



Business / Professional PoE IP Phone

VIP-350PT / VIP-550PT

User's manual

Version 1.0.0

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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.

WEEE Warning



To avoid the potential effects on the environment and human health as a result of the presence of

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Revision

User's Manual for PLANET Business/Profreesional PoE IP Phone: Model: VIP-350PT/VIP-550PT Rev: 1.0 (2007, October) Part No. EM-VIP350PT 550PTV1

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Chapter 1

Overview

Combining the cutting edge of Voice over IP and Internet telephony manufacturing experience, PLANET now introduces the latest member of mainstream business series desktop IP phone family: the VIP-350PT/550PT.

To bring the most satisfaction to customers, the VIP-350PT/550PT is the ideal choice for a business to deploy by using IP PBX service. The standard features of the VIP-350PT/550PT include two-line, dual 10/100 switched Ethernet ports and integrated IEEE power over Ethernet (802.3af) circuitry for offering a choice of powering and cabling options to help reduce cabling expenses and cord clutter.

To give most flexibility to users, the VIP-350PT/550PT platform contains a large graphic LCD with Back light, 4/8 softkeys, 8 fixed function keys and a 5-position navigation key. The PLANET VIP-350PT/550PT desktop phone is engineered to make Easy-to-install communications, cost-effective to deploy, self-contained, service-integrated, intelligent phone features offering and powerful voice processing power as possible. The VIP-350PT/550PT can effortlessly deliver toll voice quality equivalent to the regular VoIP/IP PBX connections utilizing cutting-edge Quality of Service (QoS) capabilities to encompassi IP-TOS/DiffServ, 802.1 p/q VLAN tagging, echo cancellation, comfort noise generation (CNG) and voice compensation technology. Meanwhile, the dual Ethernet interfaces on the IP phone allow users to install in an existing network location without interfering with connections of desktop PC networks.

The VIP-350PT/550PT has streamlined wired IP telephone that provides additional features such as built-in PPPoE/DHCP clients, password-protected machine management, call hold, forwarding, mute, transfer, waiting, pickup, caller ID, peed-dial, 3-way conference, last number redial, incoming message indicator, multiple call appearances and user-intuitive web administration system.

Besides, the VIP-350PT/550PT is the ideal solution for office use as well as installation for Internet Telephony Service Provider (ITSP) from leading vendors. It's the delivery platform for IP voice services that makes benefits from the VoIP technology in business class communications services.

There are models for VIP-350PT/550PT and there are:

VIP-350PT: Busniess PoE SIP IP Phone VIP-550PT: Professional PoE SIP IP Phone

Benefits

- Full-Featured business SIP Desktop Phone
- Easy-to-read and configuration via large LCD screen
- Efficient installation and deployment of PLANET IP PBX solution for professional users

Product Features

- Integrated 802.3af Power over Ethernet support
- Easy-to-read large graphical LCD display
- Dual 10/100 switched Ethernet ports
- Two-line desktop phone for general professional users

- Easy to configure and install
- Instant Message / Voice Mail Integration
- VLAN 802.1p/q / IP-TOS (IP Precedence) / DiffServ
- TFTP/HTTP for Auto Provisioning
- Full-duplex speakerphone / Dedicated RJ-9 headset port
- Reversible base stand / wall mount

VoIP Features

- SIP 2.0 (RFC3261) compliant
- Supports up to 3 service domains
- Interoperability with leading PLANET IP PBX platforms
- Voice codec support: G.711(A-Law, u-Law), G.723.1, G.729 A/B
- In-band, out-of-band DTMF Relay (RFC 2833) and SIP INFO
- Three-way conference / Caller ID / Speed Dial / Shared Line Appearance
- Call Hold / Mute / Pickup / Forward / Transfer / Waiting / Rejection / Screening
- Voice processing: VAD, CNG, AEC, Adaptive Jitter Buffer Management

Package Content

The contents of your product should contain the following items: VoIP IP Phone Power adapter Quick Installation Guide User's Manual CD Reversible base stand

Physical Details

The following figure illustrates the front/rear panel of IP Phone.

Front View



Front Panel of VIP-350PT



Front Panel of VIP-550PT

Keypad Description

1	LCD Display	Menu and all status shall be displayed for users.
2	Speaker	To send out voice of device when usage hands-free.
3	Programmable Keys	User-defined programmable keys for easy feature access. (Only VIP-550PT)
4	【 🞽 】 : Hold	To hold the conversation.
5	[(+++ (] : TRANSFER	To transfer an active call (incoming call answered or outgoing call accepted) to another devices.
6	【 🃫 】 :Redial	Press to dial the last dialed number when the IP Phone is off-hooked.
7	[🚱] :Speed Dial	To make a speed dial call by pressing.
8	【 🛠 】:Menu	To bring out the menu selection while IP Phone is in idle state.
9	Ring Lamp	The red light goes on-off when there is an incoming call.
10	Soft-button	To control context page and sip account switch function button.
11	Navigation Keys	To scroll menu item and phonebook.
12	ОК	To be used as confirm configuration or enter sub-menu.
13	Number Keypad	Use to enter numbers, characters or special characters.
14	[🖂] : MWI	Press this button can enter the voicemail service.
	\$	Message Waiting Indication, MWI: Access to voice mail system
15	【 🔜 】: Mute	
16	【	To switch between the usage of the handset and the speaker devices.
17	Volume	Volume adjustment for ringer, headset, handset, speaker.
18	Micro-Phone	Receive voice of device when usage hands-free.

Physical Interfaces

1		RJ-45 connector, for Internet access, connected directly to Switch/Hub through straight CAT-5 cable.								
	LAN	The LAN interface also can be connected with 802.3af PoE switch or converter for power supply.								
2	PC	RJ-45 connector, to maintain the existing network structure, connected directly to the PC through straight CAT-5 cable.								
3	Reset	Reset to the factory default setting.								
4	12V DC	12V DC Power input outlet.								
5	Handset Jack	RJ-22 connector, for telephone handset.								

Rear view and panel descriptions



VNote

- 1. Machine default IP is http://192.168.0.1. Press RESET button on rear panel over 5 seconds will reset the VoIP Phone Adapter to factory default value.
- (Except speed dial and call forward settings)
- 2. For VIP-350PT/550PT, either PoE or AC adapter can be deployed at one time

Chapter 2 LCD Keypad Opearting & Configurations

Configurations for Address Book

Main Menu

Main Menu keys Function Description

Menu	Enter or exit menu mode.
[\uparrow] and [\downarrow]	Navigate through menu items by up and down navigation keys.
[OK]	Traverse into selected menu or return to previous menu if no specified function.
Back	To return previous menu page.
Soft-buttons	Context-sensitive menus.

1	•	A	d	d	r	е	S	S		B	0	0	k						
2		С	a	1	1		Η	i	S	t	0	r	y						
3		Т	e	r	m	i	n	a	1		S	e	t	t	i	n	g	S	
4		C	a	1	1		F	0	r	W	a	r	d						
5		Р	r	e	f	e	r	e	n	c	e	S							
6		Ν	e	t	w	0	r	k											
7		S	Ι	Р		S	e	t	t	i	n	g	s						
8		S	e	r	v	i	c	e											
9		Α	d	v	a	n	c	e	d										
	Find						Add	l			Del					Back			

Figure 1. IP Phone LCD Main Menu

Address Book

Address Book Keys Function Description

Add	Add new contact onto address book
Find	Search a contact in address book.
Del	Delete contacts from address book.
[0K]	Enter address-book sub-menu.
[\uparrow] and [\downarrow]	Navigate through menu items.

1	•	A	d	d	r	е	S	s		B	0	0	k						
2		С	a	1	1		Η	i	S	t	0	r	y						
3		Т	e	r	m	i	n	а	1		S	e	t	t	i	n	g	S	
4		C	a	1	L		F	0	r	W	а	r	d						
5		Р	r	e	f	e	r	e	n	c	e	S							
6		N	e	t	W	0	r	k											
7		S	Ι	Р		S	e	t	t	i	n	g	S						
8		S	e	r	v	i	c	e											
9		Α	d	v	a	n	c	e	d										
	Find					Add						Del			Back				

Max size: 1,000 entries.

Figure 2. Main Menu

Find

Search for a specific entry on address book. The search criterion is a longest prefix match. If no character is entered, then it will position on the 1st entry on address book

Keys Function Description

Del	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where [Abc] indicates cureent input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Rturn previous page.
[OK]	Start to search.
[\leftarrow] and [\rightarrow]	move cursor one position in edit mode.

S	e	a	r	c	h		Ν	a	m	e	:										
Μ	i	c	h	a	e	1															
	Del		Abc/123					Clear					Back								

Figure 3. Name Search screen

Find. [OK]

Address Book List Format:

- 1. List all contacts in alphanumeric order.
- 2. First line is fixed (not scrollable) and also shows how many contacts in address book. (*VIP-550PT only*)
- 3. Position at the best matched item.
- 4. User may use keypad to jump to the first contact prefixed with entered alphanumeric character.
- 5. Alternatively, user may use navigation key \uparrow and \downarrow to scroll contact list.

6. Max size: 1,000

Keys Function Description

[Call]	Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[Del]	Delete current contact.
[Back]	Reurn without any changes.
[0K]	Edit selected contact.
[\uparrow] and [\downarrow]	Navigate through contacts list.

С	0	n	t	а	c	t	S	(1	4	3)						
Μ	i	c	h	а	e	1												
Μ	i	k	e															
Ν	i	c	k															
Р	a	t	r	i	c	k												
Р	a	u	1															
R	i	c	h	а	r	d												
	Call												Del]	Back	5	

Figure 4. Contact List

View Contact

Screen Field Description

Display	It could consist of all ASCII letters except for left and right angle parentheses, '<' and '>'. You may enter the alias of your SIP AoR, phone number of this port, or the display string of this AoR.
Protocol	Currently, we support only sip and tel URI. Note: protocol is case-sensitive!
Address	The Email-like address of your SIP address-of-record, AoR, with the user part and the domain part. For example, if your SIP AoR is "Handsome Boy <sip:micheal@yourisp.com>", please enter it as "micheal@YourISP.com".</sip:micheal@yourisp.com>

[Call]	Dial to this contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[Save]	Save changes and return
[Back]	Return with previous page.
[0K]	Edit selected field.

D	i	S	р	1	а	у	:	Μ	i	c	h	а	e	1					
Р	r	0	t	0	c	0	1	:	S	i	p								
Α	d	d	r	:	1	2	3	4	5	6	7	a	Ι	S	Р	f	0	0	c
			Call					Save	2								Back	K	

Figure 5. View Contact Screen

Edit Contact.Display

Display consist of all ASCII letters except for left and right angle parentheses, '<' and '>'. You may enter the alias of your SIP AoR, phone number of this port, or the display string of this AoR.

[Del]	Delete on character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[0K]	Save changes and return.
[\leftarrow] and [\rightarrow]	move cursor one position in edit mode.
[\uparrow] and [\downarrow]	move cursor per line in edit mode.

D	Ι	S	р	1	a	y	:														
Μ	Ι	c	h	a	e	1															
	Del				Abc/123				Clear						Back						

Figure 6. Edit Contact Display

Edit Contct.Protocol

Make IP Phone supports only sip and tel URI. Note: protocol is case-sensitive!

Keys Function Description

[Call]	Dial to this contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing 【SPK】 to make a call to selected contact as well.
[tel]/[sip]	Toggle between available protocols, including "sip" and "tel". [tel] indicates current protocol is "tel", and so on.
[Back]	Return to previous page.

D	i	S	p	1	a	у	:	М	i	c ·	h	a	e	1							
Р	r	0	t	0	c	0	I	:	S	1	р										
Α	d	d	r	:	1	2	3	4	5	(a)	Ι	S	P		f	0	0		n	e	t
			Call									t	el/sij	р				Back	K		

Figure 7. Edit Contact Protocol

Edit Contact.Email-Like Address

The Email-like address of your SIP address-of-record, AoR, with the user part and the domain part. For example, if your SIP AoR is "Handsome Boy <sip:michael@YourISP.com>", please enter it as "michael@YourISP.com".

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
\leftarrow and \rightarrow	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cusor per line in edit mode.

Е	m	a	i	1	-	L	i	k	e		Α	d	d	r	e	S	S	:			
1	2	3	4	5	a	Ι	S	Р		f	0	0		n	e	t					
	Del				Abc/123				Clear						Back						

Figure 8. Edit Contact EMail like Address

Add

Screen Field Description

Display	It could consist of all ASCII letters except for left and right angle parentheses, '<' and '>'. You may enter the alias of your SIP AoR, phone number of this port, or the display string of this AoR.
Protocol	Currently, we support only sip and tel URI. Note: protocol is <i>case-sensitive</i> !
Address	The Email-like address of your SIP address-of-record, AoR, with the user part and the domain part. For example, if your SIP AoR is "Handsome Boy <sip:michael@yourisp.com>", please enter it as "michael@YourISP.com".</sip:michael@yourisp.com>

Keys Function Description

[Save]	Add contact into address book and return.
[Back]	Return without any changes.
[OK]	Edit selected field.
[↑] and [↓]	Navigate through itmes.

If reached max allowed entries, show

А	d	d	r	e	S	S	-	В	0	0	k		А	d	d	:				
		Μ	a	х		Е	n	t	r	у	:	5	0	0	!					
]	Back	4	

Otherwise, show

D	i	S	р	1	а	у	:										
Р	r	0	t	0	c	0	1	:	S	i	р						
Α	d	d	r	:													
			Save)											Back	K	

Add Contact.Display

Display consist of all ASCII letters except for left and right angle parentheses, '<' and '>'. You may enter the alias of your SIP AoR, phone number of this port, or the display string of this AoR.

Keys Function Description

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

D	i	S	p	1	a	у	:														
Μ	i	c	h	a	e	1															
	Del				Abc/123				Clear					Back							

Figure 8. Contact Display screen

Add Contact.Protocol

Make IP Phone supports only sip and tel URI. Note: protocol is case-sensitive!

[Save]	Add contact only sip and return.
[tel]/[sip]	Toggle between available protocols, including "sip" and "tel". [tel] indicates current protocol is "tel", and so on.
[Back]	Return without any changes.

D	i	S	р	1	a	y	:	Μ	i	c	h	a	e	1							
Р	r	0	t	0	c	0	1	:	S	i	р										
Α	d	d	r	:	1	2	3	4	5	6	7	a	Ι	S	Р		f	0	0		c
	Save										tel/sip Back										

Add Contact.Email-Like Address

The Email-like address of your SIP address-of-record, AoR, with the user part and the domain part. For example, if your SIP AoR is "Handsome Boy <sip:michael@YourISP.com>", please enter it as "michael@YourISP.com".

Keys Function Description

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits alphanumeric input, where [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
\leftarrow and \rightarrow	Move cursor one position in edit mode.
	Maya autoar par lina in adit mada

 \uparrow and \downarrow Move cursor per line in edit mode.

Е	m		i	1	-	L	i	k	e		Α	d	d	r	e	S	S	:			
1	2	3	4	5	a	Ι	S	Р		f	0	0		n	e	t					
	Del					Abc/123				Clear						Back					

Figure 9. Contact EMail Address

Del

[Call]	Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.							
[Del]	Delete current contact.							
[Clear]	Delete all contacts from address book.							
[Back]	Return without any changes.							
[0K]	Edit selected contact.							
[\uparrow] and [\downarrow]	Navigate through contacts list.							

С	0	n	t	а	c	t	S	(1	4	3)						
Μ		с	h	a	e	1												
Μ	i	k	e															
Ν	i	c	k															
Р	a	t	r	i	c	k												
Р	a	u	1															
R	i	c	h	а	r	d												
		Call						Del				(Clea	r		Back	K	

Figure 10. List address book

Clear

Soft-Keys Function Description

	[`	Yes]		C	Clear	all c	onta	cts f	rom	addr	ess l	booł	۲.							
	[No]		F	Retur	n wi	thout	t any	cha	nges	5.									
С	1	e	a	r		a	1	1		с	0	n	t	a	c	t	S	?		



Figure 11. Clear Prompt

Address Book. [OK]

Keys Function Description

[Back]	Retrun to previous page.
[OK]	Enter sub-menu.

 $(\uparrow) and (\downarrow) Navigate through menu items.$

1	S	р	e	e	d		D	i	а	1							
2	С	а	1	1		S	c	r	e	e	n						
3	F	а	v	0	r	i	t	e									
4	Ι	Μ	Р	Р													
]	Back	K	

Figure 12. Address Book Main menu

Speed Dials

Speed Dials list format:

- Available entries: 0~19 (20 entries).
- User may use keypad to jump to specified entry, such as enter '9' to jump to the 9th entry.

Keys Function Description

[Call]	Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[Del]	Remove selected speed dial mapping.
[Clear]	Remove all speed dial mapping.
[Back]	Retrun to previous page.
[\uparrow] and [\downarrow]	Navigate through the list.
[OK]	Re-map selected entry.

0	М	i	c	h	а	e	1									
1	Μ	i	k	e												
2	N	i	c	k												
3	Р	а	t	r	i	c	k									
4	Р	а	u	1												
5	R	i	c	h	a	r	d									
6																
		Call					Del		(Clea	r]	Back	K	

Figure 13. Speed Dial list

Speed Dial Contacts Book Keys Function Description

[Back]	Retrun to previous page.
[0K]	Re-map selected entry.
[\uparrow] and [\downarrow]	Navigate through the list.

С	0	n	t	а	c	t	S	(1	4	3)					
Μ	i	c	h	а	e	1											
Μ	i	k	e														
Ν	i	c	k														
Р	а	t	r	i	c	k											
Р	а	u	1														
R	i	c	h	а	r	d											
															Bacl	K	



Call Screen

Call Screen List format:

1. List all contacts in black lsit in alphanumeric order.

- **2.** The First line is fixed (not scrollable), and shows how many contacts in black list. (VIP-550PT only)
- **3.** User may use keypad to jump to the first contact prefixed with the entered alphanumeric character.
- **4.** Alternatively, user may use navigation key \uparrow and \downarrow to scroll contact list.
- 5. Max size: 100.

Keys Function Description

[Call]	Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[Unblk]	Remove selected contact from black list and return (Unblock, revoke).

[Add] Add a contact from address book into black list.

С	а	1	1		S	c	r	e	e	n	(3)						
D	e	v	1	1		W	e	а	r	i	n	g		Р	r	а	d	a	
Ν	i	c	k																
S	u	a	n																
	Call						U	Jnbll	K				Add]	Back	

If reached max allowed entries, show

В	1	а	c	k		L	i	S	t	:						:			
	Μ	а	х		Е	n	t	r	у	:	1	0	0	!					
																	Back	ζ.	

Black List Keys Function Description

[Black]	Return without any changes
[0K]	Check selected record (Read only).
[\uparrow] and [\downarrow]	Navigate through black list.

Otherwise, show address book for user to pick a contact to add black list.

С	0	n	t	a	c	t	S	(1	4	3)					
Μ	i	c	h	а	e	1											
Μ	i	k	e														
Ν	i	c	k														
Р	а	t	r	i	c	k											
Р	а	u	1														
R	i	c	h	a	r	d											
															Bacl	K	

Black Information Description

Last time-of-call	This number is last time of caller.
Attempts	Accumulated call attempts originated from this contact.
AoR	Address of record.
[Call]	Dial to this contact. <i>Alternatively, You may lift the handset (offhook) or turn speaker on</i> <i>by pressing [SPK] to make a call to selected contact as well.</i>
[Black]	Return without any changes.
【↑】and【↓】	Scroll line.

1	4	:	3	7	:	2	1		1	0	/	2	9	/	2	0	0	6			
Α	t	t	e	m	р	t	S	:	2	0	1										
Α	0	R	:	D	e	v	i	1		W	e	a	r	i	n	g		Р	r	a	d
a	<	S	i	р	:	b	0	S	S	a	Ι	S	Р		f	0	0		c	0	m
>																					
	Call						J	Jnbll	K]	Back			

Figure 15. Caller information

Favorite

Configure favorite contacts for easy access during call-state, where you can activate favorite list by pressing [DSS] keys.

Favorite List Format:

- **1.** Listed in alphanumeric order.
- Fixed the 1st line (not scrollable) and indicates entry counts on the first line, such as (11). (VIP-550PT only)
- 3. Max entries: 20.

Favorite Description

[Call]	Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[Add]	Add a new entry from address book into favoriate list.
[Del]	Remove selected Favorite item.
[Back]	Return without any changes.

F	a	v	0	r	i	t	e	(1	1)						
Μ	i	c	h	а	e	1											
Μ	i	k	e														
Ν	i	c	k														
Р	a	t	r	i	c	k											
Р	a	u	1														
R	i	c	h	a	r	d											
		Call						Add				Del			Back	K	

Figure 16. Favorite list

If max allowed entries is reached, show

F	а	v	0	r	i	t	e		L	i	S	t						
	Μ	а	х		Е	n	t	r	у	:	2	0	!					
]	Back		

Otherwise, show address book for user to pick a contact to add into favoriate list.

С	0	n	t	а	c	t	S	(1	4	3)					
Μ	i	c	h	а	e	1											
Μ	i	k	e														
Ν	i	c	k														
Р	a	t	r	i	c	k											
Р	a	u	1														
R	i	c	h	a	r	d											
															Bacl	K	

Soft-Key Function Description

[Call]	Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[Del]	Remove selected Favorite item.
[Back]	Return without any changes.
[\uparrow] and [\downarrow]	Scroll Line.

Μ	i	c	h	a	e	1	<	S	i	р	:	1	2	3	4	5	a	Ι	S	Р	
f	0	0		n	e	t	>														
			Call					Del									F	Back			

Show SIP address-of-record (Read only).

IMPP

Soft-Key Function Description

[State]	Set terminal's state.
[Back]	Return previous page.
[0K]	Enter sub-menu.
[\uparrow] and [\downarrow]	Scroll Line.

1		S	р	e	e	d		D	i	a	1						
2		С	a	1	1		S	c	r	e	e	n					
3		F	а	v	0	r	i	t	e								
4		Ι	М	Р	Р												
	State														Bac	k	

Figure 17. IMPP main menu

State

Configure the Presence state of this terminal as per RFC4480 (Rich Presence Extension to the Presence Information Data Format).

Soft-Key Function Description

[Back]	Return without a	ny changes.
[OK]	On-Line	Presence state is open and user state is <other>idle</other>
	Away	Presence state is open and user state is away .
	Be Right	Presence state is open and user state <i>is <other>Be</other> Right Back</i> or any of the following states: dinner, meal, breakfast, worship such as.
	Busy	Presence state is open and user state is busy or any of the following states:

	meeting, performance, presentation, working, steering, appointment.
On the Phone	Presence state is open and user state is on-the-phone
Out to Lunch	Presence state is open and user state is lunch
Of-Line	Presence state is closed .

 \uparrow and \downarrow Navigate through menu items.

1	0	n	-	L	i	n	e											
2	Α	W	a	y														
3	В	e		R	i	g	h	t		В	a	c	k					
4	В	u	S	у														
5	0	n		t	h	e		Р	h	0	n	e						
6	0	u	t		t	0		L	u	n	c	h						
7	0	f	f	-	L	i	n	e										
]	Back		

Figure 18. IMMP Status Isit

IMPP. [OK]

IMPP List Format:

- 1. Show all subscribed Presence, Busy Lamp Filed (BLF), and Shared-Line Appearance (SLA) status.
- 2. Show display or user-part of the email-like address whenever possible and listed in alphanumeric order.
- **3.** The first line is fixed (not scrollable) and ndicates currently contacts on the first line, such as (6).(VIP-550PT only)
- 4. Max enteries: 20.
- 5. For Presence, show a for "open" state (on-line) and show 4 for "closed" state (off-line).
- 6. Busy Lamp Field (Dialog-Info)
 - SUBSCRIBE failed:
 - SUBSCRIBE ok but not received NOTIFY yet: No Symbol
 - Idle (terminated/no-dialogs): o
 - Busy (confirmed/trying/preceeding): •
 - Ringing (early):
- 7. Shared Line Appearance (Call-Info)
 - SUBSCRIBE failed:
 - SUBSCRIBE ok but not received NOTIFY yet: No Symbol
 - Idle: O
 - alerting/held: •
 - seized/progressing/active/held-private: •

Soft-Key Function Description

[Call]

Dial to selected contact.

Alternatively, You may lift the handset (offhook) or turn speaker on by pressing **[**SPK**]** to make a call to selected contact as well.

Add a new contact into IMPP list.

Ι	М	Р	Р	(6)														
8	R	i	c	h	а	r	d													
*	Р	a	t	r	i	c	k	(r	i	n	g	i	n	g)				
•	Μ	а	r	i	a	(a	c	t	i	v	e)							
۵	J	а	S	0	n	(0	n		t	h	e		р	h	0	n	e)	
	F	0	x	(U	n	k	n	0	w	n)								
٠	В	i	1	1	(0	f	f	1	i	n	e)							
			Call					Add					Del				•	Bacł	K	

Figure 19. IMMP Isit

Soft-Key Function Description

[Del]	Remove selected contact from IMPP list
[Back]	Return to previous page.
[OK]	Review selected IMPP contact.
【↑】and【↓】	Scroll IMPP list.
[\leftarrow] and [\rightarrow]	Scroll columns on the selected line to view whole records. Scroll half of the line width on each key press.

If max allowed entries is reached, show

Ι	Μ	Р	Р	Α	d	d	:										
	Μ	а	х	Е	n	t	r	у	:	2	0	!					
]	Back	K	

VIP-550PT supports the following specifications and integrated them as Instant Message and Presence Protocol, IMPP, for rich server features integration:

• Busy Lamp fileId (BLF):

This feature is to monitor the status of a contact or a list of contacts, including early (ringing)/confirmed (conversation)/terminated (idle) states. Refer to RFC3865 and RFC4235 for detail. Besides, please refer to RFC2387 for Muptipart-Related MIME data type and RFC4662 for "Event Notify for Resource List".

- SUBSCRIBE failed:
 - 1. Failed to SUBSCRIBE to the dialog state of chosen contact!
 - 2. Symbol:
- SUBSCRIBE ok but not received NOTIFY yet:
- 1. SUBSCRIBE to the dialog state of chosen contact succeeded but has not been NOTIFY yet.
- 2. Symbol: None
- Idle (terminated/no-dialogs):
 - 1. SUBSCRIBE to the dialog state of chosen contact succeeded and in *idle* mode.
 - 2. Symbol: O
- Busy (confirmed/trying/preceeding):
 - 1. the SUBSCRIBEd contact number is **busy**.
 - One of the dialogs of any dialog-info entity is in confirmed | trying | proceeding state.
 - o User off-hook is in *trying* state.
- 2. Symbol: •
- Ringing (early):
- 1. the SUBSCRIBEd contact number is ringing.
 - o Not busy
 - And one of the dialogs of **dialog-info entity** is in *early* state.
- **2.** Symbol:

Presence:

Monitor the presence state (availability or reachability to take calls) of a contact or a list of contacts. Refer to RFC3265 for SIP Event Notification, RFC3856 for Presence Event Package and RFC3863/RFC2387/RFC4662 for "application/pidf+xml"/ "MIME Muptipart-Related" / "Event Notify for Resource List" data format in NOTIFY.

- SUBSCRIBE failed:
 - 1. Failed to SUBSCRIBE to the presence state of chosen contact!
 - 2. Symbol:
- SUBSCRIBE ok but not received NOTIFY yet:
 - SUBSCRIBE to the presence state of chosen contact succeeded but has not been NOTIFY yet.
- 2. Symbol: None
- On-Line ("open"):
 - At least one of the entities of the SUBSCRIBED presence contact is available/on-line (NOTIFY status is "open").
- 2. Symbol: 0

Off-Line ("Closed"):

1. All entities of the SUBSCRIBED presence contact number are unavailable/off-line (NOTIFY status is "*closed*").

2. Symbol: 🌲

This terminal will be explicitly "unavailable" whenever:

• "Do-Not-Disturb", DND, feature is enabled

Note Note

- "All-Calls-Forward" is enabled,
- user explicitly off line by "Logout"

This terminal will respond a "closed" presence state if ever got Presence SUBSCRIBE from other terminals. Otherwise, this terminal is "available" with presence state "open".

• Shared-Line Appearance (SLA):

This feature is for the endpoint and the service delivery platform to maintain consistent "presentation" information, including

- The relative order of call appearances on a line and the current state of the call appearances.
- Enables call control clients, attendant consoles, and other applications to maintain a synchronized view of call appearance information.

Please refer to "BROADWORKS SIP ACCESS SIDE EXTENSIONS INTERFACE SPECIFICATIONS" release 13.0 version 1 from BroadSoft Inc. for detailed implementation.

- SUBSCRIBE failed:
 - 1. SUBSCRIBE to the chosen shared line failed!
 - 2. Symbol:
- SUBSCRIBE ok but not received NOTIFY yet:
 - SUBSCRIBE to the dialog state of chosen contact succeeded but has not been NOTIFY yet.
 - 2. Symbol: None
- Idle (terminated/no-dialogs):
 - 1. this appearance of chosen shared line is *idle*
 - 2. Symbol: O
- Busy (confirmed/trying/preceeding):
 - 1. this appearance of chosen shared line is:
 - o Seized
 - Progressing: making outbound calls.
 - o Active
 - o Held-private: be Held and only can be retrieved by the endpoint which

held it involved in this call.

- 2. Symbol: •
- Ringing (early):
 - 1. this appearance of chosen shared line is:
 - Alerting: receiving an incoming call.
 - $_{\odot}\,$ Held: Holding the peer and can be retrieved other endpoints.
 - 2. Symbol: •

Call to this "Shared-Line Appearance" DSS key will:

- 1. Gain an empty channel [A] / [B] and perform line-seize SUBSCRIBE-NOTIFY transaction before sending INVITE.
- 2. On receiving INVITE with Call-Info and "answer-after" parameter present, such as:

Call-Info: <sip:ProxyDNSorIP.com>;appearance-index=3;answer-after=0

- Auto Answer this call if
 - "answer-after" parameter is present in Call-Info header
 - From Header is a recognized Shared-Line AoR
 - appearance-index is configured
- Mapping Rule
 - answer-after=0: silent
 - answer-after=1: imperious
 - answer-after<3: urgent
 - Others and present: normal

Add – Type & Address-of-Record

Key Function Description

[Back]	Return Without any changes
--------	----------------------------

[OK] Pick selected item.

 \uparrow and \downarrow Navigate through the list.

1	В	u	S	у		L	а	m	р		F	i	e	1	d	,	В	L	F	
2	Р	r	e	S	e	n	c	e												
3	S	h	а	r	e	d	-	L	i	n	e	,	S	L	Α					
]	Back	ζ.		



С	0	n	t	а	c	t	S	(1	4	3)						
Μ	i	c	h	а	e	1												
Μ	i	k	e															
Ν	i	c	k															
Р	а	t	r	i	c	k												
Р	a	u	1															
R	i	c	h	а	r	d												
]	Back	K	

Figure 21. List Address Book

Add - View Status

IMPP entry format:

- 2. Show IMPP type: Presence, SLA or BLF (Fixed, not scrollable)
- 3. Show each status in the list as:
 - o Status
 - Full SIP address-of-record
- 4. Show aggregrated status and subscribed AoR on the last line.
- 5. For Presence, show ▷ for "open" state (on-line) and show ♣ for "closed" state (off-line).
- 6. Busy Lamp Field (Dialog-Info)
 - SUBSCRIBE failed:
 - SUBSCRIBE ok but not received NOTIFY yet: No Symbol
 - Idle (terminated/no-dialogs): O
 - Busy (confirmed/trying/preceeding):
 - Ringing (early):
- 7. Shared Line Appearance (Call-Info)
 - SUBSCRIBE failed:
 - SUBSCRIBE ok but not received NOTIFY yet: No Symbol
 - idle: O
 - alerting/held:
 - seized/progressing/active/held-private: •

[Call]	Dial to selected contact. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[IM]	Send instant Message to selected item.
[Renew]	Refresh status by re-SUBSCRIBE immediately.
[Back]	Return Without any changes.
[OK]	Review selected IMPP contact.
【↑】and【↓】	Scroll ines.

																	Ty Pro	pe: B esenc	BLF, e or l	SLA	
8	Р	R	e	S	e	n	c	e	-												
٠	0	Р	e	n	:	S	0	h	0	<	S	0	h	0	(a)	F	0	0		n	e
8	с	L	f	r	e	d	<	1	2	3	a	Ι	S	Р		c					
8	$\begin{array}{c c} c & L & o & s & e & d & \vdots & I & f \\ \hline o & P & e & n & \vdots & R & i & c & h \end{array}$									a	r	d	<	r	i	c	h	a	r	d	a
			Call					IM				F	Rene	W]	Back			

Show (Read only)

- 1. Type (BLF, SLA, Presence): Status
- 2. SIP address-of-record

	1	В	L	F	:	R	i	n	g	i	n	g										
	Μ	i	c	Η	a	e	1	<	S	i	р	:	1	2	3	4	5	a	Ι	S	Р	
Status Line:	f	0	0		n	e	t	>														
Type+Status																						
				Call														E	Back			

Configurations Call History

Soft-Key Function Description

[Missed]	Show the lastest 20 20 missed calls.
[Dialed]	Show the latest 20 dialed numbers.
[Recvd]	Show the latest 20 received calls.
[Back]	Return to previous page.

1		А	d	d	r	e	S	S		В	0	0	k						
2	•	С	a	l	l		Η	i	S	t	0	r	y						
3		Т	e	r	m	i	n	а	1		S	e	t	t	i	n	g	S	
4		С	а	1	1		F	0	r	W	a	r	d						
5		Р	r	e	f	e	r	e	n	c	e	S							
6		Ν	e	t	W	0	r	k											
7		S	Ι	Р		S	e	t	t	i	n	g	S						
8		S	e	r	v	i	c	e											
9		Α	d	v	a	n	c	e	d										
	Missed					Dialed				Recvd					Back				

Missed/Dialed/Received Calls

Missed Call list format:

• List all valid records with lastest call first.

By "**latest**", it refers to time sequence rather than the recorded "**time-of-call**"; because VIP-550PT has to synchronize its time with network time server by SNTP, such that an actually "**latest**" call may recorded as early as 2007 if VIP-550PT fails to synchronize its time by either SNTP or user manually configuration.

• User may use keypad to jump to specified entry, such as enter '9' to jump to 9th entry.

Key Function Description

[Call]	Dial to selected record. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[Del]	Remove selected record.
[Clear]	Remove all records in this list.
[Back]	Return to previous page.
[0K]	Review selected record.
【↑】and【↓】	Navigate through the list.

1		R	i	c	h	а	r	d								
2		Н	0	r	a	c	e									
3		М	a	r	r	y										
4		J	i	m												
	Call							Del		(Clea	r		Back	ζ	

View Record

[Call]	Dial to selected record.																				
[Del]	Delete this record.																				
[Back]	Return to previous page.																				
[OK] / [Edit]	Edit address of record to save to address book.																				
[\uparrow] and [\downarrow]	Scroll Line.																				
1	4	:	3	7	:	2	1		1	0	/	2	9	/	2	0	0	6			
---	------	---	---	---	---	---	---	-----	---	---	---	---	------	---	---	---	---	------	---	---	--
Μ	i	С	h	а	e	1	<	S	i	р	:	1	2	3	4	5	a	Ι	S	Р	
f	0	0		n	e	t	>														
	Call							Del					Edit	ţ			I	Back			

Show time-of- call and SIP address-of-record. (Read only)

Key Function Description

[Call]	Dial to selected record. Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.
[Save]	Save changes and return.
[To AB]	Save address-of-record into address book.
[0K]	Edit selected item.
[\uparrow] and [\downarrow]	Navigate through items.

D	i	S	р	1	а	у	:	Μ	i	c	h	а	e	1							
Р	R	0	t	0	c	0	1	:	S	i	р										
Α	D	d	r	:	1	2	3	4	5	a	Ι	S	Р		f	0	0		n	e	t
	Call						Save	e]	Го А	B]	Bacł	K			

Figure 22. Caller Record info

Configurations Terminal Settings

1	•	Α	d	d	r	e	S	S		В	0	0	k						
2		С	a	1	1		Η	i	S	t	0	r	у						
3	. /	T	е	r	m	i	n	a	l		S	е	t	t	i	n	g	S	
4		С	a	1	1		F	0	r	W	a	r	d						
5		Р	r	e	f	e	r	e	n	c	e	S							
6		Ν	e	t	W	0	r	k											
7		S	Ι	Р		S	e	t	t	i	n	g	S						
8		S	e	r	v	i	c	e											
9		Α	d	V	a	n	c	e	d										
\square					,												,	Back	

Figure 23. Main Menu

Key Function Description

[Back]	Return to pervious page.
[\uparrow] and [\downarrow]	Navigate through menu items.
[OK]	Enter submenu.

1		Р	а	S	S	W	0	r	d											
2		Р	r	0	g	r	а	m	m	a	b	1	e		Κ	e	y	S		
3		D	a	t	e	/	Т	i	m	e										
4		R	i	n	g	e	r													
5		Α	1	e	r	t	-	Ι	n	f	0									
6		L	C	D		В	a	c	k	-	L	i	g	h	t					
7		L a n g u a g e																		
8	. A l a r m																			
9		Р	h	0	n	e		L	0	c	k									
		A	١dmi	n				User	•									Bac	k	

Password

Set the passwords to administrative and user-level account. These passwords will be needed while unlocking the phone set (menu Lock), TELNET into the phone, logging in for web configuration, downloading or upgrading image from network by TFTP?HTTP, and while network administrator tries to remotely configure your phone set by flash update or rlogin.

[Admin]	Modify password to administrator's account.
[User]	Modify password to user's account.
[Back]	Return to previous page.
[\uparrow] and [\downarrow]	Navigate through menu items.

1	Р	а	S	S	W	0	r	d										
2	Р	r	0	g	r	а	m	m	а	b	1	e		Κ	e	у	S	
3	D	а	t	e	/	Т	i	m	e									
4	R	i	n	g	e	r												
5	Α	1	e	r	t	-	Ι	n	f	0								
6	L	С	D		В	r	i	g	h	t	n	e	s	S				
7	L	а	n	g	u	a	g	e										
8	Α	1	а	r	m													
9	Р	h	0	n	e		L	0	c	k								
	A	dmi	n				User	•]	Back	

Admin/User

Key Function Description

[Save]	Save changes and return.
[Back]	Return without and changes.
[OK]	Modified selected items.
[\uparrow] and [\downarrow]	Navigate through menu items.

Show 10 star signs, '*', for security reasons.

0	1	D	:	*	*	*	*	*	*	*	*	*	*						
Ν	e	W	:	*	*	*	*	*	*	*	*	*	*						
R	e	Т	у	р	e	:	*	*	*	*	*	*	*	*	*	*			
			Save														Bacl	K	

Passward.Admin. [OK]

Password consists of $[0-9^*#]$ only, and disply star sign, '*', for security reasons. The Default password is "**null**" (no password).

[Del]	Delete one character.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Finish editing.
【↑】and【↓】	Move cursors.

0	1	D	:		*	*	*									
Ν	e	W	:													
R	e	Т	у	p	e	:										
			Del				(Clea	r					Bacl	K	

Programable Keys

1	Р	a	s	s	W	0	r	d										
2	Р	r	0	g	r	а	m	m	а	b	1	e		Κ	e	у	S	
3	D	a	t	e	/	Т	i	m	e									
4	R	i	n	g	e	r												
5	Α	1	e	r	t	-	Ι	n	f	0								
6	L	C	D		В	a	c	k	-	L	i	g	h	t				
7	L	a	n	g	u	a	g	e										
8	Α	1	a	r	m													
9	Р	h	0	n	e		L	0	c	k								
																	Back	

Figure 24. Terminal Settings Menu

Key Function Description

[DTMF]	Configure predefined DTMF list for easy access to IVR system
[Back]	Return to previous page.
[OK]	Re-map selected programmable keys.

1		Ι	Μ	Р	Р												
2		R	e	g	i	S	t	e	r								
3		D	Т	Μ	F												
4		F	a	v	0	r	i	t	e	L	i	S	t				
5																	
6																	
7																	
8																	
9	0.																
1	0																
1	1																
1	2																
	DTMF														Back		

Configure 12 programmable keys, and the very first 4 programmable key goes to 4 softkeys on the top half of the LCD Panel.

\sqrt{DND}	√ <i>AutoAnswer</i>
√ <i>Register</i>	\sqrt{DTMF} Relay

a. Show ' $\sqrt{}$ ' to indicate activation status of "Auto-Answer", "DND", "DTMF Relay by INFO"; and unmap of these features will turn off selected feature as well .

b. "Register": Shoe ' $\sqrt{}$ ' to indicate successfully registered to all active SIP service domains and a '**X**' symbol to indicate none of the active domains are registered. No symbols shown if register to any of the active SIP service domains succeeded. c. Show display or user-part of its email-like address whenever possible for One-Touch Dial.

Besides, show Show **a** for "**One-Touch Dial**".

d. Otherwise promp to selected menu for further interaction.

The other 8 user-defined keys, excluding those NO-Function keys, will be shown during call state whenver [DSS] is pressed.

1		А	d	d	r	e	S	S		В	0	0	k							
2		С	а	1	1		Η	i	s	t	0	r	y							
3		D	Т	Μ	F		R	e	1	a	y		b	y		Ι	Ν	F	0	
4	\checkmark	Α	u	t	0	-	Α	n	s	W	e	r								
5		D	Ν	D																
6		S	Ι	Р		D	0	m	а	i	n S t a					t	u	S		
7		F	0	r	W	a	r	d												
]	Back	5	

- 1. Show ' $\sqrt{}$ ' to indicate activation status of "Auto-Answer", "DND", "DTMF Relay by INFO"; and unmap of these features will turn off selected feature as well .
- **2.** "Register": Show $\sqrt[4]{}$ to indicate successfully registered to all active SIP service domains and a 'X' symbol to indicate none of the active domains are registered. No symbols shown if register to any of the active SIP service domains succeeded.
- **3.** Show display or user-part of its email-like address whenever possible for One-Touch Dial.
- 4. Besides, show Show **a** for "One-Touch Dial".
- **5.** Otherwise promp to selected menu for further interaction.

DTMF

Specify a sequence of DTMF keys to dial in pre-dial phase or to transmit after call setup. This is shortcut to transmit a sequence of preconfigured DTMF keys, which aims to facilitate IVR system interaction. For example, you may configure your personal ID or banking account, and just activate corresponding DTMF entry to generate those pre-configured DTMF tones whenever appropriate (such as inquired by tele-banking system). If a DTMF entry is activated during digit-collecting phase, then those DTMF keys will be collected in a manner identical to those manually pressed by user.

Acceptable keys are [0-9 | * | # | p | ,], where 'p' and ',' (comma) stands for one-second pause. Depending on the configuration, such DTMF sequence may be transmitted to the peer either inbandly or outbandly (rfc2833 or INFO). You may also configure it to use as server feature access code during conversation, such as configuring it as "&76", and it will send DTMF "&76" "as is" during conversation.

[Dial]	Dial selected DTMF string.
[Add]	Add new DTMF entry.
[Back]	Return to previous page.
[0K]	Edit selected entry.
[\uparrow] and [\downarrow]	Scroll list.

D	Т	Μ	F	(3)															
С	i	t	у		В	а	n	k		Α	n	t	:	0	0	1	0	0	1	0	
Μ	a	S	t	e	r		C	a	r	d	:	1	2	3	4	5	6	7	8	9	
Р	Ι	Ν	:	a	b	c	d	e	f	g	h	i	j								
	Dial						Add					Del	-				Back	K			

DTMF List Format:

- Show all DTMF entries in alphanumeric order by tags.
 Fixed the 1st line (not scrollable) and show entry count, such as (3). (VIP-550PT only)
 Max entries count: 10

If reached max allowed entries, show

D	Т	Μ	F		А	d	d	:											
		Μ	a	Х		Е	n	t	r	у	:	1	0	!					
]	Back		

View DTMF Key Function and items Description

Тад	User-friendily display associated with this DTMF sequence numbers.
DTMF	A sequence of DTMF keys to dial in pre-dial phase or to transmit after call setup. Acceptable keys are $[0-9 * \# p ,]$, where 'p' and ',' (comma) stands for one-second pause. Depending on the configuration, such DTMF sequence may be transmitted to the peer either inbandly or outbandly (rfc2833 or INFO).
[Dial]	Dial selected DTMF string.
[Save]	Save changes and reurn.
[OK]	Edit selected entry.
[\uparrow] and [\downarrow]	Navigate through the list.

1		Т	а	g	:	С	i	t	у		В	а	n	k		А	n	t			
2		D	Т	Μ	F	:	0	0	1	0	0	1	0	0	1	0	0	1	0	0	1
	Dial						Save]	Back	ζ.			

Edit DTMF

Key Function and items Description

	7	ſag		Us	ser-fr	iend	ily d	ispla	y ass	socia	ted	with	this	DTI	MF s	eque	ence	e nui	nbe	rs.	
	D	TMF		Sp aft (co su ou fea an	ecify omm ch E tban ature d it v	y a s all se a) s DTMF dly (e acc will s	equi etup tand se (rfc2 ess end	ence Acc s for quen 833 code DTN	of E cepta one ce n or IN e dur IF "&	DTMI able -sec nay IFO) ing c .76" '	F keys ond be tr . You conve 'as is	ys to pau ans u m ersa s" du	o dia e [0- ise. mitte ay a ition uring	al in 9 * Dep ed to Ilso , suo J cor	pre- endi o the conf ch as	dial p ng c e pee igure s cor sation	ohas ,], v on th er ei e it t nfigu n.	se o whe ne co ther ther o us iring	r to f re 'p onfig inba inba inba it as	trans i and jurat andly s sei s "&	smit d ',' ion, y or rver 76",
	[Del]		De	elete	one	cha	racte	er.												
[/	Abc.	.]/[12	23]	To cu	ggle rren	bet t inpu	wee ut m	n dig ethoo	gits a d is a	and Ipha	alpha	anui ieric	meri anc	c inj I [12	put, 3] i	whe ndica	re. ∣ ates	[Abc digi] ir ts in	ndica put.	ates
	[C	lear]		Cl	ear a	all inp	out.														
	Ľ	ок]		Sa	ave c	hang	ges a	and r	eturr	า.											
	[Black]			Re	eturn	with	out	any d	chan	ges.											
【∢	$[\leftarrow]$ and $[\rightarrow]$			m	ove	curso	or on	e po	sitior	n in e	edit r	nod	e.								
1 . T a 2 . D T				g M	F	C :	i 0	t 0	y 1	0	B 0	a 1	n 0	k 0		A 0	n 0	t 1	0	0	1
	Del						Ab	oc/1	23			(Clea	r				Bac	K		

Set Programmable Keys

Key Function Description

[OK]	Map programmable key to selected function and return.	
------	---	--

[Back] Return without any changes.

А	d	d	r	e	S	S		В	0	0	k									
Α	u	t	0	-	Α	n	S	W	e	r										
С	a	1	1		D	e	t	a	i	1										
С	a	1	1		Η	i	S	t	0	r	y									
С	a	1	1		R	e	t	u	r	n										
С	h	a	n	n	e	1		Ι	n	f	0									
С	0	D	Е	С																
D	0		Ν	0	t		D	i	S	t	u	r	b	(D	Ν	D)		
D	T	M	F		L	i	S	t												
D	Т	Μ	F		R	e	1	a	у		b	y		Ι	Ν	F	0			
F	a	v	0	r	i	t	е													

F	0	r	W	а	r	d														
Ι	M	P	P																	
L	0	g	0	u	t															
М	e	S	S	a	g	e														
Ν	e	t	W	0	r	k		S	t	a	t	u	S							
Ν	0		F	u	n	c	t	i	0	n										
0	n	e		Т	0	u	c	h		D	i	a	1							
Р	а	c	k	e	t	i	z	а	t	i	0	n								
Р	h	0	n	e		L	0	c	k											
R	e	g	i	S	t	e	r													
R	e	j	e	c	t															
S	Ι	Р		D	0	m	a	i	n		S	t	a	t	u	S				
]	Back	(

Shortcut Menu

Auto-Answer (Auto Ans)

Auto-answering incoming calls on idle mode system-wide.

- Press this DSS to toggle auto-answer mode.
- Once enabled, it will show check symbol, ' $\sqrt{}$ ', along with this feature to indicate its current status, and the phone will auto-answer all incoming calls arrive on idle mode.
- Unmap this DSS feature will NOT turn auto-answer off.
- This is a shortcut key to configure
 [®] Preferences
 ₁ =>"Auto-Answer".

1	\checkmark	А	u	t	0	-	А	n	S	W	e	r								
2	\checkmark	D	Т	Μ	F		R	e	1	a	у		b	у		Ι	Ν	F	0	
3		D	Ν	D																
4	\checkmark	R	e	g	i	S	t	e	r											
5		S	Ι	Р		D	0	m	а	i	n		S	t	а	t	u	S		
6		С	а	1	1		D	e	t	а	i	1								
7		Ν	e	t	w	0	r	k		Ι	n	f	0							
]	Back		

Call History (CID)

Enter "2.Call History" / "Missed Call" menu directly if there are any unread missed call; otherwise enter "Call History" menu directly.

Call Detail (CallDetail)

Show 10 call detail records of the latest connected and finished calls (either incoming or outgoing). Those records (with their caller IDs, AoR, shown) are sorted by their finished time with latest comes first. Besides, they are volatile in memory such that they will be clean up every time the system reboots.

1	М	i	c	h	а	e	1										
2	J	a	s	0	n												
3	F	0	x														
4	Ι	n	S	u	r	a	n	c	e	Α	g	e	n	t			
5	В	r	0	k	e	r											
		Call													Bacl	k	

Key Function Description

[Call] Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.

[Back] Return to previous page.

 \uparrow and \downarrow Navigate through items / scroll Lines.

[OK] View record.

1		F	r	0	m															
2		Т	i	m	e		0	f		С	а	1	1							
3		D	u	r	a	t	i	0	n	:	2	3	:	1	0					
4		C	0	D	E	С	:	G		7	2	3		1	-	5		3	k	
5		С	0	n	t	а	c	t												
6		U	S	e	r		Α	g	e	n	t									
Call RTP										N	/ledi	a]	Bacl	K			

[RTP] : show local and remote RTP session, with port in rtp/rtcp UDP port pair.

L	0	c	a	1		R	Т	Р	:											
	Ι	Р	:	1	9	2		1	6	8		1	9	2		1	6	8		
	Р	0	r	t	:	1	2	3	4	4	/	1	2	3	4	6				
R	e	m	0	t	e		R	Т	Р	:										
	Ι	Р	:	1	0		0		2	4		3	3							
	Р	0	r	t	:	8	0	0	0	/	8	0	0	1						
															Back					

[Media] : Show media traffic information. The media-related information will be available only when the call lasted for more than 20 seconds.

Т	X	Р	а	c	k	e	Т	:	2	9							
R	х	Р	а	c	k	e	Т	:	1	8	9	6					
Т	x	В	у	t	e	:	5	6	0	Κ	В						
R	x	В	у	t	e	:	3	4	4	5	Κ	В					
]	Back	ζ	

From/To : SIP address-of-record of the peer.

М	i	c	h	а	e	1	<	s	i	р	:	1	2	3	4	a	Ι	S	Р	c
0	m	>																		
			~ 44			_														
			Call]	Back		

Time of Call

1	4	:	1	2	:	3	4	1	2	/	1	6	/	2	0	0	6		
																I	Back		

Duration: call active time (show mm:ss in-line)

CODEC: CODEC employed for the call (show in-line).

Contact: IP of the peer's CPE.

М	i	c	h	a	e	1	<	S	i	р	:	1	2	3	4	a	1	9	2	1
6	8		1		3	3	>													
-											-					-				
			Call]	Back		

User Agent: The phone tool used by the peer for this call.

Ι	Р	S	Ι	Р		Р	h	0	n	e	/	2	2	3		8		
]	Back		
		•	[E	Back]: reti	urn												

Call Return (CallReturn)

Make a call back to the last incoming numbers (missed or received call). It differs from server supported call return, such as "*69", in that it will find the latest incoming calls from "Missed calls" and "Received calls" then dial out the number.

Channel Info (A/B Info)

Show information of the current active calls on each channel. Those records (with their caller IDs, AoR, shown) are volatile in memory such that they will be clean up every time the system reboots.

[A]/[B]: Toggle between channel A and B, with [A]/[B] indicate current channel.

С	h	a	n	n	e	1		Α											
1		F	r	0	m														
2		С	0	n	t	а	c	t											
3		С	0	D	E	С	:	G		7	2	3		1	-	5	3	k	
4		U	s	e	r		Α	g	e	n	t								
	Call							RTP)				A/E	8			Bacl	K	

Do Not Disturb (DND)

Toggle switch to enable or disable Do-Not-Disturb, DND, feature. Once enabled all incoming calls will be forwarded unconditionally to the forwarded target number if applicable; otherwise, incoming calls will be turned down as "486 Busy Here". Unmap of this feature will cancel DND as well.

Show check symbol, '	$\sqrt{2}$, along with	this feature to	indicate its current	status once enabled.
----------------------	-------------------------	-----------------	----------------------	----------------------

1	\checkmark	А	u	t	0		А	n	S	W	e	r								
2	\checkmark	D	Т	Μ	F		R	e	1	а	у		b	у		Ι	Ν	F	0	
3		D	Ν	D																
4	\checkmark	R	e	g	i	S	t	e	r											
5		S	Ι	Р		D	0	m	а	i	n		S	t	а	t	u	s		
6		С	a	1	1		D	e	t	а	i	1								
7		Ν	e	t	w	0	r	k		Ι	n	f	0							
]	Back		

DTMF List

Shortcut to invoke "3.Terminal Settings / "DTMF, menu.

DTMF Relay by INFO (DTMF INFO)

This is a shortcut key to configure "Advanced $\mathcal{A}^{\mathbb{F}}$ CODEC \mathcal{A} =>"DTMF Relay by INFO" directly. Once enabled, all pressed DTMF keys (0-9*#&) during a call will be sent by SIP INFO method; otherwise those DTMF keys will be transmitted by DTMF over RTP (either by RFC2833 or DTMF tone mixed with voice stream).

Unmap of this key will not disable this feature. Show check symbol, ' $\sqrt{}$ ', along with this feature to indicate its current status once enabled.

1	\checkmark	А	u	t	0		А	n	S	W	e	r								
2	\checkmark	D	Т	Μ	F		R	e	1	а	у		b	у		Ι	Ν	F	0	
3		D	Ν	D																
4	\checkmark	R	e	g	i	S	Т	e	r											
5		S	Ι	Р		D	0	m	a	i	n		S	t	а	t	u	S		
6		С	а	1	1		D	e	t	а	i	1								
7		Ν	e	t	w	0	R	k		Ι	n	f	0							
]	Back		

Favorite

```
Shortcut to invoke "1.Address Book ] / "Favor ] menu.
```

Forward

Forward incoming ringing calls on call arrival to preconfigured forwarding target number; otherwise enter "4.Call Forward" menu. Although [Forward] will be one of the context-sensitive soft-buttons on call-waiting, it may disappear if user does not process this incoming call before this context changed, this programmable key can always be pressed at any time.

IMPP

```
Shortcut to invoke "1.Address Book ] / "IMPP ] menu.
```

Logout

Unregister all activated SIP address-of-records and cease regular auto-registration scheduling to keep user off-line until [Register] feature is activated to be on-line again. Once logged out, it appears to be in "closed" Presence state.

After reboot the phone-set will clear this status and register to all activated SIP address-of-records afte startup.

Message

Access to out-of-dialog instant messaging: on pressing, it will enter **"8.Service** / **"Message** / **"Inbox** if there are any unread new messages; otherwise enter **"8.Service** / **"Message** menu.

Network Status (Network)

Shortcut to invoke ^[]9.Advanced ^[]/ ^[]Network Status ^[] menu.

No Function (NOP)

No specific function for this key.

One-Touch Dial

Make an outbound call to the associated contact. Show the display (or user-part of its email-like address) whenver possible. Besides, show Show

for "One-Touch Dial".

1	7	Μ	i	c	h	а	e	1												
2		D	Т	Μ	F		R	e	1	а	у		b	у		Ι	Ν	F	0	
3		D	Ν	D																
4	\checkmark	R	e	g	i	S	t	e	r											
5		S	Ι	Р		D	0	m	а	i	n		S	t	а	t	u	s		
6		С	a	1	1		D	e	t	а	i	1								
7		N	e	t	W	0	r	k		Ι	n	f	0							
]	Back	2	

Register

Register to all activated domains on-demand, and appear to be in "open" Presence State unless it is in "DND"

1	\checkmark	А	u	t	0		А	n	S	w	e	r								
2	\checkmark	D	Т	Μ	F		R	e	1	a	у		b	у		Ι	Ν	F	0	
3		D	Ν	D																
4		R	e	g	i	S	t	e	r											
5		S	Ι	Р		D	0	m	a	i	n		S	t	a	t	u	S		
6		С	a	1	1		D	e	t	a	i	1								
7		Ν	e	t	w	0	r	k		Ι	n	f	0							
]	Back		

Reject

Reject incoming ringing calls on arrival as "486 Busy". Although [Busy] will be one of the context-sensitive soft-buttons on call-waiting, it may disappear if user does not process this incoming call before this context changed, this programmable key can always be pressed at any time.

SIP Domain Status (SIP Realm)

Show a symbol preceding each active SIP service domain:

- a. ' $\sqrt{}$ ' to indicate successfully registered to all active SIP service domains.
- b. 'x' to indicate none of the active domains are registered.
- c. No symbol to indicate registered to any of the active SIP service domain succeeded.

1		Α	u	t	0		А	n	S	W	e	r								
2	\checkmark	D	Т	Μ	F		R	e	1	a	у		b	у		Ι	N	F	0	
3		D	Ν	D																
4	\checkmark	S	Ι	Р		D	0	m	а	i	n		S	t	а	t	u	s		
5		R	e	j	e	c	t													
6		C	a	1	1		D	e	t	a	i	1								
7		N	e	t	w	0	r	k		Ι	n	f	0							
]	Back	x l	

If pressed (activated), it will show registration status of each active SIP service domain.

1		Ι	S	Р	f	0	0		n	e	t						
2	х	S	i	р	Ι	S	Р		c	0	m						
3		Ι	S	Р	р	r	i	v	а	t	e	b	i	z			
]	Back	

Show symbol preceding each active SIP service domain:

- a. ' $\sqrt{}$ ' to indicate a register success state
- b. 'x' to indicate a register failed state
- c. '.' To indicate a registering state.

The VIP-350PT / VIP-550PT serial IP Phone supports up to 3 SIP service domains, which this terminal may register to; and you may circle active service domains by pressing [Realm] soft-button while making outbound calls.



Settings Date/Time

IP Phone will synchronize its time by Simple Network Time Protocol, SNTP, with network time server regularly if SNTP is enabled. To keep the time you manually set here, you must disable SNTP.

[Zone]	Set time-zone.
[FMT]	Set date/time display format.
[DST]	Daylight Saving Time (DST) adjustment.
[Back]	Return to previous page.
[OK]	Adjust system date and time.

1		Р	a	s	s	w	0	r	d											
2		Р	r	0	g	r	a	m	m	а	b	1	e		Κ	e	y	S		
3		D	а	t	e	/	Т	i	m	e										
4		R	i	n	g	e	r													
5		Α	1	e	r	t	-	Ι	n	f	0									
6		L	C	D		В	a	c	k	-	L	i	g	h	t					
7		L	a	n	g	u	a	g	e											
8		Α	1	a	r	m														
9		Р	h	0	n	e		L	0	c	k									
	Zone							FMT					DST	2				Back	<u> </u>	

Zone

This terminal will synchronize its time by Simple Network Time Protocol, SNTP, with network time server regularly if SNTP is enabled. Before setting the GMT time inquired from NTP server, it will convert it into local time based on the time zone as well as daylight saving adjustment (see below).

To leave the current system date and time set from menu-"Date/Time" intact, you could either disable SNTP from menu-"8.Service" / "SNTP". **The System default is GMT+00:00.**

Key Function Description

[±]	Toggle between '+' and '-'.
[OK]	Save changes and return.
[Back]	Return without changes.
\leftarrow and \rightarrow	Navigate through fields.
[0-9]	enter values.
±	Press any key [0-9*#] to toggle between '+' and '-'.

Т	i	m	e		Ζ	0	n	e								
G	Μ	Т	:	±	0	0	:	0	0							
			±]	Back		

FMT

Specify the date format for display purpose (such as time-of-call in records of call history).

MM: 2-digit month

DD: 2-digit day of month

YYYY: 4-digit year

1	Μ	Μ	/	D	D	/	Y	Y	Y	Y						
2	D	D	/	Μ	Μ	/	Y	Y	Y	Y						
3	Y	Y	Y	Y	/	Μ	Μ	/	D	D						
4	Y	Y	Y	Y	/	D	D	/	Μ	Μ						
5	Y	Y	Y	Y	-	Μ	Μ	-	D	D						
]	Back	K	

DST

Daylight Saving Time (DST) adjustment.

1	0	f	f	S	e	t	:	+	2	:	0	0					
2	S	t	a	r	t		D	a	t	e							
3	S	t	0	p		D	a	t	e								
															Bacl	K	

DST Offset

Offset from this terminal's time when DST is in effect. When DST is over, the specified offset is no longer applied to this terminal's time. Valid values are H:MM, -H:MM or +H:MM.

The System default is 0:00 to disable DST adjustment.

[±]	Toggle between '+' and '-'.
[OK]	Save changes and return.
[Back]	Return without changes.
[\leftarrow] and [\rightarrow]	Navigate through fields.
[0-9]	enter values.

D	S	Т		0	f	f	S	e	t								
0	f	f	S	e	t	:	±	0	:	0	0						
			±]	Bacl	K	

DST Start Date

Т	i	m	e	:	0	2	:	0	0									
D	а	у	:	S	u	n	,	L	а	S	t	W	e	e	k			
Μ	0	n	t	h	:	Α	р	r	i	1								

DST Start Date.Month

J	a	n	u	a	r	y									
F	e	b	u	r	a	r	у								
Μ	a	r	c	h											
А	р	r	i	1											
Μ	а	у													
J	u	n	e												
J	u	1	y												
]	Back	5	

DST Start Date.Day

[WoM]/[DoM]: Toggle between [Week-of-Moth] and [Day-of-Month].

1	Μ	0	d	e	:	D	а	у	-	0	f	-	Μ	0	n	t	h	
2	D	а	у	-	0	f	-	М	0	n	t	h						
3	W	e	e	k	-	0	f	-	Μ	0	n	t	h					
	Wo	M/D	οM]	Back	

DST Start Date.Day.Day-Of-Month

D	a	у	-	0	f	-	Μ	0	n	t	h						
[1	-	3	1]												1
			Del												Back	κ.	

DST Start Date.Day.Week-Of-Month

W	e	e	k	:	L	a	S	t		W	e	e	k			
D	a	y	-	0	f	-	W	e	e	k	:	S	u	n		
															Back	

Week

0	L	а	S	t		W	e	e	k						
1	1	S	t		W	e	e	k							
2	2	n	d		W	e	e	k							
3	3	r	d		W	e	e	k							
4	4	t	h		W	e	e	k							
5	5	t	h		W	e	e	k							
6	6	t	h		W	e	e	k							
												I	Back		

Day-of-Week

0	S	u	n										
1	Μ	0	n										
2	Т	u	e										
3	W	e	d										
4	Т	h	u										
5	F	r	i										
6	S	а	t										
]	Back	Σ.	

DST Start Date.Time

Press **[**OK**]** to edit selected item Time of day on which DST begins in HH:MM 24 hour's format.

Μ	0	n	t	h	:	Α	р	r	i	1									
D	а	у	:	S	u	n	,	L	а	S	t	W	e	e	k				
Т	i	m	e	:	0	2	:	0	0										
			Save]	Back	ζ.	

Set Date/Time [OK]

Specify current date and time on this terminal. Enter the current date, where the time is in 24-hour format and the date format depends on the "3.Date/Time" / "FMT" configuration.

Note: this terminal will synchronize its time by Simple Network Time Protocol, SNTP, with network time server regularly if SNTP is enabled. To keep the time you manually set here, please disable SNTP from menu-"SNTP mode" first.

Key Function Description

[Save]	Save changes and return
[Back]	Return without changes.
[Back]	Return without changes.
[\leftarrow] and [\rightarrow]	Navigate through fields.
[0-9]	Enter values.

Т	0	d	a	у															
2	1	:	3	7	:	2	5	1	1	/	2	3	/	2	0	0	6		
	Save															Bacl	K	1	

Date / Time Setting Screen

Ringer

Specify the ring to play on call arrival.

Key Function Description

[Type]	Configure ringer type.
[Vol]	Configure ringer volume.
[Back]	Return to previous page.

1		Р	a	S	S	W	0	r	d												
2		Р	r	0	g	r	а	m	m	а	b	1	e		Κ	e	y	S			
3		D	a	t	e	/	Т	i	m	e											
4		R	i	n	g	e	r														
5		Α	1	e	r	t	-	Ι	n	f	0										
6		L	С	D		В	а	c	k	-	L	i	g	h	t						
7		L	a	n	g	u	a	g	e												
8		Α	1	а	r	m															
9		Р	h	0	n	e		L	0	c	k										
	Туре					Vol												Back			

Figure 25. Terminal Settings menu

1		R	i	n	g	e	r	-	0								
2		R	i	n	g	e	r	-	1								
3		R	i	n	g	e	r	-	2								
4		R	i	n	g	e	r	-	3								
5		R	i	n	g	e	r	-	4								
6		R	i	n	g	e	r	-	5								
7		R	i	n	g	e	r	-	6								
8		R	i	n	g	e	r	-	7								
9		R	i	n	g	e	r	-	8								
1	0		R	i	n	g	e	r	-	9							
]	Back	K	

\uparrow and \downarrow :	Navigate (and play	y) through ringers.
-------------------------------	--------------------	---------------------

Vol Adjustment

Adjust volume real time by playing selected ring. Use Volume Keys : [Up] and [Down] navigation keys : [\uparrow], [\downarrow], [\leftarrow] and [\rightarrow] to adjust volume.

R	i	n	g	e	r	V	0	1	u	m	e	:					
<	<	<	<	<	<												
]	Back	ζ.	

Alert-Info

The IP Phone supports "**Alert-Info**" header in the first SIP INVITE message as per RFC3261, "**Alert-Info**" header dictates the phone to use an alternative ringing tone, which is specific for that call.

The header should be in a format similar to

"Alert-Info: <http://MediaServer.ISP.com/Announce.pcmu>;AnyParameter=xxxx"

or

"Alert-Info: xxxx", where "xxxxx" (case-insensitive) is the tone tag for one of the available ringing tones (0~11). This is useful to distinguish calls, for example local calls from calls coming from PSTN. Besides, this header is ignored for re-INVITE. If the specified tone is out of range, the current ringer is used.

From this page, user may re-map which tone should be played when "xxxx" tone tag is specified. For example, if "Alert-Info: ringtone-0" is mapped to "Ringer7", then "Ringer7" will be played whenever "Alert-Info: ringtone-0" or "Alert-Info: ;anyParameter=ringtone-0">http://SIP.ISP.com/file.pcmu>;anyParameter=ringtone-0" is specified in the initial INVITE.

Default is to map "ringtone-0" to Ringer0, "ringtone-1" to Ringer1, and so on.

Key Function Description

[Tag]	Configure the tag of selected alert-info.
[Ring]	Configure the ringer type of selected alert-info.
[Back]	Return to previous page.

1		i	n	-	Η	i	:	R	i	n	g	e	r	-	1						
2		0	u	t	-	1	0	:	R	i	n	g	e	r	-	4					
3		i	n	t	r	a	-	h	i	:	R	i	n	g	e	r	-	3			
4		i	n	t	r	a	-	1	0	:	R	i	n	g	e	r	-	2			
5		i	n	t	1	:	R	i	n	g	e	r	-	9							
6																					
7																					
	Tag					Ring										Back					

Tag Settings

Specif the tag to be compared to in "Alert-Info" header. This tag is case-insensitive and can appear as header value or in any header parameter value.

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.

Т	a	g	:																	
i	n	-	Η	i																
	Del			Abc/123			Clear					Back								

Figure 26. Tag name settings

Ring Settings

Specify which ring to play when a match is found in "Alert-Info" header.

Key Function Description

[Back]	Return without changes
[\uparrow] and [\downarrow]	Navigate (and play) through ringers.
[OK]	Save and return

Play selected ring real time.

1		R	i	n	g	e	r	-	0								
2		R	i	n	g	e	r	-	1								
3		R	i	n	g	e	r	-	2								
4		R	i	n	g	e	r	-	3								
5		R	i	n	g	e	r	-	4								
6		R	i	n	g	e	r	-	5								
7		R	i	n	g	e	r	-	6								
8		R	i	n	g	e	r	-	7								
9		R	i	n	g	e	r	-	8								
1	0		R	i	n	g	e	r	-	9							
														B	Back		

LCD Back-Light

Specify how long the LCD backlite stays on after going into idle state.

]	Back	K			
9		Α	1	W	а	у	S		0	n											
8		0	f	f		а	f	t	e	r		10		Μ	i	n	u	t	e	S	
7		0	f	f		а	f	t	e	r		5		Μ	i	n	u	t	e	S	
6		0	f	f		а	f	t	e	r		3		Μ	i	n	u	t	e	S	
5		0	f	f		а	f	t	e	r		2		Μ	i	n	u	t	e	S	
4		0	f	f		а	f	t	e	r		1		Μ	i	n	u	t	e		
3		0	f	f		а	f	t	e	r		3	0		S	e	c	0	n	d	S
2		0	f	f		a	f	t	e	r		1	0		S	e	c	0	n	d	S
1		0	f	f		W	h	e	n	e	v	e	r		Ι	d	1	e			

Alarm

Set up temporary alarm and / or regular alarm.

On timer expiration (either alarm type), the phone will start ringing for 60 seconds with increasing volume (stepping from the current speaker volume). Press any key after timer expiration will stop its activation (ringing); otherwise, it will repeat the ringing process after 5 minutes 'till any key is pressed or it has been 30 minutes after its expiration.

1	Ο	n	e	-	t	i	m	e		Α	1	а	r	m				
2	R	e	g	u	1	a	r		Α	1	a	r	m					
3	R	i	n	g	e	r												
																Back	K	

Regular Alarm

Configure a timer which triggers regularly.

[On]/[Off]: Toggle switch to enable or disable alarm. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1	А	c	t	i	v	а	t	e								
2	Т	i	m	e	:	0	6	:	3	0						
3	R	e	g	u	1	a	r	i	t	у						
	C)n/O	ff				•]	Back	5	

Time Settings

Press [OK] to edit the time of this regular alarm of when it triggers.

[Back]	Return without changes
[\leftarrow] and [\rightarrow]	Navigate through fields in a item.
[0-9]	Please enter values.
[Save]/【OK】	Save and return.

1		Α	c	t	i	v	a	t	e								
2		Т	i	m	e	:	0	6	:	3	0						
3		R	e	g	u	1	a	r	i	t	у						
	Save]	Back	K	

Regularity Settings

Configure the regularity when the timer triggers.

[On]/[Off]: Toggle switch to enable or disable alarm on selected day. Show check symbol 'That ' in-line if enabled!

0		S	u	n											
1	\checkmark	Μ	0	n											
2		Т	u	e											
3		W	e	d											
4		Т	h	u											
5		F	r	i											
6		S	a	t											
	Save					0	n/O	ff					Back	K	

Alarm Ringer

Position on current values on entering.

1		R	i	n	g	e	r	-	0								
2		R	i	n	g	e	r	-	1								
3		R	i	n	g	e	r	-	2								
4		R	i	n	g	e	r	-	3								
5		R	i	n	g	e	r	-	4								
6		R	i	n	g	e	r	-	5								
7		R	i	n	g	e	r	-	6								
8		R	i	n	g	e	r	-	7								
9		R	i	n	g	e	r	-	8								
1	0		R	i	n	g	e	r	-	9							
														I	Back		

Phone Lock

Once locked in either Function Keys (security) mode or take call only mode, you need either user-level or privileged password to unlock it.

U n l	o c	k	р	a s	s w	o r	d :

Off	The terminal is not in security mode.
	This terminal is locked on security level. In security mode, the following keys will be protected (key press will be ignored):
	 4 programmable hard keys located on the upper half of LCD.
Function	Menu Key. 【Menu】
Keys	Redial key: 【Redial】
	Message Waiting Indication: 【MWI】
	 Context-sensitive soft-keys: [Call History], [Address Book] and [DSS] keys.

Take Call Only could not be configured either. That is, the phone would leave in "receive only" mode (No dial tone could be heard).

[Back]	Return without any changes.
[OK]	Save changes and return.
[\uparrow] and [\downarrow]	Navigate through items.

1	0	f	f															
2	F	u	n	c	t	i	0	n		Κ	e	у	S					
3	Т	a	k	e		С	а	1	1		0	n	1	y				
																Back	K	

Call Forward

Configure the target number to forward to while this phone is busy or not answered within a predefined guarding interval. This forwarding number is also employed while the phone is engaged in Do Not Disturb, DND, mode or while the user activate **[Forward]** feature on an incoming waiting call.

The system forwarding rules are checking Do Not Disturb mode first, then All Calls Forward, Busy Forward, finally going to No Answer Forward while no-answer timer expires.

1		Т	а	r	g	e	t		Ν	u	m	b	e	r					
2		Α	1	1		C	а	1	1	S		F	0	r	w	а	r	d	
3	\checkmark	В	u	S	y		F	0	r	W	а	r	d						
4	\checkmark	Ν	0		A	n	S	W	e	r		F	0	r	w	а	r	d	
																		Back	

Show check symbol ' $\sqrt{}$ ' in-line if enabled!

Target Number

This forwarding number is to forward incoming calls when the phone is busy or not answered within a predefined guarding interval. It is also employed while the phone is engaged in Do Not Disturb, DND, mode or while the user activates **[**Forward **]** feature on an incoming waiting call.

Key Function Description

	Dial to target number if available.
[Call]	Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.

[Reset] Reset target number from address book.

Μ	y	Α	1	t	e	r	n	a	t	i	v	e	N	0	<	S	i	р	:	1	2
3	4	4 @ I S .				•	f	0	0	•	c	0	m	>							
	Call]	Rese	t				Del]	Back	ζ.		

[Back]	Return without any changes.
[OK]	Pick selected contact.
[\uparrow] and [\downarrow]	Navigate through the list.
[Del]	Remove target number (no target number for call-forward).
[Back]	Save changes and return.
[OK]	Return.

С	0	n	t	а	c	t	S	(1	4	3)						
Μ	i	c	h	а	e	1												
Μ	i	k	e															
Ν	i	c	k															
Р	a	t	r	i	c	k												
Р	а	u	1															
R	i	c	h	а	r	d												
]	Back	ζ	

Call Forward.All Calls Forward

You can configure to unconditionally forward all incoming calls by enable the All Calls Forward feature. Forwarded calls are logged in the Missed Calls. If this feature is enabled, LCD will show "FWD:TargetAoR" to remind user that all incoming calls will be forwarded unconditionally thereafter. Default is disabled.

1	Т	a	r	g	e	t		N	u	m	b	e	r						
2	Α	1	1		С	а	1	1	S		F	0	r	W	а	r	d		
3	 В	u	S	y		F	0	r	w	a	r	d							
4	 Ν	0		Α	n	S	W	e	r		F	0	r	W	а	r	d		
	0	n/O	ff														Bacl	K	

[On] / **[Off]**: Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

Call Forward.Busy Forward

Configure to forward incoming waiting calls when the system is busy, on which time all lines are occupied. Forwarded calls are logged in the Missed Calls.

Default is enabled

1		Т	a	r	g	e	t		N	u	m	b	e	r						
2		Α	1	1		С	a	1	1	S		F	0	r	W	а	r	d		
3	\checkmark	В	u	S	у		F	0	r	W	а	r	d							
4		Ν	0		Α	n	S	W	e	r		F	0	r	W	а	r	d		
		0	n/O	ff														Bacl	K	

[On] / **[Off]**: Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

Call Forward.No Answer Forward

Configure to forward incoming waiting calls after ringing for a predefined interval. Forwarded calls are logged in the Missed Calls. When No Answer Forward feature is enabled, the incoming calls will be forwarded to the Target Number if the call has not been answered for a predefined time (see next section); otherwise the phone will keep ringing indefinitely.

Default is enabled.

[On] / **[Off]:** Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

[Timer]: Configure the time to wait (ringing) before forwarding incoming calls as No Answer. Default is 30 seconds.

1		Т	a	r	g	e	t		Ν	u	m	b	e	r						
2		Α	1	1		C	a	1	1	s		F	0	r	w	а	r	d		
3		В	u	S	y		F	0	r	W	а	r	d							
4	\checkmark	Ν	0		Α	n	S	W	e	r		F	0	r	w	а	r	d		
		0	n/O	ff			r	Гime	r									Back	K	



Figure 27. Timer Configuration

Preference

Set up personal call preferences.

1		С	а	1	1		W	а	i	t	i	n	g							
2		D	i	а	1		Т	i	m	e	0	u	t							
3		Н	0	1	d		R	e	c	а	1	1		Т	i	m	e	r		
4		Α	u	t	0	-	R	e	d	i	а	1								
5		D	i	а	1		Р	1	а	n										
6	\checkmark	Μ	e	S	S	a	g	e		Α	1	e	r	t						
7		Α	u	t	0	-	Α	n	s	W	e	r								
8		U	n	h	0	1	d		0	n		Т	r	а	n	s	f	e	r	
9	\checkmark	С	0	n	f	e	r	e	n	c	e		Α	1	e	r	t			
1	0		Χ	f	e	r		0	n		E	x	i	t		C	0	n	f	
]	Bacl	K	

Show check symbol ' $\sqrt{}$ ' in-line if enabled!



Call Waiting

Call Waiting notifies the user, who is on an established call, that an additional external call has been present and is "waiting to be answered." The waiting call receives normal ringing until it is answered, the incoming calling party abandons the call, or the ringing cycle timer expires, and the call is given No-Answer-Forward treatment (if applicable).

When the user has engaged in a call, and some new incoming calls are waiting for answer, the phone will play an auditable tone twice in 7 seconds to alert the user on active channel. You could change this setting such that the phone will reply an incoming call a "486 Busy Here" while you are busy. Default is enabled.

[On] / **[Off]:** Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1		С	a	1	1		W	а	i	t	i	n	g							
2		D	i	a	1		Т	i	m	e	0	u	t							
3		Η	0	1	d		R	e	c	a	1	1		Т	i	m	e	r		
4		Α	u	t	0	-	R	e	d	i	a	1								
5		D	i	a	1		Р	1	a	n										
6		Μ	e	S	S	a	g	e		Α	1	e	r	t						
7		Α	u	t	0	-	Α	n	S	w	e	r								
8		U	n	h	0	1	d		0	n		Т	r	a	n	s	f	e	r	
9		С	0	n	f	e	r	e	n	с	e		Α	1	e	r	t			
1	0		X	f	e	r		0	n		E	x	i	t		С	0	n	f	
	On/Off																	Bacl	K	

Dial Timeout

Configure the dial timeout if no responses (any provisional responses such as "100 Trying" or "180 Ringing") from the peer. If the timer expires before the peer starts ringing back, it will play disconnect tone and prompt the user "Dial timeout".

System default is 30 seconds.

[Del]: Delete one character

D	i	а	1		Т	i	m	e	0	u	t								
[1	0	-	6	0	0]	(S	e	с	0	n	d)				
3	0																		
			Del]	Back	K	

Hold Recall Timer

When some calls are put on hold, such as during a transfer procedure, the phone will play an auditable tone regularly to alert the user on active channel. Default alerting interval is 60 seconds. Besides, the phone will start ringing if the handset is placed on-hook and there is a call currently on hold.

The alerting process is as following:

- 1. User are in on-hook state:
 - The phone will ring.
 - The recall state continues 'till user picks up the call. However, if user does not answer the call within 1 minute, the call will be disconnected.
- 2. User are in off-hook state, either engaged in other calls or about to making calls:
 - Play hold recall tone once to remind the user that some calls are in hold.
 - Reset the hold recall timer for next time alarm.

[Del]: Delete one character

Н	0	1	d		R	e	c	a	1	1		Ι	n	t	e	r	v	а	1	
[1	0	-	6	0	0]	(S	e	с	0	n	d)					
6	0																			
	Del]	Back	κ.	

Auto-Redial

Once auto-redial is in progress, it will be automatically canceled if either the stop criterion is met or the total activation duration is expired.

1	S	t	0	р		С	r	i	t	e	r	i	0	n				
2	R	e	t	r	у		Ι	n	t	e	r	v	а	1				
3	Т	0	t	а	1		D	u	r	а	t	i	0	n				
																Bacl	ζ	

Stop Criterion Settings

Specify the circumstances to stop auto-redialing.

- 1. Stop on Ringing: only when the peer starts ringing back will auto-redialing stop (Default).
- 2. Stop on Connected: Only when the peer picks up will auto-redialing stop.

1	S	t	0	р	0	n	R	i	n	g	i	n	g				
2	S	t	0	р	0	n	С	0	n	n	e	c	t	e	d		
]	Back	K	

Retry Interval Settings

Specify the gap (measured in seconds) between two successive re-dials to avoid overflowing the networks with fast retries.

Default is 15 seconds

[\leftarrow] and [\rightarrow] : move cursor one position in edit mode.

R	e	d	i	а	1		Ι	n	t	e	r	v	а	1				
[1	-	2	5	5]	(S	e	c	0	n	d)				
1	5																	
			Del													Bacl	K	

Totoal Duration Settings

Specify the activation duration of this auto-redial feature once starts, measured in seconds. Once the timer expires, the auto-redial feature will be canceled.

Default is 1800 seconds (30 minutes).

А	c	t	i	v	e		Ι	n	t	e	r	v	a	1						
[3	0	-	8	6	4	0	0]	(S	e	с	0	n	d)			
1	8	0	0																	
			Del															Bacl	k	

Dial Plan

Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1	Ι	n	t	e	r	-	D	i	g	i	t		Т	i	m	e	0	u	t	
2	 L	Α	Ν		D	i	а	1												
3	С	а	1	1		С	0	m	m	а	n	d								
4	D	i	а	1		Р	1	а	n											
5	Η	0	t		L	i	n	e												
]	Back			

Inter-digit Timeout Settings

Configure the inter-digit timeout while dialing numbers in digit-collecting phase. **Default is 4 seconds**. If you dial the alphabets very often, such as URL, then you may increase the inter-digit timeout to, say, 6 seconds. While making calls, the first digit timeout is 15 seconds, and the inter-digit timeout after the first digit is defined here. The phone will dial out the collected digits on timer expired. To speed up the dial process, press [Call] soft-button whenever finishing dial.

Ι	n	t	e	r	-	D	i	g	i	t		Т	i	m	e	0	u	t		
[3	-	9]	(S	e	с	0	n	d)								
4																				
			Del]	Back	(

LAN Dial Settings

Configure whether a dial string starting with a star, '*', key signals a LAN dial. For example, if your IP is "192.168.3.10" and you want to make a direct IP call to another host (either a phone-set or a PC) which IP is "192.168.3.20", you may dial "*20" to make a call to it. To specify the UDP port, you may dial "*20**9999" to call to "192.168.3.20:9999".

Generally, to make LAN dial, you should use a star sign, '*', as a dot, '.'; and double stars, "**", as a colon, ':', to specify the UDP port the callee listens on. Besides, you may specify the last 1, 2 or 3 fields of the peer's IP, plus additional port.

For example, suppose the terminal's IP is 192.168.3.10.

- Example 1: to call "192.168.3.20:5060", you may dial:
 - *** 192*168*3*20**5060 (where "**" as ':' to specify UDP port)
 - *** 192*168*3*20 (default is call to UDP port 5060)
 - *** *20# (since the peer locates on the same subnet as you do, you could specify only the last portion of its IP as shorthand to "192.168.3.20:5060".
- Example 2: to call "888@192.168.4.90:5060
 - ***888*192*168*4*90**5060 (where "**" as ':' to specify UDP port)
 - ***888*192*168*4*90 (default is call to UDP port 5060)

By default, this feature is enabled. You may want to disable LAN dial if it conflicts with some dial numbers and/or you want to send those LAN dial strings "as is" (for example, "*123" may be reserved by ISP for server feature access).

[On]/[Off]: Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if

1		Ι	n	t	e	r	-	D	i	g	i	t		Т	i	m	e	0	u	t	
2	\checkmark	L	Α	Ν		D	i	а	1												
3		С	а	1	1		С	0	m	m	a	n	d								
4		D	i	а	1		Р	1	a	n											
5		Η	0	t		L	i	n	e												
		0	n/O	ff]	Back	K		

Call Command Settings

Configure various call commands, such as Calling Line Identification Restriction (CLIR), or Calling Line Identification Presentation (CLIP), on a per call basis. Alternative, you may change server feature access code to fit your local regulation, such as change the "Call Return" access call from "*69" to "#69".

1		М	S	А	С	:														
2		А	n	0	n	у	m	0	u	S		С	а	1	1	:	*	6	7	
3		С	L	Ι	Р	:	*	8	2											
4		Μ	W	Ι	:															
5		S	e	r	v	e	r		Η	0	1	d	:							
6		С	а	1	1		Р	i	c	k	u	р	:	*	*					
7		F	e	а	t	u	r	e		С	0	d	e		0	:				
8		F	e	а	t	u	r	e		С	0	d	e		1	:				
9	\checkmark	С	0	n	f	e	r	e	n	c	e		Α	1	e	r	t			
1	0		Х	f	e	r		0	n		E	x	i	t		С	0	n	f	
																		Bacł	K	

[Del]: Delete one character.

[Abc..]/[123..]: Toggle between digits and alphanumeric input, where. [Abc..] indicates current input method is alphanumeric and [123..] indicates digits input.

[Clear]: Clear all input.

1	Μ	S	Α	С	:														
2	Α	n	0	n	у	m	0	u	s		C	a	1	1	:	*	6	7	
3	С	L	Ι	Р	:	*	8	2											
4	М	W	Ι	:															
5	S	e	r	v	e	r		Η	0	1	d	:							
6	С	а	1	1		Р	i	c	k	u	p	:	*	*					
7	F	e	а	t	u	r	e		С	0	d	e		0	:				
8	F	e	а	t	u	r	e		С	0	d	e		1	:				
9	F	e	а	t	u	r	e		С	0	d	e		2	:				
		Del				Ab	c/12	23			(Clea	r]	Back	K	

Conference Server Access Code:

Whenever the [Conf] soft-key is pressed, IP Phone will try to initiate a 3-way local conferencing. However, some ISP provide media server for server-side conference support as well. Thus, if you want to send such access code to server, such as "*#", instead of launch a local conference after pressing [Conf] soft-key, you may define the access code here. If defined, this terminal will send specified MSAC code as DTMF sequence by either inband or outband (rfc2833 or SIP INFO) during conversation. Leave it blank for 3-way local conference on pressing [Conf] soft-key.

CLIR: Calling Line Identification Restriction.

System default is to enable Calling Line Identification Presentation, CLIP, all the time. You may enable the Call Line Identification Restriction, CLIR on a per-call basis by using the call command dial string. To enable CLIR on a per-call basis requires that the dial string sequence (typically ***67**) that users enter on their dial-pad prior to dialing the phone number match the specified CLIR string defined in the call command. Whenever the prefix is detected in dial string, it will be stripped off from the original dial string and send to proxy with CLIR enabled. The SIP INVITE message sent by IP Phone would look like:

```
INVITE sip:3101@192.168.3.101 SIP/2.0

Via: SIP/2.0/UDP 192.168.3.253:5060;branch=abc7801

Via: SIP/2.0/UDP 192.168.3.51:5060

From: "Anonymous" <sip:Anonymous@192.168.3.51>;tag=22516

To: <sip:3101@192.168.2.101>

Call-ID: 1157628352@192.168.3.117

CSeq: 1 INVITE

Contact: "Anonymous" <sip:Anonymous@192.168.3.51:5060>

User-Agent: IP-Phone/2.0

Content-Length: 171

Content-Type: application/sdp

v=0

o=ipr000dc300051 0 0 IN IP4 192.168.3.5
```

System default activation code is "*67".

CLIP: Calling Line Identification Presentation.

System default is to enable Calling Line Identification Presentation, CLIP, all the time. You may cancel CLIR by using the CLIP call command dial string.

To enable the Calling Line Identification Presentation, CLIR, on a per-call basis (to counter CLIR) requires that the dial string sequence (typically ***82**) that users enter on their dial-pad prior to dialing the phone number match the specified CLIP string defined in the call command. Whenever the prefix is detected in dial string, it will be stripped off from the original dial string and send to proxy with CLIP enabled.

System default activation code is "*82".

MWI: Message Waiting Indication

Whenever the [MWI] is pressed, The IP Phone will try to make a call to the SUBSCRIBEd MWI account. However, some ISPs provide voice mailbox access by specific server feature access code. Thus, if you want to send such server feature access code to server, such as "#", you may define the access code here. If defined, this terminal will make an outbound call to the defined code; for example, if "#" is configured, it will call to "sip:#@ISP.com". Leave it blank for terminal to call to the SUBSCRIBEd MWI account.

• Server Hold

Whenever the **[**HOLD**]** is pressed, The IP Phone will try to put the peer on hold by SIP re-INVITE, and retrieve the peer by SIP re-INVITE if the **[**HOLD**]** is pressed again. Besides, this terminal will play music locally whenever it is put on hold by the peer. However, some ISP provide hold operation as well as server generated music-on-hold stream through their servers. Thus, to integrate such server/IP-PBX, you may re-define the operation of **[**HOLD**]**. Once defined, such as "&", the terminal will generate the defined server hold access code as DTMF sequence by either inband or outband (rfc2833 or SIP INFO) during conversation to invoke server hold features. Leave it blank for terminal to perform HOLD/unHOLD operations by SIP re-INVITE locally.

• **Call Pickup:** Default is "**". The IP Phone supports directed call pickup as well as group call pickup.

Specify the group pickup code to pick up a ringing call of the same group (RFC4235-Dialog Event Page, RFC4462-Event Notify for Resource List, RFC2387-Muptipart-Related MIME type, and draft-ietf-sipping-service-examples-10.txt.

For example, if Bob and Peter are part of a work group at example.com that can pick up each others calls. Alice calls Bob who does not answer. Peter wishes to pick up the call then he could dial this specified "Call Pickup" code followed by Bob's number, such as "**Bob", where "**" is the assumed "Call Pickup" code. Then Peter's terminal will send a SUBSCRIBE to Bob to retrieve the dialog information. Bill then generates an INVITE with a Replaces to Alice. Alice answers the INVITE and sends a CANCEL to stop Bob's phone ringing. Note that the order of the CANCEL/ACK sequence in message-9 through message 14 is not significant.



#	Message Content
	INVITE sip:bob@192.168.3.101 SIP/2.0
	Content-Length: 223
	Content-Type: application/sdp
	VIa: SIP/2.0/0DP 192.168.3.57:5060;prancn=z9nG4pK4693120tp
	From: <sip:alice@isp.com·5060>:tag=489c13a17</sip:alice@isp.com·5060>
	Supported: replaces, timer
	Call-ID: 8dcb1544-78cd47b1-7de510c5-c0ab41b5@192.168.3.57
	CSeq: 66526 INVITE
	User-Agent: VIP-550PT/1.1.1
	Max-Forwards: 70
1	Session-Expires: 1800
	Contact. <sip.aice@192.106.3.37.3000></sip.aice@192.106.3.37.3000>
	v=0
	o=iprD0E9015189 4553807 4553807 IN IP4 192.168.3.57
	s=-
	c=IN IP4 192.168.3.57
	t=0 0
	m=audio 8004 RTP/AVP 0 8 4 18 101
	a=ntpmap. To T telephone-event/0000
	a=fmtp:101 0-16
	a=fmtp:4 ptime=30;bitrate=6.3
	SIP/2.0 180 Ringing
2	Content-Length: 0
2	Server: IPR-720E/1.0.1
	Via: SIP/2.0/UDP 192.168.3.57:5060;branch=z9hG4bK469312dtb

	To: <sip:bob@isp.com>;IPDIAL=1;tag=46f66d0e</sip:bob@isp.com>
	From: <sip:alice@ips.com:5060>;tag=489c13a17</sip:alice@ips.com:5060>
	Call-ID: 8dcb1544-78cd47b1-7de510c5-c0ab41b5@192.168.3.57
	CSeq: 66526 INVITE
	Contact: <sip:bob@192.168.3.101:5060></sip:bob@192.168.3.101:5060>
	SUBSCRIBE sip:bob@192.168.3.101 SIP/2.0
	Content-Length: 0
	Date: Fri, 23 Jun 2006 16:00:49 GMT
	Via: SIP/2.0/UDP 192.168.3.61:5060:branch=z9hG4bK41d57a19
	To: <sip:bob@isp.com></sip:bob@isp.com>
	From: <sip:peter@isp.com>;tag=41c6a9c</sip:peter@isp.com>
	Call-ID: 28c568f6-dc7fa18c-66e20de0-4c6b0c6d@192.168.3.61
3	Event: dialog
	Supported: eventlist
	CSeq: 40397 SUBSCRIBE
	Expires: 30
	User-Agent: ATA-740S/1.1.3
	Max-Forwards: 70
	Accept: multipart/related, application/rlmi+xml, application/dialog-info+xml
	Contact: <sip:peter@192.168.3.61:5060></sip:peter@192.168.3.61:5060>
	SIP/2.0 200 OK
	Content-Length: 0
	Server: AIA-740S/1.1.3
	Via: SIP/2.0/UDP 192.168.3.61:5060;branch=z9nG4bK4105/a19
4	IO: <sip:bob@isp.com>;tag=4/25161/</sip:bob@isp.com>
	F10111: <\$1p.peter@15P.com>;tag=41coa9c
	Call-ID. 200000-00710100-00020000-4000000000192.100.3.01
	Contact: <sin:bob@102 101:5060="" 168="" 3=""></sin:bob@102>
	NOTIEV sin:neter@192.168.3.61:5060 SIP/2.0
	Content-Length: 396
	Content-Type: application/dialog-info+xml
	Via: SIP/2.0/UDP 192.168.3.101:25320:branch=z9hG4bK4dae3c77
	Subscription-State: terminated
	To: <sip:perter@isp.com>;tag=41c6a9c</sip:perter@isp.com>
	From: <sip:bob@isp.com>;tag=47251617</sip:bob@isp.com>
	Call-ID: 28c568f6-dc7fa18c-66e20de0-4c6b0c6d@192.168.3.61
	Event: dialog
	CSeq: 2998 NOTIFY
	User-Agent: IPR-720E/1.0.1
	Max-Forwards: 70
	Contact: <sip:peter@192.168.3.101:5060></sip:peter@192.168.3.101:5060>
5	<2×ml version="1.0"2>
	<pre></pre>
	vorsion="0 state="full"
	entity="hob@ISP.com">
	<dialog_id="17f9388"< td=""></dialog_id="17f9388"<>
	call-id="8dcb1544-78cd47b1-7de510c5-c0ab41b5@192.168.3.57"
	local-tag="46f66d0e"
	remote-tag="489c13a17"
	direction="recipient" // [recipient initiator]
	remote-uri="alice@ISP.com"
	remote-target="alice@192.168.3.57:5060">
	<state>early</state> // [trying early confirmed terminated]
	SIP/2.0 200 OK
6	via: SIP/2.0/UDP 192.168.3.101:25320;branch=z9nG4bKdb0e6d
	IO: <sip:peter@isp.com>;tag=41c6a9c</sip:peter@isp.com>
	FIGH: <\$IP:DOD@ISP.COM>;tag=4/25161/
	Call-ID: 26C506T0-0C/TB18C-06E2U0EU-4C6DUC60(@192.168.3.61

	CSeq: 2998 NOTIFY
	Server: ATA-740S/1.1.3
	Contact: <sip:192.168.3.101:25320></sip:192.168.3.101:25320>
	SUBSCRIBE sip:bob@192.168.3.101 SIP/2.0
	Content-Length. 0 Date: Eri, 23, Jun 2006 16:00:40 GMT
	Vie: SID/2 0/LIDD 102 168 3 61:5060:branch=z0hC4bK41d57a10
	To: <sin:hob@isp.com></sin:hob@isp.com>
	From: <sip:peter@isp.com>:tag=41c6a9c</sip:peter@isp.com>
_	Call-ID: 28c568f6-dc7fa18c-66e20de0-4c6b0c6d@192.168.3.61
1	Event: dialog
	CSeq: 40398 SUBSCRIBE
	Expires: 0
	User-Agent: ATA-740S/1.1.3
	Max-Forwards: 70
	Accept: application/dialog-inio+xmi Contact: <sip:peter@102.168.3.61:5060></sip:peter@102.168.3.61:5060>
	INVITE sin alice@192.168.3.57 SIP/2.0
	Content-Length: 286
	Content-Type: application/sdp
	Via: SIP/2.0/UDP 192.168.3.61:5060;rport;branch=z9hG4bKb815e3f
	To: <sip:alice@isp.com></sip:alice@isp.com>
	From: <sip:peter@isp.com>;tag=dc8cec1</sip:peter@isp.com>
	Supported: replaces, timer
	Call-ID: 80469040-1730803C-1061C470-772e0015@192.168.3.61
	User-Agent: ATA-740S/1 1 3
	Max-Forwards: 70
	Session-Expires: 1800
~	Contact: <sip:peter@192.168.3.61:5060></sip:peter@192.168.3.61:5060>
9	Replaces: d1084a888dcb1544-78cd47b1-7de510c5-c0ab41b5@192.168.3.57;
	to-tag=489c13a17;from-tag=46f66d0e;early-only
	Require: replaces
	v=0 a=iprDC3000361 11173553 11173553 INLIP4 192 168 3 61
	s=-
	c=IN IP4 192.168.3.61
	t=0 0
	m=audio 8000 RTP/AVP 0 8 18 4 101
	a=rtpmap:101 telephone-event/8000
	a=ptime:20
	a=initp. 101 0-10 a=initp.4 ptime=30;bitrate=0.3
	Server: VIP-550PT/1.1.1
	Content-Length: 245
	Content-Type: application/sdp
	Via: SIP/2.0/UDP 192.168.3.61:5060;branch=z9hG4bKb815e3f
	To: <sip:bob@isp.com>;tag=50e14646</sip:bob@isp.com>
	Supported: replaces, timer
	From: <sip:peter@isp.com>;tag=0c8cec1</sip:peter@isp.com>
	Call-ID. 00409040-17500050-10010470-7720001500192.100.5.01
9	Session-Expires: 1800:refresher=uas
	Contact: <sip:bob@192.168.3.57:5060></sip:bob@192.168.3.57:5060>
	v=0
	o=iprD0E9015189 4553700 4553700 IN IP4 192.168.3.57
	S=- C=IN ID4 102 168 3 57
	t=0.0
	m=audio 8002 RTP/AVP 0 8 18 4 101
	a=rtpmap:101 telephone-event/8000
	a=fmtp:101 0-16
----	---
	a=ptime:20
	a=fmtp:4 ptime=30;bitrate=6.3
	CANCEL sip:bob@192.168.3.101 SIP/2.0
	Content-Length: 223
	Content-Type: application/sdp
	Via: SIP/2.0/UDP 192.168.3.57:5060;branch=z9hG4bK469312dfb
	To: <sip:bob@isp.com></sip:bob@isp.com>
11	From: <sip:alice@isp.com:5060>;tag=489c13a17</sip:alice@isp.com:5060>
	Call-ID: 8dcb1544-78cd47b1-7de510c5-c0ab41b5@192.168.3.57
	CSeq: 66526 INVITE
	User-Agent: VIP-550PT/1.1.1
	Max-Forwards: 70
	Contact: <sip:alice@192.168.3.57:5060></sip:alice@192.168.3.57:5060>

• Server Feature Access Codes.

Configure various feature access codes to activate server features, such as call return, voice mailbox, etc. On making outbound calls, those feature access codes are sequentially checked against. If the dial string matches any of the feature access code, the dial string will be dialed as "is" to facilitate server feature access. For example, if you configure Server Feature Access Code-0 to "*69", then whenever you dial "*69", The IP Phone will dial it as <u>sip:*69@sipDomain.com</u> to server instead of translating it as LAN dials.

Besides, you may configure a prefix match by append a 'v' suffix to those feature access codes. For example, if you configure "*#v", then when you dial "*#1234", the resultant dial string will be rendered as "sip:*#1234@sipDomain.com".

Note Such server features replies on the SIP proxy server support.

Dial Plan

The programmable dial plan is designed for the service provider to customize the behavior of IP Phone for collecting and sending dialed digits. The dial plan allows the terminal user to specify the events that trigger the sending of dialed digits. These events include the following:

- The termination character has been entered.
- The specified dial string pattern has been accumulated.
- The specified number of dialed digits has been accumulated.

Dial Plan consists of alphanumeric string, and the maximum number of characters is 255.

System default dial plan is empty.

For example, "911 | [1_9]>#.r5" denotes that:

- 1. 911 => dial out immediately
- **2.** [1_9]>#.r5
 - [1_9]: Any digit from 1 to 9.
 - (>#) press '#' to end dialing at any time.
 - (.r5): 6 digits from [0-9,*#]

It will dial out immediately whenever a dial string starts with any digit form 1 to 9, and at most 6 digits more has been collected (7 digits in total) or user press '#' to end dial.

Please refer to <u>Appendix A – Dial Plan</u> for syntax.

Key Function Description

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

D	i	a	1		Р	1	a	n	:									
	r	4																
	Del						Ab	c/12	23		(Clea	r]	Back	ζ	

Hot Line

Specify the number to dial out whenever the user hooks off.

For example, if you enable this feature, and fill in the number: "sip:888@ISP.com" or "888@ISP.com" or "888", then the terminal will dial to the specified number whenever user turns on the speaker phone or hooks off without waiting for user input.

Soft-Key Function Description

[On]/[Off] Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

[AoR] Specify the number to dial out whenever the user hooks off. For example, if you enable this feature, and fill in the number: "sip:888@ISP.com" or "888@ISP.com" or "888@ISP.com" or "888", then the terminal will dial to the specified number whenever user turns on the speaker phone or hooks off without waiting for user input.

[Back] Return to previous page.

1		Ι	n	t	e	r	-	D	i	g	i	t		Т	i	m	e	0	u	t	
2	\checkmark	L	Α	Ν		D	i	а	1												
3		С	а	1	1		С	0	m	m	а	n	d								
4		D	i	а	1		Р	1	а	n											
5		Н	0	t		L	i	n	e												
	On/Off							AoR										Back	K		

Soft-Key Function Description

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

Α	d	d	r	e	S	S	-	0	f	-	R	e	c	0	r	d	:			
	Del					Ab	c/12	23			(Clea	r			•	Bacl	K		

Message Alert

Configure whether the phone should play alerting tone on receiving and on sending messages. The alerting tone serves to remind the user that new messages have been received or messages have been sent successfully (failed).

Default is enabled.

[On] / [Off]: Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1	\checkmark	С	a	1	1		W	a	i	t	i	n	g							
2		D	i	а	1		Т	i	m	e	0	u	t							
3		Η	0	1	d		R	e	c	а	1	1		Т	i	m	e	r		
4		А	u	t	0	-	R	e	d	i	а	1								
5		D	i	а	1		Р	1	a	n										
6		Μ	e	S	S	а	g	e		А	1	e	r	t						
7		А	u	t	0	-	Α	n	S	W	e	r								
8		U	n	h	0	1	d		0	n		Т	r	а	n	S	f	e	r	
9		С	0	n	f	e	r	e	n	c	e		Α	1	e	r	t			
1	0		Х	f	e	r		0	n		Е	х	i	t		С	0	n	f	
	On/Off																	Bacl	K	

Auto-Answer

By "auto-answer", the phone-set will play a distinguished auditable tone shortly (unless "Silent" is specified), then answer the call by turning on speaker-phone (or headset if applicable) during idle period. By idle, it either takes no call or is putting a call on hold. Otherwise, the incoming call will proceed as normal incoming calls (play call-waiting tone to remind user of a call is waiting for answer).

[On] / **[Off]**: Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1		А	u	t	0	-	А	n	S	W	e	r		А	1	1			
2		С	0	n	t	r	0	1		L	i	S	t						
	On/Off																Back	K	

Once "Auto-Answer All" is enabled, it will auto-answer all incoming calls on idle mode. By idle, it means the phone-set is either engaging in no calls or putting a call on hold. It will play distinguished auditable ring then turn on speaker-phone (or head-set if applicable) whenever an incoming call arrives.

Default is disabled

	1.	If this featured is mapped as a programmable DSS function key, pressing the mapped DSS key will toggle the auto-answer mode. Besides, unmap of this DSS key will NOT turn off auto-answering.
	2.	If 【DND】 is on, Do-Not-Disturb feature wins; otherwise, for All-Calls-Forward and Auto-Answer features, All-Calls-Forward takes precedence.
Vote Note	3.	The IP Phone supports server-side invoked auto-answer feature, please refer to "Auto Answer" on "User's Manual" for detail.
	4.	The IP Phjone supports auto-answer based on called number as well. That is, if you registered several SIP address-of-records, you could configure to auto-answer calls arrived on specific SIP AoR(s). Please refer to "7.SIP Settings" / "Service Domain" / "Auto-Answer" for how to auto-answer incoming calls arrived on specific accounts.
	5.	On idle, all incoming calls will check against the server-side invoked auto-answer feature first, and then the global auto-answer feature, finally fall to the account-specific auto-answer feature.

Auto-Answer Control List

User controlled auto-answer list. Whenever an incoming call arrives, this terminal will check the SIP address-of-record specified in the <From> header in the received INVITE message against user-defined auto-answer control list. If a match exists, this incoming call will be auto-answered based on their configured priority.

By "auto-answer", the phone-set will play a distinguished auditable tone shortly (unless "Silent" is specified), then answer the call by turning on speaker-phone (or headset if applicable) during idle period. By idle, it either takes no call or is putting a call on hold. Otherwise, the incoming call will proceed as normal incoming calls (play call-waiting tone to remind user of a call is waiting for answer).

Soft-Key Function Description

[Level]	Modify anto-answer level.
[AoR]	Set address-of-record.
[Del]	Remove this entry.
[Back]	Return to previous page.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

1	Е	n	t	r	у	1		
2	E	n	t	r	y	2		
3	E	n	t	r	y	3		
4	Е	n	t	r	y	4		
5	E	n	t	r	y	5		
]	Leve	1			AoR	Back	K

Level

Auto-Answer Priority:

- 1. Normal
 - Respond a "486 Busy Here" if all lines are occupied.
 - Otherwise, **auto-answer** this incoming call.

2. Urgent

- Dropped an inactive call if all lines are occupied.
- Put any on-going calls to hold and auto-switch to an available line if not idle.
- **auto-answer** the newly arrived call.

3. Imperious

Identical to "**Urgent**" except that the phone-set will auto-answer this call even the do-not-disturb (DND) or all-calls-forward feature has been activated.

4. Silent

Identical to "Imperious" except that no distinguished auditable tone will ever be played. This is for supervised monitor/announce feature.

1	Ν	0	r	m	a	1								
2	U	r	g	e	n	t								
										Bacl	K			

Incoming call processing rules (by precedence):

- 1. If "silent" or "Imperious" is specified, then auto-answer it. Besides, if "silent" is specified, then no "distinguished auditable tone" will be played.
- 2. If the phone is engaged in do-not-disturb mode, then DND wins
- 3. Otherwise, if the phone has turned on unconditionally forward feature, then all incoming calls are forwarded.
- 4. Check for auto-answer feature:
 - a. Check for server-side auto-answer feature.
 - b. Check for the global switch for auto-answer feature.
 - c. Match this call against all enabled service domains for auto-answering feature, if a match is located, auto-answer it.
- 5. Otherwise, proceed as normal incoming calls.

AoR

Specify which caller (<From> header in the received INVITE SIP message) should be auto-answered.

Soft-Key Function Description

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

Α	d d r e				S	s	-	0	f	-	R	e	c	0	r	d	:			
	Del						Ab	c/12	23			(Clear	r]	Back		

Unhold on Transfer

Specify whether to unhold the transferee (as well as the transfer target in the case of attended transfer) before issuing REFER to finish transfer.

Default is unchecked (recommended).

[On] / **[Off]**: Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1		С	а	1	1		W	а	i	t	i	n	g							
2		D	i	а	1		Т	i	m	e	0	u	t							
3		Η	0	1	d		R	e	c	а	1	1		Т	i	m	e	r		
4		Α	u	t	0	-	R	e	d	i	a	1								
5		D	i	а	1		Р	1	a	n										
6		Μ	e	S	S	a	g	e		Α	1	e	r	t						
7		Α	u	t	0	-	Α	n	S	W	e	r								
8		U	n	h	0	1	d		0	n		Т	r	а	n	s	f	e	r	
9		С	0	n	f	e	r	e	n	c	e		Α	1	e	r	t			
1	0		Х	f	e	r		0	n		E	X	i	t		C	0	n	f	
		0	n/O	ff														Bacl	k	

Conference Alert

Specify whether to play alerting tone during an ad hoc 3-way conference.

Once enabled, terminal will regularly (every 60 seconds) play alerting tone during an ad hoc 3-way local conference to all attendees to reminde them a conference is undergoing.

Default is enabled

[On] / **[Off]**: Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1	\checkmark	С	a	1	1		W	a	i	t	i	n	g							
2		D	i	а	1		Т	i	m	e	0	u	t							
3		Н	0	1	d		R	e	c	а	1	1		Т	i	m	e	r		
4		Α	u	t	0	-	R	e	d	i	a	1								
5		D	i	а	1		Р	1	а	n										
6	\checkmark	Μ	e	s	s	а	g	e		Α	1	e	r	t						
7		А	u	t	0	-	А	n	S	W	e	r								
8		U	n	h	0	1	d		0	n		Т	r	а	n	S	f	e	r	
9	\checkmark	С	0	n	f	e	r	e	n	c	e		А	1	e	r	t			
1	0		Х	f	e	r		0	n		Е	х	i	t		С	0	n	f	
		0	n/O	ff														Bacl	K	

Figure 29. Preferences Menu

Xfer on Exit Conf

The Phone supports ad hoc 3-way local conference. After setting up a conference, your terminal will serve as a media mixer such that the other two parties can hear each other. However, when you hang up, all parties in this conference will be disconnected because there is no media mixer.

Under some circumstances, you may want to keep the other two parties connected even after you exit this conference. By enabling this feature, this terminal will carry out an attended transfer on exiting ad hoc 3-way local conference. As a result, the other two parties may keep conversation without your involvement.

[On] / **[Off]:** Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1		С	а	1	1		W	a	i	t	i	n	g							
2		D	i	а	1		Т	i	m	e	0	u	t							
3		Н	0	1	d		R	e	c	а	1	1		Т	i	m	e	r		
4		Α	u	t	0	-	R	e	d	i	a	1								
5		D	i	а	1		Р	1	a	n										
6		Μ	e	S	S	a	g	e		Α	1	e	r	t						
7		Α	u	t	0	-	Α	n	S	W	e	r								
8		U	n	h	0	1	d		0	n		Т	r	a	n	S	f	e	r	
9		С	0	n	f	e	r	e	n	c	e		Α	1	e	r	t			
1	0		Х	f	e	r		0	n		Е	х	i	t		С	0	n	f	
		0	n/O	ff														Bac	k	

Figure 30. Preferences Menu

Network

Soft-Key Function Description

[Ping]	To ping the remote host for its availability or to verify the network configuration of the phone. If no replies are received within a 5-second timeout period, it will show "Host unreachable."
[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
\uparrow and \downarrow	Move cursor per line in edit mode.
[OK]	Start to ping.

1	А	c	t	i	v	e		S	t	а	t	u	S					
2	G	e	n	e	r	a	1											
3	R	Т	Р		S	e	t	t	i	n	g	S						
4	Ν	Α	Т		&		F	i	r	e	W	а	1	1				
5	S	Ν	Μ	Р														
6	V	L	Α	N														
		Ping	5													Bac	k	

Figure 31. Netowrk Menu

Enter the remote host to ping by sending ICMP echo request packets. You could use either a dotted IP or a DNS name.

Р	i	n	g															
1	а	r	r	у		g	1	0	0	n	e	t						
-						1												-
	Del						Ab	c/12	23		(Clea	r]	Back		

Р	i	n	g															
Ι	С	М	Р	r	e	р	1	y	f	r	0	m	1	a	r	r	у	
g	1	0	0	n	e	t												
]	Back	2		

Active Status

Display currently active host IP, Ethernet MAC address and the active DNS IP (secondary DNS will be shown if available), LAN and PC Ethernet link status in order. The phone will set MAC as it host name in 12 hexadecimal characters.

Ι	Р	:	1	9	2		1	6	8		3		2	0	0					
М	А	С	:	0	0	0	Е	С	3	1	2	3	4	5	6					
D	Ν	S	:	1	9	2		1	6	8	•	3		1						
D	Ν	S	:	1	9	2		1	6	8	•	3		2	5	4				
L	А	Ν	:	1	0	0	Μ	,	F	u	1	1	-	D	u	р	1	e	X	
Р	С	:	:	D	0	W	n													
]	Back	ζ.	

LAN and PC Ethernet link status are one of the following:

- 1. **Down:** not connected
- 2. Up: Ethernet jack connected
 - 100M, Full: 100 MB, Full-Duplex (Usually connects to a 100-MB Ether-Switch).
 - 100M, Half: 100 MB, Half-Duplex (Usually connects to a 100-MB Ethernet hub).
 - 10M, Full: 10 MB, Full-Duplex (Usually connects to a 10-MB Ether-Switch).
 - 10M, Half: 10 MB, Half-Duplex (Usually connects to a 10-MB Ethernet hub).

General

1	М	0	d	e															
2	S	t	а	t	i	c		S	e	Т	t	i	n	g	S				
3	Р	Р	Р	0	Е		S	e	t	Т	i	n	g	S					
4	D	Ν	S		S	e	r	v	e	R									
5	Η	0	S	t		Ν	a	m	e										
6	Т	i	m	e	-	t	0	-	L	Ι	v	e							
]	Back	5	

Figure 32. General Menu

Mode

Choose the protocol to configure host IP, gateway, network mask and / or domain name server. Default is "DHCP".

1		D	Н	С	Р													
2	•	S	t	a	t	i	c	Α	S	S	i	g	n					
3		Р	Р	Р	0	Е												
															B	ack		

Figure 33. Mode select list

Static Settings

Static assign host IP, network mask and gateway IP. **Default IP is "192.168.000.001"**

1	Н	0	S	t		Ι	Р									
2	Ν	e	t	w	0	r	k	Μ	а	S	k					
3	G	а	t	e	W	a	у	Ι	Р							
]	Back		

Figure 34. Static IP Settings

Η	0	S	t		Ι	Р	:											
1	9	2		1	6	8		0	0	3	2	0	0					
]	Back		

Figure 35. Enter IP Address



You should enter "0.0.0.0" to disable gateway routing. That is, if you reside on a LAN without gateway, you must set this to "0.0.0.0" rather than a non-existent IP; otherwise the network packets may not be routed correctly (which may result in no voice packets could be sent from this phone)! This constrain applies to DHCP and PPPoE as well: DHCP and PPPoE server should not designate a non-existent or invalid gateway.

PPPoE Settings

Point-to-Point Protocol over Ethernet (PPPoE) for xDSL dialup.

1	U	S	e	r		Ν	а	m	e								
2	Р	а	S	S	W	0	r	d									
3	S	e	r	v	i	c	e		Ν	a	m	e					
]	Back		

User Name

Name used by the phone for PPPoE authentication during dialup.

[Abc..]/[123..]: Toggle between digits and alphanumeric input, where. [Abc..] indicates current input method is alphanumeric and [123..] indicates digits input.

L	0	g	i	n		U	S	e	r	Ν	а	m	e	:		
	Del						Ab	c/12	23		(Clea	r	В	ack	

Password

Password used by the phone for PPPoE authentication during dialup.

Initially show 10 '*' signs but show clear text on user input because alphanumeric characters may be needed.

Р	а	S	S	W	0	r	d	:										
1	2	3	4															
	Del					Ab	c/1 2	23		(Clear	r]	Back			

Service Name

Some ISP require the use of service name.

[Abc..]/[123..]: Toggle between digits and alphanumeric input, where. [Abc..] indicates current input method is alphanumeric and [123..] indicates digits input.

S	e	e r v i				e		Ν	а	m	e	:							
	Del						Ab	c/12	23				Clea	r]	Back		

DNS Server

Configure to obtain domain name server by DHCP, or use the one statically assigned. The configured DNS server will be activated if "Static DNS" is enabled.

Default is disable to utilize the DNS servers acquired by DHCP.

[On] / [Off]: Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1	\checkmark	S	t	а	t	i	c		D	Ν	S								
2		Р	r	i	m	a	r	у		D	Ν	S							
3		S	e	c	0	n	d	a	r	у		D	Ν	S					
		()n/()	ff]	Back		

Statically assign the primary domain name server IP.

Please enter "000.000.000" to disable this setting.

Generally, those IP(s) are invalid:

- 1. xxx.xxx.xxx.000 (this is LAN ID if network mask is 255.255.255.0)
- 2. xxx.xxx.xxx.255 (this is LAN broadcast IP if network mask is 255.255.255.0)
- 3. 255.255.255.255 (Broadcast IP)
- 4. 127.0.0.1 (localhost IP)

D	N	S	S	e	r	v	e	r	:								
1	9	2	1	6	8		0	0	3	2	5	4					
]	Back		

Host Name

Set host name of this terminal which will be used on DHCP request for better network administration as in

Η	0	S	t	Ν	a	m	e	:									
c	р	e	1														
			Del			Ab	c/1 2	23		(Clea	r]	Back	Σ.	

Time-to-Live

Configure the TTL value of IP packet. The default value is 144. For RTP packets the TTL will use a value higher than 127, and a value higher or equal to 75 for SIP signaling.

Ι	Р		Р	а	c	k	e	t	Т	Т	L					
[7	5	-	2	5	5]									
1	4	4														
			Del											Back	K	

RTP Settings

Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1	Р	0	r	t		В	а	S	e	:	8	0	0	0					
2	Р	0	r	t		R	а	n	g	e	:	6							
3	R	Т	Р		Ι	Р		Т	0	S	:	0	x	В	8				
4	R	Т	С	Р		Ι	Р		Т	0	S	:	0	x	7	4			
5	S	у	m	m	e	t	r	i	c		F	1	0	W					
																J	Back		



Port Base

Configure the starting Real-time Transport Protocol (RTP/RTCP) port for media transmission. Conventionally, the RTP will use an even UDP port, and the RTCP will use a UDP port one higher than that of RTP; such that the port base should be an even number and larger than 3000 to avoid conflict. We suggest a base higher than 3000; however, it accepts a value between 2 and 65534 to accommodate single line configuration.

Default is 8000

R	Т	Р		Р	0	r	t		В	а	S	e					
[2	-	6	5	5	3	4]									
8	0	0	0														
			Del												Back	K	

Port Range

Configure the number of ports which RTP will be used for media transport. It is an offset from the pre-configured Base port. Each channel takes 2 ports, one for RTP and the other for RTCP session. Conventionally, the RTP will use an even UDP port, and the RTCP will use a UDP port one higher than that of RTP. As a result, the range specified here should be an even number and at least doubles the lines equipped on the phone. For example, if the phone has two lines, then the least port range requirement is 4. We suggest that a value that triples the lines should be more suitable as a minimum requirement; however, it accepts a value between 2 and 100 to accommodate single line configuration.

Default is 16

R	Т	Р		Р	0	r	t	R	а	n	g	e					
[2	-	1	0	0]											
1	6																
			Del												Bacł	K	

PTP IP ToS

Specify the IP type-of-service (ToS) bits of RTP packets transmitting voice streams. The IP ToS bits consist eight bits:

7 (MSB)	6	5	4	3	2	1	0(LSB)
	Precedence		Delay	Throughput	Reliability	Reserv	ed (00)

Where MSB denotes "Most Significant Bit" and LSB denotes "Least Significant Bit".

- 1. Bits 5-7 specify the precedence:
 - 111 Network Control
 - 110 Internetwork Control
 - 101 CRITIC/ECP (RTP default)
 - 100 Flash Override
 - 011 Flash (RTCP default)
 - 010 Immediate
 - 001 Priority

- 000 Routine (normal)
- 2. Bit 4: 0 = Normal Delay, 1 = Low Delay (RTP/RTCP default).
- 3. Bit 3: 0 = Normal Throughput, 1 = High Throughput (RTP default).
- **4.** Bit 2: 0 = Normal Reliability, 1 = High Reliability (RTCP default).
- 5. Bit 0-1: Reserved for Future Use.

System default is 0xb8 (184 in decimal) for RTP and 0x74 (116 in decimal) for RTCP packets. Please enter it in decimal.

Ι	Р		Т	у	р	e	-	0	f	-	S	e	r	v	i	c	e			
[0	-	2	5	5]														
1	8	4																		
			Del															Bacl	K	

Symmetric RTP Flow

The IP Phone supports **symmetric RTP flow** for the cases where only one endpoint is behind a NAT, and RTP packet flow will be possible in at least one direction. Since a client behind a NAT can usually successfully send RTP packets to another client in the public Internet, in a symmetric mode, RTP sent in the other direction could be sent to the address and port that RTP was received from. As a result, even if you are behind a NAT and choose "**Full Access**" as your way to traverse NAT, you could still reach another IP Phone on the public internet (since the peer would adjust RTP flow accordingly).

Generally, if your terminal is behind a symmetric NAT, which cannot be traversed by STUN, or your terminal is NATed, you should activate this feature. And the other party on the WAN (or capable of traversing the NAT it currently behind without problem) should NOT enable this feature.

Once "Symmetric RTP flow" is enabled, IP Phone will send its INVITE message stipulating its desire for "Symmetric RTP flow" in SDP during call setup phase if the "UDP traversal" mode configured in [Main Menu] =>"6.Network" / "NAT & firewall" / "UDP Traversal" is either "Full access" or "Enable STUN". That is if you have configured "Static NAT map" as your way to traverse NAT, it will not request the peer to enable symmetric RTP flow). The SDP sent by this UAC would be:

o=SIP-Phone 28908445311 28908445311 IN IP4 192.168.3.51 s=t=0 0 c=IN IP4 192.168.3.51 m=audio 35000 RTP/AVP 4 18 0 8 101 a=rtpmap:101 telephone-event/8000 **a=direction:active IN IP4**

If the UAS supports this extension (which VIP-350PT/ VIP-550PT does), it will wait for RTP packets to be received from the client behind the NAT before sending, the answer SDP will be:

v=0	
o=SIP-Phone 28908445405 28908445405 IN IP4 192.168.3.101	
s=-	
t=0 0	

c=IN IP4 192.168.3.101 m=audio 35000 RTP/AVP 18 a=rtpmap:101 telephone-event/8000 a=direction:passive IN IP4

Note For backward compatibility with other SIP phones, system default is to disable "Symmetric RTP flow", thus the "a=direction" attribute will not appear in the exchanged SDP.

To make **symmetric RTP flow** works, the UAS (which sends passive RTP flow) must locate on the WAN. For IP Phone, it must comply with one of the following conditions:

- It resides on the public Internet and has public IP (Full-Access).
- It has configured to employ "2.Static NAT map" as its way to traverse its NAT from [Main Menu] =>"6.Network" / "NAT & firewall" / "UDP traversal", and you have configured your NAT to statically map the corresponding UDP ports (both SIP signaling and RTP ports) to your phone.
- It is behind a NAT which can be traverse by STUN and you have configured your IP Phone to employ STUN to traverse you NAT.
- It is behind a UPnP-capable NAT and you have configured your terminal to employ UPnP to dynamically MAP your UPnP-aware NAT.

1	Р	0	r	t		В	a	s	e	:	8	0	0	0					
2	Р	0	r	t		R	а	n	g	e	:	6							
3	R	Т	Р		Ι	Р		Т	0	S	:	0	x	В	8				
4	R	Т	С	Р		Ι	Р		Т	0	S	:	0	x	7	4			
5	S	у	m	m	e	t	r	i	с		F	1	0	W					
	On/Off																Back	K	

[On] / **[Off]:** Toggle between enable and disable this feature. Show check symbol ' $\sqrt{}$ ' in-line if enabled!

Network & Firewall

To properly configure your NAT traversal scheme, refer to "NAT Traversal" on "User's Manual".

1	S	Т	U	Ν																
2	U	D	Р		Т	r	a	v	e	r	S	a	1							
3	S	t	a	t	i	c		N	Α	Т		Ι	Р							
4	D	y	n	a	m	i	c		Ν	Α	Т		Ι	Р						
5	D	i	a	g	n	0	s	e		U	Р	n	Р							
												Back								

STUN

Simple Traversal of UDP through NATs (STUN) is a lightweight protocol that allows applications to discover the presence and types of Network Address Translators (NATs) and firewalls between them and the public Internet. It also provides the ability for applications to determine the public IP addresses allocated to them by the NAT.

1	S	Т	U	Ν		S	e	r	v	e	r						
2	D	i	a	g	n	0	S	e		Ν	А	Т					
															Back	ζ.	

STUN Server

Configure the Simple Traversal of UDP through NAT protocol server. You may use a dotted IP or a DNS name. If you fill in the DNS name, such as "isp.com", IP Phone will try to locate the STUN server of "isp.com" domain via DNS SRV query (_stun._udp.isp.com). If the STUN SRV record query failed, the IP Phone will use the IP of "isp.com" as the default STUN server with UDP port 3478. Therefore, if the DNS returns more than one STUN SRV records, IP Phone will switch to the secondary STUN server whenever it fails to reach the primary STUN server (STUN server redundancy). Besides, IP Phone will switch between STUN servers only if it fails to reach the primary STUN server for more than 30 seconds to avoid instability in switching back and forth.

Default is "larry.gloo.net".

To use a different STUN server port other than the default UDP port 3478, please specify it in the format like "STUN.isp.com:8888" or "172.16.0.1:8888".

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

S	Т	U	Ν		S	e	r	v	e	r									
1	а	r	r	у		g	1	0	0		n	e	t						
	Del						Ab	c/12	23			(Clea	r		I	Back		

Diagnose NAT

Detect the NAT IP and the NAT type this UA is currently behind (8 possibilities).

The IP Phone will communicate with the STUN server (running outside the NAT, assigned in "6.Network" / "Firewall & NAT" / "STUN Server") to determine the type of NAT it is behind and the diagnosis may take a quite while (roughly 10 seconds or so in a worse case). If the phone fails to contact a STUN server, it will report it as "No STUN server is available"; otherwise, it will show the IP of the NAT / firewall.

- Public internet (Neither NAT nor Firewall is involved)
- Firewall blocks UDP (Definitely, you cannot communicate with parties outside of your LAN!).

VNote

This may also be a false alarm if the specifed STUN server is either down or unreachable.

• Symmetric UDP Firewall (STUN may work!)

Firewall that allows UDP out and responses have to come back to the source of the request (like a symmetric NAT, but no translation. We call this a symmetric UDP Firewall).

• Full cone NAT (Enable STUN should suffice).

A full cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Furthermore, any external host can send a packet to the internal host, by sending a packet to the mapped external address.

• Restricted cone NAT (STUN may work!)

A restricted cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Unlike a full cone NAT, an external host (with IP address X) can send a packet to the internal host only if the internal host had previously sent a packet to IP address X.

Port restricted cone NAT (STUN may work!)

A port restricted cone NAT is like a restricted cone NAT, but the restriction includes port numbers. Specifically, an external host can send a packet, with source IP address X and source port P, to the internal host only if the internal host had previously sent a packet to IP address X and port P.

• Symmetric NAT

A symmetric NAT is one where all requests from the same internal IP address and port, to a specific destination IP address and port, are mapped to the same external IP address and port. If the same host sends a packet with the same source address and port, but to a different destination, a different mapping is used. Furthermore, only the external host that receives a packet can send a UDP packet back to the internal host.

Non full cone NAT

If failed to identify the NAT type, it will show "Non Full cone NAT" (such as the one bundled in Win2000).

	1.	If failed to identify the NAT type, it will show "Non Full cone NAT" (such as the
4 Note	2.	If the diagnosis is success, it will show the mapping of SIP signaling port on NAT / Firewall as well
	3.	The red font indicates that the phone has problem traversion the NAT / Firewall of that type, and the user should consider manually configure the port mappings on NAT / Firewall

[Back]	Return without any changes.
[0K]	Save changes and return.

 \uparrow and \downarrow Move cursor per line if there are too many lines to fit into one screen.

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S	у	Μ	m	e	t	r	i	c	Ν	А	Т	(2	2	2		6	8	1
8	7		1	3	5)													
]	Back		

UDP Traversal

Choose "**Full access**" if the phone is not behind a NAT / Firewall or you are behind a SIP-aware NAT. Otherwise, if you are behind a NAT / Firewall, choose "**Static NAT map**" whenever possible. Besides, if the NAT / Firewall is either a Symmetric NAT or a Firewall that blocks UDP you must choose "**Static NAT map**" as well. If the NAT/Firewall/Gateway/Router in your LAN is UPnP-aware (Universal Plug and Play), you may turn on UPnP to dynamically open and close TCP/UDP ports on these devices for SIP signaling and RTP media streaming; otherwise choose STUN, Simple Traversal of UDP through NAT, as your last resort.

Generally, if your IP falls within the ranges: 10.0.0.0 to 10.255.255.255, 172.16.0.0 to 172.31.255.255, and 192.168.0.0 to 192.168.255.255, then you may have trouble making and taking calls from the public WAN.

Default is Full access.

IP Phone supports **symmetric RTP flow** for the cases where only one endpoint is behind a NAT, and RTP packet flow will be possible in at least one direction. Since a client behind a NAT can usually successfully send RTP packets to another client in the public Internet, in a symmetric mode, RTP sent in the other direction could be sent to the address and port that RTP was received from. As a result, even if you are behind a NAT and choose "**1.Full Access**" as your way to traverse NAT, you could still reach another IP Phone on the public internet (since the peer would adjust RTP flow accordingly). Generally, if your terminal is behind a symmetric NAT, which cannot be traversed by STUN, or your terminal is NATed, you should activate this feature. And the other party on the WAN (or capable of traversing the NAT it currently behind without problem) should **NOT** enable this feature.

[Back] Return without any changes.

[OK] Save changes and return.

 \uparrow and \downarrow Move cursor per line if there are too many lines to fit into one screen.

1		F	u	1	1		А	c	c	e	S								
2		S	t	a	t	i	c		Ν	Α	Т		Μ	а	р				
3	•	Е	n	a	b	1	e		S	Т	U	Ν							
4		Е	n	a	b	1	e		U	Р	n	Р							
]	Back		

Static NAT IP

Assign the static NAT IP to use by SIP signaling and RTP media stream. You could use either a dotted IP or a DNS name. This field is valid if and only if you choose "**Static NAT map**" from the "**UDP traversal**" configuration. You may detect your NAT IP by STUN from menu-"**Diagnose NAT**" (please fill in only the IP, excluding the diagnosed SIP port).

 You should also map the SIP signaling port, RTP port base and RTP port range used for media stream from NAT to the phone to enable UDP traverse through NAT / Firewall (if there are several phones reside under the same NAT, their SIP and RTP ports must not be overlapped since they all share the same NAT resource).

Note Note

 If your NAT uses dial-up modem or connects to the WAN via ADSL by PPPoE, then the IP of your NAT may change whenever it re-dials up. Please synchronize it if the NAT does not come with a fixed IP. Besides, you may detect your NAT IP by STUN from menu- "Diagnose NAT" and re-configure the new NAT IP into menu-"Static NAT IP".

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
\leftarrow and \rightarrow	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line if there are too many lines to fit into one screen.

S	t	а	t	i	c		Ν	А	Т	Ι	Р							
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	_													-				
			Del				Ab	c/12	23		(Clear	r]	Back		

Dynamic NAT IP

If your NAT uses dial-up modem or connect to the WAN via ADSL by PPPoE, then the IP of your NAT may change whenever it re-dials up. If you choose "**1.Notify on IP Change**", the phone will notify you by sending an instant message into your inbox whenever it detects a mismatch between the NAT you configured into menu-"Static NAT IP" and the one it auto-discovered by STUN. On the contrary, if you choose "**2.Auto-Update by STUN**", it would auto-refresh the NAT IP into menu-"**Static NAT IP**" whenever it detects an inconsistency.

Default is "1.Notify on IP Change" and the detection interval is 20 minutes.

[Back]	Return without any changes.
[OK]	Save changes and return.
[↑] and [↓]	Move cursor per line if there are too many lines to fit into one screen.

1	Ν	0	t	i	f	у		0	n		Ι	Р		С	h	а	n	g	e	
2	А	u	t	0	-	U	р	d	a	t	e		b	у		S	Т	U	Ν	
															Back					

Note

Either option requires a viable STUN server has been configured into menu-"STUN server"

Diagnose UPnP

Detect the external IP of UPnP-capable NAT/Gateway/Firewall devices (support either WAN IP Connection and WAN PPP Connection). It will search for UPnP devices and show its external IP.

If such devices existed, this terminal will open UDP/TCP mapping on NAT device for media streaming and SIP signaling by UPnP as necessary; Otherwise if there is no such devices co-exist (or turn off UPnP explicitly), it may show "Not found UPnP device". Consult to your LAN administrator for further information.



The UPnP mode must have been enable before you can dignose UPnP device on your LAN.

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[OK] Save changes and return.

 \uparrow and \downarrow Move cursor per line if there are too many lines to fit into one screen.

U	Р	n	Р	:										
]	Back	K	

SNMP

The IP Phone supports Simple Network Management Protocol (SNMP) v2 and Management Information Base (MIB) v2 for SNMP Get/Set/Trap operation. It supports the following (RFC1213)

- "system" group : "sysObjectID" represents its model.
- "interface" group : Only Ethernet ports are included in the table. "ifAdminStatus" is effective. For LAN port, "ifIndex is 1, and 255 for Ethernet WAN port.
- "IP" group
- "ICMP" group
- "TCP" group
- "UDP" group
- "SNMP" group.

Besides, it supports proprietary enterprise IAD MIB to Get /Set current configuration by SNMP as well, including:

- Hardware type & version, software version
- MAC address of WAN/LAN
- Number of port (phone/Ethernet), Call-Channel 【A / B】 status
- Real-time parameters, including gateway, DNS, SNMP, RTP, CODEC, etc.
- SIP settings, Registrar server, and Proxy server.
- Image upgrade and restart.

[Back]	Return without any changes.
[OK]	Edit selected item.
[\uparrow] and [\downarrow]	Navigate through menu items.

1	R	e	а	d		С	0	m	m	u	n	i	t	у				
2	W	r	i	t	e		С	0	m	m	u	n	i	t	у			
3	Т	r	a	p		С	0	m	m	u	n	i	t	y				
4	Т	r	a	p		Μ	a	n	a	g	e	r						
																Back	K	

- 1. Read Community: Set the "read community" for SNMP Get operation. Default is "public" (case-sensitive).
- 2. Write Community: Set the "write community" for SNMP Set operation. Default is "private" (case-sensitive).

Values set by SNMP Set Operation will not take effect immediately. In order to facilitate multiple values set in a succession, IP Phone will not apply new values set by SNMP till it returns to idel mode for more than 5 seconds. Besides, a second new value application will not execute if the previous one was done within 5 seconds ago.

- **3.** Trap Community: Specify the community name used by the SNMP manager to verify traps. **Default is "public" (case-sensitive).**
- 4. Trap Manager : Specify the SNMP manager to which the SNMP trap sends. You could use either a dotted IP or a DNS name to specify the SNMP agent. The IP Phone will issue a trap to SNMP manager when:
 - System startup
 - System restart (by user/SNMP command/image upgrade)
 - Call-Channel 【A/B】 Disabled
 - Registrar Server availability.

Note

The IP Phone is capable of reporting traps to two different SNMP managers. The second trap manager (not a backup manager) could only be set via auto-provision or by SNMP browser to set it. If both trap managers are available, The IP Phone will report each trap to both of them.

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Back]	Return without any changes.
[0K]	Edit selected item.
\leftarrow and \rightarrow	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

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VLAN

The IP Phone supports IEEE 802.1Q/802.1P/802.1D VLAN and IP class of service (differentiated service in IP ToS bits). VLAN support will insert a 4-byte tag immediately after source address field in the 802.3 Ethernet frame.

Destination MAC Address (6-byte)	Source MAC Address (6-byte)	VLAN tag (4 bytes)	LAN Type 0x0800 (2-byte)	Dat	а		FCS
			<u> </u>				
VL/	AN Protocol ID:(0x8100 (16 k	oits)	Priority (3 bits)	CFI	VLAN	ID (12 bits)

Once IEEE 802.1Q VLAN tagging is enabled, it will also map IP differentiated service set in ToS bits to user priority field.

[Back] Return without any changes.	
[OK] Edit selected item.	
[↑] and [↓] Navigate through items	

1	А	c	t	i	V	а	t	i	0	n							
2	V	L	Α	Ν		Ι	D	:	4	0	9	4					
3	V	L	Α	Ν		С	0	S	:	6							
	C	n/O	ff]	Back	K	

VLAN ID

IEEE 802.1Q VLAN ID is 12-bit.

The following VLAN ID values are reserved:

VLAN ID	Meaning / Usage
0	The null VLAN ID, indicating that the tag header contains only user_priority information, no VLAN identifier is present in the frame. This VID value shall not be configured as a PVID, configured in any Filtering Database entry, or used in any Management operation.
1	The default PVID value used for classifying frames on ingress through a Bridge Port. The PVID value can be changed by management on a per-Port basis
0xFFF	Reserved for implementation use. This VID value shall not be configured as a PVID, configured in any Filtering Database entry, used in any Management operation, or transmitted in a tag header.

If you disable VLAN tagging (disable menu-Activation), the VLAN ID will not take effect and be displayed as "N/A" (not applicable).

[Del]	Delete one character.
[Back]	Return without any changes.
[OK]	Edit selected item.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.

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[1	-	4	0	9	4]								
1															
	Del										Back	K			

VLAN CoS

IEEE 802.1P VLAN User priority field, default is 6; where 0 is normal and 7 is the highest priority.

[Del]	Delete one character.
[Back]	Return without any changes.
[OK]	Edit selected item.
\leftarrow and \rightarrow	Move cursor one position in edit mode.

V	L	Α	Ν		С	1	а	S	S	-	0	f	-	S	e	r	v	i	c	e
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Chapter 3 SIP Service Configurations

Configuring SIP setting for IP Phone

SIP is a request-response protocol, dealing with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol. SIP determines the end system to be used for the session, the communication media and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination.

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[OK] Edit selected item.

[\leftarrow **] and [** \rightarrow **]** Move cursor one position in edit mode.

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2	S	e	r	v	i	c	e		D	0	m	а	i	n				
3	E	Ν	U	Μ		&		E		1	6	4						
]	Bacł	K	

SIP Service

Configure the SIP stack on the phone, including signaling port, transport protocol (TCP or UDP) to use and the default registration expiration timeout (measured in seconds).

[Back]	Return without any changes.
[OK]	Edit selected item.
[\uparrow] and [\downarrow]	Navigate through items.

Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1		Т	r	а	n	S	р	0	r	t	:	3								
2		S	Ι	Р		Р	0	r	t	:	5	0	6	0						
3		E	X	р	i	r	e	S	:	1	8	0	0							
4		r	р	0	r	t														
5		Η	0	1	d		b	y		R	F	С	3	2	6	1				
6		S	i	g	n	a	1		Ι	Р		Т	0	S		0	x	В	4	
7		S	e	S	S	i	0	n		Т	i	m	e	r		1	8	0	0	
8		Κ	e	e	p		Α	1	i	v	e		Т	i	m	e	r			
9		E	X	р	0	S	e		S	W		V	e	r						
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Transport

By default, IP Phone supports both UDP and TCP for SIP signalling, and this setting prioritize transportation protocols on making requests. UDP bears a higher priority. If you specify TCP only, then IP Phone will try to make requests to the peers/servers by TCP, and fall back to UDP if the peers/servers do not support TCP.

The default is UDP and TCP. Currently there is no TLS support.

[Back]	Return without any changes.
[OK]	Edit selected item.
[\uparrow] and [\downarrow]	Scroll menu items.

1	U	D	Р												
2	Т	С	Р												
3	U	D	Р	а	n	d	Т	С	Р						
]	Back		

SIP Port

Note, to facilitate "**Contact Dialing**", "**IP Dialing**" and "**LAN Dialing**"), where most users forget to dial the SIP signaling port of the peer, and end in no responses if the peer doesn't listen on the standard UDP port 5060 for SIP signaling, the IP Phone will always listens on UDP-5060 for SIP signaling in addition to the user configured SIP service port. That is, if you configure it to listen on port 8888, it will listen on three ports (TCP-8888, UDP-8888 and UDP-5060). However, if UDP-5060 is overlapped with RTP ports for media session (see below menu-RTP settings); it will not listen on UDP-5060 for SIP signaling to avoid conflict.

Specify both the TCP and UDP port for SIP signaling. Default is 5060.

[Del]	Delete one character.
[Back]	Return without any changes.
[OK]	Edit selected item.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.

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5	0	6	0														
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Expires

Set the default expiration time of registration; measured in seconds. We suggest a value not less than 1200 seconds to avoid heavy network traffic due to registration. This expiration may modified (shorten) by SIP registrar server, and IP Phone will re-register to server when 50% of the modified expiration time has been passed. It, however, will re-register to SIP registrar after 90 seconds on registration failure.

Default is 3600 seconds.

		[Del]]	C	Delet	e on	e ch	arac	ter.												
	[E	Back]	F	Retur	m wi	thou	t ang	y cha	ange	s.										
	ľ	OK)		E	Edit s	selec	ted	item	•												
【←	-] a	nd	(→)	N	love	ecur	sor c	one p	oosit	ion ir	ו edi	t mo	de.								
S	Ι	Р		R	e g i s t e r E x p i r e s :																
[2	0	-	8	6	4	0	0]	(S	e	c	0	n	d)				
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Rport

Enable the support of RFC3581, an extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing.

When SIP operates on UDP, responses to requests are returned to the source address the request came from, and to the port written into the topmost Via header field value of the request. This behavior is not desirable in many cases, most notably, when the client is behind a Network Address Translator (NAT). This extension defines a new parameter for the Via header field, called "rport", that allows a client to request that the server send the response back to the source IP address and port from which the request originated.

A client sends an INVITE to a proxy server which looks like, in part:

INVITE sip:user@example.com SIP/2.0 Via: SIP/2.0/UDP 10.1.1.1:4540;rport;branch=z9hG4bKkjshdyff

If the proxy does not recognize this extension, you SHOULD disable this feature such that IP Phone will not insert this "rport" parameter into Via header while sending SIP requests.

Default is enabled to help NAT traversal if your SIP proxy/soft-switch supports it (for backward compatibility.you may disable it because some server think the rport parameter bears wrong format and silently drop this reques).

[On] / [Off]	Toggle between enable and disable this feature. Show check symbol ' $\!\!\!'$ in-line if enable.
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[Back]	Return without any	changes.
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[OK] Edit selected item.

 \uparrow and \downarrow Navigate through items.

1	•	Т	r	a	n	S	р	0	r	t	:	3								
2		S	Ι	Р		Р	0	r	t	:	5	0	6	0						
3		E	x	р	i	r	e	s	:	1	8	0	0							
4	\checkmark	r	р	0	r	t														
5		Η	0	1	d		b	у		R	F	C	3	2	6	1				
6		S	i	g	n	a	1		Ι	Р		Т	0	S		0	x	В	4	
7		S	e	S	S	i	0	n		Т	i	m	e	r		1	8	0	0	
8		Κ	e	e	р		Α	1	i	v	e		Т	i	m	e	r			
9		E	x	р	0	S	e		S	W		V	e	r						
1	0		S	Ι	Р		Т	i	m	e	r		Т	1						
	On/Off]	Back	5		

Hold by RFC3261

By RFC2543, it uses "c=0.0.0.0" in SDP to perform a hold operation; but this has been depreciated in RFC3261. Instead, it recommends the use of "a=sendonly" attribute in SDP to hold the peer.

Disable this feature to be backward compatible with RFC2543 implementation.

[On] / [Off]	Toggle between enable and disable this feature. Show check symbol ' \checkmark ' in-line if enable.
[Back]	Return without any changes.
[OK]	Edit selected item.
[\uparrow] and [\downarrow]	Navigate through items.

1		Т	r	a	n	S	р	0	r	t	:	3								
2		S	Ι	Р		Р	0	r	t	:	5	0	6	0						
3		Е	х	р	i	r	e	S	:	1	8	0	0							
4	\checkmark	r	р	0	r	t														
5		Η	0	1	d		b	у		R	F	С	3	2	6	1				
6		S	i	g	n	a	1		Ι	Р		Т	0	S		0	x	В	4	
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8		Κ	e	e	р		Α	1	i	v	e		Т	i	m	e	r			
9	\checkmark	Е	х	р	0	S	e		S	W		V	e	r						
1	0		S	Ι	Р		Т	i	m	e	r		Т	1						
	On/Off																Back	K		

Signal IP ToS

Specify the IP type-of-service (ToS) bits of SIP packets transmitting signals. This setting affects both TCP and UDP signaling packets.

The IP ToS bits consist eight bits:

7 (MSB)	6	5	4	3	2	1 0(LSB)
	Precedence		Delay	Throughput	Reliability	Reserved (00)

Where MSB denotes "Most Significant Bit" and LSB denotes "Least Significant Bit".

- **1.** Bits 5-7 specify the precedence:
 - 111 Network Control
 - 110 Internetwork Control
 - 101 CRITIC/ECP (RTP default)
 - 100 Flash Override
 - 011 Flash (RTCP default)
 - 010 Immediate
 - 001 Priority
 - 000 Routine (normal)
- 2. Bit 4: 0 = Normal Delay, 1 = Low Delay (RTP/RTCP default).
- 3. Bit 3: 0 = Normal Throughput, 1 = High Throughput (RTP default).
- 4. Bit 2: 0 = Normal Reliability, 1 = High Reliability (RTCP default).
- 5. Bit 0-1: Reserved for Future Use.

System default is 0xA8 (168 in decimal) for SIP signaling packets. Please enter it in decimal.

[Del]	Delete one character.
[Back]	Return without any changes.
[OK]	Save changes and return
$\left[\leftarrow\right]$ and $\left[\rightarrow\right]$	move cursor one position in edit mode

Ι	Р		Т	у	р	e	-	0	f	-	S	e	r	v	i	c	e			
[0	-	2	5	5]														
1	8	0																		
			Del															Back	K	

Session Timer

Set the session refresh time, measured in seconds, as per RFC4028. **Default is 1800 seconds**. The minimum interval is 90 seconds. On timer expiration, it will refresh the session either by re-INVITE or UPDATE the session.

[Del]	Delete one character.
[Back]	Return without any changes.
[OK]	Save changes and return
[\leftarrow] and [\rightarrow]	move cursor one position in edit mode.

R	F	С	4	0	2	8		S	e	S	S	i	0	n		Т	i	m	e	r	
[9	0	-	1	4	4	0]	(S	e	c	0	n	d)					
1	8	0	0																		
			Del															Back	ζ.		

Keep Alive Timer

The interval to send empty UDP packets to SIP registrar server to keep NAT port mapping open after finishing registration. Set to 0 or a value smaller than Register Expire time effectively disable this feature. This option sets only the frequency to send dummy UDP packets to registrar server to keep SIP signaling channel through NAT; to turn on this feature for each service realm, go to "7. SIP Settings" / "Service Domain" / "n-th Realm" / "Keep Alive".

[Del]	Delete one character.
[Back]	Return without any changes.
[0K]	Save changes and return
(\leftarrow) and (\rightarrow)	move cursor one position in edit mode.

S	Ι	Р		Ν	А	Т		Κ	e	e	р		А	1	i	V	e			
[0	-	8	6	4	0	0]	(S	e	c	0	n	d	S)			
0																				
			Del															Back	κ.	

Expose SW Ver

Specify whether it should include "**User-Agent**" or "**Server**" SIP header in each SIP message. These headers consist of product name and version. Default is checked (expose).

[On] / [Off]	Toggle between enable and disable this feature. Show check symbol ' \checkmark ' in-line if enabled
[Back]	Return without any changes.
[OK]	Save changes and return
[↑] and [↓]	Navigate through items.

7		S K	e e	s e	s p	i	o A	n 1	i	T v	i e	m	e T	r i	m	1 e	8 r	0	0	
8 9	√	K E	e x	e p	p o	S	A e	1	i S	v W	e	V	T e	i r	m	e	r			
1	0 . S I P On/Off				Р		Т	i	m	e	r		Т	1			-	Back	4	

SIP Timer T1

By RFC3261, T1 timer is an estimated measure of network packet round-trip time, and default to be 500-ms. However, if the network link is extremely slow, such as in a satellite link, and its round trip time is more than 500-ms, you may increase T1 timer to reduce SIP messages retransmission and increase the transaction timeout. However, change of T1 will affects 7 out of 11 SIP timers (see below), which are Timer-(A, B, E, F, G, H and J). The foreseeable result of lengthen T1, such as change it from 500-ms to 1-second, will double Timer-(B, F, H and J) as well, which in turn will make the transaction timeout more time-consuming. For example, by default, at most 32 seconds a user will wait for transaction timeout (such as dial a non-existed #), now it takes as long as 64 seconds for such timeout.

And below summarizes the meaning and defaults of the various timers used by RFC3261.

Timer	Value	Meaning
T1	500ms default	Round-Trip Time, RTT, Estimate
Т2	4s	The maximum retransmit interval for non-INVITE requests and INVITE responses
T4	5s	Maximum duration of a message will remain in the network
Timer A	Initially T1, and doubles for each retransmit	INVITE request retransmit interval, for UDP only.
Timer B	64*T1	INVITE transaction timeout timer
Timer C	> 3min	Proxy INVITE transaction timeout
Timer D	> 32s for UDP and 0s for TCP/SCTP	Wait time for response retransmits
Timer E	Initially T1, and Doubles for each retransmit	Non-INVITE request retransmit interval, UDP only.

	'till hitting T2, then T2 is used instead.	
Timer F	64*T1	Non-INVITE transaction timeout timer
Timer G	Initially T1, and Doubles for each retransmit 'till hitting T2, then T2 is used instead.	INVITE response retransmit interval
Timer H	64*T1	Wait time for ACK receipt
Timer I	T4 for UDP and 0s for TCP/SCTP	Wait time for ACK retransmits
Timer J	64*T1 for UDP and 0s for TCP/SCTP	Non-INVITE request retransmits
Timer K	T4 for UDP and 0s for TCP/SCTP	Wait time for response retransmits

[Del] Delete one character.

[Back] Return without any changes.

[OK] Save changes and return

 $\left(\leftarrow \right)$ and $\left(\rightarrow \right)$ Move cursor one position in edit mode.

S	Ι	Р		Т	i	m	e	r		Т	1							
[2	0	0	-	4	0	0	0]	(m	S)					
5	0	0																
			Del]	Back		

Service Domain

The IP Phone supports up to 3 different service domains simultaneously. The default service domain of each line-N will be active service-domain-N; otherwise, it will use the 1st active service domain if not specified.

[Back]	Return without any changes.
[OK]	Edit selected item.
$ \leftarrow $ and $ \rightarrow $	Navigete through item.

Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1	 1	S	t	D	0	m	а	i	n					
2	2	n	d	D	0	m	а	i	n					
3	3	r	d	D	0	m	а	i	n					
]	Back		

N-th Domain

[Back]	Return without any changes.
[OK]	Edit selected item.
[\leftarrow] and [\rightarrow]	Navigete through item.

Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1	 А	c	t	i	V	a	t	i	0	n										
2	Α	u	t	h	e	n	t	i	c	a	t	i	0	n						
3	Α	d	d	r	e	S	S	-	0	f	-	R	e	c	0	r	d			
4	Р	r	0	X	y		S	e	r	v	e	r								
5	R	e	g	i	S	t	r	a	r											
6	Α	u	t	0	-	Α	n	S	W	e	r									
7	Κ	e	e	р		Ν	Α	Т		Α	1	i	v	e						
															Back					

Activation

Configure to activate this service domain or not.

Default is disabled and it will not take effect (neither try to register nor show it as active domains) if any of the following conditions is met:

- Address-of-record is invalid or unavailable.
- Domain part in address-of-record is missing.
- No FQDN of registrar server is specified on manual mode.
- Registrar server is configured as "Same as proxy" while proxy server is on manual mode but

no FQDN of proxy server is specified.

[On] / [Off]	Toggle between enable and disable this feature. Show check symbol ' $\!\!\!'$ in-line if enabled!
[Back]	Return without any changes.
[OK]	Edit selected item.
[\leftarrow] and [\rightarrow]	Navigate through items.

1	 А	c	t	i	v	а	t	i	0	n										
2	Α	u	t	h	e	n	t	i	c	a	t	i	0	n						
3	Α	d	d	r	e	S	S	-	0	f	-	R	e	c	0	r	d			
4	Р	r	0	x	y		S	e	r	v	e	r								
5	R	e	g	i	S	t	r	a	r											
6	Α	u	t	0	-	Α	n	S	w	e	r									
7	K	e	e	р		N	A	Т		A	1	i	v	e						
	C)n/()	ff												Back					

Authentication

[Back]	Return without any changes.
[OK]	Edit selected item.

[\leftarrow] and [\rightarrow] Navigate through items.

1		S	e	v	e	r		L	0	g	i	n							
2		S	e	v	e	r		Р	a	S	S	W	0	r	d				
3	\checkmark	q	0	р		S	u	p	р	0	r	t							
																	Bacl	k	

Server Login

Name used by the phone for authentication if a registration is challenged y the proxy server or registrar server during initialization.

	[Del]		De	Delete one character.																
[4	Abc.	.]/[12	23]	To cu	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.															es	
	[C	lear]		CI	ear a	all in	out.														
	[В	ack]		Re	Return without any changes.																
	[(0K]		Ec	Edit selected item.																
【←	-] a	nd	(→)	m	move cursor one position in edit mode.																
ľ	↑】 a	nd	(↓]	m	move cursor per line in edit mode.																
T	0	σ	i	n		TI	c	A	r		T	D									
M	i	c c	h	a	e	1	3		1		1	D	•								

111	1	U	11	a	U	1											
	Del				Abc/123					(Clear	r					
Server Password

Password used by the phone for authentication if a registration is challenged by the proxy server or registrar server during initialization.

The password will be shown on the LCD without encryption to facilitate the alphabetic input. Besides, please remember to check for ending blanks and spaces; if you erroneously input the ending spaces, the whole string (including ending spaces) will be treated as your password without trim.

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[0K]	Save changes and return.
[\leftarrow] and [\rightarrow]	move cursor one position in edit mode.
[\uparrow] and [\downarrow]	move cursor per line in edit mode.

L	0	g	i	n		Р	a	S	s	W	0	r	d								
1	2	3	4																		
	Del				Abc/123					Clear					Back						

qop Support

Some SIP servers does not support qop parameter during HTTP digest authentication (keep responding with a "401 Unauthorized" on registration), then you should consider to ignore the qop parameter in the HTTP digest authentication calculation. Note, however, in most cases, qop parameter should be included.

Once disabled, the qop parameter will be ignored and thus excluded in HTTP digest authentication calculation.

Default is to include qop calculation.

[Back]	Return without any changes.
[0K]	Save changes and return.
[\uparrow] and [\downarrow]	Scroll menu items.

1	Ι	n	c	1	u	d	e		q	0	р							
2	Ι	g	n	0	r	e		q	0	р								
									Back									

Address-of-Record

Specify the number or SIP URI to register, such as "Michael Wu <sip:200@SIP.isp.com>".

Note the domain part will be recognized as the service domain associated with the registered SIP address of record.

[Save]	Save changes and return.
[Back]	Return without any changes.
[OK]	Edit selected field.
[\uparrow] and [\downarrow]	Navigate through menu items

D	i	S	р	1	а	у	:	Μ	i	c	h	а	e	1					
Р	r	0	t	0	с	0	1	:	S	i	р								
Α	d	d	r	:	1	2	3	4	5	6	7	(a)	Ι	S	Р	f	0	0	c
		Save)]	Back	Σ.		

Proxy Server

When default outbound proxy is used, all SIP requests are sent to this default outbound proxy server instead of locating the next hop SIP proxy server by RFC 3263. All responses continue to reconcile the normal Via processing rules. The media stream is not routed through the outbound proxy. Default is auto-locating.

The IP Phone implements SIP server redundancy when the DNS server returns more than one IP for the specified domain name record. It will try the additional server IP when the first server IP has no response.

[Back]	Return without any changes.
[0K]	Save changes and return.
[\uparrow] and [\downarrow]	Scroll menu items.

1		Μ	0	d	e	:	А	u	t	0	-	L	0	c	а	t	e				
2		D	e	f	a	u	1	t		Р	r	0	x	y		F	Q	D	N		
3		Т	r	a	n	S	р	0	r	t	:	1									
4		U	D	Р		Р	0	r	t	:	5	0	6	0							
5		Т	С	Р		Р	0	r	t	:	5	0	6	0							
																Back					

Mode

Note Note

Configure to use default proxy server or not. If Auto-locating is enabled, it will acquire the SIP proxy server responsible for the target URI by enquiring DNS NAPTR records of both "SIP+D2U" and "SIP+D2T", then SRV records of the target SIP domain. For example, if the target AoR is "someone <sip:100@dummy.com>", then it would query for "_sip._udp. dummy.com" or "_sip._tcp. dummy.com" (RFC 3263). By auto-locating SIP servers, you could achieve SIP proxy redundancy on SRV records level. That is if the query for DNS SRV records returns more than one SRV record, the IP Phone will use the SRV record of higher priority as primary SIP server, whereas the one of lower priority as a backup server whenever it fails to reach the primary one. For example, you may configure your DNS server to include the following SIP "Service Record" for UDP transport " (_sip._udp); please change those host and IP in red font accordingly:

\$ORIGIN YourISP.com
;; Pref Weight Port Target
_sip_udp.YourISP.com. IN SRV 0 0 1000 proxy.sip.YourISP.com.
_sip._udp.YourISP.com IN SRV 10 0 1000 backup.sip.YourISP.com.
proxy.sip.YourISP.com IN A 192.168.192.99.
backup.sip.YourISP.com IN A 192.168.192.11.

This terminal supports DHCP option code 120 to automatically discover SIP servers as per RFC 3361. If you enable DHCP, Dynamic Host Configuration Protocol, for network configuration, this terminal supports RFC3361 for DHCP option code 120 in addition to acquire host IP, gateway, network mask, and DNS server. Besides, if DHCP option code 120 is not included in the same DHCP offer for host IP/gateway/network mask, it will send a second DHCP INFORM request for such information to automatically discover SIP server. The purpose here is to ensure it works with existing DHCP servers which do not support RFC3361 for normal host IP/gateway/netmask settings, but acquire DHCP option 120 for SIP server from another co-exist DHCP servers.

Once operate in this mode, both it will use the discovered SIP servers included in DHCP option code 120 (5 SIP servers at most) as both SIP registrar and proxy server, and treat those discovered SIP servers listening on UDP port 5060. That is Registrar server will be same as proxy and the settings including Proxy FQDN, transport, and UDP port of SIP outbound proxy server will be ignored for this service domain.

[Back]	Return without any changes.
[0K]	Save changes and return
[\uparrow] and [\downarrow]	Scroll menu items.

1	А	u	t	0	-	L	0	c	а	t	e									
2	U	S	e		D	e	f	a	u	1	t		Р	r	0	х	у			
3	R	F	С	3	3	6	1													
																Back				

Default Proxy FQDN

Specify the IP address of the outbound proxy server. You can use either a dotted IP address or a DNS name. If you specify a DNS name and the DNS server returns more than one IP for the specified hostname, the IP Phone will try them in sequence if the first server IP contacted has no response (Proxy server redundancy).

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

D	e	f	а	u	1	t		Р	r	0	X	у		F	Q	D	Ν				
S	i	р		Ι	S	Р		f	0	0		n	e	t							
	Del				Abc/123					Clear					Back						

When default outbound proxy is enabled, all SIP requests are sent to this default outbound proxy server instead of locating the next hop SIP proxy server by RFC 3263. All responses continue to reconcile the normal Via processing rules. The media stream is not routed through the outbound proxy.

NAT and outbound proxy modes can be independently enabled or disabled. The received= tag is added to the Via header of all responses if there is no received= tag in the uppermost Via header and if the source IP address is different from the IP address in the uppermost Via header. Responses are sent back to the source under the following conditions:

- If a received= tag is in the uppermost Via header, the response is sent back to the IP address contained in the received= tag (and so is rport, as specified by RFC 3581 if you turn on the rport support from menu-rport).
- If there is no received= tag and the IP address in the uppermost Via header is different from the source IP address, the response is sent back to the source IP. Otherwise, the response is sent back to the IP address in the uppermost "Via" header.

Transport

The SIP transport protocol on which the default outbound proxy listens. Default is UDP. Currently there is no TLS support.

[Back]	Return without any changes.
[OK]	Save changes and return.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

1	U	D	Р											
2	Т	С	Р											
3	U	D	Р	а	n	d	Т	С	Р					
												Back		

UDP Port

Specify the UDP port number on which the outbound proxy server listens.

Default is 5060. Use 0 to disable UDP signaling.

[Del]	Delete one character
[Back]	Return without any changes.
[0K]	Save changes and return.
\leftarrow and \rightarrow	Move cursor one position in edit mode.

Р	r	0	х	у		U	D	Р	Р	0	r	t	:				
[0	-	6	5	5	3	5]									
5	0	6	0														
	Del													Back	K		

TCP Port

Specify the TCP port number on which the outbound proxy server listens. Default is 5060. Use 0 to disable TCP signaling.

[Del]	Delete one character
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.

Р	r	0	Х	у		Т	C	Р	Р	0	r	t	:						
[0	-	6	5	5	3	5]											
5	0	6	0																
	Del													Back					

Registrar

[Back] Return witho	out any changes.
---------------------	------------------

(OK) Save changes and return.

 \uparrow and \downarrow Scroll menu items.

1	М	0	d	e	:	S	а	m	e		а	s		Р	r	0	х	у		
2	R	e	g	i	S	t	r	а	r		F	Q	D	Ν						
3	Т	r	a	n	s	р	0	r	t	:	1									
4	U	D	Р		Р	0	r	t	:	5	0	6	0							
5	Т	С	Р		Р	0	r	t	:	5	0	6	0							
															Back					

Mode

Configure how to locate the registrar server to register SIP AoR. Default is Same as Proxy.

1. Same as Proxy: Choose [Same as proxy] if you want to register on default outbound proxy server. If you do not configure default outbound proxy, then it will behave like Auto-discover (see below).

VNote

If "RFC3361 DHCP Option for SIP Servers" is enabled for "SIP Proxy Server", Registrar Server will be adjusted to "Same as Proxy", ignoring what has been configured for Registrar Server..

- 2. Manually Set: Assign a registrar server below.
- 3. Auto-discover by querying DNS NAPTR records of both "SIP+D2U" and "SIP+D2T", SRV records of SIP domain. For example, if the registered AoR is "someone <sip:100@dummy.com>", then it would query for NAPTR record of "_sip._udp. dummy.com" or "_sip._tcp. dummy.com" (RFC 3263). By auto-discover registrar server, you could achieve Registrar server redundancy in SRV record level.
- 4. Multicast (224.0.1.75) Registration: Register by sending register information to multicast address 224.0.1.75

The IP Phone implements SIP registrar redundancy when the DNS server returns more than one IP for the specified domain name record. It will try the additional server IP when the first server IP has no response.

[Back]	Return without any changes.
[OK]	Save changes and return.
【↑】and【↓】	Scroll menu items.

1	S	а	m	e		a	S		Р	r	0	Х	у					
2	Μ	а	n	u	a	1	1	у		Α	S	S	i	g	n			
3	Α	u	t	0	-	D	i	S	c	0	v	e	r					
4	Μ	u	1	t	i	c	a	s	t									
										Back								

Registrar FQDN

Specify the IP address of the registrar server. You can use either a dotted IP address or a DNS name. If you specify a DNS name and the DNS server returns more than one IP, The IP Phone will try them in sequence if the first server IP contacted has no response (Registrar server redundancy).

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

R	e	g	i	S	t	r	a	r		S	e	r	v	e	r		F	Q	D	Ν	
S	i	р		Ι	S	Р		f	0	0	•	n	e	t							
	Del				Abc/123					Clear						Back					

Transport

The SIP transport protocol on which the registrar server listens. Default is UDP. Currently there is no TLS support.

	[E	Back]		Re	eturn v	with	out a	any c	hang	ges.						
	ľ	ок]		Sa	ive ch	nang	jes a	nd re	eturn	i.						
	【↑】;	and	(↓]	Sc	roll m	nenu	ı iten	ns.								
1		U	D	Р												
2		Т	C	Р												
3		U	D	Р		а	n	d		Т	C	Р				

2	T	C	P												
3	U	D	Р	a	n	d	Т	С	Р						
												J	Back		

UDP Port

Specify the UDP port number of the registrar server. The **default is 5060**. Use 0 to disable UDP signaling.

	[Del]				Delet	e on	e ch	arac	ter											
	[E	Back]	F	Retur	n wi	thou	t any	cha	inge	s.									
	ľ	ok]	l	S	Save	cha	nges	s and	l retu	ırn.										
[•	ה [⊣	and	(→		/love	cur	sor c	one p	ositi	on ir	ı edi	t mo	de.							
D			•							TT	D	D		D						
R	e	g	1	S	t	r	a	r		U	D	Р		Р	0	r	t			
	0	-	6	5	5	3	5													
5	0	6	0																	
			Del															Bac	k	

TCP Port

Specify the TCP port number of the registrar server. The default is 5060. Use 0 to disable TCP signaling.

[Del]	Delete one character
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.

R	e	g	i	S	t	r	a	r	Т	С	Р	Р	0	r	t	:		
[0	-	6	5	5	3	5]										
5	0	6	0															
			Del													Back	K	

Auto-Answer

Auto-answer incoming calls arrive on this account (<FROM> header in the initial INVITE message) in idle mode. By idle, it means the phone-set is either engaging in no calls or putting a call on hold.

If this feature is on, it will play distinguished auditable ring then turn on speaker-phone (or head-set if applicable) whenever an incoming call arrives on this registered SIP address-of-record account during idle mode.

		1. I a	f [DND] is on, Do-Not-Disturb feature wins; otherwise, for All-Calls-Forward and Auto-Answer features, All-Calls-Forward takes precedence.
L	Note	2. 1	The IP Phone supports server-side invoked auto-answer feature
٦.	INOLE	(c	Section-"Auto Answer" on "User's Manual") and auto-answer all incoming calls on idle mode system-wide (section-"Auto-Answer" on this document)
		3. <i>A</i>	All incoming calls will check against the server-side invoked auto-answer
		f	eature first, and then the global auto-answer feature, finally fall to the
			account-specific auto-answer feature.
	[On] / [(Off]	Toggle between enable and disable this feature. Show check symbol ' $\!\!\sqrt[4]{}$ in-line if enabled!
	[Back	(]	Return without any changes.
	[OK]]	Edit selected item.
	【←】 and	[→]	Navigate through items.

1	 Α	c	t	i	V	a	t	i	0	n									
2	Α	u	t	h	e	n	t	i	c	a	t	i	0	n					
3	Α	d	d	r	e	S	s	-	0	f	-	R	e	c	0	r	d		
4	Р	r	0	x	y		S	e	r	v	e	r							
5	R	e	g	i	s	t	r	a	r										
6	А	u	t	0	-	Α	n	S	W	e	r								
7	Κ	e	e	р		Ν	Α	Т		Α	1	i	v	e					
	C)n/()) ff														Back	x	

Keep NAT Alive

Check to turn on the feature to regularly send empty UDP packets to SIP registrar server to keep NAT port mapping between this terminal and the SIP registrar server. If the SIP registrar server does NOT treat constant dummy UDP packets as denial-of-service, DOS, attack, and your terminal is behind NAT, you may turn on this feature to keep SIP signal channel passing through NAT after finishing registration.

To set the frequency of sending dummy UDP packets, please go to "7. SIP Settings" / "SIP Service" / "Keep Alive Timer".

	[On] / [C	Off]	T ir	oggl n-line	e be if er	twee nable	en er ed!	nable	e and	d dis	able	e this	fea	ture.	Sh	ow o	checł	(syi	mbol	'√'
	[E	Back]	F	Retur	n wit	hout	any	char	nges											
	[0K]				Edit s	elect	ted it	em.													
Į	[OK] [↑] and [↓]			Ν	lavig	ate t	hrou	igh it	ems.												
1		Α	c	t	i	v	а	t	i	0	n										
2		Α	u	t	h	e	n	t	i	c	a	t	i	0	n						
3		Α	d	d	r	e	S	S	-	0	f	-	R	e	c	0	r	d			
4		Р	r	0	x	v		S	e	r	v	e	r								

5	R	e	g	i	S	t	r	a	r									
6	Α	u	t	0	-	Α	n	S	w	e	r							
7	Κ	e	e	р		Ν	А	Т		Α	1	i	v	e				
	0	n/O	ff]	Back	Σ.	

ENUM & E.164

Configure the appended DNS suffix for ENUM query and the minimum digit-length of a valid ENUM dial string.

[Back]	Return without any changes.
[0K]	Edit selected item.
[\uparrow] and [\downarrow]	Navigate through items.

1	Е	Ν	U	М		D	Ν	S		S	u	f	f	i	х				
2	Μ	i	n		Е	Ν	U	Μ		L	e	n	g	t	h				
3	U	R	Ι		F	0	r	m	а	t									
4	Ι	n	t	د	1		А	c	c	e	S	S		C	0	d	e		
																	Back	K	

ENUM DNS Suffix

Set the ENUM suffix for E.164 query on DNS. Leaving it blank or setting it as "e164.arpa" to comply with RFC2916 standard ENUM suffix, otherwise you may set it to a proprietary suffix, such as "e164.net".

Default is "e164.arpa"

E.g., if the dial string is "+886-3-1234567", the ENUM query string send to DNS server to resolve will be:

- Strip off all non-digits.
- Reverse the string.
- Insert a dot, '.', between each digit.
- Append the ENUM suffix, such as "e164.arpa".
- Send to DNS to resolve.

Thus, the resultant DNS query string would be:

"7.6.5.4.3.2.1.3.6.8.8.e164.arpa".

[Del]	Delete one character
[= 0.]	
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

Е	Ν	U	М		D	Ν	S		S	u	f	f	i	x	:				
e	1	6	4		а	r	р	a											
	1										1								
	Del						Ab	c/12	23			(Clea	r]	Back		

Min ENUM Length

Configure the minimum digit length of a valid ENUM dial string.

A valid ENUM dial string must starts with a '#', which will be translated to as a '+', and follows at least "Min length" digits. If the rules does not apply, the dial string will be dial "as is" without any attempt to carry out an ENUM resolution.

Default is 6 digits

	[Del]		D	elete	e one	e cha	racte	er.												
	[E	Back]	R	eturi	n wit	hout	any	chan	ges.											
	ľ	ok]		E	dit s	elect	ted ite	em.													
$ \left(\leftarrow \right) \text{ and } \left(\rightarrow \right) $ Move cursor one position in edit mode.																					
Μ	i	n		V	a	1	i	d		E	N	U	Μ		L	e	n	g	t	h	
[1	-	1	6]	(D	i	g	i	t)									
6																					

URI Format

Del

Configure what request URI should be sent whenever IP Phone failed to resolve ENUM to SIP URI.

Back

1. SIP URI: IP Phone will send the request to the specified SIP service domain proxy in the following form:

INVITE sip: +PhoneNumber@domain:port;user=phone

For example, if you dial "+88631234567" and IP Phone fails in ENUM resolution, it will send:

INVITE sip:+88631234567@proxy.SIP.isp.com;user=phone

2. TEL URI: IP Phone will send request to the specified SIP service domain proxy in the form: INVTTE tel: +PhoneNumber

For example, if you dial "+88631234567" and IP Phone fails in ENUM resolution, it will send:

INVITE tel:+88631234567

[Back]	Return without any changes.
[OK]	Save change and return.
【↑】and【↓】	Scroll menu items.

1	S	Ι	Р	U	R	Ι								
2	Т	Е	L	U	R	Ι								
]	Back		

Int'l Access Code

Del

While making calls to PSTN, you may dial "+886-3-1234567", and this setting configures whether the leading '+' should replace with an international access code, which is "00" in most countries, before actually sending it to proxy server.

The default is leave the leading '+' intact for proxy (or PSTN gateway) to replace it with the international access code based on their geographic locations.

Thus, the leading '+' will be replaced with the string specified here in either format, SIP URI or TEL URI, when IP Phone fails to resolve an ENUM number. The format could be specified in menu-"URI format".

	[]	Del]		D	elete	one	cha	racte	r.												
[4	Abc.	.]/[12	23]	To CL	oggle irren	e bet t inpi	wee ut me	n dig ethoo	jits a d is a	and alpha	alph inum	anun Ieric	neric and	; inp [123	out, v 3] in	vher Idica	re. [/ ntes (Abc digits] in s inp	dicat out.	es
	[C	lear]	С	lear a	all in	out.														
	[B	ack]	R	eturn	with	out	any c	han	ges.											
[OK] Save changes and return.																					
【 ←	-] a	and	【→】	M	ove	curso	or on	ie po	sitior	n in e	edit r	node) .								
ľ	↑] a	and	(↓]	М	ove	curso	or pe	er line	e in e	edit n	node	-									_
R	e	р	1	a	c	e		t	e	1	:	٢	+	٤		a	S	:			
+																					-

Clear

Back

Abc../123..

Service

Configure the external services available to the phone.

[Back]	Return without any changes.
[0K]	Edit selected item.
【↑】and【↓】	Navigate through items.

1	М	W	Ι															
2	Μ	e	S	S	a	g	e											
3	S	Ν	Т	Р														
4	Α	u	t	0	-	Р	r	0	v	i	s	i	0	n				
																Back	K	

MWI

Voice mail allows you to access messages left by callers when you are unavailable to take their call. Voice mail is an optional feature configured by your system administrator. Your particular phone setup might not support accessing voice mail in this way.

To configure the voice mailbox number to dial when the message waiting indication button, [MWI], is pressed, click the "Contacts" on the right panel to pick an entry from your address book; and delete the number in input text area to remove it.

The corresponding MWI LED will be flashing whenever it receives NOTIFY messages stipulating some new unread (new) voice messages in your voice mailbox. To access voice mail, press the [MWI] button and follow the voice instructions.

The IP Phone supports messages waiting indication as per RFC3842. It accepts NOTIFY messages with or without prior SUBSCRIBE

Note

By http://pentest.tele-consulting.com/advisories/05 07 06 voip-phones.txt, there

is a weakness of unsolicited NOTIFY for MWI. You may turn off unsolicited MWI NOTIFY, thus effectively disable "Out-of-Dialog MWI Notification".

• MWI Subscription

If you have configured the voice mailbox link, the IP Phone will SUBSCRIBE to this link by sending the SUBSCRIBE message to SIP server. Please refer to the following message flow (suppose your SIP AoR is sip:7700@SIP.isp.com and the voice mailbox you configured is sip:vms7700@SIP.isp.com):

SUBSCRIBE sip:vms7700@SIP.isp.com SIP/2.0 Via: SIP/2.0/UDP 192.168.3.50 From: John <sip:7700@SIP.isp.com>;tag= 17542c1 To: John <sip:7700@SIP.isp.com> Call-ID: 0c1c7a67461@ipr.SIP.isp.com CSeq: 281 SUBSCRIBE Contact: sip:7700@192.168.3.50 Event: message-summary Accept: application/simple-message-summary Expires: 3600 User-Agent: SIP-Phone /1.1 Content-Length: 0

SIP/2.0 **202 Accepted** Via: SIP/2.0/UDP 192.168.3.50 Record-Route: <sip:192.168.3.1;Ir=1> Contact: sip:192.168.3.1:6060 Expires: 3600 User-Agent: ABS GW v5.1.0 To: John <sip:7700@SIP.isp.com>;tag=980765 From: John <sip:7700@SIP.isp.com>;tag=17542c1 Call-ID: 0c1c7a67461@ipr.SIP.isp.com CSeq: 281 SUBSCRIBE Content-Length: 0

• Out-of-Dialog MWI Notification

The IP Phone accepts unsolicited MWI NOTIFIY messages as well. The sample NOTIFY message is as follows (suppose "sip.isp.com" is one of your active service domains and your SIP AoR is sip:1234@sip.ISP.com):

NOTIFY sip:1234@192.168.3.50 SIP/2.0 Via: SIP/2.0/udp 192.168.0.1;branch=z9hG4bKcfb4 Content-Type: application/simple-message-summary Contact: sip:192.168.0.1:6060 User-Agent: ISP Soft-Switch Event: message-summary Subscription-State: active To: <sip:1234@sip.isp.com> From: <sip:1234@192.168.0.1>;tag=d8370cb Call-ID: d07b59da8e CSeq: 224493566 NOTIFY Content-Length: 39 Max-Forwards: 70

Messages-Waiting: yes Message-Account: sip: albert@sip.isp.com Voice-Message: 4/8 (1/2)

Note, when the MWI LED is off while pressing [MWI], it will dial to the voice mailbox you configured. However, when the MWI LED is flashing and you press [MWI], it will make a call to the **Message-Account** stipulated on the latest NOTIFY message it received (if this field is absent or is not a SIP AoR, the AoR in request is used instead). If there are unsolicited out-of-dialog NOTIFY messages received from different service domains, those voice mailboxes will be called in turn (in circular fashion) each time [MWI] is pressed.

[Back]	Return without any changes.
[OK]	Edit selected item.
[\uparrow] and [\downarrow]	Navigate through items.

Show check symbol ' $\sqrt{}$ ' in-line if enabled.

1	V	0	i	c	e		М	а	i	1	b	0	Х		А	0	R			
2	 U	n	S	0	1	i	c	i	t	e	d		Ν	0	Т	Ι	F	Y		
															1	-			_	
]	Back	1		

Voice Mailbox AoR

Dial to voice mailbox address-of-record if available.

[Call] Alternatively, You may lift the handset (offhook) or turn speaker on by pressing [SPK] to make a call to selected contact as well.

[Reset] Reset voice mailbox address-of-record from address book.

V	0	i	c	e		М	а	i	1	b	0	x		А	0	R			
S	i	р	:	v	m	S	a	Ι	S	Р		f	0	0		c	0	m	
	Call						J	Reset					Del]	Back	

[Back]	Return without any changes.
[0K]	Pick selected contact.
[\uparrow] and [\downarrow]	Navigate through the list.

С	0	n	t	a	c	t	S	(1	4	3)					
Μ	i	c	h	а	e	1											
Μ	i	k	e														
Ν	i	c	k														
Р	а	t	r	i	c	k											
Р	а	u	1														
R	i	c	h	a	r	d											
															Bacl	K	

Unsolicited NOTIFY

By <u>http://pentest.tele-consulting.com/advisories/05 07 06 voip-phones.txt</u>, there is a weakness of unsolicited NOTIFY for MWI. You may turn off unsolicited MWI NOTIFY, thus effectively disable "Out-of-Dialog MWI Notification".

[Back]	Return without any changes.
[OK]	Save changes and return.
[↑] and [↓]	Navigate through the list.

1	А	c	c	e	р	t								
2	R	e	j	e	c	t								
]	Back		

Message

VNote

Instant messaging service.

[Back]	Return without any changes.
[OK]	Enter submenu.
[\uparrow] and [\downarrow]	Scroll menu items.

1	W	r	i	t	e		М	e	S	S	а	g	e					
2	Ι	n	b	0	х													
3	0	u	t	b	0	х												
4	D	r	a	f	t													
]	Back		

The IP Phone supports "flashing short message" to display received out-of-dialog message on LCD without user interaction (or any alert) for 2~3 seconds. Such flashing messages are for server-side notification, and they will not be saved (thus "flashing"). To activate such feature, the received out-of-dialog instant message must carry a proprietary header "**P-Flash-SMS: on**" (case-sensitive) in received MESSAGE messages..

Write Message

Compose a new message.

Compose

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Finish composing messages.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

Η	e	1	1	0		W	0	r	1	d	~								
	Del						Ab	c/12	23			(Clea	r]	Back	2	

Finish Writing

[Send]	Deliver
[Save]	Save to draft
[Abort]	Discard message and return.
[Write]	Go back to previous step to modify message.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.
[OK]	Finish composing messages.

Η	e	1	L	0		W	0	r	1	d								
	Send							Save			A	bor	t		I	Nrit	e	

Send

	[Del]		De	elete	one	cha	racte	r.												
	[R	ealm]	Ci	rcle	serv	ice d	lomai	in (alv	ways	s en	able	[Abo	c] iı	nput	moc	le)				
	[AB]		Pio	ck a	con	tact f	rom /	Addre	ess E	300	K. [Sei	rvice	e Doi	main					
S	e	n	d		0	:			Ι	S	Р	•	f	0	2	•	n	e	t		
	Del]	Realr	n				AB					Back	K		

[Back] Return without any changes.

[OK] Pick selected contact.

 $\label{eq:constraint}$ and $\label{eq:cursor}$ move cursor one position in edit mode.

 \uparrow and \downarrow move cursor per line in edit mode.

С	0	n	t	а	c	t	S	(1	4	3)						
М	i	с	h	а	e	1		Ì										
Μ	i	k	e															
Ν	i	c	k															
Р	а	t	r	i	c	k												
Р	а	u	1															
R	i	с	h	а	r	d												
]	Back		

Inbox

List messages in the received order with the latest listed first. Besides, unread messages will have indicators \square preceeding them.

[Del]	Delete selected messages.
[Clear]	Clear all messages.
[Back]	Return to previous page.

Store 100 most recently received messages.

	1	- 0	Н	e	1	1	0		W	0	r	1	d	!							
Unread	1		Α	d	v	e	r	t	i	s	e										
Message	3		D	i	n	n	e	r		t	0	g	e	t	h	e	r				
Indicato																					
				Del					Clear	•]	Back	[

[Yes]	Clear all messages from inbox.
[No]	Return to list mode
[Back]	Return to previous page.
[OK]	Read selected message.
[\uparrow] and [\downarrow]	Navigate through messages.

D	e	1	e	t	e	а	1	1	m	e	S	S	а	g	e	S	?	
			Yes													No		

Read

Show receive time on the first line. Besides, it also show display or user-part of the email-like address on "From" item in-line.

[Call]	Call to messages sender.
[Edit]	Edit sender's number to save into address book.
[Del]	Delete this messages and return.
[Back]	Return without any changes.
[0K]	Enter submenu.
[\uparrow] and [\downarrow]	Navigate through items.

2	1	:	3	2	:	1	5		1	0	/	2	4	/	2	0	0	6			
1		F	r	0	m	:	Μ	i	с	h	а	e	1								
2		C	0	n	t	e	n	t													
3		R	e	р	1	у															
4		F	0	r	w	a	r	d													
	Call					Edit					Del					Back					

From/Edit

[Call]	Call to this contact.
[To AB]	Add modified contact into address book and return.
[Save]	Save changes and return.
[Back]	Return without any changes.
[OK]	Enter submenu.
[↑] and [↓]	Navigate through items.

D	i	S	р	1	а	у	:	М	i	с	h	а	e	1					
Р	r	0	t	0	c	0	1	:	S	i	р								
А	d	d	r	:	1	2	3	4	5	6	7	a	Ι	S	Р	f	0	0	c
			Call					Save				Т	'o Al	B]	Bacl	K	

Content

[Call]	Call to messages sender.
[Del]	Delete this message and return.
[Back]	Return without any changes.
[OK]	Enter submenu.
[\uparrow] and [\downarrow]	Navigate through items.

2	1	:	3	2	:	1	5		1	0	/	2	4	/	2	0	0	6	
1		F	r	0	m	:	Μ	i	c	h	a	e	1						
2		С	0	n	t	e	n	t											
3		R	e	р	1	у													
4		F	0	r	W	а	r	d											
	Call									Del					Back				

[Del]	Delete this message and return.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Enter submenu.
\leftarrow and \rightarrow	move cursor one position in edit mode.
[\uparrow] and [\downarrow]	move cursor per line in edit mode.

Н	e	1	1	0	W	0	r	1	d	!										
			Del		Abc/123				Clear					Back						

Reply

[Call]	Call to message sender.
[Del]	Delete this message and return.
[Back]	Return to previous page.
[\uparrow] and [\downarrow]	Navigate through items.
[OK]	Compose a new message and reply to sender.

2	1	:	3	2	:	1	5		1	0	/	2	4	/	2	0	0	6
1		F	r	0	m	:	М	i	c	h	a	e	1					
2		С	0	n	t	e	n	t										
3		R	e	р	1	У												
4		F	0	r	W	а	r	d										
	Call									Del					Back			

The process is similiart to write a brand new message, except for

- 1. It will return to "Reply" item on finishing.
- 2. The default target number on sending is the original message sender.

S	e	n	d		Т	0	:				Ι	S	Р		f	0	0		n	e	t
Μ	i	c	h	а	e	1	<	S	i	р	:	1	2	3	4	5	6	7	a	Ι	S
Р		f	0	0		с	0	m	>												
	Del						F	Realn	1				AB					Back	K		

Forward

[Call]	Call to message sender.
[Del]	Delete this message and return.
[Back]	Return to previous page.
[\uparrow] and [\downarrow]	Navigate through items.
[OK]	Compose a new message and reply to sender.

2	1	:	3	2	:	1	5		1	0	/	2	4	/	2	0	0	6	
1		F	r	0	m	:	М	i	c	h	а	e	1						
2		С	0	n	t	e	n	t											
3		R	e	р	1	у													
4		F	0	r	W	а	r	d											
	Call											Del]	Back		

The process is similiart to write a new message, except for

- 1. It will return to "Forward" item on finishing.
- 2. The default message on composing is the original message.

Н	e	1	L	0		W	0	r	1	d								
	-																	
	Del				Ab	c/12	23		C	lea	r]	Back	2			

Outbox

List all sent messages by time with the latest listed first. Keeps *100* most recently sent messages.

[Clear]	Clear all messages.
[Del]	Delete this message and return.

1		Η	e	1	1	0		W	0	r	1	d	!					
2		А	d	v	e	r	t	i	S	e								
	Del							Clear	•]	Back		

[No] Return to list mode

[Back] Return to previous page.

- **[OK]** Read selected message.
- \uparrow and \downarrow Navigate through messages.

D	e	1	e	t	e		a	1	1	m	e	S	S	a	g	e	S	?	
											\square								
			Yes	,		—					_	_			Г		No		

Read

Show sent time on the first line. Besides, it also show display or user-part of the email-like address on "To" item in-line.

	[0	Call]		Ca	all to	targ	et nı	umbe	r.										
	[E	[dit]		E	dit ta	rget	num	ber to	o sav	/e in	to ac	dres	ss bo	ook.					
	[Del]		De	elete	this	mes	sage	and	l retu	ırn.								
	[B	ack]	I	R	eturn	to p	orevio	ous p	age.										
	[ok]		Er	nter s	subn	nenu	•											
ľ	a	nd	【↓】	Na	aviga	ate th	nrouę	gh me	essa	ges.									
2	1	:	3	2	:	1	5		1	0	/	2	4	/	2	0	0	6	
1		Т	0	:	М	i	с	h	а	e	1								
2		С	0	n	t	e	n	t											
3		R	e	s	e	n	d												
4	•	F	0	r	W	a	r	d											
			Call					Edit					Del			1		Back	

To/Edit

[Call]	Call to this contact.
[To AB]	Add modified contact into address book and return.
[Save]	Save changes and return.
[Back]	Return to previous page.
[OK]	Edit selected field.
[\uparrow] and [\downarrow]	Navigate through messages.
D i s p	lay: Michael

	Call							Save				Т	o AI	3]	Back	i.	
Α	d	d	r	:	1	2	3	4	5	6	7	a	Ι	S	Р	f	0	0	c
Р	r	0	t	0	c	0	1	:	S	i	р								
D	1	S	р	1	а	У	:	Μ	1	с	h	а	e	l					

Content

[Call]	Call to this contact.
[Del]	Delete this message and return.
[Back]	Return to previous page.
[\uparrow] and [\downarrow]	Navigate through messages.
[OK]	Edit content.

2	1	:	3	2	:	1	5		1	0	/	2	4	/	2	0	0	6		
1		Т	0	:	Μ	i	c	h	a	e	1									
2		С	0	n	t	e	n	t												
3		R	e	S	e	n	d													
4		F	0	r	W	а	r	d												
	Call												Del]	Back	K	

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Back]	Return without any changes.
[OK]	Save changes and return.
(\leftarrow) and (\rightarrow)	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

Η	e	1	1	0		W	0	r	1	d	!								
	Del						Ab	c/12	23			(Clea	r		I	Back		

Resend

[Call]	Call to this contact.
[Del]	Delete this message and return.
[Back]	Return to previous page.
[\uparrow] and [\downarrow]	Navigate through messages.
[OK]	Edit original messages and resend to target number.

2	1	:	3	2	:	1	5		1	0	/	2	4	/	2	0	0	6	
1		Т	0	:	Μ	i	c	h	a	e	1								
2		C	0	n	t	e	n	t											
3		R	e	s	e	n	d												
4		F	0	r	W	а	r	d											
	Call												Del]	Back	

The process is similiart to write a brand new message, except for

- 1. It will return to "Resend" item on finishing.
- 2. The default message on composing is the original message.

Η	Е	1	L	0		W	0	r	1	d								
	Del						Ab	c/12	23		(Clea	r]	Back	2	

3. The default target number on sending is the original message sender.

S	Е	n	d		Т	0	:				Ι	S	Р		f	0	0		n	e	t
М	Ι	c	h	a	e	1	<	S	i	р	:	1	2	3	4	5	6	7	a	Ι	S
Р		f	0	0		c	0	m	>												
			Del				F	Realn	n				AB					Bacl	K		

Forward

[Call]	Call to this contact.
[Del]	Delete this message and return.
[Back]	Return to previous page.
[\uparrow] and [\downarrow]	Navigate through messages.
[OK]	Edit original messages and resend to target number.

2	1	:	3	2	:	1	5		1	0	/	2	4	/	2	0	0	6	
1		Т	0	:	Μ	i	c	h	a	e	1								
2		C	0	n	t	e	n	t											
3		R	e	s	e	n	d												
4		F	0	r	W	а	r	d											
	Call												Del]	Back	

The process is similiart to write a new message, except for

- 1. It will return to "Forward" item on finishing.
- 2. The default message on composing is the original message.

Η	e	1	L	0	W	0	r	1	d								
	Del				Ab	c/12	23		(lea	r]	Back			

Draft

List all draft messages by time order with the latest listed first. Besides, show the first line on listing. Store **50** most recently unsent messages.

[Del]	Delete selected messages.
[Clear]	Clear all messages.

1	Η	e	1	1	0		W	0	r	1	d	!					
2	А	d	v	e	r	t	i	S	e								
		Del					Clear	•]	Back	K	

[Yes]	Clear all messages from outbox.
[No]	Return to list mode
[Back]	Return to previous page.
[OK]	Read selected message.
[\uparrow] and [\downarrow]	Navigate through messages.

D	e	1	e	t	e	a	1	1	m	e	S	S	a	g	e	S	?	
			Yes													No		

[Send]	deliver => Refer to "Send" on "Write Message".
[cond]	 Draft message will be removed once sent.
[Edit]	Edit messages => Refer to "Compose" on "Write Message".
[=]	 Modified message will overwrite the original draft message.
[Del]	Delete this draft message and return.
[Back]	Return to previous page
[↑] and [↓]	Scroll lines if there are too many lines to fit into a screen.

Н	e	1	1	0		W	0	r	1	d											
	Send			Edit						Del					Back						

SNTP

Simple Network Time Protocol, SNTP, is used to acquire current date-time so as to synchronize the date-time on the phone. Once assigned, the phone will try to synchronize the date-time with the specified NTP server regularly and convert the returned GMT time to local time based on time zone and daylight saving adjustment.

[Back]	Return without any changes.
[0K]	Enter selected item.
【↑】and【↓】	Scroll menu items.

1	 Μ	0	d	e										
2	S	e	r	v	e	r								
]	Back		

Mode

The Simple Network Time Protocol is used to synchronize time with IP Phone. If you set SNTP server to Anycast mode, the phone will send SNTP query to LAN broadcast address. Otherwise, it sends a request to the specified SNTP / NTP server, extracting the reported time from the reply, and overwrites the phone's time. Typically, SNTP / NTP servers operating in broadcast mode send update messages every 64 to 1024 seconds. The default time on system starting up is 00:00, January 1, 2007, GMT. If you choose to disable SNTP, it will not synchronize current system time with NTP server, nor will it adjust daylight saving time.

Default is Unicast

	Unicast	Multicast	Anycast
Sends	SNTP request to the specified SNTP server if available.	Nothing When in multicast mode, SNTP requests are not sent.	SNTP packet to the local network broadcast address, 224.0.1.1. After the first SNTP response is received, the phone switches to unicast mode with the server being set as the one who first responded.
Receives	SNTP response from the SNTP server and ignores responses from other SNTP servers.	SNTP data via the SNTP / NTP multicast address from the local network broadcast address from any server on the network. The default multicast address is 224.0.1.1 .	Unicast SNTP data from the SNTP server that first responded to the network broadcast request.

By "**SIP Register**", this terminal will synchronize its local time based on the "Date" header present in successful 2xx SIP responses to REGISTER method. Per rfc3261, it restricts the time zone in SIP-date to "GMT", in a format similar to (case-sensitive).

Date: Sat, 13 Nov 2010 23:29:00 GMT

As a result, the parsed time will be converted to local time based on time-zone and daylight saving time (DST) adjustment as well (refer to "3.Phone Settings" / "Date/Time").

[Back]	Return without any changes.
[OK]	Save changes and return
[\uparrow] and [\downarrow]	Scroll menu items.

1	U	n	i	c	а	S	t										
2	Μ	u	1	t	i	c	a	S	t								
3	Α	n	у	c	a	S	t										
4	D	i	s	a	b	1	e										
5	S	Ι	Р		R	e	g	Ι	S	t	e	r					
]	Back		

Server

Ļ

Assign the SNTP server. You can use either a dotted IP address or a DNS name.

For available SNTP / NTP servers near your location, please refer to Appendix C – "Available NTP Servers".

No	ote	If you enable DHCP, the NTP server may be acquired by DHCP option 42.
	[Del	Delete one character.

[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

S	Ν	Т	Р		S	e	r	v	Е	r	:				
c	1	0	c	k		р	S	u		e	d	u			
	Del					Abc/123						Cle	ar	Back	

Protocol

Select which protocol to download files for auto-provision. If selected, the phone will try to download two configuration files from the provision server via either TFTP or HTTP protocol on system startup; otherwise, it will use the current settings stored in flash ROM.

Default is disabled

	[B	ack]		Re	eturn	with	out a	iny c	hanges.						-
	[эκ]		Sa	ive c	hang	jes a	nd re	eturm.						
[[↑] a	nd	(↓]	Sc	roll r	nenı	ı iten	າຣ.							_
1 2 3		D T H	i F T	S T T	a P P	b 	L	e	d						
												1	Back		

Server

Specify the IP address of provision server. You can use either a dotted IP address or a DNS name. You may include an alternative server port, such as "10.0.0.1:8000". Default TFTP server port is UDP 69 and default HTTP server port is TCP 80. If you did not configure the SIP parameters from a provision server, you can configure them directly by keypad, a web browser or a telnet client.

You may enter multiple servers separated by any of the following characters, ";,[Space][Tab]", such as "backup.isp.com, web.ISP.com" (without the quotes). And the terminal will try them in sequence 'till download successfully.

VNote

If you enable DHCP, the TFTP server may be acquired by DHCP option 66 (TFTP server IP address).

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

S	e	v	e	r														
1	9	2		1	6	8		3		1								
	Del						Ab	c/12	23			Clea	r]	Back		

Batch Default

Enter the default configuration file to download from provision server. The file generally is used for LAN administrator batch configures all the phones, such as proxy server and network environment. We suggest that you maintain the SIP parameters that are common to all your phones. By maintaining these parameters in the default configuration file, you can perform global changes without having to modify the terminal-specific configuration file for each terminal.

If the file could be fetched from provision server, the setting read from the configuration will overwrite the settings read from flash ROM.

System default is "sipDefault.cnf"

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

В	a	t	c	h		D	e	f	a	u	1	t	:						
S	i	р	D	e	f	a	u	1	t		c	n	f						
	Del						Ab	c/12	23			(Clea	r]	Back		

Terminal-Specific

Enter the terminal-specific configuration file to download from provision server. If the file could be fetched from provision server, the setting read from the configuration will overwrite the settings from [Batch Default] and those read from flash ROM.

Default is sip **\$MAC\$**.cnf, where **\$MAC\$** is the MAC address in hexadecimal representation.

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

Т	e	r	m	i	n	a	1	-	S	р	e	c	i	f	i	c	:			
S	i	р	\$	Μ	Α	С	\$		c	n	f									
	Del						Ab	c/12	23			(Clea	r]	Back		

Security

Set personal secret AES key to decipher the downloaded configuration files by AES of 192-bit key length. If you enabled the security option but left the decryption key blank, the terminal will use built-in AES default key to decipher them.

[On] / [Off]	Toggle between enable and disable this feature. Show check symbol ' \checkmark ' in-line if enabled!
[Back]	Return without any changes.
[OK]	Edit selected item.
【↑】and【↓】	Navigate through items.

1	Р	r	0	t	0	c	0	1											
2	S	e	r	v	e	r													
3	В	а	t	c	h		D	e	f	a	u	1	t						
4	Т	e	r	m	i	n	a	1	-	S	р	e	c	i	f	i	c		
5	S	e	c	u	r	i	t	у											
	C)n/()	ff														Back		

Set personal secret AES key to decipher the downloaded configuration files by AES of 192-bit key length. If you enabled the security option but left the decryption key blank, the terminal will use built-in AES default key to decipher them.

Initially show 10 '*' signs but show clear text on user input because alphanumeric characters may be needed.

		[Del]		D	elete	e one	e cha	racte	er.												
	[Abc]/[1	23]	T C	oggl urrer	e be nt inp	etwee out m	n di etho	gits a d is a	and Ilpha	alph anum	anui neric	neri and	c in [12	out, 3] i	wher ndica	re. [A ates c	vbc] digits	ind inp	icate ut.	es
	[(Clea	r]	С	lear	all ir	nput.														
	[Back	[]	R	etur	n wit	hout	any	chan	ges.											
	ľ	Ο Κ	1	S	ave	char	nges	and	returr	ı											
	【 →】	and	[→		love	curs	sor or	ne po	ositior	n in e	edit ı	nod	e.								
	[↑]	and	【↓】	Ν	love	curs	sor pe	er line	e in e	dit n	node) .									
Л	0	0	r	*7	n	+	:	0		V	0										
*	*	*	*	У	р	ι	1	0		ĸ	C	У	•								
														-							-
			Del				Ab	c/1	23				Clea	r]	Back			

Advanced

	[B	ack]		Re	turn	with	out ai	ny ch	ang	es.									
	[(эκ]		Ed	it sel	lecte	d iter	n.											
ľ	↑] a	nd	(↓]	Sc	roll n	nenu	i item	s.											
1	-	С	0	D	E	С													
23		S S	y y	S S	t t	e e	m m		A S	d t	m a	i t	n u	S					 -
4	•	L	0	g															
																-	Back	<u> </u>	Г

CODEC

The CODECs available on the terminal includes G.711A (64kbps), G.711 μ (64kbps), G.729A (8kbps), G.729A (8kbps), G.729AB (8kbps with CNG), G.723.1 (both 5.3 and 6.4 kbps), G.723.1A (both 5.3 and 6.4 kbps with CNG), and G.726 (both 32 and 40 kbps, and G721 is identical to G.726-32kbps). The default preference is to prioritize them based on their compressed voice quality: the higher quality it is, the higher priority it will be. That is the default CODEC preference is G.711 μ , G.711A, G.726, G.729A then G.723.1A.

High quality comes in a price: it consumes more precious bandwidth. Take $G.711\mu$ for example, it takes as much as 64 kbps per call, as opposed to 5.3 kbps if G.723.1A is employed. Thus, if you have limited bandwidth for VoIP, you should consider prioritize those CODECs come with lower bit-rates such that it will use your most preferred CODEC on making / taking calls.

[Back]	Return without any changes.
[OK]	Edit selected item.

 \uparrow and \downarrow Scroll menu items.

Show check symbol ' $\sqrt{}$ ' in-line if enabled!

1		Р	r	e	f	e	r	e	n	c	e	S								
2	•	Р	a	c	k	e	t	i	z	a	t	i	0	n						
3	\checkmark	С	0	m	f	0	r	t		Ν	0	i	S	e						
4		R	F	С	2	8	3	3		Р	Т	:	1	0	1					
5		D	Т	Μ	F		R	e	1	a	у		b	y		Ι	N	F	0	
																	Back			
Preferences

Prioritize between the available CODECs. CODECs will be listed based on their priorities. The smaller the value is, the higher the priority would be. Those disabled voice CODECs, which preference is zero, will be listed last. Note, however, if you disable all CODECs, it will auto-arrange the available CODECs and those come with lower bit-rates take precedence. Besides, you should just lower the priorities of G.711 in most circumstances rather than overly disabling them such that the phone could gracefully fall back to G.711A or G.711µ when the peer does not support those low bit-rate CODECs for voice compression.

Default is "Increasing voice quality": Those come with higher compressed voice quality take precedence. Such that it will prioritize those CODECs with the highest compressed voice quality first while making calls. However, it will respect the peer's preferences while taking calls.

[Up]	Increase the priority of selected CODEC
[Down]	Decrease the priority of selected CODEC
[]	Disable selected CODEC
[Back]	Return without any changes.
[OK]	Save changes and return.
[\uparrow] and [\downarrow]	Navigate through menu items.

CODEC Priority List Format: list by priority with most preferred first.

1	G		7	1	1	А	-	6	4	k	:	1						
2	G		7	1	1	μ	-	6	4	k	:	2						
3	G		7	2	3		1	-	6		4	k	:	3				
4	G		7	2	6	-	4	0	k	:	4							
5	G		7	2	6	-	3	2	Κ	:	5							
6	G		7	2	9	-	8	k	:	6								
7	G		7	2	3		1	-	5		3	k	:					
		Up]	Dowi	1]	Back		

Packetization

Configure the default voice packetization transmitted to the peer. The terminal will adjust the audio frames carried per RTP packet (voice packetization) based on this setting:

Voice Packetiz	ation	G.723.1	G.729	G.711
30(G.723.1)/10(Others)-ms	LAN		1 Frames	s (10 ms)
30(G.723.1)/20(Others)-ms	xDSL, ISDN	1 Erama (20 ma)	2 Frames	s (20 ms)
30(G723.1)/30(Others)-ms	xDSL, ISDN, Cable	T Flattle (50 fills)	3 Frames	s (30 ms)
30(G.723.1)/40(Others)-ms	56 kbps MODEM		4 Eromo	(40 mo)
60(G.723.1)/40(Others)-ms	Limited bandwidth	2 Frames (60 ms)	4 Flames	s (40 ms)

The default is [20/30(G.723.1)ms]: 1 audio frame (30 ms voice) per RTP packet for G.723.1 CODEC, and 2 frames (20 ms voice) per packet for all other CODECs such as G.729A and G.711.

Before changing this setting, please refer to the following table "Packet Rate and VoIP Bandwidth Consumption" to find out the optimal value fit into your environment. We suggest a reasonable packetization should NOT longer than 40 milliseconds. Pick the longer packetization time unless you have a very limited bandwidth access at the expense of more latency.

[Back]	Ret	urn withou	t any	changes

[OK] Save changes and return.

 \uparrow and \downarrow Navigate through menu items.

1	· _	3	0	(G	7	2	3	1)	/	1	0	m	S			
2		3	0	(G	7	2	3	1)	/	2	0	m	S			
3		3	0		m	S												
4		3	0	(G	7	2	3	1)	/	4	0	m	s			
5		6	0	(G	7	2	3	1)	/	4	0	m	S			
]	Back		

VNote

The IP Phone will dynamically adjust its voice packetization on a per call basis if the "ptime" attribute is specified in the peer's SDP. For example: if "a=ptime:10" is specified, the IP Phone will ignore the global voice packetization and adjust to the negotiated packetization time.

INVITE sip:3101@SIP.isp.com SIP/2.0 Via: SIP/2.0/UDP 192.168.3.51:5060;branch=abc7801 From: 7751 <sip:7751@SIP.isp.com>;tag=22516 To: <sip:3101@SIP.isp.com> Call-ID: 1157628352@192.168.3.51 CSeq: 1 INVITE Contact: "7751" <sip:7751@192.168.3.51:5060> User-Agent: SIP-Phone/1.1 Content-Length: 171 Content-Type: application/sdp

v=0 o=SIP-Phone 0 0 IN IP4 192.168.3.51 m=audio 5102 RTV/AVP 0 8 18 a=ptime:10

• Packet Rate and VoIP Bandwidth Consumption

6 (17 pps)

1 (100 pps)

2 (50 pps)

3 (33 pps)

4 (2<u>5 pps)</u>

5 (20 pps)

6 (17 pps)

G.711

64 kbps

38.1

100.8

84

78.4

75.6

73.9

72.8

39.7

105.6

86.4

80

76.8

74.9

73.6

Packet rate is especially important for sizing a network against a router because routers are not only constrained by bandwidth but the number of packets per second (PPS) that they can process.

	F		L	.ink	Accessed		Mean
CODEC	RTP Packet (Packet Rate)	IP (kbps)	PPP (kbps)	Ethernet 802.3	Band-width Utilization	Delay (ms)	Opinion Score (MOS ⁱ)
	1 (35 pps")	16.8	18.4	21.6	28%	37.5	
	2 (17 pps)	11.2	12	13.6	43.2%	67.5	
G.723.1	3 (12 pps)	9.3	9.9	10.9	52.8%	97.5	3537
5.3 kbps	4 (9 pps)	8.4	8.8	9.6	59.3%	127.5	5.5-5.7
	5 (7 pps)	7.8	8.2	8.8	64.1%	157.5	
	6 (6 pps)	7.5	7.7	8.3	67.7%	187.5	
	1 (35 pps)	17.9	19.5	22.7	31.9%	37.5	
	2 (17 pps)	12.3	13.1	14.7	47.9%	67.5	
G.723.1	3 (12 pps)	10.5	11	12.1	57.1%	97.5	2040
6.4 kbps	4 (9 pps)	9.5	9.9	10.7	63.8%	127.5	3.0-4.0
	5 (7 pps)	9	9.3	9.9	68.1%	157.5	
	6 (6 pps)	8.6	8.9	9.4	71.4%	187.5	
	1 (100 pps)	42	46.8	56.4	16.5%	15	
	2 (50 pps)	25.2	27.6	32.4	28.2%	25	
G.729ab	3 (35 pps)	19.6	21.2	24.4	36.8%	35	2740
8 kbps	4 (25 pps)	16.8	18	20.4	43.5%	45	3.7-4.2
	5 (20 pps)	15.1	16.1	18	48.8%	55	
	6 (17 pps)	14	14.8	16.4	53.1%	65	
	1 (100 pps)	64	73.6	75.2	43%	10	
	2 (50 pps)	48	52.8	53.6	60.2%	20	
G.726	3 (35 pps)	44.8	48.2	48.7	66%	30	11
32 kbps	4 (25 pps)	40	42.4	42.8	75.1%	40	4.1
	5 (20 pps)	38.4	40.3	40.6	79.11%	50	

This table calculates the bandwidth consumption based on uni-directional voice only. So for the total bandwidth used for media during a simple two-way call, you should multiply the value by two.

Those values in <u>blue</u> font are near optimal settings for the specific CODEC in terms of bandwidth consumption and delay. In general, the more frames carried in one RTP packet, the less bandwidth it consumes but in the cost of longer delay. We suggest a good setting should not introduce a delay for more than 60 ms.

40

115.2

91.2

83.2

79.2

76.8

75.2

80.3

59.9%

73.4%

79.5%

82.9%

85.1%

86.6%

60

10

20

30

40

50

60

4.3-4.7

Comfort Noise

You can prioritize those CODECs which are not CNG (Comfortable Noise Generation") capable over those are capable of CNG. If you disable "CNG", then IP Phone will first try those not-CNG capable CODECs (such as G.729A and G.723.1 instead of G.729AB and G.723.1A) during call setup. Note, this is only a best-effort attempt: whenever IP Phone received a silence packet from the peer (SID packets if G.723.1A is adopted), it will automatically switch to the corresponding CNG-capable CODEC, even you have disabled "CNG".

Default is enabled

	[On]	/ [0	ff]	To in-l	ggle line i	betv f ena	veen abled	ena	ble a	and (disat	ole t	his f	eatur	re. S	how	che	ck s	ymb	ol '√'
	[В	ack]		Re	turn	with	out a	ny c	hang	es.										
	[0) K]		Re	Return.															
ľ	_ ↑ 】 a	nd	(↓】	Sc	roll n	nenu	ı iterr	ıs.												
1		Р	r	e	f	e	r	e	n	c	e	S								
2		Р	a	c	k	e	t	i	z	a	t	i	0	n						
3	\checkmark	С	0	m	f	0	r	t		Ν	0	i	S	e						
4		R	F	C	2	8	3	3		Р	Т	:	1	0	1					
5		D	Т	Μ	F		R	e	1	a	у		b	y		Ι	N	F	0	
	On/Off]	Back		

RFC2833 PT

Specify the RTP payload type for the transmission of out-of-band DTMF over RTP as per RFC2833. Valid value MUST be between 96 and 127 in decimal. Specify 128 to disable RFC2833 and transmit DTMF in-band; i.e., mixing the DTMF tones with voice and transmitted in audio RTP stream.

Default is 101.

[Del]	Delete one character.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\uparrow] and [\downarrow]	Move cursor one position in edit mode.

R	F	С	2	8	3	3		Р	а	у	1	0	а	d	Т	у	р	e	:	
[9	6	-	1	2	7]													
1	0	1																		
	Del																Back	K		

DTMF Relay by INFO

This terminal supports DTMF relay over SIP signaling channel by INFO method (RFC2976) in addition to DTMF over RTP (by either RFC2833 or mixed with normal voice stream).

INFO sip:3101@ SIP.isp.com SIP/2.0
Via: SIP/2.0/UDP 192.168.3.51:5060;branch=abc7801
From: 7751 <sip:7751@ sip.isp.com="">;tag=22516</sip:7751@>
To: <sip:3101@ sip.isp.com="">;tag=99123</sip:3101@>
Call-ID: 1157628352@192.168.3.51
CSeq: 1356 INFO
Contact: "7751" <sip:7751@192.168.3.51:5060></sip:7751@192.168.3.51:5060>
User-Agent: IP PHONE/1.2.0
Content-Length: 24
Content-Type: application/dtmf-relay
Signal=1
Duration=250

Once enabled, all DTMF keys, 0-9*#&, will be sent by SIP INFO method; otherwise they will be transmitted by DTMF over RTP.

Default is disabled.

	[On]	/ [01	ff]	To in-l	ggle line i	betv f ena	veen abled	ena I.	ble a	and o	disat	ole t	his f	eatu	re. S	show	/ che	ck s	ymbo	l '√'
	[B	ack]		Re	turn	with	out a	iny c	hang	jes.										
	[0	DK]		Re	Return															
ľ	↑】a	nd 🕻	[↓]	Sc	roll n	nenu	ı iten	าร.												
1		P	r	e	f	e	r	e	n											
2	\	P C	a	c m	K f	e	t r	1	Z	a N	t	1 i	0	n						
4	V	R	F	C III	2	8	3	3		P	T	•	1	0	1					
5		D	Т	М	F	Ū	R	e	1	a	y	•	b	y	-	Ι	Ν	F	0	
		C)n/O	ff														Back	X	

System Admin

• [Re-Register]: Re-REGISTER all activated SIP address-of-records to SIP service again without user confirmation.

If the user has gone off-line (see below) explicitly, restart the auto-registration process to let user go online again.

• [Logout]: Goes off-line without user confirmation.

Unregister all SIP address-of-records to all activated SIP service immediately and stop the regular auto-registration scheduling to keep user offline. The phone will keep unregistered until the "Re-REGISTER" command is issued to go online explicitly. This is useful when you are off work and want all calls go directly to ring your home lines.

VNote

Reboot the terminal will clear this status and register all activated SIP address-of-records after startup.

- [Restart]: Save all configuration data back to NVRAM and restart (reboot) IP Phone.
- [Shut Down]: Shut down terminal for turning off power.

After shutdown, all indicators (LEDs) and status LED will be off, then you could power off your terminal safely.

• [Reset System Config]: Reset all system settings back to factory default values (such as restore both privileged password and user-level password to "0000") but leave general network settings and personal information intact.

Those unchanged user data include:

- General Network Settings: Mode, Static Setings, PPPoE Settings, DNS Settings, and IP-TTL.
- Address Book
- Speed Dials
- Call Screen
- Favorite List
- IMPP List
- Call History: Missed, Received and Dialed Calls.
- Instant Messages: Inbox, Outbox and Draft
- Call Statistics
- **[Factory Values]:** Restore everything, including network settings and personal information, back to factory values, then auto-reboot the terminal.

1	R	e	-	R	e	g	i	S	t	e	r									
2	L	0	g	0	u	t														
3	R	e	S	t	a	r	t													
4	S	h	u	t		D	0	w	n											
5	R	e	S	e	t		S	у	S	t	e	m		C	0	n	f	i	g	
6	F	a	c	t	0	r	y		V	а	1	u	e	S						
]	Back	2		

• **[OK]** : Execute command.

1	R	e	-	R	e	g	i	S	t	e	r									
2	L	0	g	0	u	t														
3	R	e	S	t	a	r	t													
4	S	h	u	t		D	0	W	n											
5	R	e	s	e	t		S	у	S	t	e	m		С	0	n	f	i	g	
6	F	a	c	t	0	r	у		V	а	1	u	e	s						
		Yes															No			

- [Yes]: Confirm execution
- [No]: Return

[Re-Register] and [Logout] will not ask for user's confirmation but executing immediately.

System Status

Display current system information.

[Back]	Return without any changes.
[OK]	View selected item.
[\uparrow] and [\downarrow]	Navigate through items

1	Ν	e	t	W	0	r	k		S	t	a	t	u	S					
2	S	y	S	t	e	m		U	р		Т	i	m	e					
3	C	a	1	1		S	t	a	t	i	S	t	i	c	s				
4	S	y	S	t	e	m		Ι	n	f	0	r	m	а	t	i	0	n	
5	V	e	r	S	i	0	n												
																	Back	<u> </u>	

Network Status

Display currently active host IP, Ethernet MAC address and the active DNS IP (secondary DNS will be shown if available), LAN and PC Ethernet link status in order. The phone will set MAC as it host name in 12 hexadecimal characters.

VNote

this item is identical to "6.Network" / "Active Status".

LAN and PC Ethernet link status are one of the following:

- 1. Down: not connected
- 2. Up: Ethernet jack connected
 - 100M, Full: 100 MB, Full-Duplex (Usually connects to a 100-MB Ether-Switch).
 - 100M, Half: 100 MB, Half-Duplex (Usually connects to a 100-MB Ethernet hub).
 - 10M, Full: 10 MB, Full-Duplex (Usually connects to a 10-MB Ether-Switch).
 - 10M, Half: 10 MB, Half-Duplex (Usually connects to a 10-MB Ethernet hub).

[Back]	Return without any changes.
[OK]	View selected item.
[↑] and [↓]	Scroll screen per line if there are too many lines to fit into one screen.

Ι	Р	:	1	9	2		1	6	8		3		2	0	0					
Μ	А	С	:	0	0	0	Е	С	3	1	2	3	4	5	6					
D	Ν	S	:	1	9	2		1	6	8		3		1						
D	Ν	S	:	1	9	2		1	6	8		3		2	5	4				
L	Α	Ν	:	1	0	0	Μ	,	F	u	1	1	-	D	u	р	1	e	X	
Р	С	:	:	D	0	W	n				u I I - D u									
																		Bacł	K	

System Up Time

Show the system up time since last boot-up in the format of "xxxD HH:MM::SS".

	[B	ack]		R	eturr	n witl	hout	any	chan	ges.									
	[0)K]		V	iew s	selec	ted i	tem.											
S 1	y 3	s D	t	e 0	m 2	:	U 1	p 9	:	T 4	i 3	m	e	:					
]	Back		

Call Statistics

	[8	Back]	R	eturr	n with	nout	any c	han	ges.										
	ľ	OK]		E	nter	selec	ted	subm	enu											
1 2	•	L L	a a	S S	t t	e	s 7	t 2		5 H	0 0	u	C r	a s	1	1	S			
3		A	c	c	u	m	u	1	a	t	e	d								
			Ping	5														Bacl	ĸ	

Lastest 50 Calls

Summarize the latest 50 (or fewer if no traffic of such amount is available) calls. Those counters will reset to zero whenever the system reboots.

- 1. Success calls: the counter of connected calls in the latest 50 calls.
- 2. Fail rate: the failed (not connected) ratio of the latest 50 calls.
- 3. Duration: the average conversation time of those successfully connected calls.
- 4. ACK timeout: the counter of inbound calls which disconnected due to ACK timeout. Only calls lasted less than 32 seconds may disconnect due to ACK timeout.

[Back]	Return without any changes.
[OK]	Enter selected submenu.
[↑] and [↓]	Scroll screen per line if there are too many lines to fit into one screen.

1	C	0	n	n	e	c	t	e	d		C	a	1	1	S	:	1	6	
2	F	а	i	1		R	a	t	i	0	:	2	5	%					
3	Α	v	g		D	u	r	а	t	i	0	n	:	1	3	:	2	1	
4	Α	С	K		Т	i	m	e	0	u	t	:	3						
																	Back	K	

Last 72 Hours/Accumulated

"Last 72 hours" summarizes call statistics during the last 72 hours, and "Accumulated" summarizes call statistics since system startup.

[Back]	Return without any changes.
[0K]	Enter selected submenu.

 \uparrow and \downarrow Scroll screen per line if there are too many lines to fit into one screen.

1	С	0	n	n	e	c	t	e	d		С	a	1	1	S				
2	C	a	1	1		А	t	t	e	m	р	t	S						
]	Back		

Connected Calls

Summarize the connected calls during the last 72 hours or since system startup. Those counters will all reset to zero whenever the system reboots.

- 1. **<3 Minutes:** Summarize those connected (either inbound or outbound) calls which lasted less than 3 minutes.
 - Total Call: the accumulated counter during the last 72 hours or since system startup
 - Avg. Duration (s): the average conversation time of this category; measured in seconds.
 - Ratio (%): the ratio of calls of this category to the total connected (both inbound and outbound) calls.
- 2. **3-20 Minutes:** Summarize those connected (either inbound or outbound) calls which lasted between 3 and 20 minutes.
- 3. **20-60 Minutes:** Summarize those connected (either inbound or outbound) calls which lasted longer than 20 minutes but shorter than one hour.
- 4. **>1 Hour:** Summarize those connected (either inbound or outbound) calls which lasted longer than one hour.
- 5. **Total:** Summarize all connected (either inbound or outbound) calls during the last 72 hours or since system startup.

[Back] Return without any changes.

(OK) View record.

 \uparrow and \downarrow Navigate through items..

1		<	3		Μ	i	n	u	t	e	S								
2		3	-	2	0		Μ	i	n	u	t	e	S						
3	•	2	0	-	6	0		М	i	n	u	t	e	S					
4	•	>	1		Н	0	u	r											
5	•	Т	0	t	a	1													
																B	Back		

<3 Minutes

- 1. Total Call: the accumulated counter during the last 72 hours or since system startup.
- 2. Avg. Duration (s): the average conversation time of this category; measured in seconds.
- 3. Ratio (%): the ratio of calls of this category to the total connected (both inbound and outbound) calls.

	[Ba	ack]		Re	eturn	with	iout a	iny c	hang	es.							 	
	[0	рк]		Re	eturn													
1		С	a	1	1	:	4	6										
2	. D u				a	t	i	0	n	:	1	:	2	8				
-		-						-	-	0 1								

-	·		u	1	u	Ľ	1	U	11	•	1	•	-	0					
3		R	а	t	i	0	:	6	3	%									
]	Back		

Total

Summarize all connected (either inbound or outbound) calls during the last 72 hours or since system startup.

	[B	ack]		Re	eturn	with	nout a	any c	hang	es.								
	[) K]		Re	eturn													
1 2	- -	C D	a u	1 r	1 a	: t	2 i	6 0	8 n	-	4	:	2	1				
																Ba	ack	

Call Attempts

Summarize the inbound and outbound call attempts during the last 72 hours or since system startup.

- 1. Inbound: Summarize the inbound calls, including
 - Total Call: the total inbound calls during the last 72 hours or since system startup.
 - Connected Call: the counter of connected (taken/picked-up) inbound calls.
 - Fail ratio (%): the ratio of those calls failed to take to the total inbound calls.
 - ACK Timeout: the accumulated counter of inbound calls which disconnected due to ACK timeout. Only inbound calls lasted less than 32 seconds may disconnect due to ACK timeout.
- 2. Outbound: Summarize the outbound calls.
- 3. Total: Summarize all call attempts during the last 72 hours or since system startup, including both inbound and outbound calls.

[Back]	Return without any changes.
[0K]	Return.

 \uparrow and \downarrow Navigate through items.

1		Ι	n	b	0	u	n	d								
2		0	u	t	b	0	u	n	d							
3	•	Т	0	t	a	1										
													B	Back		

Inbound/Outbound

- 1. Total Call: the total inbound calls during the last 72 hours or since system startup.
- 2. Connected Call: the counter of connected (taken/picked-up) inbound calls.
- 3. Fail ratio (%): the ratio of those calls failed to take to the total inbound calls.
- 4. ACK Timeout: the accumulated counter of inbound calls which disconnected due to ACK timeout.

[Back]	Return without any changes.
[OK]	Return.
[\uparrow] and [\downarrow]	Scroll screen per line if there are too many lines to fit into one screen.

1	C	a	1	1	:	4	6											
2	С	0	n	n	e	с	t	e	d	:	2	3						
3	F	а	i	1		R	a	t	e	:	5	0	%					
4	Α	С	Κ		Т	i	m	e	0	u	t	:	4					
]	Back		

Total

Summarize all call attempts during the last 72 hours or since system startup, including both inbound and outbound calls.

- 1. Call: the total calls, including both inbound and outbound calls, during the last 72 hours or since system startup.
- 2. Connected: the counter of all connected calls, which is as the same as the sum of inbound calls and outbound calls.
- 3. Fail ratio: the ratio of all failed calls to the total call attempts.

	[B	ack]		Re	eturn	with	iout a	any c	hang	es.							
	[0)к]		R	eturn	I.											
1 2 3	•	C C F	a 0 a	1 n i	1 n 1	: e	5 c R	7 t a	e t	d e	33	8 4	9%0				
															Bac	k	

System Information

Display the CPU family, hardware serial number and hardware information.

[Back]	Return without any changes.
[OK]	Return.
[\uparrow] and [\downarrow]	Scroll screen per line if there are too many lines to fit into one screen.

С	Р	U	:	Α	С	4	9	х		1	5	0	Μ	Н	Z			
D	S	Р	:	Α	С	4	9	5										
Μ	e	m	0	r	у	:	3	2	Μ	В								
F	1	a	S	h	:	4	Μ	В										
S	/	Ν	:	1	2	3	4	5	6	7	9	0						
																Back	K	

Version

Display the product version, software and firmware version of the phone.

[Back] Return without any changes.	
------------------------------------	--

[OK] Return.

 \uparrow and \downarrow Scroll screen per line if there are too many lines to fit into one screen.

F	i	r	m	W	а	r	e	:	Y	V	3		1		0		3				
А	р	р	1	i	с	а	t	i	0	n	:	1		0		3		1	2		
D	r	i	v	e	r	:	1		0		3		4								
Н	a	r	d	w	а	r	e	:	1		0		3		1						
F	а	c	t	0	r	y		V	а	1	u	e	:	1		0		3		4	
]	Back	K		

Log

Modules could log debug information to either the embedded COMM port or a network log server. Note: change log level may have negative impacts on the performance of IP Phone, such as intermittent voice and lagged responses.

	[B	ack]		Re	eturn	with	out a	any c	hang	ges.									
	[0) K]		En	iter s	ubm	ienu.												
1 2 3	-	L L N	0 0 e	g g t	g W	C e 0	0 r r	m k	p T	o y L	n p o	e e g	t g	e	r				
																	Back	5	

Log Component

Define the log verbosity of each module.

[Back]	Return without any changes.
[\uparrow] and [\downarrow]	Scroll menu items.
[OK]	Enter selected item.

1	U	t	i	1	i	t	у	:	F	а	t	а	1						
2	Ν	e	t	W	0	r	k	:	F	a	t	a	1						
3	R	Т	Р	:	F	a	t	a	1										
4	S	D	Р	:	F	a	t	a	1										
5	S	Ι	Р	:	F	a	t	a	1										
6	Ν	e	g	0	t	i	a	t	0	r	:	F	a	t	a	1			
7	D	e	v	Μ	g	R	:	F	a	t	a	1							
																	Back	K	

Valid values include:

VNote

Log Level	Verbosity
Fatal (Default)	Not recoverable error
Error	Recoverable Error
Warning	
Event	
Informative	
Trace	
Trace Level 2	
Trace Level 3	
Detail	
Debug	Everything

Change the log level for each module may have negative effect on the performance of IP Phone, such as intermittent voice and lagged response. Remember to lower the log level or set it back to default level (Fatal level) whenever you have finished tracing debug information..

[Back]	Return without any changes.
[\uparrow] and [\downarrow]	Scroll menu items.
[OK]	Save changes and return.

0	F	а	t	а	1												
1	E	r	r	0	r												
2	W	a	r	n	i	n	g										
3	E	v	e	n	t												
4	Ι	n	f	0	r	m	a	t	i	v	e						
5	Т	r	a	c	e												
6	Т	r	a	c	e	2											
7	Т	r	a	c	e	3											
8	D	e	t	a	i	1											
9	D	e	b	u	g												
														E	Back		

Logger Type

- **1. File Logger:** output logs to standard output, such as COMM port.
- 2. **TCP Logger:** output logs to a network log server by setting up a TCP stream.
- UDP Logger: output logs to a network log server by UDP packets.
 Default is File Logger.

[Back]	Return without any changes.
[\uparrow] and [\downarrow]	Scroll menu items.
[0K]	Save changes and return.

1	F	i	1	e		L	0	g	g	e	r						
2	Т	D	Р		L	0	g	g	e	r							
3	U	D	Р		L	0	g	g	e	r							
]	Back		

Network Logger

Assign the network server for log output. You can use either a dotted IP address or a DNS name. The phone will use port 901 connects to default log server port 56447 (either TCP or UDP). To connect to a different server port, please specify as "xxx.xxx.xxx:port".

[Del]	Delete one character.
[Abc]/[123]	Toggle between digits and alphanumeric input, where. [Abc] indicates current input method is alphanumeric and [123] indicates digits input.
[Clear]	Clear all input.
[Back]	Return without any changes.
[OK]	Save changes and return.
[\leftarrow] and [\rightarrow]	Move cursor one position in edit mode.
[\uparrow] and [\downarrow]	Move cursor per line in edit mode.

Ν	e	t	W	0	r	k		L	0	g	g	e	r	:					
			Del				Ab	c/12	23				Clea	r]	Back		

Appendix A Dial Plan

The programmable dial plan is designed for the service provider to customize the behavior of the IP Phone for collecting and sending dialed digits. The dial plan allows the terminal user to specify the events that trigger the sending of dialed digits. These events include the following:

- The termination character has been entered.
- The specified dial string pattern has been accumulated.
- The specified number of dialed digits has been accumulated.

PLANET

IP Phone Configuration

🔁 Configuration Menu	-	
🖶 🗀 Address Book	Preferences	
🖻 🗀 Call History		Auto Answer
🗉 🗀 Terminal Settings		Call Waiting
🖻 🕒 Call Forward		Message Alert
🖻 🔂 Preferences		LAN Dial
🕆 🖻 Voice Mailbox		Unhold on Transfer
🖻 🖻 Call Command		Play Alerting Tone during Conference
🕒 🎦 Auto Answer		Transfer after Conference Teardown
🖻 🗀 Network	Dial Timeout(s)	30
🖶 🗀 SIP Settings	Hold Recall Alerting Timeout(s)	60
Auto Provision	Inter-Digit Timeout(s)	4
🖶 🗋 Instant Message		
🖽 🗀 Advanced	L Hot Line	
	Dial Plan	
	Auto-redial	
	Stop Criterion	On Ringing
	Retry Interval(s)	15
	Total Duration Time(s)	1800
	Mo	dify

Dial Plan consists of alphanumeric string, and the maximum number of characters is 255.

System default dial plan is empty, which leads to recognize any dial strings ending with a pound, '#', sign (identical to the dial plan string ".>#-", excluding the double quotes).

Dial Plan Commands

• Commands that can be used to create you own dial plans:

Command	Description
	Wildcard, match any digit, "0-9#*", entered.
-	Additional (zero or more) digits can be entered. This command can be used only at the end of a dial plan rule (for example, 1408t5- is legal usage of the - command, but 1408t5-3 is illegal).
۵	Range, which means to match any single digit in the list. Use an underscore () to indicate a range of digits. For example, [135] matches the digits 1, 3, and 5. Also, [1_5] matches the digits 1, 2, 3, 4 and 5. The pound key (#) and asterisk (*) are not allowed in the Range command. The Range command does not allow repeat or sub-rule matching.
(subrule0) subrule1) subruleN)	Sub-rule matching. Using the () and operators allows you to specify multiple sub-rules within a dial plan rule so that a sub-rule match is reached if the entered digits fit one of the sub-rules. This can be used to reduce the length of the desired dial plan rule by concatenating the group of the sub-rules with the common rule. For example, a dial plan rule of (1900 1800 17)555.r3 or three dial plan rules of 1900555.r3 1800555.r3 17555.r3 are equivalent. A match is reached if 11 digits are entered and the first three digits are either 1900, 1800, or 17, and the fifth, sixth, and seventh digits are all 5.
>#	Defines the # character as a termination character. When the termination character is entered, the dial string is automatically sent. The termination character can be entered only after at least one user-entered digit matches a dial plan rule. Alternatively, the command >* can be used to define * as the termination character.
rn	Repeat the last pattern n times, where n is 0-9 or a-z. The values a-z indicate a range of 10 to 35. Use the repeat modifier to specify more rules in less space. Note The commands ># and tn are modifiers, not patterns, and are ignored by the rn command.
	Used to separate multiple dial plan rules.
۸	Logical not. Match any character except the character immediately following the ^ command. The ^ command can also be used as a negation instruction before the range or sub-rule matching commands.
S	Seize rule matching. If a dial plan rule matches the sequence of digits entered by the user to this point, and the modifier S is the next command in the dial plan rule, all other rules are negated for the remainder of the call (for example, a dial plan beginning with *S will be the only one in effect if the user first enters the * key). Note All rules apply in the order listed (whichever rule is completely matched first will immediately send the dial string). Note No syntax check is performed by the actual implementation. The administrator has the responsibility of making sure that the dial plan is syntactically valid.

Dial Plan Rules

Dial plan rules supported by IP Phone

- (In Rule) for Dial Plan Blocking
- 'P' Rule to Support Dial Prefix
- 'R' Rule for Enhanced Prefix
- 'A' Rule for Suffix Appending
- 'C' Rule for Call Blocking
- 'X' Rule for Call Blocking and Call Forwarding Blocking
- 'D' Rule for Displaying Caller ID

(In Rule) for Dial Plan Blocking

Dial plan blocking can be used to reduce the occurrences of invalid dialed digits being sent and can prevent the dialed string of a specified pattern from being sent. By adding dial plan blocking, dialed digits are discarded after the inter-digit timer expires unless one of the specified matching

rules is met.

In addition, the default four-second global inter-digit timeout value is also modified with the value specified in the dial plan blocking command:

• Syntax: In

where *n* specifies the global inter-digit timeout and the valid values are 1-9 and a-z (10-35).

• **Example:** *lc* | 911

This command specifies an inter-digit timeout of 12 seconds, and will discard dialed digits unless 911 is entered.

Specifying your own inter-digit timeout also changes the behavior of the dial plan so that the entire dial string, rather than being sent at timeout, is sent only as a result of a matching rule or time intended by a matching rule.

'P' Rule for Dial Prefix

This rule is for automatic pre-pending the dial string as entered by the user with a specified prefix.

• Syntax: Ptnnnn

Where *t* is a single leading trigger character; if *t* is the *first* entered digit when making a new call, it triggers the prepending of a variable-length prefix (as specified by *nnnn*) in the dial string. The *t* character can take one of the following values:

0-9,*,#, 'n' (= any of 1-9), 'N' (any of 'n' and 0), 'a' (any of 'n',* and #), or 'A' (any of 'a' and 0);

• Example: Pn12345

This rule prepends 12345 to the dial string when the first entered digit is any of 1-9. The triggered digit is not removed from the dial string.

'R' Rule for Enhanced Prefix

This enhanced prefix rule matches entire strings, whereas the 'P' rule matches only a single digit. The 'R' rule is for automatically prepending a specified prefix to the dialed string. The string must be an exact match to trigger the rule. If more than one 'R' rule matches, the first matched 'R' rule is triggered.

The 'R' rule also uses negation to exclude one or more leading digits before prepending the defined prefix string.

The number of dashes (-) after the *R* represents the number of leading digits that will be removed preceding the prefix.

• **Syntax:** Rnnnn(tttt)

Where *tttt* is a trigger string. If the dialed numbers match this string, this match triggers the prepending of a variable-length prefix (as specified by *nnnn*) to the dial string. The triggered string is not removed from the dial string. The negation, sub-rule matching and range patterns can be applied to the trigger strings.

- Example 1: R1212([2_9]-)
 This rule prepends 1212 to dial strings that have a leading digit of 2 to 9.

 Note: 'R' rules can replace most 'P' rules; for example, *Pn12345* is the same as *R12345([1 9]-)*.
- Example 2: R-0033(0[1-9].r7) This removes the first dialed digit, and then prepends 0033 to the dialed string. For example, if the number 0148336134 is dialed, the resulting string becomes 0033148336134.
- **Example 3:** R----0(0033[1-9].r7) This removes the first four dialed digits, and then prepends 0 to the dialed string. For example, if

the number 0033148336134 is dialed, the resulting string becomes 0148336134.

'A' Rule for Suffix Appending

This suffix appending rule matches entire strings. The 'A' rule is for automatically appending a specified suffix at the tail of the dialed string. The string must be an exact match to trigger the rule. If more than one 'A' rule matches, the first matched 'A' rule is triggered.

The 'A' rule also uses negation to exclude one or more leading digits before appending the defined suffix.

• Syntax: Tnnnn(tttt)

Where *tttt* is a trigger string. If the dialed numbers match this string, this match triggers the appending of a variable-length suffix (as specified by *nnnn*) to the dial string. The triggered string is not removed from the dial string. The negation, sub-rule matching and range patterns can be applied to the trigger strings.

- **Example 1:** A1234([2_9]-) This rule appends 1234 to dial strings that have a leading digit of 2 to 9.
- Example 2: A@10.0.1.100.1(8[1-9].r3) This appends "@10.0.100.1" to the dialed 4-digit string prefixed by a '8'. For example, if the number "8192" is dialed, the resulting string becomes <u>8192@10.0.1.100.1</u>
- Example 3: A@192.168.1.254(00-) This appends "@192.168.1.254" to any dialed string starts with '00'. For example, if the number 0033148336134 is dialed, the resulting string becomes <u>0033148336134@192.168.1.254</u>.

'C' Rule for Call Blocking

This rule is for blocking call numbers.

• Syntax: Cnnnn

Where *nnnn* is the leading set of digits of the blocked call number; *nnnn* can be composed with sub-rule matching and range. The rule is triggered when the leading digits of a dialed string match the string *nnnn*.

The 'C' rule does not work with negation.

• **Example:** C1900|C1888 or C(1900|1888)

This rule blocks call numbers beginning with 1900 or 1888.

'X' Rule for Call Blocking and Call Forwarding Blocking

This rule is for blocking call numbers and call forwarding numbers.

• Syntax: Xnnnn

Where *nnnn* is the leading set of digits of the blocked call number and blocked call forwarding number; *nnnn* can be composed with sub-rule matching and range. The rule is triggered when the leading set of digits of a dialed call number or forwarding number match the string *nnnn*. The 'F' rule does not work with negation.

• Example: X1900|X1888 or X(1900|1888)

This rule blocks the call numbers and call forwarding numbers beginning with 1900 or 1888.

'D' Rule for Displaying Caller ID

This rule is for displaying caller ID at the remote site. The number must be an exact match to trigger the rule.

• Syntax: Dnnnn

Where *nnnn* is the callee number. The caller ID will show to the callee; *nnnn* automatically becomes a valid calling number. Also, *nnnn* can be composed with negation, sub-rule matching and range. The 'D' rule is checked before the 'R' and 'P' rules.

• Example: D911

This rule shows the caller ID at the remote side when if the call number is 911.

Appendix B Dial Plan Samples

Example 1

The following dial plan:

*St4-/#St4-/911/1>#t8.r9t2-/0>#t811.rat4-/^1t4>#-

consists of the following rules:

- ***St4-**: If the first digit entered is *, all other dial plan rules are voided. More digits can be entered after the initial * digit, and the timeout before automatic dial string send is four seconds.
- **#St4-**: same as above, except for **#** as the initial digit entered.
- **911**—if the dial string 911 is entered, send it immediately.
- 1>#t8.r9t2-: If the first digit entered is 1, the timeout before automatic send is eight seconds. The terminating character # can be entered at any time to manually send the dial string. After the 11th digit is entered, the timeout before an automatic send changes to two seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.
- **0>#t811.rat4**-: If the first digit entered is 0, the timeout before automatic send is eight seconds, and the terminating character **#** can be entered at any time to manually send the dial string. If the first three digits entered are 011, then, after an additional 11 digits are entered, the timeout before an automatic send changes to four seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the **#** character.
- **^1t4>#**-: If the first digit entered is anything other than 1, the timeout before an automatic send is four seconds. The terminating character # can be entered at any time to manually send the dial string. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.

Example 2

The following dial plans:

.t7>#.....t4-|911|1t7>#.....t1-|0t4>#.t7-

or

.t7>#r6t4-|911|1t7>#.r9t1-|0t4>#.t7-

consist of the following rules:

- .t7>#r6t4-: You must enter at least one digit. After the first digit is entered and matched by the dial plan, the timeout before an automatic send is seven seconds, and the terminating character # can be entered at any time to manually send the dial string. After seven digits are entered, the timeout before an automatic send changes to two seconds. The symbol at the end of the rule allows further digits to be entered until the dial string is sent by the timeout or the user entering the # character.
- **911**: if the dial string 911 is entered send this string immediately.
- **1t7>#.r9t1-**: If the first digit entered is 1, the timeout before an automatic send is seven seconds, and the terminating character **#** can be entered at any time to manually send the dial string. After the 11th digit is entered, the timeout before an automatic send changes to one second. The user can enter more digits until the dial string is sent by the timeout or by the user entering the **#** character.
- **0t4>#.t7**-: If the first digit entered is 0, the timeout before an automatic send is four seconds, and the terminating character **#** can be entered at any time to manually send the dial string. After the second digit is entered, the timeout before an automatic send changes to seven seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the **#** character.

Example 3

The following dial plan:

R1408([2_9].r5|[2_9].r6)|R9^(911|.r4)|X(1900|1888)

consists of the following rules:

- **R1408([2_9].r5|[2_9].r6)** : The prefix 1408 will be added to any call numbers with seven or eight digits where the leading digit is in the range of 2 to 9. For example, 5551234 will become 14085551234, but 555123 does not match this rule.
- **R9^(911|.r4)** : The prefix 9 will added to any numbers except 911 and five-digit numbers. For example, 911 will still be 911, and 51234 will still be 51234.
- X(1900|1888) : There will be no calls or call forwarding to numbers beginning with 1900 or 1888.

Appendix C VIP-350PT Specifications

Product	Business PoE IP Phone	
Model	VIP-350PT	
Hardware		
LAN	1 x 10/100 Base-TX RJ-45 port (802.3af support)	
PC	1 x 10/100 Base-TX RJ-45 port	
Display	128 * 32-pixels LCD with Back Light	
I/O Interfaces	1 x RJ-9: Handset Connector 2 x Phone Jack: Headset Connector Embedded MIC/SPEAKER	
Function Keys	8 x Fixed Function Buttons (Hold, Speaker, VM, Transfer, Redial, Mute, Menu,	
	SPD), 2 Red LED (VM, Mute), 1 Green LED (Speaker)	
	4 x Soft Buttons	
	5 x Navigation Buttons	
	2 x Volume Buttons	
Protocols and Standard		
Standard	SIP 2.0 (RFC3261), Session Timer (RFC4028), SDP (RFC 2327), RTP/RTCP (RFC2833 and RFC3551), NARTP for SIP URI Lookup (RFC2915), STUN (RFC 3489), ARP/RAPP (RFC826/903), SNTP (RFC2030), Message Waiting Indicator (RFC3842), DHCP Option Codes for SIP (RFC3361)	
Voice Codec	G.711(A-law /µ-law), G.729 A/B, G.723.1 (6.3 Kbps / 5.3Kbps)	
Voice Standard	VAD (Voice Activity Detection) CNG (Comfort Noise Generation) AEC (Acoustic Echo Cancellation) In-band, out-of-band DTMF Relay (RFC 2833) and SIP INFO Adaptive Jitter Buffer Management	
Protocols	TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE, VLAN 802.1p/q, IP-TOS/DiffServ	
Applications	Instant Message, Voice Mail Integration, NAT Traversal (Static NAT Route/ Traversal by STUN and UPnP)	
Features		
Call Features	3-way Conference, Auto Answer/Redial, Call History (20 Most Recently Missed/ Received/ Dialed Calls), Call Hold/Forwarding/Mute/Pickup/Progress Tone/Rejection/Screening/Transfer/Waiting, Caller ID, CLIP/CLIR, Do Not Disturb (DND), Hotline, Phone Locked, Presence, Redial, Shared Line Appearance, Speed Dial, Address Book (100Entries)	
Network and Configuration		
Management	Web/LCD Menu-drive/Telnet Management Interface/Auto Provisioning	
Dimension (W x D x H)	210 mm x 190 mm x 86.4 mm	
Operating Environment	0~40 degree C, 0~95% humidity	
Power Requirement	12V DC	
EMC/EMI	CE, FCC	

Appendix D VIP-550PT Specifications

Product	Professional PoE IP Phone	
Model	VIP-550PT	
Hardware		
LAN	1 x 10/100 Base-TX RJ-45 port (802.3af support)	
PC	1 x 10/100 Base-TX RJ-45 port	
Display	132 * 64-pixels LCD with Back Light	
I/O Interfaces	1 x RJ-9: Handset Connector 2 x Phone Jack: Headset Connector Embedded MIC/SPEAKER	
Function Keys	8 x Fixed Function Buttons (Hold, Speaker, VM, Transfer, Redial, Mute, Menu,	
	SPD), 2 Red LED (VM, Mute), 1 Green LED (Speaker)	
	8 x Soft Buttons	
	5 x Navigation Buttons	
	2 x Volume Buttons	
Protocols and Standard		
Standard	SIP 2.0 (RFC3261), Session Timer (RFC4028), SDP (RFC 2327), RTP/RTCP (RFC2833 and RFC3551), NARTP for SIP URI Lookup (RFC2915), STUN (RFC 3489), ARP/RAPP (RFC826/903), SNTP (RFC2030), Message Waiting Indicator (RFC3842), DHCP Option Codes for SIP (RFC3361)	
Voice Codec	G.711(A-law /µ-law), G.729 A/B, G.723.1 (6.3 Kbps / 5.3Kbps)	
Voice Standard	VAD (Voice Activity Detection) CNG (Comfort Noise Generation) AEC (Acoustic Echo Cancellation) In-band, out-of-band DTMF Relay (RFC 2833) and SIP INFO Adaptive Jitter Buffer Management	
Protocols	TCP/IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE, VLAN 802.1p/q, IP-TOS/DiffServ	
Applications	Instant Message, Voice Mail Integration, NAT Traversal (Static NAT Route/ Traversal by STUN and UPnP)	
Features		
Call Features	3-way Conference, Auto Answer/Redial, Call History (20 Most Recently Missed/ Received/ Dialed Calls), Call Hold/Forwarding/Mute/Pickup/Progress Tone/Rejection/Screening/Transfer/Waiting, Caller ID, CLIP/CLIR, Do Not Disturb (DND), Hotline, Phone Locked, Presence, Redial, Shared Line Appearance, Speed Dial, Address Book (100Entries)	
Network and Configuration		
Management	Web/LCD Menu-drive/Telnet Management Interface/Auto Provisioning	
Dimension (W x D x H)	210 mm x 190 mm x 86.4 mm	
Operating Environment	0~40 degree C, 0~95% humidity	
Power Requirement	12V DC	