



Intercom Quick Installation Guide



Introduction

Package Contents



i12 Intercom



Connector



Installation diagram



Quick Installation Guide



Installation Size Map



Screw and tool

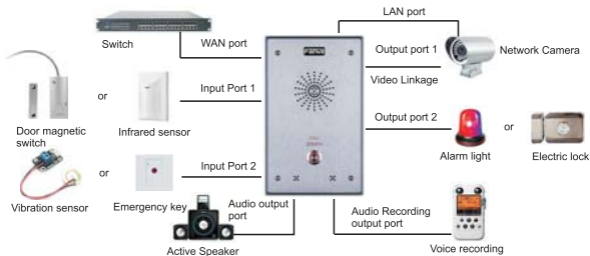


CD



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.



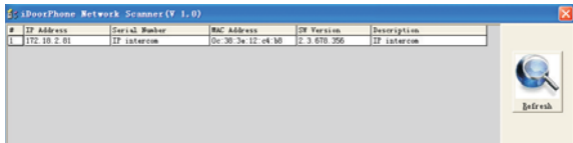
Users can call the same group of people through the VOIP phone, PC or mobile phone SIP phone software, and realizes remote control to the device (such as a door lock, Alarm lamp etc.)

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: English

Logon

Voice Intercom Configuration

Step Four: Modify the device description

The screenshot shows the 'INTERCOM' configuration page. The left sidebar has 'INTERCOM' selected. The main area contains the following settings:

| FEATURE | VALUE |
|----------------------------|-------------------------------------|
| Auto Headset | <input checked="" type="checkbox"/> |
| Enable Intercom | <input checked="" type="checkbox"/> |
| Enable Intercom Tone | <input checked="" type="checkbox"/> |
| FQP IP Profile | |
| Turn Off Power Light | <input checked="" type="checkbox"/> |
| Emergency Call Number | (11) |
| Enable Password Dial | <input type="checkbox"/> |
| Password Dial Prefix | |
| Password Length | 0-31 (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 |
| Ring from Headset | <input type="checkbox"/> |
| Enable Intercom Mute | <input checked="" type="checkbox"/> |
| Enable Intercom Barge | <input checked="" type="checkbox"/> |
| DND Return Code | 480(Temporarily Not Available) |
| Busy Return Code | 480(Busy Here) |
| Reject Return Code | 403(Decline) |
| Active IMS List IP | |
| Push IMS Server | |
| Enable Call Waiting Tone | <input checked="" type="checkbox"/> |
| Description | IP Intercom |
| Auto Answer Timeout | 0 (seconds) |
| Status Led Reuse Mode | Disable |
| Time of Dial Switch | 25 (5 Side) |

An 'Apply' button is located at the bottom right of the configuration area.

Step Five: Add SIP account

The screenshot shows the 'SIP' configuration page. The left sidebar has 'SIP' selected. The main area shows the configuration for 'SIP Line: SIP_1'.

SIP Line: SIP_1

Basic Settings >>>

| | | | |
|-------------------------|-------------------------------------|-----------------------------|----------|
| Status | Registered | Domain Name | test.com |
| Server Address | 172.18.1.212 | Proxy Server Address | |
| Server Port | 5060 | Proxy Server Port | |
| Authentication User | 001 | Proxy User | |
| Authentication Password | ***** | Proxy Password | |
| SIP User | 001 | Backup Proxy Server Address | |
| Display Name | 001 | Backup Proxy Server Port | 5060 |
| Enable Registration | <input checked="" type="checkbox"/> | Server Name | |

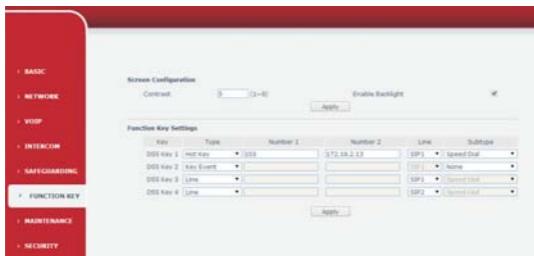
Codecs Settings >>>

Advanced SIP Settings >>>

An 'Apply' button is located at the bottom right of the configuration area.

Voice Intercom Configuration

Step Six: DSS key Configuration method



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 <small>Enable Speed Dial Handdown</small> and <small>Trans. W</small> 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



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) |

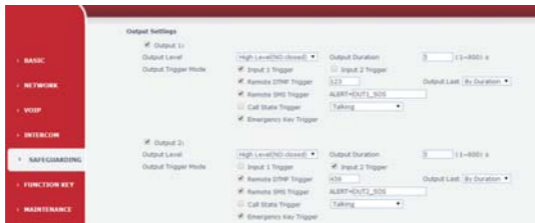
> The two short circuits input configuration method
 WEB→Safeguarding, As shown in the figure below
 The input port Settings



| 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



| 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 | | 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

The screenshot shows the 'Tamper Alarm Settings' interface. It includes a checked checkbox for 'Tamper Alarm', an 'Alarm command' field with the value 'Tamper_Alarm', a 'Reset command' field with the value 'Tamper_Reset', and a 'Reset' button.

| 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

The screenshot shows the 'Server & Trigger Ring Type Settings' interface. It includes a 'Server Address' field with '0.0.0.0', 'Input 1 Trigger Ring' set to 'User 1', 'Remote DTMF Trigger Ring' set to 'Enable', 'Input 2 Trigger Ring' set to 'User 3', 'Remote SMS Trigger Ring' set to 'User 2', 'Tamper Alarm Ring' set to 'User 3', and 'Alarm Ring Duration' set to '5' seconds. An 'Apply' button is at the bottom.

| 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.

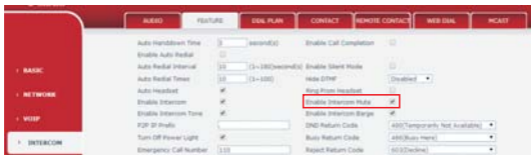
The file name, ring1: 1.wav(the ring2 replacement, file name: 2.wav)



>The broadcast terminal configuration notice

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

When 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.



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.



>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.



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 | MCADT |
|-------|----------------------------|-------------------------------------|--------------------------|-------------------------------------|----------|-------|
| | Turn Off Power Light | <input checked="" type="checkbox"/> | Busy Return Code | 480(Busy Here) | | |
| | Emergency Call Number | 110 | Reject Return Code | 603(Decline) | | |
| | Enable Password Dial | <input type="checkbox"/> | Active URI Line ID | | | |
| | Password Dial Prefix | | Push IM, Server | | | |
| | Password Length | 0 (0~31) | Enable Call Waiting Tone | <input checked="" type="checkbox"/> | | |
| | Enable Multi Line | <input checked="" type="checkbox"/> | IP Description | Web(12.001) | | |
| | Enable Auto Answer | <input checked="" type="checkbox"/> | Auto Answer Timeout | 0 second(s) | | |
| | Enable Speed Dial Handdown | Enable | Status Led Route Mode | Disable | | |
| | Dial Number Voice Play | Disable | Time of Dial Switch | 0s (0~300s) | | |

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>.

The screenshot shows the 'Advanced SIP Settings >>' configuration page. The left sidebar has a red background with a white menu containing: BASIC, NETWORK, VOIP (highlighted), INTERCOM, SAFEGUARDING, FUNCTION KEY, MAINTENANCE, SECURITY, and LOGOUT. The main content area has a red header with tabs: SIP, FAX2, STUN, and DIAL PEER. The 'SIP' tab is active. The settings are organized into two columns. The right column contains the following settings:

| | |
|----------------------|-------------------------------------|
| Enable Hotline | <input type="checkbox"/> |
| Hotline Number | <input type="text"/> |
| Warm Line Wait Time | 0 [0-1000000] |
| Keep Alive Type | SIP Option |
| Keep Alive Interval | 30 [0-1000] |
| SIP Server | <input type="text"/> |
| Transfer Timeout | 0 [0-1000] |
| Enable Auto Answer | <input checked="" type="checkbox"/> |
| Auto Answer Timeout | 30 [0-1000] |
| Enable Session Timer | <input type="checkbox"/> |
| Session Timeout | 0 [0-1000] |
| Session Refresh | UAS |
| Conference Type | Local |
| Conference Number | <input type="text"/> |
| Registration Expires | 30 [0-1000] |

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>.

The screenshot shows the 'INTERCOM' configuration page. The left sidebar is the same as in the first screenshot, with 'INTERCOM' highlighted. The main content area has a red header with tabs: HERO, FEATURE, DIAL PLAN, CONTACT, REMOVE CONTACT, WEB DIAL, and HCAST. The 'FEATURE' tab is active. The settings are organized into two columns. The right column contains the following settings:

| | |
|----------------------------|-------------------------------------|
| Ring From Headset | <input type="checkbox"/> |
| Enable Intercom Mute | <input checked="" type="checkbox"/> |
| Enable Intercom Barge | <input checked="" type="checkbox"/> |
| DND Return Code | 400(Temporarily Not Available) |
| Busy Return Code | 400(Busy Here) |
| Reject Return Code | 403(Decline) |
| DND Limit SP | <input type="text"/> |
| Push HMI Server | <input type="text"/> |
| Enable Call Waiting Time | <input checked="" type="checkbox"/> |
| SP Description | SP Intercom |
| Enable Auto Answer | <input checked="" type="checkbox"/> |
| Auto Answer Timeout | 0 [0-1000] |
| Enable Speed Dial Handdown | Enable |
| Status Led Reuse Mode | Disable |
| Dial Number Voice Play | Disable |
| Time of Dial Switch | 10 [0-300] |

At the bottom of the page, there is an 'Apply' button.

Intercom Configuration

>The other function settings

The screenshot shows a web interface for configuring intercom settings. The interface has a red sidebar on the left with navigation options: BASIC, NETWORK, VOIP, INTERCOM (selected), SAFEGUARDING, and FUNCTION KEY. The main content area is divided into several tabs: RING, FEATURE, DIAL PLAN, CONTACT, REMOTE CONTACT, WEB DIAL, and NCALL. The INTERCOM tab is active, displaying a list of settings. Two settings are highlighted with red boxes: 'Dual Number Voice Play' and 'Status Led Reuse Mode', both set to 'Enable'. Other visible settings include 'Auto Headset', 'Enable Intercom', 'Emergency Call Number', 'Password Dial Prefix', 'Enable Multi-Line', 'Enable Auto Answer', 'Ring From Headset', 'Enable Intercom Mute', 'DND Return Code', 'Busy Return Code', 'Reject Return Code', 'Active SNG Limit SP', 'Push IRL Server', 'Enable Call Waiting Tone', 'SP Description', 'Auto Answer Timeout', and 'Dial Number Voice Play'.

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.

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