PL-300A 1FXS ATA User's Manual

<Version: V2.0 >



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1 Overview

The popularization of the Internet drives the rapid development of a wide variety of IP-based applications. The IP telephone technology has become the major means for operators to develop voice services now. Especially, IP technology becomes the core of the next generation network (NGN), so the IP-based voice technology will keep soaring speeding the future and become the No. 1 choice of new operators in exploring services.

As an Integrated Access Device, the upstream port of the ATA can be directly connected to the IP network, to provide basic accesses for POTS users. This user port gateway can support four telephone lines. It is also applicable for small-size enterprises and IP telephone bars.

2 Packing

The ATA is packed with color chassis. Upon receiving the product, please confirm whether the fittings are complete. The packing box contains a set of IAD, 1 piece of RJ45 network cable, one IAD power adapter and user's manual.

3 Safety Instructions

To ensure your safety and safe use of this product, please pay attention to the following items:

- **n** Follow the instructions in the user's manual.
- **n** Keep the device far away from chemicals and regent.
- **n** Store/use the equipment in dry and well-ventilated environment.
- **n** Never open the chassis lest the device is short-circuited or damaged.

4 Introduction to ATA

The IAD works with the most popular LINUX embedded operation system and has special CPU and DSP compression algorithms, featuring universal functions and applicable to a wide variety of needs.

Basic features:

- n One 10/100 BASE-T WAN port, used to connect broadband data network
- n 1 analog loops starts the FXS interface (RJ-11), used to connect 1 telephones
- n Supporting DHCP Client or static IP address allocation plan

- n Supporting 802.1Q VLAN and VLAN Tag
- n Mute compression and comfort tone generation technology ensure clear conversation quality.
- **n** Self-adaptive jitter cache ensures smooth voice function
- n Lost-packet compensation guarantee mechanism provides a better voice quality.
- **n** Built-in Internet gateway function
- n Supporting NAT (Network Address Translation) and NAPT
- **n** Supporting remote configuration of Web mode and remote software downloading/upgrading

5 Performance Indices

Description of Product Mode	1						
ATA-100	1FXS port IP voice gateway, SIP protocol						
Physical Specifications							
Size	90mm (L) × 65mm (W) × 25mm (H)						
Power supply	AC/DC power adapter, 12V DC DC input: 12V DC/0.8A						
Power consumption	< 10W						
Weight	About 0.1 kg						
Reliability	System availability > 99.999%, MTBF > 100,000 hours, MTTR < 5 min						
Ambient requirements							
Working temperature	0°C ~ 50°C						
Storage temperature	-10°C ~ 50°C						
Relative humidity	5% ~ 95%, non-condensing						
Technical Specifications for	the interface attribute gateway						
Supporting SIP protocol							
Mute processing/four wave p	processing						
RTP/RTCP voice channel							
Voice compression algorithm	n G.729, G.711, G.723 and G.726						
Analog voice port (FXS), 1 p	ports						
Signal format: DTMF							
Echo suppression: G.165/G.168							
DTMF signal detection/gene	DTMF signal detection/generation						
Compatible to the Internet protocols, such as TCP/IP, UDP, ARP, TFTP and ICMP							

Supporting SNMP Version II

Compatible to IEEE 802.3 10BASE-TX Ethernet

Compatible to IEEE 802.3u 100BASE-TX fast Ethernet

6 Networking Mode



Fig. 1

6.1 Typical Applications

The IAD user gateway integrates the Internet gateway and VoIP gateway into a box. They can use category-5 cables to connect the ISP switch (as shown in Fig. A). Four FXS can be connected to Four ordinary analog telephone sets to provide conversation based on IP network.



Internet + VOIP System Architecture for Small-size Enterprise/Branch (Copper Cable DSL)

Fig .A

7 Appearance Description



WAN ----RJ45 PHONE----RJ11 Power----12V 1A Power Adapter Reset --- button

8 Configuration Description

The IAD provides two ways to modify WEB parameters: through LAN interface and WAN interface. Below describes how to enter the WEB interface configuration parameter (refer to Chapter 9 *WEB configuration interface description*) through WAN interface (refer to 8.1).

8.1 WAN Configuration Environment

- **n** Configure the "TCP/IP Protocol" of PC according to Fig. 1 with the PC and WAN interface in the same network segment.
- **n** Configure the device according to Fig. 2.
- **n** Use straight-through cables in the figure.
- **n** After configuration, input the IAD default IP address in IE address bar. Each IAD will be allocated with an initial Wan IP address before delivery, assumed to be 192.168.1.200.

Internet 协议(TCP/IP)	属性 ? 💈									
常规										
如果网络支持此功能,则可以家 您需要从网络系统管理员处获得	灰取自动指派的 IP 设置。否则, 影适当的 IP 设置。									
○ 自动获得 IP 地址 @)										
┌──④使用下面的 IP 地址(S):										
IP 地址(I):	192 .168 . 0 .115									
子网掩码(U):	255 . 255 . 255 . 0									
默认网关 (2):	192 . 168 . 0 . 1									
○自动获得 DMS 服务器地址	(<u>B</u>)									
┌──③使用下面的 DNS 服务器地	址(2):									
首选 DNS 服务器(P):	202 . 96 . 134 . 133									
备用 DNS 服务器(A):	202 . 96 .128 .68									
	高级 (V)									
	确定 取消									

Fig. 1





9 Configuration in WEB Mode

9.1 LOGIN WEB

Through IAD's Wan IP to login web, Default Admin Username: admin , Default Admin Password: 888888 , Default User Username: user , Default User Password: 888888.

When login as admin user, will see like figure 3:



Fig.3

When login as user, will see like figure 4:

Home	^	Home
WAN System Configuration		Welcome to VOIP produce configuration utility. Select from the configuration options in the menu on the left.
LogOut Reboot		System Information
		System Uptime: 0 days, 0h 17m 26s
		Image Version: cos-sz-vr070901-rel1.bin
		NTP time: 08:52PM 09/17/2007 (GMT-8) DST
		LAN IP Address: 192.168.1.156 (Static)
		MAC Address: 00:90:ff:50:10:38 [Licence OK]
		Security: Password installed
		Copyright @ 2003-2008,All Right Reserved.

Fig. 4

9.2 WAN Configuration

Home	^	WAN Status	WAN Setting	VLAN
WAN				
SIP		LAN Status		
CODECS		Interface Status		
AdvanceSet		Enabled:	Yes	
System		Protocol:	Ethernet	
Configuration		Link Status:	100M bps, Fu	III Duplex
Upgrade		Network Settings		
Reboot		IP Address:	192.168.1.15	6
LogOut		MAC Address:	00:90:ff:50:10):38
Logodi		Subnet Mask:	255.255.255.	0
		Default Gateway:	192.168.1.1	
		Host Name:		
		Domain Name:		
		Priority Tag:	Not set	
		Update		

This page is the first page displayed when the device's web pages are accessed. It shows how long the device has been running since its last reboot, the IP address the device is currently using, whether or not the device is password protected, and also displays the main application and downloader application firmware versions. In addition, MAC address of the WAN port, and serial number of the device, if it has one, are also displayed in this page.

Home	^	WAN Status WAN Se	tting VLAN							
WAN		WAN Configuration								
SIP		WAN Configuration								
CODECS		O Llos DUCD to obtain W/	Neonfiguration							
AdvanceSet			w conngui auon							
System		Specify fixed WAN cont	figuration							
Configuration		IP Address: 192.168	.1.156							
Upgrade		IP Netmask: 255.255	.255.0							
Reboot		IP Gateway: 192, 168	- 1 - 1							
LogOut		1 Outomaj. 1911100	• • • •							
		O Automatically obtain DN	O Automatically obtain DNS server settings							
		 Manual DNS server sett 	Manual DNS server settings							
		IP DNS Server:		202.96.134.133						
		IP DNS Server2:								
		Host Name:								
		Domain Name:								
	¥	Save WAN Settings								

This page and its sub-pages are available on device that supports routing/bridging and allows viewing and configuration of the WAN interface status/settings. The default WAN interface IP address is set to 192.168.0.200.Please note that any actions/modifications which alter the topology of the Ethernet Bridge will result in the spanning tree protocol to relearn.

9.3 SIP Configuration

WAN	
SIP SIP Server Configuration CODECS AdvanceSet Primary Server Settings Secondary Server Settings System (Current Server: 192.168.1.232 : 5060 ; Domain: 192.168.1.232) (Current Server: 0 ; Domain:) Upgrade Address: 192.168.1.232 (IP or FQDN) Reboot Port: 5060 Domain Name: Send Registration Request with Expire Send Registration Request with Expire Time 3600 Outbound (IP or Outbound (IP or Proxy IP: FQDN) Outbound Send Registration Request with Expire Time Outbound (IP or Proxy Port: 5082 Proxy Port: 5082 Proxy Port: 5082	

This page allows configuration of the SIP server and endpoint settings.

Enter the address and port value of the SIP server. The address may be an IP address or the name of the server. If no SIP server address is entered, the device will attempt to self provision a SIP server using a DNS query. For this to be successful, ensure that the DNS settings on the device include a DNS server address which is configured with the SIP server address and will respond to the query, and the appropriate domain name of the network.

If you wish to specify a special SIP domain name, you may enter the domain name here. If no domain name is entered, the SIP domain name will be set to that of the network (i.e. that which is obtained via DHCP, or specified on the WAN settings page, section 9.2).

The currently provisioned SIP Server and Domain are displayed beside "SIP Server Settings" for informational purposes.

Select whether or not to send a Registration Request to the SIP server by checking the box next to "Send Registration Request".

For the endpoint, set the dial plan to be used by all lines (refer to "Appendix D" for details on the dialplan representation), and select the transport method to be used for SIP signaling (either UDP or TCP).

For each line on the endpoint (NOTE: The IP Phone has a single line), enter the Line Phone Number, Caller-ID Name, signaling port value, authentication Username and Password, and select if AEC is to be performed on this line.

Press "Save SIP Settings" to save the new values.

Licme	~			Constantiana	A Second Discourse House	at 1 materia	Longer Martin	Missis Maria	51 A.L.	
wen		Delvar	058.	iewenalona	OOD SIGNAIIING	COLD THEY	Service Code	Pric 18 Bock	A THUA	
SIP		SIP Ext	ensions							
CODECS										
AdvanceCet										
Evetern		¥ 1	Frid the SIP							
C milig malicu		¥ \$	See arm Tir	ua ea liPCATE e s	ltu I					
Updrade			Sal – dri ea	är μα=0 101(REC)	2543) in FCP					
Hi and LearCrub	-	\square	enat e Cilco	al Number support;	E 164)					
nodeat		<u>s</u> 1	iend NOTIF	Y for REFER reques	:					
	 send Messade (Vallind indicator (MKV) C JEDCRIDE command Nic Authorization Lieader in re-RECIGNER Check existence of licitizi in INVITE devices on se 									
		ore run	ICI S							
		= ,			olue de sur	-				
				l us usbur	alde Detblic	2				
	-		DIF DESS DI	TT THE VAILUE:	Seconds					
		ş	SIH Keep Mi	və imeriyaluq	Seccies					
		× 1	Coditiona	Call Fe warding Lim	er 15 Beconds					
		-uh s	C jil Timi ∙	Brock:						
		SIP 7	Tin a 🗵	JU ¥ilica m	r ;					
		SIP 7	S Tin a 💵	UU Yilisa m	r ;					
		DIP 1	4 Timer 50	0CD v Illsecon	сз					
		283	e SiFiLxtens	on Setting s						
	~									

This page allows specification of the SIP signaling stack behavior under certain scenarios.

If you wish for the SIP stack to implement reliable transmission of provisional responses according to RFC 3262 (using the PRACK method), check the option "Support PRACK method with provisional response reliability".

If you wish for the SIP stack to include the user parameter "user=phone" in the SIP URI header(s), check the option "Encode SIP URI with user parameter".

If you wish for the SIP stack to send INVITE messages with the "Timer" header field present, check the option "Send INVITE with Timer header value" and enter the Timer header value.

If you wish for the SIP stack to implement a session timer according to "draft-sip-session-timer", select the option "SIP Session Timer value", and enter the session time-out value.

Home	Server	User 1 Exte	nsions OOB	Signalling	ToS/DiffServ	Service Code	Phone Book	VLAN		
WAN SIP	User 1 C	onfiguration								
AdvanceSet System	Line 1	Phone Number	CallerID Name	Port	User Name	Password ^F	Register State			
Configuration	Server	101	101	5060 1	01	•	DOWN			
Upgrade Reboot	Secondary Server			5060						
LogOut	Line1 AEC C	ontrol 🛛 🔽								
	Line1 Gain (Line1 Gain Control								
	Input Gain (Input Gain (-12 ~ 18)db -2 db Output Gain (-12 ~ 18)db -2 db								
	Supplement	ary Service Subs								
Enable Call Waiting (Reject second incoming call)										
	🗸 🗹 Enable	Caller-ID Display								

Press "Save SIP Extension Settings" to save the new values.

This sub-page and the next User2,User3,User4 sub-page allow the user to configure the device with phone number, caller ID, username and password specified by the service provider.

Horne Mark	= Dervel	Jeer	Extensions	008 Signalling	TOQUE M ORIN	Service Ocde	Phone Dock	MLAN
SIP	RTP							
CODECO AdrishosOct Oystam Configuration Upgrade	30 F: E	nd DTMF Eva 10.2939 olynu] Regenerate	nts Tr. F. wii Il no using paylos : SOD CTk T long	23. Jul ≫L				
kebool Logoul	20	re CCU Setting	15					

This sub-page allows configuration of the out-of-band signaling options for SIP. Select whether OOB telephone event signaling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signaling

- 000 9940	Ê	Cerver	User 1	Extensions	CODIC gralline	To\$/DiffServ	CerMce Code	Finche Dock	VLAN
812		TaS/DH	TServ						
00000									
/stanceSel		(a :	Bighalling P	ackets: 10 (2	ex digit byte value)				
0ystem		КY	Mackets:	11 0	ex digit the value)				
Contiguiation				<u> </u>	a contraction of the state of t				
U) quinh		Sav	a tusiciirisa	iv Selli us					
rocder-									
njõul	1								

This sub-page is used to configure the Type-of-Service/Diffserv byte values which are to be used in the IP header of all transmitted SIP signaling packets and RTP packets. The ToS/DiffServ byte values are entered as two-digit hexadecimal values. If no special ToS/DiffServ value is to be used for a particular traffic type, enter "00" or leave the setting empty.

Press "Save ToS/DiffServ Settings" to save these new settings.

Home	Server	User 1	Edensions	008 Signalling	ToSIDIffServ	Service Code	Phone Book	MLAN
SID CODECS	Service	Code C	onfiguration	1				
CODECS AdvanceSet System Configuration Upgrade Rehoot LogOut	Cond Call F Call F Do N Do N Call F Spee - USE	Ittional Call Forwarding Forwarding Forwarding of Disturb C of Disturb C of Disturb C of Disturb C Inanster: Return: rd Dial: 7004 or door	Forwarding: *74 On Busy: *7: On: *7; Off: #7; m: *7; m: *7; #7; #7; #7; #6; *6; *6; *6; *6; *6; *6; *6; *6; *6; *	0# 1# 2# 2# 4# 4# 0 99				

You need to set the service code for using value add service. For example, I set the service code as the above picture.

Condition Call Forwarding: (the call will transfer if no one answer)

- a) Set forwarding number: pick up the handset(press *70# (then you will hear the dial tone(press the forwarding number(then you will here three beeps indicating setting finish.
- b) Set the timeout: go to the "sip extensions(Conditional call Forwarding timer" and

set the timeout before forwarding, unit: second, and then active this option.

c) Then the call will automatically transfer to the forwarding number if no one answers the call in the timeout.

Call Forwarding on busy

Enable call forwarding on busy: pick up the handset(press *71# (then you will hear the dial tone(press the forwarding number(then you will here three beeps indicating setting finish, then all incoming call will forward to this number automatically when the Phone is busy

Call Forwarding: (forwarding always)

- a) Enable call forwarding: pick up the handsetàpress *72# à then you will hear the dial toneàpress the forwarding numberà then you will here three beeps indicating setting finish, then all incoming call will forward to this number automatically.
- b) Disable call forwarding: pick up the handset à press #72# à then you will here three beeps indicating setting finish

Do not disturb: (DND)

- a) Enable DND: pick up the handset à press *74# (then you will here three beeps indicating setting finish(then the phone won't ringing when there is an incoming call.
- **b**) Disable DND: pick up the handset (press #74# (then you will here three beeps indicating setting finish

Call transfer:

- a) Unattended transfer: A call B(B press *98# and then enter C number(then B will hear three beeps indicating the transfer successfully.
- b) Attended transfer: A calls B(B push the hook flash to hold A(B then dial C number to talk with C (then B press *98# to transfer the call(then A can talk with C_{\circ}

Call Return:

Pick up the handset (and then press *69# to dial the latest received call

3 way conference call:

A calls and talks with B (B push the hook flash to hold A B then dial C number to talk with C (B then push the hook flash again to enable three way conference call(C will leave the call is B push the hook-flash again.

Tone

Γ

This sub-page is used to configure Tones which applies in order to acknowledge users.

Dial Tone: The tone you hear when you pick up handset

Recall Dial Tone: The tone when you hold callee and prepare to make another call.

Confirm Tone: The tone after you've set up some service, like DND (Do Not Disturb), Call Forwarding, etc.

Ring Back Tone: The audible ringing you hear before callee picks up and answers your call.

Busy Tone: The tone indicates the number you dialed is in busy now.

Reorder Tone: The tone you hear if you dial an invalid number or the call is not available.

Receiver-Off-Hook Tone: The tone to alert you to place the handset on-hook.

Message-Waiting-Indicator Tone: The tone to notify you to call for message box.

Call-Waiting-Indicator Tone: The tone to make you aware of the second incoming call while you're in conversations.

Ring

Ring Configuratioin					
Default Ring: Call-Waiting Reminder Ring:	ON(1000),OFF(2000),R ON(125),OFF(625),ON(2000),OFF(2875),R				
Save Tone and Ri	ing Settings				

This sub-page is used to configure Ring Cadences required by Rings, Call-Waiting-Indicator, and Distinctive Ring features.

1. Ring Configuration:

Default Ring: Default ring cadence when the phone rings.

Call-Waiting Reminder Ring: Ring cadence of Call-Waiting Reminder Ring.

9.4 CODECS Setting

Home	CODECS
VVAN	
SIP	Selected Silence Suppression
CODECS	✓ G711U ON ✓
AdvanceSet	🗹 G711A ON 🔽
System	G723 ON v
Configuration	6726
Upgrade	
Reboot	✓ G729 ON ✓
LogOut	Packetization 10ms
	Jitter Buffer
	⊙ Adaptive Jitter Buffer: 100ms 👻 (maximum playout delay in milliseconds)
	40ms 👻 (minimum playout delay in milliseconds)
	◯ Fixed Jitter Buffer: 40ms 👻 (fixed playout delay in milliseconds)
	Automatically switch to Fixed Jitter Buffer upon fax/modem tone detection
	Save CODEC Configuration

If the device is running one of the four VoIP applications, this page is available for configuring the audio CODEC parameters, as well as the Jitter Buffer settings for the CODEC decoders.

Enter which CODECs are to be supported.

Select which complex codec is to be supported. Due to memory limitations, it is not possible to select G723 and G729 complex codec at the same time.

Select the packetization period to be used for each selected CODEC. For MGCP, a range of packetizations may be provided for each CODEC (to be advertised in the device's "capabilities" set).

Select whether Silence Suppression is to be supported for each CODEC.

The Jitter Buffer settings apply to all active CODEC decoders. You may choose between an adaptive jitter buffer and a fixed jitter buffer. For an adaptive jitter buffer, choose the maximum allowable playout delay (in milliseconds). For a fixed jitter buffer, choose the fixed playout delay (in milliseconds).

Finally, select whether or not a decoder should automatically switch from an adaptive jitter buffer to a fixed jitter buffer upon fax/modem tone detection. Adaptive jitter buffers are sometimes detrimental to fax transmission over G711 CODECs if they have to adapt too rapidly or too extensively due to inconsistent and widespread packet delays. In these adverse network conditions, a fixed jitter buffer provides superior performance when handling incoming fax transmissions over G711 CODECs.

Press "Save CODEC Settings" to save the new CODEC parameters.

9.5 System Setting

Home	^	Security	HTTPSet	Localization	Handset	Tone/Ring	Ringer Tone	Gain	SNMP
WAN SIP		Set Securi	ity Passwo	ord					
CODECS AdvanceSet		Password is	currently inst	alled					
System Configuration		Account: Old password	4. aq	min					
Upgrade Reboot		New passwo	rd:						
LogOut		Confirm new	password:						
		Change Pa	assword						

1. Set Security Password:

Click "System" item on the top menu.

Click "Security" on the left menu.

In Old password field, input old password , if you have it.

In New password field, input new password .

In confirm password field, input confirm password

Press "Change Password" button to save changes.

Home	^	Security	HTTPSet	Localization	Handset	Tone/Ring	Ringer Tone	Gain	SNMP
SIP		Localiza	tion						
AdvanceSet		Country:	China	♥					
Configuration		NTP Server Time Zone	r: : (GMT+08:0	0) Beijin, Chongqing	ı, Hong Kong,	Urumqi	~		
Reboot		🗌 Adjust	clock for dayli	ght savings					
LogOut		Save	Localization Set	tings					

Timezone:

Find the current time from a list of cities.

Country Caller ID:

The caller ID can find out who's calling you and keep track of how often they call.

Users should set the country field according to their geographical location, otherwise the Caller ID function might not work properly.

1. Timezone setting:

Click "System" on the top menu.

Click "Localization" on the left menu.

In NTP Server field, enter a NTP server IP address . If you want to use the default NTP server, this field should be blank .

In Time Zone drop down menu, select one time zone .

In Adjust clock for daylight savings checkbox, if your country has daylight savings time, you can enable it

Press Save Localization Settings button , then system will redirect to the web page of reset.

Home	Security	HTTPSet	Localization	Handset	Tone/Ring	Ringer Tone	Gain	SNMP
SIP	SNMP	Configurati						
AdvanceSet	SNMP Tr	ap Configuratio	n					
System Configuration	IP ac	dress:		Tra	p Community:			
- Upgrade	SNMP Co	mmunity Confi	guration					
Reboot	Read	l Community:	public	Wri	te Community:	private		
LogOut	SNMP Sy	stem Configura	ation					
	Syste	em Description:						
	Syste	em Objectld:	4528					
	Sa	we SNMP Sett	ings					

This sub-page is used for configuring the device's SNMP manager. Configure the SNMP Trap Host IP address and community, the SNMP read and write community parameters, and the SNMP System Description and System Object ID parameters.

Press "Save SNMP Settings" to apply the new values. These settings will only take effect when the device is rebooted.

9.6 Download Setting

This page provides two options for downloading a new firmware application image to the

device. If you wish to download the new firmware image using TFTP, enter the filename of the ROM image and enter the IP address of the TFTP server on which this file resides.

To initiate the TFTP download process, press "Start TFTP Download."

If the ROM image is stored on the same local machine you are using to access the device's web pages, you can choose to download the ROM file to the device using an HTTP post. Enter the filename of the ROM image or press "Browse" to help locate the file.

To initiate the HTTP download process, press "Start HTTP Download."

If the main application is executing at the time, the device will automatically reboot itself into the downloader mode and begin the download process. If the downloader application is executing at the time, the download process will begin. The download status will be displayed when the image download process is complete. Please refer to Section A "The Downloader Application" for more details on the download process.

Home	a Download
WAN SIP CODECS AdvanceSet System Configuration	TFTP Download method (Select remote TFTP server IP address and filename) TFTP Server IP: Filename: Start TFTP Download
Upgrade Reboot LogOut	■ HTTP Download method (Select filename on local browser machine) Filename:
	URL Download method (Currently tftp://, http:// and https:// are supported) URL: Security Protocol: NONE Start URL Download

HTTP Download method:

When using http to upgrade firmware, it will check firmware version before starting download process.

In Filename field, press Browsing Button.

Press Start HTTP Download button to start downloading file.

If firmware version doesn't fit in with old version, it won't allow updating.

9.7 Advance Setting

Home	^	Dial Plan Adv	anceSet			
WAN						
SIP		Dial Plan Settings				
CODECS						
AdvanceSet		Name Digit	s for matching	Operation	Digits for operation	
System		Digit Map1		dropped 🛛 🖌		
Configuration		Digit Map2		dropped 🔽		
Upgrade		Digit Man3		dropped 🗸		
Reboot		Digit Maps				
LogOut		Digit Map4		aroppea 🛛 💙		
		Digit Map5		dropped 💙		
		Digit Map6		dropped 🛛 🗸		
		Digit Map7		dropped 🖌 🖌		
		Digit Map8		dropped 🖌 🖌		
		Dial Plan:	Т			
		Prefix Add:				
		Local Prefix :				
		Local dial numb	er:	Local dial Pi	efix :	
		In Prefix:		Out prefi	x:	
		notice :The fields	s must be set to	'null' if this field will o	lo nothing.	
		💌 #use as a q	uick dial functio	n	📃 * use as a quick dial function	1
		🔲 To enable #	to be recognize	d as dial number	To enable * to be recognized	l as dial number
		Save SIP Settings				

- Dial Plan such as 12.1.1
- I Digit Map 1: For the No prefixed with 13, 013 will substitute.

eg., when call 1368611111, the send no will be 01368611111

I Digit Map 2: For the No prefixed with 8, 07558 will substitute

But due to the setting of Digit Map 1 & Digit Map 2, this rule

exclude the No prefixed with 86 & 88.

eg., when call 81971111, the send no will be 075581971111

I Digit Map 3: For the No prefixed with 86, 86 will be dropped.

eg., when call 8607552647xxxx, the send no will be 07552647xxxx

I Digit Map 4: For the No prefixed with 88, no change

eg., when call 88011000, the send no will be 88011000

Home	Dial Plan AdvanceSet
SIP CODECS	Use Radius server IP : Port: Password:
AdvanceSet System Configuration	HotLine Setting
Upgrade Reboot LogOut	Pstn dialout NUM Setting
	Encrypt Setting Use Encrypt Encrypt Way SinoVoip Encrypt KEY: Use RTP Encrypt
	Voip server use time limit USE Time Limit:(e.g:20060504)
	Polarity billing Setting
	Save Setting

- 1. Radius billing Setting: Start billing software, Enable "Use Radius server Ip" check button, enter the IP address of target computer
- 2. Use FXS Hotline Number: Enable "Use FXS Hotline Number" check button, when you pick up telephone receiver, the HotLine No will be send automatically immediately.
- 3. PSTN Dialout Number: Lifelive setting
- 4. Encrytp setting: association server setting
- 5. Use Time Limit: Setting the expire date of account No
- 6. Polarity billing setting: Enable "Use Polarity billing" check button will Start Polarity billing function

9.8 Reboot Setting

Home	^	Reboot
WAN		
SIP		Reset
CODECS		You must reboot to make your changes active.
AdvanceSet		Warning! Resetting the system will terminate all network connections and reset your browser connection.
System		
Configuration		 Reset and execute Main Application
Upgrade		Reset and execute Downloader Application
Reboot		
LogOut		Reset

This page provides options for resetting the device. Select whether you wish to reset the device and start executing the main (default) application, or whether you wish to reset the device and start

executing the internal downloader application.

Press "Reset" to reset the device

9.9 Config Backup

Home	^	Backup	Restore	RemoteAutoUpdate
WAN				
SIP		Configur	e File Back	up
CODECS				
AdvanceSet			Backup Cont	figure File
System				
Configuration				
Upgrade				
Reboot				
LogOut				

Backup configuration values of system settings to a file from the device.

Click "Configuration" item on the top menu.

Click "Backup" item on the left menu.

Press Backup Configure File button to save configuration file.

9.10 Config Restore

Home	^	Backup Restore RemoteAutoUpdate
WAN		Configure Destance
CODECS		Configure Restore
AdvanceSet		Configure Restore method (Select filename on local browser machine)
System		Filename: 浏览
Configuration		Start Download
Upgrade		
Reboot		
LogOut		Restore Factory Default
		Start Restore Default Factory

This page is used to restore configuration values of system settings from a previously saved configuration file, or default factory values that stored inside the device.

Restore configuration from a file.

Click "Configuration" item on the top menu.

Click "Restore" item on the left menu.

Press Browsing button to select file by backup from local machine.

Press Start Download button to process downloading file.

After downloading file is finished, the web system will redirect to restart device.

Restore default factory values form device.

Click "Configuration" item on the top menu.

Click "Restore" item on the left menu.

Press "Start Restore Default Factory" button.

After restoring default factory, the web system will redirect to restart device.

10 Dial Model

10. 1 Direct Dial:

Dial through VoIP No: Directly pick up telephone receiver connected with FXS port, and then dial telephone No

11 Troubleshooting

1. Confirm all cables are connected properly.

- 2. Check whether there is the connection through Ping action of PC
- 3. If the fault cannot be solved yet, please contact the technicians.

12 Dial Plan And IVR

12.1.1 Dial Plans

The SIP code will allow provisioning (via web browser) of the dial plan. A dial plan gives the unit a map to determine when a complete number has been entered and should be passed to the gatekeeper for resolution into an IP address. Dial plans are expressed using the same syntax as used by MGCP NCS specification.

The formal syntax of the dial plan is described by the following notation:

Digit ::= "0" | "1" | "2" | "3" | "4" | "5" | "6" | "7" | "8" | "9"

Timer ::= "T" | "t"

Letter ::= Digit | Timer | "#" | "*" | "A" | "a" | "B" | "b" | "C" | "c"

| "D" | "d"

Range ::= "X" | "x" -- matches any digit

| "[" Letters "]" -- matches any of the specified letters

Letters::= Subrange | Subrange Letters

Subrange::= Letter -- matches the specified letter

| Digit "-" Digit -- matches any digit between first and last

Position::= Letter | Range

StringElement::= Position -- matches any occurrence of the position

| Position "." -- matches an arbitrary number of occurrences including 0

String ::= StringElement | StringElement String

StringList::= String | String "|" StringList

DialPlan::= String | "(" StringList ")"

A dial plan, according to this syntax, is defined either by a (case insensitive) string or by a list of strings. Regardless of the above syntax a timer is only allowed if it appears in the last position in a string (12T3 is not valid). Each string is an alternate numbering scheme. The unit will process the dial plan by comparing the current dial string against the dial plan, if the result is under-qualified (partial matches at least one entry) then it will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match) then send the string to the gatekeeper and clear the dial string.

The Timer T is activated when it is all that is required to produce a match. The period of timer T is 4 seconds. For example a dial plan of (xxxT|xxxx) will match immediately if 5 digits are entered, it will also match after a 4 second pause when 3 digits are entered.

D.1 Sample Dial Plans

Simple Dial Plan

This allows dialing of 7 digit numbers (e.g. 5551234) or an operator on 0. Dial plan is (0T|xxxxxx)

Non-dialed Line Dial Plan

As soon as handset is lifted the unit contacts the gatekeeper (used for systems where DTMF detection is done in-call). Dial plan is (x.) i.e. match against 0 (or more) digits. Note: the dot '.'

Complex Dial Plan

Local operator on 0, long distance operator on 00, four digit local extension number starting with 3,4 or 5, seven digit local numbers are prefixed by an 8, two digit star services (e.g. 69), ten digit long distance prefixed by 91, and international numbers starting with 9011+variable number of digits.

Dial plan for this is:

```
(0T|00T|[3-5]xxx|8xxxxxx|*xx|91xxxxxxxxx|9011x.T)
```

12.1.2 IVR

To use IVR, one should pick up the phone, and then dial four consecutive asterisks (****.) And hang up the phone will stop IVR.

Code	Status	User input
****	Menu	Enter choice code.
100#	Network status	None.
110#	DHCP setting	1# to enable
		2# to disable
		or # back to menu
120#	IP address setting	Use "*" to instant of ".", and "#" to end.
		Ex: 172*16*230*227#
		Or # back to menu.
130#	Gateway setting	Use "*" to instant of ".", and "#" to end.
		Or # back to menu.
140#	Net mask setting	Use "*" to instant of ".", and "#" to end.
		Or # back to menu.