

IP7 Endpoint eSIP Configuration

Digital Acoustics' IP7 IP Endpoints support the SIP 2.0 Protocol. The TalkMaster FOCUS Administrator software must be used to configure the IP Endpoints. An IP7 can be configured as a:

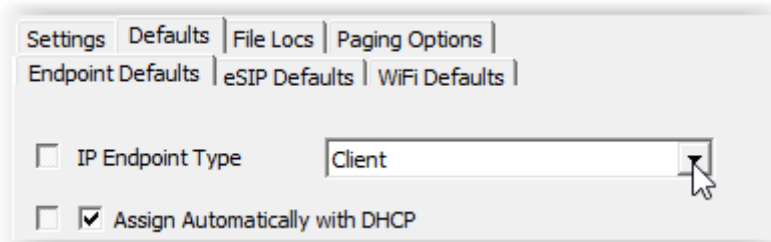
- **TalkMaster FOCUS Client** – The device will initiate a connection to the TalkMaster FOCUS Server
- **Stand-alone SIP endpoint** – The device will register with the SIP Server
- **Hybrid TalkMaster FOCUS Client and SIP Endpoint** - The IP7 Endpoint will connect to the TalkMaster FOCUS Server and then register with the SIP Server. This option provides the ability for the device to be used in either mode and to be managed by using the Administrator Console and Firmware Console

Configure an extension on the SIP Server

To begin, refer to the documentation included with the SIP Server for information on setting up a SIP Extension. Refer to the end of this document for **Cisco Universal Call Manager** considerations.

Setup the defaults – The **Defaults** tab provide the ability to setup a “template” to be used when configuring an IP7 IP Endpoint

- Install the **TalkMaster Server/Admin Console** software from the Digital Acoustics Website
- Open the Administrator Console by clicking the Windows **Start** menu and selecting **TalkMaster FOCUS Administrator**
- Select the **Setup → Defaults → Endpoint Defaults** tab



- Enable the **IP Endpoint Type** option and select either **Client** or **SIP Stand-alone** from the drop-down box.
- Enable the **Assign Automatically with DHCP** option. Check the box to enable DHCP or uncheck it to assign static IP Addresses

- Select the **Setup → Defaults → eSIP Defaults** tab

Setting	Value
<input type="checkbox"/> Primary Registrar	AsteriskNowpjSIP.digitala 5060
<input type="checkbox"/> Primary SIP Server	AsteriskNowpjSIP.digitala 5060
<input type="checkbox"/> Secondary Registrar	Grandstream.digitalacous 5060
<input type="checkbox"/> Secondary SIP Server	Grandstream.digitalacous 5060
<input type="checkbox"/> Local SIP Port	5060
<input type="checkbox"/> RTP Port	46000
<input type="checkbox"/> Door Code	5
<input type="checkbox"/> Call Timeout	5

- Enable the **Primary Registrar** and **Primary SIP Server** options and enter the IP Address or DNS and Port of the SIP Server. Typically, these have the same IP Address or DNS (Note that DNS is only supported in IP7 Firmware V7 or greater)
- Enable the **Local SIP Port** option and enter the default SIP Port for the IP Endpoint
- Enable the **RTP Port** option and enter the default RTP Port for the IP Endpoint
- If the IP7's onboard relay will be used to open a door, enable the **Door Code** option and enter the 1 to 4 digit value to be entered thru the Phones keypad to activate the relay
- Enable the **Call Timeout** option and enter the number of seconds to timeout an active call. While in a SIP call, if the network connection with the other party in the conversation is interrupted, this is the number of seconds the IP7 will wait for RTP audio to be received before automatically ending the call

Discover and configure the IP Endpoints on the network:

- Make sure the IP7 IP Endpoints are connected to the network and powered up
- 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# printed on the IP7's label

Auth	Location Name	Icom ID	IP Address	Queue
○		049875	10.3.3.24	Unass

Repeat the following to configure each IP Endpoint:

- Select the IP Endpoint to be updated by clicking on it
- On the Configuration sub tab, press the **Copy Defaults** button at the bottom of the right-hand pane to copy in the previously setup defaults
- Optionally, enter a **Location Name**. This will only be displayed in the Admin Console. It is not used by the SIP Server
- The **Authorized IP Endpoint** and **Server IP** fields only need to be filled out if **Client** has been selected for the **Type** field

The screenshot shows the 'General' configuration tab for an IP endpoint. The 'Type' is set to 'Client'. The 'Location Name' is 'Main Floor Lobby'. The 'Adapter In Use' is 'Ethernet'. The IP configuration fields are: IP Address (10.3.3.24), Subnet Mask (255.255.255.0), Gateway (10.3.3.1), and DNS Address (192.168.254.237). The 'Server-Peer Port' is 3000. The 'Authorized IP Endpoint' checkbox is checked, with 'Server IP' set to 10.3.3.220 and port 3000. A 'Copy Defaults' button is visible at the bottom right. System information at the bottom left includes: NIC Version 7.3.0.19, ICOM Version 7.3.0.19, 17-1.0-8.E2, ZC Version Unknown - No Board Found, and WiFi Adapter Not Found.

- Select the **Adapters** tab
- If static IP Addresses are being assigned to the endpoints, uncheck the DHCP option and enter the **IP Address, Subnet Mask** and **Gateway**

The screenshot shows the 'Adapters' configuration tab. It is divided into two sections: 'IP Endpoint's Ethernet Adapter' and 'IP Endpoint's Wireless Adapter'. In the Ethernet section, the 'Assign IP Automatically with DHCP' checkbox is checked, and the IP configuration fields (IP Address: 10.3.3.24, Subnet Mask: 255.255.255.0, Gateway: 10.3.3.1, DNS Address: 192.168.254.237) are visible. In the Wireless section, the 'Assign IP Automatically with DHCP' checkbox is unchecked, and the 'Alt. Network Set' checkbox is also unchecked. The IP configuration fields for the wireless adapter are all set to 0.0.0.0.

- Select the **eSIP** sub tab

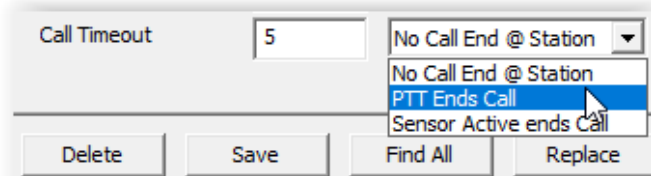
The screenshot shows the configuration window for the eSIP sub-tab. The 'Primary Registrar' dropdown menu is open, with 'Dial SIP Extension On PTT' selected. The 'SIP Full Duplex' checkbox is checked. The 'Out Dial Extension' field is empty. The 'Door Open Code' is set to 5, and the 'Call Timeout' is set to 5. The 'RTP Port' is set to 46000.

- 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** that was defined in the SIP Server
- Enter the **SIP Password** that was defined in the SIP Server
- Enter the **SIP Authentication** or User associated with the Extension that was defined in the SIP Server. Typically, this is the same as the Extension
- If an IP7 supports full-duplex, check the **SIP Full Duplex** box
- If the **Dial SIP Extension on PTT** option has been selected, enter the number to automatically dialed in the **Out Dial Extension**
- Press the **Save** button

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 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 SIP 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



- 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 are 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**