WCS REST API

API methods

HTTP: http://host:9091   HTTPS: https://host:8888

Method example: http://host:9091/RestCall/call

REST method	Params	Returns	Errors	Description
SIP as RTMP				
/RESTCall/call	CallConnection	{}	409 - Conflict 500 - Internal error	Сделать SIP звонок для кейса SIP as RTMP
/RESTCall/getCalls	{}	List<Call>	404 - Call not found 500 - Internal error	Получить список всех SIP-звонков
/RESTCall/getStatus	Call	String	404 - Call not found 500 - Internal error	Получить статус SIP-звонка
/RESTCall/hangup	Call	{}	404 - Call not found 500 - Internal error	Сбросить SIP-звонок
/RESTCall/sendDTMF	DTMF	{}	404 - Call not found 500 - Internal error	Отправить DTMF внутри SIP-звонка
RTSP				
/RESTCall/findRtspAgents	RtspAgentFilter	List<RtspAgentFilter>	404 - RTSP not found 500 - Internal error	Получить все RTSP сессии
/RESTCall/findAllRtspAgents	{}	List<RtspAgentFilter	500 - Internal error	Получить все RTSP

		>		сессии
/RESTCall/shutdownRtspAgent	RtspAgentFilter	{}	404 - RTSP not found 500 - Internal error	Остановить RTSP сессию
/RESTCall/startupRtspAgent	{uri=”rtsp://”}	{}	500 - Internal error	Создать RTSP-сессию
Streaming				
/RESTCall/findStreams	Stream	List<Stream>	404 - Stream not found 500 - Internal error	Найти потки
/RESTStream/getStream	Stream	Stream	404 - Stream not found 500 - Internal error	Найти один поток
/RESTStream/terminate	Stream	{}	404 - Stream not found	Завершить работу потока

Params

CallConnection	String callId; String parentCallId; Boolean incoming; String status; Integer sipStatus; String rtmpUrl; String rtmpStream; RtmpStreamStatus rtmpStreamStatus; String caller; String callee; Boolean hasAudio = true;

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Boolean hasVideo = false;
String sdp;
String visibleName;
Map<String,String> inviteParameters;
String mediaProvider;
String sipMessageRaw;
Boolean isMsrp = false;
String target;
Boolean holdForTransfer = false;
        Boolean sipRegisterRequired;
        String sipLogin;
        String sipAuthenticationName;
        String sipPassword;
        String sipVisibleName;
        String sipDomain;
        String sipOutboundProxy;
        Integer sipPort;
        String sipContactParams;
        Integer width;
        Integer height;
        String supportedResolutions;
        Boolean useProxy;
        Boolean useDTLS;
        Boolean useWsTunnel = false;
        Boolean useWsTunnelPacketization2 = false;
        Boolean useBase64BinaryEncoding = false;
        List<String> mediaProviders;
        String appMainClass;
        String appCallbackClass;
        String authToken;
        String status;
        Map<String,RestMethodConfig> restClientConfig;
        String clientVersion;
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	String clientOSVersion; String clientBrowserVersion;
Call	String callId; String parentCallId; Boolean incoming; String status; Integer sipStatus; String rtmpUrl; String rtmpStream; RtmpStreamStatus rtmpStreamStatus; String caller; String callee; Boolean hasAudio = true; Boolean hasVideo = false; String sdp; String visibleName; Map<String,String> inviteParameters; String mediaProvider; String sipMessageRaw; Boolean isMsrp = false; String target; Boolean holdForTransfer = false;
DTMF	String callId; String dtmf; DTMFType type;
RtspAgentFilter	uri status
Stream	String mediaSessionId; String remoteMediaElementId;

	String name; boolean published; boolean hasVideo; boolean hasAudio = true; StreamStatus status = StreamStatus.NEW; String sdp; String info; boolean record; String recordName; int width; int height; int bitrate; int quality; String rtmpUrl;