

Akuvox SDP-R25 User Manual

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Production Overview

1. Production Description

The Akuvox SPD-R25 is the video door phone, that you can connect with your Akuvox IP Phones for remote unlock control and monitoring. You can operate the indoor handset to communicate with visitors via voice and video, and unlock the door if you wish. It's applicable in apartment, villas, Office, building and so on.



2. Features

- Key Features
 - HD Voice
 - Compatible with Asterisk and Broadsoft platforms
- Physical Features
 - Camera: 2.0 Mega Pixels
 - White balance:Auto

- Lens: 4.0mm/F2.8
- Viewing Angle (Diagonal): 50
- Minimum Illumination: 1 LUX (without LED Illumination)
- LED: 6 LEDs
- Power Requirement: DC12V
- Operating Temperature; -30C ~ 40C
- Weight: 180g
- Size (WXHXD)185 x 68 x 50 mm
- Phone Features
 - Video resolution: 320 x 240, with 20pics per second
 - Wide angle lens and IR LEDs for night vision Crystal sound quality
 - Remote door opening
 - Integrated microphone and speaker
 - Water-proof outdoor unit: IP55
 - Support all the VoIP Phones
- ➢ IP-PBX Features
 - Video Codec: H.264
 - Audio Codec: PCMU
 - VAD, CNG , Echo Canceller
- Network Features
 - SIP v1(RFC2543), V2(RFC3261)
 - Static IP/DHCP for IP conguration
 - 3 DTMF modes: In-Band, RFC2833, SIP INFO
 - HTTP/HTTPS Web Server for Management
 - NTP for Auto Time Setting
 - TFTP/FTP/HTTP/HTTPS client API
- Administration Features
 - Auto provisioning using FTP/TFTP/HTTP/HTTPS/PnP
 - Dial through IP PBX Using Phone Number
 - Dial through IP PBX Using URL Address
 - Conguration Managements with Web, keypad on the phone, and Auto Provisioning
- Security Features
 - Support HTTPS (SSL)
 - Support SRTP for Voice Data Encryption
 - Support Login for Administration
 - Sip Over TLS

Configuration

1. Web Login

1.1. Obtaining the IP address

The Akuvox R25 uses Static IP by default, and the default IP address is 192.168.1.100.

If the IP address is unknown, press the call button when the door phone is initialing, after a short period of time, the phone will announce its IP.

1.2. Login the Web

Open a Web Browser, enter the corresponding IP address. Then, type the default user name and password to log in. The default User Name and Password are as below,

User name: admin

Password: admin



2. Status

Status, including product information, network information and Account information, can be viewed from, Status > Basic.

Basic

Product Information Model

Status

MAC Address Firmware Version Hardware Version

- Network Information
- Phone
- PhoneBook
 - rade
- LAN Sublet Ma LAN Gateway LAN DNS1 LAN DNS2 Brimony NTR
 - Primary NTP Secondary NTP Account Information

LAN Link Status LAN IP Address LAN Subnet Mask

Account1 Account2

Account3

SP-R25 0c:11:05:00:17:5d 25.0.1.27 25.0.0.0.0.0.0.0

DHCP Auto
Connected
192.168.1.40
255.255.255.0
192.168.1.1
192.168.1.1

0.pool.ntp.org 1.pool.ntp.org

101@192.188.1.126 Registration Failed None@None UnRegistered None@None UnRegistered box: 255: Broadsoft Phonebook server address 127: Remote Phonebook URL & AUTOP Manual Update Server URI

URL 63: The rest of input boxes

Help

Max length of characters for input

LogOu

Warning :

Note :

Field Description :

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

3. Language

Web Language can be configured from, Phone > Time/Lang.

					LogOut
► Status	Time/Lang			Help	
Push Button	Web Language		Note :		
Account	Туре	English	Max lengt	h of characters for input	

Select the desire language from the pull-down list of Type. The default language is English.

4. Network configuration

To configure the basic network settings, Go to Network > Basic.

The static IP is set as default, and its IP address is 192.168.1.100.

122 M		
Status	Network-Basic	Help
ush Button	LAN Port	Note
ccount	DHCP	Max length of characters for inpu
Internets	O Static IP	box: 255: Broadsoft Phonebook serve
IELWOIK	IP Address	address
Basic	Default Gateway	127: Remote Phonebook URL & AUTOP Manual Uodate Server
Advanced	LAN DNS1	URL
Phone	LAN DNS2	63: The rest of input boxes
	O PPPoE	Warning :
honeBook	User Name	Field Description :
Jpgrade	Password	
Security		
	Submit Cancel	

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

For advanced settings, go to Network > Advanced

Ne	twork-Adva	nced			Help
utton L	ocal				Note :
t RIP		Max RTP Port	12000	(1024-85535)	Max length of characters for input
k i		Min RTP Port	11800	(1024~65535)	255: Broadsoft Phonebook server
т	R069				address 127: Remote Phonebook URL &
		Active	Disabled	~	AUTOP Manual Update Server
nced		Version	1.0	~	URL
	ACS	URL			53: The rest of input boxes
		User Name			Warning :
look		Password	******		
e Inform	Periodic	Active	Disabled	~	Field Description :
		Periodic Interval	1800	(3~3600s)	
	CPE	URL			
		User Name			
		Password			

Sections	Description
Local RTP	To display and configure Local RTP settings.
	 Max RTP Port: Determine the maximum port that RTP stream can use. Min RTP Port: Determine the minimum port that RTP stream can use.
TR069	To display and configure TR069 settings.
	• Active: To enable or disable TR069 feature.

• Version: To select supported TR069 version (version 1.0 or 1.1)
1.0 01 1.1J.
• ACS/CPE: ACS is short for Auto configuration servers
as server side, CPE is short for Customer-premise
equipment as client side devices.
• URL: To configure URL address for ACS or CPE.
• User name: To configure username for ACS or CPE.
• Password: To configure Password for ACS or CPE.
• Periodic Inform: To enable periodically inform.
• Periodic Interval: To configure interval for periodic
inform.
Note : TR-069(Technical Report 069) is a technical specification entitled CPE WAN Management Protocol (CWMP).It defines an application layer protocol for remote management of end-user devices.

5. Account

To configure your SIP account, go to Account > Basic.

5	Account-Basic		Help
Button	SIP Account		
unt	Status Account	Registration Failed	
ic	Account Active	Enabled V	
anaad	Display Label	101	
anceu	Display Name	101	Note :
ork	Register Name	101	Max length of characters for input
	User Name	101	255: Broadsoft Phonebook serve
	Password	•••••	address
Book	SIP Server 1		127: Remote Phonebook URL & AUTOP Manual Update Server
de	Server IP	192.168.1.126 Port 5060	URL
ue	Registration Period	1800 (30~65535s)	63: The rest of input boxes
ty	SIP Server 2		Warning :
	Server IP	Port 5060	Field Description :
	Registration Period	1800 (30~65535s)	
	Outbound Proxy Server		
	Enable Outbound	Disabled V	
	Server IP	Port 5060	
	Backup Server IP	Port 5060	
	Transport Type		
	Transport Type	UDP 🗸	
	NAT		
	NAT	Disabled V	
	Stun Server Address	Port 3478	
		Foil	
	Submit	Cancel	

Sections	Description
SIP Account	To display and configure the specific Account settings.
	• Status: To display register result.
	• Display Name: Which is sent to the other call party for displaying.
	• Register Name: Allocated by SIP server provider, used for authentication.
	 User Name: Allocated by your SIP server provide, used for authentication.
	• Password: Used for authorization.
SIP Server 1	To display and configure Primary SIP server settings.
	• Server IP: SIP server address, it could be an URL or IP address.

	• Registration Period: The registration will expire after Registration period, the IP phone will re-register automatically within registration period.	
SIP Server 2	To display and configure Secondary SIP server settings.	
	This is for redundancy, if registering to Primary SIP server fails, the IP phone will go to Secondary SIP server for registering.	
	Note : Secondary SIP server is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.	
Outbound Proxy Server	To display and configure Outbound Proxy server settings.	
	An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server.	
	Note : If configured, all SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.	
Transport Type	To display and configure Transport type for SIP message	
	 UDP: UDP is an unreliable but very efficient transport layer protocol. TCP: Reliable but less-efficient transport layer protocol. TLS: Secured and Reliable transport layer protocol. DNS-SRV: A DNS RR for specifying the location of services. 	
NAT	To display and configure NAT(Net Address Translator) settings.	
	• STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues.	
	Note : By default, NAT is disabled.	

For advance account settings, go to Account > Advanced.

Account-Advanced Help sh Button Codecs Note : count Disabled Codecs Enabled Codecs PCMA G723_53 G723_53 G729 G729 255: Broadsoft Phonebook server address dvanced G726-24 >> work G726-24 >> G726-40 < 1 G726-40 < Max Local SIP Port 5062 Min Local SIP Port 5062 Max Local SIP Port 5062 Vice Encryption Vice Encryption NAT VDP Keep Alive Messages UDP Keep Alive Messages Disabled <						
sh Button count sasic dvanced preduces dvanced corre count sasic dvanced corre cone	tus	Account-Advanced				Help
grade Call Field Description : Fueld Description : Field Description :	sh Button count Basic kdvanced twork one oneBook	Codecs PCMA G723_53 G723_63 G729 G726-16 G726-32 G726-40	Enabled Codec PCMU	- 		Note : Max length of characters for input box: 255: Broadsoft Phonebook server address 127: Remote Phonebook URL & AUTOP Manual Update Server URL 63: The rest of input boxes Warning :
Max Local SIP Port 5062 (1024-65535) Min Local SIP Port 5062 (1024-65535) Encryption Voice Encryption Disabled V NAT UDP Keep Alive Messages Disabled V	grade	Call				Field Description :
Min Local SIP Port 5062 (1024~65535) Encryption Voice Encryption NAT UDP Keep Alive Messages Disabled UDP Keep Internation		Max Local SIP Port	506	2	(1024~65535)	
Encryption Voice Encryption NAT UDP Keep Alive Messages UDP Alive Messages UDP Alive Messages UDP Alive Messages UDP Alive Messages	urity	Min Local SIP Port	506	2	(1024~65535)	
Voice Encryption Disabled V NAT UDP Keep Alive Messages Disabled V		Encryption			-	
UDP Keep Alive Messages Disabled		Voice Encryption	Disa	bled	~	
UDP Keep Alive Messages Disabled		NAT		11.4		
		UDP Keep Alive Messages	Disa	bled	×	

Sections	Description
Codecs	To display and configure available/unavailable codecs list.
	Codec means coder-decoder which is used to transfer analog signal to digital signal or vice versa.
	Familiar codecs are PCMU(G711U), PCMA(G711A), G722 (wid-bandth codecs), G723,G726,G729 and so on.
Call	To display and configure call-related features.
	 Max Local SIP Port: To configure maximum local sip port for designated account. Min Local SIP Port: To configure minimum local sip port for designated account.
Encryption	To enable or disabled SRTP feature.
	• Voice Encryption (SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.
NAT	To display NAT-related settings.
	• UDP Keep Alive message: If enabled, IP phone will

send UDP keep-alive message periodically to router to
keep NAT port alive.
• UDP Alive Msg Interval: Keepalive message interval.
• Rport: Remote Port, if enabled, it will add Remote
Port into outgoing SIP message for designated
account.

6. Push Button

To configure Push Button, go to Push Button.

Puch Putton	-			Hala
Tusir Button				Help
Push Button				Note :
	Rey Push Button	1003	r	Max length of characters for inp
DTMF Code		1005	102	255: Broadsoft Phonebook serv
	DTMF Code	2	~	address 127: Remote Phonebook URL 8
Lock Deset				AUTOP Manual Update Server
LUCK NUSCI	Relay action time	1000	~	URL 62: The cert of input heres
Mary Call Tim	i telby action and	1000		oo. merestormpatooxes
Max Call Tim		40		Warning :
	Max Call Time	18	(2~30Minutes)	Field Description :
Push to Hang	up			
	Push to Hang up	Enabled	~	
	Submit		Cancel	
1.1				
1.1				

Sections	Description
Push Botton	To configure the destination number you want to contact with.
DTMF Code	To select the desired DTMF Code
Lock Reset	To set the lock reset time

Max Call Time	To configure the max call time
Push to Hang up	To enable or disable the Push to Hang up function

7. Phone

7.1. Call Feature

Call feature can be configured from, Phone > Call Feature.

			L
Status	Call Feature		Help
Push Button	Call Waiting		New -
► Account	Call Waiting Enable	Disabled V	Max length of characters for input
Notwork	Call Waiting Tone	Disabled 🗸	box: 255: Broadsoft Phonebook server
Network	Auto Redial		address
▼ Phone	Auto Redial	Disabled V	127: Remote Phonebook URL & AUTOP Manual Undate Server
Time/Lang	Auto Redial Interval	10(1~300s)	URL
Preference	Auto Redial Times	3 (1~100)	63: The rest of input boxes
Treference	DND		Warning :
Call Feature	Return Code When DND	486(Busy Here)	Field Description :
Voice	DND On Code		Tield Description .
Key/Display	DND Off Code		
Tanaa	Remote Control		
Tones	Allowed Access IP List		
Dial Plan	Others		
► PhoneBook	Return Code When Refuse	486(Busy Here)	
▶ Upgrade	Auto Answer Delay	0(0~5s)	
 Security 	Submit	Cancel	

Sections	Description
Call Waiting	To enable or disable Call Waiting.
	 Call Waiting Enable: If enabled, it allows IP phones to receive a new incoming call when there is already an active call. Call Waiting Tone: If enabled, it allows IP phones to play the call waiting tone to the waiting callee.

Auto Redial	 Auto redial allows IP phones to redial an unsuccessful call for designated times within designated interval. Auto Redial: To enable or disable auto redial feature. Auto Redial Interval: Determine the interval between two consecutive attempts. Auto Redial Times: Determine how many times to redial. 	
DND	 DND(Do Not Disturb) allows IP phones to ignore any incoming calls. Return Code when DND: Determine what response code should be sent back to server when there is an incoming call if DND on. DND On Code: The Code used to turn on DND on server's side, if configured, IP phone will send a SIP message to server to turn on DND on server side if you press DND when DND is off. DND Off Code: The Code used to turn off DND on server's side, if configured,IP phone will send a SIP message to server to turn off DND on server's side, if configured,IP phone will send a SIP message to server to turn off DND on server's side, if configured,IP phone will send a SIP message to server to turn off DND on server side if you press DND when DND is on. 	
Remote Control	 Remote Control allows specific host to interact with phone by sending HTTP or HTTPS requests. The spec action could be answering an incoming call, hangup ongoing call and so on. Allowed Access IP List: To configure the allowed h address. Note: For now, IP phone can only support IP address, address list and IP address pattern as allowed hosts 	
Others	 Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected. Auto Answer Delay: To configure delay time before an incoming call is automatically answered. 	

7.2. Voice

Voice can be configured from, Phone > Voice

tatus	Voice		Help
ush Button	Echo Canceller		
ccount	Echo Canceller	Enabled V	Max length of characters for input
	VAD	Disabled V	box:
etwork	CNG	Enabled V	255: Broadsoft Phonebook server address
hone	Jitter Buffer		127: Remote Phonebook URL &
	Jitter Type	Fixed 🗸	AUTOP Manual Update Server
Time/Lang	Min Delay	0 (0~1000ms)	63: The rest of input boxes
Preference	Nominal Delay	120 (0~1000ms)	
Call Feature	Max Delay	300 (0~1000ms)	Warning :
Voice	Mic Volume		Field Description :
VOICE		-	

Sections	Description
Echo Canceller	Echo Canceller: To remove acoustic echo from a voice
	communication in order to improve the voice quality .
	 VAD (Voice Activity Detection): Allow IP phone to detect the presence or absence of human speech during a call. When detecting period of "silence", VAD replaces that silence efficiently with special packets that indicate silence is occurring. It can facilitate speech processing, and deactivate some processes during non-speech section of an audio session. It can avoid unnecessary coding or transmission of silence packets in VoIP applications, saving on computation and network bandwidth. CNG (Comfort Noise Generation): Allow IP phone to
	generate comfortable background noise for voice communications during periods of silence in a conversation. It is a part of the silence suppression or
	VAD handling for VoIP technology. CNG, in conjunction with VAD algorithms, quickly responds when periods of silence occur and inserts artificial noise until voice activity resumes. The insertion of
	artificial noise gives the illusion of a constant transmission stream, so that background sound is
	consistent throughout the call and the listener does not think the line has released.

Jitter Buffer	Jitter buffer is a shared data area where voice packets can be collected, stored, and sent to the voice processor in even intervals. Jitter is a term indicating variations in packet arrival time, which can occur because of network congestion, timing drift or route changes. The jitter buffer, located at the receiving end of the voice connection, intentionally delays the arriving packets so that the end user experiences a clear connection with very little sound distortion. IP phones support two types of jitter buffers: fixed and adaptive. Fixed: Add the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones. Adaptive: Capable of adapting the changes in the network's delay. The range of the delay time for the dynamic jitter buffer added to packets can be also configured on IP phones.
Mic Volume	To configure Microphone volume

7.3. Country Ringtone

Country Ringtone can be configured from, Phone > Tone.

Select the desired country ringtone from the pull-down list of Select Country.

					LogOut
► Status					
	Tone		0.	Help	
Push Button	Select Country	China	~		

8. PhoneBook

8.1. Call Log

			.01						1111111
									LogOu
► Status		and and the		Indiate		Min Male			
	Cal	l Log						Help	
Push Button	C	all Histor	v	All	✓ Hand Up				
A	Index	Type	Date	Time	Local Identity	Name	Number	Note : May length of characters for input	
 Account 	1	Dialed	2014-08-25	05:40:55	101@192.168.1.126	Unknown	100@192.168.1.126	hox.	
▶ Network	2	Dialed	2014-08-19	03:56:53	101@192.168.1.126	Unknown	100@192.168.1.126	255: Broadsoft Phonebook server	
	3	Dialed	2014-08-19	02:57:48	101@192.168.1.126	Unknown	100@192.168.1.126	address	
▶ Phone	4	Dialed	2014-08-19	02:57:25	101@192.168.1.126	Unknown	100@192.168.1.126	127: Remote Phonebook URL &	
	5	Dialed	2014-08-19	02:51:02	101@192.168.1.126	Unknown	100@192.168.1.126	AUTOP Manual Update Server	
PhoneBook	6	Dialed	2014-08-19	02:30:33	101@192.168.1.126	Unknown	100@192.168.1.126	URL 63: The rest of input haves	
Logal Desk	7	Dialed	2014-08-19	02:24:35	101@192.168.1.126	Unknown	100@192.168.1.126	03. The rest of input boxes	
LOCAI DOOK	8	Dialed	2014-08-19	02:19:38	101@192.168.1.126	Unknown	100@192.168.1.126	Warning :	
Remote Book	9	Dialed	2014-08-19	02:17:49	101@192.168.1.126	Unknown	100@192.168.1.126		
	10	Dialed	2014-08-19	02:15:33	101@192.168.1.126	Unknown	100@192.168.1.126	Field Description :	
Call Log	11	Dialed	2014-08-19	02:14:24	101@192.168.1.126	Unknown	100@192.168.1.126		
	12	Dialed	2014-08-19	02:13:39	101@192.168.1.126	Unknown	100@192.168.1.126		
LUAI	13	Dialed	2014-08-19	02:11:52	101@192.168.1.126	Unknown	100@192.168.1.126		
Broadsoft	14	Received	2014-08-19	02:11:40	101@192.168.1.126	100	100@192.168.1.126		
	15	Dialed	2014-08-19	02:10:35	101@192.168.1.126	Unknown	100@192.168.1.126		
► Upgrade		Page 1 🗸	P	rev	Next	elete	Delete All		
▶ Security									
occurry									

Sections	Description
Call History	To display call history records.
	Available call history type are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls.
	HangUp: To click to hangup ongoing call on the IP phone.
	Note : For "HangUp" feature, you need to have the remote control privilege to control IP phone via Web UI. Please refer to section "Remote Control" in the Web UI->Phone->Call Feature page.

9. Security

9.1. Web Password Modify

To modify web passoword, go to Security > Basic

	A Contraction		
b. Status			LogOut
Status	Security-Basic		Help
Push Button	Web Password Modify		
► Account	User Name Current Password	admin	Note : Max length of characters for input box:
► Network	New Password		255: Broadsoft Phonebook server
▶ Phone	Confirm Password		address 127: Remote Phonebook URL & AUTOP Manual Update Server
▶ PhoneBook	Submit	Cancel	URL 63: The rest of input boxes
► Upgrade			Warning :
▼ Security			Field Description :
Basic			
Advanced			

Sections	Description
Web Password Modify	To modify user's password.
	 Current Password: The current password you used. New Password: Input new password you intend to use. Confirm Password: Repeat the new password
	Note : For now, IP phone can only support user admin.

9.2. Web Server Certificate

To check or upload your web server certificate, go to Security > Advanced

	Advanced					Help
on	Web Serv	er Certificate				
	Index	Issue To	Issuer	Expire Time	Delete	Note : Max length of characters for inn
	1	Ringslink	Ringslink	Sun Jun 27 07:14:32 2037	Delete	box:
						255: Broadsoft Phonebook serv
E	Web Serv	er Certificate				address
Ŀ	Upload					AUTOP Manual Undate Server
		浏览		Submit Cancel		URL
						63: The rest of input boxes
	l seureer					Warning .
	Client Cer	rtificate				warning .
Ŀ	Index	Issue To	Issu	er Expire Time		Field Description :
E	1					
E	2					
L	3					
	4					
	5					
	6					
l	6					
	6 7 8					
	6 7 8 9					

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

10. Upgrade

10.1. Basic upgrade

To upgrade your device, go to Upgrade > Basic.

IS U	Ipgrade-Basic		Help
Button unt ork eeBook ade sic vanced	Upgrade Firmware Version Hardware Version Reset To Factory Setting Reboot	Submit Cancel 25.0.1.27 25.0.0.0.0.0 Submit Submit Submit Submit	Note : Max length of characters for input box: 265: Broadsoft Phonebook server address 127: Remote Phonebook URL & AUTOP Manual Update Server URL 63: The rest of input boxes Warning : Field Description :

Sections	Description
Upgrade	To select upgrading rom file from local or a remote server automatically. Note: Please make sure it's right file format for right model.
Firmware version	To display firmware version, firmware version starts with MODEL name.
Hardware Version	To display Hardware version.

Reset to Factory Setting	To enable you to reset IP phone's setting to factory settings.
Reboot	To reboot IP phone remotely from Web UI.

10.2. Advanced Upgrade

To do the advanced upgrade for your device, go to Upgrade > Advanced.

Status	Upgrade-Advanced		Help
Push Button Account letwork Phone PhoneBook	PNP Option PNP Config DHCP Option Custom Option Manual Update Server URL	Disabled (128~254) (tftp://192.168.1.34	Note : Max length of characters for input box: 255: Broadsoft Phonebook server address 127: Remote Phonebook URL & AUTOP Manual Update Server URL
Upgrade Basic Advanced Security	User Name Password Common AES Key AES Key(MAC) AutoP Mode Schedule AutoP Immediately Clear MD5 Submit Cancel	Power On V Sunday V 22 Hour(0~23) AutoProvision	83: The rest of input boxes Warning : Field Description :
	System Log LogLevel Export Log PCAP PCAP Others Config File(.tgz)	3 V Export Start Stop Export	

Sections	Description
PNP Option	To display and configure PNP setting for Auto Provisioning.
	• PNP: Plug and Play, once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get Auto Provisioning server's

	address. By default, this SIP message is sent to multicast address.
	224.0.1.75(PNP server address by standard).
DHCP Option	To display and configure custom DHCP option.
	• DHCP option: If configured, IP Phone will use designated DHCP option to get Auto Provisioning server's address via DHCP.
	This setting require DHCP server to support corresponding option.
Manual Update Server	To display and configure manual update server's settings.
	 URL: Auto provisioning server address. User name: Configure if server needs an username to access, otherwise left blank. Password: Configure if server needs a password to access, otherwise left blank. Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file. AES Key (MAC): Used for IP phone to decipher MAC-oriented auto provisioning configuration file(for example, file name could be 0c1105888888.conf if IP phone's MAC address is 0c1105888888). Note: AES is one of many encryption, it should be configure only configure filed is ciphered with AES, otherwise left blank.
AutoP	To display and configure Auto Provisioning mode settings.
	This Auto Provisioning mode is actually self-explanatory.
	For example, mode "Power on" means IP phone will go to do Provisioning every time it powers on.
System Log	To display syslog level and export syslog file.
	Syslog level: From level $0 \sim 7$. The higher level means the more specific syslog is saved to a temporary file.
	By default, it's level 3.

	Export Log: Click to export temporary syslog file to local PC.
РСАР	To start, stop packets capturing or to export captured Packet file.
	 Start: To start capturing all the packets file sent or received from IP phone. Stop: To stop capturing packets.
	Note: IP phone will save captured packets file to a temporary file, this file maximum size is 1M(mega bytes), and will top capturing once reaching this maximum size.
Others	To display or configure others features from this page.
	Config file: To export or import configure file for IP phone.