



Intercom Quick Installation Guide



Introduction

Package Contents



DP20 Intercom



Connector



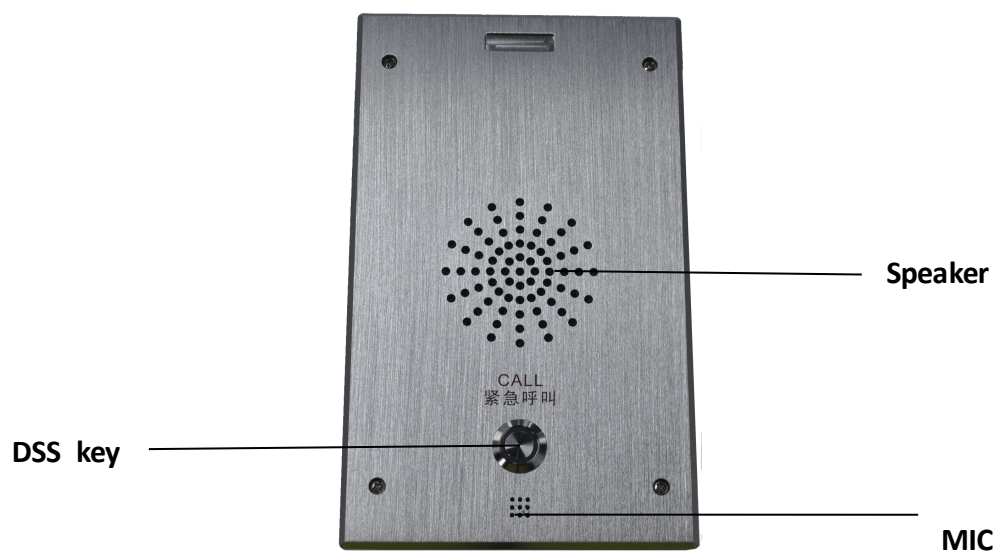
**Installation
diagram**



**Quick Installation
Guide**

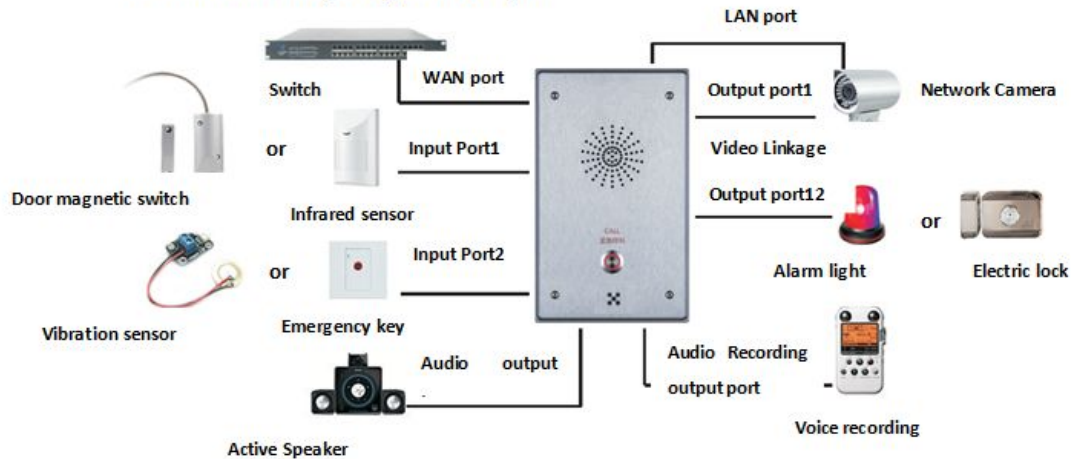


Screw and tool



Voice Intercom Configuration

IP intercom Topological Graph



Step One: Connect to the network

Connect the end of network cable to the device WAN port, another end is connected to the LAN port of the router, then the hardware connection is completed. Normally, you should set your network to DHCP mode.



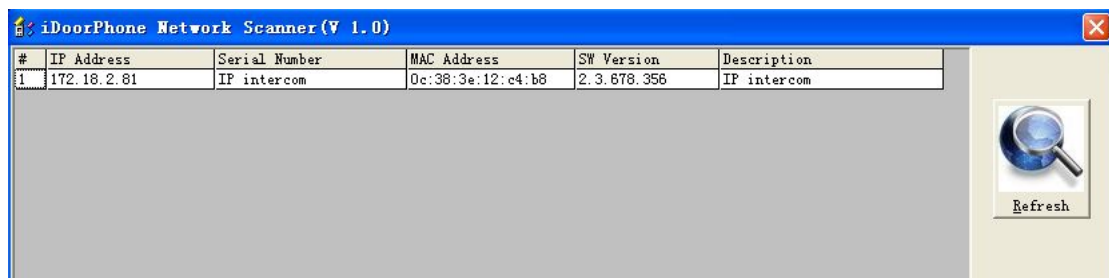
Users can call the same group of people through the VOIP phone, PC or mobile phone SIP phone software, and realizes remote control to the device(such as a door lock, Alarm lamp etc.)

Voice Intercom Configuration

Step Two: Get the device IP Address:

Methods:

1. Use the default IP scanner tool to get it: iDoorPhoneNetworkScanner
 - 1) Install the scanner tool: iDoorPhoneNetworkScanner;
 - 2) Ensure the working computer (installing IP scanner tool, exe.) is in the same local network with the corresponding device;
 - 3) Run the tool (iDoorPhoneNetworkScanner.exe), to search the IP address of corresponding device within the network.



Method 2: Long Press “#”key for 3 seconds, the intercom will report the IP numbers by itself.

Step Three: Log in the WEB admin interface of the device

Input IP address (e.g.: <http://192.168.1.149>) the Web browser, the default user name: admin, password: admin.

User:

Password:

Language: English

Voice Intercom Configuration

Step Four: Modify the device description

	AUDIO	FEATURE	DIAL PLAN	CONTACT	REMOTE CONTACT	WEB DIAL	MCAST
<ul style="list-style-type: none">> BASIC> NETWORK> VOIP> INTERCOM> SAFEGUARDING> FUNCTION KEY> MAINTENANCE> SECURITY> LOGOUT	Auto Handdown Time	3	second(s)	Enable Call Completion	<input type="checkbox"/>		
	Enable Auto Redial	<input type="checkbox"/>		Enable Silent Mode	<input type="checkbox"/>		
	Auto Redial Interval	10	(1~180)second(s)	Hide DTMF	Disabled	<input type="button" value="v"/>	
	Auto Redial Times	10	(1~100)	Ring From Headset	<input type="checkbox"/>		
	Auto Headset	<input checked="" type="checkbox"/>		Enable Intercom Mute	<input type="checkbox"/>		
	Enable Intercom	<input checked="" type="checkbox"/>		Enable Intercom Barge	<input checked="" type="checkbox"/>		
	Enable Intercom Tone	<input checked="" type="checkbox"/>		DND Return Code	480(Temporarily Not Available)	<input type="button" value="v"/>	
	P2P IP Prefix			Busy Return Code	486(Busy Here)	<input type="button" value="v"/>	
	Turn Off Power Light	<input checked="" type="checkbox"/>		Reject Return Code	603(Dedline)	<input type="button" value="v"/>	
	Emergency Call Number	110		Active URI Limit IP			
	Enable Password Dial	<input type="checkbox"/>		Push XML Server			
	Password Dial Prefix			Enable Call Waiting Tone	<input checked="" type="checkbox"/>		
	Password Length	0	(0~31)	IP Description	IP intercom		
	Enable Multi Line	<input checked="" type="checkbox"/>		Auto Answer Timeout	0	second(s)	
	Enable Auto Answer	<input checked="" type="checkbox"/>		Status Led Reuse Mode	Disable	<input type="button" value="v"/>	
	Enable Speed Dial Handdown	Enable	<input type="button" value="v"/>	Time of Dial Switch	16	(5-50)s	
	Dial Number Voice Play	Disable	<input type="button" value="v"/>				
					<input type="button" value="Apply"/>		

Step Five: Add SIP account

	SIP	IAX2	STUN	DIAL PEER
<ul style="list-style-type: none">> BASIC> NETWORK> VOIP> INTERCOM> SAFEGUARDING> FUNCTION KEY> MAINTENANCE> SECURITY> LOGOUT	SIP Line	SIP 1		
	Basic Settings >>			
	Status	Unapplied	Domain Realm	
	Server Address	172.18.1.212	Proxy Server Address	
	Server Port	5060	Proxy Server Port	
	Authentication User	607	Proxy User	
	Authentication Password	*****	Proxy Password	
	SIP User	607	Backup Proxy Server Address	
	Display Name	607	Backup Proxy Server Port	5060
	Enable Registration	<input checked="" type="checkbox"/>	Server Name	
	Codecs Settings >>			
	Advanced SIP Settings >>			
				<input type="button" value="Apply"/>

Voice Intercom Configuration

Step Six: DSS key Configuration method

The screenshot shows the 'Function Key Settings' section of the configuration interface. It includes a table for configuring four DSS keys. The table has columns for Key, Type, Number 1, Number 2, Line, and Subtype. The current settings are as follows:

Key	Type	Number 1	Number 2	Line	Subtype
DSS Key 1	Key Event			SIP1	None
DSS Key 2	Key Event			SIP1	None
DSS Key 3	Line			SIP1	Speed Dial
DSS Key 4	Line			SIP2	Speed Dial

Below the table is an 'Apply' button. Above the table, there is a 'Screen Configuration' section with a 'Contrast' field set to 5 (range 1~9) and an 'Enable Backlight' checkbox checked. An 'Apply' button is also present in this section.

Intercom software can support up to four DSS key functions

1) The Subtype configuration of Hot key

DSS type	key	Number	Line	Subtype	usage
Hot key		Fill the called party's SIP account or address	The SIP account corresponding lines	Speed Dial	In Speed dial mode, with Enable Speed Dial Handdown <input type="checkbox"/> Enable can define whether this call is allowed to be hang up by re-press the speed dial
				Intercom	In Intercom mode, if the caller's IP phone support intercom feature, can realize auto answer

Each DSS key can be configured two numbers, when the first number is busy or no answer within the set time, the call will be forwarded to the second number automatically. The Switching time of the setting: WEB→Intercom→Feature

Time of Dial Switch (5-50)s

2) The Subtype configuration of key Event

The screenshot shows the 'Function Key Settings' section with the 'Subtype' dropdown for DSS Key 1 open. The dropdown menu lists the following options: None, Speed Dial, Release, OK, and Handfree. The 'Speed Dial' option is currently selected.

Key	Type	Number 1	Number 2	Line	Subtype
DSS Key 1	Hot Key	8201		SIP1	Speed Dial
DSS Key 2	Key Event			SIP1	None
DSS Key 3	Line			SIP1	None
DSS Key 4	Line			SIP2	Release

An 'Apply' button is located at the bottom of the section.

Voice Intercom Configuration

DSS key type	Subtype	usage
key Event	None	No Answer
	Dial	Dial function
	Release	End calls
	OK	Identify key
	Hand free	The hand-free key(with hook dial, hang up)

➤ The two short circuits input configuration method

WEB→Safeguarding, As shown in the figure below

The input port Settings

The screenshot shows the 'Safeguarding' configuration page. On the left is a red sidebar with navigation links: NETWORK, VOIP, INTERCOM, SAFEGUARDING (selected), and FUNCTION KEY. The main content area is titled 'Input Settings' and 'Output Settings'. The 'Input Settings' section is highlighted with a red box and contains the following options:

- ☒ Input 1: Trigger Mode: Low Level Trigger(Close Trigger), Response Mode: ☒ Remote Response
- ☐ Input 2: Trigger Mode: Low Level Trigger(Close Trigger), Response Mode: ☒ Remote Response

The 'Output Settings' section contains the following options:

- ☐ Output 1: Output Level: High Level(NO:closed), Output Duration: 5 (1~600) s
- Output Trigger Mode: ☒ Input 1 Trigger, ☐ Input 2 Trigger

Function		Description
Trigger mode	Low Level Trigger(Close Trigger)	Double short circuit detection port(If it is single port, is the low level)Detection to trigger when closed
	High Level Trigger(Disconnect Trigger)	Double short circuit detection port (If it is single port, is the high level)) Detection to trigger when disconnect
	Remote Response	When meet the input port to trigger condition, to the server sends the alarm information correspondence. [note] Input port1 trigger, to send command format: The trigger device the IP; Port=Input1 Input port2 trigger, to send command format: The trigger device the IP; Port=Input2

Voice Intercom Configuration

➤ The two short circuits output configuration method

The screenshot shows the 'Output Settings' configuration page. On the left is a navigation menu with options: BASIC, NETWORK, VOIP, INTERCOM, SAFEGUARDING, FUNCTION KEY, MAINTENANCE, and SECURITY. The main area is titled 'Output Settings' and contains two sections for 'Output 1' and 'Output 2'. Each section has a dropdown for 'Output Level' (set to 'High Level(NO:closed)'), a dropdown for 'Output Trigger Mode' (set to 'High Level(NO:closed)'), and a section for 'Output Duration' (set to '5' seconds). Below these are checkboxes for 'Input 1 Trigger', 'Input 2 Trigger', 'Remote DTMF Trigger', 'Remote SMS Trigger', 'Call State Trigger', and 'Emergency Key Trigger'. The 'Remote DTMF Trigger' section shows 'ALERT=OUT1_SOS' and 'Talking' as options. The 'Remote SMS Trigger' section shows 'ALERT=OUT2_SOS' and 'Talking' as options.

Function		Description	
Output level	Low Level(NO: always on)	When meet the trigger condition, trigger the NO port disconnected.	
	High Level(NO: always off)	When meet the trigger condition, trigger the NO port close.	
Output Duration	1~600S	Define the output Duration change of output port.	
Output trigger mode	Input port1 trigger		When the input port1 meet to trigger condition, the output port1 will trigger(The Port level time change, By < Output Duration> control)
	Input port2 trigger		When the input port2 meet to trigger condition, the output port2 will trigger(The Port level time change, By < Output Duration> control)
	Remote DTMFtrigger	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 Duration> control)
		By Calling State	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 trigger		In the remote device or server to send instructions to ALERT=[instructions], if correct, which triggers the corresponding output port
	Call state trigger		The port output continuous time synchronization and trigger state changes, including the trigger conditions: 1,call; 2,call and singing; 3,singing; three models. (for example: the call trigger output port, will be in conversation state continued to output the corresponding level)
	Emergency key trigger		When the emergency call button to trigger the equipment shell, which triggers the corresponding output port(after the end of the call, port to return the default state)

Voice Intercom Configuration

➤ The tamper detection configuration method

Tamper Alarm Settings

☒ Tamper Alarm
 Alarm command:
 Reset command:

Function	Describe
Tamper Alarm	When the selection is enabled, the tamper detection enabled
Alarm command	When detected someone tampering the equipment, will be sent alarm to the corresponding server
Reset command	When the equipment receives the command of reset from server, the equipment will stop alarm
Reset	Directly stop the alarm from equipment in the Webpage

➤ The trigger ring type setting

Server & Trigger Ring Type Settings

Server Address:
 Input 1 Trigger Ring:
 Input 2 Trigger Ring:

Remote DTMF Trigger Ring:
 Remote SMS Trigger Ring:

Tamper Alarm Ring:
 Alarm Ring Duration: (1~600) s

Function	Description
Server Address	Configure remote response server address(including remote response server address and tamper alarm server address)
Input 1 trigger ring	When the input port 1 triggering condition is satisfied, the corresponding ring tone or alarm
Input 2 trigger ring	When the input port 2 triggering condition is satisfied, the corresponding ring tone or alarm
Remote DTMF trigger ring	When received the remote DTMF command, whether to output the ringtone
Remote SMS trigger ring	When receiving the remote SMS instructions, whether to output the ringtone
Tamper alarm ring	When the detected someone tampering the equipment, plays the corresponding ringtone or alarm
Alarm duration	duration of alarm ring(not including tamper alarm)

Voice Intercom Configuration

Notice: You can access to webpage to change the ringtone: WEB → Maintenance → Update

File format: wav, single channel 8Khz sampling.

The file name, ring1: 1.wav(the ring2 replacement, file name: 2.wav)

➤ The broadcast terminal configuration notice

1) How to avoid an incoherency sound when the broadcast playing?

When interrupt to use as broadcast, the speak is loud, if 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 radio should be set as intercom mode, and activate the intercom mute, so as to ensure the broadcast quality.

Auto Headset	<input checked="" type="checkbox"/>	Ring From Headset	<input type="checkbox"/>
Enable Intercom	<input checked="" type="checkbox"/>	Enable Intercom Mute	<input checked="" type="checkbox"/>
Enable Intercom Tone	<input checked="" type="checkbox"/>	Enable Intercom Barge	<input checked="" type="checkbox"/>
P2P IP Prefix	<input type="text" value="."/>	DND Return Code	480(Temporarily Not Available)
Turn Off Power Light	<input checked="" type="checkbox"/>	Busy Return Code	486(Busy Here)
Emergency Call Number	<input type="text" value="110"/>	Reject Return Code	603(Decline)
Enable Password Dial	<input type="checkbox"/>	Active URI Limit IP	<input type="text"/>
Password Dial Prefix	<input type="text"/>	Push XML Server	<input type="text"/>
Password Length	<input type="text" value="0"/> (0~31)	Enable Call Waiting Tone	<input checked="" type="checkbox"/>
Enable Multi Line	<input checked="" type="checkbox"/>	IP Description	IP intercom
Enable Auto Answer	<input checked="" type="checkbox"/>	Auto Answer Timeout	0 second(s)
Enable Speed Dial Handdown	Enable	Status Led Reuse Mode	Disable
Dial Number Voice Play	Disable	Time of Dial Switch	16 (5-50)s
<input type="button" value="Apply"/>			

Voice Intercom Configuration

2) 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.722) of 4 KHz, is extended to broadband (G.722)7 KHz, when combined with the active speaker, the effect will be better.

Audio Settings	
First Codec	G.711A
Third Codec	G.729AB
Fifth Codec	None
Onhook Time	200 millisecond(s)
G.729AB Payload Length	20ms
G.722 Timestamps	160/20ms
Enable VAD	<input type="checkbox"/>
Second Codec	G.711U
Fourth Codec	G.722
Sixth Codec	None
Default Ring Type	Type 1
Tone Standard	China
G.723.1 Bit Rate	6.3kb/s
DTMF Payload Type	101 (96~127)

➤ The volume adjustment method

Method one: To adjust the volume of speaker and microphone by webpage.

Click “apply” to take effect (even in the call status), and it will save automatically.

AUDIO		FEATURE	DIAL PLAN	CONTACT	REMOTE CONTACT	WEB DIAL	MCAST																																								
<div> <div> > BASIC > NETWORK > VOIP > INTERCOM > SAFEGUARDING > FUNCTION KEY > MAINTENANCE > SECURITY > LOGOUT </div> <div> <h3>Audio Settings</h3> <table border="1"> <tbody> <tr> <td>First Codec</td> <td>G.711A</td> <td>Second Codec</td> <td>G.711U</td> </tr> <tr> <td>Third Codec</td> <td>G.729AB</td> <td>Fourth Codec</td> <td>G.722</td> </tr> <tr> <td>Fifth Codec</td> <td>None</td> <td>Sixth Codec</td> <td>None</td> </tr> <tr> <td>Onhook Time</td> <td>200 millisecond(s)</td> <td>Default Ring Type</td> <td>Type 1</td> </tr> <tr> <td>G.729AB Payload Length</td> <td>20ms</td> <td>Tone Standard</td> <td>China</td> </tr> <tr> <td>G.722 Timestamps</td> <td>160/20ms</td> <td>G.723.1 Bit Rate</td> <td>6.3kb/s</td> </tr> <tr> <td>Enable VAD</td> <td><input type="checkbox"/></td> <td>DTMF Payload Type</td> <td>101 (96~127)</td> </tr> </tbody> </table> </div> <div> <h3>Volume Settings</h3> <table border="1"> <tbody> <tr> <td>Handset/Handsfree Input Volume</td> <td>5 (1~9)</td> <td>Handset Output Volume</td> <td>5 (1~9)</td> </tr> <tr> <td>Handsfree Output Volume</td> <td>4 (1~9)</td> <td>Ring Volume</td> <td>5 (0~9)</td> </tr> </tbody> </table> </div> <div> <h3>Codec Gain Settings</h3> <table border="1"> <tbody> <tr> <td>Handsfree Hardware Mic Gain</td> <td>9 (1~11)</td> <td>Handsfree Hardware Speakerphone Gain</td> <td>5 (1~8)</td> </tr> </tbody> </table> </div> <div> <input type="button" value="Apply"/> </div> </div>								First Codec	G.711A	Second Codec	G.711U	Third Codec	G.729AB	Fourth Codec	G.722	Fifth Codec	None	Sixth Codec	None	Onhook Time	200 millisecond(s)	Default Ring Type	Type 1	G.729AB Payload Length	20ms	Tone Standard	China	G.722 Timestamps	160/20ms	G.723.1 Bit Rate	6.3kb/s	Enable VAD	<input type="checkbox"/>	DTMF Payload Type	101 (96~127)	Handset/Handsfree Input Volume	5 (1~9)	Handset Output Volume	5 (1~9)	Handsfree Output Volume	4 (1~9)	Ring Volume	5 (0~9)	Handsfree Hardware Mic Gain	9 (1~11)	Handsfree Hardware Speakerphone Gain	5 (1~8)
First Codec	G.711A	Second Codec	G.711U																																												
Third Codec	G.729AB	Fourth Codec	G.722																																												
Fifth Codec	None	Sixth Codec	None																																												
Onhook Time	200 millisecond(s)	Default Ring Type	Type 1																																												
G.729AB Payload Length	20ms	Tone Standard	China																																												
G.722 Timestamps	160/20ms	G.723.1 Bit Rate	6.3kb/s																																												
Enable VAD	<input type="checkbox"/>	DTMF Payload Type	101 (96~127)																																												
Handset/Handsfree Input Volume	5 (1~9)	Handset Output Volume	5 (1~9)																																												
Handsfree Output Volume	4 (1~9)	Ring Volume	5 (0~9)																																												
Handsfree Hardware Mic Gain	9 (1~11)	Handsfree Hardware Speakerphone Gain	5 (1~8)																																												

Method two: to adjust the volume by the remote command

Remote adjustment by active URL commands to complete the speaker and microphone gain.

Voice Intercom Configuration

➤ The speed Dial key configuration method

AUDIO	FEATURE	DIAL PLAN
Turn Off Power Light	<input checked="" type="checkbox"/>	
Emergency Call Number	110	
Enable Password Dial	<input type="checkbox"/>	
Password Dial Prefix		
Password Length	0 (0~31)	
Enable Multi Line	<input checked="" type="checkbox"/>	
Enable Auto Answer	<input checked="" type="checkbox"/>	
Enable Speed Dial Handdown	Enable	
Dial Number Voice Play	Disable	

Enable the <Speed Dial Hand down> and set DSS key as speed dial, whether allow DSS key to hang up the call (SIP call or P2P call)

➤ The incoming call settings

By default, all calls are automatically answered, including SIP or P2P.

AUDIO	FEATURE	DIAL PLAN	CONTACT	REMOTE CONTACT	WEB DIAL	MCAST
Turn Off Power Light	<input checked="" type="checkbox"/>		Busy Return Code	486(Busy Here)		
Emergency Call Number	110		Reject Return Code	603(Dedline)		
Enable Password Dial	<input type="checkbox"/>		Active URI Limit IP			
Password Dial Prefix			Push XML Server			
Password Length	0 (0~31)		Enable Call Waiting Tone	<input checked="" type="checkbox"/>		
Enable Multi Line	<input checked="" type="checkbox"/>		IP Description	IP intercom		
Enable Auto Answer	<input checked="" type="checkbox"/>		Auto Answer Timeout	0 second(s)		
Enable Speed Dial Handdown	Enable		Status Led Reuse Mode	Disable		
Dial Number Voice Play	Disable		Time of Dial Switch	16 (5-50)s		

The definitions of the red box part are effective for all incoming calls. When disable the <Enable Auto Answer> function, SIP call or P2P calls will be ringing tone hint.

Voice Intercom Configuration

1) How to set SIP account incoming call

The incoming call will be automatically answered after a period of time, you only need to set <auto answer enable> and fill in the needed answer period of time, (If set to 0, the call automatically answer). Click< apply>.

SIP Line SIP 1

Basic Settings >>

Codecs Settings >>

Advanced SIP Settings >>

Always Forward	<input type="checkbox"/>	Enable Hotline	<input type="checkbox"/>
Always Fwd Number	<input type="text"/>	Hotline Number	<input type="text"/>
Busy Forward	<input type="checkbox"/>	Warm Line Wait Time	<input type="text"/> (0~9)second(s)
Busy Fwd Number	<input type="text"/>	Keep Alive Type	SIP Option
No Answer Forward	<input type="checkbox"/>	Keep Alive Interval	<input type="text"/> second(s)
NoAnswer Fwd Number	<input type="text"/>	BLF Server	<input type="text"/>
No Ans. Fwd Wait Time	<input type="text"/> (0~120)second(s)	Transfer Timeout	<input type="text"/> second(s)
SIP Encryption	<input type="checkbox"/>	Enable Auto Answer	<input checked="" type="checkbox"/>
SIP Encryption Key	<input type="text"/>	Auto Answer Timeout	<input type="text"/> second(s)
RTP Encryption	<input type="checkbox"/>	Enable Session Timer	<input type="checkbox"/>
RTP Encryption Key	<input type="text"/>	Session Timeout	<input type="text"/> second(s)
		Session Refresher	UAS

2) How to set the P2P(IP to IP) incoming call

When incoming call need to be auto answered after a period of time, enable the <auto answer enable> and fill in the needed auto answer time, (If set to 0, the call will answer automatically). Click< apply>.

AUDIO **FEATURE** **DIAL PLAN** **CONTACT** **REMOTE CONTACT** **WEB DIAL** **MCAST**

Turn Off Power Light	<input checked="" type="checkbox"/>	Busy Return Code	486(Busy Here)
Emergency Call Number	<input type="text"/>	Reject Return Code	603(Decline)
Enable Password Dial	<input type="checkbox"/>	Active URI Limit IP	<input type="text"/>
Password Dial Prefix	<input type="text"/>	Push XML Server	<input type="text"/>
Password Length	<input type="text"/> (0~31)	Enable Call Waiting Tone	<input checked="" type="checkbox"/>
Enable Multi Line	<input checked="" type="checkbox"/>	IP Description	IP intercom
Enable Auto Answer	<input checked="" type="checkbox"/>	Auto Answer Timeout	<input type="text"/> second(s)
Enable Speed Dial Handdown	Enable	Status Led Reuse Mode	Disable
Dial Number Voice Play	Disable	Time of Dial Switch	<input type="text"/> (5-50)s

Apply

Intercom Configuration

➤ The other function settings

	AUDIO	FEATURE	DIAL PLAN	CONTACT	REMOTE CONTACT	WEB DIAL	MCAST
> BASIC > NETWORK > VOIP > INTERCOM > SAFEGUARDING > FUNCTION KEY	Enable Intercom	<input checked="" type="checkbox"/>		Enable Intercom Mute	<input type="checkbox"/>		
	Enable Intercom Tone	<input checked="" type="checkbox"/>		Enable Intercom Barge	<input checked="" type="checkbox"/>		
	P2P IP Prefix	<input type="text"/>		DND Return Code	<input type="text" value="480(Temporarily Not Available)"/>		
	Turn Off Power Light	<input checked="" type="checkbox"/>		Busy Return Code	<input type="text" value="486(Busy Here)"/>		
	Emergency Call Number	<input type="text" value="110"/>		Reject Return Code	<input type="text" value="603(Decline)"/>		
	Enable Password Dial	<input type="checkbox"/>		Active URI Limit IP	<input type="text"/>		
	Password Dial Prefix	<input type="text"/>		Push XML Server	<input type="text"/>		
	Password Length	<input type="text" value="0"/> (0~31)		Enable Call Waiting Tone	<input checked="" type="checkbox"/>		
	Enable Multi Line	<input checked="" type="checkbox"/>		IP Description	<input type="text" value="IP intercom"/>		
	Enable Auto Answer	<input checked="" type="checkbox"/>		Auto Answer Timeout	<input type="text" value="0"/> second(s)		
	Enable Speed Dial Handdown	<input type="text" value="Enable"/>		Status Led Reuse Mode	<input type="text" value="Disable"/>		
	Dial Number Voice Play	<input type="text" value="Disable"/>		Time of Dial Switch	<input type="text" value="16"/> (5-50)s		

Apply

1) Status Led reuse mode

Enable this function, the registered status indicator will reuse the call instructions function, which means the LED will flash in the call state.

2) Dialing tone prompt

Enable this function; it will have corresponding key tone of voice when operating the digital keyboard

3) Call switching time

This function is used to define the time interval when use speed dial key making call, and call switching from number 1 to number 2.