User's Manual

AudioCodes WebRTC Solutions for Enterprises

WebRTC Web Softphone

Version 2.6.0

WebRTC

audiocodes

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Related Documentation

Document Name
https://www.audiocodes.com/solutions-products/solutions/enterprise-voice/webrtc-connectivity
WebRTC Softphone Client Quick Guide
WebRTC Client Installation Manual
WebRTC Click-to-Call Widget Installation and Configuration Guide
WebRTC Android Client SDK API Reference Guide
WebRTC iOS Client SDK API Reference Guide
WebRTC Web Browser Client SDK API Reference Guide

Document Revision Record

LTRT	Description
14110	Initial document release for Version 1.1.0
14113	Updated for Ver. 1.2; Multiple Calls Management added
14114	Updated for Ver. 1.3; Video Calls added
14115	Updated for Ver. 1.5; OAuth 2.0 added
14130	Updated for Ver. 1.8; screen sharing added, updated default ICE servers, updated call controls
14131	Updated for Ver. 1.9; Added Azure Active Directory support for OAuth
14132	Updated for Ver. 2.0; Added Call History and Automatic Call Distribution support. Changed menu GUI to a side-bar.
14133	Updated to Ver 2.1; Updated “Server addresses” field.
14134	Updated to Ver 2.2; Updated OAuth login process. Updated error handling.
14135	Updated to Ver 2.4; new OAuth support for SIP user-password retrieval. Created section for auto-play policy handling.
14136	Added Virtual Desktop integration; Added media device selection settings; Updated ACD Not-Ready state values.

LTRT	Description
14137	Added section for the Agent Assist feature. Updated call history UI. Updated media device settings.
14138	Updated to Version 2.4.6. Updated Virtual Background Settings.
14139	Updated to Version 2.4.8. Updated the “choose file” option in the Virtual Background Settings section.
14140	Updated to Version 2.5.4. Updated the Server Configuration section with the addition of the 'Configure Automatically' check box.
14141	Updated to Version 2.6.0. Updated the following sections: ■ **General:** Most references to “web client” have been changed to “WebRTC softphone”. ■ **Feature Overview:** Added detailed support on CTI integration, Genesys Cloud sign-in and video calls on Genesys Cloud. ■ **Settings UI:** The whole chapter was re-written in accordance with the 2.6.0 new Settings features. ■ **Dialer UI:** Some UI updates have been added. ■ **Call History:** Modified to reflect the updated UI design. ■ **ACD Agent for CTI:** Moved the ACD Settings section to the Settings UI section.

Table of Contents

1 Introduction

AudioCodes provides the WebRTC Web Softphone, which utilizes the WebRTC SDK, to perform various telephony functions (listed later in this section). The Web Softphone URL is located at https://demo.webrtc.audiocodes.com/webrtc_client.

The softphone's general features and their relationship to the its user experience is described below. Throughout this document, the WebRTC Web Softphone is referred to as the Web client or WebRTC softphone.

2 Feature Overview

This section provides an overview of the WebRTC softphone features.

Calls

The Web softphone calling features are listed below:

- Supports audio and video calls
- Manages call media and RTP streams using WebRTC
- Manages multiple calls
- Starts / stops audio conferences
- Starts / stops screen sharing (for single call, multiple calls or audio conference)

> Video conferencing is currently not supported.

- Blind and attended call transfer
- Voice quality indication (see Voice Quality Indicator)
- Video filtering: Apply virtual background effects to the user's video
- Multiple SIP Authentication schemes:

  The softphone supports the following user authentication methods:

  - Standard SIP user-password login for SIP Digest authentication
  - OAuth2.0 access management providing SIP user-password for SIP Digest authentication
  - OAuth 2.0 access management providing SIP OAuth token authentication
- OAuth2.0 access management:

  The OAuth-2.0 token and SIP credentials management and implementation is performed using authentication flows with well-known identity platforms. The softphone currently supports the following platforms:

  - Keycloak
  - Azure Active Directory (Azure AD)
  - IdentityServer4
  - Genesys Cloud
- Agent Assist integration: Integrated chatbot capabilities, utilizing speech recognition to display call transcript and facilitate chatbot features.
- Web softphone user interface (UI):

- Displays a Login screen for configuring the softphone and performing SIP registration (REGISTER message).
- Displays the call history for a registered SIP user.
- Allows for saving softphone configuration, enabling later page reloads to perform automatic registration and displays the dialer.
- Allows for un-registration by using the **Unregister** button in the **Settings** menu.
- The Dialer GUI displays:
  - Dialer keypad for making audio calls.
  - Incoming call screen for accepting calls with audio or rejecting calls.
  - List of established calls, with one of them selected, or displays only the selected call with the other calls hidden.
  - Call-option buttons that allow the following operations on the selected call:

    Established Calls:
    > Mute / unmute audio
    > Hold / resume call
    > Add video to an audio call
    > Pause / resume sending captured video
    > Display DTMF keypad
    > Display keypad for starting blind or attended call transfer
    > Display keypad for making a new call
    > All call states: End call
- Play sounds for the following events:
  - Incoming call ringing tone (when it is the only existing call)
  - Incoming call-waiting tone (when other calls exist)
  - Outgoing call ring-back tone
  - Call disconnected beeping tone
  - Incoming DTMF tones (currently only via the RTP stream)
- Voice quality information is displayed when it’s available during a call, or after a call ends
- Errors are displayed as an alert to the user

- First-Party Call Control Actions:
  - REGISTER / UNREGISTER
  - Make outbound audio calls
  - Reject incoming calls
  - Answer incoming calls with audio
  - Mute or un-mute audio

- Hold, resume, transfer, toggle conference on or off
- Send DTMF tones via RTP

- Third-Party Call Control Actions (can be invoked by third-party agents):
    - SIP Alert-Info header support: When the incoming call INVITE message includes the Alert-Info header with 'info=alert-autoanswer', the softphone automatically accepts the call.
    - Hold or resume for incoming SIP NOTIFY messages:
        - A NOTIFY with the 'talk' event in early dialog triggers the softphone to accept a call (OK response to NOTIFY followed by an OK response to INVITE).
        - A NOTIFY with the 'hold' event during a call triggers the softphone to hold the call (OK response to a received NOTIFY and INVITE request for hold).
        - A NOTIFY with the 'talk' event during a held call triggers the softphone to send a re-INVITE to resume the call (OK response to a received NOTIFY and INVITE request to resume call).

Networking

- The WebRTC softphone supports SIP connection through WebSocket only (server URL must be wss://…).
- The WebRTC softphone uses WebRTC to manage network connectivity for the RTP stream, using ICE to establish and maintain RTP connection.
- For idle state with no calls, reloading a web page reconnects the WebSocket and performs SIP REGISTER again. Alternatively, reconnection is performed upon REGISTER expiration.
- Networking errors in idle state triggers periodic retries for performing SIP REGISTER through WebSocket.

Integration with Computer Telephony Integration (CTI) Systems

- The WebRTC softphone supports integration with Automatic Call Distribution for a CTI agent.
- The WebRTC softphone supports integration with Genesys Pure Connect / Pure Engage / Pure Cloud systems.
- The WebRTC softphone can be embedded within the Genesys Workspace Desktop Edition application and the Genesys Cloud web application, as well as any other CTI application which features integrated web apps / plugins with the Genesys platform.
- The WebRTC softphone supports video calls on the Genesys Cloud platform.

Integration with VDI (Virtual Desktop Infrastructure) Systems

- The WebRTC softphone supports integration with the Citrix Workspace VDI solution.

Supported Browsers

The following browsers are supported and were tested with the WebRTC softphone:

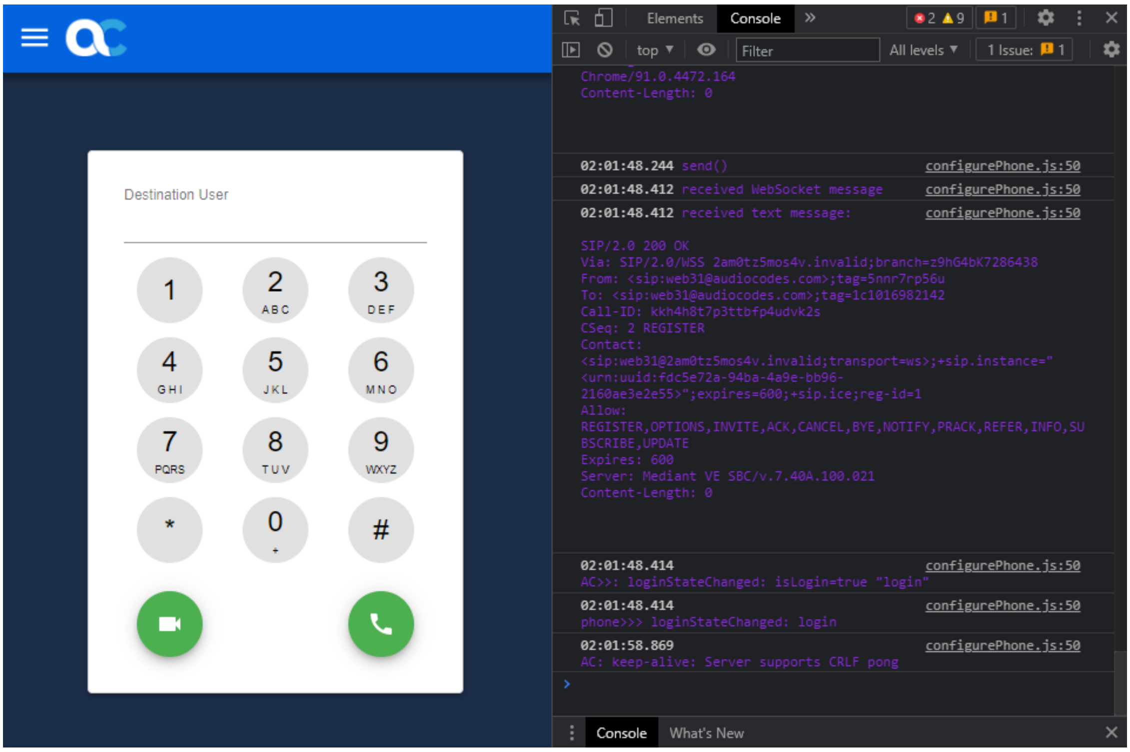
- Google Chrome
- Mozilla Firefox
- Safari

Logging

WebRTC softphone logs are forwarded to the web browser's developer console, which is accessible in all supported web browsers, using the Developer Tools panel.

To gather console logs, open the developer console from the browser's developer tools panel, and export the logs from there.

Figure 2-1: Developer Console



3 Settings UI

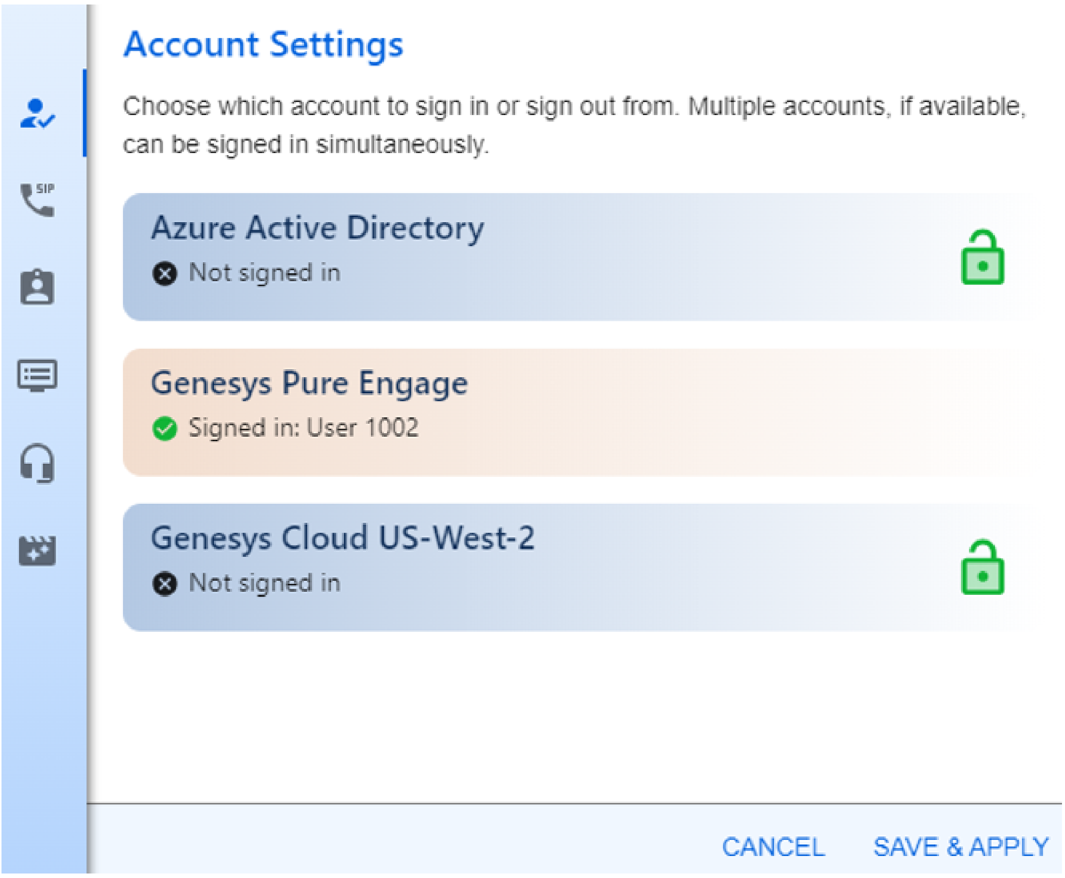
The Settings view is used to configure the WebRTC softphone. Upon launch, if the WebRTC softphone has no available configuration for making and receiving calls, the Settings view is displayed to the user.

Changes are applied only when the **Save & Apply** button is pressed, unless stated otherwise in specific setting sections.

Account Settings

The Account Settings dialog is available in the softphone application, only if the site administrator configured the available accounts to sign into. The Account Settings dialog lists possible systems for the user to sign into, each also functioning as an authentication provider. It may be possible to be signed into multiple accounts simultaneously, if the authentication providers allow for it.

Some authentication providers can already be signed in without the user interaction at all (See the example image).



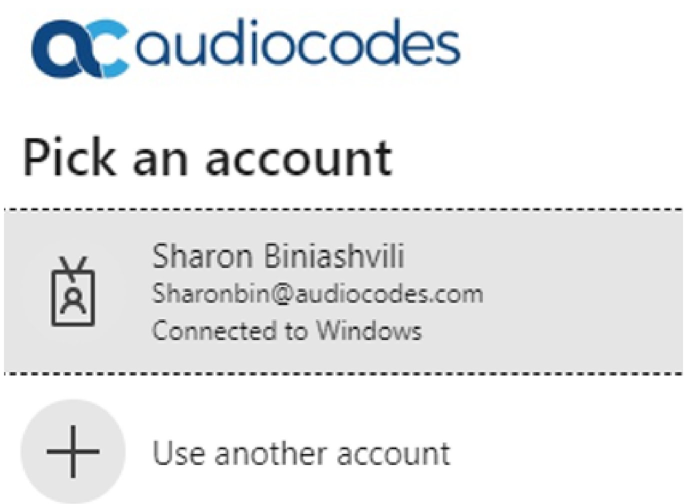
Mandatory Account Sign-In

One of the accounts can be configured to be the main account by the system administrator, in which case the user will not be able to use the softphone without logging into it first. When the user loads the softphone and is not logged in, or after they logged out, they will be redirected to the authentication provider login page, and after a successful login they will be redirected back to the softphone.

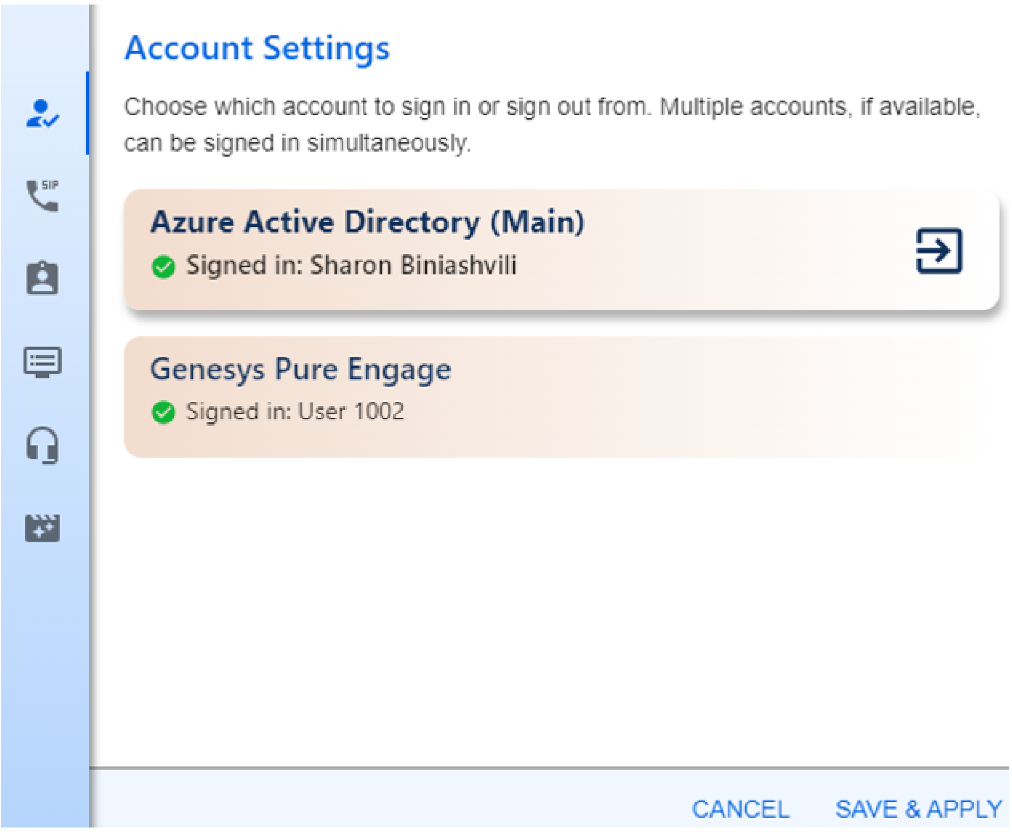
Supporting OAuth 2.0 and OpenID Connect

Signing into an authentication provider, grants the softphone access to relevant protected user resources such as SIP / ACD credentials (See SIP Settings, Agent Settings). Most supported account types provide authentication using the OpenID Connect protocol based on OAuth 2.0.

The following is an example of Redirect to the Login Page of a Main Account (Azure Active Directory):



The following is an example of Account Settings Displaying a Main Account:



For each listed account, the following buttons might be available:

Prompts/Buttons	Description
	Perform sign-in to the account
	Sign out of the account.
No buttons	This account sign-in does not involve user interaction

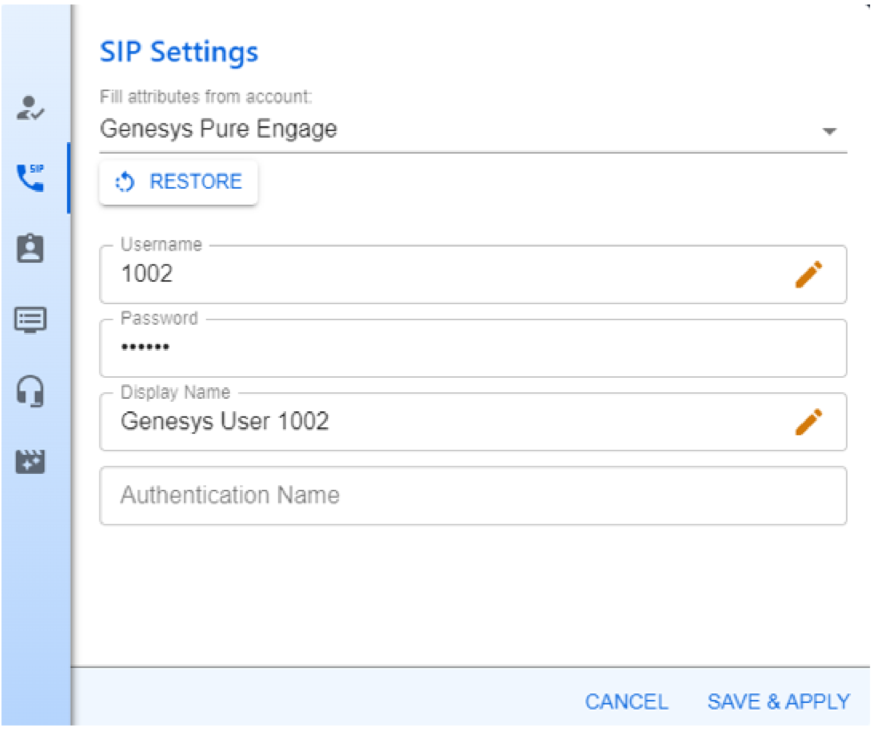
SIP Settings

For making and receiving calls, the WebRTC Softphone must first establish SIP registration using SIP credentials. SIP credentials can be obtained by the softphone in the following manners:

- Manually submitted by the user
- Provided from an account login if such is available (See "Accounts Settings").

The user can choose to manually edit SIP credentials even if they are provided via account login.

When the WebRTC softphone has obtained SIP credentials, either during startup / login or after applying new credentials in SIP Settings, it attempts to perform SIP registration automatically.



The SIP Settings dialog provides the means to configure the SIP credentials to perform SIP registration.

The Login / Settings dialog box includes the following controls, given they are allowed in the softphone configuration by the system administrator:

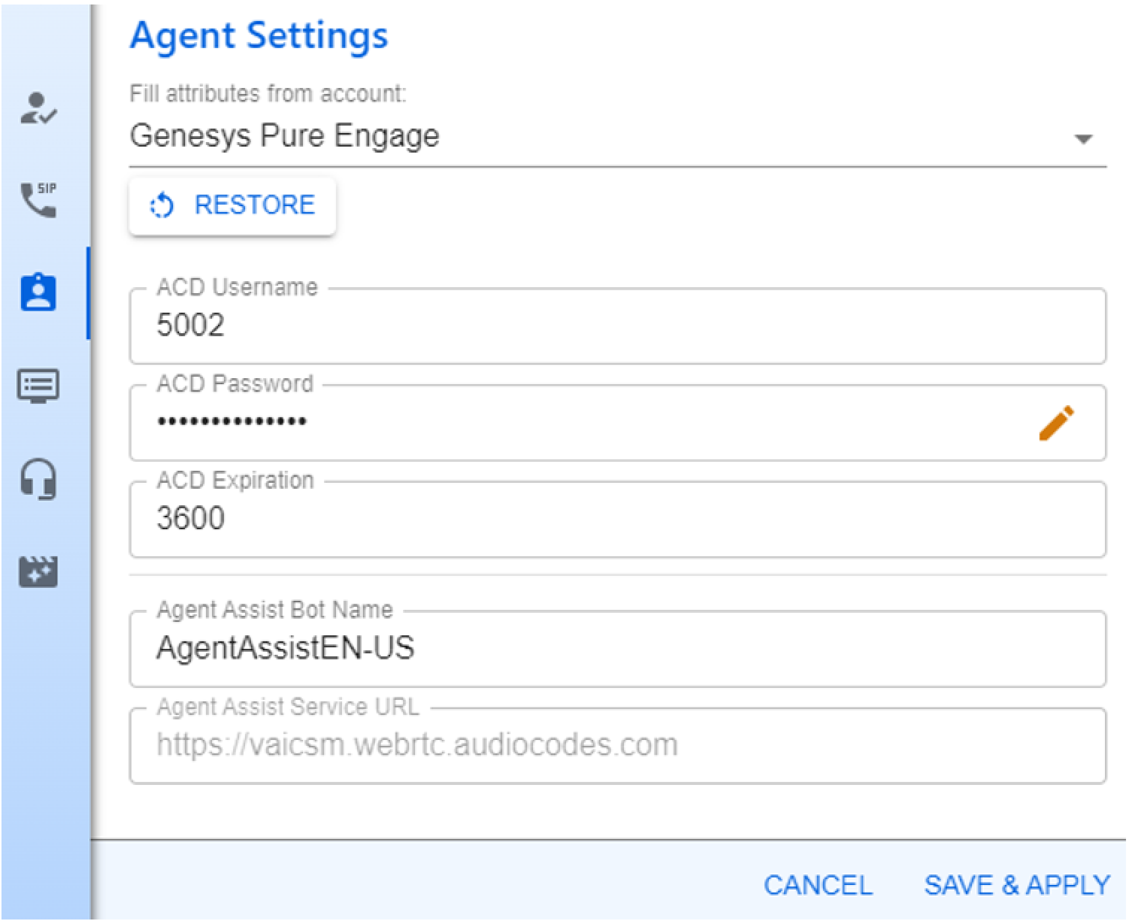
Prompts/Buttons	Description
"Fill attributes from account" drop-down selection	Choose an account from which SIP credentials are obtained. Each account listed here provides its own SIP credentials to perform SIP registration. Once selected, the fields below are filled automatically, except for the manually edited ones (see "manual edit indicator").
RESTORE	Clears all SIP credential fields and re-fills their values from the selected account. This also clears the manually edited user fields.
Manual Edit Indicator 	Indicates that this field has been edited by the user. It is not determined by the selected account, unless **RESTORE** is pressed. **NOTE:** Each manually submitted field value is stored locally within the WebRTC softphone local storage, making it non-secure.

Prompts/Buttons	Description
Username	Defines the SIP username (This does not include the SIP domain name).
Display Name (Optional)	Defines the SIP Display Name (If it exists, it is added to the user's SIP URI).
Password	Defines the SIP authentication password.
Authentication Name (Optional)	Defines the SIP Authentication Name (This field can be left empty, in which case, the softphone authenticates using the SIP username).

Agent Settings

The Agent Settings screen allows the user to control the following features, if made available by the site administrator:

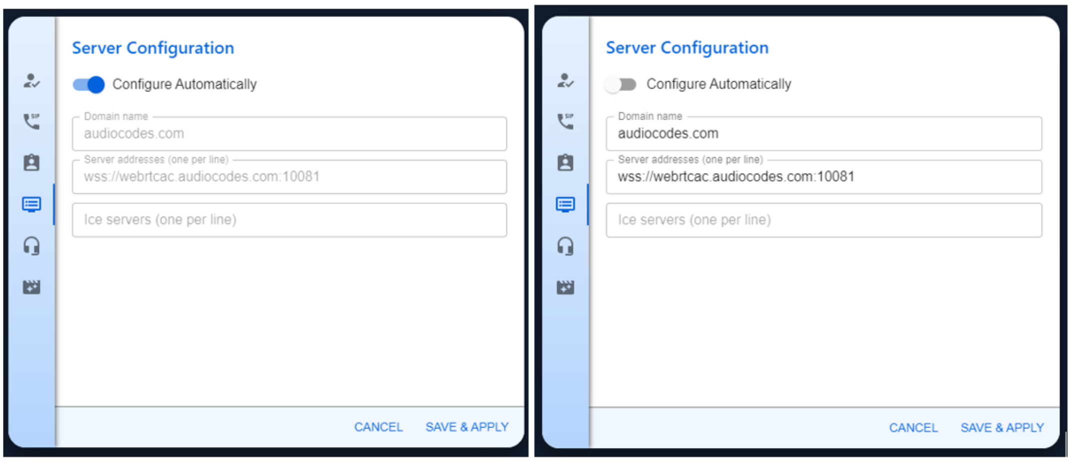
- ACD Settings (See ACD Agent for CTI)
- Agent Assist settings (See Agent Assist)



Prompts/Buttons	Description
"Fill attributes from account" drop-down selection	Choose an account from which the ACD Username and ACD Password field values are obtained. Each account listed here provides its own ACD credentials to perform ACD subscription. Once selected, the fields below are filled automatically except for the manually edited ones (see "Manual Edit Indicator").
RESTORE	Clears the ACD Username / Password fields, and re-fills their values from the selected account. This also clears the manually edited user fields.
Manual Edit Indicator	Indicates that this field has been edited by the user. It is determined by the selected account, unless **RESTORE** is pressed. **NOTE:** Each manually submitted field value is stored locally within the WebRTC softphone local storage, making it non-secure.
ACD Username	Defines the agent guest address username for the BroadWorks Hoteling ACD subscription. See ACD Agent for CTI.
ACD Password (Optional)	Defines the agent guest address password for the BroadWorks Hoteling ACD subscription. See ACD Agent for CTI.
ACD Expiration	Defines expiration interval for the ACD subscription services in seconds (Both BroadWorks Hoteling and ac-feature-events).
Agent Assist Bot Name	Defines the Agent Assist bot name that is used for every bot session that the agent activates. See Agent Assist.
Agent Assist Service URL	Defines the URL to which the softphone connects to the Agent Assist service. Configured by the site administrator. See Agent Assist.

Server Configuration

The following describes the Server configuration.



The Server Configuration dialog box includes the following server configuration fields:

Prompts/Buttons	Description
Configure Automatically	**On:** Sets the application to obtain server configuration values from the provider. Whenever the configuration is updated on the provider, it is applied automatically on the application. **Off:** Allows the user to modify sever configuration. Further updates from the provider are ignored.
Domain Name	Defines the SIP domain name, corresponding to the domain part of the user's SIP URI: <sip:user@domain>.
Server Addresses	Defines a list of server addresses, namely the SBC / SIP Proxy URLs to which the softphone attempts to establish the WebSocket connection for SIP transport. The list is prioritized from the highest to the lowest priority, so if a server's URL in the list is not responding, the next one is attempted. For multiple servers, enter each server address in a new line. Only wss://(WebSocket) URLs are

Prompts/Buttons	Description
	supported. For example: wss://webrtclab.audiocodes.com
ICE Server (Optional)	Defines an array of ICE (STUN or TURN) server URLs used by WebRTC for NAT traversal and connectivity. For more than one URL, enter each one in a new line. The default value is the Google STUN servers: stun.l.google.com:19302 stun1.l.google.com:19302

Login / Authentication Errors

On SIP registration errors, the softphone manages its SIP credentials configuration automatically, unless the user submitted manual values. If the SIP credentials are associated with a signed-in account, then the softphone attempts to refresh authentication and obtain new SIP credentials if they exist, for a retry.

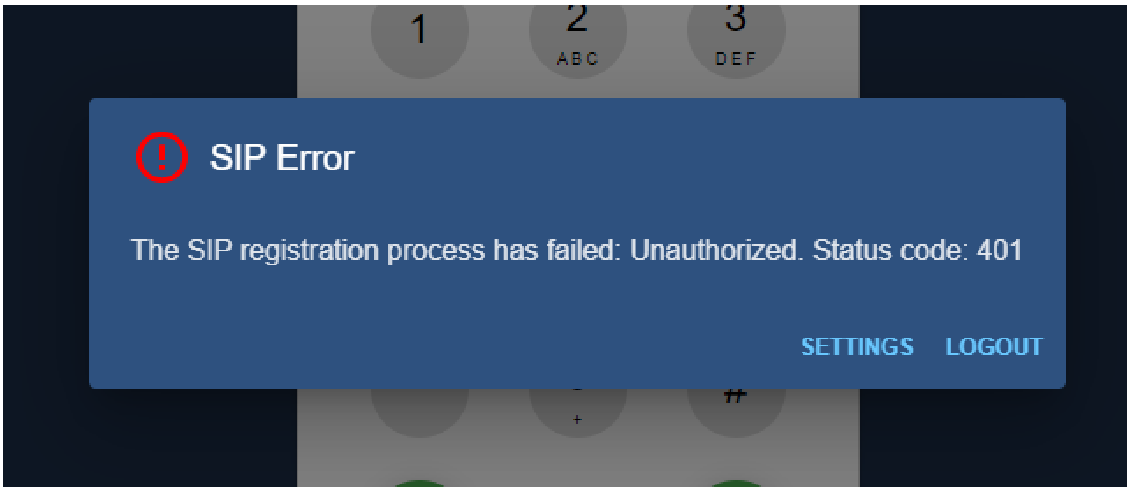
A SIP authentication error message is displayed in when:

- No signed-in account is associated with SIP credentials
- The softphone has exhausted its retry attempts to renew SIP credentials with the signed-in account
- The user has entered SIP credentials manually

If an account-related error occurs during active calls, then the softphone can be configured to postpone handling the error until all calls have terminated. Once all calls have terminated, the softphone terminates its SIP connection and displays the last error that occurred.

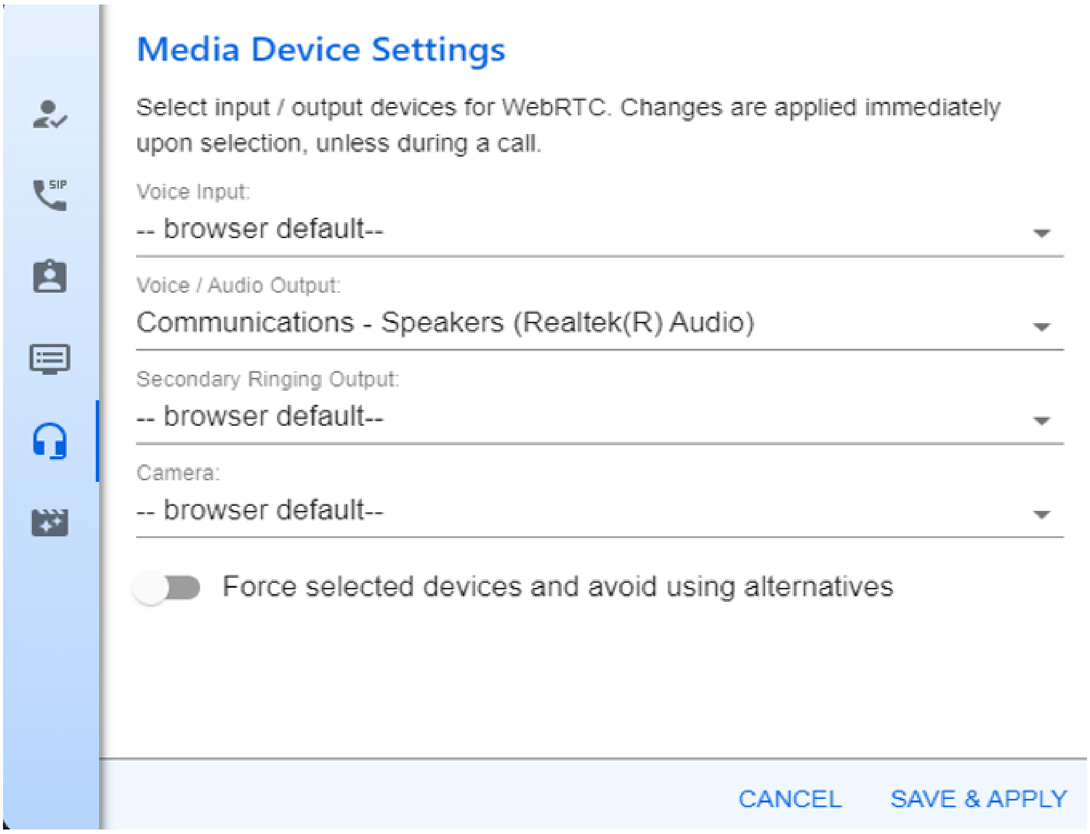
- When the page reloads after a login error has occurred, without displaying an error alert dialog, the softphone displays the login UI with the latest error message.

Figure 3-1: SIP Register Error



Media Device Settings

The softphone may be configured by the ystem administrator to allow for media device selection, which shows the dialog box below:



When the softphone is configured to allow media device selection for WebRTC input / output, the Login / Settings UI displays the following entry:

Figure 3-2: Media Device Settings

Device selection full support only exists for Google Chrome, Microsoft Edge, and the WebView2 Edge-based embedded browser. With other browsers, some or all media device selections are unavailable (most notably audio output devices).

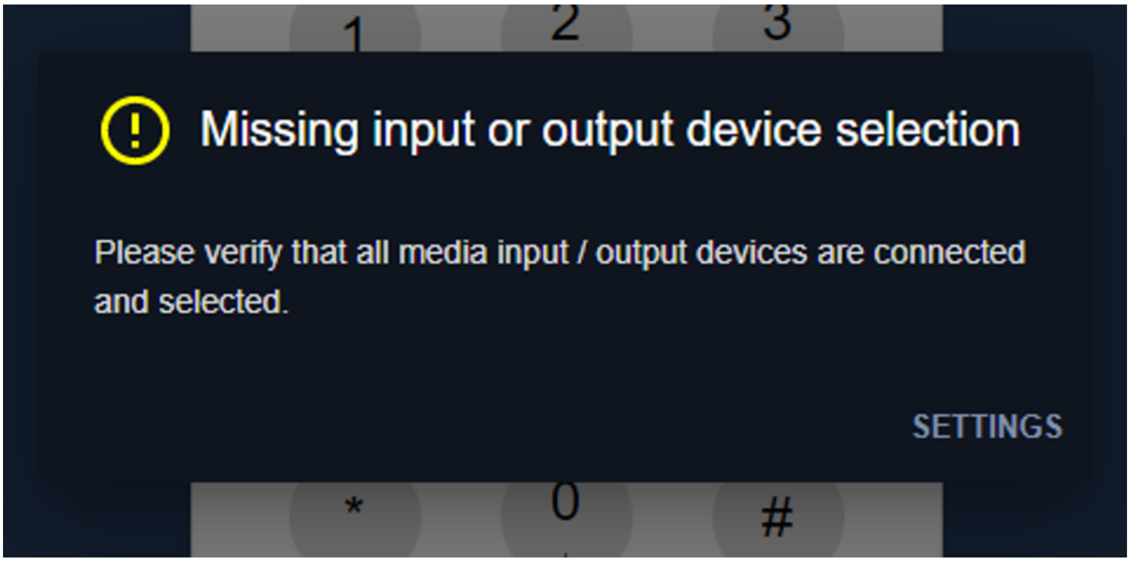
Changes take effect immediately upon selection, unless during a call. No need to press **SAVE & Apply**.

The following device selection may or may not appear, according to the site administrator configuration:

Feature	Description
Voice Input	Defines the microphone that will be used for all calls.
Voice / Audio Output	Defines the headset / speaker device from which incoming voice is heard, as well as ringing playback. This feature is currently available only on Chrome / Edge.
Secondary Ringing Output	Defines the secondary device that plays incoming call alerts, other than the voice / audio output device. For example, it allows the user to play the ringing of an incoming call to both the headset and the speaker. This feature is currently available only on Chrome / Edge.
Select Camera	Defines the camera that captures the user's video for video calls.
Force selected devices and avoid using alternatives	■ If switched off and a default device becomes unavailable, you can select alternative devices. ■ If switched on and a default device becomes unavailable, an error is generated.

Mandatory Device Selection

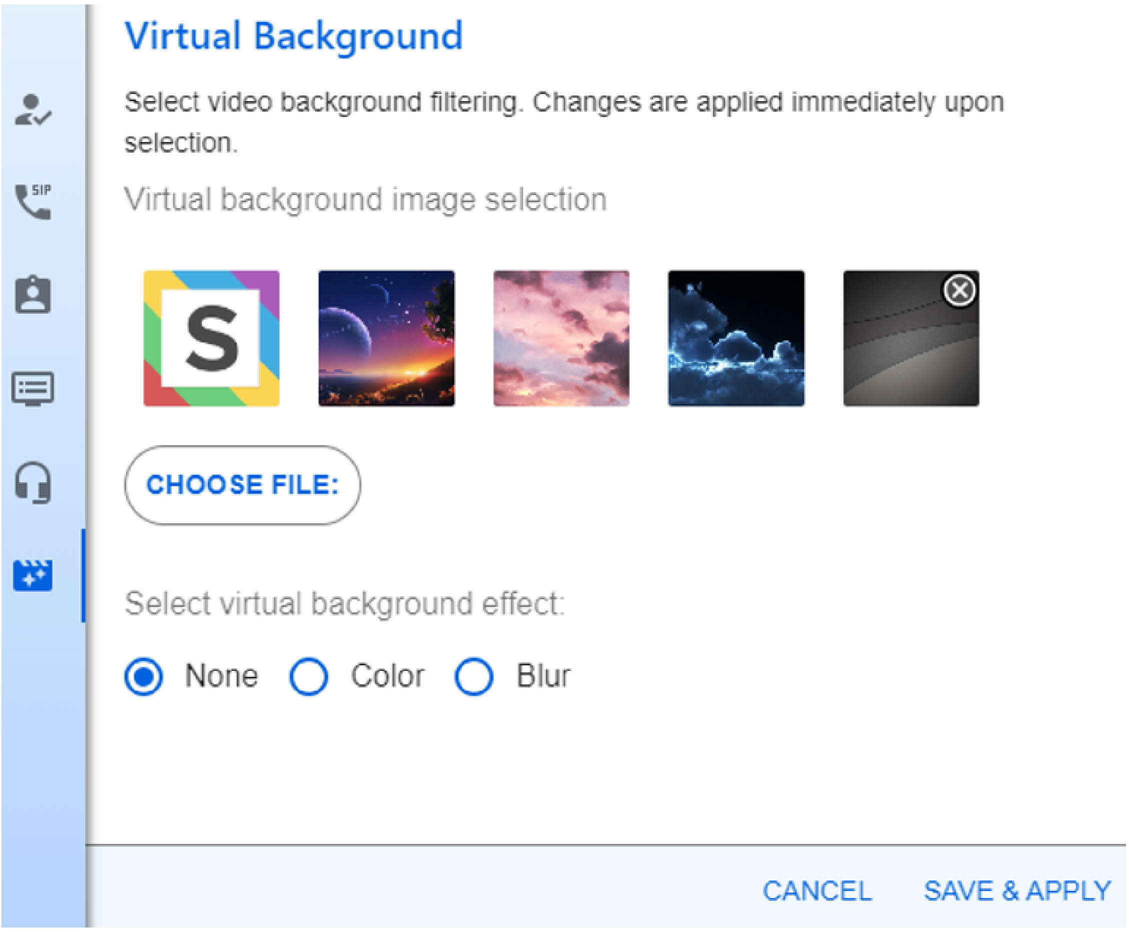
If the softphone is configured by the site administrator not to allow default device selection, and one or more devices are not selected (see the previous figure), then the softphone displays a warning on startup, to prompt the user to select an audio device.



Virtual Background Settings

The softphone may be configured by the system administrator, to allow virtual background video filtering. This feature adds user settings, that enable the user to select one of several available effects, to be applied to their video background when in video calls.

To modify virtual background filter settings, press the **Virtual Background** button on the Settings screen, to open the following dialog:



- **Select background for image mode:** The user video background is painted over with the selected image.
- **Choose File:** Select a different background image file, to add to the possible video background image options.
- **Select other modes:**
  - **None:** No filter will be applied.
  - **Color:** The user video background will be painted over with a uniform green color.
  - **Blur:** The user video background will be blurred.

- Changes take effect immediately. No need to press **Save & Apply**.
- On most browsers, If the browser tab is not in focus, then the video feed will pause because of privacy concerns and timer throttling that the browser applies.
- Currently this feature is not supported in iOS.

4 Dialer UI

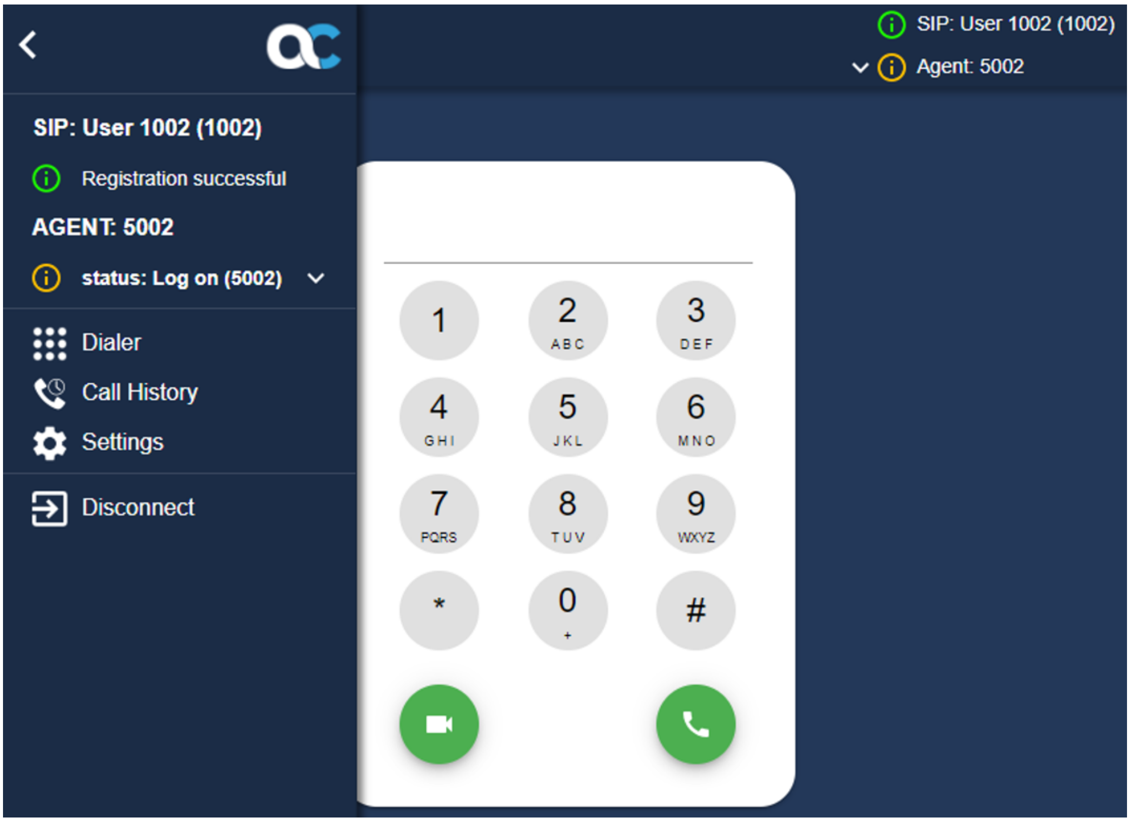
This section describes the Dialer user interface.

App Menu

The App Menu includes the following:

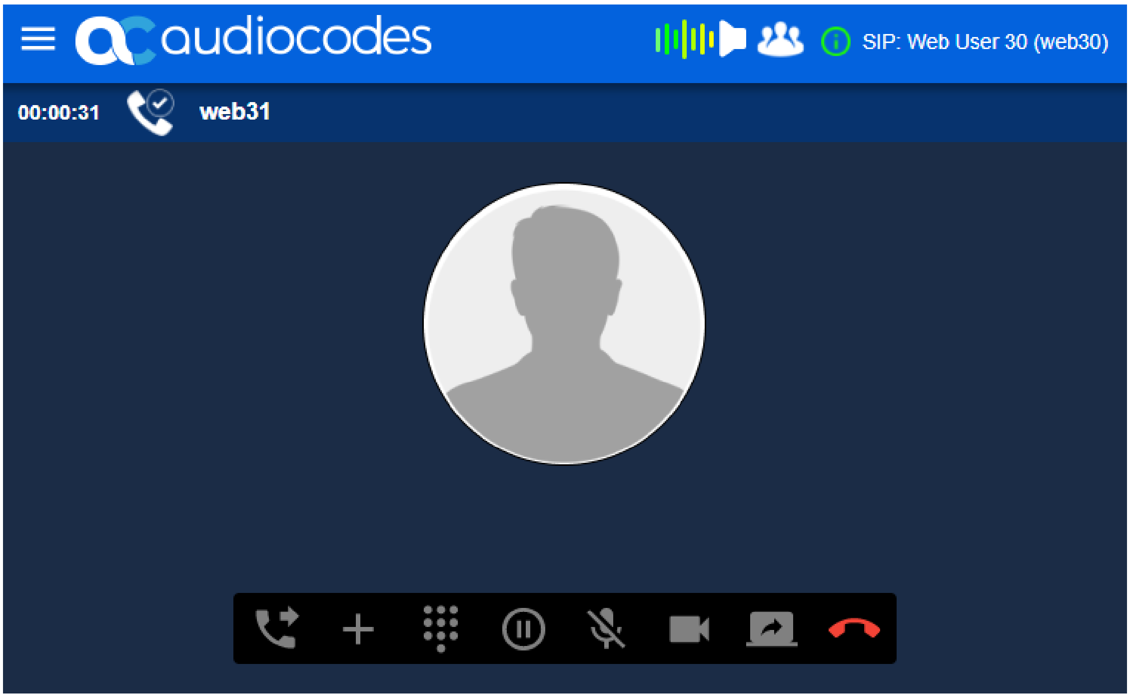
- **Top Header:** Displays the user's SIP / ACD connection status. This is hidden when the screen width is small. ACD connection status is displayed only if ACD is enabled by the site administrator.
- **App Menu pane:**
  - **SIP Status:** Displays the SIP connection status to the VoIP system.
  - **Agent Status:** Displays the ACD connection status, given that ACD support is enabled by the site administrator.
  - **Dialer:** Displays the softphone Dialer Keypad and active calls.
  - **Call History:** Displays the user's call history (See Call History UI).
  - **Settings:** Displays the Settings screen (See Login Settings UI).
  - **Disconnect / Logout:** The softphone disconnects from the SIP and ACD services and signs-out from the account that is associated with SIP the registration, if such exists.

Figure 4-1: App Menu



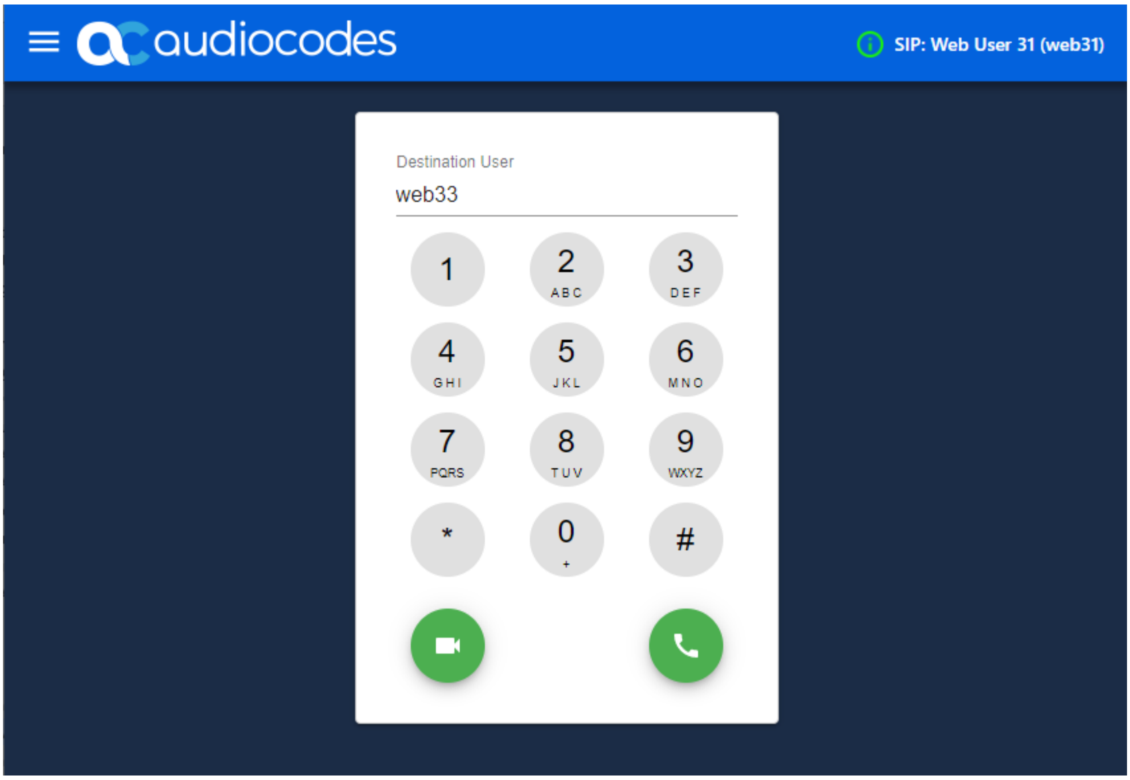
- **Conference Menu:** (This menu is hidden when there are no existing calls). For more information, see Conference Call on page 33.
- **Voice quality indicator:** If the softphone is configured to display voice quality information and the voice quality information is available, then the softphone displays an indicator for the following quality levels:
  - **Green:** High quality
  - **Orange:** Medium quality
  - **Red:** Low quality
  - **Gray:** Quality level unavailable

Figure 4-2: Voice Quality Indicator



Dialer Keypad (Main Application GUI)

Figure 4-3: Dialer UI



- **Destination User Field:** Defines the destination SIP username to call. This can be defined directly in the text field for alphanumeric text, or using the keypad buttons for numbers, '*', and '#' characters only.
- **Phone** button: Starts an audio call.
- **Video** button: Starts a video call.

Check for Available Media

When an incoming or outgoing call is initiated, the Web softphone checks for media availability. If any of the following is not available, an error alert is displayed to the user and the call is terminated:

- A connected audio input device (microphone)
- A connected audio output device (speakers) – only for browsers supporting this availability check
- WebRTC supported by the web browser

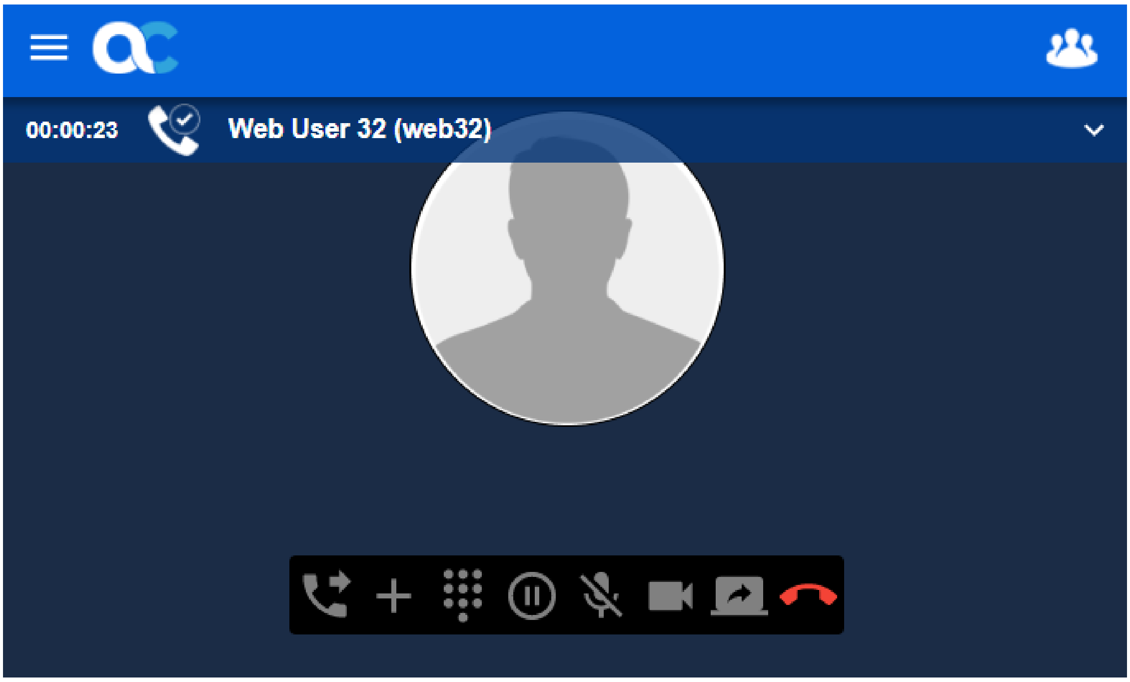
Call Information Display

When the Web softphone manages existing calls, the Dialer keypad disappears (by default), and call information is displayed as a list of all current calls and their data (or as a single item showing only the currently selected call, with a drop-down button).

Single Call Display

When a single call is displayed and there are other calls, the call item displays a drop-down button on the right-hand side, which if clicked, displays a drop-down list of all the existing calls.

Figure 4-4: Single Call Display with Multiple Existing Calls

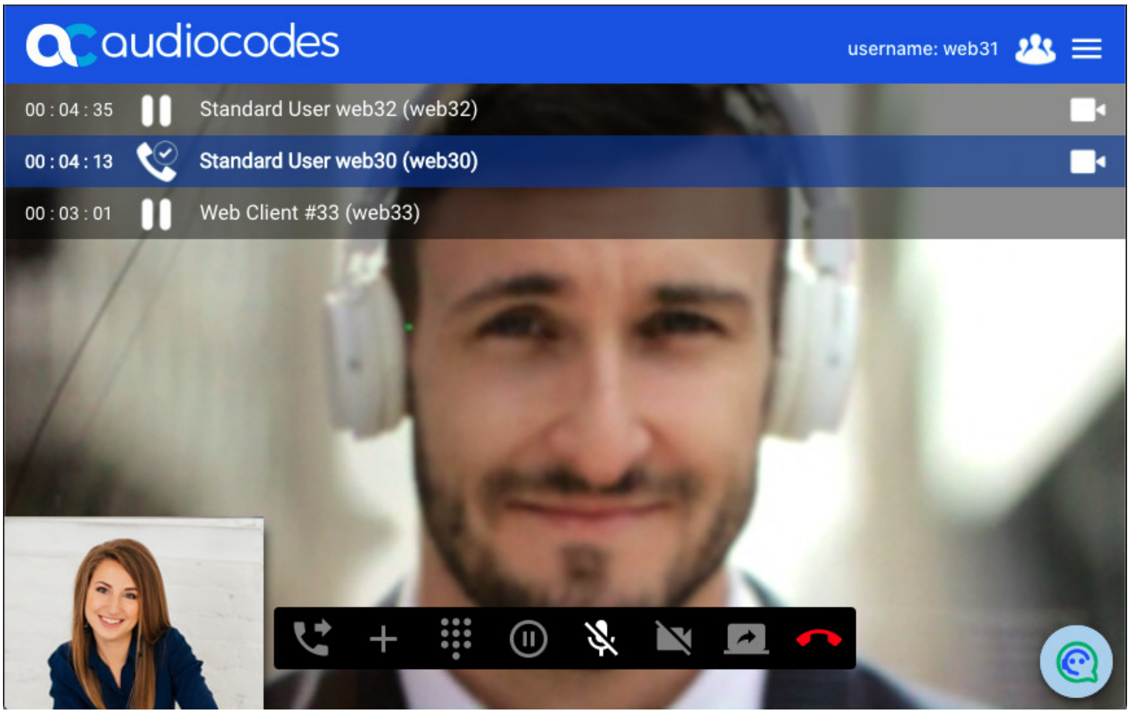


Call List Display

- When the list of calls is displayed, the user can hide it by clicking anywhere outside the area of the call list (e.g., the bottom toolbar or dialer keypad).
- The currently selected call is highlighted in the list and is the call that responds to the call-related user interface (e.g., bottom call option buttons).

- The user can select any other call in the list, and that call is then brought into focus. When hiding the list, the newly selected call item is the only one displayed.
- If the currently selected call is a video call, then its local and remote video streams are displayed, if available.

Figure 4-5: Call List Display



Selected (Focused) Call

At any given time, only one call is selected/highlighted. Only this call responds to any user interaction with the UI. This is true, regardless of the state of the existing calls. For example, when an incoming call is received and automatically highlighted, the user can select other calls and interact with them, before accepting or rejecting the incoming call.

> ⚠ When the user selects a call, this has no effect on the call state (i.e., no hold or mute operations occur implicitly). Selecting a call only highlights the call and renders that selected call as the one to respond to user interactions.

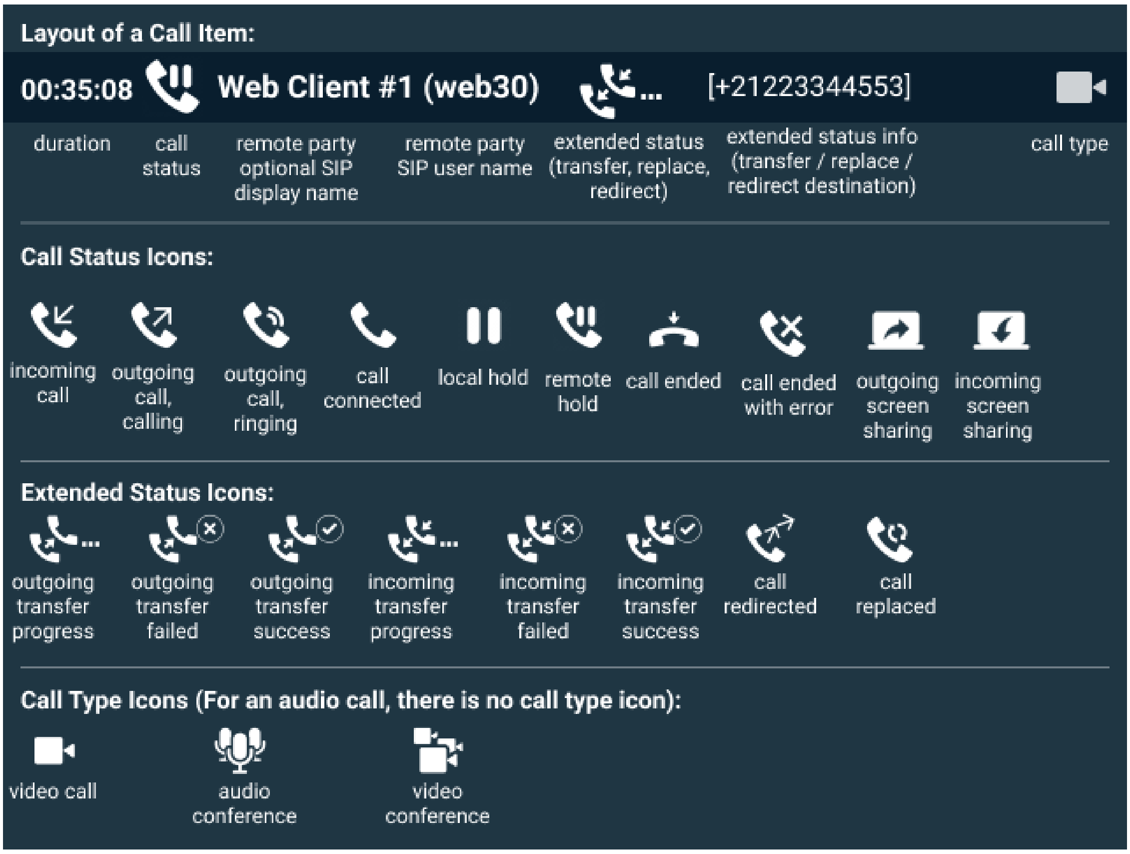
A selected call will lose highlighting, when one of the following occurs:

- The user clicks a call in the call list.
- The user makes a new outgoing call (see Initiating Outgoing Calls on the next page).
- An incoming call is received.
- The selected call is terminated. In this case, the previously selected call becomes selected, or the first one in the list.

Call Information Status – Icons and Textual Data

The Web softphone uses as few words as possible to display information regarding calls. Most of the call-state data is provided using graphic elements, while the textual data only describes language-independent information such as source and destination SIP username and display name.

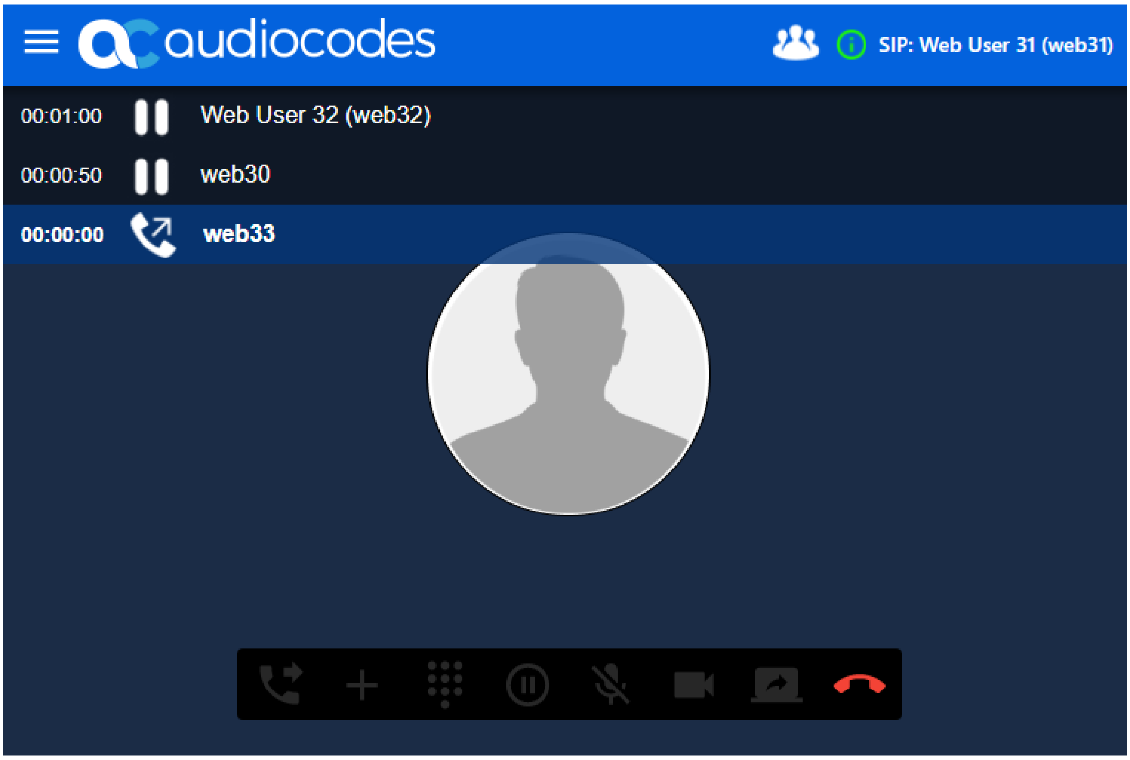
Figure 4-6: Call Information Legend



Initiating Outgoing Calls

The following figure displays the Outgoing Call screen.

Figure 4-7: Outgoing Calls



The user can initiate a new outgoing call, either from the Dialer keypad (see Dialer Keypad (Main Application GUI) on page 20), or from a currently active call (see Call Option Toolbar Buttons on page 28). Any current call is put on hold.

The Outgoing Call Progress screen is displayed when an outgoing SIP INVITE transaction is initiated for a new call, until the final response.

The outgoing Call Progress screen includes the following:

- Outgoing call indicator: Displays an outgoing call icon and the remote destination's SIP username.
- End Call Button: Cancels the outgoing call (sends a SIP CANCEL message).
- Call Responses:
  - Call Progress (SIP 18x response): For call progress response messages before the final response to the outgoing INVITE, the softphone plays a ring-back tone (locally), and then displays the Call Ringing icon.
  - Call Answered: When the remote party accepts the call (SIP 200 OK), the softphone establishes the audio stream connection using WebRTC, and the call is activated (see Call Controls and Features on page 27).
  - Call Redirected (SIP 3xx response): For SIP call redirect response, which includes a new call destination, a call redirection progress indicator is displayed with the SIP URI of the new destination in square brackets.

Internally, this initiates a new SIP dialog, by sending a SIP INVITE to the new destination.

Figure 4-8: Call Redirection

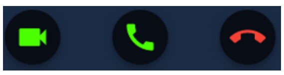
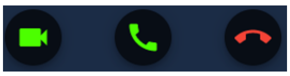
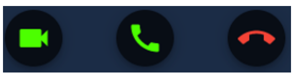
Responding to Incoming Calls

The following dialog displays an incoming call.

Figure 4-9: Incoming Calls

- When no other calls exist, an incoming call screen is shown, displaying the calling SIP username, and buttons for accepting the call with audio / video or rejecting it. The softphone plays an incoming call ringing tone.
- When other calls exist, the incoming call becomes focused, and the softphone plays the call-waiting beeping tone.

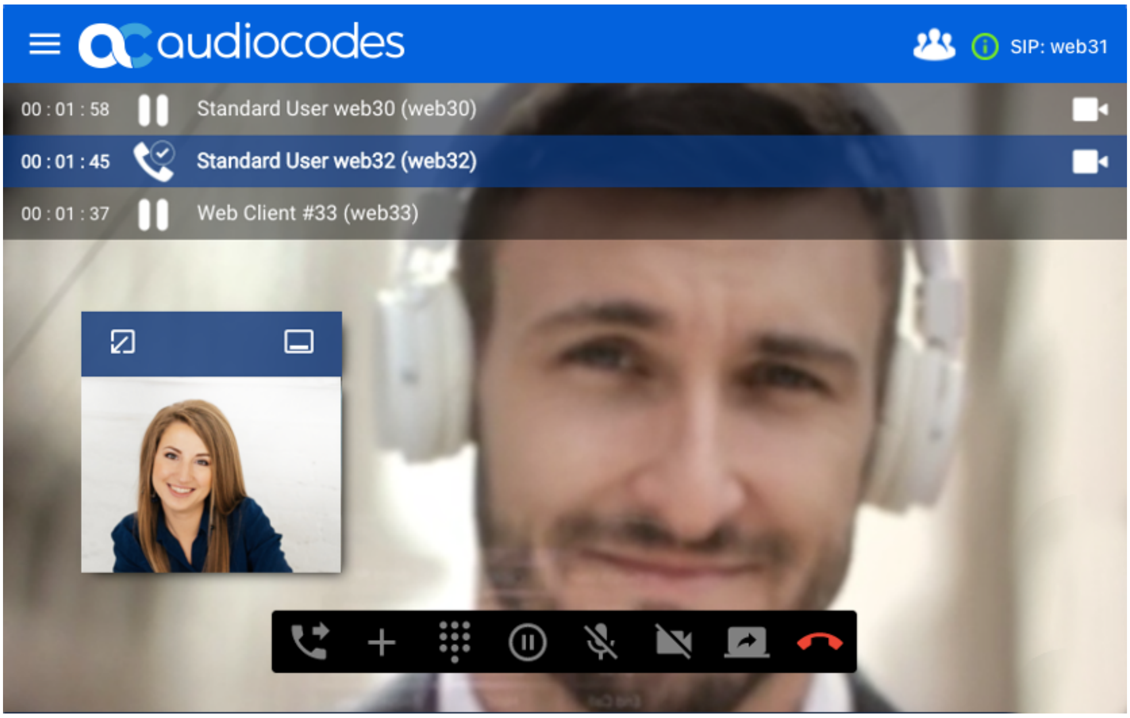
At this stage, the Web softphone can perform the following on an incoming call:

- **Answer Call Buttons:** When accepting the call, the softphone establishes the media streams using WebRTC, and the call is established (see Call Controls and Features on the next page).
  - Click the **audio** button  for incoming audio and video calls; the softphone answers with audio only. The user can click the **Add Video** button to begin video transmission.
  - Click the **video** button  for incoming video calls; the softphone answers with video. If the incoming call was an audio call, the softphone answers with audio only.
- **Reject Call Button** : When rejecting the call, it is terminated with a SIP 486 Busy Here response.

- **Auto Answer:** See 'Third-Party Call Control Actions' in the Calls on page 2 section.

Call Controls and Features

Figure 4-10: Established (Active) Call



When a call is established, by accepting an incoming call or when the remote party accepts an outgoing call, the following occurs:

- If it is an incoming call, all other established calls are put on hold.
- The established call indicator is displayed and the call duration timer starts counting.
- The Call Option buttons are enabled (lower toolbar). For more information, see Call Option Toolbar Buttons on the next page.
- Various call features described below become available for the call.

Call Controls and Information:

The call controls and information consist of the following:

- Call Options toolbar buttons: The lower bar provides the available call operations, see Call Option Toolbar Buttons on the next page.
- Call info display: See Call Information Display on page 21.
- Hide / Show call controls: When clicking on the remote video view, the call options toolbar and call info display are hidden. Moving the mouse around displays them again.
- Local video display: On video calls, the video captured by the camera is displayed in a small draggable window, with the following controls:
    - Left button: Toggle minimize

- Right button: Dock to bottom-left corner

Call Option Toolbar Buttons

The following call-control buttons are available in the call options toolbar (from left to right):

- Transfer Call  button: Displays/Hides a keypad that allows initiating a blind or attended call transfer for the selected call (see Call Transfer below).
- Add Call  button: Displays/Hides a keypad that allows initiating a new outgoing call (see Initiating Outgoing Calls on page 23).
- DTMF  button: Displays/Hides a keypad for sending DTMF tones. DTMF tones are sent within the RTP stream using WebRTC.
- Hold Call  button: Places a call on hold, or resumes a call that was placed on hold.
- Mute Audio  button: Mutes/Unmutes your audio from being heard.
- Video  button: Mutes/Unmutes your video from being seen. If the call was an audio call, then clicking this button allows the user to turn the call into a video call.
- Screen Sharing  button: Starts/Stops a screen sharing session (see Screen Sharing on page 34).
- End Call  button: Sends a SIP BYE message to terminate the call.

Call Transfer

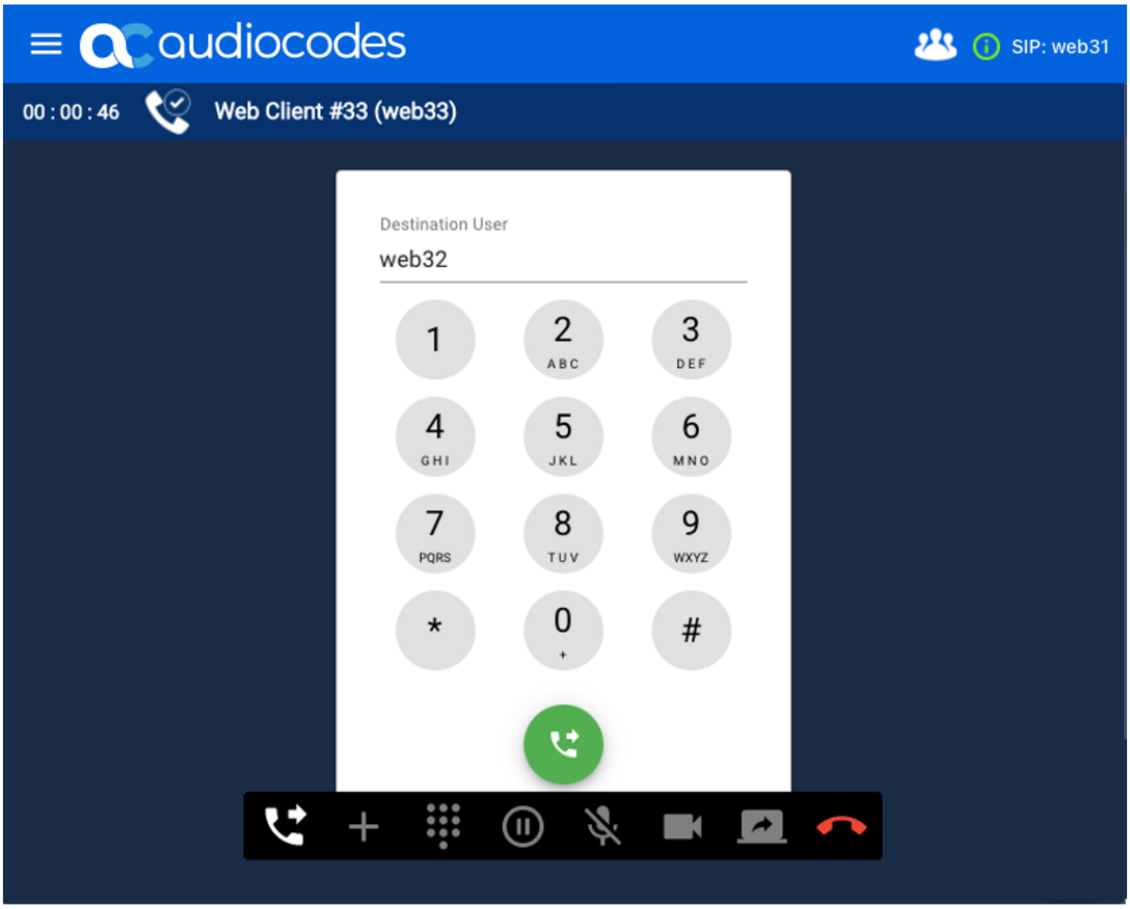
When a call is established, the user can select it and perform a call transfer to a desired destination. This destination can either be:

- A different SIP username (blind transfer)
- A different established call (attended transfer)

Initiating a Blind Transfer

The user can initiate a blind transfer, by displaying the Transfer keypad and then entering a destination user.

Figure 4-11: Initiating Blind Transfer

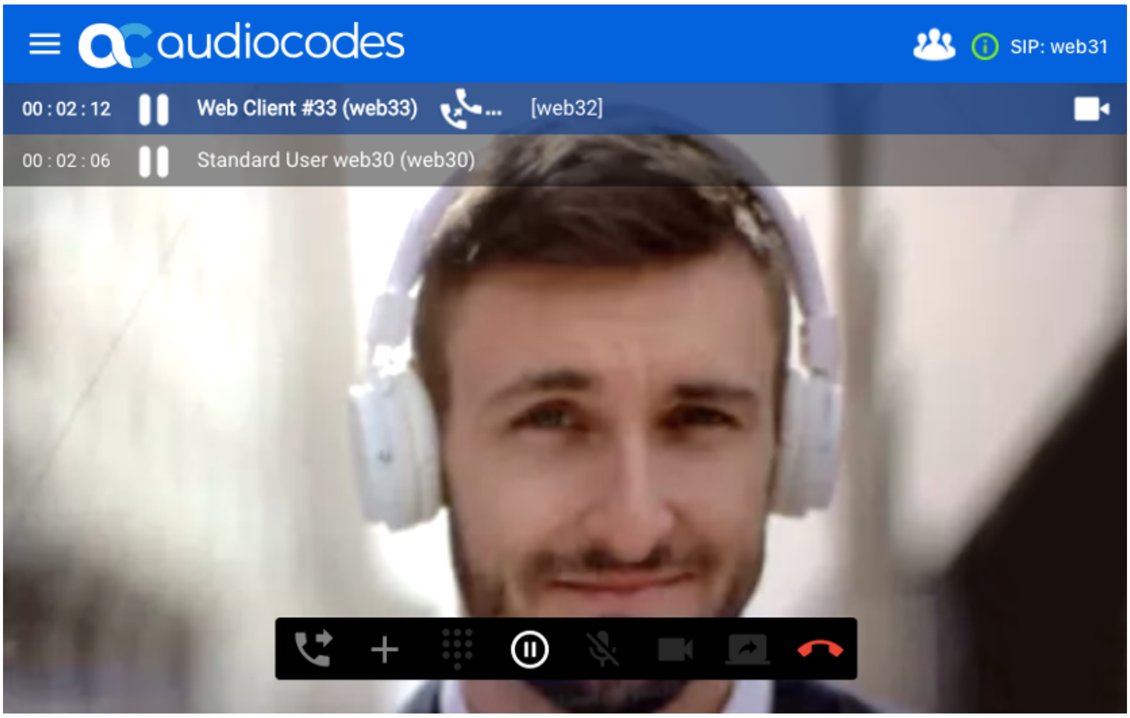


Once the user has initiated a transfer, the call is put on hold, and then the transfer progress indicator is displayed, showing the transfer destination in square brackets.

Figure 4-12: Blind Transfer Progress

A transfer can be initiated even if other non-related calls exist.

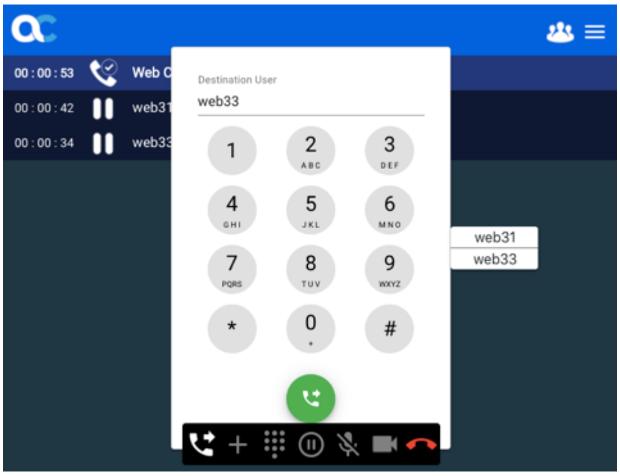
Figure 4-13: Blind Transfer Progress with Multi-call



Initiating an Attended Call Transfer

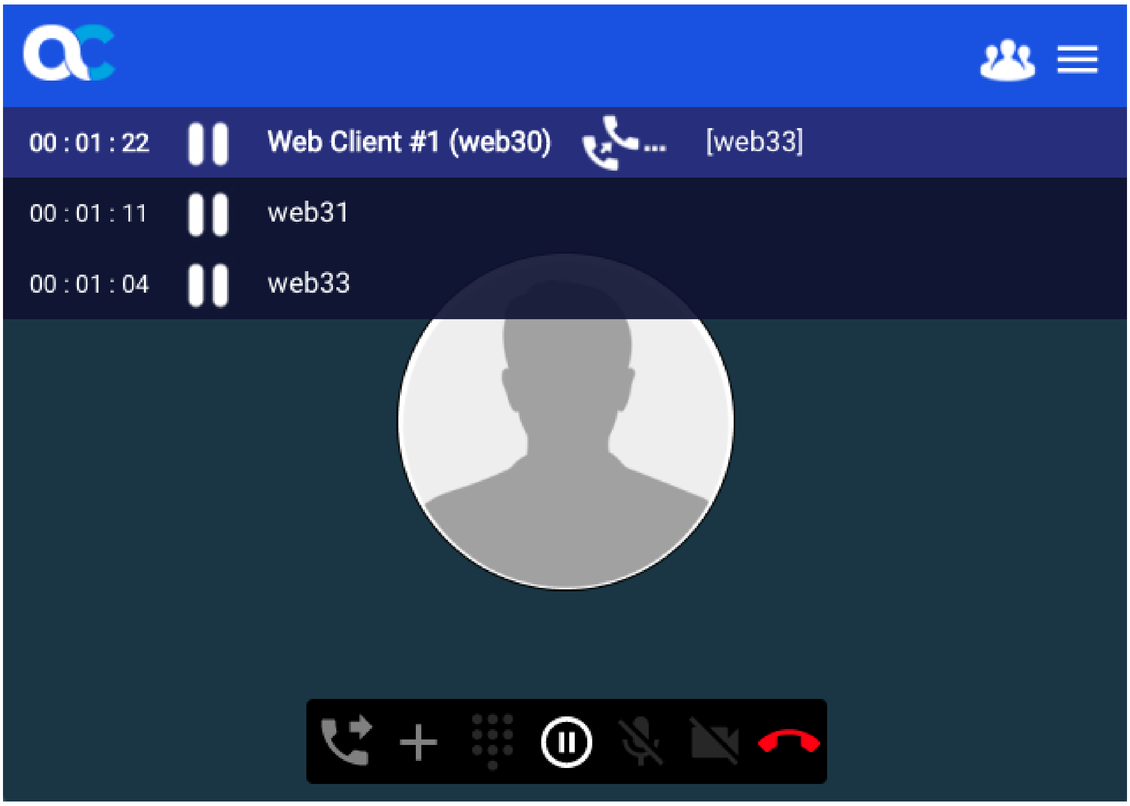
The user can initiate an attended transfer by entering the existing call's SIP username in the 'Destination User' field, or by selecting the desired destination from the menu located to the right of the transfer keypad.

Figure 4-14: Attended Transfer Keypad



The attended call transfer progress looks like the Blind Call transfer progress.

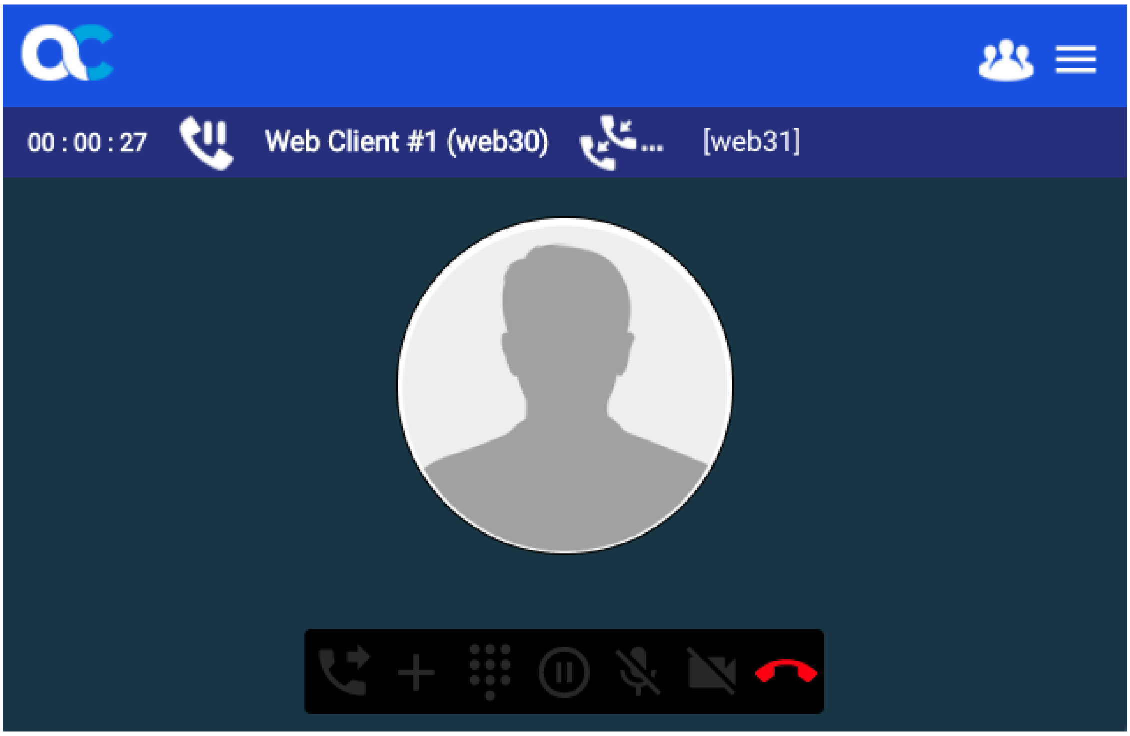
Figure 4-15: Attended Call Transfer Progress



Receiving Incoming Transfer

When the remote party transfers an existing call, it is put on remote hold, and then an incoming transfer progress indicator is displayed with the transfer destination in square brackets.

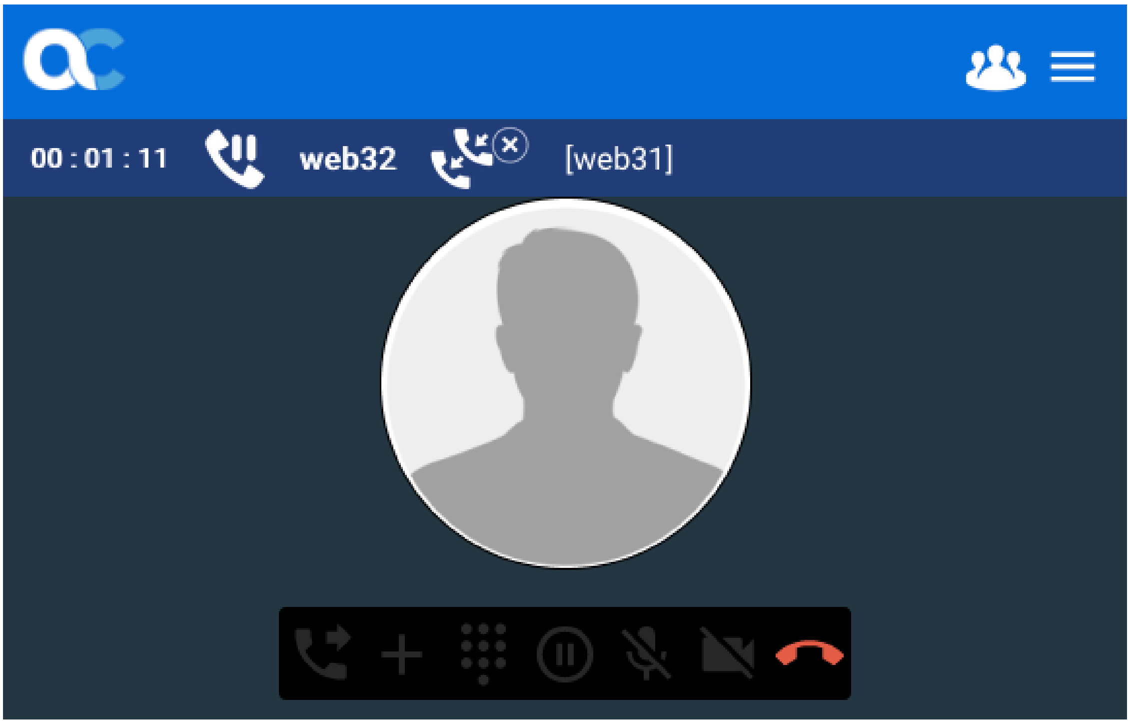
Figure 4-16: Incoming Transfer Progress



Success or Failure of Call Transfer

- Initiating a transfer:
    - Success: Successful transfer is indicated by the termination of the call, without an error message. This indicates that the transfer operation is complete.
    - Failure: Transfer failure results in un-holding the call and resuming it. A transfer failure indicator might appear for an instance.
- Receiving a transfer (remote end transfers the call):
    - Success: Successful incoming transfer is indicated by the call display text changing to represent the new remote party that the call is transferred to.
    - Failure: Incoming transfer failure is indicated by resuming the original call. A transfer failure indicator might appear for an instance.

Figure 4-17: Incoming Transfer Fail Indicator



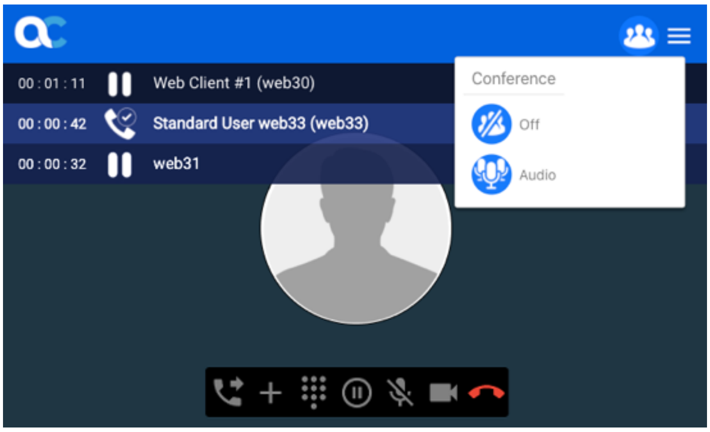
Conference Call

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

Video conferencing is currently not supported.

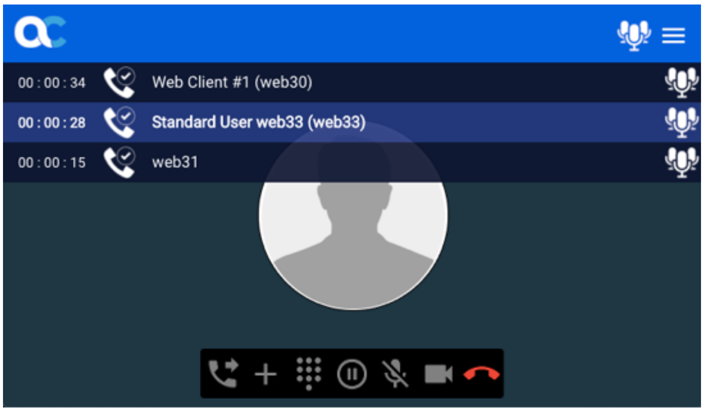
- Off Button: Ends the conference call. All existing calls resume as regular calls.
- Audio Button: Un-holds all established calls and adds them to the conference call. Any new (incoming or outgoing) calls automatically join the conference once they are established.

Figure 4-18: Conference Menu



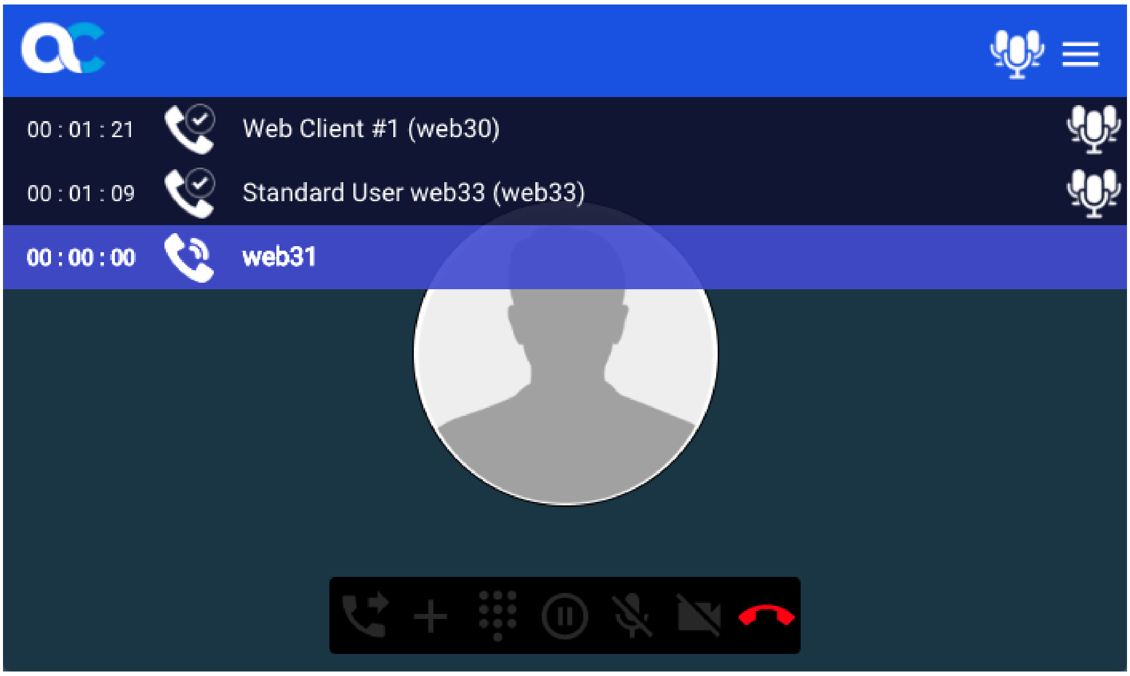
Once the calls are joined in a conference, the conference type indicator is displayed next to every call in the conference, and the conference button icon:

Figure 4-19: Conference Joined



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

Figure 4-20: Adding New Call to Conference

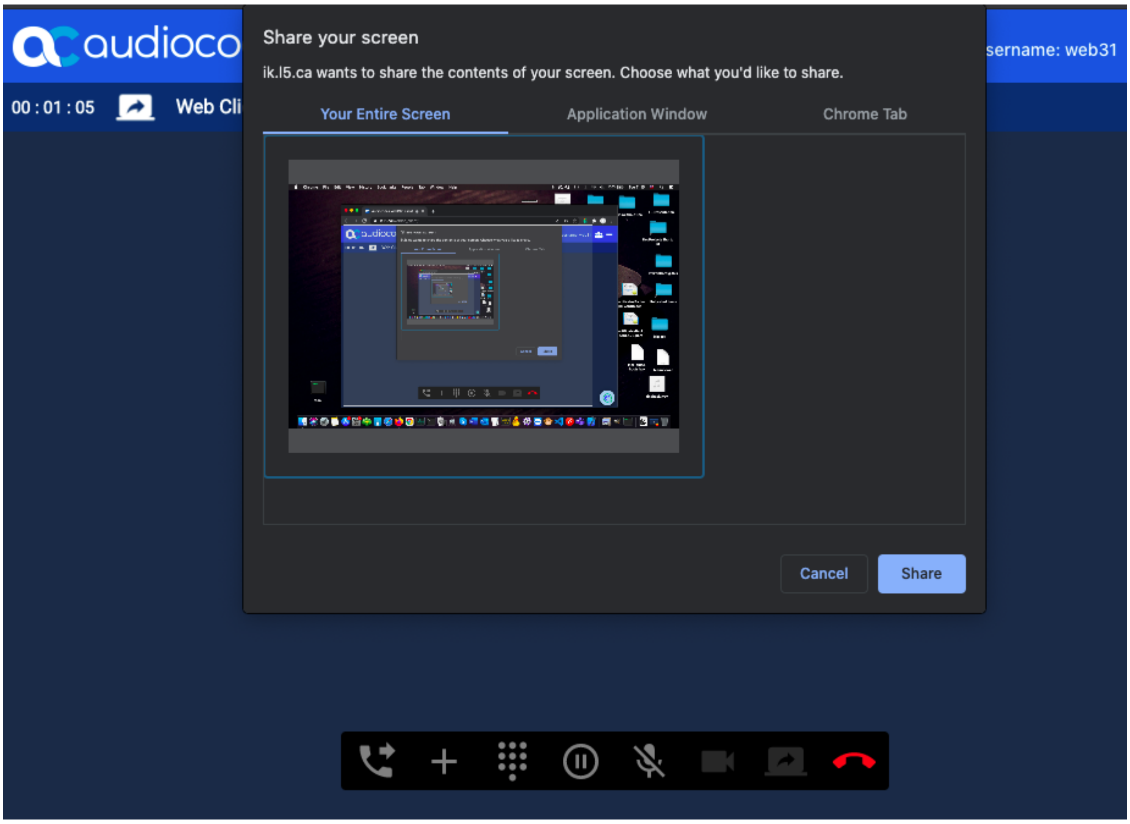


Screen Sharing

Initiator Side

The screen sharing toggle button allows the user to start / stop a screen sharing session, in which the user can share a screen capture of either an entire screen, an application window, or a browser tab. The way of selecting the source of the screen capture varies from browser to browser.

Figure 4-21: Selecting a Screen Capture Source with Google Chrome



Once the user has approved of the selected screen capture source, the call media is re-negotiated to include the screen capture media, and the call status icon changes to outgoing screen sharing.

The browser displays a floating indicator, notifying the user that a screen capture session is being performed by the web softphone.

If the call has outgoing video captured from the camera, then it is replaced with the screen capture media, so that the local video is no longer sent. Once the screen capture has finished, the call resumes sending camera video to the remote party.

Figure 4-22: Screen Sharing is On - Screen Capture Sent to Remote Party

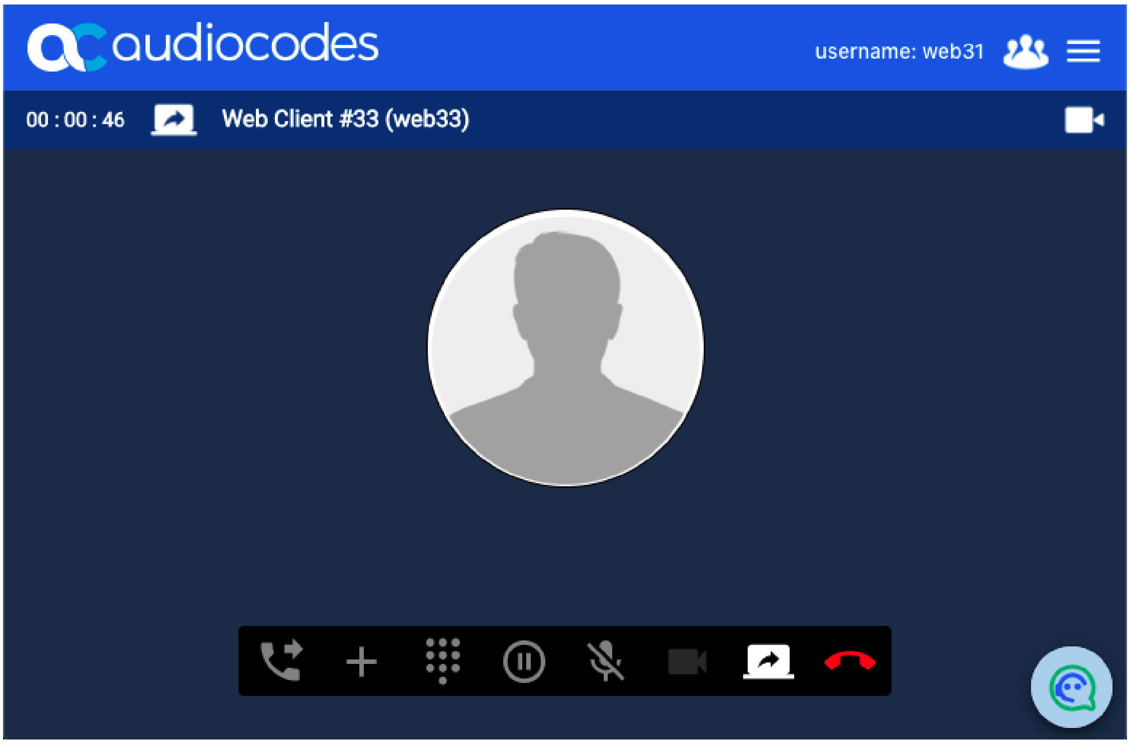


Figure 4-23: Screen Sharing Indicator Provided by Browser

The screen sharing session can be stopped either by clicking the screen sharing toggle button, or by stopping it via the browser's indicator.

- When the call has terminated, the screen sharing session automatically ends.
- If there are currently multiple calls in place, the user can initiate different screen sharing sessions for different calls at the same time.

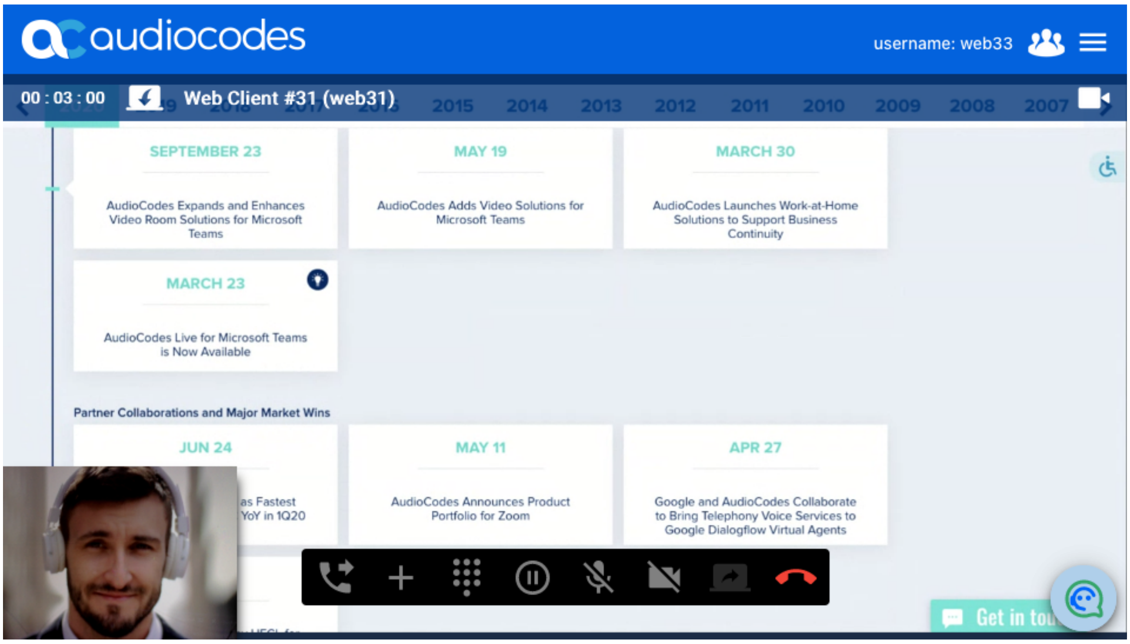
Receiving Side

The call that is receiving screen sharing, is receiving it as an incoming video. Therefore, the call is marked as a video call and the incoming screen sharing status icon appears.

The softphone can be configured by the system administrator, who decides whether to allow the user to share the screen while the other party performs screen sharing.

For regular incoming video from the remote party's camera, the video is displayed in a manner which covers the entire softphone window. However, for incoming screen sharing, the video fits inside the window rather than covers it, so that no portion of the captured screen is lost.

Figure 4-24: Call Status Icon Indicating Incoming Screen Sharing



If the call has local video, then the user can drag the local video view around or minimize it, so that it doesn't interfere with the incoming screen sharing.

Figure 4-25: Local Video Dragged Around

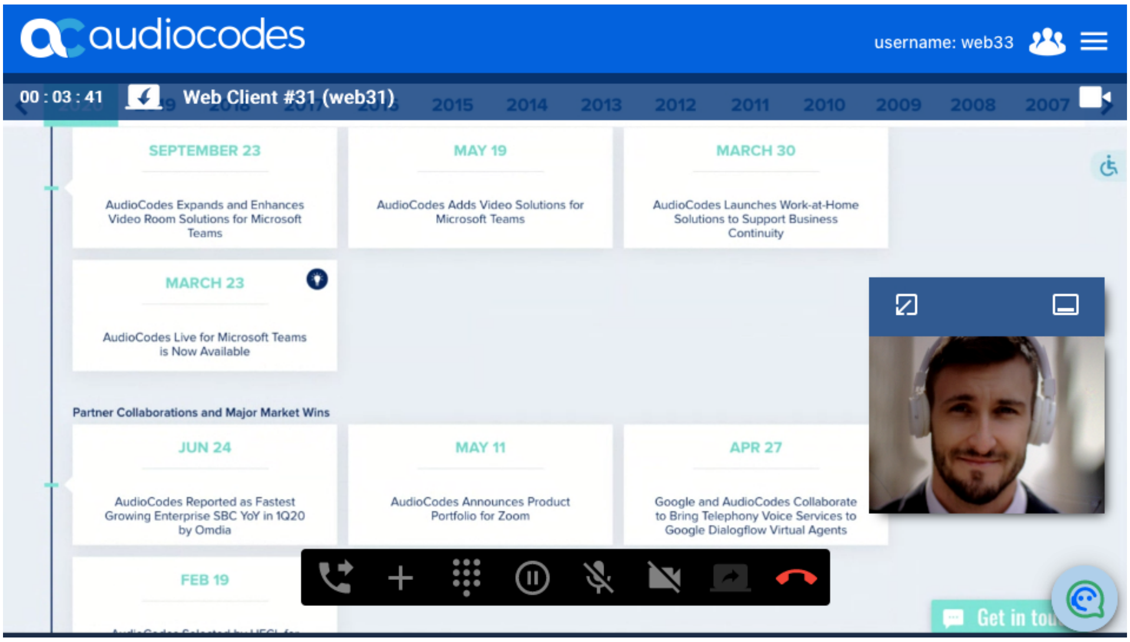
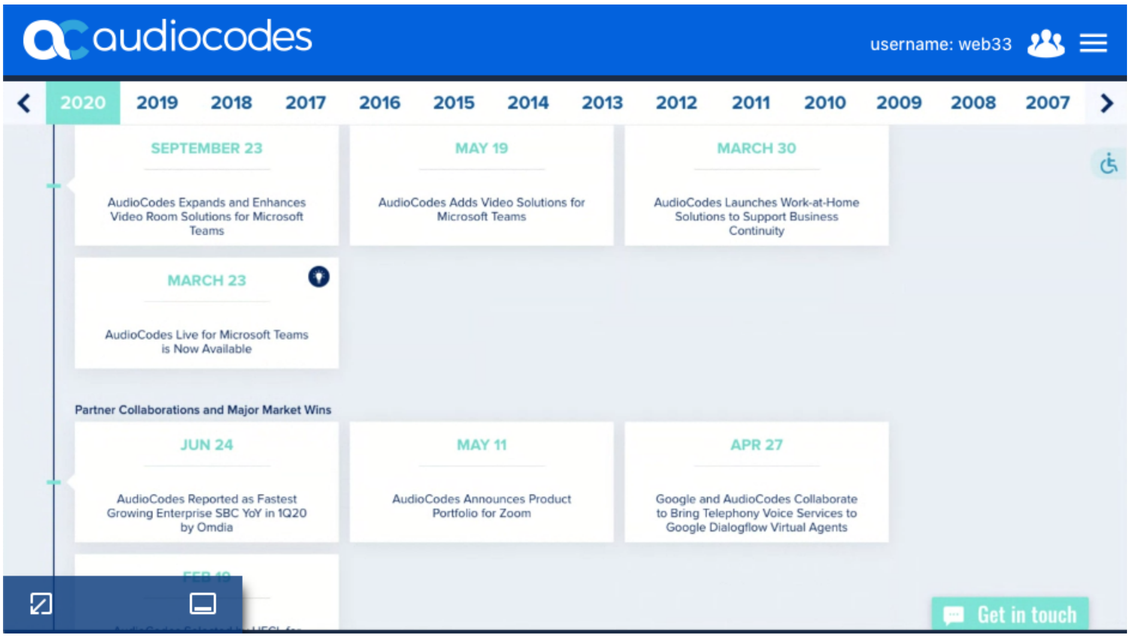


Figure 4-26: User Hides Call Controls and Information



Screen Sharing with Conference

The softphone features audio conference by mixing the audio of the call participants, while for each participant, the conference is transparent. This means that the conference is applied only at the side of the conference initiator, while the web softphones of other participants regard it as a regular call.

This also applies to screen sharing during an audio conference. Only the conference initiator can share their screen with all conference parties. When another party attempts to share the screen, the screen capture media reaches only the conference initiator, but doesn’t arrive at the other parties.

Starting / Stopping Screen Sharing when Conference in Place

The starting and stopping a screen sharing session GUI is similar to the one described in Screen Sharing on page 34. However, when a conference is in place, the following also occurs:

- Starting screen sharing: The media captured from the screen is shared between all calls in conference.
- Stopping screen sharing: Stops the screen sharing session with all calls.

Starting / Stopping Audio Conference During Active Screen Sharing Session

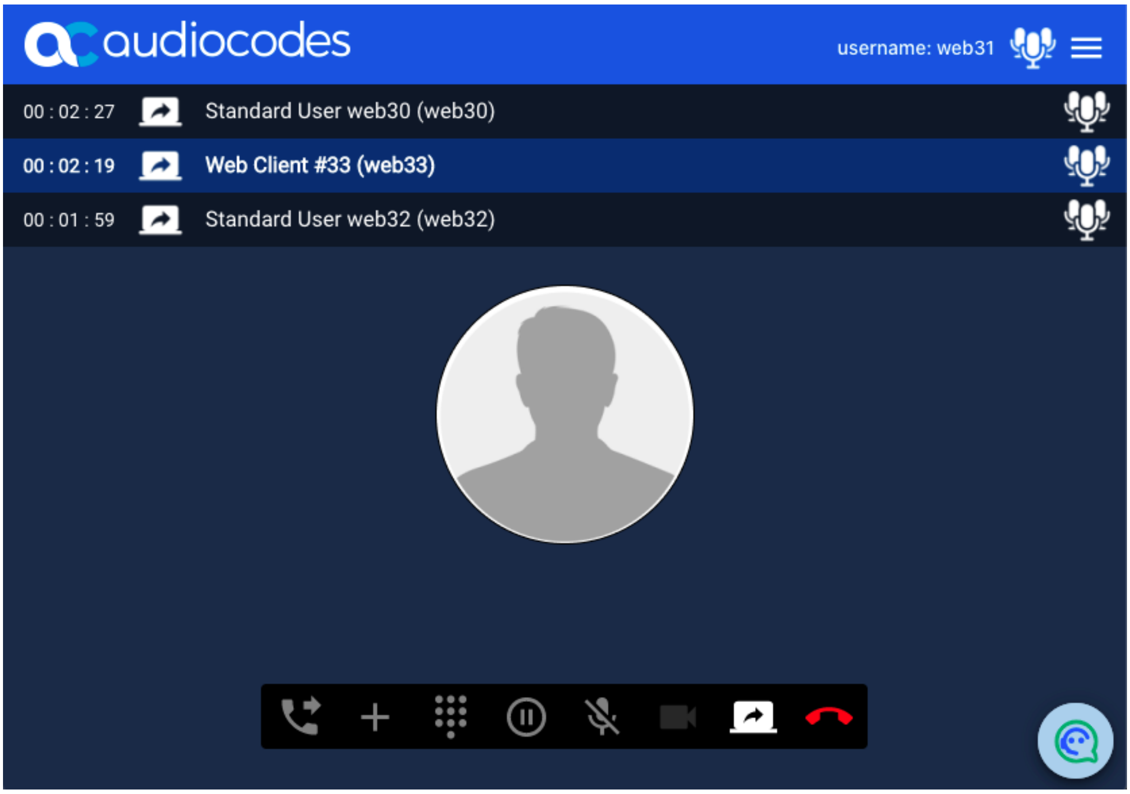
- When one of the currently active calls is performing screen sharing, starting an audio conference will begin sharing the screen with all the other active calls.
- When the audio conference stops, the screen sharing session is stopped altogether, for all calls.

New Calls Joining Conference with Screen Sharing

When an audio conference with screen sharing is in place, and a new incoming or outgoing call is established, the following occurs:

- The call joins the conference automatically.
- The screen sharing session is also being shared with the new call.

Figure 4-27: Audio Conference with Screen Sharing



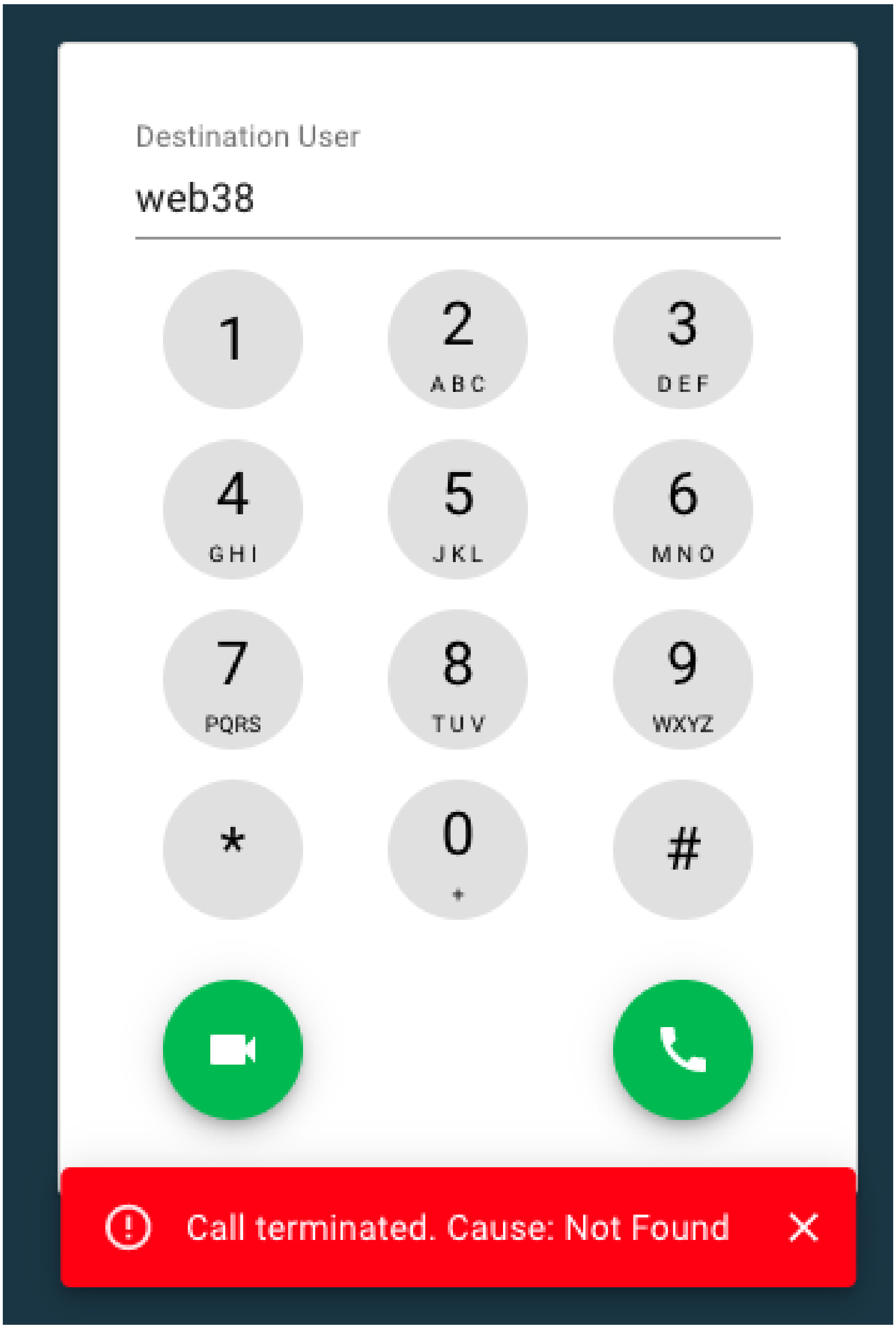
Call Errors

Generally, an error during the call displays an alert to the user and terminates the call.

Call errors can involve the following:

- SIP error responses for an outbound request (e.g., SIP 4xx response for a re-INVITE)
- Media stream errors
- Network transport errors
- Unsupported or unavailable media

Figure 4-28: Call Error Alert



OAuth Authentication Errors

If the softphone uses OAuth (see Settings UI) and there has been an authentication challenge response to a SIP request when in the Dialer, the softphone attempts to refresh the access token and proceed.

Call Termination

The following are related to call termination:

- **Call Terminated Tone:** Upon call termination, the softphone plays a call disconnected beeping tone.
- **Call Terminated with Error:** If the call is terminated with an error, an alert is displayed on the softphone.
- **OAuth, Call Rejected with Authentication Challenge:** When the softphone uses OAuth-token authentication scheme for SIP registration, if a call is terminated with a 401-authentication challenge, the softphone attempts to refresh authentication and restore the call.

In general, upon call termination, one of the current calls is focused and if no other calls exist, the Dialer appears.

Page Reload

When reloading the page, the following occurs:

- If the user has selected 'Keep me logged in':
  - SIP Registration: The softphone performs SIP REGISTER with the account configurations from before the page reloaded, with the server address that was last used.
  - Call Restoration: For each call that existed prior to the page reload, the softphone initiates an outgoing call to the corresponding remote party, and when that call is established, the softphone resumes the previous call state (hold, mute, or conference).
- Otherwise:
  - The login screen is displayed again to the user.

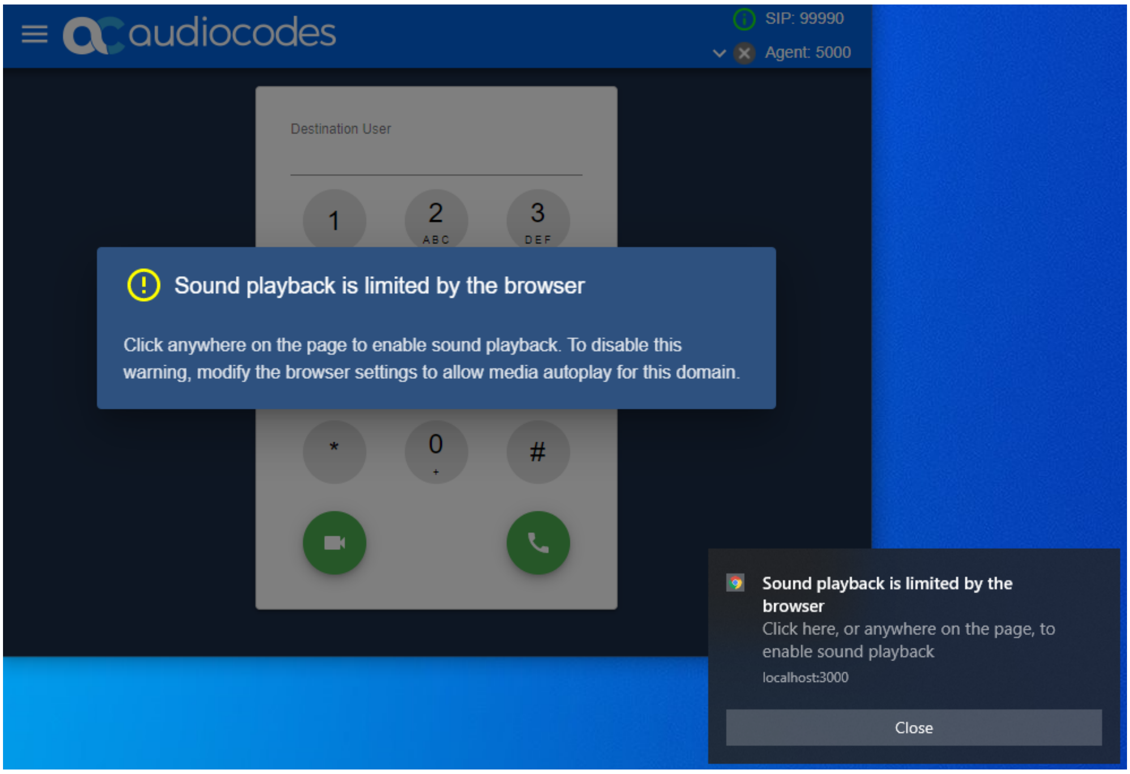
Auto-Play Policy Considerations

Most browsers, as part of their auto-play policy, block media playback without user interaction by default, until the user interacts with the page. This might prevent tone playback for call events, such as playing the incoming call sound, busy tone etc.

To handle the case when the browser blocks media playback:

- The softphone checks upon loading the page, whether audio auto-play is enabled.
- If auto-play is disabled, the site administrator can configure the softphone to:
  - Display a warning on the page
  - Display a browser desktop notification if possible (currently not supported in Firefox).
- Once the user clicks anywhere on the page, or clicks the notification, the warning alert and notification disappear, and media playback is enabled.

Figure 4-29: Warning for Auto-play Being Disabled



5 Call History UI

The softphone can display the call history for a SIP user once the user has successfully registered. The Call History screen is accessible from the sidebar.

Figure 5-1: Selecting Call History

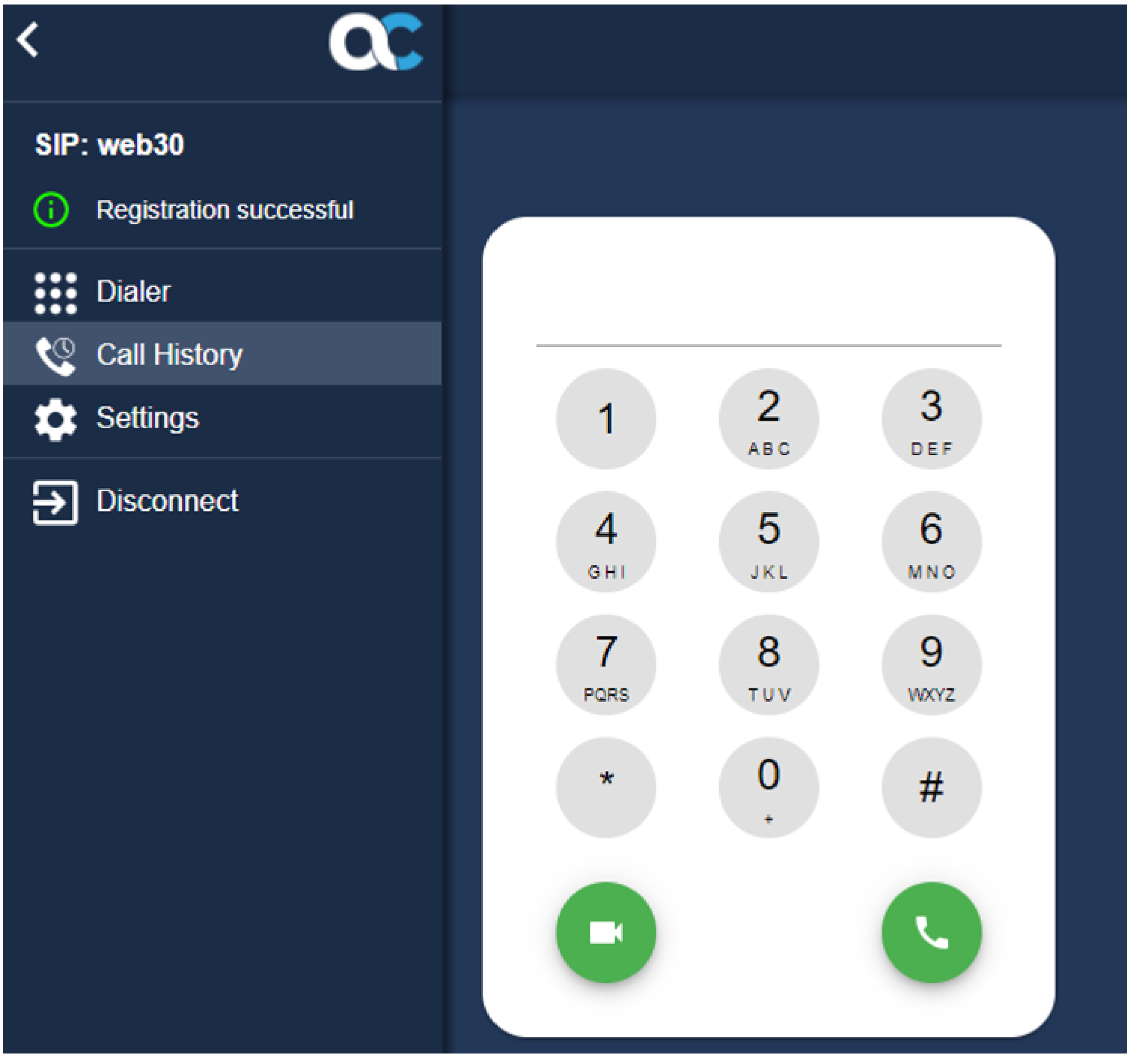
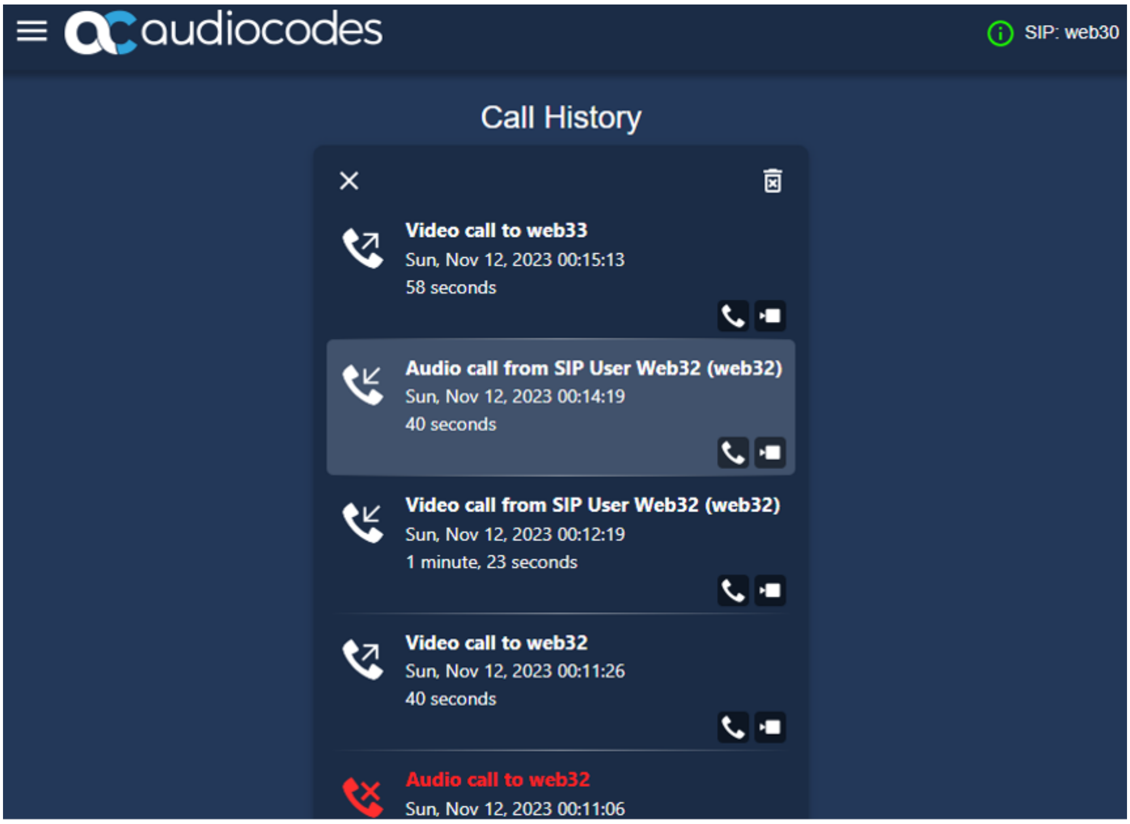


Figure 5-2: Call History



- The user can choose to delete all call logs by clicking the trash button.
- For each call history entry, the user can choose to start audio or video call by pressing the corresponding call button.
- Unless the user explicitly clears the call history, it will be available when the user logs out and logs in again.

6 ACD Agent for CTI

This section introduces the softphones's integration with an Automatic Call Distributor (ACD) of a computer telephony integration (CTI) system, such as Genesys.

Feature Overview

The site administrator can configure the softphone to enable integration with ACD agents. This enables the softphone to display GUI features related to ACD agent configuration and status management.

See the figures in sections Softphone ACD Flows on the next page, Agent Settings, and Agent State Control via the Web Softphone Header.

Supported ACD Agent Operations

The web softphone is capable of performing the following operations in various sequences according to the GUI triggers defined in ACD GUI Features on the next page. The following definitions provide a way to refer to these sequences throughout the document:

- ACD Start:
  - Subscribe to the ACD BroadWorks Hoteling event packages.
  - Subscribe to the ACD as-feature event packages.
- ACD Stop:
  - Unsubscribe to the ACD BroadWorks Hoteling event packages.
  - Unsubscribe to the ACD as-feature event packages.
- ACD Log On:
  - Subscribe to the hoteling service, to set hoteling guest address to a specific username.
  - Set agent state to “not ready”, with a default reason-code (see ACD Set Agent State).
- ACD Log Off:
  - Set agent state to “logged off” (see ACD Set Agent State).
  - Subscribe to the hoteling service, to set hoteling guest address to be empty.
- ACD Set Agent State: “ready” / “not ready” / “working after call”
  - Subscribe to as-feature-event with a state change request to the appropriate state.
  - For “not ready” state, the user may have the option to select a reason code, which defaults to a pre-configured default value that is determined by the system administrator (typically 0).

ACD GUI Features

If the web softphone is configured to have the ACD feature enabled by the site administrator, then the following capabilities are available to the user:

- Log On: Performs ACD Start + ACD Log On.
- Log Off: Performs ACD Log Off + ACD Stop.
- Set agent state (ready / not ready / working after call): Corresponds to ACD Set Agent State. Not Ready state is set with a reason code that is either selected by the user, or defaults to a pre-determined value.
- Configure ACD Settings: The user can configure the agent guest address username and password, as well as the ACD subscription expiration interval.
- Display ACD Subscription Errors: When the softphone initiates an ACD operation, which involves a SUBSCRIBE request to either BroadWorks Hoteling or to ac-feature-event, an error response is displayed to the user.

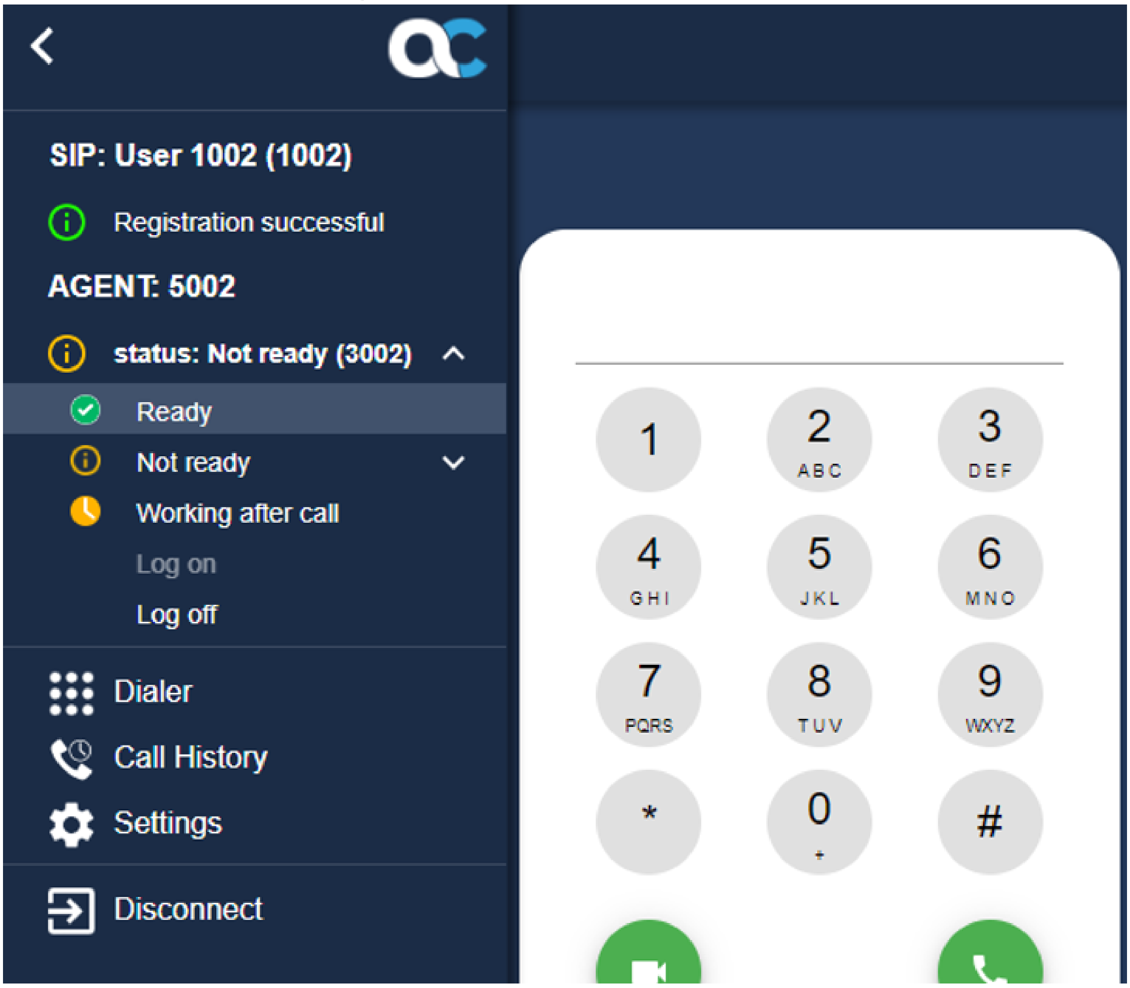
> Unless the ACD feature is enabled, these capabilities, as well as the ACD agent status display, are not visible or accessible to the user.

Softphone ACD Flows

The web softphone allows the user to set the ACD agent configuration, as well as perform the ACD status updates, via the side bar or the application header.

Figure 6-1: ACD Control From the Sidebar

Figure 6-2:



ACD Agent States Control

The following describes ACD Agent States Control.

Agent Logged-off State

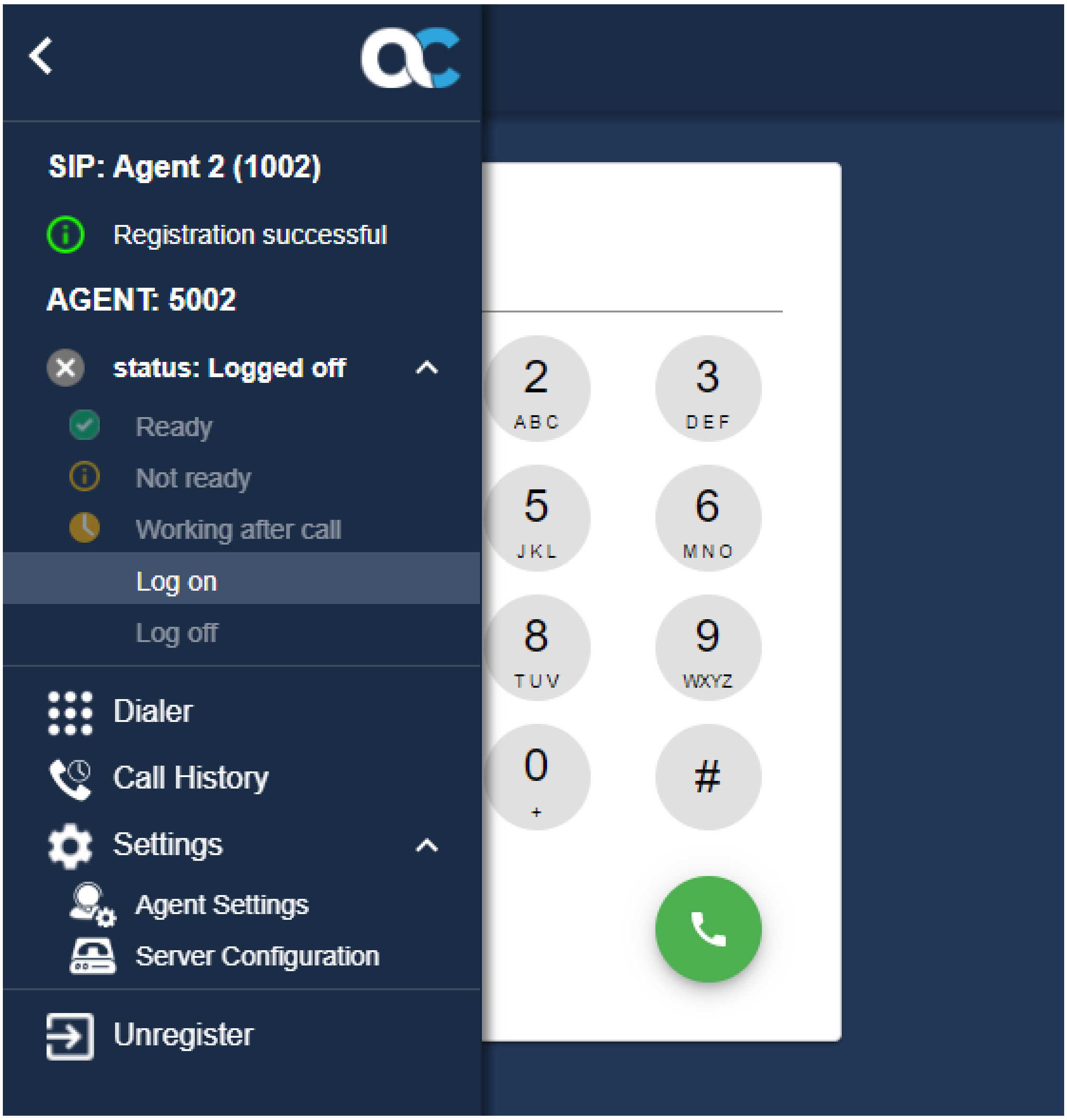
Here the softphone displays the agent status with an empty username (N/A), and the state to be logged-off.

> In this state, the ACD feature is not started.

From the logged-off state, the user can change the agent state only to “Log on”.

Change the Agent State to "Log on"

Figure 6-3: ACD Agent State Changed to "Log on"



This triggers initiating ACD Start + ACD Log On, as defined in Supported ACD Agent Operations on page 45. If successful, this eventually results in the agent state being set to "not ready", with a reason code which is determined by the system administrator as the default "not ready" reason.

Agent Logged-on ("not ready") State

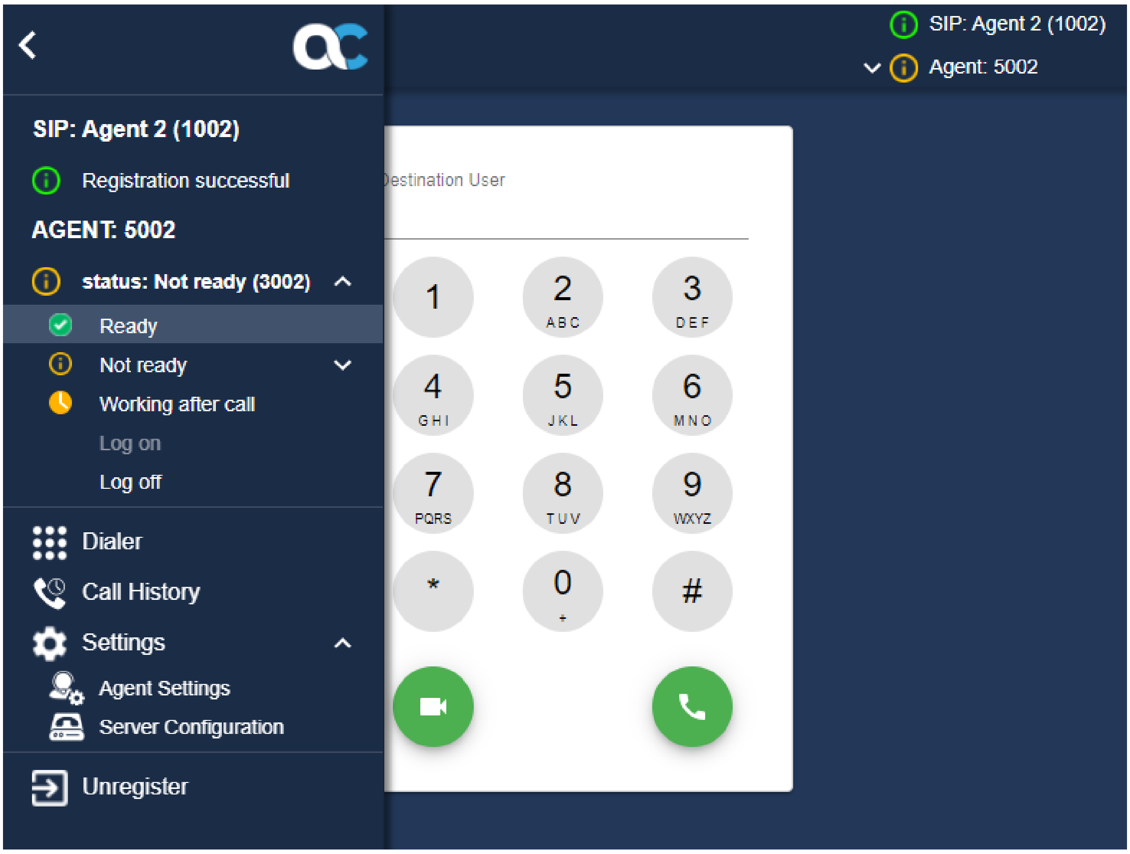
From the logged-on state, which automatically places the agent as "not ready" with a default reason code, the user can:

- Change the state to: "Log off", "ready", "working after call", or "not ready" with a different reason code
- Update the agent settings and re-login

Change the Agent State to “Ready / Working after call”

This triggers the ACD Set Agent State operation to the corresponding state. For example, setting the state to “ready”:

Figure 6-4: ACD Agent State Change from “Not Ready” to “Ready”



Change Agent State to “Log Off”:

This triggers the softphone to perform ACD Log Off + ACD Stop (see Supported ACD Agent Operations on page 45).

Agent Ready / Not Ready / Working After Call States

Switching between these states performs a subscription to the as-feature-event service, with the appropriate state change request. See **ACD Set Agent State** under the Supported ACD Agent Operations on page 45. From either of these states, the user can also change to "Log Off".

Figure 6-5: ACD Agent State Changed "Working after call"

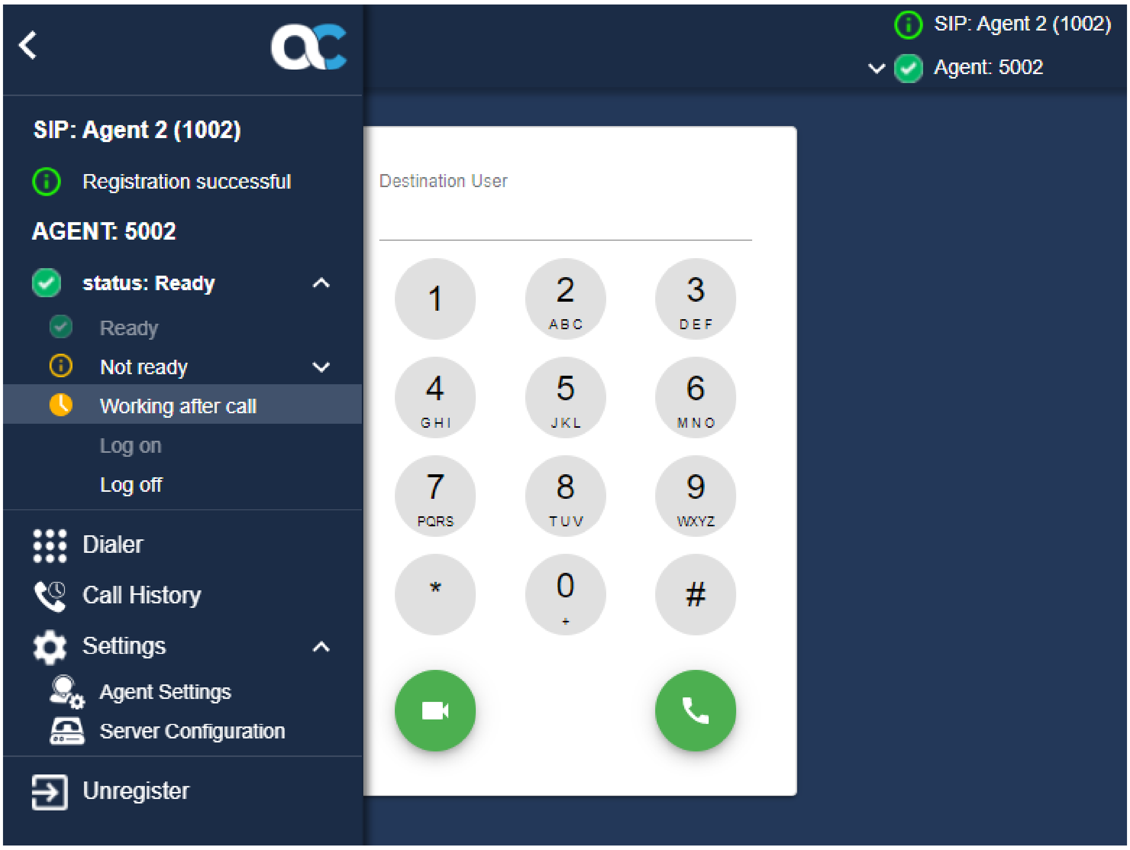
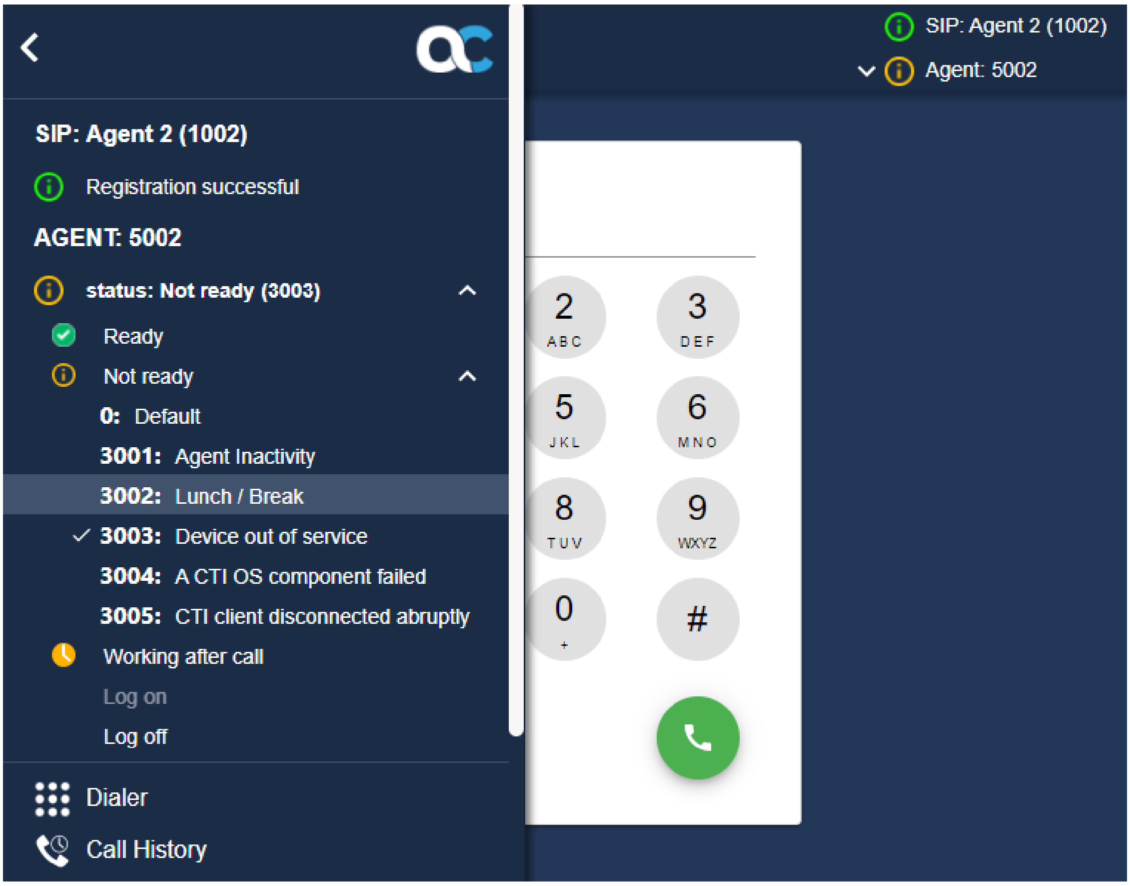


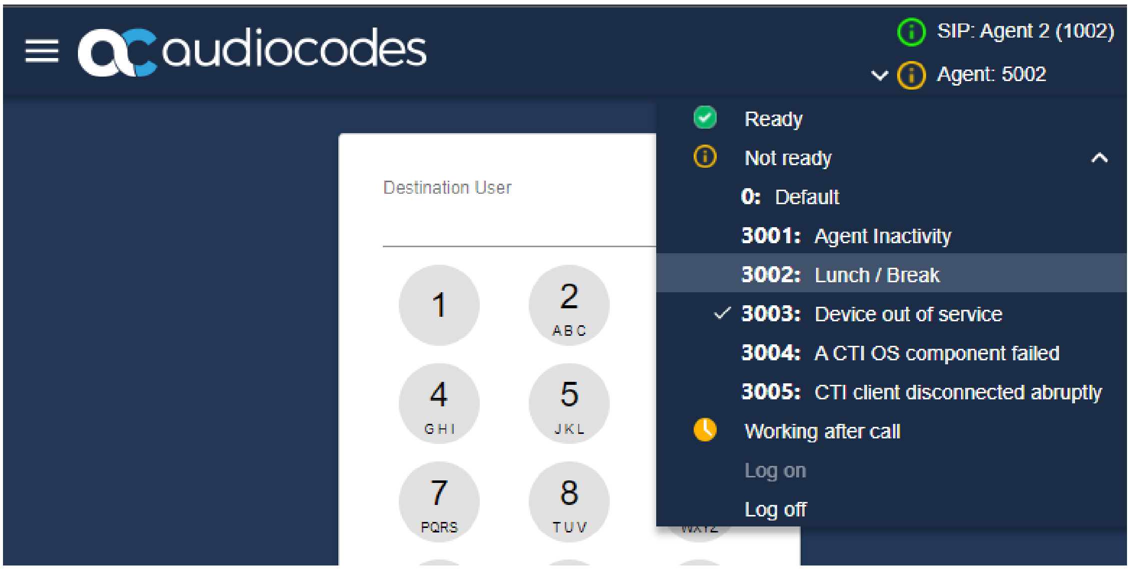
Figure 6-6: ACD Agent Selecting Not-Ready State with a Reason Code



Agent State Control via the Web Softphone Header

The user can also change the agent state from the softphone header:

Figure 6-7: ACD Agent State Changed from Softphone Header

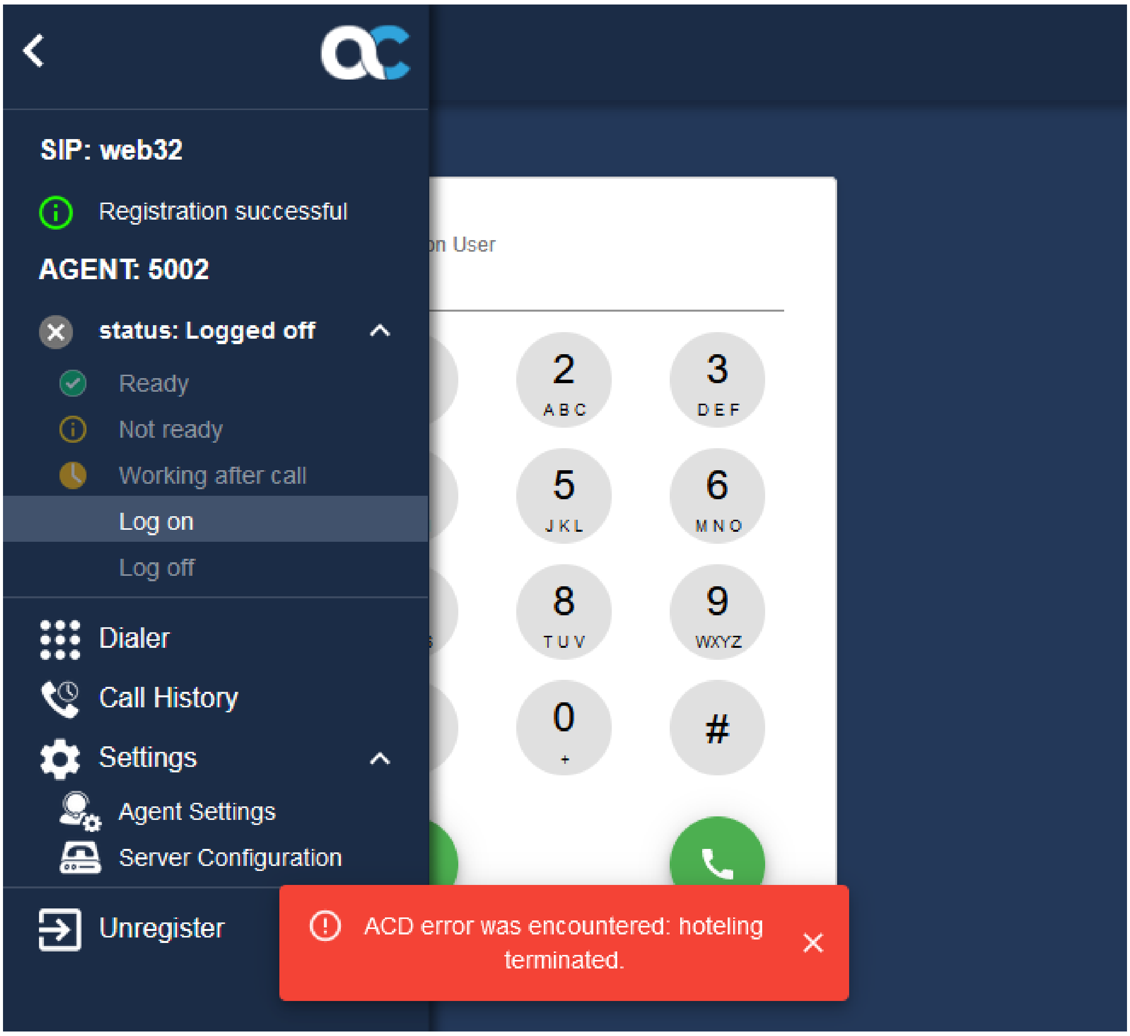


ACD Error Handling

Upon receiving an error as a response to a ACD subscription request, the softphone displays a corresponding error message.

If the error causes the subscription to be terminated, the softphone shows the agent state as "logged off":

Figure 6-8: Softphone Encountered Hoteling Subscription Error During Log On



Softphone Startup / Shutdown

The following describes the softphone startup and shutdown issues.

Softphone Shutdown

Upon softphone shutdown on page unload, it automatically performs an ACD Stop (see Supported ACD Agent Operations on page 45) and stores the last agent state and configuration, except for the agent password.

Softphone Startup

Upon softphone startup, it automatically attempts to perform ACD Start + ACD Log On as defined in Supported ACD Agent Operations on page 45, if all of the following conditions apply:

- The user has selected the 'Keep me logged in' check-box.
- The ACD agent has logged in without a password.
- ACD was previously stopped due to page unload.

7 Agent Assist

This section introduces the softphone's integration with the Agent Assist service, which allows utilizing a chatbot conversation assistant for an agent using the softphone.

Feature Overview

Agent Assist uses the VoiceAI Connect speech recognition capabilities, along with integrated chatbot flows, to deliver the call transcript as well as conversational insights during a call.

For a call to be eligible to use Agent Assist, it must include certain attributes that serve as the call context for the Agent Assist service. Typically, the call context consists of:

- Bot name: The bot name to request for the Agent Assist bot session
- Call UUID: A unique identifier that conforms to the Agent Assist service requirements, which is established for each call according to the call center platform being used. For example, when integrating with Genesys, the Call UUID is the Genesys Call UUID.

The site administrator can configure the softphone to enable integration with the Agent Assist service. This configures the softphone to display the Agent Assist chatbot GUI for calls that are eligible for utilizing the service, as well as chatbot history for recent calls.

Agent Assist can choose to start / stop chatbot activation during a call.

Agent Assist During Calls

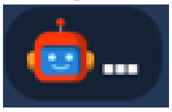
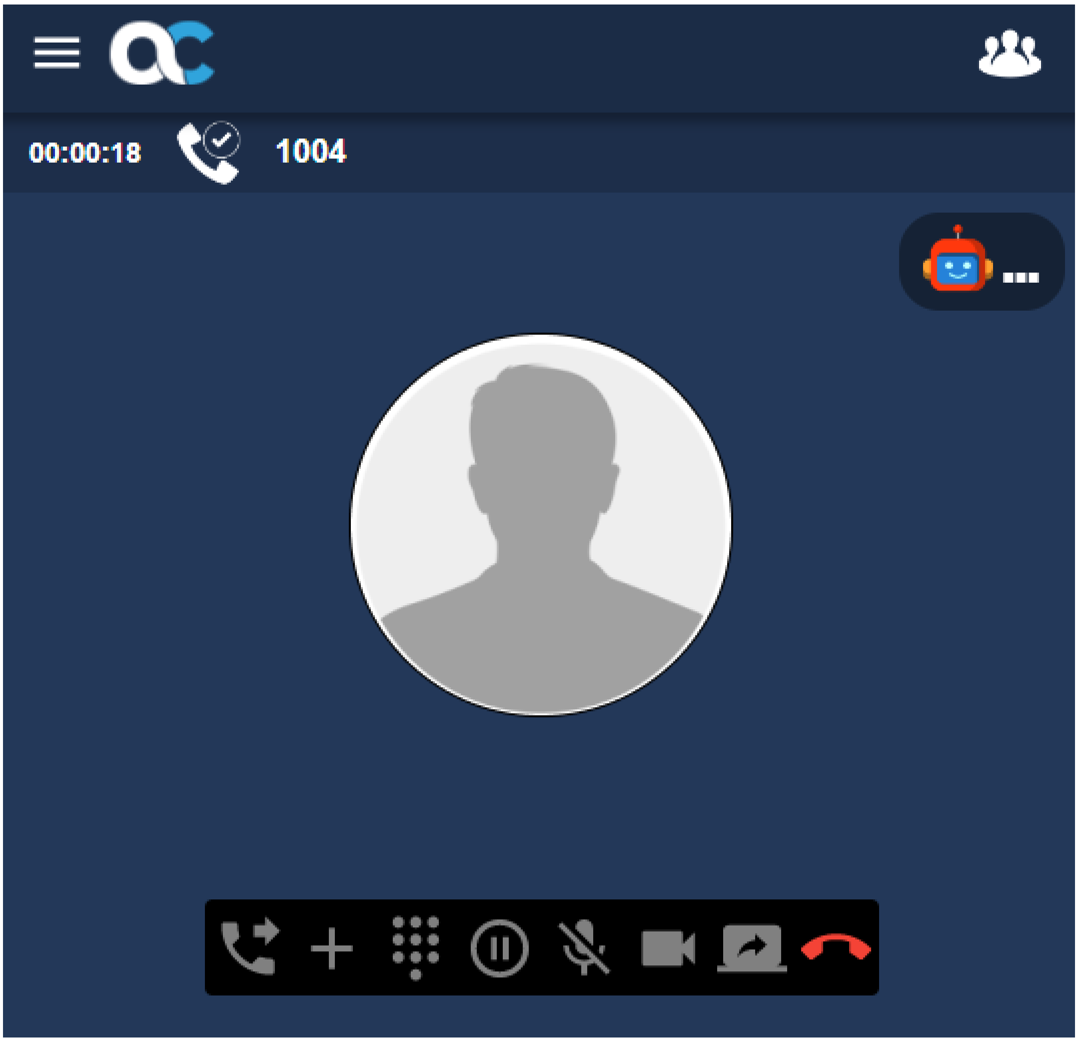
When a call is active, if that call has the appropriate information to be used as a call context for the Agent Assist service, the call display includes the Agent Assist UI, initially as a chatbot  button at the top-right corner of the call display:

Figure 7-1: Active Call Showing the Agent Assist Chatbot Button:



Clicking it displays the chatbot UI, showing the following call details:

Prompts/Buttons	Description
Title	<bot-name> Agent Assist Indicating the bot name, which is part of the calling context for Agent Assist
Call UUID	A unique identifier for the call which serves as part of the calling context for Agent Assist
Connection	The connection status to the Agent Assist service idle / disconnected / connected / initiated, where “initiated” means that the service is connected and ready to start bot sessions.
Bot session	The status of the current call bot session (idle / starting / started / stopping)

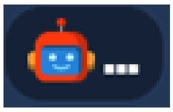
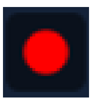
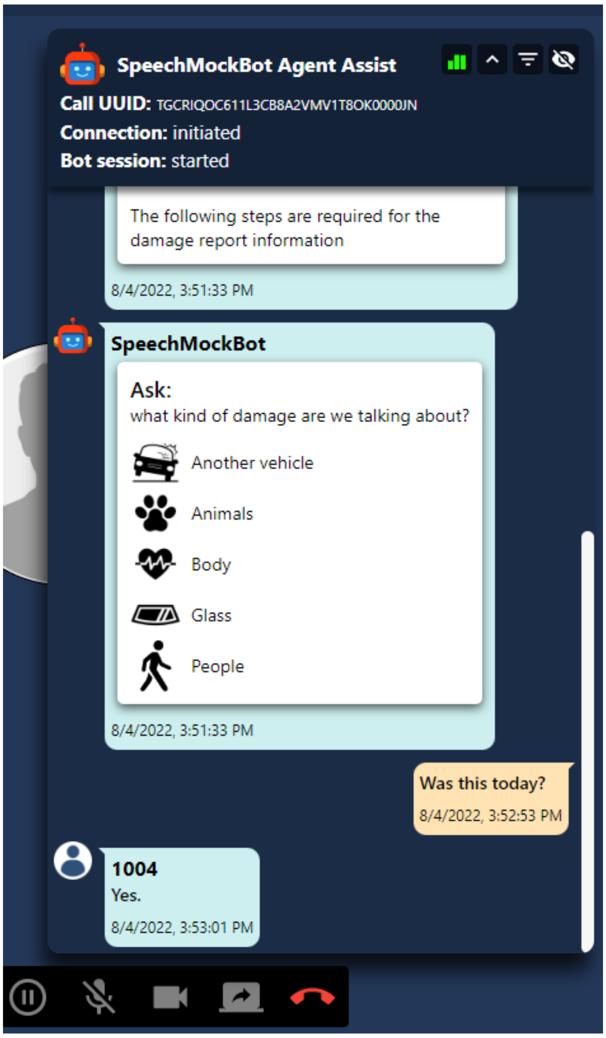
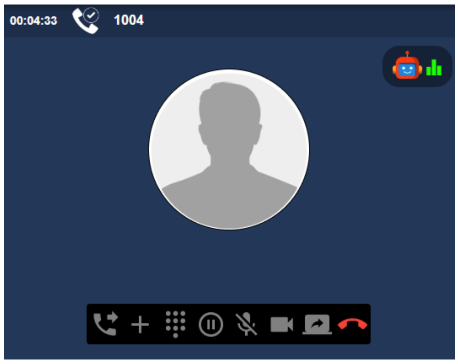
Prompts/Buttons	Description
Start / stop bot session	Start / stop the Agent Assist bot session for the current call
Collapse / Expand details	Collapse or Expand details
Filter	Filter the displayed chatbot messages to allow only bot feedback, only call transcript, or both
Hide chat	Hide the chat GUI, showing only the chatbot  button

Figure 7-2: Chatbot UI in idle state

Activating the Agent Assist bot session

When clicking the start  button, Agent Assist delivers the call transcript as chat messages where the speaker is recognized, and so the chat bubbles are identified as either incoming or outgoing, according to the speaker.

In addition, the Agent Assist delivers bot feedback messages, whose sender is shown to be the bot name (in this example, the sender is “SpeechMockBot”).

Chatbot UI in active state, showing bot feedback and call transcript messages	Chatbot UI in active state, When the chat screen is hidden
	

The bot session can be started and stopped multiple times during a call.

The Agent Assist chat messages can be filtered via the Filter  button as follows:

Figure 7-3: Chat UI filtered to show only transcript messages

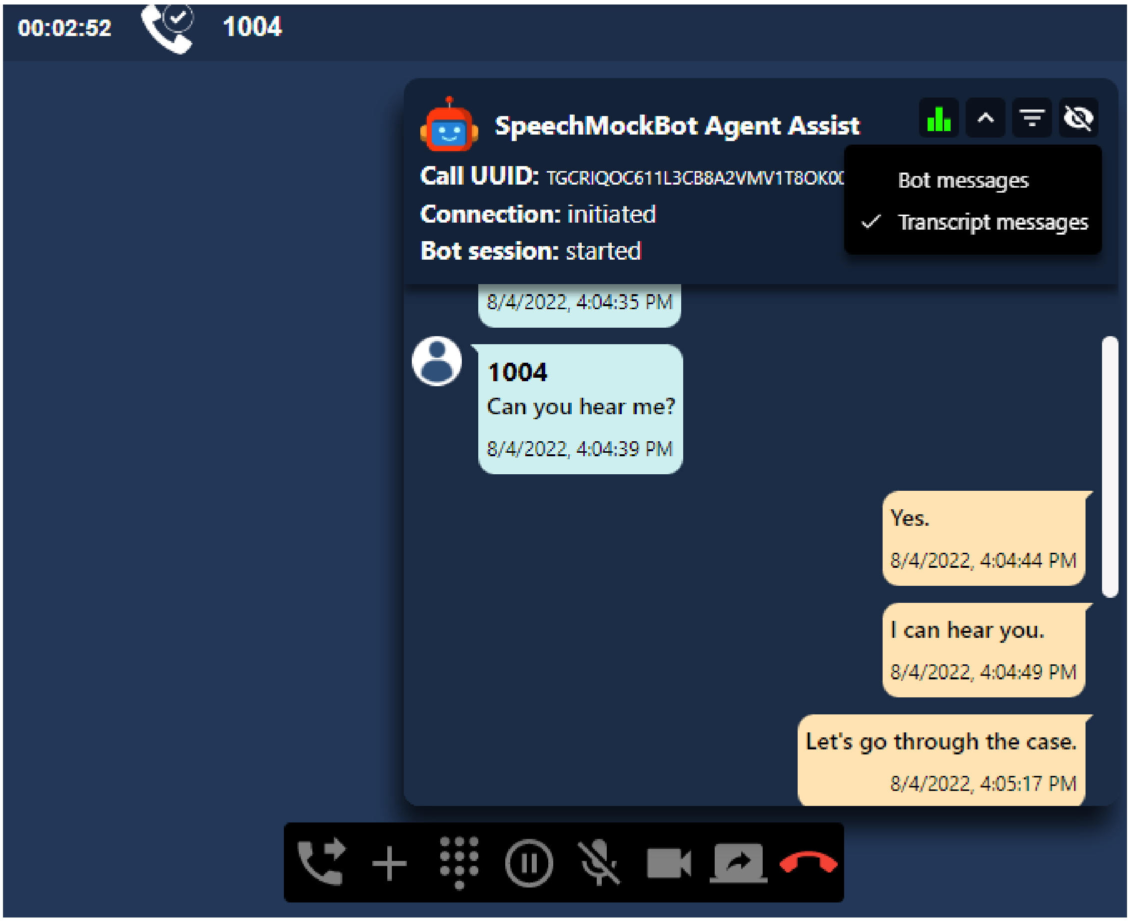
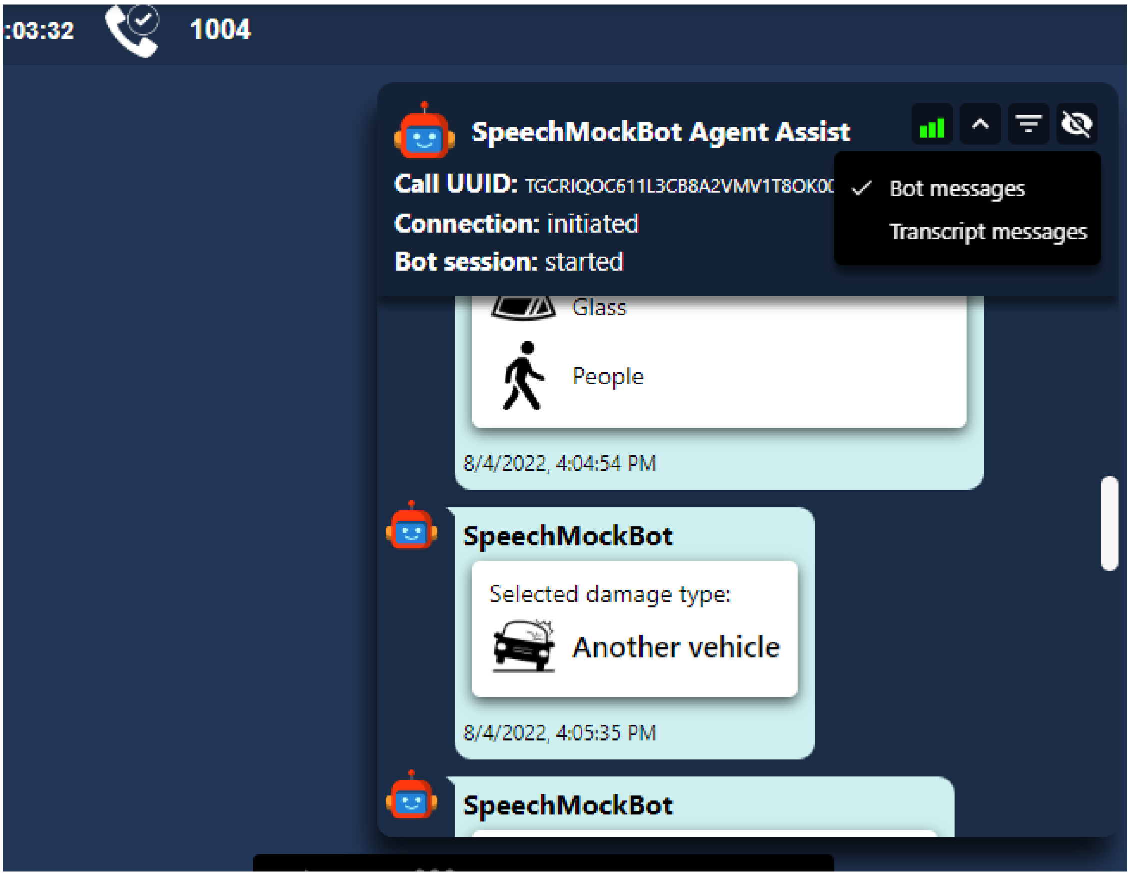
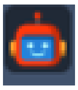


Figure 7-4: Chat UI filtered to show only bot feedback messages



Agent Assist Chat History

Accessing the Agent Assist chat history is done from the call history entries. Each call history entry from the recent calls that has call-context information eligible for agent assist, includes a chatbot  button that can display the Agent Assist chat for that call.

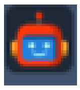
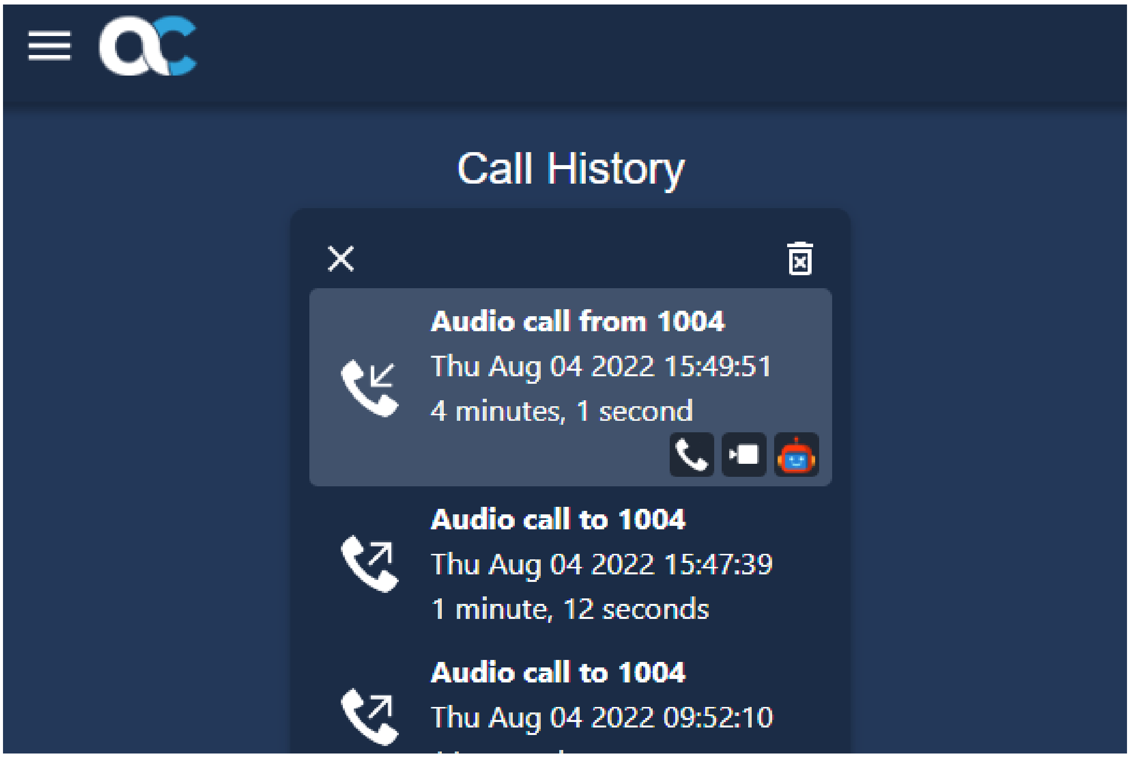
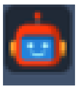
For calls that do not include valid calling context, the chatbot  button is disabled.

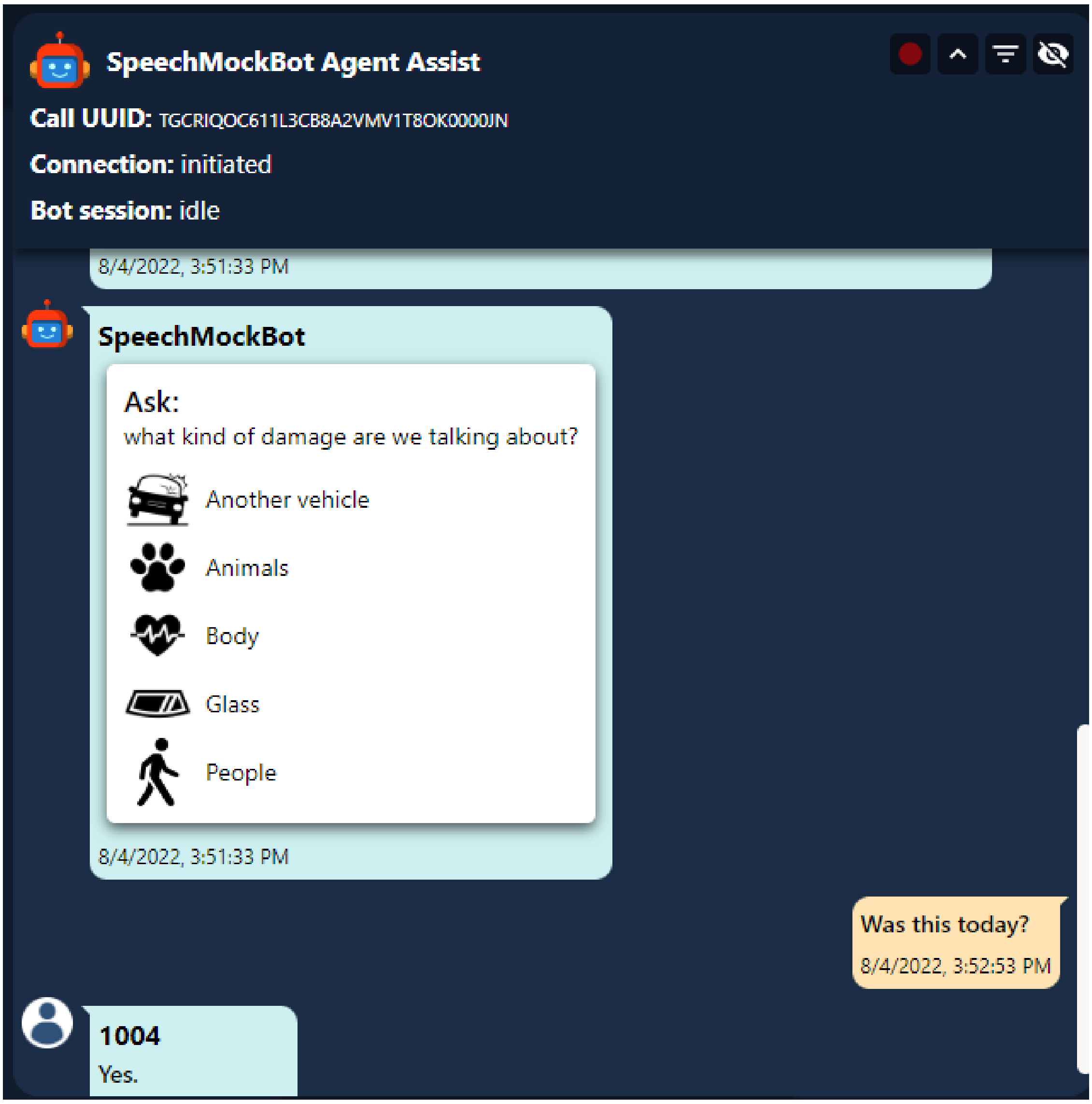
Figure 7-5: Call history entries with Agent Assist chat button



Clicking the Agent Assist chatbot  button shows the chatbot history for that call.

Note that in this display, the chat GUI can also be filtered according to bot / transcript messages.

Figure 7-6: Agent Assist history displayed for the call history entry



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