

The list of methods and their parameters

The complete list of methods and parameters

Grey denotes parameters described above or below in the table.

Depending on the direction and destination of the call, different subsets of parameters for the same invocation can be used. For example, in case of the invocation of `ConnectionStatusEvent`, `sipLogin`, `sipPassword`, and other corresponding parameters are passed. In case of an error, the same event `ConnectionStatusEvent` will have only two parameters: `status` and `info` when sending to a client, and `status`, `info`, `nodeId`, `sessionId`, `appKey` when sending to the Web-server.

connect	Establishes connection with the WCS Server
urlServer	This parameter is used by WCS JavaScript API to connect to the server.
appKey	This parameter passes the REST - URL for the given application to WCS. To view and add applications use the command line interface (CLI).
sipRegisterRequired	If this parameter is true, registration on the SIP server is performed by invoking SIP REGISTER. If the parameter is false, registration on the SIP server is not performed. In this case, a web page cannot accept incoming SIP calls, but still can make outgoing calls if the SIP server allows outgoing calls without SIP registration.
sipLogin	SIP login of a user
sipAuthenticationName	SIP name of a user used for SIP authentication. Can be different from sipLogin.
sipPassword	SIP password. Used for SIP authentication.
sipVisibleName	SIP user name displayed to other users receiving an incoming call from this user.
sipDomain	SIP domain. FQDN or IP address.
sipOutboundProxy	SIP proxy server. FQDN or IP address. Can be different from sipDomain.
sipPort	SIP port the SIP server uses to handle SIP traffic.
sipContactParams	A string of custom parameters added to the SIP Connect header of the REGISTER query.
status	
mediaProviders	Array of available types of media on WCS JavaScript API: ['WebRTC', 'Flash'].
restClientConfig	A JSON-object describing web-server interaction control configuration. If the object isn't passed, the default values are used. See also: RestClientConfig .
width	Maximal video width, in pixels
height	Maximal video height, in pixels
ConnectionStatusEvent	Connection status change

sipR egist erRe quired	
sipL ogin	
sipP ass word	
sipVi sible Name	
sipD omain	
sipO utbo und Proxy	
sipP ort	
sipC onta ctPa rams	
status	WCS Server connection status:PENDING,ESTABLISHED,FAILED,DISCONNECTED.
info	Additional information can be added to this field. For example, if status==FAILED, the info contains the description of the reason.
auth Token	A key being used in WCS JavaScript API for connection of Flash implementation to WCS server using RTMFP protocol.
medi aPro viders	
node Id	
sessi onId	
app Key	
Regi strati onSt atus Event	SIP registration status change
status	Registration statuses:REGISTERED,UNREGISTERED,FAILED.
info	
sipM essa geR aw	Original SIP-message with headers. SIP Response to the REGISTER Request.
node Id	
sessi onId	
app Key	
call	Outgoing call
callId	Unique id of the call.
callee	A callee in the SIP URI format, tel URI or a telephone number.

caller	A caller in the SIP URI format.
visibleName	A label displayed to the callee.
hasVideo	If true, this is a video call.
inviteParameters	Parameters added to the SIP INVITE Request URI.
isMsrp	If true, this is not a voice call, but establishing of MSRP-connection to transmit data.
status	
incoming	If true, it is an incoming call from SIP side.
mediaProvider	Media technology used on WCS JavaScript API, possible values: "WebRTC", "Flash".
sdp	SDP, created on WCS JavaScript API side, will be placed when mediaProvider is WebRTC
OnCallEvent	Incoming call
callId	`
callee	
caller	
visibleName	
hasVideo	
inviteParameters	
sipMessageRaw	SIP INVITE message the incoming call even is based upon.
incoming	
status	
mediaProvider	
sdp	
nodeId	
sessionId	
appKey	
CallStatusEvent	Call status change
callId	

incoming	If true, the call is incoming
status	Call statuses: TRYING, RING, SESSION_PROGRESS, BUSY, ESTABLISHED, HOLD, FINISH, FAILED
info	
sipMessageRaw	Original message corresponding to the message being sent. For example, in case of TRYING, this would be SIP 100 TRYING Response, in case of ESTABLISHED this would be SIP 200 OK Response, and in case of HOLD this would be SIP 200 OK Response to re-INVITE, and so on.
sipStatus	Response status received from SIP side
caller	
callee	
hasVideo	
visibleName	
mediaProvider	
nodeId	
sessionId	
appKey	
answer	Answer incoming call
callId	
incoming	
sipStatus	
caller	
callee	
hasVideo	
visibleName	
mediaProvider	
sdp	
status	
nodeId	
sessionId	
appKey	
hangup	Hangs up the call
callId	

hasV ideo	
node Id	
sessi onId	
app Key	
hold	Puts the call on hold
callId	
hasV ideo	
node Id	
sessi onId	
app Key	
unhold	Unhold the call
callId	
hasV ideo	
node Id	
sessi onId	
app Key	
transfer	Transfer the call
callId	
targ et	The number or the SIP URI of the subscriber the call is transferred to.
node Id	
sessi onId	
app Key	
Transfer StatusEvent	Call transfer status change
callId	
inco ming	If true, the transfer was initiated by the other side.
status	Call transfer statuses:ACCEPTED,TRYING,COMPLETED,FAILED.If the status is not recognized, then status received from SIP side will be passed.
info	
sipM essa geR aw	

hasV ideo	
node Id	
sessi onId	
app Key	
send DTMF	Sends DTMF signal
callId	
dtmf	A symbol to pass in DTMF as text: 1-16, *, #.
type	The type of the DTMF signal:INFO,INFO_RELAY,RFC2833.
node Id	
sessi onId	
app Key	
send Mes sage	Sends a message
id	The unique id of the message.
from	The number of the SIP URI of the sender.
to	The number, the login or the SIP URI of the recipient.
body	The text of the message.
cont entT ype	<p>text/plain- the message is sent as SIP MESSAGE with the 'Content-Type : text/plain' header and with text in the body of the message.</p> <p>message/cpim- the message is sent as SIP MESSAGE with the 'Content-Type:message/cpim' header and a text/plain message in the body of the CPIM-message.</p> <p>multipart/mixed- the message is sent as SIP MESSAGE with the 'Content-Type:multipart/mixed' header and CPIM messages in the body, each one containing one text/plain message.</p>
isIm dnR equir ed	If the flag is set to true, for message/cpim and multipart/mixed message types, the information asking for a delivery notification via IMDN will be added to the body of the CPIM message.
recip ients	The list of recipients separated by commas. SIP URI, tel URI or SIP logins of recipients must be specified and separated by commas. The field is used only if ContentType is set to multipart/mixed. This field works correctly only when the SIP-server supports sending messages to multiple subscribers based on multipart/mixed. WCS sends a multipart/mixed message with multiple recipients to the SIP-server. If your SIP-server doesn't support such sending, leave this field blank and try sending several individual messages.
node Id	
sessi onId	
app Key	
OnM essa geEv ent	Incoming message
id	
from	

to	
body	
contentType	
isImDnRequired	If the incoming message has this flag, an IMDN delivery notification will be sent.
sipMessageRaw	A SIP MESSAGE message that corresponds to the OnMessageEvent event of the incoming message.
status	
nodeId	
sessionId	
appKey	
MessageStatusEvent	Message status change
id	
from	
to	
contentType	
isImDnRequired	
body	
status	Message statuses:RECEIVED,ACCEPTED,FAILED,IMDN_DELIVERED,IMDN_FAILED, IMDN_NOTIFICATION_SENT
info	
sipMessageRaw	<p>SIP message corresponding to the status:</p> <p>ACCEPTED- SIP 200 OK Response to SIP MESSAGE Request.</p> <p>FAILED- SIP 4xx Response from the SIP-server.</p> <p>DELIVERED- received a SIP MESSAGE delivery notification with the status 'Delivered'.</p> <p>DELIVERY_FAILED- received a delivery notification with the status 'Delivery Failed'.</p>
nodeId	
sessionId	

app Key	
send IMDN	Sends IMDN delivery notification
message Id	
node Id	
sessionId	
app Key	
subscribe	SIP subscribe - subscribe to notification of the SIP-server. RFC3265 .
event	Event type:reg
expires	Time interval in seconds. During this interval the WCS-server performs re-SUBSCRIBE.
terminate	
node Id	
sessionId	
app Key	
Subscription StatusEvent	SIP-subscription status change
event	
expires	
terminate	If true, the subscription should be deactivated
requestBody	XML received from SIP side
status	Subscription statuses:Active,Terminated
info	
sipMessageRaw	<p>SIP message changing the status of the subscription:</p> <p>Active- SIP 200 OK Response on SUBSCRIBE Request.</p> <p>Terminated- SIP 200 OK Response on SUBSCRIBE Request with expires:0.</p> <p>Terminated- SIP NOTIFY Request with the 'terminated' status in the body of the NOTIFY message.</p>
node Id	
sessionId	

app Key	
send XcapRequest	Send an XCAP request
url	URL for the XCAP request
node Id	
sessi onId	
app Key	
XcapStatusEvent	Receiving XCAP response
url	
xcap Res ponse	The body of the XCAP response
publishStream	Publishing the stream to the server
name	The name of the published stream. Must be unique. If a stream with such name already published, the publishing of the stream is prohibited.
medi aSes sionId	Identifier of media session
publi shed	If true, the stream is being published
hasV ideo	If true, the stream has video
status	
sdp	SDP received from client
node Id	
sessi onId	
app Key	
reco rd	If true, the published stream is being recorded
unpublishStream	Unpublishing the stream
name	
medi aSes sionId	
publi shed	
hasV ideo	
status	
sdp	

node Id	
sessi onId	
app Key	
reco rd	
play Stre am	Play the stream
name	The name of the played stream.
medi aSes sion	
publi shed	
hasV ideo	
status	
sdp	
node Id	
sessi onId	
app Key	
play HLS	Play the stream via HLS
name	
medi aSes sionId	
medi aPro vider	
node Id	
sessi onId	
app Key	
token	Client authentication token
play RTSP	Play the stream via RTSP
name	
medi aSes sionId	
medi aPro vider	
node Id	

sessi onId	
app Key	
rtsp Url	RTSP stream URL
User - Age nt	Client User-Agent
stop Stream	Stop playback of the stream
name	
medi aSes sionId	
publi shed	
hasV ideo	
status	
sdp	
node Id	
sessi onId	
app Key	
Stream Status Event	Stream status change
name	
status	Stream status:PUBLISHING,UNPUBLISHED,PLAYING,STOPPED
medi aSes sionId	
publi shed	
hasV ideo	
sdp	
info	
node Id	
sessi onId	
app Key	
reco rd	

StreamKeepAliveEvent	Stream keep-alive REST request
nodeId	
appKey	
sessionId	
mediaSessionId	
name	
published	
hasVideo	
status	Stream status:PLAYING, PUBLISHING
info	
mediaProvider	Media technology used on WCS JavaScript API, possible values: "WebRTC", "Flash"
record	
sendData	Sends data
operationId	Unique id of the data to send.
payload	JSON object containing data.
nodeId	
sessionId	
appKey	
OnDataEvent	Receiving of input data
operationId	
payload	
nodeId	
sessionId	
appKey	
DataStatusEvent	Sent data status change

operationId	
status	ACCEPTED,FAILED
info	
nodeId	
sessionId	
appKey	
Error Event	Unclassified error
info	Additional information about the error.
send Bug Report	Sends an error report to save on the server
text	Brief custom description of the error.
type	If the type is no_media, the server enables traffic dump before creating a bug report to make sure the traffic goes properly for that user. Sending bug reports of this type can help diagnose problems with sound going one side only.
nodeId	
sessionId	
appKey	
Bug Report Status Event	Error report sending confirmation with the name of the saved file as the output
fileName	The name of the file on the server where the bug report was saved.
nodeId	
sessionId	
appKey	
Context Parameters	Context parameters. Used for all calls from WCS to the Web-server.
nodeId	Unique id of the WCS server instance.
sessionId	Unique id of the client connect in that instance.
appKey	Application id on the WCS server the user has established connection with.