forward-->Enter, or just press FWD key to enter Call Forward interface.

2. There are 3 options: Always, Busy, and No Answer.

3. If user chooses one of them, enter the phone number user wants to forward to the receiving party. Press Save to save the changes.

3.8 Call Transfer

User can use the following two ways to transfer call to the other party :

- Blind Transfer: Transfer call directly to the other party without any negotiation.
- Consultation Transfer: Transfer call to the other person involved after the other person involved answers the incoming call and with consultation.

3.8.1 Blind Transfer

1. Press the Trans soft key during the talking;



2. Enter the Trans number interface, and then input the number you will transfer to;

3. Press the FWD key or the Trans soft key to transfer the call to the number you want to transfer to;

4. Return to the Idle automatically ;

Note: The UI will display Hold status interface when the number you want to transfer to does not exist.

3.8.2 Consultation Transfer

1. Press the Trans soft key to enter the number you want to transfer to during the talking. Input the number you want to transfer to.

2. Press the OK key on the phone keyboard or the Dial key to make a call.

3. Press the Trans soft key to finish transfer after the other person involved answers the incoming call and with consultation. You can finish transfer via putting down the handset or press the Cancel soft key to cancel transfer if you currently use handset to make or answer a call.

3.9 Three-way conference call

User can use the Local conference feature to hold a 3-way conference by pressing the Conference soft key to invite the current talking line and the line on hold to attend the conference. The Local conference feature of IP phone VIP-1010PT can invite two parties at most to attend conference. The conference type of IP phone VIP-1010PT is Local conference with default.

- 1. Create talking with first party.
- 2. Press the New soft key to create a new talking.
- 3. Press the Back soft key of dial interface to hold talking with first party.
- 4. Input the number of the second party and press the OK key on the phone keyboard or the Dial key or the Send soft key to make a call. When the



second party answers your call, inquire whether they want to attend the conference.

- 5. Press the Conference soft key to start the 3-way conference.
- 6. Press the Split soft key to split to two lines standalone talking, then these two parties talking are under Hold status.
- 7. Press the Resume soft key to resume the current talking.
- 8. Press the Cancel soft key or the to cancel the conference talking and return to Idle.

3.10 Call Park

User can use Call Park feature to park the current talking, and then resume the Parking talking with another phone (For example, with another phone of another office or conference). Press the Call Park key to park the current talking during the talking. If successful, you will hear voice announce or see the reserved extension number on the phone LCD. Dial the reserved extension number in another phone to resume the Parking talking.



Not all servers can support Call Park feature.

To configure Call Park via Phone interface:

PATH: Press Menu-->Features-->Call Park-->Press Left or Right key or Switch soft key to enable Call Park--> Press the Down key to set Target number-->Press the Down key to set Account-->Press the Save soft key to save

3.11 Pickup

User can use pickup to answer other users' incoming call. The VIP-1010PT IP phone



supports specified pickup and group pickup.



Press the group pickup only to answer line 1 (incoming call) if there are many lines (incoming calls) in group.

3.11.1 Specified Pickup

Specified pickup can answer specified user's incoming calls

1. Set specified pickup key via phone interface,

PATH: Press Menu-->Features-->Programmable keys-->Soft

Keys-->PickUp-->Press Down key to set label/Value--> Save soft key.

2. Use specified pickup feature

When the user of specified pickup number is off or busy, you can press the Pickup key to answer incoming call.

3.11.2 Group Pickup

Group pickup can answer group's user incoming calls. Group pickup needs to set group members.

Set group pickup via phone interface

PATH: Press Menu-->Features-->Programmable keys-->Soft Keys-->Group PickUp -->Press Down key to set label/Value/Account--> Save soft key.

Use group pickup feature

When anyone in group receives an incoming call, you can press the group Pickup key to answer.

3.12 Speed Dial

User can use the Speed Dial feature to dial the specified contact speedily

PATH: Press Menu-->Features-->Programmable keys-->Soft Keys-->Speed Dial -->Press Down key to set label/Value/Account--> Save soft key



3.13 Auto-redial

When hung up by the other party during calling, the phone will enter the auto-redial screen. Press OK for redial now or wait for the timeout to cancel Auto-redial.

To configure Auto Redial via Phone interface:

- 1. Press Menu -->Features-->Auto Redial-->Enter.
- 2. Use the Left or Right key to activate or deactivate Auto Redial.
- 3. Use the Up or Down key to configure Interval and Times.
- 4. Then press the Save key to save the changes.

3.14 Hot line

The Hot line refers to the number you often dial. You can set hot lines in the phone. The phone will dial the hotline number automatically when you pick up the handset and press the hand-free or the account key. Also you can set the timeout of dialing the hotline number, and then the phone will dial the hotline number automatically after the timeout.

To configure Hot line via Phone interface:

- 1. Press Menu -->Features-->Hot line-->Enter.
- 2. Use the Left or Right key to activate or deactivate Hot line.
- 3. Use the Up or Down key to configure Number and Timeout.
- 4. Then press the Save key to save the changes.

3.15 Intercom

To configure Intercom via Phone interface:

PATH: Press Menu-->Features-->Programmable keys-->Soft keys-->Intercom--> Press Down key to set label/Value/Account--> Save soft key

1. Press the Intercom key when the phone is available. The phone will connect the



extension number of remote user automatically.

- 2. Press the Intercom key or the Back soft key to end the intercom.
- 3. Answer the intercom incoming call.
- 4. By default, the IP phone VIP-1010PT will answer the intercom incoming call automatically and make a noise. You can set the phone to enable the silent mode when picking up the intercom call so that the other will not hear you. The features of intercom :

Intercom Feature	Note
Allow Intercom	Enable or disable Auto-receive intercom
Intercom Mute	Enable or disable Mute mode after receiving intercom incoming

3.16 HotDesking

In some working place, the people are always walking around. HotDesking feature will make the staff login his account on any computer in the company. In some public places, the working people are not fixed in one place. Anyone can use HotDesking for logging his account, and setting the phones to the familiar mode, such as the remote function of the computer.

3.16.1 Set the HotDesking Key

To configure Intercom via Phone interface: PATH: Press Menu-->Features-->Programmable keys-->Soft keys-->HotDesking--> Press Down key to set label--> Save soft key

3.16.2 HotDesking Feature :

- 1. After setting the HotDesking on Soft-key, it will return to the idle screen.
- 2. Press the HotDesking, and enter the HotDesking screen.
- 3. If you press clear on the screen, the phone will begin to clear the information stored on the phone.



- 4. After clearing the setting, the phone will enter the account setting screen.
- 5. After entering the account information, it will go back to the home screen, and begin to use the new account.

3.17 Application

3.17.1 Text Message

The VIP-1010PT IP Phone can send and answer text message. The phone will make a "Du" sound and present "N piece of new message" on the LCD (For example, 1 new message), and a twinkling message icon will appear.



Note: Not all servers support message feature.

Read Text Message

- 1. Access Menu->Message->Text Message-> In box.
- 2. Press the OK key on the phone keyboard or the Enter soft key to enter the Text Message interface. Press the OK key on the phone keyboard or the Enter soft key to enter the in-box interface.
- 3. Select the message you want to read and press the OK key on the phone keyboard or the Enter soft key to read.

Send Text Message

- 1. In the Idle, press the Menu soft key.
- 2. In the mail menu interface, press the Down key on the phone keyboard to select Message. Press the OK key on the phone keyboard or the Enter soft key to enter Message interface.



- 3. In the Text Message interface, select "New Message". Press the OK key on the phone keyboard or the Enter soft key to enter new message and edit it. Press the "abc" soft key to switch the input methods.
- 4. Press the OK key on the phone keyboard or the Send soft key to send message;
- 5. Press the Left or Right key on the phone keyboard or the Switch soft key to switch to the relevant addresser.



6. Input the number of addresser :

Send Setting(2/2)					
To:					
1015					
Back	123	Delete	Send		

7. Press the Send soft key to send message.

Delete Text Message

- 1. In the Idle, press the Menu soft key.
- 2. Press the main menu interface. Press the Down key on the phone keyboard to select message. Press the OK key on the phone keyboard or the Enter soft key to enter the Message interface.
- 3. In the Text Message interface, press the Down key on the phone keyboard to select in-box.
- 4. Press the OK key on the phone keyboard or the Enter soft key to enter the in-box interface.
- 5. Select the message you want to delete and press the Delete soft key.
- 6. Delete all the text messages in the in-box. Press the Delete soft key and select "Delete All". Press the OK soft key and then all the messages in the in-box will be deleted.



3.17.2 Voice Message

The VIP-1010PT IP Phone can send or answer voice message. The phone will make a "Du Du" sound, the LED light of message flashes green, and the LCD presents "New Voice Message" on the LCD with a twinkling voice message icon.

8 100:	3		
Voice Message			
			
Exit			View

Note: Not all servers support voice message.

Voice Message

You can leave a message when the user whom you called is busy or unavailable.

Leave a message according to the voice prompt of server, and then hang up after

leaving the message.

Set Visit account number of voice message via phone interface.

- 1. In the Idle, press the Menu soft key.
- 2. In the Idle, press the Down key on the phone keyboard to select message, press the OK key on the phone keyboard or the Enter soft key to enter the Message interface.
- 3. In the Message interface, press the Down key on the phone keyboard to select the voice message. Press the OK key on the phone keyboard or the Enter soft key to enter the Voice Message interface.
- 4. Select the Voice Message Setting.
- 5. Press the OK key on the phone keyboard or the Enter soft key to set account 1, input the Visit account number of voice message (For example, *97). Press 123 soft key to switch the input methods.

Voice Message Setting(1/3)			
Account1 NO.			
*97			
	¢	n 18	
Back	123	Delete	Save

6. Press the OK key on the phone keyboard or the Save soft key to save and return to message interface.



Check voice message

- 1. Press the Message key or the Connect soft key to call the Visit account number of voice message.
- 2. Check voice message according to voice prompt.
 - Set the Visit account number of voice message first before checking voice message. The LED light of Message will darken after all the voice messages are checked.
- 3. Check voice message via phone interface
 - Access Menu-> Message->Voice Message-> New Message. The LCD displays new messages and old messages of every account.



• 2. Select the account you will check and press the Connect soft key to check

📢 Talki	ng		1/1		
2 *97					
00:00:05					
Trans	Hold	New	Cancel		



4 LCD Advanced Settings

4.1 Basic Settings

4.1.1 Language

You can change the language through the method below:

Press Menu -> Settings -> Basic Setting -> Language

4.1.2 Date & Time

- The IP phone displays Time and Date in Idle status. You can set the Time and Date obtained from SNTP server automatically or you can set the time and date manually.
- Set SNTP via phone interface: Access Menu -> Settings -> Basic Setting -> Date & Time -> SNTP Setting.
- 3. To set the date & time format via the phone interface, access Menu -> Settings -> Basic Setting -> Date & Time -> Format Setting:
- Access the Time Format in Format Setting interface and then press the Left or Right key on the phone keyboard, or the Switch soft key to select the time format (12 Hours or 24 Hours).
- In the Date &Time Format interface, press the Up or Down key on the phone keyboard to access the Date Format. Press the Left or Right key on the phone keyboard or the Switch soft key to select the date format to process setting.
- The phone supports four Date formats. The selected date format will appear in the Idle. For example, if the time was "2013-09-13", the date formats in the menu and the corresponding formats displayed in the Idle would be as follows:



Date Format	e.g.(2013-09-13)	
YYYY-MM-DD	2013-09-13	
YYYY/MM/DD	2013/09/13	
DD-MM-YYYY	13-09-2013	
DD/MM/YYYY	13/09/2013	

4.1.3 Backlight

Set the screen backlight level and duration of backlight Press Menu -> Settings -> Basic Setting ->Backlight

4.1.4 Password Setting

This function is to set into the advanced Settings password Press Menu -> Settings -> Advanced Setting ->Password Setting A dialog box "Enter Password:" appears, enter the password: admin (default) and then press the OK key on the phone keyboard. Input the current password and the new password, and then confirm the new password to modify the current password.

4.2 Sound Settings

4.2.1 Phone Volume

1. The Volume key can be used to adjust the volume of handset, hand-free or headset during a call. Also, the key can be used to adjust the ring tone volume in the Idle mode.

2. Adjust the volume via the phone interface; access Menu -> Settings -> Basic Setting -> Phone Volume. In the Volume Setting interface, access the Handset Volume, Hands-free Volume or Headset Volume interface and then press the "+" or "-" soft key, or Left or Right key to adjust the volume. Press the Save soft key



to save the operation or press the Back soft key to cancel operation.

4.2.2 Ring Tones

- 1. The Ring Tone refers to the incoming ring tone, which reminds the user that a new call is coming with the phone. The VIP-1010PT supports phone ring tone to distinguish the incoming call from the other surrounding phone's ring tone. At the same time, the VIP-1010PT also supports setting of a specific incoming ring tone for contacts.
- To set the ring tone via the phone interface, access Menu -> Settings -> Basic Setting -> Ring Tones.

4.3 Phone Book

4.3.1 Local Phone Book

The Local Phone Book is used for storing the contact names and numbers. The VIP-1010PT can store up to 500 entries. You can add, edit, delete, search, or call any contact from the Local Phone Book \circ

4.3.2 Add contacts manually

Add contacts manually from the Local phone book via Phone interface: Press Phone book -> Local phone book -> Add to Contacts.

Select the relevant group (For example, contacts) and press the OK key on the phone keyboard or the Enter soft key in the UI to enter all Contacts:

- 1. Press the Add soft key to enter the Add Contact interface.
- 2. Input name in the relevant area.
- 3. Press the Down key on the phone keyboard to input the office number in the relevant area.
- 4. Press the Down key on the phone keyboard to input mobile number in the relevant area.
- 5. Press the Down key on the phone keyboard to input other numbers in the relevant area.



6. Press the Down key on the phone keyboard to enter Account selection; Press the Left or Right key on the phone keyboard or the Switch soft key to select the relevant account. If Auto selected, the phone will select the current available account automatically when the contact called from Local phone book.

4.3.3 Add Contacts from All Calls History :

Add contacts from All Calls History in the phone interface:

1. Press the History soft key.

2. Press the Up or Down key on the phone keyboard to select the contact you want to add.

3. Press the Option soft key to add to contacts.

4.3.4 Search Contacts

 Press the Book soft key in the Idle interface to enter the Phone Book menu.
 Select the Local Phone Book and press the OK key on the phone keyboard or the Enter soft key to enter the Local Phone Book.

3. Press the Search soft key to search contacts.

4. Input keywords such as name, any character of number or whole phone number and press the Search soft key or the OK key to enter the Search Contacts interface.

4.3.5 Blacklists

100 Blacklist contacts are available with the VIP-1010PT IP Phone. You can add,

edit, delete, search or call contact. The phone will reject to answer automatically

within the blacklist contacts' incoming call.

PATH: Press Phone book -> Blacklist -> Add.

4.3.6 Remote Phone Book

1. Access the remote phone book, add the contacts to the local phone book from the remote phone book or make calls from the remote phone book. Five URLs of remote phone book are available to set.

- 2. Set the remote phone book via web interface.
- 3. Access Book-> Remote Phone Book.
- 4. Input URL of phone book.


- 5. Input the phone book name.
- 6. Click the Submit soft key to submit.
- 7. Access the remote phone book via phone interface.
- 8. Access Book->Remote phone book.

9. Select the relevant Remote Group and press the Enter soft key. The phone will load the remote group information, and the LCD will display the contacts of this remote group.

10. Press the 🕩 key or the Back soft key to unlink.

11. Press the Book soft key to enter the Phone Book Menu.

4.4 History Management

The History management of the VIP-1010PT contains dialed calls, received calls,

missed calls and forwarded calls, and supports 100 storage logs at most. You

can check the history, make calls from the calls history and delete the calls

history.

- 1. Press the History key and the LCD will display all the recent calls.
- 2. Press the Left or Right key on the phone keyboard to switch the lists of All Calls, Dialed Calls, Received Calls, Missed Calls and Forwarded Calls.
- 3. Press the Up or Down key on the phone keyboard to select the log.
- Press the Option soft key and select the detail. The LCD will display the detailed information of this log; Press the Dial soft key to make a call from the History.
- Press the Option soft key to add to contacts(Move to Blacklists) from the History.
- Press the Delete soft key to delete calls log from the History.
- Press the Option soft key to select "Delete all" to delete all the calls log from the History.

4.5 System Customizations

4.5.1 Programmable keys

1. Press the Menu soft key in the Idle interface and access Menu->Features-> Programmable keys.

2. Select the programmable key you will set and press the Enter soft key.



- 3. Select key style in the type area.
- 4. Input suitable value in the label area.
- 5. (Optional) Select the relevant account in the account ID area.
- 6. (Optional) Input suitable value in Value blank.
- 7. Press the Save soft key to save or the Cancel soft key to cancel.

4.5.2 SIP Account management

4.5.2.1 Register an Account

Register an account via phone interface :

1. Press the Menu soft key to enter setting interface to select Advanced setting,

and input password(password: admin) to enter the Account setting.

2. Press Enter key to enter the account activation status area.

3. Input the label, display name, register name, account, password and SIP separately.

4. Press the Save soft key to save, or the Back soft key to cancel.

4.5.2.2 Disable an Account

- 1. Access Menu->Settings->Advanced setting->Account (password: admin).
- 2. Press Enter key to enter the account activation status area.
- 3. Select "Disable" in the account active status area.
- 4. Press the Save soft key to save or the Back soft key to cancel.

4.6 Basic Network Settings

Through the Basic Network setting, you can set the IP phones to get the IP address by three ways : DHCP, static IP and PPPoE.

PATH: Menu -> Settings -> Advanced Setting -> Network

4.6.1 DHCP Mode

1. In the Network Settings interface, Press the OK key on the phone keyboard or the Enter soft key to enter LAN Port.

2. In the LAN Port interface, press the Up or Down key on the phone keyboard to select DHCP (default is DHCP).

3. Press the Enter on the soft key or the OK key on the phone keyboard to enter the DHCP switch interface. It will automatically return to the last interface after seconds.



4.6.2 Static IP Mode

1. In the LAN Port interface, press the Up or Down key on the phone keyboard to select Static IP and then press the OK key on the phone keyboard or the Enter soft key to enter Static IP Setting interface and input IP address.

2. Press the Down key on the phone keyboard to enter the Subnet Mask of Static IP Setting and input the subnet mask.

3. Input the IP address, Subnet mask, Gateway, DNS 1 and DNS 2 in the corresponding area and press the OK key on the phone keyboard or the Save soft key to save.

4.6.3 PPPoE Mode

1. In the LAN Port interface, press the Up or Down key on the phone keyboard to select PPPoE and then press the OK key on the phone keyboard or the Enter soft key to enter PPPoE Setting interface.

2. Press the Up or Down key on the phone keyboard to enter User Name and Password.

3. In the related fields, input User Name and Password.

4.7 Reset to Factory

In the Advanced Setting interface, press the Up or Down key on the phone keyboard to select "Reset to factory". Press the OK key on the phone keyboard or the Enter soft key to access the reset to factory interface.

4.8 Reboot

This is a function to set the phone reboot.

1. In the Advanced Setting interface, press the Up or Down key on the phone keyboard to select Reboot.

2. Press the OK key or the Enter soft key to on the phone keyboard to enter the reboot warning interface.



5 Web Configuration

5.1 Ways to configure

The VIP-1010PT has two different ways for different users.

- Use phone keypad.
- Use web browser (recommended way).

5.2 Setting via web browser

When this phone and PC are connected to network, enter the IP address of the WAN port in this phone as the URL (e.g. http://xxx.xxx.xxx/ or http://xxx.xxx.xxx.xxx/).

If user does not know the IP address, he can look it up on the phone's display by pressing the Status button.

The login page is shown below: A

PLANET Networking & Communication	High De	finition VolP Ph	ONC VIP-1010PT	à
Login L U P R User	ogin Status Iser Name 'assword temember 'name/Password 🗖			Help Login Page

- Default user with root level:
 - User Name: admin
 - Password: **123**

The default password of phone screen menu is **123**.

After user configures the IP phone, he needs to click the Save button in the configuration under Maintenance on the left side of the screen to save user configuration. Otherwise, the phone will lose user modification after power is off and on.



5.3 Status / Basic

PLA Networking & Ca	NET High L	Definition VolP Phone	VIP-1010PT
▼ Status Basic	Status Draduct Information		Help
► Account	Model MAC Address	VIP-1010PT 00:30:4F:00:29:6D	Note : Max length of characters for input box:
▶ Network	Firmware Version Hardware Version	50.141.2.15 50.0.1.0.0.0.0	255: Broadsoft Phonebook server address
 Phone Dhome Dhome 	Network Information	Static IP	127: Remote Phonebook URL & AUTOP Manual Update Server URL 63: The rest of input boxes
 PhoneBook Upgrade 	INTERNET Link Status INTERNET IP Address	Connected 192.168.1.101	Warning :
► Security	INTERNET Subrechask INTERNET Gateway INTERNET DNS1	192.168.1.254 192.168.1.254	Field Description :
	INTERNET DNS2 Primary NTP Secondary NTP	0.pool.ntp.org 1.pool.ntp.org	
	Account Information		
	Account1	204@192.168.1.21 Registered	

Status->Basic page is used to display some basic information for IP phone. Please refer to the corresponding page for any further information.

Status

Field Name	Explanation
Product	To display the device's information, such as Model name,
Information	MAC address (IP device's physical address), Firmware
	version and Hardware firmware.
Network	To display the device's Networking status(LAN Port), such
Information	as Port Type(which could be DHCP/Static/PPPoE), Link
	Status, IP Address, Subnet Mask, Gateway, Primary DNS
	server, Secondary DNS server, and Primary NTP server
	and Secondary NTP server (NTP server is used to
	synchronize time from Internet automatically.).
Account	To display device's Account information and Registration
Information	status (account user name, registered server's address,
	and register result).



5.4 Account / Basic



Field Name	Explanation		
SIP Account	To display and configure the specific Account settings.		
	Status: To display register result.		
	Display Label: Label is displayed on the phone's LCD		
	screen.		
	Display Name: Name is sent to the other call party for		
	displaying.		
	Register Name: Allocated by SIP server provider, used for		
	authentication.		
	User Name: Allocated by your SIP server provider, used		
	for authentication.		
	Password: Used for authorization.		
SIP Server 1	To display and configure Primary SIP server settings.		
	Server IP: SIP server address; it could be an URL or IP		
	address.		
	Registration Period: The registration will expire after		
	registration period. The IP phone will re-register		
	automatically within registration period.		
SIP Server 2	To display and configure Secondary SIP server settings.		



	This is for redundancy; if registering to Primary SIP server		
	fails, the IP phone will go to Secondary SIP server for		
	registration.		
	Secondary SIP server is used for redundancy;		
	it can be left blank if there is no redundancy		
	Note SIP server in user's environment.		
Outbound Proxy	To display and configure Outbound Proxy server settings.		
Server	An outbound proxy server is used to receive all initiating		
	request messages and route them to the designated SIP		
	server.		
	If configured, all SIP request messages from		
	the IP phone will be sent to the outbound		
	Note proxy server forcefully.		
Transport Type	To display and configure Transport type for SIP message		
Transport Type			
	UDP: UDP is an unreliable but very efficient transport		
Transport Type	UDP: UDP is an unreliable but very efficient transport layer protocol.		
Transport Type	UDP: UDP is an unreliable but very efficient transport layer protocol. TCP: Reliable but less-efficient transport layer protocol.		
Transport Type	UDP: UDP is an unreliable but very efficient transport layer protocol.TCP: Reliable but less-efficient transport layer protocol.TLS: Secured and Reliable transport layer protocol.		
Transport Type	 UDP: UDP is an unreliable but very efficient transport layer protocol. TCP: Reliable but less-efficient transport layer protocol. TLS: Secured and Reliable transport layer protocol. DNS-SRV: A DNS RR for specifying the location of 		
	 UDP: UDP is an unreliable but very efficient transport layer protocol. TCP: Reliable but less-efficient transport layer protocol. TLS: Secured and Reliable transport layer protocol. DNS-SRV: A DNS RR for specifying the location of services. 		
NAT	 UDP: UDP is an unreliable but very efficient transport layer protocol. TCP: Reliable but less-efficient transport layer protocol. TLS: Secured and Reliable transport layer protocol. DNS-SRV: A DNS RR for specifying the location of services. To display and configure NAT (Net Address Translator) 		
NAT	 UDP: UDP is an unreliable but very efficient transport layer protocol. TCP: Reliable but less-efficient transport layer protocol. TLS: Secured and Reliable transport layer protocol. DNS-SRV: A DNS RR for specifying the location of services. To display and configure NAT (Net Address Translator) settings. 		
NAT	 UDP: UDP is an unreliable but very efficient transport layer protocol. TCP: Reliable but less-efficient transport layer protocol. TLS: Secured and Reliable transport layer protocol. DNS-SRV: A DNS RR for specifying the location of services. To display and configure NAT (Net Address Translator) settings. STUN: Short for Simple Traversal of UDP over NATS, a 		



5.5 Account->Advanced

PLAN Networking & Com	IET High Defi	nition VolP Phone 🛛	-1010PT
► Status	Account-Advanced		
Account	Codecs		нер
Pacie	Disabled Codecs Enabled	Codecs	Note : Max length of characters
Adversel	G723_53 PCMU G723_63 PCMA		for input box: 255: Broadsoft Phonebook
Advanced	G726-32 G729 G722		server address 127: Remote Phonebook
Network			URL & AUTOP Manual Update Server URL
Phone	>>		63: The rest of input boxes
PhoneBook	<<	L	Warning :
► Upgrade			Field Description :
Security			Submit Shortcut
	Durk surk to		
	Subscribe MW/L Cubscribe	Disphied	
	MWI Subscribe Deried	1900 (190065525c)	
	Voice Mail Number	(1800~033535)	
	DTME		
	Tupo	PE02822	
	How To Notify DTME	Disabled	
	DTME Payload	101 (96~127)	
	Call		
	Max Local SID Dort	5062 (1024-65525)	
	Min Local SIP Port	5062 (1024~65535)	
	Caller ID Header	EROM V	
	Auto Answer	Disabled T	
	Ringtones	Default 🔻	
	Provisional Response ACK	Disabled •	
	user=phone	Disabled 🗸	box:
	PTime	20	255: Broadsoft Phonebook server
	Anonymous Call	Disabled 🗸	127: Remote Phonebook URL &
	Anonymous Call Rejection	Disabled Y	AUTOP Manual Update Server URL
	Missed Call Log		os. merescon input boxes
	Music Server Address		Warning :
	Active	Disabled	Field Description :
	Music Server Address		Submit Shortcut
	Session Timer		Submit Cancel
	Active	Disabled 💙	
	Session Expire	1800 (90~7200s)	
	Session Refresher	UAC 👻	
	Broadsoft		
	AOC	Disabled	
	Encryption		
	Voice Encryption(SRTP)	Disabled	
	NAT	To the second	
	UDP Alive Msg Interval	30 (5~60s)	
	RPort	Disabled	
	Submit	Cancel	



Field Name	Explanation	
Codecs	To display and configure available/unavailable codecs list.	
	Codec means coder-decoder which is used to transfer	
	analog signal to digital signal or vice versa.	
	Familiar codecs are G723_53, G723_63, G726_32,	
	PCMA, PCMU, G.729 and G722.	
Subscribe	To display and configure MWI, subscription settings.	
	MWI: Message Waiting Indicator which is used to indicate	
	whether there is unread new voice messages.	
DTMF	To display and configure DTMF settings.	
	Type: Supports Inband, Info, RFC2833 or their	
	combination.	
	How To Notify DTMF: Only available when it is DTMF tone	
	information.	
	DTMF Payload: To configure payload type for DTMF.	
	By default, DTMF type is RFC2833 which is the	
	standard. Inband type uses inband frequency	
	to indicate DTMF tone which is most used to be	
	Note compatible with the traditional telephone	
	server. Info type uses SIP Info message to	
	indicate DTMF message.	
Call	To display and configure call-related features.	
	Max Local SIP Port: To configure maximum local sip port	
	for designated account.	
	Min Local SIP Port: To configure minimum local sip port	
	for designated account.	
	for the displaying on Phone III	
	Auto Answer: If enabled, IP phone will be auto-answered	
	when there is an incoming call for designated account	
	Provisioning Response ACK : 100% reliability for all	
	provisional mossages meaning it will sond ACK every	
	time the IP phone receives a provisional SIP message	
	from SIP server	
	User=phone [.] If enabled IP phone will send user=phone	
	within SIP message.	
	PTime: Interval time between two consecutive RTP	
	packets.	
	Anonymous Call: If enabled, all outgoing calls for the	
	designated account will be anonymous numbers.	
	Anonymous Call Rejection: If enabled, all incoming	
	anonymous calls for the designated account will be	
	rejected.	



Music Server	To display or configure third-party MOH (music-on-hold)		
Address	server.		
	Active: To enable or disable this MOH server. If enabled,		
	the IP phone will play MOH from configured server.		
	Music Server Address: To configure MOH server address.		
Session Timer	To display or configure session timer settings.		
	Active: To enable or disable this feature. If enabled, the		
	ongoing call will be disconnected automatically once the		
	session expired unless it's been refreshed by UAC or UAS.		
	Session Expire: Configure session expiry time.		
	Session Refresher: To configure who should respond for		
	refreshing a session.		
	UAC means User Agent Client and here stands		
	for IP phone. UAS means User Agent Server		
	Note and here stands for SIP server.		
Broadsoft	To display or configure Broadsoft AOC feature.		
	AOC: A feature is used to be accounted on Broadsoft		
	platform.		
	Please consult your administrator or Planet Technical		
	support for further information.		
Encryption	To enable or disable SRTP feature.		
	Voice Encryption (SRTP): If enabled, all audio signals		
	(technically speaking it's RTP streams) will be encrypted		
	for more security.		
NAT	To display NAT-related settings.		
	UDP keepalive message: If enabled, IP phone will send		
	UDP keepalive message periodically to router to keep NAT		
	port alive.		
	UDP Alive Msg Interval: Keepalive message interval.		
	Rport: Remote Port; if enabled, it will add Remote Port		
	into outgoing SIP message for designated account.		



5.6 Network / Basic

PLAN Networking & Cor	JET High Do	efinition VolP Phon	C VIP-1010PT
► Status	Network-Basic		LogOut Help
Account	INTERNET Port		
▼ Network	DHCPStatic IP		Max length of characters for input box:
Basic	IP Address	192.168.1.101	255: Broadsoft Phonebook server
Advanced	Subnet Mask	255.255.255.0	127: Remote Phonebook URL &
▶ Phone	Default Gateway INTERNET DNS1	192.168.1.254 192.168.1.254	AUTOP Manual Update Server URL 63: The rest of input boxes
► PhoneBook	INTERNET DNS2		Warning :
▶ Upgrade	PPPoE User Name		Field Description :
Security	Password	*****	Submit Shortcut Submit Cancel
	Submit	Cancel	

LANGUAGE		
Field Name	Explanation	
LAN Port	To display and configure LAN Port settings.	
	DHCP: If selected, IP phone will get IP address, Subnet	
	Mask, Default Gateway and DNS server address from	
	DHCP server automatically.	
	Static IP: If selected, you have to set IP address, Subnet	
	Mask, Default Gateway and DNS server manually.	
	PPPoE: Use PPPoE username/password to connect to	
	PPPoE server.	



5.7 Network / Advanced

PLAN Networking & Co	NET	High Defi	inition Ve	olP Phone	/IP-1010PT	
▶ Status	1.					<u>LogOut</u>
Jialus	Network-Advan	iced			Help	
Account	Local RTP				Note :	
▼ Network		Max RTP Port	12000	(1024~65535)	Max length of characters for input	
Pasic		Min RTP Port	11800	(1024~65535)	box: 255: Broadsoft Phonebook server	
Dasic	SNMP				address	
Advanced		Active	Disabled	¥	127: Remote Phonebook URL &	
▶ Phone		Port		(0~65535)	63: The rest of input boxes	
▶ PhoneBook		Trusted IP			Warning :	
10 - 10 - 10 - 10 - 10 - 10 - 10 - 10 -	TR069					
Upgrade		Active	Disabled	T	Field Description :	
Security		Version	1.0	•	Submit Shortcut	
	ACS	URL			Submit Cancel	
		User Name				
		Password	•••••			
	Periodic Inform	Active	Disabled	¥		
		Periodic Interval	1800	(3~3600s)		
	CPE	URL				
		User Name				
		Password	•••••			
	DDNS					
		Туре	Disabled	¥		
		Easy Domain Name	pl00296D.planetdo	dns.com		
		Provider	planetddns.com	•		
		Account				
		Password	******			
		DDNS				
		Submit	Ca	ancel		

Network Configuration		
Field Name	Explanation	
Local RTP	To display and configure Local RTP settings.	
	Max. RTP Port: Determine the maximum port that RTP	
	stream can use.	
	Min. RTP Port: Determine the minimum port that RTP	
	stream can use.	
SNMP	To display and configure SNMP settings.	
	Active: To enable or disable SNMP feature.	
	Port: To configure SNMP server's port.	
	Trusted IP: To configure allowed SNMP server address, it	
	could be an IP address or any valid URL domain name.	
	SNMP (Simple Network Management Protocols) is	
	Internet-standard protocol for managing devices on IP	



	networks.		
TR069	To display and configure TR069 settings.		
	Active: To enable or disable TR069 feature.		
	Version: To select supported TR069 version (version 1.0		
	or 1.1).		
	ACS/CPE: ACS is short for Auto configuration servers as		
	server side. CPE is short for Customer-premise equipment		
	as client-side devices.		
	URL: To configure URL address for ACS or CPE.		
	User Name: To configure username for ACS or CPE.		
	Password: To configure Password for ACS or CPE.		
	Periodic Inform: To enable periodically inform.		
	Periodic Interval: To configure interval for periodic inform.		
	TR-069(Technical Report 069) is a technical		
	specification entitled CPE WAN Management		
	Protocol (CWMP). It defines an application		
	Note layer protocol for remote management of		
	end-user devices.		
DDNS	The VIP-1010PT supports Planet DDNS and Easy DDNS		
	function.		



For now, the VIP-1010PT can only support Planet DDNS.





5.8 Phone/Time/Language

Status Time/Lang Help Account Web Language Note : Network Type English Max length of characters for input book server address Phone LCD Language 255: Broadsoft Phonebook server address				mmunication HIGN DETIN	PLFIN Networking & Col
Status Time/Lang Account Web Language Network Type Type English LCD Language Dox: 255: Broadsoft Phonebook server address	ogout	L			
Account Web Language Note : Network Type English Max length of characters for input box: Phone LCD Language 255: Broadsoft Phonebook server Type English address		Help		Time/Lang	► Status
Network Type English Note : Vertwork Type English Max length of characters for input box: Vertwork LCD Language box: Type English 255: Broadsoft Phonebook server Type English address				Web Language	► Account
V Phone LCD Language box: 255: Broadsoft Phonebook server Type English address		Note : Max length of characters for input	jlish 🔻	Туре	▶ Network
Y Phone English address		box: 255: Brandroft Dhamahaali aanvar		LCD Language	
		address	lish 🔹	Туре	 Phone
Time/Lang 127: Remote Phonebook URL &		127: Remote Phonebook URL & AUTOP Manual Lindate Server URL		Format Setting	Time/Lang
Preference Time Format 12Hour		63: The rest of input boxes	lour 🔹	Time Format	Preference
Call Feature Date Format DD-MM-YYYY Warning :		Warning :	MM-YYYY	Date Format	Call Feature
Display Mode Day T			•	Display Mode	Voice
Type		Field Description -		Туре	VOICE
Key/Display O Manual Submit Shortcut		Submit Shortcut		Manual	Key/Display
Ringtones Date Year Mon Day Submit Cancel		Submit Cancel	Mon Day	Date Year	Ringtones
Tones Time Hour Min Sec			Min Sec	Time Hour	Tones
Dial Plan O Auto				Auto	Dial Plan
NTP I I I I I I I I I I I I I I I I I I I				NTP	Letter L D
Time Zone +1 France(Paris)			•	Time Zone +1 France(Pa	ACIONORL
PhoneBook Primary Server 0.pool.ntp.org				Primary Server 0.pool.ntp.org	PhoneBook
► Upgrade Secondary Server 1.pool.ntp.org				Secondary Server 1.pool.ntp.org	▶ Upgrade
Update Interval 3600 (>= 3600s)		<u>,</u>	(>= 3600s)	Update Interval 3600	N D N
Daylight Saving Time		Field Description :		Daylight Saving Time	
Active Auto		Field Description -) •	Active	
OffSet 60 (-300~300Minutes) Submit Shortcut		Submit Shortcut	(-300~300Minutes)	OffSet	
By Date Submit Cancel		Submit Cancel		By Date	
Start Time			I Mon I Day U Hour	Start Time	
End Time 12 Mon 31 Day 23 Hour			12 Mon 31 Day 23 Hour	End Time	
By Week				By Week Start Month	
Start Week Of Month			In Month	Start Week Of Month	
Start Day Of Week Monday			day 🔻	Start Day Of Week	
Start Hour 0 (0~23)			(0~23)	Start Hour	
End Month Dec				End Month	
End Week Of Month Fourth In Month			th In Month	End Week Of Month	
End Day Of Week Sunday			day 🔻	End Day Of Week	
End Hour 23 (0~23)			(0~23)	End Hour	
Submit			Cancel	Submit	

Time / Language Configuration		
Field Name	Explanation	
Web Language	To switch to designated language.	
Format Setting	To configure time display settings.	
	Time Format: Determine what format to display on Phone	
	UI (12 hour/24 hour).	
	Date Format: Determine what format to display on Pho	
	UI for Date.	
	Display Mode: Determine what mode to display Time &	



	Date on Phone UI.		
Туре	To select how to configure time. It could be set manually		
	or get from Internet automatically via NTP server.		
	Manual: To set Time and Date manually.		
	Auto: To get Time via NTP server.		
	If user sets time to be Manually, it will only		
	take effect till the next reboot; after reboot,		
	the phone will switch to Auto mode		
	Note automatically because there is no way for IP		
	phone to record time during power off.		
NTP	To configure NTP server related settings.		
	Time Zone: To select local Time Zone for NTP server.		
	Primary Server: To configure primary NTP server address.		
	Secondary Server: To configure secondary NTP server		
	address; it takes effect if primary NTP server is		
	unreachable.		
	Update Interval: To configure interval between two		
	consecutive NTP requests. Network Time Protocol (NTP) is used to		
	Network Time Protocol (NTP) is used to		
	automatically synchronize local time with		
	Internet time, since NTP server only responds to GMT time, so that you need to specify the		
	Time Zone for IP phone to decide the local		
	time.		
Daylight Saving	To display or configure DST settings.		
Time	DST is short for daylight saving time, which		
	means the time on the summer days when the		
	sun rises early will be adjusted forward to save		
	daylight. The DST will take effect during the		
	Note period that is set by user. (All the settings for		
	DST are all self-explanatory. Please consult		
	with your administrator for local DST details.)		



5.9 Phone / Preference

PLAN Networking & Con	JET High Def	inition VolP Phone	VIP-1010PT
▶ Status			LogOut
otatas	Preference		Help
Account	Headset Mode		Note :
Network	Active	Enabled	Max length of characters for input box:
▼ Phone	Key Press Sound		255: Broadsoft Phonebook server
Time/Lang	Volume Ringtone Volume	8 (0~15)	address 127: Remote Phonebook URL & AUTOP Manual Update Server URL
Preference	Volume	5 (0~15)	63: The rest of input boxes
Call Feature	Submit	Cancel	Warning :
Voice			Field Description :
Key/Display			Submit Shortcut
Ringtones			Submit Cancel

Preference Configuration		
Field Name	Explanation	
Headset Mode	To enable or disable Headset Mode.	
	Active: If enabled, the default audio track will be headset	
	mode; if audio track is changed during a call, it will be	
	back to headset mode after you hang up the call.	
Key Press Sound	To configure the sound volume for key press.	
	Volume: The valid volume range is from 0 to15; by	
	default, it's 8.	



5.10 Phone / Call Feature

PLA Networking & Co	NET High De	finition VolP Phone 🖻	/IP-1010PT
			LogOut
► Status	Call Feature		Help
► Account	Forward Transfer		
▶ Network	Always Forward	Disabled	Note : Max length of characters for input
-	Target Number	205	box:
* Phone	On Code		address
Time/Lang	Off Code		127: Remote Phonebook URL & AUTOP Manual Update Server URL
Preference	Busy Forward	Disabled •	63: The rest of input boxes
Call Feature	Target Number		Warning :
Voice	On Code		Field Description :
Key Dipplay	Off Code	Disphlad	
кеу/ыбріау	No Answer Porward		Submit Shortcut
Ringtones	Target Number		
Tones	On Code	, 	
Dial Plan	Off Code		
Action URL	Call Waiting		
Disas Disale	Call Waiting Enable	Enabled	
• РПОПЕВООК	Call Waiting Tone	Enabled	
▶ Socuritu	Auto Redial		Note :
· Security	Auto Redial	Disabled •	Max length of characters for input
	Auto Redial Interval	10 (1~300s)	box: 255: Broadsoft Phonebook server
	Auto Redial Times	3 (1~100)	address
	DND		AUTOP Manual Update Server URL
	DND	Disabled •	63: The rest of input boxes
	Return Code When DND	486(Busy Here) ▼	Warning :
	DND On Code		Field Description :
			Cubmit Charteut
	Call Park	Dischlad	Submit Cancel
	Target	Disabled	
	Taiget		
	Activo	Enabled	
	Intercom Mute	Disabled T	
	HotLine	Dischlad	
	Number	Disabled	
	Delay Time	4 (0~5s)	
	Remote Control		
	Allowed Access IP List		
	Keypadlock		
	Keypad Lock	Disabled	
	Kawad Lialack RIN	(0.15)	
	Keypad onlick Pill	0~13)	
	Kou As Sond		
	Key As Send	#	
	Others	π ,	
	Return Code When Refuse	486(Busy Here)	
	Auto Answer Delay	0 (0~5s)	
	,		
	Submit	Cancel	



Call Feature Configuration		
Field Name	Explanation	
Forward Transfer	To display and configure Forward setting.	
	There are three types of forward: Always	
	Forward, Busy Forward and No Answer Forward.	
	Always Forward: Any incoming call will be	
	forwarded in any situation.	
	Busy Forward: An incoming call will be	
	forwarded if IP phone is busy.	
Call Waiting	To enable or disable Call Waiting.	
	Call Waiting Enable: If enabled, it allows IP phones to	
	receive a new incoming call when there is already an	
	active call.	
Auto Redial	Auto redial allows IP phones to redial an unsuccessful call	
	for designated times within designated interval.	
	Auto Redial: To enable or disable auto redial feature.	
	Auto Redial Interval: Determine the interval between two	
	consecutive attempts.	
	Auto Redial Times: Determine how many times to redial.	
DND	DND (Do Not Disturb) allows IP phones to ignore any	
	incoming calls.	
	Return Code when DND: Determine what response code	
	should be sent back to server when there is an incoming	
	call if DND is on.	
	DND on Code: The Code is used to turn on DND on	
	server's side; if configured, IP phone will send a SIP	
	message to server to turn on DND on server side if you	
	press DND when DND is off.	
	DND Off Code: The Code is used to turn off DND on	
	server's side; if configured, IP phone will send a SIP	
	message to server to turn off DND on server side if you	
	press DND when DND is on.	
Call Park	Call park allows users to park a call to a special extension	
	and then retrieve it via any other phone within the same	
	system.	
	Active: To enable or disable call park feature.	
	Account: To determine which account to take effect.	
	larget: To configure a designated target extension.	
	Please consult with your telephony system	
	administrator for special extension on your	
	Note system.	
Intercom	Intercom allows user to establish a call directly with the	



	callee.		
	Active: To enable or disable Intercom feature.		
	Intercom Mute: If enabled, once the call established, the		
	callee will be muted.		
Hot Line	Hot Line allows user to call out the defined number		
	automatically without dialing any number.		
	Active: To enable or disable Hot Line feature.		
	Number: The number you want to dial out automatically.		
	Delay Time: The delay time before calling out.		
Remote Control	Remote Control allows specific host to interact with IP		
	phone by sending HTTP or HTTPS request. The specific		
	action could be answering an incoming call, hang up an		
	ongoing call and so on.		
	Allowed Access IP List: To configure the allowed host		
	address.		
	For now, IP phone can only support IP address,		
	IP address list and IP address pattern as		
	Note allowed hosts		
Key As Send	Key As Send allows you to disable send key or assign		
	pound key as send key.		
Others	Return Code When Refuse: Allows user to assign specific		
	code as returned code to SIP server when an incoming		
	call is rejected.		
	Auto Answer Delay: To configure delay time before an		
	incoming call is automatically answered.		



5.11 Phone / Voice

PLAN Networking & Cor	JET High	Definition Vo	IP Phone	/IP-1010PT
► Status	Voice			LogOut
 Account Network 	Echo Canceller Echo Canceller	Enabled	T	Note : Max length of characters for input
▼ Phone	VAD	Disabled Enabled	T T	box: 255: Broadsoft Phonebook server address
Time/Lang	Jitter Buffer			127: Remote Phonebook URL & AUTOP Manual Update Server URL
Preference Call Feature	Jitter Type Min Delay	Fixed	• (0~1000ms)	63: The rest of input boxes
Voice	Nominal Delay Max Delay	120 300	(0~1000ms) (0~1000ms)	Field Description :
Key/Display	Mic Volume		X	Submit Shortcut
Ringtones	Handset Volume	8	(1~15)	
Dial Plan	Headset Volume Hand Free Volume	8	(1~15)	
Action URL	Submit	Can	cel	

Voice Configuration		
Field Name	Explanation	
Echo Canceller	Echo Canceller: To remove acoustic echo from a voice	
	communication in order to improve the voice quality.	
	VAD (Voice Activity Detection): Allows IP phone to detect	
	the presence or absence of human speech during a call.	
	When detecting period of "silence", VAD replaces that	
	silence efficiently with special packets that indicate	
	silence is occurring. It can facilitate speech processing,	
	and deactivate some processes during non-speech	
	section of an audio session. It can avoid unnecessary	
	coding or transmission of silent packets in VoIP	
	applications, saving on computation and network	
	bandwidth.	
	CNG (Comfort Noise Generation): Allows IP phone to	
	generate comfortable background noise for voice	
	communications during periods of silence in a	
	conversation. It is a part of the silence suppression or	
	VAD handling for VoIP technology. CNG, in conjunction	
	with VAD algorithms, quickly responds when periods of	
	silence occur and inserts artificial noise until voice activity	
	resumes. The insertion of artificial noise gives the illusion	
	of a constant transmission stream, so that background	



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	sound is consistent throughout the call and the listener	
	does not think the line has released.	
Jitter Buffer	Jitter buffer is a shared data area where voice packets can	
	be collected, stored, and sent to the voice processor in	
	even intervals. Jitter is a term indicating variations in	
	packet arrival time, which can occur because of network	
	congestion, timing drift or route changes. The jitter	
	buffer, located at the receiving end of the voice	
	connection, intentionally delays the arriving packets so	
	that the end user experiences a clear connection with	
	very little sound distortion.	
	IP phones support two types of jitter buffers: fixed and	
	adaptive.	
	Fixed: Add the fixed delay to voice packets. You can	
	configure the delay time for the static jitter buffer on IP	
	phones.	
	Adaptive: Capable of adapting the changes in the	
	network's delay. The range of the delay time for the	
	dynamic jitter buffer added to packets can be also	
	configured on IP phones.	
Mic Volume	To configure Microphone volume for headset, handset or	
	speaker mode.	



5.12 Phone/Key/Display

PLAN Networking & Co	NET High Definition VolP Ph	ONC VIP-1010PT
		LogOut
► Status	Key/Display	Help
Account	Soft Kau	
	Kev Tyne Lahel Value	Note : Max length of characters for input
Network	Soft Key 1 History T	box:
▼ Phone	Soft Key 2 Book T	255: Broadsoft Phonebook server
Time/Lang	Soft Key 3 DND V	address 127: Remote Phonebook URL &
Preference	Soft Key 4 Menu V	AUTOP Manual Update Server URL 63: The rest of input boxes
Call Feature	Function Key	Warning :
	Key Type Value	indrining -
Voice	OK Status 🔻	Field Description :
Key/Display	Cancel N/A 🔹	Submit Shortcut
Rinatones	Forward Fwd	Submit Cancel
4	Book T	
Tories	RD Redial	
Dial Plan	Mute N/A V	
Action URL	Others	
▶ PhoneBook	Backlight Intensity 4	
▶ Upgrade	Backlight Time 20 🔹	
▶ Security	Submit	

TIME & DATE	
Field Name	Explanation
Soft Key	Allows user to assign specific feature to the designated
	soft keys.
	For softkey, the available features list:
	DND, Menu, MSG, Status, Book, Fwd, PickUp, Group,
	PickUp, Intercom, Speed Dial, History, Favorites, Redial,
	CallReturn and HotDesking.
Function Key	Allows user to assign specific feature to the designated
	function keys.
	For function keys, the available features list:
	N/A, DND, Menu, MSG, Status, Book, Fwd, PickUp, Group
	PickUp, Intercom, Speed Dial, History, Favorites, Redial,
	CallReturn and HotDesking.
Others	Backlight Intensity: To adjust the backlight intensity of
	Phone UI.
	Backlight Time: To adjust backlight on timer; once
	expired, the backlight of Phone UI will go off.



5.13 Phone / Ring Tones

PLAN Networking & Con		igh Defin	nition VolP Pl	hone we	-1010PT
					LogOut
Status	Ringtones				Help
Account	All Ringtones				Note -
► Network	Upload(Max Total Siz	e: 100K)	Upload No file		Max length of characters for
▼ Phone	Uploaded Ringtones		· · · · · · · · · · · · · · · · · · ·		255: Broadsoft Phonebook server address
Time/Lang			Delete		127: Remote Phonebook URL & AUTOP Manual Update Server
Preference	System Ringtones		Bellcore-dr1.wav 🔻		URL
Call Feature	Distinctive Ringers Index	Keyword	Rinatone		63: The fest of input boxes
Voice	0		Ring1.wav	Y	warning -
Key/Display	1		Ring1.wav Ring1.wav	• •	Field Description :
Ringtones	3		Ring1.wav	•	Submit Shortcut
Topos	4		Ring1.wav	*	Submit Cancer
runes	5		Ring1.wav	*	
Dial Plan	7		Ring1.wav	• •	
Action URL	8		Ring1.wav	·	
N DhamaDa ala	9		Ring1.wav	•	
 рпопевоок 	10		Ring1.wav	۲	
Upgrade	11		Ring1.wav	•	
Security		Submit	Cancel		

Ring Tone Configuration			
Field Name E	Explanation		
All Ringtones	Allows user to upload and view ringtone files or delete		
	uploaded ringtone files.		
	Ringtone files must be .wav format and		
	has some specific requirements. Please		
	Note contact Planet technical support team for		
	instructions on how to make ringtone files.		
Distinctive Ringers	Distinctive ringers allow different incoming calls to		
	trigger distinctive ringtones. The IP phone will check		
	"Alert-Info" header inside the incoming "invite" SIP		
	message. And strip out the URL or keyword inside the		
	"Alert-Info" header, from the specific URL or keyword		
	to match or download designated ringtones to play.		



5.14 Phone / Tones

PLAN Networking & Co	NET High	Definition VolP	Phone we	1010PT
				LogOut
Status	Tone			Help
Account	Select Country	Default	*	
	Ring Back			Note :
Network	Dial			Max length of characters for
▼ Dhone	Call Waiting			255: Broadsoft Phonebook server
FIIONE	DTMF 0			address
Time/Lang	DTMF 1			127: Remote Phonebook URL &
Droference	DTMF 2			AUTOP Manual Update Server
Preierence	DTMF 3			63: The rest of input boxes
Call Feature	DTMF 4			
Q-1	DTMF 5			Warning :
VOICE	DTMF 6			Field Description -
Key/Display	DTMF 7			Tield Description -
Print and a	DTMF 8			Submit Shortcut
Ringtones	DTMF 9			Submit Cancel
Tones	DTMF *			
	DTMF #			
Dial Plan	Cubroit	Concol	18	
Action URL	Submit	Cancer		

IAX2 Configuration				
Field Name	Explana	ation		
Tone	Allows user to select a specialized tone sets (classified by			
	countries) or to customize own tones.			
		Available country tone sets are:		
	China, Spain, Luxembourg, Sweden,			
	Taiwan, Belgium, Denmark, Finland,			
	Note	Germany, Netherlands, Norway and		
		Portugal.		



5.15 Phone / Dial Plan

		3.65	
Dial Plan			Help
Rules	Replace Rule 🔻		
Index F	Prefix Replace	Max length of	characters for
1		input box:	B Dhanal
3		255: Broadsot address	τ Phonebook serv
4		127: Remote	Phonebook URL &
6		AUTOP Manu	al Update Server
7		63: The rest	of input boxes
9			
10		warriing -	
Add	Edit	Delete Field Descri	iption :
Rules Modify 3	>>	Submit Sho	ortcut
Area Code		Submit	Cancel
Code			
Min Length	1	(1~15)	
Max Length	1	(1~15)	
	Submit Cancel		
Dial Plan			Help
Dial Plan Rules	Dial Now 🔻	Note :	Help
Dial Plan Rules Index	Dial Now Dial Now Rule	Note : Max length of	Help characters for
Dial Plan Rules Index 1 2	Dial Now 💌 Dial Now Rule	Note : Max length of input box: 255: Broaded	Help characters for
Dial Plan Rules Index 1 2 3 4	Dial Now Dial Now Rule	Note : Max length of input box: 255: Broadsof address	Help characters for t Phonebook serv
Dial Plan Rules Index 1 2 3 4 5	Dial Now Cial Now Rule	Note : Max length of input box: 255: Broadsol address 127: Remote AUTCO Manu	Help characters for t Phonebook serv Phonebook URL 8 al Update Server
Dial Plan Rules Index 1 2 3 4 5 6 6	Dial Now Dial Now Rule	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manu URL	Help characters for t Phonebook serv Phonebook URL & al Update Server
Dial Plan Rules Index 1 2 3 4 5 6 7 8	Dial Now Clai Now Rule	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manu URL 63: The rest of	Help characters for t Phonebook serve Phonebook URL & al Update Server of input boxes
Dial Plan Rules Index 1 2 3 4 5 6 7 8 9 9	Dial Now Cial Now Rule	Note : Max length of input box: 255: Broadsol address 127: Remote AUTOP Manu- URL 63: The rest of Warning :	Help characters for it Phonebook serve Phonebook URL & al Update Server of input boxes
Dial Plan Rules Index 1 2 3 4 5 6 7 7 8 9 10 10	Dial Now Dial Now Rule Edit	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manu. URL 63: The rest of Warning : Delete	Help characters for it Phonebook servi Phonebook URL & al Update Server of input boxes
Dial Plan Rules Index 1 2 3 4 5 6 7 8 9 10 Add Dial Now Dela	Dial Now Dial Now Rule	Note : Max length of input box: 255: Broadsot address 127: Remote AUTOP Manu. URL 63: The rest of Warning : Delete Field Descri	Help characters for t Phonebook serve Phonebook URL & al Update Server of input boxes ption :
Dial Plan Rules Index 1 2 3 4 5 6 7 8 9 10 Control Add Dial Now Dela 1	Dial Now Dial Now Rule Edit V (0~15s)	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manue URL 63: The rest u Barring : Pelete Pelete Field Descri Submit Sho	Help characters for t Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes ption :
Dial Plan Rules Index 1 2 3 4 5 6 7 8 9 10 Add Dial Now Delat 1 2	Dial Now Dial Now Rule Dial Now Rule Edit y (0~15s)	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manue URL 63: The rest of 63: The rest of 63: The rest of Warning : Field Descri Submit Sho Submit Sho	Help characters for it Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes ption : cancel
Dial Plan Rules Index 1 2 3 4 5 6 7 8 9 10 Dial Now Delate 1 1 2 3 6 7 8 9 10 Add	Dial Now Dial Now Rule Dial Now Rule Edit (0~15s) >>	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manu URL 63: The rest of Warning : Delete Field Descri Submit Sho	Help characters for t Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes ption : prtcut Cancel
Dial Plan Rules Index 1 2 3 4 5 6 7 8 9 10 Dial Now Dela 1 1 Rules Modify : Area Code	Dial Now Dial Now Rule Edit (0~15s) >>	Note : Max length of input box: 255: Broadsol address 127: Remote AUTOP Manu- URL 63: The rest of Warning : Delete Field Descri Submit Sho	Help characters for it Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes ption : prtcut Cancel
Dial Plan Rules Index 1 2 3 4 4 5 6 6 7 8 9 10 8 9 10 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0	Dial Now Dial Now Rule Edit V (0~15s) >>	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manu. URL 63: The rest of 63: The rest of 63: The rest of 63: The rest of 64: The rest	Help characters for it Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes iption : <u>cancel</u>
Dial Plan Rules Index 1 2 3 4 5 6 6 7 8 9 10 Add Dial Now Dela 1 1 Rules Modify : Rules Modify : Area Code Code Min Length Max Length	Dial Now Dial Now Rule Edit	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manu. URL 63: The rest of Warning : Delete Field Descri Submit Sho Submit ((1~15) (1~15)	Help characters for it Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes ption : cancel
Dial Plan Rules Index 1 2 3 4 5 6 7 8 9 10 Add 11 12 13 4 5 6 7 8 9 10 Add 11 12 Add 13 Add 14 15 10 Add 10 Add 11 12 Rules Modify I Area Code Max Length	Dial Now Dial Now Rule Dial Now Rule Edit y (0~15s) >>	Note : Max length of input box: 255: Broadsof address 127: Remote AUTOP Manu URL 63: The rest of Warning : Delete Field Descri Submit Sho Submit Sho Submit (1/~15) (1/~15)	Help characters for t Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes ption : <u>ortcut</u> <u>Cancel</u>
Dial Plan Rules Index 1 2 3 4 5 6 6 7 8 9 10 10 10 10 10 10 10 10 10 10 10 10 10	Dial Now Dial Now Rule Dial N	Image: Stress of the stress	Help characters for t Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes ption : <u>cancel</u>
Dial Plan Rules Index 1 2 3 4 5 6 7 8 9 10 Add 11 2 3 4 5 6 7 8 9 10 Add 1 Code Min Length Max Length	Dial Now Dial Now Rule Dial Now Rule Edit y (0~15s) >> Submit Cancel	Image: Stream of the stream	Help characters for t Phonebook serve Phonebook URL & al Update Server of input boxes of input boxes ption : <u>cancel</u>

or edit.



Rules Modified	Allow user to modify selected rules information To		
	replace rules, you can modify related account, prefix		
	and replacement.		
Area Code	Area codes are also known as NPAs (Numbering Plan		
	Areas). They usually indicate different geographical		
	areas within one country. If entered numbers match		
	the predefined area code rule, the IP phone will		
	automatically prefix outgoing number with area code.		
	There is only one area code rule		
	Note supported.		
Dial Now			
Rules	Allow user to select Replace rule or Dial-now to display		
	or edit.		
Dial Now Delay	Allow user to configure dial now delay time for dial		
	now. It means user can configure the IP phone to dial		
	out the phone number automatically after the		
	designated delay time if it matches any dial now rule.		
Rules Modify	Allow user to modify selected rules information, for		
	dial-now rule, user can modify related account, Dial		
	now Rule itself.		
Area Code	Area codes are also known as NPAs (Numbering Plan		
	Areas). They usually indicate different geographical		
	areas within one country. If entered numbers match		
	the predefined area code rule, the IP phone will		
	automatically prefix outgoing number with area code.		
	There is only one area code rule		
	Note supported.		



5.16 Phone –>Action URL

PLAN Networking & Co	NET High Def	finition VolP Phone	VIP-1010PT
	7		LogOut
Status	Action URI		Halp
Account	Action IRI		Top
▶ Network	Active [Disabled •	Note : Max length of characters for
▼ Phone	Registered		255: Broadsoft Phonebook server
Time/Lang	Unregistered Registered Failed		address 127: Remote Phonebook URL & AUTOP Manual Undate Server
Preference	Off Hook		URL
Call Feature	On Hook Incoming Call		63: The rest of input boxes
Voice	Outgoing Call Established		Field Description :
Key/Display	Terminated		
Ringtones	Open DND Close DND		Submit Shortcut Submit Cancel
Tones	Open Always FWD		
Dial Plan	Close Always FWD Open Busy FWD		
Action URL	Close Busy FWD		
▶ PhoneBook	Open No Answered FWD Close No Answered FWD		
▶ Upgrade	Transfer Call Blind Transfer		
Security	Attended Transfer		
	Hold		
	UnHold		-
	Mute UnMute		
	MissedCall		
	IP Changed		_
	FWD Incoming Call		_
	Reject Incoming Call		_
	Answer New Call		
	Transfer Finished		
	Transfer Failed		
	Idle To Busy		
	Busy to tule		
	Submit	Cancel	

DIAL PEER	
Field Name	Explanation
Action URL	To display and configure Action URL settings.
	Setup Completed: When the IP phone completes startup.
	Registered: When the IP phone successfully registers an
	account.
	Unregistered: When the IP phone logs off the registered
	account.
	Register Failed: When the IP phone fails to register an
	account.
	Off Hook: When the IP phone is off hook.
	On Hook: When the IP phone is on hook.





Incoming Call: When the IP phone receives an incoming call
Outgoing Call: When the IP phone places a call
Established: When the IP phone establishes a call
Terminated: When the IP phone terminates a call
Open DND: When the IP phone enables the DND mode
Close DND: When the IP phone disables the DND mode.
Close DND. When the IP phone disables the DND mode.
always forward.
Close Always Forward: When the IP phone disables the
always forward.
Open Busy Forward: When the IP phone enables the busy
forward.
Close Busy Forward: When the IP phone disables the busy
forward.
Open No Answer Forward: When the IP phone enables the
no answer forward.
Close No Answer Forward: When the IP phone disables
the no answer forward
Iransfer Call: When the IP phone transfers a call.
Blind Iransfer: When the IP phone blind transfers a call.
Attended Iransfer: When the IP phone performs the
semi-attended/attended transfer.
Hold: When the IP phone places a call on hold.
UnHold: When the IP phone retrieves a hold call.
Mute: When the IP phone mutes a call.
UnMute: When the IP phone un-mutes a call.
Missed Call: When the IP phone misses a call.
IP Changed: When the IP address of the IP phone
changes.
FWD Incoming Call: When the IP phone forwards an
incoming call.
Reject Incoming Call: When the IP phone rejects an
incoming call.
Answer New Call: When the IP phone answers a new call.
Transfer Finished: When the IP phone completes to
transfer a call.
Transfer Failed: When the IP phone fails to transfer a call.
Idle To Busy: When the state of the IP phone changes
from idle to busy.
Busy To Idle: When the state of phone changes from busy
to idle.



5.17 Phone Book->Local Phone Book

PLAN Networking & Con	High Definition VolP Phone	IO10PT
		LogOut
► Status	Local Rook	
Account		нер
		Note : May longth of characters for
 Network 	Search Keset	input box;
Phone	Dial Hand Up	255: Broadsoft Phonebook server address
PhoneBook	index warne once warn woolle warn other warn king Group	127: Remote Phonebook URL &
Local Book	3	URL
Domoto Dool/	4	63: The rest of input boxes
Kennote Book	6	Warning :
Call Log	7	Field Description :
LDAP	9	•
Broadsoft	10 Page 1 Prev Next Move To All Contacts Delete All	
▶ Upgrade	Contact Setting	
Recuritu	Name Office Num	
security	Mobile Num Other Num	
	Group Default	
	Add Edit Cancel	
	Group	
	Index Name Ring Description	
	2	
	4	
	5 Delete all	
	Group Setting	
	Name	
	Ring Auto V	
	Description	
	Add Edit Cancel	
	Import/Export	
	Contact Upload No file (.XML)	
	Import Export Cancel	
	Black List Upload No file (.XML)	
	Import Export Cancel	

AUDIO Configuration	
Field Name	Explanation
Contact	To display and select local contact type.
	All Contacts: To display or edit all local contacts.
	Favorites: To display or edit favorite contacts.
	Black List: To display black list contacts.
Search	To search designated contacts from local phonebook.
Dial	To dial out a call or hang up an ongoing call from Web UI.



		For this feature, you need to have the	
	remote control privilege to control IP		
		phone via Web UI. Please refer to section	
	Note "Remote Control" on the Web		
		UI->Phone->Call Feature page.	
Group	To display or edit Group contacts.		
Group Setting	To display or change Group name, related ringtone or		
	descript	ion.	

5.18 Phone Book->Remote Phone Book

PLAN Networking & Com	IET	High Definition	VolP Phone	1010PT
Status	Remote Book			Help
 Network 	Remote Book Index 1	Local Book URL	Local Book Name	Note : Max length of characters for input box:
 Phone PhoneBook 	3			255: Broadsoft Phonebook server address 127: Remote Phonebook URL & AUTOP Manual Update Server
Local Book <mark>Remote Book</mark>	5	Submit	Cancel	URL 63: The rest of input boxes Warning :
Call Log LDAP				Field Description :
Broadsoft				Submit Cancel

FEATURE			
Field Name	Explanation		
Remote	To display and configure Remote Book settings.		
Book	Index: To select desired Remote Book item to display and		
	configure.		
	Local Book URL: To configure remote book server address		
	Local Book Name: To configure display remote book name on Phone		
	UI		
	IP phone supports at most 5 remote		
	books. Please refer to your administrator		
	or Planet Technical Support team for how		
	Note to establish a remote book server and how		
	to create remote book xml file.		



5.19 Phone Book->Call log

PLAN Networking & Com	High Definition VolP Phone	1010PT
		LogOut
Status	Call Log	Help
Account	Call History All V Hand Up	Nata -
Network	Index Type Date Time Local Identity Name Number 🗖	Max length of characters for
Network	1 Received 2014-07- 11:31:34 204@192.168.1.21 201 201@192.168.1.21	input box:
Phone	2 Forwarded 2014-07- 2 Forwarded 07 11:31:09 204@192.168.1.21 201 201@192.168.1.21	255: Broadsoft Phonebook server address
PhoneBook	3	127: Remote Phonebook URL &
14 - 14 - 14	4 8	AUTOP Manual Update Server
Local Book	5	63: The rest of input boxes
Remote Book	6 0	
Contraction of the	7	Warning :
Call Log		Field Description -
LDAP	10	Field Description -
14 14 2012	11	
Broadsoft	12 0	
▶ Upgrade	13	
	14	
Security	15	
	Page 1 V Prev Next Delete Delete All	

Call Log			
Field Name	Explanation		
Basic Setting			
Call History	To display call history records.		
	Available call history types are All calls, Dialed calls,		
	Received calls, Missed calls and Forwarded calls.		
	HangUp: To click to hang up ongoing call on the IP phone.		
	For "HangUp" feature, you need to have		
	the remote control privilege to control IP		
	phone via Web UI. Please refer to section		
	Note "Remote Control" on the Web		
	UI->Phone->Call Feature page.		



5.20 Phone Book->LDAP

PLA Networking & Co	NET High L	Definition VolP	Phone wa	1010PT
				LogOut
Status	LDAP			Help
Account	LDAP			Nieto -
Network	Name Filter Number Filter			Max length of characters for
▶ Phone	Server			255: Broadsoft Phonebook server
PhoneBook	Port Base DN		(1~65535)	address 127: Remote Phonebook URL & ALITOP Manual Lindate Server
Local Book	User Name Password	•••••		URL 63: The rest of input boxes
Remote Book	Name Attribute			141
Call Log	Number Attribute Display Name			Warning -
LDAP	Max Hits	50	(1~500)	ricid Beschpaorre
Broadsoft	Search Delay Time	1000	(200~3000)ms	Submit Shortcut Submit Cancel
▶ Upgrade	Submit	Cancel		

LDAP	
Field Name	Explanation
LDAP	To display and configure LDAP phonebook settings.
	Name Filter: The settings are used to tell LDAP server
	what name attributes to search.
	Number Filter: The settings are used to tell LDAP server
	what number attributes to search.
	Server: To configure LDAP server's address.
	Port: To configure LDAP server's port.
	Base DN: To configure searching base DN on LDAP server.
	User Name: To configure user name for accessing LDAP
	server.
	Password: To configure password for accessing LDAP
	server.
	Name Attribute: To configure which name attributes
	should be feedback from LDAP server.
	Number Attribute: To configure which number attributes
	should be fedback from LDAP server.
	Display Name: To configure display name on Phone UI
	when there is any searching result from LDAP server.
	Max. Hits: To configure the maximum size of result
	response from LDAP server.
	Search Delay Time: To configure delay time before
	initiating LDAP searching request after you input a value
	from Phone UI.



5.21 Phone Book->BroadSoft

PLAN Networking & Cor	High Definition VolP Phone	/IP-1010PT
▶ Status		LogOut
orartas	Broadsoft	Help
Account	Broadsoft PhoneBook	Note -
▶ Network	PhoneBaok Item Item1 Display Name	Max length of characters for input box:
▶ Phone	Server Address	255: Broadsoft Phonebook server
▼ PhoneBook	Server Port (1~65535) User Name	auuress 127: Remote Phonebook URL & AUTOP Manual Update Server
Local Book	Password ••••••	URL 62: The rest of input haves
Remote Book	Submit	Warning :
Call Log		Field Description :
LDAP		
Broadsoft		Submit Shortcut Submit Cancel
▶ Upgrade		
► Security		

Broadsoft			
Field Name	Explanation		
Broadsoft PhoneBook	To display and configure Broadsoft PhoneBook settings.		
	PhoneBook Item: To select specific item to configure.		
	Display Name: The name displayed on IP phone's LCD		
	screen when accessed via Phone UI.		
	Server Address: Broadsoft PhoneBook server's		
	address.		
	Server Port: Broadsoft PhoneBook server's port.		
	User Name: Username used to access Broadsoft		
	PhoneBook server.		
	Password: Password used to access Broadsoft		
	PhoneBook server.		
	IP phone supports at most 5 Broadsoft		
	PhoneBook items. For Broadsoft Phone		
	Book's server address, port, username and		
	password, you need to consult your		
	Broadsoft service provider for further		
	information.		



5.22 Upgrade->Basic

PLAN Networking & Co	NET High De	finition VolP Phone	VIP-1010PT
Ctatue			LogOut
Status	Upgrade-Basic		Help
Account	Upgrade	Bowse	Note -
Network		Submit Cancel	Max length of characters for input
▶ Phone	Firmware Version Hardware Version	50.141.2.15 50.0.1.0.0.0.0	box: 255: Broadsoft Phonebook server
i none	Reset To Factory Setting	Submit	address
PhoneBook	Reboot	Submit	127: Remote Phonebook URL & AUTOP Manual Update Server URL
▼ Upgrade			63: The rest of input boxes
Basic			Warning :
Advanced			Field Description :

Upgrade Configuration	
Field Name	Explanation
Upgrade	To select upgrading ROM file from local or a remote
	server automatically.
	Note: Please make sure it's the right file format for the
	right model.
Firmware Version	To display firmware version; firmware version starts
	with MODEL name.
	For example, VIP-1010PT firmware version should be
	like 50.xxx.xxx.xxx.
Hardware Version	To display Hardware version.
Reset to Factory	To enable you to reset IP phone's setting to factory
Setting	settings.
Reboot	To reboot IP phone remotely from Web UI.



5.23 Upgrade->Advanced

PLANET High Definition VolP Phone VIPETOTOPT			
			LogOut
Status	Upgrade-Advanced		Help
Account	PNP Option		
Network	PNP Config	Enabled	Note : Max length of characters for input
▶ Phone	DHCP Option	(128~254)	box: 255: Broadsoft Phonebook server address
PhoneBook	Manual Update Server		127: Remote Phonebook URL &
▼ Upgrade	URL User Name		AUTOP Manual Update Server URL 63: The rest of input boxes
Basic	Password Common AES Key	•••••	Warning :
Advanced	AES Key(MAC)	•••••	Field Description :
► Security	AutoP Mode Schedule AutoP Immediately Clear MD5 Export Autop Template Submit Cancel	Power On Sunday 22 Hour(0~23) AutoProvision Submit Export	Submit Shortcut Submit Cancel
	System Log LogLevel Export Log PCAP PCAP Others Config File(.tgz)	3 Export Start Stop Export Browse Export (Encrypted) Export (Unencrypted)	

Upgrade- Advance		
Field Name	Explanation	
PNP Option	To display and configure PNP setting for Auto	
	Provisioning.	
	PNP: Plug and Play; Once PNP is enabled, the phone will	
	send SIP subscription message to PNP server	
	automatically to get Auto Provisioning server's address.	
	By default, this SIP message is sent to multicast	
	address 224.0.1.75 (PNP server address by standard).	
DHCP Option	To display and configure custom DHCP option.	
	DHCP option: If configured, IP phone will use	
	designated DHCP option to get Auto Provisioning	
	server's address via DHCP. This setting requires DHCP	
	server to support the corresponding option.	
Manual Update	To display and configure manual update server's	
Server	settings.	
	URL: Auto provisioning server address.	



	User name: Configure if server needs a username to
	access, otherwise left blank.
	Password: Configure if server needs a password to
	access, otherwise left blank.
	Common AES Key: Used for IP phone to decipher
	common Auto Provisioning configuration file (For the
	VIP-1010PT, this configuration file is
	r0000000053.conf.).
	AES Key(MAC): Used for IP phone to decipher
	MAC-oriented auto portioning configuration file(for
	example, file name could be 00304f8888888.conf if IP
	phone's MAC address is 00304f888888.).
	Note: AES is one of many encryptions. It should be
	configured only if the configure file is ciphered with AES,
	otherwise left blank.
AutoP	To display and configure Auto Provisioning mode
	settings.
	This Auto Provisioning mode is actually
	self-explanatory.
	For example, mode "Power on" means IP phone will go
	to do Provisioning every time it powers on.
System Log	To display syslog level and export syslog file.
	Syslog level: From level 0~7. The higher level means
	the more specific syslog is saved to a temporary file.
	By default, it's level 3.
	Export Log: Click to export temporary syslog file to local
	PC.
PCAP	To start, stop packets capturing or to export captured
	Packet file.
	Start: To start capturing all the packets file sent or
	received from IP phone.
	Stop: To stop capturing packets.
	Note: IP phone will save captured packets file to a
	temporary file; this file maximum size is 1M
	(megabytes), and will stop capturing once this
	maximum size is reached.
Others	To display or configure other features from this page.
	Configure file: To export or import configure file for IP
	phone.


5.24 Security->Basic

PLAN Networking & Co	NET High Definition VolP Phone	VIP-1010PT
 Status Account Network Phone PhoneBook 	Security-Basic Web Password Modify User Name admin Current Password New Password Confirm Password	Help Note : Max length of characters for input box: 255: Broadsoft Phonebook server address 127: Remote Phonebook UBL & ALITOP Manual
 Upgrade Security Basic Advanced 	Submit Cancel	Update Server URL 63: The rest of input boxes Warning : Field Description : Submit Shortcut Submit Cancel

Upgrade- Advance			
Field Name	Explanation		
Web Password	To modify user's password.		
Modify	Current Password: The current password you used.		
	New Password: Input new password you intend to use.		
	Confirm Password: Repeat the new password.		
	For now, IP phone can only support user administrator		
	mode.		



5.25 Security->Advanced

PLAN Retworking & Comm	High Definition Vo	DIP Phone VIP-1010PT
 Status Account Network Phone PhoneBook 	Advanced Web Server Certificate Index Issue To Issuer Ex 1 Ringslink Ringslink Sun Jun 2 Web Server Certificate Upload Erowse Subr Client Certificate	Help
 Upgrade Security Basic Advanced 	Client Certificate Upload Index Auto Erowse	Expire Time Warning : Field Description :

Sections	Description		
Web Server Certificate	To display or delete Certificate which is used when IP		
	phone is connected from any incoming HTTPs request.		
	The default certificate could not be deleted.		
Web Server Certificate	To upload a certificate file which will be used as server		
Upload	certificate.		
Client Certificate	To display or delete Certificates which is used when IP		
	phone is connecting to any HTTPs server.		
Client Certificate	To upload certificate files which is used as client		
Upload	certificate.		



6 Appendix

6.1 Digit-character Map Table

Keypad	Character	Keypad	Character
	1 @	Pors	7 P Q R S p q r s
2 ABC	2 A B C a b c	8 FV	8 T U V t u v
(Ber	3 D E F d e f	9 WXYZ	9 W X Y Z w x y z
4 GH1	4 G H I g h i	*.	*/.
5. JKL	5 J K L j k I	0	0
6 MN0	6 M N O m n o	# send	#/SEND

6.2 Frequently Asked Questions List

Q1: No operation after power on?
 A1: Check if the power adapter is properly connected. If applicable, check if the PoE (Power over Ethernet) switch behind the IP phone is set correctly.
Q2: No dial tone?
A2: Check if the handset cord is properly connected.
Q3: Cannot make a call?
A3: Check the status of User SIP registration status or contact user administrator, supplier, or ITSP for more information or assistance.
Q4: Cannot receive any phone call?
A4 : Check the status of user SIP registration status, or contact user administrator, supplier, or ITSP for more information or assistance
Q5: No voice during an active call?
A5: Check if the servers support the current audio codec type, or contact user administrator, supplier, or ITSP for more information or assistance.





Q6: Cannot connect to the configuration website?

A6: Check if the Ethernet cable is properly connected.

Check if the URL is right; the format of URL is: http:// the Internet port IP address. Check if user firewall/NAT settings are correct.

Check if the version of IE is IE8, or use other browser such as Firefox or Mozilla, or contact user administrator, supplier, or ITSP for more information or assistance.

Q7: Forget the password?

A7: Default password of website and menu is null.

If user changes the password and then forget it, or user cannot access to the configuration website or the menu items need password.

Solution:

Factory default: press the Menu button and choose 16Factory Default and then a notice will appear. Choose OK by using the corresponding softkey button.

If user chooses the factory default, user will return the phone to the original factory settings and will erase ALL current settings, including the directory and call logs.





	1000					
Home	Extens			Edit		Х
Operator		General				
asic		SIP:	1	IAX2:		
Extensions	Exten	Name:	204	Extension:	204	
• Trunks		Password:	123456	Outbound CID:		_
Outbound Poutes	New	DialPlan:	DialPlan1	 Analog Phone: 	None 🔻	
- Outboaria Roates	Exten	Voicemail			1001	
Indound Control		Voicemail:		VM Password:	1234 av.	
Advanced		Other Ontio	. L.	Email(Fax/Vuicema		-
Network Settings		Web Manage	er: 🗹 Aae	nt: 📃 Call Waitin	a: 🖉	
Security		Allow Being	Spied: 📃 Pick	up Group: 1		
Report	0 5	Mobility Exte	ension: 🔟 Mot	oility Extension Number:		i.
System		NAT:	ys Tror		CDTD	
				Bormit ID:	SKIP:	
		Video Ontio	RFC2833 •	Permit IP	5	
	1	Video Cally				i
		Audio Codeo	CS			
	1	🗹 alaw 🗹 ula	aw 🗆 G.722 🗹 G.	729 🛛 G.726 🔍 GSM 🔍 Sp	Deex	
				Save Cancel		
		8				- Internet

Display Label = Choose any Display message you want.

Display name: Choose any Display name you want.

Register Name: 204 (in this case)

User name : 204 (in this case)

Password: 123456 (in this case)

Server IP: 192.168.1.21

After saving, user can check the register status in "Status" item.



PLAN Networking & Con	NET High D	efinition Voll	Phone 🛛	IP-1010PT
▶ Status	Account-Basic			LogOu Help
 Account Basic 	SIP Account Status	Registered		Note : Max length of characters for input
Advanced	Account Active Display Label	Enabled IPX-2100-204		box: 255: Broadsoft Phonebook server address
NetworkPhone	Display Name Register Name	204		127: Remote Phonebook URL & AUTOP Manual Update Server URL 63: The rest of input boxes
PhoneBook	User Name Password	•••••		Warning :
UpgradeSecurity	SIP Server 1 Server IP	192.168.1.21	Port 5060	Field Description : Submit Shortcut
	Registration Period SIP Server 2	1800	(30~65535s)	Submit Cancel
	Server IP Registration Period	1800	Port 5060 (30~65535s)	