

E10S Secondary Entry Phone

Content

1 Production Overview	3
1.1 Production Description1.2 Dimension1.3 Connector1.4 Installation	
2 Basic Function	8
2.1 Make a call 2.2 Monitor	8
3 Configuration	9
3.1 Web login 3.1.1 Obtaining IP address	9
3.1.2 Login the web 3.2 Status-Basic	
3.3 Intercom- Basic 3.4 Intercom-Advanced	
3.5 Network-Basic 3.6 Network-Advanced	
3.7 Phone-Time/Lang 3.8 Phone-Call Feature	
3.9 Phone-Voice 3.10 Phone-Call Log	
3.11 Upgrade-Basic 3.12 Upgrade-Advanced	
3.13 Security-Basic	

1 Production Overview

1.1 Production Description



E10S is a smart SIP-based secondary entry phone. It can be connected with Akuvox indoor phone for unlock and monitor. It is more convenient and safe for residents to check the visitor identity through E10S. E10S is often applicable in villas , apartments.

IFCC Caution:

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to

provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

-Reorient or relocate the receiving antenna.

-Increase the separation between the equipment and receiver.

-Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

-Consult the dealer or an experienced radio/TV technician for help.

Specific Absorption Rate (SAR) information

SAR tests are conducted using standard operating positions accepted by the FCC with the device transmitting at its highest certified power level in all tested frequency bands, although the SAR is determined at the highest certified power level, the actual SAR level of the device while operating can be well below the maximum value. Before a new product is a available for sale to the public, it must be tested and certified to the FCC that it does not exceed the exposure limit established by the FCC, tests for each phone are performed in positions and locations as required by the FCC. For headset, this part has been tested and meets the FCC RF exposure guidelines when used with an accessory designated for this product or when used with an accessory that contains no metal.

For baseband, this equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment .This equipment should be installed and operated with minimum distance 20cm between the radiator& your body.

1.2 Dimension



- ① Mic
- 2 LED
- ③ Key Light 1
- (4) Indicator Light
- (5) Button
- 6 Preformed hole
- ⑦ Camera
- 8 Key Light 2
- (9) Infrared sensor
- 10 Loudspeaker

1.3 Connector



Note: Akuvox IT81 indoor phone can also provide power for E10S through Ethernet port.

1.4 Installation



- 1. Use screws 2 to fix the mainboard into the 86 embedded box of the wall.(the screw of length: 10mm, diameter: 4mm)
- 2. Use screw 1(M2.5X5) Lock the surface at the mainboard.

Warning :

- 1. To protect the product from crashing, knocking or shaking.
- 2. Please don't place the E10S under the sun, high temperature, snow or chemical corrosion or dust.
- 3. Please install the E10S in a good visual level. (about 160cm)
- 4. Please cut down the power if you find the product is not working properly.
- 5. If the stairway phone is broken, you should cut down the power immediately and check for fault. Otherwise call the customer service manager to help.
- 6. Please protect the IC card from water and broken, antimagnetic.



2 Basic Function

2.1 Make a call

Once you setup push button number(please refer to chapter Intercom- Basic), press it to call the indoor phone. After the call is answered, the LED and Key light will turn up. Resident can pickup the call in video or audio mode.

2.2 Monitor

Users can press Monitor button in indoor phone to get the live video from E10S any time.

3 Configuration

3.1 Web login

3.1.1 Obtaining IP address

Hold the call button about 5s, the phone will announce its IP. Press again to stop.

3.1.2 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

00,00		
Login		Help Login Page
User Name Password	Remember Username/Password Login	
Al.		1

3.2 Status-Basic

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

	Status		Help
sic	Pi	roduct Information	Note ·
ercom	Model	E10S	Max length of characters for input
	MAC Address	C4:09:38:E2:F3:D7	box:
ount	Firmware Version	110.0.0.39	255: Broadsoft Phonebook server
	Hardware Version	110.0.0.0.0.0.0	address
work			127: Remote Phonebook URL &
	Ne	atwork Information	AUTOP Manual Update Server URL
ne	INC		63: The rest of input boxes
	LAN Port Type	Static IP	Warning
rade	LAN Link Status	Connected	warning .
	LAN IP Address	192.168.35.100	Field Description :
urity	LAN Subnet Mask	255.255.255.0	rield bescription .
	LAN Gateway	192.168.35.1	
	LAN DNS1	192.168.35.1	
	LAN DNS2		
	A	ccount Information	
	Account1	110@192.168.35.254	
	, cecourter	Registered	
	Account2	None@None	
		Disabled	

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).

3.3 Intercom- Basic

Account Dasic			Help
	SIP Account		Noto :
Status	Registered		Max length of characters for input
Account	Account 1	•	box:
Account Active	Enabled	•	255: Broadsoft Phonebook serve
Display Label	110		127: Remote Phonebook URL &
Display Name	110		AUTOP Manual Update Server U
Register Name	110		63: The rest of input boxes
User Name	110		Warning :
Password			
			Field Description :
	SIP Server 1		Submit Shortcut
Comment The	102 102 25 254	Dart Soco	Submit Cancel
Server IP	192.108.33.234	(20) (5525a)	
Registration Period	1800	(30~655355)	
	SIP Server 2		
Sapur IP		Port 5060	
Peoletration Period	1900	(200655250)	
registration renou	1000	(30 033335)	
Outb	ound Proxy Server		
Enable Outbound	Disabled	•	
Server IP		Port 5060	
Backup Server IP		Port 5060	
1	Transport Type		
Transport Type	UDP	•	
	NAT		
	Disabled	•	
NAT			

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
	display.
	• 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.

3.4 Intercom-Advanced

Account-Advanced		Help
5	SIP Account	Note ·
Account	Account 1	Max length of characters for inpu
		255: Broadsoft Phonebook server
	Codecs	address
Disabled Codecs Enab	led Codecs	127: Remote Phonebook URL & AUTOP Manual Update Server UR
 PCMU PCMU 	*	63: The rest of input boxes
G729		Warning :
0722		
>>	1	Field Description :
<<		Submit Shortcut
		Submit Cancel
-	-	
١	/ideo Codec	
Codec Name	✓ H264	
Codec Resolution	4CIF •	
Codec Bitrate Codec Pavload	104 *	
	Subscribe	
MWI Subscribe	Disabled 🔹	
MWI Subscribe Period	1800 (120~65535s)	
Voice Mail Number		
BLF Expire	1800 (120~65535s)	
ACD Expire	1800 (120~655355)	
	DTMF	
Tune	PE(2922	
How To Notify DTMF	Disabled	
DTMF Payload	101 (96~127)	
	Call	
Max Local SIP Port	5062 (1024~65535)	
Min Local SIP Port	5062 (1024~65535)	
Caller ID Header	FROM *	
Provisional Response ACK	Disabled •	
Register with user=phone	Disabled •	
Invite with user=phone	Disabled •	
Anonymous Call Anonymous Call Rejection	Disabled •	
Missed Call Log	Enabled •	
Prevent SIP Hacking	Disabled •	
S	ession Timer	
Active	Disabled v	
Session Expire Session Refresher	1800 (90~7200s)	
a contract contract		
	BLFList	
BLFList URI		
BLFList PickUp Code		
BLFList BargeIn Code		
	Encryption	
Voice Frankin (corre)	Disabled	
voice encryption(SRTP)		
	NAT	
UDP Keep Alive Messages	Disabled	
UDP Alive Msg Interval	30 (5~60s)	
RPort	Disabled •	
	User Agent	
User Agent		
Submit	Cancel	

Sections	Description
SIP Account	To display current Account settings or to select which account to display
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 (wide-bandth codecs), G729 and so on.
Video Codec	 To configure the video quality Codec Name: The default video codec is H264. Codec Resolution: It can support QCIF, CIF, VGA, 4CIF, 720P. Codec Bitrate: The lowest bitrate is 128, the highest bitrate is 2048. Codec payload: From 90-119.
Subscribe	 To display and configure MWI, BLF, ACD subscription settings. MWI: Message Waiting Indicator which is used to indicate whether there is unread new voice message. BLF: BLF is short for Busy Lamp Field which is used to monitor the designated extension status. ACD: Automatic Call Distribution is often used in offices for customer service, such as call center. The setting here is to negotiate with the server about expire time of ACD subscription.
DTMF	 To display and configure DTMF settings. Type: Support Inband,Info, RFC2833 or their combination. How To Notify DTMF: Only available when DTMF Type is Info. DTMF Payload: To configure payload type for DTMF. Note: By default, DTMF type is RFC2833 which is the standard. Type Inband uses inband frequency to indicate DTMF tone which is most used to be compatible to traditional telephone server. Type Info use SIP Info message to indicate DTMF message.
Call	 To display and configure call-related features. Max Local SIP Port: To configure maximum local sip port for designated account. Min Local SIP Port: To configure minimum local sip port for designated account. Caller ID Header: To configure which Caller ID format to fetch for displaying on Phone UI. Auto Answer: If enabled, IP phone will be auto-answered when there is an incoming call for

	designated account.
	 Ringtones: Choose the ringtone for each account.
	 Provisioning Response ACK: 100% reliability for all
	provisional messages this means it will send ACK every
	time the IP phone receives a provisional SIP message
	from SIP server.
	 User=phone: If enabled, IP phone will send user=phone within SIP message
	 DTime: Interval time between two consecutive RTP.
	packets.
	• Anonymous Call: If enabled, all outgoing call for the
	designated account will be anonymous number.
	• Anonymous Call Rejection: If enabled, all incoming
	anonymous-out call for the designated account will be
	 Is escape non Ascii character: To transfer the symbol to
	Ascii character
	 Miscod Call Log: To display the miss call log
	Inissed Call Log. To display the miss call log.
	Frevent Sir Hacking. Enable to prevent Sir Holf Hacking.
Session Timer	To display or configure session timer settings.
	• Active: To enable or disable this feature, if enable, the
	on going call will be disconnected automatically once
	the session expired unless it's been refreshed by UAC or
	UAS.
	 Session Expire: Configure session expire time.
	• Session Refresher: To configure who should be response
	for refreshing a session.
	Note: UAC means User Agent Client, here stands for IP
	phone. UAS means User Agent Server, here stands for SIP
	server.
BLF List	To display or configure BLF List URI address.
	• BLF List URI: BLF List is short for Busy Lamp Field List.
	• BLFList PickUp Code: To set the BLF pick up code.
	• BLFList BargeIn Code : To set the BLF barge in code.
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: Keenalive message interval
	 Rport: Remote Port, if enabled, it will add Remote Port

	into outgoing SIP message for designated account.
User Agent	One can customize User Agent field in the SIP message; If
	user agent is set to specific value, user could see the
	information from PCAP. If user agent is not set by default,
	user could see the company name, model number and
	firmware version from PCAP

3.5 Network-Basic

tatus	Network-Basic		Help
itercom		LAN Port	Note :
Iccount Ietwork Basic Advanced Advanced Ipgrade Iecurity	DHCP Static IP IP Address Subnet Mask Default Gateway LAN DNS1 LAN DNS2 Submit	192.168.35.100 255.255.255.0 192.168.35.1 192.168.35.1 Cancel	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 : Field Description : Submit Shortcut 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.

3.6 Network-Advanced

Status Netwo	ork-Advanced			Help
Intercom	Lo	cal RTP		Note :
Account	Starting RTP Port	11800	(1024~65535)	Max length of characters for input
etwork	Max RTP Port	12000	(1024~65535)	255: Broadsoft Phonebook server
asic	Submit		Cancel	address 127: Remote Phonebook URL & AUTOP Manual Update Server URL
Advanced				63: The rest of input boxes
hone				Warning :
pgrade				Field Description :
ecurity				Submit Shortcut
				Submit Cancel

Sections	Description
Local RTP	To display and configure Local RTP settings.
	• Max RTP Port: Determine the maximum port that RTP
	stream can use.
	• Starting RTP Port: Determine the minimum port that RTP
	stream can use.

3.7 Phone-Time/Lang

tatus	Time/Lang				Help
ntercom		NTP			Note :
ccount	Time Zone	0 GMT		•	Max length of characters for input
etwork	Primary Server	0.pool.ntp.org			255: Broadsoft Phonebook server
	Secondary Server	1.pool.ntp.org	(address
hone	Update Interval	3600	(>= 3600s)		AUTOP Manual Update Server URL
Time/Lang					63: The rest of input boxes
Cal Feature	Sut	omit	Cancel		Warning :
10/00					Field Description :
VOLC					ricia Description .
Call Log					Submit Shortcut
pgrade					Submit Cancel
curity					

Sections	Description
NTP	To configure NTP server related settings.
	• Time Zone: To select local Time Zone for NTP server.
	• Primary Server: To configure primary NTP server
	address.
	• Secondary Server: To configure secondary NTP server
	address, it takes effect if primary NTP server is
	unreachable.
	• Update interval: To configure interval between two
	consecutive NTP requests.
	Note: NTP, Network Time Protocol is used to automatically
	synchronized local time with INTERNET time, since NTP
	server only response GMT time, so that you need to specify
	the Time Zone for IP phone to decide the local time.

3.8 Phone-Call Feature

status	Phone-Call Feature		Help
Intercom	M	Note :	
Account	Mode	Max length of characters for input	
Network		255: Broadsoft Phonebook server address	
Phone	Assessed	All Assount	127: Remote Phonebook URL &
	DND	Disabled	AUTOP Manual Update Server URL
Time/Lang	Return Code When DND	486(Busy Here)	us. The rest of input boxes
Call Feature	DND On Code		Warning :
Voice	DND Off Code		Field Description :
Call Log		Submit Shortcut	
Ingrade		Submit Cancel	
-pg. auc	Active	Enabled V	
Security	Intercom Mute		
		Others	
	Return Code When Refuse	486(Busy Here)	
	Auto Answer Delay	0 (0~5s)	
	Auto Answer Mode:	Video 🔻	
	Multicast Codec	PCMU 🔻	
	Direct IP	Enabled 🔻	
	Submit	Cancel	

Sections	Description
Mode	• Mode: Select the desired mode.
DND	DND (Do Not Disturb) allows IP phones to ignore any
	• 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 side if you
	press DND when DND is on.
Intercom	Intercom allows user to establish a call directly with the
	callee.
	 Active: To enable or disable Intercom feature.
	• Intercom Mute: If enabled, once the call established, the

	callee will be muted.
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.
	• Auto Answer Mode: To set video or audio mode for auto
	answer by default.
	• Direct IP: Direct IP call without SIP proxy.

3.9 Phone-Voice

atus	Voice			Help
tercom		Mic Volume		Note :
count	Mic Volume	12	(1~15)	Max length of characters for input
etwork				255: Broadsoft Phonebook server
ione	S	peaker Volume		127: Remote Phonebook URL &
Fime/Lang	Speaker Volume	12	(1~15)	AUTOP Manual Update Server URL 63: The rest of input boxes
Call Feature	Submit	Ca	ncel	warning :
/oice				Field Description :
Call Log				Submit Shortcut
grade				Submit Cancel
curity				

Sections	Description
Mic Volume	To configure Microphone volume , from 1-15. 12 by default.
Speaker Volume	To configure Speaker Volume, from 1-15, 12 by default.

3.10 Phone-Call Log

Ca	ll Log				Help
com	all History	All			Note :
int Index	Type Date	Time Local Identity	Name	Number	Max length of characters for input
1	Dialed 1970-01-01	00:25:51 110@192.168.35.25	54 Unknown	109@192.168.35.254	255: Broadsoft Phonebook server
ork 2	Dialed 1970-01-01	00:04:37 110@192.168.35.25	54 Unknown	109@192.168.35.254	address
3	Dialed 1970-01-01	00:04:31 110@192.168.35.25	54 Unknown	109@192.168.35.254	127: Remote Phonebook URL &
4	Dialed 1970-01-01	00:03:32 110@192.168.35.25	54 Unknown	109@192.168.35.254	AUTOP Manual Update Server URI
/Lang 5					63: The rest of input boxes
6					Warning
ature 7					warning .
8					Field Description :
9					
.og 10					
11					
1e 12					
13					
14					
15					
Pa	ge 1 V Pre	ev Next	Delete	Delete All	
14 15 Pa	g <mark>e 1 ▼</mark> Pre	ev Next	Delete	Delete All	

Sections	Description
Call History	To display call history records.
	Available call history types are All calls, Dialed calls, Received
	calls, Missed calls, Forwarded calls.
	Users can check the call history in detail. Tick the number to
	delete or delete all logs. E10S supports 100 call logs.

3.11 Upgrade-Basic

us	Jpgrade-Basic		Help
rcom vunt vork ae ade ade anced urity	Firmware Version Hardware Version Upgrade Reset To Factory Setting Reboot	110.0.09 110.0.0.0.0.0 <u>速接文件</u> 未选择任何文件 Submit Cancel Submit Submit	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 : 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.

3.12 Upgrade-Advanced

tus	Upgrade-Advanced		Help
rcom		PNP Option	Note :
ount	PNP Config	Enabled v	Max length of characters for input box:
work	Submit Cancel		255: Broadsoft Phonebook server address
ne			127: Remote Phonebook URL & AUTOP Manual Undate Server URL
rade		System Log	63: The rest of input boxes
sic	LogLevel	3 •	Warning :
vanced	Export Log	Export	Field Description :
urity			Submit Shortcut
			Submit Cancel

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).
System Log	To display system log level and export system log file.
	• System log level: From level 0~7.The higher level means
	the more specific system log is saved to a temporary file.
	By default, it's level 3.
	• Export Log: Click to export temporary system log file to
	local PC.

3.13 Security-Basic

► Status	Help
▶ Intercom	Net a
 Account Network Phone Upgrade Security Basic 	Max length of characters for input box: 235: Broadsoft Phonebook server address 127: Remote Phonebook URL & AUTOP Manual Update Server URL 63: The rest of input boxes Warning : Field Description : Submit Cancel
Sections	Description

Confirm Password: Repeat the new password.

•