

# WebRTC Android Client SDK

Version 2.0.1

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## Notice

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## Stay in the Loop with AudioCodes



## Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

## Related Documentation

Document Name
<a href="https://www.audiocodes.com/solutions-products/solutions/enterprise-voice/webrtc-connectivity">https://www.audiocodes.com/solutions-products/solutions/enterprise-voice/webrtc-connectivity</a>
WebRTC Client SDK API Reference Guide
WebRTC Click-to-Call Widget Installation and Configuration Guide

## Document Revision Record

LTRT	Description
14060	Initial document release for Version 1.0
14061	Added functionality for SDK Version 1.0.1
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14063	<ul style="list-style-type: none"> <li>■ Updated to software Version 1.1.0.</li> <li>■ Blind transfer: <ul style="list-style-type: none"> <li>• New function - transferCall(RemoteContact transferTo)</li> <li>• New function - getTransferContact</li> <li>• New function - getTransferState</li> </ul> </li> <li>■ Attended transfer - New function - transferCall(AudioCodesSession transferToSession)</li> <li>■ OAuth authorization - New function setOAuthToken</li> <li>■ 64 bit support (arm64-v8a)</li> <li>■ IPv6 support</li> <li>■ Click to call support (calls without registration) - Added new function - login(Context, Boolean autologin)</li> <li>■ Push support: <ul style="list-style-type: none"> <li>• New function - setPushToken</li> <li>• New function - updatePushToken</li> </ul> </li> <li>■ Support for Google WebRTC SDK 1.0.26885</li> <li>■ Support for delayed offer</li> </ul>
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14068	<ul style="list-style-type: none"> <li>■ Added support for Android tablet devices</li> </ul>
14069	<ul style="list-style-type: none"> <li>■ Updated software to Version 1.2.1</li> <li>■ Updated SIP stack to pjsip2.10</li> <li>■ Support for Android API Level 31</li> </ul>
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LTRT	Description
14071	<ul style="list-style-type: none"><li>■ Updated software to Version 2.0.1</li><li>■ Updated SIP stack to PJSIP 2.13</li><li>■ Updated WebRTC library to M123 (branch 6312)</li><li>■ Updated OpenSSL to 3.2.0</li><li>■ Updated setAccount description</li><li>■ Added permission description</li></ul>



## Documentation Feedback

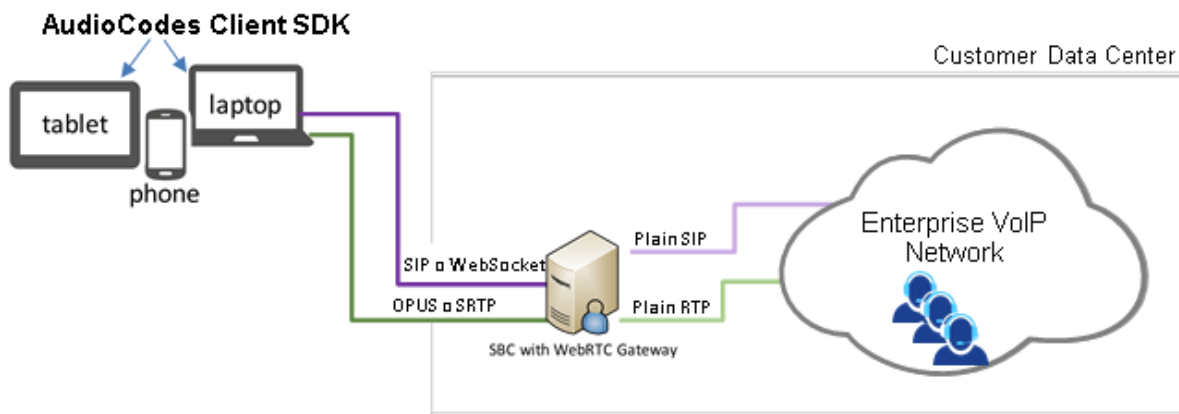
AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our Website at <https://online.audiocodes.com/documentation-feedback>.



# 1 Introduction

WebRTC technology enriches user experience by adding voice, video and data communication to the browser, as well as to mobile applications. AudioCodes WebRTC gateway provides seamless connectivity between WebRTC clients and existing VoIP deployments.

A typical WebRTC solution is comprised of a WebRTC Gateway, which is an integrated functionality on AudioCodes SBCs, and a client application running on a browser or a mobile app. AudioCodes WebRTC Android client SDK is a Java code based API that allows Android developers to integrate WebRTC functionality into Android applications for placing calls from the Android device to the SBC.



For a simple click-to-call button use case, a WebRTC widget is offered which can be easily integrated into websites and blogs without any JavaScript knowhow. See the *WebRTC Widget Installation and Configuration Guide*.

## 1.1 Purpose

This *Reference Guide* defines the Application Programming Interface (API) use of the Web Real-Time Communications (RTC) SDK.

## 1.2 Scope

The guide describes the API that must be implemented to use AudioCodes' WebRTC Android SDK to build an Android application that will interact with AudioCodes' server to establish voice and video calls.

The guide may be used by Android developers who want to use the AudioCodes-provided SDK to build Web RTC clients.

## 1.3 Benefits

Here's a summary of the benefits:

- Simple deployment - a single WebRTC gateway device for both signaling and media
- Strong security and interoperability capabilities resulting from integration with SBC
- Client SDK for Android application.
- OPUS powered IP phones for superb, transcoder-less voice quality
- Optional recording of WebRTC calls

## 2 Android SDK

The sections below describe how to install the Android SDK.

### 2.1 Before you Start



This document has references to the Democlient (which includes code examples), accompanying this documentation.

Here's what you require before you start using the Android SDK:

- Android Studio (tested with Android Studio 3.1)
- Android device with minimum OS 6.0
- Provided with the SDK:
  - **WebRTC SDK AAR File:** This is the WebRTC SDK AAR file. The file must be included in a project to use the SDK.
  - **Android Demo Client Project:** This is the Android Studio project that can be opened from the Android Studio. This is a fully working client and shows how to use the SDK and the AAR file.
  - **Javadocs:** Located in the Android demo client. These are javadocs that can be used as a reference to the WebRTC SDK.

### 2.2 Installation

Here's what you need to do to install the SDK.

**To install the SDK:**

1. Install Android Studio.
2. Open a project for the demo client.
3. Follow the instructions from the Android Studio (necessary Android SDK files will need to be downloaded and installed.)
4. The WebRTC SDK is currently required to add the following lines to the application build.gradle files (see Demo client):

```
ndk {  
    abiFilters 'armeabi-v7a', 'arm64-v8a'  
}
```

5. The SDK needs specific Android permissions. The SDK does not check if these permissions are present for the Android application. Any Android application that uses the SDK needs to make sure the permission has been requested and provided by the user. The demo client provides examples on how to do this.

## 6. Permission description:

### Required permissions:

- Internet permission:  
<uses-permission android:name="android.permission.INTERNET" />
- Microphone permission:  
<uses-permission android:name="android.permission.RECORD\_AUDIO" />
- Permission to route audio to speaker/headset/Bluetooth  
<uses-permission android:name="android.permission.MODIFY\_AUDIO\_SETTINGS" />
- Used for audio routing from speaker/headset/bluetooth  
<uses-permission android:name="android.permission.BLUETOOTH" />  
<uses-permission android:name="android.permission.BLUETOOTH\_CONNECT" />

### Recommended permissions:

- Handling of incoming GSM calls (prevent simultaneous voip and gsm calls)  
<uses-permission android:name="android.permission.READ\_PHONE\_STATE" />
- Is used to access WifiManager and receive events of connection/disconnection, which allows calls to be transferred from Wifi to Cellular (and vice versa)  
<uses-permission android:name="android.permission.ACCESS\_WIFI\_STATE" />
- Same as 2, but allows access to ConnectivityManager to get events for network changes  
<uses-permission android:name="android.permission.ACCESS\_NETWORK\_STATE" />
- To keep the phone awake during calls (the call will be dropped if the device goes to sleep)  
<uses-permission android:name="android.permission.WAKE\_LOCK" />
- For incoming call vibrate  
<uses-permission android:name="android.permission.VIBRATE" />
- To be used if custom certificates (PEM files) are to be used  
<uses-permission android:name="android.permission.WRITE\_EXTERNAL\_STORAGE" />
- Not required but older devices might still require this:  
<uses-permission android:name="android.permission.BLUETOOTH\_ADMIN" />

## 3 API Classes

The API consists of:

- Main Classes
  - AudioCodesUA – AudioCodes User Agent (*Singleton*) – see Section 3.1
  - AudioCodesSession – For call representation (*Interface*) – see Section 3.1.1
  - WebRTCAudioManager – Class for managing audio routes – see Section 3.3
  - [TerminationInfo](#) – [Class for information about Call Termination](#) – see Section 3.8
- Helper Classes
  - ACConfiguration – (Optional) class for general configuration options – see Section 3.4
  - VideoConfiguration – (Optional) class for video configuration – see Section 3.5
  - DTMFOptions – Class for settings DTMF options – see Section 3.6
  - [RemoteContact](#) – [Class representing the remote contact](#) – see Section 3.7
- Listener Interfaces
  - AudioCodesEventListener – Event listener for incoming calls and login state changes – see Section 4.1
  - AudioCodesSessionEventListener – Event listener for call related events – see Section 4.2
  - WebRTCAudioRouteListener – Event listener for audio route events – see Section 4.3

## 3.1 AudioCodesUA

Used to initialize the framework before starting to make and receive calls. Mostly used to initialize the Web RTC engine and register to the service.

```
Class AudioCodesUA{
    AudioCodesUA getInstance();
    void setServerConfig(String proxyAddress, int port, String
domain, Transport, List<PeerConnection.IceServer>
iceServerList);
    void setAccount (String userName, String displayName, String
password);
    void setAccount (String userName, String displayName, String
password, String authName);
    void setListeners (AudioCodesEventListener listener);
    void login(Context context);
    void login(Context, Boolean autoLogin);
    void logout();
    AudioCodesSession call(RemoteContact call_to, Boolean
withVideo, Hashtable<String, String> inviteHeaders);
    void setRegisterExtraHeaders (Hashtable<String, String>
headers);
    void setInviteExtraHeaders (Hashtable<String, String> headers);
    String getUserAgent();
    void setUserAgent(String userAgent);
    int getRegExpires();
    void setRegExpires(int registerExpires);
    void setUseSessionTimer(Boolean use);
    void setLogLevel(LogLevel level);
    void setLogger(ILog log);
    void handleNetworkChange();
    ArrayList<AudioCodesSession> getSessionList();
    void setContactRewrite(Boolean enable);
    void setOAuthToken(String accessToken);
    void setPushToken(Context context, String registrationToken, String
projectID)
    void updatePushToken(Context context, String registrationToken)
    long sendInstantMessage(RemoteContact contact, String
message);
    void disconnectOnBrokenConnection(Boolean disconnect);
}
    void setVerifyServer(Boolean verifyServer)
    void setAllowHeader(String[] events);
```

### 3.1.1 Standard Methods

#### 3.1.1.1 getInstance

Returns the singleton object instance of class *AudioCodesUA*.

#### 3.1.1.2 setServerConfig

Configures the server.

---

**Parameters**

- proxyAddress [*String*, address of server]
- port [*int*, port of the proxy server address]
- serverDomain [*String*, domain name to register to]
- transport [*Transport*, transport for connection to the server – UDP/TCP/TLS]
- iceServerList [*List<PeerConnection.IceServer>*, list of STUN and TURN servers – null value may be supplied]

---

**Return Values**

N/A

#### 3.1.1.3 setAccount (1)

Defines the account details. For this method, the authorization name is the same as the username.

---

**Parameters**

- userName [*String*, authenticating user name]
- displayName [*String*, display name for the userpassword [*String*, authenticating user password]

---

**Return Values**

N/A



### 3.1.1.4 setAccount (2)

Defines the account details. This is the same as setAccount (1) but with the option of having a different authorization name.

---

**Parameters**

- userName [*String*, user name]
- displayName [*String*, display name for the user]
- password [*String*, authenticating user password]
- authName [*String*, authorization user name]

---

**Return Values**

N/A

### 3.1.1.5 setListeners

Defines the listeners object.

---

**Parameters**

listener [*AudioCodesEventListener*, instance implementation of the *AudioCodesEventListener* interface that holds the methods to be triggered; see Section 4.1 for details on how it is defined; see also Section 5.2 for an example] [API Callbacks/ Listeners Interfaces User Agent: Set](#)

---

**Return Values**

N/A

### 3.1.1.6 login

Performs registration to the service.

---

**Parameters**

**context** [*Context*, Android application context]

---

**Return Values**

N/A

### 3.1.1.7 login (click to call)

Performs initialization to the service. The SDK will not register to the service but will allow outgoing calls to be made.

---

**Parameters**

- **context** [*Context*, Android application context]
- **AutoLogin** (Boolean):
  - **true**: The SDK registers to the service (same as login 3.1.1.6)
  - **false**: The SDK does not register to the service, but calls can be made (click to call feature)

---

**Return Values**

N/A

### 3.1.1.8 logout

Performs de-registration from the service.

---

**Parameters**

N/A

---

**Return Values**

N/A

### 3.1.1.9 call

Initiates an outgoing call.

---

**Parameters**

- **call\_to** [*RemoteContact*, destination address/number]
- **withVideo** [*Boolean*, 'true' if the call is initiated with video]
- **inviteHeaders** [*Hashtable<String, String>*, list of headers with a key/value where each key is added as a header to the SIP INVITE with the specified value]

---

**Return Values**

- **session** [*AudioCodesSession*, a call session object defined [here](#).]

### 3.1.1.10 setVerifyServer

Enables/disables the certificate verification when establishing a TLS connection; the default state is disabled.

---

**Parameters**

verifyServer [*Boolean*, 'true' to enable verification]

---

**Return Values**

N/A

## 3.1.2 Advanced Methods

The advanced methods are optional. They provide an extra level of flexibility to the API, which is based on SIP (Session Initiation Protocol). Developers who are familiar with SIP can make use of the advanced methods.

### 3.1.2.1 setRegisterExtraHeaders

Allows adding additional headers to the registration request.



The headers must be SIP headers that conform to RFC 3261.

---

**Parameters**

headers [*Hashtable<String, String>*, list of headers with a key/value where each key is added as a header to the registration request with the specified value]

---

**Return Values**

N/A

### 3.1.2.2 setInviteExtraHeaders

Allows adding additional headers to the INVITE request or response.



The headers must be SIP headers that conform to RFC 3261.

---

**Parameters**

headers [*Hashtable<String, String>*, list of headers with a key/value where each key is added as a header to the SIP INVITE with the specified value]

---

**Return Values**

N/A

### 3.1.2.3 `getUserAgent`

Gets the current user-agent string, used to build the SIP header User-Agent.

---

**Parameters**

N/A

---

**Return Values**

User agent [*String*, text describing the SIP user agent]

### 3.1.2.4 `setUserAgent`

Sets the user-agent string, used to build the SIP header User-Agent.

---

**Parameters**

User agent [*String*, text describing the SIP User Agent]

---

**Return Values**

N/A

### 3.1.2.5 `getRegExpires`

Gets the current default registration interval.

---

**Parameters**

N/A

---

**Return Values**

expires [*integer*, seconds]

### 3.1.2.6 `setRegExpires`

Changes the default registration interval from the default value (600).

---

**Parameters**

expires [*integer*, seconds]

---

**Return Values**

N/A

### 3.1.2.7 `setUseSessionTimer`

Allows enabling session timers in the call session.

---

#### Parameters

- **UseSessionTimer** (Boolean):
  - **true**: Session timers are optionally supported. [e.g., the SBC initiates session timers if configured to do so (default value)]
  - **false**: The session timers are not enabled

---

#### Return Values

N/A

### 3.1.2.8 `setLogLevel`

Changes the log level used by the application. Release builds might want to set the log level higher for security reasons. WebRTC internal logs are only enabled if the debug level is lower than or equal to DEBUG level.

---

#### Parameters

logLevel [*LogLevel*, log level enum]

---

#### Return Values

N/A

### 3.1.2.9 `setLogger`

Changes the logger used by the SDK. Logs for the WebRTC SDK and logs for PjSIP are written to the custom logger. WebRTC internal logs are still be written to the console (if the log level is lower than debug)

---

#### Parameters

logger [*ILog*, instance object of the ILog interface]

---

#### Return Values

N/A

### 3.1.2.10 `handleNetworkChange`

Handles network changes when called. This re-registers the client and reestablishes the audio sessions when the network has been changed.

It must be explicitly called by the client application. The SDK does not automatically detect a network change. Ideally, this must be called when the network is reconnected, not when disconnected.

---

**Parameters**

- N/A

---

**Return Values**

N/A

### 3.1.2.11 `getSessionList`

Gets the current session list.

---

**Parameters**

- N/A

---

**Return Values**

### 3.1.2.12 `List of Sessions [ArrayList<AudiocodesSession>, List of Active Sessions]``disconnectOnBrokenConnection`

This method changes the way call handover is being handled by the SDK. Default behavior is to disconnect a call when there is no RTP for a period of time. This method (when set to 'false') allows for the call to continue even when the connection is broken.

---

**Parameters**

- **DisconnectOnBrokenConnection**(Boolean):
  - **true**: Calls are terminated when a network connection error occurs in the audio stream (default value)
  - **false**: Calls continue when a network connection error occurs

**Note:** Setting this to 'false' forces enable contact rewrite.

---

**Return Values**

N/A

### 3.1.2.13 setContactRewrite

This option is used to update the transport address and the Contact header of REGISTER request. When this option is enabled, the SDK keeps track of the public IP address from the response of the REGISTER request. Once it detects that the transport address has changed, it will unregister the current Contact, update the Contact with the transport address learned from the Via header, and register a new Contact to the registrar. It will also update the public name of the UDP transport if STUN (Session Traversal Utilities for NAT) is configured.

---

#### Parameters

- **ContactRewrite**(Boolean):
  - **true**: The library tracks the public IP address from the response of the REGISTER request
  - **false**: The library does not track the public IP address from the response of the REGISTER request. (Default value)

---

#### Return Values

N/A

### 3.1.2.14 setOauthToken

Optional method to allow the SDK to use Oauth authentication for registration to the service. The SDK will add an authorization header with the supplied access token. (The SBC will need to be configured to use Oauth authorization as well.)

---

#### Parameters

**accessToken** [*String*]- access token as received from the Oauth server (refer to the Democlient for an example on Oauth registration)

---

#### Return Values

N/A

### 3.1.2.15 setPushToken

Optional method to allow the SDK to use push for incoming calls. If set then the SDK will send the push credentials to the SBC which will allow the SBC to send push messages for incoming calls. Refer to the Democlient for an example.

This method sets the push parameters for the PNS according to:

*Push Notification with the Session Initiation Protocol (SIP)* see: <https://www.ietf.org/id/draft-ietf-sipcore-sip-push-20.txt>

The values are stored in permanent memory and will be used by the WebRTC SDK until the values are reset to null values. It is recommend to call upon this method before calling upon login as calling upon this method after will cause a re-register of the SIP stack.

---

#### Parameters

- **context** [*Context*, Android application context]
- **registrationToken** [*String*, Google push registration token]
- **projectID** [*String*, Google project ID]

---

#### Return Values

N/A

### 3.1.2.16 updatePushToken

This method should be called if the client's registration token has been changed. Refer to the Democlient for an example.

This method is the same as setPushToken except the projectID that is emitted. This method uses the projectID as set in setPushToken. Therefore, setPushToken must have been called once before.

---

#### Parameters

- **context** [*Context*, Android application context]
- **registrationToken** [*String*, Google push registration token]

---

#### Return Values

N/A



### 3.1.2.17 `sendInstantMessage`

This method sends a message to the remote contact. This is done according to RFC 3428 - Session Initiation Protocol (SIP) Extension for Instant Messaging. The status of the message will be returned in `AudioCodesEventListener` (see `onInstantMessageStatus`).

---

**Parameters**

- `contact` [`RemoteContact`, destination address/number]
- `message` [`String`, message that will be sent to the remote contact]

---

**Return Values**

- message ID [long, ID for the message, this message ID will be used in the callback for the message status]

### 3.1.2.18 `setAllowHeader`

Sets/adds a new allow header to the SDP in case of 180 ringing message

---

**Parameters**

User agent [`String[]`, comma separated value for the header]

---

**Return Values**

N/A

## 3.2 AudioCodesSession

Represents a call session. Used in two scenarios:

- When initiating a call via the AudioCodesUA
- When receiving a callback of an incoming call

---

### Syntax

```
class AudioCodesSession {
    int getSessionID();
    void answer(Hashtable<String, String> inviteHeaders, Boolean
withVideo) ;
    void reject(Hashtable<String, String> inviteHeaders);
    void terminate();
    void muteAudio(Boolean mute);
    void muteVideo(Boolean mute);
    boolean isAudioMuted();
    boolean isVideoMuted();
    void sendDTMF(DTMF dtmf);
    void sendInfo(String info);
    Boolean isOutgoing();
    Boolean hasVideo();
    RemoteContact getRemoteNumber();
    CallState getCallState();
    int duration();
    long getCallStartTime();
    CallTermination getTermination();
    Boolean isLocalHold();
    Boolean isRemoteHold();
    Object getUserData();
    void setUSerData(Object object);
    void hold(Boolean hold);
    void switchCamera();
    void showVideo(Activity);
    void showVideo(SurfaceViewRenderer localView,
SurfaceViewRenderer remoteView);
    void stopVideo();
    void setLocalRenderPosition(int xPercentage, int yPercentage);
    void addSessionEventListener(AudioCodesSessionEventListener
sessionEventListener);
    void removeSessionEventListener(AudioCodesSessionEventListener
sessionEventListener);
    ACCallStatistics getStats();
    void redirect(RemoteContact contact, Hashtable<String,String>
inviteHeaders);
    void reinvokeWithVideo();
    Boolean transferCall(RemoteContact transferTo);
    Boolean transferCall(AudioCodesSession transferToSession);
    RemoteContact getTransferContact();
    CallTransferState getTransferState();
}
```

## 3.2.1 Standard Methods

### 3.2.1.1 getSessionID

Retrieves the internal identifier for the session. This identifier can be used in case there is more than one session.

---

**Parameters**

N/A

---

**Return Values**

sessionID [*int*, ID of the session]

### 3.2.1.2 answer

Initiates the object and establishes the call. Only valid for incoming calls.

---

**Parameters**

- headers [*Hashtable<String, String>*, list of headers with a key/value where each key is added as a header to the SIP response with the specified value]
- **AnswerWithVideo** (Boolean):
  - **true**: The call is answered with video and video is unmuted. Both sides see each other
  - **false**: The call is answered with video, but video muted. This side will see the remote video, but the remote side cannot see the local side video

---

**Return Values**

N/A

### 3.2.1.3 reject

Rejects a call. Only valid for incoming calls.

---

**Parameters**

headers [*Hashtable<String, String>*, list of headers with a key/value where each key is added as a header to the SIP response with the specified value]

---

**Return Values**

N/A

### 3.2.1.4 Terminate

Terminates an active call. Only valid for outgoing and established calls.

---

**Parameters**

N/A

---

**Return Values**

N/A

### 3.2.1.5 muteAudio

Defines the status of the audio mute (on/off).

---

**Parameters**

- **MuteAudio** (Boolean):
  - **true**: To mute audio
  - **false**: To unmute audio

---

**Return Values**

N/A

### 3.2.1.6 muteVideo

Defines the status of the video mute (on/off).

---

**Parameters**

- **MuteVideo** (Boolean):
  - **true**: To mute video
  - **false**: To unmute video

---

**Return Values**

N/A

### 3.2.1.7 `isAudioMuted`

Checks audio mute status.

---

**Parameters**

N/A

---

**Return Values**

- **isAudioMuted** (Boolean):
  - **true**: If muted
  - **false**: If not

### 3.2.1.8 `isVideoMuted`

Checks video mute status.

---

**Parameters**

N/A

---

**Return Values**

- **isVideoMuted**(Boolean):
  - **true**: If muted
  - **false**: If not

### 3.2.1.9 `sendDTMF`

Sends a DTMF character

---

**Parameters**

- `dtmf` [*DTMF*, an enum containing a DTMF character]

---

**Return Values**

N/A

### 3.2.1.10 sendInfo

Sends a SIP info message

---

**Parameters**

- info [*String*, message in JSON format]

---

**Return Values**

N/A

### 3.2.1.11 isOutgoing

Checks if a call is outgoing.

---

**Parameters**

N/A

---

**Return Values**

- **IsCallOutgoing** (Boolean):
  - **true**: If outgoing
  - **false**: If incoming

### 3.2.1.12 hasVideo

Checks if a call has video.

---

**Parameters**

N/A

---

**Return Values**

- **IsCallWithVideo** (Boolean):
  - **true**: If the call has video
  - **false**: If the call is audio only

### 3.2.1.13 `getCallState`

Gets the call state of the session.

---

**Parameters**

N/A

---

**Return Values**

- `callstate` [*CallState*, enum containing the call state]

### 3.2.1.14 `getTermination`

Gets the termination reason of the session.

---

**Parameters**

N/A

---

**Return Values**

- `termination` [*CallTermination*, enum containing the termination reason]

### 3.2.1.15 `duration`

Defines the call duration, in seconds. It will be -1 if the call has still not been established.

---

**Parameters**

N/A

---

**Return Values**

- `duration` [*int*, call duration in seconds, -1 if the call is not yet established]

### 3.2.1.16 `isLocalHold`

Checks if this side of the call has put the call on hold.

---

**Parameters**

N/A

---

**Return Values**

- `isCallLocallyOnHold` (Boolean):
  - **true**: If call is on local hold
  - **false**: if otherwise

### 3.2.1.17 isRemoteHold

Checks if the remote side of the call has put the call on hold.

---

**Parameters**

N/A

---

**Return Values**

- **IsCallRemotelyOnHold** (Boolean):
  - **true**: If the call is on local hold
  - **false**: If otherwise

### 3.2.1.18 setUserData

Allows the optional setting of a user-created object / data to be attached to a specific session. The reference is removed from the session after the session has been terminated.

---

**Parameters**

userData [*Object*, Java object for option user data]

---

**Return Values**

N/A

### 3.2.1.19 getUserData

Allows the optional retrieval of a user-created object / data which was attached to a certain session.

---

**Parameters**

N/A

---

**Return Values**

- userData [*Object*, Java object for option user data]



### 3.2.1.20 hold

Set call on hold (or un-hold). The callProgress callback in AudioCodesSessionEventListener indicates when the call has been placed on hold/unhold.

---

**Parameters**

**Hold** [*Boolean*, set call to hold]

---

**Return Values**

N/A

### 3.2.1.21 switchCamera

Switches the camera between front and back camera. This method requires the device to have two cameras. A successful camera switch is returned in the cameraSwitched callback in AudioCodesSessionEventListener.

---

**Parameters**

N/A

---

**Return Values**

N/A

### 3.2.1.22 showVideo (1)

Displays video during a call. This requires the UI element *ac\_webrtc\_video* to be included in the layout of the activity. This is the recommended method to be used when using video. For a more advanced and customized option, see *showVideo(2)* in the section below.

---

**Parameters**

Activity [*Activity*, Android activity that is used to display the video. This activity must contain the *ac\_webrtc\_video* UI element in its content view.

---

**Return Values**

N/A

### 3.2.1.23 showVideo (2)

Displays video during a call. This does not require the UI element `ac_webrtc_video` to be included in the layout of the activity. The method is mutually exclusive to `showVideo (1)`.

---

**Parameters**

- `localView` [`SurfaceViewRender`, Android `SurfaceViewRender`er UI element; this element is used to show the local stream.]
- `remoteView` [`SurfaceViewRender`, Android `SurfaceViewRender`er UI element; this element is used to show the remote stream.]

---

**Return Values**

N/A

### 3.2.1.24 stopVideo

Stops the capturing of video and removes the remote and local renderer. To start video again, `showVideo` needs to be called.

---

**Parameters**

N/A

---

**Return Values**

N/A

### 3.2.1.25 setLocalRenderPosition

Sets the local render position. Only relevant if `showVideo(1)` method is used. This method should be called before `showVideo`.

---

**Parameters**

- `xPercentage` [`int`, the horizontal position of the top left side of the local render view. This is set as a percentage of the `ac_webrtc_video` screen. E.g., if the local render view has a width of 1000 pixels. Setting this item to 66 will place the top left side of the local render view at pixel 660 of the x-axis]
- `yPercentage` [`int`, the vertical position of the top left side of the local render view. This is set as a percentage of the `ac_webrtc_video` screen. For example, if the local render view has a height of 1500 pixels. Setting this item to 66 will place the top left side of the local render view at pixel 1000 of the y-axis]

---

**Return Values**

N/A

### 3.2.1.26 addSessionEventListener

Adds an event listener to listen for session events. The client application might add multiple listeners. The listeners will receive events until they are other removed or the session was terminated.

---

**Parameters**

sessionEventListener[*AudioCodesSessionEventListener*, implementation object of the *AudioCodesSessionEventListener* interface]

---

**Return Values**

N/A

### 3.2.1.27 removeSessionEventListener

Removes a sessionEventListener that was previously added in the addSessionEventListener method.

---

**Parameters**

sessionEventListener[*AudioCodesSessionEventListener*, implementation object of the *AudioCodesSessionEventListener* interface]

---

**Return Values**

N/A

### 3.2.1.28 getStats

Retrieves statistics on the entire call session.

---

**Parameters**

- N/A

---

**Return Values**

ACCallStatistics[*ACCallStatistics*, object containing the call statistics]

### 3.2.1.29 redirect

Redirects an incoming call to a new number. Indicates to the calling party that the caller should try to call the number specified in RemoteContact (see also under [here](#)).

---

**Parameters**

- contact [RemoteContact, redirect destination address/number]
- inviteHeaders [*Hashtable<String, String>*, list of headers with a key/value where each key is added as a header to the SIP INVITE with the specified value]

---

**Return Values**

- session [*AudioCodesSession*, a call session object defined [here](#)]

### 3.2.1.30 reinvokeWithVideo

Enables video during a call. Sends a REINVITE with video enabled. Note that showVideo will need to be called (see code example delivered with the SDK).

---

**Parameters**

N/A

---

**Return Values**

N/A

### 3.2.1.31 transferCall (blind transfer)

Transfers the other side of the current AudioCodesSession to the remote contact supplied.

This is a blind transfer and should be used when there is only one session. Refer to the Democlient for an example.

---

**Parameters**

contact [RemoteContact, transfer destination address/number. Remote Contact to which the other side of the current call will be transferred to].

---

**Return Values**

**TransferCall** (Boolean):

- **true:** If the current call is in connected state
- **false:** In any other case

### 3.2.1.32 transferCall (attended transfer)

Transfers the other side of the current AudioCodesSession to the AudioCodesSession supplied. This is a attended transfer and should be used when there is more than one session. Refer to the Democlient for an example.

---

#### Parameters

transferToSession [AudioCodesSession - Transfers the destination call AudioCodesSession to which the other side of the current call will be transferred to. For example, the other side of the current AudioCodesSession will try to replace this call with a call to the number of the supplied AudioCodesSession.]

---

#### Return Values

- **TransferCall** (Boolean):
  - **true:** If the current call (AudioCodesSession) is on hold or connected and the supplied call (AudioCodesSession) is on hold
  - **false:** In any other case

### 3.2.1.33 getTransferContact

Gets the contact to which the call is being transferred to. After a successful transfer the transfer contact becomes the remote contact (see getRemoteContact).

---

#### Parameters

- NA

---

#### Return Values

- Transfer contact [RemoteContact, the transfer contact. This can be the contact to which the other side of the current call is being transferred to, or the remote contact to which this call is being transferred to.]

### 3.2.1.34 getTransferState

Gets the state of the transfer. This method is only applicable for calls that are in the process of transfer or being transferred. See `getCallState` to know when an `AudioCodesSession` is in the state of transfer.

---

#### Parameters

- N/A

---

#### Return Values

Transfer state [`CallTransferState`, the state of the transfer for this `AudioCodesSession`. Possible states:

- **NONE:** No transfer in progress.
- **TRANSFER\_REQUEST\_RECEIVED\_IN\_PROGRESS:** The other side has sent a transfer request, `getTransferContact` which returns the contact to whom this call is being transferred to.
- **TRANSFER\_REQUEST\_RECEIVED\_FAILED:** The other side has sent a transfer request, but the transfer did not succeed.
- **TRANSFER\_REQUEST\_RECEIVED\_SUCCEEDED:** The other side has sent a transfer request, and the transfer succeeded.
- **TRANSFER\_REQUEST\_SEND\_IN\_PROGRESS:** This side has sent a transfer request which is being processed. `getTransferContact` returns the contact to whom the other side of this call is being transferred to.
- **TRANSFER\_REQUEST\_SEND\_FAILED:** This side has sent a transfer request which has failed.
- **TRANSFER\_REQUEST\_SEND\_SUCCEEDED:** This side has sent a transfer request which has succeeded.
- **TRANSFER\_REPLACED:** This `AudioCodesSession` is in a call with a remote party and the remote party has been replaced (side C in an attended transfer). `getRemoteContact` returns the new number to where the call has been transferred to.

## 3.3 WebRTCAudioManager

This defines WebRTC SDK Audio management. This class handles audio routing during WebRTC calls.

---

### Syntax

```
class AudioCodesSession {
    WebRTCAudioManager getInstance ();
    void setWebRTCAudioRouteListener (WebRTCAudioRouteListener
    listener) ;
    void setAudioRoute (AudioRoute route);
    AudioRoute getAudioRoute ();
    List<AudioRoute> getAvailableAudioRoutes ();
}
```

### 3.3.1 Standard Methods

#### 3.3.1.1 getInstance

Gets the singleton instance of the WebRTCAudioManager class.

---

#### Parameters

N/A

---

#### Return Values

- instance [*WebRTCAudioManager*, singleton object instance]

#### 3.3.1.2 setWebRTCAudioRouteListener

Sets a listener for listening to updates in the current audio route and available audio routes.

There must always be only one listener; setting a new listener will overwrite the previous listener.

---

#### Parameters

listener [*WebRTCAudioRouteListener*, implementation instance of the WebRTCAudioRouteListener interface]

---

#### Return Values

- N/A

### 3.3.1.3 **setAudioRoute**

Sets the audio route. This method changes the audio route of the device. This generally should be used during a call. The audio will be only be routed if the new audio route is available.

For Tablet devices, it is recommended to change to speaker route if available.

---

**Parameters**

audioRoute [*AudioRoute*, enum describing the new audio route]

---

**Return Values**

- **NewAudioRouteFound** (Boolean):
  - **true**: If the new audio route was found
  - **false**: If the new audio route is unavailable

### 3.3.1.4 **getAudioRoute**

Gets the current audio route.

---

**Parameters**

N/A

---

**Return Values**

- audioRoute [*AudioRoute*, enum describing the new audio route]

### 3.3.1.5 **getAvailableAudioRoutes**

Gets the available audio routes.

For Tablet devices, it is recommended to check if a speaker route is available.

---

**Parameters**

N/A

---

**Return Values**

- audioRouteList [*List<AudioRoute>*, list of enums with the available audio routes]



## 3.4 ACConfiguration

Used to provide additional configuration options for the WebRTC SDK. Using this class is optional. The class is a singleton object. The configuration object can be retrieved through `getConfiguration`. Any changes in that object will be applied to the SDK. It is recommended to apply any changes before calling the `AudioCodesUA` login method.

```
Class ACConfiguration{
    ACConfiguration getConfiguration() ;
    String version() ;
    int getLocalServerPort() ;
    void setLocalServerPort(int port);
    DTMFOptions getDTMFOptions() ;
    void setDTMFOptions(DTMFOptions dtmfOptions);
    VideoConfiguration getVideoConfiguration() ;
    void setVideoConfiguration(VideoConfiguration configuration);
    void setAutomaticCallOnRedirect(Boolean automaticRedirect);
    Boolean getAutomaticCallOnRedirect() ;
    void setRedirect(Boolean redirect, RemoteContact
    redirectContact);
    RemoteContact getRedirectContact() ;
    boolean getRedirectEnabled() ;
}
```

### 3.4.1 Standard Methods

#### 3.4.1.1 `getConfiguration`

Defines the static method that returns the current used configuration object.

---

**Parameters**

N/A

---

**Return Values**

- configuration [*ACConfiguration*, current used configuration object; see Section 3.4]

#### 3.4.1.2 `version`

Defines the static method that returns the current version of the SDK.

---

**Parameters**

N/A

---

**Return Values**

- Version [*String*, version of the SDK, e.g., 1.x]

### 3.4.1.3 `getLocalServerPort`

Gets the current default local port used by the SIP stack.

---

**Parameters**

N/A

---

**Return Values**

- port [*integer*, default local user port (default 6000)]

### 3.4.1.4 `setLocalServerPort`

Changes the default local SIP server port from the default value (6000).

---

**Parameters**

- port [*integer*, default local server port]

---

**Return Values**

N/A

### 3.4.1.5 `getDtmfOptions`

Gets the current default local port used by the SIP stack

---

**Parameters**

N/A

---

**Return Values**

dtmfOptions [*DTMFOptions*, DTMFOptions class for setting the handling of DTMF tones; the default value is for the WebRTC to handle DTMF tones]

### 3.4.1.6 `setDtmfOptions`

Changes the DTMFOptions class used by the SDK. This allows for sending DTMF through either the WebRTC or SIP INFO. The class allows for changing DTMF duration and interval (if applicable for the chosen method). See Section 3.6 for more information.

---

**Parameters**

- dtmfOptions [*DTMFOptions*, DTMFOptions class for setting the handling of DTMF tones]

---

**Return Values**

N/A

### 3.4.1.7 `getVideoConfiguration`

Gets the current video configuration used by the SDK.

---

**Parameters**

N/A

---

**Return Values**

- configuration [*VideoConfiguration*, class containing video configuration options]

### 3.4.1.8 `setVideoConfiguration`

Changes the video configurations options used by the SDK. See also Section 3.5.

---

**Parameters**

- configuration [*VideoConfiguration*, class containing video configuration options]

---

**Return Values**

N/A

### 3.4.1.9 `setAutomaticCallOnRedirect`

Sets automatic call redirection for outgoing call redirection. If enabled, the SDK will automatically attempt to call the number supplied in the 3XX response on the INVITE.

If not, the call will be terminated, and the redirect number will be provided in the callProgress (redirect) callback. See the code example delivered with the SDK. In this case, the application using the SDK will be responsible for placing a new call to the new number, if necessary.

---

**Parameters**

- **CallRedirected** (Boolean):
  - **true**: automatic redirect is enabled
  - **false**: automatic redirect is disabled

---

**Return Values**

N/A

### 3.4.1.10 `getAutomaticCallOnRedirect`

Gets the current setting for automatic call redirection.

---

#### Parameters

N/A

---

#### Return Values

**IsCallRedirected** (Boolean):

- **true**: Automatic redirect is enabled
- **false**: Redirect is disabled

### 3.4.1.11 `setRedirect`

This method allows for the setting of incoming call redirection. If enabled, *all* incoming calls will be answered with a 302 response and the redirect contact provided in the `redirectContact` field. If the `redirect` field is enabled, there will be no incoming call callbacks

---

#### Parameters

- `redirect` [*Boolean*, if true, *all* incoming call redirection is enabled; if not, then redirect is disabled]
- `redirectContact` [*RemoteContact*, the contact to whom the call should be redirected. Only username is mandatory. The domain and scheme fields will be taken from the user account if not provided.

---

#### Return Values

N/A

### 3.4.1.12 `getRedirectContact`

Gets the current contact that was set for incoming call redirection

---

#### Parameters

N/A

---

#### Return Values

- Redirect contact [*RemoteContact*, current contact set for incoming call redirection]

### 3.4.1.13 `getRedirectEnabled`

Gets the current setting for incoming call redirection.

---

**Parameters**

N/A

---

**Return Values**

**IncomingCallRedirected** (Boolean):

- **true**: Incoming call redirect is enabled
- **false**: Incoming call redirect is disabled

## 3.5 VideoConfiguration

Used to provide additional configuration options for the WebRTC SDK. Using this class is optional. The class provides access to public parameters that can be changed if needed.

The configuration object can be retrieved through `getVideoConfiguration` in `ACConfiguration` class. Calling `setVideoConfiguration` in the `ACConfiguration` class will apply the changes. It is recommend to call `setVideoConfiguration` before `showVideo` is called.

```
Class VideoConfiguration{  
    }  
}
```

### 3.5.1 Camera Parameters

- **cameraWidth** – captures the width of the camera (default 640)
- **cameraHeight** - captures the height of the camera (default 480)
- **cameraFrameRate** - captures the framerate of the camera (default 15)

### 3.5.2 Rendering Views Parameters

- **LOCAL\_X\_CONNECTING** – Uppermost left corner (% of the screen width) of the local video screen when a remote video stream is not yet available.
- **LOCAL\_Y\_CONNECTING** - Uppermost left corner (% of the screen height) of the local video screen when a remote video stream is not yet available.
- **LOCAL\_WIDTH\_CONNECTING** – Lowermost right corner (% of the screen width) of the local video screen when a remote video stream is not yet available.
- **LOCAL\_HEIGHT\_CONNECTING** – Lowermost right corner (% of the screen height) of the local video screen when a remote video stream is not yet available.
- **LOCAL\_X\_CONNECTED** - Uppermost left corner (% of the screen width) of the local video screen when a remote video stream is available.
- **LOCAL\_Y\_CONNECTED** - Uppermost left corner (% of the screen height) of the local video screen when a remote video stream is available.
- **LOCAL\_WIDTH\_CONNECTED** – Lowermost right corner (% of the screen width) of the local video screen when a remote video stream is available.
- **LOCAL\_HEIGHT\_CONNECTED** – Lowermost right corner (% of the screen height) of the local video screen when a remote video stream is available.
- **REMOTE\_X** - Uppermost left corner (% of the screen width) of the remote video screen when a remote video stream is available.
- **REMOTE\_Y** - Uppermost left corner (% of the screen height) of the remote video screen when a remote video stream is available.
- **REMOTE\_WIDTH** - Lowermost right corner (% of the screen width) of the remote video screen when a remote video stream is available
- **REMOTE\_HEIGHT** - Lowermost right corner (% of the screen height) of the remote video screen when a remote video stream is available

## 3.6 DTMFOptions

Used to provide additional configuration options for the WebRTC SDK. Using this class is optional. The class provides access to public parameters that can be changed if needed. The class allows configuration of sending DTMF events.

```
Class DTMFOptions{  
    }  
}
```

### 3.6.1 DTMF Parameters

- **dtmfMethod** - DTMFMethod enum parameter that supports sending of DTMF through:
  - **WEBRTC** - DTMF is sent through media by telephone-event using the WebRTC engine. This is the default method.
  - **SIP\_INFO** - DTMF events are sent through SIP\_INFO events.
- **duration** - duration of the DTMF event, in milliseconds. When using SIP\_INFO, the minimum is 100, which is the default value.
- **intervalGap** - the interval gap in milliseconds between sending DTMF events. This is only relevant for WEBRTC DTMF events. Default: 70.

## 3.7 RemoteContact

Used to represent a remote contact. This can be either a dialed number or a remote contact received through an incoming call.

```
Class RemoteContact{
    String getDisplayname();
    String getUsername();
    String getDomain();
    void setDisplayname(String displayName);
    void setUsername(String username);
    void setDomain(String domain);
}
```

### 3.7.1 Standard Methods

#### 3.7.1.1 `getDisplayname`

Gets the contact display name. Note that this might not be available since the remote contact did not set a display name.

---

**Parameters**

N/A

---

**Return Values**

- `displayName` [*String*, display name of the remote contact]

#### 3.7.1.2 `getUsername`

Gets the contact user name.

---

**Parameters**

N/A

---

**Return Values**

`userName` [*String*, user name of the remote contact]

#### 3.7.1.3 `getDomain`

Gets the contact domain.

---

**Parameters**

N/A

---

**Return Values**

- `domain` [*String*, domain of the remote contact]



### 3.7.1.4 `setDisplay_name`

(Optional) Sets the contact display name. Since this does not affect SIP signaling, it's optional; it allows for easier retrieval of the display name used in the call.

---

**Parameters**

- `displayName` [*String*, display name of the remote contact]

---

**Return Values**

N/A

### 3.7.1.5 `setUsername`

Sets the contact user name.

---

**Parameters**

- `userName` [*String*, user name of the remote contact]

---

**Return Values**

N/A

### 3.7.1.6 `setDomain`

(Optional) Sets the contact domain.

---

**Parameters**

`domain` [*String*, domain of the remote contact. This value doesn't usually have to be set as the remote contact is likely to reside in the same domain]

---

**Return Values**

N/A

## 3.8 TerminationInfo

This section describes Call Termination information.

```
Class TerminationInfo{
    CallTermination;
    int statusCode;
    String reason;
    String reasonHeader;
    String sipMessage;
```

### 3.8.1 TerminationInfo attribute

- **CallTermination** – enum indicating the reason for the termination. This is a simplified version of the status code and should be enough for most call cases
- **statusCode** – SIP status code of the termination (e.g. 404 or 500, etc...). This can be useful in specific use cases
- **reason** – SIP reason, e.g. Server Internal Error or User Not Found
- **reasonHeader** – SIP reason header – describes a specific reason for the call termination, e.g. in case of 500 Server Internal Error, the reason header could be: Reason: SIP ; cause=500; tet="General Routing Failure"
- **sipMessage** – the last sip message, this will be mostly be used for debugging or for future uses

## 3.9 InfoAlert

This section describes Call Termination information.

```
Class InfoAlert{
    boolean autoAnswer;
    int delay;
```

### 3.9.1 InfoAlert attribute

- **autoAnswer** – Boolean indicating whether the call should be answered automatically after the *delay* property value
- **delay** – The amount of time, in seconds, that needs to pass before the call is answered automatically.

## 4 API Callbacks/ Listeners Interfaces

The API provides capability to register in order to listen to different types of events. Here's a list of the interfaces that must be implemented to receive an event:

### 4.1 AudioCodesEventListener

Interface for receiving SDK events. This interface must be implemented and set through the AudioCodesUA class to receive these event.

#### 4.1.1 Login state changed event

Triggered when the login state has been changed.

---

**Syntax**

```
void loginStateChanged(Boolean isLogin, String cause);
```

---

**Parameters**

- isLogin [*Boolean*, 'true' if logged in and 'false' if not logged in]
- cause [*String*, text describing the received SIP reason. This can be mostly used if more information on a login failure is required]

#### 4.1.2 Incoming call event

Triggered when receiving an incoming call.

---

**Syntax**

```
void incomingCall(AudioCodesSession call, InfoAlert infoAlert);
```

---

**Parameters**

- Session [*AudioCodesSession*, the incoming call session object]
- infoAlert [*InfoAlert*, the incoming call auto answer object]

#### 4.1.3 Incoming IM message event

Triggered when receiving an incoming message.

---

**Syntax**

```
void onIncomingInstantMessage(RemoteContact contact, String message);
```

---

**Parameters**

- contact [*RemoteContact*, remote contact that send the message]
- message [*String*, the message that the remote contact send]

### 4.1.4 IM Message status event

When a message has been sent, the status of the message can be tracked using the callback `onInstantMessageStatus`.

---

#### Syntax

```
void onInstantMessageStatus (InstanceMessageStatus status, long id);
```

---

#### Parameters

- `status` [*InstanceMessageStatus*, status of the message. This is an enum with the following values:
  - **SUCCESS** – the message has been delivered to the remote contact
  - **ACCEPTED** – the message has been delivered to the server (remote contact might not have received the message yet though)
  - **NOT\_FOUND** – the remote contact does not exist or the server is not able to deliver the message to the remote contact
  - **UNKNOWN\_ERROR** – an unknown error occurred, the message was not delivered
- Message ID [long, the ID of the message, this corresponds to the ID used in `sendInstantMessage`]

## 4.2 AudioCodesSessionEventListener

### 4.2.1 callTerminated

Callback for when the session is terminated by the local or the remote side.

---

**Syntax**

```
void callTerminated(AudioCodesSession session, TerminationInfo info);
```

---

**Parameters**

- `session` [*AudioCodesSession* refers to the call session object that was terminated. The object will be removed at the end of the `callTerminated` method]
- `info` [*TerminationInfo* – Refers to the object containing information about the termination reason. e.g., termination reason, status code,. See [TerminationInfo](#).

### 4.2.2 callProgress

Callback for changes in the state of the call. The call progress state can be retrieved by `getCallState` in the `AudioCodesSession` object.

---

**Syntax**

```
void callProgress(AudioCodesSession call);
```

---

**Parameters**

- `session` [*AudioCodesSession*, the call session object]

### 4.2.3 cameraSwitched

Callback for when the camera has been switched between the front or the back camera.

---

**Syntax**

```
void cameraSwitched(Boolean frontCamera);
```

---

**Parameters**

**FrontCamera** (Boolean):

- **true**: The camera has switched to the front camera
- **false**: The camera has switched to the back camera

---

**Return Values**

- N/A

### 4.2.4 reinvokeWithVideoCallback

Callback for when a video is added during a call. This callback can be used to call `showVideo` (from the UI thread). See the code example delivered with the SDK.

---

**Syntax**

```
void reinvokeWithVideoCallback(AudioCodesSession session);
```

---

**Parameters**

- `session` [*AudioCodesSession*, the call session object]

### 4.2.5 mediaFailed

Callback for a failure in the media. This callback is for information purposes only and can be used to inform the user that there currently is no media. This can be most useful in the case of a network change, where a media fail might be expected.

---

**Syntax**

```
void mediaFailed(AudioCodesSession call);
```

---

**Parameters**

- `session` [*AudioCodesSession*, the call session object]

### 4.2.6 incoming Notify

Callback for when an incoming notify state is received, video is added during the call. This callback can be used to execute the incoming notify state (from the UI thread if needed). See the code example delivered with the SDK.

---

**Syntax**

```
void incomingNotify(NotifyEvent notifyEvent, String dtmfValue);
```

---

**Parameters**

- `notifyEvent` [*NotifyEvent*, the call notify event type, This is an enum with the following values:
  - **TALK**– If the call is incoming and is not answered yet, then the client application is required to answer the call. If the call is already active and on hold, then the client application is required to un-hold it.
  - **HOLD**– If the call is active, then the client application is required to put the call on hold.
  - **DTMF**– The client application is required to send DTMF characters provided in the *dtmf* string parameter. This should be performed using calls to the `sendDtmf` method consecutively for each character in the DTMF string, and in a way that is non-blocking to the current thread. The interval between sending each DTMF character should be `MAX(DTMFOptions.intervalGap, DTMFOptions.duration)`.
  - **CONFERENCE** – Currently not supported
- **MESSAGE – Incoming notification for a SIP message**`dtmfValue` [*String*, the call dtmf value or message body, null in case `notifyEvent` is not DTMF or **MESSAGE**]

## 4.3 WebRTCAudioRouteListener

Interface for receiving audio routes events. The interface must be implemented and set through the WebRTCAudioManager class to receive these events.

### 4.3.1 audioRoutesChanged

Callback for when the list of available audio routes has been changed, for example, if the user is connected to a Bluetooth audio device

---

**Syntax**

```
void audioRoutesChanged (List<WebRTCAudioManager.AudioRoute>  
audioRouteList);
```

---

**Parameters**

- audioRouteList [*List<WebRTCAudioManager.AudioRoute>*, a list of available audio routes]

### 4.3.2 currentAudioRouteChanged

Callback for when the currently used audio route has been changed. If the user adds a Bluetooth audio device, for example, the SDK routes the audio to the Bluetooth device and this callback will be called.

---

**Syntax**

```
void currentAudioRouteChanged (WebRTCAudioManager.AudioRoute  
newAudioRoute);
```

---

**Parameters**

newAudioRoute [*WebRTCAudioManager.AudioRoute*, the new audio route where the audio is being routed to]

## 5 Use Examples

Here are some user examples for your reference.

### 5.1 User Agent: Create Instance, Set Server and Account

```
AudioCodesUA phone = new AudioCodesUA(); // phone API
ArrayList<PeerConnection.IceServer> iceServerList = new
ArrayList<PeerConnection.IceServer>();
phone.setServerConfig("webrtclab.audiocodes.com", 5080,
"example.com", Transport.TCP, iceServerList);
phone.setAccount("John", "*****", "John Smit", "jsmit");
```

### 5.2 User Agent: Set Listeners (Callbacks)

```
phone.setListener(new AudioCodesEventListener() {
    @Override
    public void loginStateChanged(Boolean isLogin, String cause) {
// place your code here (remember that this is being called
//on the WebRTC SDK thread, not on the UI thread)
    }

    @Override
    public void incomingCall(AudioCodesSession session, InfoAlert
infoAlert) {
// place your code here (remember that this is being called
//on the WebRTC SDK thread, not on the UI thread)
    }
});
```

### 5.3 User Agent login: Connection to SBC Server and Login

```
phone.login(getApplicationContext()); // getApplicationContext is
an Android method available for Activities and Services
```

### 5.4 Make a Call

```
Boolean withVideo = true;
AudioCodesSession activeCall = phone.call("jane", withVideo,
null);
```

### 5.5 Send DTMF During a Call

```
activeCall.sendDTMF(DTMF.NINE);
```

### 5.6 Send SIP Message During a Call

```
activeCall.sendInfo(String);
```

### 5.7 Mute / Unmute During a Call

```
activeCall.muteAudio(true);
```



```
activeCall.muteAudio(false);
```

## 5.8 Accept Incoming Call (with video)

```
incomingCall.answer(null, true);
```

## 5.9 Reject Incoming Call

```
incomingCall.reject(null);
```

## 5.10 Terminate a Call

```
activeCall.terminate();
```

## 5.11 Use of Video

Include the `ac_webrtc_video` UI element in the XML file for your call activity:

```
<include
    android:id="@+id/my_ac_webrtc_video"
    layout="@layout/ac_webrtc_video"
    android:layout_width="match_parent"
    android:layout_height="match_parent" />
```

Update the position of your local render screen:

```
activeCall.setLocalRenderPosition(70, 60);
```

Call `showVideo` with your call activity; the WebRTC SDK will locate the `ac_webrtc_video` UI element and use it to display the remote and the local video:

```
activeCall.showVideo(CallActivity.this);
```

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