Quick Guide

AudioCodes WebRTC Solutions for Enterprises

WebRTC Web Softphone

Version 1.3.0

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Document Revision Record

LTRT	Description
14111	Initial document release (Ver. 1.1.0).
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14116	Updated for Ver. 1.3.0

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1 Introduction

AudioCodes offers a WebRTC-based softphone client that can be used by contact center agents in a Genesys PureEngage/PureConnect environment. This WebRTC softphone client is used by agents, in conjunction with Genesys agent desktop applications—such as Workspace Desktop Edition (WDE) or Workspace Web Edition (WWE) for PureEngage and Integration Desktop for PureConnect —as their telephony device that enables them to make and receive calls and perform other telephony functions.

1.1 Targeted Audience

This document is intended for contact center agents.

1.2 Purpose of Document

The purpose of this document is to provide instructions for the agent on how to use the WebRTC softphone client.

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2 Getting Started

This section provides information and instructions on how to get started with the WebRTC softphone.

2.1 Prerequisites

Before you can start using the softphone, make sure that you have met all the requirements mentioned in this section.

2.1.1 Required Information from Your IT Administrator

Your company's IT Administrator must provide you with the following information:

- The Web address (URL) of AudioCodes WebRTC softphone. When you browse to this address, the WebRTC Softphone Client page is displayed.
- Your login username and password for authenticating you when logging into the softphone. (This requirement depends on your company's security policy.)
- Your agent DN (phone number).
- The user and password for registering you to Genesys SIP server. (This requirement depends on your company's security policy.)

2.1.2 Connect Your Audio Devices

When there is an incoming call or you initiate an outgoing call, AudioCodes WebRTC softphone client checks availability of your audio devices:

- Audio input device (microphone) is connected to your computer
- Audio output device (speakers) is connected to your computer (only for web browsers that support this)

If any of the above audio devices are not available, the softphone displays an error message and the call is rejected.

2.1.3 Supported Web Browsers

AudioCodes WebRTC softphone client supports the following web browsers:

- Google Chrome
- Mozilla Firefox
- Safari
- Microsoft Edge (based on Chromium)

2.2 Logging into WebRTC Softphone Client

Once you have all the required information (as listed in Section Prerequisites), you can log in to the WebRTC softphone client.

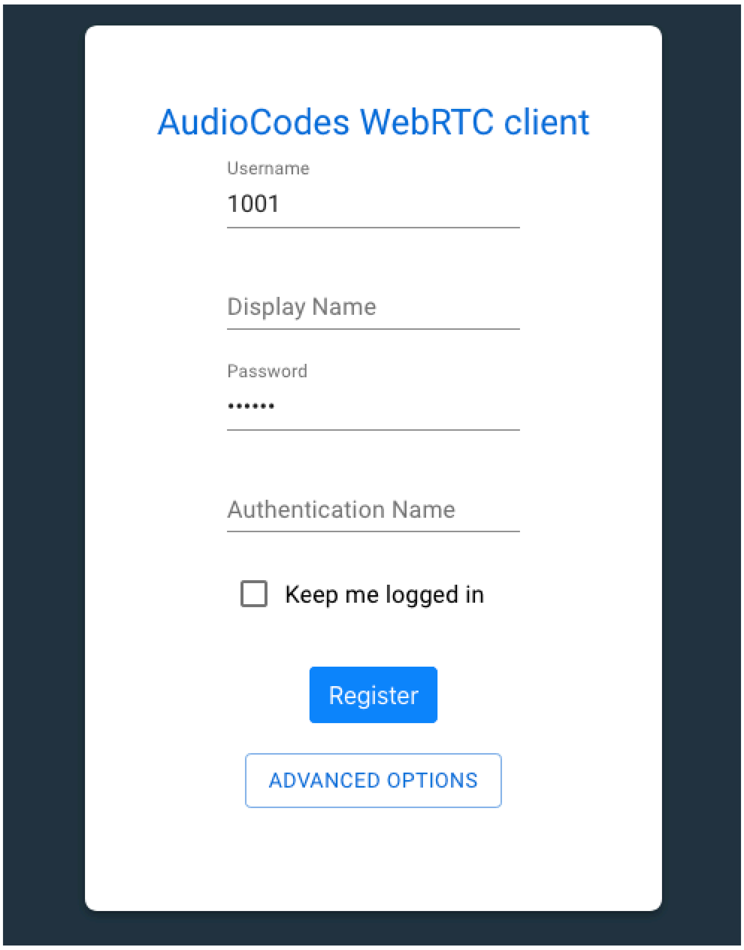
➢ **To log in to WebRTC softphone client:**

1. Using a standard Web browser, browse to the URL address of AudioCodes WebRTC softphone client.

Note: Depending on your company's security policy, you may be prompted by the Web browser for your username-password credentials for authentication to continue to the WebRTC Softphone Client page.

AudioCodes WebRTC Softphone Client login page is displayed.

Figure 2-1: WebRTC Softphone Client Page



2. In the 'Username' field, type your agent DN (phone number).
3. (Optional) In the 'Display Name' field, type the name that you want displayed on the telephone screen of the person that you call. This is referred to as *caller ID* in the Telephony world. You can change this name whenever you log in.
4. In the 'Password' and 'Authentication Name' fields, type the password and username for registering you with your company's SIP server.

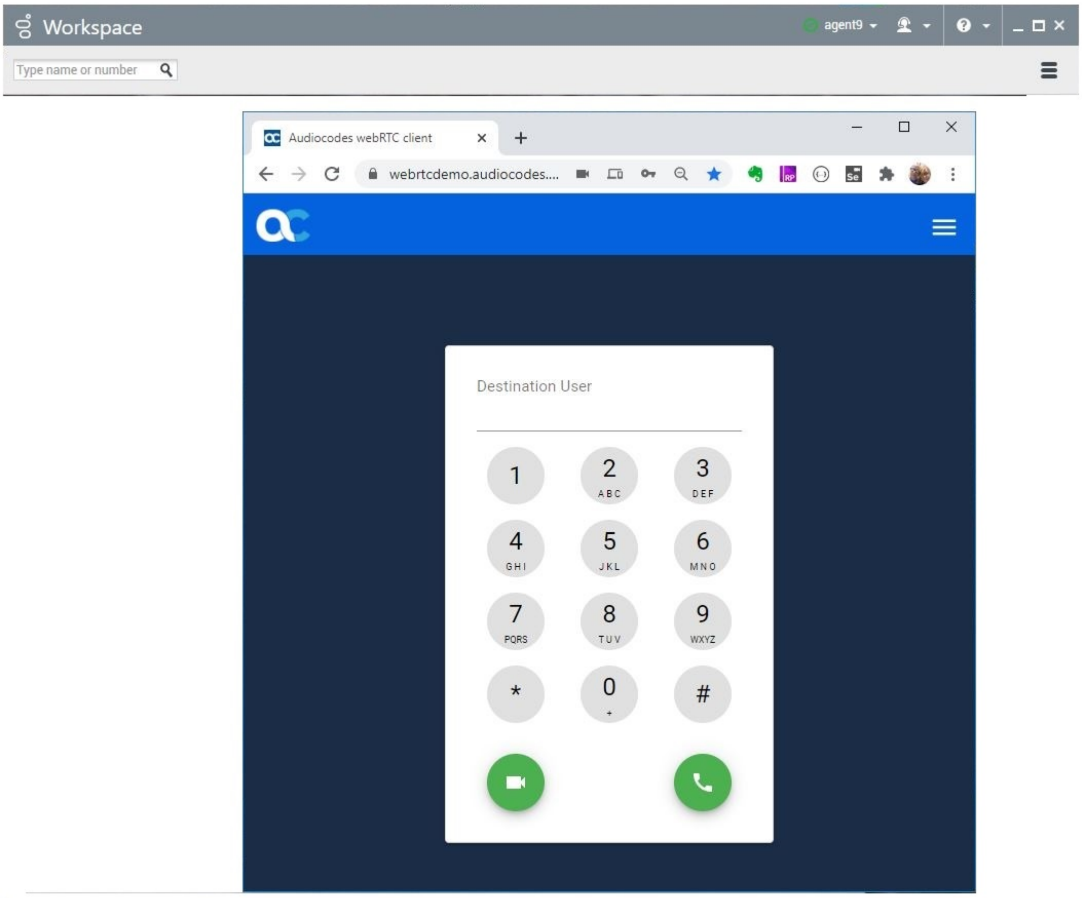
5. If you want to remain logged in with the above login information even if you refresh your web browser or close it, then select the **Keep me logged in** check box. Selecting this option stores all the above login information in your browser's cache and each time you open this page, the browser automatically attempts to register you with the SIP server using the stored attributes.

> **Note:** The settings accessed by clicking the **Advanced Options** button are typically done by your company's IT Administrator and are automatically populated with information. Do not modify them unless requested by the IT Administrator.

6. Click the **Register** button to register the Softphone client to the Genesys SIP server.

If your registration has succeeded and you are logged in, the following web-based softphone appears. You can now start making and receiving calls. You can use all the telephony functions offered by Genesys desktop application such as WDE or WWE (see Section Genesys Desktop Application) as well as other functions offered by AudioCodes WebRTC softphone client (see section AudioCodes WebRTC Softphone Client).

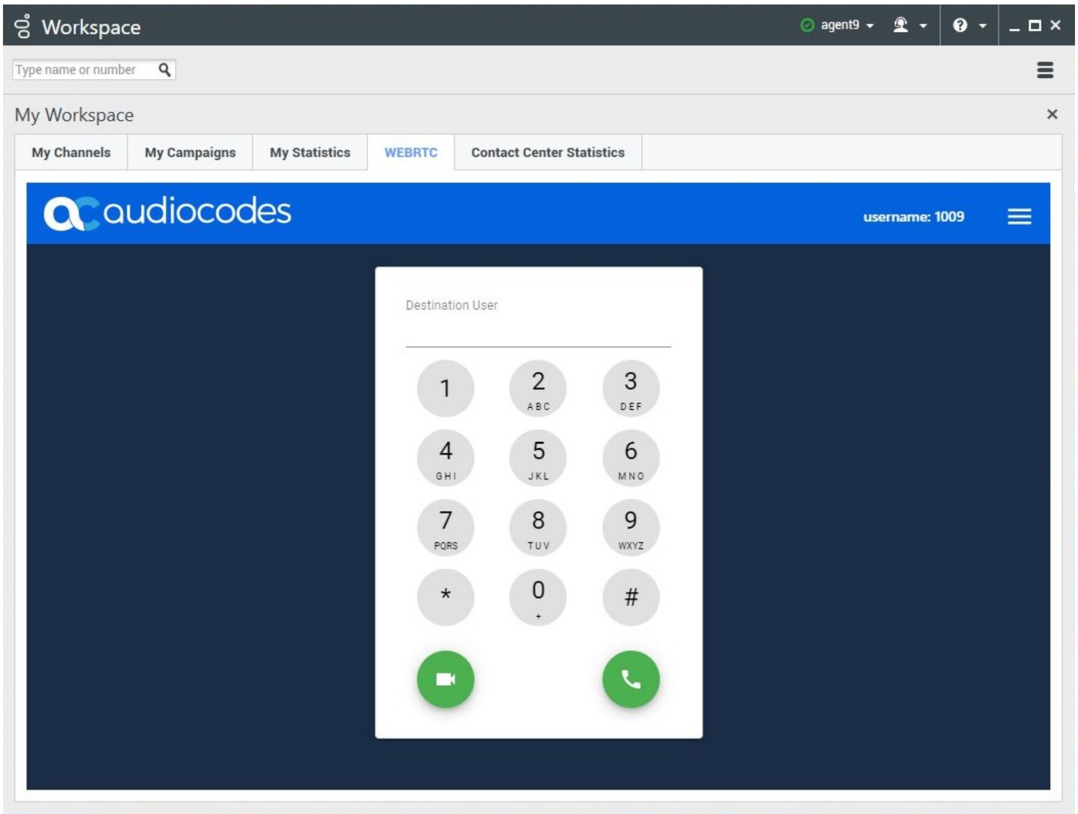
Figure 2-2: Logged into Genesys WDE and AudioCodes WebRTC Softphone Client



> **Note:** If a login error message is displayed, make sure that you have typed in the correct username, password and DN. If you still cannot log in, the cause may be a loss of network connectivity with the SIP server. In such a scenario, contact your IT Administrator for support.

If the client is embedded into the WDE, there is no need for the above procedure, logging in to WDE will log in automatically the WebRTC client:

Figure 2-3: Embedded Client in WDE



3 Softphone Features

This section lists the telephony functions offered by Genesys desktop application (WDE or WWE) and AudioCodes WebRTC softphone client.

3.1 Genesys Desktop Application

You can perform the following telephony operations using your Genesys desktop application (third-party call control) when integrated into AudioCodes WebRTC softphone client solution:

- Incoming calls when you are not in a call (*idle* state):
    - Incoming calls are answered automatically (if enabled by your IT Administrator). Your softphone doesn't ring for such calls and only a beep is heard when the call is connected (answered).
    - If automatic answer is not enabled by your IT Administrator, you can answer or reject incoming calls manually. Ringing tone is heard for these incoming calls.
- If you are in a call, a new incoming call will be displayed, and a call-waiting beeping sound is heard.
- Initiate outgoing calls.
- Place a call on hold and then resume the call.
- Disconnect an active call.
- When in a call, you can transfer it to another contact.
- When in a call, you can perform an instant conference call with another contact.
- Start a consultation with another contact and then conclude with a consultative transfer.

The following two figures show an example of an incoming call that is answered by an agent, where the calling and answered (connected) stages are indicated by both Genesis WDE and AudioCodes WebRTC softphone client:

Figure 3-1: Incoming Call Indication by Genesys WDE and AudioCodes Softphone Client

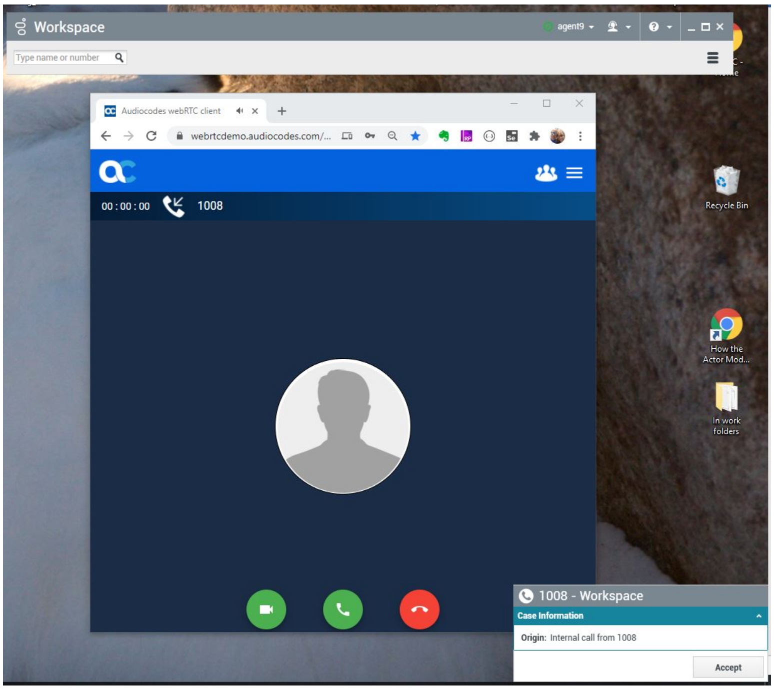
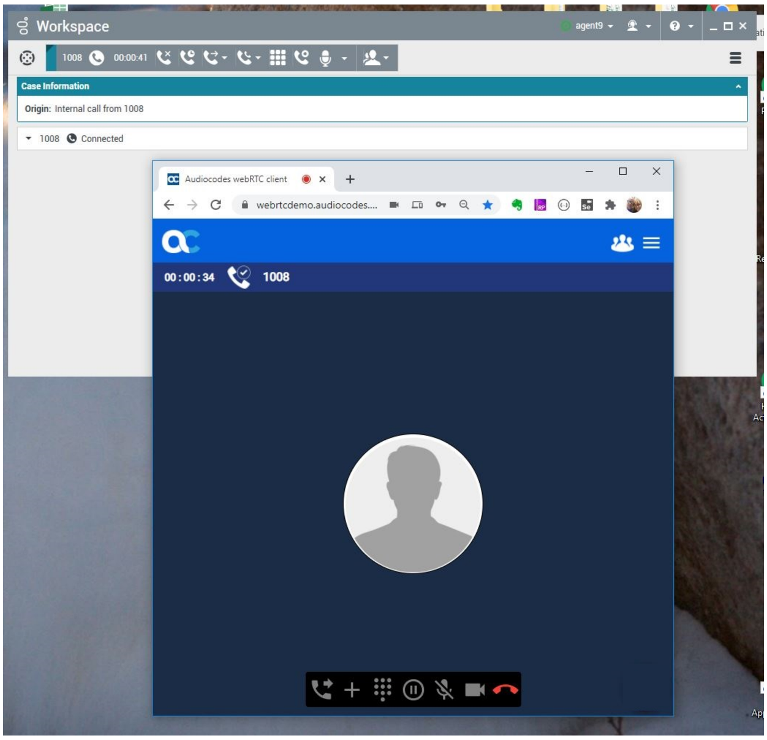


Figure 3-2: Connected Call Indication by Genesys WDE and AudioCodes Softphone Client



3.2 AudioCodes WebRTC Softphone Client

This section describes AudioCodes WebRTC softphone client.

> **Note:** Typically, you will not be using AudioCodes WebRTC softphone client for making or answering calls. Instead, you will be using Genesys desktop application for performing your various telephony operations.

3.2.1 Main Display and Functions in Idle State

When you are not in a call, the WebRTC softphone displays the dial pad, as shown in the example below:

Figure 3-3: Dial Pad of WebRTC Softphone Client

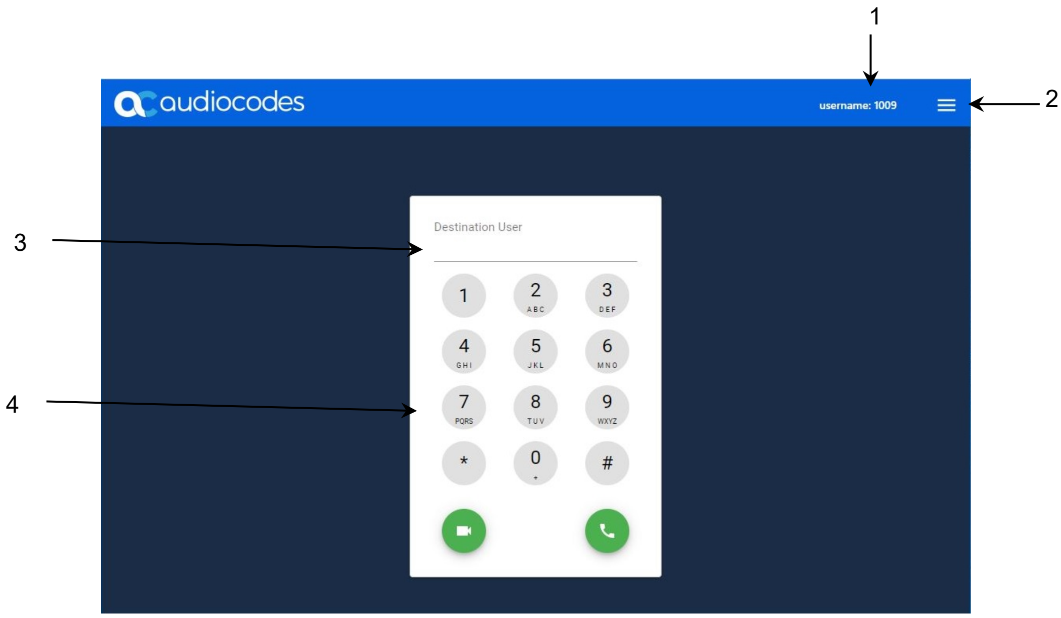


Figure 3-4: Description of Main Page (Dialpad) of WebRTC Softphone Client

Item #	Description
1	Displays your agent username (Directory Number or DN).
2	Menu button, which if clicked, displays a drop-down list with the following commands: ▪ **Settings**: Lets you view or modify the client's settings. Clicking **Register** applies the new settings and registers to the SIP server. ▪ **Unregister**: Lets you unregister the softphone client from the SIP server. When unregistered, you will not be able to make or receive any calls.
3	The 'Destination User' field displays the destination user that you want to call. For more information, see Section Making a Call (Outgoing Calls).

Item #	Description
4	Dialpad used for making calls (numbers and "*" and "#" symbols only), as described in Making a Call (Outgoing Calls). It can also be used for sending dual-tone multi-frequency signaling (DTMF) tones during a call. **Note:** The dialpad doesn't support letters.

3.2.2 Making a Call (Outgoing Calls)

➢ **To make a call:**

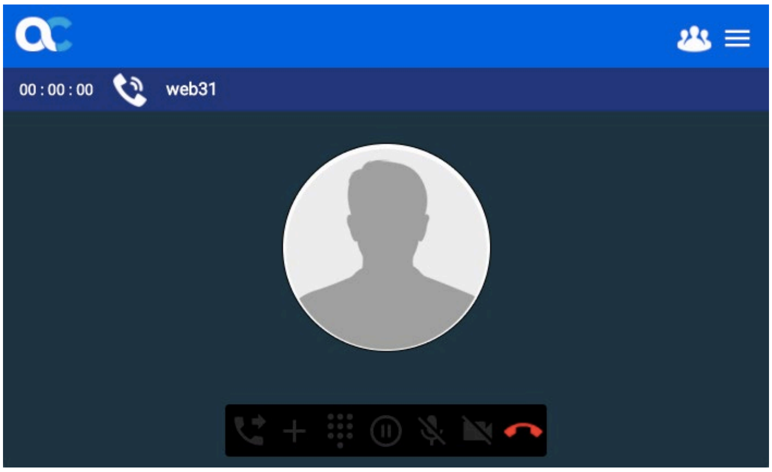
1. Dial the required user, by performing the following:

- To enter numbers or the "*" and "#" symbols, click the relevant buttons on the dialpad.
- To enter letters, use your computer's keyboard.

As you enter each alphanumerical character, it is displayed in the 'Destination User' field, located above the dial pad.

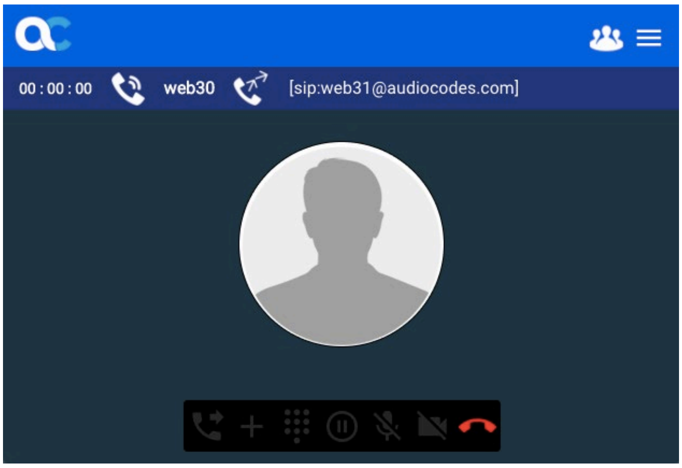
2. Click the button; ringing from the called destination (ringback tone) is heard on your audio device and the softphone displays a calling indicator with the destination user, as shown below:

Figure 3-5: Outgoing Call Display on WebRTC Softphone Client



If the call is redirected to a different destination, a call redirect progress indication is displayed with the new destination, as shown below. Once the call is established, the redirect destination is applied as the remote party.

Figure 3-6: Outgoing Call Redirection Progress on WebRTC Softphone Client



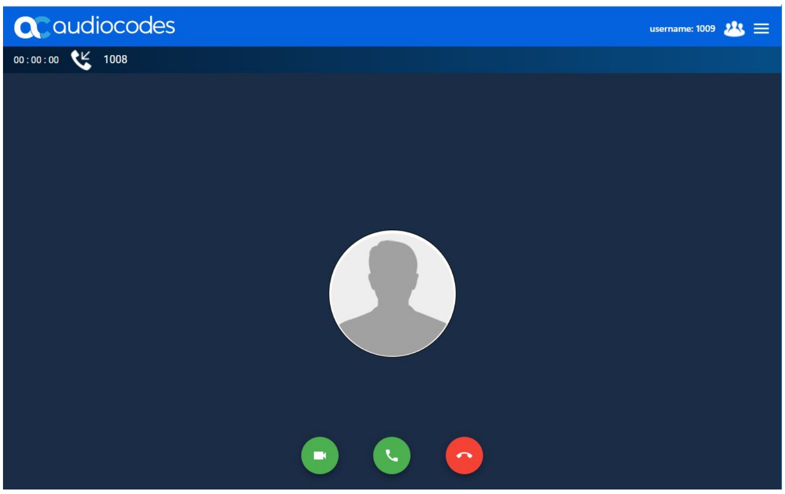
To cancel the call (before the called party has answered), click the  button. A beep sound is heard on your audio device.

If an error occurs, an error message is displayed at the bottom of the page.

3.2.3 Answering a Call (Incoming Calls)

When there is an incoming call and you are not already in another active call, the softphone displays the incoming call indication, as shown below:

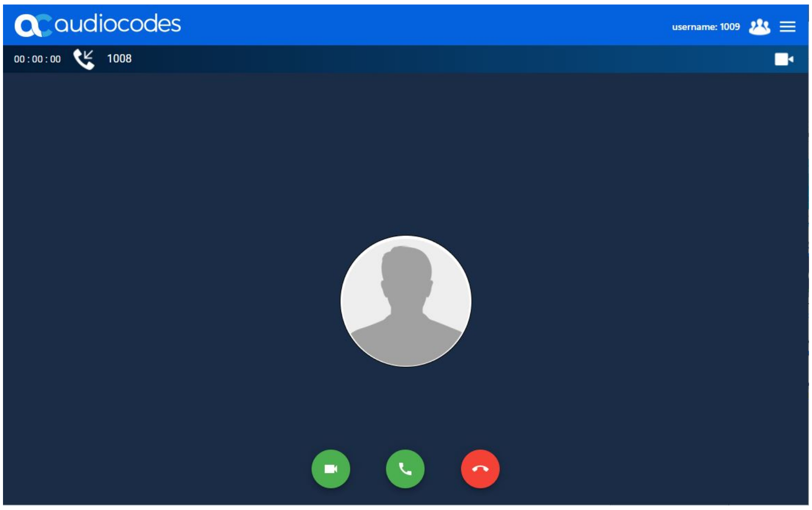
Figure 3-7: Incoming Call Display on WebRTC Softphone Client



For such calls, the softphone plays a ringing tone to your audio device.

- To accept the call, click the button.
- To reject the call, click the button.

Figure 3-8: Incoming Video Call Display on WebRTC Softphone Client

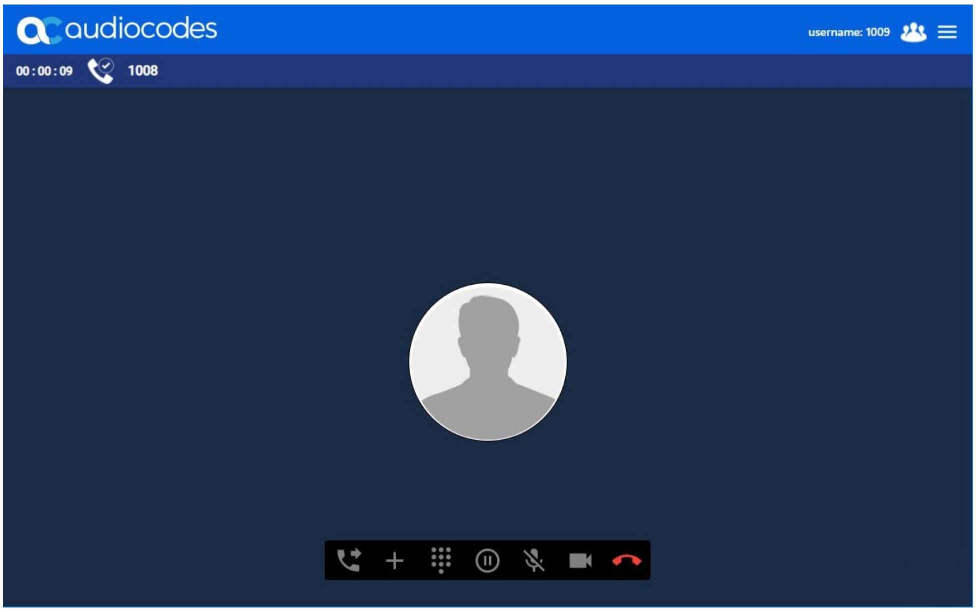


- To accept the audio call, click the button.
- To accept the video call, click the button.

3.2.4 Softphone Operations During a Call

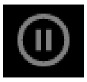
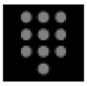
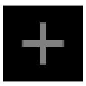
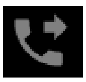
After you have answered the call, the softphone displays the connected call indicator, and buttons to perform various phone operations:

Figure 3-9: Connected (Active) Call Display on WebRTC Softphone Client



The buttons are described in the following table:

Table 3-1: Description of Buttons During Call

Button	Description
	Mutes your microphone so that the other person cannot hear you, but you can still hear that person. To unmute, click the button again.
	Enables/Disables video.
	Places the call on hold and the softphone displays the "On Hold" message. To resume conversation, click the button again.
	Displays the keypad for entering numbers, alphabetical letters, and symbols.
	Ends the call. When you end the call, a beep sound is heard and you are returned to the Dialer Keypad page. If the call is terminated with an error, an alert is shown on the page.
	Displays a keypad to add a new call.
	Displays a keypad to perform a call transfer.
	Displays the Conference menu. For more information, see Section Conference Call.

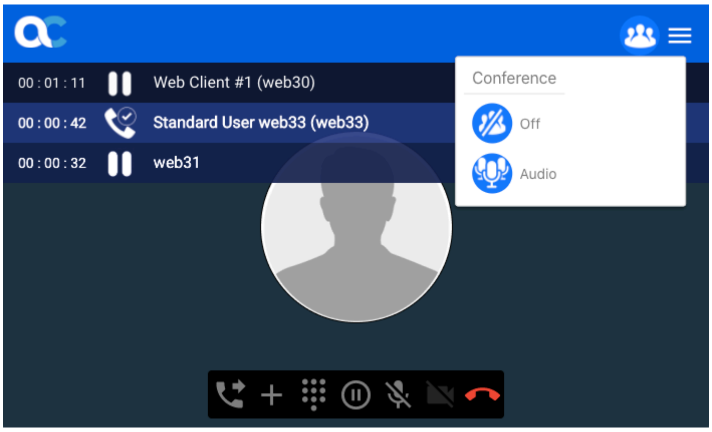
3.2.4.1 Conference Call

When there are existing calls, the user can choose to start or stop a conference call, using the Conference menu.

➢ **To make a conference call:**

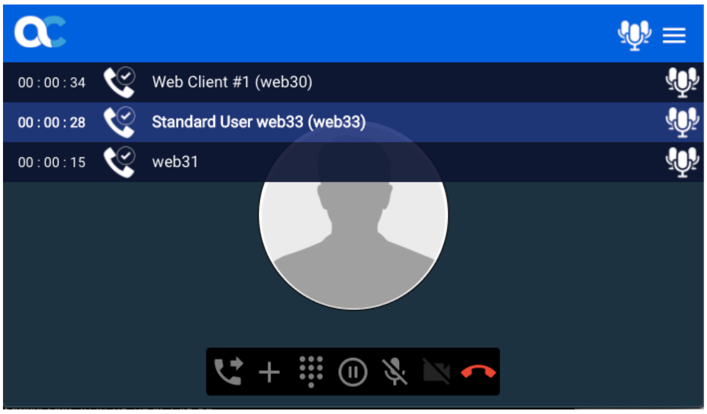
1. Open the Conference menu, and then chose one of the following:
   - **Off:** Ends the conference call. All existing calls will resume as regular calls.
   - **Audio:** Un-holds all established calls and adds them to the conference call. Any new (incoming or outgoing) calls are automatically joined to the conference call once established.

Figure 3-10: Conference Menu



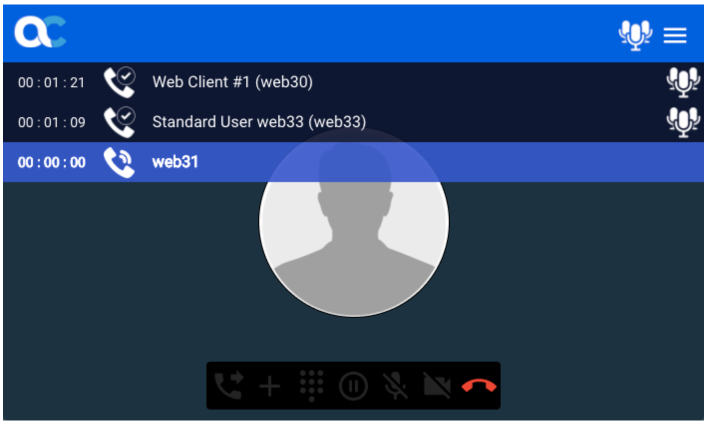
Once the calls are joined to a conference call, the conference icon indicator is displayed next to every call in the conference, and the conference button:

Figure 3-8: Calls Joined to Conference Call



When in conference mode, new calls are automatically joined once they are established.

Figure 3-9: Adding New Call to Conference



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