

INT-IP-2GANG

SIP Getting Started Guide

Introduction

The INT-IP-2GANG is a PoE powered (802.3af) IP Intercom Call Station that can be configured as a SIP extension for use with a SIP based phone system. The unit fits in a standard 2-gang electrical box with plaster ring that has a minimum depth of 2" and is suitable for installation indoors or outdoors in damp locations. It includes:

- Integrated Microphone - water resistant
- Integrated Speaker – waterproof cone
- Security screens protect Microphone and Speaker
- Integrated vandal resistant Call button with LED
- Integrated Relay for door/gate activation
- PoE (802.3af)
- Includes four stainless steel security screws (T10)
- User configurable Full or Half duplex operation
- 12 Gauge Stainless Steel Call Station panel

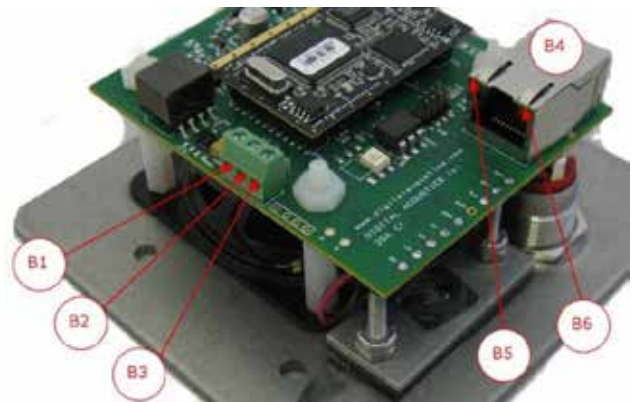


The INT-IP-2GANG is configured using the Digital Acoustics **eSIP Configuration** utility. This [Getting Started Guide](#) covers basic hardware installation and software configuration of the INT-IP-2GANG.

Installation

The INT-IP-2Gang can be flush mounted into a standard 2-gang electrical box or surface mounted in an optional surface mount box. For outdoor installations, mount under an eave or use a hooded surface mount box. Please note, the INT-IP-2Gang can be configured before being installed in the 2-gang box.

- Connect RJ-45 (B4) to a PoE (802.3af) network switch or injector. The INT-IP-2GANG will power up within 5 seconds and request 7.0 watts from the Power Source Equipment
- If the relay will be used to activate an electronic door strike, connect the COM (B2) and either the NO (B1) or NC (B3) to the door strike
- The green **Link** LED (B5) will turn solid. The yellow **Ready** LED (B6) and the Call button LED will flash 4x per second
- Record the **ID#** from the label on back of the INT-IP-2GANG for use during software configuration.
- Optionally, label the panel with one of the extra **ID#** labels
- Install the Call Station in the 2-gang electrical box with the call button in the upper left-hand corner so the RJ-45 connector faces down
- Fasten using the four security screws (T10)
- Use a high-quality silicone sealant between the top and sides of the unit and the surface it is mounted to
- The Call Station LED will continue to flash until configuration is complete
- Please refer to the **INT-IP-2GANG_ReferenceManual** available at <https://www.digitalacoustics.com/support/ip7-hardware-talkmaster-software-manuals/> for additional information on installation

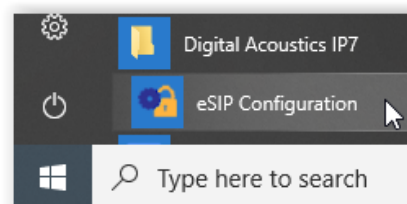


SIP Configuration

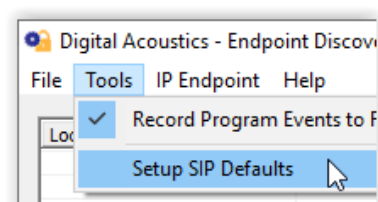
If the **eSIP Configuration** utility has already been installed and the defaults have been setup, skip to the **Select the General Settings tab** step.

Download and install the **eSIP Configuration** utility from <https://www.digitalacoustics.com/support/software-firmware-downloads/>

- After the configuration utility has been installed, open it from the Windows® **Start** menu by selecting **Digital Acoustics IP7 → eSIP Configuration**



- Click menu option **Tools → Setup SIP Defaults** to setup the default SIP based Phone system



- In the **Primary Registrar** (also know as the **Domain**), enter the IP Address or DNS and the port of the SIP based phone system
- The **Primary Server** (also known as the **Proxy Address**) is typical the same as the **Primary Registrar**. If this is different than the **Primary Registrar**, then the **Registrar is Proxy** box must be checked
- The **Secondary Registrar** and **Secondary Server** are only specified if a backup VOIP based phone system is available in case the Primary fails
- The **SIP Local Port** and **RTP Port** specify the ports used by the IP7 when communicating with the VOIP based Phone system
- Press the **OK** button to save the defaults

Select the **General Settings** tab

- Press the **Find Devices** button to discover any new IP7 devices that are connected to the network. The IP7 devices will be listed on the left hand side of the screen
- Click on an IP7 Endpoint to be configured (match the **Device ID** column to the Intercom's previously recorded ID#)

Location Name	Device ID	Address
	04635d	192.168.5.179

- On the right hand side of the screen, select **SIP Standalone Device** from the dropdown
- Enter a descriptive name into the **Location Name** field (this name is only used by this utility. The description from the phone system will be used to identify this endpoint during a call)
- Check the DHCP option to automatically assign the IP Address information or uncheck it to enter static IP address information

Select the **SIP Settings** tab

- Press the **Apply Defaults** button on the bottom right to copy in the default SIP information that was previously setup
- Enter the **SIP Extension**, **SIP Password** (also known as the **Secret**) that have been setup on the SIP based phone system for this IP7 device
- In the **SIP Authentication** (also known as the **SIP Authorization Name**), enter the User that has been associated with this extension in the SIP based phone system. If no User has been setup, use the same value as the SIP Extension
- Check the **Full Duplex** option to enable full duplex operation or leave it blank for of half duplex
 - In a half-duplex phone call, one party has their microphone active while the other party has their speaker active. By default, when a call is initiated, the INT-IP-2Gang starts with the speaker active and the phone operator starts their microphone active. The phone operator can toggle the modes by pressing 0 on the phone's keypad. In half duplex mode, the INT-IP-2Gang volume will always be louder, and the microphone gain will always be higher
 - In full duplex mode, both parties have their speaker and microphone active at the same time. In full duplex mode, the INT-IP-2Gang volume will always be softer, and the microphone gain will always be lower. This is due to the algorithms used by the IP7 for echo cancellation
- Check the **Outdial Extension** and enter the number to be automatically dialed when the integrated Call button is pressed
- If the integrated relay will be used to open a door during a call, specify a 1 – 4 digit code in the **Door Open Code** to be entered on the phone's keypad. If the specified code has multiple digits, the phone operator will need to enter a # after the code (using this example, phone operator would need to enter the digits 65# during the call to activate the relay). If only a single digit is specified, the phone operator only needs to enter that digit.

The screenshot shows the 'SIP Settings' tab with the following fields and options:

- ☐ Peer-to-Peer
- Primary Registrar: asterisk.digitalacoustics.com, Port: 5060
- Primary Server: asterisk.digitalacoustics.com, Port: 5060
- Secondary Registrar: (empty), Port: 0
- Secondary Server: (empty), Port: 0
- SIP Local Port: 5060, RTP Port: 21000
- SIP Extension: 5802
- SIP Password: (masked with asterisks)
- SIP Authentication: 5802
- ☐ Full Duplex, ☐ Default to Listen, ☐ Registrar is Proxy
- ☐ SIP / TCP, ☐ Always Connected
- ☒ Outdial Extension: 5600
- Door Open Code: 65, Then press #: (empty)
- Call Timeout: 5, No Call End @ Station (dropdown)
- Apply Defaults button

Select the **Advanced Settings** tab

- Click the **Set** button next to **Audio Profile** and select the **INT-IP-2Gang** audio profile if using full duplex. Otherwise select the **Half Duplex** audio profile
- Click the **Save Settings** button. The endpoint will reboot with the changes and register with the SIP Server within 30 seconds. The red Call button LED on the panel will blink rapidly for 30 seconds and then turn solid
- There are 7 volume steps that will vary based on whether full or half duplex has been configured. Volume can be adjusted when a SIP call is in progress by pressing:
 - #00 – Set volume down one level
 - #01 – Set volume level to 1
 - #02 – Set volume level to 2
 - #03 – Set volume level to 3
 - #04 – Set volume level to 4
 - #05 – Set volume level to 5
 - #06 – Set volume level to 6
 - #07 – Set volume level to 7
 - #08 – Set volume up one level
- For Full Duplex, the volume level should be set to 4 or 5 for best results

The screenshot shows the 'Advanced Settings' tab with the following sections and options:

- Device Options:**
 - ☐ Call Button, ☐ Handset, ☐ +Speaker, ☐ ST/ADA Mode
- Relay Activation:**
 - Factory Defaults (dropdown)
 - Door Relay Activate: 5 Sec
 - ☐ Mic, ☐ Speaker
- Microphone Options:**
 - ☐ Disable AGC, Mic Level (0-63): 0
 - Microphone Input: Microphone (dropdown)
 - Default Mic Type: uLaw (dropdown)
 - Audio Profile: Half Duplex
 - Set button