WebRTC Android Client SDK API Reference Guide
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Abbreviations and Terminology

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

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Documentation Feedback

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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.

Note: 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 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
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2 Android SDK

2.1 Before you Start

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 5.0

Provided with the SDK:

- **WebRTC SDK AAR file.** This is the WebRTC SDK aar file. The file must be included in a project in order to use the SDK.
- **Android demo client project.** This is the Android Studio project that can be opened from 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 used as a reference to the WebRTC SDK.

2.2 Installation

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

1. **To install the SDK:**
   1. Install Android Studio
   2. Open a project for the demo client
   3. Follow instructions from Android Studio (necessary Android SDK files will need to be downloaded and installed.)
   4. Disable instant run in Android Studio. (Android Studio 3.1 has a bug where it will not compile correctly.)
      a. Disable instant run in Android studio because of Android Studio issue: https://issuetracker.google.com/issues/72811718
   5. Make sure the AAR file / module is included (see demo client build.gradle file). For example
      a. implementation project(':webrtcsdk-release')
   6. Make sure the WebRTC files are included:
      a. implementation 'org.webrtc:google-webrtc:1.0.+'
      b. The current WebRTC supported SDK is 1.0.22920
   7. The WebRTC SDK is currently required to add the following lines to the application build.gradle files (see Demo client):
      ```
      ndk {
        abiFilters 'armeabi', 'armeabi-v7a'
        abiFilters 'armeabi-v7a'
      }
      ```
   8. The SDK will need certain Android permissions. The SDK will not check if these permissions are present for the Android application. Any Android application that uses the SDK will need to make sure the permission is requested and provided by the user. The demo client provides examples on how to do this.
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3 API Classes

The API consists of:

Main Classes
- AudioCodesUA – Audio Codes User Agent (Singleton) – see Section 3.1
- AudioCodesSession – for call representation (Interface) – see Section 3.2
- WebRTCAudioManager – class for managing audio routes – see Section 3.3

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 transport, List<PeerConnection.IceServer> iceServerList);
    void setAccount (String userName, String password, String displayName);
    void setAccount (String userName, String password, String displayName, String authName);
    void setListeners (AudioCodesEventListener listener);
    void login(Context context);
    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);
    }

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]

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

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]
- password [String, authenticating user password]
- displayName [String, display name for the user]
- 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_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 logout

Performs de-registration from the service.

**Parameters**
- N/A

**Return Values**
- N/A

### 3.1.1.8 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.]
3.1.2 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.

**Note:** 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.

**Note:** 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**
- use [boolean, if false then the session timers will not be enabled, otherwise (default value) session timers will be optionally supported. E.g., the SBC will initiate session timers if configured to do so]

**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.

*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
List of sessions [ArrayList<AudiocodesSession>, list of active sessions]

3.1.2.12 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.

Default: false

Parameters

- enable [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.

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

```java
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);
    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 addSessionEvenListener(AudioCodesSessionEventListener sessionEventListener);
    void removeSessionEvenListener(AudioCodesSessionEventListener sessionEventListener);
    ACCallStatistics getStats();
    void redirect(RemoteContact contact, Hashtable<String,String> inviteHeaders);
    void reinviteWithVideo();
}
```
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]
- withvideo [boolean, True - the call will be answered with video and video unmuted. Both sides will see each other. False - the call will be 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
- Mute [boolean, value. ‘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
- mute [boolean, value; ‘True’ to mute video; ‘False’ to unmute video]

Return Values
N/A

3.2.1.7 isAudioMute

Checks audio mute status.

Parameters
N/A

Return Values
- mute [boolean, ‘True’ if mute, ‘False’ if not]

3.2.1.8 isVideoMute

Checks video mute status.

Parameters
N/A

Return Values
- mute [boolean, ‘True’ if mute, ‘False’ if not]
3.2.1.9 **sendDTMF**

Sends a DTMF character

**Parameters**
- dtmf \([DTMF, \text{an enum containing a DTMF character}]\)

**Return Values**
N/A

3.2.1.10 **isOutgoing**

Checks if a call is outgoing.

**Parameters**
N/A

**Return Values**
- \([\text{boolean}, \text{‘True’ if outgoing, ‘False’ if incoming.}]\)

3.2.1.11 **hasVideo**

Checks if a call has video.

**Parameters**
N/A

**Return Values**
- \([\text{boolean}, \text{‘True’ if the call has video, ‘False’ if the call is audio only.}]\)

3.2.1.12 **getCallState**

Gets the call state of the session.

**Parameters**
N/A

**Return Values**
- callstate \([\text{CallState, enum containing the call state}]\)
3.2.1.13 getTermination

Gets the termination reason of the session.

Parameters
N/A

Return Values
- termination [CallTermination, enum containing the termination reason]

3.2.1.14 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]

3.2.1.15 isLocalHold

Parameters
N/A

Return Values
- [boolean 'True' if call is on local hold, 'False' if otherwise]

3.2.1.16 isRemoteHold

Parameters
N/A

Return Values
- [boolean 'True' if the call is on local hold, 'False' if otherwise]
3.2.1.17 **setUserData**

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

**Parameters**

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

**Return Values**

N/A

3.2.1.18 **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]

3.2.1.19 **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

3.2.1.20 **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
3.2.1.21 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

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

**Return Values**
N/A

3.2.1.23 stopVideo

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

**Parameters**
N/A

**Return Values**
N/A
3.2.1.24 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. E.g., 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

3.2.1.25 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 otherwise removed or the session was terminated.

Parameters

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

Return Values

N/A

3.2.1.26 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
3.2.1.27 getStats

Retrieves statistics on the entire call session.

Parameters
N/A

Return Values
- ACCallStatistics[ACCallStatistics, object containing the call statistics]

3.2.1.28 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]

3.2.1.29 reinviteWithVideo

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
3.3 WebRTC Audio Manager

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

Syntax
```java
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.

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

Return Values
- [Boolean, true – if the new audio route was found
false – if the new audio route is not available ]

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.

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.

```java
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
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
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
- automatic redirect [boolean, if True, automatic redirect will be enabled; otherwise redirect will be disabled]

Return Values
N/A

3.4.1.10 getAutomaticCallOnRedirect

Gets the current setting for automatic call redirection.

Parameters
N/A

Return Values
- automatic redirect [boolean, if True, automatic redirect will be enabled; otherwise redirect will be 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
- incoming call redirect [boolean, if True, then incoming call redirection is enabled; if not, then incoming call redirect is disabled]
3.5 Video Configuration

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 DTMF Options

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 Remote Contact

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  setDisplayName
(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
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 in order to receive an event:

4.1 AudioCodes Event Listener

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

4.1.1 LoginStateChanged

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 incomingCall

Triggered when receiving an incoming call.

Syntax
void incomingCall(AudioCodesSession call);

Parameters
- Session [AudioCodesSession, the incoming call session object]
4.2 AudioCodes Session Event Listener

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

**Syntax**
```java
void callTerminated(AudioCodesSession session);
```

**Parameters**
- `session [AudioCodesSession, the call session object that was terminated. The object will be removed at the of the callTerminated method]`

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**
```java
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**
```java
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]`
4.2.4 **reinviteWithvideoCallback**

Callback for when 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**

```java
void reinviteWithVideoCallback(AudiocodesSession session);
```

**Parameters**

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

4.3 **Web RTC Audio Route Listener**

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**

```java
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. E.g., if the user adds a Bluetooth audio device, the SDK will route the audio to the Bluetooth device and this callback will be called.

**Syntax**

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

**Parameters**

- `newAudioRoute` [WebRTCAudioManager.AudioRoute, the new audio route where the audio is being routed to]
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5 Use Examples

Here are some use 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) {
        // 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 Call

activeCall.sendDTMF(DTMF.NINE);

5.6 Mute / Unmute During Call

activeCall.muteAudio(true);
activeCall.muteAudio(false);

5.7 Accept Incoming Call (with Video)

incomingCall.answer(null, true);
5.8  Reject Incoming Call

incomingCall.reject(null);

5.9  Terminate a Call

activeCall.terminate();

5.10 Use of Video

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

```xml
<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:

```java
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:

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