



# Intercom Quick Installation Guide



# Introduction

## Package Contents



i12 Intercom



Connector



Installation diagram



Quick Installation Guide



Installation Size Map



Screw and tool

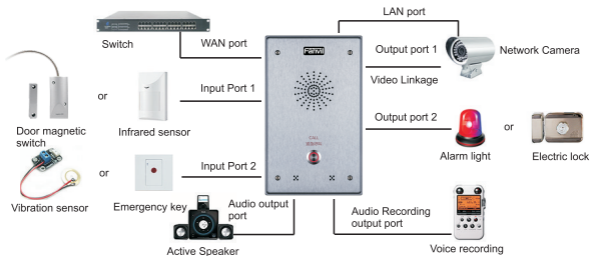


CD



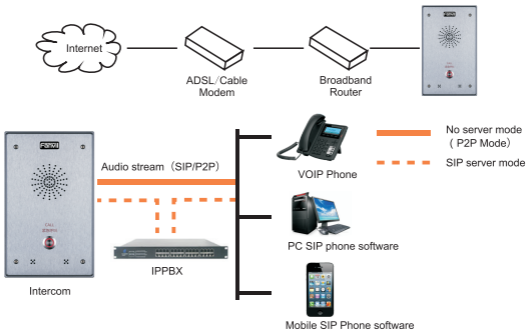
# Voice Intercom Configuration

## 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.



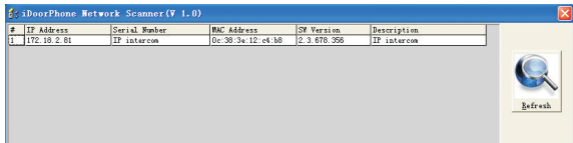
Users can call the same group of people through the VOIP phone, PC or mobile phone SIP phone software, and realizes remote control to the device(such as a door lock, Alarm lamp etc.)

# Voice Intercom Configuration

## Step Two: Get the device IP Address:

Methods 1:

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:

# Voice Intercom Configuration

## Step Four: Modify the device description

The screenshot shows the 'FEATURE' tab of the configuration interface. The left sidebar has 'INTERCOM' selected. The main content area is divided into two columns of settings. The 'IP Description' field in the right column is highlighted with a red box.

AUDIO	FEATURE	DIAL PLAN	CONTACT	REMOTE CONTACT	WEB DIAL	MCAST
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"/>		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(Dedline)"/>		
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="10"/> [5-30]s		

Apply

## Step Five: Add SIP account

The screenshot shows the 'SIP' tab of the configuration interface. The left sidebar has 'VOIP' selected. The 'SIP Line' dropdown is set to 'SIP 1'. The 'Basic Settings >>' section is expanded, showing various configuration fields.

SIP	IAX2	STUN	DIAL PEER
SIP Line			
SIP Line	<input type="text" value="SIP 1"/>		
Basic Settings >>			
Status	<span>Registered</span>	Domain Realm	<input type="text" value="test.com"/>
Server Address	<input type="text" value="172.18.1.212"/>	Proxy Server Address	<input type="text"/>
Server Port	<input type="text" value="5060"/>	Proxy Server Port	<input type="text"/>
Authentication User	<input type="text" value="601"/>	Proxy User	<input type="text"/>
Authentication Password	<input type="text" value="*****"/>	Proxy Password	<input type="text"/>
SIP User	<input type="text" value="601"/>	Backup Proxy Server Address	<input type="text"/>
Display Name	<input type="text" value="601"/>	Backup Proxy Server Port	<input type="text" value="5060"/>
Enable Registration	<input checked="" type="checkbox"/>	Server Name	<input type="text"/>
Codex Settings >>			
Advanced SIP Settings >>			

Apply

# Voice Intercom Configuration

## Step Six: DSS key Configuration method

The screenshot shows a configuration page with a red sidebar on the left containing menu items: BASIC, NETWORK, VOIP, INTERCOM, SAFEGUARDING, FUNCTION KEY (highlighted), MAINTENANCE, and SECURITY. The main content area is titled 'Screen Configuration' and includes a 'Contrast' field set to 5 (range 1-9) and an 'Enable Backlight' checkbox which is checked. Below this is the 'Function Key Settings' section, which contains a table with columns: Key, Type, Number 1, Number 2, Line, and Subtype. The table lists four DSS keys with their respective configurations. An 'Apply' button is located below the table.

Key	Type	Number 1	Number 2	Line	Subtype
DSS Key 1	Hot Key	103	172.18.2.13	SIP1	Speed Dial
DSS Key 2	Key Event			SIP1	None
DSS Key 3	Line			SIP1	Speed Dial
DSS Key 4	Line			SIP2	Speed Dial

Intercom software can support up to four DSS key functions

1)The Subtype configuration of Hot key

DSS key type	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

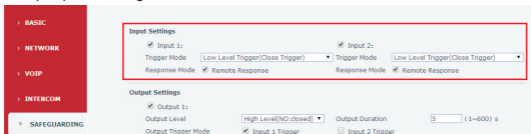
This screenshot shows the 'Function Key Settings' table from the previous image. The 'Subtype' dropdown for 'DSS Key 2' is open, showing a list of options: OK, None, Dial, Release, OK, and Handfree. The 'Apply' button is visible at the bottom of the table.

Key	Type	Number 1	Number 2	Line	Subtype
DSS Key 1	Hot Key	103	172.18.2.13	SIP1	Speed Dial
DSS Key 2	Key Event			SIP1	OK
DSS Key 3	Line			SIP1	None
DSS Key 4	Line			SIP2	Dial
					Release
					OK
					Handfree

# Voice Intercom Configuration

DSS key type	Subtype	Usage
Key Event	None	No Answer
	Dial	Dial function
	Release	End calls
	OK	Identify key
	Handfree	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



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 (selected), FUNCTION KEY, and MAINTENANCE. The main content area is divided into two sections for 'Output 1' and 'Output 2'. Each section has a 'High Level(NO:closed)' dropdown menu, an 'Output Duration' input field (set to 5), and an 'Output Last' dropdown menu (set to 'By Duration'). Under 'Output 1', the 'Output Level' is 'High Level(NO:closed)', 'Output Trigger Mode' is 'High Level(NO:closed)', and several triggers are checked: 'Input 1 Trigger', 'Remote DTMF Trigger', 'Remote SMS Trigger', and 'Emergency Key Trigger'. The 'Input 2 Trigger' is unchecked. The 'Alert' field is set to 'OUT1\_SOS' and the 'Talking' dropdown is set to 'Talking'. The 'Output 2' section has similar settings, with 'Input 2 Trigger' checked and 'Alert' set to 'OUT2\_SOS'.

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		
	Input port2 trigger		
	Remote DTMF trigger	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		
	Call state trigger		
	Emergency key trigger		

# 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)

Web Update

Select File:  Browse ("z","txt","xml","au","vcl","csv","wav" Update)

TFTP/FTP Update

Server Address

User

Password

File Name

Type Application Update

Protocol FTP

Apply

Update Logo File

Select File:  Browse Update

Delete Logo File

Select File:  Delete

Logo File

>The broadcast terminal configuration notice

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

When the terminal 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.

AUDIO FEATURE DIAL PLAN CONTACT REMOTE CONTACT WEB DIAL MCAST

Auto Handdown Time 3 second(s) Enable Call Completion

Enable Auto Redial

Auto Redial Interval 10 (1-180)second(x) Enable Silent Mode

Auto Redial Times 10 (1-100) Hide DTNF Disabled

Auto Headset  Ring From Headset

Enable Intercom  Enable Intercom Mute

Enable Intercom Tone  Enable Intercom Barge

P2P IP Prefix  DND Return Code 480(Temporarily Not Available)

Turn Off Power Light  Busy Return Code 486(Busy Here)

Emergency Call Number 110 Reject Return Code 603(Decline)

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

## >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
<b>Volume Settings</b>						
Handset/Handsfree Input Volume	5 (1~9)	Handset Output Volume	5 (1~9)			
Handsfree Output Volume	4 (1~9)	Ring Volume	5 (0~9)			
<b>Codec Gain Settings</b>						
Handsfree Hardware Mic Gain	0 (1~11)	Handsfree Hardware Speakerphone Gain	4 (1~8)			
<input type="button" value="Apply"/>						

MIC gain of software (points to Handsfree Output Volume)

Speaker gain of software (points to Handsfree Output Volume)

MIC hardware gain (points to Handsfree Hardware Mic Gain)

Speaker hardware gain (points to Handsfree Hardware Speakerphone Gain)

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

The screenshot shows the 'AUDIO' configuration tab. On the left is a navigation menu with 'INTERCOM' selected. The main area contains several settings:

Setting	Value
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.

The screenshot shows the 'FEATURE' configuration tab. The 'Enable Auto Answer' setting is highlighted with a red box. Other settings include:

Setting	Value
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
Auto Answer Timeout	0 second(s)
Status Led Reuse Mode	Disable
Time of Dial Switch	16 (5-30)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	IAX2	STUN	DIAL PEER
<b>Advanced SIP Settings &gt;&gt;</b>				
BASIC	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	0 (0-9)second(s)
	Busy Fwd Number	<input type="text"/>	Keep Alive Type	SIP Option
	No Answer Forward	<input type="checkbox"/>	Keep Alive Interval	60 second(s)
	NoAnswer Fwd Number	<input type="text"/>	BLF Server	<input type="text"/>
	No Ans. Fwd Wait Time	60 (0-120)second(s)	Transfer Timeout	0 second(s)
	SIP Encryption	<input type="checkbox"/>	Enable Auto Answer	<input checked="" type="checkbox"/>
	SIP Encryption Key	<input type="text"/>	Auto Answer Timeout	50 second(s)
	RTP Encryption	<input type="checkbox"/>	Enable Session Timer	<input type="checkbox"/>
RTP Encryption Key	<input type="text"/>	Session Timeout	0 second(s)	
Session Refresher			UAS	
Subscribe For MWI	<input type="checkbox"/>	Conference Type	Local	
MWI Number	<input type="text"/>	Conference Number	<input type="text"/>	
Subscribe Period	3600 second(s)	Registration Expires	60 second(s)	
Enable Service Code	<input type="checkbox"/>			
DND On Code	<input type="text"/>	DND Off Code	<input type="text"/>	

## 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	HCAST	
BASIC	Auto Headset	<input checked="" type="checkbox"/>	Ring From Headset	<input type="checkbox"/>	Enable Intercom Mute	<input checked="" type="checkbox"/>	Enable Intercom Barge	<input checked="" type="checkbox"/>
	Enable Intercom	<input checked="" type="checkbox"/>	DND Return Code	480(Temporarily Not Available)	Busy Return Code	485(Busy Here)	Reject Return Code	603(Decline)
	Enable Intercom Tone	<input checked="" type="checkbox"/>	Turn Off Power Light	<input checked="" type="checkbox"/>	Active URI Limit IP	<input type="text"/>	Push XML Server	<input type="text"/>
	P2P IP Prefix	<input type="text"/>	Emergency Call Number	110	Enable Call Waiting Tone	<input checked="" type="checkbox"/>	IP Description	IP Intercom
	Password Dial Prefix	<input type="text"/>	Enable Password Dial	<input type="checkbox"/>	Auto Answer	<input checked="" type="checkbox"/>	Auto Answer Timeout	0 second(s)
	Password Length	0 (0-31)	Enable Multi Line	<input checked="" type="checkbox"/>	Enable Speed Dial	Enable	Status Led Reuse Mode	Disable
	Enable Auto Answer	<input checked="" type="checkbox"/>	Dial Number Voice Play	Disable	Time of Dial Switch	16 (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	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"/>		
	PDP IP Prefix	<input type="text"/>		DND Return Code	480(Temporarily Not Available)		
	Turn Off Power Light	<input checked="" type="checkbox"/>		Busy Return Code	485(Busy Here)		
	Emergency Call Number	110		Reject Return Code	603(Dedine)		
	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	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 (15-50)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.

**Fanvil**