

AOR-IP-CSE Revision 1 Call station Configuration

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① This article assumes that the suggested default IP scheme for the AOR-IP series is being used. The IP scheme can be found [here](#).

① This article is for the first version of the AOR-IP-CSE That is based on the PCB with P/N 68820, It has two network ports and is configured via a web browser. Talkaphone switched to a different device after 8/2022. The new device has only a single nic and is configured via a configuration utility.

Configuration Via Browser

Accessing the Call Station On the Default IP Address

The default IP address of the AOR-IP-CSE series call stations is 192.168.1.10

Username: admin

Password: admin@123

⚠ The default IP address of the AOR-IP-CSE overlaps with an IP address used in the recommended IP scheme. It is advised to configure all of the AOR-IP-CSE series call stations before connecting to the final network. If a station needs to be added or replaced, it is recommended that the IP address be changed before it is connected to the actual network. Alternatively the Call station that uses the IP address 192.168.1.10 can be disconnected from the network temporarily to allow the configuration of a new device

1. Open a web browser and enter the default IP address into the address bar.
2. When the Login Page appears select **Secure Login** or **Unsecure Login**, the difference in these two login methods is that the password is encrypted during transmission when the **Secure Login** option is selected.

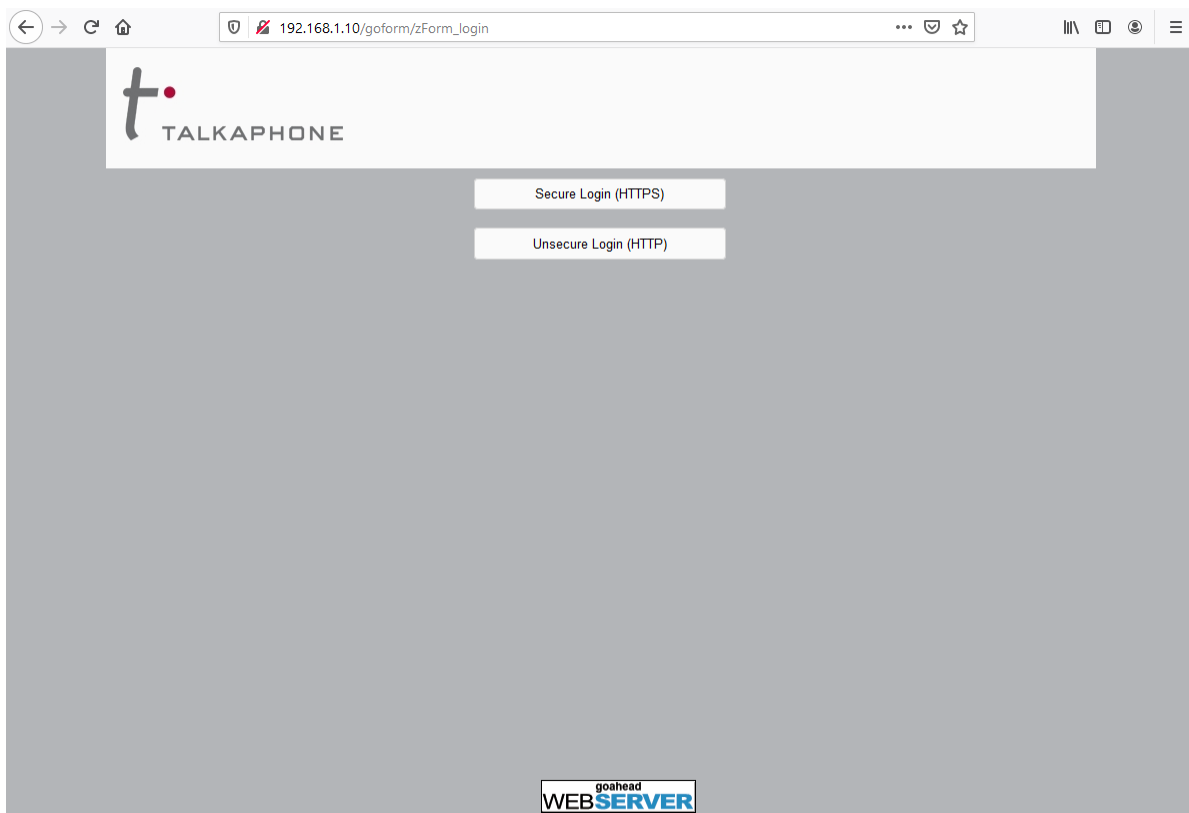


Figure 1 Station Web Login Page

3. When the Login Box appears enter the default credentials, and click ok.

Import Recommended Base Configuration.

1. Download and extract the provided Configurations from the link below.

[AOR-IP_Station_Configurations.zip](#)

2. In the Web GUI for the AOR-IP-CSE navigate to Administration>Configuration Updates.
3. Click on the **Browse** button and navigate to the extracted files, and the folder for the station you are configuring. The station number corresponds to the last octet of the IP address. Select [ipst_config.xml](#) and

click **Open** .

4. Click on Upload

5. Reboot the call station for the changes to take effect.

Manual Configuration

1. Navigate to Configuration>Buttons.

2. On the first line for Button 1 Call to, in the Options field select **Numberlist 1** .

3. Under Numberlist Settings, Ringlist 1, Value 1, Enter "1000" as the destination phone number for the Command Unit phone, or 9<phone number> for an outside number. If a secondary number is needed, enter that for Value 2 and continue for each additional number.

4. The **Ringer Timer** at the bottom of the page, controls how many rings before it changes to the next number in the list or returns to the first. 5 seconds is approximately 1 ring.

5. Click save.

The screenshot shows the Talkphone web interface for configuring buttons. The left sidebar contains navigation links: IP Settings, SIP Settings, Audio Settings, Buttons (selected), Auxiliary Output, Digital Outputs Scripts, Digital Outputs Events, Voice Messages Played to User, Voice Messages Played to Remote Side, and Time Settings.

Buttons (Idle)

	Function	Value	Option
Button 1	Call To		Numberlist 1
Button 2	Call To		None
Button 3	Call To		None

Buttons (Active Call)

	Function	Activated	Deactivated
Button 1	Do Nothing		
Button 2	Do Nothing		
Button 3	Do Nothing		

Numberlist Settings

	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	1000	☐		☐		☐
Value 2		☐		☐		☐
Value 3		☐		☐		☐
Value 4		☐		☐		☐
Value 5		☐		☐		☐
Value 6		☐		☐		☐
Value 7		☐		☐		☐
Value 8		☐		☐		☐
Value 9		☐		☐		☐
Value 10		☐		☐		☐
Value 11		☐		☐		☐
Value 12		☐		☐		☐
Value 13		☐		☐		☐
Value 14		☐		☐		☐

Call Until Answer ☒ (loops the numberlist)

Ringer Time seconds, (0=unlimited)

Call Conversation Timer seconds, (Range: 0 to 9999 seconds, 0 = unlimited)

Local Interdigit Timer seconds, (Range: 5 to 20 seconds)

Figure 2. Buttons

6. Click on **SIP Settings** .
7. In **Display Name** enter a description of the location for this call station, in this example it is "Station 1".
8. In **Directory Number (SIP ID)** enter the Extension for this call station, in this example it is "1001". See the [default IP Scheme](#) for the extensions and IP addresses
9. For the **Primary SIP Server** enter the IP address of the Command Unit, in the default configuration it is 192.168.1.199.
10. For **Username** enter the extension number, "1001" in this example. For the **Password** the default is "sip123".
11. Click **Save** .

The screenshot shows the 'SIP Settings' page in the Talkphone web interface. The browser window title is 'Station Web' and the address bar shows '192.168.1.1/goform/zForm_header'. The interface includes a sidebar with navigation links: Home, Configuration, Administration, Diagnostics, and Network Security. Under 'Configuration', there are sub-links for IP Settings, SIP Settings (selected), Audio Settings, Buttons, Auxiliary Output, Digital Outputs Scripts, Digital Outputs Events, Voice Messages Played to User, Voice Messages Played to Remote Side, and Time Settings.

The main content area is divided into two sections: 'Registration Settings' and 'Call Settings'.

Registration Settings

Description	Configuration
Display Name:	Station 1
Directory Number (SIP ID):	1001
Primary SIP Server:	192.168.1.199
Secondary SIP Server:	
Tertiary SIP Server:	
Registration Method:	Serial
Username:	1001
Password:	•••••
Re-registration Time:	1800 (Range: 60-14400 seconds)
Outbound Proxy 1 (optional):	Port: 5060
Outbound Proxy 2 (optional):	Port: 5060
Outbound Proxy 3 (optional):	Port: 5060

Call Settings

Description	Configuration
Enable auto-answer:	☒
Auto-answer Delay:	0 seconds (Range: 0 to 30 seconds)
Provisional Timer:	0 seconds (Range: 0 to 60 seconds) Delays call setup using input buttons
Overlap dialing:	☐
DTMF method:	RFC 2833
Call LED off during ringing:	☐
Hang-up on Silence Timer:	0 seconds (0 = Hang-up on Silence disabled)
Codec G.711 PCM u-law:	High Priority
Codec G.711 PCM A-law:	Low Priority
Codec G.722:	Low Priority
Codec G.729:	Low Priority

At the bottom of the 'Call Settings' section is a 'Save' button.

Figure 3. SIP Settings

12. Click on the **Back to config page** button.
13. Click on **IP Settings**.
14. Set the IP address according to the Default IP Scheme.
15. Click on the **Save** Button. Then click **Apply** to allow the station to reboot.