Settings file flashphoner.properties

Main server settings

Outdated or invalid setting are highlighted in grey. They were used in previous versions. These settings will be probably deleted in further WCS updates.

Option	Default	Type	Need restart	Description
aac_bitrate	128000	Integer	false	AAC encoding bitrate
aac_encoder_sync_drop_threshold	1000	Long	true	JitterBuffer will be reset upon reaching this number of dropped sync packets
aac_test_start_codec	20	Integer	true	AAC test codecs count
aac_test_transcode_iterations	1000	Integer	true	AAC test interval
add_register_auth_headers	false	Boolean	false	If true, then add Authorization header in REGISTER request when first registering. Some SIP servers are configured so that they do not accept such requests. In that case this setting should be set to 'false'
agent_set_local_session_debug	false	Boolean	false	If true, enable local agent session debug
allow_domains	null	String	false	If set, then WebSocket connections from these domains only will be allowed
allow_outside_codecs	true	Boolean	false	If false, don't add outside (browser) codecs to SDP
allow_reinvite_in_hold_state	true	Boolean	false	If true, process re-INVITE requests within the session even if the call is in hold state
answer_with_one_codec_in_sdp	false	Boolean	false	If true, answer with one codec only in SDP. It can be useful in cases of improper operation of SIP equipment from some vendors, which incorrectly interpret two or more codecs in SDP during a connection establishment in Offer-Answer model
audio_frames_per_packet	6	Integer	false	RTMFP. Audio will be flushed after this number of audio frames in the packet is reached
audio_incoming_buffer_size	50	Integer	false	Waiting for RTCP sync packet on this interval in packets, for audio
audio_incoming_min_buffer_size	2	Integer	false	Waiting for RTCP sync packet at least on this interval in packets, for audio
audio_mixer_max_delay	300	Integer	false	Audio mixer max delay in milliseconds
audio_mixer_output_codec	opus	String	false	Audio mixer output codec (multiple codecs not allowed)
audio_mixer_output_sample_rate	48000	Integer	false	Audio mixer output samle rate in Hz
audio_reliable	partial	on partial off	false	RTMFP, reliability for audio
audio_stream_mode_udp	true	Boolean	true	Not in use
auto_login_url	null	String	false	Not in use
aws_s3_credentials	null	String	true	AWS s3 credentials: region;accessKey;secretKey
balance_header	balance	String	false	This SIP header will be sent to client as a balance
burst_avoidance_count	100	String	false	Burst avoidance count
busy_state	null	String	false	Used if send_busy_when_on_call=true, and an incoming call comes during another established call. Caller will receive this status. If true, the value of ip_local= setting will be used in SIP and SDP. If false, then the value of ip= will be used

call_record_listener	com.flashphoner.
	server.client.
	DefaultCallRecord
	Listener
		String	false	Full name of Java class that implements interface ICallRecordListener
				public interface ICallRecordListener {
				void onRecordReport(RecordReport recordReport);
				}

cdn_advertise_pulled	false	Boolean	true	If true, pulls CDN advertise
cdn_allowed_ips		ArrayList	true	Comma-separated list of allowed IPs or networks for CDN. Example: 88.198.98.1/24, 88.198.99.219
cdn_enabled	false	Boolean	true	If true, enables CDN
cdn_groups		ArrayList	true	CDN groups for this node
cdn_inbound_auditor_interval	1000	Integer	true	Time interval to check inbound connections, in milliseconds
cdn_inbound_connection_unanswered_pings	3	Integer	true	Inbound connection unanswered pings number. Connection considered to be lost when this number is reached
cdn_ip	null	String	true	CDN node IP address (or domain name when cdn_nodes_resolve_ip=true)
cdn_nodes_auditor_interval	1000	Integer	true	Time interval to check available CDN nodes, in milliseconds

cdn_nodes_resolve_ip	false	Boolean	true	If true, resolve CDN node domain names to IP addresses
cdn_nodes_route_refresh_interval	60000	Integer	true	Time interval to refresh CDN routes, in milliseconds
cdn_nodes_timeout	-1	Integer	true	CDN nodes timeout in seconds. -1 means nodeTimeout disabled
cdn_outbound_auditor_interval	2000	Integer	true	Time interval to check outbound connections, in milliseconds
cdn_outbound_connection_timeout	6000	Integer	true	Outbound connection timeout, in milliseconds
cdn_point_of_entry		String	true	CDN point of entry node IP address (or domain name when cdn_nodes_resolve_ip=true)
cdn_port	8084	Integer	true	CDN server port
cdn_role	EDGE	ORIGIN EDGE TRANSCODER	true	CDN role: origin - the source of media streams for other CDN nodes edge (default) pulls media streams from origin CDN node(s)
cdn_skip_pulled_streams	true	Boolean	true	If true, skip pulled streams
cdn_ssl	false	Boolean	true	If true, enables SSL

chat_listener	null	String	false	Full name of Java class that implements interface IChatListener
				public interface IChatListener {
				void onMessage(InstantMessage message);
				}

check_incoming_video_negotiated	true	Boolean	false	If true, check video negotiated and if it is not force RTP synchronization for audio
check_receiver_origin	false	Boolean	false	If true, check origin of RTCP packet and discard if unknown
cli_auth_keys	/usr/local /FlashphonerWeb CallServer/.ssh /authorized_keys	String	true	CLI Auth keys file path
cli_enabled	true	Boolean	true	If true, enables CLI
cli_port	2001	Integer	true	CLI server port
client_dump_level	0	Integer	false	If tcpdump is installed in the system, it will be started and will capture client session traffic: 0 - do not capture traffic 1 - capture SIP traffic only 2 - capture SIP and media traffic: ICE, RTP, SRTP, RTCP, WebRTC
client_handler	null	String	true	Not in use
client_log_exclude		String	false	Do not log events listed
client_log_level	INFO	String	false	Log4j level. Logs related to client sessions will be recorded on the server in /usr/local/FlashphonerWebCallServer/logs/client_logs directory with the set logging level. Will work only if enable_extended_logging=true
client_mode	true	Boolean	false	If true, the value of ip_local= setting will be used in SIP and SDP. If false, then the value of ip= will be used
client_timeout	3600000	Integer	false	Client timeout value in milliseconds
codec_terminator_timeout	5000	Integer	false	Codec release timeout, in seconds. Default: If codec has been marked as terminated, and if no new packets went through this codec in 5 seconds, the codec will be released
codecs	null	String	false	List of supported codecs ordered by priority
codecs_exclude_cdn	null	String	false	Comma-separated list of codecs which will not be used for CDN
codecs_exclude_sip	null	String	false	Comma-separated list of codecs which will not be used for SIP phone cases
codecs_exclude_sip_rtmp	null	String	false	Comma-separated list of codecs which will not be used for SIP as RTMP case
codecs_exclude_streaming	null	String	false	Comma-separated list of codecs which will not be used for streaming
complex_test_config		String	false	Complex transcoder test configuration
complex_test_decode	false	Boolean	false	Enable decoding during complex transcoding test
complex_test_fps	15	Integer	true	Complex transcoder test FPS
complex_test_replay	3	Integer	true	Complex transcoder test repeats count
complex_test_thread	3	Integer	true	Complex transcoder test threads count
cost_header	cost	String	false	This SIP header will be sent to client as a call cost
cps_client	null	String	false	Comma-separated list of IPs or networks with corresponding CPS limits. Example: 192.168.88.2:10,192.168.88.0/16:15
cps_interval	1000	Long	false	Time window for measuring CPS, in milliseconds
cps_node	2147483647	Integer	false	Global CPS limitation for node
cpu_load_avg_size	20	Integer	true	CPU load average size
cpu_load_refresh	50	Integer	true	CPU load refresh rate

cpu_load_reject	false	Boolean	false	If true, reject streams when CPU load exceeds treshold
cpu_load_threshold	80	Integer	true	CPU load treshold
cpu_load_window	2000	Integer	true	Timeslice to estimate CPU load
custom_ice_agent	true	Boolean	false	If true, use custom ICE agent
custom_watermark_filename	null	String	false	Sets custom PNG file for watermark. The file should be placed in /usr /local/FlashphonerWebCallServer/conf directory. The feature is not available for Trial license
data_packet_decoder_fire_null_messages	true	Boolean	false	If true, pass special data packet up the RTP process chain when original received data failed to decode
datagram_channel_factory	NioDatagramChannelFactory	String	true	NioDatagramChannelFactory, OioDatagramChannelFactory - channel factory used for server sockets
decoded_frame_interceptor	null	String	false	Full name of Java class that implements interface IDecodedFrameInterceptor. This class should be wrapped to .jar file placed in /usr/local/FlashphonerWebCallServer/lib directory
decoder_binary_log_enable	false	Boolean	false	Binary log decoder
decoder_binary_log_size	5	Integer	true	Binary log decoder size
decoder_mode	JNI	QUEUE JNI	false	Decoder mode
decoder_priority	FF,OPENH264	String	false	Decoder priority
decoder_stat_log	false	Boolean	false	Enable decoder statistics logging
default_sdp_state	sendrecv	String	false	If SDP from SIP side comes without sendrecv, recvonly, or sendonly attribute, then it is assumed that the attribute defined in this setting was received
degraded_streams_threshold	20	Integer	true	Degraded streams threshold
degraded_streams_window	2000	Integer	true	Timeslice to estimate stream degradation
delta_threshold	100	Integer	false	RTMFP. If delta between UDP media packets is greater than the threshold, it will be reported
detect_flash_2_flash_calls	true	Boolean	false	If true, WCS server will use an RTP extension header in RTP packets, which can be used for designation of WCS's own streams, even if they are traced through third-party PBX, e.g. Asterisk
disable_drop_aac_frame	true	Boolean	false	If true, disables dropping AAC frames
disable_manager_rmi	true	Boolean	true	If true, disables RMI communications between WCS Core and WCS Manager
disable_rest_auth	true	Boolean	false	If true, disables authorization in rest api
disable_rest_requests	false	Boolean	true	If true, disables Rest requests to application
disable_rtc_ata	null	String	false	By default WCS server will try to avoid transcoding and send its supported codec to the other side, even if codecs will be chosen asymmetrically. This behaviour is called Avoid Transcoding Algorithm (ATA). This option defines comma-separated list of SIP User Agents, for which the algorithm will be disabled. It means that if codecs are asymmetrical, then for these User Agents transcoding will proceed
disable_rtc_avoid_transcoding_alg	false	Boolean	false	If true, disables RTC ATA (see above)
disable_streaming_proxy	false	Boolean	false	If true, disable proxy and enable transcoding for all streams. For debug only
disable_streaming_proxy_aac	false	Boolean	false	If false, enable AAC proxying
domain	null	String	false	SIP domain. If this parameter is set, it will redefine values that were transmitted during connection
dtls_close_socket_after_tries	0	Integer	false	Disable / enable DTLS session termination after the specified number of connection attempts. By default, DTLS session will not be terminated: dtls_close_socket_after_tries=0
dtls_message_timeout	120	Integer	false	DTLS handshake timeout in seconds, must be set to a non-zero value
dtls_socket_timeout_ms	1000	Integer	false	DTLS socket SO_TIMEOUT in milliseconds. With this option set to a non-zero value, a read() call on the InputStream associated with this Socket will block for only this amount of time
dtls_use_socket_timeout	true	Boolean	false	If true, enable DTLS socket SO_TIMEOUT
dtmf	null	String	false	This type will be used if DTMF type (INFO, INFO_RELAY, RFC2833) was not specified when DTMF was sent
dump_avcc_relay	false	Boolean	false	If true, write outgoing MSE packets to file. That file can afterwards be processed as VoD at client side. Used for MSE development tests
enable_candidate_harvester	false	Boolean	false	If true, gather ICE candidates using external STUN server
enable_empty_shift_writer	false	Boolean	false	Enable empty shift writer for conference
enable_extended_logging	true	Boolean	false	When extended logging is enabled, these settings are used: - client_log_level - client_dump_level Then logs for all client sessions are saved in /usr/local /FlashphonerWebCallServer/logs/client_logs directory

enable_flight_recorder	false	Boolean	false	Enable flight recorder
enable_flight_recorder_test	false	Boolean	false	Enable flight recorder test
enable_local_videochat	false	Boolean	false	Not in use
enable_new_client_logger	true	Boolean	false	If true, enable new client logger
enable_rtc_video_generator	false	Boolean	false	Designed to avoid video negotiation issue in SIP cases. If true, generated video will be sent once session is established. It is a workaround and should not be used in normal situation
enable_sip_stack_thread_audit	true	Boolean	false	If true, enable audit of SIP stack
encode_record_name	null	String	true	Encode record name setting
encoder_buffer_length_sec	1	Integer	false	Encoding buffer for audio, in seconds
encoder_default_video_resolution	640x480	String	false	encoder_default_video_resolution
encoder_mode	JNI	QUEUE JNI	false	Encoder mode
encoder_priority	FF,OPENH264	String	false	Encoder priority
encoder_stat_log	false	Boolean	false	Enable encoder statistics logging
event_scanner_cached_pool	false	Boolean	false	If true, use event scanner cached pool
event_scanner_pool_size	10	Integer	false	Event scanner pool size
exclude_record_name_characters	null	String	true	Exclude characters from record name
fetch_caller_from_pai	false	Boolean	false	If true, then for an incoming call the caller should be taken from PAI (P-Asserted-Identity) header. If that header is empty, the caller will be displayed as Unknown/Anonymous
fetch_caller_from_pai_set_from_if_empty	false	Boolean	false	If true, fetch caller from PAI 'from' when caller is empty
flash_codecs	alaw,ulaw, speex16,h264, vp8	String	false	This set of codecs (if it is not empty) will be used if either party of a call is Flash
flash_policy.port	843	Integer	true	Listening port for flash policy requests to crossdomain.xml file
flash_streaming_enable	true	Boolean	false	Not in use
flight_recorder_capacity	500	Integer	false	Flight recorder's buffer capacity in records
flight_recorder_categories	NONE	NONE WCS1438	true	Flight recorder categories
flush_audio_interval	80	Integer	true	RTMFP flush interval in milliseconds for flash-audio data from server
flush_video_interval	0	Integer	true	RTMFP flush interval in milliseconds for flash-video data from server
force_client_requested_video_resolution	true	Boolean	false	If true, use client-specified resolution passed in Stream object
force_expires	-1	Integer	false	If this parameter is set, WCS server will assume that Expires header had this value in 200 OK received in response to SIP REGISTER request
force_local_audio_codec	null	String	false	This setting is used for Flash SIP calls. You can enforce audio codec, e.g. ulaw, and Flash client should switch to that audio codec
force_periodic_fir_request_for_sip_as_rtmp	true	Boolean	false	If true, FIR request will be sent to SIP endpoint every 5 seconds
force_profile_level	null	String	false	If set, this profile will be used regardless of profiles which figured in H. 264 codec negotiation. Example: force_profile_level=420020
force_rtmp_audio_codec	null	String	false	Forced codec for old as-RTMP cases using RTMPOutputWriter and for the latest HLS writer
force_sendrecv_for_outgoing_calls	false	Boolean	false	If true, force 'sendrecv' for audio and video for outgoing SIP calls
generate_av_for_ua	null	String	true	WCS server generates RTP traffic (inaudible audio and video with Flashphoner logo) when SIP session is established if detected that the other party's SIP User Agent name is specified in the setting. Required in case of 'SIP as RTMP' stream with Zoom or Twilio SIP Domain as the SIP endpoint. Example: generate_av_for_ua = Twilio Media Gateway
get_callee_url	null	String	false	Not in use
global_bandwidth_check_enabled	false	Boolean	false	If true, enable global bitrate in and out in statistics
h264_buffer_nack_list_threshold	30	Integer	false	JitterBuffer will be reset upon reaching this number of NACK packets
h264_max_nalu_size	1346	Integer	true	Maximum size of outgoing NALU while H.264 is encoded. The option is used to prevent MTU excess while encoding high resolution video
h264_new_buffer	false	Boolean	false	Not in use
handler_async_disconnect	true	Boolean	false	If true, enable asynchronous disconnect handler
hangup_incoming_call_state	null	String	false	Send BUSY_HERE by default. It is also possible to set custom status that should be returned as BUSY response. This can be used for IMS use cases. If true, do not send SIP messages to browser

hide_all	false	Boolean	false	If true, do not send SIP messages to browser
hls.address	0.0.0.0	InetAddress	true	Listening address for HLS server
hls.http.port	8082	Integer	true	HLS server HTTP port
hls.https.port	8445	Integer	true	HLS server HTTPS port
hls_access_control_headers	null	String	true	HLS response headers
hls_auth_enabled	false	Boolean	false	Enable check auth tokens for hls
hls_auth_token_cache	10	Integer	false	Timeout for cache auth tokens in seconds
hls_auto_start	false	Boolean	false	If true, enable HLS autostart
hls_disable_cleanup	false	Boolean	false	Do not remove inactive hls files from hdd
hls_discontinuity_enabled	false	Boolean	false	If true, enables HLS discontinuity
hls_enabled	true	Boolean	false	If true, enable HLS support
hls_list_size	10	Integer	false	Maximum number of segments in playlist
hls_manager_provider_timeout	300	Integer	false	HLS manager provider timeout
hls_min_list_size	1	Integer	false	Minimum number of segments in playlist (should be less than 11)
hls_player_height	480	Integer	false	HLS player height
hls_player_width	640	Integer	false	HLS player width
hls_server_enabled	true	Boolean	true	If true, activate HLS server
hls_static_dir	client2/examples /demo/streaming /hls_static	String	false	HLS static dir
hls_static_enabled	false	Boolean	false	If true, enables HLS static content
hls_test_interval	182000	Integer	true	HLS test interval
hls_test_run_for	180	Integer	true	HLS test duration in seconds
hls_test_start_streams	10	Integer	true	HLS test streams count
hls_test_start_writers	10	Integer	true	HLS test writers count
hls_time	4	Integer	false	Size of one HLS segment in seconds
hls_time_min	2000	Long	false	Minimal size of one HLS segment in milliseconds
hls_wrap	20	Integer	false	Maximum number of ts-files. The option is necessary to prevent disc overflow
http.address	0.0.0.0	InetAddress	true	Listening address for HTTP server (statistics)
http.port	8081	Integer	true	WCS server HTTP port
http_client_connection_read_timeout	2000	Integer	false	HTTP client connection read timeout in milliseconds
http_client_connection_timeout	2000	Integer	false	HTTP client connection timeout in milliseconds
https.address	0.0.0.0	InetAddress	true	Listening address for HTTPS server (statistics)
https.port	8444	Integer	true	WCS server HTTPS port
https_server_enabled	true	Boolean	true	If true, activate HTTPS server
ice_authorize_by_address	false	Boolean	false	If true, authorize ICE by IP address only. So, if we receive packets from authorized address but another port, the packets will be accepted even though the port was not authorized
ice_consent_freshness	true	Boolean	false	If true, send binding request instead of binding indication for consent freshness
ice_keep_alive_enabled	true	Boolean	false	If true, enables ICE keep-alive
ice_keep_alive_timeout	15	Integer	false	ICE establishing timeout in seconds. By default, if ICE is in running (waiting COMPLETE) state after 15 seconds, the session will be terminated
ice_tcp_receive_buffer_size	1048576	Integer	true	Receive buffer size for ice tcp channels
ice_tcp_send_buffer_size	1048576	Integer	true	Send buffer size for ice tcp channels
ice_tcp_transport	false	Boolean	false	If true, use tcp transport only
ice_timeout	15	Integer	false	ICE keep-alive timeout in seconds. By default, ICE session will be terminated if no refresh packets from browser in 15 seconds
ice_udp_transport_new	true	Boolean	false	If true, use new udp transport
ignore_incoming_call_if_sip_login_port_does_not_match_request_uri	false	Boolean	false	If true, terminate incoming call if the SIP port does not correspond to the user indicated in Request-URI
in_jitter_buffer_enabled	false	Boolean	false	If true, switch on intermediary buffer on server side, which will reset downstream packets according to reset algorithm and min_drop_rate=, max_drop_rate=, min_queue_size=, max_queue_size= and in_jitter_buffer_enabled= settings
increase_equals_timestamp	100	Integer	false	Timestamps are equal within this interval in milliseconds

ip	0.0.0.0	String	true	External IP address. This IP address will differ from specified with ip_local option when WCS server is behind NAT
ip_local	0.0.0.0	String	true	WCS server will create sockets and listen on this interface
jitter_threshold	50	Integer	false	RTMFP. If jitter between UDP media packets is greater than the threshold, it will be reported
jni_cache_class	true	Boolean	false	If true, cache JNI Class object
keep_alive.algorithm	HIGH_LEVEL	INTERNAL NONE HIGH_LEVEL	true	Keep-alive algorithm: INTERNAL, NONE, or HIGH_LEVEL
keep_alive.enabled	websocket,rtmp, rtmfp	String	true	Enable keep-alive for the listed protocols
keep_alive.peer_interval	2000	Integer	true	Keep-alive peer interval (Not in use)
keep_alive.probes	10	Integer	true	Number of unsuccessfull attempts to ping connected client (WebSocket, RTMP, RTMFP). If reached, server will consider the client as disconnected and will release the associated resources.
keep_alive.server_interval	5000	Integer	true	Interval in milliseconds between attempts to ping connected client (WebSocket, RTMP, RTMFP)
keep_alive_streaming_sessions_enabled	false	Boolean	true	If true, server sends keep-alive REST requests to check if stream playback is allowed to continue / resume
kill_event_scanner	false	Boolean	false	Debug option, for development only
load_balancing_acao_header		String	true	Use this value for Access-Control-Allow-Origin (ACAO) header in the response when cross-domain HTTP request to the loadbalancer received
load_balancing_enabled	false	Boolean	true	If true, activate loadbalancer
max_callid_length	32	Integer	false	Maximum length of SIP callID. If the length of generated callID exceeds this value, it will be cut to this length
max_drop_rate	null	String	false	Queue size will be increased if loss raises up to this value. Is used only if out_jitter_buffer_enabled=true or in_jitter_buffer_enabled=true
max_queue_size	null	String	false	Packets will be reset if queue size exceeds this maximum value. Is used only if out_jitter_buffer_enabled=true or in_jitter_buffer_enabled=true
media_port_from	31001	Integer	true	Beginning of media ports range for ICE, RTP, SRTP, RTCP
media_port_stress_test_iterations	1	Integer	false	Media port stress test iterations
media_port_stress_test_thread_sleep	5	Integer	false	Media port stress test thread sleeping interval
media_port_stress_test_threads	5	Integer	false	Media port stress test threads count
media_port_to	32000	Integer	true	End of media ports range for ICE, RTP, SRTP, RTCP
media_ports_auditor_interval	5000	Integer	true	Audit interval for busy and free ports, in milliseconds
media_ports_auditor_max_attempts	3	Integer	true	Number of audits to make sure freed port is not bound. Freed port will be returned to the pool of free ports if this number of successfull audits is reached
min_drop_rate	null	String	false	Queue size will be decreased if loss reduces to this value. Is used only if out_jitter_buffer_enabled=true or in_jitter_buffer_enabled=true
min_queue_size	null	String	false	Queue size will not be decreased lower that this minimum value. Is used only if out_jitter_buffer_enabled=true or in_jitter_buffer_enabled=true
mixer_activity_timer_cool_off_period	1	Integer	false	Mixer will be terminated after {mixer_activity_timer_cool_off_period * mixer_activity_timer_timeout} since last stream activity for the corresponding mixer
mixer_activity_timer_timeout	60000	Integer	false	If there is no streams added to mixer within this timeout in milliseconds, corresponding mixer will be terminated
mixer_auto_create_delimiter	#	String	false	Mixer auto create stream/room delimiter
mixer_auto_start	false	Boolean	false	If true, enable mixer autostart
mixer_idle_timeout	60000	Long	false	Mixer idle timeout in milliseconds
mixer_layout_class	com.flashphoner. media.mixer. video. presentation. GridLayout	String	true	Name of class for custom mixer layout
mixer_lossless_video_processor_enabled	false	Boolean	false	Enable custom video processor for mixer incoming streams, setting this to true may degrade realtime part
mixer_lossless_video_processor_max_mixer_buffer_size_ms	200	Integer	false	Max size that is allowed for mixer's incoming buffer, after reaching this point processor will use own buffer instead
mixer_lossless_video_processor_wait_time_ms	20	Integer	false	How long to wait before checking mixer's incoming buffer again in case it was full
mixer_out_buffer_enabled	false	Boolean	false	If true, enable buffer for out mixer streams

mixer_out_buffer_initial_size	2000	Long	false	Initial size of output mixer buffer in milliseconds
mixer_out_buffer_max_bufferings_allowed	-1	Integer	false	mixer_out_buffer_max_bufferings_allowed
mixer_out_buffer_polling_time	100	Long	false	Output mixer buffer polling time in milliseconds
mixer_out_buffer_start_size	150	Long	false	Start size of output mixer buffer in milliseconds
mixer_prune_streams	false	Boolean	false	When true, prune mixer stream
mixer_thread_priority	5	Integer	false	Mixer thread priority, min 1 max 10
mixer_thread_timeout_ms	33	Integer	false	Mixer thread timeout
mixer_video_bitrate_kbps	2000	Integer	false	Encoded video bitrate kbps
mixer_video_buffer_length	10	Integer	false	Video buffer length for decoded frames
mixer_video_desktop_layout_inline_padding	10	Integer	false	Padding between video streams in bottom row (under screen sharing stream)
mixer_video_desktop_layout_padding	30	Integer	false	Padding between top row (screen sharing stream) and bottom row (other streams)
mixer_video_enabled	true	Boolean	false	When false, mixer stream has audio-only
mixer_video_fps	30	Integer	false	Fps constraint for mixer stream
mixer_video_grid_layout_middle_padding	10	Integer	false	Padding between video streams in one row (when there is no screen sharing stream)
mixer_video_grid_layout_padding	30	Integer	false	Padding between rows of video streams (when there is no screen sharing stream)
mixer_video_height	720	Integer	false	Height constraint for mixer stream
mixer_video_layout_desktop_key_word	desktop	String	false	Keyword for screen sharing streams
mixer_video_quality	24	Integer	false	Encoded video quality (CRF)
mixer_video_width	1280	Integer	false	Width constraint for mixer stream
mpeg1.gop_size	60	Integer	false	GOP size or k-frame interval
mpeg1.qmax	24	Integer	false	Maximum value of quality parameter. The lower the value, the better is quality, and the higher is bitrate. If it is too low (e.g. 1), bitrate is too high and vice versa
mpeg1.qmin	4	Integer	false	Minimum value of quality parameter. The lower the value, the better is quality, and the higher is bitrate. If it is too low (e.g. 1), bitrate is too high and vice versa
mpeg1.trellis	0	Integer	false	Trellis quantization
msrp_port	2855	Integer	false	Port for receiving MSRP / TCP connections
multipart_message_service_uri	null	String	false	SIP URI for sending message to multiple destinations. A message is sent from client with Content-Type:multipart/mixed and then sent by SIP server to multiple destinations
native_test_aac	true	Boolean	true	If true, enable AAC native test
native_test_decoder	true	Boolean	true	If true, enable decoder native test
native_test_encoder	true	Boolean	true	If true, enable encoder native test
native_test_opus	true	Boolean	true	If true, enable Opus native test
native_test_resampler	true	Boolean	true	If true, enable native test resampler
native_test_run_for	180	Integer	true	Native test duration
native_test_start_threads	10	Integer	true	Native test threads count
native_test_thread_interval	200	Integer	true	Native test interval
netty_deadlock_aware_server_workers	true	Boolean	false	If true, enable Netty SSH deadlock server workers
netty_deadlock_aware_worker_timeout	10000	Integer	false	Timeout to detect SSL connection with Netty deadlock
no_media_dump_interval	15000	Long	false	Period in milliseconds, within which media traffic should be captured by tcpdump when client sends bug report with no_media type
notify_message_call_timeout	null	String	false	Timeout in milliseconds to wait for client confimation of receiving an incoming message. When an incoming message is received, it is sent to the destination client, and the confirmation timeout is started. If the client does not confirm receiving the message within the timeout, WCS server responds to the sender that the message was not received and delivered (in cases when delivery report is required)

on_record_hook_script	on_record_hook.sh	String	false	This option points to shell script located in /usr/local/FlashphonerWebCallServer/bin directory, which is started when stream is unpublished, if a recording of the stream has been created. Two parameters will be passed to the script: $1 - the stream name $2 - absolute name of the file with recording of audio and video of the stream This script can be used to copy or move a stream record from /usr/local/FlashphonerWebCallServer/records directory to another location as soon as the recording is completed. By default, the script does not contain such commands and should be edited as required. Example: STREAM_NAME=$1 SRC_FILE=$2 SRC_DIR=/usr/local/FlashphonerWebCallServer/records/ REPLACE_STR=/var/www/html/stream_records/$STREAM_NAME- DST_FILE=${SRC_FILE/$SRC_DIR/$REPLACE_STR} cp $SRC_FILE $DST_FILE Make sure the script works correctly: start it manually, e.g. ./on_record_hook.sh streamName /usr/local/FlashphonerWebCallServer/records/stream-a58aea39-6333-4cb2-8jtn93gtmgr6mrq0nilk6l958j.mp4
options2flash_delegate	null	String	false	If true, then wait for a client response prior to responding with 200 OK to an OPTIONS request
opus.encoder.bitrate	-1	Integer	false	Target bitrate for Opus encoder, in bps
opus.encoder.complexity	-1	Integer	false	Target complexity for Opus encoder
opus_formats	null	String	false	Comma-separated list of Opus formats (name=value). Example: maxaveragebitrate=20000. These formats will be listed in SDP
order_threads_by_seq	true	Boolean	false	If true, order incoming SIP messages by sequence number and wait if number is out of order
out_jitter_buffer_enabled	null	String	false	If true, switch on intermediary buffer on server side, which will reset upstream packets according to reset algorithm and min_drop_rate=, max_drop_rate=, min_queue_size=, max_queue_size= and in_jitter_buffer_enabled= settings
outbound_port	null	String	false	SIP port. If this parameter is set, it will redefine values that were transmitted during connection
outbound_proxy	null	String	false	SIP outbound proxy. If this parameter is set, it will redefine values that were transmitted during connection
parse_system_stats	false	Boolean	false	If true, gather system level statistics such as netstat, lsof, etc. The parsing may take a lot of time
periodic_fir_request	false	Boolean	false	If true, then every 5 seconds WCS server sends an RTCP Full Intra Request (FIR) message to the input stream source and then forwards its response to the RTMP CDN. Required in case of 'SIP as RTMP' stream with Zoom as the SIP Endpoint and the input stream source, so that every new subscriber receives video keyframe (otherwise, stream video may be not played)
periodic_fir_request_interval	5000	Integer	false	Interval to send RTCP FIR in milliseconds
port_from	30000	Integer	false	Beginning of range of ports for SIP signaling
port_to	31000	Integer	false	End of range of ports for SIP signaling
preserve_non_mixed_recorded_files	false	Boolean	false	Two files are created when recording: one for incoming sound, and another for outgoing. Then those files are mixed in one resulting recording. If this setting is false, the temporary files will be deleted after mixing. If true, the files will be saved
print_publication_tables	false	Boolean	false	RTMFP. If true, print statistics of streams in logs
print_rtcp_stats	false	Boolean	false	If true, print RTCP report on end of session
priority_outside_codecs	false	Boolean	false	If true, then outside (browser) codecs will be in first place
process_remote_sdp_candidates	true	Boolean	false	If true, process candidates from SDP
profiles	42e01f	String	false	Comma-separated list of H.264 profiles. These profiles will be used in SDP for video calls
proxy_propagate_fir	true	Boolean	false	Propagate FIR requests through proxy
proxy_use_h264_packetization_mode_1_only	true	Boolean	false	If true, use H.264 packetization mode 1
ptime	20	Integer	false	Packetization time. Use carefully
ptime_corrector_enabled	true	Boolean	false	Enabling corrector by required packetization time
publication_report_format	null	String	false	RTMFP. Sets format for statistics. Possible value: csv
pull_streams	null	String	true	Comma separated list of urls to pull from at server startup
queue_ping_period	2000	Integer	true	Queue ping interval in ms
queue_stat_log	true	Boolean	false	Enable queue statistics logging
queue_transcoder_core_router_uri	tcp://127.0.0.1:5555	String	false	Queue transcoder core router URI

queue_transcoder_receive_timeout	500	Integer	true	Queue transcoder receive timeout
queue_transcoder_shm_path	/dev/shm/	String	false	Path to shared memory objects for queue transcoder
queue_transcoder_shm_size	5	Integer	true	Shared memory object size for queue transcoder
queue_transcoder_transmit_timeout	500	Integer	true	Queue transcoder transmit timeout
queue_transcoder_worker_router_uri	ipc:///tmp /flashphoner.pipe	String	false	Queue transcoder core router URI
record	null	String	false	Path to the directory for audio call recordings. If this path is designated, then audio call recordings will be saved to that directory in WAV Track format. Also, this is used for recording PCM audio on streams for debug needs (see record_audio_processor_pcm= setting)
record_audio_codec_sample_rate	44100	Integer	false	Codec sample rate used for recording streams
record_audio_processor_pcm	false	Boolean	false	If true, record audio on stream as PCM16. (Then record= option should point to a valid path, e.g. record=/tmp/)
record_filename_template	null	String	false	Filename template for an audio call recording. Besides the default fields, {date} field can also be used
record_flash_published_streams	false	Boolean	false	If true, record streams published from native Flash clients and RTMP live encoders such as Wirecast, FFmpeg, FMLE, etc.
record_mixer_streams	false	Boolean	false	When true, mixer streams are recorded
record_rotation	null	String	false	If set, rotation for stream recording files is enabled, in seconds or in Megabytes. Example: 3600 - rotate every hour Example: 10M - rotate after every 10 Megabytes
record_rtsp_streams	false	Boolean	false	If true, record RTSP streams
record_streams	true	Boolean	false	If true, WebRTC and RTMFP streams published will be recorded if stream recording is enabled for the publishing client as well: session. createStream({record:true,...}). The records will be saved to /usr/local/FlashphonerWebCallServer /records directory
recording_by_user	false	Boolean	true	If true, call is recorded for the initiator of the call only
reg_expires	3600	Integer	false	Value in seconds, which will be used in Expires header when SIP REGISTER request is sent
remove_ssrc_attr	null	Boolean	false	If true, remove ssrc attribute
replace_cached_pool_with_default_pool	false	Boolean	true	If true, replaces cached thread pool with default
resample_video	true	Boolean	false	If true, enable video rescaling. Example: 1. Publish video as 640x480 (4:3) 2. Play video as 400x225 (16:9) If resample_video=true, WCS server will rescale video from 640x480 to 400x225 and it will be flattened vertically. If resample_video=false, video will be cut down to 400x225, and part of the video will be lost. So, when setting playback width and height, you should specify appropriate ratio (e.g., 320x240 for 640x480 published stream); then, if resample_video=true, video will be rescaled properly
rest_access_control_allow_headers	content-type,x-requested-with	String	false	Rest-api response access_control_allow_headers header
rest_access_control_allow_methods	POST	String	false	Rest-api response access_control_allow_methods header
rest_access_control_allow_origin	*	String	false	Rest-api response access_control_allow_origin header
rest_hook_secret_key	null	String	false	Rest hook secret key
rest_hook_send_hash	false	Boolean	false	Rest hook send hash
rest_max_connections	200	Integer	true	Rest max connextions
rest_request_timeout	15	Integer	true	Rest request timeout in seconds
rfc2833_packets_count	null	String	false	Number of RTP packets for sending one DTMF
rmi.port	1098	Integer	true	Internal RMI port for communications with WCS Manager
rtc_ice_add_local_component	true	Boolean	false	If true, add local component for ICE candidates
rtc_ice_add_local_interface	false	Boolean	false	If true, ip_local= address will be added to ICE candidates as another candidate. (External IP address specified in ip= setting is added to ICE candidates by default)
rtc_ip	null	String	false	External IP address for WebRTC. Can be used for WebRTC deployment on particular network interface having external address different from the one specified with ip= setting
rtc_ip_local	null	String	false	Local IP address for WebRTC. Can be used for WebRTC deployment on particular network interface having local address different from the one specified with ip_local= setting
rtcp_disabled	false	Boolean	false	If true, do not process sender reports
rtcp_sender_interval	0.1	Double	false	Guard RTCP interval based on the specified fraction of RTCP bitrate
rtmfp.address	0.0.0.0	InetAddress	true	Listening address for RTMFP server

rtmfp.port	1935	Integer	true	RTMFP server port, UDP
rtmp.address	0.0.0.0	InetAddress	true	Listening address for RTMP server
rtmp.port	1935	Integer	true	RTMP server port, TCP
rtmp.server_buffer_enabled	false	Boolean	false	Enable/disable buffering rtmp data on java's heap if socket buffer is full
rtmp.server_channel_high_water_mark	52428800	Integer	true	High watermark for connected rtmp channels
rtmp.server_channel_low_water_mark	5242880	Integer	true	Low watermark for connected rtmp channels
rtmp.server_channel_send_buffer_size	1048576	Integer	true	Send buffer size for rtmp channels
rtmp.server_read_socket_timeout	0	Integer	true	TCP socket write timeout for RTMP server, in seconds
rtmp.server_socket_timeout	0	Integer	true	TCP socket write and read timeout for RTMP server for, in seconds
rtmp.server_write_socket_timeout	0	Integer	true	TCP socket write timeout for RTMP server, in seconds
rtmp.use_server_socket_timeout	false	Boolean	true	DEPRECATED (use rtmp.server_socket_timeout, rtmp.server_read_socket_timeout, rtmp.server_write_socket_timeout). If true, use for RTMP server TCP socket timeout set with rtmp.server_socket_timeout option
rtmp_activity_timer_cool_off_period	1	Integer	false	RTMP agent will be terminated after {rtmp_activity_timer_cool_off_period * rtmp_activity_timer_timeout} since last subscriber activity for the corresponding RTMP stream
rtmp_activity_timer_timeout	60000	Integer	false	If there is no subscribers for an RTMP stream within this timeout in milliseconds, corresponding RTMP session will be terminated
rtmp_appkey_source	default	String	false	RTMP appkey source: default/app
rtmp_command_amf3	true	Boolean	true	rtmp_command_amf3
rtmp_flash_ver_publisher	FMLE/3.0	String	false	RTMP publisher Flash version
rtmp_flash_ver_subscriber	LNX 9,0,124,2	String	false	RTMP subscriber Flash version
rtmp_in_buffer_enabled	false	Boolean	false	If true, enable buffer for incoming RTMP streams
rtmp_in_buffer_initial_size	2000	Long	false	Initial size of incoming RTMP buffer in milliseconds
rtmp_in_buffer_max_bufferings_allowed	-1	Integer	false	rtmp_in_buffer_max_bufferings_allowed
rtmp_in_buffer_polling_time	100	Long	false	Incoming RTMP buffer polling time in milliseconds
rtmp_in_buffer_start_size	300	Long	false	Start size of incoming RTMP buffer in milliseconds
rtmp_output_writer_bufsize	0	Integer	false	Buffer time for FFRtmpOutputWriter old outbound buffer for as-RTMP cases
rtmp_port_from	33001	Integer	false	First port in RTMP ports range, for RTMP republisher
rtmp_port_to	34000	Integer	false	Last port in RTMP ports range, for RTMP republisher
rtmp_ports_auditor_interval	10000	Integer	false	Audit interval for RTMP ports, in milliseconds
rtmp_ports_auditor_max_attempts	3	Integer	false	Number of audits to make sure freed port is not bound. Freed port will be returned to the pool of free ports if this number of successfull audits is reached
rtmp_publisher_start_time_ts	1000	Long	false	RTMP publisher start time
rtmp_pull_agent_account_for_lost_audio	false	Boolean	false	If true, enable RTMP pull agent account for lost audio
rtmp_pull_rtp_activity_detection	true	Boolean	false	If true, enable RTP activity detection while RTMP pulling
rtmp_push_auto_start	false	Boolean	false	If true, enable RTMP push autostart for newly published streams
rtmp_push_auto_start_url	null	String	false	RTMP server address to auto start pushing to
rtmp_send_video_first	false	Boolean	true	Send video first in RTMP
rtmp_transponder_force_kframe_interval	true	Boolean	false	If true, force k-frame interval for transponder in latest cases 'as-RTMP'. It is implemented sending RTCP PLI, if that is supported
rtmp_transponder_full_url	false	Boolean	false	If true, ignore streamName and use full rtmpUrl in transponders and 'as RTMP' cases. If false, streamName will be used as RTMP stream name and rtmpUrl will be treated as URL to RTMP application, e.g. rtmp://host:1935/live
rtmp_transponder_kframe_interval	60	Integer	false	Forced k-frame interval in frames. See also rtmp_transponder_force_kframe_interval= setting.
rtmp_transponder_metadata	null	String	false	RTMP transponder metadata
rtmp_transponder_send_metadata	false	Boolean	false	If true, RTMP transponder will send metadata
rtmp_transponder_stream_name_prefix	rtmp_	String	false	The specified prefix is added for all as-RTMP stream names. By default, stream named stream1 will be republished as RTMP stream with name rtmp_stream1
rtp_activity_detecting	null	String	true	Disables / enables and sets value of RTP activity timeout, in seconds. By default, RTP session will be closed if there is no media traffic in 60 seconds period (rtp_activity_detecting=true,60)
rtp_activity_video	true	Boolean	false	If true, RTP activity check is enabled for video. If false, this check is enabled for audio only
rtp_elapsed_time_threshold	10000	Long	false	RTP elapsed time threshold, in milliseconds
rtp_force_synchronization	false	Boolean	false	If true, force RTP synchronization

rtp_in_buffer_initial_size	2000	Integer	false	Initial size of incoming RTP buffer in milliseconds
rtp_in_buffer_polling_time	100	Long	false	Incoming RTP buffer polling time in milliseconds
rtp_in_reset_marker	false	Boolean	false	If true, use RTP in reset marker
rtp_paced_sender	false	Boolean	false	If true, enable paced sender for WebRTC video session. EXPERIMENTAL
rtp_paced_sender_capacity	200000000	Long	false	RTP paced sender capacity
rtp_paced_sender_increase_interval	50	Integer	false	Paced sender increase interval
rtp_paced_sender_initial_rate	200000	Integer	false	Paced sender initial rate
rtp_paced_sender_k_deviation	0.02	Double	false	Paced sender K deviation
rtp_paced_sender_k_down	0.02	Double	false	Paced sender K down
rtp_paced_sender_k_up	0.04	Double	false	Paced sender K up
rtp_paced_sender_period	1000	Long	false	RTP paced sender period
rtp_paced_sender_queue_size	2000	Integer	false	Outgoing queue maximum size
rtp_paced_sender_refill	200000000	Long	false	RTP paced sender refill
rtp_packet_cache_size	250	Integer	false	Cache size for sent packets. This is used only on video sessions to provide response to NACK requests
rtp_receive_buffer_predicator_size	1500	Integer	false	DatagramSocket constructing: receiveBufferSizePredictorFactory size
rtp_receive_buffer_size	65536	Integer	false	Buffer size for incoming RTP and SRTP (WebRTC). DatagramSocket constructing: receiveBufferSize
rtp_send_buffer_size	65536	Integer	false	Buffer size for outgoing RTP and SRTP (WebRTC). DatagramSocket constructing: sendBufferSize
rtp_session_init_always	false	Boolean	false	If true init rtp session for all media providers
rtsp.address	0.0.0.0	InetAddress	true	Listening address for RTSP server
rtsp.port	554	Integer	true	RTSP server port
rtsp_activity_timer_cool_off_period	1	Integer	false	RTSP agent will be terminated after {rtsp_activity_timer_cool_off_period * rtsp_activity_timer_timeout} since last subscriber activity for the corresponding RTSP stream
rtsp_activity_timer_timeout	60000	Integer	false	If there is no subscribers for an RTSP stream within this timeout in milliseconds, corresponding RTSP session will be terminated
rtsp_auth_cnonce	1234567890	String	true	RTSP server port
rtsp_client_address	0.0.0.0	InetAddress	true	RTSP client address
rtsp_client_strip_audio_codecs	null	String	false	Comma-separated list of audio codecs which will not be used for RTSP
rtsp_fail_on_error_track	true	Boolean	true	If true, RTSP pulling fails on error in any track
rtsp_in_buffer	false	Boolean	false	If true, use RTSP in buffer
rtsp_interleaved_enable_rtcp	true	Boolean	false	If true, enable replying to RTCP packets on the RTSP interleaved channel
rtsp_interleaved_mode	true	Boolean	false	If true, interleaved mode for RTSP (RTP over RTSP/TCP) is enabled
rtsp_port_from	32001	Integer	false	First TCP port in the port range for RTSP pooling agent
rtsp_port_to	33000	Integer	false	Last TCP port in the port range for RTSP pooling agent
rtsp_ports_auditor_interval	10000	Integer	false	Audit interval for RTSP ports, in milliseconds
rtsp_ports_auditor_max_attempts	3	Integer	false	Number of audits to make sure freed port is not bound. Freed port will be returned to the pool of free ports if this number of successfull audits is reached
rtsp_refresh_requests_limit	5	Integer	false	Maximum number of non-answered GET_PARAMETER refresh requests. Stop sending refresh requests if the limit has been reached
rtsp_server_auth_enabled	false	Boolean	false	If true, enable RTSP server authentication
rtsp_server_enabled	true	Boolean	true	If true, activate RTSP server
rtsp_server_forse_interleave	false	Boolean	false	If true, force interleaved mode for RTSP server and answer with interleaved mode SDP
rtsp_server_packetization_mode	null	String	false	H.264 packetization mode for RTSP server. FU-A by default
rtsp_server_profile_level_id	null	String	false	H.264 profile-level-id for RTSP server
rtsp_user_agent		String	false	User agent indicated in RTSP packets
rvg_timer_activity	500	Integer	false	RVG timer interval in milliseconds
rvg_timer_delay	500	Integer	false	RVG timer initial delay in milliseconds
scheduling_service_core_threads	5	Integer	true	Core threads count for scheduling service
send_receive_buffer_size	1600	Integer	true	RTMFP buffer size in bytes
send_receive_on_incoming_re_invite	true	Boolean	false	If true, send 'receive' on incoming re-INVITE

session_idle_timeout	300000	Integer	true	RTMFP server-side timeout in milliseconds if no UDP messages received over RTMFP/UDP session
set_sync_time_from_ts	false	Boolean	false	Workaround for SIP audio only
sip.pre_init	true	Boolean	true	If true, use SIP pre-initiation
sip_as_rtmp_java_client	true	Boolean	false	If true, then the latest RTMP transponder implementation will be used for as-RTMP cases. See also use_rtmp_java_client option
sip_as_rtmp_stream_type	live	String	false	Sets RTMP AMF stream type for as-RTMP cases
sip_auditor_dialog_timeout	10000	Integer	false	SIP auditor dialog timeout
sip_auditor_transaction_timeout	50000	Integer	false	SIP auditor transaction timeout
sip_dns_failover	false	Boolean	false	If true, enable DNS failover. See also sip_srv_lookup= option
sip_force_session_expires	1800	Integer	false	Forced session expiration timeout in seconds. WCS server will send refresh request before the timeout is reached
sip_force_tcp	false	Boolean	false	If true, force TCP usage for SIP messaging
sip_invite_params_to_headers	false	Boolean	false	If true, place SIP INVITE parameters to headers

sip_msg_listener	com.flashphoner.sdk.sip.ChangeCallIdListener	String	false	Full name of Java class that implements interface ISipMessageListener
				public interface ISipMessageListener {
				void processMessage(SIPMessage sipMessage);
				}

sip_ports_auditor_interval	10000	Integer	false	Audit interval for SIP ports, in milliseconds
sip_ports_auditor_max_attempts	3	Integer	false	Number of audits to make sure freed port is not bound. Freed SIP port will be returned to the pool of free ports if this number of successfull audits is reached
sip_record_stream	false	Boolean	false	If true, record SIP as RTMP stream and SIP as stream
sip_remove_video_sdp_section_instead_of_adding_inactive_with_zero_port	false	Boolean	false	If true, fully remove video part of SDP. If false, just set video part to inactive
sip_session_expires_header	true	Boolean	false	If true, use Expires header
sip_single_route_only	false	Boolean	false	If true, then traffic is passed only to the streaming engine, and is not passed to the SIP caller
sip_srv_lookup	false	Boolean	false	If true, enable DNS SRV lookup. See also sip_dns_failover= option
sip_thread_pool_size	null	String	false	Size of SIP thread pool
sip_timer	null	String	false	Value of timer T1 according to RFC 3261, in milliseconds
sip_traffic_class	null	String	false	QoS class for SIP traffic
sip_use_netty	false	Boolean	false	If true, use Netty
sip_use_reentrant_listener	false	Boolean	false	If true, enable SIP reentrant listener
sip_use_tls	false	Boolean	false	If true, TLS used for SIP connections
speex_g711_speex_transcoding	false	Boolean	false	If true, then Speex16 codec is forcedly deleted from the list of supported codecs, which leads to double transcoding. The option was used for debugging
speex_in_policy	null	String	false	Speex encoding settings used in transcoding featuring the codec. Default: 8 - Quality false - VBR encoding 8 - Quality of VBR 4 - Algorithmic complexity
start_test	false	Boolean	false	If true, tests listed in streaming_tests= setting will be launched after WCS server startup
stats	false	Boolean	true	If true, enable sampling for streams. The sampling is used for charts
stats_bitrate_window	1000	Integer	false	Window size to collect bitrate statistics
stats_fps_window	1000	Integer	false	Window size to collect FPS statistics
stats_incoming_stream_monitor_deviation_threshold	20	Integer	false	If deviation between audio and video is greater than the threshold in milliseconds, it will be logged
stats_sampling_frequency	1000	Long	true	Interval in milliseconds. Stream sampling will be taken with the specified frequency
stream_record_policy		String	false	Available values: streamName, template. By default, WCS server generates filename based on mediaSessionId and login. If set to 'streamName', recorded file will have the exact name of stream with extension .mp4 or .webM (depending on the video codec). If set to 'template', filename will be built using template. See also stream_record_policy_template= option
stream_record_policy_template		String	false	If set, name of recorded file will be built using the specified template. Example: {streamName}-{startTime}-{sessionId}-{mediaSessionId}-{login}-{audioCodec}-{videoCodec} Note that if filename length exceeds system limit, recording may be not created. See also stream_record_policy= option

streaming_custom_stream_stress_test_encoding_subscriber_groups	1	String	false	StreamingCustomStreamStressTest / Number of subscribers for transcoded stream, per encoding groups E.g., three encoding groups with three subscribers in each streaming_custom_stream_stress_test_encoding_subscriber_groups=3,3,3
streaming_custom_stream_stress_test_max_proxy_subscribers	1	Integer	false	StreamingCustomStreamStressTest / Number of subscribers for non-transcoded stream (codecs, resolution and bitrate are the same for publisher and subscriber)
streaming_custom_stream_stress_test_rate	1000	Long	false	StreamingCustomStreamStressTest / Period in milliseconds. Each period a new subscriber will be added
streaming_custom_stream_stress_test_stream_name	STRESS_TEST_STREAM	String	false	StreamingCustomStreamStressTest / Name of stream published on WCS server, which will be used for the test
streaming_custom_stream_stress_test_subscriber_ttl_sec	30	Long	false	StreamingCustomStreamStressTest / Lifetime of subscriber in seconds
streaming_distributor_queue_max_waiting_time	5000	Integer	true	Maximum time that distributor thread will wait for frame arrival before executing next iteration
streaming_distributor_queue_size	300	Integer	true	Size of queue. Processor will block distributor queue upon it reaching this size (i.e., no more space for new frames)
streaming_load_test_duration_minutes	5	Long	false	StreamingLoadTest / Test duration in minutes
streaming_load_test_encoding_subscriber_groups	1	String	false	StreamingLoadTest / Number of subscribers for transcoded stream, per encoding groups E.g., two encoding groups: one with two subscribers and another with five streaming_load_test_encoding_subscriber_groups =2,5
streaming_load_test_proxy_subscribers	1	Integer	false	StreamingLoadTest / Number of subscribers for non-transcoded stream (codecs, resolution and bitrate are the same for publisher and subscriber)
streaming_processor_queue_max_waiting_time	5000	Integer	true	Maximum time that processor thread will wait for frame arrival before executing next iteration
streaming_processor_queue_size	300	Integer	true	Size of queue. Feeding thread (e.g., RTP thread in case of WebRTC) will block processor queue upon it reaching this size (i.e., no more space for new frames)
streaming_sessions_keep_alive_app_keys		String	false	Comma-separated list of appKeys of server-side applications. If set, WCS server will periodically send StreamKeepAliveEvent for all streams within the listed applications. For example, if set 'defaultApp,myApp', the event will be sent for all streams connected to those two applications. See also streaming_sessions_keep_alive_interval= option
streaming_sessions_keep_alive_interval	10000	Long	false	StreamKeepAliveEvent sending interval. See also streaming_sessions_keep_alive_app_keys= option
streaming_stress_test_duration_minutes	5	Long	false	StreamingStressTest / Test duration in minutes
streaming_stress_test_encoding_subscriber_groups	1	String	false	StreamingStressTest / Number of subscribers for transcoded stream, per encoding groups E.g., five encoding groups with five or ten subscribers in each streaming_stress_test_encoding_subscriber_groups=5,5,5,10,10
streaming_stress_test_max_proxy_subscribers	1	Integer	false	StreamingStressTest / Number of subscribers for non-transcoded stream (codecs, resolution and bitrate are the same for publisher and subscriber)
streaming_stress_test_rate	1000	Long	false	StreamingStressTest / Period in milliseconds. Each period a new subscriber will be added
streaming_stress_test_subscriber_ttl_sec	30	Long	false	StreamingStressTest / Lifetime of subscriber in seconds
streaming_tests		String	false	Comma-separated list of tests which will be launched after WCS server startup if start_test=true. Available tests: - MP4AgentTest - StreamingCustomStreamStressTest - StreamingLoadTest - StreamingStressTest
streaming_video_decoder_fast_start	false	Boolean	false	If true, all incoming streams are decoded. If false, incoming stream is decoded only on demand, when codecs, resolution or bitrate are different for the stream publisher and subscriber
strict_get_callee_policy	false	Boolean	false	Not in use
stun_freshness_period	1500	Integer	false	STUN freshness period in milliseconds
stun_freshness_timeout	15000	Integer	false	STUN freshness timeout in milliseconds
stun_server	stun1.l.google.com:19302	String	false	STUN server, which is used for WebRTC ICE, if enable_candidate_harvester=true
stun_socket_buffer_size	100	Integer	false	Size of STUN socket buffer
stun_socket_queue_size	100	Integer	false	Size of STUN socket queue
stun_socket_queue_timeout	1500	Integer	false	STUN socket queue timeout in milliseconds
stun_stack_default_thread_pool_size	0	Integer	false	STUN default thread pool size

suppress_audio	false	Boolean	false	If true, globally suppress audio on server. This feature is not available for Trial license
suppress_dynamic_logs	false	Boolean	false	If true, suppress dynamic logs update
suppress_dynamic_logs_to_server_log	false	Boolean	false	If true, suppress dynamic server logs update
tcp_relay_packetization2	true	Boolean	false	If true, enable TCP relay packetization for WSPlayer. Should be false when WSPLayer 1.0 is used
tcp_relay_packetization_time	20	Integer	false	Experimental option, allows to send audio packets with custom ptime to WSPlayer 1.0. This property was not tested with new versions and should be removed
tcp_relay_rtcp_interval	2000	Integer	false	RTCP packets generation interval for TCP relay in milliseconds. RTCP is used to carry stream synchronization
thread_pool_default_core_threads	2	Integer	true	Default core threads count in thread pool (equal to CPUs count)
thread_pool_default_max_threads	4	Integer	true	Maximum core threads count in thread pool
thread_pool_default_queue_size	100	Integer	true	Default thread pool queue size
thread_pool_default_thread_timeout_sec	300	Integer	true	Default thread timeout, in seconds
throughput_test_receivers_qty	1	Integer	false	Throughput test receivers quantity
throughput_test_sender_dst	localhost	String	false	Throughput test sender destination host
throughput_test_senders_qty	1	Integer	false	Throughput test senders quantity
timing_shift	null	String	false	Timer ambiguity in milliseconds, which is used in a stream stagnation (in case the stream is too fast in relation to timestamps) and compensates inaccuracy of system timers. Is used only if in_jitter_buffer_enabled=true
trace_socket_fd	false	Boolean	true	If true, trace usage of socket file descriptors for HLS, HTTP, RTSP, WebSockets and HTTP LB client
transcoder_activity_timer_cool_off_period	1	Integer	false	Transcoder agent will be terminated after {rtsp_activity_timer_cool_off_period * rtsp_activity_timer_timeout} since last subscriber activity for the corresponding RTSP stream
transcoder_activity_timer_timeout	60000	Integer	false	If there is no subscribers for an Transcoder agent stream within this timeout in milliseconds, corresponding RTSP session will be terminated
transcoding_disabled	false	Boolean	false	Force transcoding disabling
turn.server_channel_receive_buffer_size	1048576	Integer	true	Receive buffer size for turn channels
turn.server_channel_send_buffer_size	1048576	Integer	true	Send buffer size for turn channels
turn_ip	null	String	true	TURN IP address
turn_life_time	600	Integer	true	TURN Allocation life time
turn_media_port_from	36001	Integer	true	Beginning of media ports range for turn
turn_media_port_to	37000	Integer	true	End of media ports range for turn
turn_media_ports_auditor_interval	5000	Integer	true	Audit interval for busy and free ports, in milliseconds
turn_media_ports_auditor_max_attempts	3	Integer	true	Number of audits to make sure freed port is not bound. Freed port will be returned to the pool of free ports if this number of successfull audits is reached
turn_password	coM77EMrV7Cwhyan	String	true	TURN password
turn_port	3478	Integer	true	TURN server port
unsupported_messages	null	String	false	If a message has body noted in this list, then such incoming message will be rejected. Can be useful for some service messages, when delivery to client is not required. The list consists of strings, divided by three colons :::
use_alaw_ulaw_speex_switch	true	Boolean	false	If true, switch to the local codec according to content received from SIP side. If false, use Speex16
use_control_destination_from_incoming_rtcp	true	Boolean	false	If true, set RTCP destination by received RTCP packets
use_fdk_aac	true	Boolean	false	If true, use the fdk-aac fro encoding and decoding
use_ip_local_in_call_id	true	Boolean	false	If true, use value of ip_local= option when forming callID
use_java_hls_writer	true	Boolean	false	If true, use Java HLS implementation
use_mp4_h264_aac	true	Boolean	false	If true, use H.264 + AAC in MP4 container
use_new_aac_encoder	true	Boolean	false	If true, use the latest AAC encoder
use_new_rtcp	true	Boolean	false	If true, use the latest RTCP module
use_rtcp_synch	true	Boolean	false	If true, use RTCP synchronization for audio and video
use_rtmp_java_client	true	Boolean	false	If true, use the latest implementation of RTMP agent for republishing
use_speex_java_impl	true	Boolean	true	If true, use Java implementation for Speex codec

use_tcp_for_long_sip_messages	false	Boolean	false	If true, and size of SIP message is more than 1350 bytes, then such message will be sent via TCP. By default, SIP messages are sent over UDP
use_trying_notification	false	Boolean	false	If true, then broadcast SIP response TRYING to client as a call status TRYING
user_agent	Flashphoner/1.0	String	true	User-Agent header value
video_distributor_multi_test	false	Boolean	false	Enable video distributor multi test
video_enabled	true	Boolean	false	Not in use
video_encoder_h264_gop	60	Integer	false	GOP size for H.264 encoder
video_encoder_second_thread_threshold	777000	Integer	false	Resolution threshold. Once it is reached, encoder should start using second thread. Example: 800x600 = 480000, 1280x720=921600. So, by default all 720p streams will be encoded using two CPU threads
video_encoder_vp8_gop	900	Integer	false	GOP size for VP8 encoder
video_encoding_quality	30	Integer	false	See information on FFmpeg CRF
video_filter_enable_fps	false	Boolean	true	Enable video filter
video_filter_enable_rotate	false	Boolean	true	Enable video rotate filter
video_filter_fps	30	Long	true	Video filter output fps
video_filter_fps_gap_coefficient	2.0	Double	true	Video filter gap coefficient (max gap C x FPS)
video_incoming_buffer_size	50	Integer	false	Waiting for RTCP sync packet on this interval in packets, for video
video_processor_multi_test	false	Boolean	false	Enable video processor multi test
video_reliable	partial	on partial off	false	RTMFP, reliability for video
video_stream_mode_udp	false	Boolean	true	Not in use
video_streamer_generate_seq	false	Boolean	false	Should be set to true for transfer of video calls. Otherwise, there may be no video after transfer
vod_activity_timer_cool_off_period	1	Integer	false	VOD agent will be terminated after {vod_activity_timer_cool_off_period * vod_activity_timer_timeout} since last subscriber activity for the corresponding RTSP stream
vod_live_loop	false	Boolean	false	If true, loop streaming MP4 file as VoD. EXPERIMENTAL
vod_mp4_container_new	false	Boolean	false	Use new implementation of mp4 container for vod
vod_mp4_test_file	null	String	false	Path to MP4 file. If start_test=true and streaming_tests=MP4AgentTest, VoD stream playing the file will be published when WCS server is started
vod_mp4_test_loop	true	Boolean	false	If true, loop streaming MP4 file. Not in use, replaced by vod_live_loop=
vod_mp4_test_stream_name	null	String	false	This name will be used as name of VoD stream published for playing MP4 file for test MP4AgentTest. See also vod_mp4_test_file= setting
vod_sink_ready_checks	50	Integer	false	Waiting for first packet on audio streamer. If no packets within the specified number of checks, then audio injection is aborted
vod_stream_timeout	30000	Integer	false	VoD stream with no subscribers will be terminated after this timeout in milliseconds
vow_wait_for_sync	false	Boolean	false	If true, session will wait for audio AND video before sending stream to client
vp8_buffer_nack_list_threshold	200	Integer	false	JitterBuffer will be reset upon reaching this number of NACK packets
vp8_max_rtp_packet_size	1400	Integer	true	Maximum size of VP8 carrying packet
vp8_new_buffer	false	Boolean	false	Not in use
wcs_activity_timer_cool_off_period	1	Integer	false	WCS agent will be terminated after {wcs_agent_activity_timer_cool_off_period * wcs_agent_activity_timer_timeout} since last activity for the corresponding WCS agent session
wcs_activity_timer_timeout	60000	Integer	false	If there is no activity within this timeout in milliseconds, corresponding WCS agent session will be terminated
wcs_agent_port_from	34001	Integer	false	Beginning of range of ports for WCS agent
wcs_agent_port_to	35000	Integer	false	End of range of ports for WCS agent
wcs_agent_ports_auditor_interval	10000	Integer	false	Audit interval for WCS agent ports, in milliseconds
wcs_agent_ports_auditor_max_attempts	3	Integer	false	Number of audits to make sure freed port is not bound. Freed WCS agent port will be returned to the pool of free ports if this number of successfull audits is reached
wcs_agent_session_alive_check_interval	30000	Integer	false	Interval in milliseconds to check if WCS agent session is alive
wcs_agent_session_audit	true	Boolean	false	If true, enable WCS agent session audit
wcs_agent_session_connect_timeout	10000	Integer	false	Connect timeout in milliseconds
wcs_agent_session_timeout	30000	Integer	false	WCS agent session timeout in milliseconds

wcs_agent_session_use_keep_alive_timeout	true	Boolean	true	If true, WCS agent session will use keep alive timeout
wcs_agent_ssl	false	Boolean	false	If true, enable SSL for pulling/re-publishing streams
webrtc_agent_use_webrtc	true	Boolean	false	If true, switch WebRTC push and pull to AVP profile
webrtc_cc2	true	Boolean	false	If true, the latest congestion control CC2 is used
webrtc_cc2_bitrate_overuse_event	true	Boolean	false	If true, enable NBE evant raising
webrtc_cc2_bitrate_overuse_event_interval	5000	Long	false	NBE event will be raised periodically with this interval in milliseconds
webrtc_cc2_bitrate_overuse_event_threshold	0.05	Double	false	NBE event will be raised when loss on stream being played reaches this value (5% by default)
webrtc_cc2_cc	false	Boolean	false	If true, react upon WebRTC playback endpoint (e.g. Chrome) requests, e.g. request the publisher to decrease bitrate
webrtc_cc2_cc_interval	500	Long	false	Congestion control interval, not in use
webrtc_cc2_cc_k_noise	0.1	Double	false	Congestion control noise value, not in use
webrtc_cc2_cc_retransmit_rate_threshold	0.15	Double	false	Fraction of send bitrate that retransmit bitrate can raise to. By default, retransmit bitrate can use 15% of send bitrate
webrtc_cc2_cc_track_joined_retransmit_bitrate	true	Boolean	false	If true, enable tracking of retransmit bitrate across all media groups
webrtc_cc2_enable_burst_grouping	false	Boolean	false	Internal bitrate estimation configuration, must not be exposed to public. CC2 estimation will account for packet burst
webrtc_cc2_local_congestion_event_interval	2000	Long	false	Not in use, legacy code
webrtc_cc2_local_k_threshold	0.1	Double	false	Not in use, legacy code
webrtc_cc2_min_remb_bitrate_bps	100000	Long	false	Minimum value for received REMB (Receiver Estimated Max Bitrate) boundary in bps. Ignore the boundary if the received value is less than the minimum defined
webrtc_cc2_receiver_state_window	1000	Long	false	Window size for receiver state, in milliseconds. Default: 1000 - keep and account reports received in last second
webrtc_cc2_twcc	false	Boolean	false	If true, enable TWCC reports. EXPERIMENTAL
webrtc_cc_bitrate_window	1000	Integer	false	Time window in milliseconds. Bitrate estimator works on this time frame
webrtc_cc_initial_avg_noise	0.0	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_e_0_0	100.0	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_e_0_1	0.0	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_e_1_0	0.0	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_e_1_1	0.1	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_offset	0.0	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_process_noise_0	1.0E-13	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_process_noise_1	0.001	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_slope	0.015625	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_threshold	15.0	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_initial_var_noise	50.0	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_k_down	1.8E-4	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_k_up	0.01	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_max_bitrate	10000000	Long	false	Maximum global bitrate for publishing WebRTC streams
webrtc_cc_min_bitrate	30000	Long	false	Minimum global bitrate for publishing WebRTC streams
webrtc_cc_overusing_threshold	10.0	Double	false	Internal bitrate estimation configuration, must not be exposed to public
webrtc_cc_use_sync_ts	true	Boolean	false	If true, timestamp is used as synchronization source
work_around	false	Boolean	false	Not in use
ws.address	0.0.0.0	InetAddress	true	Listening address for WebSocket server
ws.map_custom_headers	false	Boolean	true	If true, parse and inject custom HTTP headers to REST requests
ws.port	8080	Integer	true	WebSocket connection port
ws_connections_test_run_for	1800	Integer	true	Websocket connections test duration in seconds
ws_connections_test_uri	ws://192.168.88.100:8080	String	true	Websocket connections test URI
ws_read_socket_timeout	true	Boolean	true	Enable WebSocket read timeout
ws_read_socket_timeout_sec	120	Integer	true	WebSocket read timeout value (if enabled)
wss.address	0.0.0.0	InetAddress	true	Listening address for WebSocket SSL server
wss.cert.password	password	String	true	Key password to the SSL certificate in keystore

wss.keystore.file	/usr/local /FlashphonerWebCallServer/conf /wss.jks	String	true	Keystore file containing SSL certificate for secure WebSocket connection
wss.keystore.password	password	String	true	SSL certificate keystore password
wss.port	8443	Integer	true	WebSocket SSL connection port
wss.ssl.cache_size	0	Integer	true	SSL session objects cache size
wss.ssl.session_timeout	0	Integer	true	Cached SSL session objects timeout, in seconds