

# **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.

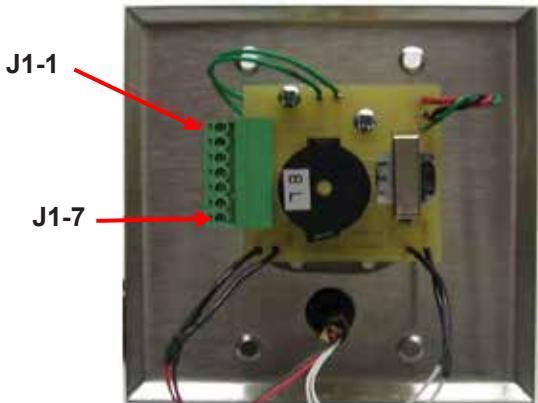
The HS-A1 can place a call by removing the handset from the switch hook. If configured for TalkMaster, it will place an Incoming Call. If the IP7 is configured to dial out to a SIP Extension, it will initiate a call to the specified phone number.

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**

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
Port	3000

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

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

Fail Forward IP 1: 0.0.0.0 : 0  
 Fail Forward IP 2: 0.0.0.0 : 0  
 Fail Forward IP 3: 0.0.0.0 : 0

Enable Automatic Return to primary

Relay Mode

Factory Defaults

Security

Sensor: Activate when grounded (Door)

Input Source: Microphone

Enable ST/ADA Mode  Disable Sensor Repeat

Use UDP/RTP  Fixed RTP Port: 0

Supports Full-Duplex Audio

- 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

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

Remote Listen Disable  Show on Paging Tab

Call Button

Handset  Speaker

IP7 Traling Enabled

Use IGMP V3 instead of IGMP V2

Multicast Background Music  PTT Sends Audio

225.0.0.254 : 4000 8000

Server-Peer Settings

No Beep Tone After PTT Released

Default to Listen (Direct Mode)

Server First Hands Free Mode

Send Sensor State to Client

Audio Settings

Disable AGC (0-63) 16

Default Microphone: uLaw

Audio Profile: None

Set