

# HS-A1

## Getting Started Guide

### Introduction

The HS-A1 is an analog handset with an armored cable and a hook switch mechanism designed to be connected to an IP7 intercom module to enable full duplex audio operations with a TalkMaster operator console or SIP Phone.

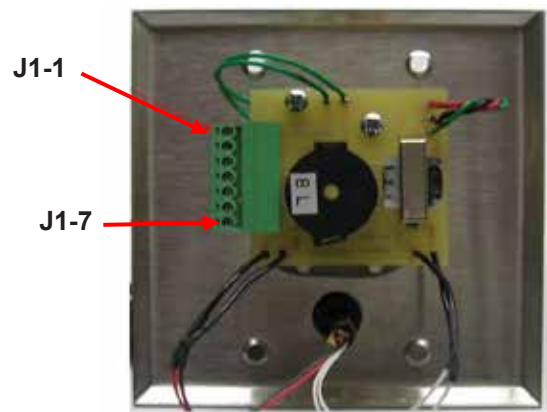
The HS-A1 can place a call by removing the handset from the switch hook. If configured for TalkMaster, it will place an Incoming Call. If the IP7 is configured to dial out to a SIP Extension, it will initiate a call to the specified phone number.

If the HS-A1 is called from TalkMaster or from a SIP phone, the internal piezo buzzer will sound to indicate there is an incoming call. The HS-1A must be connected to either an IP7-SE8 or IP7-FD IP Module.



# Installation

The Handsets are designed to be flush mounted in a 2-gang box. A 3-pair cable is used to connect the Handset to an IP7-SE8. On the back of the handset is a seven position 3.81mm DIN connector. The connection details are:



Connector	Signal	IP7 Connections
J1-1	SPKR 8Ω+	Connect to J2-6 - SPKR 8Ω+ on the IP7
J1-2	SPKR 8Ω-	Connect to J2-7 - SPKR 8Ω- on the IP7
J1-3	RESERVED	
J1-4	Hook Switch	Connect to J3-5 – SENSOR on the IP7
J1-5	GND	Connect to J2-4 – GND on the IP7
J1-6	MIC -	Connect to J2-2 - MIC- on the IP7
J1-7	MIC +	Connect to J2-1 - MIC+ on the IP7

For best audio quality, install the Handset as close as possible to the IP7 to reduce the chance of picking up audio noise. Use an 18 AWG 3-pair multi-conductor with a shield. Refer to the IP7 Reference Manual for instructions on connecting the shield and reducing audio noise.

# Configuration

TalkMaster FOCUS Software is a suite of Windows® based applications used to configure and manage Digital Acoustics IP Intercoms and Paging endpoints. TalkMaster FOCUS is available for download on the Digital Acoustics Website. The TalkMaster Server/Admin Console must be installed to configure IP Endpoints.

The following procedure is for configuring endpoints as TalkMaster Clients. For SIP Configuration, see the **eSIP Stand-alone Configuration Steps** topic in the Admin Console online help.

To enable Handset support for an IP7:

- Power up the IP7 and connect it to the network
- Open the TalkMaster FOCUS Administrator console
- Select the **IP Endpoints** tab
- Click the **Find All** button. This will discover any new intercoms added to the network
- Select the IP7 used for the Handset by matching the ICOM ID on back of the unit with the ICOM ID displayed in the left-hand pane of the **IP Endpoints** tab
- Enter the **Location Name**, **IP Address** information, check the **Authorized IP Endpoint** box and enter the **Server IP** and **port**

The screenshot shows the 'eSIP' configuration tab in the TalkMaster FOCUS software. The 'Configuration' sub-tab is active. The 'Type' is set to 'Client'. The 'Location Name' is 'Handset'. The 'Assign IP automatically with DHCP' checkbox is checked. The IP configuration fields are: IP Address (10.3.3.157), Subnet Mask (255.255.255.0), Gateway (10.3.3.1), DNS Address (192.168.254.237), and Port (0). The 'Authorized IP Endpoint' checkbox is also checked. The 'Server IP' is set to 10.3.3.220 and the port is 3000.

General	eSIP	ZC
Configuration	Options 1	Options 2   Options 3
Type	Client	
Location Name	Handset	
<input checked="" type="checkbox"/> Assign IP automatically with DHCP		
IP Address	10.3.3.157	
Subnet Mask	255.255.255.0	
Gateway	10.3.3.1	
DNS Address	192.168.254.237	
Port	0	
<input checked="" type="checkbox"/> Authorized IP Endpoint		
Server IP	10.3.3.220	: 3000

- Select the **Options1** tab
- Check the **Disable Sensor Repeat** box
- Check the **Use UDP/RTP** box
- Check the **Supports Full Duplex** box

The screenshot shows the 'Options 1' configuration tab. It includes fields for 'Fail Forward IP 1', 'Fail Forward IP 2', and 'Fail Forward IP 3', each with a port dropdown set to 0. There is a checkbox for 'Enable Automatic Return to primary'. The 'Relay Mode' is set to 'Factory Defaults'. A 'Security' section has a 'Set' button. The 'Sensor' is set to 'Activate when grounded (Door)' and the 'Input Source' is 'Microphone'. Under 'Audio Settings', 'Enable ST/ADA Mode' is unchecked, 'Disable Sensor Repeat' is checked, 'Use UDP/RTP' is checked, 'Fixed RTP Port' is 0, and 'Supports Full-Duplex Audio' is checked.

- Select the **Options 2** tab
- Place a check next to the **Handset** option
- Under **Audio Settings**, if paired with an IP7-SE8, check the **Disable AGC** box and enter 16 for the Microphone Sensitivity
- Under **Audio Settings**, change the **Default Microphone** to **uLaw**
- If paired with an IP7-SE8, check **Disable AGC** and enter **16** for the microphone sensitivity
- If paired with an IP7-FD, press the **SET** button and select **HS-A1 Handset** from the list
- Click the **Save** button to save the changes to the Intercom

The screenshot shows the 'Options 2' configuration tab. It includes checkboxes for 'Remote Listen Disable', 'Call Button', 'Handset' (checked), 'IP7 Tracking Enabled', 'Use IGMP v3 instead of IGMP v2', and 'Multicast Background Music'. There are also checkboxes for 'Show on Paging Tab', 'Speaker', and 'PTT Sends Audio'. The 'Server-Peer Settings' section includes checkboxes for 'No Beep Tone After PTT Released', 'Default to Listen (Direct Mode)', 'Server-Peer Hands Free Mode', and 'Send Sensor State to Client'. The 'Audio Settings' section includes 'Disable AGC (0-63)' checked with a value of 16, 'Default Microphone' set to 'uLaw', and 'Audio Profile' set to 'None'. A 'Set' button is at the bottom right.