

SIP Video Intercom i18S

USER MANUAL

V1.0



Wall mounting



Flush mounting

www.fanvil.com



Document	Firmware	Explanation	Time
VER	VER		
V1.0	2.1.1.3390	Initial issue	20180208
V1.1	2.1.1.3445	Change some description	20180514





Safety Notices

- 1. Please use the specified power adapter. If you need to use the power adapter provided by other manufacturers under special circumstances, please make sure that the voltage and current provided is in accordance with the requirements of this product, meanwhile, please use the safety certificated products, otherwise may cause fire or get an electric shock.
- 2. When using this product, please do not damage the power cord either by forcefully twist it, stretch pull, banding or put it under heavy pressure or between items, otherwise it may cause damage to the power cord, lead to fire or get an electric shock.
- 3. Before using, please confirm that the temperature and environment is humidity suitable for the product to work. (Move the product from air conditioning room to natural temperature, which may cause this product surface or internal components produce condense water vapor, please open power use it after waiting for this product is natural drying).
- 4. Please do not let non-technical staff to remove or repair. Improper repair may cause electric shock, fire, malfunction, etc. It will lead to injury accident or cause damage to your product.
- 5. Do not use fingers, pins, wire, other metal objects or foreign body into the vents and gaps. It may cause current through the metal or foreign body, which may even cause electric shock or injury accident. If any foreign body or objection falls into the product please stop using.
- 6. Please do not discard the packing bags or store in places where children could reach, if children trap his head with it, may cause nose and mouth blocked, and even lead to suffocation.
- 7. Please use this product with normal usage and operating, in bad posture for a long time to use this product may affect your health.
- 8. Please read the above safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.





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A.Product introduction

i18S Voice Access is a digital network door phone. Its core part adopts mature VoIP solution (Broadcom chip), which can perform stably and reliably, it is hands-free, adopting digital full-duplex mode. The voice is loud and clear. It has a series of advantages, such as generous appearance, comfortable keypad and low power consumption, etc. i31S is easy to install. It is solid and durable.



1. Appearance of the product

Single button



Dual button

2. Description

Picture	Description Function			
		Network error: Blink with 2s		
	DSS Key LED	Network running: Off		
		Registration failed: Blink with 6s		
		Registration succeeded: On		



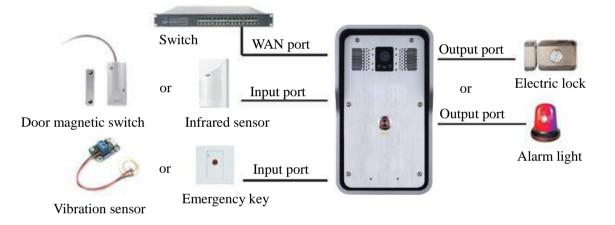
B.Start Using

Before you start to use the equipment, please make the following installation:

1. Confirm the connection

Please confirm the power cord, network cable, electric lock control line connected and the boot-up is normal. (Check the network state of light)

1) Power port



2) Power, Security functions Input, Security functions Output port

Power supply ways: 12v/DC or PoE.

			CN7				
1	2	3	4	5	6	7	elele ele ele
+12V	VSS	NC	COM	NO	S_IN	S_OUT	****
10)/ 10/00		Security functions Output			Security functions		ERRE
12V 1A/DC			port		Input	t port	

3) Wiring instructions

- NO: Normally Open Contact.
- COM: Common Contact.
- NC: Normally Close Contact.

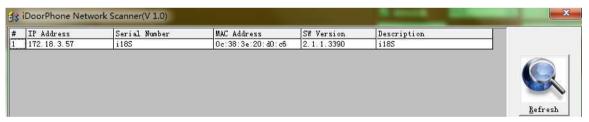


2. Quick Setting

The product provides completed functions and parameter settings. To understand all meaning of parameters well, it is better for users to have knowledge of network and SIP protocol. In order to let users, enjoy the high-quality voice service and low-cost advantage immediately, here we list some basic but compulsory setting options in this section. Users can use it without understanding such complicated knowledge of SIP protocols.

In prior to this step, please make sure your broadband Internet online can work normally and complete the connection of the network hardware. The product default factory setting of network mode is STATIC IP. Before the entering of Web setting, pls connec the PC to the same LAN network with i18S or set the network segment of PC's Static IP address in the same segment of i18S.

- The default IP address is static IP address: 192.168.1.128. User can also use the software"iDoorPhoneNetworkScanner.exe" to find the IP address of the device. (download address http://download.fanvil.com/tool/iDoorPhoneNetworkScanner.exe)
- > Note: Waiting for 30s to run the device when it is power on.
- Login to the WEB to configure the device
- Configure the service account, user name, server registered address and other parameters on the web page of SIP.
- Set DSS key in the Webpage (Intercom -> function key).
- > Set function parameters in the Webpage (Safeguarding).



C.Basic operation

1. Answer a call

By default, the incoming call will be answered automatically without any ringing. User MAY want to hear ring before answer the incoming call. This could be configured under EGS setting -> Features -> Basic Settings -> Auto Answer timeout. This parameter is the ringing time. Auto answered could be disabled under EGS setting -> Features -> Basic settings -> Enable auto Answer.



2. Call

Configure shortcut key as hot key then setup a number. The configured number will be called when user press the shortcut key.

3. End Call

Enable the DSS key to hang up the call.

D.Page settings

1. Browser configuration

When the device and your computer are successfully connected to the network, enter the IP address of the device on the browser as http://xxx.xxx.xxx/, Then you can see the login interface of the web page management.

Input the user name and password. Then click the [logon] button to enter the settings screen.

llcom		
User: Password:		
Language:	English 💌	
	Logon	

After configuring the equipment, remember to click SAVE under the Maintenance tab. If this is not done, the equipment will lose the modifications when it has been rebooted.

2. Password Configuration

There are two levels of access: root level and general level. A user with root level access can browse and set all configuration parameters. While a user with general level can set all configuration parameters except server parameters for SIP

- Default user with general level:
 - Username: guest
 - Password: guest



- Default user with root level:
 - Username: admin
 - Password: admin

3. Configuration via WEB

- (1) System
- a) Information

	Information	Account	Configurations U	pgrade Auto	Provision FDMS	5 Tools
> System	System Information					
	Model:		i18S			
> Network	Hardware:		2.1			
	Software:		2.1.1.3432			
> Line	Uptime:		03:35:04			
	Last uptime:		00:15:05			
› Intercom settings	MEMInfo:		ROM: 0.8/8(M)	RAM: 2.3/16(M)		
intercom settings	System Time:		2018-03-03 15:5	0		
> Security settings	Network					
	Network mode:		DHCP			
Function Key	MAC:		Oc:38:3e:1e:61:0	ld		
	IP:		172.18.3.40			
	Subnet mask:		255.255.0.0			
	Default gateway:		172.18.1.1			
	SIP Accounts					
	Line 1	5528	Registere	ed		
	Line 2	N/A	Inactive			

Information				
Field Name	Explanation			
System	Display equipment model, hardware version, software version, uptime, Last			
Information	uptime and MEMinfo.			
Network	Shows the configuration information for WAN port, including connection mode of			
INELWOIK	WAN port (Static, DHCP, PPPoE), MAC address, IP address of WAN port.			
SIP Accounts	Shows the phone numbers and registration status for the 2 SIP LINES.			



b) Account

On this page, user can add or remove users depending on their needs and can modify existing user permission.

	Information	Account Configura	ations Upgrade	Auto Provision	FDMS	Tools
> System	Change Web Auther	ntication Password				
› Network	Old Password: New Password:					
> Line	Confirm Passwor	rd:	Apply			
> Intercom settings	Add New User					
› Security settings	Username Web Authenticat	ion Password				
> Function Key	Confirm Passwor Privilege	ď	Administrators V			
	User Accounts					
	User		Privilege			
	admin	A	dministrators		Delete	

Account				
Field Name	Explanation			
Change Web Authentication Password				
You Can modif	You Can modify the login password to the account			
Add New User				
You can add new user				
User Accounts				
Show the existi	Show the existing user information			



c) Configurations

	Information	Account	Configurations	Upgrade	Auto Provision	FDMS	Tools
> System							
> Network	Export Configu	rations	Right click her	e to SAVE configura	ations in 'txt' format.		
> Line	Import Configu	rations	Right click her	e to SAVE configura	ations in 'xml' format.		
› Intercom settings			Configuration	file:	Select	Import	
Security settings	Reset to factory	y defaults					
› Function Key			-	et] button to reset t	the phone to factory de AFTER RESET!	efaults.	

Configurations	
Field Name	Explanation
Export	Save the equipment configuration to a txt or xml file. Please Right click on
Configurations	the choice and then choose "Save Link As."
Import	Prowee to the config file, and proce Lindete to lead it to the equipment
Configurations	Browse to the config file, and press Update to load it to the equipment.
Reset to factory	This will reset factory default settings and remove all configuration
defaults	information.

d) Upgrade

	Information	Account Configurations	Upgrade	Auto Provision	FDMS	Tools
> System						
> Network	Software upgrade					
		Current Software Version:	2.1.1.3432			
› Line		System Image File		Select	Upgrade	
Upgrade						
Field Name	Explanation					
Software upg	rade					

Browse to the firmware, and press Update to load it to the equipment.



e) Auto Provision

	Information	Account	Configurations	Upgrade	Auto Provision	FDMS	Tools
> System							
	Common Settin	gs					
> Network	Current Cor	nfiguration Version					
	General Co	nfiguration Version					
> Line	CPE Serial I	Number	00100400FV02	2001000000c383e1	e61dd		
	Authenticat	ion Name					
Intercom settings	Authenticat	ion Password					
	Configuratio	on File Encryption Ke	γ				
Security settings	General Cor Key	nfiguration File Encry	ption				
Function Key	Save Auto I	Provision Information					
	DHCP Option >	>					
	SIP Plug and Pl	ay (PnP) >>					
	Static Provision	ing Server >>					
	TR069 >>						
			Apply				
DHCP Option >>							
Option Value		Option 66	\checkmark				
Custom Optio	n Value	66	(12	8~254)			
SIP Plug and Play	(DnD) >>						
Enable SIP Pr	ιP						
Server Addres	SS	224.0.1.75					
Server Port		5060					
Transportatio	n Protocol	UDP 🗸					
Update Interv	/al	1	Ηοι	ır			
Static Provisionir	g Server >>						
Server Address		0.0.0.0					
Configuration File Name							
		FTP V					
Protocol Type							
Update Interv	al	1	Hou	r			
Update Mode		Disabled	~				

Auto Provisio	n
Field Name	Explanation
Common Setting	js
	Show the current config file's version. If the version of configuration
Current	downloaded is higher than this, the configuration will be upgraded. If the
Configuration	endpoints confirm the configuration by the Digest method, the
Version	configuration will not be upgraded unless it differs from the current
	configuration
General	Show the common config file's version. If the configuration downloaded
Configuration	and this configuration is the same, the auto provision will stop. If the
Version	endpoints confirm the configuration by the Digest method, the



	configuration will not be upgraded unless it differs from the current configuration.
CPE Serial Number	Serial number of the equipment
Authentication Name	Username for configuration server. Used for FTP/HTTP/HTTPS. If this is blank the phone will use anonymous
Authentication Password	Password for configuration server. Used for FTP/HTTP/HTTPS.
Configuration File Encryption Key	Encryption key for the configuration file
General Configuration File Encryption Key	Encryption key for common configuration file
Save Auto Provision Information	Save the auto provision username and password in the phone until the server url changes
DHCP Option	
Option Value	The equipment supports configuration from Option 43, Option 66, or a Custom DHCP option. It may also be disabled.
Custom Option Value	Custom option number. Must be from 128 to 254.
SIP Plug and Play	(PnP)
Enable SIP PnP	If this is enabled, the equipment will send SIP SUBSCRIBE messages to a multicast address when it boots up. Any SIP server understand that message will reply with a SIP NOTIFY message containing the Auto Provisioning Server URL where the phones can request their configuration.
Server Address	PnP Server Address
Server Port	PnP Server Port
Transportation Protocol	PnP Transfer protocol – UDP or TCP
Update Interval	Interval time for querying PnP server. Default is 1 hour.
Static Provisioning	g Server
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be an IP address or Domain name with subdirectory.
Configuration File Name	Specify configuration file name. The equipment will use its MAC ID as the config file name if this is blank.
Protocol Type	Specify the Protocol type FTP, TFTP or HTTP.
Update Interval	Specify the update interval time. Default is 1 hour.
Update Mode	 Disable – no update Update after reboot – update only after reboot.



	3. Update at time interval – update at periodic update interval
TR069	
Enable TR069	Enable/Disable TR069 configuration
ACS Server Type	Select Common or CTC ACS Server Type.
ACS Server URL	ACS Server URL.
ACS User	User name for ACS.
ACS Password	ACS Password.
TR069 Auto Login	Enable/Disable TR069 Auto Login.
INFORM Sending	Time between transmissions of "Inform" is 3600 seconds.
Period	

f) FDMS

	Information Account	Configurations Upgrade	Auto Provision	FDMS Tools
> System				
> Network	FDMS Settings Enable FDMS			
> Line	FDMS Interval	3600		
› Intercom settings	Doorphone Info Settings			
	Community Name			
Security settings	Building Number			
	Room Number			
Function Key		Apply		

FDMS Settings

T Blife Cottinge		
Enable FDMS	Enable/Disable FDMS configuration	
FDMS Interval	The time to send sip Subscribe information to the FDMS server on a regular	
FDIVIS IIItervai	basis. Unit is in second.	
Doorphone Info Se	ettings	
Community Name	The name of the community where the device is installed	
Building Number	The name of the building where the equipment is installed	
Room Number	The name of the room where the equipment is installed	



g) Tools

	Information Account	Configurations	Upgrade	Auto Provision	FDMS	Tools
ystem						
	Syslog					
twork	Enable Syslog					
	Server Address	0.0.0.0				
Line	Server Port	514	1			
	APP Log Level	None				
Intercom settings	SIP Log Level	None	\sim			
curity settings		Apply				
	Network Packets Capture					
ion Key		Start				
	Auto Reboot Setting					
	Reboot Mode	Disable 🔽				
	Fixed Time	2	(0~23)			
	Uptime	72	(h)			
	Sip Reg Fail Reboot					
	Waiting Time	180	(s)			
	Network Fail Reboot					
	Waiting Time	300	(5)			
		Apply				
	Reboot Phone					
		Click [Reboot]	button to restart th	e phone!		
		Reboot				

Syslog provide a client/server mechanism for the log messages which is recorded by the system. The Syslog server receives the messages from clients and classifies them based on priority and type. Then these messages will be written into a log by rules which the administrator has configured.

There are 8 levels of debug information.

- Level 0: emergency; System is unusable. This is the highest debug info level.
- Level 1: alert; Action must be taken immediately.
- Level 2: critical; System is probably working incorrectly.
- Level 3: error; System may work incorrectly.
- Level 4: warning; System may work correctly but needs attention.
- Level 5: notice; It is the normal but significant condition.
- Level 6: Informational; It is the normal daily messages.
- Level 7: debug; Debug messages normally used by system designer. This level can only be displayed via telnet.

Tools	
Field Name	Explanation
Syslog	
Enable	Enable or disable system log.



Syslog			
Server	System log conver ID address		
Address	System log server IP address.		
Server Port	System log server port.		
APP Log	Sat the lovel of ADD log		
Level	Set the level of APP log.		
SIP Log Level	Set the level of SIP log.		
Network Packets Capture			
Capture a packet stream from the equipment. This is normally used to troubleshoot problems.			
Reboot Phone			
Some configuration modifications require a reboot to become effective. Clicking the Reboot			
button will lead to reboot immediately.			
Note: Be sure to save the configuration before rebooting.			

(2) network a) Basic

	Basic VPN		
> System	Network Status		
	IP:	172.18.3.40	
> Network	Subnet mask:	255.255.0.0	
- Hetwork	Default gateway:	172.18.1.1	
0.14020	MAC:	0c:38:3e:1e:61:dd	
> Line	MAC Timestamp:	20170301	
› Intercom settings	Settings		
	Static IP 〇	DHCP	PPPoE O
> Security settings	DNS Server Configured by	DHCP	
	Primary DNS Server		
> Function Key	Secondary DNS Server		
		Apply	
	Service Port Settings 😡		
	Web Server Type	HTTP V	
	HTTP Port	80	
	HTTPS Port	443	
		Apply	
Field Name	Explanation		
Network Status			
Р	The current IP address of	f the equipment	



Subnet mask	The current Subnet Mask	
Default	The current Gateway IP address	
gateway		
MAC	The MAC address of the equipment	
MAC	Cat the MAC address of time	
Timestamp	mp Get the MAC address of time.	
Settings		
Select the appro	opriate network mode. The equipment supports three network modes:	
Static IP	Network parameters must be entered manually and will not change. All parameters are provided by the ISP.	
DHCP	Network parameters are provided automatically by a DHCP server.	
PPPoE	Account and Password must be input manually. These are provided by your ISP.	
If Static IP is cho	osen, the screen below will appear. Enter values provided by the ISP.	
DNS Server	Select the Configured mode of the DNS Server.	
Configured by	Select the Configured mode of the DNS Server.	
Primary DNS Server	Enter the server address of the Primary DNS.	
Secondary DNS Server	Enter the server address of the Secondary DNS.	
	I vertice the new settings. The equipment will save the new settings.	
	If a new IP address was entered for the equipment, it must be used to login to	
the phone after	clicking the APPLY button.	
Service Port Se	ettings	
Web Server Type	Specify Web Server Type – HTTP or HTTPS	
	Port for web browser access. Default value is 80. Change this from the default	
	to enhance security. Setting this port to 0 will disable HTTP access.	
HTTP Port	Example: The IP address is 192.168.1.70 and the port value is 8090, The	
	accessing address is http://192.168.1.70:8090.	
	Port for HTTPS access. An https authentication certification must be	
HTTPS Port	downloaded into the equipment before using https.	
	Default value is 443. Change this from the default to enhance security.	
Note:		
1) Any changes	made on this page require a reboot to become active.	
2) It is suggeste	d that the make the values bigger than 1024 if users change the port to HTTPS.	

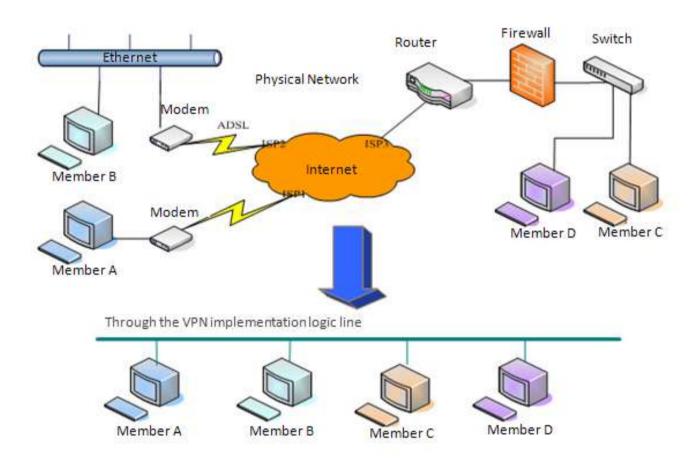
Values less than 1024 are reserved.



3) If the HTTP port is set to 0, HTTP service will be disabled.

b) VPN

The device supports remote connection via VPN. It supports both Layer 2 Tunneling Protocol (L2TP) and Open VPN protocol. This allows users securely connect from public network to local network remotely.





	Basic	4		
	Virtual Private Network (V	/PN) Status		
System		VPN IP Addre	ess:	0.0.0.0
Network	VPN Mode			
		Enable VPN		
Line		L2TP O		OpenVPN 💿
Intercom settings	Layer 2 Tunneling Protoco	ol (L2TP)		
		L2TP Server	Address	
Security settings		Authenticatio	n Name	
		Authenticatio	n Password	
Function Key			_	
			-0	Apply
	OpenVPN Files			
	OpenVPN Configuration file:	client.ovpn	N/A	Upload Delete
	CA Root Certification:	ca.crt	N/A	Upload Delete
	Client Certification:	client.crt	N/A	Upload Delete
	Client Key:	client.key	N/A	Upload Delete

Field Name	Explanation			
VPN IP Address	Shows the current VPN IP address.			
VPN Mode				
Enable VPN	Enable/Disable VPN.			
L2TP	Select Layer 2 Tunneling Protocol			
	Select OpenVPN Protocol. (Only one protocol may be activated. After the			
OpenVPN	selection is made, the configuration should be saved and the phone be			
	rebooted.)			
Layer 2 Tunnelin	ng Protocol (L2TP)			
L2TP Server	Set VPN L2TP Server IP address.			
Address	Set VFIN LZTF Server IF address.			
Authentication	Set User Name access to VPN L2TP Server.			
Name	Set User Maine access to VFN LZTF Server.			
Authentication	Set Password access to VPN L2TP Server.			
Password	Set Password access to VPN LZTP Server.			
Open VPN Files				
Upload or delete Open VPN Certification Files				

(3) Line a) SIP

Configure a SIP server on this page.



	SIP	Basic Settings				
-	512	Basic Setungs				
> System						
	Line	SIP 1 💙				
> Network	Basic Settings >					
> Line	Line Status		Registered		SIP Proxy Server Address	172.18.1.88
	Phone numb	ber	5528	×	SIP Proxy Server Port	5060
> Intercom settings	Display nam		5528		Backup Proxy Server Address	
. Cocurity cottings	Authenticati	on Name on Password	5528		Backup Proxy Server Port Outbound proxy address	5060
 Security settings 	Activate	on Fassword			Outbound proxy address Outbound proxy port	
› Function Key					Realm	
	Codecs Settings	; >>				
	Advanced Setti	ngs >>				
		-	Apply			
Codecs Settings >>						
Disabled Codecs				Enabled Cod	laca	
Disabled Codecs		_		G.722		
		*		G.711U G.711A		
		-		G.729AB	✓ ↓	
Advanced Settings >:	>					
Subscribe For Void	e Message					
Voice Message Nu	mber]		
Voice Message Sul	bscribe Period	3600	Second(s)			
Enable DND				Ring Typ	e	Default 🗸
Blocking Anonymo	ous Call			Conferer	псе Туре	Local 🗸
Use 182 Response	for Call waiting			Server C	onference Number	
Anonymous Call S	Anonymous Call Standard		None 🖌		Timeout	0 Second(s)
Dial Without Regis	Dial Without Registered				ong Contact	
Click To Talk	Click To Talk		En En		se Inactive Hold	
User Agent	User Agent		Use Quote in Dis		te in Display Name	
Response Single C	odec					
		_				
Use Feature Code	Use Feature Code					
Enable DND				DND Dis		
Enable Blocking A		Disable Blocking Anonymous Call				



Specific Server Type	COMMON 🗸	Enable DNS SRV	
Registration Expiration	3600 Second(s)	Keep Alive Type	SIP Option 🗸
Use VPN		Keep Alive Interval	60 Second(s)
Use STUN		Sync Clock Time	
Convert URI		Enable Session Timer	
DTMF Type	RFC2833 🗸	Session Timeout	0 Second(s)
DTMF SIP INFO Mode	Send */# 🗸	Enable Rport	\checkmark
Transportation Protocol	UDP 🔽	Enable PRACK	\checkmark
Local Port	5060	Auto Change Port	\checkmark
SIP Version	RFC3261 🗸	Keep Authentication	
Caller ID Header	PAI-RPID-	Auto TCP	
Enable Strict Proxy		Enable Feature Sync	
Enable user=phone		Enable GRUU	
Enable SCA			
RTP Encryption		RTP Encryption Key	
A	Apply		

SIP					
Field Name	Explanation				
Basic Settings (Choose the SIP line to configured)					
Line Status	 Display the current line status at page loading. To get the up to date line status, user has to refresh the page manually. There are a few status here: 1) Inactive, indicates that this line is not activated yet, user can activate the line by selecting the option "activate". 2) Timeout, indicates the SIP registration status timeout. It means that there's no response from SIP server. User may need to check the network or SIP server IP address and port. 3) Registered, indicates the SIP account is registered to SIP server successfully, is able to send or receive calls. 4) 403 forbidden, indicates the SIP error code 403, means SIP server rejected the SIP registration because the username and password are incorrect. User will need to check the username and password, they must be matched with the username and password which were provided by SIP server. 5) Other SIP error code, check SIP protocol standard, or contact support. 				
Username	Enter the username of the service account, assigned by IPPBX administrator, or provided by ISP provider.				
Display name	Enter the display name to be sent in a call request.				
Authentication Name	Name Enter the authentication name of the service account, which is assigned by IPPBX administrator, or provided by ISP provider.				
Authentication	Enter the authentication password of the service account, which is				
Password	assigned by IPPBX administrator, or provided by ISP provider.				
Activate	Whether the service of the line should be activated				

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SIP Proxy Server	Enter the IP or FQDN address of the SIP proxy server				
Address SIP Proxy Server Port	Enter the SIP proxy server port, default is 5060				
Outbound proxy	Enter the IP or FQDN address of outbound proxy server provided by the				
address	service provider				
Outbound proxy port Realm	Enter the outbound proxy port, default is 5060				
	Enter the SIP domain if requested by the service provider				
Codecs Settings	ilability of the endoes by adding or remove them from the list				
	ilability of the codecs by adding or remove them from the list.				
Advanced Settings					
Subscribe For Voice	Enable the device to subscribe a voice message waiting notification, if				
Message	enabled, the device will receive notification from the server if there is voice				
	message waiting on the server				
Voice Message	Set the number for retrieving voice message				
Number	5 5				
Voice Message	Set the interval of voice message notification subscription				
Subscribe Period	Set the interval of voice message notification subscription				
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically				
Blocking Anonymous	automationly				
Call	Reject any incoming call without presenting caller ID				
Use 182 Response for					
Call waiting	Set the device to use 182 response code at call waiting response				
Anonymous Call					
Standard	Set the standard to be used for anonymous				
Dial Without					
Registered	Set call out by proxy without registration				
Click To Talk	Set Click To Talk				
User Agent	Set the user agent, the default is Model with Software Version.				
Response Single	If setting enabled, the device will use single codec in response to an				
Codec	incoming call request				
Ring Type	Set the ring tone type for the line				
	Set the type of call conference, Local=set up call conference by the device				
Conference Type	itself, maximum supports two remote parties, Server=set up call				
	conference by dialing to a conference room on the server				
Server Conference	Set the conference room number when conference type is set to be				
Number	Server				
Transfer Timeout	Set the timeout of call transfer process				
Enable Long Contact	Allow more parameters in contact field per RFC 3840				
Use Quote in Display	Whether to add quote in display name				



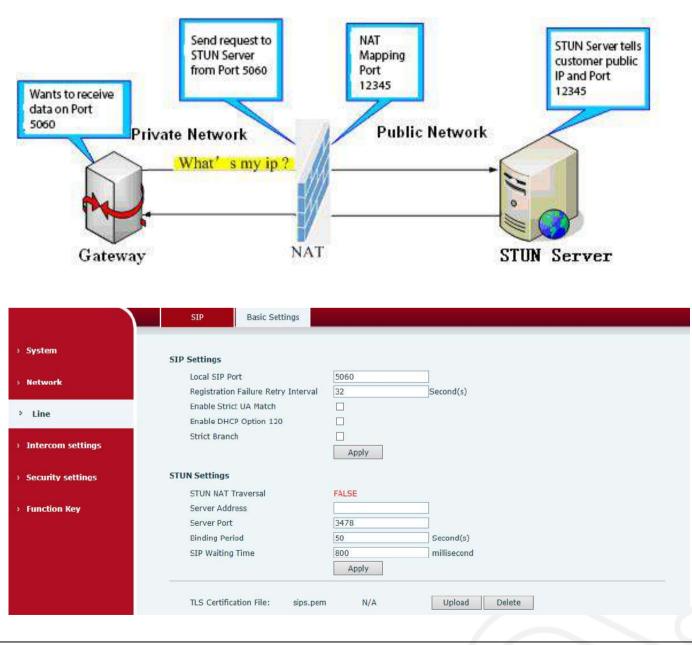
Name		
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field.	
Specific Server Type	Set the line to collaborate with specific server type	
Registration Expiration	Set the SIP expiration interval	
Use VPN	Set the line to use VPN restrict route	
Use STUN	Set the line to use STUN for NAT traversal	
Convert URI	Convert not digit and alphabet characters to %hh hex code	
DTMF Type	Set the DTMF type to be used for the line	
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'	
Transportation Protocol	Set the line to use TCP or UDP for SIP transmission	
Local Port	Set the Local Port	
SIP Version	Set the SIP version	
Caller ID Header	Set the Caller ID Header	
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server, it will use the source IP address, not the address in via field.	
Enable user=phone	Sets user=phone in SIP messages.	
Enable SCA	Enable/Disable SCA (Shared Call Appearance)	
Enable BLF List	Enable/Disable BLF List	
Enable DNS SRV	Set the line to use DNS SRV which will resolve the FQDN in proxy server into a service list	
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened	
Keep Alive Interval	Set the keep alive packet transmitting interval	
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if there is not new session timer event update received after the timeout period	
Session Timeout	Set the session timer timeout period	
Enable rport	Set the line to add rport in SIP headers	
Enable PRACK	Set the line to support PRACK SIP message	
Enable DNS SRV	Set the line to use DNS SRV which will resolve the FQDN in proxy server into a service list	
Auto Change Port	Enable/Disable Auto Change Port	



Auto TCP	Using TCP protocol to guarantee usability of transport for SIP messages
AULOTOP	above 1500 bytes
Enable Feature Sync	Feature Sycn with server
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
RTP Encryption	Enable RTP encryption such that RTP transmission will be encrypted
RTP Encryption Key	Set the pass phrase for RTP encryption

b) Basic Settings

STUN -Simple Traversal of UDP through NAT -A STUN server allows a phone in a private network to know its public IP and port as well as the type of NAT being used. The equipment can then use this information to register itself to a SIP server so that it can make and receive calls while in a private network.



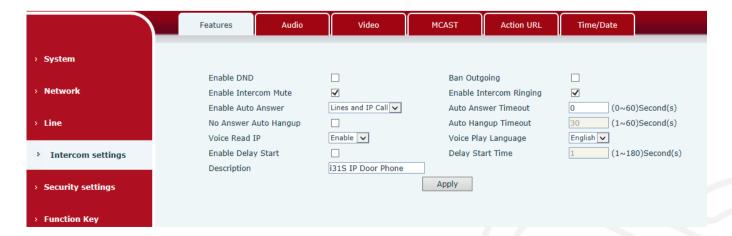
Basic Settings



Field Name	Explanation				
SIP Settings					
Local SIP Port	Set the local SIP port used to send/receive SIP messages.				
Registration Failure Retry Interval	ailure Set the retry interval of SIP REGISTRATION when registration failed.				
Enable Strict UA Match	Enable or disable Strict UA Match				
Enable DHCP Option 120	DHCP Server would respond an OPTION message to the request from DHCP client. To working with the terminal device, Access device and DHCP policy server would be able to implement the zero configuration and auto provisioning. OPTION 120 is one of the OPTIONS in which the device could obtain the SIP server address from the ACK response sent back by the DHCP server. Then the SIP Agent of terminal device starts register with the SIP server address.				
Strict Branch The value determined whether it's exactly matched the Branch					
STUN Settings					
Server Address	STUN Server IP address				
Server Port	STUN Server Port – Default is 3478.				
Binding Period	STUN blinding period – STUN packets are sent at this interval to keep the NAT mapping active.				
SIP Waiting Time Waiting time for SIP. This will vary depending on the network.					
TLS Certification File					
Upload or delete the TLS certification file used for encrypted SIP transmission.					
Note: the SIP STUN is used to achieve the SIP penetration of NAT, and the realization of a					
service, when the equipment configuration of the STUN server IP and port (usually the default is 3478), and select the Use Stun SIP server, the use of NAT equipment to achieve penetration.					

(4) Intercom settings







Features				
Field Name	Explanation			
Basic Settings				
Enable DND	DND might be disabled phone for all SIP lines, or line for SIP individually.			
	But the outgoing calls will not be affected			
Ban Outgoing	If enabled, no outgoing calls can be made.			
Enable Intercom	If anabled, mutae incoming calle during an intercom call			
Mute	If enabled, mutes incoming calls during an intercom call.			
Enable Intercom	If anabled, playe intercom ring tang to glart to an intercom call			
Ringing	If enabled, plays intercom ring tone to alert to an intercom call.			
Enable Auto Answer	Enable Auto Answer function			
Auto Answer Timeout	Set Auto Answer Timeout			
No Answer Auto	Enchle outemetically hend up when he enouver			
Hangup	Enable automatically hang up when no answer			
Auto Hangup	Configuration in a set time, automatically hand up when no answer			
Timeout	Configuration in a set time, automatically hang up when no answer			
Voice Read IP	Enable or disable voice broadcast IP address			
Voice Play Language	Set language of the voice prompt			
Enable Delay Start	Enable or disable the start delay			
Delay Start Time	Set start delay time			
Description	Device description displayed on IP scanning tool software. Initial Value is			
Description	"i18S".			

b) Audio

This page configures audio parameters such as voice codec, speak volume, mic volume and ringer volume.



	Features Audio	Video	MCAST Action URL	Time/Date:	
System					
A	udio Settings				
- metwork	First Codec	G.722 ¥	Second Codec	G.711A 💙	
	Third Codec	G.711U 💌	Fourth Codec	G.729AB	
Line	Fifth Codec	None	Sixth Codec	None 🖌	
Intercom settings	DTMF Payload Type	101 (96~127)	Default Ring Type	Type 1 💙	
· Intercom settings	G.729AB Payload Length	20ms 🔽	Tone Standard	United St: 🔽	
a construction and a construction of the const	G.722 Timestamps	160/20ms	G.723.1 Bit Rate	6.3kb/s	
Security settings	Speakerphone Volume	5 (1~9)	MIC Input Volume	5 (1~9)	
Function Key	Broadcast Output Volume	5 (1~9)	Signal Tone Volume	4 (0~9)	
- Function Key	Enable VAD				
		Apply			
s	ipeaker Settings				
	Speaker	Panel Spe	External Speaker Power	10 💟 W	
		Apply			
A	EC Settings				
	Speaker Limit in Double Talk	12 💟	Local Noise Inhibition in No Talking	18	
	Speaker Inhibition in Double Talk	8	Mic Inhibition in Double Talk	6	
		Apply	Reset		

Audio Setting				
Field Name	Explanation			
First Codec	The first codec choice: G.711A/u, G.722, G.723.1, G.729AB, G.726-32			
Second Codec	The second codec choice: G.711A/u, G.722, G.723.1, G.729AB, G.726-32,			
Second Codec	None			
Third Codec	The third codec choice: G.711A/u, G.722, G.723.1, G.729AB, G.726-32,			
	None			
Fourth Codec	The forth codec choice: G.711A/u, G.722, G.723.1, G.729AB, G.726-32,			
	None			
DTMF Payload	The RTP Payload type that indicates DTMF. Default is 101			
Туре				
Default Ring Type	Ring Sound – There are 9 standard types and 3 User types.			
G.729AB Payload	G.729AB Payload Length – Adjusts from 10 – 60 ms.			
Length				
Tone Standard	Configure tone standard area.			
G.722 Timestamps	Choices are 160/20ms or 320/20ms.			
G.723.1 Bit Rate	Choices are 5.3kb/s or 6.3kb/s.			
Speakerphone	Set the speaker calls the volume level.			
Volume				
MIC Input Volume	Set the MIC calls the volume level.			
Broadcast Output	Set the broadcast the output volume level.			
Volume				
Signal Tone	Set the audio signal the output volume level.			
Volume				
Enable VAD	Enable or disable Voice Activity Detection (VAD). If VAD is enabled, G729			
	Payload length cannot be set greater than 20 ms.			
Speaker Settings				



These settings are only for the devices which support multiple output power. Be aware of that, the selected output power must be less than the real output power of the external speak, otherwise the external speak might be damaged.

	The embedded speaker can be set to use static output power mode, and the external					
Speaker	speak can be set as 10W, 20W, 30W output power. NOTE: this device support					
	embedded speaker					
External Speaker	Set the external speaker power, it must be lower than the real power of the external					
Power	speaker, otherwise the external speaker might be damaged.					
AEC Settings						
Speaker Limit in	Limit maximum volume of the speaker while it's in the two-way					
Double Talk	conversation, the bigger the value, the loader the volume allowed.					
	While there's no talking on the conversation, the background noise will be inhibited,					
Local Noise	this value determined how much it's inhibited. The higher the value, the more					
Inhibition in No	background noise will be inhibited. It's not recommended to set it too big, because					
Talking	there will be more background noise while talking in the conversation.					
Speaker Inhibition in	Set the speaker inhibition while it's in the two-way conversation, the higher of the					
Double Talk	inhibition value, the smaller of the volume.					
Mic Inhibition in	Set the MIC inhibition while it's in the two-way conversation, the higher of the inhibition					
Double Talk	value, the smaller of the volume.					

c) Video

This page allows you to set the video capture and video encode.

	Features A	udio Video	MCAST A	ction URL Time/Date	
> System					
› Network	Camera Status	Active			
› Line	Max Access Num Max M Num	5	Use	0	
> Intercom settings	Max S Num Video Capture>>	5	Use	0	
Security settings	Video Encode>>				
› Function Key	Advanced Settings >> RTSP Information				
	Main Stream Url :	real_stream	admin&password=tlJwpbo6&c	-	Preview
	Sub Stream Url :	rtsp://172.18.3.40/user= real_stream	admin&password=tlJwpbo6&c	hannel=1&stream=1.sdp?	Preview



Video Capture>>			
IRCUT Mode	Automatic 🔽	Day/Night Mode	Automatic 🗸
White Balance	Automatic 🗸	Horizon Flip	Enable 🗸
Anti Flicker	Disable	Vertical Flip	Enable 🗸
IR Swap	Disable	DNC Threshold	29 (10~50)
Backlight Compensat	ion Disable V	AutoFill Sensitivity	5 (1~10)
wide dynamic	Enable	Wide dynamic upper limit	30 (0~100)
Fill Light	Enable V		
	Default	Apply	
	Default		
Video Encode>>			
Video Elicode>>			
	Main Stream	Sub Stream	
Encode Format	H264 V	H264 🗸	
Resolution	720P 🗸	CIF	
Frame Rate	20 🗸	20 🗸	
Bitrate Control	VBR	VBR	
Quality	General	General	
Bitrate	1700 🗸	318 🗸	
I Frame Interval	2 (1~12)S	2 (1~12)S	
Activate	\checkmark	\checkmark	
	Default	Apply	
Encode Static config	Base line		
Encode Static comy			
	Apply		
	Арргу		
Advanced Settings >>			
Video Direction	Sendonly 🗸		
H.264 Payload Type	117 (96~127)		
	Default	Apply	
RTSP Information			
Main Stream Url :	rtsp://172.18.3.40/user=admin&passv real_stream	word=tlJwpbo6&channel=1&stream	=0.sdp? Preview
Sub Stream Url :	rtsp://172.18.3.40/user=admin&passv	word=tlJwpbo6&channel=1&stream	=1.sdp? Preview
Sub Stream Off :	real_stream		Fleview

Video					
Field Name	Explanation				
Camera Status: Display the relevant information of the camera, including maximum access,					
maximum stream, maximum sub stream, and the status.					
Video Capture					
	Auto: IRCUT switches according to the actual ambient light level of the camera				
IRCUT Mode Synchronization: The switching of the IRCUT is determined by the actual					
	brightness of the IR lamp.				



Day/Night Mode	Automatic: automatically switches according to the DNC Threshold and the brightness of the actual environment where the camera is located Day Mode: The camera's video screen is always colored, if there is IR-cut will be synchronized to switch. Night Mode: the camera's video screen is always black and white, if there is IR-cut will be synchronized switch.					
White Balance Automatic: Automatically adjusts according to the actual environment the camera is located. Outdoor: installed in the outdoor preferred. Indoor: installed in the room preferred.						
Horizon Flip	The video is flipped horizontally					
Anti Flicker	Enable the option. In a fluorescent environment can eliminate the video horizontal scroll					
Vertical Flip	The video is flipped horizontally					
IR Swap	IR-cut filter switch					
DNC	In the Day / Night mode Auto option, the color switching black and white					
Threshold	threshold is set					
Backlight Compensation In front of a very strong background light can see people or objects clearly						
AutoFill	In the environment changes in light and shade, the higher the sensitivity the					
Sensitivity	faster the video changes					
wide Dynamic	Set wide dynamic					
Wide Dynamic Upper Limit	Change the brightness of the background image, the higher the brighter.					
Fill Light	Enable or disable Fill Light					
Video Encode	·					
Encode Format	Only H.264 encoding format is supported					
Resolution	Main stream: support 720P Sub-stream: you can select CIF (352 * 288), D1 (720 * 576)					
Frame Rate The larger the value is, the more coherent the video would be recommend adjusted.						
Bitrate Control CBR: If the code rate (bandwidth) is insufficient, it is preferred. VBR: Image quality is preferred, not recommended.						
Quality	Video quality adjustment, the better the quality needs to transfer faster					
Bit rate	It is proportional to video file size, not recommend adjusted.					
I Frame	The greater the value is, the worse the video quality would be, otherwise the					
Interval	better video quality would be; not recommend adjusted.					
Activate	When you selected it, the stream is enabled, otherwise disabled					
Encode Static	config					



Select the video codec type, it's recommended to use "Base Line" to stay the same as the video output or stream receiver.

output of stream receiver.							
Advanced Se	Advanced Settings						
Video	Select the transport type of the video streem						
Direction	Select the transport type of the video stream						
H.264 Payloa	d Set the periled type of H 264						
Туре	Set the payload type of H.264						
RTSP Information							
Main Stream	Access the main address of PTSP						
Url	Access the main address of RTSP						
Sub Stream	Access the child address of RTSP						
Url							

d) MCAST

	Features	Audio	Video	MCAST	Action URL	Time/Date	
> System							
	MCAST Settings						
> Network	Priority		1	v			
› Line	Enable Page Prio	rity					
/ Line	Index/Prior	ity	Name			Host:port	
> Intercom settings	1						
· Intercom settings	2						
Committee on Minner	3						
 Security settings 	4						
The Market	5						
• Function Key	7						
	8						
	9						
	10						
			Apply				

It is easy and convenient to use multicast function to send notice to each member of the multicast via setting the multicast key on the device and sending multicast RTP stream to pre-configured multicast address. By configuring monitoring multicast address on the device, monitor and play the RTP stream which sent by the multicast address.

MCAST Settings

Equipment can be set up to monitor up to 10 different multicast address, used to receive the multicast RTP stream sent by the multicast address.

Here are the ways to change equipment receiving multicast RTP stream processing mode in the Web interface: set the ordinary priority and enable page priority.

• Priority:

In the drop-down box to choose priority of ordinary calls the priority, if the priority of the



incoming flows of multicast RTP, lower precedence than the current common calls, device will automatically ignore the group RTP stream. If the priority of the incoming flow of multicast RTP is higher than the current common calls priority, device will automatically receive the group RTP stream, and keep the current common calls in state. You can also choose to disable in the receiving threshold drop-down box, the device will automatically ignore all local network multicast RTP streams.

- The options are as follows:
 - ♦ 1-10: To definite the priority of the common calls, 1 is the top level while 10 is the lowest
 - ♦ Disable: ignore all incoming multicast RTP stream
 - \diamond Enable the page priority:

Page priority determines the device how to deal with the new receiving multicast RTP stream when it is in multicast session currently. When Page priority switch is enabled, the device will automatically ignore the low priority multicast RTP stream but receive top-level priority multicast RTP stream and keep the current multicast session in state; If it is not enabled, the device will automatically ignore all receiving multicast RTP streams.

• Web Settings:

MCA	ST Settings		
	Priority	1 💙	
	Enable Page Priority		
	Index/Priority	Name	Host:port
	1	SS	239.1.1.1:1366
	2	ee	239.1.1.1:1367

The multicast SS priority is higher than that of EE, which is the highest priority.

Note: when pressing the multicast key for multicast session, both multicast sender and receiver will beep.



Listener configuration

MCAST Settings

CA	AST Settings							
	Priority	3 💌						
Í	Enable Page Priority							
	Index/Priority	Name	Host:port					
	1	group 1	224.0.0.2:2366					
	2	group 2	224.0.0.2:1366					
	3	group 3	224.0.0.6:3366					
	4							
	5							
	6							
	7							
	8							
	9							
	10							

• Blue part (name)

"Group 1", "Group 2" and "Group 3" are the names of the monitoring multicast which you set. The group name will be displayed on the screen when you answer the multicast. If you have not set, the screen will display the IP: port directly.

• Purple part (host: port)

It is a set of addresses and ports to listen, separated by a colon.

• Pink part (index / priority)

Multicast is a sign of listening, but also the monitoring multicast priority. The smaller number refers to higher priority.

• Red part (priority)

It is the general call, non-multicast call priority. The smaller number refers to higher priority. The followings will explain how to use this option:

- ♦ The purpose of setting monitoring multicast "Group 1" or "Group 2" or "Group 3" is able to launch a multicast call.
- ♦ All equipment has one or more common non-multicast communications.
- When you set the Priority for the disable, any level of multicast will not answer. Multicast call is rejected.
- When you set the Priority to a value, only higher than the priority of multicast can get access. If you set the Priority is 3, group 2 and group 3 for priority level equal to 3 or less than 3 were rejected, 1 priority is 2 higher than ordinary call priority. Device can answer the multicast message and hold the other call at the same time

• Green part (Enable Page priority)

- User can set whether to open more priority to be is the priority of multicast, Multicast is the pink part number. Explain how to use:
- The purpose of setting monitoring multicast "group 1" or "3" is to set up listening "group of 1" or "3" as multicast address multicast call.
- ♦ All equipment has been a path or multi-path to multicast phone, such as listening to



"multicast information group 2".

- If multicast is a new "group of 1", then the call will come in. Because "the priority group 1" is 2, higher than the current call "priority group 2" 3
- If multicast is a new "group of 3", "1" will listen to the equipment and maintain the "group of 2".
 Because "the priority group 3" is 4, lower than the current call "priority group 2" 3,

Multicast service

- **Send:** When configured done, our key will press shell on the corresponding equipment. The equipment will turn into the talking interface directly. The premise is to ensure no current multicast call and 3-way of the case. Then the multicast can be established.
- **Monitor:** It is the IP port and priority configuration monitoring device. When the call is initiated and incoming multicast, it will turn into the Talking interface equipment directly.

e) Action URL

()	Features Audio	Video	MCAST	Action URL	Time/Date
System					
Network	Action URL Event Settings				
100000000	Active URI Limit IP				
• Line	Setup Completed				
- Marrie	Registration Succeeded				
> Intercom settings	Registration Disabled	ļ.			
· Intercom seconds	Registration Failed				
Security settings	Off Hooked				
seconcy seconds	On Hooked				
and the second second	Incoming Call				
Function Key	Outgoing calls				
	Call Established				
	Call Terminated	-			
	DND Enabled	1			
	DND Disabled	1			
	Mute				
	Unmute				
	Missed calls				
	IP Changed	-			
	Idle To Busy	-			
	Busy To Idle		1		
		Apply			

Action URL Settings

URL for various actions is performed by the phone. These actions are recorded and sent as xml files to the server. Sample format is http://InternalServer /FileName.xml



f) Time/Date

	Features	Audio	video	MCAST	Action URL	Time/Date
System						
	Network Time S	erver Settings				
Network	Time Synchr	ronized via SNTP	2			
N7/1		onized via DHCP				
10	Primary Tim	e Server	time.nist.gov			
and the second second	Secondary T	îme Server	pool.ntp.org			
Intercom settings	Time zone		(UTC+8) 中国	,新加坡,澳大利亚,Ru	* •	
Accession and the second	Resync Perio	bd	60	(1~500	0)Second(s)	
ecurity settings	Date Format					
inction Key	Date Formal		1 JAN M	N V		
1339-1227-1229-1229						
			Apply	A%		
	Daylight Saving	Time Settings				
	Location		中国(北京)	\sim		
	OST Set Typ	ić.	Disabled	Y		
			Apply			
	Manual Time Se	ttings 0				
	2018-03-03	17 🔽	39	Apply		
	System Tim	e: 2018-03- 17:40	03	all all these		

Time/Date						
Field Name	Explanation					
Network Time Server Settings						
Time Synchronized via SNTP	Enable time-sync through SNTP protocol					
Time Synchronized via DHCP	Enable time-sync through DHCP protocol					
Primary Time Server	Set primary time server address					
Secondary Time	Set secondary time server address, when primary server is not reachable, the device					
Server	will try to connect to secondary time server to get time synchronization.					
Time zone	Select the time zone					
Resync Period	Time of re-synchronization with time server					
Date Format						
Date Format	Select the time/date display format					
Daylight Saving Tim	e Settings					
Location	Select the user's time zone specific area					
DST Set Turce	Select automatic DST according to the preset rules of DST, or the manually input					
DST Set Type	rules					
Manual Time Setting	Manual Time Settings					
The time set by hand, need to disable SNTP service first.						



(5) Security settings

System	Input Settings			
oy strain	Input Detect			
Network	Trigger Mode	Low Level Trigger(Close Trigger)	Alert messag	ge send to server
	Output Settings			
Line	Output Response			
the second second second	Output Level	High Level(NC:closed)	Output Duration	5 (1~600)s
tercom settings	Alert Trigger Setting			
Security settings	Alarm Ring Duration	5 (1~600)s		
Security securitys	🗹 Input Trigger	Enable Ring 💌	Output Last	By Duration
unction Key	Remote DTMF Trigger	Enable Ring 🔽	Trigger Code	1234
includin Koya	Remote SMS Trigger	Enable Ring 🔽	Trigger Message	ALERT=OUT1_SOS
	Call State Trigger	Falking 🔽		
			Apply	
	Tamper Alarm Settings			
	Tamper Alarm			
	Alarm command	Tamper_Alarm	Reset command	Tamper_Reset
	Reset Alerting Status	Reset	Ring Type	Default 🗸
			Apply	
	Server Settings			
	Server Address		Send message to the	server when the alarm is triggered
	Message:Alarm_Info:Descr	iption=i315 IP Door Phone;SIP U	ser=5528:Mac=0c:38:30	e:1e:61:dd;IP=172,18.3.40;port=Input1
			Apply	

Security setting	Security settings				
Field Name	Explanation				
Input settings					
Input Detect	Enable input detection				
	Low Level Trigger(Close Trigger), Double short circuit detection port(If it is				
Triggor Modo	single port, is the low level)Detection to trigger when closed				
Trigger Mode	High Level Trigger(Disconnect Trigger),Double short circuit detection port(If it is				
	single port, is the high level)Detection to trigger when disconnect.				
Alert message	When meet the input port to trigger condition, to the server sends the alarm				
send to server	information correspondence.				
Output Setting	S				
Output	Enable output port detection				
Response					
	Low Level(NO: always on)When meet the trigger condition, trigger the NO port				
Output Level	disconnected.				
Oulpul Level	High Level(NC: always off)When meet the trigger condition, trigger the NC				
	port close.				
Output	Define the output Duration change of output port. (1~600S)				
Duration					
Alert Trigger S	etting				
Alarm Ring	Define the output Duration change of output port. (1~600S)				
Duration	Define the output Duration change of output port. (1~0005)				
Input Trigger	When the input port meet to trigger condition, the output port will be triggered				



	
	By Duration:
	Received the terminal equipment to send the DTMF password, if correct, which
	triggers the corresponding output port (The Port level time change, By < Output
Remote DTMF	Duration> control)
Trigger	By Calling State:
пудеі	During the call, receive the terminal equipment to send the DTMF password, if
	correct, which triggers the corresponding output port (The Port level time
	change, by call state control, after the end of the call, port to return the default
	state)
Remote SMS	In the remote device or server to send instructions to ALERT=[instructions], if
Trigger	correct, which triggers the corresponding output port
	The port output continuous time synchronization and trigger state changes,
Call State	including the trigger conditions: 1, call; 2, call and singing; 3, singing; three
Trigger	models. (for example: the call trigger output port, will be in conversation state
	continued to output the corresponding level)
Tamper Alarm	Settings
Tamper Alarm	When the selection is enabled, the tamper detection enabled
Alarm	When detected someone tampering the equipment, will be sent alarm to the
command	corresponding server
Reset	When the equipment receives the command of reset from server, the
command	equipment will stop alarm
Reset Alerting	Directly stop the alarm from equipment in the Webpage
Status	Directly stop the alarm nom equipment in the webpage
Server Settings	S
Server	Configure remote response server address (including remote response server
Address	address and tamper alarm server address)



(6) Function Key

> System							
> Network	Function Key S	ettings					
	Кеу	Туре	Number 1	Number 2	Line	Subtype	
› Line	DSS Ke	1 Hot Key	8102		SIP1 🗸	Speed Dial	~
	DSS Ke	2 None			SIP1 🗸	Speed Dial	~
Intercom settings							
› Security settings	Advanced Sett	-	Enable 🔽	Use Hot Key to Hangu	D Enab	ole V	
> Function Key	Hot K	ey Dial Mode Select	Main-Secondary 🗸				
	Call Swit	ched Time 16 (5~	50)S Day Start Time 06:	00 (00:00~23:59) Day	y End Time	18:00 (00:00~23	:59)
			Ар	ply			

> Key Event

You might set up the key type with the Key Event.

Key	Туре	Number 1	Number 2	Line	Subtype	
DSS Key 1	Key Event			SIP1 .	OK	2
		A	pply		None Dial Release	
					OK	
					Handfree	

Туре	Subtype	Usage
	None	No responding
	Dial	Dialing function
Key Event	Release	Delete password input, cancel dialing input and end
		call
	OK	Identification key

> Hot Key

You might enter the phone number in the input box. When you press the shortcut key, equipment would dial preset telephone number. This button can also be used to set the IP address: you can press the shortcut key to make a IP call directly.

Key	Туре	Number 1	Number 2	Line	Subtype
DSS Key 1	Hot Key 🔻			SIP1 V	Speed Dial 🔹
					Speed Dial
		Ap	oply		Intercom

Туре	Number	Line	Subtype	Usage



Hot Key	Fill the called ot Key party's SIP	The SIP account correspond	Speed Dial	Using Speed Dial mode together with Enable Speed Dial Hangup Enable, can define whether this call is allowed to be hung up by re-pressing the speed dial key.
	account or IP address	ing lines	Intercom	In Intercom mode, if the caller's IP phone supports Intercom feature, the device can automatically answer the Intercom calls

> Multicast

Multicast function is to deliver voice streams to configured multicast address; all equipment monitored the multicast address can receive and play it.

The DSS Key multicast web configuration for calling party is as follow:

Key	Туре	Number 1	Number 2	Line	Subtype	
DSS Key 1	Multicast 🔹			SIP1 T	G.722	
		A	pply		G.711A G.711U G.722	
					G.723.1 G.726-32 G.729AB	

Туре	Number	Subtype	Usage
		G.711A	Narrowhand apageh adding (4Khz)
	Set the host IP address	G.711U	Narrowband speech coding (4Khz)
Multicast r	and port number; they	G.722	Wideband speech coding (7Khz)
	must be separated by a	G.723.1	
	colon	G.726-32	Narrowband speech coding (4Khz)
		G.729AB	

♦ operation mechanism

You can define the DSS Key configuration with multicast address, port and used codec. The device can configure via WEB to monitor the multicast address and port. When the device makes a multicast, all devices monitoring the address can receive the multicast data.

♦ calling configuration

If the device is in calls, or it is three-way conference, or in initiated multicast communication, the device would not be able to launch a new multicast call.



E.Appendix

1. Technical parameters

Communica	ation protocol	SIP 2.0(RFC-3261)		
Main chipse	et	Broadcom		
Kau	DSS key materials	Stainless steel		
Кеу	DSS Key	1 or 2		
	Audio amplifier	3W		
	Volume control	Adjustable		
	Full duplex	Support (AEC)		
Speech	speakerphone	Support (AEC)		
flow	DTMF TYPE	In-band, Out-of-band(RFC 2833), SIP INFO		
110 W	wideband speech	G.722		
	code	6.722		
	Narrowband	G711A/u, G.723.1, G.729AB, ILBC, AMR		
	speech code	07 11A/d; 0.723.1, 0.723AB, 1600; AMIK		
	Security linkage	1 embedded short circuit input interface		
Port	Security initiage	1 embedded short circuit output interface		
	WAN	10/100BASE-TX s Auto-MDIX, RJ-45		
Camera		1/3 "color CMOS, wide angle		
Power supp	oly mode	12V / 1A DC or PoE		
Cables		CAT5 or better		
Shell Mater	ial	Cast aluminium panel, Cast aluminium back shell		
Working ter	mperature	-40°C to 70°C		
Working hu	imidity	10% - 90%		
Storage ten	nperature	-40°C to 70°C		
Installation	way	Wall-mounting or Flush-mounting		
Dimension		Wall-mounting: 223*130*74mm		
DIIICIISIOII		Flush-mounting: 270*150*61mm		



2. Basic functions

- 2 SIP Lines
- PoE Enabled
- Full-duplex speakerphone (HF)
- Intelligent DSS Keys (Speed Dial/intercom etc.)
- Wall-mounting / Flush-mounting
- Special integrated noise reduction module
- Dual microphone Omnidirectional voice pickup
- 1 embedded short circuit input interface
- 1 embedded short circuit output interface. Support 4 controlled events: remote DTMF; remote server's commands; interact with short circuit input; talking status
- Anti-tamper switch
- Record voice and video during calls (Optional)
- All in ONE: Radio and intercom, intelligent security function
- Industrial standard certifications: IP65, IK10, CE/FCC





3. Schematic diagram



4. The broadcast terminal configuration notice

♦ How to avoid an incoherency sound when the broadcast playing?

When the terminal used as broadcast, the speaker is loud. If do not set mute for microphone, the AEC (echo cancellation) of equipment will be activated, which leads the sound incoherence. In order to avoid such circumstance, when the equipment turn to use as a radio should be set as intercom mode. Then activate the intercom mute, so as to ensure the broadcast quality.

	Features Audio	Video	MCAST	Action URL	Time/D	Time/Date	
› System							
Network	Enable DND Enable Intercom Mute			rcom Ringing			
Line	Enable Auto Answer No Answer Auto Hangup Voice Read IP	Enable	Auto Answe Auto Hangu Voice Play L	p Timeout	0 30 English V] (0~60)Second(s)] (1~60)Second(s)	
> Intercom settings	Enable Delay Start Description	i31S IP Door Phone	Delay Start	Time	1] (1~180)Second(s)	
Security settings			Apply				
Function Key							

♦ How to improve broadcasting tone quality?

In order to obtain better broadcast quality, recommend the use of the HD (G.722) mode for broadcast. Voice bandwidth will be by the narrow width (G.711) of 4 KHz, which is extended to broadband (G.722)7 KHz. When combined with the active speaker, the effect will be better.

Broadcast Output Volume 5 (1~9) Signal Tone Volume 4 (0-		Features Audio	Video	MCAST Action URL	Time/Date
I Network First Codec G.722 Sacond Codec G.711A Y I Line Third Codec G.711U Fourth Codec G.72AB Y I Line Fifth Codec None Social Codec None Y I Intercom settings DTMF Payload Type 101 (96~127) Default Ring Type Type 1 I Security settings G.723AB Payload Length 20ms/Y G.723A Bit Pathe 6.3kb/e ^C Speakerphone Yolume 5 (1~9) MC Input Volume 5 (1~9) Broadcast Output Volume 5 (1~9) Signal Tone Volume 4 (0~1000)	1 System				
Line First Codec C.722 Saccad Codec C.711A V Third Codec G.711U Fourth Codec G.729AB Fourth Codec G.729AB V Intercom settings Fifth Code Bione Saccad Codec None V Intercom settings G.729AB Payload Type 101 (96~127) Default Ring Type Type 1 Security settings G.722 Timestampe 160/20ms G. 22.21. Bit Bate 6.3bc/s/C Security settings Speakerphone Volume 5 (1~9) MC Liput Volume 5 (1~9)	1. Notwork	Audio Settings			
Punc Fifth Codec None Suth Codec None Codec > Intercom settings DTMF Payload Type 101	1. Network	First Codec	G.722 V	Second Codec	G.711A ¥
Fifth Codec None Stath Codec None None DTMF Payload Type 101 (96~127) Default Ring Type Type 1 > Intercom settings G.729AB Payload Length 20ms Tone Standard United Str > Security settings G.729 mistamps 160/20ms G.729.1 kit Rate 6.3kb/s > Broadcast Output Volume 5 (1~9) MIC Input Volume 5 (1~9)	an anna an	Third Codec	G.711U 👻	Fourth Codec	G.729AB
Intercom settings G.725AB Payload Length Z0ms V Tone Standard United St.V J. Security settings G.722 Timestamps 160/20ms V G.722.1 Bit Rate 6.3kb/e V J. Security settings Speakerphone Volume S (1~9) MIC Input Volume S (1~9) Broadcast Output Volume 5 (1~9) Signal Tone Volume 4 (0~9)	Line	Fifth Codec	None 🔽	Sixth Codec	None 🔽
Security settings C.72x89 Sayload Length ZUmit V Torie Standard United St.V J Security settings 6.722 Timestamps 160/20mit V G.722.1 Bit Rate 6.3kb/eV Breadcast Output Volume 5 (1~9) MtC tiput Volume 5 (1~9) Breadcast Output Volume 5 (1~9) Signal Tone Volume 4 (0~9)	> Intercom settings	DTMF Payload Type		Default Ring Type	Type 1
3 Security settings Speakerphone Volume S (1~9) MIC Input Volume S (1~9) Broadcast Output Volume 5 (1~9) Signal Tone Volume 4 (0~9)		G.729AB Payload Length	20ms 💌	Tone Standard	United Sti 👻
Broadcast Output Volume 5 (1~9) Signal Tone Volume (4 (0~9)	> Security settings	G.722 Timestamps	160/20ms 💙	G.723.1 Bit Rate	6.3kb/s 💙
		Speakerphone Volume	5 (1~9)	MIC Input Volume	5 (1~9)
Function Key Enable VAD	the second s	Broadcast Output Volume	5 (1~9)	Signal Tone Volume	4 (0~9)
	Function Key	Enable VAD			
			Apply		