

Introduction

The IP7-FX is a PoE powered (802.3af) IP Audio Module that can be configured as a SIP extension for use with a SIP based phone system. The module can connect to an analog audio device such as a call station or speaker and provides the following features

- Ethernet 10/100 Network Connector
- Port 2 10/100 available to connect another device to the network
- Eight-watt amplifier
- Microphone or Line In
- Speaker or Line Out
- Two input signals (Call and Sensor)
- Normally Open or Normally Closed Dry Contact Relay (250 VAC / 30VDC @ 60W / 1500 VAC Isolation)
- PoE (802.3af), 12VDC or 24VDC
- User configurable Full or Half duplex operation
- Standard 35MM DIN Rail Mount or optional Surface Mount Plate (included)



The IP7-FX can be configured using the **eSIP Configuration** utility or the **TalkMaster Administrator** console. This Getting Started Guide covers basic hardware installation and software configuration of the IP7-FX.

Replacing an IP7-ST/STx, SS8/SE8, FD or SS20

If the IP7-FX is being used to replace an existing IP7-ST/STX, IP7-SS8/SE8, IP7-FD or an IP7-SS20, please note the following:

- The J3-7 (Case) connector on older models was an extra ground. If replacing an older model with an IP7-FX, any wire connected to J3-7 should be moved to J3-6.
- If replacing an IP7-ST/STx, make sure the internal **ST-MIC** switch (located on the DIN clip side) is set in the same position. The IP7-FX does not support the Aiphone IE/ IF series call station. An IP7-ST/STx must be used
- If the older IP7's J2-6 (SPKR 8Ω+) and J2-7 (SPKR 8Ω-) is connected to an amplifier or amplified speaker, use the IP7-FX's J2-3 (Line Out) and J2-4 (GND) instead
- If replacing an IP7-SE8 / IP7-MSR-BRD with an IP7-FX, the IP7-MSR-BRD can be eliminated by changing the IP7-FX's ST-MIC internal switch to the ST position
- Other than the exceptions noted above, the J1, J2, J3, and network can be moved over to the IP7-FX

New Installations

The IP7-FX can be mounted indoors or outdoors in a weather tight enclosure.

 If PoE (Power over Ethernet) is not available, connect the IP7-FX's Ethernet port to a network switch using a standard RJ-45 cable. Connect the IP7-FX to an external 12VDC (1.2 Amps) or 24VDC (2.5 Amps) power supply using either J1-1 (+) and J1-2 (-) terminals or a 2.5mm barrel connector (center tip positive)



- If PoE is available, connect the IP7-FX's Ethernet port to a PoE (802.3af) network switch or injector using a standard RJ-45 network cable. It will request 15.4 watts from the power source equipment.
- The Link LED will turn solid. The Ready LED will begin flashing 4x per second
- The IP7-FX can use either a mic level or line level microphone. Connect a mic level microphone to the J2-1 (Mic+) and J2-2 (Mic-) terminals. Optionally, connect a line level microphone using J3-4 (Line In), J3-6 (Ground) and J3-7 (Mic Power 12V). Use 20-22 AWG twisted, shielded wire. The Microphone wire cannot be in the same jacket as the speaker wire. The internal ST-MIC switch (located on the DIN clip side) must be in the MIC position
- The IP7-FX can also use the speaker (connected to the J2-6 and J2-7 connectors) as a microphone (half duplex mode only). To enable this mode, slide the internal **ST-MIC** switch (located on the DIN clip side) to the **ST** position
- The IP7-FX can drive either an 8 Ohm speaker or an amplifier. Connect an 8 Ohm speaker to the J2-6 and J2-7 (polarity independent) terminals using 18 AWG. To connect to an amplifier or self-amplified speaker, connect the J2-3 (Line Out) to the amplifier's "+" terminal and J2-4 (ground) to its "-" terminal using 18 22 AWG. Follow any recommendations by the amplifier's manufacture for balanced vs unbalanced audio signals.
- The IP7-FX can initiate a SIP call by grounding the J2-5 (Talk) connector. Connect a button or switch to J2-5 (Talk) and J2-4 (Ground) using 18 22 AWG wire. When J2-5 is grounded, the IP7-FX will dial a number stored in the IP7-FX configuration
- If the relay will be used to activate an electronic door strike, connect the J3-2 (COM) and either the J3-1 (NO) or J3-3 (NC) to the door strike. The relay can be activated during a call by entering a code on a telephone keypad that is stored in the IP7-FX configuration
- The IP7-FX can be DIN rail or surface mounted. To install on a standard 35mm DIN rail using the integrated DIN clip, tilt the top of the unit (J1, J2, J3 connectors facing up with Volume buttons facing forward) back towards the DIN Rail until the IP7 DIN clip catches the top of the rail. Press in at the bottom of the IP7 to snap into place). To surface mount, snap the included Mounting Plate in half, line the two pieces (beveled holes facing up) with the screw holes on the bottom of the IP7-FX case and secure it using the provided screws so that the mounting holes extend past the J1, J2 and J3 edge of the case
- Record the **ID#** from the label on back of the IP7-FX for use during software configuration.
- Optionally, the extra ID# label can be used for easy identification of this location
- Please refer to the *IP7-FX_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 <u>https://www.digitalacoustics.com/support/software-firmware-downloads/</u>

• 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 typically 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#)
- 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 automatcally assign the IP Address infromation or uncheck it to enter static IP address information





SIP Defaults		×
Primary Registrar	asterisk.digitalacoustics.com	5060
Primary Server	asterisk.digitalacoustics.com	5060
Secondary Registrar		0
Secondary Server		0
SIP Local Port	5060 RTP Port	21000
🗌 Full Duplex 🗌 D	efault to Listen 📃 Registra	ar is Proxy



Location Name	Device ID	Address
	04635d	192.168.5.179

General Settings SIP Se	ettings Advanced Settings
SIP Stand Alone	Device 💌
Location Name	Classroom 107
Assign IP Auto	omatically with DHCP
IP Address	45
Subnet Mask	
Gateway	
DNS Address	

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 IP7 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 IP7's 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 IP7's 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.

Select the Advanced Settings tab

- Click the **Set** button next to **Audio Profile** and **select** the **IP7-FX** audio profile if using full duplex. Otherwise select the **Half Duplex** audio profile
- If a microphone was connected to the Line In connectors, select Line In from the Microphone Input dropdown box
- 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 a full or half duplex audio profile has been configured. Volume can be adjusted when a SIP call is in progress by pressing:
 - $\circ~$ #00 Set volume down one level
 - \circ #01 Set volume level to 1
 - \circ #02 Set volume level to 2
 - #03 Set volume level to 3
 - \circ #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

General Settings SIP Settings Advanced Settings				
Call Button Handset Style="background-color: gray;"> Style="background-color: Gray;"/>				
Relay Activation Factory Defaults Mic Speaker				
Microphone Options				
Disable AGC Mic Level (0-63)				
Microphone Input Microphone				
Default Mic Type ULaw				
Audio Profile Half Duplex Set				

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General Settings SIP Setting	Advanced Settings			
Peer-to-Peer				
Primary Registrar	asterisk.digitalacoustics.com	5060		
Primary Server	asterisk.digitalacoustics.com	5060		
Secondary Registrar		0		
Secondary Server		0		
SIP Local Port	5060 RTP Port	21000		
SIP Extension	5802			
SIP Password	********			
SIP Authentication	5802			
🗌 Full Duplex 🔲 Default to Listen 🔲 Registrar is Proxy				