

atl *IP300S*

User Guide



Directory

DIRECTORY	2
1. OVERVIEW.....	6
1.1. FEATURES.....	6
1.2. TECHNICAL SPECIFICATIONS	7
1.2.1. CALL CONTROL CAPABILITY	7
1.2.2. REAL-TIME VOICE STREAMING.....	8
1.2.3. NAT AND FIREWALL	8
1.2.4. MANAGEMENT.....	8
2. LAYOUT.....	10
2.1. HARDWARE	10
2.1.1. FRONT VIEW	10
2.1.2. REAR VIEW.....	10
2.1.3. BACK VIEW	11
2.2. KEYS	11
2.3. KEYPAD.....	14
3. OPERATION.....	15
3.1. KEY DEFINITIONS IN MENU MODE	15
3.2. ENTER ALPHABETS AND NUMBERS.....	16
3.3. ADDRESS-OF-RECORD (SIP AOR)	16
4. STARTUP.....	17
4.1. PREREQUISITE.....	17
4.1.1. NETWORK	17
4.1.1.1. DHCP	18
4.1.1.2. STATIC IP (FIXED IP)	18
4.1.1.3. PPPoE	18
4.1.1.4. VERIFY NETWORK CONFIGURATION.....	19
4.1.2. SIP SERVICE	19
4.1.3. CONFIGURE NAT AND FIREWALL.....	20
4.2. INITIALIZATION.....	21
4.3. REGISTRATION.....	22
5. SHUTDOWN	22
5.1. UNREGISTRATION	22
6. IDLE.....	23
6.1. REGISTERED	23
6.2. NOT REGISTERED YET OR REGISTRATION EXPIRES	23
6.3. REGULAR REGISTRATION	23
7. TAKE CALLS.....	24

7.1.	RINGING	24
7.2.	REJECT CALL	25
7.3.	FORWARD CALL	25
7.4.	ANSWER CALL	25
7.5.	CONNECTED	26
7.6.	DISCONNECTED	26
7.7.	FORWARD AND DND	28
7.7.1.	DO NOT DISTURB (DND)	28
7.7.2.	CALL FORWARD	28
7.7.2.1.	ALL CALLS FORWARD	29
7.7.2.2.	BUSY FORWARD	29
7.7.2.3.	NO ANSWER FORWARD	29
7.7.3.	FORWARDING RULES	29
8.	MAKE CALLS	30
8.1.	DIAL SCHEME	31
8.1.1.	GUARDING TIME	34
8.1.2.	ENUM SAMPLE	35
8.2.	REDIAL	37
8.3.	ADDRESS BOOK	37
8.4.	CALL HISTORY	38
8.5.	SPEED DIAL	39
8.6.	CALL RETURN	41
8.7.	CALLING	41
8.8.	CALL FAILURE	42
8.9.	AUTO-REDIAL	42
8.10.	ONE-TOUCH DIAL	43
9.	CALL PROCESSING	45
9.1.	HANDSET, SPEAKER-PHONE, EAR-PHONE AND LOUD-SPEAKER	45
9.2.	HOLD	45
9.3.	MUTE	46
9.4.	TRANSFER	46
9.4.1.	CONSULTATIVE TRANSFER	47
9.4.2.	BLIND TRANSFER	48
9.4.3.	PHONE LOCKED	48
9.5.	CONFERENCE	49
9.5.1.	HEURISTICS AND CONSTRAINT	49
9.5.2.	CONFERENCE TIPS	50
9.6.	BLOCK CALLS	51
10.	CALL PREFERENCE	54

10.1.	CALL WAITING	54
10.2.	DIAL TIMEOUT.....	55
10.3.	HOLD RECALL	55
10.4.	AUTO HOLD ON CALL SWITCH	56
10.5.	AUTO REDIAL	56
10.6.	SILENTLY FOLLOW REDIRECTION.....	57
10.7.	DIAL PLAN.....	57
10.7.1.	INTER-DIGIT TIMEOUT.....	57
10.7.2.	DIAL KEY	58
10.7.3.	LAN DIAL	58
10.7.4.	CALL COMMAND	59
10.7.4.1.	CALL RETURN.....	60
10.7.4.2.	ANONYMOUS CALL (CLIP & CLIR).....	60
10.8.	MESSAGE ALERT	61
10.9.	AUTO-ANSWER.....	61
10.10.	CODEC PREFERENCE	64
10.10.1.	ENABLE PERSONAL PREFERENCE.....	67
10.11.	COMFORT NOISE GENERATION	67
10.12.	REGISTRATION ON DEMAND	68
10.13.	MULTI-DOMAIN REGISTRATION	70
11.	VOICE VOLUME ADJUSTMENT	72
11.1.	RINGER	72
11.2.	HANDSET	72
11.3.	SPEAKER PHONE	72
11.4.	EAR PHONE.....	73
12.	SERVICE.....	74
12.1.	VOICE MAIL	74
12.1.1.	SET UP VOICE MAIL	75
12.1.2.	ACCESS VOICE MAIL	76
12.2.	INSTANT MESSAGING	76
12.3.	SYNCHRONIZE TIME	77
12.4.	AUTO PROVISION.....	79
12.5.	SOFT-SWITCH (PBX) FEATURE ACCESS	82
13.	NAT TRAVERSAL	85
13.1.	PUBLIC INTERNET CONFIGURATION.....	85
13.2.	LAN CONFIGURATION TO TRAVERSE NAT AND FIREWALL	86
13.2.1.	STATIC NAT ROUTE	86
13.2.2.	NAT TRAVERSAL BY STUN	89
APPENDIX A - AVAILABLE NTP SERVERS		91

APPENDIX B – TROUBLE SHOOTING 96

APPENDIX C – TONES..... 100

1. Overview

1.1. Features

- DHCP or PPPoE for host IP, gateway, network mask, DNS (optional, 2 DNS at most), TFTP server, NTP server, and TTL; all those settings could be static assigned as well.
- If provided by DHCP server, it could use DHCP to get NTP server.
- Ear-phone, speaker-phone for hand-free, handset and loud-speaker support (all are volume adjustable). Changeable ringing tone.
- Supports 2 concurrent calls.
- Multiple service domains for easy access to different ISPs (3 domains at most).
- Address book (up to 500 entries) and call history (10 most recently received calls, 10 most recently missed calls and 10 most recently dialed numbers).
- Call return, speed dials (20 numbers), redial, auto-redial, call-screening (20 numbers) and detail records of the latest three calls.
- Call forwarding: configurable forwarding number, unconditionally forward all calls, forward calls on busy and forward calls on no response (adjustable waiting time).
- Call processing includes: hold (music on hold), mute, Caller ID, Call waiting (alerting tone, LED indication and screen popup), call transfer (blind transfer, consultative transfer, semi-supervised transfer, and take-back), call forward, call reject, do not disturb (DND).
- Call preferences include: call waiting, auto-answer (server-side invoked, locally activated and selectively auto-answer), dial-timeout, adjustable hold recall timer, auto-hold on call switch, auto-redial criterion (stop-on-ringing or stop-on-connected), accept call diversion or not, inter-digit time-out, message alerting, and per call Calling Line Identification Restriction.
- 3-way local conferencing
- Message Waiting Indication (MWI)
- 180 locally generated ringing and 183 remotely generated call progress tone.
- Out-of-dialog instant messaging, and flashing SMS without user interaction.
- NAT & firewall support by STUN or pre-configured NAT Gateway port mapping; Auto-update or notify the change of NAT IP by STUN if NAT employs DHCP as well (such as xDSL dial-up).
- Symmetric RTP flow for cases where only one endpoint is behind a NAT.
- Voice activity detection to reduce network bandwidth consumption.
- Comfort noise generation and dynamic de-jitter buffer to deliver better voice quality.
- 8 Programmable DSS keys.
- One-touch dialing (Hot lines)
- Configurable dial-key (#, *, & or dedicated DSS key as **【Dial-key】**) to facilitate the inclusion of “#, *, or &” in dial strings.

- Regular alarm and one-time alarm.
- Prompt user on call diversion for better security support (Configurable)
- Menu driven configuration by keypad, Web browser or TELNET.
- Use of Simple Network Time Protocol (SNTP) to synchronize time with network time server and adjust to time-zone (configurable) and daylight saving time (configurable).
- Use of Trivial File Transport Protocol (TFTP) and HTTP for auto-provisioning and image update
- IEEE 802.1Q VLAN tagging support
- Support both 802.1P link layer precedence bits and IP layer type-of-service (ToS) bits for voice streaming, such that *IP SIP Phone* would perfectly replace your desktop analog phone within a switched network.
- SNMPv2 for network management and supervision.
- Call-related statistics, including total inbound/outbound calls, average conversation duration, connected ratio of the last 50 calls, and the conversation time distribution (less than 3 minutes, 3-20 minutes and longer than 20 minutes) during the last 72 hours, or since system startup.

1.2. Technical Specifications

1.2.1. Call Control Capability

- Fully complies with RFC 3261 (SIP) with RFC 2543 backward compatible.
- Fully complies with RFC 2327 (SDP) and RFC 3264 for capability negotiation based on SDP offer and answer model.
- Multiple Outbound Proxy, Registrar and Redirect server support, up to 3 different service domains.
- Support SIP server authentication procedure (HTTP digest authentication scheme)
- On-demand registration and re-registration on network configuration changed (auto-detect the changed IP of host, NAT, Dynamic DNS)
- Auto-locating SIP server (RFC 3263) by DNS NAPTR/SRV record (RFC 2782) lookup
- Supports SIP multicast registration to 224.0.1.75
- Call Transfer (RFC3515 for REFER method, RFC3420 for sipfrag support, RFC3891 for replaces header and RFC3892 for Referred-by header).
- Message waiting indication, MWI, (RFC 3842).
- Fully Implementation of RFC 2916 (E.164 and DNS) for ENUM translation by NAPTR (RFC 2915)
- Configurable SIP signaling port (default 5060), support both UDP and TCP.
- Support rport and received in VIA header (RFC3581) (Configurable)
- Redundancy SIP proxy server support by DNS NAPTR/SRV/AAAA records.

- Support REGISTER, INVITE, ACK, CANCEL, BYE, OPTION, REFER, MESSAGE, SUBSCRIBE, NOTIFY, INFO methods
- Support “alert-info” header for distinctive ring.

1.2.2. Real-time Voice Streaming

- Fully complies with RFC 1889 (RTP / RTCP), RFC 1890 (AVT profiles), RFC 3551 (RTP Profile for Audio and Video Conference with Minimal Control) and RFC 3555 (MIME Type Registration of RTP Payload Formats).
- Support both in-band DTMF mixed with RTP voice stream and out-of-band DTMF over RTP (RFC2833).
- Dynamic RTP de-jitter buffer and lost packets concealment management.
- Speech CODEC supports: G.711 (A-law and μ -law), G.723.1/G.723.1A (both 5.3 and 6.4 kbps), and G.729A/G.729AB. CODEC precedence is configurable to adjust to your network link speed.
- 3-way local conferencing
- Voice activity detection (VAD) and comfort noise generation (CNG).
- Voice and ringer volume control
- Real-time acoustic echo canceller.
- IP Type of Service (ToS) bits set for RTP/RTCP packet prioritization
- 802.1P precedence bits support to prioritize RTP voice frames within switched network.
- Configurable RTP / RTCP ports.

1.2.3. NAT and Firewall

- Support static NAT mapping (both NAT IP and SIP/RTP ports are configurable)
- Support Simple Traversal of UDP through NAT, (RFC 3489 STUN).
- Support auto-detect (auto-update) the change of NAT IP by STUN (in case the NAT has no static IP and employ dial-up to public internet).
- draft-ietf-mmusic-sdp4nat-03.txt (RTCP attribute in SDP)
- Symmetric RTP flow for the cases where only one endpoint behind NAT.
- Support STUN server redundancy by DNS SRV/AAAA records

1.2.4. Management

- TFTP and HTTP for Auto-Provision
- HTTP configuration by web browser
- Key-pad configuration
- TELNET configuration

- SNMPv2 for network management:
 - MIB2: RFC1213
 - Get and Set operation for internal state (Proprietary Enterprise MIB for system configuration access).
 - Trap:
 - ◆ System startup
 - ◆ System shutdown (by command/SNMP/Image upgrade)
 - ◆ SIP Registrar availability
 - ◆ Call-Channel Status.

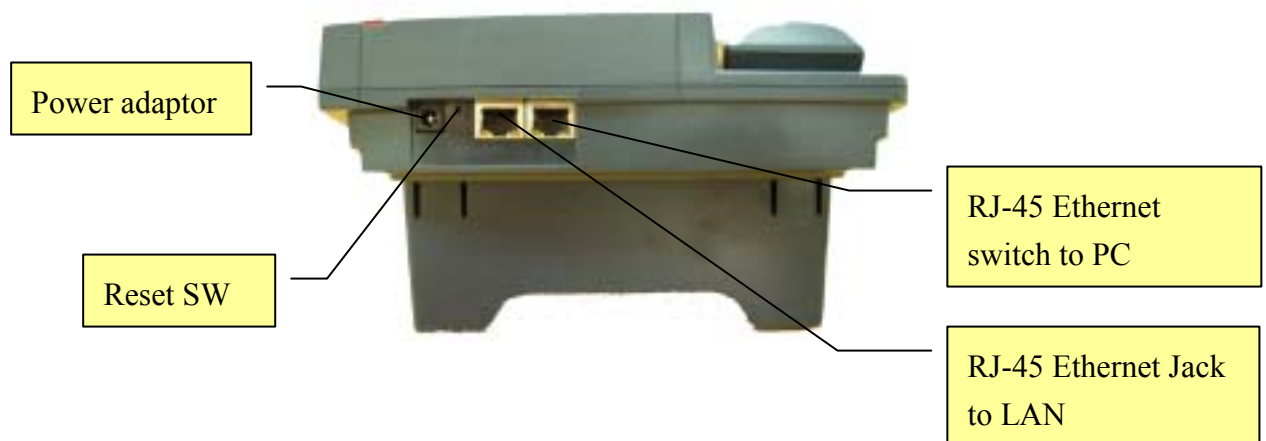
2. Layout

2.1. Hardware

2.1.1. Front View



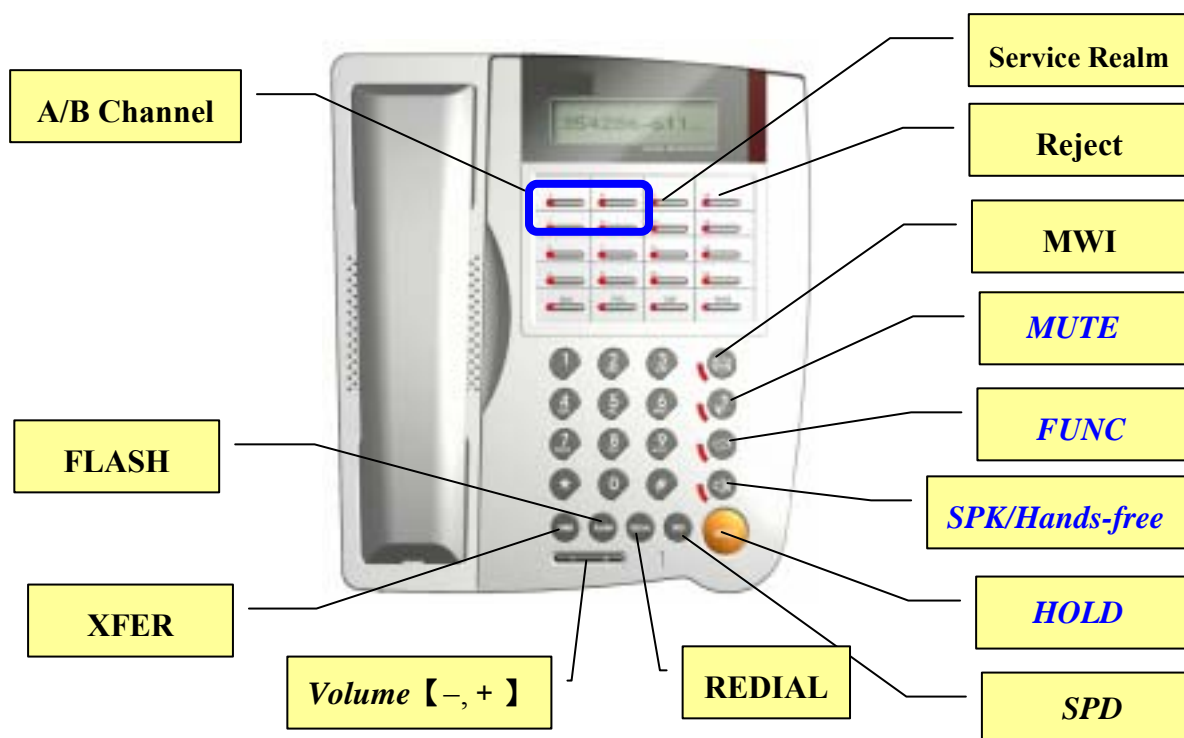
2.1.2. Rear View



2.1.3. Back View



2.2. Keys



【Keys】 : Function when in-call or idle / *Function on menu mode*

【A / B Channel】: Call lines (2 concurrent calls at most) / Review the calling information on this channel during conversation.

【Service Realm】 : Display the registration status of each active service domain on idle; switch target service domain (ISP) while making calls.

- 【Reject】** : Reject incoming waiting calls
【MWI】 : Message Waiting Indication, MWI: Access to voice mail system
【MUTE】 : Mute | *Delete character*
【FUNC】 : Menu | *Return to upper level submenu*
【SPK】 : Hands-free | *Exit menu*
【HOLD】 : Hold | *Confirm, Save*
【SPD】 : Speed Dial
【Redial】 : Redial the last dialed number.
【FLASH】 : Take back or cancel transferring calls; cancel conference.
【XFER】 : Transfer
【Volume】 :
【 + 】 : Volume UP (Ringer, headset, handset, speaker) | *Next item, Move cursor to right*
【 - 】 : Volume down (Ringer, headset, handset, speaker) | *Previous item, Move cursor to left*



【Registration】 : Re-register. The LED indicates the registration status of each active service domain:

Green LED On: Successfully register to all active service domains.

Red LED On: At least one service domain could not be registered.

Green LED Flashes: Registration is in progress. Note, IP SIP Phone will regularly refresh SIP Address-or-Record registration as necessary.

Red LED Flashes: No service domains have ever registered successfully.

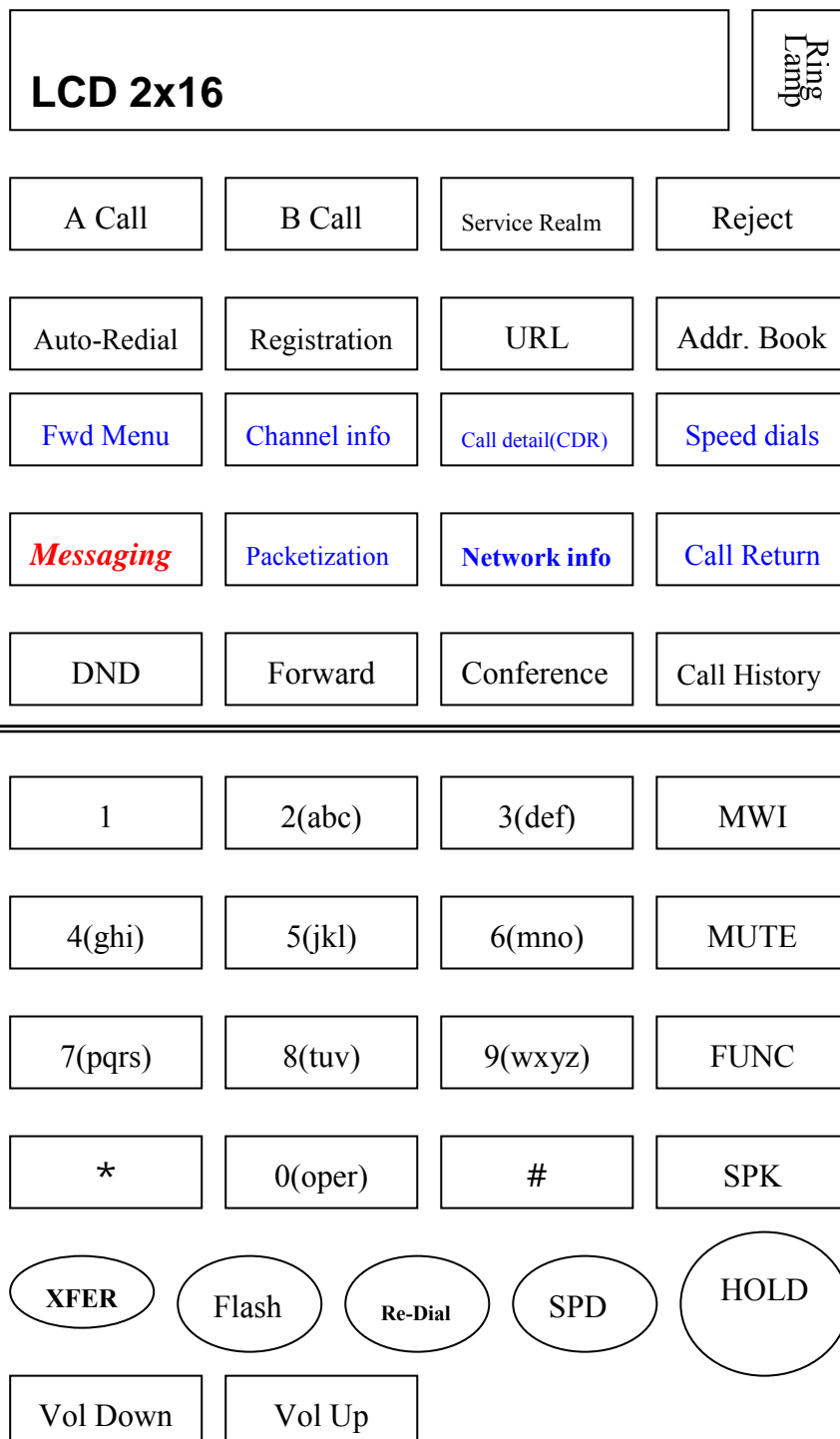
LED is off: Users explicitly logs out all SIP service and goes off-line 'till user presses the 【Registration】 key again to go on-line (re-register to all SIP service).

【Auto-redial】 : Auto-redial the last dialed number 'till connected (ringing)

【DND】 : Do Not Disturb (red LED indicates on)

- 【Forward】 : Forward incoming waiting calls
- 【URL】 : Use keypad to enter alphabets and numbers (red LED on).
- 【Address Book】 : Access to address book (search an entry or list all entries).
- 【Call History】 : Missed calls / Received calls / Dialed Number. If the red LED is on, it indicates there are unread records of missed calls.
- 【Conference】 : Three-way local conferencing
- 【F1-F8】 : User programmable DSS keys for easy access to various phone features. The default mappings of these function keys are:
 - 【F1】 : Forward menu – shortcut to activate incoming calls forwarding menu.
 - 【F2】 : Channel info–show information of the last call on each call channel 【A / B】 .
 - 【F3】 : Call Detail - show detail records of the latest three connected and *finished* calls.
 - 【F4】 : Speed dials – activate speed dial list.
 - 【F5】 : *Messaging; Out-of-dialog instant messaging.***
 - 【F6】 : Packetization - adjust the voice packetization based on your network link speed.
 - 【F7】 : Network Info - show the current active host IP, MAC address and DNS IP(s).
 - 【F8】 : Call Return - place a call to the last incoming call, either a missed or received one.

2.3. Keypad



Those keys in **blue** font (which are referred as DSS function keys hereafter) can be dynamically re-configured by user from menu-3.2 DSS Functions.

3. Operation

3.1. Key Definitions in Menu Mode



【HOLD】	<i>Enter</i> the selected menu item or confirm the modification.
【MUTE】	<i>Delete</i> the current character or the previous character if the cursor is positioned at the end.
【FUNC】	<i>Return</i> to upper level menu.
【SPK】	<i>Exit</i> the menu
【⇐】 / 【⇒】	Circle through the selected menu items and adjust volume.
【0 - 9】	Direct-selection / setting / jump-to-specific-item.
Cursor	<ol style="list-style-type: none"> 1. Insert mode only 2. Positioned on the currently setting value. 3. Positioned on the 1st menu item.

3.2. Enter Alphabets and Numbers

Circular input by pressing the same key

Circular Input by pressing the same key																																	
Key	Alphabet & Number																																
1	1																																
2	2->a->b->c->A->B->C																																
3	3->d->e->f->D->E->F																																
4	4-> g->h->i->G-> H->I																																
5	5->j->k->l->J->K->L																																
6	6->m->n->o->M->N->O																																
7	7->p->q->r->s->P->Q->R->S																																
8	8->t->u->v->T->U->V																																
9	9->w->x->y->z->W->X->Y->Z																																
0	0->[Space]																																
*	Punctuation Table: <table><tr><td>.</td><td>@</td><td>-</td><td>*</td><td>#</td><td>_</td><td>?</td><td>&</td><td>\$</td><td>/</td><td>\</td><td>,</td><td>:</td><td>;</td><td>+</td><td>(</td></tr><tr><td>)</td><td>'</td><td>!</td><td>“</td><td>¥</td><td>%</td><td><</td><td>></td><td> </td><td>§</td><td>=</td><td>%</td><td>{</td><td>}</td><td>Φ</td><td>±</td></tr></table>	.	@	-	*	#	_	?	&	\$	/	\	,	:	;	+	()	'	!	“	¥	%	<	>		§	=	%	{	}	Φ	±
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)	'	!	“	¥	%	<	>		§	=	%	{	}	Φ	±																		
#	#																																

3.3. Address-of-Record (SIP AoR)

The general form of SIP address-of-record is:

“Display” <protocol:email-like-address>;tag=param

- The “Display” field is optional. If present, it consists of any ASCII characters except for ‘<’ and ‘>’. If the “Display” is present, the following address must be enclosed in a paired ‘<’ and ‘>’.
- Protocol: Usually in lower case, such as “sip”, “tel” or “sips”. Note, “sip”, “tel” and “sips” protocol names MUST be specified in lower case, which is stipulated on RFC3261.
- email-like-address: in the form of “user-part@domain” where user-part is optional and the domain part could be either a dotted IP or a domain name record, such as:
 - 3200@SIP.isp.com
 - mike@192.168.192.100
 - 192.168.3.100 (Note, the user-part is optional in direct IP dialing mode)
 - +886-3-5639025
- “tag=param”: Multiple parameters could be present (separated by ‘;’)

Example	Note
Michael <sip:Michael@SIP.isp.com>	

Mike Jackson <sip:3200@SIP.isp.com>	
"Voice Mailbox" <sip:8888@vms.SIP.isp.com>	Display with enclosing ""
sip:300@SIP.isp.com	AoR without display
sip:192.168.3.100	AoR without user part (in dotted IP)
tel:+886-3-5639025	ENUM AoR
sip:+88635639025@SIP.isp.com;user=phone	ENUM AoR with SIP proxy support

4. Startup

Basically you have the following ways to configure your *IP SIP Phone*.

- Press **【FUNC】** + **【#】** to activate the configuration menu and configure it via keypad.
- Use any modern web browser to configure the phone from a PC. The default login password for both privileged and user-level password is "0000".
- TELNET into the phone by any TELNET client. The default TELNET port is TCP port 23, login password is as the same as your phone password set in menu-3.1 on "*IP SIP Phone v2 Keypad-TELNET Administration*" (the default phone password is "0000") and the max concurrency is 4. IP-Range will not apply the changes until user presses [Ctrl] + 's' to apply the modifications or the client disconnected by [Ctrl] + 'c'.
- Auto-provision on phone startup. Please refer to "Menu-7.4 Auto-provision." on "*IP SIP Phone v2 Keypad- TELNET Administration*".

Note, before you can configure your phone-set from your PC, such as by a TELNET client or point your browser to the phone-set, you must have configured its IP via keypad properly.

4.1. Prerequisite

Initially, your phone can only be configured via keypad since it bears no valid IP yet. After finishing configuring your network, you could use either a web browser (HTTP port 80) or a TELNET client (TCP port 23) if you have a small number phone-sets to configure.

However, we recommend you to use TFTP for auto-provision if you have to administer large amounts of phone-sets. For TFTP provision, please refer to [12.4 Auto Provisioning](#) on this document.

4.1.1. Network

To configure your network:

- Press **【FUNC】** + **【#】**
- Go to **【6.Network】** \ **【1.General】**

Please configure the phone based on your network configuration: DHCP, static IP or

PPPoE.

4.1.1.1. DHCP

- Pick **【1.Mode】 \ 【1.DHCP】**
- Disable **【4.Use Static DNS】** by choosing **【2.DHCP】**

Note: if you want to assign a different domain name server instead of using those obtained by DHCP, you should choose **【1.Static DNS】** and set the IP of your specific DNS into **【5.Static DNS】**, such as “1.Primary DNS” = 192.168.3.254 (*modify this as necessary*)

The supported DHCP options are:

- Client PC address
- DHCP option 1—Client Subnet Mask
- DHCP option 3—Gateway IP on the client's subnet
- DHCP option 6—One or two Domain Name servers
- DHCP option 15—Domain name
- DHCP option 42—Network Time Protocol servers
- DHCP option 66 (TFTP server name)

Note: To make DHCP option 6 take effect, you must disable **【MENU】 => 【6.Network】 / 【1.General】 / 【4.Use static DNS】 by picking **【2.DHCP】****

Note: If DHCP option 42 is present, it will overwrite the SNTP server in menu-7.3.2 **【Server IP】**

Note: DHCP option 66 will overwrite the Auto-provision server in Menu-7.4.2 **【TFTP server】**

4.1.1.2. Static IP (Fixed IP)

- Pick **【1.Mode】 \ 【2.Static assign】**
- Go to **【2.Static Settings】**, and enter your network configurations based on your ISP.

For example:

1. Host IP = 210.201.210.132 (*modify this as necessary*)
 2. Network mask = 255.255.255.0 (*modify this as necessary*)
 3. Gateway IP = 210.201.210.128 (*modify this as necessary*)
- Enable **【4.Use Static DNS】** by choosing **【1.Static DNS】**
 - Assign **【5.Static DNS】**, such as: **【1.Primary DNS】** = “168.95.1.1” (*modify this as necessary*)

4.1.1.3. PPPoE

- Pick **【1.Mode】 \ 【3.PPPoE】**
- Go to **【3.PPPoE settings】** and enter your PPPoE authentication information, such as:
 1. Login ID = MyPPPoEAccount (*modify this as necessary*)
 2. Password = PPPoEDialupPassword (*modify this as necessary*)

3. Service Name = Optional, some ISP requires it (*modify this as necessary*).

4.1.1.4. Verify Network Configuration

Press **【F7】** (which default is a shortcut to menu **【8.Advanced】 \ 【4.System Status】 \ 【1.Network】**) to check current active network settings:

1	9	2	.	1	6	8	.	2	1	0	.	1	1	3	
MA	C	:													
1	9	2	.	1	6	8	.	2	1	0	.	1			

It will display the host IP, Ethernet MAC address and the active DNS IP (secondary DNS IP will be shown if available) in order.

Once finishing network configuration, you should be able to place a point-to-point call. For example, if your phone IP is “192.168.1.10” and you want to dial another SIP phone which IP is “192.168.1.20”, please dial “*20*5060” (or just “*20” if the target phone listens on UDP port 5060; otherwise you must dial the target UDP port as well). This is “LAN dialing” (Refer to section 8.1-“Dialing Scheme” on this document). If the call could be set up correctly, then your network configuration is fine; otherwise, please refer to B-1 on Appendix B-“Trouble Shooting”.

Note, if you reside on a LAN without gateway, you should specify the gateway IP as “0.0.0.0” rather than assigning a non-existent or an invalid 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.

4.1.2. SIP Service

Before you start, you should ensure you have SIP-related data from your ISP. For example, if you get the following information from your SIP ISP:

- i. Account: Michael
- ii. Password: secret
- iii. SIP address-of-record: 8888@isp.com
- iv. SIP Proxy / Registrar Server: sip.isp.com, which serves on UDP port 5060

Then you could configure your *IP SIP Phone* by a web browser as below:

Activate

Authentication

Login User ID: Michael

Login Password: ••••

☐ Ignore qop Parameter in Digest Authentication

SIP Address-of-Record

Alias/Nickname/Display: Michael Wu

Protocol: sip

Username/Phone-No: 8888@ISP.com

☐ Auto Answer

Proxy Server

Proxy Server: Default Outbound Proxy

Proxy FQDN: sip.SIP.com

Transport: UDP

UDP Port: 5060

TCP Port: 5060

Registrar Server

Registrar Server: Same as Proxy

Registrar FQDN:

Registrar Transport: UDP

Registrar UDP Port: 5060

Registrar TCP Port: 5060

Modify

Alternatively, you may to 『Main Menu』=>”6.Network” / ”2.SIP settings” / ”1.1st Realm” to configure these information by keypad.

- i. If you have applied for more than one service domains, please repeat step I and II until all active domains are properly configured. *IP SIP Phone* supports three different service domains at most.
- ii. After saving the configuration, the system will try to register to those activated SIP domains. The flashing green LED of 【Registration】 key indicates that the registration is undergoing. Once the green LED stops flashing, you could know the registration result by the LED. Please refer to section-10.11 “Registration on Demand” and section-10.12 “Multi-domain Registration” on this user’s guide for detail.
- iii. If you failed to register your phone to SIP registrar, please refer to Appendix B-“Trouble Shooting” on this document.

4.1.3. Configure NAT and Firewall

If your SIP server locates on public internet, whereas your phone resides on a local area

network, please refer to chapter 13-“NAT Traversal” on this user's guide if your phone-set is behind network address translator (NAT) and / or firewall.

4.2. Initialization

(a) Startup:

	I	P	S	I	P	P	h	o	n	e	
			V	e	r	s	i	o	n		

(b) Check for auto-provision. Please refer to section [12.4 Auto Provisioning](#) on this document for detail

i. Auto-provision is on but no TFTP server specified:

A	u	t	o	-	p	r	o	v	i	s	i	o	n	
N	o		T	F	T	P		s	e	r	v	e	r	!

ii. Download configuration files from TFTP server

T	F	T	P		s	e	r	v	e	r	n	a	m	e
i	p	r	a	n	g	e	r	.	c	f	g	.	.	.

iii. Download failed:

T	F	T	P		i	p	r	a	n	g	e	r	.	c	F
T	i	m	e	d		o	u	t							

iv. Download successfully and apply changes:

A	u	t	o	-	p	r	o	v	i	s	i	o	n	
i	p	r	a	n	g	e	r	.	c	f	G	.	.	.

v. If auto-provision does not apply any changes to current settings:

A	u	t	o	-	p	r	o	v	i	s	i	o	n		
U	s	e		R	O	M		s	e	t	t	i	n	g	s

(c) Check Date and Time:

C	u	r	R	e	n	t		d	a	t	e	:			
							M	/	D	D	/	Y	Y	Y	Y

C	u	r	R	e	n	t		t	i	m	e	:			
							h	:	m	m	:	s	s		

Note:

- ✧ IP SIP Phone will go the Idle ready Mode after 5 seconds if user dose not enter any digit. The default system time on start-up is January 1, 1970, 00:00, GMT.

- ✧ User entry any digit for Time & Date. It must press **【HOLD】** Key to be Idle Ready Mode Display.
- ✧ The phone will synchronize its time by Simple Network Time Protocol, SNTP, with network time server regularly if SNTP is enabled. If you want to keep the time you manually set previously, you must disable SNTP. Please refer to section-4.4.1 “Date/Time” on “*IP SIP Phone v2 Web Administration*” for detail.
- ✧ You can just ignore the date & time settings on boot, leaving the phone to synchronize its clock with network time server. Please check menu-3.3.4: “Time zone” (by TELNET or keypad) to adjust your time zone otherwise the synchronized time may be several hours late (earlier) than your local time. Please refer to section-4.4.1 “Date/Time” on “*IP SIP Phone v2 Web Administration*” for detail.

4.3. Registration

(a) Registering

R	e	g	t	o	r	e	g	i	s	t	r	a	r	
.	s	i	p	.	f	c	i	.	c	o	m	.	t	w

The registered on-line timeout is 3600 seconds (1 hour).

(b) Registration Done

R	e	g	i	s	t	e	r	e	d					
E	x	p	i	r	e	s		i	n	3	6	0	0	s

System will refresh registration after 90% of the expiration interval elapsed.

(c) Registration Failed

R	e	g		e	r	r	:	h	o	s	t		u	n	r
r	a	c		h	a	b		l	e						

This message will freeze the screen for a while, such as 5 seconds, for the user to figure out the reason. Failed registration will shorten the re-registration interval to 90 seconds.

5. Shutdown

5.1. Unregistration

U	n	r	e	g	i	s	t	e	r	i	n	g	.	.	.

6. Idle

6.1. Registered

F	r	i		M	M	/	D	D		h	h	:	m	m
M	i	c	h	a	e	l		W	u					

6.2. Not registered yet or registration expires

F	r	i		M	M	/	D	D		h	h	:	m	m
M	i	c	h	a	e	l		W	u					

The registration LED will flash to indicate that the phone set has not been registered to any service domain. That is, the phone set could only be reached via “Intra-domain Dialing”, “IP Dialing” or “LAN Dialing” methods; please refer to section 8.1- “Dial Scheme” on this user's guide for detail.

6.3. Regular Registration

1. Registering

Reg	to	regist	rar
.sip	.fci	.com	.tw

2. Registration Done

Registered																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																																					
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3. Registration Failed

R	e	g		e	r	r	:	h	o	s	t		u	n	r
r	a	c	h	a	b	l	e								

This error message will freeze the screen for a while on idle state, such as 5 seconds, for the user to figure out the reason. Failed registration will shorten the re-registration interval to 90 seconds. You could perform registration on demand by pressing **【Registration】** to force an immediate registration.

In menu mode, you cannot receive calls. If there is incoming calls when you are configuring your phone, the call will be silently rejected with a 486 busy response.

7.1. Ringing

(a) On Idle State

From	mm:ss
Scott Sun	

1. Pop up the caller ID for 7 seconds. It may show the display name if the caller ID could be found on the address book. The “mm:ss” keeps track of the time elapsed after the call arrived.

From	mm:ss
Scott	Sun

2. The channel LED of this incoming waiting call, 【A / B / C Call】, will flash fast.
3. 【Forward】 and 【Reject】 LEDs will be on for the user to handle this incoming call.
4. Call waiting alarm will alert the user twice in 7 seconds.

Note: If the caller ID is too long to accommodate into a line or you want to review the caller ID after the screen pop-up disappeared, you can review the call information on each channel by pressing **【F2】**, which is mapped to “Channel info” by default.

A. Caller ID (AoR)
B. Caller ID (AoR)

Those records (with their caller IDs, AoR, shown) are volatile in memory such that they will be clean up every time the system reboots. If you review them on idle mode, you may press **【Redial】** to dial a specific number. Each record keeps the following call-related information:

1.	C A l l e r I D (A o R)
2.	From (C O n t a c t)
3.	C O D E C
4.	U s e r A g e n t
5.	M e d i a S e s s i o n

- | | | | | |
|----|--------|-----|--|--|
| 1. | Local | RTP | | |
| 2. | Remote | RTP | | |

1	9	2	.	1	6	8	.	3	.	1	:	6	2	2	6
2	/	6	2	2	6	3									

7.2. Reject Call

7.3. Forward Call

This may also be a network feature, handled from a network configuration by the system administrator.

7.4. Answer Call

[illegible]

Scott Sun

The “mm:ss” keeps track of the time elapsed after answered.

(A) On Idle State:

- a. Press **【A / B / C Call】** to answer the call
- b. Pick up the handset or turn on the speaker phone to answer it. Note: if you turn the ear-phone on, the voice will be output to ear-phone and the speaker LED will flash to notify that ear-phone is active.

(B) Other calls are in progress

- Press **【A / B / C Call】** to answer the call
- The **【Reject】** and **【Forward】** LEDs will be on such that you may reject the incoming waiting call as busy by pressing **【Reject】** or forward it to a predefined target number by pressing **【Forward】**.

7.5. Connected

mm:ss
Scott Sun DTMF

7.6. Disconnected

(a) Peer hangs up

Released mm:ss
Scott Sun

Where “mm:ss” indicates the duration of the call. If the user is on speaker phone mode, it will return to IDLE state in 5 seconds.

(b) User hands up

1. Ringing calls are waiting => Back to Ringing State
2. Answered calls are waiting:

The “Holding calls recall” will be triggered on expiry to alarm the user that there are still some holding calls. User could pick up the holding call by hooking up again; otherwise the holding call will be disconnected after ringing for 1 minute.

After disconnected, you may review the call detail record (CDR) by pressing **【F3】**, which is mapped to “Call detail” by default. “Call detail” keeps track of CDR of the latest three 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.	C	A	I	I	D	(A	o	R)
2.	C	A	I	I	D	(A	o	R)
3.	C	A	I	I	D	(A	o	R)

You may press **【Redial】** to dial the target's SIP address-of-record.

Each record keeps the following call-related information:

1.	C	A	I	I	D	(A	o	R)
2.	T	i	m	e						
3.	D	u	r	a	t	i	O	n		
4.	F	r	o	m	(C	O	n	t	a
5.	C	O	D	E	C					
6.	U	s	e	r		a	g	e	n	t
7.	M	e	d	i	a		T	r	a	f

1. Caller ID: address-of-record of the peer (press **【Redial】** to dial this number).
2. Time of call
3. Duration: call active time.
4. From: Contact IP of the peer for SIP signaling (press **【Redial】** to dial this number).
5. CODEC: CODEC employed for the call.
6. User agent: The phone tool used by the peer for this call.
7. Media traffic:

The media-related information will be available only when the call lasted for more than 20 seconds. But the information about RTP session such as local and remote IP / port for RTP session is always available.

1.	R	T	P	s	e	s	s	i	o	n
2.	P	a	c	k	e	t	s			
3.	B	y	t	e	s					
4.	P	a	c	k	e	t				
5.	D	e	l	a	y					

7-1 RTP session:

1.	L	o	c	a	l	R	T	P		
2.	R	e	m	o	t	e	R	T	P	

Display the RTP IP and port pair of CPE for RTP media stream.

1	9	2		1	6	8		3	1	:	6	2	2	6
2	/	6	2	2	6	3								

The format comes in the format “xxx.xxx.xxx.xxx:RTP/RTCP”, which specifies the IP of CPE and the UDP port used for RTP and RTCP session.

7-2 Packets: Received / Sent RTP packets

S	e	n	t	:	1	,	2	3	4	,	5	6	7		
R	c	v	d	:	1	,	2	3	4	,	5	6	7		

7-3 Bytes: Received / Sent bytes, including IP / UDP / RTP header and payload

S	e	n	t	:	1	,	2	3	4	,	5	6	7	K	B
R	c	v	d	:	1	,	2	3	4	,	5	6	7	K	B

7-4 Packet lost: Received / Sent RTP packet lost ratio.

S	e	n	t	:	1	.	2	1	%						
R	c	v	d	:	9	.	3	%							

7-5 Delay: One-way / Round-trip delay

O	n	e	-	w	a	y	:	1	.	2	3	4	m	S	
R	o	u	n	d	-	t	r	i	p	:	N	/	A		

7.7. Forward and DND

You can 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 presses **【Forward】** key on an incoming waiting call.

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

7.7.1. Do Not Disturb (DND)

You can block incoming calls by configuring the Do Not Disturb feature. Blocked calls are logged in the Missed Calls.

Press **【DND】** to toggle Do Not Disturb feature.

7.7.2. Call Forward

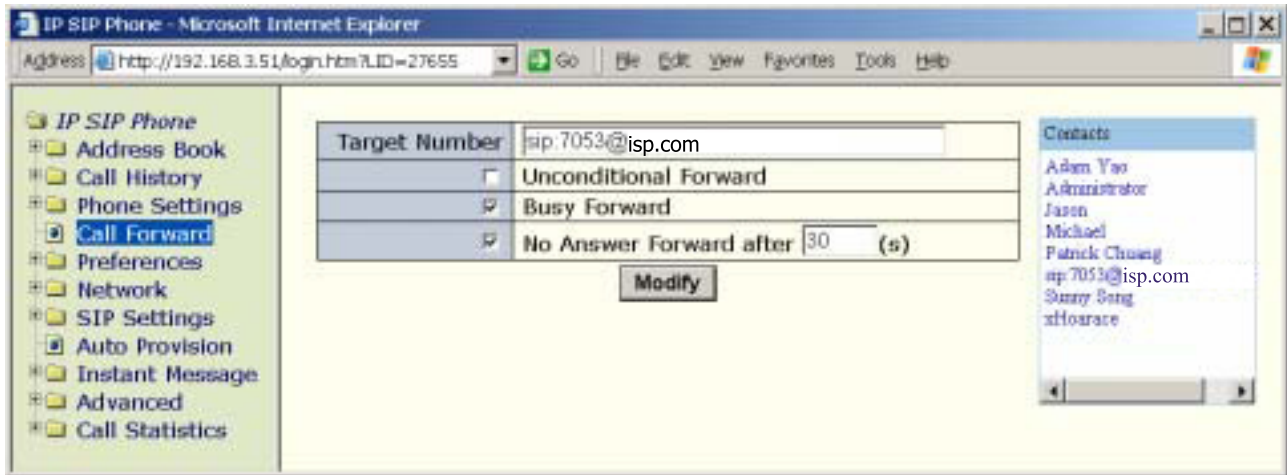
To configure the target number to forward to while this phone is busy or not answered within a predefined guarding interval:

- By keypad or TELNET:
 - Press **【FUNC】** + **【#】** to activate menu.
 - Go to submenu “4. Call forward” / “1. Target number”.
 - Press **【HOLD】** to pick an entry from address book. . Use **【←】** and **【→】** to view the current number.

M	i	c	h	a	e	l	<	s	i	p	:	8	8	8	8
@	Q	u	a	n	t	i	e	r	T	e	c	h	.	c	o

- Press **【MUTE】** to remove the current mapping of forward number.

- By web browser:
 - Go to 『Call Forward』 page, then click the “Contacts” on the right panel to pick an entry from the address book to set it as “Target Number”.
 - Delete the number in the text input to remove it.



Note: *by default, 【F1】 is mapped to the “Call Forward” menu.*

This target forwarding number is also employed while the phone is engaged in Do Not Disturb (DND) mode or while the user presses 【Forward】 key on an incoming waiting call.

7.7.2.1. All Calls Forward

You can configure to unconditionally forward all incoming calls by enable the All Calls Forward feature from menu 4-2 “All Calls Fwd” or on 『Call Forward』 web page. Forwarded calls are logged in the Missed Calls, menu 2-1. If this feature is enabled, the corresponding 【Forward】 LED will blink slowly to remind user. Default is disabled.

7.7.2.2. Busy Forward

You can configure to forward incoming waiting calls when the system is busy, on which time all lines are occupied, from menu 4-3 “Busy Forward” or on 『Call Forward』 web page. Forwarded calls are logged in the Missed Calls, menu 2-1. Default is disabled.

7.7.2.3. No Answer Forward

You can configure to forward incoming waiting calls after ringing for a predefined interval from menu 4-4 “No answer fwd” or on 『Call Forward』 web page. Forwarded calls are logged in the Missed Calls, menu 2-1.

7.7.3. Forwarding Rules

Priority	Conditions	Results
----------	------------	---------

All Calls Forward	【DND】 is on ¹	Forwarding number is available	Forward the incoming call to the target forwarding number.
		Forwarding number is unavailable	1. Reply as 480 Temporarily unavailable. 2. Recorded as a missed call
	Both 1. All calls forward feature is on 2. Forwarding number is available		Forward the incoming call to the target forwarding number.
	Others		Normal incoming call
Busy Forward	System call capacity is full loaded	Both 1. Busy forward feature is on 2. The forwarding number is available.	Forward the incoming call to the target forwarding number.
		others	1. Reply as busy. 2. Record as a missed call.
	Press 【Forward】 on a ringing call.	Forwarding number is available	Forward the incoming call to the target forwarding number.
		Forwarding number is unavailable	1. Reply as busy 2. Record as a received call.
No Answer Forward	Both 1. No answer forward feature is on 2. No-answer timer expires.	Forwarding number is available	Forward the incoming call to the target forwarding number.
		Forwarding number is unavailable	1. Reply as no-answer. 2. Recorded as a missed call.

Note: Those not-forwarded calls should be recorded as “Missed Calls” and retrievable at a later time to review. The call history LED should turn on to inform users and turn off after the user has reviewed them.

Note: Whenever there are un-reviewed missed calls, the Call history LED will be on. Moreover, it will directly enter into Missed calls menu while the call history LED is on and go into Call history menu if the LED is off.

8. Make Calls

If system is busy or there is no enough resource available, such as no suitable codec or no call manager available, to make a call, it will play “Network congestion tone” and show the following prompt.

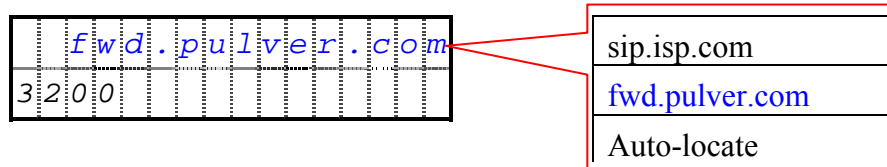
S Y S T E M B U S Y

¹ 【DND】 , Do Not Disturb, will turn on the corresponding LED.

T r y a g a I n l a t t e r

8.1. Dial Scheme

- A. Press **【URL】** to activate alphabets typing via keypad on which time the corresponding LED will be on, or type numbers directly via DTMF. Press **【MUTE】** to delete typos.
- B. The serving domain will appear on the upper-right corner. Press [Domain Switch] to circle the available service domains. With *IP SIP Phone*, you could configure up to 3 different service domains and the default service domain of line-N is service-domain-N. For example, the default service domain of **【B Call】** is the 2nd service domain. If no corresponding service domain is available, it will use the 1st service domain by default.



- C. Press **【#】** to dial the number immediately, or the phone will dial out the number after 4 seconds (the inter-digit timed-out is programmable).

Method	Rule	Example
Pick from address book	<ol style="list-style-type: none"> 1. Enter address book. 2. Search for entry. 3. Press 【Redial】 <p>Note: The phone will not use the domain you specified if the domain part of the dialing AoR is different from the one you specified and “Auto-locate” as picked domain, based on the domain part in AoR.</p>	<ol style="list-style-type: none"> 1. sip:albert@SIP.isp.com 2. <sip:felix@SIP.isp.com:3000> 3. Scott<sip:3100@SIP.isp.com> 4. felix@SIP.isp.com, which is equal to sip:felix@SIP.isp.com 5. tel:+886-3-5639025 <p>+88635639025, which is equal to “tel:+886-3-5639025”</p>
ENUM dialing ²	<ol style="list-style-type: none"> 1. Starts with a 【#】, which will be translated as ‘+’ (if you enter via keypad). 2. Dial numbers including country code and area code (Use 【MUTE】 to delete previous character on typos). 3. Dial 【#】 to confirm. 	<p>To dial “tel:+886-3-5639025”, please enter “#88635639025#” and it will show on the LCD as:</p> <p>The LCD display shows the word 'Call' in the top row and the number '+88635639025' in the bottom row.</p>

² ENUM capable DNS must be configured and set as the primary DNS of IP SIP Phone.

³ If the caller does not use default outbound proxy but the callee does, then the conversation may not be hanged up properly if the callee hangs up first (but the other direction could). This is due to the nature of stateful proxy to reply a “Call/Transaction doesn’t exist” response for those calls whose INVITE messages not flowing through the default outbound proxy.

$[33/100]$

	<i>numbers should send “as is” by disabling this feature from menu “5.Preferences” -> “7.Dial plan” -> “3.LAN dial”.</i>	
--	--	--

- *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), IP SIP Phone always listens on UDP-5060 for SIP signaling in addition to the user configured SIP service port. However, if UDP-5060 is overlapped with RTP ports for media session, it will not listen on UDP-5060 for SIP signaling to avoid conflict. This help when both parties are IP SIP Phone and reside on the same LAN.*
- The **【#】** is used to denote end of dialing phase only while the user performs a dialing via keypad. If the number is picked from the address book or call history, the **【#】** will not be translated. As a result, if you dialing “#8863” via key pad, it will be translated as “+8863”, but if you redial an entry from address book, which number is “#8863”(where the **【#】** is entered as an punctuation), it will be dialed as is (no translation will be done). If your intention is to dial “+8863” on address book, please enter ‘+’ via punctuation table rather than a **【#】** when you add such entry into your address book.
- **Multi-domain note:**
 - A. If the dialed AoR has no user part, it will use auto-locating (ignore the user specified domain) to facilitate IP-dialing.
 - B. If the dialed AoR has no domain part (such as intra-domain dialing or the number in address book specify no domain), it will use the domain you specified. Besides, if you pick “Auto-locate”, then the default domain of the current channel (A / B / C call) will be used.
 - C. If the dialed AoR has both user part and domain part, and you don’t specify “Auto-locate”, then:
 - i. If the domain part of dialing AoR could match one of those registered domains, it will use the matched domain instead.
 - ii. If the domain part of dialing AoR matches none of those registered domains
 1. If the domain part is in dotted IP format, it will use the user specified domain.
 2. Otherwise, use “Auto-locate”.
 - D. For all others, it will use the domain you specified while making calls.

8.1.1. Guarding Time

- User hooks up already:
 - ◆ The first digit timeout is 15 seconds and it will play network fail tone on expiry. Besides, the default inter-digit timeout is 4 seconds and it will dial out the collected

digits on expiry as well. To speed up the dialing process, press **【#】** key whenever finishing dialing.

- User does not hook up yet:
 - ◆ The default inter-digit timeout is 4 seconds and the phone will dial out the collected digits either on timer expiry or user presses the pound, **【#】**, key.
- If the call is made by speaker phone, it will return to IDLE state on call finish.

The inter-digit timeout is configurable:

- To change it by TELNET or keypad, go to **【MENU】** => “5. Preferences” / “7. Dial Plan” / “1. Inter-Digit”.
 - To configure it by web browser, go to 『Preferences』 => “Inter-Digit Timeout (s)”
- Please refer to section-10.7.1 “Inter-Digit Timeout” on this document for detail.

To speed up the dialing process, press **【#】** key whenever finishing dialing.

8.1.2. ENUM Sample

1. Configure your *IP SIP Phone* to use a NAPTR capable DNS server. If you use auto-locating SIP server instead of default outbound proxy for next hop delivery, your DNS must have SRV and A records like the following (please change those host and IP in red font accordingly):

```
$ORIGIN SIP.isp.com
;; Pref Weight Port Target
_sip_udp.SIP.isp.com. IN SRV 0 0 1000 proxy.sip.SIP.isp.com.
_sip_tcp.SIP.isp.com IN SRV 10 0 1000 proxy.sip.SIP.isp.com.
proxy.sip.SIP.isp.com IN A 192.168.192.99.
```

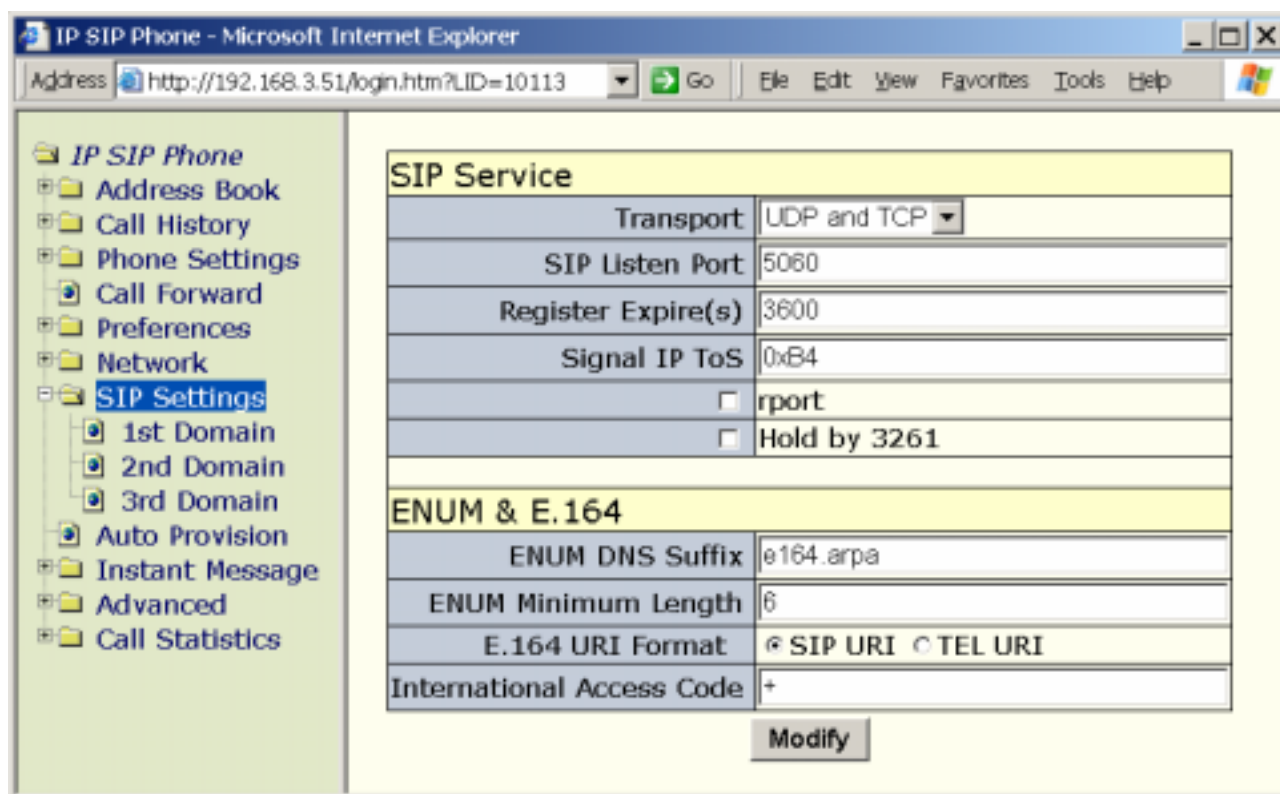
2. Suppose you want to register your mobile number, +886-939342017 as an ENUM number associated with your SIP AoR, sip:Michael@SIP.isp.com. Moreover, we assume that you configure your SIP phone to use the standard E.164 suffix for ENUM, “e164.arpa”, then you should add a NAPTR record under the domain “7.1.0.2.4.3.9.3.9.6.8.8.e164.arpa” (which is the reverse of your mobile phone number appended with “e164.arpa” suffix) like the following:

```
$ORIGIN 7.1.0.2.4.3.9.3.9.6.8.8.e164.arpa
;; order pref flags service regexp replacement
IN NAPTR 100 10 "u" "E2U+sip" " !^.*$!sip:Michael@SIP.isp.com!" .
```

Note: *IP SIP Phone* accepts both “sip+E2U” (RFC2916) and “E2U+sip” (draft-ietf-sipping-e164-04.txt) as NAPTR service strings.

3. The E.164 suffix for ENUM is configurable from menu-6.2.3 “ENUM & E.164”.

Alternatively, you may go to 『IP SIP Phone』 / 『SIP Settings』 page by web browser and configure “ENUM & E.164”.



Below illustrates how to configure them by keypad (TELNET).

3.1. ENUM suffix for DNS query:

E	N	U	M	D	N	S	s	u	f	f	i	x	:
e	1	6	4	.	a	r	P	a					

Set the ENUM suffix for E.164 query on DNS. Leaving it blank or setting it as “e164.arpa” to comply with RFC 2916 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 “+886-3-5639025”, 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: “5.2.0.9.3.6.5.3.6.8.8.e164.arpa”.

3.2. Minimum Valid ENUM Digit Length:

M	i	n		v	a	l	i	d		l	e	n	g	t	h
[1	-	1	6]	d	i	g	i	t	s	:			6

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 this rule failed does not comply, the dial string will be dial "as is" without any attempt to carry out an ENUM resolution. Default is 6 digits.

- 3.3. If *IP SIP Phone* fails to resolve an ENUM number to SIP address-of-record, AoR, it will render the request line based on the URI format you specified in menu-6.2.3.3 “URI Format”.

1.	SIP	URI							
2.	TEL	URI							

- 3.3.1. SIP URI: *IP SIP 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 “+88635639025” and *IP SIP Phone* fails in ENUM resolution, it will send:

INVITE sip:+88635639025@proxy.SIP.isp.com;user=phone

- 3.3.2. TEL URI: *IP SIP Phone* will send request to the specified SIP service domain proxy in the form:

INVTE tel: +PhoneNumber

For example, if you dial “+88635639025” and *IP SIP Phone* fails in ENUM resolution, it will send: ***INVITE tel:+88635639025***

- 3.3.3. In either format, you could also configure whether the leading ‘+’ in request line should be replaced or not by configure menu-6.2.3.4 “Int’l access”.

[illegible]

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.

8.2. Redial

Press **【Redial】** to dial the last dialed number. It will turn on the speaker phone automatically if the user has not done so yet.

8.3. Address Book

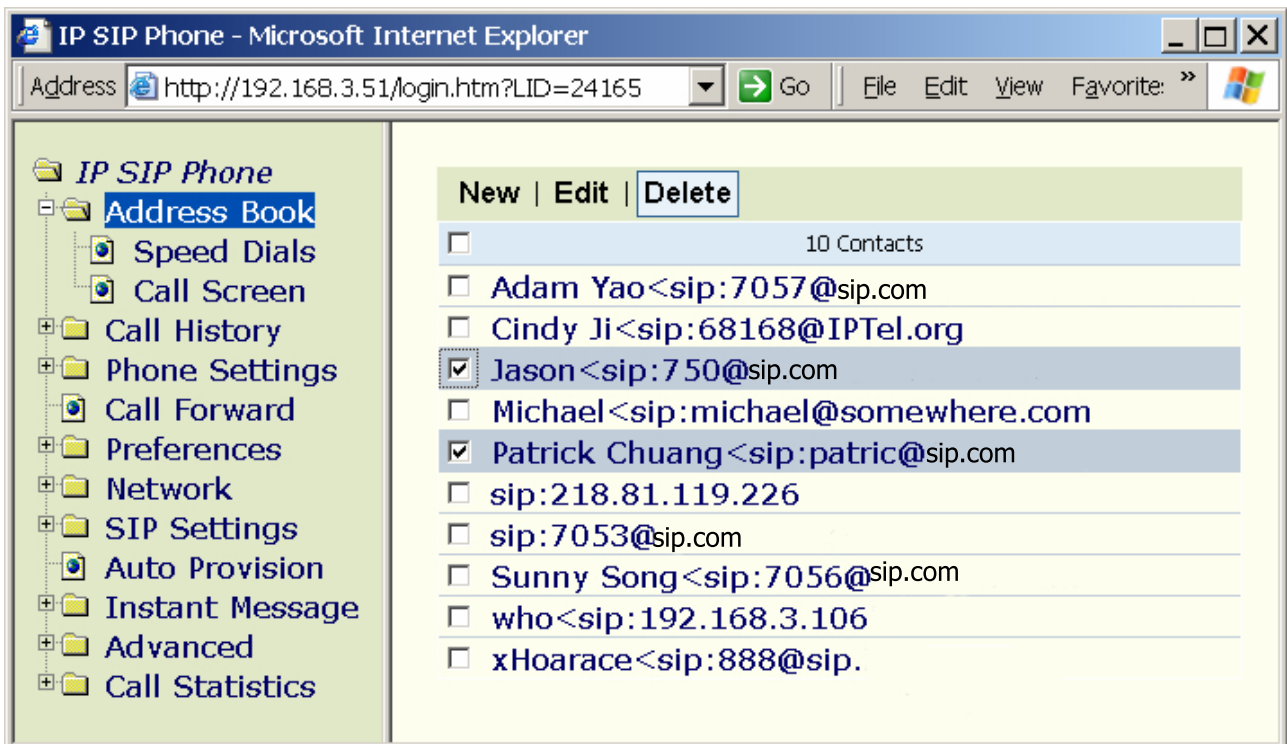
You can pick an entry from the address book, press **【Redial】** to dial out the specified number.

- 『Main Menu』 => 『1.Address book』 \ 『1.Search』 (Or press DSS key **【Address Book】** to activate it).
- Enter the prefix of the target number (display) to search or press **【HOLD】** with

any input to go to the first entry on address book.

- On listing mode, press **【2】** twice to first entry prefixed with an 'A' (or press **【8】** consecutively for 3 times will jump to the first entry prefixed with a 'U'. etc.)
- Press **【Redial】** to dial the selected number.

You may configure your address book by pointing web browser to your terminal's IP and go to page **『IP SIP Phone』 / 『Address Book』**.



Please refer to section-4.2 “Address Book” on “*IP SIP Phone v2 Web Administration*” for detail.

8.4. Call History

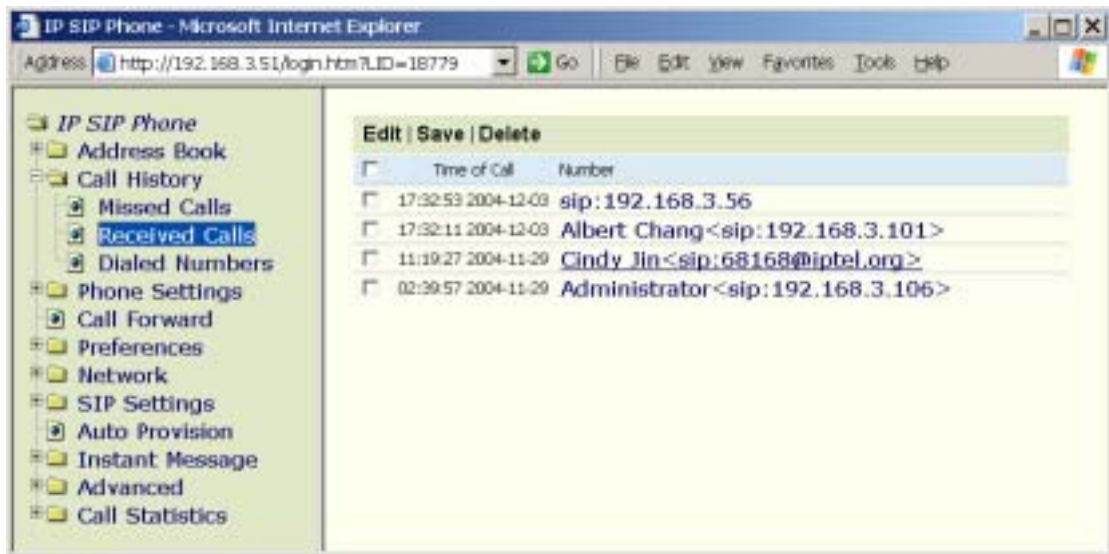
You can pick an entry from the call history, either missed calls, received calls or dialed numbers, press **【Redial】** to dial out the specified number on list mode. Those calls are listed on a last come first shown manner.

- **『Main Menu』 => 『2.Call History』** (or press **【Call History】** button directly)
- Choose the desired call list, such as “Missed Calls”.
- Traverse the list, and locate the target number.
- Press **【Redial】** to dial the selected number.

Note: if you press **【Call History】** while its red LED is flashing (indicating there are unviewed missed calls), it will enter the “Missed calls” list directly.

You may configure your address book by pointing web browser to your terminal's IP and

go to page 『IP SIP Phone』 / 『Call History』 => 『Missed Calls』 , 『Received Calls』 or 『Dialed Numbers』 .

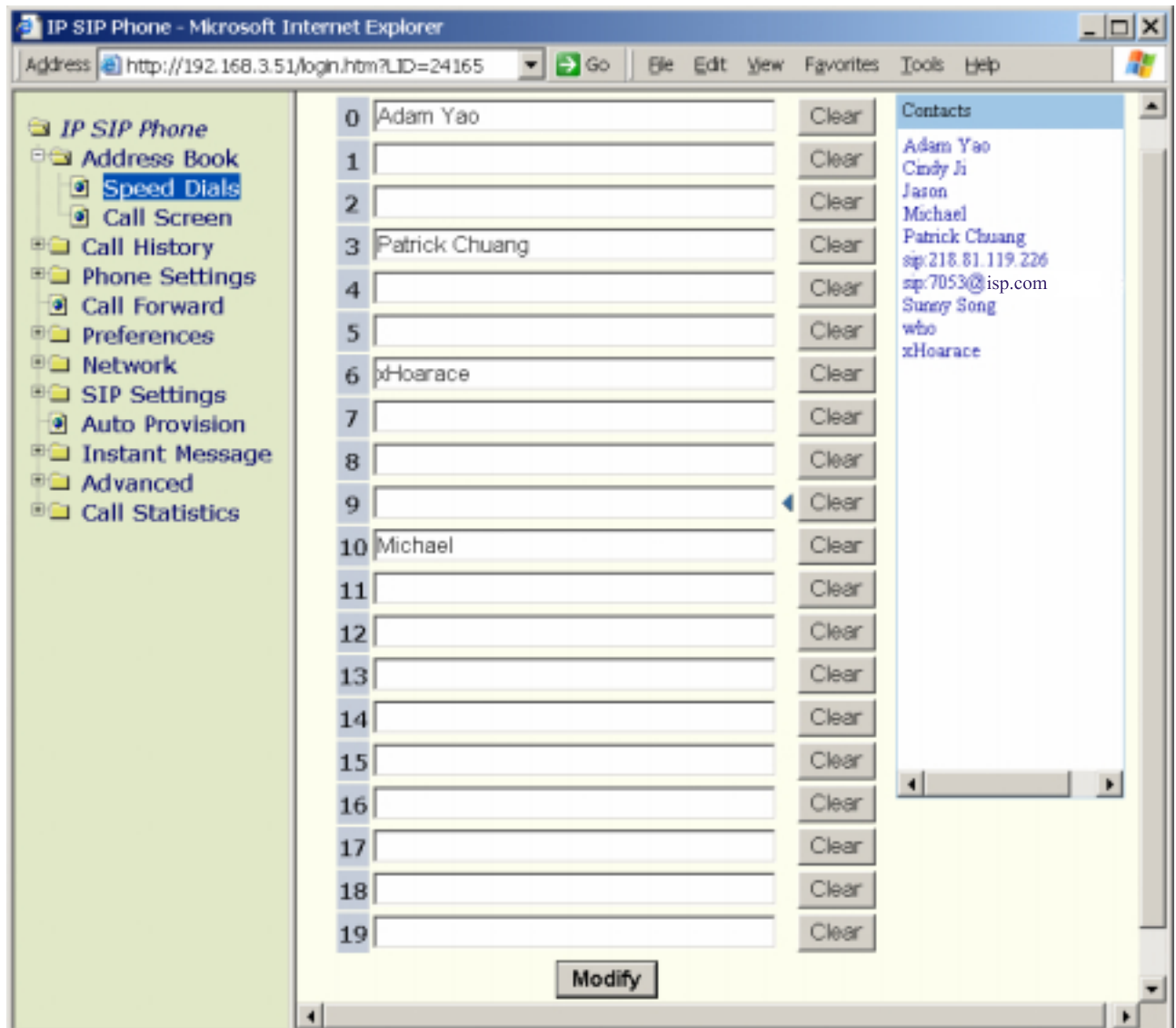


Please refer to section-4.3 “Call History” on “IP SIP Phone v2 Web Administration” for detail.

8.5. Speed Dial

- (1) Add speed dial number:
 - 『Main Menu』 => 『1.Address Book』 \ 『4.Speed Dials』
 - Go to the digit you want to set for speed dial and press **【Hold】**
 - Pick the target number from address book to associate with this speed dial digit. Press **【Hold】** to save.
- (2) Remove a specific speed dial entry:
 - 『Main Menu』 => 『1.Address Book』 \ 『4.Speed Dials』
 - Go to the digit you want to set for speed dial.
 - Press **【MUTE】** to remove the current speed dial mapping.
- (3) Alternatively, you may configure the speed dial numbers by pointing web browser to your terminal's IP and go to page 『IP SIP Phone』 / 『Address Book』 / 『Speed Dials』 . Please refer to section-4.2.1 “Speed Dials” on “IP SIP Phone v2 Web Administration” for detail.

On default, all entries are unassigned. If the assigned entry has been erased from address book, the mapping will be removed from speed dials automatically.



- To assign a speed dial entry:
 - Position cursor on the text input which you want to assign a speed dial mapping.
 - Click the “Contacts” on the right panel to pick an entry from address book to map to a specific speed dial entry.
- To clear a speed dial mapping:
 - Position cursor on the text input which you want to clear the speed dial mapping.
 - Click “Clear” collocates with each speed dial entry to remove the mapping.

(4) Perform Speed Dial:

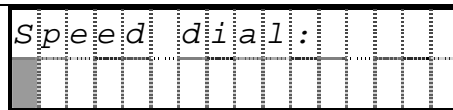
IP SIP Phone supports two different ways to perform speed dial:

- i. [0-19] + 【SPD】 ; e.g., 【1】 + 【SPD】 , or “01” + 【SPD】
- ii. 【SPD】 + [00-19]; e.g., 【SPD】 + “01”

Note: if you want to dial single digit only, such as 【SPD】+ 【1】, please press an additional 【#】 to finish dialing; otherwise, it will wait for the second digit until inter-digit timeout, which default is 4 seconds. For example, 【SPD】

+ **【1】** + **【#】**.

Key	Description
【SPD】 + '0', '0' + 【SPD】 , 【SPD】 + '00' or '00' + 【SPD】	Dial the 0 th speed dial number.
【SPD】 + '1', '1' + 【SPD】 , 【SPD】 + '01' or '01' + 【SPD】	Dial the 1 st speed dial number.
【SPD】 + '2', '2' + 【SPD】 , 【SPD】 + '02' or '02' + 【SPD】	Dial the 2 nd speed dial number.
...	...
【SPD】 + '9', '9' + 【SPD】 , 【SPD】 + '09' or '09' + 【SPD】	Dial the 9 ^d speed dial number.
【SPD】 + '10' or '10' + 【SPD】	Dial the 10 th speed dial number.
...	...
【SPD】 + '19' or '19' + 【SPD】	Dial the 19 th speed dial number.



If the phone has not been hooked off yet, the phone will turn on the speaker phone automatically.

8.6. Call Return

Call return allows the user to call back the last caller. The user can either dial a special feature code such as “*69” to dial up the last caller or select from a list of callers on his or her phone. You could use it as either a network core system feature or a phone-set feature.

If you dial the special feature code “*69” (which is configurable, see below), the phone will send the dialing string “sip:*69@yourdomain.com” to default outbound proxy server, and depends on proxy server to keep the latest call history for each user to cover the phone off-line interval. Moreover, the phone must have been configured to use the call-return capable default SIP outbound proxy server.

On the other hand, if you mapped the DSS key to the phone-set feature “Call Return”, it will find the latest incoming calls from call history of “Missed Calls” and “Received Calls” then dial out the latest incoming call number. *By default, **【F8】** is mapped to phone-set feature “Call Return”.*

The phone-set feature access code can be configured by

- TELNET or keypad: **【MENU】** => “5.Preferences” / “7.Dial Plan” / “3.Call Command” / “1.Call Return”)
- Web browser: 『IP SIP Phone』 / 『Preferences』 / 『Call Command』 => “Call Return”.

8.7. Calling

Dialing state

C	a	l	l	i	n	g				m	m	:	s	s
M	i	c	h	a	e	l	W	u						

Callee reached

T	r	y	i	n	g					m	m	:	s	s
M	i	c	h	a	e	l	W	u						

Ringing

R	i	n	g	i	n	g				m	m	:	s	s
M	i	c	h	a	e	l	W	u						

8.8. Call Failure

Busy

M	i	c	h	a	e	l	W	u						
B	u	s	y		h	e	r	e						

No Answer

M	i	c	h	a	e	l	W	u						
N	o		a	n	s	w	e	r						

Unavailable

M	i	c	h	a	e	l	W	u							
T	e	m	p	o	r	a	r	i	l	y		u	n	a	v

Others

M	i	c	h	a	e	l	W	u						
N	o	t		F	o	u	n	d						

M	i	c	h	a	e	l	W	u						
5	8	8												

No error description available only error ID is shown.

8.9. Auto-redial

You can press **【Auto-redial】** after finishing dialing while making a phone call but before connected or hanging up. If you press **【Auto-redial】** on idle mode, it is effectively as the same as press **【Redial】** follows by a **【Auto-redial】**. After activating auto-redial, the system will lighten its corresponding LED, launch the auto-redial process and re-dial the target number regularly 'till "connected". To manually cancel auto-redial, press **【Auto-redial】** again. This feature will be automatically canceled if either the stop criterion is met or the total activation duration is expired. Pressing the **【Auto-redial】** button has no effect if no active auto-redial.

The semantics of "connected" could be configured from 『Main Menu』=>『5.Preferences』/『5.Auto-Redial』/『1.Stop Criterion』

1.	R	i	n	g	i	n	g				
2.	C	o	n	n	e	c	t	e	d		

- Note: active auto-redial will be canceled whenever the phone is locked out.

To specify the activation duration of this auto-redial feature once starts (measured in seconds), please go to 『Main Menu』 => 『5.Preferences』 / 『5.Auto-Redial』 / 『3.Total Duration』 . Once this duration expires, the auto-redial feature will be silently canceled. System default is 1800 seconds (3 minutes).

IP SIP Phone - Microsoft Internet Explorer

Address <http://192.168.3.51/login.htm?LID=27655> Go File Edit View Favorites Tools Help

- IP SIP Phone
 - Address Book
 - Call History
 - Phone Settings
 - Call Forward
 - Preferences**
 - Voice Mailbox
 - Call Command
 - Network
 - SIP Settings
 - Auto Provision
 - Instant Message
 - Advanced
 - Call Statistics

Preferences

<input type="checkbox"/>	Auto Answer
<input checked="" type="checkbox"/>	Call Waiting
<input checked="" type="checkbox"/>	Auto Hold on Call Switch
<input checked="" type="checkbox"/>	Follow Redirect
<input checked="" type="checkbox"/>	Message Alert
<input checked="" type="checkbox"/>	LAN Dial
Dial Timeout(s)	30
Hold Recall Alerting Timeout(s)	60
Inter-digit Timeout(s)	4
Dial Key	#

Auto-redial

Stop Criterion	On Ringing
Retry Interval(s)	15
Total Duration Time(s)	1800

Modify

IP SIP Phone supports one-touch dial by mapping some DSS keys as hot lines such that the user could dial out frequently used numbers by pressing the associated function key(s) directly.

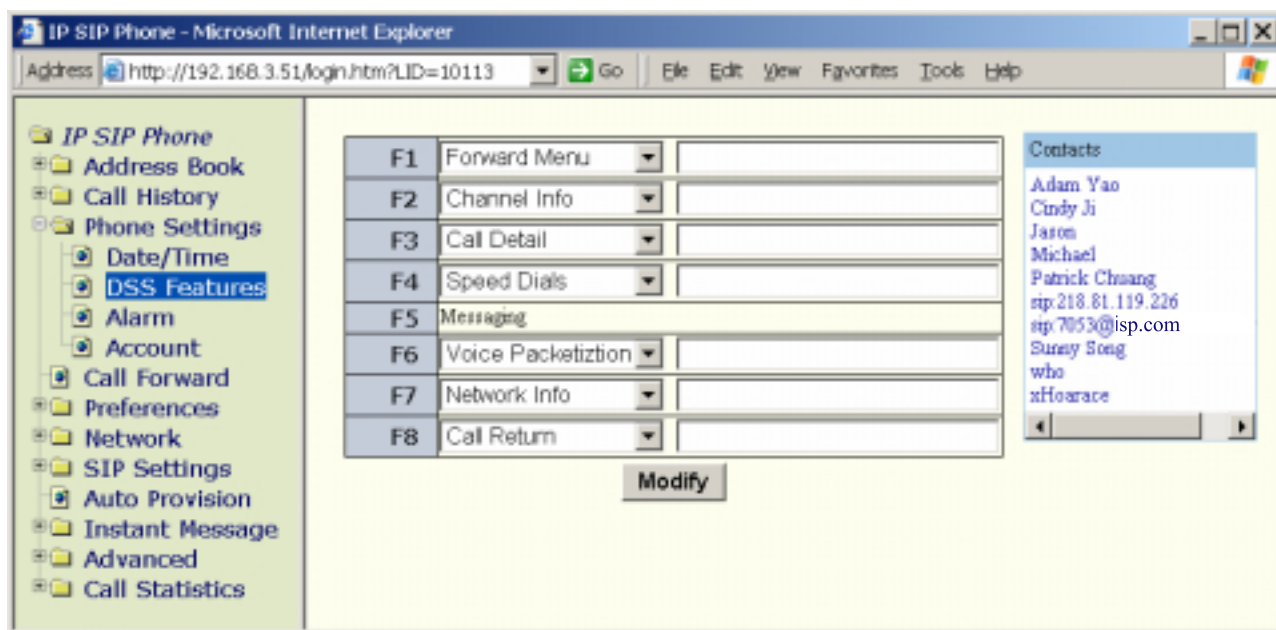
[43/100]

- | | | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|--|--|--|--|--|
| A | d | a | m | | Y | a | o | | | | | | | | |
| A | b | e | r | t | | C | h | a | n | g | | | | | |
| F | e | l | i | x | | C | h | a | n | g | | | | | |
| H | o | r | a | c | e | | F | u | | | | | | | |
| J | a | c | k | y | | W | a | n | g | | | | | | |
| J | u | n | | D | e | n | g | | | | | | | | |
| M | a | x | y | | M | a | | | | | | | | | |
| M | i | c | h | a | e | l | | W | u | | | | | | |
| S | c | o | t | t | | S | u | n | | | | | | | |
| S | u | n | n | y | | S | o | n | g | | | | | | |
| W | i | l | l | i | a | m | | G | u | | | | | | |

- Repeat from step-b to step-d if you want to set up more than one hot line.

Note: the associated hot line number will be invalidated whenever the corresponding address book entry has been erased.

You may also configure it from web browser by going to 『*IP SIP Phone*』 / 『Phone Settings』 / 『DSS Features』 .



Hold Recall” by keypad or web configuration page **IP SIP Phone** / **Preferences**
 => “Hold Recall Alerting Timeout (s) “):

- If you are in on-hook state:
 - i. The phone will ring.
 - ii. The green LED of **【A Call】** flash fast.
 - iii. 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.
- If you are in off-hook state, either engaged in other calls or about to making calls:
 - i. The green LED of **【A Call】** flashes slowly but the ringer does not ring.
 - ii. Play hold recall tone once to remind the user that some calls are in hold.
 - iii. Reset the hold recall timer for next time alarm.

3. If the user is put on hold by the peer, LCD will show:

B	e		H	e	l	d					m	m	:	s	s
M	i	c	H	a	e	L									

Note: if you are held by the peer, *IP SIP Phone* will play music-on-hold.

9.3. Mute

													m	m	:	s	s
M	i	c	H	a	e	L											

- Set up a connected call.
- Press **【MUTE】** to toggle your voice transmission. On muting state, the red LED of **【MUTE】** will be on, and you can hear still the peer’s voice; however they cannot hear you.

9.4. Transfer

Call Transfer allows a user, on any existing two-party call, to place the existing call on hold and originate another call to a third party. The user may consult privately or connect the original call to the third party.

There are two types of transfers:

- Blind Transfer—With blind transfer you transfer the caller to another party without announcing the caller to the called party. To blind transfer a caller, you would press **【XFER】** to place the caller on hold. You will hear a dial tone and can dial the number you want to transfer the call to. It will immediately transfer the call and you could hang up as soon as finishing dialing or wait until the 3rd party picks up then

disconnect from the other party.

- Consultative Transfer -With consult transfer you consult the other party before transferring the call. To consult transfer a caller, you would press the 【XFER】 button to place the caller on hold. You will hear a dial tone and can dial the number you want to transfer the call to. Wait until the other party answers the call and inform them you are transferring the call. Press the 【XFER】 button on the phone set to complete the transfer or just hangs up the call it will also perform the transfer.

Note, you can not transfer two inbound calls to each other. Instead, you can blindly transfer each call to a third party. On which time, the channel LED in question, 【A / B】, will be red (v.s. green).

9.4.1. Consultative Transfer

Follow the following steps to do a consultative transfer: announce it first before actually transfer the call.

- Set up a connected call to party-A
- Press 【XFER】 to put A on hold and gain dial-tone.
- Dial the target number of party-B and press 【#】 to finish dial.
- Wait 'till B picks up, announcing the call from A (consultation).
- If B is unwilling to take this call, you may press 【FLASH】 to cancel the transfer operation and take back A (alternatively, you may press 【A/B Channel】 to cancel transfer and take back A as well).
- Press 【XFER】 again (*or you could just hang up the call to B*) to transfer party-A to party-B. Note, you may do so even when B has not picked up the consultation call yet (such as it is in ringing back phase) to perform a call-transfer; and this is the so-called *Semi-attended transfer or ring-transfer*.
- (Optional) Alternatively, you may press 【Conference】 to hold a 3-way conference with A and B instead of transferring A to B.

Alternatively, you may do a consultative transfer as follows:

- Take an incoming call from party-A on 【A Channel】.
- Get an empty channel by pressing 【B Channel】 to perform a call switch (effectively put A on hold), then make an outbound call to party-B.
- After talking to party-B, you decide to transfer A to B, press 【XFER】 on 【B Channel】 to perform a call-transfer. Note, you may do so even when B has not picked up the consultation call yet (such as it is in ringing back phase) to perform a call-transfer; and this is the so-called *Semi-attended transfer or ring-transfer*
- However, if you decide to host a conference among party-A and party-B rather than

transfer from call-A to call-B, you should press **【Conference】** to conference both parties in.

9.4.2. Blind Transfer

Follow the following steps to do a blind transfer: transfer the call without asking the transferring target first.

- Set up a connected call to party-A
- Press **【XFER】** to put A on hold and gain dial-tone.
- Dial the target number of party-B *without* an ending **【#】** ; instead, you **SHOULD** press **【XFER】** immediately to transfer party-A to party-B; alternatively, you **MAY** just **hang up** (on B-channel) as soon as finishing dialing B's number *without* an ending **【#】** .
- (Optional) During dialing phase, if you decide not to transfer but to take back party-A, press **【FLASH】** to abandon blind transfer and take back A (alternative, you may press **【A/B Channel】** to cancel blind transfer and take back A as well).

*Note, if you press **【#】** to finish dialing, it behaves just the same as consultative transfer.*

Alternatively, you may do a blind transfer as follows:

- Take an incoming call from party-A on **【A Channel】** .
- Get an empty channel by pressing **【B Channel】** to perform a call switch (effectively put A on hold), and gain dial tone.
- Dial the target number of party-B *without* an ending **【#】** ; instead, you **SHOULD** press **【XFER】** immediately to transfer party-A to party-B; alternatively, you **MAY** just **hang up** (on B-channel) as soon as finishing dialing B's number *without* an ending **【#】** .

*Note, you can not transfer two inbound calls to each other. Instead, you can blindly transfer each call to a third party. On which time, the channel LED in question, **【A / B】**, will be **red** (v.s. **green**).*

9.4.3. Phone Locked

Under station lock, you can receive calls but not making calls. Moreover, you cannot perform call transfer either. The phone will show the following alarm to warn you:

- on Case 2: you have two calls at hand already, and you press **【Conference】** on either active channel:
 - a. The green LED of **【Conference】** will be on if *IP SIP Phone* can host a conference among them.
 - b. Otherwise, the red LED of **【Conference】** will flash for a while to indicate a conference cannot be set up properly. This may due to the fact that both connected parties demand a low bit-rate compressed voice CODEC, such as G.723.1(A) or G.729 (A/B), and thus a conference cannot be held between them since the low bit-rate compressed voice CODEC is shared between them (IP SIP Phone equips with only one hardware low bit-rate G.723/G.729 CODEC).*
 - c.

Note: When a conference is undergoing, *IP SIP Phone* will take or receive no more calls (just like in busy state).

9.5.2. Conference Tips

- Once the conference call initiator disconnects, the calls are terminated between all parties.
- To disconnect a specific party in a three-way local conference, you may ask the party to whom you want to disconnect to hang up first. Alternatively, you may press **【FLASH】** to tear down the conference and separate them into 2 independent calls first, then disconnect each of them as your wish.
- To place a conference call on hold, press the **【HOLD】** key. The other parties can not talk among themselves. To avoid disrupting the other callers, consider muting the call instead. To mute the call, press the **【MUTE】** button.
- To place a conference call on speakerphone, press **【SPK】**.
- Press **【MUTE】** to mute the speakerphone. The conference parties cannot hear you but you can hear them.
- The phone only allows three parties in a conference call.
- Call waiting calls cannot be conferenced in; the conferencing node must establish both legs of the 3-Way conference.
- A holding call will be unhold automatically before adding into a conference. Besides, you could put all parties on hold after setting up conference.
- Muting calls will be un-mute automatically before being conferenced in.
- You cannot perform transfer on conference calls.
- To conference in additional callers, the last person called with an *IP SIP Phone* can call an additional person, that new person can then call someone else, and so on. This is called “daisy-chaining”.

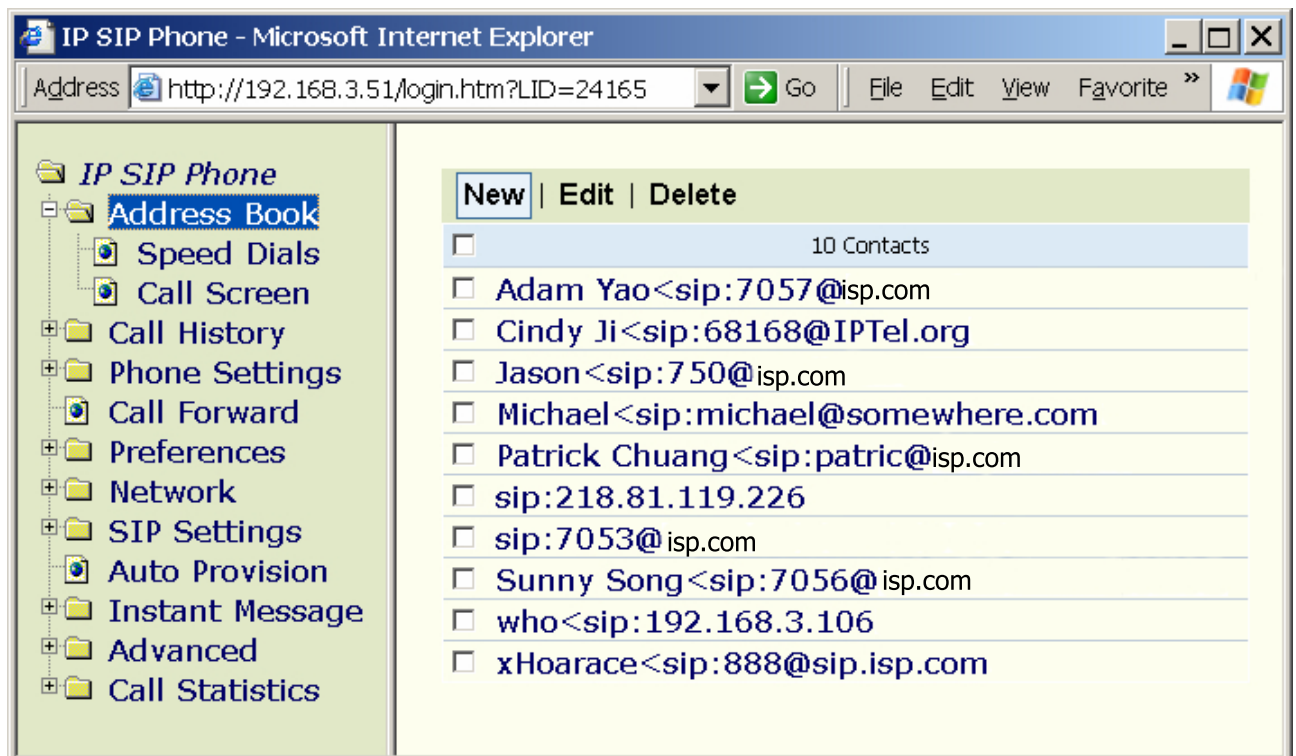
9.6. Block Calls

IP SIP Phone can add entries on address book into blocking list and filter out calls from those parties. On receiving calls from those parties, *IP SIP Phone* will drop them silently with a response “480 Temporarily unavailable” (same as DND). Currently the max blocking list capacity is 20.

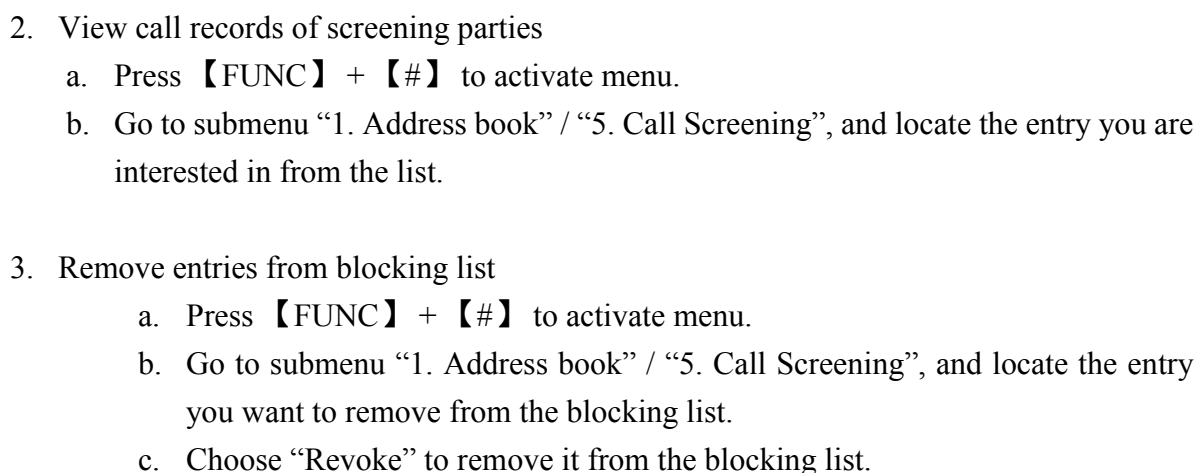
1. Add entries into blocking list
 - a. Press **【FUNC】** + **【#】** to activate menu.
 - b. Go to submenu “1. Address book” / “1. Search”, and locate the party you want to block.
 - c. Choose “screen call” to add it into blocking list (press **【HOLD】** again to toggle between addition and removal).

1	M o d i f y
2	E r a s e
3	S c r e e n c a l l s

Alternatively, you may point your web browser from you PC to the phone, then go to 『Address Book』 and click on the contact which you want to block out.

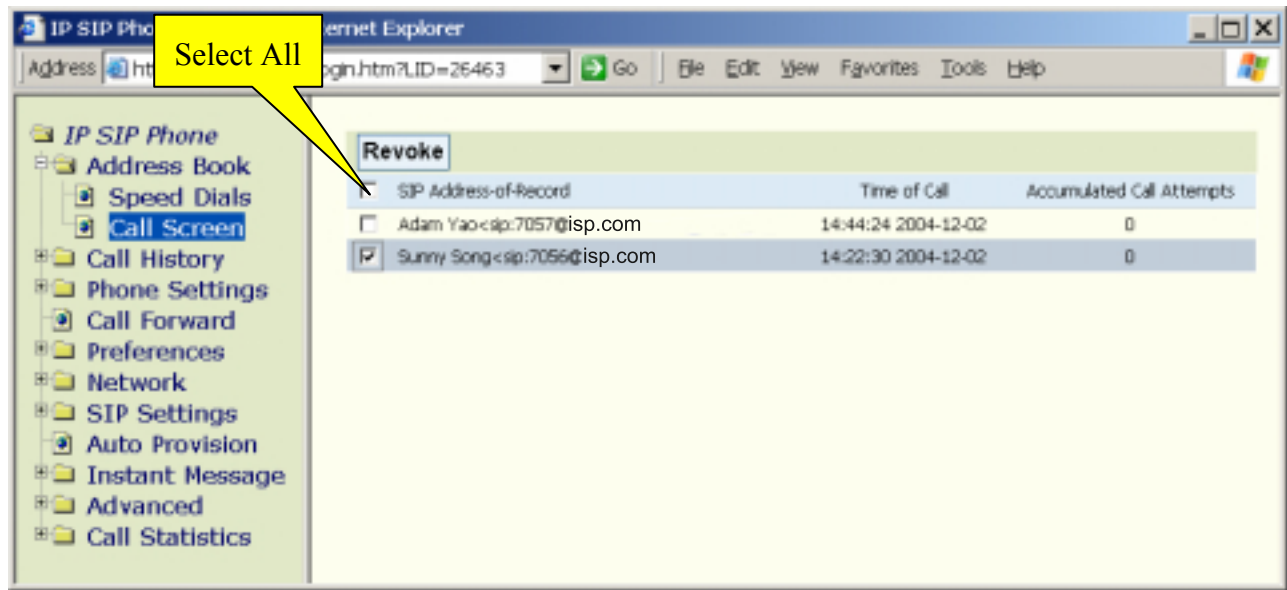


Then click “Add to Call Screen” to add this contact into black list.



1.	Time	of	call		
2.	Call	at	t	e	m
3.	<i>Revoke</i>				

[52/100]



System default is enabled.

10.2. Dial Timeout

You could configure the dialing timeout if no response (no ringing back) 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

To configure Dial timeout:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “2. Dial Timeout”.
- Set the timer.

D	i	a	l	i	n	g	T	i	m	e	o	u	t	:
[1	0	-	6	0	0]	s	e	c		3	0	

- Alternatively, you may go to web page **『IP SIP Phone』** / **『Preferences』** => “Dial Timeout (s)” to configure it.

10.3. Hold Recall

When some calls are put on hold, the phone will play an audible tone regularly to alert user on active channel and the indicating LED will blink as well. The 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.

Note: *During a conference, an audible tone will be played regularly from the hosting phone to notify all parties that a conference is undergoing. The alerting tone is as the same as the hold-recall tone and the alerting interval is identical to the configured hold-recall interval.*

The alerting process is as following:

- User are in on-hook state:
 - The phone will ring.
 - The green LED of **【A Call】** flash fast.
 - 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.
- User are in off-hook state, either engaged in other calls or about to making calls:
 - The green LED of **【A Call】** flashes slowly but the ringer does not ring.
 - Play hold recall tone once to remind the user that some calls are in hold.
 - Reset the hold recall timer for next time alarm.

To configure the alerting interval:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “3. Hold Recall”.

c. Set the interval.

A	l	a	r	t	I	n	t	e	r	v	A	l	:
[1	0	-	6	0	0]	s	e	c		6	0

d. Alternatively, you may go to web page 『IP SIP Phone』 / 『Preferences』 => “Hold Recall Alerting Timeout (s)” to configure it.

10.4. Auto Hold on Call Switch

Enable the “Hold on call switch” feature will automatically put calls on hold whenever the user performs call switch by pressing **【A / B / C Call】** . If this feature is disabled, the phone treats call-switch as disconnection signal: hanging up the active call then switch to designated call.

Note: Press the button standing for current call has no effect, such as pressing **【A Call】** during A-line is active has no effect.

To configure Dial timeout:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “4. Hold on Switch”.
- Set your preference.

1.	Enabled
2.	Disabled

d. Alternatively, you may go to web page *IP SIP Phone* / *Preferences* => “Auto Hold on Call Switch” to configure it.

System default is enabled.

10.5. Auto Redial

You can configure when to stop auto-redial (once activated) by going to **【MENU】 => “5.Preferences” / “ 5.Auto-Redial” / “1. Stop Criterion”** to specify the stop criterion:

1.	Ring in g
2.	Connect Ed

1. Only when the peer starts ringing back will auto-redialing stops (Default).
2. Only when the peer picks up will auto-redialing stops.

When you dial a number and fail to reach the peer, you will hear a busy tone. You can press **【Auto-redial】** to activate this feature. When the auto-redial is in progress, the corresponding LED will be on, and it will re-dial the failed number 'till the stop criterion is met or this feature is canceled. To cancel auto-redial manually, press **【Auto-redial】** again. This feature will be automatically canceled if either the stop criterion is met or the total activation duration is expired.

The total activation duration is configurable by **【MENU】** => “5.Preferences” / “5.Auto-Redial” / “3. Total Duration”. Default is 1800 seconds (30 minutes).

A	c	t	i	v	a	t	i	o	n	t	i	m	e	:
[3	0	-	8	6	4	0	0]			1	8	0

To avoid overflow the network with redial retries, the phone will wait a gap between each redial (default is 15 seconds). You can configure this redial interval from **【MENU】** => “5.Preferences” / “5.Auto-Redial” / “2.Retry Interval”.

R	e	d	i	a	l		I	n	t	e	r	v	a	l
[1	-	2	5	5]	s	e	c				1	5

Alternatively, you may go to web page [IP SIP Phone](#) / [Preferences](#) => “Auto-redial” to configure it.

10.6. Silently Follow Redirection

You can configure the phone to ask user to confirm the redirection before continuing calling attempts. For example,

F	W	D		t	o		x	y	z	@	s	o	m	e	w
(1	=	Y	,	2	=	N)							1

It will stop making calls whenever it is redirected to somewhere else and the user choose not to accept the redirection such that the outbound calls are more secure. On the contrary, you may configure it to automatically follow those redirections until the target number is reached or a loop (ping-pong redirect) is detected.

The default is to silently follow the redirections on making calls without user's interference.

To configure Silently Follow Redirection:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “6. Accept Divert”.
- Configure “Accept” or “Reject” redirections on making calls.
- Alternatively, you may go to web page [IP SIP Phone](#) / [Preferences](#) => “Follow Redirect” to configure it.

10.7. Dial Plan

10.7.1. Inter-Digit Timeout

You can configure the inter-digit timeout while dialing numbers. The 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 **【#】** key whenever finishing dial.

To configure inter-digit timeout:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “7.Dial Plan” / “1. Inter-Digit”.

- c. Enter the preferred inter-digit timeout, measured in seconds.

I	n	t	e	r	-	D	i	g	i	t	T	i	m	e
[3	-	9]	s	e	c	o	n	d	s			4

- d. Alternatively, you may go to web page 『IP SIP Phone』 / 『Preferences』 => “Inter-Digit Timeout (s)” to configure it.

System default is 4 seconds.

10.7.2. Dial key

You can configure which key denotes the end of a dialing string (exclusive). You may use the pound sign, **【#】**, the flash button, **【FLASH】**, or assign a DSS key as “Dial key”.

Default is the pound sign, **【#】**

To configure the “Dial-key”:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “7.Dial Plan / “2. Dial Key”.
- Pick your favorite dial key.

1	P	o	u	n	d	K	e	y	:	#				
2	S	t	a	r		K	e	y	:	*				
3	F	l	a	s	h	K	e	y	:	&				
4	D	S	S	:	D	i	a	l	K	e	y			

- d. To enable “4. DSS Dial key”, you SHOULD map one of the DSS keys (**【F1】** ~ **【F8】**) as **【Dial-key】** from **【MENU】** => “3.Phone Settings” / “2. DSS Features”. For example, you may **【F4】** as **【Dial-Key】** by picking “Dial Key” on **【F4】**.

*Note, the DSS dial key will function as well even you pick **【#】**, **【*】** or **【FLASH】** as dial key.*

- e. Alternatively, you may go to web page 『IP SIP Phone』 / 『Preferences』 => “Dial Key” to configure it.

10.7.3. LAN Dial

You can 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.

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

Please refer to section 8.1 - “Dial Scheme” for LAN dialing, and section 12.5 - “Soft-Switch (PBX) Feature Access”.

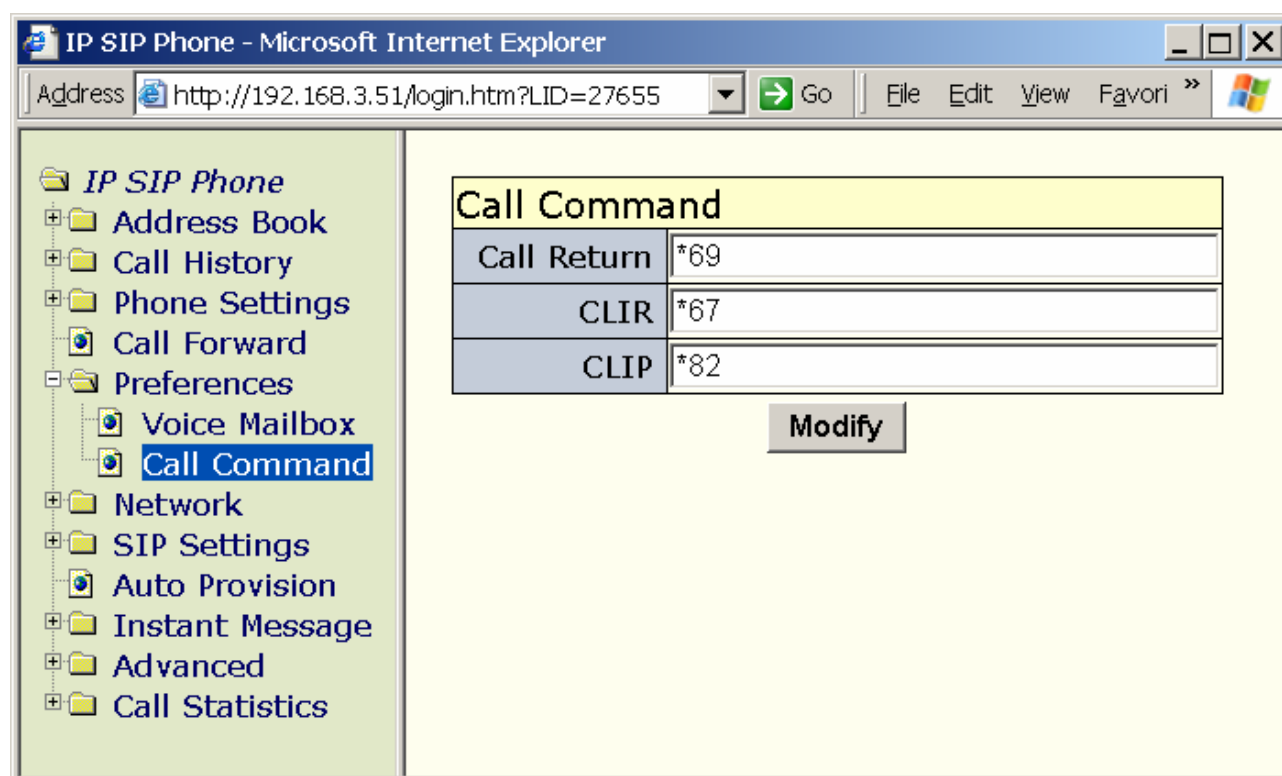
- a. Press **【FUNC】** + **【#】** to activate menu.
- b. Go to submenu “5. Preferences” / “7.Dial Plan / “3. LAN Dial” to configure this feature.

<i>1</i>	E n a b l e d						
<i>2</i>	D i s a b l e d						

Default is enabled

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

You may point web browser to your terminal's IP and go to page 『*IP SIP Phone*』 / 『Preferences』 / 『Call Command』 to configure available call commands.



Subsections below illustrate how to configure them by keypad (TELNET).

10.7.4.1. Call Return

Configure the code to access soft-switch feature: “call return”. Whenever your dial string matched the specified call-return string, *IP SIP Phone* will send the dial-string “as is” (with SIP domain appended) to the SIP proxy server, such as “sip:*69@SIP.isp.com”, and depends on proxy server to keep the latest call history for each user to cover the phone off-line interval.

To configure the access code:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “7.Dial Plan / “4. Call Command” / “1. Call Return”.
- Enter the call return access code to meet your local regulation.

C	a	l	l		R	e	t	u	r	n	:							
*	6	9																

System default is “*69”.

Note: this is a server feature which relies on the SIP proxy server support. This differs from the DSS “Call return”, which is a phone-set feature. If you map one of the DSS keys (**【F1】** ~ **【F8】**) to “Call return” function, it will find the latest incoming calls from call history of “Missed calls” and “Received calls” (which both are kept locally) then dial out the latest incoming call number. *By default, **【F8】** is mapped to “Call return”.*

10.7.4.2. Anonymous Call (CLIP & CLIR)

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 SIP 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=m000dc300051 0 0 IN IP4 192.168.3.5
```

To configure the access code:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “7.Dial Plan / “4. Call Command” / “2. Anonymous Call”.
- Enter your customized CLIR access code, following the destination number:

C	L	I	R	A	c	c	e	s	s	C	o	d	e
*	6	7											

Note: Once the CLIR activation code is detected, the **RED** channel LED (**【A/B Call】**) will be on (instead of the green one) to remind user that CLIR is currently on. To cancel CLIR, you may hang on then restart the whole dial process or alternatively, you may enter the CLIP activation code to counter CLIR.

System default activation code for CLIR is “*67”, and “*82” for CLIP.

10.8. Message alert

After sending an instant message or receiving a new message, the phone will play an auditable tone to remind user. You can configure the phone whether the auditable tone should be played or not.

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “5. Preferences” / “8.Message Alert”.

1	E	n	a	b	l	e	d						
2	D	i	s	a	b	l	e	d					

- Alternatively, you may go to web page [IP SIP Phone](#) / [Preferences](#) => “Message Alert” to configure it.

System default is enabled

10.9. Auto-Answer

If auto-answer feature is enabled when an incoming call arrives, the phone-set will play distinguished auditable tone shortly, then answer the call by turning on speaker-phone (or headset if it is enabled) on idle mode; otherwise proceed as a normal incoming call.

By idle, it means the phone-set is either engaging in no calls or putting a call on hold.

IP SIP Phone implements various ways to support auto-answer features:

- Server-side invoked auto-answer for urgent calls:

Use of a private header “P-Auto-answer” in the initial INVITE message for the server-side invoked auto-answer extension:

 - P-Auto-answer: normal*
 - Respond a “486 Busy Here” if all lines are occupied.

- Otherwise, **auto-answer^(l)** this incoming call.
- b. *P-Auto-answer: **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^(l)** the newly arrived call.
- c. *P-Auto-answer: **imperious***
- *Identical to “P-Auto-answer: **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.*

Note, the header, “P-Auto-Answer”, and its keyword, “normal”, “urgent” or “imperious”, are all case-sensitive.

- ii. Phone-set locally configured to auto-answer all incoming calls unconditionally (suitable for attendant in call center service).
 - a. Unconditionally auto-answer all incoming calls on idle mode can be achieved by either configuring the global phone-set setting and/or map the auto-answer feature as a programmable key.
 - b. To configure the global phone-set setting for auto-answer:
 - Press **【FUNC】 + 【#】** to activate menu.
 - Go to submenu **▣5. Preferences ▣ / ▣9.Auto-Answer ▣**.

1	. E n a b l e d .								
2	. D i s a b l e d								

System default is Disabled

- c. Alternatively, you may configure this system-wide feature from web page 『IP SIP Phone』 / 『Preferences』 => “Auto Answer”.
- d. If you have mapped one of your DSS keys to “Auto-Answer”, press this DSS to toggle auto-answer mode. Once enabled, the corresponding red LED of mapped DSS key will be on, and the phone will **auto-answer**⁽¹⁾ all incoming calls arrive on idle mode. Besides, unmap this DSS feature will turn off auto-answer configuration as well.

- iii. Auto-Answer incoming calls arrive on specific registered SIP address-of-records.
For each service domain, you can configure to auto-answer all incoming calls to this SIP account (<FROM> header in the initial INVITE message). This is particularly useful for IP-PBX to implement broadcast feature. Alternatively this feature is also useful when the user have 2 SIP accounts (or more), one for public and one for private use only, and he/she wants to auto-answer all calls to his/her private account.

To configure the account-specific auto-answer feature:

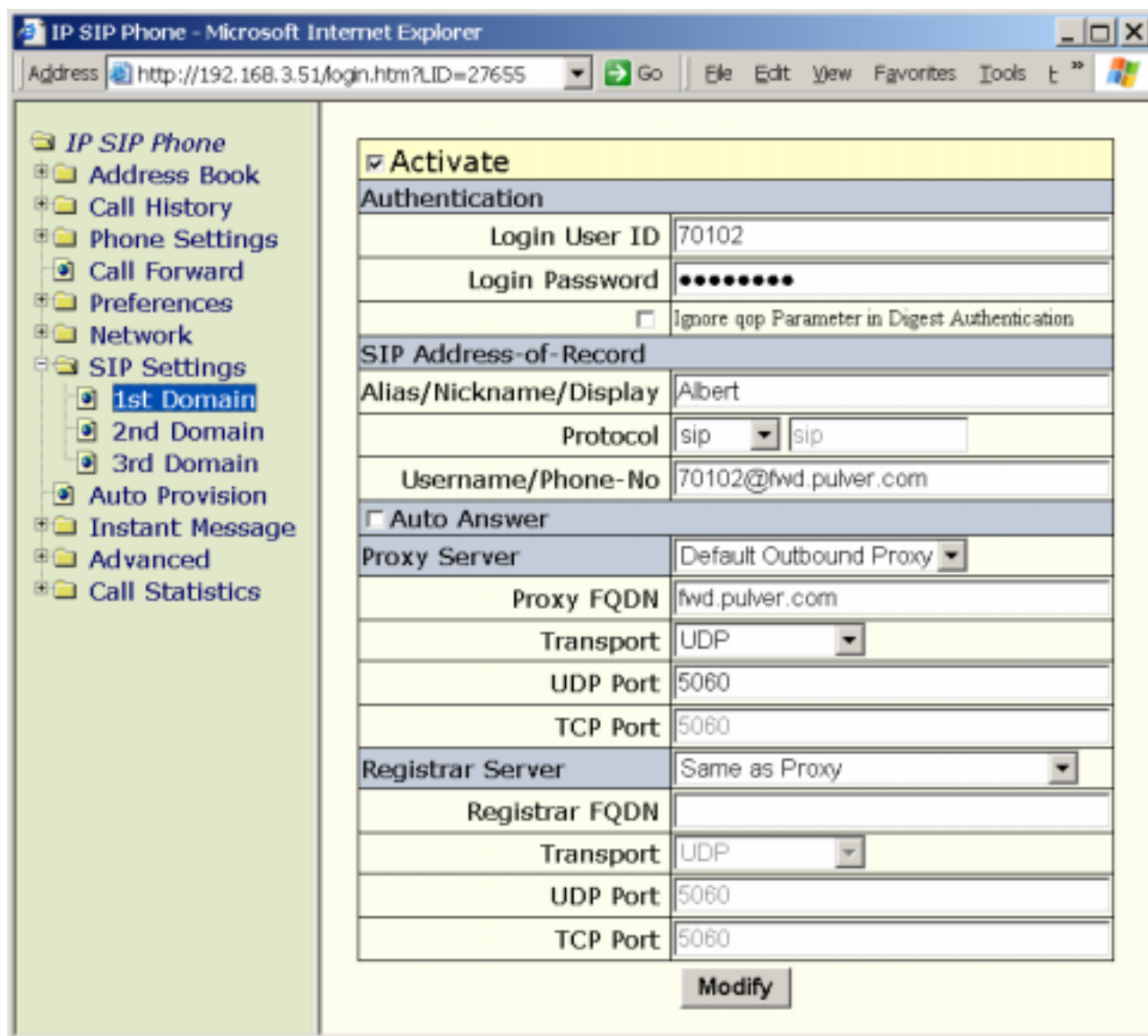
- Press **【FUNC】 + 【#】** to activate menu.
- Go to submenu **⌂6. Network⌂ / ⌂2.SIP Settings⌂ / ⌂2.Service Domain⌂ / ⌂Nth**

Realm』 / 『Auto-Answer』.

1.	Enabled
2.	Disabled

Default is Disabled.

- Alternatively, you may configure this account-specific feature from web page `IP SIP Phone` / `SIP Settings` / `Nth Domain` => “Auto Answer”.



- a. Once enabled, all calls destined to this specific service account will be auto-answered on idle mode.
 - b. This works even when the system-wide auto-answering is off.
- **auto-answer⁽¹⁾** operation:
 - i. On idle mode, the phone-set will play a distinguished audible tone shortly, then answer the call by turning on speaker-phone (or headset if applicable). By idle, it either takes no call or is putting a call on hold.

- ii. Otherwise, proceed as normal incoming calls
- Incoming call processing rules (by precedence):
 - i. Check for the server-side invoked auto-answer feature, if “*P-Auto-answer: imperious*” is present, auto-answer it.
 - ii. If the phone is engaged in do-not-disturb mode, then DND wins
 - iii. Otherwise, if the phone has turned on unconditionally forward feature, then all incoming calls are forwarded.
 - iv. 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.
 - v. Otherwise, proceed as normal incoming calls.

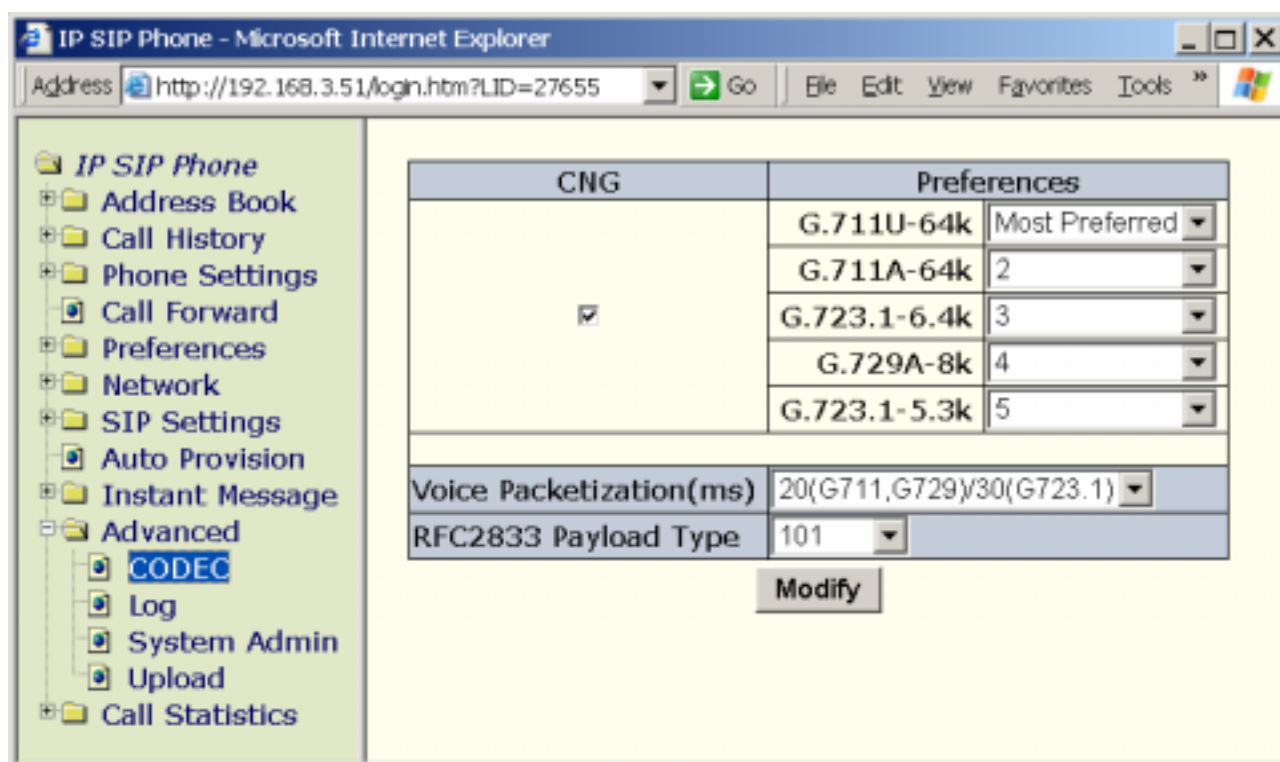
10.10. CODEC Preference

The CODECs available on the phone includes G.711A (64kbps), G.711 μ (64kbps), G.729A / G.729AB (8 kbps), G.723.1 / G.723.1A (both 5.3 and 6.4 kbps). 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.711A, G.711 μ , G.729A then G.723.1A (in that order).

High quality comes in a price: it consumes more precious bandwidth. Take G.711 μ 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 the most preferred CODEC on making / taking calls.

You could also configure the default voice packetization transmitted to the peer:

- By keypad (TELNET):
 - a. Press **【FUNC】** + **【#】** to activate menu.
 - b. Go to submenu “8. Advanced” / ”1. CODEC” / “2.Packetization”.
 - c. Configure the default voice packetization transmitted from your *IP SIP Phone* to the peer.
- By web configuration:
 - a. go to 『IP SIP Phone』 / 『Advanced』 / 『CODEC』



The phone will adjust the audio frames carried per RTP packet (voice packetization) based on this setting:

Voice Packetization		G.723.1	G.729A	G.711
10/30(G.723.1)-ms	LAN	1 Frame (30 ms)	1 Frames (10 ms)	
20/30(G.723.1)-ms	xDSL, ISDN		2 Frames (20 ms)	
30-ms	xDSL, ISDN, Cable		3 Frames (30 ms)	
40/30(G.723.1)-ms	56 kbps MODEM		4 Frames (40 ms)	
40/60(G.723.1)-ms	Limited bandwidth	2 Frames (60 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.

Note: IP SIP 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 SIP 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=m000dc300051 0 0 IN IP4 192.168.3.51
m=audio 5102 RTV/AVP 0 8 18
a=ptime:10

```

● VoIP Bandwidth

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.

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.

CODEC	Frame per RTP Packet (Packet Rate)	IP (kbps)	Link		Averaged Band-width Utilization	Delay (ms)	Mean Opinion Score (MOS ⁶)
			PPP (kbps)	Ethernet 802.3			
G.723.1 5.3 kbps	1 (35 pps ⁷)	16.8	18.4	21.6	28%	37.5	3.5-3.7
	2 (17 pps)	11.2	12	13.6	43.2%	67.5	
	3 (12 pps)	9.3	9.9	10.9	52.8%	97.5	
	4 (9 pps)	8.4	8.8	9.6	59.3%	127.5	
	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	
G.723.1 6.4 kbps	1 (35 pps)	17.9	19.5	22.7	31.9%	37.5	3.8-4.0
	2 (17 pps)	12.3	13.1	14.7	47.9%	67.5	
	3 (12 pps)	10.5	11	12.1	57.1%	97.5	
	4 (9 pps)	9.5	9.9	10.7	63.8%	127.5	
	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	
G.729ab	1 (100 pps)	42	46.8	56.4	16.5%	15	3.7-4.2

⁶ **Mean Opinion Score** is very subjective and varies from one scoring episode to another depending on a variety of things, e.g., sample size, acoustic environment, and methodology. The values presented here are for audio with no packet loss. Some codecs fare better than others under packet-loss conditions.

⁷ PPS: Packet per Second.

8 kbps	2 (50 pps)	25.2	27.6	32.4	28.2%	25	
	3 (35 pps)	19.6	21.2	24.4	36.8%	35	
	4 (25 pps)	16.8	18	20.4	43.5%	45	
	5 (20 pps)	15.1	16.1	18	48.8%	55	
	6 (17 pps)	14	14.8	16.4	53.1%	65	
G.711 64 kbps	1 (100 pps)	100.8	105.6	115.2	59.9%	10	4.3-4.7
	2 (50 pps)	84	86.4	91.2	73.4%	20	
	3 (33 pps)	78.4	80	83.2	79.5%	30	
	4 (25 pps)	75.6	76.8	79.2	82.9%	40	
	5 (20 pps)	73.9	74.9	76.8	85.1%	50	
	6 (17 pps)	72.8	73.6	75.2	86.6%	60	

Those values in **blue** 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 more delay. We suggest a good setting should not introduce a delay for more than 60 ms.

10.10.1. Enable Personal Preference

To set up personal CODEC preference:

- Press **【FUNC】** + **【#】** to activate menu.
- Go to submenu “8. Advanced” / ”1. CODEC” / “1. Preferences”.
- The 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.
- Edit the priority value you want to set: the smaller the value is, the higher the priority will be.

Note: 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.711 μ when the peer does not support those low bit-rate CODECs for voice compression.

10.11. Comfort Noise Generation

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 SIP 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 SIP 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”.

To configure Voice Activity Detection

- a. Press **【FUNC】** + **【#】** to activate menu.
- b. Go to submenu “8. Advanced” / ”3. CNG”.
- c. Configure to “Enable” or “Disable” silence suppression.

10.12. Registration on Demand

You can dictate the terminal to refresh registration immediately to all service domains by pressing **【Registration】**. If you have gone offline (see-below), by pressing **【Registration】** key, it will restart regular auto-registration scheduling as necessary too. During registration, the corresponding green LED will flash. Based on the registration result, the LED will have different indications:

***Green LED On:** Successfully register to all activated service domains.*

***Red LED On:** At least one activated service domain could not be registered.*

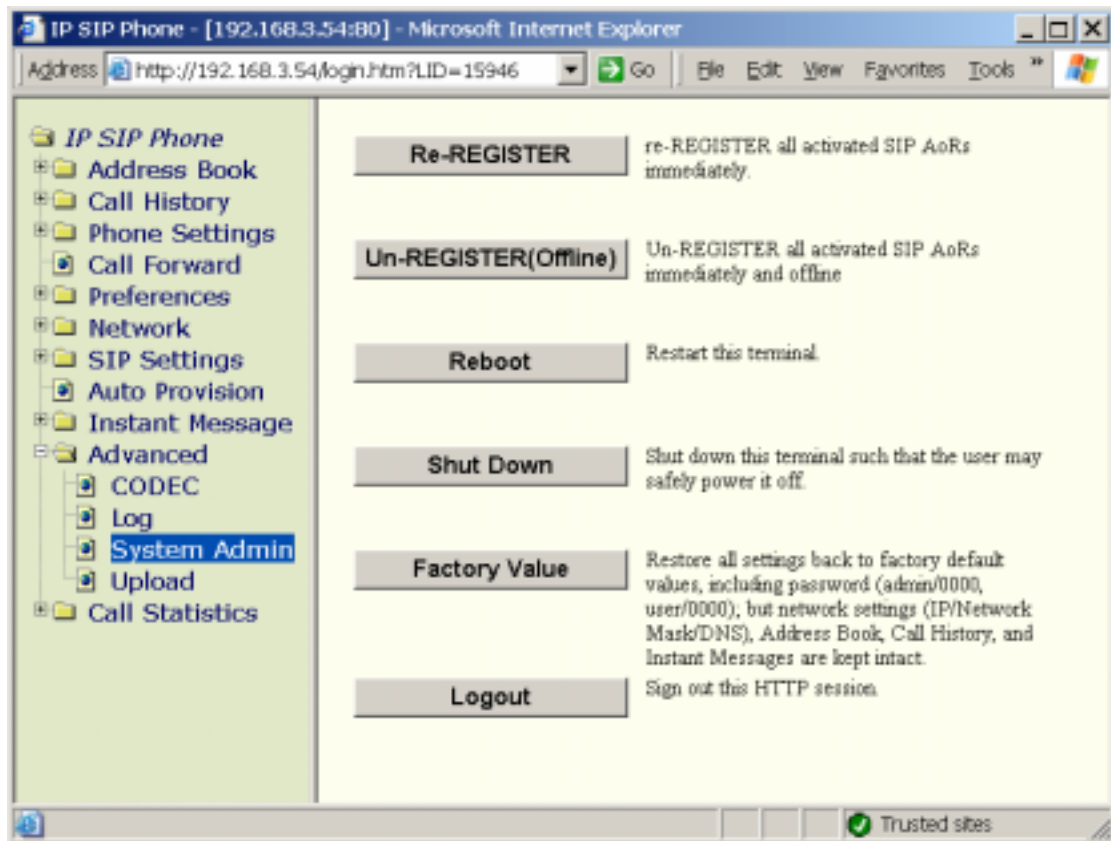
***Green LED Flashes:** Registration is in progress. **Note**, IP SIP Phone **will regularly refresh SIP Address-of-Record registration as necessary**.*

***Red LED Flashes:** No activated service domains have ever registered successfully.*

***LED is off:** Users explicitly logs out all SIP service and goes off-line ‘till user presses the **【Registration】** key again to go on-line (re-register to all SIP service).*

I. To can go off-line explicitly by:

- From keypad (TELNET): going to **【Main Menu】** / **【8.Advanced】** / **【3.System Admin】** and pick **【1.Logout】**.
- From web browser:
 - Go to 『IP SIP Phone』 / 『Advanced』 / 『System Admin』 => “Un-REGISTER” to go offline.



- Alternatively you may issue the following HTTP Get command from your web browser to go online:

http://terminal_ip_address/unregister

where terminal_ip_address is the IP address of your terminal. Those command web pages are password protected.

Please refer to section 4.11.3.1- “Issue Commands by HTTP Get” on “IP SIP Phone v2 Web Administration” for detail.

Once off-line, the terminal will unregister to all activated SIP service domains and cease regular auto-registration scheduling until the **【Registration】** key is explicitly pressed again.

Note: reboot the phone-set will clear this status and register the SIP address-of-record after boot-up.

II. To can go on-line explicitly by:

- From keypad: press **【Registration】** key explicitly.
- From web browser:
 - Point the web browser to your terminal and go to 『IP SIP Phone』 / 『Advanced』 / 『System Admin』 => “Re-REGISTER” to go online again.
 - Alternatively you may issue the following HTTP Get command from your web browser to go online:

http://terminal_ip_address/register,

where terminal_ip_address is the IP address of your terminal. Those command web

pages are password protected.

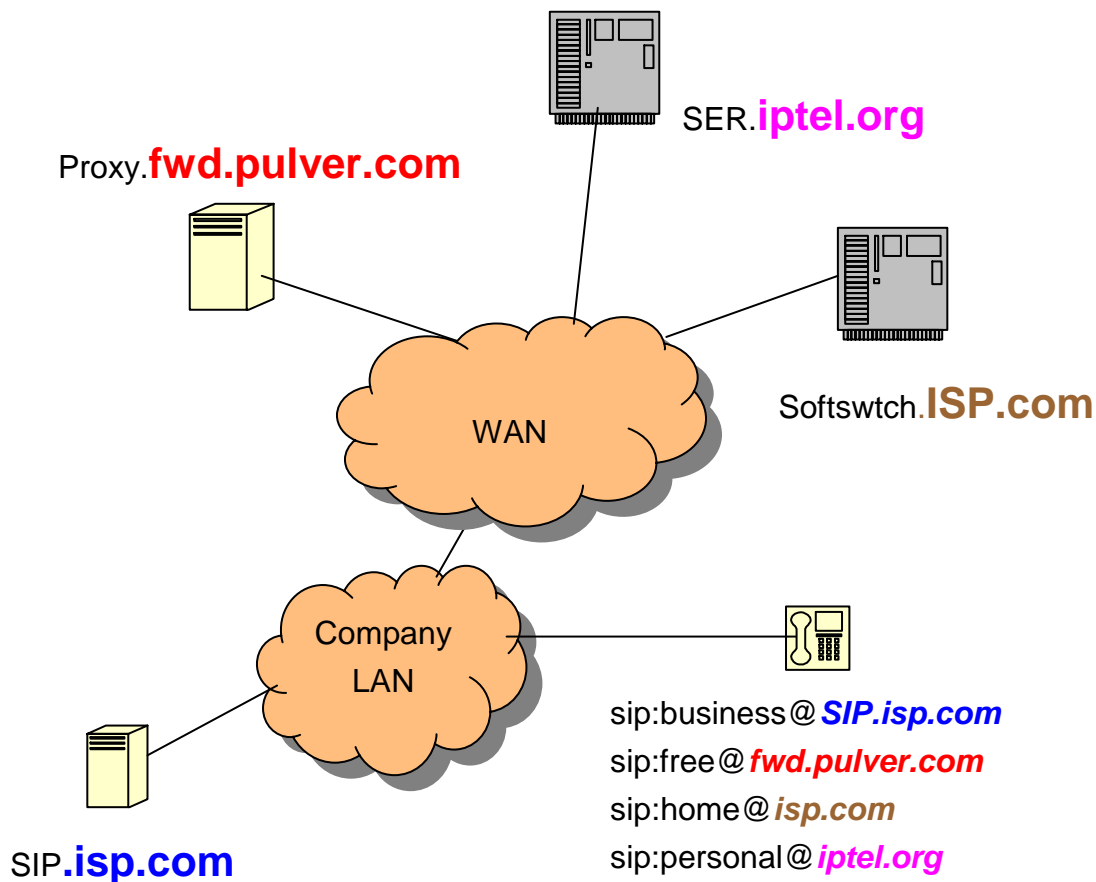
Please refer to section 4.11.3.1- “Issue Commands by HTTP Get” on “*IP SIP Phone v2 Web Administration*” for detail.

You could view registration status of each service domain by pressing **【Service Realm】** on idle status. It will prefix a ‘X’ mark on the unregistered domain, and a ‘0’ mark on the registered domain.

0	:	Q	u	a	n	t	i	e	r	T	e	c	h	.	c	
0	:	f	w	d	.	p	u	l	v	e	r	.	c	o	m	
X	:	u	n	k	n	o	w	n	.	c	o	m				

10.13. Multi-Domain Registration

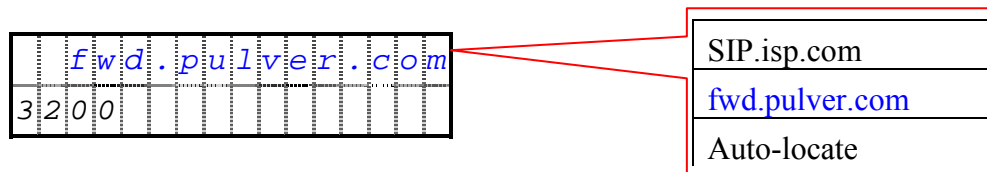
You could register to multiple domains simultaneously such that you could receive calls from those registered domains and make calls to users belonging to different realm. For example, you may have several SIP addresses from different ISPs. Your configuration may be similar to the following:



By default, you can gain access to active Service-Domain-N by line-N whenever applicable.

That is, the default service domain of **【B Call】** is 2nd active service domain if available; otherwise, it will use the 1st active service domain as default.

Moreover, you could change the target service domain while making calls by pressing **【Service Realm】** to circle those active service domains (no matter whether they have registered or not), but must before dialing the **【#】**, which finishes the dialing. The active service domain will appear on the upper-right corner.

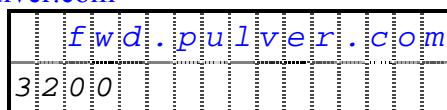


In addition to those service domains, you could also specify the call as “Auto-locating”. That is, if you have registered 2 service domains, you have 3 items to pick from:

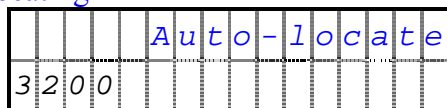
a. [SIP.isp.com](#)



b. [fwd.pulver.com](#)



c. [Auto-locating](#)



11. Voice Volume Adjustment

11.1. Ringer

You could adjust the ringing volume either during the phone set is ringing or from the menu.

- Press 【⇐】 and 【⇒】 to adjust the ringing volume while the phone set is ringing.



- To adjust ringer volume from menu:
 - a. Press 【FUNC】 + 【#】 to active menu.
 - b. Go to submenu “3. Phone settings” / ”4. Ringer” / ”1.Ringer Volume”.
 - c. Adjust the ringing volume by 【⇐】 and 【⇒】 keys, and it will play the ringing tone on adjustment.

Besides, you may pick another ringing tone by to play on calls arrival:

- Press 【FUNC】 + 【#】 to active menu.
- Go to submenu “3. Phone settings” / ”4. Ringer” / ”2.Ringer Type”.
- Pick your favorite ringing tone to use when incomng calls arrive.

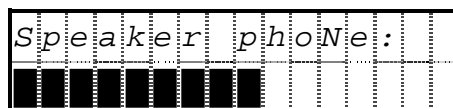
Note, *IP SIP Phone* supports the “alert-info” header in the first INVITE message as per RFC3261, “alert-info” header dictates the phone to use an alternative ringing tone, which is specific for that call. The header value should be something like “ringtone-X” (case-sensitive), where X is the number of 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.

11.2. Handset



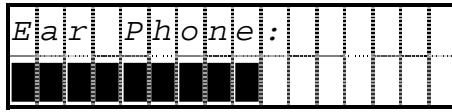
Activate when speaker phone is off and the phone set is hooked off or at least one call is engaged. Use 【⇐】 and 【⇒】 keys to adjust volume.

11.3. Speaker Phone



Activate when speaker phone is on or while at least one call is engaged. Use [⇐] and [⇒] keys to adjust volume.

11.4. Ear Phone



Activate when ear phone is on and the phone set is hooked off or while at least one call is engaged. Use 【⇐】 and 【⇒】 keys to adjust volume.

12. Service

12.1. Voice Mail

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.

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

- MWI Subscription

If you have configured the voice mailbox link, *IP SIP 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;lr=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

IP SIP Phone accepts unsolicited MWI NOTIFY 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)

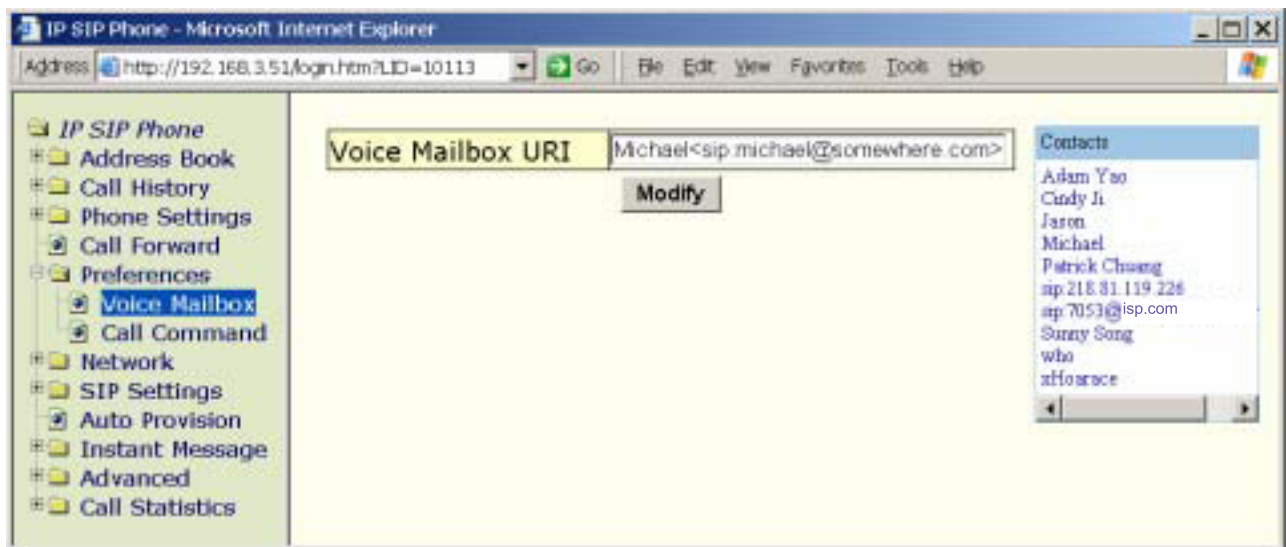
```

12.1.1. Set up Voice Mail

- To set up voice mail access number by TELNET or keypad:
 - a. Press **【FUNC】** + **【#】** key to activate menu.
 - b. Go to submenu “7. Service” / “1. Voice mail URI”
 - c. Configure the voice-mail number to dial when the **【MWI】** button is pressed. Press **【HOLD】** to pick an entry from address book. Use **【⇐】** and **【⇒】** to view the whole link.

M	i	c	h	a	e	l	<	s	i	p	:	8	8	8	8
@	Q	u	a	n	t	i	e	r	T	e	c	h	.	c	o

- d. Press **【MUTE】** to remove current voice mailbox mapping.
- To set up voice mail access number from web browser, go to 『Preferences』 / 『Voice Mailbox』, then click the “Contacts” on the right panel to pick an entry from your address book.



12.1.2. Access Voice Mail

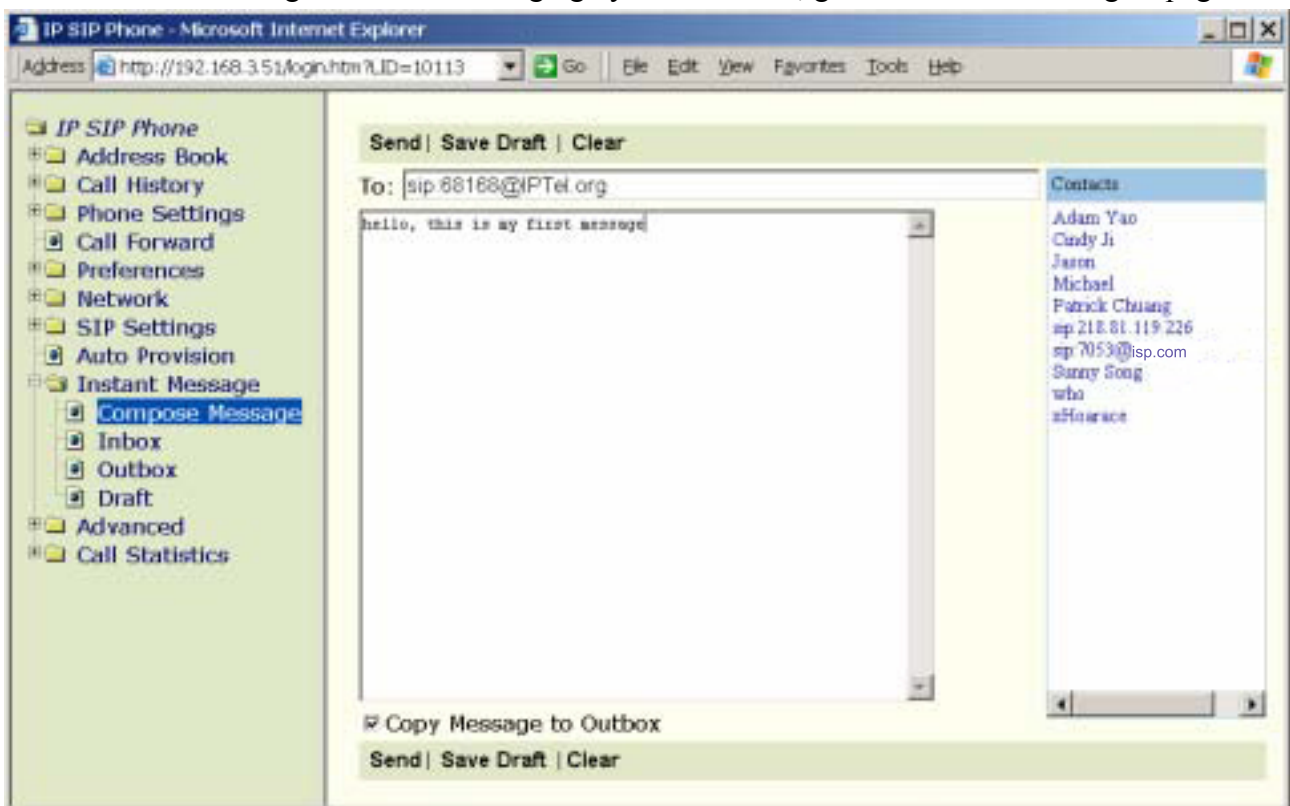
To access voice mail, press the 【MWI】 button and follow the voice instructions. The red LED adjunct to the 【MWI】 button also will be flashing whenever you have unread and/or new voice messages (Message Waiting Indication) in your voice mailboxes.

Note, when the MWI LED is off while pressing 【MWI】, it will dial to the voice mailbox your 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.

12.2. Instant Messaging

IP SIP Phone supports instant short message (255 characters). Its operation is very similar to the short message on modern mobile phones.

- To take advantage of instant messaging by TELNET or keypad:
 - a. Press 【FUNC】 + 【#】 key to activate menu.
 - b. Go to submenu “7. Service” / “2. Message”.
- To take advantage of instant messaging by web browser, go to 『Instant Message』 page.



Please refer to section-4.10 “Instant Message” on “IP SIP Phone v2 Web Administration” for detail.

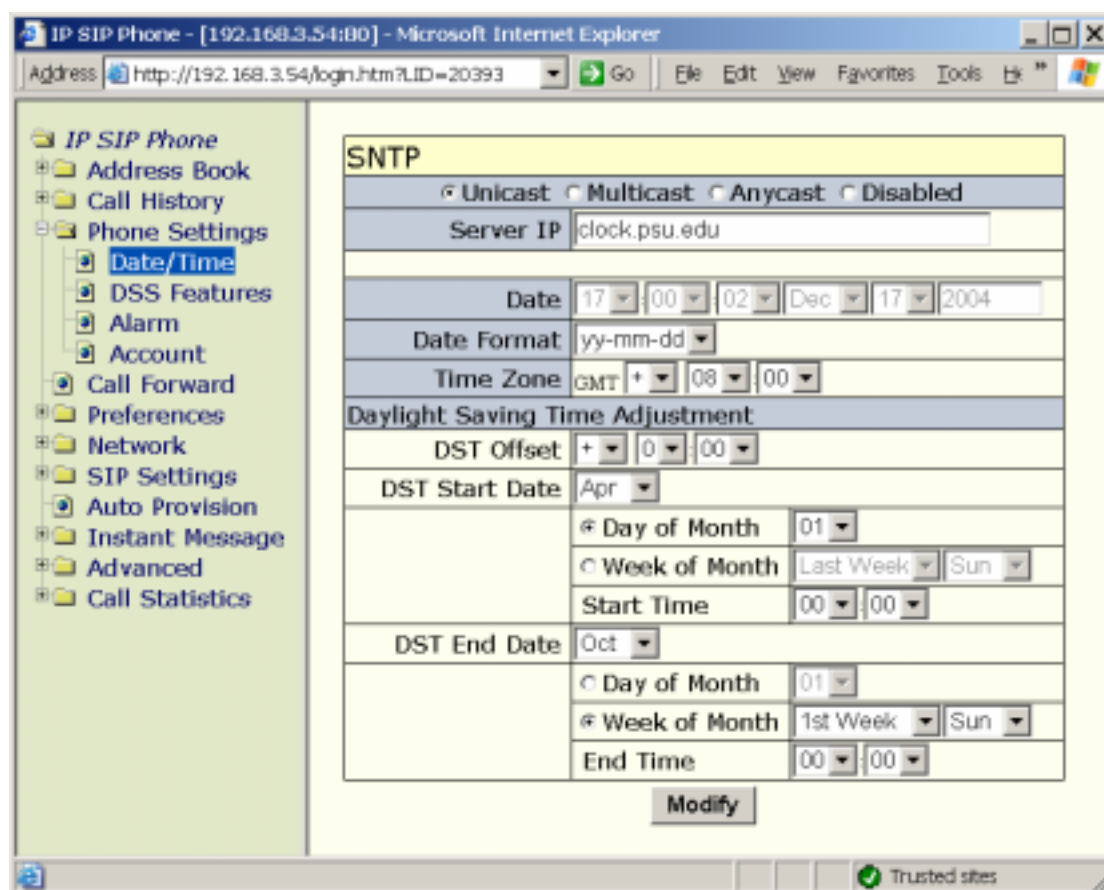
Note, IP SIP Phone supports “flashing short message” to display received out-of-dialog message on LCD without user interaction (or any alert) for 2~3 seconds. Such flashing messages are for server-side notification, and they will not be saved (thus “flashing”). To activate such feature, the received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

You could synchronize the time on your phone with external time server by Simple Network Time Protocol, SNTP.

- | | | | | | | | | |
|----|---|---|---|---|---|---|---|---|
| 1. | S | N | T | P | m | o | d | e |
| 2. | S | e | r | v | e | r | I | P |

- | <i>I</i> | <i>Un</i> | <i>i</i> | <i>c</i> | <i>a</i> | <i>s</i> | <i>t</i> | | | |
|----------|-----------|----------|----------|----------|----------|----------|---|---|--|
| 2 | Mu | l | t | i | c | a | s | t | |
| 3 | An | y | c | a | s | t | | | |
| 4 | Di | s | a | b | l | e | | | |

- To configure NTP / SNTP from web browser, go to 『IP SIP Phone』 / 『Phone Settings』 / 『Date / Time』



The Simple Network Time Protocol is used to synchronize time with *IP SIP 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, 1970, GMT.

	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 data via the SNTP / NTP multicast address from the local network	Unicast SNTP data from the SNTP server that first responded to the network

	SNTP servers.	broadcast address from any server on the network. The default multicast address is 224.0.1.1 .	broadcast request.
--	---------------	--	--------------------

- | | | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|--|
| 2 | 1 | 1 | . | 1 | 7 | 9 | . | 1 | 9 | . | 1 | 3 | 3 | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|--|

Note: If you enable DHCP, the NTP server may be acquired by DHCP option 42.

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

12.4. Auto Provision

You can configure your phone to download configuration files from provision server for auto-provisioning. This feature is available for administrator batch configures the phones in a whole and still leave room for customized service for each phone. If the feature is enabled, the phone will try to download two configuration files from the provisioning server via either TFTP or HTTP on system startup; otherwise, it will use the current settings stored in flash ROM. ***Please refer to section-4.9 “Auto Provision” and section-4.9.6 “Available Configuration” on “IP SIP Phone v2 Web Administration” for configuration file format and available tags.***

- To enable auto-provisioning on system startup:
 - a. Press **【FUNC】** + **【#】** key to activate menu.
 - b. Go to submenu: “7. Service” / “4 Auto provision”

[illegible]

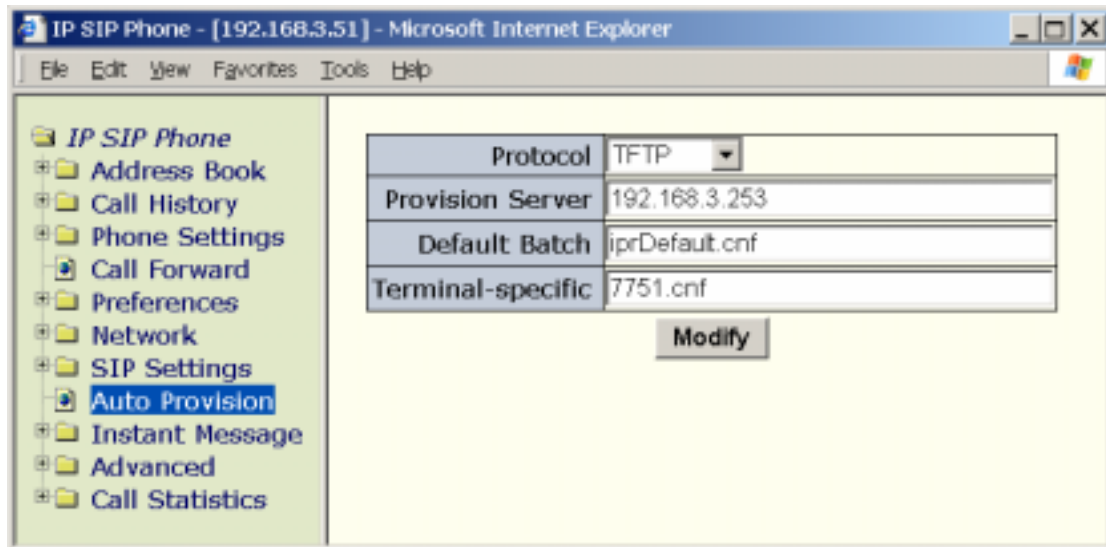
On this menu, you could select which protocol to use for auto-provision, specify where the auto-provision server locates and which configuration files to download. The applying rule is: settings read from “Phone-specific” configuration file will overwrite those read from “Batch default” which in turn overwrites that of Flash ROM settings.

Note: If you enable DHCP, the TFTP server may be acquired by DHCP option 66 (TFTP server name).

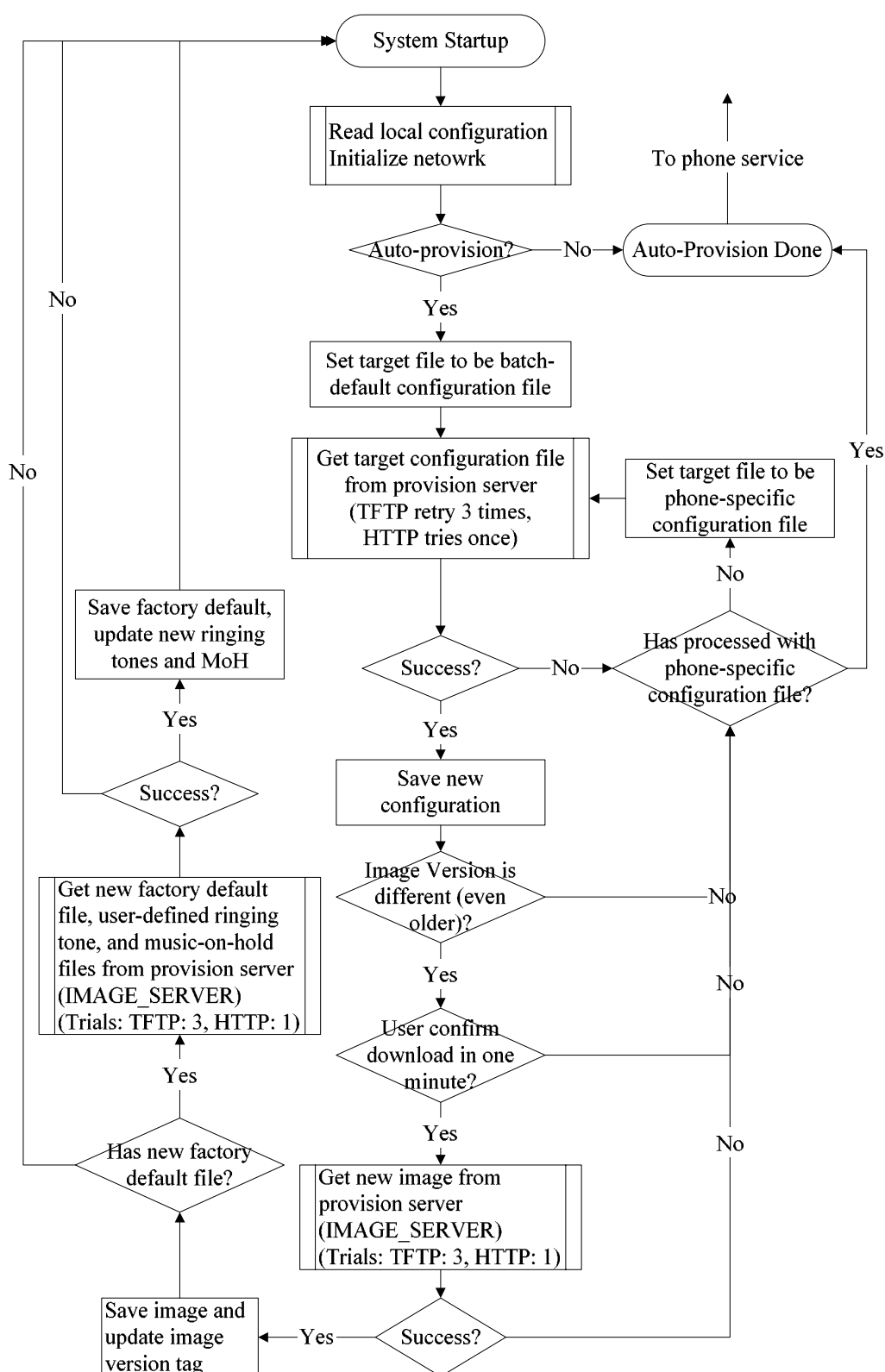
That is, the phone will first load configuration from flash ROM, then download the “Batch default” configuration file from provision server and apply it (if applicable) immediately. Finally, it will try to download the “Phone-specific” configuration file from the provision server (The server location and Phone-specific file may have been changed after applying “Batch default”) again. If the customized file could be fetched from provision server, the setting read from the “Customized” file will overwrite the settings

from “Batch settings” and those read from flash ROM.

- To configure auto-provisioning by web browser:
 - a. Go to 『IP SIP Phone』 / 『Auto Provision』



● Auto-Provision Flow



We suggest that you use the default configuration file to define values for SIP parameters that are common to all phones. Doing so will make controlling and maintaining your network easier. You can then define only those parameters that are specific to a phone in the phone-specific configuration file. Phone-specific parameters should be defined only in a phone-specific configuration file, or they should be manually configured.

12.5. Soft-Switch (PBX) Feature Access

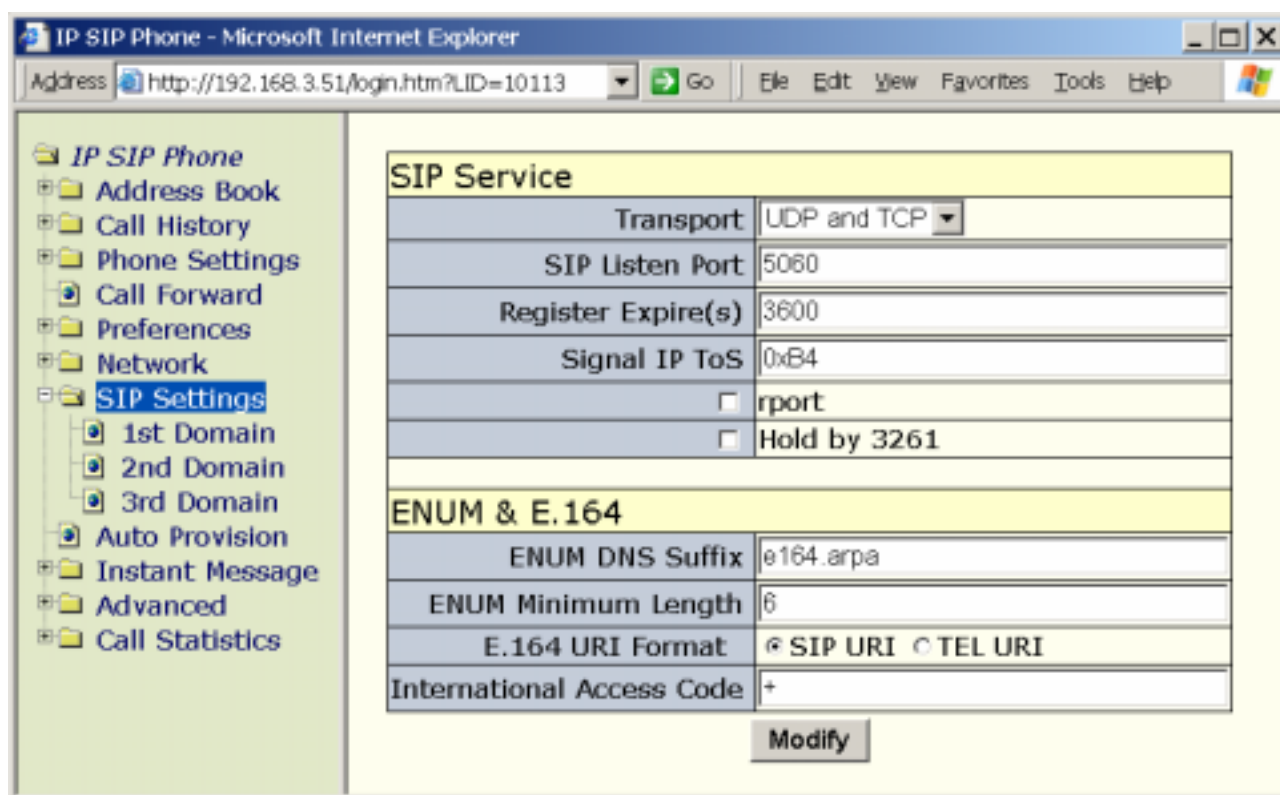
Many Soft-switches, including IP-PBX and SIP proxies implement various proprietary features such as group pick up or call return. Most often, end users on such systems could gain access to these supplant features by dialing special numbers such as “*69” or “#7”.

The star sign, ‘*’, and the pound sign, 【#】, bear special meaning on *IP SIP Phone* (where a dialing string starts with a ‘*’ denotes LAN dialing, whereas a dialing string starts with a 【#】 denotes ENUM dialing), but it reserves “*”, “#”, “*0”, “#0”, “#1”, “#2”, “#3”, “#4”, “#5”, “#6”, “#7”, “#8”, “#9” and all dialing strings starting with a star sign except for “*XXX” where XXX is between 1 and 255 (denotes LAN dialing) for server features access. ***You could send all LAN dial strings “as is” by disable this feature from 【MENU】 => “5.Preferences” / “7.Dial plan” / “3.LAN dial”. Alternatively, you may disable this feature by pointing your web browser to your terminal and go to 『IP SIP Phone』 / 『Preferences』 => “LAN Dial”.***

Preferences	
<input type="checkbox"/>	Auto Answer
<input checked="" type="checkbox"/>	Call Waiting
<input checked="" type="checkbox"/>	Auto Hold on Call Switch
<input checked="" type="checkbox"/>	Follow Redirect
<input checked="" type="checkbox"/>	Message Alert
<input checked="" type="checkbox"/>	LAN Dial
Dial Timeout(s)	30
Hold Recall Alerting Timeout(s)	60
Inter-digit Timeout(s)	4
Dial Key	#
Auto-redial	
Stop Criterion	On Ringing
Retry Interval(s)	15
Total Duration Time(s)	1800
<input type="button" value="Modify"/>	

Note, ENUM dial strings must longer than the minimum valid digit length configured in 【MENU】 => “6.Networ” / “2.SIP settings” / “4.ENUM &E.164” / “2.Min length” (web page

『IP SIP Phone』 / 『SIP Settings』 => “ENUM & E.164” / “ENUM Minimum Length”), which default is 6-digits, and only consists of digits (optionally a leading ‘+’). Therefore, any string starts with a ‘#’ but is not recognized as ENUM dial string will dial “as is”, too.



Moreover, *IP SIP Phone* users can also employ the **【FLASH】** key as well. During dialing phase, whenever the user press **【FLASH】** key, the system will interpret it as an ampersand, ‘&’, and put it into dialing string. For example, if the group pick-up access code is “&7”, user could dial **【FLASH】** +’7’ to gain access to such server feature.

For most servers (IP-PBXs), the feature access code is configurable; please consult to system administrator for prefix reconfiguration.

This table summarizes the heuristics taken by *IP SIP Phone* (suppose your SIP domain is “SIP.isp.com” and your phone resides on “192.168.1.xxx” network segment).

Category	Explanation	Example
Call Return	“*69” => send as is to activate server SIP supplementary service: Call return.	*69 -> *69@SIP.isp.com Note: the server access code to activate “Call Return” feature is configurable from 【MENU】 => “5.Preferences” / “7.Dial Plan” / “3.Call Command” / “Call Return”, or from web page 『IP SIP Phone』 / 『Preferences』 / 『Call Command』 => “Call Return”.
LAN dialing	1. Pattern: “*ddd”, where “ddd” is between 1 and 255.	1. *10 => sip:192.168.1.10 2. *020 => sip:192.168.1.20

	<p>2. This is a short-hand to directly dial the IP of a SIP CPE residing on the same LAN segment.</p>	<p>3. *069 => sip:192.168.1.69</p> <p>4. *50**3000 => sip:192.168.1.50:3000</p> <p>5. *1*20**9999 => sip:192.168.1.20:9999</p>
ENUM Dialing	<p>1. Pattern: #NNNN[N]*.</p> <p>2. The dialing string must longer than the minimum valid digit length configured in 【MENU】 => “6.Network” / “2.SIP Settings” / “4.ENUM & E.164” / “2.Min length” (which default is 6-digits), and only consists of digits (Web page 『IP SIP Phone』 / 『SIP Settings』 => “ENUM & E.164” / “ENUM Minimum Length”.</p>	<p>1. #5639025 => tel:+5639025</p> <p>2. #3-5639025 -> tel:+35639025</p> <p>3. #(86) 3 5639025 => tel:+88635639025</p>
Soft-switch supplementary service activation	<p>1. *, #</p> <p>2. Any string starts with a ‘*’ and is not recognized as LAN dialing.</p> <p>3. Any string stars with a ‘#’ and is not recognized as ENUM dialing.</p> <p>4. Any string starts with a ‘&’ (flash).</p>	<p>1. * => sip:*@SIP.isp.com</p> <p>2. ** => sip:**37@SIP.isp.com</p> <p>3. *311 => sip:*311@SIP.isp.com</p> <p>4. #72 => sip:72@SIP.isp.com</p> <p>5. &911 => sip:&911@SIP.isp.com</p>

13. NAT Traversal

If this terminal locates within a local area network and you want to place a call to public internet, you must configure your terminal to traverse the NAT and firewall it currently behind.

To learn determine whether you resides on public internet or local area network, please click 『IP SIP Phone』 to show the current Host IP:

Service Domain Status		
1st Domain	Registered	Albert<sip:70102@fwd.pulver.com>
2nd Domain	N/A	91888<sip:albert.chang@iptel.org>
3rd Domain	N/A	

Active Network Status	
WAN Link	Up
LAN Link	Down
Host IP	192.168.3.51
Network Mask	255.255.0.0
MAC	000DC300EF51
Primary DNS Server	192.168.3.253
Alternate DNS Server	0.0.0.0

System Information	
CPU	ARM940T 100M
DSP	DSP1600
Memory	16 MB
Flash ROM	2 MB
Serial Number	IPRv2-RD-0054

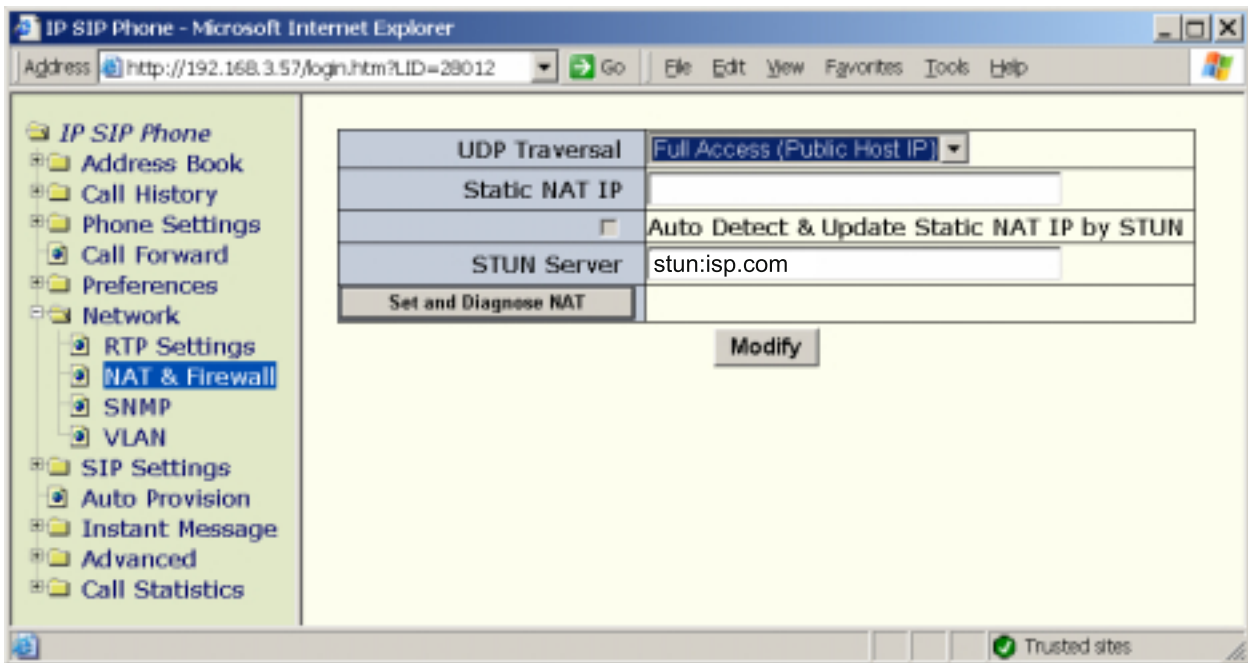
Version	
Product	IP SIP Phone/1.2.1
Software	1.2.1.3
Firmware	1.1.9.3
Hardware	1.0

If your host IP is within any of the listed ranges, then your terminal resides on LAN, otherwise, it locates on public internet.

- 10.0.0.0 ~ 10.255.255.255 (10.xxx.xxx.xxx).
- 172.16.0.0 ~ 172.31.255.255 (172.16.xxx.xxx ~ 172.31.xxx.xxx).
- 192.168.0.0 ~ 192.168.255.255 (192.168.xxx.xxx).

Note: if your host IP is 127.0.0.1, you should reconfigure your network to acquire a valid IP.

13.1. Public Internet Configuration



- UDP Traversal: Full Access (Public Host IP)

13.2. LAN Configuration to Traverse NAT and Firewall

There are basically two options for CPE to traverse NAT and Firewall:

- Option 1: Set up a static route in the NAT gateway (Recommend).
- Option 2: Use STUN to measure out ports.

Please adopt suitable option based on your network configuration.

Note: some SIP ISPs may provide SIP-aware routers (NAT/Firewall) for their customers. When you use a SIP-aware router, NAT detection should be set to "Off" as if you were on the public internet and the configuration is the same as "Public Internet Configuration": set 『UDP Traversal』 to be 『Full Access (Public Host IP)』, which is to say that you do not have to worry about the NAT, nor firewall.

13.2.1. Static NAT Route

If you have access to your NAT/Firewall, you may contact your network administrator to configure your NAT/Firewall route.

- Contact your network administrator and acquire consecutive UDP ports (at least 5 UDP ports in case of *IP SIP Phone*) mapped from the NAT to your terminal. For example:
 - NAT IP = 218.81.107.51 (you may detect your NAT IP by clicking 『Diagnose NAT』)

- The network administrator has mapped 7 consecutive UDP ports, 45700 ~ 5706, from NAT to your terminal, which the terminal IP is 192.168.3.57.

Note 1: Since the network administrator has to configure the NAT/firewall to map those UDP ports to your terminal statically, thus you should use static IP instead of DHCP as your network configuration. Otherwise you take the risk that the ATA would get a different IP from the currently set into the mapping when it reboot and re-get its IP by DHCP.

Note 2: If there are several terminals reside under the same NAT, their NAT port mappings must not be overlapped since they all share the same NAT resource!

- Configure RTP ports

The screenshot shows a web browser window titled "IP SIP Phone - Microsoft Internet Explorer". The address bar shows "http://192.168.3.57/login.htm?LID=28012". The left sidebar contains a tree view with the following items: IP SIP Phone, Address Book, Call History, Phone Settings, Call Forward, Preferences, Network (expanded), RTP Settings (selected), NAT & Firewall, SNMP, VLAN, SIP Settings, Auto Provision, Instant Message, Advanced, and Call Statistics. The main content area displays the "RTP Settings" form with the following fields:

RTP Port Base	45700
RTP Port Range	6
RTP IP ToS	0xB8
RTCP IP ToS	0x74
<input type="checkbox"/> Symmetric RTP Flow	

Below the form is a "Modify" button. The status bar at the bottom shows "Done" and "Trusted sites".

- RTP Port Base: 45700 (Must be an even number and between 2 and 65534).
- RTP Port Range: 6 (This value must be an even number, and larger than or equal to 2); for *IP SIP Phone*, it should be at least 4 to support two concurrent calls.

Such RTP setting effectively designates 6 consecutive UDP ports ranging from 45700 to 45705 (inclusive) for media transaction.

Note: RTP/RTCP ports are used for media transmission. Those ports must be consecutive and start with an even number. Besides, the total allocated UDP ports SHOULD at least double the max concurrent calls supported by your terminal since each active call takes 2 ports for RTP/RTCP.

- Configure SIP service signaling port:

Take the scenario above as an example:

The screenshot shows the 'SIP Service' configuration page in a web browser. The left sidebar contains a tree view with 'SIP Settings' selected. The main content area has two sections: 'SIP Service' and 'ENUM & E.164'.

SIP Service	
Transport	UDP and TCP
SIP Listen Port	45706
Register Expire(s)	3600
Signal IP ToS	0xB4
<input type="checkbox"/> rport	
<input type="checkbox"/> Hold by 3261	

ENUM & E.164	
ENUM DNS Suffix	e164.arpa
ENUM Minimum Length	6
E.164 URI Format	<input checked="" type="radio"/> SIP URI <input type="radio"/> TEL URI
International Access Code	+

Modify

- Transport: UDP and TCP (or “UDP”, you must include UDP anyway).
- SIP Listen port: 45706

- Assign static NAT IP:

The screenshot shows the 'NAT & Firewall' configuration page in a web browser. The left sidebar contains a tree view with 'NAT & Firewall' selected. The main content area has a table for NAT settings.

UDP Traversal	Static NAT IP/UDP MAP
Static NAT IP	218.81.107.51
<input checked="" type="checkbox"/> Auto Detect & Update Static NAT IP by STUN	
STUN Server	stun:isp.com
Set and Diagnose NAT	

Modify

- Diagnose NAT (optional): you may detect your NAT IP by clicking 『Set and Diagnose NAT』.

Note, this diagnosis utilize STUN server, you must have assigned a valid/viable STUN server first.

- Static NAT IP: Fill in the acquired NAT IP from network administrator, such as 218.81.107.51 mentioned above.
- UDP traversal: Static NAT IP/UDP Map

Note, if your NAT equipped with no fixed IP, such as those NATs dial into WAN by PPPoE, then you must synchronize the NAT IP currently set into IP SIP Phone whenever your NAT IP changed (such as NAT re-dialup). By default, the system will regularly (every 10 minutes) auto-detect the NAT IP (if the 『STUN server』 set a valid and viable server) and notify you on the 『IP SIP Phone』 status page whenever it determines a mismatch between the static NAT IP you configured and the one it auto-discovered by STUN protocol.

Alternatively, you may enable 『Auto Detect & Update Static NAT IP by STUN』 to auto-refresh the newly changed NAT IP into 『Static NAT IP』 field whenever it detects an inconsistency.

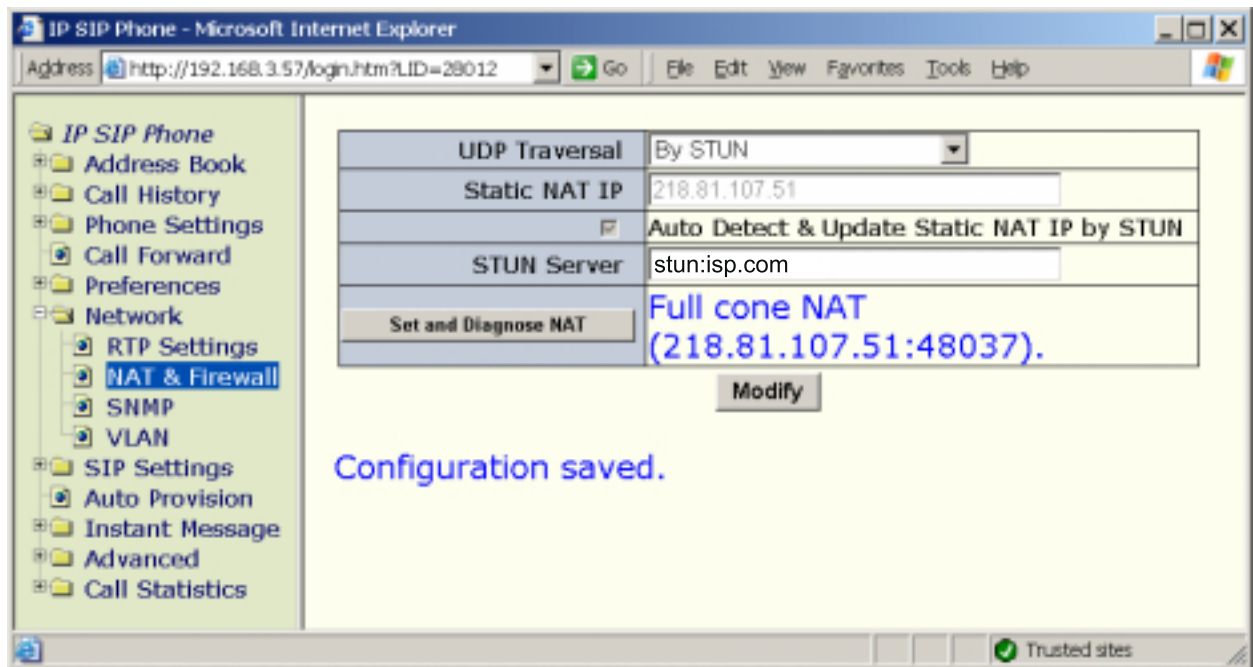
13.2.2. NAT Traversal by STUN

Setting up the NAT router is impossible in many cases, and new equipment may be too expensive. For these environments, "Simple Traversal of UDP Through NATs" (STUN) has come to rescue. To enable STUN, you must acquire the IP of your served STUN server and diagnose your NAT to see if STUN is viable.

- To determine whether you are currently NATed or behind a firewall:
 - STUN server: configure your STUN server, such as "qt.com" (IP SIP Phone will try to locate the STUN server of "qt.com" domain via DNS SRV query (_stun._udp.qt.com) or you may explicitly specify the STUN server as "STUN.qt.com" or 61.222.26.124. You can enter both dotted IP and DNS record.
 - Click 『Set and Diagnose NAT』 to detect your NAT type. If the diagnosis is one of the following, you could consider to enable STUN as a way to traverse NAT; otherwise you should adopt option 1- 『Static NAT Route』.
 - (a) Public internet (neither NAT nor Firewall is involved).
 - (b) Symmetric UDP Firewall
 - (c) Full cone NAT
 - (d) Restricted cone NAT
 - (e) Port restricted cone NAT.

Note: The NAT bundled with Windows 2000/XP/2003 can NOT be traversed by STUN protocol, even though it may wrongly detected by STUN as Port-restricted NAT.

- Activate STUN Mode



- STUN server: Enter a functional and reachable STUN server IP for STUN to work.
- UDP Traversal: Enable STUN

Appendix A - Available NTP servers

Service	Level 2 NTP Server	Service Area
<i>North America</i>		
Ontario, Canada: University of Toronto	tick.utoronto.ca tock.utoronto.ca	Eastern Canada
Quebec, Canada: Canadian Meteorological Center	ntp1.cmc.ec.gc.ca ntp2.cmc.ec.gc.ca	Eastern Canada
Ontario, Canada: National Research Council of Canada Service Area	time.chu.nrc.ca; time.nrc.ca	Canada
Saskatchewan, Canada: University of Regina	142.3.100.15: timelord.uregina.ca	Canada
Mexico: Audiotel office	ntp2a.audiotel.com.mx; ntp2c.audiotel.com.mx; ntp2b.audiotel.com.mx	Avantel, MCINet, and Mexico
Santa Cruz, CA: Scrutz-net, inc.	165.227.1.1: ns.scrutz.net	Western U.S.
San Diego, CA: UCSD Academic Computing Services / Network Operations	132.239.254.49: ntp.ucsd.edu	CERFNET; NSFNET, SDSC region, and nearby
Quincy, California	ntp1.mainecoon.com; ntp2.mainecoon.com	North America
Newark, DE: University of Delaware	128.175.1.3: louie.udel.edu	CAIRN, Abilene, and vBNS
Atlanta, GA: CNSG	ntp.shorty.com	Southeast U.S.
Kennesaw, GA: PeachNet	rolex.peachnet.edu; timex.peachnet.edu	PeachNet (Georgia) and Southeast U.S.
Urbana-Champaign, IL: University of Illinois	ntp-0.cso.uiuc.edu; ntp-1.cso.uiuc.edu; ntp-2.cso.uiuc.edu	CICNET, Midwest, and NCSA region
Chicago, IL: Argonne National Laboratory	ntp-1.mcs.anl.gov; ntp-2.mcs.anl.gov	NSF / ANSNet, CICNet, NetIllinois, and ESNet
West Lafayette, IN: Purdue University	gilbreth.ecn.purdue.edu; harbor.ecn.purdue.edu; molecule.ecn.purdue.edu	NSFNET and CICNET area
Manhattan, KS: KansasNet OnLine Services	199.240.130.1: ntp1.kansas.net 199.240.130.12: ntp2.kansas.net	Central U.S. and Great Plains
Boston, MA: UMass-Boston CS	158.121.104.4:	New England

dept.	timeserver.cs.umb.edu	
Minneapolis / St Paul, MN: University of Minnesota	ns.nts.umn.edu; nss.nts.umn.edu	CICNET region
Columbia, MO: University of Missouri-Columbia	128.206.206.12: everest.cclabs.missouri.edu	MOREnet
Omaha, NE: Radiks Internet Access	205.138.126.83: allison.radiks.net	Midwest U.S.
Las Vegas, NV: University of Nevada System Computing Services	131.216.1.101: cuckoo.nevada.edu	NevadaNet, NSFNET, and SDSC region
Las Vegas, NV: UNLV College of Engineering	tick.cs.unlv.edu; tock.cs.unlv.edu	Sprintnet
New York City, NY: Columbia University	ntp.ctr.columbia.edu	Sprintlink and NYSERnet
Ithaca, NY: Cornell University	192.35.82.50: ntp0.cornell.edu	NSFNET and NYSER region
New York, NY: Morningside Campus, Columbia University	sundial.columbia.edu	NYSERnet
New York City, NY: Columbia University Computer Science Department	timex.cs.columbia.edu	PSINET; NSFNET, and NYSER region
Norman, Oklahoma: University of Oklahoma	129.15.22.8: constellation.ecn.uoknor.edu	Midnet
Coos Bay, OR: Koala Computers	207.48.109.6: tick.koalas.com	Northwestern U.S.
University Park, PA: Penn State University	128.118.25.3: clock.psu.edu	Internet2, vBNS, CERFnet(AT&T IP Services), PSC / NCNE, and CASC
Pittsburgh, PA: Pittsburgh Supercomputing Center	128.182.58.100: fuzz.psc.edu	NSFNET and PSC region
Pittsburgh, PA: Carnegie Mellon Electrical and Computer Engineering	ntp-1.ece.cmu.edu; ntp-2.ece.cmu.edu	PREPNET and PSC region
Dallas, TX: Cox School of Business, Southern Methodist University	129.119.80.126: ntp.cox.smu.edu	NSFNET and SESQUI region
Hughes Springs, TX: First National Bank	209.144.20.76: ntp.fnbhs.com	Northeast Texas
Houston, Tx: Baylor College of	128.249.1.10: ntp.tmc.edu	NSFNET and SESQUI

Medicine		region
College Station, TX: Texas A&M University	165.91.52.110: ntp5.tamu.edu	NSFNET, SESQUI region, THENet, and TAMUSDSN
Plano, TX: Greyware Automation Products	tick.greyware.com; tock.greyware.com	South-Central U.S.
Blacksburg, VA: Virginia Tech Computing Center	ntp-1.vt.edu; ntp-2.vt.edu	Southeast U.S.
Arlington, VA: Center for Seismic Studies	140.162.1.3: ntp.cmr.gov	NSFNET and SURA region
Richland, WA: Washington State University	192.31.216.30: clock.tricity.wsu.edu	NSFNET and NorthWestNet
Washington, U.S.: Mill Creek	time.ultimeth.net	Northwest U.S.
Wisconsin, U.S.: Computer Science Department, University of Wisconsin-Madison	ntp1.cs.wisc.edu; ntp2.cs.wisc.edu; ntp3.cs.wisc.edu	U.S. and any
<i>South America</i>		
Buenos Aires, Argentina: Network Access Point	200.49.40.1: tick.nap.com.ar 200.49.32.1: tock.nap.com.ar	Argentina
Buenos Aires, Argentina: Sinectis S.A.	time.sinectis.com.ar	Argentina
Brazil: Brazilian Research Network	200.144.121.33: ntp.cais.rnp.br	Brazil
Venezuela: VELUG, Grupo de Usuarios Linux de Venezuela	150.185.192.250: ntp.linux.org.ve	Arica
<i>Europe</i>		
Zurich, Switzerland: Swiss Fed. Inst. of Technology	129.132.98.11: bernina.ethz.ch	Switzerland and Europe
Location: Copenhagen, Denmark, Europe	clock.netcetera.dk; clock2.netcetera.dk	Denmark, Scandinavia, and Northern Europe
Spain: Altea	slug.ctv.es	Spain
Pieksamaki, Finland: Keski-Savon Oppimiskeskus	tick.keso.fi; tock.keso.fi	Finland
Meudon, France: Observatoire de Paris-Meudon	ntp.obspm.fr	France and Europe
Lyon, France: CISM	ntp.univ-lyon1.fr	France, Switzerland, Italy, and Europe
Paris, France: VIA, Ecole Centrale	138.195.130.70: ntp.via.ecp.fr	France and Europe

Budapest, Hungary: KFKI Research Institute for Particle and Nuclear Physics	148.6.0.1: time.kfki.hu	HUNGARNET
Italy: Net4u Srl, Vercelli, Italy	195.32.52.129: ntps.net4u.it	Italy
Oslo, Norway: University of Oslo	129.240.64.3: fartein.ifi.uio.no	NORDUnet
Oslo, Norway: Alcanet International	time.alcanet.no	Europe
Krakow, Poland: Academic Computer Centre	149.156.4.11: info.cyf-kr.edu.pl	Poland and Europe
Lund, Sweden: Lund Institute of Technology	130.235.20.3: ntp.lth.se	Sweden and NORDUnet
Ljubljana, Slovenia: Institute of Biophysics, University of Ljubljana	biofiz.mf.uni-lj.si 193.2.69.11	Slovenia and Europe
Ljubljana, Slovenia: Hydrometeorological Institute of Slovenia	hmljhp.rzs-hm.si	Slovenia and Europe
Ljubljana, Slovenia: Academic and Research Network of Slovenia	ntp1.arnes.si; ntp2.arnes.si	Slovenia and Europe
Ljubljana, Slovenia: J. Stefan Institute	time.ijs.si	Slovenia and Europe
British Isles		
Dublin, Ireland: School of Mathematics, Trinity College	ntp.cs.tcd.ie; ntp.maths.tcd.ie; ntp.tcd.ie	Ireland and U.K.
Glasgow, Scotland: Strathclyde University	ntp.cs.strath.ac.uk	U.K., Europe, and any
Cambridge, U.K.	ntp0.uk.uu.net ntp1.uk.uu.net; ntp2.uk.uu.net	UUNET (formerly known in the U.K. as PIPEX) and U.K.
Manchester, England: University of Manchester	ntp2a.mcc.ac.uk; ntp2b.mcc.ac.uk; ntp2c.mcc.ac.uk; ntp2d.mcc.ac.uk	U.K.
Buckinghamshire U.K.: Wibble U.K., Aylesbury	tick.tanac.net	Service area: U.K.
Asia		
Moscow, Russia: Landau Institute	193.233.9.7: ntp.landau.ac.ru	Service area: Russia

for Theoretical Physics		
Russia: Pushchino (near Moscow)	194.149.67.130: ntp.psn.ru	Service area: Russia
Chernogolovka, Russia: Chernogolovka Scientific Center (near Moscow)	193.233.46.10: sign.chg.ru	Russia
<i>Far East and Pacific Rim</i>		
Tokyo, Japan: Cyber Fleet, Inc.	203.139.30.195: ntp.cyber-fleet.net	Japan and East Asia
Seoul, Korea: Inet, Inc.	time.nuri.net	Korea, Japan, Hong Kong, and East Asia
Hamilton, New Zealand: The University of Waikato	truechimer.waikato.ac.nz; truechimer1.waikato.ac.nz; truechimer2.waikato.ac.nz; truechimer3.waikato.ac.nz	New Zealand
Singapore and the Philippines	ntp.shim.org	Singapore
<i>Africa</i>		
Lagos, Nigeria: Supernet300	ntp.supernet300.com	Western Africa (primarily Nigerian NITEL backbone)
Pietermaritzburg, South Africa: Natal University	143.128.82.200: ntp.cs.unp.ac.za	South Africa
<i>Australia</i>		
Adelaide, South Australia: The University of Adelaide	129.127.28.4: augean.eleceng.adelaide.edu.au 129.127.40.3: ntp.adelaide.edu.au 203.21.37.18: ntp.saard.net	AARNet
Carlton, Victoria, Australia: eSec Limited	203.21.84.4: time.esec.com.au	

Appendix B – Trouble shooting

1. To verify your network, please go to 『Advanced』, then type a domain name to ping its reachability and/or aliveness, like “yahoo.com”, “iptel.org” or “fwd.pulver.com”. If the response is “Host unreachable”:
 - i. Check your network link, make sure it works normally (RJ-45 jack plugs into the right hole and the LAN indication LED should be on).
 - ii. Check your IP, DNS, and gateway settings on 『Network』 setting page. Note: if you do not statically assign domain name server by picking 『Static DNS Server』, you must enable either 『DHCP』 or 『PPPoE』; otherwise, it will use the static DNS IP configured into the 『Primary DNS Server』 and 『Alternate DNS server』 item.
 - iii. If you reside on LAN without gateway, you should specify “0.0.0.0” as your gateway IP to disable gateway routing rather than assign a non-existent or an invalid 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 an invalid gateway.
 - iv. Go to 『IP SIP Phone』 page to make sure that those 『Active Network Status』 matches those configured. Specifically, if the active DNS is 0.0.0.0, then you may have wrongly configured to 『Static DNS Server』 without setting a valid DNS IP in either 『Primary DNS server』 or 『Alternate DNS Server』 (see point-2 for detail).
2. To verify SIP settings:
 - i. Go to 『SIP Settings』 page:
 - a. Your 『Transport』 setting should include 『UDP』. Verify the network settings to see whether it works normally.
 - b. Your 『SIP Listen Port』 setting must be less than 65536 and greater than 0. We suggest a value greater than 5000 to avoid accidentally conflicting with system service ports. System default is 5060.
 - ii. Go to 『Network』 / 『RTP Settings』:
 - a. Your 『RTP Port Base』 should be an even number ranges between 2 and 65534.
 - b. Your 『RTP Port Range』 setting should be an even number and larger than or equal to 2. At least 4 ports should be configured for *IP SIP Phone* since it has a maximum capacity of two concurrent calls and each call consumes two consecutive UDP port pair (one for RTP and the other for RTCP). We suggest a range value of 6
 - c. The sum of 『RTP Port Base』 and 『RTP Port Range』 must be less than 65536, and must not overlap with 『SIP Settings』 / 『SIP Listen Port』.
 - iii. Go to 『Nth Domain』 and configure
 - a. Authentication
 - b. SIP Address-of-Record

- c. Proxy Server
- d. Registrar

Configure SIP domain applied from service provider as appropriate. To apply for a public domain account, you may go to www.freeworlddialup.com or www.iptel.org.

3. Check for NAT and Firewall settings:

Go to 『Network』 / 『NAT & firewall』, enter the STUN server IP provided by your service provider. For example, you may enter “stun.qt.com” or “qt.com” or “larry.gloo.net” for test purpose.

Please refer to chapter 6 – 『NAT Traversal』 to determine the best way to traverse your network address translator or firewall deployed by your ISP or company.

4. Experience long-delay or poor response time on making calls (roughly 10 seconds):

The most possible cause is that you have enabled STUN to traverse NAT, but the STUN server is either down or unreachable.

To verify the exact reason, please go to 『Network』 / 『NAT & firewall』 and click 『Set and Diagnose NAT』 to figure it out. If the result is 『Firewall blocks UDP』 and you are pretty sure that you are not under such a network condition, you should check for the DNS and STUN server, making sure they work normally by Pinging for their viability. Otherwise, you should pick 『Static NAT IP/UDP Map』 as the way to traverse NAT.

5. After setting up a call, it always disconnects automatically after 32 seconds.

This may be due to the fact that either the involved SIP proxy servers and / or the terminals (either the caller or callee) are behind NAT, such that the ACK signal cannot be transmitted through the signaling path.

If only the terminals are behind the NAT, please refer to previous parts for NAT & firewall traversal. On the contrary, if it is the SIP proxy that is behind NAT, which has a private unroutable IP address, then it hardly leaves room to solve this problem except for direct IP dialing (because most proxy servers demand record-route, but it fails if any of the involved servers bear private IP).

6. Experience one-way voice (or no voice could be heard by either party) after call setup.

This is because one of the terminals resides behind NAT and it does not specify a way to traverse it reliably. Normally, the one who cannot hear voice mainly because it specifies a private IP address to receive voice media on call setup phase; as a result, the peer starts to send voice media to this unroutable private IP address and the voice packets will get lost since private IP are not routable. To verify the real cause:

- i. If both parties have connected and engage in calls already, please go to 『Call Statistics』 / 『Channel Info』 page to view the following information and check

whether they convey private IPs (*alternatively, you may activate by alternatively, you may press DSS key **[F2]**, which default maps to “Channel Info”, to activate **『Channel Info』***):

- a. From(Contact)
- b. Media Session => both local and remote RTP session.
- ii. If you have hanged up the call, please go to **『Call Statistics』** / **『Call Detail』** to check for the following information and figure out whether they convey private IPs (*alternatively, you may activate by alternatively, you may press DSS key **[F3]**, which default maps to “Call Detail”, to activate **『Call Detail』***):
 - a. From(Contact)
 - b. Media Traffic / RTP Session => both local and remote RTP session.
- iii. The format of **『Contact』** is xxx.xxx.xxx.xxx:port, which specifies the peer's contact information for SIP signaling.
- iv. The format of RTP/RTCP session comes in the format “xxx.xxx.xxx.xxx:RTP/RTCP”, which specifies the IP of CPE and the UDP port used for RTP and RTCP session.

For *IP SIP Phone*, if either side is behind NAT and conveying its private IP, please specify a way to solve this problem from page **『Network』** / **『NAT & firewall』**, and refer to chapter 6 – **『NAT Traversal』** to determine the best way to traverse your network address translator or firewall deployed by your ISP or company.

7. Sometimes there would be only one party be held successfully on conference mode when the conference master presses **【HOLD】** :

This happens only on some SIP proxy server implementations, such as the VOCAL server (both 1.4 and 1.5) from www.vovida.com. The problem is due to the fact that those implementations will increase their “nonce” on authorization (www-authorization), therefore forbid multiple concurrent transactions originated from the same host.

The same problem happens on those servers when you have multiple calls on hold!

8. Direct IP dialing or LAN dialing cannot work:

Most often it is due to the SIP service port does not match. For example, if your dial string is something like “*100”, then the default SIP service port of the called party is UDP 5060. If the called party does not run its SIP service port on UDP 5060, then it definitely will not ring. To solve it, please dial the port as well if the called party listens on port other than UDP 5060. For example, dial “192*168*1*200**6666” to reach “192.168.1.200:6666” (you first dial the IP dial prefix to activate IP dial plan as appropriate).

IP SIP Phone by default will always listen on UDP port 5060 in addition to the user configured SIP Listen port on **『SIP Settings』** page.

9. Received “Unsupported Media” or “Not Acceptable” while making calls.

This message will be shown if both sides fail to negotiate a CODEC for media session. The

failure cause is due to the mutual exclusion of both parties' CODEC capabilities. For example, if you specify explicitly to use only G.723.1 for voice stream whereas the peer is only capable of G.711, then the conversation cannot proceed.



















To ensure the phone will gracefully fall back to G.711, either party of the call should not disable G.711 CODEC. To check CODEC settings on *IP SIP Phone*, please go to 『Advanced』 / 『CODEC』 page to check for 『Preference』 section. If you prefer to use G.729 or G.723.1 for voice compression, consider lower the priority of G.711 rather than disabling them.

10. I can call others, but some of them or all others can not ring me.

Possible cause:

- i. You are in All Call Forward mode (You could hear stuttering dial tone while picking up handset).
- ii. Both of you are under on the same NAT, and either one of you employs 『Enable STUN』 or 『Static NAT IP/UDP Map』 to traverse NAT. This is largely because some NATs will not loop packets back if the source and destination are on the same NAT.
- iii. You use 『Static NAT IP/UDP Map』 to traverse NAT but the assigned static NAT IP on 『NAT & Firewall』 page does not match the real IP of your NAT. (This is possible especially when your NAT does not come with a fixed IP but employ PPPoE or DHCP to gain access to WAN instead). By default, *IP SIP Phone* will auto-detect the NAT IP (if the STUN server configured into 『NAT & Firewall』 page is viable) and notify you whenever you hit the 『Network』 / 『NAT & Firewall』 page if it detects a mismatch between the static NAT IP you configured and the one it auto-discovered by STUN protocol. Alternatively, you enable the option – 『Auto Update Static NAT IP by STUN』 to auto-refresh the newly changed NAT IP whenever it detects an inconsistency.

Appendix C – Tones

TONE	Usage
<i>NULL TONE</i> 	Stop tone playing
<u>DIAL TONE</u>  Dialtone.wav	Dial tone
<u>RECALLDIAL TONE</u>  Re-call tone.wav	Dial tone on call transfer
<u>ALERT RING TONE</u>  Alerting Tone.wav	Ringing on incoming calls
<u>RINGBACK TONE</u>  RingbackTone.wav	Ring back
<u>BUSY TONE</u>  Busy.wav	Peer busy
<u>REORDER TONE</u>  Reorder.wav	Network failure
<u>FAST BUSY TONE</u>  Fastbusy.wav	Call transfer failed. (600 ms)
<u>FAST BUSY 2 TONE</u>  FastBusy2.wav	1 st digit timeout
<u>CALLWAITING TONE</u>  CallWaiting.wav	Hold recall Conference alerting
<u>COVERAGE TONE</u>  Coverage.wav	Call transfer succeeded
<i>INTERCEPTION ERROR TONE</i> 	X (No voice)
<i>LOCAL ERROR TONE</i> 	X (No voice)
<u>CONFIRMATION TONE</u>  Confirmation.wav	Call-waiting alert
<u>STUTTERDIAL TONE</u>  StutterDialTone.wav	
<u>REGISTRATION FAILURE TONE</u>  RegistrationFailureTone.wav	System busy
<u>CONNECTION LOST TONE</u>  ConnectionLostTone.wav	Peer Hangs up
<u>AUDIBLE RINGING TONE</u>  AudibleRingingTone.wav	
<i>CALLWAITING INSIDE TONE</i>	X (No voice)
<i>CALLWAITING OUTSIDE TONE</i>	X (No voice)