

PortSIP VoIP SDK Manual for iOS

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Welcome to PortSIP VoIP SDK

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), Eclipse (Java for Android) 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 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

Development using the PortSIP VoIP/IMS SDK for iOS requires an Intel-based Macintosh running Snow Leopard (OS X 10.8 or higher)

Apple's iOS SDK

If you are not yet a registered Apple developer, to be able to develop applications for the iOS, you do need to become a registered Apple developer. After registered, Apple grants you free access to a select set of technical resources, tools, and information for developing with iOS, Mac OS X, and Safari. You can open [registration page](#) and enroll.

Once registered, you can then go to the [iOS Dev Center](#), login and download the iOS SDK. The SDK contains documentation, frameworks, tools, and a simulator to help develop iOS applications. XCode (the developer toolset for iOS application development) is included in the download as well so you do not need to purchase any developer tools to build iOS

applications - that is included in the enrollment fee. You will need to use a minimum of iOS SDK 10 for developing iPhone and iPod Touch applications. At the time of writing this document, iOS SDK 10 was the most recent version available and supported.

Note:

Beta and GM seed versions of the iOS SDK are generally not supported unless noted otherwise. Regardless of the iOS SDK you're using for development, you can still target your application for devices running on an older iOS version by configuring your Xcode project's iOS Deployment Target build settings. Be sure to add runtime checks where appropriate to ensure that you use only those iOS features available on the target platform/device. If your application attempts to use iOS features that are not available on the device, your application may crash.

Device Requirements

Applications built with PortSIP VoIP/IMS SDK for iOS can be run on iPhone 4S or higher, iPod touch 4 or higher, and iPad 2 or higher devices. These devices must be running iOS8 or higher. We strongly recommend that you test your applications on actual devices to ensure that they work as expected and perform well. Testing on the simulator alone does not provide a good measure of how the application will perform on the physical device.

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 with your Xcode:
4. Compile the sample project directly. The trial version SDK allows you a 2-3 minutes conversation.

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 directory.
2. Run the Xcode and create a new iOS Project.
3. Drag and drop PortSIPVoIPSDK.framework from Finder to XCode->Frameworks.
4. Add depend Frameworks:
Build Phases->Link Binary With Libraries, add libc++.tbd, libresolv.tbd, VideoToolbox.framework, GLKit.framework, MetalKit.framework.
5. Add "-ObjC" to "Build Settings"-> "Other Linker Flags"
6. Add the code in .h file to import the SDK, example:
`#import <PortSIPVoIPSDK/PortSIPVoIPSDK.h>`
7. Inherit the interface PortSIPEventDelegate to process the callback events. For example:
`@interface AppDelegate : UIResponder <UIApplicationDelegate,PortSIPEventDelegate>{
 PortSIPSDK* mPortSIPSDK;
}
@end`
8. Initialize sdk. For example:
`mPortSIPSDK = [[PortSIPSDK alloc] init];`

```
mPortSIPSDK.delegate = self;
```

9. For more details, please read the Sample project source code.

4. How to upgrade existing iOS Project from v15 to V16?

1. Remove PortSIPLib.framework from XCode->Frameworks.
2. Drag and drop PortSIPVoIPSDK.framework from Finder to XCode->Frameworks.
3. Remove libstdc++.6.tbd from Build Phases->Link Binary with Libraries.
4. Add GLKit.framework, MetalKit.framework to Build Phases->Link Binary with Libraries.
5. Add "-ObjC" to "Build Settings"-> "Other Linker Flags".
6. Replace #import <PortSIPLib/PortSIPSDK.h> with #import <PortSIPVoIPSDK/PortSIPVoIPSDK.h>.
7. Remove APIs setVideoOrientation, enableVideoDecoderCallback.
8. Remove events onVideoDecoderCallback and updateRemoteVideoWindowSize.
9. Remove applicationWillEnterForeground and applicationDidBecomeActive process code.
10. Display video refer VideoViewController.

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

1. Download and extract the SDK sample project ZIP file in 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. 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.
4. The log box will appear "Initialized" if the SDK is successfully initialized.
5. 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".

6. 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", "onReceivedRtpPacket", "onSendingRtpPacket" callbacks.

7. Does the SDK support native 64-bit?

Yes, both 32-bit and 64-bit are supported for SDK.

7. Does the SDK support VoIP PUSH?

Yes, please refer to <https://www.portsip.com/knowledge-base/> for more details.

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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:

NSObject	
PortSIPSDK	76
<NSObject>	
<PortSIPEventDelegate >	74
UIView	
PortSIPVideoRenderView	83

Class Index

Class List

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

<u><PortSIPEventDelegate></u> (PortSIP SDK Callback events Delegate)	74
<u>PortSIPSDK</u> (PortSIP VoIP SDK functions class)	76
<u>PortSIPVideoRenderView</u> (PortSIP VoIP SDK Video Render View class)	83

Module Documentation

SDK Callback events

Modules

- [Register events](#)
- [Call events](#)
- [Refer events](#)
- [Signaling events](#)
- [MWI events](#)
- [DTMF events](#)
- [INFO/OPTIONS message events](#)
- [Presence events](#)
- [MESSAGE message events](#)
- [Play audio and video file finished events](#)
- [RTP callback events](#)
- [Audio and video stream callback events](#)

Detailed Description

SDK Callback events

Register events

Functions

- (void) - [<PortSIPEventDelegate >::onRegisterSuccess:statusCode:sipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onRegisterFailure:statusCode:sipMessage:](#)

Detailed Description

Register events

Function Documentation

- (void PortSIPEventDelegate) **onRegisterSuccess:** (char *) *statusText* **statusCode:** (int) *statusCode* **sipMessage:** (char *) *sipMessage*

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

Parameters:

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

- (void PortSIPEventDelegate) **onRegisterFailure:** (char *) *statusText* **statusCode:** (int) *statusCode* **sipMessage:** (char *) *sipMessage*

If registration to SIP server fails, this event will be triggered.

Parameters:

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

Call events

Functions

- (void) - [<PortSIPEventDelegate >::onInviteIncoming:callerDisplayName:caller:calleeDisplayName:callee:audioCodecs:videoCodecs:existsAudio:existsVideo:sipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onInviteTrying:](#)
- (void) - [<PortSIPEventDelegate >::onInviteSessionProgress:audioCodecs:videoCodecs:existsEarlyMedia:existsAudio:existsVideo:sipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onInviteRinging:statusText:statusCode:sipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onInviteAnswered:callerDisplayName:caller:calleeDisplayName:callee:audioCodecs:videoCodecs:existsAudio:existsVideo:sipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onInviteFailure:reason:code:sipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onInviteUpdated:audioCodecs:videoCodecs:existsAudio:existsVideo:sipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onInviteConnected:](#)
- (void) - [<PortSIPEventDelegate >::onInviteBeginningForward:](#)
- (void) - [<PortSIPEventDelegate >::onInviteClosed:](#)
- (void) - [<PortSIPEventDelegate >::onDialogStateUpdated:BLFDialogState:BLFDialogId:BLFDialogDirection:](#)
- (void) - [<PortSIPEventDelegate >::onRemoteHold:](#)
- (void) - [<PortSIPEventDelegate >::onRemoteUnHold:audioCodecs:videoCodecs:existsAudio:existsVideo:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onInviteIncoming: (long) *sessionId* callerDisplayName: (char *) *callerDisplayName* caller: (char *) *caller* calleeDisplayName: (char *) *calleeDisplayName* callee: (char *) *callee* audioCodecs: (char *) *audioCodecs* videoCodecs: (char *) *videoCodecs* existsAudio: (BOOL) *existsAudio* existsVideo: (BOOL) *existsVideo* sipMessage: (char *) *sipMessage*

When the 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>caller</i>	The caller.
<i>calleeDisplayName</i>	The display name of callee.
<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 indicates that this call includes the audio.
<i>existsVideo</i>	By setting to true, it indicates that this call includes the video.
<i>sipMessage</i>	The SIP message received.

- (void PortSIPEventDelegate) 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.
------------------	-----------------------------

**- (void PortSIPEventDelegate) onInviteSessionProgress: (long) *sessionId*
audioCodecs: (char *) *audioCodecs* videoCodecs: (char *) *videoCodecs*
existsEarlyMedia: (BOOL) *existsEarlyMedia* existsAudio: (BOOL) *existsAudio*
existsVideo: (BOOL) *existsVideo* sipMessage: (char *) *sipMessage***

Once the caller received the "183 session in progress" message, 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 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 indicates that the call has early media.
<i>existsAudio</i>	By setting to true, it indicates that this call includes the audio.
<i>existsVideo</i>	By setting to true, it indicates that this call includes the video.
<i>sipMessage</i>	The SIP message received.

**- (void PortSIPEventDelegate) onInviteRinging: (long) *sessionId* statusText: (char *)
statusText statusCode: (int) *statusCode* sipMessage: (char *) *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 PortSIPEventDelegate) onInviteAnswered: (long) *sessionId* callerDisplayName:
(char *) *callerDisplayName* caller: (char *) *caller* calleeDisplayName: (char *)
calleeDisplayName callee: (char *) *callee* audioCodecs: (char *) *audioCodecs*
videoCodecs: (char *) *videoCodecs* existsAudio: (BOOL) *existsAudio* existsVideo:
(BOOL) *existsVideo* sipMessage: (char *) *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>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 indicates that this call includes the audio.
<i>existsVideo</i>	By setting to true, it indicates that this call includes the video.
<i>sipMessage</i>	The SIP message received.

- (void PortSIPEventDelegate) onInviteFailure: (long) *sessionId* reason: (char *)
reason code: (int) *code* sipMessage: (char *) *sipMessage*

If the outgoing call fails, this event will be triggered.

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 PortSIPEventDelegate) onInviteUpdated: (long) *sessionId* audioCodecs: (char *)
audioCodecs videoCodecs: (char *) *videoCodecs* existsAudio: (BOOL)
existsAudio existsVideo: (BOOL) *existsVideo* sipMessage: (char *) *sipMessage*

This event will be triggered when remote party updates this 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>existsAudio</i>	By setting to true, it indicates that this call includes the audio.
<i>existsVideo</i>	By setting to true, it indicates that this call includes the video.
<i>sipMessage</i>	The SIP message received.

- (void PortSIPEventDelegate) 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 is connected,
otherwise it will return error.

Parameters:

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

- (void PortSIPEventDelegate) onInviteBeginingForward: (char *) *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 for forwarding.
------------------	------------------------------------

- (void PortSIPEventDelegate) onInviteClosed: (long) *sessionId*

This event will be triggered once remote side closes the call.

Parameters:

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

- (void PortSIPEventDelegate) onDialogStateUpdated: (char *) *BLFMonitoredUri*
BLFDialogState: (char *) *BLFDialogState* *BLFDialogId*: (char *) *BLFDialogId*
BLFDialogDirection: (char *) *BLFDialogDirection*

If a user subscribed and his dialog status monitored, when the monitored user is holding a
call or 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 PortSIPEventDelegate) onRemoteHold: (long) *sessionId*

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

Parameters:

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

- (void PortSIPEventDelegate) onRemoteUnHold: (long) *sessionId* audioCodecs: (char *) *audioCodecs* videoCodecs: (char *) *videoCodecs* existsAudio: (BOOL) *existsAudio* existsVideo: (BOOL) *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 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 indicates that this call includes the audio.
<i>existsVideo</i>	By setting to true, it indicates that this call includes the video.

Refer events

Functions

- (void) - [<PortSIPEventDelegate >::onReceivedRefer:referId:to:from:referSipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onReferAccepted:](#)
- (void) - [<PortSIPEventDelegate >::onReferRejected:reason:code:](#)
- (void) - [<PortSIPEventDelegate >::onTransferTrying:](#)
- (void) - [<PortSIPEventDelegate >::onTransferRinging:](#)
- (void) - [<PortSIPEventDelegate >::onACTVTransferSuccess:](#)
- (void) - [<PortSIPEventDelegate >::onACTVTransferFailure:reason:code:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onReceivedRefer: (long) *sessionId* referId: (long) *referId* to: (char *) *to* from: (char *) *from* referSipMessage: (char *) *referSipMessage*

This event will be triggered once receiving 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 acceptRefer or rejectRefer.
<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 PortSIPEventDelegate) 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 PortSIPEventDelegate) onReferRejected: (long) *sessionId* reason: (char *)
reason code: (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>	Reason for rejecting.
<i>code</i>	Rejecting code.

- (void PortSIPEventDelegate) 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.
------------------	-----------------------------

- (void PortSIPEventDelegate) onTransferRinging: (long) *sessionId*

When the refer call rings, this event will be triggered.

Parameters:

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

- (void PortSIPEventDelegate) onACTVTransferSuccess: (long) *sessionId*

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

Parameters:

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

**- (void PortSIPEventDelegate) onACTVTransferFailure: (long) *sessionId* reason: (char *)
reason code: (int) *code***

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

Parameters:

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

Signaling events

Functions

- (void) - [<PortSIPEventDelegate >::onReceivedSignaling:message:](#)
- (void) - [<PortSIPEventDelegate >::onSendingSignaling:message:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onReceivedSignaling: (long) *sessionId* message: (char *) *message*

This event will be triggered when receiving an 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 SIP message received.

- (void PortSIPEventDelegate) onSendingSignaling: (long) *sessionId* message: (char *) *message*

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

Parameters:

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

MWI events

Functions

- (void) - [<PortSIPEventDelegate >::onWaitingVoiceMessage:urgentNewMessageCount:urgentOldMessageCount:newMessageCount:oldMessageCount:](#)
- (void) - [<PortSIPEventDelegate >::onWaitingFaxMessage:urgentNewMessageCount:urgentOldMessageCount:newMessageCount:oldMessageCount:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onWaitingVoiceMessage: (char *) *messageAccount* urgentNewMessageCount: (int) *urgentNewMessageCount* urgentOldMessageCount: (int) *urgentOldMessageCount* newMessageCount: (int) *newMessageCount* oldMessageCount: (int) *oldMessageCount*

If there are any waiting voice messages (MWI), this event will be triggered.

Parameters:

<i>messageAccount</i>	Account for voice message.
-----------------------	----------------------------

<i>urgentNewMessageCount</i>	Count of new urgent messages.
<i>urgentOldMessageCount</i>	Count of old urgent messages.
<i>newMessageCount</i>	Count of new messages.
<i>oldMessageCount</i>	Count of old messages.

- (void PortSIPEventDelegate) onWaitingFaxMessage: (char *) *messageAccount* urgentNewMessageCount: (int) *urgentNewMessageCount* urgentOldMessageCount: (int) *urgentOldMessageCount* newMessageCount: (int) *newMessageCount* oldMessageCount: (int) *oldMessageCount*

If there are any waiting fax messages (MWI), this event will be triggered.

Parameters:

<i>messageAccount</i>	Account for fax message.
<i>urgentNewMessageCount</i>	Count of new urgent messages.
<i>urgentOldMessageCount</i>	Count of old urgent messages.
<i>newMessageCount</i>	Count of new messages.
<i>oldMessageCount</i>	Count of old messages.

DTMF events

Functions

- (void) - [<PortSIPEventDelegate >::onRecvDtmfTone:tone:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onRecvDtmfTone: (long) *sessionId* tone: (int) *tone*

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

Parameters:

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

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) - [<PortSIPEventDelegate >::onRecvOptions:](#)
- (void) - [<PortSIPEventDelegate >::onRecvInfo:](#)
- (void) - [<PortSIPEventDelegate >::onRecvNotifyOfSubscription:notifyMessage:messageData:messageDataLength:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onRecvOptions: (char *) *optionsMessage*

This event will be triggered when receiving the OPTIONS message.

Parameters:

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

- (void PortSIPEventDelegate) onRecvInfo: (char *) *infoMessage*

This event will be triggered when receiving the INFO message.

Parameters:

<i>infoMessage</i>	The whole received INFO message in text format.
--------------------	---

- (void PortSIPEventDelegate) onRecvNotifyOfSubscription: (long) *subscribeId* notifyMessage: (char *) *notifyMessage* messageData: (unsigned char *) *messageData* messageDataLength: (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 can be either text or binary data.
<i>messageDataLength</i>	The length of "messageData".

Presence events

Functions

- (void) - [<PortSIPEventDelegate >::onPresenceRecvSubscribe:fromDisplayName:from:subject:](#)
- (void) - [<PortSIPEventDelegate >::onPresenceOnline:from:stateText:](#)
- (void) - [<PortSIPEventDelegate >::onPresenceOffline:from:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onPresenceRecvSubscribe: (long) *subscribeId* fromDisplayName: (char *) *fromDisplayName* from: (char *) *from* subject: (char *) *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 PortSIPEventDelegate) onPresenceOnline: (char *) *fromDisplayName* from: (char *) *from* stateText: (char *) *stateText*

This event will be triggered when the contact is online or changes presence status.

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 PortSIPEventDelegate) onPresenceOffline: (char *) *fromDisplayName* from: (char *) *from*

When the contact status is changed to 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

MESSAGE message events

Functions

- (void) - [<PortSIPEventDelegate >::onRecvMessage:mimeType:subMimeType:messageData:messageDataLength:](#)

- (void) - [<PortSIPEventDelegate >::onRecvOutOfDialogMessage:from:toDisplayName:to:mimeType:subMimeType:messageData:messageDataLength:sipMessage:](#)
- (void) - [<PortSIPEventDelegate >::onSendMessageSuccess:messageId:](#)
- (void) - [<PortSIPEventDelegate >::onSendMessageFailure:messageId:reason:code:](#)
- (void) - [<PortSIPEventDelegate >::onSendOutOfDialogMessageSuccess:fromDisplayName:from:toDisplayName:to:](#)
- (void) - [<PortSIPEventDelegate >::onSendOutOfDialogMessageFailure:fromDisplayName:from:toDisplayName:to:reason:code:](#)
- (void) - [<PortSIPEventDelegate >::onSubscriptionFailure:statusCode:](#)
- (void) - [<PortSIPEventDelegate >::onSubscriptionTerminated:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onRecvMessage: (long) *sessionId* mimeType: (char *) *mimeType* subMimeType: (char *) *subMimeType* messageData: (unsigned char *) *messageData* messageDataLength: (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 either text or binary data.
<i>messageDataLength</i>	The length of "messageData".

- (void PortSIPEventDelegate) onRecvOutOfDialogMessage: (char *) *fromDisplayName* from: (char *) *from* toDisplayName: (char *) *toDisplayName* to: (char *) *to* mimeType: (char *) *mimeType* subMimeType: (char *) *subMimeType* messageData: (unsigned char *) *messageData* messageDataLength: (int) *messageDataLength* sipMessage: (char *) *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 recipient.
<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.
<i>messageDataLength</i>	The length of "messageData".
<i>sipMessage</i>	The SIP message received.

- (void PortSIPEventDelegate) onSendMessageSuccess: (long) *sessionId* messageId: (long) *messageId*

This event will be triggered when the message is sent successfully in dialog.

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 PortSIPEventDelegate) onSendMessageFailure: (long) *sessionId* messageId: (long) *messageId* reason: (char *) *reason* code: (int) *code*

This event will be triggered when the message fails to be sent out of dialog.

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 PortSIPEventDelegate) onSendOutOfDialogMessageSuccess: (long) *messageId* fromDisplayName: (char *) *fromDisplayName* from: (char *) *from* toDisplayName: (char *) *toDisplayName* to: (char *) *to*

This event will be triggered when the message is sent successfully out of dialog.

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 PortSIPEventDelegate) onSendOutOfDialogMessageFailure: (long) *messageId* fromDisplayName: (char *) *fromDisplayName* from: (char *) *from* toDisplayName: (char *) *toDisplayName* to: (char *) *to* reason: (char *) *reason* code: (int) *code*

This event will be triggered when the message fails to be sent out of dialog.

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 recipient.
<i>reason</i>	The failure reason.
<i>code</i>	The failure code.

- (void PortSIPEventDelegate) onSubscriptionFailure: (long) *subscribeId* statusCode: (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 PortSIPEventDelegate) 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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Play audio and video file finished events

Functions

- (void) - [<PortSIPEventDelegate >::onPlayAudioFileFinished:fileName:](#)
- (void) - [<PortSIPEventDelegate >::onPlayVideoFileFinished:](#)

Detailed Description**Function Documentation**

- (void PortSIPEventDelegate) onPlayAudioFileFinished: (long) *sessionId* fileName: (char *) *fileName*

If playAudioFileToRemote function is called 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 PortSIPEventDelegate) onPlayVideoFileFinished: (long) *sessionId*

If playVideoFileToRemote function is called 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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RTP callback events

Functions

- (void) - [<PortSIPEventDelegate >::onReceivedRTPPacket:isAudio:RTPPacket:packetSize:](#)
- (void) - [<PortSIPEventDelegate >::onSendingRTPPacket:isAudio:RTPPacket:packetSize:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onReceivedRTPPacket: (long) *sessionId* isAudio: (BOOL) *isAudio* RTPPacket: (unsigned char *) *RTPPacket* packetSize: (int) *packetSize*

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

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>isAudio</i>	If the received RTP packet is of audio, this parameter is true, otherwise false.
<i>RTPPacket</i>	The memory of whole RTP packet.
<i>packetSize</i>	The size of received RTP Packet.

Note:

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.

- (void PortSIPEventDelegate) onSendingRTPPacket: (long) *sessionId* isAudio: (BOOL) *isAudio* RTPPacket: (unsigned char *) *RTPPacket* packetSize: (int) *packetSize*

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

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>isAudio</i>	If the received RTP packet is of audio, this parameter returns true, otherwise false.
<i>RTPPacket</i>	The memory of whole RTP packet.
<i>packetSize</i>	The size of received RTP Packet.

Note:

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.

Audio and video stream callback events

Functions

- (void) - [<PortSIPEventDelegate >::onAudioRawCallback:audioCallbackMode:data:dataLength:samplingFreqHz:](#)
- (int) - [<PortSIPEventDelegate >::onVideoRawCallback:videoCallbackMode:width:height:data:dataLength:](#)

Detailed Description

Function Documentation

- (void PortSIPEventDelegate) onAudioRawCallback: (long) *sessionId*
audioCallbackMode: (int) *audioCallbackMode* data: (unsigned char *) *data*
dataLength: (int) *dataLength* samplingFreqHz: (int) *samplingFreqHz*

This event will be triggered once receiving the audio packets when enableAudioStreamCallback function is called.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>audioCallbackMode</i>	The type that is passed in enableAudioStreamCallback function.
<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, it could be 8000 or 16000.

Note:

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.

- (int PortSIPEventDelegate) onVideoRawCallback: (long) *sessionId*
videoCallbackMode: (int) *videoCallbackMode* width: (int) *width* height: (int) *height*
data: (unsigned char *) *data* dataLength: (int) *dataLength*

This event will be triggered once received the video packets if called enableVideoStreamCallback function.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>videoCallbackMode</i>	The type passed in enableVideoStreamCallback function.
<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, such as YV12.
<i>dataLength</i>	The data size.

Returns:

If you changed the sent video data, dataLength should be returned, otherwise 0.

Note:

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.

SDK functions

Modules

- [Initialize and register functions](#)
- [NIC and local IP functions](#)
- [Audio and video codecs functions](#)
- [Additional settings functions](#)
- [Access SIP message header functions](#)
- [Audio and video functions](#)
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- [RTP packets, audio stream and video stream callback functions](#)
- [Record functions](#)
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- [Conference functions](#)
- [RTP and RTCP QOS functions](#)
- [Media statistics functions](#)
- [Audio effect functions](#)
- [Send OPTIONS/INFO/MESSAGE functions](#)
- [Presence functions](#)
- [Keep awake functions](#)
- [Audio Controller](#)

Detailed Description

SDK functions

Initialize and register functions

Functions

- (int) - [PortSIPSDK::initialize:localIP:localSIPPort:loglevel:logPath:maxLine:agent:audioDeviceLayer:videoDeviceLayer:TLSCertificatesRootPath:TLSCipherList:verifyTLSCertificate:](#)
Initialize the SDK.
- (int) - [PortSIPSDK::setInstanceId:](#)
Set the instance Id, the outbound instanceId((RFC5626)) used in contact headers.
- (void) - [PortSIPSDK::unInitialize](#)
Un-initialize the SDK and release resources.
- (int) - [PortSIPSDK::setUser:displayName:authName:password:userDomain:SIPServer:SIPServerPort:STUNServer:STUNServerPort:outboundServer:outboundServerPort:](#)
Set user account info.
- (void) - [PortSIPSDK::removeUser](#)
Remove user account info.
- (int) - [PortSIPSDK::registerServer:retryTimes:](#)
Register to SIP proxy server (login to server)
- (int) - [PortSIPSDK::refreshRegistration:](#)
Refresh the registration manually after successfully registered.
- (int) - [PortSIPSDK::unRegisterServer](#)
Un-register from the SIP proxy server.
- (int) - [PortSIPSDK::setLicenseKey:](#)
Set the license key. It must be called before setUser function.

Detailed Description

Initialize and register functions

Function Documentation

- (int) initialize: (TRANSPORT_TYPE) *transport* localIP: (NSString *) *localIP*
localSIPPort: (int) *localSIPPort* loglevel: (PORTSIP_LOG_LEVEL) *logLevel* logPath:

**(NSString *) logFilePath maxLine: (int) maxCallLines agent: (NSString *) sipAgent
 audioDeviceLayer: (int) audioDeviceLayer videoDeviceLayer: (int) videoDeviceLayer
 TLSCertificatesRootPath: (NSString *) TLSCertificatesRootPath TLSCipherList:
 (NSString *) TLSCipherList verifyTLSCertificate: (BOOL) verifyTLSCertificate**

Initialize the SDK.

Parameters:

<i>transport</i>	Transport for SIP signaling. TRANSPORT_PERS_UDP/TRANSPORT_PERS_TCP is the PortSIP private transport for anti SIP blocking. It must be used with PERS Server.
<i>localIP</i>	The local computer IP address to be bounded (for example: 192.168.1.108). It will be used for sending and receiving SIP messages and RTP packets. If this IP is transferred in IPv6 format, the SDK will use IPv6. If you want the SDK to choose correct network interface (IP) automatically, please pass the "0.0.0.0"(for IPv4) or "::" (for IPv6).
<i>localSIPPort</i>	The SIP message transport listener port (for example: 5060).
<i>logLevel</i>	Set the application log level. The SDK will generate "PortSIP_Log_datetime.log" file if the log enabled.
<i>logFilePath</i>	The log file path. The path (folder) MUST be existent.
<i>maxCallLines</i>	Theoretically, unlimited lines are supported depending on the device capability. For SIP client recommended value ranges 1 - 100.
<i>sipAgent</i>	The User-Agent header to be inserted in SIP messages.
<i>audioDeviceLayer</i>	0 = Use OS default device 1 = Set to 1 to use the virtual audio device if the no sound device installed.
<i>videoDeviceLayer</i>	0 = Use OS default device 1 = Set to 1 to use the virtual video device if 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. By setting to false, the SDK will not verify the validity of TLS certificate.

Returns:

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

- (int) setInstanceId: (NSString *) 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) setUser: (NSString *) userName displayName: (NSString *) displayName
 authName: (NSString *) authName password: (NSString *) password userDomain:
 (NSString *) userDomain SIPServer: (NSString *) sipServer SIPServerPort: (int)**

***sipServerPort* STUNServer: (NSString *) *stunServer* STUNServerPort: (int)
stunServerPort outboundServer: (NSString *) *outboundServer* outboundServerPort:
(int) *outboundServerPort***

Set user account info.

Parameters:

<i>userName</i>	Account (username) of the SIP. It's usually provided by an IP-Telephony service provider.
<i>displayName</i>	The display name of user. 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>	The password of user. It's optional.
<i>userDomain</i>	User domain. This parameter is optional. It allows to pass an empty string if you are not using 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, used for NAT traversal. It's optional and can pass an empty string to disable STUN.
<i>stunServerPort</i>	STUN server port. It will be ignored if the <i>outboundServer</i> is empty.
<i>outboundServer</i>	Outbound proxy server. For example: sip.domain.com. It's optional and allows to pass an empty string if not using outbound server.
<i>outboundServerPort</i>	Outbound proxy server port. It will be ignored if the <i>outboundServer</i> is empty.

Returns:

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

- (int) registerServer: (int) expires retryTimes: (int) retryTimes

Register to SIP proxy server (login to server)

Parameters:

<i>expires</i>	Registration refreshment interval in seconds. Maximum of 3600 allowed. It will be inserted into SIP REGISTER message headers.
<i>retryTimes</i>	The retry times if failed to refresh the registration. Once set to <= 0, the retry will be disabled and onRegisterFailure callback triggered for retry failure.

Returns:

If the function succeeds, it will return value 0. If the function fails, it will return a specific error code. If registration to server succeeds, onRegisterSuccess will be triggered, otherwise onRegisterFailure triggered.

- (int) refreshRegistration: (int) expires

Refresh the registration manually after successfully registered.

Parameters:

<i>expires</i>	Registration refreshment interval in seconds. Maximum of 3600 supported. It will be inserted into SIP REGISTER message header. If it's set to 0, default value will be used.
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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. If registration to server succeeds, onRegisterSuccess will be triggered, otherwise onRegisterFailure triggered.

- (int) unRegisterServer

Un-register from the SIP proxy server.

Returns:

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

- (int) setLicenseKey: (NSString *) 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.

NIC and local IP functions

Functions

- (int) - [PortSIPSDK::getNICNums](#)
Get the Network Interface Card numbers.
- (NSString *) - [PortSIPSDK::getLocalIpAddress:](#)
Get the local IP address by Network Interface Card index.

Detailed Description

Function Documentation

- (int) getNICNums

Get the Network Interface Card numbers.

Returns:

If the function succeeds, it will return NIC numbers, which are greater than or equal to 0. If the function fails, it will return a specific error code.

- (NSString*) getLocalIpAddress: (int) *index*

Get the local IP address by Network Interface Card index.

Parameters:

<i>index</i>	The IP address index. For example, suppose the PC has two NICs. If we want to obtain the second NIC IP, please set this parameter as 1, and the first NIC IP index 0.
--------------	---

Returns:

The buffer for receiving the IP.

Audio and video codecs functions

Functions

- (int) - [PortSIPSDK::addAudioCodec:](#)
Enable an audio codec. It will appear in SDP.
- (int) - [PortSIPSDK::addVideoCodec:](#)
Enable a video codec. It will appear in SDP.
- (BOOL) - [PortSIPSDK::isAudioCodecEmpty](#)
Detect if the enabled audio codecs is empty.
- (BOOL) - [PortSIPSDK::isVideoCodecEmpty](#)
Detect if enabled video codecs is empty or not.
- (int) - [PortSIPSDK::setAudioCodecPayloadType:payloadType:](#)
Set the RTP payload type for dynamic audio codec.
- (int) - [PortSIPSDK::setVideoCodecPayloadType:payloadType:](#)
Set the RTP payload type for dynamic Video codec.
- (void) - [PortSIPSDK::clearAudioCodec](#)
Remove all enabled audio codecs.
- (void) - [PortSIPSDK::clearVideoCodec](#)
Remove all enabled video codecs.
- (int) - [PortSIPSDK::setAudioCodecParameter:parameter:](#)
Set the codec parameter for audio codec.
- (int) - [PortSIPSDK::setVideoCodecParameter:parameter:](#)
Set the codec parameter for video codec.

Detailed Description

Function Documentation

- (int) addAudioCodec: (AUDIOCODEC_TYPE) *codecType*

Enable an audio codec. It will appear in SDP.

Parameters:

<i>codecType</i>	Audio codec type.
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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) addVideoCodec: (VIDEOCODEC_TYPE) *codecType*

Enable a video codec. It will appear in SDP.

Parameters:

<i>codecType</i>	Video codec type.
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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.

- (BOOL) isAudioCodecEmpty

Detect if the enabled audio codecs is empty.

Returns:

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

- (BOOL) isVideoCodecEmpty

Detect if enabled video codecs is empty or not.

Returns:

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

**- (int) setAudioCodecPayloadType: (AUDIOCODEC_TYPE) *codecType* payloadType:
(int) *payloadType***

Set the RTP payload type for dynamic audio codec.

Parameters:

<i>codecType</i>	Audio codec type defined in the PortSIPTypes file.
<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) setVideoCodecPayloadType: (VIDEOCODEC_TYPE) *codecType* payloadType:
(int) *payloadType***

Set the RTP payload type for dynamic Video codec.

Parameters:

<i>codecType</i>	Video codec type defined in the PortSIPTypes file.
<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) **setAudioCodecParameter: (AUDIOCODEC_TYPE) codecType parameter: (NSString *) parameter**

Set the codec parameter for audio codec.

Parameters:

<i>codecType</i>	Audio codec type defined in the PortSIPTypes file.
<i>parameter</i>	The parameter 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:

```
[myVoIPsdk setAudioCodecParameter:AUDIOCODEC AMR parameter:"mode-set=0;
octet-align=1; robust-sorting=0"];
```

- (int) **setVideoCodecParameter: (VIDEOCODEC_TYPE) codecType parameter: (NSString *) parameter**

Set the codec parameter for video codec.

Parameters:

<i>codecType</i>	Video codec type defined in the PortSIPTypes file.
<i>parameter</i>	The parameter in string format.

Returns:

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

Remarks:

Example:

```
[myVoIPsdk setVideoCodecParameter:VIDEOCODEC H264
parameter:"profile-level-id=420033; packetization-mode=0"];
```

Additional settings functions

Functions

- (NSString *) - [PortSIPSDK::getVersion](#)
Get the current version number of the SDK.
- (int) - [PortSIPSDK::enableRport:](#)
Enable/disable rport(RFC3581).
- (int) - [PortSIPSDK::enableEarlyMedia:](#)

- Enable/disable Early Media.*

 - (int) - [PortSIPSDK::enableReliableProvisional:](#)
Enable/disable PRACK.
 - (int) - [PortSIPSDK::enable3GppTags:](#)
Enable/disable the 3Gpp tags, including "ims.icsi.mmtel" and "g.3gpp.smsip".
 - (void) - [PortSIPSDK::enableCallbackSignaling:enableReceived:](#)
Enable/disable to callback the SIP messages.
 - (int) - [PortSIPSDK::setSrtpPolicy:](#)
Set the SRTP policy.
 - (int) - [PortSIPSDK::setRtpPortRange:maximumRtpAudioPort:minimumRtpVideoPort:maximumRtpVideoPort:](#)
Set the RTP ports range for audio and video streaming.
 - (int) - [PortSIPSDK::setRtcpPortRange:maximumRtcpAudioPort:minimumRtcpVideoPort:maximumRtcpVideoPort:](#)
Set the RTCP ports range for audio and video streaming.
 - (int) - [PortSIPSDK::enableCallForward:forwardTo:](#)
Enable call forwarding.
 - (int) - [PortSIPSDK::disableCallForward](#)
Disable the call forwarding. The SDK is not forwarding any incoming calls once this function is called.
 - (int) - [PortSIPSDK::enableSessionTimer:refreshMode:](#)
Allows to periodically refresh Session Initiation Protocol (SIP) sessions by sending INVITE requests repeatedly.
 - (int) - [PortSIPSDK::disableSessionTimer](#)
Disable the session timer.
 - (void) - [PortSIPSDK::setDoNotDisturb:](#)
Enable the "Do not disturb" to enable/disable.
 - (void) - [PortSIPSDK::enableAutoCheckMwi:](#)
Enable the CheckMwi to enable/disable.
 - (int) - [PortSIPSDK::setRtpKeepAlive:keepAlivePayloadType:deltaTransmitTimeMS:](#)
Enable or disable to send RTP keep-alive packet when the call is established.
 - (int) - [PortSIPSDK::setKeepAliveTime:](#)
Enable or disable to send SIP keep-alive packet.
 - (int) - [PortSIPSDK::setAudioSamples:maxPtime:](#)
Set the audio capturing sample.
 - (int) - [PortSIPSDK::addSupportedMimeType:mimeType:subMimeType:](#)
Set the SDK to receive the SIP message that includes special mime type.

Detailed Description

Function Documentation

- (NSString*) getVersion

Get the current version number of the SDK.

Returns:

Return a current version number MAJOR.MINOR.PATCH of the SDK.

- (int) enableRport: (BOOL) *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.

- (int) enableEarlyMedia: (BOOL) *enable*

Enable/disable Early Media.

Parameters:

<i>enable</i>	Set to true to enable the SDK to 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) enableReliableProvisional: (BOOL) *enable*

Enable/disable PRACK.

Parameters:

<i>enable</i>	Set to true to enable the SDK to support PRACK. By 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) enable3GppTags: (BOOL) *enable*

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

Parameters:

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

Returns:

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

- (void) **enableCallbackSignaling: (BOOL) enableSending enableReceived: (BOOL) enableReceived**

Enable/disable to callback 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.

- (int) **setSrtpPolicy: (SRTP_POLICY) srtpPolicy**

Set the SRTP policy.

Parameters:

<i>srtpPolicy</i>	The SRTP policy.
-------------------	------------------

Returns:

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

- (int) **setRtpPortRange: (int) minimumRtpAudioPort maximumRtpAudioPort: (int) maximumRtpAudioPort minimumRtpVideoPort: (int) minimumRtpVideoPort maximumRtpVideoPort: (int) maximumRtpVideoPort**

Set the RTP ports range for audio and video streaming.

Parameters:

<i>minimumRtpAudioPort</i>	The minimum RTP port for audio stream.
<i>maximumRtpAudioPort</i>	The maximum RTP port for audio stream.
<i>minimumRtpVideoPort</i>	The minimum RTP port for video stream.
<i>maximumRtpVideoPort</i>	The maximum RTP port for video stream.

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) **setRtcpPortRange: (int) minimumRtcpAudioPort maximumRtcpAudioPort: (int) maximumRtcpAudioPort minimumRtcpVideoPort: (int) minimumRtcpVideoPort maximumRtcpVideoPort: (int) maximumRtcpVideoPort**

Set the RTCP ports range for audio and video streaming.

Parameters:

<i>minimumRtcpAudioPort</i>	The minimum RTCP port for audio stream.
<i>maximumRtcpAudioPort</i>	The maximum RTCP port for audio stream.
<i>minimumRtcpVideoPort</i>	The minimum RTCP port for video stream.
<i>maximumRtcpVideoPort</i>	The maximum RTCP port for video stream.

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) **enableCallForward: (BOOL) forBusyOnly forwardTo: (NSString *) forwardTo**

Enable call forwarding.

Parameters:

<i>forBusyOnly</i>	If this parameter is set as true, the SDK will forward all incoming calls when currently it's busy. If it's set as false, the SDK forward all incoming calls anyway.
<i>forwardTo</i>	The target of call forwarding. It must in the format of sip: xxx@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) **disableCallForward**

Disable the call forwarding. The SDK is not forwarding any incoming calls once this function is called.

Returns:

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

- (int) **enableSessionTimer: (int) timerSeconds refreshMode: (SESSION_REFRESH_MODE) refreshMode**

Allows to periodically refresh Session Initiation Protocol (SIP) sessions by sending INVITE requests repeatedly.

Parameters:

<i>timerSeconds</i>	The value of the refreshment interval in seconds. Minimum of 90 seconds required.
<i>refreshMode</i>	Allow to set the session refreshment by UAC or UAS: SESSION_REFRESH_UAC or SESSION_REFRESH_UAS;

Returns:

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

Remarks:

The INVITE requests, or re-INVITEs, are sent repeatedly 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 for remembering incoming and outgoing requests (stateful proxies) may continue to retain call state needlessly. If a UA fails to send a BYE message at the end of a session or if the BYE message is lost because of network problems, a stateful proxy does not know that the session has ended. The re-INVITEs ensure that active sessions stay active and completed sessions are terminated.

- (int) disableSessionTimer

Disable the session timer.

Returns:

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

- (void) setDoNotDisturb: (BOOL) state

Enable the "Do not disturb" to enable/disable.

Parameters:

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

- (void) enableAutoCheckMwi: (BOOL) state

Enable the CheckMwi to enable/disable.

Parameters:

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

- (int) setRtpKeepAlive: (BOOL) state keepAlivePayloadType: (int) keepAlivePayloadType deltaTransmitTimeMS: (int) deltaTransmitTimeMS

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

Parameters:

<i>state</i>	Set as true to allow 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 keep-alive RTP packet sending interval, in milliseconds. 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) setKeepAliveTime: (int) *keepAliveTime*

Enable or disable to send SIP keep-alive packet.

Parameters:

<i>keepAliveTime</i>	This is the SIP keep-alive time interval in seconds. By setting 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) setAudioSamples: (int) *ptime* maxPtime: (int) *maxPtime*

Set the audio capturing sample.

Parameters:

<i>ptime</i>	It should be a multiple of 10 between 10 - 60 (with 10 and 60 inclusive).
<i>maxPtime</i>	For the "maxptime" attribute, it should be a multiple of 10 between 10 - 60 (with 10 and 60 inclusive). It cannot 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.

Remarks:

It will appear in the SDP of INVITE and 200 OK message as "ptime and "maxptime" attribute.

- (int) addSupportedMimeType: (NSString *) *methodName* mimeType: (NSString *) *mimeType* subMimeType: (NSString *) *subMimeType*

Set the SDK to receive the SIP message that includes 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 the RFC3261.
<i>mimeType</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:

By default, PortSIP VoIP SDK supports the media types (mime types) listed 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.  
"application/media_control+xml" in INFO message.
```

The SDK allows to receive SIP messages that include above mime types. Now if remote side sends an INFO SIP message with its "Content-Type" header value "text/plain", SDK will reject this INFO message, for "text/plain" of INFO message is not included in the default

supported list. How should we enable the SDK to receive the SIP INFO messages that include "text/plain" mime type? The answer is to use `addSupportedMimyType`:

```
[myVoIPSdk addSupportedMimeType:@"INFO" mimeType:@"text" subMimeType:@"plain"];
```

To receive the NOTIFY message with "application/media_control+xml":

```
[myVoIPSdk addSupportedMimeType:@"NOTIFY" mimeType:@"application"
subMimeType:@"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

- (NSString *) - [PortSIPSDK::getSipMessageHeaderValue:headerName:](#)
Access the SIP header of SIP message.
- (long) - [PortSIPSDK::addSipMessageHeader:methodName:msgType:headerName:headerValue:](#)
Add the SIP Message header into the specified outgoing SIP message.
- (int) - [PortSIPSDK::removeAddedSipMessageHeader:](#)
Remove the headers (custom header) added by `addSipMessageHeader`.
- (void) - [PortSIPSDK::clearAddedSipMessageHeaders](#)
Clear the added extension headers (custom headers)
- (long) - [PortSIPSDK::modifySipMessageHeader:methodName:msgType:headerName:headerValue:](#)
Modify the special SIP header value for every outgoing SIP message.
- (int) - [PortSIPSDK::removeModifiedSipMessageHeader:](#)
Remove the extension header (custom header) from every outgoing SIP message.
- (void) - [PortSIPSDK::clearModifiedSipMessageHeaders](#)
Clear the modified headers value, and do not modify every outgoing SIP message header values any longer.

Detailed Description

Function Documentation

- (NSString*) **getSipMessageHeaderValue:** (NSString *) *sipMessage* headerName:
(NSString *) *headerName*

Access the SIP header of SIP message.

Parameters:

<i>sipMessage</i>	The SIP message.
<i>headerName</i>	The header with which to access the SIP message.

Returns:

If the function succeeds, it will return value `headerValue`. If the function fails, it will return value `null`.

Remarks:

When receiving an SIP message in the onReceivedSignaling callback event, and user wishes to get SIP message header value, use getExtensionHeaderValue:

```
NSString* headerValue = [myVoIPSdk getSipMessageHeaderValue:message
headerName:name];
```

**- (long) addSipMessageHeader: (long) sessionId methodName: (NSString *)
methodName msgType: (int) msgType headerName: (NSString *) headerName
headerValue: (NSString *) 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>	Just add the header to the SIP message with specified method name. 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 that will appear in SIP message.
<i>headerValue</i>	The custom header value.

Returns:

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

- (int) 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) clearAddedSipMessageHeaders

Clear the added extension headers (custom headers)

Remarks:

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

```
[myVoIPSdk addedSipMessageId:-1 methodName:@"ALL" msgType:3
headerName:@"Billing" headerValue:@"usd100.00"];
[myVoIPSdk addedSipMessageId:-1 methodName:@"ALL" msgType:3
headerName:@"ServiceId" headerValue:@"8873456"];
[myVoIPSdk clearAddedSipMessageHeaders];
```


- (long) **modifySipMessageHeader:** (long) *sessionId* **methodName:** (NSString *) *methodName* **msgType:** (int) *msgType* **headerName:** (NSString *) *headerName* **headerValue:** (NSString *) *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.

- (int) **removeModifiedSipMessageHeader:** (long) *modifiedSipMessageId*

Remove the extension header (custom header) from every outgoing SIP message.

Parameters:

<i>modifiedSipMessageId</i>	The <code>modifiedSipMessageId</code> is returned 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) **clearModifiedSipMessageHeaders**

Clear the modified headers value, and do not modify every outgoing SIP message header values any longer.

Remarks:

For example, to modify two headers' value for every outgoing SIP message and wish to clear it:

```
[myVoIPSdk removeModifiedSipMessageHeader:-1 methodName:@"ALL" msgType:3
headerName:@"Expires" headerValue:@"1000"];
[myVoIPSdk removeModifiedSipMessageHeader:-1 methodName:@"ALL" msgType:3
headerName:@"User-Agent" headerValue:@"MyTest Softphone 1.0"];
[myVoIPSdk clearModifiedSipMessageHeaders];
```

Audio and video functions

Functions

- (int) - [PortSIPSDK::setVideoDeviceId:](#)
Set the video device that will be used for video call.

- (int) - [PortSIPSDK::setVideoResolution:height:](#)
Set the video capturing resolution.
- (int) - [PortSIPSDK::setVideoCropAndScale:](#)
When the camera does not support specified resolution, enable or disable SDK to crop and scale to the specified resolution.
- (int) - [PortSIPSDK::setAudioBitrate:codecType:bitrateKbps:](#)
Set the audio bit rate.
- (int) - [PortSIPSDK::setVideoBitrate:bitrateKbps:](#)
Set the video bitrate.
- (int) - [PortSIPSDK::setVideoFrameRate:frameRate:](#)
Set the video frame rate.
- (int) - [PortSIPSDK::sendVideo:sendState:](#)
Send the video to remote side.
- (void) - [PortSIPSDK::setLocalVideoWindow:](#)
Set the window on which the local video image will be displayed.
- (int) - [PortSIPSDK::setRemoteVideoWindow:remoteVideoWindow:](#)
Set the window for a session to display the received remote video image.
- (int) - [PortSIPSDK::displayLocalVideo:](#)
Start/stop displaying the local video image.
- (int) - [PortSIPSDK::setVideoNackStatus:](#)
Enable/disable the NACK feature (RFC4585) to help to improve the video quality.
- (void) - [PortSIPSDK::muteMicrophone:](#)
Mute the device microphone. It's unavailable for Android and iOS.
- (void) - [PortSIPSDK::muteSpeaker:](#)
Mute the device speaker. It's unavailable for Android and iOS.
- (int) - [PortSIPSDK::setLoudspeakerStatus:](#)
Set the audio device that will be used for audio call.
- (int) - [PortSIPSDK::setChannelOutputVolumeScaling:scaling:](#)
- (int) - [PortSIPSDK::setChannelInputVolumeScaling:scaling:](#)

Detailed Description

Function Documentation

- (int) **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) **setVideoResolution:** (int) *width* height: (int) *height*

Set the video capturing resolution.

Parameters:

<i>width</i>	Video width.
<i>height</i>	Video height.

Returns:

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

- (int) setVideoCropAndScale: (BOOL) *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 Video 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) setAudioBitrate: (long) *sessionId* codecType: (AUDIOCODEC_TYPE) *codecType* bitrateKbps: (int) *bitrateKbps*

Set the audio bit rate.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>codecType</i>	Audio codec type.
<i>bitrateKbps</i>	The Audio bit rate in KBPS.

Returns:

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

- (int) setVideoBitrate: (long) *sessionId* bitrateKbps: (int) *bitrateKbps*

Set the video bitrate.

Parameters:

<i>sessionId</i>	The session ID of the call. Set it to -1 for all calls.
<i>bitrateKbps</i>	The video bit rate in KBPS.

Returns:

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

- (int) setVideoFrameRate: (long) *sessionId* frameRate: (int) *frameRate*

Set the video frame rate.

Parameters:

<i>sessionId</i>	The session ID of the call. Set it to -1 for all calls.
<i>frameRate</i>	The frame rate value, with its minimum value 5, and maximum value 30. Greater value renders better video quality but requires more bandwidth.

Returns:

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

Remarks:

Usually you do not need to call this function to set the frame rate, as the SDK uses default frame rate.

- (int) sendVideo: (long) sessionId sendState: (BOOL) sendState

Send the video to remote side.

Parameters:

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

- (void) setLocalVideoWindow: ([PortSIPVideoRenderView](#) *) localVideoWindow

Set the window on which the local video image will be displayed.

Parameters:

<i>localVideoWindow</i>	The PortSIPVideoRenderView for displaying local video image from camera.
-------------------------	--

- (int) setRemoteVideoWindow: (long) sessionId remoteVideoWindow: ([PortSIPVideoRenderView](#) *) remoteVideoWindow

Set the window for a session to display the received remote video image.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>remoteVideoWindow</i>	The PortSIPVideoRenderView for displaying 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) displayLocalVideo: (BOOL) state

Start/stop displaying the local video image.

Parameters:

<i>state</i>	Set to true to display local video image.
--------------	---

Returns:

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

- (int) setVideoNackStatus: (BOOL) state

Enable/disable the NACK feature (RFC4585) to help 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.

- (void) muteMicrophone: (BOOL) mute

Mute the device microphone. It's unavailable for Android and iOS.

Parameters:

<i>mute</i>	If the value is set to true, the microphone is muted, or set to false to be un-muted.
-------------	---

- (void) muteSpeaker: (BOOL) mute

Mute the device speaker. It's unavailable for Android and iOS.

Parameters:

<i>mute</i>	If the value is set to true, the speaker is muted, or set to false to be un-muted.
-------------	--

- (int) setLoudspeakerStatus: (BOOL) enable

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

Parameters:

<i>enable</i>	By setting to true the SDK uses loudspeaker for audio call. This is 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.

Remarks:

Allow to switch between earphone and loudspeaker only.

- (int) setChannelOutputVolumeScaling: (long) sessionId scaling: (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) setChannelInputVolumeScaling: (long) sessionId scaling: (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.

Call functions

Functions

- (long) - [PortSIPSDK::call:sendSdp:videoCall:](#)
Make a call.
- (int) - [PortSIPSDK::rejectCall:code:](#)
rejectCall Reject the incoming call.
- (int) - [PortSIPSDK::hangUp:](#)
hangUp Hang up the call.
- (int) - [PortSIPSDK::answerCall:videoCall:](#)
answerCall Answer the incoming call.
- (int) - [PortSIPSDK::updateCall:enableAudio:enableVideo:](#)
Use the re-INVITE to update the established call.
- (int) - [PortSIPSDK::hold:](#)
Place a call on hold.
- (int) - [PortSIPSDK::unHold:](#)
Take off hold.
- (int) - [PortSIPSDK::muteSession:muteIncomingAudio:muteOutgoingAudio:muteIncomingVideo:muteOutgoingVideo:](#)
Mute the specified session audio or video.
- (int) - [PortSIPSDK::forwardCall:forwardTo:](#)
Forward the call to another user once received an incoming call.
- (long) - [PortSIPSDK::pickupBLFCall:videoCall:](#)
This function will be used for picking up a call based on the BLF (Busy Lamp Field) status.
- (int) - [PortSIPSDK::sendDtmf:dtmfMethod:code:dtmfDuration:playDtmfTone:](#)
Send DTMF tone.

Detailed Description

Function Documentation

- (long) call: (NSString *) *callee* sendSdp: (BOOL) *sendSdp* videoCall: (BOOL) *videoCall*

Make a call.

Parameters:

<i>callee</i>	The callee. It can be a name only or full SIP URI. For example, user001, sip: user001@sip.iptel.org or sip: user002@sip.yourdomain.com :5068.
<i>sendSdp</i>	If set to false, the outgoing call will not include the SDP in INVITE message.
<i>videoCall</i>	If set to true and at least one video codec was added, the outgoing call will include the video codec into 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 being processed, and you need to detect the final state of calling in onInviteTrying, onInviteRinging, onInviteFailure callback events.

- (int) rejectCall: (long) *sessionId* code: (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 and 480.

Returns:

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

- (int) 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) answerCall: (long) *sessionId* videoCall: (BOOL) *videoCall*

answerCall Answer the incoming call.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>videoCall</i>	If the incoming call is a video call and the video codec is matched, set it to true to answer the video call. If set to false, the answer call will 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) updateCall: (long) *sessionId* enableAudio: (BOOL) *enableAudio* enableVideo: (BOOL) *enableVideo*

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 updated 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 a specific error code.

Remarks:

Example usage:

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

```
[myVoIPSdk clearVideoCodec];
[myVoIPSdk addVideoCodec:VIDEOCODEC_H264];
[myVoIPSdk updateCall:sessionId enableAudio:true enableVideo:true];
```

Example 2: Remove video stream from current conversation.

```
[myVoIPSdk updateCall:sessionId enableAudio:true enableVideo:false];
```

- (int) hold: (long) *sessionId*

Place a call on hold.

Parameters:

<i>sessionId</i>	The 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) 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) muteSession: (long) *sessionId* muteIncomingAudio: (BOOL) *muteIncomingAudio* muteOutgoingAudio: (BOOL) *muteOutgoingAudio* muteIncomingVideo: (BOOL) *muteIncomingVideo* muteOutgoingVideo: (BOOL) *muteOutgoingVideo*

Mute the specified session audio or video.

Parameters:

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

Returns:

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

- (int) forwardCall: (long) *sessionId* forwardTo: (NSString *) *forwardTo*

Forward the call to another user once received an incoming call.

Parameters:

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

Returns:

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

- (long) pickupBLFCall: (const char *) *replaceDialogId* videoCall: (BOOL) *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 to be picked up. It comes with onDialogStateUpdated callback.
<i>videoCall</i>	Indicates if it is video call or audio call to be picked up.

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");

- When 100 hold 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) sendDtmf: (long) *sessionId* dtmfMethod: (DTMF_METHOD) *dtmfMethod* code: (int) *code* dtmfDuration: (int) *dtmfDuration* playDtmfTone: (BOOL) *playDtmfTone*

Send DTMF tone.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>dtmfMethod</i>	Support sending DTMF tone with two methods: DTMF_RFC2833 and DTMF_INFO. The DTMF_RFC2833 is recommended.
<i>code</i>	The DTMF tone (0-16).

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>	By setting to true, the SDK plays local DTMF tone sound when sending 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) - [PortSIPSDK::refer:referTo:](#)
Refer the current call to another one.
- (int) - [PortSIPSDK::attendedRefer:replaceSessionId:referTo:](#)
Make an attended refer.

- (int) - [PortSIPSDK::attendedRefer2:replaceSessionId:replaceMethod:target:referTo:](#)
Make an attended refer with specified request line and specified method embedded into the "Refer-To" header.
- (int) - [PortSIPSDK::outOfDialogRefer:replaceMethod:target:referTo:](#)
Send an out of dialog REFER to replace the specified call.
- (long) - [PortSIPSDK::acceptRefer:referSignaling:](#)
Once the REFER request accepted, a new call will be made if called this function. The function is usually called after onReceivedRefer callback event.
- (int) - [PortSIPSDK::rejectRefer:](#)
Reject the REFER request.

Detailed Description

Function Documentation

- (int) refer: (long) *sessionId* referTo: (NSString *) *referTo*

Refer the current call to another one.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>referTo</i>	Target of the refer. It could 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:

```
[myVoIPSdk refer:sessionId referTo:@"sip:testuser12@sip.portsip.com"];
```

You can watch the video on YouTube at <https://www.youtube.com/watch?v=2w9EGgr3FY>. It will demonstrate the transfer.

- (int) attendedRefer: (long) *sessionId* replaceSessionId: (long) *replaceSessionId* referTo: (NSString *) *referTo*

Make an attended refer.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>replaceSessionId</i>	Session ID of the replaced call.
<i>referTo</i>	Target 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 for more details, or you can watch the video on YouTube at <https://www.youtube.com/watch?v=2w9EGgr3FY>, which will demonstrate the transfer.

- (int) attendedRefer2: (long) *sessionId* replaceSessionId: (long) *replaceSessionId* replaceMethod: (NSString *) *replaceMethod* target: (NSString *) *target* referTo: (NSString *) *referTo*

Make an attended refer with specified request line and specified method embedded into the "Refer-To" header.

Parameters:

<i>sessionId</i>	Session ID of the call.
<i>replaceSessionId</i>	Session ID of the replaced call.
<i>replaceMethod</i>	The SIP method name which will be embedded in the "Refer-To" header, usually INVITE or BYE.
<i>target</i>	The target to which the REFER message will be sent. It appears in the "Request Line" of REFER message.
<i>referTo</i>	Target of the refer that appears in the "Refer-To" header. 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 refer to the sample project source code for more details. Or you can watch the video on YouTube at <https://www.youtube.com/watch?v=2w9EGgr3FY>. It will demonstrate the transmission.

- (int) outOfDialogRefer: (long) *replaceSessionId* replaceMethod: (NSString *) *replaceMethod* target: (NSString *) *target* referTo: (NSString *) *referTo*

Send an out of dialog REFER to replace the specified call.

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 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.

- (long) acceptRefer: (long) *referId* referSignaling: (NSString *) *referSignaling*

Once the REFER request accepted, a new call will be made if called this function. 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 that is greater than 0 to the new call for REFER, otherwise returns a specific error code that is less than 0.

- (int) **rejectRefer:** (long) *referId*

Reject the REFER request.

Parameters:

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

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) - [PortSIPSDK::enableSendPcmStreamToRemote:state:streamSamplesPerSec:](#)
Enable the SDK to send PCM stream data to remote side from another source instead of microphone.
- (int) - [PortSIPSDK::sendPcmStreamToRemote:data:](#)
Send the audio stream in PCM format from another source instead of audio device capturing (microphone).
- (int) - [PortSIPSDK::enableSendVideoStreamToRemote:state:](#)
Enable the SDK to send video stream data to remote side from another source instead of camera.
- (int) - [PortSIPSDK::sendVideoStreamToRemote:data:width:height:](#)
Send the video stream to remote side.

Detailed Description

Function Documentation

- (int) **enableSendPcmStreamToRemote:** (long) *sessionId* **state:** (BOOL) *state*
streamSamplesPerSec: (int) *streamSamplesPerSec*

Enable the SDK to send PCM stream data to remote side from another source instead of microphone.

Parameters:

<i>sessionId</i>	The session ID of call.
<i>state</i>	Set to true to enable the sending 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.

Remarks:

To send the PCM stream data to another side, this function MUST be called first.

- (int) sendPcmStreamToRemote: (long) *sessionId* data: (NSData *) *data*

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 in 16bit, mono.

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:

```
[myVoIPSdk enableSendPcmStreamToRemote:sessionId state:YES
streamSamplesPerSec:16000];
[myVoIPSdk sendPcmStreamToRemote:sessionId data:data];
```

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) enableSendVideoStreamToRemote: (long) *sessionId* state: (BOOL) *state*

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

Parameters:

<i>sessionId</i>	The session ID of call.
<i>state</i>	Set to true to enable the sending 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) sendVideoStreamToRemote: (long) *sessionId* data: (NSData *) *data* width: (int) *width* height: (int) *height*

Send the video stream to remote side.

Parameters:

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

Returns:

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

Remarks:

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. Usually we should use it like below:

```
[myVoIPSdk enableSendVideoStreamToRemote:sessionId state:YES];
[myVoIPSdk sendVideoStreamToRemote:sessionId data:data width:352 height:288];
```

RTP packets, audio stream and video stream callback functions

Functions

- (int) - [PortSIPSDK::setRtpCallback:](#)
Set the RTP callbacks to allow to access the sent and received RTP packets.
- (int) - [PortSIPSDK::enableAudioStreamCallback:enable:callbackMode:](#)
Enable/disable the audio stream callback.
- (int) - [PortSIPSDK::enableVideoStreamCallback:callbackMode:](#)
Enable/disable the video stream callback.

Detailed Description

Function Documentation

- (int) setRtpCallback: (BOOL) enable

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

Parameters:

<i>enable</i>	Set to true to enable the RTP callback for received and sent RTP packets. The onSendingRtpPacket and onReceivedRtpPacket events will be triggered.
---------------	--

Returns:

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

- (int) enableAudioStreamCallback: (long) sessionId enable: (BOOL) enable callbackMode: (AUDIOSTREAM_CALLBACK_MODE) callbackMode

Enable/disable the audio stream callback.

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>callbackMode</i>	The audio stream callback mode.

Type	Description
AUDIOSTREAM_LOCAL_PER_CHANNEL	Callback the audio stream from microphone for one channel based on the given sessionId.
AUDIOSTREAM_REMOTE_PER_CHANNE	Callback the received audio stream for one

L	channel based on the given <code>sessionId</code> .
AUDIOSTREAM_BOTH	Callback both local and remote audio stream on the given <code>sessionId</code> .

Returns:

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

Remarks:

onAudioRawCallback event will be triggered if the callback is enabled.

- (int) **enableVideoStreamCallback:** (long) *sessionId* **callbackMode:** (VIDEOSTREAM_CALLBACK_MODE) *callbackMode*

Enable/disable the video stream callback.

Parameters:

<i>sessionId</i>	The session ID of call.
<i>callbackMode</i>	The video stream callback mode.
Mode	Description
VIDEOSTREAM_NONE	Disable video stream callback.
VIDEOSTREAM_LOCAL	Local video stream callback.
VIDEOSTREAM_REMOTE	Remote video stream callback.
VIDEOSTREAM_BOTH	Both of local and remote video stream callback.

Returns:

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

Remarks:

The onVideoRawCallback event will be triggered if the callback is enabled.

Record functions

Functions

- (int) - [PortSIPSDK::startRecord:recordFilePath:recordFileName:appendTimeStamp:audioFileFormat:audioRecordMode:aviFileCodecType:videoRecordMode:](#)
Start recording the call.
- (int) - [PortSIPSDK::stopRecord:](#)
Stop recording.

Detailed Description

Function Documentation

- (int) startRecord: (long) *sessionId* recordFilePath: (NSString *) *recordFilePath* recordFileName: (NSString *) *recordFileName* appendTimeStamp: (BOOL) *appendTimeStamp* audioFileFormat: (AUDIO_FILE_FORMAT) *audioFileFormat* audioRecordMode: (RECORD_MODE) *audioRecordMode* aviFileCodecType: (VIDEOCODEC_TYPE) *aviFileCodecType* videoRecordMode: (RECORD_MODE) *videoRecordMode*

Start recording the call.

Parameters:

<i>sessionId</i>	The session ID of call conversation.
<i>recordFilePath</i>	The filepath to which the recording will be saved. It must be existent.
<i>recordFileName</i>	The filename of the recording. For example audiorecord.wav or videorecord.avi.
<i>appendTimeStamp</i>	Set to true to append the timestamp to the filename of the recording.
<i>audioFileFormat</i>	The file format for the audio recording.
<i>audioRecordMode</i>	The audio recording mode.
<i>aviFileCodecType</i>	The codec that is used for compressing the video data to save into video recording.
<i>videoRecordMode</i>	Allow to set video recording 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) stopRecord: (long) *sessionId*

Stop recording.

Parameters:

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

Play audio and video files to remote party

Functions

- (int) - [PortSIPSDK::playVideoFileToRemote:aviFile:loop:playAudio:](#)
Play an AVI file to remote party.
- (int) - [PortSIPSDK::stopPlayVideoFileToRemote:](#)
Stop playing video file to remote party.
- (int) - [PortSIPSDK::playAudioFileToRemote:filename:fileSamplesPerSec:loop:](#)
Play a wave file to remote party.
- (int) - [PortSIPSDK::stopPlayAudioFileToRemote:](#)
Stop playing wave file to remote party.
- (int) - [PortSIPSDK::playAudioFileToRemoteAsBackground:filename:fileSamplesPerSec:](#)

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

- (int) - [PortSIPSDK::stopPlayAudioFileToRemoteAsBackground:](#)
Stop playing a wave file to remote party as conversation background sound.
- (void) - [PortSIPSDK::audioPlayLoopbackTest:](#)
Used for the loop back testing against audio device.

Detailed Description

Function Documentation

- (int) **playVideoFileToRemote:** (long) *sessionId* **aviFile:** (NSString *) *aviFile* **loop:** (BOOL) *loop* **playAudio:** (BOOL) *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 "/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 it is set to true, audio and video will be played together, or false with video played only.

Returns:

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

- (int) **stopPlayVideoFileToRemote:** (long) *sessionId*

Stop playing video file to remote party.

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) **playAudioFileToRemote:** (long) *sessionId* **filename:** (NSString *) *filename* **fileSamplesPerSec:** (int) *fileSamplesPerSec* **loop:** (BOOL) *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 "/test.wav".
<i>fileSamplesPerSec</i>	The sample wave file in seconds. It should 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) stopPlayAudioFileToRemote: (long) *sessionId*

Stop playing wave file to remote party.

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) playAudioFileToRemoteAsBackground: (long) *sessionId* filename: (NSString *) *filename* fileSamplesPerSec: (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 "/test.wav".
<i>fileSamplesPerSec</i>	The sample wave file 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) stopPlayAudioFileToRemoteAsBackground: (long) *sessionId*

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

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) audioPlayLoopbackTest: (BOOL) *enable*

Used for the loop back testing against audio device.

Parameters:

<i>enable</i>	Set to true to start audio look back test, or false to stop.
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Conference functions

Functions

- (int) - [PortSIPSDK::createAudioConference](#)
Create an audio conference.
- (int) - [PortSIPSDK::createVideoConference:videoWidth:videoHeight:displayLocalVideo:](#)
Create a video conference.
- (void) - [PortSIPSDK::destroyConference](#)
Destroy the existent conference.
- (int) - [PortSIPSDK::setConferenceVideoWindow:](#)
Set the window for a conference that is used to display the received remote video image.
- (int) - [PortSIPSDK::joinToConference:](#)
Join a session into existent conference. If the call is in hold, please un-hold first.
- (int) - [PortSIPSDK::removeFromConference:](#)
Remove a session from an existent conference.

Detailed Description

Function Documentation

- (int) createAudioConference

Create an audio conference.

Returns:

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

- (int) createVideoConference: ([PortSIPVideoRenderView *](#)) *conferenceVideoWindow* *videoWidth:* (int) *videoWidth* *videoHeight:* (int) *videoHeight* *displayLocalVideo:* (BOOL) *displayLocalVideoInConference*

Create a video conference.

Parameters:

<i>conferenceVideoWindow</i>	The PortSIPVideoRenderView used for displaying the conference video.
<i>videoWidth</i>	The conference video width.
<i>videoHeight</i>	The conference video height.
<i>displayLocalVideoInConference</i>	Display the local video on video window.

Returns:

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

- (int) **setConferenceVideoWindow:** ([PortSIPVideoRenderView](#) *)
conferenceVideoWindow

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

Parameters:

<i>conferenceVideoWindow</i>	The PortSIPVideoRenderView used to display 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) **joinToConference:** (long) *sessionId*

Join a session into existent conference. If the call is in hold, please un-hold first.

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) **removeFromConference:** (long) *sessionId*

Remove a session from an existent 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) - [PortSIPSDK::setAudioRtcpBandwidth:BitsRR:BitsRS:KBitsAS:](#)
Set the audio RTCP bandwidth parameters as the RFC3556.
- (int) - [PortSIPSDK::setVideoRtcpBandwidth:BitsRR:BitsRS:KBitsAS:](#)
Set the video RTCP bandwidth parameters as the RFC3556.
- (int) - [PortSIPSDK::enableAudioQos:](#)
Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for audio channel.
- (int) - [PortSIPSDK::enableVideoQos:](#)
Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for video channel.
- (int) - [PortSIPSDK::setVideoMTU:](#)
Set the MTU size for video RTP packet.

Detailed Description

Function Documentation

- (int) **setAudioRtcpBandwidth:** (long) *sessionId* **BitsRR:** (int) *BitsRR* **BitsRS:** (int) *BitsRS* **KBitsAS:** (int) *KBitsAS*

Set the audio 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) **setVideoRtcpBandwidth:** (long) *sessionId* **BitsRR:** (int) *BitsRR* **BitsRS:** (int) *BitsRS* **KBitsAS:** (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) **enableAudioQos:** (BOOL) *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) enableVideoQos: (BOOL) *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) setVideoMTU: (int) *mtu*

Set the MTU size for video RTP packet.

Parameters:

<i>mtu</i>	Set MTU value. Allowed value ranges 512-65507. Other values will be automatically changed to the default 1400.
------------	--

Returns:

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

Media statistics functions

Functions

- (int) - [PortSIPSDK::getAudioStatistics:sendBytes:sendPackets:sendPacketsLost:sendFractionLost:sendRttMS:sendCodecType:sendJitterMS:sendAudioLevel:recvBytes:recvPackets:recvPacketsLost:recvFractionLost:recvCodecType:recvJitterMS:recvAudioLevel:](#)
Obtain the statistics of audio channel.
- (int) - [PortSIPSDK::getVideoStatistics:sendBytes:sendPackets:sendPacketsLost:sendFractionLost:sendRttMS:sendCodecType:sendFrameWidth:sendFrameHeight:sendBitrateBPS:sendFramerate:recvBytes:recvPackets:recvPacketsLost:recvFractionLost:recvCodecType:recvFrameWidth:recvFrameHeight:recvBitrateBPS:recvFramerate:](#)
Obtain the statistics of video channel.

Detailed Description

Function Documentation

- (int) getAudioStatistics: (long) *sessionId* sendBytes: (int *) *sendBytes*
sendPackets: (int *) *sendPackets* sendPacketsLost: (int *) *sendPacketsLost*
sendFractionLost: (int *) *sendFractionLost* sendRttMS: (int *) *sendRttMS*

**sendCodecType: (int *) sendCodecType sendJitterMS: (int *) sendJitterMS
sendAudioLevel: (int *) sendAudioLevel recvBytes: (int *) recvBytes recvPackets:
(int *) recvPackets recvPacketsLost: (int *) recvPacketsLost recvFractionLost: (int *)
recvFractionLost recvCodecType: (int *) recvCodecType recvJitterMS: (int *)
recvJitterMS recvAudioLevel: (int *) recvAudioLevel**

Obtain the statistics of audio channel.

Parameters:

<i>sessionId</i>	The session ID of call conversation.
<i>sendBytes</i>	The number of sent bytes.
<i>sendPackets</i>	The number of sent packets.
<i>sendPacketsLost</i>	The number of sent but lost packet.
<i>sendFractionLost</i>	Fraction of sent but lost packet in percentage.
<i>sendRttMS</i>	The round-trip time of the session, in milliseconds.
<i>sendCodecType</i>	Send Audio codec Type.
<i>sendJitterMS</i>	The sent jitter, in milliseconds.
<i>sendAudioLevel</i>	The sent audio level.It ranges 0 - 9.
<i>recvBytes</i>	The number of received bytes.
<i>recvPackets</i>	The number of received packets.
<i>recvPacketsLost</i>	The number of received but lost packet.
<i>recvFractionLost</i>	Fraction of received but lost packet in percentage.
<i>recvCodecType</i>	Received Audio codec Type.
<i>recvJitterMS</i>	The received jitter, in milliseconds.
<i>recvAudioLevel</i>	The received audio level.It ranges 0 - 9.

Returns:

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

**- (int) getVideoStatistics: (long) sessionId sendBytes: (int *) sendBytes sendPackets:
(int *) sendPackets sendPacketsLost: (int *) sendPacketsLost sendFractionLost: (int *)
sendFractionLost sendRttMS: (int *) sendRttMS sendCodecType: (int *)
sendCodecType sendFrameWidth: (int *) sendFrameWidth sendFrameHeight: (int *)
sendFrameHeight sendBitrateBPS: (int *) sendBitrateBPS sendFramerate: (int *)
sendFramerate recvBytes: (int *) recvBytes recvPackets: (int *) recvPackets
recvPacketsLost: (int *) recvPacketsLost recvFractionLost: (int *) recvFractionLost
recvCodecType: (int *) recvCodecType recvFrameWidth: (int *) recvFrameWidth
recvFrameHeight: (int *) recvFrameHeight recvBitrateBPS: (int *) recvBitrateBPS
recvFramerate: (int *) recvFramerate**

Obtain the statistics of video channel.

Parameters:

<i>sessionId</i>	The session ID of call conversation.
<i>sendBytes</i>	The number of sent bytes.
<i>sendPackets</i>	The number of sent packets.
<i>sendPacketsLost</i>	The number of sent but lost packet.
<i>sendFractionLost</i>	Fraction of sent lost in percentage.
<i>sendRttMS</i>	The round-trip time of the session, in milliseconds.
<i>sendCodecType</i>	Send Video codec Type.
<i>sendFrameWidth</i>	Frame width for the sent video.
<i>sendFrameHeight</i>	Frame height for the sent video.
<i>sendBitrateBPS</i>	Bitrate in BPS for the sent video.
<i>sendFramerate</i>	Frame rate for the sent video.

<i>recvBytes</i>	The number of received bytes.
<i>recvPackets</i>	The number of received packets.
<i>recvPacketsLost</i>	The number of received but lost packet.
<i>recvFractionLost</i>	Fraction of received but lost packet in percentage.
<i>recvCodecType</i>	Received Video codec Type.
<i>recvFrameWidth</i>	Frame width for the received video.
<i>recvFrameHeight</i>	Frame height for the received video.
<i>recvBitrateBPS</i>	(This parameter is not implemented yet) Bitrate in BPS for the received video.
<i>recvFramerate</i>	Framerate for the received video.

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) - [PortSIPSDK::enableVAD:](#)
Enable/disable Voice Activity Detection (VAD).
- (void) - [PortSIPSDK::enableCNG:](#)
Enable/disable Comfort Noise Generator (CNG).

Detailed Description

Function Documentation

- (void) enableVAD: (BOOL) state

Enable/disable Voice Activity Detection (VAD).

Parameters:

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

- (void) enableCNG: (BOOL) state

Enable/disable Comfort Noise Generator (CNG).

Parameters:

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

Send OPTIONS/INFO/MESSAGE functions

Functions

- (int) - [PortSIPSDK::sendOptions:sdp:](#)
Send *OPTIONS* message.
- (int) - [PortSIPSDK::sendInfo:mimeType:subMimeType:infoContents:](#)
Send an *INFO* message to remote side in dialog.
- (long) - [PortSIPSDK::sendMessage:mimeType:subMimeType:message:messageLength:](#)
Send a *MESSAGE* message to remote side in dialog.
- (long) - [PortSIPSDK::sendOutOfDialogMessage:mimeType:subMimeType:isSMS:message:messageLength:](#)
Send an out of dialog *MESSAGE* message to remote side.

Detailed Description

Function Documentation

- (int) **sendOptions: (NSString *) to sdp: (NSString *) sdp**

Send *OPTIONS* message.

Parameters:

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

Returns:

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

- (int) **sendInfo: (long) sessionId mimeType: (NSString *) mimeType subMimeType: (NSString *) subMimeType infoContents: (NSString *) infoContents**

Send an *INFO* message to remote side in dialog.

Parameters:

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

Returns:

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

- (long) **sendMessage: (long) sessionId mimeType: (NSString *) mimeType subMimeType: (NSString *) subMimeType message: (NSData *) message messageLength: (int) messageLength**

Send a MESSAGE message to remote side in dialog.

Parameters:

<i>sessionId</i>	The session ID of the call.
<i>mimeType</i>	The mime type of MESSAGE message.
<i>subMimeType</i>	The sub mime type of MESSAGE message.
<i>message</i>	The contents to 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 less than 0.

Remarks:

Example 1: Send a plain text message. Note: to send other languages text, please use the UTF-8 to encode the message before sending.

```
[myVoIPsdk sendMessage:sessionId mimeType:@"text" subMimeType:@"plain" message:data messageLength:dataLen];
```

Example 2: Send a binary message.

```
[myVoIPsdk sendMessage:sessionId mimeType:@"application" subMimeType:@"vnd.3gpp.sms" message:data messageLength:dataLen];
```

- (long) **sendOutOfDialogMessage: (NSString *) to mimeType: (NSString *) mimeType subMimeType: (NSString *) subMimeType isSMS: (BOOL) isSMS message: (NSData *) message messageLength: (int) messageLength**

Send an out of dialog MESSAGE message to remote side.

Parameters:

<i>to</i>	The message recipient, such as sip: receiver@portsip.com .
<i>mimeType</i>	The mime type of MESSAGE message.
<i>subMimeType</i>	The sub mime type of MESSAGE message. <code>isSMS</code> Set to YES to specify "messagetype=SMS" in the To line, or NO to disable.
<i>message</i>	The contents 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 less than 0.

Remarks:

Example 1: Send a plain text message. Note: to send text in other languages, please use UTF-8 to encode the message before sending.

```
[myVoIPsdk sendOutOfDialogMessage:@"sip:user1@sip.portsip.com" mimeType:@"text" subMimeType:@"plain" message:data messageLength:dataLen];
```

Example 2: Send a binary message.

```
[myVoIPsdk sendOutOfDialogMessage:@"sip:user1@sip.portsip.com" mimeType:@"application" subMimeType:@"vnd.3gpp.sms" isSMS:NO message:data messageLength:dataLen];
```

Presence functions

Functions

- (int) - [PortSIPSDK::setPresenceMode:](#)
Indicate that SDK uses the P2P mode for presence or presence agent mode.
- (int) - [PortSIPSDK::setDefaultSubscriptionTime:](#)
Set the default expiration time to be used when creating a subscription.
- (int) - [PortSIPSDK::setDefaultPublicationTime:](#)
Set the default expiration time to be used when creating a publication.
- (long) - [PortSIPSDK::presenceSubscribe:subject:](#)
Send a SUBSCRIBE message for subscribing the contact's presence status.
- (int) - [PortSIPSDK::presenceTerminateSubscribe:](#)
Terminate the given presence subscription.
- (int) - [PortSIPSDK::presenceAcceptSubscribe:](#)
Accept the presence SUBSCRIBE request which is received from contact.
- (int) - [PortSIPSDK::presenceRejectSubscribe:](#)
Reject a presence SUBSCRIBE request which is received from contact.
- (int) - [PortSIPSDK::setPresenceStatus:statusText:](#)
Send a NOTIFY message to contact to notify that presence status is online/offline/changed.
- (long) - [PortSIPSDK::sendSubscription:eventName:](#)
Send a SUBSCRIBE message to remote side.
- (int) - [PortSIPSDK::terminateSubscription:](#)
Terminate the given subscription.

Detailed Description

Function Documentation

- (int) setPresenceMode: (int) *mode*

Indicate that 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>

- (int) 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.

- (int) 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) presenceSubscribe: (NSString *) contact subject: (NSString *) subject

Send a SUBSCRIBE message for subscribing the contact's presence status.

Parameters:

<i>contact</i>	The target contact. It must be like 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 in UTF-8 format. You should use UTF-8 to decode it.

Returns:

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

- (int) 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) presenceAcceptSubscribe: (long) subscribeId

Accept the presence SUBSCRIBE request which is received from contact.

Parameters:

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

	subscription ID.
--	------------------

Returns:

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

Remarks:

If the P2P presence mode is enabled, when someone subscribes your presence status, you will receive the subscription request in the callback, and you can use this function to reject it.

- (int) presenceRejectSubscribe: (long) *subscribeId*

Reject a presence SUBSCRIBE request which is 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.

Remarks:

If the P2P presence mode is enabled, when someone subscribe your presence status, you will receive the subscribe request in the callback, and you can use this function to accept it.

- (int) setPresenceStatus: (long) *subscribeId* statusText: (NSString *) *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 value a specific error code.

- (long) sendSubscription: (NSString *) *to* eventName: (NSString *) *eventName*

Send a SUBSCRIBE message to remote side.

Parameters:

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

Returns:

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

Remarks:

Example 1, to subscribe the MWI (Message Waiting notifications), You can use this code:
long 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 is the one to be monitored. Once being monitored, when extension 100 hold a call or is ringing, the `onDialogStateUpdated` callback will be triggered.

- (int) `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);
```

Keep awake functions

Functions

- (BOOL) - [PortSIPSDK::startKeepAwake](#)
Keep VoIP awake in the background. If you want your application to be able to receive the incoming call while it's running in background, you should call this function in `applicationDidEnterBackground`.
- (BOOL) - [PortSIPSDK::stopKeepAwake](#)
Keep VoIP awake in the background. Call this function in `applicationWillEnterForeground` once your application comes back to foreground.

Detailed Description

Function Documentation

- (BOOL) `startKeepAwake`

Keep VoIP awake in the background. If you want your application to be able to receive the incoming call while it's running in background, you should call this function in `applicationDidEnterBackground`.

Returns:

If the function succeeds, it will return value true. If the function fails, it will return value false.

- (BOOL) `stopKeepAwake`

Keep VoIP awake in the background. Call this function in `applicationWillEnterForeground` once your application comes back to foreground.

Returns:

If the function succeeds, it will return value `true`. If the function fails, it will return value `false`.

Audio Controller

Functions

- (BOOL) - [PortSIPSDK::startAudio](#)
Start Audio Device. Call it as `AVAudioSessionInterruptionTypeEnded`.
- (BOOL) - [PortSIPSDK::stopAudio](#)
Stop Audio Device. Call it as `AVAudioSessionInterruptionTypeBegan`.
- (int) - [PortSIPSDK::enableCallKit](#)
Enable/disable CallKit(Native Integration).

Detailed Description

Function Documentation

- (BOOL) `startAudio`

Start Audio Device. Call it as `AVAudioSessionInterruptionTypeEnded`.

Returns:

If the function succeeds, it will return value `true`. If the function fails, it will return value `false`.

- (BOOL) `stopAudio`

Stop Audio Device. Call it as `AVAudioSessionInterruptionTypeBegan`.

Returns:

If the function succeeds, it will return value `true`. If the function fails, it will return value `false`.

- (int) `enableCallKit`: (BOOL) *state*

Enable/disable CallKit(Native Integration).

Parameters:

<i>state</i>	Set to true to enable CallKit, 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.

Class Documentation

<PortSIPEventDelegate > Protocol Reference

PortSIP SDK Callback events Delegate.

```
#import <PortSIPEventDelegate.h>
```

Inherits <NSObject>.

Instance Methods

- (void) - [onRegisterSuccess:statusCode:sipMessage:](#)
- (void) - [onRegisterFailure:statusCode:sipMessage:](#)
- (void) - [onInviteIncoming:callerDisplayName:caller:calleeDisplayName:callee:audioCodecs:videoCodecs:existsAudio:existsVideo:sipMessage:](#)
- (void) - [onInviteTrying:](#)
- (void) - [onInviteSessionProgress:audioCodecs:videoCodecs:existsEarlyMedia:existsAudio:existsVideo:sipMessage:](#)
- (void) - [onInviteRinging:statusText:statusCode:sipMessage:](#)
- (void) - [onInviteAnswered:callerDisplayName:caller:calleeDisplayName:callee:audioCodecs:videoCodecs:existsAudio:existsVideo:sipMessage:](#)
- (void) - [onInviteFailure:reason:code:sipMessage:](#)
- (void) - [onInviteUpdated:audioCodecs:videoCodecs:existsAudio:existsVideo:sipMessage:](#)
- (void) - [onInviteConnected:](#)
- (void) - [onInviteBeginningForward:](#)
- (void) - [onInviteClosed:](#)
- (void) - [onDialogStateUpdated:BLFDialogState:BLFDialogId:BLFDialogDirection:](#)
- (void) - [onRemoteHold:](#)
- (void) - [onRemoteUnHold:audioCodecs:videoCodecs:existsAudio:existsVideo:](#)
- (void) - [onReceivedRefer:referId:to:from:referSipMessage:](#)
- (void) - [onReferAccepted:](#)
- (void) - [onReferRejected:reason:code:](#)
- (void) - [onTransferTrying:](#)
- (void) - [onTransferRinging:](#)
- (void) - [onACTVTransferSuccess:](#)
- (void) - [onACTVTransferFailure:reason:code:](#)
- (void) - [onReceivedSignaling:message:](#)
- (void) - [onSendingSignaling:message:](#)
- (void) - [onWaitingVoiceMessage:urgentNewMessageCount:urgentOldMessageCount:newMessageCount:oldMessageCount:](#)
- (void) - [onWaitingFaxMessage:urgentNewMessageCount:urgentOldMessageCount:newMessageCount:oldMessageCount:](#)
- (void) - [onRecvDtmfTone:tone:](#)
- (void) - [onRecvOptions:](#)
- (void) - [onRecvInfo:](#)
- (void) - [onRecvNotifyOfSubscription:notifyMessage:messageData:messageDataLength:](#)
- (void) - [onPresenceRecvSubscribe:fromDisplayName:from:subject:](#)
- (void) - [onPresenceOnline:from:stateText:](#)
- (void) - [onPresenceOffline:from:](#)
- (void) - [onRecvMessage:mimeType:subMimeType:messageData:messageDataLength:](#)
- (void) - [onRecvOutOfDialogMessage:from:toDisplayName:to:mimeType:subMimeType:messageData:messageDataLength:sipMessage:](#)

- (void) - [onSendMessageSuccess:messageId:](#)
 - (void) - [onSendMessageFailure:messageId:reason:code:](#)
 - (void) - [onSendOutOfDialogMessageSuccess:fromDisplayName:from:toDisplayName:to:](#)
 - (void) - [onSendOutOfDialogMessageFailure:fromDisplayName:from:toDisplayName:to:reason:code:](#)
 - (void) - [onSubscriptionFailure:statusCode:](#)
 - (void) - [onSubscriptionTerminated:](#)
 - (void) - [onPlayAudioFileFinished:fileName:](#)
 - (void) - [onPlayVideoFileFinished:](#)
 - (void) - [onReceivedRTPPacket:isAudio:RTPPacket:packetSize:](#)
 - (void) - [onSendingRTPPacket:isAudio:RTPPacket:packetSize:](#)
 - (void) - [onAudioRawCallback:audioCallbackMode:data:dataLength:samplingFreqHz:](#)
 - (int) - [onVideoRawCallback:videoCallbackMode:width:height:data:dataLength:](#)
-

Detailed Description

PortSIP SDK Callback events Delegate.

Author:

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Version:

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See also:

<http://www.PortSIP.com> PortSIP SDK Callback events Delegate description.

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

- PortSIPEventDelegate.h

PortSIPSDK Class Reference

PortSIP VoIP SDK functions class.

```
#import <PortSIPSDK.h>
```

Inherits NSObject.

Instance Methods

- (int) - [initialize:localIP:localSIPPort:loglevel:logPath:maxLine:agent:audioDeviceLayer:videoDeviceLayer:TLSCertificatesRootPath:TLSCipherList:verifyTLSCertificate:](#)
Initialize the SDK.
- (int) - [setInstanceId:](#)
Set the instance Id, the outbound instanceId((RFC5626)) used in contact headers.
- (void) - [unInitialize](#)
Un-initialize the SDK and release resources.
- (int) - [setUser:displayName:authName:password:userDomain:SIPServer:SIPServerPort:STUNServer:STUNServerPort:outboundServer:outboundServerPort:](#)
Set user account info.
- (void) - [removeUser](#)
Remove user account info.
- (int) - [registerServer:retryTimes:](#)
Register to SIP proxy server (login to server)
- (int) - [refreshRegistration:](#)
Refresh the registration manually after successfully registered.
- (int) - [unRegisterServer](#)
Un-register from the SIP proxy server.
- (int) - [setLicenseKey:](#)
Set the license key. It must be called before setUser function.
- (int) - [getNICNums](#)
Get the Network Interface Card numbers.
- (NSString *) - [getLocalIpAddress:](#)
Get the local IP address by Network Interface Card index.
- (int) - [addAudioCodec:](#)
Enable an audio codec. It will appear in SDP.
- (int) - [addVideoCodec:](#)
Enable a video codec. It will appear in SDP.
- (BOOL) - [isAudioCodecEmpty](#)
Detect if the enabled audio codecs is empty.
- (BOOL) - [isVideoCodecEmpty](#)
Detect if enabled video codecs is empty or not.
- (int) - [setAudioCodecPayloadType:payloadType:](#)
Set the RTP payload type for dynamic audio codec.
- (int) - [setVideoCodecPayloadType:payloadType:](#)
Set the RTP payload type for dynamic Video codec.
- (void) - [clearAudioCodec](#)
Remove all enabled audio codecs.
- (void) - [clearVideoCodec](#)
Remove all enabled video codecs.
- (int) - [setAudioCodecParameter:parameter:](#)

- Set the codec parameter for audio codec.*

 - (int) - [setVideoCodecParameter:parameter:](#)
Set the codec parameter for video codec.
- (NSString *) - [getVersion](#)
Get the current version number of the SDK.
- (int) - [enableRport:](#)
Enable/disable rport(RFC3581).
- (int) - [enableEarlyMedia:](#)
Enable/disable Early Media.
- (int) - [enableReliableProvisional:](#)
Enable/disable PRACK.
- (int) - [enable3GppTags:](#)
Enable/disable the 3Gpp tags, including "ims.icsi.mmtel" and "g.3gpp.smsip".
- (void) - [enableCallbackSignaling:enableReceived:](#)
Enable/disable to callback the SIP messages.
- (int) - [setSrtpPolicy:](#)
Set the SRTP policy.
- (int) - [setRtpPortRange:maximumRtpAudioPort:minimumRtpVideoPort:maximumRtpVideoPort:](#)
Set the RTP ports range for audio and video streaming.
- (int) - [setRtcpPortRange:maximumRtcpAudioPort:minimumRtcpVideoPort:maximumRtcpVideoPort:](#)
Set the RTCP ports range for audio and video streaming.
- (int) - [enableCallForward:forwardTo:](#)
Enable call forwarding.
- (int) - [disableCallForward](#)
Disable the call forwarding. The SDK is not forwarding any incoming calls once this function is called.
- (int) - [enableSessionTimer:refreshMode:](#)
Allows to periodically refresh Session Initiation Protocol (SIP) sessions by sending INVITE requests repeatedly.
- (int) - [disableSessionTimer](#)
Disable the session timer.
- (void) - [setDoNotDisturb:](#)
Enable the "Do not disturb" to enable/disable.
- (void) - [enableAutoCheckMwi:](#)
Enable the CheckMwi to enable/disable.
- (int) - [setRtpKeepAlive:keepAlivePayloadType:deltaTransmitTimeMS:](#)
Enable or disable to send RTP keep-alive packet when the call is established.
- (int) - [setKeepAliveTime:](#)
Enable or disable to send SIP keep-alive packet.
- (int) - [setAudioSamples:maxPtime:](#)
Set the audio capturing sample.
- (int) - [addSupportedMimeType:mimeType:subMimeType:](#)
Set the SDK to receive the SIP message that includes special mime type.
- (NSString *) - [getSipMessageHeaderValue:headerName:](#)
Access the SIP header of SIP message.
- (long) - [addSipMessageHeader:methodName:msgType:headerName:headerValue:](#)
Add the SIP Message header into the specified outgoing SIP message.
- (int) - [removeAddedSipMessageHeader:](#)
Remove the headers (custom header) added by addSipMessageHeader.
- (void) - [clearAddedSipMessageHeaders](#)

Clear the added extension headers (custom headers)

- (long) - [modifySipMessageHeader:methodName:msgType:headerName:headerValue:](#)
Modify the special SIP header value for every outgoing SIP message.
- (int) - [removeModifiedSipMessageHeader:](#)
Remove the extension header (custom header) from every outgoing SIP message.
- (void) - [clearModifiedSipMessageHeaders](#)
Clear the modified headers value, and do not modify every outgoing SIP message header values any longer.
- (int) - [setVideoDeviceId:](#)
Set the video device that will be used for video call.
- (int) - [setVideoResolution:height:](#)
Set the video capturing resolution.
- (int) - [setVideoCropAndScale:](#)
When the camera does not support specified resolution, enable or disable SDK to crop and scale to the specified resolution.
- (int) - [setAudioBitrate:codecType:bitrateKbps:](#)
Set the audio bit rate.
- (int) - [setVideoBitrate:bitrateKbps:](#)
Set the video bitrate.
- (int) - [setVideoFrameRate:frameRate:](#)
Set the video frame rate.
- (int) - [sendVideo:sendState:](#)
Send the video to remote side.
- (void) - [setLocalVideoWindow:](#)
Set the window on which the local video image will be displayed.
- (int) - [setRemoteVideoWindow:remoteVideoWindow:](#)
Set the window for a session to display the received remote video image.
- (int) - [displayLocalVideo:](#)
Start/stop displaying the local video image.
- (int) - [setVideoNackStatus:](#)
Enable/disable the NACK feature (RFC4585) to help to improve the video quality.
- (void) - [muteMicrophone:](#)
Mute the device microphone. It's unavailable for Android and iOS.
- (void) - [muteSpeaker:](#)
Mute the device speaker. It's unavailable for Android and iOS.
- (int) - [setLoudspeakerStatus:](#)
Set the audio device that will be used for audio call.
- (int) - [setChannelOutputVolumeScaling:scaling:](#)
- (int) - [setChannelInputVolumeScaling:scaling:](#)
- (long) - [call:sendSdp:videoCall:](#)
Make a call.
- (int) - [rejectCall:code:](#)
rejectCall Reject the incoming call.
- (int) - [hangUp:](#)
hangUp Hang up the call.
- (int) - [answerCall:videoCall:](#)
answerCall Answer the incoming call.
- (int) - [updateCall:enableAudio:enableVideo:](#)
Use the re-INVITE to update the established call.
- (int) - [hold:](#)
Place a call on hold.

- (int) - [unHold:](#)
Take off hold.
- (int) - [muteSession:muteIncomingAudio:muteOutgoingAudio:muteIncomingVideo:muteOutgoingVideo:](#)
Mute the specified session audio or video.
- (int) - [forwardCall:forwardTo:](#)
Forward the call to another user once received an incoming call.
- (long) - [pickupBLFCall:videoCall:](#)
This function will be used for picking up a call based on the BLF (Busy Lamp Field) status.
- (int) - [sendDtmf:dtmfMethod:code:dtmfDuration:playDtmfTone:](#)
Send DTMF tone.
- (int) - [refer:referTo:](#)
Refer the current call to another one.

- (int) - [attendedRefer:replaceSessionId:referTo:](#)
Make an attended refer.
- (int) - [attendedRefer2:replaceSessionId:replaceMethod:target:referTo:](#)
Make an attended refer with specified request line and specified method embedded into the "Refer-To" header.
- (int) - [outOfDialogRefer:replaceMethod:target:referTo:](#)
Send an out of dialog REFER to replace the specified call.
- (long) - [acceptRefer:referSignaling:](#)
Once the REFER request accepted, a new call will be made if called this function. The function is usually called after onReceivedRefer callback event.
- (int) - [rejectRefer:](#)
Reject the REFER request.
- (int) - [enableSendPcmStreamToRemote:state:streamSamplesPerSec:](#)
Enable the SDK to send PCM stream data to remote side from another source instead of microphone.
- (int) - [sendPcmStreamToRemote:data:](#)
Send the audio stream in PCM format from another source instead of audio device capturing (microphone).
- (int) - [enableSendVideoStreamToRemote:state:](#)
Enable the SDK to send video stream data to remote side from another source instead of camera.
- (int) - [sendVideoStreamToRemote:data:width:height:](#)
Send the video stream to remote side.
- (int) - [setRtpCallback:](#)
Set the RTP callbacks to allow to access the sent and received RTP packets.
- (int) - [enableAudioStreamCallback:enable:callbackMode:](#)
Enable/disable the audio stream callback.
- (int) - [enableVideoStreamCallback:callbackMode:](#)
Enable/disable the video stream callback.
- (int) - [startRecord:recordFilePath:recordFileName:appendTimeStamp:audioFileFormat:audioRecordMode:aviFileCodecType:videoRecordMode:](#)
Start recording the call.
- (int) - [stopRecord:](#)
Stop recording.
- (int) - [playVideoFileToRemote:aviFile:loop:playAudio:](#)
Play an AVI file to remote party.
- (int) - [stopPlayVideoFileToRemote:](#)

- *Stop playing video file to remote party.*
- (int) - [playAudioFileToRemote:filename:fileSamplesPerSec:loop:](#)
Play a wave file to remote party.
- (int) - [stopPlayAudioFileToRemote:](#)
Stop playing wave file to remote party.
- (int) - [playAudioFileToRemoteAsBackground:filename:fileSamplesPerSec:](#)
Play a wave file to remote party as conversation background sound.
- (int) - [stopPlayAudioFileToRemoteAsBackground:](#)
Stop playing a wave file to remote party as conversation background sound.
- (void) - [audioPlayLoopbackTest:](#)
Used for the loop back testing against audio device.
- (int) - [createAudioConference](#)
Create an audio conference.
- (int) - [createVideoConference:videoWidth:videoHeight:displayLocalVideo:](#)
Create a video conference.
- (void) - [destroyConference](#)
Destroy the existent conference.
- (int) - [setConferenceVideoWindow:](#)
Set the window for a conference that is used to display the received remote video image.
- (int) - [joinToConference:](#)
Join a session into existent conference. If the call is in hold, please un-hold first.
- (int) - [removeFromConference:](#)
Remove a session from an existent conference.
- (int) - [setAudioRtcpBandwidth:BitsRR:BitsRS:KBitsAS:](#)
Set the audio RTCP bandwidth parameters as the RFC3556.
- (int) - [setVideoRtcpBandwidth:BitsRR:BitsRS:KBitsAS:](#)
Set the video RTCP bandwidth parameters as the RFC3556.
- (int) - [enableAudioQos:](#)
Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for audio channel.
- (int) - [enableVideoQos:](#)
Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for video channel.
- (int) - [setVideoMTU:](#)
Set the MTU size for video RTP packet.
- (int) - [getAudioStatistics:sendBytes:sendPackets:sendPacketsLost:sendFractionLost:sendRttMS:sendCodecType:sendJitterMS:sendAudioLevel:recvBytes:recvPackets:recvPacketsLost:recvFractionLost:recvCodecType:recvJitterMS:recvAudioLevel:](#)
Obtain the statistics of audio channel.
- (int) - [getVideoStatistics:sendBytes:sendPackets:sendPacketsLost:sendFractionLost:sendRttMS:sendCodecType:sendFrameWidth:sendFrameHeight:sendBitrateBPS:sendFramerate:recvBytes:recvPackets:recvPacketsLost:recvFractionLost:recvCodecType:recvFrameWidth:recvFrameHeight:recvBitrateBPS:recvFramerate:](#)
Obtain the statistics of video channel.
- (void) - [enableVAD:](#)
Enable/disable Voice Activity Detection (VAD).
- (void) - [enableCNG:](#)
Enable/disable Comfort Noise Generator (CNG).
- (int) - [sendOptions:sdp:](#)
Send OPTIONS message.

- (int) - [sendInfo:mimeType:subMimeType:infoContents:](#)
Send an INFO message to remote side in dialog.
- (long) - [sendMessage:mimeType:subMimeType:message:messageLength:](#)
Send a MESSAGE message to remote side in dialog.
- (long) - [sendOutOfDialogMessage:mimeType:subMimeType:isSMS:message:messageLength:](#)
Send an out of dialog MESSAGE message to remote side.
- (int) - [setPresenceMode:](#)
Indicate that SDK uses the P2P mode for presence or presence agent mode.
- (int) - [setDefaultSubscriptionTime:](#)
Set the default expiration time to be used when creating a subscription.
- (int) - [setDefaultPublicationTime:](#)
Set the default expiration time to be used when creating a publication.
- (long) - [presenceSubscribe:subject:](#)
Send a SUBSCRIBE message for subscribing the contact's presence status.
- (int) - [presenceTerminateSubscribe:](#)
Terminate the given presence subscription.
- (int) - [presenceAcceptSubscribe:](#)
Accept the presence SUBSCRIBE request which is received from contact.
- (int) - [presenceRejectSubscribe:](#)
Reject a presence SUBSCRIBE request which is received from contact.
- (int) - [setPresenceStatus:statusText:](#)
Send a NOTIFY message to contact to notify that presence status is online/offline/changed.
- (long) - [sendSubscription:eventName:](#)
Send a SUBSCRIBE message to remote side.
- (int) - [terminateSubscription:](#)
Terminate the given subscription.
- (BOOL) - [startKeepAwake](#)
Keep VoIP awake in the background. If you want your application to be able to receive the incoming call while it's running in background, you should call this function in applicationDidEnterBackground.
- (BOOL) - [stopKeepAwake](#)
Keep VoIP awake in the background. Call this function in applicationWillEnterForeground once your application comes back to foreground.
- (BOOL) - [startAudio](#)
Start Audio Device. Call it as AVAudioSessionInterruptionTypeEnded.
- (BOOL) - [stopAudio](#)
Stop Audio Device. Call it as AVAudioSessionInterruptionTypeBegan.
- (int) - [enableCallKit:](#)
Enable/disable CallKit(Native Integration).

Properties

- id< PortSIPEventDelegate > **delegate**

Detailed Description

PortSIP VoIP SDK functions class.

Author:

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Version:

16.3

See also:

<http://www.PortSIP.com>

PortSIP SDK functions class description.

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

- PortSIPSDK.h

PortSIPVideoRenderView Class Reference

PortSIP VoIP SDK Video Render View class.

```
#import <PortSIPVideoRenderView.h>
```

Inherits UIView.

Instance Methods

- (void) - [initWithVideoRender](#)
Initialize the Video Render view. Render should be initialized before using.
- (void) - [releaseVideoRender](#)
Release the Video Render.
- (void *) - [getVideoRenderView](#)
Don't use this. Just call by SDK.
- (void) - [updateVideoRenderFrame:](#)
Change the Video Render size.

Detailed Description

PortSIP VoIP SDK Video Render View class.

Author:

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Version:

16

See also:

<http://www.PortSIP.com>

PortSIP VoIP SDK Video Render View class description.

Method Documentation

- (void) updateVideoRenderFrame: (CGRect) *frameRect*

Change the Video Render size.

Remarks:

Example:

```
CGRect rect = videoRenderView.frame;
rect.size.width += 20;
rect.size.height += 20;

videoRenderView.frame = rect;
[videoRenderView setNeedsDisplay:YES];

CGRect renderRect = [videoRenderView bounds];
[videoRenderView updateVideoRenderFrame:renderRect];
```

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

- PortSIPVideoRenderView.h

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