



**AVM-1**  
**IP Video Attendant Station**  
Configuration and Operation Manual  
v1.0.0

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## Acronyms and Abbreviations

The following acronyms and abbreviations are commonly used throughout the document:

Acronyms	Definitions
AEC	Acoustic Echo Cancellation
AGC	Automatic Gain Control
CNG/VAD	Comfort Noise Generator/Voice Activity Detector. It is used to reduce the transmission rate during inactive speech periods while maintaining an acceptable level of output quality.
DHCP	Dynamic Host Configuration Protocol — protocol for assigning dynamic IP addresses to devices on a network.
DNS	Domain Name Server
DTMF	Dual Tone Multi Frequency signaling is used for telecommunication signaling over telephone lines.
FTP	File Transfer Protocol
GUI	Graphical User Interface
G.711	G.711 is codec also known as Pulse Code Modulation (PCM). It is the ITU-T international standard for encoding telephone audio on a 64 kbps channel.
G.723	G.723 is an ITU-T standard speech codec.
G.729	G.729 is an audio data compression algorithm. It is the ITU-T international standard for encoding telephone audio on 8 kbps channel.
IP-PBX	It is an IP based switch for call handling through public and private exchanges.
IE	Internet Explorer
IETF	The Internet Engineering Task Force (IETF) develops and promotes Internet standards.
PoE	Power over Ethernet, IEEE 802.3af standard.
QoS	Quality of Service is of particular concern for the continuous transmission of high-bandwidth video and multimedia information.
SIP	Session Initiation Protocol is a signaling protocol, widely used for setting up and tearing down multimedia communication sessions over network.
VoIP	Voice over Internet Protocol

## 1. Introduction

### 1.1. Overview of the Manual

This manual provides detailed instructions for the configuration and operation of AVM-1 IP Video Attendant Station. It is recommended to read this instructional manual completely before performing any configuration.

### 1.2. Intended Audience

This manual is targeted towards systems administrators, or any person who would configure and maintain AVM-1 IP Video Attendant Station. Fundamental knowledge in computer networking and Voice over Internet Protocol (VoIP) technologies is recommended for understanding this manual.

### 1.3. Objective

This manual provides a detailed examination of the features included in AVM-1 IP Video Attendant Station. It guides an administrator through the configuration and optimization of phone features. While configuration of the AVM-1 IP Video Attendant Station is covered in detail, configuration of other peripheral VoIP network elements is beyond the scope of this document.

### 1.4. Typographic Conventions

The following guidelines are used as typographic conventions in this user manual:

Item	Convention	Sample
Acronyms	All uppercase	SIP
Chapter titles	Title caps	See Chapter 3 Getting Started
Command-line commands and options (switches)	All lowercase, bold	<b>ifconfig</b> command <b>/a</b> option
Device names	All uppercase	AVM-1
Directories	All lowercase	/flash
Error message names	Initial caps	Update failed
File names	Title caps (internal caps in short file names are acceptable for readability)	MainLogFile.txt BackupLogFile.txt
Menu names	Bold; title caps	<b>Insert</b> menu
Programs and applications	Usually title caps	<b>HyperTerminal</b>
Toolbar button names	Usually title caps (follow the interface); bold	<b>Apply</b> <b>Reset</b>
URLs	All lowercase; break long URLs before a forward slash, if necessary to break; do not hyphenate.	<a href="http://www.talkaphone.com/">http://www.talkaphone.com/</a>
User input	Usually title caps; bold	<b>Enter Password</b>

## 1.5. Related Documents

VOIP-200 Series IP Call Station Configuration and Operation Manual:

[http://talkaphone.com/wp-content/uploads/2015/03/VOIP-200\\_Manual\\_Rev\\_1\\_0\\_2\\_09032015.pdf](http://talkaphone.com/wp-content/uploads/2015/03/VOIP-200_Manual_Rev_1_0_2_09032015.pdf)

VOIP-500 Series IP Call Station Configuration and Operation Manual:

[http://talkaphone.com/wp-content/uploads/2012/08/VOIP-500\\_Manual\\_Rev\\_3\\_0\\_2\\_07312012.pdf](http://talkaphone.com/wp-content/uploads/2012/08/VOIP-500_Manual_Rev_3_0_2_07312012.pdf)

VOIP-600 Series IP Call Station Configuration and Operation Manual:

[http://talkaphone.com/wp-content/uploads/2014/09/VOIP-600\\_Manual\\_Rev\\_1\\_0\\_1\\_09172014.pdf](http://talkaphone.com/wp-content/uploads/2014/09/VOIP-600_Manual_Rev_1_0_1_09172014.pdf)

VOIP-500 Quick Installation Guide:

[http://talkaphone.com/wp-content/uploads/2014/03/VOIP-500\\_Quick\\_Installation\\_Guide\\_Rev\\_2\\_0\\_0\\_02042014.pdf](http://talkaphone.com/wp-content/uploads/2014/03/VOIP-500_Quick_Installation_Guide_Rev_2_0_0_02042014.pdf)

VOIP-600 Quick Installation Guide:

[http://talkaphone.com/wp-content/uploads/2014/09/VOIP-600\\_Quick\\_Installation\\_Guide\\_Rev\\_1\\_0\\_09102014.pdf](http://talkaphone.com/wp-content/uploads/2014/09/VOIP-600_Quick_Installation_Guide_Rev_1_0_09102014.pdf)

AVM-1 Quick Installation Guide:

[http://www.grandstream.com/sites/default/files/Resources/gxv3275\\_QIG\\_0.pdf](http://www.grandstream.com/sites/default/files/Resources/gxv3275_QIG_0.pdf)

Grandstream Networks GXV3275 Administrator Guide:

[http://www.grandstream.com/sites/default/files/Resources/gxv3275\\_administration\\_guide.pdf](http://www.grandstream.com/sites/default/files/Resources/gxv3275_administration_guide.pdf)

## 1.6. Technical Support / Help Desk

For technical assistance beyond the scope of this document, contact your distributor or Talkaphone Technical Support for further information.

Talk-A-Phone Co.

7530 North Natchez Avenue

Niles, Illinois 60714

**Phone:** 773.539.1100

**Fax:** 773.539.1241

**Email:** [support@talkaphone.com](mailto:support@talkaphone.com)

**Web:** [www.talkaphone.com](http://www.talkaphone.com)

## **2. Overview**

### **2.1. AVM-1 IP Video Attendant Station Overview**

The AVM-1 is an IP Video Attendant Station that provides peer-to-peer (P2P-SIP) communication with Talkaphone's VOIP-200/VOIP-500/VOIP-600 Series IP Call Stations. The AVM-1 can also display video from VOIP-200/VOIP-500/VOIP-600 Series IP Call Stations equipped with a camera. Up to thirty (30) VOIP-200/VOIP-500/VOIP-600 Series IP Call Stations are supported by the AVM-1.

The AVM-1 is a SIP compliant telephony device (RFC 3261) and provides basic telephony features such as hold, transfer, call waiting, and call history.

As a result, the AVM-1 will also function as a SIP endpoint on a SIP-based Private Branch Exchange (PBX). Please note that the video functionality of the AVM-1 will not interoperate with a SIP-based PBX.

## **3. Getting Started**

### **3.1. Pre-requisites**

Prior to configuring a AVM-1 IP Video Attendant Station, ensure the unit is powered on and connected to the network. When powered on and connected to a network, the AVM-1 IP Video Attendant Station will display its IP address through the "Account" widget located on the home screen.

The AVM-1 can be configured from a computer with a Web browser such as Google Chrome, Microsoft Edge, Microsoft Internet Explorer, and Mozilla Firefox.

### **3.2. Configuration Through the Web GUI**

1. Ensure both the AVM-1 and your PC are connected to the Local Area Network or a direct connection with an Ethernet network cable via the PC port on the rear of the AVM-1.

The AVM-1 IP Video Attendant Station is pre-configured with the following login credentials:

**IP Address:** 192.168.1.200  
**Username:** admin  
**Password:** admin

2. Configure the IP address of your PC to be on the same subnet as the AVM-1. For example, 192.168.1.3
3. Open a supported Web browser and direct it to the IP address of the AVM-1. The AVM-1 will prompt for authentication:



**Figure 1.** Web GUI authentication request.

4. Enter the default Username and Password. After authentication is successful, you are redirected to the main page of the Web GUI.
5. The AVM-1 also supports DHCP and can be reconfigured from the factory-configured static IP address to DHCP under **Maintenance** → **Network Settings** → **Address Type**.
6. Further configuration of AVM-1 settings is examined in Section 4: Factory Configuration.

## **4. Factory Configuration**

This section describes the various configuration parameters of the AVM-1 as programmed from the factory. Performing a **Factory Reset** or defaulting the AVM-1 will clear all factory programming. Factory programming can be restored by downloading the factory configuration file from Talkaphone.com and using the **Upload Device Configuration** feature.

### **4.1. Account**

#### **4.1.1. General Settings**

The AVM-1 has six (6) lines that can be configured to accommodate up to six (6) independent SIP accounts. Each SIP account has an individual configuration page.

**Account 1** is the only SIP account configured at time of manufacture.

Account → General Settings → Account

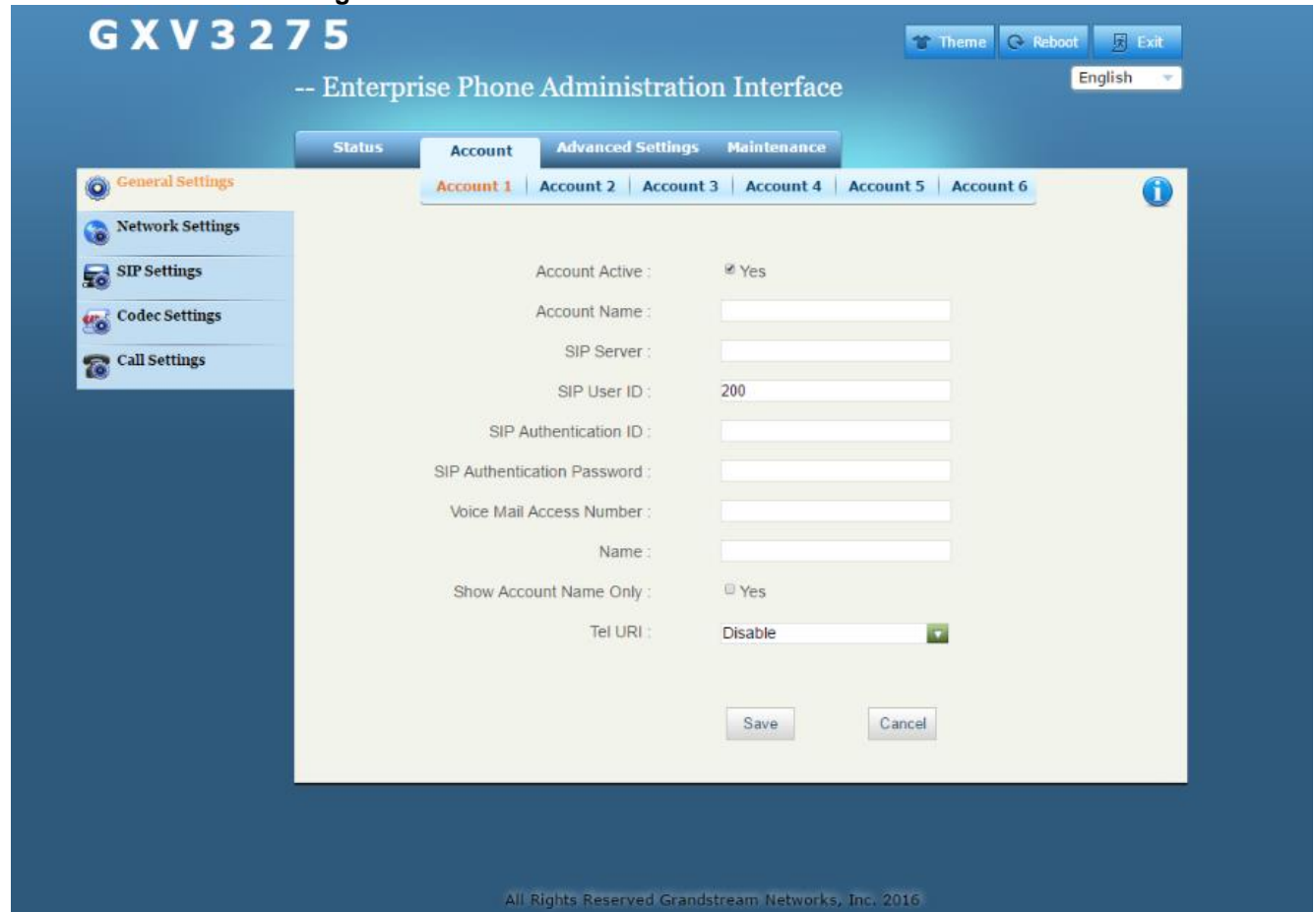
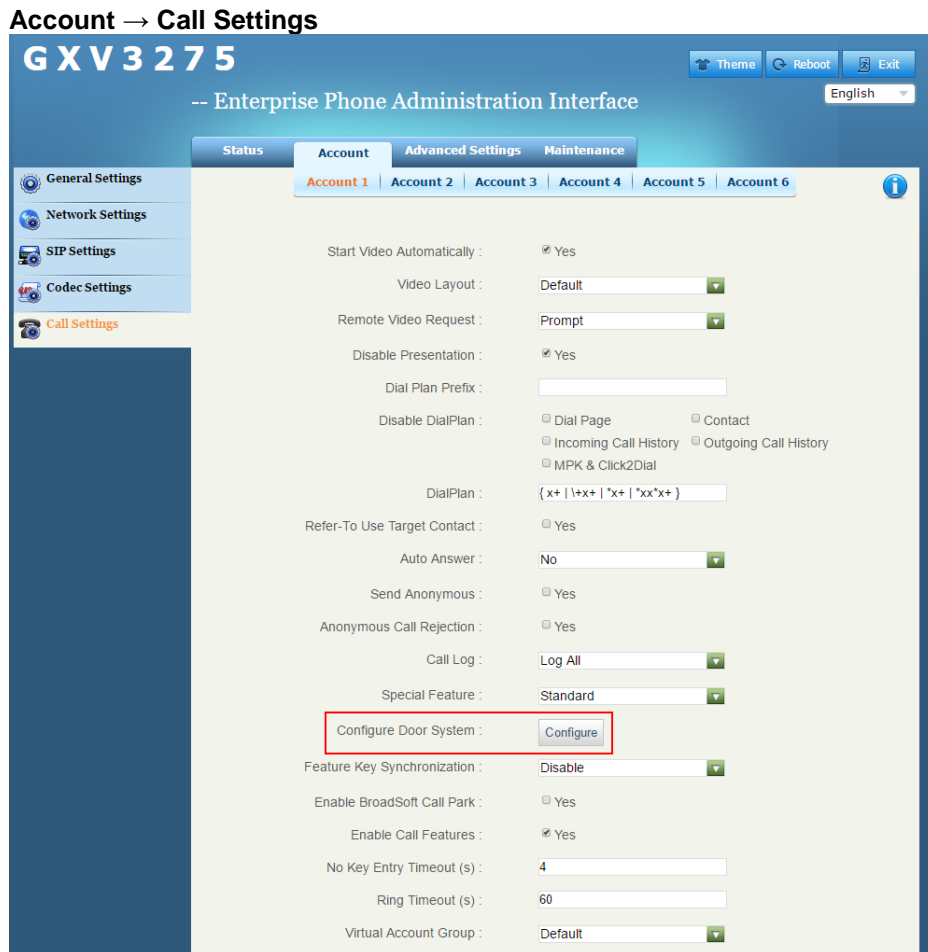


Figure 2. General settings.

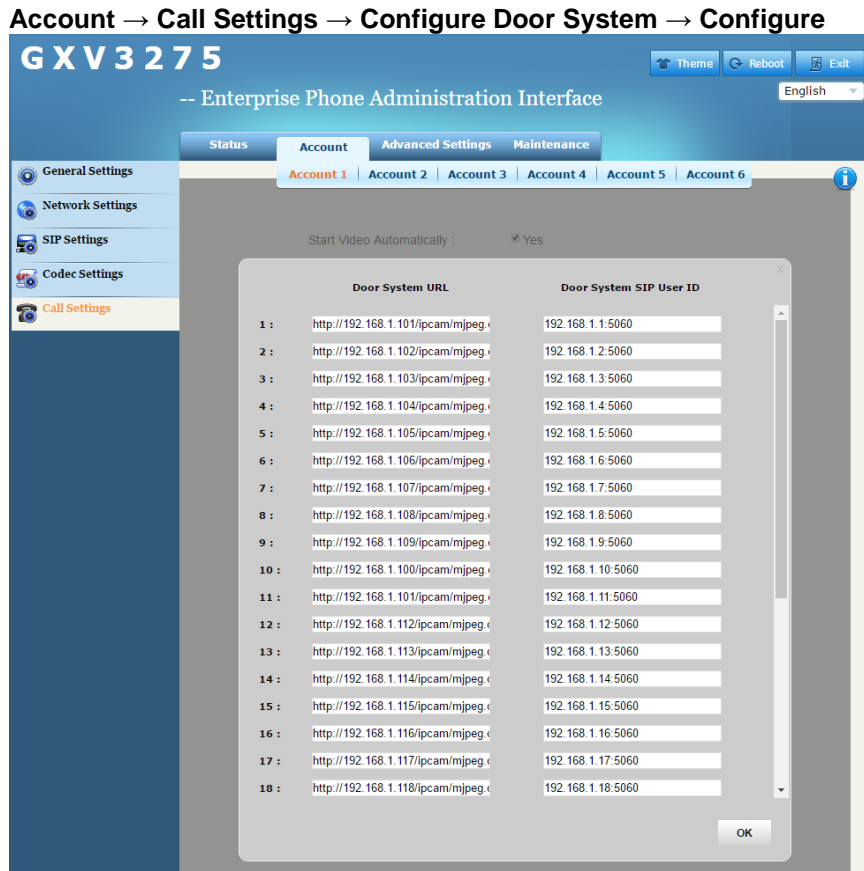
Parameter Name	Description
Account Active	Indicates whether the account is active. The factory value for <b>Account 1</b> is <b>Yes</b> . The factory value for the IPVideoTalk account ( <b>Account 6</b> ) is <b>Yes</b> . The factory value for <b>Account 2</b> , <b>Account 3</b> , <b>Account 4</b> , and <b>Account 5</b> is <b>No</b> .
Account Name	Configures the name associated with each account to be displayed on the LCD.
SIP User ID	Configures the phone number or SIP User ID of the AVM-1. This parameter is usually in the form of digits similar to a phone number or is actually a phone number. The factory value is <b>200</b> .



4.1.2. Call Settings



**Figure 3.** Call settings and the location of the **Configure** button for configuring a “door system”.



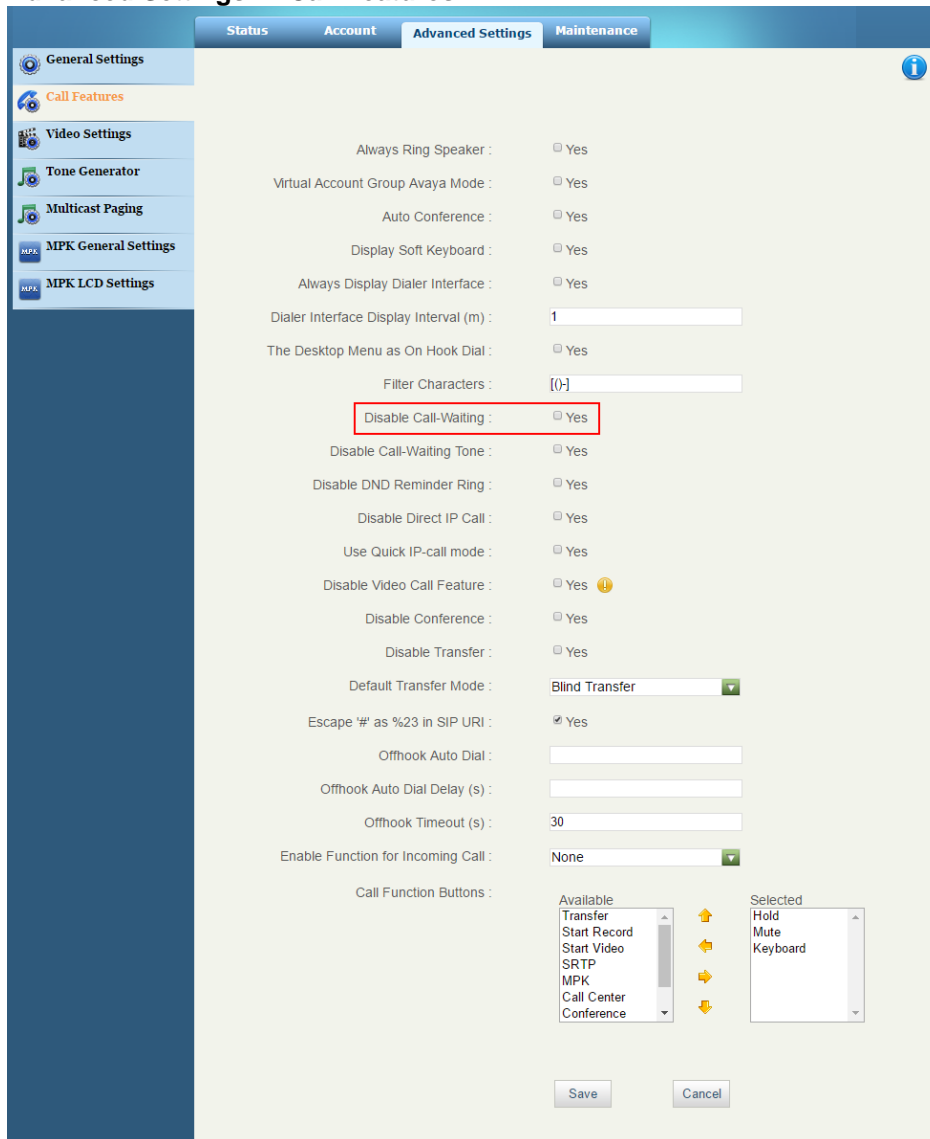
**Figure 4.** The directory for configuration of a “door system”.

Parameter Name	Description
Door System URL	<p>Specifies the URL of a camera for a VOIP-200/VOIP-500/VOIP-600 Series IP Call Station equipped with a camera.</p> <p>The factory values will specify up to thirty (30) VOIP-200/VOIP-500/VOIP-600 Series IP Call Station cameras in the following format:</p> <p><b>http://&lt;ip_address&gt;/ipcam/mjpeg.cgi</b> where &lt;ip_address&gt; is a value from <b>192.168.1.101</b> through <b>192.168.1.130</b></p>
Door System SIP User ID	<p>When used in conjunction with a SIP proxy/registrar, specify the <b>Directory Number</b> or phone number of each VOIP-200/VOIP-500/VOIP-600 Series IP Call Station. The value can be specified as a SIP directory number (DN) with port number.</p> <p>At the factory, the AVM-1 is configured in peer-to-peer mode. The factory values will specify up to thirty (30) VOIP-200/VOIP-500/VOIP-600 Series IP Call Stations in the following format:</p> <p><b>&lt;ip_address&gt;:5060</b> where &lt;ip_address&gt; is a value from <b>192.168.1.1</b> through <b>192.168.1.30</b></p>

## 4.2. Advanced Settings

### 4.2.1. Call Features

#### Advanced Settings → Call Features



The screenshot shows the 'Advanced Settings' configuration page for 'Call Features'. The 'Disable Call-Waiting' option is checked, and the checkbox is highlighted with a red box. Other settings include 'Always Ring Speaker', 'Virtual Account Group Away Mode', 'Auto Conference', 'Display Soft Keyboard', 'Always Display Dialer Interface', 'Dialer Interface Display Interval (m)', 'The Desktop Menu as On Hook Dial', 'Filter Characters', 'Disable Call-Waiting Tone', 'Disable DND Reminder Ring', 'Disable Direct IP Call', 'Use Quick IP-call mode', 'Disable Video Call Feature', 'Disable Conference', 'Disable Transfer', 'Default Transfer Mode', 'Escape '#' as %23 in SIP URI', 'Offhook Auto Dial', 'Offhook Auto Dial Delay (s)', 'Offhook Timeout (s)', 'Enable Function for Incoming Call', and 'Call Function Buttons'.

Figure 5. Disabling call waiting.

Parameter Name	Description
Disable Call Waiting	Disables the call waiting feature. If it is checked, the phone system will reject a secondary incoming call during an active session without notifying the attendant. However, this missed call record will be saved to the local call history of the AVM-1.  The factory value is <b>Yes</b> (i.e. <b>checked</b> ).

## **5. Operations**

### **5.1. Answering Calls**

One of the following scenarios occurs when answering calls from a call station at the AVM-1:

**A. Incoming Video Call:**

When the phone rings, the video feed corresponding to the VOIP-200/VOIP-500/VOIP-600 Series IP Call Station (equipped with a camera) phone number will display on the LCD. Tap on one of the following softkeys to answer the incoming call: “Audio Answer” or “Reject”

**B. Incoming Audio Call:**

When the phone rings, tap on the “Answer” or “Reject” softkey.

**C. Missed Call:**

If a call is not answered, a missed call message will show up in on the idle screen. Users can tap on the missed call to access further details.

### **5.2. Ending a Call**

A call can be ended at the AVM-1 by either (1) tapping on the “End” softkey or (2) hanging up the handset (i.e. placing back on-hook).

### **5.3. Remotely Activating/Deactivating Call Station Auxiliary Outputs**

During an active call, DTMF operation codes can be entered via the AVM-1 keypad to remotely activate or deactivate a specific IP Call Station Auxiliary Output. The DTMF operation code will be defined by the configuration of the IP Call Station.

### **5.4. Remotely Activating/Deactivating Call Station “Help on the Way” LED**

For IP Call Stations equipped with a “Help on the Way” LED, the AVM-1 is capable of activating/deactivating this LED remotely via a DTMF operation code. During an active call, this DTMF operation code can be entered via the AVM-1 keypad to remotely activate or deactivate the “Help on the Way” LED. The DTMF operation code will be defined by the configuration of the IP Call Station.

## 6. Appendix A: Network Ports

### 6.1. Requirements

The network for the AVM-1 IP Video Attendant Station requires the following:

- A. IPv4 enabled
- B. Allow the following protocols:
  1. SIP
  2. RTP
  3. HTTP/HTTPS
- C. Routed network
- D. DHCP Server (if applicable)

#### Ports:

Depending on the needed functionality, the following ports may need to be enabled and allowed across the firewall and routers in the network.

Service/Module	Port	Port Type	Mode
HTTP	80	TCP	Inbound
HTTPS	443	TCP	Inbound
TFTP	69	UDP	Inbound
SSH	22	TCP	Inbound
SIP Signaling	5060	UDP	Inbound/Outbound
SIP Secure	5061	UDP	Inbound/Outbound
SIP Start RTP	5004	UDP	Inbound/Outbound
STUN	3478	TCP/UDP	Inbound/Outbound

### 6.2. Optional

The network for the AVM-1 IP Video Attendant Station can optionally provide the following:

- A. SIP Proxy/Registrar