PortSIP VoIP SDK Manual for Windows

Version 19.4
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Welcome to PortSIP VoIP SDK For Windows

Create your SIP-based application for multiple platforms (iOS, Android, Windows, macOS 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 macOS), 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.

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/

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 IDE: C#, VB.NET, VC++: Visual Studio 2012 or higher.
4. Compile the sample project directly. The trial version SDK allows a 2-3 minutes conversation.

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

C#/VB.NET:

```
1) Download the Sample project and extract it for C#/VB.NET.
2) Create a new "Windows application" project.
3) Copy the PortSIP_sdk.dll to project output directory: bin\release and bin\debug.
4) Copy the "PortSIP" folder to project folder and add into Solution.
5) Inherit the interface "SIPCallbackEvents" to process the callback events.
6) Right-click the project, choose "Properties". Click "Build" tab, and then check the
"Allow unsafe code" checkbox.

For more details please read the Sample project source code.
```

VC++:

```
1) Download and extract the sample project.
2) Create a new "MFC Application" project.
3) Copy the PortSIP_sdk.dll to project output directories.
4) Copy the "include/PortSIPLib" folder to project folder and add the ".hxx" files into
project.
5) Copy the "lib" folder to project folder and link "PortSIP_sdk.lib" into project.

For more details please read the Sample project source code.
```

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

1. Download and extract the SDK sample project ZIP file, compile and run the "P2PSample" project.
2. Run the P2Psample on two devices. For example, run it on device A and device B, and IP address for A is 192.168.1.10, IP address for B is 192.168.1.11.
3. Enter a user name and password on A. For example, user name 111, and password aaa (you can enter anything for the password as the SDK will ignore it). Enter a user name and password on B, for example: user name 222, password aaa.
4. Click the "Initialize" button on A and B. If the default port 5060 is already in use, the P2PSample will prompt "Initialize failure". In case of this, please click the "Uninitialize" button and change the local port, and click the "Initialize" button again.
5. The log box will appear "Initialized." if the SDK is successfully initialized.
6. To make call from A to B, enter: sip:222@192.168.1.11 and click "Dial" button; while to make call from B to A, enter: sip:111@192.168.1.10.

5. Is the SDK thread safe?

Yes, the SDK is thread safe. You can call any of the API functions without the need to consider the multiple threads. Note: the SDK allows to call API functions in callback events directly - except for the "onAudioRawCallback", "onVideoRawCallback", "onRTPPacketCallback" callbacks.

6. Does the SDK support native 64-bit?

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

Module Index

Modules

Here is a list of all modules:

Namespace Index

Packages

Here are the packages with brief descriptions (if available):

Class Index

Class List

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

Module Documentation

SDK functions

Modules

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Detailed Description

SDK functions

Initialize and register functions

Functions

- Int32 PortSIP.PortSIPLib.initialize (TRANSPORT_TYPE transportType, String localIp, Int32 localSIPPort, PORTSIP_LOG_LEVEL logLevel, String logFilePath, Int32 maxCallSessions, String sipAgentString, Int32 audioDeviceLayer, Int32 videoDeviceLayer, String TLSCertificatesRootPath, String TLSCipherList, Boolean verifyTLSCertificate)
  Initialize the SDK.

- void PortSIP.PortSIPLib.unInitialize ()
  Un-initialize the SDK and release resources.

- String PortSIP.PortSIPLib.getVersion ()
  Get the current version number of the SDK.

- Int32 PortSIP.PortSIPLib.setLicenseKey (String key)
  Set the license key. It must be called before setUser function.

Detailed Description

Initialize and register functions

Function Documentation

Int32 PortSIP.PortSIPLib.initialize (TRANSPORT_TYPE *transportType*, String *localIp*, Int32 *localSIPPort*, PORTSIP_LOG_LEVEL *logLevel*, String *logFilePath*, Int32 *maxCallSessions*, String *sipAgentString*, Int32 *audioDeviceLayer*, Int32 *videoDeviceLayer*, String *TLSCertificatesRootPath*, String *TLSCipherList*, Boolean *verifyTLSCertificate*)

Initialize the SDK.

Parameters

transport	Transport for SIP signaling. TRANSPORT_PERS is the PortSIP private transport for anti SIP blocking. It must be used with PERS.
localIP	The local computer IP address to be bound (for example: 192.168.1.108). It will be used for sending and receiving SIP messages and RTP packets. If this IP is passed in IPv6 format, the SDK will be using 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).
localSIPPort	The SIP message transport listener port, for example: 5060.
logLevel	Set the application log level. The SDK will generate "PortSIP_Log_datatime.log" file if the log enabled.
logFilePath	The log file path. The path (folder) MUST be existent.
maxCallLines	Theoretically, unlimited lines could be supported depending on the device capability. For SIP client recommended value ranges 1 - 100;
sipAgent	The User-Agent header to be inserted into SIP messages.
audioDeviceLayer	Specify the audio device layer to be used:

	0 = Use the OS default device. 1 = Virtual device, usually use this for the device which has no sound device installed.
videoDeviceLayer	Specifies the video device layer that should be used: 0 = Use the OS default device. 1 = Use Virtual device. Usually use this for the device which has no camera installed.
TLSCertificatesRootPath	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.
TLSCipherList	Specify the TLS cipher list. This parameter is usually passed as empty so that the SDK will offer all available ciphers.
verifyTLSCertificate	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

String PortSIP.PortSIPLib.getVersion ()

Get the current version number of the SDK.

Returns

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

Int32 PortSIP.PortSIPLib.setLicenseKey (String   *key*)

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

Parameters

key	The SDK license key, please purchase from PortSIP.

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

- Int32 PortSIP.PortSIPLib.getNICNums ()
  Get the Network Interface Card numbers.

- Int32 PortSIP.PortSIPLib.getLocalIpAddress (Int32 index, StringBuilder ip, Int32 ipSize)
  Get the local IP address by Network Interface Card index.

- Int32 PortSIP.PortSIPLib.setUser (String userName, String displayName, String authName, String password, String sipDomain, String sipServerAddr, Int32 sipServerPort, String stunServerAddr, Int32 stunServerPort, String outboundServerAddr, Int32 outboundServerPort)
  Set user account info.

- void PortSIP.PortSIPLib.removeUser ()
  Remove the user. It will un-register from SIP server given that the user is already registered.

- Int32 PortSIP.PortSIPLib.setDisplayName (String displayName)
  Set the display name of user.

- Int32 PortSIP.PortSIPLib.setInstanceId (String uuid)
  Set outbound (RFC5626) instanceId to be used in contact headers.

- Int32 PortSIP.PortSIPLib.registerServer (Int32 expires, Int32 retryTimes)
  Register to SIP proxy server (login to server).

- Int32 PortSIP.PortSIPLib.unRegisterServer (Int32 waitMS)
  Un-register from the SIP proxy server.

- Int32 PortSIP.PortSIPLib.enableRport (Boolean enable)
  Enable/disable rport(RFC3581).

- Int32 PortSIP.PortSIPLib.enableEarlyMedia (Boolean enable)
  Enable/disable Early Media.

- Int32 PortSIP.PortSIPLib.enablePriorityIPv6Domain (Boolean enable)
  Enable/disable which allows specifying the preferred protocol when a domain supports both IPV4 and IPV6 simultaneously.

- Int32 PortSIP.PortSIPLib.setUriUserEncoding (String character, Boolean enable)
  Modifies the default URI user character needs to be escaped.

- Int32 PortSIP.PortSIPLib.setReliableProvisional (Int32 mode)
  Enable/disable PRACK.

- Int32 PortSIP.PortSIPLib.enable3GppTags (Boolean enable)
  Enable/disable the 3Gpp tags, including "ims.icsi.mmtel" and "g.3gpp.smsip".

- void PortSIP.PortSIPLib.enableCallbackSignaling (Boolean enableSending, Boolean enableReceived)

  Enable/disable to callback the SIP messages.

- Int32 PortSIP.PortSIPLib.enableRtpCallback (Int32 sessionId, Int32 mediaType, Int32 directionMode)

  Set the RTP callbacks to allow access to the sent and received RTP packets. The onRTPPacketCallback events will be triggered.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.getNICNums ()

Get the Network Interface Card numbers.

Returns

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

Int32 PortSIP.PortSIPLib.getLocalIpAddress (Int32 *index*, StringBuilder *ip*, Int32 *ipSize*)

Get the local IP address by Network Interface Card index.

Parameters

index	The IP address index. For example, if the PC has two NICs, and we wish to obtain the second NIC IP. Set this parameter to 1 and the first NIC IP index is 0.
ip	The buffer that is used to receive the IP.
ipSize	The IP buffer size, which cannot be less than 32 bytes.

Returns

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

Int32 PortSIP.PortSIPLib.setUser (String *userName*, String *displayName*, String *authName*, String *password*, String *sipDomain*, String *sipServerAddr*, Int32 *sipServerPort*, String *stunServerAddr*, Int32 *stunServerPort*, String *outboundServerAddr*, Int32 *outboundServerPort*)

Set user account info.

Parameters

userName	Account (User name) of the SIP. Usually provided by an IP-Telephony service provider.
displayName	The display name of user. You can set it as your like, such as "James Kend". It's optional.
authName	Authorization user name (usually equal to the username).
password	The password of user. It's optional.
localIp	The local computer IP address to be bound. For example: 192.168.1.108. It will be used for sending and receiving SIP message and RTP packet. If pass this IP as the IPv6 format, the SDK will use IPv6.
localSipPort	The SIP message transport listener port. For example: 5060.
userDomain	User domain; this parameter is optional that allows to pass an empty string if you are not using the domain.
sipServer	SIP proxy server IP or domain. For example: xx.xxx.xx.x or sip.xxx.com.
sipServerPort	Port of the SIP proxy server. For example: 5060.
stunServer	Stun server used for NAT traversal. It's optional and can pass empty string to disable STUN.
stunServerPort	STUN server port. It will be ignored if the outboundServer is empty.
outboundServer	Outbound proxy server. For example: sip.domain.com. It's optional and allows to pass a empty string if not using outbound server.
outboundServerPort	Outbound proxy server port, it will be ignored if the outboundServer is empty.

Returns

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

Int32 PortSIP.PortSIPLib.setDisplayName (String *displayName*)

Set the display name of user.

Parameters

displayName	that will appear in the From/To Header.

Returns

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

Int32 PortSIP.PortSIPLib.setInstanceId (String *uuid*)

Set outbound (RFC5626) instanceId to be used in contact headers.

Parameters

uuid	The ID that will appear in the contact header. Please make sure it's a unique ID.

Returns

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

Int32 PortSIP.PortSIPLib.registerServer (Int32 *expires*, Int32 *retryTimes*)

Register to SIP proxy server (login to server).

Parameters

expires	Registration refresh Interval in seconds with maximum 3600. It will be inserted into SIP REGISTER message headers.
retryTimes	The retry times if failed to refresh the registration. If it's set to be less than or equal to 0, the retry will be disabled and onRegisterFailure callback triggered when retry failed.

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 succeeded, onRegisterSuccess will be triggered, otherwise onRegisterFailure triggered.

Int32 PortSIP.PortSIPLib.unRegisterServer (Int32 *waitMS*)

Un-register from the SIP proxy server.

Parameters

waitMS	Wait for the server to reply that the un-registration is successful, waitMS is the longest waiting milliseconds, 0 means not waiting.

Returns

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

Int32 PortSIP.PortSIPLib.enableRport (Boolean *enable*)

Enable/disable rport(RFC3581).

Parameters

enable	Set to true to enable the SDK to support rport. By default it is enabled.

Returns

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

Int32 PortSIP.PortSIPLib.enableEarlyMedia (Boolean *enable*)

Enable/disable Early Media.

Parameters

enable	Set to true to enable the SDK to support Early Media. By default 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.

Int32 PortSIP.PortSIPLib.enablePriorityIPv6Domain (Boolean *enable*)

Enable/disable which allows specifying the preferred protocol when a domain supports both IPV4 and IPV6 simultaneously.

Parameters

enable	Set to true to enable priority IPv6 Domain. with the default priority being IPV4.

Returns

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

Int32 PortSIP.PortSIPLib.setUriUserEncoding (String *character*, Boolean *enable*)

Modifies the default URI user character needs to be escaped.

Parameters

character	The character to be modified, set one at a time.
enable	Whether escaping is required, true for allowing escaping, false for disabling escaping.

Returns

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

Int32 PortSIP.PortSIPLib.setReliableProvisional (Int32 *mode*)

Enable/disable PRACK.

Parameters

mode	Modes work as follows: 0 - Never, Disable PRACK,By default the PRACK is disabled. 1 - SupportedEssential, Only send reliable provisionals if sending a body and far end supports. 2 - Supported, Always send reliable provisionals if far end supports. 3 - Required Always send reliable provisionals.

Returns

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

Int32 PortSIP.PortSIPLib.enable3GppTags (Boolean *enable*)

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

Parameters

enable	Set to true to enable 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 PortSIP.PortSIPLib.enableCallbackSignaling (Boolean *enableSending*, Boolean *enableReceived*)

Enable/disable to callback the SIP messages.

Parameters

enableSending	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.
enableReceived	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.

Int32 PortSIP.PortSIPLib.enableRtpCallback (Int32   *sessionId*, Int32   *mediaType*, Int32 *directionMode*)

Set the RTP callbacks to allow access to the sent and received RTP packets. The onRTPPacketCallback events will be triggered.

Parameters

sessionId	The session ID of call.
mediaType	0 -audo 1-video 2-screen.
directionMode	The RTP stream callback mode.

Type	Description
DIRECTION_SEND	Callback the send RTP stream for one channel based on the given sessionId.
DIRECTION_RECV	Callback the received RTP stream for one channel based on the given sessionId.
DIRECTION_SEND_RECV	Callback both local and remote RTWP stream on the given sessionId.

Returns

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

Audio and video codecs functions

Functions

- Int32 PortSIP.PortSIPLib.addAudioCodec (AUDIOCODEC_TYPE codecType)
  Enable an audio codec. It will be appears in SDP.

- Int32 PortSIP.PortSIPLib.addVideoCodec (VIDEOCODEC_TYPE codecType)
  Enable a video codec. It will appear in SDP.

- Boolean PortSIP.PortSIPLib.isAudioCodecEmpty ()
  Detect if enabled audio codecs is empty or not.

- Boolean PortSIP.PortSIPLib.isVideoCodecEmpty ()
  Detect if enabled video codecs is empty or not.

- Int32 PortSIP.PortSIPLib.setAudioCodecPayloadType (AUDIOCODEC_TYPE codecType, Int32 payloadType)
  Set the RTP payload type for dynamic audio codec.

- Int32 PortSIP.PortSIPLib.setVideoCodecPayloadType (VIDEOCODEC_TYPE codecType, Int32 payloadType)
  Set the RTP payload type for dynamic Video codec.

- void PortSIP.PortSIPLib.clearAudioCodec ()
  Remove all enabled audio codecs.

- void PortSIP.PortSIPLib.clearVideoCodec ()
  Remove all enabled video codecs.

- Int32 PortSIP.PortSIPLib.setAudioCodecParameter (AUDIOCODEC_TYPE codecType, String parameter)
  Set the codec parameter for audio codec.

- Int32 PortSIP.PortSIPLib.setVideoCodecParameter (VIDEOCODEC_TYPE codecType, String parameter)
  Set the codec parameter for video codec.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.addAudioCodec (AUDIOCODEC_TYPE *codecType*)

Enable an audio codec. It will be appears in SDP.

Parameters

codecType	Audio codec type.

Returns

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

Int32 PortSIP.PortSIPLib.addVideoCodec (VIDEOCODEC_TYPE *codecType*)

Enable a video codec. It will appear in SDP.

Parameters

codecType	Video codec type.

Returns

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

Boolean PortSIP.PortSIPLib.isAudioCodecEmpty ()

Detect if enabled audio codecs is empty or not.

Returns

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

Boolean PortSIP.PortSIPLib.isVideoCodecEmpty ()

Detect if enabled video codecs is empty or not.

Returns

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

Int32 PortSIP.PortSIPLib.setAudioCodecPayloadType (AUDIOCODEC_TYPE *codecType*, Int32 *payloadType*)

Set the RTP payload type for dynamic audio codec.

Parameters

codecType	Audio codec type, which is defined in the PortSIPTypes file.
payloadType	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 value a specific error code.

Int32 PortSIP.PortSIPLib.setVideoCodecPayloadType ([VIDEOCODEC_TYPE](#) *codecType*, Int32 *payloadType*)

Set the RTP payload type for dynamic Video codec.

Parameters

codecType	Video codec type, which is defined in the PortSIPTypes file.
payloadType	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.

Int32 PortSIP.PortSIPLib.setAudioCodecParameter ([AUDIOCODEC_TYPE](#) *codecType*, String *parameter*)

Set the codec parameter for audio codec.

Parameters

codecType	Audio codec type, defined in the PortSIPTypes file.
parameter	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:

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

Int32 PortSIP.PortSIPLib.setVideoCodecParameter ([VIDEOCODEC_TYPE](#) *codecType*, String *parameter*)

Set the codec parameter for video codec.

Parameters

codecType	Video codec type, defined in the PortSIPTypes file.
parameter	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:

```
setVideoCodecParameter(VIDEO_CODEC_H264, "profile-level-id=420033;
packetization-mode=0");
```

Additional setting functions

Functions

- Int32 PortSIP.PortSIPLib.setSrtpPolicy (SRTP_POLICY srtpPolicy, Boolean allowSrtpOverUnsecureTransport)
  Set the SRTP policy.

- Int32 PortSIP.PortSIPLib.setRtpPortRange (Int32 minimumRtpPort, Int32 maximumRtpPort)
  Set the RTP ports range for RTP streaming.

- Int32 PortSIP.PortSIPLib.enableCallForward (Boolean forBusyOnly, String forwardTo)
  Enable call forward.

- Int32 PortSIP.PortSIPLib.disableCallForward ()
  Disable the call forwarding. The SDK is not forwarding any incoming call after this function is called.

- Int32 PortSIP.PortSIPLib.enableSessionTimer (Int32 timerSeconds, SESSION_REFRESH_MODE refreshMode)
  Allows to periodically refresh Session Initiation Protocol (SIP) sessions by sending INVITE requests repeatedly.

- Int32 PortSIP.PortSIPLib.disableSessionTimer ()
  Disable the session timer.

- void PortSIP.PortSIPLib.setDoNotDisturb (Boolean state)
  Enable the "Do not disturb" to enable/disable.

- Int32 PortSIP.PortSIPLib.enableAutoCheckMwi (Boolean state)
  Allows to enable/disable the check MWI (Message Waiting Indication) automatically.

- Int32 PortSIP.PortSIPLib.setRtpKeepAlive (Boolean state, Int32 keepAlivePayloadType, Int32 deltaTransmitTimeMS)
  Enable or disable to send RTP keep-alive packet when the call is established.

- Int32 PortSIP.PortSIPLib.setKeepAliveTime (Int32 keepAliveTime)
  Enable or disable to send SIP keep-alive packet.

- Int32 PortSIP.PortSIPLib.getSipMessageHeaderValue (String sipMessage, String headerName, StringBuilder headerValue, Int32 headerValueLength)
  Access the SIP header of SIP message.

- Int32 PortSIP.PortSIPLib.addSipMessageHeader (Int32 sessionId, String methodName, Int32 msgType, String headerName, String headerValue)
  Add the SIP Message header into the specified outgoing SIP message.

- Int32 PortSIP.PortSIPLib.removeAddedSipMessageHeader (Int32 sipMessageHeaderId)
  Remove the headers (custom header) added by addSipMessageHeader.

- Int32 [PortSIP.PortSIPLib.clearAddedSipMessageHeaders](#) ()
  Clear the added extension headers (custom headers)

- Int32 [PortSIP.PortSIPLib.modifySipMessageHeader](#) (Int32 sessionId, String methodName, Int32 msgType, String headerName, String headerValue)
  Modify the special SIP header value for every outgoing SIP message.

- Int32 [PortSIP.PortSIPLib.removeModifiedSipMessageHeader](#) (Int32 sipMessageHeaderId)
  Remove the extension header (custom header) from every outgoing SIP message.

- Int32 [PortSIP.PortSIPLib.clearModifiedSipMessageHeaders](#) ()
  Clear the modified headers value, and do not modify every outgoing SIP message header values any longer.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.setSrtpPolicy ([SRTP_POLICY](#) *srtpPolicy*, Boolean *allowSrtpOverUnsecureTransport*)

Set the SRTP policy.

Parameters

srtpPolicy	The SRTP policy.

Returns

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

Int32 PortSIP.PortSIPLib.setRtpPortRange (Int32 *minimumRtpPort,* Int32 *maximumRtpPort*)

Set the RTP ports range for RTP streaming.

Parameters

minimumRtpPort	The minimum RTP port.
maximumRtpPort	The maximum RTP port.

Returns

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

Remarks

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

Int32 PortSIP.PortSIPLib.enableCallForward (Boolean *forBusyOnly*, String *forwardTo*)

Enable call forward.

Parameters

forBusyOnly	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.
forwardTo	The call forward target. It must be like sip:xxxx@sip.portsip.com.

Returns

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

Int32 PortSIP.PortSIPLib.disableCallForward ()

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

Returns

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

Int32 PortSIP.PortSIPLib.enableSessionTimer (Int32 *timerSeconds*, SESSION_REFRESH_MODE *refreshMode*)

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

Parameters

timerSeconds	The value of the refreshment interval in seconds. Minimum value of 90 seconds required.
refreshMode	Allow to set the session refresh by UAC or UAS: SESSION_REFERESH_UAC or SESSION_REFERESH_UAS;

Returns

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

Remarks

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

Int32 PortSIP.PortSIPLib.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 PortSIP.PortSIPLib.setDoNotDisturb (Boolean *state*)

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

Parameters

state	If it is set to true, the SDK will reject all incoming calls anyway.

Int32 PortSIP.PortSIPLib.enableAutoCheckMwi (Boolean *state*)

Allows to enable/disable the check MWI (Message Waiting Indication) automatically.

Parameters

state	If it is set as true, MWI will be checked automatically once successfully registered to a SIP proxy server.

Returns

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

Int32 PortSIP.PortSIPLib.setRtpKeepAlive (Boolean *state*, Int32 *keepAlivePayloadType*, Int32 *deltaTransmitTimeMS*)

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

Parameters

state	Set to true to allow to send the keep-alive packet during the conversation.
keepAlivePayload Type	The payload type of the keep-alive RTP packet. It's usually set to 126.
deltaTransmitTime MS	The keep-alive RTP packet sending interval, in millisecond. Recommend 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.

Int32 PortSIP.PortSIPLib.setKeepAliveTime (Int32 *keepAliveTime*)

Enable or disable to send SIP keep-alive packet.

Parameters

keepAliveTime	This is the SIP keep alive time interval in seconds. Set it to 0 to disable the SIP keep alive. Recommend to set as 30 or 50.

Returns

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

Int32 PortSIP.PortSIPLib.getSipMessageHeaderValue (String *sipMessage*, String *headerName*, StringBuilder *headerValue*, Int32 *headerValueLength*)

Access the SIP header of SIP message.

Parameters

sipMessage	The SIP message.
headerName	The header which wishes to access the SIP message.
headerValue	The buffer to receive header value.
headerValueLength	The headerValue buffer size. Usually we recommend to set it more than 512 bytes.

Returns

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

Remarks

When receiving a SIP message in the onReceivedSignaling callback event, and wishes to get SIP message header value, please use getSipMessageHeaderValue:

```
StringBuilder value = new StringBuilder();
value.Length = 512;
getSipMessageHeaderValue(message, name, value);
```

Int32 PortSIP.PortSIPLib.addSipMessageHeader (Int32 *sessionId*, String *methodName*, Int32 *msgType*, String *headerName*, String *headerValue*)

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

Parameters

sessionId	Add the header to the SIP message with the specified session Id only. By setting to -1, it will be added to all messages.
methodName	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.
msgType	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.
headerName	The custom header name that will appears in every outgoing SIP message.
headerValue	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.

Int32 PortSIP.PortSIPLib.removeAddedSipMessageHeader (Int32 *sipMessageHeaderId*)

Remove the headers (custom header) added by addSipMessageHeader.

Parameters

addedSipMessageId	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.

Int32 PortSIP.PortSIPLib.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.

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

Int32 PortSIP.PortSIPLib.modifySipMessageHeader (Int32   *sessionId*, String *methodName*, Int32   *msgType*, String   *headerName*, String   *headerValue*)

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

Parameters

sessionId	The header to the SIP message with the specified session Id. By setting to -1, it will be added to all messages.
methodName	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.
msgType	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.

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.

Int32 PortSIP.PortSIPLib.removeModifiedSipMessageHeader (Int32 *sipMessageHeaderId*)

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

Parameters

modifiedSipMessageId	The modifiedSipMessageId is returned by modifySipMessageHeader.

Returns

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

Int32 PortSIP.PortSIPLib.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:

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

Audio and video functions

Functions

- Int32 PortSIP.PortSIPLib.addSupportedMimeType (String methodName, String mimeType, String subMimeType)
  Set the SDK to receive the SIP messages that include special mime type.

- Int32 PortSIP.PortSIPLib.setAudioSamples (Int32 ptime, Int32 maxPtime)
  Set the audio capture sample.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.addSupportedMimeType (String *methodName*, String *mimeType*, String *subMimeType*)

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

Parameters

methodName	Method name of the SIP message, such as INVITE, OPTION, INFO, MESSAGE, UPDATE, ACK etc. For more details please read the RFC3261.
mimeType	The mime type of SIP message.
subMimeType	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 these media types (mime types) below for 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 send an INFO SIP message with its "Content-Type" header value "text/plain". SDK will reject this INFO message, as "text/plain" of INFO message is not included in the default support list. How should we enable the SDK to receive the SIP INFO message that includes "text/plain" mime type? The answer is addSupportedMimyType:

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

If we want to receive the NOTIFY message with "application/media_control+xml", please:

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

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

Int32 PortSIP.PortSIPLib.setAudioSamples (Int32 *ptime*, Int32 *maxPtime*)

Set the audio capture sample.

Parameters

ptime	It should be a multiple of 10 between 10 - 60 (with 10 and 60 inclusive).
maxPtime	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.

Access SIP message header functions

Functions

- Int32 PortSIP.PortSIPLib.setAudioDeviceId (Int32 recordingDeviceId, Int32 playoutDeviceId)
  Set the audio device that will be used for audio call.

- Int32 PortSIP.PortSIPLib.setVideoOrientation (Int32 rotation)
  Set the video device that will be used for video call.

- Int32 PortSIP.PortSIPLib.setVideoDeviceId (Int32 deviceId)
  Set the video device that will be used for video call.

- Int32 PortSIP.PortSIPLib.setVideoResolution (Int32 width, Int32 height)
  Set the video capturing resolution.

- Int32 PortSIP.PortSIPLib.setAudioBitrate (Int32 sessionId, AUDIOCODEC_TYPE audioCodecType, Int32 bitrateKbps)
  Set the audio bitrate.

- Int32 PortSIP.PortSIPLib.setVideoBitrate (Int32 sessionId, Int32 bitrateKbps)
  Set the video bitrate.

- Int32 PortSIP.PortSIPLib.setVideoFrameRate (Int32 sessionId, Int32 frameRate)
  Set the video frame rate.

- Int32 PortSIP.PortSIPLib.sendVideo (Int32 sessionId, Boolean sendState)
  Send the video to remote side.

- void PortSIP.PortSIPLib.muteMicrophone (Boolean mute)
  Mute the device microphone. It's unavailable for Android and iOS.

- void PortSIP.PortSIPLib.muteSpeaker (Boolean mute)
  Mute the device speaker. It's unavailable for Android and iOS.

- void PortSIP.PortSIPLib.setChannelOutputVolumeScaling (Int32 sessionId, Int32 scaling)
- void PortSIP.PortSIPLib.setChannelInputVolumeScaling (Int32 sessionId, Int32 scaling)
- Int32 PortSIP.PortSIPLib.setRemoteVideoWindow (Int32 sessionId, IntPtr remoteVideoWindow)
  Set the window for a session that is used to display the received remote video image.

- Int32 PortSIP.PortSIPLib.displayLocalVideo (Boolean state, Boolean mirror, IntPtr localVideoWindow)
  Start/stop displaying the local video image.

- Int32 PortSIP.PortSIPLib.setVideoNackStatus (Boolean state)
  Enable/disable the NACK feature (rfc6642) that helps to improve the video quality.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.setAudioDeviceId (Int32 *recordingDeviceId*, Int32 *playoutDeviceId*)

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

Parameters

recordingDeviceId	Device ID (index) for audio recording. (Microphone).
playoutDeviceId	Device ID (index) for audio playback (Speaker).

Returns

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

Int32 PortSIP.PortSIPLib.setVideoOrientation (Int32 *rotation*)

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

Parameters

rotation	for video

Returns

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

Int32 PortSIP.PortSIPLib.setVideoDeviceId (Int32 *deviceId*)

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

Parameters

deviceId	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.

Int32 PortSIP.PortSIPLib.setVideoResolution (Int32 *width*, Int32 *height*)

Set the video capturing resolution.

Parameters

width	Video width.
height	Video height.

Returns

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

Int32 PortSIP.PortSIPLib.setAudioBitrate (Int32 *sessionId*, AUDIOCODEC_TYPE *audioCodecType,* Int32 *bitrateKbps*)

Set the audio bitrate.

Parameters

bitrateKbps	The audio bitrate in KBPS.

Returns

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

Int32 PortSIP.PortSIPLib.setVideoBitrate (Int32 *sessionId,* Int32 *bitrateKbps*)

Set the video bitrate.

Parameters

bitrateKbps	The video bitrate in KBPS.

Returns

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

Int32 PortSIP.PortSIPLib.setVideoFrameRate (Int32 *sessionId,* Int32 *frameRate*)

Set the video frame rate.

Parameters

frameRate	The frame rate value with minimum value 5, and maximum value 30. A greater value will enable you 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. The SDK uses default frame rate.

Int32 PortSIP.PortSIPLib.sendVideo (Int32 *sessionId,* Boolean *sendState*)

Send the video to remote side.

Parameters

sessionId	The session ID of the call.
sendState	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 PortSIP.PortSIPLib.muteMicrophone (Boolean *mute*)

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

Parameters

mute	If the value is set to true, the microphone will be muted. You may also set it to false to un-mute it.

void PortSIP.PortSIPLib.muteSpeaker (Boolean *mute*)

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

Parameters

mute	If the value is set to true, the speaker is muted. You may also set it to false to un-mute it.

void PortSIP.PortSIPLib.setChannelOutputVolumeScaling (Int32 *sessionId*, Int32 *scaling*)

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

Parameters

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

Returns

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

void PortSIP.PortSIPLib.setChannelInputVolumeScaling (Int32 *sessionId*, Int32 *scaling*)

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

Parameters

sessionId	The session ID of the call.
scaling	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.

Int32 PortSIP.PortSIPLib.setRemoteVideoWindow (Int32 *sessionId*, IntPtr *remoteVideoWindow*)

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

Parameters

sessionId	The session ID of the call.
remoteVideoWindo	The window to display received remote video image.

w	

Returns

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

Int32 PortSIP.PortSIPLib.displayLocalVideo (Boolean *state,* Boolean *mirror,* IntPtr *localVideoWindow*)

Start/stop displaying the local video image.

Parameters

state	Set to true to display local video image.
mirror	Set to true to display the mirror image of local video.
localVideoWindow	The window on which the local video image from camera will be displayed.

Returns

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

Int32 PortSIP.PortSIPLib.setVideoNackStatus (Boolean *state*)

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

Parameters

state	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.

Call functions

Functions

- Int32 PortSIP.PortSIPLib.call (String callee, Boolean sendSdp, Boolean videoCall)
  Make a call.

- Int32 PortSIP.PortSIPLib.rejectCall (Int32 sessionId, int code)
  rejectCall Reject the incoming call.

- Int32 PortSIP.PortSIPLib.hangUp (Int32 sessionId)
  hangUp Hang up the call.

- Int32 PortSIP.PortSIPLib.answerCall (Int32 sessionId, Boolean videoCall)
  answerCall Answer the incoming call.

- Int32 PortSIP.PortSIPLib.updateCall (Int32 sessionId, bool enableAudio, bool enableVideo, bool enableScreen)
  Use the re-INVITE to update the established call.

- Int32 PortSIP.PortSIPLib.hold (Int32 sessionId)
  To place a call on hold.

- Int32 PortSIP.PortSIPLib.unHold (Int32 sessionId)
  Take off hold.

- Int32 PortSIP.PortSIPLib.muteSession (Int32 sessionId, Boolean muteIncomingAudio, Boolean muteOutgoingAudio, Boolean muteIncomingVideo, Boolean muteOutgoingVideo)
  Mute the specified session audio or video.

- Int32 PortSIP.PortSIPLib.forwardCall (Int32 sessionId, String forwardTo)
  Forward call to another one when receiving the incoming call.

- Int32 PortSIP.PortSIPLib.pickupBLFCall (String replaceDialogId, Boolean videoCall)
  This function will be used for picking up a call based on the BLF (Busy Lamp Field) status.

- Int32 PortSIP.PortSIPLib.sendDtmf (Int32 sessionId, DTMF_METHOD dtmfMethod, int code, int dtmfDuration, bool playDtmfTone)
  Send DTMF tone.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.call (String *callee*, Boolean *sendSdp*, Boolean *videoCall*)

Make a call.

Parameters

callee	The callee. It can be either name or full SIP URI. For example: user001, sip:user001@sip.iptel.org or sip:user002@sip.yourdomain.com:5068
sendSdp	If it's set to false, the outgoing call doesn't include the SDP in INVITE message.
videoCall	If it's set to true with at least one video codecs 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 that 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. You need to detect the call final state in onInviteTrying, onInviteRinging, onInviteFailure callback events.

Int32 PortSIP.PortSIPLib.rejectCall (Int32 *sessionId*, int *code*)

rejectCall Reject the incoming call.

Parameters

sessionId	The sessionId of the call.
code	Reject code. For example, 486, 480 etc.

Returns

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

Int32 PortSIP.PortSIPLib.hangUp (Int32 *sessionId*)

hangUp Hang up the call.

Parameters

sessionId	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.

Int32 PortSIP.PortSIPLib.answerCall (Int32 *sessionId*, Boolean *videoCall*)

answerCall Answer the incoming call.

Parameters

sessionId	The session ID of call.
videoCall	If the incoming call is a video call and the video codec is matched, set it to true to answer the video call.

	If it's set to false, the answered 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.

Int32 PortSIP.PortSIPLib.updateCall (Int32   *sessionId*, bool   *enableAudio*, bool *enableVideo*, bool   *enableScreen*)

Use the re-INVITE to update the established call.

Parameters

sessionId	The session ID of call.
enableAudio	Set to true to allow the audio in updated call, or false to disable audio in updated call.
enableVideo	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 value a specific error code.

Remarks

Example usage:

Example 1: A called B with the audio only, B answered A, there has an audio conversation between A, B. Now A wants to see B visually, A could use these functions to do it.

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

Example 2: Remove video stream from current conversation.

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

Int32 PortSIP.PortSIPLib.hold (Int32   *sessionId*)

To place a call on hold.

Parameters

sessionId	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.

Int32 PortSIP.PortSIPLib.unHold (Int32   *sessionId*)

Take off hold.

Parameters

sessionId	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.

Int32 PortSIP.PortSIPLib.muteSession (Int32 *sessionId*, Boolean *muteIncomingAudio*, Boolean *muteOutgoingAudio*, Boolean *muteIncomingVideo*, Boolean *muteOutgoingVideo*)

Mute the specified session audio or video.

Parameters

sessionId	The session ID of the call.
muteIncomingAudio	Set it to true to mute incoming audio stream, and remote side audio cannot be heard.
muteOutgoingAudio	Set it to true to mute outgoing audio stream, and the remote side can't hear the audio.
muteIncomingVideo	Set it to true to mute incoming video stream, and the remote side video will be invisible.
muteOutgoingVideo	Set it to true to mute outgoing video stream, and the remote side can't see the video.

Returns

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

Int32 PortSIP.PortSIPLib.forwardCall (Int32 *sessionId*, String *forwardTo*)

Forward call to another one when receiving the incoming call.

Parameters

sessionId	The session ID of the call.
forwardTo	Target of the 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 value a specific error code.

Int32 PortSIP.PortSIPLib.pickupBLFCall (String *replaceDialogId*, Boolean *videoCall*)

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

Parameters

replaceDialogId	The session ID of the call.
videoCall	Target of the forwarding. It can be "sip:number@sipserver.com" or "number" only.

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("100", "dialog");
2. 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.

Int32 PortSIP.PortSIPLib.sendDtmf (Int32   *sessionId*, DTMF_METHOD   *dtmfMethod*, int   *code*, int   *dtmfDuration*, bool   *playDtmfTone*)

Send DTMF tone.

Parameters

sessionId	The session ID of the call.
dtmfMethod	DTMF tone could be sent with two methods: DTMF_RFC2833 and DTMF_INFO, of which DTMF_RFC2833 is recommend.
code	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

| *dtmfDuration* | The DTMF tone samples. Recommended value 160. |
| *playDtmfTone* | If it is set 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

- Int32 PortSIP.PortSIPLib.refer (Int32 sessionId, String referTo)
  Refer the current call to another one.

- Int32 PortSIP.PortSIPLib.attendedRefer (Int32 sessionId, Int32 replaceSessionId, String referTo)
- Int32 PortSIP.PortSIPLib.attendedRefer2 (IntPtr libSDK, Int32 sessionId, Int32 replaceSessionId, String replaceMethod, String target, String referTo)
  Make an attended refer with specified request line and specified method embedded into the "Refer-To" header.

- Int32 PortSIP.PortSIPLib.outOfDialogRefer (Int32 replaceSessionId, String replaceMethod, String target, String referTo)
  Send an out of dialog REFER to replace the specified call.

- Int32 PortSIP.PortSIPLib.acceptRefer (Int32 referId, String referSignalingMessage)
  Accept the REFER request, and a new call will be made if called this function. The function is usually called after onReceivedRefer callback event.

- Int32 PortSIP.PortSIPLib.rejectRefer (Int32 referId)
  Reject the REFER request.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.refer (Int32 *sessionId*, String *referTo*)

Refer the current call to another one.

Parameters

sessionId	The session ID of the call.
referTo	Target of the refer, which can be either "sip:number@sipserver.com" or "number".

Returns

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

Remarks

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

You can watch the video on YouTube at https://www.youtube.com/watch?v=_2w9EGgr3FY. It will demonstrate the transfer.

Int32 PortSIP.PortSIPLib.attendedRefer (Int32 *sessionId*, Int32 *replaceSessionId*, String *referTo*)

```
    @brief  Make an attended refer.

    @param sessionId      The session ID of the call.
    @param replaceSessionId Session ID of the repferred call.
    @param referTo        Target of the refer, which can be either
"sip:number@sipserver.com" or "number".

    @return If the function succeeds, it will return value 0. If the function fails,
it will return a specific error code.
    @remark
    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=NezhIZW4lV4,
```

which will demonstrate the transfer.

Int32 PortSIP.PortSIPLib.attendedRefer2 (IntPtr *libSDK*, Int32 *sessionId*, Int32 *replaceSessionId*, String *replaceMethod*, String *target*, String *referTo*)

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

Parameters

sessionId	Session ID of the call.
replaceSessionId	Session ID of the replaced call.
replaceMethod	The SIP method name which will be embeded in the "Refer-To" header, usually INVITE or BYE.
target	The target to which the REFER message will be sent. It appears in the "Request Line" of REFER message.
referTo	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 read the sample project source code for more details, or you can watch the video on YouTube at https://www.youtube.com/watch?v=NezhIZW4lV4, which will demonstrate the transfer.

Int32 PortSIP.PortSIPLib.outOfDialogRefer (Int32 *replaceSessionId*, String *replaceMethod*, String *target*, String *referTo*)

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

Parameters

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

Int32 PortSIP.PortSIPLib.acceptRefer (Int32 *referId*, String *referSignalingMessage*)

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

Parameters

referId	The ID of REFER request that comes from onReceivedRefer callback event.
referSignalingMessage	The SIP message of REFER request that comes from onReceivedRefer callback event.

Returns

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

Int32 PortSIP.PortSIPLib.rejectRefer (Int32 *referId*)

Reject the REFER request.

Parameters

referId	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

- Int32 [PortSIP.PortSIPLib.enableSendPcmStreamToRemote](#) (Int32 sessionId, Boolean state, Int32 streamSamplesPerSec)
  Enable the SDK to send PCM stream data to remote side from another source instead of microphone.

- Int32 [PortSIP.PortSIPLib.sendPcmStreamToRemote](#) (Int32 sessionId, byte[] data, Int32 dataLength)
  Send the audio stream in PCM format from another source instead of audio device capturing (microphone).

- Int32 [PortSIP.PortSIPLib.enableSendVideoStreamToRemote](#) (Int32 sessionId, Boolean state)
  Enable the SDK send video stream data to remote side from another source instead of camera.

- Int32 [PortSIP.PortSIPLib.sendVideoStreamToRemote](#) (Int32 sessionId, byte[] data, Int32 dataLength, Int32 width, Int32 height)
  Send the video stream to remote side.

- Int32 [PortSIP.PortSIPLib.enableSendScreenStreamToRemote](#) (Int32 sessionId, Boolean state)
  Enable the SDK send Screen stream data to remote side from selected screen source instead of camera.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.enableSendPcmStreamToRemote (Int32 *sessionId*, Boolean *state*, Int32 *streamSamplesPerSec*)

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

Parameters

sessionId	The session ID of call.
state	Set to true to enable the send stream, or false to disable.
streamSamplesPerSec	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

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

Int32 PortSIP.PortSIPLib.sendPcmStreamToRemote (Int32 *sessionId*, byte[] *data*, Int32 *dataLength*)

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

Parameters

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

Returns

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

Remarks

Usually it should be used as below:

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

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

Int32 PortSIP.PortSIPLib.enableSendVideoStreamToRemote (Int32 *sessionId*, Boolean *state*)

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

Parameters

sessionId	The session ID of call.
state	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.

Int32 PortSIP.PortSIPLib.sendVideoStreamToRemote (Int32 *sessionId*, byte[] *data*, Int32 *dataLength*, Int32 *width*, Int32 *height*)

Send the video stream to remote side.

Parameters

sessionId	Session ID of the call conversation.
data	The video stream data. It must be in i420 format.
dataLength	The size of data.
width	The video image width.
height	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 it should be used as below:

```
enableSendVideoStreamToRemote(sessionId, true);
sendVideoStreamToRemote(sessionId, data, dataSize, 352, 288);
```

Int32 PortSIP.PortSIPLib.enableSendScreenStreamToRemote (Int32 *sessionId*, Boolean *state*)

Enable the SDK send Screen stream data to remote side from selected screen source instead of camera.

Parameters

sessionId	The session ID of call.
state	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.

```
enableSendVideoStreamToRemote(sessionId, true);
sendVideoStreamToRemote(sessionId, data, dataSize, 352, 288);
```

RTP packets, Audio stream and video stream callback functions

Functions

- Int32 PortSIP.PortSIPLib.enableAudioStreamCallback (Int32 sessionId, Boolean enable, DIRECTION_MODE direction)
  Enable/disable the audio stream callback.

- Int32 PortSIP.PortSIPLib.enableVideoStreamCallback (Int32 sessionId, DIRECTION_MODE direction)
  Enable/disable the video stream callback.

- Int32 PortSIP.PortSIPLib.enableScreenStreamCallback (Int32 sessionId, DIRECTION_MODE direction)
  Enable/disable the video stream callback.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.enableAudioStreamCallback (Int32 *sessionId*, Boolean *enable*, DIRECTION_MODE *direction*)

Enable/disable the audio stream callback.

Parameters

sessionId	The session ID of call.
enable	Set to true to enable audio stream callback, or false to stop the callback.
direction	The audio stream callback direction.

Type	Description
DIRECTION_SEND	Callback the send audio stream for one channel based on the given sessionId.
DIRECTION_RECV	Callback the received audio stream for one channel based on the given sessionId.
DIRECTION_SEND_RECV	Callback both send & received audio stream for one channel based on the given sessionId.

Returns

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

Remarks

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

Int32 PortSIP.PortSIPLib.enableVideoStreamCallback (Int32   *sessionId*,
DIRECTION_MODE   *direction*)

Enable/disable the video stream callback.

Parameters

callbackObject	The callback object that you passed in can be accessed once callback function triggered.
sessionId	The session ID of call.
direction	The video stream callback direction.

Type	Description
DIRECTION_SEND	Callback the send video stream for one channel based on the given sessionId.
DIRECTION_RECV	Callback the received video stream for one channel based on the given sessionId.
DIRECTION_SEND_RECV	Callback both send & received video stream for one channel based on the given sessionId.

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.

Int32 PortSIP.PortSIPLib.enableScreenStreamCallback (Int32   *sessionId*,
DIRECTION_MODE   *direction*)

Enable/disable the video stream callback.

Parameters

callbackObject	The callback object that you passed in can be accessed once callback function triggered.
sessionId	The session ID of call.
direction	The video stream callback direction.

Type	Description
DIRECTION_SEND	Callback the send video stream for one channel based on the given sessionId.
DIRECTION_RECV	Callback the received video stream for one channel based on the given sessionId.
DIRECTION_SEND_RECV	Callback both send & received video stream for one channel based on the given sessionId.

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

- Int32 PortSIP.PortSIPLib.startRecord (Int32 sessionId, String recordFilePath, String recordFileName, Boolean appendTimestamp, Int32 channels, FILE_FORMAT recordFileFormat, RECORD_MODE audioRecordMode, RECORD_MODE videoRecordMode)
  Start recording the call.

- Int32 PortSIP.PortSIPLib.stopRecord (Int32 sessionId)
  Stop record.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.startRecord (Int32 *sessionId*, String *recordFilePath*, String *recordFileName*, Boolean *appendTimestamp*, Int32 *channels*, FILE_FORMAT *recordFileFormat*, RECORD_MODE *audioRecordMode*, RECORD_MODE *videoRecordMode*)

Start recording the call.

Parameters

sessionId	The session ID of call conversation.
recordFilePath	The file path to which the record file will be saved. It must be existent.
recordFileName	The file name of record file. For example: audiorecord.wav or videorecord.avi.
appendTimestamp	Set to true to append the timestamp to the recording file name.
channels	Set to record file audio channels, 1 - mono 2 - stereo.
recordFileFormat	The file format for the recording.
audioRecordMode	Allow to set audio recording mode. Support to record received and/or sent audio.
videoRecordMode	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.

Int32 PortSIP.PortSIPLib.stopRecord (Int32 *sessionId*)

Stop record.

Parameters

sessionId	The session ID of call conversation.

Returns

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

Play audio and video file to remote functions

Functions

- Int32 [PortSIP.PortSIPLib.startPlayingFileToRemote](#) (Int32 sessionId, String fileName, Boolean loop, Int32 playAudio)
  Play a file to remote party.

- Int32 [PortSIP.PortSIPLib.stopPlayingFileToRemote](#) (Int32 sessionId)
  Stop playing file to remote party.

- Int32 [PortSIP.PortSIPLib.startPlayingFileLocally](#) (String fileUrl, Boolean loop, IntPtr playVideoWindow)
  Play a file to remote party.

- Int32 [PortSIP.PortSIPLib.stopPlayingFileLocally](#) ()
  Stop playing file to locally.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.startPlayingFileToRemote (Int32 *sessionId*, String *fileName*, Boolean *loop*, Int32 *playAudio*)

Play a file to remote party.

Parameters

sessionId	Session ID of the call.
fileUrl	url or file name, such as "/test.mp4","/test.wav".
loop	Set to false to stop playing video file when it is ended, or true to play it repeatedly.
playAudio	0 - Not play file audio. 1 - Play file audio, 2 - Play file audio mix with Mic.

Returns

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

Int32 PortSIP.PortSIPLib.stopPlayingFileToRemote (Int32 *sessionId*)

Stop playing file to remote party.

Parameters

sessionId	Session ID of the call.

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

Int32 PortSIP.PortSIPLib.startPlayingFileLocally (String *fileUrl*, Boolean *loop*, IntPtr *playVideoWindow*)

Play a file to remote party.

Parameters

sessionId	Session ID of the call.
fileUrl	url or file name, such as "/test.mp4","/test.wav".
loop	Set to false to stop playing video file when it is ended, or true to play it repeatedly.
playVideoWindow	The PortSIPVideoRenderView used for displaying the video.

Returns

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

Int32 PortSIP.PortSIPLib.stopPlayingFileLocally ()

Stop playing file to locally.

Returns

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

Conference functions

Functions

- Int32 [PortSIP.PortSIPLib.createAudioConference](#) ()
  Create an audio conference. It will be failed if the existent conference is not ended yet.

- Int32 [PortSIP.PortSIPLib.createVideoConference](#) (IntPtr conferenceVideoWindow, Int32 width, Int32 height, Int32 layout)
  Create a video conference. It will be failed if the existent conference is not ended yet.

- void [PortSIP.PortSIPLib.destroyConference](#) ()
  End the existent conference.

- Int32 [PortSIP.PortSIPLib.setConferenceVideoWindow](#) (IntPtr videoWindow)
  Set the window for a conference that is used to display the received remote video image.

- Int32 [PortSIP.PortSIPLib.joinToConference](#) (Int32 sessionId)
  Join a session into existent conference. If the call is in hold, it will be un-hold automatically.

- Int32 [PortSIP.PortSIPLib.removeFromConference](#) (Int32 sessionId)
  Remove a session from an existent conference.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.createAudioConference ()

Create an audio conference. It will be failed if the existent conference is not ended yet.

Returns

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

Int32 PortSIP.PortSIPLib.createVideoConference (IntPtr *conferenceVideoWindow,* Int32 *width,* Int32 *height,* Int32 *layout*)

Create a video conference. It will be failed if the existent conference is not ended yet.

Parameters

conferenceVideoW indow	The UIView used to display the conference video.

videoResolution	The conference video resolution.
layout	Conference Video layout, default is 0 - Adaptive. 0 - Adaptive(1,3,5,6) 1 - Only Local Video 2 - 2 video,PIP 3 - 2 video, Left and right 4 - 2 video, Up and Down 5 - 3 video 6 - 4 split video 7 - 5 video

Returns

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

Int32 PortSIP.PortSIPLib.setConferenceVideoWindow (IntPtr *videoWindow*)

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

Parameters

videoWindow	The UIView 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.

Int32 PortSIP.PortSIPLib.joinToConference (Int32 *sessionId*)

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

Parameters

sessionId	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.

Int32 PortSIP.PortSIPLib.removeFromConference (Int32 *sessionId*)

Remove a session from an existent conference.

Parameters

sessionId	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

- Int32 PortSIP.PortSIPLib.setAudioRtcpBandwidth (Int32 sessionId, Int32 BitsRR, Int32 BitsRS, Int32 KBitsAS)
  Set the audio RTCP bandwidth parameters to the RFC3556.

- Int32 PortSIP.PortSIPLib.setVideoRtcpBandwidth (Int32 sessionId, Int32 BitsRR, Int32 BitsRS, Int32 KBitsAS)
  Set the video RTCP bandwidth parameters as the RFC3556.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.setAudioRtcpBandwidth (Int32 *sessionId*, Int32 *BitsRR*, Int32 *BitsRS*, Int32 *KBitsAS*)

Set the audio RTCP bandwidth parameters to the RFC3556.

Parameters

sessionId	The session ID of call conversation.
BitsRR	The bits for the RR parameter.
BitsRS	The bits for the RS parameter.
KBitsAS	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.

Int32 PortSIP.PortSIPLib.setVideoRtcpBandwidth (Int32 *sessionId*, Int32 *BitsRR*, Int32 *BitsRS*, Int32 *KBitsAS*)

Set the video RTCP bandwidth parameters as the RFC3556.

Parameters

sessionId	The session ID of call conversation.
BitsRR	The bits for the RR parameter.
BitsRS	The bits for the RS parameter.
KBitsAS	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.

RTP statistics functions

Functions

- Int32 PortSIP.PortSIPLib.getStatistics (Int32 sessionId)
  Obtain the statistics of channel. the event onStatistics will be triggered.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.getStatistics (Int32 *sessionId*)

Obtain the statistics of channel. the event onStatistics will be triggered.

Parameters

sessionId	The session ID of call conversation.

Returns

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

Audio effect functions

Functions

- void PortSIP.PortSIPLib.enableVAD (Boolean state)
  Enable/disable Voice Activity Detection (VAD).

- void PortSIP.PortSIPLib.enableAEC (Boolean state)
  Enable/disable AEC (Acoustic Echo Cancellation).

- void PortSIP.PortSIPLib.enableCNG (Boolean state)
  Enable/disable Comfort Noise Generator (CNG).

- void PortSIP.PortSIPLib.enableAGC (Boolean state)
  Enable/disable Automatic Gain Control (AGC).

- void PortSIP.PortSIPLib.enableANS (Boolean state)
  Enable/disable Audio Noise Suppression (ANS).

- Int32 PortSIP.PortSIPLib.enableAudioQos (Boolean state)
  Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for audio channel.

- Int32 PortSIP.PortSIPLib.enableVideoQos (Boolean state)
  Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for video channel.

- Int32 PortSIP.PortSIPLib.setVideoMTU (Int32 mtu)
  Set the MTU size for video RTP packet.

Detailed Description

Function Documentation

void PortSIP.PortSIPLib.enableVAD (Boolean *state*)

Enable/disable Voice Activity Detection (VAD).

Parameters

state	Set to true to enable VAD, or false to disable.

void PortSIP.PortSIPLib.enableAEC (Boolean *state*)

Enable/disable AEC (Acoustic Echo Cancellation).

Parameters

state	Set it to true to enable AEC, or false to disable.

void PortSIP.PortSIPLib.enableCNG (Boolean *state*)

Enable/disable Comfort Noise Generator (CNG).

Parameters

state	Set it to true to enable CNG, or false to disable.

void PortSIP.PortSIPLib.enableAGC (Boolean *state*)

Enable/disable Automatic Gain Control (AGC).

Parameters

state	Set it to true to enable AGC, or false to disable.

void PortSIP.PortSIPLib.enableANS (Boolean *state*)

Enable/disable Audio Noise Suppression (ANS).

Parameters

state	Set it to true to enable ANS, or false to disable.

Int32 PortSIP.PortSIPLib.enableAudioQos (Boolean *state*)

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

Parameters

state	Set to true to enable audio QoS, and DSCP value will be 46; or false 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.

Int32 PortSIP.PortSIPLib.enableVideoQos (Boolean *state*)

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

Parameters

state	Set as true to enable video QoS and DSCP value will be 34; or false 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.

Int32 PortSIP.PortSIPLib.setVideoMTU (Int32 *mtu*)

Set the MTU size for video RTP packet.

Parameters

mtu	Set MTU value. Allow value ranges (512-65507). Other value will be modified to the default 1450.

Returns

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

Send OPTIONS/INFO/MESSAGE functions

Functions

- Int32 PortSIP.PortSIPLib.sendOptions (String to, String sdp)
  Send OPTIONS message.

- Int32 PortSIP.PortSIPLib.sendInfo (Int32 sessionId, String mimeType, String subMimeType, String infoContents)
  Send a INFO message to remote side in a call.

- Int32 PortSIP.PortSIPLib.sendSubscription (String to, String eventName)
  Send a SUBSCRIBE message to subscribe an event.

- Int32 PortSIP.PortSIPLib.terminateSubscription (Int32 subscribeId)
  Terminate the given subscription.

- Int32 PortSIP.PortSIPLib.sendMessage (Int32 sessionId, String mimeType, String subMimeType, byte[] message, Int32 messageLength)
  Send a MESSAGE message to remote side in dialog.

- Int32 PortSIP.PortSIPLib.sendOutOfDialogMessage (String to, String mimeType, String subMimeType, Boolean isSMS, byte[] message, Int32 messageLength)
  Send an out of dialog MESSAGE message to remote side.

- Int32 PortSIP.PortSIPLib.setDefaultSubscriptionTime (Int32 secs)
  Set the default expiration time to be used when creating a subscription.

- Int32 PortSIP.PortSIPLib.setDefaultPublicationTime (Int32 secs)
  Set the default expiration time to be used when creating a publication.

- Int32 PortSIP.PortSIPLib.setPresenceMode (Int32 mode)
  Indicate the SDK uses the P2P mode for presence or presence agent mode.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.sendOptions (String *to*, String *sdp*)

Send OPTIONS message.

Parameters

to	The recipient of OPTIONS message.

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

Returns

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

Int32 PortSIP.PortSIPLib.sendInfo (Int32 *sessionId*, String *mimeType*, String *subMimeType*, String *infoContents*)

Send a INFO message to remote side in a call.

Parameters

sessionId	The session ID of call.
mimeType	The mime type of INFO message.
subMimeType	The sub mime type of INFO message.
infoContents	The contents that is sent with INFO message.

Returns

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

Int32 PortSIP.PortSIPLib.sendSubscription (String *to*, String *eventName*)

Send a SUBSCRIBE message to subscribe an event.

Parameters

to	The user/extension to be subscribed.
eventName	The event name to be subscribed.

Returns

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

Remarks

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

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

Int32 PortSIP.PortSIPLib.terminateSubscription (Int32 *subscribeId*)

Terminate the given subscription.

Parameters

subscribeId	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);
```

Int32 PortSIP.PortSIPLib.sendMessage (Int32 *sessionId*, String *mimeType*, String *subMimeType*, byte[] *message*, Int32 *messageLength*)

Send a MESSAGE message to remote side in dialog.

Parameters

sessionId	The session ID of the call.
mimeType	The mime type of MESSAGE message.
subMimeType	The sub mime type of MESSAGE message.
message	The contents which is sent with MESSAGE message. Binary data allowed.
messageLength	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 text in other languages, please use the UTF-8 to encode the message before sending.

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

Example 2: send a binary message.

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

Int32 PortSIP.PortSIPLib.sendOutOfDialogMessage (String *to*, String *mimeType*, String *subMimeType*, Boolean *isSMS*, byte[] *message*, Int32 *messageLength*)

Send an out of dialog MESSAGE message to remote side.

Parameters

to	The message recipient, such as sip:receiver@portsip.com
mimeType	The mime type of MESSAGE message.
subMimeType	The sub mime type of MESSAGE message. @isSMS isSMS Set to YES to specify "messagetype=SMS" in the To line, or NO to disable.
message	The contents which is sent with MESSAGE message. Binary data allowed.
messageLength	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 the UTF-8 to encode the message before sending.

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

Example 2: send a binary message.

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

Int32 PortSIP.PortSIPLib.setDefaultSubscriptionTime (Int32 *secs*)

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

Parameters

secs	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.

Int32 PortSIP.PortSIPLib.setDefaultPublicationTime (Int32 *secs*)

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

Parameters

secs	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.

Int32 PortSIP.PortSIPLib.setPresenceMode (Int32 *mode*)

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

Parameters

mode	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 and PortSIP PBX support this feature. For more details please visit: https://www.portsip.com/portsip-pbx

Presence functions

Functions

- Int32 PortSIP.PortSIPLib.presenceSubscribe (String to, String subject)
  Send a SUBSCRIBE message for subscribing the contact's presence status.

- Int32 PortSIP.PortSIPLib.presenceTerminateSubscribe (Int32 subscribeId)
  Terminate the given presence subscription.

- Int32 PortSIP.PortSIPLib.presenceRejectSubscribe (Int32 subscribeId)
  Reject a presence SUBSCRIBE request which is received from contact.

- Int32 PortSIP.PortSIPLib.presenceAcceptSubscribe (Int32 subscribeId)
  Accept the presence SUBSCRIBE request which is received from contact.

- Int32 PortSIP.PortSIPLib.setPresenceStatus (Int32 subscribeId, String stateText)
  Set the presence status.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.presenceSubscribe (String *to*, String *subject*)

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

Parameters

to	The target contact. It must be like sip:contact001@sip.portsip.com.
subject	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 value subscribeId. If the function fails, it will return a specific error code.

Int32 PortSIP.PortSIPLib.presenceTerminateSubscribe (Int32 *subscribeId*)

Terminate the given presence subscription.

Parameters

subscribeId	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.

Int32 PortSIP.PortSIPLib.presenceRejectSubscribe (Int32 *subscribeId*)

Reject a presence SUBSCRIBE request which is received from contact.

Parameters

subscribeId	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.

Int32 PortSIP.PortSIPLib.presenceAcceptSubscribe (Int32 *subscribeId*)

Accept the presence SUBSCRIBE request which is received from contact.

Parameters

subscribeId	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.

Int32 PortSIP.PortSIPLib.setPresenceStatus (Int32 *subscribeId*, String *stateText*)

Set the presence status.

Parameters

subscribeId	Subscription ID.
stateText	The state text of presence online. For example: "I'm here". If you want to appear as offline to others, please pass the "Offline" to "statusText" parameter.

Returns

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

Remarks

With P2P presence mode, when receiving a SUBSCRIBE request from contact, the event onPresenceRecvSubscribe will be triggered. The event includes the subscription ID. This

function will cause the SDK sending a NOTIFY message to update your presence status, and you must pass the correct subscribeId.

With presence agent mode, this function will cause the SDK to send a PUBLISH message to update your presence status, and you must pass 0 to the "subscribeId" parameter.

Device Manage functions.

Functions

- Int32 PortSIP.PortSIPLib.getNumOfRecordingDevices ()
  Gets the count of audio devices available for audio recording.

- Int32 PortSIP.PortSIPLib.getNumOfPlayoutDevices ()
  Gets the number of audio devices available for audio playout.

- Int32 PortSIP.PortSIPLib.getRecordingDeviceName (Int32 deviceIndex, StringBuilder nameUTF8, Int32 nameUTF8Length)
  Gets the name of a specific recording device given by an index.

- Int32 PortSIP.PortSIPLib.getPlayoutDeviceName (Int32 deviceIndex, StringBuilder nameUTF8, Int32 nameUTF8Length)
  Get the name of a specific playout device given by an index.

- Int32 PortSIP.PortSIPLib.setSpeakerVolume (Int32 volume)
  Set the speaker volume level.

- Int32 PortSIP.PortSIPLib.getSpeakerVolume ()
  Gets the speaker volume level.

- Int32 PortSIP.PortSIPLib.setMicVolume (Int32 volume)
  Sets the microphone volume level.

- Int32 PortSIP.PortSIPLib.getMicVolume ()
  Retrieves the current microphone volume.

- Int32 PortSIP.PortSIPLib.getScreenSourceCount ()
  Retrieves the current number of screen.

- Int32 PortSIP.PortSIPLib.getScreenSourceTitle (Int32 deviceIndex, StringBuilder nameUTF8, Int32 nameUTF8Length)
  Retrieves the current screen title .

- Int32 PortSIP.PortSIPLib.selectScreenSource (Int32 nDeviceIndex)
  Sets the Screen to share .

- Int32 PortSIP.PortSIPLib.SetScreenFrameRate (Int32 nFrameRate)
  Sets the Screen video framerate .

- Int32 PortSIP.PortSIPLib.setScreenVideoWindow (Int32 sessionId, IntPtr screenVideoWindow)
  Set the window for a session that is used to display the received screen video .

- void PortSIP.PortSIPLib.audioPlayLoopbackTest (Boolean enable)
  Use it for the audio device loop back test.

- Int32 [PortSIP.PortSIPLib.getNumOfVideoCaptureDevices](#) ()
  Get the number of available capturing devices.

- Int32 [PortSIP.PortSIPLib.getVideoCaptureDeviceName](#) (Int32 deviceIndex, StringBuilder uniqueIdUTF8, Int32 uniqueIdUTF8Length, StringBuilder deviceNameUTF8, Int32 deviceNameUTF8Length)
  Get the name of a specific video capture device given by an index.

- Int32 [PortSIP.PortSIPLib.showVideoCaptureSettingsDialogBox](#) (String uniqueIdUTF8, Int32 uniqueIdUTF8Length, String dialogTitle, IntPtr parentWindow, Int32 x, Int32 y)
  Display the capture device property dialog box for the specified capture device.

Detailed Description

Function Documentation

Int32 PortSIP.PortSIPLib.getNumOfRecordingDevices ()

Gets the count of audio devices available for audio recording.

Returns

It will return the count of recording devices. If the function fails, it will return a specific error code less than 0.

Int32 PortSIP.PortSIPLib.getNumOfPlayoutDevices ()

Gets the number of audio devices available for audio playout.

Returns

It will return the count of playout devices. If the function fails, it will return a specific error code less than 0.

Int32 PortSIP.PortSIPLib.getRecordingDeviceName (Int32 *deviceIndex*, StringBuilder *nameUTF8*, Int32 *nameUTF8Length*)

Gets the name of a specific recording device given by an index.

Parameters

deviceIndex	Device index (0, 1, 2, ..., N-1), where N is given by getNumOfRecordingDevices (). Also -1 is a valid value and will return the name of the default recording device.
nameUTF8	A character buffer to which the device name will be copied as a null-terminated string in UTF-8 format.

nameUTF8Length	The size of nameUTF8 buffer. It cannot be less than 128.

Returns

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

Int32 PortSIP.PortSIPLib.getPlayoutDeviceName (Int32   *deviceIndex*, StringBuilder *nameUTF8*, Int32   *nameUTF8Length*)

Get the name of a specific playout device given by an index.

Parameters

deviceIndex	
deviceIndex	Device index (0, 1, 2, ..., N-1), where N is given by getNumOfRecordingDevices (). Also -1 is a valid value and will return the name of the default recording device.
nameUTF8	A character buffer to which the device name will be copied as a null-terminated string in UTF-8 format.
nameUTF8Length	The size of nameUTF8 buffer. It cannot be less than 128.

Returns

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

Int32 PortSIP.PortSIPLib.setSpeakerVolume (Int32   *volume*)

Set the speaker volume level.

Parameters

volume	Volume level of speaker. Valid value ranges 0 - 255.

Returns

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

Int32 PortSIP.PortSIPLib.getSpeakerVolume ()

Gets the speaker volume level.

Returns

If the function succeeds, it will return the speaker volume with valid range 0 - 255. If the function fails, it will return a specific error code.

Int32 PortSIP.PortSIPLib.setMicVolume (Int32   *volume*)

Sets the microphone volume level.

Parameters

volume	The microphone volume level. Valid value ranges 0 - 255.

Returns

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

Int32 PortSIP.PortSIPLib.getMicVolume ()

Retrieves the current microphone volume.

Returns

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

Int32 PortSIP.PortSIPLib.getScreenSourceCount ()

Retrieves the current number of screen.

Returns

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

Int32 PortSIP.PortSIPLib.getScreenSourceTitle (Int32 *deviceIndex*, StringBuilder *nameUTF8*, Int32 *nameUTF8Length*)

Retrieves the current screen title .

Returns

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

Int32 PortSIP.PortSIPLib.selectScreenSource (Int32 *nDeviceIndex*)

Sets the Screen to share .

Returns

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

Int32 PortSIP.PortSIPLib.SetScreenFrameRate (Int32 *nFrameRate*)

Sets the Screen video framerate .

Returns

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

Int32 PortSIP.PortSIPLib.setScreenVideoWindow (Int32 *sessionId*, IntPtr *screenVideoWindow*)

Set the window for a session that is used to display the received screen video .

Parameters

sessionId	The session ID of the call.
remoteVideoWindow	The window to display 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.

void PortSIP.PortSIPLib.audioPlayLoopbackTest (Boolean *enable*)

Use it for the audio device loop back test.

Parameters

enable	Set to true to start audio look back test; or fase to stop.

Int32 PortSIP.PortSIPLib.getNumOfVideoCaptureDevices ()

Get the number of available capturing devices.

Returns

It will return the count of video capturing devices. If it fails, it will return a specific error code less than 0.

Int32 PortSIP.PortSIPLib.getVideoCaptureDeviceName (Int32 *deviceIndex*, StringBuilder *uniqueIdUTF8*, Int32 *uniqueIdUTF8Length*, StringBuilder *deviceNameUTF8*, Int32 *deviceNameUTF8Length*)

Get the name of a specific video capture device given by an index.

Parameters

deviceIndex	Device index (0, 1, 2, ..., N-1), where N is given by getNumOfVideoCaptureDevices (). Also -1 is a valid value and will return the name of the default capturing device.
uniqueIdUTF8	Unique identifier of the capturing device.
uniqueIdUTF8Length	Size in bytes of uniqueIdUTF8.
deviceNameUTF8	A character buffer to which the device name will be copied as a null-terminated string in UTF-8 format.
deviceNameUTF8Length	The size of nameUTF8 buffer. It cannot be less than 128.

Returns

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

Int32 PortSIP.PortSIPLib.showVideoCaptureSettingsDialogBox (String *uniqueIdUTF8*, Int32 *uniqueIdUTF8Length*, String *dialogTitle*, IntPtr *parentWindow*, Int32 *x*, Int32 *y*)

Display the capture device property dialog box for the specified capture device.

Parameters

uniqueIdUTF8	Unique identifier of the capture device.
uniqueIdUTF8Length	Size in bytes of uniqueIdUTF8.
dialogTitle	The title of the video settings dialog.
parentWindow	Parent window used for the dialog box. It should originally be a HWND.
x	Horizontal position for the dialog relative to the parent window, in pixels.
y	Vertical position for the dialog relative to the parent window, in pixels.

Returns

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

SDK Callback events

Modules

- [Register events](#)
- [Call events](#)
- [Refer events](#)
- [Signaling events](#)
- [MWI events](#)
- [DTMF events](#)
- [INFO/OPTIONS message events](#)
- [Presence 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

- Int32 PortSIP.SIPCallbackEvents.onRegisterSuccess (String statusText, Int32 statusCode, StringBuilder sipMessage)
- Int32 PortSIP.SIPCallbackEvents.onRegisterFailure (String statusText, Int32 statusCode, StringBuilder sipMessage)

Detailed Description

Register events

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onRegisterSuccess (String *statusText*, Int32 *statusCode*, StringBuilder *sipMessage*)

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

Parameters

callbackIndex	This is a callback index passed in when creating the SDK library.
callbackObject	This is a callback object passed in when creating the SDK library.
statusText	The status text.
statusCode	The status code.
sipMessage	The SIP message received.

Int32 PortSIP.SIPCallbackEvents.onRegisterFailure (String *statusText*, Int32 *statusCode*, StringBuilder *sipMessage*)

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

Parameters

statusText	The status text.
statusCode	The status code.
sipMessage	The SIP message received.

Call events

Functions

- Int32 [PortSIP.SIPCallbackEvents.onInviteIncoming](#) (Int32 sessionId, String callerDisplayName, String caller, String calleeDisplayName, String callee, String audioCodecNames, String videoCodecNames, Boolean existsAudio, Boolean existsVideo, StringBuilder sipMessage)
- Int32 [PortSIP.SIPCallbackEvents.onInviteTrying](#) (Int32 sessionId)
- Int32 [PortSIP.SIPCallbackEvents.onInviteSessionProgress](#) (Int32 sessionId, String audioCodecNames, String videoCodecNames, Boolean existsEarlyMedia, Boolean existsAudio, Boolean existsVideo, StringBuilder sipMessage)
- Int32 [PortSIP.SIPCallbackEvents.onInviteRinging](#) (Int32 sessionId, String statusText, Int32 statusCode, StringBuilder sipMessage)
- Int32 [PortSIP.SIPCallbackEvents.onInviteAnswered](#) (Int32 sessionId, String callerDisplayName, String caller, String calleeDisplayName, String callee, String audioCodecNames, String videoCodecNames, Boolean existsAudio, Boolean existsVideo, StringBuilder sipMessage)
- Int32 [PortSIP.SIPCallbackEvents.onInviteFailure](#) (Int32 sessionId, String callerDisplayName, String caller, String calleeDisplayName, String callee, String reason, Int32 code, StringBuilder sipMessage)
- Int32 [PortSIP.SIPCallbackEvents.onInviteUpdated](#) (Int32 sessionId, String audioCodecNames, String videoCodecNames, Boolean existsAudio, Boolean existsVideo, Boolean existsScreen, StringBuilder sipMessage)
- Int32 [PortSIP.SIPCallbackEvents.onInviteConnected](#) (Int32 sessionId)
- Int32 [PortSIP.SIPCallbackEvents.onInviteBeginingForward](#) (String forwardTo)
- Int32 [PortSIP.SIPCallbackEvents.onInviteClosed](#) (Int32 sessionId)
- Int32 [PortSIP.SIPCallbackEvents.onDialogStateUpdated](#) (String BLFMonitoredUri, String BLFDialogState, String BLFDialogId, String BLFDialogDirection)
- Int32 [PortSIP.SIPCallbackEvents.onRemoteHold](#) (Int32 sessionId)
- Int32 [PortSIP.SIPCallbackEvents.onRemoteUnHold](#) (Int32 sessionId, String audioCodecNames, String videoCodecNames, Boolean existsAudio, Boolean existsVideo)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onInviteIncoming (Int32 *sessionId*, String *callerDisplayName*, String *caller*, String *calleeDisplayName*, String *callee*, String *audioCodecNames*, String *videoCodecNames*, Boolean *existsAudio*, Boolean *existsVideo*, StringBuilder *sipMessage*)

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

Parameters

sessionId	The session ID of the call.
callerDisplayName	The display name of caller
caller	The caller.
calleeDisplayName	The display name of callee.
callee	The callee.
audioCodecNames	The matched audio codecs. It's separated by "#" if there are more than one codecs.
videoCodecNames	The matched video codecs. It's separated by "#" if there are more than one codecs.

existsAudio	If it's true, it indicates that this call includes the audio.
existsVideo	If it's true, it indicates that this call includes the video.
sipMessage	The SIP message received.

Int32 PortSIP.SIPCallbackEvents.onInviteTrying (Int32 *sessionId*)

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

Parameters

sessionId	The session ID of the call.

Int32 PortSIP.SIPCallbackEvents.onInviteSessionProgress (Int32 *sessionId*, String *audioCodecNames*, String *videoCodecNames*, Boolean *existsEarlyMedia*, Boolean *existsAudio*, Boolean *existsVideo*, StringBuilder *sipMessage*)

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

Parameters

sessionId	The session ID of the call.
audioCodecNames	The matched audio codecs. It's separated by "#" if there are more than one codecs.
videoCodecNames	The matched video codecs. It's separated by "#" if there are more than one codecs.
existsEarlyMedia	If it's true, it indicates that the call has early media.
existsAudio	If it's true, it indicates that this call includes the audio.
existsVideo	If it's true, it indicates that this call includes the video.
sipMessage	The SIP message received.

Int32 PortSIP.SIPCallbackEvents.onInviteRinging (Int32 *sessionId*, String *statusText*, Int32 *statusCode*, StringBuilder *sipMessage*)

If the outgoing call was ringing, this event would be triggered.

Parameters

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

Int32 PortSIP.SIPCallbackEvents.onInviteAnswered (Int32 *sessionId*, String *callerDisplayName*, String *caller*, String *calleeDisplayName*, String *callee*, String *audioCodecNames*, String *videoCodecNames*, Boolean *existsAudio*, Boolean *existsVideo*, StringBuilder *sipMessage*)

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

Parameters

sessionId	The session ID of the call.
callerDisplayName	The display name of caller
caller	The caller.
calleeDisplayName	The display name of callee.
callee	The callee.
audioCodecNames	The matched audio codecs. It's separated by "#" if there are more than one codecs.
videoCodecNames	The matched video codecs. It's separated by "#" if there are more than one codecs.
existsAudio	If it's true, it indicates that this call includes the audio.
existsVideo	If it's true, it indicates that this call includes the video.

sipMessage	The SIP message received.

Int32 PortSIP.SIPCallbackEvents.onInviteFailure (Int32 *sessionId*, String *callerDisplayName*, String *caller*, String *calleeDisplayName*, String *callee*, String *reason*, Int32 *code*, StringBuilder *sipMessage*)

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

Parameters

sessionId	The session ID of the call.
callerDisplayName	The display name of caller
caller	The caller.
calleeDisplayName	The display name of callee.
callee	The callee.
reason	The failure reason.
code	The failure code.
sipMessage	The SIP message received.

Int32 PortSIP.SIPCallbackEvents.onInviteUpdated (Int32 *sessionId*, String *audioCodecNames*, String *videoCodecNames*, Boolean *existsAudio*, Boolean *existsVideo*, Boolean *existsScreen*, StringBuilder *sipMessage*)

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

Parameters

sessionId	The session ID of the call.
audioCodecNames	The matched audio codecs. It's separated by "#" if there are more than one codecs.
videoCodecNames	The matched video codecs. It's separated by "#" if there are more than one codecs.
existsAudio	If it's true, it indicates that this call includes the audio.
existsVideo	If it's true, it indicates that this call includes the video.
sipMessage	The SIP message received.

Int32 PortSIP.SIPCallbackEvents.onInviteConnected (Int32 *sessionId*)

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

Parameters

sessionId	The session ID of the call.

Int32 PortSIP.SIPCallbackEvents.onInviteBeginingForward (String *forwardTo*)

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

Parameters

forwardTo	The forwarding target SIP URI.

Int32 PortSIP.SIPCallbackEvents.onInviteClosed (Int32 *sessionId*)

This event is triggered once remote side closes the call.

Parameters

sessionId	The session ID of the call.

Int32 PortSIP.SIPCallbackEvents.onDialogStateUpdated (String *BLFMonitoredUri*, String *BLFDialogState*, String *BLFDialogId*, String *BLFDialogDirection*)

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

Parameters

BLFMonitoredUri	the monitored user's URI
BLFDialogState	- the status of the call
BLFDialogId	- the id of the call
BLFDialogDirecti on	- the direction of the call

Int32 PortSIP.SIPCallbackEvents.onRemoteHold (Int32 *sessionId*)

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

Parameters

sessionId	The session ID of the call.

Int32 PortSIP.SIPCallbackEvents.onRemoteUnHold (Int32 *sessionId*, String *audioCodecNames*, String *videoCodecNames*, Boolean *existsAudio*, Boolean *existsVideo*)

If the remote side un-hold the call, this event would be triggered.

Parameters

sessionId	The session ID of the call.
audioCodecNames	The matched audio codecs. It's separated by "#" if there are more than one codecs.
videoCodecNames	The matched video codecs. It's separated by "#" if there are more than one codecs.
existsAudio	If it's true, it indicates that this call includes the audio.
existsVideo	If it's true, it indicates that this call includes the video.

Refer events

Functions

- Int32 [PortSIP.SIPCallbackEvents.onReceivedRefer](#) (Int32 sessionId, Int32 referId, String to, String from, StringBuilder referSipMessage)
- Int32 [PortSIP.SIPCallbackEvents.onReferAccepted](#) (Int32 sessionId)
- Int32 [PortSIP.SIPCallbackEvents.onReferRejected](#) (Int32 sessionId, String reason, Int32 code)
- Int32 [PortSIP.SIPCallbackEvents.onTransferTrying](#) (Int32 sessionId)
- Int32 [PortSIP.SIPCallbackEvents.onTransferRinging](#) (Int32 sessionId)
- Int32 [PortSIP.SIPCallbackEvents.onACTVTransferSuccess](#) (Int32 sessionId)
- Int32 [PortSIP.SIPCallbackEvents.onACTVTransferFailure](#) (Int32 sessionId, String reason, Int32 code)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onReceivedRefer (Int32 *sessionId*, Int32 *referId*, String *to*, String *from*, StringBuilder *referSipMessage*)

This event will be triggered once receiving a REFER message.

Parameters

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

Int32 PortSIP.SIPCallbackEvents.onReferAccepted (Int32 *sessionId*)

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

Parameters

sessionId	The session ID of the call.

Int32 PortSIP.SIPCallbackEvents.onReferRejected (Int32 *sessionId*, String *reason*, Int32 *code*)

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

Parameters

sessionId	The session ID of the call.
reason	Reject reason.
code	Reject code.

Int32 PortSIP.SIPCallbackEvents.onTransferTrying (Int32 *sessionId*)

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

Parameters

sessionId	The session ID of the call.

Int32 PortSIP.SIPCallbackEvents.onTransferRinging (Int32 *sessionId*)

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

Parameters

sessionId	The session ID of the call.

Int32 PortSIP.SIPCallbackEvents.onACTVTransferSuccess (Int32 *sessionId*)

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

Parameters

sessionId	The session ID of the call.

Int32 PortSIP.SIPCallbackEvents.onACTVTransferFailure (Int32 *sessionId*, String *reason*, Int32 *code*)

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

Parameters

sessionId	The session ID of the call.
reason	The error reason.
code	The error code.

Signaling events

Functions

- Int32 [PortSIP.SIPCallbackEvents.onReceivedSignaling](#) (Int32 sessionId, StringBuilder signaling)
- Int32 [PortSIP.SIPCallbackEvents.onSendingSignaling](#) (Int32 sessionId, StringBuilder signaling)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onReceivedSignaling (Int32 *sessionId*, StringBuilder *signaling*)

This event will be triggered when receiving an SIP message.

Parameters

sessionId	The session ID of the call.
signaling	The SIP message received.

Int32 PortSIP.SIPCallbackEvents.onSendingSignaling (Int32 *sessionId*, StringBuilder *signaling*)

This event will be triggered when a SIP message sent.

Parameters

sessionId	The session ID of the call.
signaling	The SIP message sent.

MWI events

Functions

- Int32 [PortSIP.SIPCallbackEvents.onWaitingVoiceMessage](#) (String messageAccount, Int32 urgentNewMessageCount, Int32 urgentOldMessageCount, Int32 newMessageCount, Int32 oldMessageCount)
- Int32 [PortSIP.SIPCallbackEvents.onWaitingFaxMessage](#) (String messageAccount, Int32 urgentNewMessageCount, Int32 urgentOldMessageCount, Int32 newMessageCount, Int32 oldMessageCount)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onWaitingVoiceMessage (String *messageAccount*, Int32 *urgentNewMessageCount*, Int32 *urgentOldMessageCount*, Int32 *newMessageCount*, Int32 *oldMessageCount*)

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

Parameters

messageAccount	Voice message account
urgentNewMessageCount	Urgent new message count.
urgentOldMessageCount	Urgent old message count.
newMessageCount	New message count.
oldMessageCount	Old message count.

Int32 PortSIP.SIPCallbackEvents.onWaitingFaxMessage (String *messageAccount*, Int32 *urgentNewMessageCount*, Int32 *urgentOldMessageCount*, Int32 *newMessageCount*, Int32 *oldMessageCount*)

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

Parameters

messageAccount	Fax message account
urgentNewMessageCount	Urgent new message count.
urgentOldMessageCount	Urgent old message count.
newMessageCount	New message count.
oldMessageCount	Old message count.

DTMF events

Functions

● Int32 PortSIP.SIPCallbackEvents.onRecvDtmfTone (Int32 sessionId, Int32 tone)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onRecvDtmfTone (Int32 *sessionId*, Int32 *tone*)

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

Parameters

sessionId	The session ID of the call.
tone	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

- Int32 PortSIP.SIPCallbackEvents.onRecvOptions (StringBuilder optionsMessage)
- Int32 PortSIP.SIPCallbackEvents.onRecvInfo (StringBuilder infoMessage)
- Int32 PortSIP.SIPCallbackEvents.onRecvNotifyOfSubscription (Int32 subscribeId, StringBuilder notifyMsg, byte[] contentData, Int32 contentLenght)
- Int32 PortSIP.SIPCallbackEvents.onSubscriptionFailure (Int32 subscribeId, Int32 statusCode)
- Int32 PortSIP.SIPCallbackEvents.onSubscriptionTerminated (Int32 subscribeId)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onRecvOptions (StringBuilder *optionsMessage*)

This event will be triggered when receiving the OPTIONS message.

Parameters

optionsMessage	The received whole OPTIONS message in text format.

Int32 PortSIP.SIPCallbackEvents.onRecvInfo (StringBuilder *infoMessage*)

This event will be triggered when receiving the INFO message.

Parameters

infoMessage	The received whole INFO message in text format.

Int32 PortSIP.SIPCallbackEvents.onRecvNotifyOfSubscription (Int32 *subscribeId*, StringBuilder *notifyMsg*, byte[] *contentData*, Int32 *contentLenght*)

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

Parameters

subscribeId	The ID of SUBSCRIBE request.
notifyMessage	The received INFO message in text format.
contentData	The received message body. It's can be either text or binary data.
contentLenght	The length of "messageData".

Int32 PortSIP.SIPCallbackEvents.onSubscriptionFailure (Int32 *subscribeId*, Int32 *statusCode*)

This event will be triggered on sending SUBSCRIBE failure.

Parameters

subscribeId	The ID of SUBSCRIBE request.
statusCode	The status code.

Int32 PortSIP.SIPCallbackEvents.onSubscriptionTerminated (Int32 *subscribeId*)

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

Parameters

subscribeId	The ID of SUBSCRIBE request.

Presence events

Functions

- Int32 [PortSIP.SIPCallbackEvents.onPresenceRecvSubscribe](#) (Int32 subscribeId, String fromDisplayName, String from, String subject)
- Int32 [PortSIP.SIPCallbackEvents.onPresenceOnline](#) (String fromDisplayName, String from, String stateText)
- Int32 [PortSIP.SIPCallbackEvents.onPresenceOffline](#) (String fromDisplayName, String from)
- Int32 [PortSIP.SIPCallbackEvents.onRecvMessage](#) (Int32 sessionId, String mimeType, String subMimeType, byte[] messageData, Int32 messageDataLength)
- Int32 [PortSIP.SIPCallbackEvents.onRecvOutOfDialogMessage](#) (String fromDisplayName, String from, String toDisplayName, String to, String mimeType, String subMimeType, byte[] messageData, Int32 messageDataLength)
- Int32 [PortSIP.SIPCallbackEvents.onSendMessageSuccess](#) (Int32 sessionId, Int32 messageId)
- Int32 [PortSIP.SIPCallbackEvents.onSendMessageFailure](#) (Int32 sessionId, Int32 messageId, String reason, Int32 code)
- Int32 [PortSIP.SIPCallbackEvents.onSendOutOfDialogMessageSuccess](#) (Int32 messageId, String fromDisplayName, String from, String toDisplayName, String to)
- Int32 [PortSIP.SIPCallbackEvents.onSendOutOfDialogMessageFailure](#) (Int32 messageId, String fromDisplayName, String from, String toDisplayName, String to, String reason, Int32 code)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onPresenceRecvSubscribe (Int32 *subscribeId*, String *fromDisplayName,* String *from*, String *subject*)

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

Parameters

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

Int32 PortSIP.SIPCallbackEvents.onPresenceOnline (String *fromDisplayName,* String *from*, String *stateText*)

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

Parameters

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

Int32 PortSIP.SIPCallbackEvents.onPresenceOffline (String *fromDisplayName,* String *from*)

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

Parameters

fromDisplayName	The display name of contact.
from	The contact who sends the SUBSCRIBE request

Int32 PortSIP.SIPCallbackEvents.onRecvMessage (Int32   *sessionId*, String   *mimeType*, String   *subMimeType*, byte[]   *messageData*, Int32   *messageDataLength*)

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

Parameters

sessionId	The session ID of the call.
mimeType	The message mime type.
subMimeType	The message sub mime type.
messageData	The received message body. It can be text or binary data.
messageDataLength	The length of "messageData".

Int32 PortSIP.SIPCallbackEvents.onRecvOutOfDialogMessage (String   *fromDisplayName*, String   *from*, String   *toDisplayName*, String   *to*, String   *mimeType*, String   *subMimeType*, byte[]   *messageData*, Int32   *messageDataLength*)

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

Parameters

fromDisplayName	The display name of sender.
from	The message sender.
toDisplayName	The display name of recipient.
to	The recipient.
mimeType	The message mime type.
subMimeType	The message sub mime type.
messageData	The received message body. It can be text or binary data.
messageDataLength	The length of "messageData".

Int32 PortSIP.SIPCallbackEvents.onSendMessageSuccess (Int32   *sessionId*, Int32   *messageId*)

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

Parameters

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

Int32 PortSIP.SIPCallbackEvents.onSendMessageFailure (Int32   *sessionId*, Int32   *messageId*, String   *reason*, Int32   *code*)

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

Parameters

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

Int32 PortSIP.SIPCallbackEvents.onSendOutOfDialogMessageSuccess (Int32   *messageId*, String   *fromDisplayName*, String   *from*, String   *toDisplayName*, String   *to*)

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

Parameters

messageId	The message ID. It's equal to the return value of SendOutOfDialogMessage function.
fromDisplayName	The display name of message sender.
from	The message sender.

toDisplayName	The display name of message recipient.
to	The message receiver.

Int32 PortSIP.SIPCallbackEvents.onSendOutOfDialogMessageFailure (Int32 *messageId***, String** *fromDisplayName***, String** *from***, String** *toDisplayName***, String** *to***, String** *reason***, Int32** *code***)**

If the message was sent failure out of dialog, this event will be triggered.

Parameters

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

Play audio and video file finished events

Functions

- Int32 PortSIP.SIPCallbackEvents.onPlayFileFinished (Int32 sessionId, String fileName)
- Int32 PortSIP.SIPCallbackEvents.onStatistics (Int32 sessionId, String stat)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onPlayFileFinished (Int32 *sessionId*, String *fileName*)

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

Parameters

sessionId	The session ID of the call.
fileName	The name of the file played.

Int32 PortSIP.SIPCallbackEvents.onStatistics (Int32 *sessionId*, String *stat*)

If getStatistics function is called, this event will be triggered once receiving a RTP statistics .

Parameters

sessionId	The session ID of the call.
stat	RTP statistics of a json format.

RTP callback events

Functions

- Int32 PortSIP.SIPCallbackEvents.onRTPPacketCallback (IntPtr callbackObject, Int32 sessionId, Int32 mediaType, Int32 direction, byte[] RTPPacket, Int32 packetSize)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onRTPPacketCallback (IntPtr *callbackObject*, Int32 *sessionId*, Int32 *mediaType*, Int32 *direction*, byte[] *RTPPacket*, Int32 *packetSize*)

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

Parameters

sessionId	The session ID of the call.
mediaType	If the received RTP packet is audio this parameter is 0 video this parameter is 1 screen this parameter is 2
direction	The RTP stream callback direction.

Type	Description
DIRECTION_SEND	Callback the send RTP stream for one channel based on the given sessionId.
DIRECTION_RECV	Callback the received RTP stream for one channel based on the given sessionId.

Parameters

RTPPacket	The memory of whole RTP packet.
packetSize	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

- Int32 PortSIP.SIPCallbackEvents.onAudioRawCallback (IntPtr callbackObject, Int32 sessionId, Int32 callbackType, byte[] data, Int32 dataLength, Int32 samplingFreqHz)
- Int32 PortSIP.SIPCallbackEvents.onVideoRawCallback (IntPtr callbackObject, Int32 sessionId, Int32 callbackType, Int32 width, Int32 height, byte[] data, Int32 dataLength)

Detailed Description

Function Documentation

Int32 PortSIP.SIPCallbackEvents.onAudioRawCallback (IntPtr *callbackObject*, Int32 *sessionId*, Int32 *callbackType*, byte[] *data*, Int32 *dataLength*, Int32 *samplingFreqHz*)

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

Parameters

sessionId	The session ID of the call.
audioCallbackMode	The type that is passed in enableAudioStreamCallback function.
data	The memory of audio stream. It's in PCM format.
dataLength	The data size.
samplingFreqHz	The audio stream sample in HZ. For example, it's 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.

Int32 PortSIP.SIPCallbackEvents.onVideoRawCallback (IntPtr *callbackObject*, Int32 *sessionId*, Int32 *callbackType*, Int32 *width*, Int32 *height*, byte[] *data*, Int32 *dataLength*)

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

Parameters

sessionId	The session ID of the call.
videoCallbackMode	The type which is passed in enableVideoStreamCallback function.
width	The width of video image.
height	The height of video image.
data	The memory of video stream. It's in YUV420 format, YV12.
dataLength	The data size.

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.

Namespace Documentation

PortSIP Namespace Reference

Classes

- class PortSIPLib
- interface SIPCallbackEvents
- class PortSIP_Errors

Enumerations

- enum class AUDIOCODEC_TYPE : int { **AUDIOCODEC_NONE** = -1, AUDIOCODEC_G729 = 18, AUDIOCODEC_PCMA = 8, AUDIOCODEC_PCMU = 0, AUDIOCODEC_GSM = 3, AUDIOCODEC_G722 = 9, AUDIOCODEC_ILBC = 97, AUDIOCODEC_AMR = 98, AUDIOCODEC_AMRWB = 99, AUDIOCODEC_SPEEX = 100, AUDIOCODEC_SPEEXWB = 102, AUDIOCODEC_ISACWB = 103, AUDIOCODEC_ISACSWB = 104, AUDIOCODEC_G7221 = 121, AUDIOCODEC_OPUS = 105, AUDIOCODEC_DTMF = 101 }
  Audio codec type.

- enum class VIDEOCODEC_TYPE : int { VIDEO_CODE_NONE = -1, VIDEO_CODEC_I420 = 113, VIDEO_CODEC_H263 = 34, VIDEO_CODEC_H263_1998 = 115, VIDEO_CODEC_H264 = 125, VIDEO_CODEC_VP8 = 120, VIDEO_CODEC_VP9 = 122 }
  Video codec type.

- enum class FILE_FORMAT : int { FILEFORMAT_NONE = 0, FILEFORMAT_WAVE = 1, FILEFORMAT_AMR, FILEFORMAT_MP3, FILEFORMAT_MP4 }
  The record file format.

- enum class RECORD_MODE : int { RECORD_NONE = 0, RECORD_RECV = 1, RECORD_SEND, RECORD_BOTH }
  The audio/Video record mode.

- enum class DIRECTION_MODE : int { DIRECTION_NONE = 0, DIRECTION_SEND_RECV = 1, DIRECTION_SEND, DIRECTION_RECV, DIRECTION_INACTIVE }
  The direction.

- enum class PORTSIP_LOG_LEVEL : int { **PORTSIP_LOG_NONE** = -1, **PORTSIP_LOG_ERROR** = 1, **PORTSIP_LOG_WARNING** = 2, **PORTSIP_LOG_INFO** = 3, **PORTSIP_LOG_DEBUG** = 4 }
  Log level.

- enum class SRTP_POLICY : int { SRTP_POLICY_NONE = 0, SRTP_POLICY_FORCE, SRTP_POLICY_PREFER }
  SRTP Policy.

- enum class TRANSPORT_TYPE : int { TRANSPORT_UDP = 0, TRANSPORT_TLS, TRANSPORT_TCP, TRANSPORT_PERS }
  Transport for SIP signaling.

- enum class SESSION_REFRESH_MODE : int { SESSION_REFERESH_UAC = 0, SESSION_REFERESH_UAS }
  The session refreshment by UAC or UAS.

- enum class DTMF_METHOD { DTMF_RFC2833 = 0, DTMF_INFO = 1 }
  send DTMF tone with two methods

Detailed Description

PortSIP The PortSIP VoIP SDK namespace

Enumeration Type Documentation

enum PortSIP.AUDIOCODEC_TYPE : int`[strong]`

Audio codec type.

Enumerator:

AUDIOCODEC_ G729	G729 8KHZ 8kbit/s.
AUDIOCODEC_P CMA	PCMA/G711 A-law 8KHZ 64kbit/s.
AUDIOCODEC_P CMU	PCMU/G711 Î¼-law 8KHZ 64kbit/s.
AUDIOCODEC_ GSM	GSM 8KHZ 13kbit/s.
AUDIOCODEC_ G722	G722 16KHZ 64kbit/s.
AUDIOCODEC_I LBC	iLBC 8KHZ 30ms-13kbit/s 20 ms-15kbit/s
AUDIOCODEC_ AMR	Adaptive Multi-Rate (AMR) 8KHZ (4.75,5.15,5.90,6.70,7.40,7.95,10.20,12.20)kbit/s.
AUDIOCODEC_ AMRWB	Adaptive Multi-Rate Wideband (AMR-WB)16KHZ (6.60,8.85,12.65,14.25,15.85,18.25,19.85,23.05,23.85)kbit/s.
AUDIOCODEC_S PEEX	SPEEX 8KHZ (2-24)kbit/s.
AUDIOCODEC_S PEEXWB	SPEEX 16KHZ (4-42)kbit/s.
AUDIOCODEC_I SACWB	internet Speech Audio Codec(iSAC) 16KHZ (32-54)kbit/s
AUDIOCODEC_I SACSWB	internet Speech Audio Codec(iSAC) 16KHZ (32-160)kbit/s
AUDIOCODEC_ G7221	G722.1 16KHZ (16,24,32)kbit/s.
AUDIOCODEC_ OPUS	OPUS 48KHZ 32kbit/s.
AUDIOCODEC_ DTMF	DTMF RFC 2833.

enum PortSIP.VIDEOCODEC_TYPE : int`[strong]`

Video codec type.

Enumerator:

VIDEO_CODE_N ONE	Do not use Video codec.

VIDEO_CODEC_I420	I420/YUV420 Raw Video format. Used with startRecord only.
VIDEO_CODEC_H263	H263 video codec.
VIDEO_CODEC_H263_1998	H263+/H263 1998 video codec.
VIDEO_CODEC_H264	H264 video codec.
VIDEO_CODEC_VP8	VP8 video codec.
VIDEO_CODEC_VP9	VP9 video codec.

enum [PortSIP.FILE_FORMAT](#) : int[strong]

The record file format.

Enumerator:

FILEFORMAT_NONE	Not Recorded.
FILEFORMAT_WAVE	The record audio file is in WAVE format.
FILEFORMAT_AMR	The record audio file is in AMR format - all voice data are compressed by AMR codec. Mono.
FILEFORMAT_MP3	The record audio file is in MP3 format.
FILEFORMAT_MP4	The record video file is in MP4(AAC and H264) format.

enum [PortSIP.RECORD_MODE](#) : int[strong]

The audio/Video record mode.

Enumerator:

RECORD_NONE	Not Record.
RECORD_RECV	Only record the received data.
RECORD_SEND	Only record the sent data.
RECORD_BOTH	Record both received and sent data.

enum [PortSIP.DIRECTION_MODE](#) : int[strong]

The direction.

Enumerator:

DIRECTION_NO NE	NOT EXIST.
DIRECTION_SE ND_RECV	both received and sent.
DIRECTION_SE ND	Only the sent.
DIRECTION_RE CV	Only the received .
DIRECTION_INA CTIVE	INACTIVE.

enum PortSIP.SRTP_POLICY : int[strong]

SRTP Policy.

Enumerator:

SRTP_POLICY_N ONE	Do not use SRTP. The SDK can receive the encrypted call(SRTP) and unencrypted call both, but can't place outgoing encrypted call.
SRTP_POLICY_F ORCE	All calls must use SRTP. The SDK allows to receive encrypted call and place outgoing encrypted call only.
SRTP_POLICY_P REFER	Top priority for using SRTP. The SDK allows to receive encrypted and decrypted call, and to place outgoing encrypted call and unencrypted call.

enum PortSIP.TRANSPORT_TYPE : int[strong]

Transport for SIP signaling.

Enumerator:

TRANSPORT_U DP	UDP Transport.
TRANSPORT_TL S	Tls Transport.
TRANSPORT_TC P	TCP Transport.
TRANSPORT_PE RS	PERS is the PortSIP private transport for anti SIP blocking. It must be used with the PERS Server http://www.portsip.com/pers.html.

enum PortSIP.SESSION_REFRESH_MODE : int[strong]

The session refreshment by UAC or UAS.

Enumerator:

SESSION_REFER ESH_UAC	The session refreshment by UAC.
SESSION_REFER ESH_UAS	The session refreshment by UAS.

enum PortSIP.DTMF_METHOD`[strong]`

send DTMF tone with two methods

Enumerator:

DTMF_RFC2833	Send DTMF tone with RFC 2833. Recommended.
DTMF_INFO	Send DTMF tone with SIP INFO.

Class Documentation

PortSIP.PortSIP_Errors Class Reference

Static Public Attributes

- static readonly int **INVALID_SESSION_ID** = -1
- static readonly int **CONFERENCE_SESSION_ID** = 0x7FFF
- static readonly int **ECoreAlreadyInitialized** = -60000
- static readonly int **ECoreNotInitialized** = -60001
- static readonly int **ECoreSDKObjectNull** = -60002
- static readonly int **ECoreArgumentNull** = -60003
- static readonly int **ECoreInitializeWinsockFailure** = -60004
- static readonly int **ECoreUserNameAuthNameEmpty** = -60005
- static readonly int **ECoreInitializeStackFailure** = -60006
- static readonly int **ECorePortOutOfRange** = -60007
- static readonly int **ECoreAddTcpTransportFailure** = -60008
- static readonly int **ECoreAddTlsTransportFailure** = -60009
- static readonly int **ECoreAddUdpTransportFailure** = -60010
- static readonly int **ECoreMiniAudioPortOutOfRange** = -60011
- static readonly int **ECoreMaxAudioPortOutOfRange** = -60012
- static readonly int **ECoreMiniVideoPortOutOfRange** = -60013
- static readonly int **ECoreMaxVideoPortOutOfRange** = -60014
- static readonly int **ECoreMiniAudioPortNotEvenNumber** = -60015
- static readonly int **ECoreMaxAudioPortNotEvenNumber** = -60016
- static readonly int **ECoreMiniVideoPortNotEvenNumber** = -60017
- static readonly int **ECoreMaxVideoPortNotEvenNumber** = -60018
- static readonly int **ECoreAudioVideoPortOverlapped** = -60019
- static readonly int **ECoreAudioVideoPortRangeTooSmall** = -60020
- static readonly int **ECoreAlreadyRegistered** = -60021
- static readonly int **ECoreSIPServerEmpty** = -60022
- static readonly int **ECoreExpiresValueTooSmall** = -60023
- static readonly int **ECoreCallIdNotFound** = -60024
- static readonly int **ECoreNotRegistered** = -60025
- static readonly int **ECoreCalleeEmpty** = -60026
- static readonly int **ECoreInvalidUri** = -60027
- static readonly int **ECoreAudioVideoCodecEmpty** = -60028
- static readonly int **ECoreNoFreeDialogSession** = -60029
- static readonly int **ECoreCreateAudioChannelFailed** = -60030
- static readonly int **ECoreSessionTimerValueTooSmall** = -60040
- static readonly int **ECoreAudioHandleNull** = -60041
- static readonly int **ECoreVideoHandleNull** = -60042
- static readonly int **ECoreCallIsClosed** = -60043
- static readonly int **ECoreCallAlreadyHold** = -60044
- static readonly int **ECoreCallNotEstablished** = -60045
- static readonly int **ECoreCallNotHold** = -60050
- static readonly int **ECoreSipMessaegEmpty** = -60051
- static readonly int **ECoreSipHeaderNotExist** = -60052
- static readonly int **ECoreSipHeaderValueEmpty** = -60053
- static readonly int **ECoreSipHeaderBadFormed** = -60054
- static readonly int **ECoreBufferTooSmall** = -60055
- static readonly int **ECoreSipHeaderValueListEmpty** = -60056
- static readonly int **ECoreSipHeaderParserEmpty** = -60057
- static readonly int **ECoreSipHeaderValueListNull** = -60058
- static readonly int **ECoreSipHeaderNameEmpty** = -60059
- static readonly int **ECoreAudioSampleNotmultiple** = -60060
- static readonly int **ECoreAudioSampleOutOfRange** = -60061
- static readonly int **ECoreInviteSessionNotFound** = -60062

- static readonly int **ECoreStackException** = -60063
- static readonly int **ECoreMimeTypeUnknown** = -60064
- static readonly int **ECoreDataSizeTooLarge** = -60065
- static readonly int **ECoreSessionNumsOutOfRange** = -60066
- static readonly int **ECoreNotSupportCallbackMode** = -60067
- static readonly int **ECoreNotFoundSubscribeId** = -60068
- static readonly int **ECoreCodecNotSupport** = -60069
- static readonly int **ECoreCodecParameterNotSupport** = -60070
- static readonly int **ECorePayloadOutofRange** = -60071
- static readonly int **ECorePayloadHasExist** = -60072
- static readonly int **ECoreFixPayloadCantChange** = -60073
- static readonly int **ECoreCodecTypeInvalid** = -60074
- static readonly int **ECoreCodecWasExist** = -60075
- static readonly int **ECorePayloadTypeInvalid** = -60076
- static readonly int **ECoreArgumentTooLong** = -60077
- static readonly int **ECoreMiniRtpPortMustIsEvenNum** = -60078
- static readonly int **ECoreCallInHold** = -60079
- static readonly int **ECoreNotIncomingCall** = -60080
- static readonly int **ECoreCreateMediaEngineFailure** = -60081
- static readonly int **ECoreAudioCodecEmptyButAudioEnabled** = -60082
- static readonly int **ECoreVideoCodecEmptyButVideoEnabled** = -60083
- static readonly int **ECoreNetworkInterfaceUnavailable** = -60084
- static readonly int **ECoreWrongDTMFTone** = -60085
- static readonly int **ECoreWrongLicenseKey** = -60086
- static readonly int **ECoreTrialVersionLicenseKey** = -60087
- static readonly int **ECoreOutgoingAudioMuted** = -60088
- static readonly int **ECoreOutgoingVideoMuted** = -60089
- static readonly int **ECoreFailedCreateSdp** = -60090
- static readonly int **ECoreTrialVersionExpired** = -60091
- static readonly int **ECoreStackFailure** = -60092
- static readonly int **ECoreTransportExists** = -60093
- static readonly int **ECoreUnsupportTransport** = -60094
- static readonly int **ECoreAllowOnlyOneUser** = -60095
- static readonly int **ECoreUserNotFound** = -60096
- static readonly int **ECoreTransportsIncorrect** = -60097
- static readonly int **ECoreCreateTransportFailure** = -60098
- static readonly int **ECoreTransportNotSet** = -60099
- static readonly int **ECoreECreateSignalingFailure** = -60100
- static readonly int **ECoreArgumentIncorrect** = -60101
- static readonly int **ECoreSipMethodNameEmpty** = -60102
- static readonly int **ECoreSipAlreadySubscribed** = -60103
- static readonly int **EAudioFileNameEmpty** = -70000
- static readonly int **EAudioChannelNotFound** = -70001
- static readonly int **EAudioStartRecordFailure** = -70002
- static readonly int **EAudioRegisterRecodingFailure** = -70003
- static readonly int **EAudioRegisterPlaybackFailure** = -70004
- static readonly int **EAudioGetStatisticsFailure** = -70005
- static readonly int **EAudioIsPlaying** = -70006
- static readonly int **EAudioPlayObjectNotExist** = -70007
- static readonly int **EAudioPlaySteamNotEnabled** = -70008
- static readonly int **EAudioRegisterCallbackFailure** = -70009
- static readonly int **EAudioCreateAudioConferenceFailure** = -70010
- static readonly int **EAudioOpenPlayFileFailure** = -70011
- static readonly int **EAudioPlayFileModeNotSupport** = -70012
- static readonly int **EAudioPlayFileFormatNotSupport** = -70013
- static readonly int **EAudioPlaySteamAlreadyEnabled** = -70014
- static readonly int **EAudioCreateRecordFileFailure** = -70015
- static readonly int **EAudioCodecNotSupport** = -70016
- static readonly int **EAudioPlayFileNotEnabled** = -70017
- static readonly int **EAudioPlayFileUnknowSeekOrigin** = -70018

- static readonly int **EAudioCantSetDeviceIdDuringCall** = -70019
- static readonly int **EAudioVolumeOutOfRange** = -70020
- static readonly int **EVideoFileNameEmpty** = -80000
- static readonly int **EVideoGetDeviceNameFailure** = -80001
- static readonly int **EVideoGetDeviceIdFailure** = -80002
- static readonly int **EVideoStartCaptureFailure** = -80003
- static readonly int **EVideoChannelNotFound** = -80004
- static readonly int **EVideoStartSendFailure** = -80005
- static readonly int **EVideoGetStatisticsFailure** = -80006
- static readonly int **EVideoStartPlayAviFailure** = -80007
- static readonly int **EVideoSendAviFileFailure** = -80008
- static readonly int **EVideoRecordUnknowCodec** = -80009
- static readonly int **EVideoCantSetDeviceIdDuringCall** = -80010
- static readonly int **EVideoUnsupportCaptureRotate** = -80011
- static readonly int **EVideoUnsupportCaptureResolution** = -80012
- static readonly int **ECameraSwitchTooOften** = -80013
- static readonly int **EMTUOutOfRange** = -80014
- static readonly int **EDeviceGetDeviceNameFailure** = -90001

The documentation for this class was generated from the following file:
- PortSIP_Errors.cs

PortSIP.PortSIPLib Class Reference

Public Member Functions

- **PortSIPLib** ([SIPCallbackEvents](#) calbackevents)
- Boolean **createCallbackHandlers** ()
- void **releaseCallbackHandlers** ()
- Int32 [initialize](#) ([TRANSPORT_TYPE](#) transportType, String localIp, Int32 localSIPPort, [PORTSIP_LOG_LEVEL](#) logLevel, String logFilePath, Int32 maxCallSessions, String sipAgentString, Int32 audioDeviceLayer, Int32 videoDeviceLayer, String TLSCertificatesRootPath, String TLSCipherList, Boolean verifyTLSCertificate)
  Initialize the SDK.

- void [unInitialize](#) ()
  Un-initialize the SDK and release resources.

- String [getVersion](#) ()
  Get the current version number of the SDK.

- Int32 [setLicenseKey](#) (String key)
  Set the license key. It must be called before setUser function.

- Int32 [getNICNums](#) ()
  Get the Network Interface Card numbers.

- Int32 [getLocalIpAddress](#) (Int32 index, StringBuilder ip, Int32 ipSize)
  Get the local IP address by Network Interface Card index.

- Int32 [setUser](#) (String userName, String displayName, String authName, String password, String sipDomain, String sipServerAddr, Int32 sipServerPort, String stunServerAddr, Int32 stunServerPort, String outboundServerAddr, Int32 outboundServerPort)
  Set user account info.

- void [removeUser](#) ()
  Remove the user. It will un-register from SIP server given that the user is already registered.

- Int32 [setDisplayName](#) (String displayName)
  Set the display name of user.

- Int32 [setInstanceId](#) (String uuid)
  Set outbound (RFC5626) instanceId to be used in contact headers.

- Int32 [registerServer](#) (Int32 expires, Int32 retryTimes)
  Register to SIP proxy server (login to server).

- Int32 [unRegisterServer](#) (Int32 waitMS)
  Un-register from the SIP proxy server.

- Int32 [enableRport](#) (Boolean enable)

Enable/disable rport(RFC3581).

- Int32 enableEarlyMedia (Boolean enable)
Enable/disable Early Media.

- Int32 enablePriorityIPv6Domain (Boolean enable)
Enable/disable which allows specifying the preferred protocol when a domain supports both IPV4 and IPV6 simultaneously.

- Int32 setUriUserEncoding (String character, Boolean enable)
Modifies the default URI user character needs to be escaped.

- Int32 setReliableProvisional (Int32 mode)
Enable/disable PRACK.

- Int32 enable3GppTags (Boolean enable)
Enable/disable the 3Gpp tags, including "ims.icsi.mmtel" and "g.3gpp.smsip".

- void enableCallbackSignaling (Boolean enableSending, Boolean enableReceived)
Enable/disable to callback the SIP messages.

- Int32 enableRtpCallback (Int32 sessionId, Int32 mediaType, Int32 directionMode)
Set the RTP callbacks to allow access to the sent and received RTP packets. The onRTPPacketCallback events will be triggered.

- Int32 addAudioCodec (AUDIOCODEC_TYPE codecType)
Enable an audio codec. It will be appears in SDP.

- Int32 addVideoCodec (VIDEOCODEC_TYPE codecType)
Enable a video codec. It will appear in SDP.

- Boolean isAudioCodecEmpty ()
Detect if enabled audio codecs is empty or not.

- Boolean isVideoCodecEmpty ()
Detect if enabled video codecs is empty or not.

- Int32 setAudioCodecPayloadType (AUDIOCODEC_TYPE codecType, Int32 payloadType)
Set the RTP payload type for dynamic audio codec.

- Int32 setVideoCodecPayloadType (VIDEOCODEC_TYPE codecType, Int32 payloadType)
Set the RTP payload type for dynamic Video codec.

- void clearAudioCodec ()
Remove all enabled audio codecs.

- void clearVideoCodec ()
Remove all enabled video codecs.

- Int32 setAudioCodecParameter (AUDIOCODEC_TYPE codecType, String parameter)
  Set the codec parameter for audio codec.

- Int32 setVideoCodecParameter (VIDEOCODEC_TYPE codecType, String parameter)
  Set the codec parameter for video codec.

- Int32 setSrtpPolicy (SRTP_POLICY srtpPolicy, Boolean allowSrtpOverUnsecureTransport)
  Set the SRTP policy.

- Int32 setRtpPortRange (Int32 minimumRtpPort, Int32 maximumRtpPort)
  Set the RTP ports range for RTP streaming.

- Int32 enableCallForward (Boolean forBusyOnly, String forwardTo)
  Enable call forward.

- Int32 disableCallForward ()
  Disable the call forwarding. The SDK is not forwarding any incoming call after this function is called.

- Int32 enableSessionTimer (Int32 timerSeconds, SESSION_REFRESH_MODE refreshMode)
  Allows to periodically refresh Session Initiation Protocol (SIP) sessions by sending INVITE requests repeatedly.

- Int32 disableSessionTimer ()
  Disable the session timer.

- void setDoNotDisturb (Boolean state)
  Enable the "Do not disturb" to enable/disable.

- Int32 enableAutoCheckMwi (Boolean state)
  Allows to enable/disable the check MWI (Message Waiting Indication) automatically.

- Int32 setRtpKeepAlive (Boolean state, Int32 keepAlivePayloadType, Int32 deltaTransmitTimeMS)
  Enable or disable to send RTP keep-alive packet when the call is established.

- Int32 setKeepAliveTime (Int32 keepAliveTime)
  Enable or disable to send SIP keep-alive packet.

- Int32 getSipMessageHeaderValue (String sipMessage, String headerName, StringBuilder headerValue, Int32 headerValueLength)
  Access the SIP header of SIP message.

- Int32 addSipMessageHeader (Int32 sessionId, String methodName, Int32 msgType, String headerName, String headerValue)
  Add the SIP Message header into the specified outgoing SIP message.

- Int32 removeAddedSipMessageHeader (Int32 sipMessageHeaderId)

Remove the headers (custom header) added by addSipMessageHeader.

- Int32 clearAddedSipMessageHeaders ()
  Clear the added extension headers (custom headers)

- Int32 modifySipMessageHeader (Int32 sessionId, String methodName, Int32 msgType, String headerName, String headerValue)
  Modify the special SIP header value for every outgoing SIP message.

- Int32 removeModifiedSipMessageHeader (Int32 sipMessageHeaderId)
  Remove the extension header (custom header) from every outgoing SIP message.

- Int32 clearModifiedSipMessageHeaders ()
  Clear the modified headers value, and do not modify every outgoing SIP message header values any longer.

- Int32 addSupportedMimeType (String methodName, String mimeType, String subMimeType)
  Set the SDK to receive the SIP messages that include special mime type.

- Int32 setAudioSamples (Int32 ptime, Int32 maxPtime)
  Set the audio capture sample.

- Int32 setAudioDeviceId (Int32 recordingDeviceId, Int32 playoutDeviceId)
  Set the audio device that will be used for audio call.

- Int32 setVideoOrientation (Int32 rotation)
  Set the video device that will be used for video call.

- Int32 setVideoDeviceId (Int32 deviceId)
  Set the video device that will be used for video call.

- Int32 setVideoResolution (Int32 width, Int32 height)
  Set the video capturing resolution.

- Int32 setAudioBitrate (Int32 sessionId, AUDIOCODEC_TYPE audioCodecType, Int32 bitrateKbps)
  Set the audio bitrate.

- Int32 setVideoBitrate (Int32 sessionId, Int32 bitrateKbps)
  Set the video bitrate.

- Int32 setVideoFrameRate (Int32 sessionId, Int32 frameRate)
  Set the video frame rate.

- Int32 sendVideo (Int32 sessionId, Boolean sendState)
  Send the video to remote side.

- void muteMicrophone (Boolean mute)

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

- void muteSpeaker (Boolean mute)
  Mute the device speaker. It's unavailable for Android and iOS.

- void setChannelOutputVolumeScaling (Int32 sessionId, Int32 scaling)
- void setChannelInputVolumeScaling (Int32 sessionId, Int32 scaling)
- Int32 setRemoteVideoWindow (Int32 sessionId, IntPtr remoteVideoWindow)
  Set the window for a session that is used to display the received remote video image.

- Int32 displayLocalVideo (Boolean state, Boolean mirror, IntPtr localVideoWindow)
  Start/stop displaying the local video image.

- Int32 setVideoNackStatus (Boolean state)
  Enable/disable the NACK feature (rfc6642) that helps to improve the video quality.

- Int32 call (String callee, Boolean sendSdp, Boolean videoCall)
  Make a call.

- Int32 rejectCall (Int32 sessionId, int code)
  rejectCall Reject the incoming call.

- Int32 hangUp (Int32 sessionId)
  hangUp Hang up the call.

- Int32 answerCall (Int32 sessionId, Boolean videoCall)
  answerCall Answer the incoming call.

- Int32 updateCall (Int32 sessionId, bool enableAudio, bool enableVideo, bool enableScreen)
  Use the re-INVITE to update the established call.

- Int32 hold (Int32 sessionId)
  To place a call on hold.

- Int32 unHold (Int32 sessionId)
  Take off hold.

- Int32 muteSession (Int32 sessionId, Boolean muteIncomingAudio, Boolean muteOutgoingAudio, Boolean muteIncomingVideo, Boolean muteOutgoingVideo)
  Mute the specified session audio or video.

- Int32 forwardCall (Int32 sessionId, String forwardTo)
  Forward call to another one when receiving the incoming call.

- Int32 pickupBLFCall (String replaceDialogId, Boolean videoCall)
  This function will be used for picking up a call based on the BLF (Busy Lamp Field) status.

- Int32 sendDtmf (Int32 sessionId, DTMF_METHOD dtmfMethod, int code, int dtmfDuration, bool playDtmfTone)

Send DTMF tone.

- Int32 refer (Int32 sessionId, String referTo)
  Refer the current call to another one.

- Int32 attendedRefer (Int32 sessionId, Int32 replaceSessionId, String referTo)
- Int32 attendedRefer2 (IntPtr libSDK, Int32 sessionId, Int32 replaceSessionId, String replaceMethod, String target, String referTo)
  Make an attended refer with specified request line and specified method embedded into the "Refer-To" header.

- Int32 outOfDialogRefer (Int32 replaceSessionId, String replaceMethod, String target, String referTo)
  Send an out of dialog REFER to replace the specified call.

- Int32 acceptRefer (Int32 referId, String referSignalingMessage)
  Accept the REFER request, and a new call will be made if called this function. The function is usually called after onReceivedRefer callback event.

- Int32 rejectRefer (Int32 referId)
  Reject the REFER request.

- Int32 enableSendPcmStreamToRemote (Int32 sessionId, Boolean state, Int32 streamSamplesPerSec)
  Enable the SDK to send PCM stream data to remote side from another source instead of microphone.

- Int32 sendPcmStreamToRemote (Int32 sessionId, byte[] data, Int32 dataLength)
  Send the audio stream in PCM format from another source instead of audio device capturing (microphone).

- Int32 enableSendVideoStreamToRemote (Int32 sessionId, Boolean state)
  Enable the SDK send video stream data to remote side from another source instead of camera.

- Int32 sendVideoStreamToRemote (Int32 sessionId, byte[] data, Int32 dataLength, Int32 width, Int32 height)
  Send the video stream to remote side.

- Int32 enableSendScreenStreamToRemote (Int32 sessionId, Boolean state)
  Enable the SDK send Screen stream data to remote side from selected screen source instead of camera.

- Int32 enableAudioStreamCallback (Int32 sessionId, Boolean enable, DIRECTION_MODE direction)
  Enable/disable the audio stream callback.

- Int32 enableVideoStreamCallback (Int32 sessionId, DIRECTION_MODE direction)
  Enable/disable the video stream callback.

- Int32 enableScreenStreamCallback (Int32 sessionId, DIRECTION_MODE direction)
  Enable/disable the video stream callback.

- Int32 startRecord (Int32 sessionId, String recordFilePath, String recordFileName, Boolean appendTimestamp, Int32 channels, FILE_FORMAT recordFileFormat, RECORD_MODE audioRecordMode, RECORD_MODE videoRecordMode)
  Start recording the call.

- Int32 stopRecord (Int32 sessionId)
  Stop record.

- Int32 startPlayingFileToRemote (Int32 sessionId, String fileName, Boolean loop, Int32 playAudio)
  Play a file to remote party.

- Int32 stopPlayingFileToRemote (Int32 sessionId)
  Stop playing file to remote party.

- Int32 startPlayingFileLocally (String fileUrl, Boolean loop, IntPtr playVideoWindow)
  Play a file to remote party.

- Int32 stopPlayingFileLocally ()
  Stop playing file to locally.

- Int32 createAudioConference ()
  Create an audio conference. It will be failed if the existent conference is not ended yet.

- Int32 createVideoConference (IntPtr conferenceVideoWindow, Int32 width, Int32 height, Int32 layout)
  Create a video conference. It will be failed if the existent conference is not ended yet.

- void destroyConference ()
  End the existent conference.

- Int32 setConferenceVideoWindow (IntPtr videoWindow)
  Set the window for a conference that is used to display the received remote video image.

- Int32 joinToConference (Int32 sessionId)
  Join a session into existent conference. If the call is in hold, it will be un-hold automatically.

- Int32 removeFromConference (Int32 sessionId)
  Remove a session from an existent conference.

- Int32 setAudioRtcpBandwidth (Int32 sessionId, Int32 BitsRR, Int32 BitsRS, Int32 KBitsAS)
  Set the audio RTCP bandwidth parameters to the RFC3556.

- Int32 setVideoRtcpBandwidth (Int32 sessionId, Int32 BitsRR, Int32 BitsRS, Int32 KBitsAS)
  Set the video RTCP bandwidth parameters as the RFC3556.

- Int32 getStatistics (Int32 sessionId)
  Obtain the statistics of channel. the event onStatistics will be triggered.

- void enableVAD (Boolean state)
  Enable/disable Voice Activity Detection (VAD).

- void enableAEC (Boolean state)
  Enable/disable AEC (Acoustic Echo Cancellation).

- void enableCNG (Boolean state)
  Enable/disable Comfort Noise Generator (CNG).

- void enableAGC (Boolean state)
  Enable/disable Automatic Gain Control (AGC).

- void enableANS (Boolean state)
  Enable/disable Audio Noise Suppression (ANS).

- Int32 enableAudioQos (Boolean state)
  Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for audio channel.

- Int32 enableVideoQos (Boolean state)
  Set the DSCP (differentiated services code point) value of QoS (Quality of Service) for video channel.

- Int32 setVideoMTU (Int32 mtu)
  Set the MTU size for video RTP packet.

- Int32 sendOptions (String to, String sdp)
  Send OPTIONS message.

- Int32 sendInfo (Int32 sessionId, String mimeType, String subMimeType, String infoContents)
  Send a INFO message to remote side in a call.

- Int32 sendSubscription (String to, String eventName)
  Send a SUBSCRIBE message to subscribe an event.

- Int32 terminateSubscription (Int32 subscribeId)
  Terminate the given subscription.

- Int32 sendMessage (Int32 sessionId, String mimeType, String subMimeType, byte[] message, Int32 messageLength)
  Send a MESSAGE message to remote side in dialog.

- Int32 sendOutOfDialogMessage (String to, String mimeType, String subMimeType, Boolean isSMS, byte[] message, Int32 messageLength)
  Send an out of dialog MESSAGE message to remote side.

- Int32 setDefaultSubscriptionTime (Int32 secs)
  Set the default expiration time to be used when creating a subscription.

- Int32 setDefaultPublicationTime (Int32 secs)
  Set the default expiration time to be used when creating a publication.

- Int32 setPresenceMode (Int32 mode)
  Indicate the SDK uses the P2P mode for presence or presence agent mode.

- Int32 presenceSubscribe (String to, String subject)
  Send a SUBSCRIBE message for subscribing the contact's presence status.

- Int32 presenceTerminateSubscribe (Int32 subscribeId)
  Terminate the given presence subscription.

- Int32 presenceRejectSubscribe (Int32 subscribeId)
  Reject a presence SUBSCRIBE request which is received from contact.

- Int32 presenceAcceptSubscribe (Int32 subscribeId)
  Accept the presence SUBSCRIBE request which is received from contact.

- Int32 setPresenceStatus (Int32 subscribeId, String stateText)
  Set the presence status.

- Int32 getNumOfRecordingDevices ()
  Gets the count of audio devices available for audio recording.

- Int32 getNumOfPlayoutDevices ()
  Gets the number of audio devices available for audio playout.

- Int32 getRecordingDeviceName (Int32 deviceIndex, StringBuilder nameUTF8, Int32 nameUTF8Length)
  Gets the name of a specific recording device given by an index.

- Int32 getPlayoutDeviceName (Int32 deviceIndex, StringBuilder nameUTF8, Int32 nameUTF8Length)
  Get the name of a specific playout device given by an index.

- Int32 setSpeakerVolume (Int32 volume)
  Set the speaker volume level.

- Int32 getSpeakerVolume ()
  Gets the speaker volume level.

- Int32 setMicVolume (Int32 volume)
  Sets the microphone volume level.

- Int32 getMicVolume ()

Retrieves the current microphone volume.

- Int32 getScreenSourceCount ()
  Retrieves the current number of screen.

- Int32 getScreenSourceTitle (Int32 deviceIndex, StringBuilder nameUTF8, Int32 nameUTF8Length)
  Retrieves the current screen title .

- Int32 selectScreenSource (Int32 nDeviceIndex)
  Sets the Screen to share .

- Int32 SetScreenFrameRate (Int32 nFrameRate)
  Sets the Screen video framerate .

- Int32 setScreenVideoWindow (Int32 sessionId, IntPtr screenVideoWindow)
  Set the window for a session that is used to display the received screen video .

- void audioPlayLoopbackTest (Boolean enable)
  Use it for the audio device loop back test.

- Int32 getNumOfVideoCaptureDevices ()
  Get the number of available capturing devices.

- Int32 getVideoCaptureDeviceName (Int32 deviceIndex, StringBuilder uniqueIdUTF8, Int32 uniqueIdUTF8Length, StringBuilder deviceNameUTF8, Int32 deviceNameUTF8Length)
  Get the name of a specific video capture device given by an index.

- Int32 showVideoCaptureSettingsDialogBox (String uniqueIdUTF8, Int32 uniqueIdUTF8Length, String dialogTitle, IntPtr parentWindow, Int32 x, Int32 y)
  Display the capture device property dialog box for the specified capture device.

The documentation for this class was generated from the following file:
- PortSIPLib.cs

PortSIP.SIPCallbackEvents Interface Reference

Public Member Functions

- Int32 onRegisterSuccess (String statusText, Int32 statusCode, StringBuilder sipMessage)
- Int32 onRegisterFailure (String statusText, Int32 statusCode, StringBuilder sipMessage)
- Int32 onInviteIncoming (Int32 sessionId, String callerDisplayName, String caller, String calleeDisplayName, String callee, String audioCodecNames, String videoCodecNames, Boolean existsAudio, Boolean existsVideo, StringBuilder sipMessage)
- Int32 onInviteTrying (Int32 sessionId)
- Int32 onInviteSessionProgress (Int32 sessionId, String audioCodecNames, String videoCodecNames, Boolean existsEarlyMedia, Boolean existsAudio, Boolean existsVideo, StringBuilder sipMessage)
- Int32 onInviteRinging (Int32 sessionId, String statusText, Int32 statusCode, StringBuilder sipMessage)
- Int32 onInviteAnswered (Int32 sessionId, String callerDisplayName, String caller, String calleeDisplayName, String callee, String audioCodecNames, String videoCodecNames, Boolean existsAudio, Boolean existsVideo, StringBuilder sipMessage)
- Int32 onInviteFailure (Int32 sessionId, String callerDisplayName, String caller, String calleeDisplayName, String callee, String reason, Int32 code, StringBuilder sipMessage)
- Int32 onInviteUpdated (Int32 sessionId, String audioCodecNames, String videoCodecNames, Boolean existsAudio, Boolean existsVideo, Boolean existsScreen, StringBuilder sipMessage)
- Int32 onInviteConnected (Int32 sessionId)
- Int32 onInviteBeginingForward (String forwardTo)
- Int32 onInviteClosed (Int32 sessionId)
- Int32 onDialogStateUpdated (String BLFMonitoredUri, String BLFDialogState, String BLFDialogId, String BLFDialogDirection)
- Int32 onRemoteHold (Int32 sessionId)
- Int32 onRemoteUnHold (Int32 sessionId, String audioCodecNames, String videoCodecNames, Boolean existsAudio, Boolean existsVideo)
- Int32 onReceivedRefer (Int32 sessionId, Int32 referId, String to, String from, StringBuilder referSipMessage)
- Int32 onReferAccepted (Int32 sessionId)
- Int32 onReferRejected (Int32 sessionId, String reason, Int32 code)
- Int32 onTransferTrying (Int32 sessionId)
- Int32 onTransferRinging (Int32 sessionId)
- Int32 onACTVTransferSuccess (Int32 sessionId)
- Int32 onACTVTransferFailure (Int32 sessionId, String reason, Int32 code)
- Int32 onReceivedSignaling (Int32 sessionId, StringBuilder signaling)
- Int32 onSendingSignaling (Int32 sessionId, StringBuilder signaling)
- Int32 onWaitingVoiceMessage (String messageAccount, Int32 urgentNewMessageCount, Int32 urgentOldMessageCount, Int32 newMessageCount, Int32 oldMessageCount)
- Int32 onWaitingFaxMessage (String messageAccount, Int32 urgentNewMessageCount, Int32 urgentOldMessageCount, Int32 newMessageCount, Int32 oldMessageCount)
- Int32 onRecvDtmfTone (Int32 sessionId, Int32 tone)
- Int32 onRecvOptions (StringBuilder optionsMessage)
- Int32 onRecvInfo (StringBuilder infoMessage)
- Int32 onRecvNotifyOfSubscription (Int32 subscribeId, StringBuilder notifyMsg, byte[] contentData, Int32 contentLenght)
- Int32 onSubscriptionFailure (Int32 subscribeId, Int32 statusCode)
- Int32 onSubscriptionTerminated (Int32 subscribeId)
- Int32 onPresenceRecvSubscribe (Int32 subscribeId, String fromDisplayName, String from, String subject)
- Int32 onPresenceOnline (String fromDisplayName, String from, String stateText)
- Int32 onPresenceOffline (String fromDisplayName, String from)
- Int32 onRecvMessage (Int32 sessionId, String mimeType, String subMimeType, byte[] messageData, Int32 messageDataLength)

- Int32 onRecvOutOfDialogMessage (String fromDisplayName, String from, String toDisplayName, String to, String mimeType, String subMimeType, byte[] messageData, Int32 messageDataLength)
- Int32 onSendMessageSuccess (Int32 sessionId, Int32 messageId)
- Int32 onSendMessageFailure (Int32 sessionId, Int32 messageId, String reason, Int32 code)
- Int32 onSendOutOfDialogMessageSuccess (Int32 messageId, String fromDisplayName, String from, String toDisplayName, String to)
- Int32 onSendOutOfDialogMessageFailure (Int32 messageId, String fromDisplayName, String from, String toDisplayName, String to, String reason, Int32 code)
- Int32 onPlayFileFinished (Int32 sessionId, String fileName)
- Int32 onStatistics (Int32 sessionId, String stat)
- Int32 onRTPPacketCallback (IntPtr callbackObject, Int32 sessionId, Int32 mediaType, Int32 direction, byte[] RTPPacket, Int32 packetSize)
- Int32 onAudioRawCallback (IntPtr callbackObject, Int32 sessionId, Int32 callbackType, byte[] data, Int32 dataLength, Int32 samplingFreqHz)
- Int32 onVideoRawCallback (IntPtr callbackObject, Int32 sessionId, Int32 callbackType, Int32 width, Int32 height, byte[] data, Int32 dataLength)
- Int32 onScreenRawCallback (IntPtr callbackObject, Int32 sessionId, Int32 callbackType, Int32 width, Int32 height, byte[] data, Int32 dataLength)

Detailed Description

SIPCallbackEvents PortSIP VoIP SDK Callback events

Member Function Documentation

Int32 PortSIP.SIPCallbackEvents.onScreenRawCallback (IntPtr *callbackObject*, Int32 *sessionId*, Int32 *callbackType*, Int32 *width*, Int32 *height*, byte[] *data*, Int32 *dataLength*)

This event will be triggered once receiving the screen video packets if called enableScreenStreamCallback function.

Parameters

sessionId	The session ID of the call.
videoCallbackMode	The type which is passed in enableVideoStreamCallback function.
width	The width of video image.
height	The height of video image.
data	The memory of video stream. It's in YUV420 format, YV12.
dataLength	The data size.

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.

The documentation for this interface was generated from the following file:
- SIPCallbackEvents.cs

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