

PortSIP VoIP SDK Manual for Android

Version 18.1
5/21/21

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Welcome to PortSIP VoIP SDK For Android

Create your SIP-based application for multiple platforms (iOS, Android, Windows, Mac OS and Linux) with our SDK.

The rewarding PortSIP VoIP SDK is a powerful and versatile set of tools that dramatically accelerate SIP application development. It includes a suite of stacks, SDKs, and some Sample projects, with each of them enables developers to combine all the necessary components to create an ideal development environment for every application's specific needs.

The PortSIP VoIP SDK complies with IETF and 3GPP standards, and is IMS-compliant (3GPP/3GPP2, TISPAN and PacketCable 2.0). These high performance SDKs provide unified API layers for full user control and flexibility.

Getting Started

You can download PortSIP VoIP SDK Sample projects at our [Website](#). Samples include demos for VC++, C#, VB.NET, Delphi XE, Xcode (for iOS and Mac OS), Android Studio with the sample project source code provided (with SDK source code exclusive). The sample projects demonstrate how to create a powerful SIP application with our SDK easily and quickly.

Contents

The sample package for downloading contains almost all of materials for PortSIP VoIP SDK: documentation, Dynamic/Static libraries, sources, headers, datasheet, and everything else a SDK user might need!

SDK User Manual

To be started with, it is recommended to read the documentation of PortSIP VoIP SDK, [SDK User Manual page](#), which gives a brief description of each API function.

Website

Some general interest or often changing PortSIP SDK information will be posted on the [PortSIP website](#) in real time. The release contains links to the site, so while browsing you may see occasional broken links if you are not connected to the Internet. To be sure everything needed for using the PortSIP VoIP SDK has been contained within the release.

Support

Please send email to our [Support team](#) if you need any help.

Installation Prerequisites

To use PortSIP VoIP/IMS SDK for Android for development, SDK version with later than API-16 is required.

Frequently Asked Questions

1. Where can I download the PortSIP VoIP SDK for test?

All sample projects of the PortSIP VoIP SDK can be found and downloaded at:
<https://www.portsip.com/download-portsip-voip-sdk/>
<https://www.PortSIP.com/portsip-voip-sdk>

2. How can I compile the sample project?

1. Download the sample project from PortSIP website.
2. Extract the .zip file.
3. Open the project by your Android studio:
4. Compile the sample project directly.

3. How can I create a new project with PortSIP VoIP SDK?

1. Download the sample project and evaluation SDK and extract it to a specified directory
2. Run Android Studio and create a new Android Application Project
3. Copy all files from libs directory under extracted directory to the libs directory of your new application.
4. Import the dependent class from the SDK. For example: import [com.portsip.OnPortSIPEvent](#);
import [com.portsip.PortSipSdk](#);
5. Inherit the interface OnPortSIPEvent to process the callback events.
6. Initialize SDK. For example: mPortSIPSDK = new PortSipSdk();
mPortSIPSDK.setOnPortSIPEvent(instanceofOnPortSIPEvent);
mPortSIPSDK.CreateCallManager(context); mPortSIPSDK.initialize(...); For more details please refer to the Sample project source code.

4. How can I test the P2P call (without SIP server)?

1. Download and extract the SDK sample project ZIP file into local. Compile and run the "P2PSample" project.
2. Run the P2PSample on two devices. For example, run it on device A and device B, and IP address for A is 192.168.1.10, IP address for B is 192.168.1.11.
3. Enter a user name and password on A. For example, enter user name 111, and password aaa (you can enter anything for the password as the SDK will ignore it). Enter a user name and password on B. For example, enter user name 222, and password aaa.
4. Click the "Initialize" button on A and B. If the default port 5060 is already in use, the P2PSample will prompt "Initialize failure". In case of this, please click the "Uninitialize" button and change the local port, and click the "Initialize" button again.
5. The log box will appear "Initialized" if the SDK is successfully initialized.
6. To make call from A to B, enter "sip:222@192.168.1.11" and click "Dial" button; while to make call from B to A, enter "sip:111@192.168.1.10".

Note: If the local sip port is changed to other port, for example, A is using local port 5080, and B is using local port 6021, to make call from A to B, please enter "sip:222@192.168.1.11:6021" and dial; while to make call from B to A, enter "sip:111@192.168.1.10:5080".

5. Is the SDK thread safe?

Yes, the SDK is thread safe. You can call any of the API functions without the need to consider the multiple threads.

Note: the SDK allows to call API functions in callback events directly - except for the "onAudioRawCallback", "onVideoRawCallback", "onRTPPacketCallback" callbacks.

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Modules

Here is a list of all modules:

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Hierarchical Index

Class Hierarchy

This inheritance list is sorted roughly, but not completely, alphabetically:

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com.portsip.PortSipErrorcode	69
CapturerObserver	
com.portsip.PortSIPCameraCapturer	66
SurfaceViewRenderer	
com.portsip.PortSIPVideoRenderer	76

Class Index

Class List

Here are the classes, structs, unions and interfaces with brief descriptions:

<u>com.portsip.PortSipEnumDefine.AUDIOCODEC</u>	62
<u>com.portsip.PortSipEnumDefine.AudioDevice</u>	63
<u>com.portsip.OnPortSIPEvent</u>	64
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<u>com.portsip.PortSipSdk</u>	72
<u>com.portsip.PortSIPVideoRenderer</u>	76

Module Documentation

SDK Callback events

SDK Callback events.

Modules

- [Register events](#)
Register events.
 - [Call events](#)
 - [Refer events](#)
 - [Signaling events](#)
 - [MWI events](#)
 - [DTMF events](#)
 - [INFO/OPTIONS message events](#)
 - [Presence events](#)
 - [audio device changed, Play audio and video file finished events](#)
 - [RTP callback events](#)
-

Detailed Description

SDK Callback events.

Register events

Register events.

Functions

- void [com.portsip.OnPortSIPEvent.onRegisterSuccess](#) (String *reason*, int *code*, String *sipMessage*)
 - void [com.portsip.OnPortSIPEvent.onRegisterFailure](#) (String *reason*, int *code*, String *sipMessage*)
-

Detailed Description

Register events.

Function Documentation

void com.portsip.OnPortSIPEvent.onRegisterSuccess (String *reason*, int *code*, String *sipMessage*)

When successfully registered to server, this event will be triggered.

Parameters

<i>reason</i>	The status text.
<i>code</i>	The status code.
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onRegisterFailure (String *reason*, int *code*, String *sipMessage*)

If failed to register to SIP server, this event will be triggered.

Parameters

<i>reason</i>	The status text.
<i>code</i>	The status code.
<i>sipMessage</i>	The SIP message received.

Call events

Functions

- void [com.portsip.OnPortSIPEvent.onInviteIncoming](#) (long *sessionId*, String *callerDisplayName*, String *caller*, String *calleeDisplayName*, String *callee*, String *audioCodecs*, String *videoCodecs*, boolean *existsAudio*, boolean *existsVideo*, String *sipMessage*)
- void [com.portsip.OnPortSIPEvent.onInviteTrying](#) (long *sessionId*)
- void [com.portsip.OnPortSIPEvent.onInviteSessionProgress](#) (long *sessionId*, String *audioCodecs*, String *videoCodecs*, boolean *existsEarlyMedia*, boolean *existsAudio*, boolean *existsVideo*, String *sipMessage*)
- void [com.portsip.OnPortSIPEvent.onInviteRinging](#) (long *sessionId*, String *statusText*, int *statusCode*, String *sipMessage*)
- void [com.portsip.OnPortSIPEvent.onInviteAnswered](#) (long *sessionId*, String *callerDisplayName*, String *caller*, String *calleeDisplayName*, String *callee*, String *audioCodecs*, String *videoCodecs*, boolean *existsAudio*, boolean *existsVideo*, String *sipMessage*)
- void [com.portsip.OnPortSIPEvent.onInviteFailure](#) (long *sessionId*, String *reason*, int *code*, String *sipMessage*)
- void [com.portsip.OnPortSIPEvent.onInviteUpdated](#) (long *sessionId*, String *audioCodecs*, String *videoCodecs*, String *screenCodecs*, boolean *existsAudio*, boolean *existsVideo*, boolean *existsScreen*, String *sipMessage*)
- void [com.portsip.OnPortSIPEvent.onInviteConnected](#) (long *sessionId*)
- void [com.portsip.OnPortSIPEvent.onInviteBeginingForward](#) (String *forwardTo*)
- void [com.portsip.OnPortSIPEvent.onInviteClosed](#) (long *sessionId*, String *sipMessage*)
- void [com.portsip.OnPortSIPEvent.onDialogStateUpdated](#) (String *BLFMonitoredUri*, String *BLFDialogState*, String *BLFDialogId*, String *BLFDialogDirection*)
- void [com.portsip.OnPortSIPEvent.onRemoteHold](#) (long *sessionId*)
- void [com.portsip.OnPortSIPEvent.onRemoteUnHold](#) (long *sessionId*, String *audioCodecs*, String *videoCodecs*, boolean *existsAudio*, boolean *existsVideo*)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onInviteIncoming (long *sessionId*, String *callerDisplayName*, String *caller*, String *calleeDisplayName*, String *callee*, String *audioCodecs*, String *videoCodecs*, boolean *existsAudio*, boolean *existsVideo*, String *sipMessage*)

When a call is coming, this event will be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>callerDisplayName</i>	The display name of caller

<i>e</i>	
<i>caller</i>	The caller.
<i>calleeDisplayName</i>	The display name of callee.
<i>e</i>	
<i>callee</i>	The callee.
<i>audioCodecs</i>	The matched audio codecs. It's separated by "#" if there are more than one codecs.
<i>videoCodecs</i>	The matched video codecs. It's separated by "#" if there are more than one codecs.
<i>existsAudio</i>	By setting to true, it means that this call include the audio.
<i>existsVideo</i>	By setting to true, it means that this call include the video.
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onInviteTrying (long *sessionId*)

If the outgoing call is being processed, this event will be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
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void com.portsip.OnPortSIPEvent.onInviteSessionProgress (long *sessionId*, String *audioCodecs*, String *videoCodecs*, boolean *existsEarlyMedia*, boolean *existsAudio*, boolean *existsVideo*, String *sipMessage*)

Once the caller received the "183 session progress" message, this event would be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>audioCodecs</i>	The matched audio codecs. It's separated by "#" if there are more than one codecs.
<i>videoCodecs</i>	The matched video codecs. It's separated by "#" if there are more than one codecs.
<i>existsEarlyMedia</i>	By setting to true it means the call has early media.
<i>existsAudio</i>	By setting to true it means this call include the audio.
<i>existsVideo</i>	By setting to true it means this call include the video.
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onInviteRinging (long *sessionId*, String *statusText*, int *statusCode*, String *sipMessage*)

If the outgoing call is ringing, this event will be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>statusText</i>	The status text.
<i>statusCode</i>	The status code.
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onInviteAnswered (long *sessionId*, String *callerDisplayName*, String *caller*, String *calleeDisplayName*, String *callee*, String *audioCodecs*, String *videoCodecs*, boolean *existsAudio*, boolean *existsVideo*, String *sipMessage*)

If the remote party answered the call, this event would be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>callerDisplayName</i>	The display name of caller
<i>caller</i>	The caller.
<i>calleeDisplayName</i>	The display name of callee.

<i>e</i>	
<i>callee</i>	The callee.
<i>audioCodecs</i>	The matched audio codecs. It's separated by "#" if there are more than one codecs.
<i>videoCodecs</i>	The matched video codecs. It's separated by "#" if there are more than one codecs.
<i>existsAudio</i>	By setting to true, this call includes the audio.
<i>existsVideo</i>	By setting to true, this call includes the video.
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onInviteFailure (long *sessionId*, String *reason*, int *code*, String *sipMessage*)

This event will be triggered if the outgoing call fails.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>reason</i>	The failure reason.
<i>code</i>	The failure code.
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onInviteUpdated (long *sessionId*, String *audioCodecs*, String *videoCodecs*, String *screenCodecs*, boolean *existsAudio*, boolean *existsVideo*, boolean *existsScreen*, String *sipMessage*)

This event will be triggered when remote party updates the call.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>audioCodecs</i>	The matched audio codecs. It's separated by "#" if there are more than one codecs.
<i>videoCodecs</i>	The matched video codecs. It's separated by "#" if there are more than one codecs.
<i>screenCodecs</i>	The matched screen codecs. It's separated by "#" if there are more than one codecs.
<i>existsAudio</i>	By setting to true, this call includes the audio.
<i>existsVideo</i>	By setting to true, this call includes the video.
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onInviteConnected (long *sessionId*)

This event will be triggered when UAC sent/UAS received ACK (the call is connected). Some functions (hold, updateCall etc...) can be called only after the call connected, otherwise the functions will return error.

Parameters

<i>sessionId</i>	The session ID of the call.
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void com.portsip.OnPortSIPEvent.onInviteBeginingForward (String *forwardTo*)

If the enableCallForward method is called and a call is incoming, the call will be forwarded automatically and this event will be triggered.

Parameters

<i>forwardTo</i>	The target SIP URI of the call forwarding.
------------------	--

void com.portsip.OnPortSIPEvent.onInviteClosed (long *sessionId*, String *sipMessage*)

This event is triggered once remote side ends the call.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onDialogStateUpdated (String *BLFMonitoredUri*, String *BLFDialogState*, String *BLFDialogId*, String *BLFDialogDirection*)

If a user subscribed and his dialog status monitored, when the monitored user is holding a call or is being rang, this event will be triggered.

Parameters

<i>BLFMonitoredUri</i>	the monitored user's URI
<i>BLFDialogState</i>	- the status of the call
<i>BLFDialogId</i>	- the id of the call
<i>BLFDialogDirection</i>	- the direction of the call

void com.portsip.OnPortSIPEvent.onRemoteHold (long *sessionId*)

If the remote side places the call on hold, this event will be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
------------------	-----------------------------

void com.portsip.OnPortSIPEvent.onRemoteUnHold (long *sessionId*, String *audioCodecs*, String *videoCodecs*, boolean *existsAudio*, boolean *existsVideo*)

If the remote side un-holds the call, this event will be triggered

Parameters

<i>sessionId</i>	The session ID of the call.
<i>audioCodecs</i>	The matched audio codecs. It's separated by "#" if there are more than one codec.
<i>videoCodecs</i>	The matched video codecs. It's separated by "#" if there are more than one codec.
<i>existsAudio</i>	By setting to true, this call includes the audio.
<i>existsVideo</i>	By setting to true, this call includes the video.

Refer events

Functions

- void [com.portsip.OnPortSIPEvent.onReceivedRefer](#) (long sessionId, long referId, String to, String from, String referSipMessage)
- void [com.portsip.OnPortSIPEvent.onReferAccepted](#) (long sessionId)
- void [com.portsip.OnPortSIPEvent.onReferRejected](#) (long sessionId, String reason, int code)
- void [com.portsip.OnPortSIPEvent.onTransferTrying](#) (long sessionId)
- void [com.portsip.OnPortSIPEvent.onTransferRinging](#) (long sessionId)
- void [com.portsip.OnPortSIPEvent.onACTVTransferSuccess](#) (long sessionId)
- void [com.portsip.OnPortSIPEvent.onACTVTransferFailure](#) (long sessionId, String reason, int code)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onReceivedRefer (long *sessionId*, long *referId*, String *to*, String *from*, String *referSipMessage*)

This event will be triggered once received a REFER message.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>referId</i>	The ID of the REFER message. Pass it to <code>acceptRefer</code> or <code>rejectRefer</code>
<i>to</i>	The refer target.
<i>from</i>	The sender of REFER message.
<i>referSipMessage</i>	The SIP message of "REFER". Pass it to "acceptRefer" function.

void com.portsip.OnPortSIPEvent.onReferAccepted (long *sessionId*)

This callback will be triggered once remote side calls "acceptRefer" to accept the REFER

Parameters

<i>sessionId</i>	The session ID of the call.
------------------	-----------------------------

void com.portsip.OnPortSIPEvent.onReferRejected (long *sessionId*, String *reason*, int *code*)

This callback will be triggered once remote side calls "rejectRefer" to reject the REFER

Parameters

<i>sessionId</i>	The session ID of the call.
<i>reason</i>	Reject reason.
<i>code</i>	Reject code.

void com.portsip.OnPortSIPEvent.onTransferTrying (long *sessionId*)

When the refer call is being processed, this event will be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
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void com.portsip.OnPortSIPEvent.onTransferRinging (long *sessionId*)

When the refer call is ringing, this event will be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
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void com.portsip.OnPortSIPEvent.onACTVTransferSuccess (long *sessionId*)

When the refer call succeeds, this event will be triggered. The ACTV means Active. For example, A establishes the call with B, A transfers B to C, C accepts the refer call, and A will receive this event.

Parameters

<i>sessionId</i>	The session ID of the call.
------------------	-----------------------------

void com.portsip.OnPortSIPEvent.onACTVTransferFailure (long *sessionId*, String *reason*, int *code*)

When the refer call fails, this event will be triggered. The ACTV means Active. For example, A establish the call with B, A transfers B to C, C rejects this refer call, and A will receive this event.

Parameters

<i>sessionId</i>	The session ID of the call.
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<i>reason</i>	The error reason.
<i>code</i>	The error code.

Signaling events

Functions

- void [com.portsip.OnPortSIPEvent.onReceivedSignaling](#) (long sessionId, String message)
- void [com.portsip.OnPortSIPEvent.onSendingSignaling](#) (long sessionId, String message)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onReceivedSignaling (long sessionId, String message)

This event will be triggered when receiving a SIP message. This event is disabled by default. To enable, use enableCallbackSignaling.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>message</i>	The received SIP message.

void com.portsip.OnPortSIPEvent.onSendingSignaling (long sessionId, String message)

This event will be triggered when sent a SIP message. This event is disabled by default. To enable, use enableCallbackSignaling.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>message</i>	The sent SIP message.

MWI events

Functions

- void [com.portsip.OnPortSIPEvent.onWaitingVoiceMessage](#) (String messageAccount, int urgentNewMessageCount, int urgentOldMessageCount, int newMessageCount, int oldMessageCount)
- void [com.portsip.OnPortSIPEvent.onWaitingFaxMessage](#) (String messageAccount, int urgentNewMessageCount, int urgentOldMessageCount, int newMessageCount, int oldMessageCount)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onWaitingVoiceMessage (String *messageAccount*, int *urgentNewMessageCount*, int *urgentOldMessageCount*, int *newMessageCount*, int *oldMessageCount*)

If there is the waiting voice message (MWI), this event will be triggered.

Parameters

<i>messageAccount</i>	Voice message account
<i>urgentNewMessageCount</i>	Count of new urgent messages.
<i>urgentOldMessageCount</i>	Count of history urgent message.
<i>newMessageCount</i>	Count of new messages.
<i>oldMessageCount</i>	Count of history messages.

void com.portsip.OnPortSIPEvent.onWaitingFaxMessage (String *messageAccount*, int *urgentNewMessageCount*, int *urgentOldMessageCount*, int *newMessageCount*, int *oldMessageCount*)

If there is waiting fax message (MWI), this event will be triggered.

Parameters

<i>messageAccount</i>	Fax message account
<i>urgentNewMessageCount</i>	Count of new urgent messages.
<i>urgentOldMessageCount</i>	Count of history urgent messages.
<i>newMessageCount</i>	Count of new messages.
<i>oldMessageCount</i>	Count of old messages.

DTMF events

Functions

- void [com.portsip.OnPortSIPEvent.onRecvDtmfTone](#) (long *sessionId*, int *tone*)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onRecvDtmfTone (long *sessionId*, int *tone*)

This event will be triggered when receiving a DTMF tone from remote side.

Parameters

<i>sessionId</i>	Session ID of the call.
<i>tone</i>	

code	Description
0	The DTMF tone 0.
1	The DTMF tone 1.
2	The DTMF tone 2.

3	The DTMF tone 3.
4	The DTMF tone 4.
5	The DTMF tone 5.
6	The DTMF tone 6.
7	The DTMF tone 7.
8	The DTMF tone 8.
9	The DTMF tone 9.
10	The DTMF tone *.
11	The DTMF tone #.
12	The DTMF tone A.
13	The DTMF tone B.
14	The DTMF tone C.
15	The DTMF tone D.
16	The DTMF tone FLASH.

INFO/OPTIONS message events

Functions

- void [com.portsip.OnPortSIPEvent.onRecvOptions](#) (String optionsMessage)
- void [com.portsip.OnPortSIPEvent.onRecvInfo](#) (String infoMessage)
- void [com.portsip.OnPortSIPEvent.onRecvNotifyOfSubscription](#) (long subscribeId, String notifyMessage, byte[] messageData, int messageDataLength)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onRecvOptions (String optionsMessage)

This event will be triggered when receiving the OPTIONS message.

Parameters

<i>optionsMessage</i>	The received whole OPTIONS message in text format.
-----------------------	--

void com.portsip.OnPortSIPEvent.onRecvInfo (String infoMessage)

This event will be triggered when receiving the INFO message.

Parameters

<i>infoMessage</i>	The whole INFO message received in text format.
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void com.portsip.OnPortSIPEvent.onRecvNotifyOfSubscription (long subscribeId, String notifyMessage, byte[] messageData, int messageDataLength)

This event will be triggered when receiving a NOTIFY message of the subscription.

Parameters

<i>subscribeId</i>	The ID of SUBSCRIBE request.
<i>notifyMessage</i>	The received INFO message in text format.
<i>messageData</i>	The received message body. It's can be either text or binary data.

<i>messageDataLength</i> <i>h</i>	The length of "messageData".
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Presence events

Functions

- void [com.portsip.OnPortSIPEvent.onPresenceRecvSubscribe](#) (long subscribeId, String fromDisplayName, String from, String subject)
- void [com.portsip.OnPortSIPEvent.onPresenceOnline](#) (String fromDisplayName, String from, String stateText)
- void [com.portsip.OnPortSIPEvent.onPresenceOffline](#) (String fromDisplayName, String from)
- void [com.portsip.OnPortSIPEvent.onRecvMessage](#) (long sessionId, String mimeType, String subMimeType, byte[] messageData, int messageDataLength)
- void [com.portsip.OnPortSIPEvent.onRecvOutOfDialogMessage](#) (String fromDisplayName, String from, String toDisplayName, String to, String mimeType, String subMimeType, byte[] messageData, int messageDataLength, String sipMessage)
- void [com.portsip.OnPortSIPEvent.onSendMessageSuccess](#) (long sessionId, long messageId)
- void [com.portsip.OnPortSIPEvent.onSendMessageFailure](#) (long sessionId, long messageId, String reason, int code)
- void [com.portsip.OnPortSIPEvent.onSendOutOfDialogMessageSuccess](#) (long messageId, String fromDisplayName, String from, String toDisplayName, String to)
- void [com.portsip.OnPortSIPEvent.onSendOutOfDialogMessageFailure](#) (long messageId, String fromDisplayName, String from, String toDisplayName, String to, String reason, int code)
- void [com.portsip.OnPortSIPEvent.onSubscriptionFailure](#) (long subscribeId, int statusCode)
- void [com.portsip.OnPortSIPEvent.onSubscriptionTerminated](#) (long subscribeId)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onPresenceRecvSubscribe (long *subscribeId*, String *fromDisplayName*, String *from*, String *subject*)

This event will be triggered when receiving the SUBSCRIBE request from a contact.

Parameters

<i>subscribeId</i>	The ID of SUBSCRIBE request.
<i>fromDisplayName</i>	The display name of contact.
<i>from</i>	The contact who sends the SUBSCRIBE request.
<i>subject</i>	The subject of the SUBSCRIBE request.

void com.portsip.OnPortSIPEvent.onPresenceOnline (String *fromDisplayName*, String *from*, String *stateText*)

When the contact is online or changes presence status, this event will be triggered.

Parameters

<i>fromDisplayName</i>	The display name of contact.
<i>from</i>	The contact who sends the SUBSCRIBE request.
<i>stateText</i>	The presence status text.

void com.portsip.OnPortSIPEvent.onPresenceOffline (String *fromDisplayName*, String *from*)

When the contact is offline, this event will be triggered.

Parameters

<i>fromDisplayName</i>	The display name of contact.
<i>from</i>	The contact who sends the SUBSCRIBE request

void com.portsip.OnPortSIPEvent.onRecvMessage (long *sessionId*, String *mimeType*, String *subMimeType*, byte[] *messageData*, int *messageDataLength*)

This event will be triggered when receiving a MESSAGE message in dialog.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>mimeType</i>	The message mime type.
<i>subMimeType</i>	The message sub mime type.
<i>messageData</i>	The received message body. It can be text or binary data. Use the mimeType and subMimeType to differentiate them. For example, if the mimeType is "text" and subMimeType is "plain", "messageData" is text message body. If the mimeType is "application" and subMimeType is "vnd.3gpp.sms", "messageData" is binary message body.
<i>messageDataLength</i>	The length of "messageData".

void com.portsip.OnPortSIPEvent.onRecvOutOfDialogMessage (String *fromDisplayName*, String *from*, String *toDisplayName*, String *to*, String *mimeType*, String *subMimeType*, byte[] *messageData*, int *messageDataLength*, String *sipMessage*)

This event will be triggered when receiving a MESSAGE message out of dialog. For example pager message.

Parameters

<i>fromDisplayName</i>	The display name of sender.
<i>from</i>	The message sender.
<i>toDisplayName</i>	The display name of receiver.
<i>to</i>	The receiver.
<i>mimeType</i>	The message mime type.
<i>subMimeType</i>	The message sub mime type.
<i>messageData</i>	The received message body. It can be text or binary data. Use the mimeType and subMimeType to differentiate them. For example, if the mimeType is "text" and subMimeType is "plain", "messageData" is text message body. If the mimeType is "application" and subMimeType is "vnd.3gpp.sms", "messageData" is binary message body.
<i>messageDataLength</i>	The length of "messageData".
<i>sipMessage</i>	The SIP message received.

void com.portsip.OnPortSIPEvent.onSendMessageSuccess (long *sessionId*, long *messageId*)

If the message is sent successfully in dialog, this event will be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>messageId</i>	The message ID. It's equal to the return value of sendMessage function.

void com.portsip.OnPortSIPEvent.onSendMessageFailure (long *sessionId*, long *messageId*, String *reason*, int *code*)

If the message is failed to be sent out of dialog, this event will be triggered.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>messageId</i>	The message ID. It's equal to the return value of sendMessage function.
<i>reason</i>	The failure reason.
<i>code</i>	Failure code.

void com.portsip.OnPortSIPEvent.onSendOutOfDialogMessageSuccess (long *messageId*, String *fromDisplayName*, String *from*, String *toDisplayName*, String *to*)

If the message is sent successfully out of dialog, this event will be triggered.

Parameters

<i>messageId</i>	The message ID. It's equal to the return value of SendOutOfDialogMessage function.
<i>fromDisplayName</i>	The display name of message sender.
<i>from</i>	The message sender.
<i>toDisplayName</i>	The display name of message receiver.
<i>to</i>	The message receiver.

void com.portsip.OnPortSIPEvent.onSendOutOfDialogMessageFailure (long *messageId*, String *fromDisplayName*, String *from*, String *toDisplayName*, String *to*, String *reason*, int *code*)

If the message failed to be sent out of dialog, this event would be triggered.

Parameters

<i>messageId</i>	The message ID. It's equal to the return value of SendOutOfDialogMessage function.
<i>fromDisplayName</i>	The display name of message sender
<i>from</i>	The message sender.
<i>toDisplayName</i>	The display name of message receiver.
<i>to</i>	The message receiver.
<i>reason</i>	The failure reason.
<i>code</i>	The failure code.

void com.portsip.OnPortSIPEvent.onSubscriptionFailure (long *subscribeId*, int *statusCode*)

This event will be triggered on sending SUBSCRIBE failure.

Parameters

<i>subscribeId</i>	The ID of SUBSCRIBE request.
<i>statusCode</i>	The status code.

void com.portsip.OnPortSIPEvent.onSubscriptionTerminated (long *subscribeId*)

This event will be triggered when a SUBSCRIPTION is terminated or expired.

Parameters

<i>subscribeId</i>	The ID of SUBSCRIBE request.
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audio device changed, Play audio and video file finished events

Functions

- void [com.portsip.OnPortSIPEvent.onPlayAudioFileFinished](#) (long sessionId, String fileName)
- void [com.portsip.OnPortSIPEvent.onPlayVideoFileFinished](#) (long sessionId)
- void [com.portsip.OnPortSIPEvent.onAudioDeviceChanged](#) (PortSipEnumDefine.AudioDevice audioDevice, Set< PortSipEnumDefine.AudioDevice > devices)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onPlayAudioFileFinished (long sessionId, String fileName)

If called playAudioFileToRemote function with no loop mode, this event will be triggered once the file play finished.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>fileName</i>	The play file name.

void com.portsip.OnPortSIPEvent.onPlayVideoFileFinished (long sessionId)

If called playVideoFileToRemote function with no loop mode, this event will be triggered once the file play finished.

Parameters

<i>sessionId</i>	The session ID of the call.
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void com.portsip.OnPortSIPEvent.onAudioDeviceChanged (PortSipEnumDefine.AudioDevice audioDevice, Set< PortSipEnumDefine.AudioDevice > devices)

fired When available audio devices changed or audio device currently in use changed.

Parameters

<i>audioDevice</i>	device currently in use
<i>devices</i>	devices useable. If a wired headset is connected, it should be the only possible option. When no wired headset connected, the devices set may contain speaker, earpiece, Bluetooth devices. can be set by PortSipSdk#setAudioDevice to switch to current device.

RTP callback events

Functions

- void [com.portsip.OnPortSIPEvent.onRTPPacketCallback](#) (long sessionId, int mediaType, int enum_direction, byte[] RTPPacket, int packetSize)

- void [com.portsip.OnPortSIPEvent.onAudioRawCallback](#) (long sessionId, int enum_direction, byte[] data, int dataLength, int samplingFreqHz)
- void [com.portsip.OnPortSIPEvent.onVideoRawCallback](#) (long sessionId, int enum_direction, int width, int height, byte[] data, int dataLength)

Detailed Description

Function Documentation

void com.portsip.OnPortSIPEvent.onRTPPacketCallback (long sessionId, int mediaType, int enum_direction, byte[] RTPPacket, int packetSize)

If enableRtpCallback function is called to enable the RTP callback, this event will be triggered once a RTP packet is received or sent.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>mediaType</i>	RTP packet media type, 0 for audio, 1 for video , 2 for screen.
<i>enum_direction</i>	RTP packet direction ENUM_DIRECTION_SEND , ENUM_DIRECTION_RECV .
<i>RTPPacket</i>	The size of the RTP packet.
<i>packetSize</i>	The size of the received RTP packet. Remarks

Donot call any SDK API functions in this event directly. If you want to call the API functions or other code which is time-consuming, you should post a message to another thread and execute SDK API functions or other code in another thread.

void com.portsip.OnPortSIPEvent.onAudioRawCallback (long sessionId, int enum_direction, byte[] data, int dataLength, int samplingFreqHz)

This event will be triggered once receiving the audio packets if called [enableAudioStreamCallback](#) function.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>enum_direction</i>	The type passed in enableAudioStreamCallback function. Below types allowed: ENUM_DIRECTION_SEND , ENUM_DIRECTION_RECV .
<i>data</i>	The memory of audio stream. It's in PCM format.
<i>dataLength</i>	The data size.
<i>samplingFreqHz</i>	The audio stream sample in HZ. For example, 8000 or 16000. Remarks

Don't call any SDK API functions in this event directly. If you want to call the API functions or other code which is time-consuming, you should post a message to another thread and execute SDK API functions or other code in another thread.

See also

[PortSipSdk::enableAudioStreamCallback](#)

void com.portsip.OnPortSIPEvent.onVideoRawCallback (long sessionId, int enum_direction, int width, int height, byte[] data, int dataLength)

This event will be triggered once receiving the video packets if [enableVideoStreamCallback](#) function is called.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>enum direction</i>	The type which is passed in enableVideoStreamCallback function. Below

	types allowed: ENUM_DIRECTION_SEND , ENUM_DIRECTION_RECV .
<i>width</i>	The width of video image.
<i>height</i>	The height of video image.
<i>data</i>	The memory of video stream. It's in YUV420 format, YV12.
<i>dataLength</i>	The data size.

See also

[PortSipSdk::enableVideoStreamCallback](#)

SDK functions

Modules

- [Initialize and register functions](#)
- [Audio and video codecs functions](#)
- [Additional settings functions](#)
- [Access SIP message header functions](#)
- [Audio and video functions](#)
- [Call functions](#)
- [Refer functions](#)
- [Send audio and video stream functions](#)
- [RTP packets, Audio stream and video stream callback functions](#)
- [Record functions](#)
- [Play audio and video file to remote functions](#)
- [Conference functions](#)
- [RTP and RTCP QOS functions](#)
- [RTP statistics functions](#)
- [Audio effect functions](#)
- [Send OPTIONS/INFO/MESSAGE functions](#)

Detailed Description

Initialize and register functions

Functions

- void [com.portsip.PortSipSdk.CreateCallManager](#) (Context context)
- PortSipEnumDefine.AudioDevice [com.portsip.PortSipSdk.getSelectedAudioDevice](#) ()
- void [com.portsip.PortSipSdk.DeleteCallManager](#) ()
- int [com.portsip.PortSipSdk.initialize](#) (int enum_transport, String localIP, int localSIPPort, int enum_LogLevel, String LogPath, int maxLines, String agent, int audioDeviceLayer, int videoDeviceLayer, String TLSCertificatesRootPath, String TLSCipherList, boolean verifyTLSCertificate, String dnsServers)
- int [com.portsip.PortSipSdk.setInstanceId](#) (String instanceId)
- int [com.portsip.PortSipSdk.setUser](#) (String userName, String displayName, String authName, String password, String userDomain, String SIPServer, int SIPServerPort, String STUNServer, int STUNServerPort, String outboundServer, int outboundServerPort)
- void [com.portsip.PortSipSdk.removeUser](#) ()
remove user account info.
- int [com.portsip.PortSipSdk.registerServer](#) (int expires, int retryTimes)

- int [com.portsip.PortSipSdk.refreshRegistration](#) (int expires)
- int [com.portsip.PortSipSdk.unregisterServer](#) ()
- int [com.portsip.PortSipSdk.setDisplayName](#) (String displayName)
- int [com.portsip.PortSipSdk.setLicenseKey](#) (String key)

Detailed Description

Function Documentation

void com.portsip.PortSipSdk.CreateCallManager (Context context)

Create the callback handlers.

Parameters

<i>context</i>	The context of application.
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PortSipEnumDefine.AudioDevice com.portsip.PortSipSdk.getSelectedAudioDevice ()

Returns

the audio output device current used. this function is Deprecated.the better way to get current used and available audio devices is [OnAudioManagerEvents](#)

void com.portsip.PortSipSdk.DeleteCallManager ()

Release the callback Handlers.

int com.portsip.PortSipSdk.initialize (int enum_transport, String localIP, int localSIPPort, int enum_LogLevel, String LogPath, int maxLines, String agent, int audioDeviceLayer, int videoDeviceLayer, String TLSCertificatesRootPath, String TLSCipherList, boolean verifyTLSCertificate, String dnsServers)

Initialize the SDK.

Parameters

<i>enum_transport</i>	Transport for SIP signaling, which can be set as: ENUM_TRANSPORT_UDP , ENUM_TRANSPORT_TCP , ENUM_TRANSPORT_TLS , ENUM_TRANSPORT_PERS . The ENUM_TRANSPORT_PERS_UDP is the private PortSIP transport for anti-SIP blocking, which must work with the PERS. ENUM_TRANSPORT_PERS . The ENUM_TRANSPORT_PERS_TCP is the private PortSIP transport for anti-SIP blocking, which must work with the PERS.
<i>localIP</i>	The local PC IP address (for example: 192.168.1.108). It will be used for sending and receiving SIP messages and RTP packets. If the local IP is provided in IPv6 format, the SDK will use IPv6. If you want the SDK to choose correct network interface (IP) automatically, please use "0.0.0.0" for IPv4, or ":::" for IPv6.
<i>localSIPPort</i>	The listening port for SIP message transmission, for example 5060.
<i>enum_LogLevel</i>	Set the application log level. The SDK will generate "PortSIP_Log_datetime.log" file if the log is enabled. ENUM_LOG_LEVEL_NONE ENUM_LOG_LEVEL_DEBUG ENUM_LOG_LEVEL_ERROR ENUM_LOG_LEVEL_WARNING ENUM_LOG_LEVEL_INFO ENUM_LOG_LEVEL_DEBUG
<i>LogPath</i>	The path for storing log file. The path (folder) specified MUST be existent.
<i>maxLines</i>	Theoretically, unlimited count of lines are supported depending on the device

	capability. For SIP client, it is recommended to limit it as ranging 1 - 100.
<i>agent</i>	The User-Agent header to be inserted in to SIP messages.
<i>audioDeviceLayer</i>	Specifies the audio device layer that should be using: 0 = Use the OS defaulted device. 1 = Virtual device, usually use this for the device that has no sound device installed.
<i>videoDeviceLayer</i>	Specifies the video device layer that should be using: 0 = Use the OS defaulted device. 1 = Use Virtual device, usually use this for the device that has no camera installed.
<i>TLSCertificatesRootPath</i>	Specify the TLS certificate path, from which the SDK will load the certificates automatically. Note: On Windows, this path will be ignored, and SDK will read the certificates from Windows certificates stored area instead.
<i>TLSCipherList</i>	Specify the TLS cipher list. This parameter is usually passed as empty so that the SDK will offer all available ciphers.
<i>verifyTLSCertificate</i>	Indicate if SDK will verify the TLS certificate or not. By setting to false, the SDK will not verify the validity of TLS certificate.
<i>dnsServers</i>	Additional Nameservers DNS servers. Value null indicates system DNS Server. Multiple servers will be split by ";", e.g "8.8.8.8;8.8.4.4"

Returns

If the function succeeds, it returns value 0. If the function fails, it will return a specific error code

int com.portsip.PortSipSdk.setInstanceId (String *instanceId*)

Set the instance Id, the outbound instanceId((RFC5626)) used in contact headers.

Parameters

<i>instanceId</i>	The SIP instance ID. If this function is not called, the SDK will generate an instance ID automatically. The instance ID MUST be unique on the same device (device ID or IMEI ID is recommended). Recommend to call this function to set the ID on Android devices.
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Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setUser (String *userName*, String *displayName*, String *authName*, String *password*, String *userDomain*, String *SIPServer*, int *SIPServerPort*, String *STUNServer*, int *STUNServerPort*, String *outboundServer*, int *outboundServerPort*)

Set user account info.

Parameters

<i>userName</i>	Account (username) of the SIP, usually provided by an IP-Telephony service provider.
<i>displayName</i>	The name displayed. You can set it as your like, such as "James Kend". It's optional.
<i>authName</i>	Authorization user name (usually equal to the username).
<i>password</i>	User's password. It's optional.
<i>userDomain</i>	User domain; this parameter is optional, which allows to transfer an empty string if you are not using the domain.
<i>SIPServer</i>	SIP proxy server IP or domain, for example xx.xxx.xx.x or sip.xxx.com.
<i>SIPServerPort</i>	Port of the SIP proxy server, for example 5060.
<i>STUNServer</i>	Stun server for NAT traversal. It's optional and can be used to transfer empty string to disable STUN.
<i>STUNServerPort</i>	STUN server port. It will be ignored if the outboundServer is empty.
<i>outboundServer</i>	Outbound proxy server, for example sip.domain.com. It's optional and allows

	to transfer an empty string if not using the outbound server.
<i>outboundServerPort</i>	Outbound proxy server port, it will be ignored if the outboundServer is empty.

Returns

If this function succeeds, it will return value 0. If it fails, it will return a specific error code.

void com.portsip.PortSipSdk.removeUser ()

remove user account info.

int com.portsip.PortSipSdk.registerServer (int *expires*, int *retryTimes*)

Register to SIP proxy server (login to server)

Parameters

<i>expires</i>	Time interval for registration refreshment, in seconds. The maximum of supported value is 3600. It will be inserted into SIP REGISTER message headers.
<i>retryTimes</i>	The maximum of retry attempts if failed to refresh the registration. By setting to <= 0, the attempt will be disabled and onRegisterFailure callback will be triggered when facing retry failure.

Returns

If this function succeeds, it will return value 0. If fails, it will return a specific error code.

If the registration to server succeeds, onRegisterSuccess will be triggered; otherwise onRegisterFailure will be triggered.

int com.portsip.PortSipSdk.refreshRegistration (int *expires*)

Refresh the registration manually after successfully registered.

Parameters

<i>expires</i>	Time interval for registration refreshment, in seconds. The maximum of supported value is 3600. It will be inserted into SIP REGISTER message headers.
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Returns

If this function succeeds, it will return value 0. If fails, it will return a specific error code.

If the registration to server succeeds, onRegisterSuccess will be triggered; otherwise onRegisterFailure will be triggered.

int com.portsip.PortSipSdk.unregisterServer ()

Un-register from the SIP proxy server.

Returns

If this function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setDisplayName (String *displayName*)

Set the display name of user.

Parameters

<i>displayName</i>	That will appear in the From/To Header.
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Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setLicenseKey (String key)

Set the license key. It must be called before setUser function.

Parameters

<i>key</i>	The SDK license key. Please purchase from PortSIP.
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Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Audio and video codecs functions

Functions

- int [com.portsip.PortSipSdk.addAudioCodec](#) (int enum_audiocodec)
- int [com.portsip.PortSipSdk.addVideoCodec](#) (int enum_videocodec)
- boolean [com.portsip.PortSipSdk.isAudioCodecEmpty](#) ()
- boolean [com.portsip.PortSipSdk.isVideoCodecEmpty](#) ()
- int [com.portsip.PortSipSdk.setAudioCodecPayloadType](#) (int enum_audiocodec, int payloadType)
- int [com.portsip.PortSipSdk.setVideoCodecPayloadType](#) (int enum_videocodec, int payloadType)
- void [com.portsip.PortSipSdk.clearAudioCodec](#) ()
- void [com.portsip.PortSipSdk.clearVideoCodec](#) ()
- int [com.portsip.PortSipSdk.setAudioCodecParameter](#) (int enum_audiocodec, String sdpParameter)
- int [com.portsip.PortSipSdk.setVideoCodecParameter](#) (int enum_videocodec, String sdpParameter)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.addAudioCodec (int enum_audiocodec)

Enable an audio codec, and it will be shown in SDP.

Parameters

<i>enum_audiocodec</i>	Audio codec type, including: ENUM_AUDIOCODEC_G729 , ENUM_AUDIOCODEC_PCMA , ENUM_AUDIOCODEC_PCMU , ENUM_AUDIOCODEC_GSM , ENUM_AUDIOCODEC_G722 , ENUM_AUDIOCODEC_ILBC , ENUM_AUDIOCODEC_AMR , ENUM_AUDIOCODEC_AMRWB , ENUM_AUDIOCODEC_SPEEX , ENUM_AUDIOCODEC_SPEEXWB , ENUM_AUDIOCODEC_ISACWB , ENUM_AUDIOCODEC_ISACSWB , ENUM_AUDIOCODEC_OPUS , ENUM_AUDIOCODEC_DTMF .
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Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.addVideoCodec (int *enum_videocodec*)

Enable a video codec, and it will be shown in SDP.

Parameters

<i>enum_videocodec</i>	Video codec type. Supported types include ENUM_VIDEOCODEC_H264 , ENUM_VIDEOCODEC_VP8 , ENUM_VIDEOCODEC_VP9 .
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Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

boolean com.portsip.PortSipSdk.isAudioCodecEmpty ()

Detect if the audio codecs are enabled.

Returns

If no audio codec enabled, it will return value true; otherwise it returns false.

boolean com.portsip.PortSipSdk.isVideoCodecEmpty ()

Detect if the video codecs are enabled.

Returns

If no video codec enabled, it will return value true; otherwise it returns false.

int com.portsip.PortSipSdk.setAudioCodecPayloadType (int *enum_audiocodec*, int *payloadType*)

Set the RTP payload type for dynamic audio codec.

Parameters

<i>enum_audiocodec</i>	Audio codec types. Supported types include: ENUM_AUDIOCODEC_G729 , ENUM_AUDIOCODEC_PCMA , ENUM_AUDIOCODEC_PCMU , ENUM_AUDIOCODEC_GSM , ENUM_AUDIOCODEC_G722 , ENUM_AUDIOCODEC_ILBC , ENUM_AUDIOCODEC_AMR , ENUM_AUDIOCODEC_AMRWB , ENUM_AUDIOCODEC_SPEEX , ENUM_AUDIOCODEC_SPEEXWB , ENUM_AUDIOCODEC_ISACWB , ENUM_AUDIOCODEC_ISACSWB , ENUM_AUDIOCODEC_OPUS , ENUM_AUDIOCODEC_DTMF
<i>payloadType</i>	The new RTP payload type that you want to set.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setVideoCodecPayloadType (int *enum_videocodec*, int *payloadType*)

Set the RTP payload type for dynamic video codec.

Parameters

<i>enum_videocodec</i>	Video codec type. Supported types include: ENUM_VIDEOCODEC_H264 , ENUM_VIDEOCODEC_VP8 , ENUM_VIDEOCODEC_VP9 .
<i>payloadType</i>	The new RTP payload type that you want to set.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.clearAudioCodec ()

Remove all the enabled audio codecs.

void com.portsip.PortSipSdk.clearVideoCodec ()

Remove all the enabled video codecs.

int com.portsip.PortSipSdk.setAudioCodecParameter (int *enum_audiocodec*, String *sdpParameter*)

Set the codec parameter for audio codec.

Parameters

<i>enum_audiocodec</i>	Audio codec type. Supported types include: ENUM_AUDIOCODEC_G729 , ENUM_AUDIOCODEC_PCMA , ENUM_AUDIOCODEC_PCMU , ENUM_AUDIOCODEC_GSM , ENUM_AUDIOCODEC_G722 , ENUM_AUDIOCODEC_ILBC , ENUM_AUDIOCODEC_AMR , ENUM_AUDIOCODEC_AMRWB , ENUM_AUDIOCODEC_SPEEX , ENUM_AUDIOCODEC_SPEEXWB , ENUM_AUDIOCODEC_ISACWB , ENUM_AUDIOCODEC_ISACSWB , ENUM_AUDIOCODEC_OPUS , ENUM_AUDIOCODEC_DTMF
<i>sdpParameter</i>	The parameter is in string format.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

See also

[PortSipEnumDefine](#)

Remarks

Example:

```
setAudioCodecParameter(AUDIOCODEC_AMR, "mode-set=0; octet-align=1; robust-sorting=0")
```

int com.portsip.PortSipSdk.setVideoCodecParameter (int *enum_videocodec*, String *sdpParameter*)

Set the codec parameter for video codec.

Parameters

<i>enum_videocodec</i>	Video codec types. Supported types include: ENUM_VIDEOCODEC_H264 , ENUM_VIDEOCODEC_VP8 , ENUM_VIDEOCODEC_VP9 .
<i>sdpParameter</i>	The parameter is in string format.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

Example:

```
setVideoCodecParameter(PortSipEnumDefine.ENUM_VIDEOCODEC_H264, "profile-level-id=420033; packetization-mode=0");
```

Additional settings functions

Functions

- String [com.portsip.PortSipSdk.getVersion \(\)](#)

- int [com.portsip.PortSipSdk.enableRport](#) (boolean enable)
- int [com.portsip.PortSipSdk.enableEarlyMedia](#) (boolean enable)
Enable/disable rport(RFC3581).
- int [com.portsip.PortSipSdk.enableReliableProvisional](#) (boolean enable)
- int [com.portsip.PortSipSdk.enable3GppTags](#) (boolean enable)
- void [com.portsip.PortSipSdk.enableCallbackSignaling](#) (boolean enableSending, boolean enableReceived)
- void [com.portsip.PortSipSdk.setSrtpPolicy](#) (int enum_srtpolicy)
- int [com.portsip.PortSipSdk.setRtpPortRange](#) (int minimumRtpPort, int maximumRtpPort)
- int [com.portsip.PortSipSdk.enableCallForward](#) (boolean forBusyOnly, String forwardTo)
- int [com.portsip.PortSipSdk.disableCallForward](#) ()
- int [com.portsip.PortSipSdk.enableSessionTimer](#) (int timerSeconds)
- void [com.portsip.PortSipSdk.disableSessionTimer](#) ()
- void [com.portsip.PortSipSdk.setDoNotDisturb](#) (boolean state)
- void [com.portsip.PortSipSdk.enableAutoCheckMwi](#) (boolean state)
- int [com.portsip.PortSipSdk.setRtpKeepAlive](#) (boolean state, int keepAlivePayloadType, int deltaTransmitTimeMS)
- int [com.portsip.PortSipSdk.setKeepAliveTime](#) (int keepAliveTime)
- int [com.portsip.PortSipSdk.setAudioSamples](#) (int ptime, int maxptime)
- int [com.portsip.PortSipSdk.addSupportedMimeType](#) (String methodName, String mimeType, String subMimeType)

Detailed Description

Function Documentation

String com.portsip.PortSipSdk.getVersion ()

Get the version number of the current SDK.

Returns

String with version description

int com.portsip.PortSipSdk.enableRport (boolean *enable*)

Enable/Disable rport(RFC3581).

Parameters

<i>enable</i>	enable Set to true to enable the SDK to support rport. By default it is enabled.
---------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.enableEarlyMedia (boolean *enable*)

Enable/disable rport(RFC3581).

Parameters

<i>enable</i>	Set to true to enable the SDK to support rport. By default it is enabled.
---------------	---

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code. Enable/Disable Early Media.

Parameters

<i>enable</i>	Set to true to enable the SDK support Early Media. By default the Early Media is disabled.
---------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.enableReliableProvisional (boolean *enable*)

Enable/Disable PRACK.

Parameters

<i>enable</i>	Set to true to enable the SDK support PRACK. In default the PRACK is disabled.
---------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.enable3GppTags (boolean *enable*)

Enable/disable the 3Gpp tags, including "ims.icsi.mmtel" and "g.3gpp.smsip".

Parameters

<i>enable</i>	Set to true to enable 3Gpp tags for SDK.
---------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.enableCallbackSignaling (boolean *enableSending*, boolean *enableReceived*)

Enable/disable the callback of the SIP messages.

Parameters

<i>enableSending</i>	Set as true to enable to callback the sent SIP messages, or false to disable. Once enabled, the "onSendingSignaling" event will be triggered when the SDK sends a SIP message.
<i>enableReceived</i>	Set as true to enable to callback the received SIP messages, or false to disable. Once enabled, the "onReceivedSignaling" event will be triggered when the SDK receives a SIP message.

void com.portsip.PortSipSdk.setSrtpPolicy (int *enum_srtppolicy*)

Set the SRTP policy.

Parameters

<i>enum_srtppolicy</i>	The SRTP policy.allow: ENUM_SRTPPOLICY_NONE , ENUM_SRTPPOLICY_FORCE , ENUM_SRTPPOLICY_PREFER .
------------------------	--

int com.portsip.PortSipSdk.setRtpPortRange (int *minimumRtpPort*, int *maximumRtpPort*)

This function allows to set the RTP port range for audio and video streaming.

Parameters

<i>minimumRtpPort</i>	The minimum RTP port.
-----------------------	-----------------------

<i>maximumRtpPort</i>	The maximum RTP port.
-----------------------	-----------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

The port range ((max - min) % maxCallLines) should be greater than 4.

int com.portsip.PortSipSdk.enableCallForward (boolean *forBusyOnly*, String *forwardTo*)

Enable call forwarding.

Parameters

<i>forBusyOnly</i>	If this parameter is set to true, the SDK will forward incoming calls when the user is currently busy. If set it to false, SDK will forward all incoming calls.
<i>forwardTo</i>	The target to which the call will be forwarded. It must be in the format of sip: xxxx@sip.portsip.com .

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.disableCallForward ()

Disable the call forwarding. The SDK will not forward any incoming call when this function is called.

Returns

If the function succeeds, it will not return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.enableSessionTimer (int *timerSeconds*)

This function allows to periodically refresh Session Initiation Protocol (SIP) sessions by sending repeated INVITE requests.

Parameters

<i>timerSeconds</i>	The value of the refresh interval in seconds. A minimum of 90 seconds required.
---------------------	---

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

The repeated INVITE requests, or re-INVITES, are sent during an active call log to allow user agents (UA) or proxies to determine the status of a SIP session. Without this keep-alive mechanism, proxies that remember incoming and outgoing requests (stateful proxies) may continue to retain call state in vain. If a UA fails to send a BYE message at the end of a session, or if the BYE message is lost due to network problems, a stateful proxy will not know that the session has ended. The re-INVITES ensure that active sessions stay active and completed sessions are terminated.

void com.portsip.PortSipSdk.disableSessionTimer ()

Disable the session timer.

void com.portsip.PortSipSdk.setDoNotDisturb (boolean *state*)

Enable/disable the "Do not disturb" status.

Parameters

<i>state</i>	If it is set to true, the SDK will reject all incoming calls.
--------------	---

void com.portsip.PortSipSdk.enableAutoCheckMwi (boolean *state*)

Enable/disable the "Auto Check MWI" status.

Parameters

<i>state</i>	If it is set to true, the SDK will check Mwi automatically.
--------------	---

int com.portsip.PortSipSdk.setRtpKeepAlive (boolean *state*, int *keepAlivePayloadType*, int *deltaTransmitTimeMS*)

Enable or disable to send RTP keep-alive packet when the call is ongoing.

Parameters

<i>state</i>	When it's set to true, it's allowed to send the keep-alive packet during the conversation;
<i>keepAlivePayloadType</i>	The payload type of the keep-alive RTP packet. It's usually set to 126.
<i>deltaTransmitTimeMS</i>	The interval for sending keep-alive RTP packet, in millisecond. Recommended value ranges 15000 - 300000.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setKeepAliveTime (int *keepAliveTime*)

Enable or disable to send SIP keep-alive packet.

Parameters

<i>keepAliveTime</i>	This is the time interval for SIP keep-alive, in seconds. When it is set to 0, the SIP keep-alive will be disabled. Recommended value is 30 or 50.
----------------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setAudioSamples (int *ptime*, int *maxptime*)

Set the audio capture sample, which will be present in the SDP of INVITE and 200 OK message as "ptime" and "maxptime" attribute.

Parameters

<i>ptime</i>	It should be a multiple of 10 between 10 - 60 (included 10 and 60).
<i>maxptime</i>	The "maxptime" attribute should be a multiple of 10 between 10 - 60 (included 10 and 60). It can't be less than "ptime".

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.addSupportedMimeType (String *methodName*, String *imeType*, String *subMimeType*)

Set the SDK to receive SIP messages that include special mime type.

Parameters

<i>methodName</i>	Method name of the SIP message, such as INVITE, OPTION, INFO, MESSAGE, UPDATE, ACK etc. For more details please refer to RFC3261.
<i>imeType</i>	The mime type of SIP message.
<i>subMimeType</i>	The sub mime type of SIP message.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

In default, PortSIP VoIP SDK supports media types (mime types) included in the below incoming SIP messages:

```
"message/sipfrag" in NOTIFY message.  
"application/simple-message-summary" in NOTIFY message.  
"text/plain" in MESSAGE message. "application/dtmf-relay" in INFO  
message. <br> "application/media_control+xml" in INFO message.
```

The SDK allows to receive SIP messages that include above mime types. Now if remote side send an INFO SIP message with its "Content-Type" header value "text/plain", SDK will reject this INFO message, because "text/plain" of INFO message is not included in the default type list. How should we enable the SDK to receive SIP INFO messages that include "text/plain" mime type? The answer is `addSupportedMimeType`:

```
addSupportedMimeType("INFO", "text", "plain");
```

If the user wishes to receive the NOTIFY message with "application/media_control+xml", it should be set as below:

```
addSupportedMimeType("NOTIFY", "application", "media_control+xml");
```

For more details about the mime type, please visit:
<http://www.iana.org/assignments/media-types/>

Access SIP message header functions

Functions

- String [com.portsip.PortSipSdk.getSipMessageHeaderValue](#) (String sipMessage, String headerName)
- long [com.portsip.PortSipSdk.addSipMessageHeader](#) (long sessionId, String methodName, int msgType, String headerName, String headerValue)
- int [com.portsip.PortSipSdk.removeAddedSipMessageHeader](#) (long addedSipMessageId)
- void [com.portsip.PortSipSdk.clearAddedSipMessageHeaders](#) ()
- long [com.portsip.PortSipSdk.modifySipMessageHeader](#) (long sessionId, String methodName, int msgType, String headerName, String headerValue)
- int [com.portsip.PortSipSdk.removeModifiedSipMessageHeader](#) (long modifiedSipMessageId)
- void [com.portsip.PortSipSdk.clearModifiedSipMessageHeaders](#) ()

Detailed Description

Function Documentation

String `com.portsip.PortSipSdk.getSipMessageHeaderValue` (String *sipMessage*, String *headerName*)

Access the SIP header of SIP message.

Parameters

<i>sipMessage</i>	The SIP message.
-------------------	------------------

<i>headerName</i>	The header of which user wishes to access the SIP message.
-------------------	--

Returns

String. The SIP header of SIP message.

long com.portsip.PortSipSdk.addSipMessageHeader (long sessionId, String methodName, int msgType, String headerName, String headerValue)

Add the SIP Message header into the specified outgoing SIP message.

Parameters

<i>sessionId</i>	Add the header to the SIP message with the specified session Id only. By setting to -1, it will be added to all messages.
<i>methodName</i>	Add the header to the SIP message with specified method name only. For example: "INVITE", "REGISTER", "INFO" etc. If "ALL" specified, it will add all SIP messages.
<i>msgType</i>	1 refers to apply to the request message, 2 refers to apply to the response message, 3 refers to apply to both request and response.
<i>headerName</i>	The header name which will appear in SIP message.
<i>headerValue</i>	The custom header value.

Returns

If the function succeeds, it will return the addedSipMessageId , which is greater than 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.removeAddedSipMessageHeader (long addedSipMessageId)

Remove the headers (custom header) added by addSipMessageHeader.

Parameters

<i>addedSipMessageId</i>	The addedSipMessageId return by addSipMessageHeader.
--------------------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.clearAddedSipMessageHeaders ()

Clear the added extension headers (custom headers)

Remarks

For example, we have added two custom headers into every outgoing SIP message and want to have them removed.

```
addSipMessageHeader(-1,"ALL",3,"Blling", "usd100.00");
addSipMessageHeader(-1,"ALL",3,"ServiceId", "8873456");
clearAddedSipMessageHeaders();
```

If this function is called, the added extension headers will no longer appear in outgoing SIP message.

long com.portsip.PortSipSdk.modifySipMessageHeader (long sessionId, String methodName, int msgType, String headerName, String headerValue)

Modify the special SIP header value for every outgoing SIP message.

Parameters

<i>sessionId</i>	The header to the SIP message with the specified session Id. By setting to -1, it will be added to all messages.
<i>methodName</i>	Modify the header to the SIP message with specified method name only. For example: "INVITE", "REGISTER", "INFO" etc. If "ALL" specified, it will add all SIP messages.

<i>msgType</i>	1 refers to apply to the request message, 2 refers to apply to the response message, 3 refers to apply to both request and response.
<i>headerName</i>	The SIP header name of which the value will be modified.
<i>headerValue</i>	The header value to be modified.

Returns

If the function succeeds, it will return `modifiedSipMessageId`, which is greater than 0. If the function fails, it will return a specific error code.

Remarks

Example: modify "Expires" header and "User-Agent" header value for every outgoing SIP message:

```
modifySipMessageHeader(-1, "ALL", 3, "Expires", "1000");
modifySipMessageHeader(-1, "ALL", 3, "User-Agent", "MyTest Softphone 1.0");
```

int com.portsip.PortSipSdk.removeModifiedSipMessageHeader (long modifiedSipMessageId)

Remove the headers (custom header) added by `modifiedSipMessageId`.

Parameters

<i>modifiedSipMessageId</i>	The <code>modifiedSipMessageId</code> return by <code>modifySipMessageHeader</code> .
-----------------------------	---

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.clearModifiedSipMessageHeaders ()

Clear the modify headers value. Once cleared, it will no longer modify every outgoing SIP message header values.

Remarks

Example: modify two headers value for every outgoing SIP message and then clear it:

```
modifySipMessageHeader(-1, "ALL", 3, "Expires", "1000");
modifySipMessageHeader(-1, "ALL", 3, "User-Agent", "MyTest Softphone 1.0");
clearModifyHeaders();
```

Audio and video functions

Functions

- int [com.portsip.PortSipSdk.setVideoDeviceId](#) (int deviceId)
- int [com.portsip.PortSipSdk.setVideoOrientation](#) (int rotation)
- int [com.portsip.PortSipSdk.enableVideoHardwareCodec](#) (boolean enableHWEncoder, boolean enableHWDecoder)
- int [com.portsip.PortSipSdk.setVideoResolution](#) (int width, int height)
- int [com.portsip.PortSipSdk.setVideoCropAndScale](#) (boolean enable)
- int [com.portsip.PortSipSdk.setAudioBitrate](#) (long sessionId, int enum_audiocodec, int bitrateKbps)
- int [com.portsip.PortSipSdk.setVideoBitrate](#) (long sessionId, int bitrateKbps)
- int [com.portsip.PortSipSdk.setVideoFrameRate](#) (long sessionId, int frameRate)
- int [com.portsip.PortSipSdk.sendVideo](#) (long sessionId, boolean send)
- int [com.portsip.PortSipSdk.setRemoteVideoWindow](#) (long sessionId, [PortSIPVideoRenderer](#) renderer)
- int [com.portsip.PortSipSdk.setRemoteScreenWindow](#) (long sessionId, [PortSIPVideoRenderer](#) renderer)

- void [com.portsip.PortSipSdk.displayLocalVideo](#) (boolean state, boolean mirror, [PortSIPVideoRenderer](#) renderer)
- int [com.portsip.PortSipSdk.setVideoNackStatus](#) (boolean state)
- int [com.portsip.PortSipSdk.setChannelOutputVolumeScaling](#) (long sessionId, int scaling)
- int [com.portsip.PortSipSdk.setChannelInputVolumeScaling](#) (long sessionId, int scaling)
- void [com.portsip.PortSipSdk.enableAudioManager](#) (boolean state)
- Set< PortSipEnumDefine.AudioDevice > [com.portsip.PortSipSdk.getAudioDevices](#) ()
- int [com.portsip.PortSipSdk.setAudioDevice](#) (PortSipEnumDefine.AudioDevice defaultDevice)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.setVideoDeviceId (int *deviceId*)

Set the video device that will be used for video call.

Parameters

<i>deviceId</i>	Device ID (index) for video device (camera).
-----------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setVideoOrientation (int *rotation*)

Setting the video Device Orientation.

Parameters

<i>rotation</i>	Device Orientation for video device (camera), e.g 0,90,180,270.
-----------------	---

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.enableVideoHardwareCodec (boolean *enableHWEncoder*, boolean *enableHWDDecoder*)

Set enable/disable video Hardware codec.

Parameters

<i>enableHWEncoder</i>	If it is set to true, the SDK will use video hardware encoder when available. By default it is true.
<i>enableHWDDecoder</i>	If it is set to true, the SDK will use video hardware decoder when available. By default it is true.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setVideoResolution (int *width*, int *height*)

Set the video capturing resolution.

Parameters

<i>width</i>	Video resolution, width
<i>height</i>	Video resolution, height

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setVideoCropAndScale (boolean *enable*)

When the camera does not support specified resolution, enable or disable SDK to crop and scale to the specified resolution.

Parameters

<i>enable</i>	Enable or disable to crop or scale the video to fit in specified resolution. By default it is disabled.
---------------	---

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setAudioBitrate (long *sessionId*, int *enum_audiocodec*, int *bitrateKbps*)

Set the audio bitrate.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>enum_audiocodec</i>	Audio codec type allowed: ENUM_AUDIOCODEC_OPUS
<i>bitrateKbps</i>	The Audio bitrate in KBPS.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setVideoBitrate (long *sessionId*, int *bitrateKbps*)

Set the video bitrate.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>bitrateKbps</i>	The video bitrate in KBPS.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setVideoFrameRate (long *sessionId*, int *frameRate*)

Set the video frame rate. Usually you do not need to call this function to set the frame rate since the SDK uses default frame rate.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>frameRate</i>	The frame rate value, with its minimum of 5, and maximum value of 30. The greater the value is, the better video quality enabled and more bandwidth required;

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.sendVideo (long *sessionId*, boolean *send*)

Send the video to remote side.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>send</i>	Set to true to send the video, or false to stop sending.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setRemoteVideoWindow (long *sessionId*, [PortSIPVideoRenderer](#) *renderer*)

Set the window for a session that is used for displaying the received remote video image.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>renderer</i>	SurfaceView a SurfaceView for displaying the received remote video image.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setRemoteScreenWindow (long *sessionId*, [PortSIPVideoRenderer](#) *renderer*)

Set the window for a session that is used for displaying the received remote screen image.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>renderer</i>	SurfaceView a SurfaceView for displaying the received remote screen image.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.displayLocalVideo (boolean *state*, boolean *mirror*, [PortSIPVideoRenderer](#) *renderer*)

Start/stop displaying the local video image.

Parameters

<i>state</i>	Set to true to display local video image.
<i>mirror</i>	Set to true to display the mirror image of local video.
<i>renderer</i>	SurfaceView a SurfaceView for displaying local video image from camera.

int com.portsip.PortSipSdk.setVideoNackStatus (boolean *state*)

Enable/disable the NACK feature (rfc6642) which helps to improve the video quality.

Parameters

<i>state</i>	Set to true to enable.
--------------	------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setChannelOutputVolumeScaling (long *sessionId*, int *scaling*)

Set a volume |scaling| to be applied to the outgoing signal of a specific audio channel.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>scaling</i>	Valid scale ranges [0, 1000]. Default is 100.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setChannelInputVolumeScaling (long *sessionId*, int *scaling*)

Set a volume *|scaling|* to be applied to the microphone signal of a specific audio channel.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>scaling</i>	Valid scale ranges [0, 1000]. Default is 100.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.enableAudioManager (boolean *state*)

enable/disable sdk audio manager,when enable sdk will auto manager audio device input/output. if the state is enabled, the onAudioDeviceChanged event will be triggered when available audio devices changed or audio device currently in use changed .

Parameters

<i>state</i>	@true enable sdk audio manager @false disable audio manager
--------------	---

Set<PortSipEnumDefine.AudioDevice> com.portsip.PortSipSdk.getAudioDevices ()

Get current set of available/selectable audio devices.

Returns

Current set of available/selectable audio devices.

int com.portsip.PortSipSdk.setAudioDevice (PortSipEnumDefine.AudioDevice *defaultDevice*)

Set the audio device that will used for audio call. For Android and iOS, switch between earphone and Loudspeaker allowed.

Parameters

<i>defaultDevice</i>	Set to true the SDK use loudspeaker for audio call, this just available for mobile platform only.
----------------------	---

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Call functions

Functions

- long [com.portsip.PortSipSdk.call](#) (String callee, boolean sendSdp, boolean videoCall)
- int [com.portsip.PortSipSdk.rejectCall](#) (long sessionId, int code)
- int [com.portsip.PortSipSdk.hangUp](#) (long sessionId)
- int [com.portsip.PortSipSdk.answerCall](#) (long sessionId, boolean videoCall)
- int [com.portsip.PortSipSdk.updateCall](#) (long sessionId, boolean enableAudio, boolean enableVideo)
- int [com.portsip.PortSipSdk.hold](#) (long sessionId)
- int [com.portsip.PortSipSdk.unHold](#) (long sessionId)

- int [com.portsip.PortSipSdk.muteSession](#) (long sessionId, boolean muteIncomingAudio, boolean muteOutgoingAudio, boolean muteIncomingVideo, boolean muteOutgoingVideo)
- int [com.portsip.PortSipSdk.forwardCall](#) (long sessionId, String forwardTo)
- long [com.portsip.PortSipSdk.pickupBLFCall](#) (String replaceDialogId, boolean videoCall)
- int [com.portsip.PortSipSdk.sendDtmf](#) (long sessionId, int enum_dtmfMethod, int code, int dtmfDuration, boolean playDtmfTone)

Detailed Description

Function Documentation

long com.portsip.PortSipSdk.call (String *callee*, boolean *sendSdp*, boolean *videoCall*)

Make a call

Parameters

<i>callee</i>	The callee. It can be a name only or full SIP URI, for example: user001 or sip: user001@sip.iptel.org or sip: user002@sip.yourdomain.com :5068
<i>sendSdp</i>	If it is set to false, the outgoing call will not include the SDP in INVITE message.
<i>videoCall</i>	If it is set to true and at least one video codec was added, the outgoing call will include the video codec into SDP. Otherwise no video codec will be added into outgoing SDP.

Returns

If the function succeeds, it will return the session ID of the call, which is greater than 0. If the function fails, it will return a specific error code.

Note: the function success just means the outgoing call is processing, you need to detect the call final state in onInviteTrying, onInviteRinging, onInviteFailure callback events.

int com.portsip.PortSipSdk.rejectCall (long *sessionId*, int *code*)

rejectCall Reject the incoming call.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>code</i>	Reject code, for example, 486, 480 etc.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.hangUp (long *sessionId*)

hangUp Hang up the call.

Parameters

<i>sessionId</i>	Session ID of the call.
------------------	-------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.answerCall (long *sessionId*, boolean *videoCall*)

answerCall Answer the incoming call.

Parameters

<i>sessionId</i>	The session ID of call.
<i>videoCall</i>	If the incoming call is a video call and the video codec is matched, set to true to answer the video call. If set to false, the answer call does not include video codec answer anyway.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.updateCall (long sessionId, boolean enableAudio, boolean enableVideo)

updateCall Use the re-INVITE to update the established call.

Parameters

<i>sessionId</i>	The session ID of call.
<i>enableAudio</i>	Set to true to allow the audio in updated call, or false to disable audio in updated call.
<i>enableVideo</i>	Set to true to allow the video in update call, or false to disable video in updated call.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return specific error code.

Remarks

Example usage:

Example 1: A called B with the audio only, and B answered A, there would be an audio conversation between A and B. Now A want to see B through video, A could use these functions to fulfill it.

```
clearVideoCodec();  
addVideoCodec(VIDEOCODEC_H264);  
updateCall(sessionId, true, true);
```

Example 2: Remove video stream from the current conversation.

```
updateCall(sessionId, true, false);
```

int com.portsip.PortSipSdk.hold (long sessionId)

To place a call on hold.

Parameters

<i>sessionId</i>	The session ID of call.
------------------	-------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.unHold (long sessionId)

Take off hold.

Parameters

<i>sessionId</i>	The session ID of call.
------------------	-------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.muteSession (long sessionId, boolean muteIncomingAudio, boolean muteOutgoingAudio, boolean muteIncomingVideo, boolean muteOutgoingVideo)

Mute the specified audio or video session.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>muteIncomingAudio</i>	Set it to true to mute incoming audio stream. Once set, remote side audio cannot be heard.
<i>muteOutgoingAudio</i>	Set it to true to mute outgoing audio stream. Once set, the remote side cannot hear the audio.
<i>muteIncomingVideo</i>	Set it to true to mute incoming video stream. Once set, remote side video cannot be seen.
<i>muteOutgoingVideo</i>	Set it to true to mute outgoing video stream, the remote side cannot see the video.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.forwardCall (long sessionId, String forwardTo)

Forward call to another one when receiving the incoming call.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>forwardTo</i>	Target of the forward. It can be either "sip:number@sipserver.com" or "number".

Returns

If the function succeeds, it will return value 0. If the function fails, it will return value a specific error code.

long com.portsip.PortSipSdk.pickupBLFCall (String replaceDialogId, boolean videoCall)

This function will be used for picking up a call based on the BLF (Busy Lamp Field) status.

Parameters

<i>replaceDialogId</i>	The ID of the call which will be pickup. It comes with onDialogStateUpdated callback.
<i>videoCall</i>	Indicates pickup video call or audio call

Returns

If the function succeeds, it will return a session ID that is greater than 0 to the new call, otherwise returns a specific error code that is less than 0.

Remarks

The scenario is:

1. User 101 subscribed the user 100's call status: sendSubscription(mSipLib, "100", "dialog");
2. When 100 holds a call or 100 is ringing, onDialogStateUpdated callback will be triggered, and 101 will receive this callback. Now 101 can use pickupBLFCall function to pick the call rather than 100 to talk with caller.

int com.portsip.PortSipSdk.sendDtmf (long sessionId, int enum_dtmfMethod, int code, int dtmfDuration, boolean playDtmfTone)

Send DTMF tone.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>enum_dtmfMethod</i>	DTMF tone could be sent via two methods: DTMF_RFC2833 or DTMF_INFO. The DTMF_RFC2833 is recommend.
<i>code</i>	The DTMF tone. Values include:

code	Description
0	The DTMF tone 0.
1	The DTMF tone 1.
2	The DTMF tone 2.
3	The DTMF tone 3.
4	The DTMF tone 4.
5	The DTMF tone 5.
6	The DTMF tone 6.
7	The DTMF tone 7.
8	The DTMF tone 8.
9	The DTMF tone 9.
10	The DTMF tone *.
11	The DTMF tone #.
12	The DTMF tone A.
13	The DTMF tone B.
14	The DTMF tone C.
15	The DTMF tone D.
16	The DTMF tone FLASH.

Parameters

<i>dtmfDuration</i>	The DTMF tone samples. Recommended value 160.
<i>playDtmfTone</i>	Set to true the SDK play local DTMF tone sound during send DTMF.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Refer functions

Functions

- int [com.portsip.PortSipSdk.refer](#) (long sessionId, String referTo)
- int [com.portsip.PortSipSdk.attendedRefer](#) (long sessionId, long replaceSessionId, String referTo)
- int [com.portsip.PortSipSdk.attendedRefer2](#) (long sessionId, long replaceSessionId, String replaceMethod, String target, String referTo)
- int [com.portsip.PortSipSdk.outOfDialogRefer](#) (long replaceSessionId, String replaceMethod, String target, String referTo)
- long [com.portsip.PortSipSdk.acceptRefer](#) (long referId, String referSignaling)
- int [com.portsip.PortSipSdk.rejectRefer](#) (long referId)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.refer (long *sessionId*, String *referTo*)

Transfer the current call to another callee.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>referTo</i>	Target callee of the transfer. It can be either "sip:number@sipserver.com" or "number".

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

```
refer(sessionId, "sip:testuser12@sip.portsip.com");
```

You can refer to the video on Youtube at:

https://www.youtube.com/watch?v=_2w9EGgr3FY, which will demonstrate how to complete the transfer.

int com.portsip.PortSipSdk.attendedRefer (long *sessionId*, long *replaceSessionId*, String *referTo*)

Make an attended refer.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>replaceSessionId</i>	Session ID of the replace call.
<i>referTo</i>	Target callee of the refer. It can be either "sip:number@sipserver.com" or "number".

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

Please read the sample project source code to get more details, or you can refer to the video on YouTube at:

https://www.youtube.com/watch?v=_2w9EGgr3FY

Note: Please use Windows Media Player to play the AVI file, which demonstrates how to complete the transfer.

int com.portsip.PortSipSdk.attendedRefer2 (long *sessionId*, long *replaceSessionId*, String *replaceMethod*, String *target*, String *referTo*)

Make an attended refer.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>replaceSessionId</i>	The session ID of the session to be replaced.
<i>replaceMethod</i>	The SIP method name to be added in the "Refer-To" header, usually INVITE or BYE.
<i>target</i>	The target to which the REFER message will be sent.
<i>referTo</i>	The URI to be added into the "Refer-To" header.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.outOfDialogRefer (long *replaceSessionId*, String *replaceMethod*, String *target*, String *referTo*)

Make an attended refer.

Parameters

<i>replaceSessionId</i>	The session ID of the session which will be replaced.
<i>replaceMethod</i>	The SIP method name which will be added in the "Refer-To" header, usually INVITE or BYE.
<i>target</i>	The target to which the REFER message will be sent.
<i>referTo</i>	The URI which will be added into the "Refer-To" header.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

long com.portsip.PortSipSdk.acceptRefer (long *referId*, String *referSignaling*)

By accepting the REFER request, a new call will be made if this function is called. The function is usually called after onReceivedRefer callback event.

Parameters

<i>referId</i>	The ID of REFER request that comes from onReceivedRefer callback event.
<i>referSignaling</i>	The SIP message of REFER request that comes from onReceivedRefer callback event.

Returns

If the function succeeds, it will return a session ID greater than 0 to the new call for REFER; otherwise it will return a specific error code less than 0;

int com.portsip.PortSipSdk.rejectRefer (long *referId*)

Reject the REFER request.

Parameters

<i>referId</i>	The ID of REFER request that comes from onReceivedRefer callback event.
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Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Send audio and video stream functions

Functions

- int [com.portsip.PortSipSdk.enableSendPcmStreamToRemote](#) (long sessionId, boolean state, int streamSamplesPerSec)
- int [com.portsip.PortSipSdk.sendPcmStreamToRemote](#) (long sessionId, byte[] data, int dataLength)
- int [com.portsip.PortSipSdk.enableSendVideoStreamToRemote](#) (long sessionId, boolean state)
- int [com.portsip.PortSipSdk.sendVideoStreamToRemote](#) (long sessionId, byte[] data, int dataLength, int width, int height)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.enableSendPcmStreamToRemote (long *sessionId*, boolean *state*, int *streamSamplesPerSec*)

Enable the SDK send PCM stream data to remote side from another source instead of microphone. This function MUST be called first to send the PCM stream data to another side.

Parameters

<i>sessionId</i>	The session ID of call.
<i>state</i>	Set to true to enable the send stream, or false to disable.
<i>streamSamplesPerSec</i>	The PCM stream data sample, in seconds. For example 8000 or 16000.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.sendPcmStreamToRemote (long *sessionId*, byte[] *data*, int *dataLength*)

Send the audio stream in PCM format from another source instead of audio device capturing (microphone).

Parameters

<i>sessionId</i>	Session ID of the call conversation.
<i>data</i>	The PCM audio stream data. It must be 16bit, mono.
<i>dataLength</i>	The size of data.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

Usually we should use it like below:

```
enableSendPcmStreamToRemote(sessionId, true, 16000);  
sendPcmStreamToRemote(sessionId, data, dataSize);
```

You can't have too much audio data at one time as we have 100ms audio buffer only. Once you put too much, data will be lost. It is recommended to send 20ms audio data every 20ms.

int com.portsip.PortSipSdk.enableSendVideoStreamToRemote (long *sessionId*, boolean *state*)

Enable the SDK to send video stream data to remote side from another source instead of camera.

This function MUST be called first to send the video stream data to another side.

Parameters

<i>sessionId</i>	The session ID of call.
<i>state</i>	Set to true to enable the send stream, or false to disable.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.sendVideoStreamToRemote (long *sessionId*, byte[] *data*, int *dataLength*, int *width*, int *height*)

Send the video stream in i420 from another source instead of video device capturing (camera).

Before calling this function, you MUST call the `enableSendVideoStreamToRemote` function.

Parameters

<i>sessionId</i>	Session ID of the call conversation.
<i>data</i>	The video stream data. It must be in i420 format.
<i>dataLength</i>	The size of data.
<i>width</i>	The width of the video image.
<i>height</i>	The height of video image.

Returns

If the function succeeds, it will return value is 0. If the function fails, it will return a specific error code.

RTP packets, Audio stream and video stream callback

functions

Functions

- long [com.portsip.PortSipSdk.enableRtpCallback](#) (long sessionId, int mediaType, int directionMode)
- void [com.portsip.PortSipSdk.enableAudioStreamCallback](#) (long sessionId, boolean enable, int enum_direction)
- void [com.portsip.PortSipSdk.enableVideoStreamCallback](#) (long sessionId, int enum_direction)

Detailed Description

functions

Function Documentation

long com.portsip.PortSipSdk.enableRtpCallback (long sessionId, int mediaType, int directionMode)

Set the RTP callbacks to allow access to the sent and received RTP packets.

Parameters

<i>sessionId</i>	The session ID of the call.
<i>mediaType</i>	RTP packet media type, 0 for audio, 1 for video , 2 for screen.
<i>directionMode</i>	RTP packet direction, 0 for sending, 1 for receiving.

Returns

If the function succeeds, it will return value is 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.enableAudioStreamCallback (long sessionId, boolean enable, int enum_direction)

Enable/disable the audio stream callback. The `onAudioRawCallback` event will be triggered if the callback is enabled.

Parameters

<i>sessionId</i>	The session ID of call.
------------------	-------------------------

<i>enable</i>	Set to true to enable audio stream callback, or false to stop the callback.
<i>enum_direction</i>	The audio stream callback mode. Supported modes include ENUM_DIRECTION_NONE , ENUM_DIRECTION_SEND , ENUM_DIRECTION_RECV ENUM_DIRECTION_SEND_RECV .

void com.portsip.PortSipSdk.enableVideoStreamCallback (long sessionId, int enum_direction)

Enable/disable the video stream callback, the onVideoRawCallback event will be triggered if the callback is enabled.

Parameters

<i>sessionId</i>	The session ID of call.
<i>enum_direction</i>	The video stream callback mode. Supported modes include ENUM_DIRECTION_NONE , ENUM_DIRECTION_SEND , ENUM_DIRECTION_RECV , ENUM_DIRECTION_SEND_RECV .

Record functions

Functions

- int [com.portsip.PortSipSdk.startRecord](#) (long sessionId, String recordFilePath, String recordFileName, boolean appendTimeStamp, int enum_audioFileFormat, int enum_audioRecordMode, int enum_videocodec, int enum_videoRecordMode)
- int [com.portsip.PortSipSdk.stopRecord](#) (long sessionId)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.startRecord (long sessionId, String recordFilePath, String recordFileName, boolean appendTimeStamp, int enum_audioFileFormat, int enum_audioRecordMode, int enum_videocodec, int enum_videoRecordMode)

Start recording the call.

Parameters

<i>sessionId</i>	The session ID of call conversation.
<i>recordFilePath</i>	The file path to save record file. It must be existent.
<i>recordFileName</i>	The file name of record file. For example audiorecord.wav or videorecord.avi.
<i>appendTimeStamp</i>	Set to true to append the timestamp to the name of the recording file.
<i>enum_audioFileFormat</i>	The audio record file format, allow below values: FILEFORMAT_WAVE = 1, /// The record audio file is WAVE format. FILEFORMAT_AMR, /// The record audio file is in AMR format with all voice data compressed by AMR codec.
<i>enum_audioRecordMode</i>	The audio record mode, allow below values: RECORD_NONE = 0, /// Not Record. RECORD_RECV = 1, /// Only record the received data. RECORD_SEND, /// Only record send data. RECORD_BOTH /// The record audio file is WAVE format.
<i>enum_videocodec</i>	The codec used for compressing the video data to save into video record file.
<i>enum_videoRecordMode</i>	Allow to set video record mode. Support to record received and/or sent video.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.stopRecord (long sessionId)

Stop recording.

Parameters

<i>sessionId</i>	The session ID of call conversation.
------------------	--------------------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Play audio and video file to remote functions

Functions

- int [com.portsip.PortSipSdk.playVideoFileToRemote](#) (long sessionId, String aviFile, boolean loop, boolean playAudio)
- int [com.portsip.PortSipSdk.stopPlayVideoFileToRemote](#) (long sessionId)
- int [com.portsip.PortSipSdk.playAudioFileToRemote](#) (long sessionId, String filename, int fileSamplesPerSec, boolean loop)
- int [com.portsip.PortSipSdk.stopPlayAudioFileToRemote](#) (long sessionId)
- int [com.portsip.PortSipSdk.playAudioFileToRemoteAsBackground](#) (long sessionId, String filename, int fileSamplesPerSec)
- int [com.portsip.PortSipSdk.stopPlayAudioFileToRemoteAsBackground](#) (long sessionId)
- void [com.portsip.PortSipSdk.audioPlayLoopbackTest](#) (boolean enable)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.playVideoFileToRemote (long sessionId, String aviFile, boolean loop, boolean playAudio)

Play an AVI file to remote party.

Parameters

<i>sessionId</i>	Session ID of the call.
<i>aviFile</i>	The full filepath, such as "/mnt/sdcard/test.avi".
<i>loop</i>	Set to false to stop playing video file when it is ended, or true to play it repeatedly.
<i>playAudio</i>	If set to true, audio and video will be played together; or false to play the video only.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.stopPlayVideoFileToRemote (long sessionId)

Stop play video file to remote side.

Parameters

<i>sessionId</i>	Session ID of the call.
------------------	-------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.playAudioFileToRemote (long *sessionId*, String *filename*, int *fileSamplesPerSec*, boolean *loop*)

Play a wave file to remote party.

Parameters

<i>sessionId</i>	Session ID of the call.
<i>filename</i>	The full filepath, such as "/mnt/sdcard/test.wav".
<i>fileSamplesPerSec</i>	The wave file sample in seconds. It could be 8000, 16000 or 32000.
<i>loop</i>	Set to false to stop playing audio file when it is ended, or true to play it repeatedly.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.stopPlayAudioFileToRemote (long *sessionId*)

Stop playing wave file to remote side.

Parameters

<i>sessionId</i>	Session ID of the call.
------------------	-------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.playAudioFileToRemoteAsBackground (long *sessionId*, String *filename*, int *fileSamplesPerSec*)

Play a wave file to remote party as conversation background sound.

Parameters

<i>sessionId</i>	Session ID of the call.
<i>filename</i>	The full filepath, such as "/mnt/sdcard/test.wav".
<i>fileSamplesPerSec</i>	The wave file sample, in seconds. It should be 8000, 16000 or 32000.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.stopPlayAudioFileToRemoteAsBackground (long *sessionId*)

Stop playing a wave file to remote party as background sound for the conversation.

Parameters

<i>sessionId</i>	Session ID of the call.
------------------	-------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.audioPlayLoopbackTest (boolean *enable*)

Used for testing loopback for the audio device.

Parameters

<i>enable</i>	Set to true to start testing audio loopback test; or set to false to stop.
---------------	--

Conference functions

Functions

- int [com.portsip.PortSipSdk.createAudioConference](#) ()
- int [com.portsip.PortSipSdk.createVideoConference](#) ([PortSIPVideoRenderer](#) conferenceVideoWindow, int videoWidth, int videoHeight, boolean displayLocalVideoInConference)
- void [com.portsip.PortSipSdk.destroyConference](#) ()
- int [com.portsip.PortSipSdk.setConferenceVideoWindow](#) ([PortSIPVideoRenderer](#) conferenceVideoWindow)
- int [com.portsip.PortSipSdk.joinToConference](#) (long sessionId)
- int [com.portsip.PortSipSdk.removeFromConference](#) (long sessionId)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.createAudioConference ()

Create an audio conference. It will fail if the existing conference is not ended yet.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.createVideoConference ([PortSIPVideoRenderer](#) conferenceVideoWindow, int videoWidth, int videoHeight, boolean displayLocalVideoInConference)

Create a video conference. It will fail if the existing conference is not ended yet.

Parameters

<i>conferenceVideoWindow</i>	SurfaceView The window used for displaying the conference video.
<i>videoWidth</i>	Width of conference video resolution
<i>videoHeight</i>	Height of conference video resolution
<i>displayLocalVideoInConference</i>	Display local video during conference

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

void com.portsip.PortSipSdk.destroyConference ()

End the exist conference.

int com.portsip.PortSipSdk.setConferenceVideoWindow ([PortSIPVideoRenderer](#) conferenceVideoWindow)

Set the window for a conference that is used for displaying the received remote video image.

Parameters

<i>conferenceVideoWindow</i>	SurfaceView The window which is used for displaying the conference video
------------------------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.joinToConference (long sessionId)

Join a session into existing conference. If the call is in hold, it will be un-hold automatically.

Parameters

<i>sessionId</i>	Session ID of the call.
------------------	-------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.removeFromConference (long sessionId)

Remove a session from an existing conference.

Parameters

<i>sessionId</i>	Session ID of the call.
------------------	-------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

RTP and RTCP QOS functions

Functions

- int [com.portsip.PortSipSdk.setAudioRtcpBandwidth](#) (long sessionId, int BitsRR, int BitsRS, int KBitsAS)
- int [com.portsip.PortSipSdk.setVideoRtcpBandwidth](#) (long sessionId, int BitsRR, int BitsRS, int KBitsAS)
- int [com.portsip.PortSipSdk.enableAudioQos](#) (boolean state)
- int [com.portsip.PortSipSdk.enableVideoQos](#) (boolean state)
- int [com.portsip.PortSipSdk.setVideoMTU](#) (int mtu)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.setAudioRtcpBandwidth (long *sessionId*, int *BitsRR*, int *BitsRS*, int *KBitsAS*)

Set the audio RTCP bandwidth parameters as RFC3556.

Parameters

<i>sessionId</i>	Set the audio RTCP bandwidth parameters as RFC3556.
<i>BitsRR</i>	The bits for the RR parameter.
<i>BitsRS</i>	The bits for the RS parameter.
<i>KBitsAS</i>	The Kbits for the AS parameter.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setVideoRtcpBandwidth (long *sessionId*, int *BitsRR*, int *BitsRS*, int *KBitsAS*)

Set the video RTCP bandwidth parameters as the RFC3556.

Parameters

<i>sessionId</i>	The session ID of call conversation.
<i>BitsRR</i>	The bits for the RR parameter.
<i>BitsRS</i>	The bits for the RS parameter.
<i>KBitsAS</i>	The Kbits for the AS parameter.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.enableAudioQos (boolean *state*)

Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for audio channel.

Parameters

<i>state</i>	Set to YES to enable audio QoS and DSCP value will be 46; or NO to disable audio QoS and DSCP value will be 0.
--------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.enableVideoQos (boolean *state*)

Set the DSCP(differentiated services code point) value of QoS(Quality of Service) for video channel.

Parameters

<i>state</i>	Set to YES to enable video QoS and DSCP value will be 34; or NO to disable video QoS and DSCP value will be 0.
--------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setVideoMTU (int *mtu*)

Set the MTU size for video RTP packet.

Parameters

<i>mtu</i>	Set MTU value. Allow values range 512 - 65507. Default is 14000.
------------	--

Returns

If the function succeeds, the return value is 0. If the function fails, the return value is a specific error code.

RTP statistics functions

Functions

- int [com.portsip.PortSipSdk.getAudioStatistics](#) (long sessionId, int[] statistics)
- int [com.portsip.PortSipSdk.getVideoStatistics](#) (long sessionId, int[] statistics)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.getAudioStatistics (long sessionId, int[] statistics)

Obtain the statistics of audio channel.

Parameters

<i>sessionId</i>	The session ID of call conversation.
<i>statistics</i>	Return audio statistics statistics[0] - The number of sent bytes. statistics[1] - The number of sent packets. statistics[2] - The number of sent but lost packet. statistics[3] - Fraction of sent but lost packet in percentage. statistics[4] - The round-trip time of the session, in milliseconds. statistics[5] - Sent Audio codec type. statistics[6] - The sent jitter, in milliseconds. statistics[7] - The sent audio level.It ranges 0 - 9. statistics[8] - The number of received bytes. statistics[9] - The number of received packets. statistics[10] - The number of received but lost packets. statistics[11] - Fraction of received but lost packet in percentage. statistics[12] - Received Audio codec type. statistics[13] - The received jitter, in milliseconds. statistics[14] - The received audio level.It ranges 0 - 9.

Returns

If the function succeeds, it will return value is 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.getVideoStatistics (long sessionId, int[] statistics)

Obtain the statistics of video channel.

Parameters

<i>sessionId</i>	The session ID of call conversation.
<i>statistics</i>	Return Video statistics statistics[0] - The number of sent bytes.

	<p>statistics[1] - The number of sent packets. statistics[2] - The number of sent but lost packet. statistics[3] - Fraction of sent lost in percentage. statistics[4] - The round-trip time of the session, in milliseconds. statistics[5] - Send Video codec type. statistics[6] - Frame width for the sent video. statistics[7] - Frame height for the sent video. statistics[8] - Bitrate in BPS for the sent video. statistics[9] - Frame rate for the sent video. statistics[10] - The number of received bytes. statistics[11] - The number of received packets. statistics[12] - The number of received but lost packet. statistics[13] - Fraction of received but lost packet in percentage. statistics[14] - Received Video codec type. statistics[15] - Frame width for the received video. statistics[16] - Frame height for the received video. statistics[17] - (This parameter is not implemented yet)Bitrate in BPS for the received video. statistics[18] - Framerate for the received video. statistics[19] - The number of sent bytes.</p>
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Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Audio effect functions

Functions

- void [com.portsip.PortSipSdk.enableVAD](#) (boolean state)
- void [com.portsip.PortSipSdk.enableAEC](#) (boolean state)
- void [com.portsip.PortSipSdk.enableCNG](#) (boolean state)
- void [com.portsip.PortSipSdk.enableAGC](#) (boolean state)
- void [com.portsip.PortSipSdk.enableANS](#) (boolean state)

Detailed Description

Function Documentation

void com.portsip.PortSipSdk.enableVAD (boolean state)

Enable/disable Voice Activity Detection(VAD).

Parameters

<i>state</i>	Set to true to enable VAD, or false to disable.
--------------	---

void com.portsip.PortSipSdk.enableAEC (boolean state)

Enable/disable AEC (Acoustic Echo Cancellation).

Parameters

<i>state</i>	Set to true to enable AEC, or false to disable.
--------------	---

void com.portsip.PortSipSdk.enableCNG (boolean state)

Enable/disable Comfort Noise Generator(CNG).

Parameters

<i>state</i>	Set to true to enable CNG, or false to disable.
--------------	---

void com.portsip.PortSipSdk.enableAGC (boolean state)

Enable/disable Automatic Gain Control(AGC).

Parameters

<i>state</i>	Set to true to enable AEC, or false to disable.
--------------	---

void com.portsip.PortSipSdk.enableANS (boolean state)

Enable/disable Audio Noise Suppression(ANS).

Parameters

<i>state</i>	Set to true to enable ANS, or false to disable.
--------------	---

Send OPTIONS/INFO/MESSAGE functions

Functions

- int [com.portsip.PortSipSdk.sendOptions](#) (String to, String sdp)
 - int [com.portsip.PortSipSdk.sendInfo](#) (long sessionId, String mimeType, String subMimeType, String infoContents)
 - long [com.portsip.PortSipSdk.sendMessage](#) (long sessionId, String mimeType, String subMimeType, byte[] message, int messageLength)
 - long [com.portsip.PortSipSdk.sendOutOfDialogMessage](#) (String to, String mimeType, String subMimeType, boolean isSMS, byte[] message, int messageLength)
 - long [com.portsip.PortSipSdk.setPresenceMode](#) (int mode)
 - long [com.portsip.PortSipSdk.setDefaultSubscriptionTime](#) (int secs)
 - long [com.portsip.PortSipSdk.setDefaultPublicationTime](#) (int secs)
 - long [com.portsip.PortSipSdk.presenceSubscribe](#) (String contact, String subject)
 - int [com.portsip.PortSipSdk.presenceTerminateSubscribe](#) (long subscribeId)
 - int [com.portsip.PortSipSdk.presenceAcceptSubscribe](#) (long subscribeId)
 - int [com.portsip.PortSipSdk.presenceRejectSubscribe](#) (long subscribeId)
 - int [com.portsip.PortSipSdk.setPresenceStatus](#) (long subscribeId, String statusText)
 - long [com.portsip.PortSipSdk.sendSubscription](#) (String to, String eventName)
Send a SUBSCRIBE message to subscribe an event.
-
- int [com.portsip.PortSipSdk.terminateSubscription](#) (long subscribeId)

Detailed Description

Function Documentation

int com.portsip.PortSipSdk.sendOptions (String to, String sdp)

Send OPTIONS message.

Parameters

<i>to</i>	The recipient of OPTIONS message.
<i>sdp</i>	The SDP of OPTIONS message. It's optional if user does not want to send the SDP with OPTIONS message.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return value a specific error code.

int com.portsip.PortSipSdk.sendInfo (long *sessionId*, String *contentType*, String *subMimeType*, String *infoContents*)

Send a INFO message to remote side in dialog.

Parameters

<i>sessionId</i>	The session ID of call.
<i>contentType</i>	The mime type of INFO message.
<i>subMimeType</i>	The sub mime type of INFO message.
<i>infoContents</i>	The contents that will be sent with INFO message.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

long com.portsip.PortSipSdk.sendMessage (long *sessionId*, String *contentType*, String *subMimeType*, byte[] *message*, int *messageLength*)

Send a MESSAGE message to remote side in dialog.

Parameters

<i>sessionId</i>	The session ID of call.
<i>contentType</i>	The mime type of MESSAGE message.
<i>subMimeType</i>	The sub mime type of MESSAGE message.
<i>message</i>	The contents that will be sent with MESSAGE message. Binary data allowed.
<i>messageLength</i>	The message size.

Returns

If the function succeeds, it will return a message ID that allows to track the message sending state in onSendMessageSuccess and onSendMessageFailure. If the function fails, it will return a specific error code that is less than 0.

Remarks

Example 1: Send a plain text message. Note: to send other languages text, please use the UTF8 to encode the message before sending.

```
sendMessage(sessionId, "text", "plain", "hello", 6);
```

Example 2: Send a binary message.

```
sendMessage(sessionId, "application", "vnd.3gpp.sms", binData, binDataSize);
```

long com.portsip.PortSipSdk.sendOutOfDialogMessage (String *to*, String *contentType*, String *subMimeType*, boolean *isSMS*, byte[] *message*, int *messageLength*)

Send a out of dialog MESSAGE message to remote side.

Parameters

<i>to</i>	The message receiver. Likes sip: receiver@portsip.com
<i>contentType</i>	The mime type of MESSAGE message.
<i>subMimeType</i>	The sub mime type of MESSAGE message.
<i>isSMS</i>	Set to YES to specify "messagetype=SMS" in the To line, or NO to disable.
<i>message</i>	The contents that will be sent with MESSAGE message. Binary data allowed.
<i>messageLength</i>	The message size.

Returns

If the function succeeds, it will return a message ID that allows to track the message sending state in `onSendOutOfMessageSuccess` and `onSendOutOfMessageFailure`. If the function fails, it will return a specific error code that is less than 0.

Remarks

Example 1: Send a plain text message. Note: to send other languages text, please use the UTF8 to encode the message before sending.

```
sendOutOfDialogMessage("sip:user1@sip.portsip.com", "text", "plain", "hello", 6);
```

Example 2: Send a binary message.

```
sendOutOfDialogMessage("sip:user1@sip.portsip.com", "application",  
"vnd.3gpp.sms", binData, binDataSize);
```

long com.portsip.PortSipSdk.setPresenceMode (int *mode*)

Indicate the SDK uses the P2P mode for presence or presence agent mode.

Parameters

<i>mode</i>	0 - P2P mode; 1 - Presence Agent mode. Default is P2P mode.
-------------	---

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

Since presence agent mode requires the PBX/Server support the PUBLISH, please ensure you have your server and PortSIP PBX support this feature. For more details please visit:

<https://www.portsip.com/portsip-pbx>

long com.portsip.PortSipSdk.setDefaultSubscriptionTime (int *secs*)

Set the default expiration time to be used when creating a subscription.

Parameters

<i>secs</i>	The default expiration time of subscription.
-------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

long com.portsip.PortSipSdk.setDefaultPublicationTime (int *secs*)

Set the default expiration time to be used when creating a publication.

Parameters

<i>secs</i>	The default expiration time of publication.
-------------	---

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

long com.portsip.PortSipSdk.presenceSubscribe (String *contact*, String *subject*)

Send a SUBSCRIBE message for presence to a contact.

Parameters

<i>contact</i>	The target contact, it must be in the format of sip: contact001@sip.portsip.com .
<i>subject</i>	This subject text will be inserted into the SUBSCRIBE message. For example: "Hello, I'm Jason". The subject maybe is in UTF8 format. You should use UTF8 to decode it.

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.presenceTerminateSubscribe (long *subscribeId*)

Terminate the given presence subscription.

Parameters

<i>subscribeId</i>	The ID of the subscription.
--------------------	-----------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.presenceAcceptSubscribe (long *subscribeId*)

Accept the presence SUBSCRIBE request which received from contact.

Parameters

<i>subscribeId</i>	Subscription ID. When receiving a SUBSCRIBE request from contact, the event onPresenceRecvSubscribe will be triggered. The event includes the subscription ID.
--------------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.presenceRejectSubscribe (long *subscribeId*)

Reject a presence SUBSCRIBE request received from contact.

Parameters

<i>subscribeId</i>	Subscription ID. When receiving a SUBSCRIBE request from contact, the event onPresenceRecvSubscribe will be triggered. The event includes the subscription ID.
--------------------	--

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

int com.portsip.PortSipSdk.setPresenceStatus (long *subscribeId*, String *statusText*)

Send a NOTIFY message to contact to notify that presence status is online/offline/changed.

Parameters

<i>subscribeId</i>	Subscription ID. When receiving a SUBSCRIBE request from contact, the event onPresenceRecvSubscribe that includes the Subscription ID will be triggered.
<i>statusText</i>	The state text of presence status. For example: "I'm here", offline must use "offline"

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

long com.portsip.PortSipSdk.sendSubscription (String *to*, String *eventName*)

Send a SUBSCRIBE message to subscribe an event.

Parameters

<i>to</i>	The user/extension will be subscribed.
<i>eventName</i>	The event name to be subscribed.

Returns

If the function succeeds, it will return the ID of that SUBSCRIBE which is greater than 0. If the function fails, it will return a specific error code which is less than 0.

Remarks

Example 1, below code indicates that user/extension 101 is subscribed to MWI (Message Waiting notifications) for checking his voicemail: `int32 mwiSubId = sendSubscription("sip:101@test.com", "message-summary");`

Example 2, to monitor a user/extension call status, You can use code: `sendSubscription(mSipLib, "100", "dialog");` Extension 100 refers to the user/extension to be monitored. Once being monitored, when extension 100 hold a call or is ringing, the `onDialogStateUpdated` callback will be triggered.

int com.portsip.PortSipSdk.terminateSubscription (long *subscribeId*)

Terminate the given subscription.

Parameters

<i>subscribeId</i>	The ID of the subscription.
--------------------	-----------------------------

Returns

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code.

Remarks

For example, if you want stop check the MWI, use below code:

```
terminateSubscription (mwiSubId) ;
```

Class Documentation

com.portsip.PortSipEnumDefine.AUDIOCODEC Interface Reference

The documentation for this interface was generated from the following file:

- PortSipEnumDefine.java

com.portsip.PortSipEnumDefine.AudioDevice Enum Reference

Public Attributes

- SPEAKER_PHONE
- WIRED_HEADSET
- EARPIECE
- BLUETOOTH
- NONE

Detailed Description

[AudioDevice](#) list possible audio devices that we currently support.

The documentation for this enum was generated from the following file:

- PortSipEnumDefine.java

com.portsip.OnPortSIPEvent Interface Reference

Public Member Functions

- void [onRegisterSuccess](#) (String reason, int code, String sipMessage)
- void [onRegisterFailure](#) (String reason, int code, String sipMessage)
- void [onInviteIncoming](#) (long sessionId, String callerDisplayName, String caller, String calleeDisplayName, String callee, String audioCodecs, String videoCodecs, boolean existsAudio, boolean existsVideo, String sipMessage)
- void [onInviteTrying](#) (long sessionId)
- void [onInviteSessionProgress](#) (long sessionId, String audioCodecs, String videoCodecs, boolean existsEarlyMedia, boolean existsAudio, boolean existsVideo, String sipMessage)
- void [onInviteRinging](#) (long sessionId, String statusText, int statusCode, String sipMessage)
- void [onInviteAnswered](#) (long sessionId, String callerDisplayName, String caller, String calleeDisplayName, String callee, String audioCodecs, String videoCodecs, boolean existsAudio, boolean existsVideo, String sipMessage)
- void [onInviteFailure](#) (long sessionId, String reason, int code, String sipMessage)
- void [onInviteUpdated](#) (long sessionId, String audioCodecs, String videoCodecs, String screenCodecs, boolean existsAudio, boolean existsVideo, boolean existsScreen, String sipMessage)
- void [onInviteConnected](#) (long sessionId)
- void [onInviteBeginingForward](#) (String forwardTo)
- void [onInviteClosed](#) (long sessionId, String sipMessage)
- void [onDialogStateUpdated](#) (String BLFMonitoredUri, String BLFDialogState, String BLFDialogId, String BLFDialogDirection)
- void [onRemoteHold](#) (long sessionId)
- void [onRemoteUnHold](#) (long sessionId, String audioCodecs, String videoCodecs, boolean existsAudio, boolean existsVideo)
- void [onReceivedRefer](#) (long sessionId, long referId, String to, String from, String referSipMessage)
- void [onReferAccepted](#) (long sessionId)
- void [onReferRejected](#) (long sessionId, String reason, int code)
- void [onTransferTrying](#) (long sessionId)
- void [onTransferRinging](#) (long sessionId)
- void [onACTVTransferSuccess](#) (long sessionId)
- void [onACTVTransferFailure](#) (long sessionId, String reason, int code)
- void [onReceivedSignaling](#) (long sessionId, String message)
- void [onSendingSignaling](#) (long sessionId, String message)
- void [onWaitingVoiceMessage](#) (String messageAccount, int urgentNewMessageCount, int urgentOldMessageCount, int newMessageCount, int oldMessageCount)
- void [onWaitingFaxMessage](#) (String messageAccount, int urgentNewMessageCount, int urgentOldMessageCount, int newMessageCount, int oldMessageCount)
- void [onRecvDtmfTone](#) (long sessionId, int tone)
- void [onRecvOptions](#) (String optionsMessage)
- void [onRecvInfo](#) (String infoMessage)
- void [onRecvNotifyOfSubscription](#) (long subscribeId, String notifyMessage, byte[] messageData, int messageDataLength)
- void [onPresenceRecvSubscribe](#) (long subscribeId, String fromDisplayName, String from, String subject)
- void [onPresenceOnline](#) (String fromDisplayName, String from, String stateText)
- void [onPresenceOffline](#) (String fromDisplayName, String from)
- void [onRecvMessage](#) (long sessionId, String mimeType, String subMimeType, byte[] messageData, int messageDataLength)
- void [onRecvOutOfDialogMessage](#) (String fromDisplayName, String from, String toDisplayName, String to, String mimeType, String subMimeType, byte[] messageData, int messageDataLength, String sipMessage)
- void [onSendMessageSuccess](#) (long sessionId, long messageId)
- void [onSendMessageFailure](#) (long sessionId, long messageId, String reason, int code)

- void [onSendOutOfDialogMessageSuccess](#) (long messageId, String fromDisplayName, String from, String toDisplayName, String to)
- void [onSendOutOfDialogMessageFailure](#) (long messageId, String fromDisplayName, String from, String toDisplayName, String to, String reason, int code)
- void [onSubscriptionFailure](#) (long subscribeId, int statusCode)
- void [onSubscriptionTerminated](#) (long subscribeId)
- void [onPlayAudioFileFinished](#) (long sessionId, String fileName)
- void [onPlayVideoFileFinished](#) (long sessionId)
- void [onAudioDeviceChanged](#) (PortSipEnumDefine.AudioDevice audioDevice, Set<PortSipEnumDefine.AudioDevice > devices)
- void [onRTPPacketCallback](#) (long sessionId, int mediaType, int enum_direction, byte[] RTPPacket, int packetSize)
- void [onAudioRawCallback](#) (long sessionId, int enum_direction, byte[] data, int dataLength, int samplingFreqHz)
- void [onVideoRawCallback](#) (long sessionId, int enum_direction, int width, int height, byte[] data, int dataLength)

The documentation for this interface was generated from the following file:

- OnPortSIPEvent.java

com.portsip.PortSIPCameraCapturer Class Reference

Inherits CapturerObserver.

Public Member Functions

- void **InitializeVideoCapturer** (Context applicationContext)
- void **UnInitializeVideoCapturer** ()
- void **switchCamera** (int newDeviceId)
- void [onCapturerStarted](#) (boolean success)
- void [onCapturerStopped](#) ()
- void [onFrameCaptured](#) (VideoFrame frame)

Public Attributes

- CameraVideoCapturer **capturer**
-

Member Function Documentation

void com.portsip.PortSIPCameraCapturer.onCapturerStarted (boolean *success*)

Notify if the capturer have been started successfully or not.

void com.portsip.PortSIPCameraCapturer.onCapturerStopped ()

Notify that the capturer has been stopped.

void com.portsip.PortSIPCameraCapturer.onFrameCaptured (VideoFrame *frame*)

Delivers a captured frame.

The documentation for this class was generated from the following file:

- PortSIPCameraCapturer.java

com.portsip.PortSipEnumDefine Class Reference

Classes

- interface [AUDIOCODEC](#)
- enum [AudioDevice](#)

Static Public Attributes

- static final int **ENUM_AUDIOCODEC_G729** = 18
 - static final int **ENUM_AUDIOCODEC_PCMA** = 8
 - static final int **ENUM_AUDIOCODEC_PCMU** = 0
 - static final int **ENUM_AUDIOCODEC_GSM** = 3
 - static final int **ENUM_AUDIOCODEC_G722** = 9
 - static final int **ENUM_AUDIOCODEC_ILBC** = 97
 - static final int **ENUM_AUDIOCODEC_AMR** = 98
 - static final int **ENUM_AUDIOCODEC_AMRWB** = 99
 - static final int **ENUM_AUDIOCODEC_SPEEX** = 100
 - static final int **ENUM_AUDIOCODEC_SPEEXWB** = 102
 - static final int **ENUM_AUDIOCODEC_ISACWB** = 103
 - static final int **ENUM_AUDIOCODEC_ISACSWB** = 104
 - static final int **ENUM_AUDIOCODEC_OPUS** = 105
 - static final int **ENUM_AUDIOCODEC_DTMF** = 101
 - static final int [ENUM_VIDEOCODEC_NONE](#) = -1
 - static final int [ENUM_VIDEOCODEC_I420](#) = 133
 - static final int **ENUM_VIDEOCODEC_H264** = 125
 - static final int **ENUM_VIDEOCODEC_VP8** = 120
 - static final int **ENUM_VIDEOCODEC_VP9** = 122
 - static final int **ENUM_SRTTPOLICY_NONE** = 0
 - static final int **ENUM_SRTTPOLICY_FORCE** = 1
 - static final int **ENUM_SRTTPOLICY_PREFER** = 2
 - static final int **ENUM_TRANSPORT_UDP** = 0
 - static final int **ENUM_TRANSPORT_TLS** = 1
 - static final int **ENUM_TRANSPORT_TCP** = 2
 - static final int **ENUM_TRANSPORT_PERS_UDP** = 3
 - static final int **ENUM_TRANSPORT_PERS_TCP** = 4
 - static final int **ENUM_LOG_LEVEL_NONE** = -1
 - static final int **ENUM_LOG_LEVEL_ERROR** = 1
 - static final int **ENUM_LOG_LEVEL_WARNING** = 2
 - static final int **ENUM_LOG_LEVEL_INFO** = 3
 - static final int **ENUM_LOG_LEVEL_DEBUG** = 4
 - static final int **ENUM_DTMF_MOTHOD_RFC2833** = 0
 - static final int **ENUM_DTMF_MOTHOD_INFO** = 1
 - static final int **ENUM_DIRECTION_NONE** = 0
 - static final int [ENUM_DIRECTION_SEND_RECV](#) = 1
 - static final int [ENUM_DIRECTION_SEND](#) = 2
 - static final int [ENUM_DIRECTION_RECV](#) = 3
 - static final int [ENUM_RECORD_MODE_NONE](#) = 0
 - static final int [ENUM_RECORD_MODE_RECV](#) = 1
 - static final int [ENUM_RECORD_MODE_SEND](#) = 2
 - static final int [ENUM_RECORD_MODE_BOTH](#) = 3
 - static final int **ENUM_AUDIO_FILE_FORMAT_WAVE** = 1
 - static final int **ENUM_AUDIO_FILE_FORMAT_AMR** = 2
-

Member Data Documentation

final int com.portsip.PortSipEnumDefine.ENUM_VIDEOCODEC_NONE = -1 [static]

Used in startRecord only

final int com.portsip.PortSipEnumDefine.ENUM_VIDEOCODEC_I420 = 133 [static]

Used in startRecord only

final int com.portsip.PortSipEnumDefine.ENUM_DIRECTION_SEND_RECV = 1 [static]

both received and sent

final int com.portsip.PortSipEnumDefine.ENUM_DIRECTION_SEND = 2 [static]

Only the sent

final int com.portsip.PortSipEnumDefine.ENUM_DIRECTION_RECV = 3 [static]

Only the received

final int com.portsip.PortSipEnumDefine.ENUM_RECORD_MODE_NONE = 0 [static]

Not Recorded.

final int com.portsip.PortSipEnumDefine.ENUM_RECORD_MODE_RECV = 1 [static]

Only record the received data.

final int com.portsip.PortSipEnumDefine.ENUM_RECORD_MODE_SEND = 2 [static]

Only record the sent data.

final int com.portsip.PortSipEnumDefine.ENUM_RECORD_MODE_BOTH = 3 [static]

Record both received and sent data.

The documentation for this class was generated from the following file:

- PortSipEnumDefine.java

com.portsip.PortSipErrorcode Class Reference

Static Public Attributes

- static final int **ECoreErrorNone** = 0
- static final int **INVALID_SESSION_ID** = -1
- static final int **ECoreAlreadyInitialized** = -60000
- static final int **ECoreNotInitialized** = -60001
- static final int **ECoreSDKObjectNull** = -60002
- static final int **ECoreArgumentNull** = -60003
- static final int **ECoreInitializeWinsockFailure** = -60004
- static final int **ECoreUserNameAuthNameEmpty** = -60005
- static final int **ECoreInitiazeStackFailure** = -60006
- static final int **ECorePortOutOfRange** = -60007
- static final int **ECoreAddTcpTransportFailure** = -60008
- static final int **ECoreAddTlsTransportFailure** = -60009
- static final int **ECoreAddUdpTransportFailure** = -60010
- static final int **ECoreNotSupportMediaType** = -60011
- static final int **ECoreNotSupportDTMFValue** = -60012
- static final int **ECoreAlreadyRegistered** = -60021
- static final int **ECoreSIPServerEmpty** = -60022
- static final int **ECoreExpiresValueTooSmall** = -60023
- static final int **ECoreCallIdNotFound** = -60024
- static final int **ECoreNotRegistered** = -60025
- static final int **ECoreCalleeEmpty** = -60026
- static final int **ECoreInvalidUri** = -60027
- static final int **ECoreAudioVideoCodecEmpty** = -60028
- static final int **ECoreNoFreeDialogSession** = -60029
- static final int **ECoreCreateAudioChannelFailed** = -60030
- static final int **ECoreSessionTimerValueTooSmall** = -60040
- static final int **ECoreAudioHandleNull** = -60041
- static final int **ECoreVideoHandleNull** = -60042
- static final int **ECoreCallsClosed** = -60043
- static final int **ECoreCallAlreadyHold** = -60044
- static final int **ECoreCallNotEstablished** = -60045
- static final int **ECoreCallNotHold** = -60050
- static final int **ECoreSipMessaegEmpty** = -60051
- static final int **ECoreSipHeaderNotExist** = -60052
- static final int **ECoreSipHeaderValueEmpty** = -60053
- static final int **ECoreSipHeaderBadFormed** = -60054
- static final int **ECoreBufferTooSmall** = -60055
- static final int **ECoreSipHeaderValueListEmpty** = -60056
- static final int **ECoreSipHeaderParserEmpty** = -60057
- static final int **ECoreSipHeaderValueListNull** = -60058
- static final int **ECoreSipHeaderNameEmpty** = -60059
- static final int **ECoreAudioSampleNotmultiple** = -60060
- static final int **ECoreAudioSampleOutOfRange** = -60061
- static final int **ECoreInviteSessionNotFound** = -60062
- static final int **ECoreStackException** = -60063
- static final int **ECoreMimeTypeUnknown** = -60064
- static final int **ECoreDataSizeTooLarge** = -60065
- static final int **ECoreSessionNumsOutOfRange** = -60066
- static final int **ECoreNotSupportCallbackMode** = -60067
- static final int **ECoreNotFoundSubscribeId** = -60068
- static final int **ECoreCodecNotSupport** = -60069
- static final int **ECoreCodecParameterNotSupport** = -60070
- static final int **ECorePayloadOutofRange** = -60071
- static final int **ECorePayloadHasExist** = -60072
- static final int **ECoreFixPayloadCantChange** = -60073

- static final int **ECoreCodecTypeInvalid** = -60074
- static final int **ECoreCodecWasExist** = -60075
- static final int **ECorePayloadTypeInvalid** = -60076
- static final int **ECoreArgumentTooLong** = -60077
- static final int **ECoreMiniRtpPortMustIsEvenNum** = -60078
- static final int **ECoreCallInHold** = -60079
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- static final int **ECoreTrialVersionLicenseKey** = -60087
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- static final int **ECoreTrialVersionExpired** = -60091
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- static final int **ECoreUnsupportTransport** = -60094
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- static final int **ECoreUserNotFound** = -60096
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- static final int **ECoreCreateTransportFailure** = -60098
- static final int **ECoreTransportNotSet** = -60099
- static final int **ECoreECreateSignalingFailure** = -60100
- static final int **ECoreArgumentIncorrect** = -60101
- static final int **ECoreIVRObjectNull** = -61001
- static final int **ECoreIVRIndexOutOfRange** = -61002
- static final int **ECoreIVRReferFailure** = -61003
- static final int **ECoreIVRWaitingTimeOut** = -61004
- static final int **EAudioFileNameEmpty** = -70000
- static final int **EAudioChannelNotFound** = -70001
- static final int **EAudioStartRecordFailure** = -70002
- static final int **EAudioRegisterRecodingFailure** = -70003
- static final int **EAudioRegisterPlaybackFailure** = -70004
- static final int **EAudioGetStatisticsFailure** = -70005
- static final int **EAudioPlayFileAlreadyEnable** = -70006
- static final int **EAudioPlayObjectNotExist** = -70007
- static final int **EAudioPlaySteamNotEnabled** = -70008
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- static final int **EAudioCreateAudioConferenceFailure** = -70010
- static final int **EAudioOpenPlayFileFailure** = -70011
- static final int **EAudioPlayFileModeNotSupport** = -70012
- static final int **EAudioPlayFileFormatNotSupport** = -70013
- static final int **EAudioPlaySteamAlreadyEnabled** = -70014
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- static final int **EAudioPlayFileUnknowSeekOrigin** = -70018
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- static final int **EAudioVolumeOutOfRange** = -70020
- static final int **EVideoFileNameEmpty** = -80000
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- static final int **EVideoUnsupportCaptureRotate** = -80011
- static final int **VideoUnsupportCaptureResolution** = -80012
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- static final int **EMTUOutOfRange** = -80014
- static final int **EDeviceGetDeviceNameFailure** = -90001

The documentation for this class was generated from the following file:

- PortSipErrorcode.java

com.portsip.PortSipSdk Class Reference

Inherits AudioManagerEvents.

Classes

- class **MainHandler**

Public Member Functions

- void **onAudioDeviceChanged** (AppRTCAudioManager.AudioDevice innerAdioDevice, Set<AppRTCAudioManager.AudioDevice > innerSet)
- void **CreateCallManager** (Context context)
- PortSipEnumDefine.AudioDevice **getSelectedAudioDevice** ()
- void **DeleteCallManager** ()
- int **initialize** (int enum_transport, String localIP, int localSIPPort, int enum_LogLevel, String LogPath, int maxLines, String agent, int audioDeviceLayer, int videoDeviceLayer, String TLSCertificatesRootPath, String TLSCipherList, boolean verifyTLSCertificate, String dnsServers)
- int **setInstanceId** (String instanceId)
- int **setUser** (String userName, String displayName, String authName, String password, String userDomain, String SIPServer, int SIPServerPort, String STUNServer, int STUNServerPort, String outboundServer, int outboundServerPort)
- void **removeUser** ()
remove user account info.
- int **registerServer** (int expires, int retryTimes)
- int **refreshRegistration** (int expires)
- int **unRegisterServer** ()
- int **setDisplayname** (String displayName)
- int **setLicenseKey** (String key)
- int **addAudioCodec** (int enum_audiocodec)
- int **addVideoCodec** (int enum_videocodec)
- boolean **isAudioCodecEmpty** ()
- boolean **isVideoCodecEmpty** ()
- int **setAudioCodecPayloadType** (int enum_audiocodec, int payloadType)
- int **setVideoCodecPayloadType** (int enum_videocodec, int payloadType)
- void **clearAudioCodec** ()
- void **clearVideoCodec** ()
- int **setAudioCodecParameter** (int enum_audiocodec, String sdpParameter)
- int **setVideoCodecParameter** (int enum_videocodec, String sdpParameter)
- String **getVersion** ()
- int **enableRport** (boolean enable)
- int **enableEarlyMedia** (boolean enable)
Enable/disable rport(RFC3581).
- int **enableReliableProvisional** (boolean enable)
- int **enable3GppTags** (boolean enable)
- void **enableCallbackSignaling** (boolean enableSending, boolean enableReceived)
- void **setSrtpPolicy** (int enum_srtpolicy)
- int **setRtpPortRange** (int minimumRtpPort, int maximumRtpPort)
- int **enableCallForward** (boolean forBusyOnly, String forwardTo)
- int **disableCallForward** ()
- int **enableSessionTimer** (int timerSeconds)
- void **disableSessionTimer** ()
- void **setDoNotDisturb** (boolean state)
- void **enableAutoCheckMwi** (boolean state)
- int **setRtpKeepAlive** (boolean state, int keepAlivePayloadType, int deltaTransmitTimeMS)
- int **setKeepAliveTime** (int keepAliveTime)

- int [setAudioSamples](#) (int ptime, int maxptime)
- int [addSupportedMimeType](#) (String methodName, String mimeType, String subMimeType)
- String [getSipMessageHeaderValue](#) (String sipMessage, String headerName)
- long [addSipMessageHeader](#) (long sessionId, String methodName, int msgType, String headerName, String headerValue)
- int [removeAddedSipMessageHeader](#) (long addedSipMessageId)
- void [clearAddedSipMessageHeaders](#) ()
- long [modifySipMessageHeader](#) (long sessionId, String methodName, int msgType, String headerName, String headerValue)
- int [removeModifiedSipMessageHeader](#) (long modifiedSipMessageId)
- void [clearModifiedSipMessageHeaders](#) ()
- int [setVideoDeviceId](#) (int deviceId)
- int [setVideoOrientation](#) (int rotation)
- int [enableVideoHardwareCodec](#) (boolean enableHWEncoder, boolean enableHWD decoder)
- int [setVideoResolution](#) (int width, int height)
- int [setVideoCropAndScale](#) (boolean enable)
- int [setAudioBitrate](#) (long sessionId, int enum_audiocodec, int bitrateKbps)
- int [setVideoBitrate](#) (long sessionId, int bitrateKbps)
- int [setVideoFrameRate](#) (long sessionId, int frameRate)
- int [sendVideo](#) (long sessionId, boolean send)
- int [setRemoteVideoWindow](#) (long sessionId, [PortSIPVideoRenderer](#) renderer)
- int [setRemoteScreenWindow](#) (long sessionId, [PortSIPVideoRenderer](#) renderer)
- void [displayLocalVideo](#) (boolean state, boolean mirror, [PortSIPVideoRenderer](#) renderer)
- int [setVideoNackStatus](#) (boolean state)
- int [setChannelOutputVolumeScaling](#) (long sessionId, int scaling)
- int [setChannelInputVolumeScaling](#) (long sessionId, int scaling)
- void [enableAudioManager](#) (boolean state)
- Set< PortSipEnumDefine.AudioDevice > [getAudioDevices](#) ()
- int [setAudioDevice](#) (PortSipEnumDefine.AudioDevice defaultDevice)
- long [call](#) (String callee, boolean sendSdp, boolean videoCall)
- int [rejectCall](#) (long sessionId, int code)
- int [hangUp](#) (long sessionId)
- int [answerCall](#) (long sessionId, boolean videoCall)
- int [updateCall](#) (long sessionId, boolean enableAudio, boolean enableVideo)
- int [hold](#) (long sessionId)
- int [unHold](#) (long sessionId)
- int [muteSession](#) (long sessionId, boolean muteIncomingAudio, boolean muteOutgoingAudio, boolean muteIncomingVideo, boolean muteOutgoingVideo)
- int [forwardCall](#) (long sessionId, String forwardTo)
- long [pickupBLFCall](#) (String replaceDialogId, boolean videoCall)
- int [sendDtmf](#) (long sessionId, int enum_dtmfMethod, int code, int dtmfDuration, boolean playDtmfTone)
- int [refer](#) (long sessionId, String referTo)
- int [attendedRefer](#) (long sessionId, long replaceSessionId, String referTo)
- int [attendedRefer2](#) (long sessionId, long replaceSessionId, String replaceMethod, String target, String referTo)
- int [outOfDialogRefer](#) (long replaceSessionId, String replaceMethod, String target, String referTo)
- long [acceptRefer](#) (long referId, String referSignaling)
- int [rejectRefer](#) (long referId)
- int [enableSendPcmStreamToRemote](#) (long sessionId, boolean state, int streamSamplesPerSec)
- int [sendPcmStreamToRemote](#) (long sessionId, byte[] data, int dataLength)
- int [enableSendVideoStreamToRemote](#) (long sessionId, boolean state)
- int [sendVideoStreamToRemote](#) (long sessionId, byte[] data, int dataLength, int width, int height)
- long [enableRtpCallback](#) (long sessionId, int mediaType, int directionMode)
- void [enableAudioStreamCallback](#) (long sessionId, boolean enable, int enum_direction)
- void [enableVideoStreamCallback](#) (long sessionId, int enum_direction)
- int [startRecord](#) (long sessionId, String recordFilePath, String recordFileName, boolean appendTimeStamp, int enum_audioFileFormat, int enum_audioRecordMode, int enum_videocodec, int enum_videoRecordMode)
- int [stopRecord](#) (long sessionId)

- int [playVideoFileToRemote](#) (long sessionId, String aviFile, boolean loop, boolean playAudio)
- int [stopPlayVideoFileToRemote](#) (long sessionId)
- int [playAudioFileToRemote](#) (long sessionId, String filename, int fileSamplesPerSec, boolean loop)
- int [stopPlayAudioFileToRemote](#) (long sessionId)
- int [playAudioFileToRemoteAsBackground](#) (long sessionId, String filename, int fileSamplesPerSec)
- int [stopPlayAudioFileToRemoteAsBackground](#) (long sessionId)
- void [audioPlayLoopbackTest](#) (boolean enable)
- int [createAudioConference](#) ()
- int [createVideoConference](#) ([PortSIPVideoRenderer](#) conferenceVideoWindow, int videoWidth, int videoHeight, boolean displayLocalVideoInConference)
- void [destroyConference](#) ()
- int [setConferenceVideoWindow](#) ([PortSIPVideoRenderer](#) conferenceVideoWindow)
- int [joinToConference](#) (long sessionId)
- int [removeFromConference](#) (long sessionId)
- int [setAudioRtcpBandwidth](#) (long sessionId, int BitsRR, int BitsRS, int KBitsAS)
- int [setVideoRtcpBandwidth](#) (long sessionId, int BitsRR, int BitsRS, int KBitsAS)
- int [enableAudioQos](#) (boolean state)
- int [enableVideoQos](#) (boolean state)
- int [setVideoMTU](#) (int mtu)
- int [getAudioStatistics](#) (long sessionId, int[] statistics)
- int [getVideoStatistics](#) (long sessionId, int[] statistics)
- void [enableVAD](#) (boolean state)
- void [enableAEC](#) (boolean state)
- void [enableCNG](#) (boolean state)
- void [enableAGC](#) (boolean state)
- void [enableANS](#) (boolean state)
- int [sendOptions](#) (String to, String sdp)
- int [sendInfo](#) (long sessionId, String mimeType, String subMimeType, String infoContents)
- long [sendMessage](#) (long sessionId, String mimeType, String subMimeType, byte[] message, int messageLength)
- long [sendOutOfDialogMessage](#) (String to, String mimeType, String subMimeType, boolean isSMS, byte[] message, int messageLength)
- long [setPresenceMode](#) (int mode)
- long [setDefaultSubscriptionTime](#) (int secs)
- long [setDefaultPublicationTime](#) (int secs)
- long [presenceSubscribe](#) (String contact, String subject)
- int [presenceTerminateSubscribe](#) (long subscribeId)
- int [presenceAcceptSubscribe](#) (long subscribeId)
- int [presenceRejectSubscribe](#) (long subscribeId)
- int [setPresenceStatus](#) (long subscribeId, String statusText)
- long [sendSubscription](#) (String to, String eventName)
Send a SUBSCRIBE message to subscribe an event.
- int [terminateSubscription](#) (long subscribeId)
- void **receiveSIPEvent** (long sipCommand)
- void **rtpPacketCallback** (long sessionId, int mediaType, int directionMode, byte[] RTPPacket, int packetSize)
- void **audioRawCallback** (long sessionId, int enum_audioCallbackMode, byte[] data, int dataLength, int samplingFreqHz)
- void **videoRawCallback** (long sessionId, int enum_videoCallbackMode, int width, int height, byte[] data, int dataLength)
- void **setOnPortSIPEvent** ([OnPortSIPEvent](#) l)

Detailed Description

Author

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Version

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The documentation for this class was generated from the following file:

- PortSipSdk.java

com.portsip.PortSIPVideoRenderer Class Reference

Inherits `SurfaceViewRenderer`.

Classes

- enum `ScalingType`

Public Member Functions

- [PortSIPVideoRenderer](#) (`Context context`)
- [PortSIPVideoRenderer](#) (`Context context`, `AttributeSet attrs`)
- void `setScalingType` (`ScalingType scalingType`)
- void [setVideoRotation](#) (`int rotation`)
- void [release](#) ()
- void `surfaceDestroyed` (`SurfaceHolder holder`)

Detailed Description

Display the video stream on a `SurfaceView`.

Constructor & Destructor Documentation

com.portsip.PortSIPVideoRenderer.PortSIPVideoRenderer (`Context context`)

Standard View constructor. In order to render something, you must first call `init()`.

com.portsip.PortSIPVideoRenderer.PortSIPVideoRenderer (`Context context`, `AttributeSet attrs`)

Standard View constructor. In order to render something, you must first call `init()`.

Member Function Documentation

void com.portsip.PortSIPVideoRenderer.setVideoRotation (`int rotation`)

Standard View constructor. In order to render something, you must first call `init()`.

void com.portsip.PortSIPVideoRenderer.release ()

Block until any pending frame is returned and all GL resources released, even if an interrupt occurs. If an interrupt occurs during [release\(\)](#), the interrupt flag will be set. This function should be called before the Activity is destroyed and the `EGLContext` is still valid. If you don't call this function, the GL resources might leak.

The documentation for this class was generated from the following file:

- `PortSIPVideoRenderer.java`

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