

IP7 Endpoint eSIP Standalone Configuration

Digital Acoustics' IP7 IP Endpoints support the TalkMaster FOCUS Protocol and the SIP 2.0 Protocol. The IP7 Endpoints can be configured as:

- TalkMaster FOCUS Endpoint
- SIP 2.0 Endpoint
- TalkMaster FOCUS / SIP 2.0 hybrid endpoint


These instructions explain how to configure IP7s to function as Standalone SIP Endpoints that register with a SIP Server.

Configure an extension on the SIP Server

Refer to the documentation included with the SIP Server for information on setting up SIP Extensions. Create an extension for each IP7 that will be configured as a Standalone SIP device. Refer to the end of this document for **Cisco Universal Call Manager** considerations.

Discover and configure the IP Endpoints on the network:

- Make sure the IP7 IP Endpoints are connected to the network and powered up
- Open the **TalkMaster FOCUS Admin Console**
- Select the **IP Endpoints** tab
- Press the **Find All** button in the right-hand corner of the screen. The IP Endpoints will be discovered on the network and will be listed in the left-hand pane. Match the **ICOM ID** to the ID# on the IP7's label

Setup	Queues	IP Endpoints	Messages	Paging Groups	Operators	Audit	Status	IP Video Sources
Auth	Location Name	Icom ID	IP Address	Queue	General	eSIP	ZC	
		0451AF	10.3.3.111	Unassigned	Configuration	Options 1	Options 2	Options 3

Repeat the following to configure each IP Endpoint as a SIP Extension:

- Select the IP Endpoint to be updated by clicking on it
- On the **Configuration** sub tab, set the Type to **SIP Stand-alone**
- Enter a **Location Name**. This will only be displayed in the Admin Console. It is not used by the SIP Server
- Select **Assign IP automatically with DHCP** or enter the static IP address you wish to assign the device
- If static IP Addresses are being assigned to the endpoints, enter the **IP Address, Subnet Mask and Gateway**

General | eSIP | ZC |
Configuration | Options 1 | Options 2 | Options 3 |

Type: SIP Stand-alone

Location Name:

☒ Assign IP automatically with DHCP

IP Address: 10.3.3.111

Subnet Mask: 255.255.255.0

Gateway: 10.3.3.1

Port: 0

☐ Authorized IP Endpoint

Server IP: 0.0.0.0 : 3000

- Select the **eSIP → Options 1** sub tab
- Select the **Primary Registrar** and **Primary SIP Server** options individually and enter the IP Address or DNS and Port of the SIP Server. Typically, these have the same IP Address
- Select the Local **SIP Port** option and enter the default SIP Port for the IP Endpoint
- Select the **RTP Port** option and enter the default RTP Port for the IP Endpoint
- For "Paging" endpoints, select **SIP Extension Only** from the drop-down list. If the endpoint should dial an extension when its **TALK** button is engaged, select **Dial SIP Extension on PTT** from the drop-down list
- Enter the **SIP Extension**
- Enter the **SIP Password**
- Enter the **SIP Authentication** or the SIP Extension if a User is not associated with the Extension
- Check the **SIP Full Duplex** box if an IP7-FD or an IP7 with a handset or headphone is being used
- If the **Dial SIP Extension on PTT** option has been selected, enter the number to be automatically dialed in the **Out Dial Extension**.
- Press the **Save** button

General | eSIP | ZC |
Options 1 | Options 2 |

SIP Extension Only

Primary Registrar: asterisknow.digitalacousti 5060

Primary Server: asterisknow.DigitalAcoust 5060

Secondary Registrar: 0

Secondary Server: 0

SIP Local Port: 5060 RTP Port: 46000

SIP Extension: 6002

SIP Password: tle5)12!

SIP Authentication: 6002

SIP Full Duplex: ☒ Out Dial on Queue Overflow: ☐

Out Dial Extension:

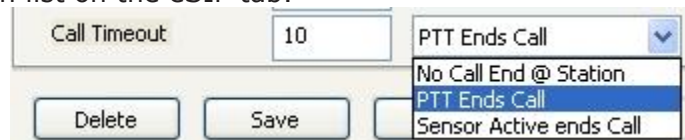
Door Open Code: 5 Then press #

Call Timeout: 5 No Call End @ Station

After the IP Endpoint restarts, the IP Endpoint will register with the Primary SIP Server.

IP7 IP Endpoint Operation Considerations

- IP7s automatically answer incoming SIP calls.
- For Half-Duplex IP7s in a two-way audio call with a Phone:
 - When an IP7 connects to a SIP call, the IP7's speaker is active and will play any audio received from the Phone's microphone. The Phone user must press the Phone's Zero key to change the audio direction
 - When an IP7 is configured with the **Default to Listen** option on the **eSIP** → **Options 2** tab and connects to a SIP call, the Phone's speaker or earpiece is active and will play any audio received from the IP7's microphone. The Phone user must press the Phone's Zero key to change the audio direction
- When an IP7 is configured with a **Door Open Code** on the eSIP tab:
 - If the **Door Open Code** is one digit long, enter the digit on the phone to open the door
 - If the **Door Open Code** is more than one digit long, enter the digits followed by the # key to open the door
 - Each IP7 can be configured with a different **Door Open Code**
- The IP7 can be configured to hang up a call by activating either the IP7's **TALK** (J2-4, J2-5) button or the **Sensor** (J3-5, J3-6). The corresponding **PTT Ends Call** or **Sensor Active Ends Call** option must be selected from the drop down list on the eSIP tab.



The screenshot shows a configuration window with a 'Call Timeout' field set to '10'. Below it are 'Delete' and 'Save' buttons. To the right is a dropdown menu labeled 'PTT Ends Call' which is open, showing three options: 'No Call End @ Station', 'PTT Ends Call' (which is highlighted in blue), and 'Sensor Active ends Call'.

- If the IP7 is unable to contact its **Primary SIP Server/Registrar**, it will automatically register with its **Secondary SIP Server/Registrar** if configured. When the **Primary SIP Server** is available, the IP7 will un-register with the **Secondary SIP Server** and re-register with the **Primary SIP Server**

Cisco Universal Call Manager Configuration Considerations:

IP7's must be configured as **Third-party SIP Device (Basic)** on the Cisco Call Manager. Make sure sufficient **Device License Units** are available.

- Create a **Phone Security Profile** that has the **Enable Digest Authentication** option selected and a **Transport type** of **UDP**
- Create a **User** for each IP7. Enter the same value for the **Password, PIN** and **Digest Credentials**
- Create a **Third-party SIP Device (Basic)** for each IP7:
 - The first six digits of the IP7's MAC Address are 00:1B:61. The last six digits are the IP7's ID#. The ID# is printed on the IP7's outside carton and on the IP7's back label and side label
 - Specify the **User** created in the previous step as the **Owner User ID**
 - Specify a codec of **711uLaw** or **711aLaw**
 - Specify the **Phone Security Profile** created in the first step
 - Specify the **User** created in the previous step as the **Digest User**
- Add a **DN** to the device - No Voice Mail box should be associated with the IP7
- Associate the **User** with the **Device** and set the **Primary Extension**

The CUCM **User** should be used in the IP7's **SIP Authentication** field