

# flashphoner.properties config file

Main server settings

## **aac\_bitrate**

*Default:*

128000

*Type:*

Integer

*Need restart:* No

*Description:*

AAC encoding bitrate

## **aac\_encoder\_sync\_drop\_threshold**

*Default:*

1000

*Type:*

Long

*Need restart:* Yes

*Description:*

JitterBuffer will be reset upon reaching this number of dropped sync packets

## **aac\_test\_start\_codec**

*Default:*

20

*Type:*

Integer

*Need restart:* Yes

*Description:*

AAC test codecs count

## **aac\_test\_transcode\_iterations**

*Default:*

1000

*Type:*

Integer

*Need restart:* Yes

*Description:*

AAC test interval

### **add\_register\_auth\_headers**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

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**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, enable local agent session debug

### **agent\_use\_subscriber\_listener**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, does the agent have to wait for the subscriber

### **allow\_domains**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

If set, then WebSocket connections from these domains only will be allowed

### **allow\_domains\_allow\_empty\_origin**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If set, then WebSocket connections with empty origin will be allowed

### **allow\_outside\_codecs**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If false, dont add outside (browser) codecs to SDP'

### **allow\_reinvite\_in\_hold\_state**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, process re-INVITE requests within the session even if the call is in hold state

### **allow\_stream\_names**

*Default:*

`*`

*Type:*

`String`

*Need restart:* No

*Description:*

If set, then client stream name from these stream names only will be allowed

### **answer\_with\_one\_codec\_in\_sdp**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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\_force\_sync\_timeout**

*Default:*

`100`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Waiting for RTCP sync packet on this interval in ms, for audio

### **audio\_frames\_per\_packet**

*Default:*

`6`

Type:

Integer

Need restart: No

Description:

RTMFP. Audio will be flushed after this number of audio frames in the packet is reached

### **audio\_incoming\_min\_buffer\_size**

Default:

2

Type:

Integer

Need restart: No

Description:

Waiting for RTCP sync packet at least on this interval in packets, for audio

### **audio\_mixer\_max\_delay**

Default:

300

Type:

Integer

Need restart: No

Description:

Audio mixer max delay in milliseconds

### **audio\_mixer\_output\_channels**

Default:

1

Type:

Integer

Need restart: No

Description:

Audio mixer output channels, default is 1 channel, mono

### **audio\_mixer\_output\_codec**

Default:

opus

Type:

`String`

Need restart: No

Description:

Audio mixer output codec (multiple codecs not allowed)

### **audio\_mixer\_output\_sample\_rate**

Default:

`48000`

Type:

`Integer`

Need restart: No

Description:

Audio mixer output sample rate in Hz

### **audio\_reliable**

Default:

`partial`

Type:

`on`

`partial`

`off`

Need restart: No

Description:

RTMFP, reliability for audio

### **audio\_stream\_mode\_udp**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

`true`

Type:

`Boolean`

*Need restart:* Yes

*Description:*

Not in use

### **auto\_login\_url**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Not in use

### **av\_paced\_sender**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable paced sender for output stream. EXPERIMENTAL

### **av\_paced\_sender\_max\_buffer\_size**

*Default:*

`5000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Max size of audio or video buffer. Once size is reached buffers are cleared

### **avatar\_dir**

*Default:*

`avatar`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Avatar base folder

### **avcc\_buffer\_wait\_frames\_count**

*Default:*

`5`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Wait until the buffer is filled with frames

### **avcc\_send\_buffer\_size**

*Default:*

`500000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Avcc send buffer size in bytes

### **aws\_s3\_credentials**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

AWS s3 credentials: region;accessKey;secretKey

### **balance\_header**



*Default:*

`balance`

*Type:*

`String`

*Need restart:* No

*Description:*

This SIP header will be sent to client as a balance

### **burst\_avoidance\_count**

*Default:*

`100`

*Type:*

`String`

*Need restart:* No

*Description:*

Burst avoidance count

### **busy\_state**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

*Default:*

`com.flashphoner.server.client.DefaultCallRecordListener`

*Type:*

`String`

*Need restart:* No

*Description:*

Full name of Java class that implements interface ICallRecordListener

```
public interface ICallRecordListener {  
    void onRecordReport(RecordReport recordReport);  
}
```

### **case\_sensitive\_auth\_match**

*Default:*

**true**

*Type:*

**Boolean**

*Need restart:* No

*Description:*

If false, ignore case on url auth

### **cdn\_advertise\_pulled**

*Default:*

**false**

*Type:*

**Boolean**

*Need restart:* Yes

*Description:*

If true, pulls CDN advertise

### **cdn\_advertise\_streams\_by\_kframe**

*Default:*

**false**

*Type:*

**Boolean**

*Need restart:* No

*Description:*

Advertise stream to CDN by key frame

### **cdn\_allowed\_ips**

*Default:*

**|**

Type:

`ArrayList`

Need restart: Yes

Description:

Comma-separated list of allowed IPs or networks for CDN.

Example: 88.198.98.1/24, 88.198.99.219

### **cdn\_auto\_pull**

Default:

`false`

Type:

`Boolean`

Need restart: Yes

Description:

Pull CDN stream once it becomes available

### **cdn\_connection\_quality\_calculation\_timeout\_ms**

Default:

`10000`

Type:

`Integer`

Need restart: Yes

Description:

Connection quality calculation update timeout ms

### **cdn\_connection\_tcp\_no\_delay**

Default:

`true`

Type:

`Boolean`

Need restart: Yes

Description:

Turns on tcp no delay for CDN signalling connections

### **cdn\_controller\_request\_timeout**

*Default:*

5000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Timeout for requests sent to CDN controller

### **cdn\_controller\_response\_cache\_expire**

*Default:*

10000

*Type:*

Integer

*Need restart:* Yes

*Description:*

TTL for cached records received from CDN controller

### **cdn\_dtls\_force\_version\_0**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Force DTLS version 1.0 inside cdn

### **cdn\_enabled**

*Default:*

false

*Type:*

Boolean

*Need restart:* Yes

*Description:*

If true, enables CDN

### **cdn\_force\_version**

*Default:*

`2.0`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Force to set CDN version

### **cdn\_group\_origin\_to\_transcoder\_relation**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Use CDN group indications to relate origin to transcoder rather than transcoder to edge

### **cdn\_groups**

*Default:*

`[]`

*Type:*

`ArrayList`

*Need restart:* Yes

*Description:*

CDN groups for this node

### **cdn\_inbound\_auditor\_interval**

*Default:*

`1000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Time interval to check inbound connections, in milliseconds

### **cdn\_inbound\_connection\_unanswered\_pings**

*Default:*

`3`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Inbound connection unanswered pings number.  
Connection considered to be lost when this number is reached

### **cdn\_inbound\_ws\_read\_socket\_timeout**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Enable WebSocket read timeout for inbound cdn connections

### **cdn\_inbound\_ws\_read\_socket\_timeout\_sec**

*Default:*

`60`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

WebSocket read timeout value (if enabled) for inbound cdn connections

### **cdn\_ip**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

CDN node IP address (or domain name when `cdn_nodes_resolve_ip=true`)

### **cdn\_load\_interval**

*Default:*

**500**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

load interval

### **cdn\_load\_node**

*Default:*

**false**

*Type:*

**Boolean**

*Need restart:* Yes

*Description:*

Turn on cdn load behaviour

### **cdn\_load\_pool\_size**

*Default:*

**500**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

load pool

### **cdn\_load\_pool\_size\_change\_interval**

*Default:*

**-1**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

Change pool size every interval

### **cdn\_load\_pool\_size\_lower\_threshold**

*Default:*

**-1**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

Lower threshold for pool size change

### **cdn\_load\_pool\_size\_upper\_threshold**

*Default:*

**-1**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

Upper threshold for pool size change

### **cdn\_load\_proto\_pull**

*Default:*

**websocket**

*Type:*

**String**

*Need restart:* Yes

*Description:*

CDN load protocol stream

### **cdn\_load\_reserved\_stream**

*Default:*

**1**

*Type:*

**String**

*Need restart:* Yes

*Description:*

CDN load reserved stream



### **cdn\_load\_step**

*Default:*

`10`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

load step

### **cdn\_load\_unique\_streams**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Pull only unique streams

### **cdn\_load\_use\_profile\_name**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Put profile name in stream name. Use if entry point is edge

### **cdn\_load\_use\_profiles**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Pull with profiles

### **cdn\_node\_load\_average\_threshold**

*Default:*

1.0

*Type:*

Double

*Need restart:* Yes

*Description:*

If threshold reached node will advertise it's state as GROUP\_CONNECTIONS

### **cdn\_nodes\_acl\_refresh\_interval**

*Default:*

60000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Time interval to refresh CDN node acl list, in milliseconds

### **cdn\_nodes\_auditor\_interval**

*Default:*

1000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Time interval to check available CDN nodes, in milliseconds

### **cdn\_nodes\_group\_refresh\_interval**

*Default:*

60000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Time interval to refresh CDN node group, in milliseconds

### **cdn\_nodes\_on\_single\_server**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, cdn nodes can be located on single server

### **cdn\_nodes\_resolve\_ip**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, resolve CDN node domain names to IP addresses

### **cdn\_nodes\_role\_refresh\_interval**

*Default:*

`60000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Time interval to refresh CDN node role, in milliseconds

### **cdn\_nodes\_route\_refresh\_interval**

*Default:*

`60000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Time interval to refresh CDN routes, in milliseconds

### **cdn\_nodes\_state\_refresh\_interval**

*Default:*

`60000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Time interval to refresh CDN node state, in milliseconds

### **cdn\_nodes\_timeout**

*Default:*

`-1`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

CDN nodes timeout in seconds. -1 means nodeTimeout disabled

### **cdn\_nodes\_version\_refresh\_interval**

*Default:*

`90000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Time interval to refresh CDN node version, in milliseconds

### **cdn\_origin\_allowed\_to\_transcode**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

In case no transcoders left node will request transcoding profile from origin

### **cdn\_origin\_to\_origin\_route\_propagation**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, origin sends routes to other origins

### **cdn\_outbound\_auditor\_interval**

*Default:*

`2000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Time interval to check outbound connections, in milliseconds

### **cdn\_outbound\_connection\_timeout**

*Default:*

`6000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Outbound connection timeout, in milliseconds

### **cdn\_outbound\_ws\_read\_socket\_timeout**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Enable WebSocket read timeout for outbound cdn connactions

### **cdn\_outbound\_ws\_read\_socket\_timeout\_sec**

*Default:*

60

*Type:*

Integer

*Need restart:* Yes

*Description:*

WebSocket read timeout value (if enabled) for outbound cdn connections

### **cdn\_point\_of\_entry**

*Default:*

1

*Type:*

String

*Need restart:* Yes

*Description:*

CDN point of entry node IP address (or domain name when cdn\_nodes\_resolve\_ip=true)

### **cdn\_port**

*Default:*

8084

*Type:*

Integer

*Need restart:* Yes

*Description:*

CDN server port

### **cdn\_role**

*Default:*

EDGE

*Type:*

ORIGIN

EDGE

TRANSCODER

CONTROLLER

*Need restart:* Yes

*Description:*

CDN role:

origin - the source of media streams for other CDN nodes

edge (default) pulls media streams from origin CDN node(s)

### **cdn\_role\_strict**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, sets restrictions on publishing streams by role

### **cdn\_role\_strict\_stream\_name**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Reserved name for setting cdn\_role\_strict bypass

### **cdn\_skip\_pulled\_streams**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, skip pulled streams

### **cdn\_ssl**

*Default:*

`false`

Type:

Boolean

Need restart: Yes

Description:

If true, enables SSL

### **cdn\_standalone**

Default:

false

Type:

Boolean

Need restart: Yes

Description:

If true, start server in CDN standalone mode, streaming will not available

### **cdn\_strict\_transcoding\_boundaries**

Default:

false

Type:

Boolean

Need restart: Yes

Description:

Prevent transcoding to the same or higher resolution of original stream by placing resolution boundary

### **cdn\_strict\_transcoding\_throws\_exception**

Default:

false

Type:

Boolean

Need restart: Yes

Description:

Whether to fail play or substitute requested profile with original stream if profile hit the strict transcoding boundary

### **cdn\_test\_enabled**



*Default:*

false

*Type:*

Boolean

*Need restart:* Yes

*Description:*

Turn on cdn tests

### **cdn\_test\_interval**

*Default:*

500

*Type:*

Integer

*Need restart:* Yes

*Description:*

test interval

### **cdn\_test\_max\_subscribers\_for\_stream**

*Default:*

10

*Type:*

Integer

*Need restart:* Yes

*Description:*

Max subscribers for each CDN stream. Edge-only setting

### **cdn\_test\_pool\_size**

*Default:*

500

*Type:*

Integer

*Need restart:* Yes

*Description:*

test pool

### **cdn\_test\_step**

*Default:*

10

*Type:*

Integer

*Need restart:* Yes

*Description:*

test step

### **cdn\_transcoder\_degraded\_streams\_threshold**

*Default:*

-1

*Type:*

Integer

*Need restart:* Yes

*Description:*

If threshold reached node will advertise it's state as GROUP\_CONNECTIONS. Threshold is the percent of degraded streams

### **cdn\_transcoder\_for\_new\_connects\_expire**

*Default:*

10000

*Type:*

Integer

*Need restart:* Yes

*Description:*

CDN transcoder cache expire for new stream requests

### **cdn\_transcoder\_threshold\_state**

*Default:*

GROUP\_CONNECTIONS\_ALLOWED

*Type:*

UNKNOWN

PASSIVE

GROUP\_CONNECTIONS\_ALLOWED

CONNECTIONS\_ALLOWED

NEW\_STREAMS\_ALLOWED

*Need restart:* Yes

*Description:*

If threshold reached node will change state to provided value

#### **cdn\_transcoder\_video\_decoders\_load\_threshold**

*Default:*

**-1**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

If decoders load threshold reached node will advertise it's state as GROUP\_CONNECTIONS.

Threshold is the sum of  $decoderWidth \cdot decoderHeight \cdot decoderFPS$

#### **cdn\_transcoder\_video\_encoders\_load\_threshold**

*Default:*

**-1**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

If encoders load threshold reached node will advertise it's state as GROUP\_CONNECTIONS.

Threshold is the sum of  $encoderWidth \cdot encoderHeight \cdot encoderFPS$

#### **cdn\_transcoder\_video\_encoders\_threshold**

*Default:*

**10000**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

If threshold reached node will advertise it's state as GROUP\_CONNECTIONS

#### **cdn\_transport**

*Default:*

**udp**

Type:

String

Need restart: Yes

Description:

CDN internal transport (udp, tcp, srt)

### **chat\_listener**

Default:

null

Type:

String

Need restart: No

Description:

Full name of Java class that implements interface IChatListener

```
public interface IChatListener {  
    void onMessage(InstantMessage message);  
}
```

### **check\_receiver\_origin**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, check origin of RTCP packet and discard if unknown

### **cli.address**

Default:

localhost

Type:

String

Need restart: Yes

Description:

Listening address for CLI SSH server

### **cli\_auth\_keys**

*Default:*

`/usr/local/FlashphonerWebCallServer/.ssh/authorized_keys`

*Type:*

`String`

*Need restart:* Yes

*Description:*

CLI client auth keys file path

### **cli\_auth\_server\_keys**

*Default:*

`/usr/local/FlashphonerWebCallServer/conf/cli-hostkey.pem`

*Type:*

`String`

*Need restart:* Yes

*Description:*

CLI host identification key file path

### **cli\_enabled**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, enables CLI

### **cli\_port**

*Default:*

`2001`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

CLI server port

### **cli\_v2.address**

*Default:*

`localhost`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Listening address for CLI\_V2 SSH server

### **cli\_v2\_port**

*Default:*

`2002`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

CLI V2 server port

### **client\_acl\_property\_name**

*Default:*

`aclAuth`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Access list identifier property key that server should look for in custom config when client connects

### **client\_appkey\_property\_name**

*Default:*

`appKey`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Property name to get application name used in authentication

### **client\_dump\_level**

*Default:*

`0`

*Type:*

`Integer`

*Need restart:* No

*Description:*

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**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Not in use

### **client\_log\_exclude**

*Default:*

`■`

*Type:*

`String`

*Need restart:* No

*Description:*

Do not log events listed

### **client\_log\_force\_debug**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable client logs for every newly connected client for a period of time specified by client\_log\_force\_debug\_timeout regardless of other settings

### **client\_log\_force\_debug\_timeout**

*Default:*

`60`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Timeout after which client logs will be turned off

### **client\_log\_level**

*Default:*

`INFO`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

*Default:*

`true`

*Type:*

`Boolean`



*Need restart:* No

*Description:*

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\_subscribe\_streams\_max**

*Default:*

10

*Type:*

Integer

*Need restart:* No

*Description:*

Max subscribe streams allowed for client

#### **client\_timeout**

*Default:*

3600000

*Type:*

Integer

*Need restart:* No

*Description:*

Client timeout value in milliseconds

#### **client\_timeout\_check\_interval**

*Default:*

300000

*Type:*

Integer

*Need restart:* No

*Description:*

Client timeout interval value in milliseconds

#### **codec\_terminator\_timeout**

*Default:*

5000

Type:

Integer

Need restart: No

Description:

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**

Default:

null

Type:

String

Need restart: No

Description:

List of supported codecs ordered by priority

### **codecs\_exclude\_cdn**

Default:

null

Type:

String

Need restart: No

Description:

Comma-separated list of codecs which will not be used for CDN

### **codecs\_exclude\_sfu**

Default:

alaw, ulaw, g729, speex16, g722, mpeg4-generic, telephone-event, h265, flv, mpv, vp9, h264

Type:

String

Need restart: No

Description:

Comma-separated list of codecs which will not be used for SIP as RTMP case

### **codecs\_exclude\_sip**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Comma-separated list of codecs which will not be used for SIP phone cases

### **codecs\_exclude\_sip\_rtmp**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Comma-separated list of codecs which will not be used for SIP as RTMP case

### **codecs\_exclude\_streaming**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Comma-separated list of codecs which will not be used for streaming

### **compact\_media\_port\_usage**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Use odd media ports for transferring data (requires rtcpMux)

### **complex\_test\_config**

*Default:*

`1`

*Type:*

`String`

*Need restart:* No

*Description:*

Complex transcoder test configuration

### **complex\_test\_decode**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable decoding during complex transcoding test

### **complex\_test\_fps**

*Default:*

`15`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Complex transcoder test FPS

### **complex\_test\_replay**

*Default:*

`3`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Complex transcoder test repeats count

### **complex\_test\_thread**

*Default:*

`3`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Complex transcoder test threads count

### **core\_standalone\_web\_dir**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Web directory for standalone mode

### **cost\_header**

*Default:*

`cost`

*Type:*

`String`

*Need restart:* No

*Description:*

This SIP header will be sent to client as a call cost

### **cps\_client**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

*Default:*

1000

*Type:*

Long

*Need restart:* No

*Description:*

Time window for measuring CPS, in milliseconds

### **cps\_node**

*Default:*

2147483647

*Type:*

Integer

*Need restart:* No

*Description:*

Global CPS limitation for node

### **cpu\_load\_avg\_size**

*Default:*

20

*Type:*

Integer

*Need restart:* Yes

*Description:*

CPU load average size

### **cpu\_load\_refresh**

*Default:*

50

*Type:*

Integer

*Need restart:* Yes

*Description:*

CPU load refresh rate

### **cpu\_load\_reject**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, reject streams when CPU load exceeds treshold

### **cpu\_load\_threshold**

*Default:*

`80`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

CPU load treshold

### **cpu\_load\_window**

*Default:*

`2000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Timeslice to estimate CPU load

### **custom\_ice\_agent**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`true`

Type:

`Boolean`

Need restart: No

Description:

If true, use custom ICE agent

### **custom\_stats\_script**

Default:

`1`

Type:

`String`

Need restart: No

Description:

Script can be used to provide custom stat params with action=stat request

### **custom\_watermark\_filename**

Default:

`null`

Type:

`String`

Need restart: No

Description:

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

### **custom\_watermark\_mix**

Default:

`false`

Type:

`Boolean`

Need restart: No

Description:

Enables watermark mixing for alpha-layer

### **data\_packet\_decoder\_fire\_null\_messages**



*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, pass special data packet up the RTP process chain when original received data failed to decode

### **decoded\_frame\_interceptor**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Full name of Java class that implements interface IDecodedFrameInterceptor. This class should be wrapped to .jar file placed in /usr/local/FlashphonerWebCallServer/lib directory

### **decoded\_pcm\_interceptor**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Full name of Java class that implements interface IDecodedPcmInterceptor. This class should be wrapped to .jar file placed in /usr/local/FlashphonerWebCallServer/lib directory

### **decoder\_binary\_log\_enable**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Binary log decoder

### **decoder\_binary\_log\_size**

*Default:*

`5`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Binary log decoder size

### **decoder\_buffer\_pool**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Enable buffer pool usage during video decoding

### **decoder\_buffer\_pool\_stats**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable buffer pool stats, might slow down video transcoding

### **decoder\_mode**

*Default:*

`JNI`

*Type:*

`QUEUE`

`JNI`

*Need restart:* No

*Description:*

Decoder mode

### **decoder\_priority**

*Default:*

`FF, OPENH264`

*Type:*

`String`

*Need restart:* No

*Description:*

Decoder priority

### **decoder\_stat\_log**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable decoder statistics logging

### **default\_packetization\_mode**

*Default:*

`0`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Packetization mode default value if incoming SDP doesn't contains packetization\_mode parameter.

### **default\_sdp\_state**

*Default:*

`sendrecv`

Type:

String

Need restart: No

Description:

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**

Default:

20

Type:

Integer

Need restart: Yes

Description:

Degraded streams threshold

### **degraded\_streams\_window**

Default:

2000

Type:

Integer

Need restart: Yes

Description:

Timeslice to estimate stream degradation

### **delta\_threshold**

Default:

100

Type:

Integer

Need restart: No

Description:

RTMFP. If delta between UDP media packets is greater than the threshold, it will be reported

### **depacketizer\_dump\_dir**

*Default:*

`depacketizer_dump_dir`

*Type:*

`String`

*Need restart:* Yes

*Description:*

H264 RTP stream dump base folder

### **detect\_flash\_2\_flash\_calls**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, WCS server will use an RTP extension header in RTP packets, which can be used for designation of WCSs own streams, even if they are traced through third-party PBX, e.g. Asterisk'

### **disable\_drop\_aac\_frame**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, disables dropping AAC frames

### **disable\_manager\_rmi**

*Default:*

`true`

Type:

Boolean

Need restart: Yes

Description:

If true, disables RMI communications between WCS Core and WCS Manager

### **disable\_rest\_auth**

Default:

true

Type:

Boolean

Need restart: No

Description:

If true, disables authorization in rest api

### **disable\_rest\_requests**

Default:

false

Type:

Boolean

Need restart: Yes

Description:

If true, disables Rest requests to application

### **disable\_rtc\_ata**

 **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

null

Type:

String

Need restart: No

*Description:*

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**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, disables RTC ATA (see above)

**disable\_streaming\_proxy**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, disable proxy and enable transcoding for all streams. For debug only

**disable\_streaming\_proxy\_aac**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If false, enable AAC proxying

### **dns\_test\_enable**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Enable dns test statistic

### **dns\_test\_name**

*Default:*

`''`

*Type:*

`String`

*Need restart:* Yes

*Description:*

DNS name for time resolved statistic

### **domain**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

SIP domain. If this parameter is set, it will redefine values that were transmitted during connection

### **dtls0\_ua\_match\_substring**

*Default:*

`false`

*Type:*

`Boolean`



*Need restart:* No

*Description:*

If true, DTLS User-Agent matching will be by substring. Ex: Chrome/70.0

### **dtls\_close\_socket\_after\_tries**

*Default:*

10

*Type:*

Integer

*Need restart:* No

*Description:*

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\_force\_version\_0**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Force DTLS version 1.0

### **dtls\_message\_timeout**

*Default:*

15

*Type:*

Integer

*Need restart:* No

*Description:*

DTLS handshake timeout in seconds, must be set to a non-zero value

### **dtls\_socket\_timeout\_ms**

*Default:*

1000

Type:

Integer

Need restart: No

Description:

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**

Default:

true

Type:

Boolean

Need restart: No

Description:

If true, enable DTLS socket SO\_TIMEOUT

#### **dtmf**

Default:

null

Type:

String

Need restart: No

Description:

This type will be used if DTMF type (INFO, INFO\_RELAY, RFC2833) was not specified when DTMF was sent

#### **dump\_avcc\_relay**

Default:

false

Type:

Boolean

Need restart: No

Description:

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**

### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, gather ICE candidates using external STUN server

## **enable\_empty\_shift\_writer**

### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable empty shift writer for conference

## **enable\_extended\_logging**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Enable flight recorder

### **enable\_flight\_recorder\_test**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Enable flight recorder test

### **enable\_local\_videochat**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Not in use

### **enable\_network\_address\_cache**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Enable cache for resolved addresses.

### **enable\_new\_client\_logger**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable new client logger

### **enable\_rtc\_video\_generator**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable audit of SIP stack

### **enable\_sync\_time\_normalizer**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, then enable sync time normalizer

### **encode\_record\_name**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Encode record name setting

### **encoder\_buffer\_length\_sec**

*Default:*

`1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Encoding buffer for audio, in seconds

### **encoder\_default\_video\_resolution**

*Default:*

`640x480`

*Type:*

`String`

*Need restart:* No

*Description:*

encoder\_default\_video\_resolution

### **encoder\_mode**

*Default:*

JNI

*Type:*

QUEUE

JNI

*Need restart:* No

*Description:*

Encoder mode

### **encoder\_priority**

*Default:*

FF, OPENH264

*Type:*

String

*Need restart:* No

*Description:*

Encoder priority

### **encoder\_stat\_log**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Enable encoder statistics logging

### **event\_scanner\_cached\_pool**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, use event scanner cached pool

### **event\_scanner\_pool\_size**

*Default:*

10

*Type:*

Integer

*Need restart:* No

*Description:*

Event scanner pool size

### **exclude\_record\_name\_characters**

*Default:*

null

*Type:*

String

*Need restart:* Yes

*Description:*

Exclude characters from record name

### **fetch\_caller\_from\_pai**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

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**

*Default:*

false

*Type:*

Boolean



*Need restart:* No

*Description:*

If true, fetch caller from PAI from' when caller is empty'

### **file\_recorder\_error\_interval**

*Default:*

60

*Type:*

Integer

*Need restart:* No

*Description:*

Error counter's interval in minutes

### **file\_recorder\_max\_errors\_per\_interval**

*Default:*

3

*Type:*

Integer

*Need restart:* No

*Description:*

Max errors per interval

### **file\_recorder\_min\_space**

*Default:*

1g

*Type:*

String

*Need restart:* No

*Description:*

Minimum available disk space for recording in GiB(G|g), MiB(M|m) or KiB(K|k). By default, GiB is used if no suffix specified

### **file\_recorder\_thread\_pool\_max\_size**

*Default:*

4

Type:

Integer

Need restart: Yes

Description:

Maximum core threads count in record thread pool

### **file\_recorder\_thread\_queue\_initial\_size**

Default:

50

Type:

Integer

Need restart: Yes

Description:

Initial size of queue of samples in record thread pool

### **flash\_codecs**

#### Warning

Deprecated parameter. Will be deleted in future releases

Default:

alaw, ulaw, speex16, h264, vp8

Type:

String

Need restart: No

Description:

This set of codecs (if it is not empty) will be used if either party of a call is Flash

### **flash\_detect\_metadata\_by\_traffic**

Default:

true

Type:

Boolean

Need restart: No

*Description:*

If true try to detect flash SDP by incoming traffic

### **flash\_detect\_metadata\_by\_traffic\_timeout**

*Default:*

1000

*Type:*

Integer

*Need restart:* No

*Description:*

Traffic metadata waiting time (ms), if no metadata has been received after this time, the media (video or audio) will be excluded from the SDP.

### **flash\_handler\_play\_sdp\_filename**

*Default:*

flash\_handler\_play.sdp

*Type:*

String

*Need restart:* Yes

*Description:*

Filename of RTMP subscriber sdp

### **flash\_handler\_publish\_sdp\_filename**

*Default:*

flash\_handler\_publish.sdp

*Type:*

String

*Need restart:* Yes

*Description:*

Filename of RTMP publisher sdp

### **flash\_policy.port**

*Default:*

843

*Type:*

Integer

*Need restart:* Yes

*Description:*

Listening port for flash policy requests to crossdomain.xml file

### **flash\_rtp\_activity\_enabled**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, enable RTP activity for Flash streams

### **flash\_streaming\_enable**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

Not in use

### **flight\_recorder\_capacity**

*Default:*

500

*Type:*

Integer

*Need restart:* No

*Description:*

Flight recorder's buffer capacity in records

### **flight\_recorder\_categories**

*Default:*

NONE

*Type:*

NONE

WCS1438

*Need restart:* Yes

*Description:*

Flight recorder categories

### **flush\_audio\_interval**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

80

*Type:*

Integer

*Need restart:* Yes

*Description:*

RTMFP flush interval in milliseconds for flash-audio data from server

### **flush\_video\_interval**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

0

*Type:*

Integer

*Need restart:* Yes

*Description:*

RTMFP flush interval in milliseconds for flash-video data from server

### **force\_client\_requested\_video\_resolution**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, use client-specified resolution passed in Stream object

### **force\_expires**

*Default:*

`-1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

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**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, FIR request will be sent to SIP endpoint every 5 seconds

### **force\_profile\_level**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Forced codec for old as-RTMP cases using RTMPOutputWriter and for the latest HLS writer

### **force\_sendrecv\_for\_outgoing\_calls**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, force sendrecv' for audio and video for outgoing SIP calls'

### **frame\_cnt\_to\_determine\_their\_type**

*Default:*

`10`

*Type:*

`Integer`

*Need restart:* No

*Description:*

How long to wait for frames to determine their type

### **frep\_database\_address**

*Default:*

`jdbc:mysql://localhost/wcs?user=wcs&password=wcs`

*Type:*

`String`

*Need restart:* No

*Description:*

Address of database that will be used for FREP data storing

### **frep\_enabled**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enables Flashphoner remote event protocol

### **frep\_filter\_events**

*Default:*

`CONNECT, CONNECTION_STATUS_EVENT, STREAM, CONNECTION_STATUS_EVENT`

*Type:*

`ArrayList`

*Need restart:* Yes

*Description:*

List of allowed events, which client can send and server can handle

### **frep\_port**

*Default:*

`8085`

*Type:*

`Integer`



*Need restart:* No

*Description:*

FREP port

### **frep\_role**

*Default:*

`CLIENT`

*Type:*

`CLIENT`

`SERVER`

*Need restart:* No

*Description:*

Role of the frep stack, client or server

### **frep\_secret\_key**

*Default:*

`dsjfoiewqhriywqtrfewfiuewqiufh`

*Type:*

`String`

*Need restart:* No

*Description:*

Secret key for FREP authentication

### **frep\_server\_ip**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Address of FREP server. Has no effect in server mode.

### **generate\_av\_for\_ua**

*Default:*

`null`

Type:

String

Need restart: Yes

Description:

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

### generate\_av\_start\_delay

Default:

0

Type:

Integer

Need restart: Yes

Description:

Generator start delay in ms, 0 - no delay

### get\_callee\_url

#### Warning

Deprecated parameter. Will be deleted in future releases

Default:

null

Type:

String

Need restart: No

Description:

Not in use

### global\_bandwidth\_check\_enabled

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, enable global bitrate in and out in statistics

### **gpu\_load\_switch\_encoding\_percentage\_limit**

Default:

80

Type:

Integer

Need restart: No

Description:

Switch to another GPU or CPU if GPU has reached it's encoding load limit

### **gpu\_load\_switch\_min\_available\_memory\_limit**

Default:

734003200

Type:

Long

Need restart: No

Description:

Switch to another GPU or CPU if GPU has reached it's memory limit

### **gpu\_load\_switch\_overall\_percentage\_limit**

Default:

80

Type:

Integer

Need restart: No

Description:

Switch to another GPU or CPU if GPU has reached it's overall load limit

### **gpu\_max\_encoding\_sessions\_per\_gpu**

Default:

0

Type:

Integer

Need restart: No

Description:

How many sessions can open on a GPU

### **gpu\_max\_encoding\_sessions\_per\_host**

Default:

-1

Type:

Integer

Need restart: No

Description:

How many concurrent gpu sessions supported on host

### **h264\_allowed\_nal\_types**

Default:

1, 5, 7, 8, 12

Type:

String

Need restart: No

Description:

List of NAL unit types allowed for decoding

### **h264\_b\_frames\_force\_transcoding**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, force transcoding by higher profile

### **h264\_buffer\_nack\_list\_threshold**

Default:

30

Type:

Integer

Need restart: No

Description:

JitterBuffer will be reset upon reaching this number of NACK packets

#### **h264\_buffer\_reset\_on\_flush\_indicator**

Default:

true

Type:

Boolean

Need restart: No

Description:

Clear h264 buffer state upon flush indication

#### **h264\_encoder\_filler\_data\_padding**

Default:

false

Type:

Boolean

Need restart: No

Description:

Fill frames with Filler Data NAL Units to always maintain max bitrate

#### **h264\_encoder\_rc\_buffer\_size**

Default:

2

Type:

Integer

Need restart: No

Description:

Coefficient for rc buffer

#### **h264\_max\_nalu\_size**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`1346`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

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**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Not in use

### **h264\_remove\_filler\_data**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enables condition to remove FILLER\_DATA from h264 bitstream

### **h264\_sps\_buff\_scale**

*Default:*

1.6

*Type:*

Double

*Need restart:* No

*Description:*

Buffer scale for H264 SPS

### **h264\_sps\_default\_size**

*Default:*

100

*Type:*

Integer

*Need restart:* No

*Description:*

Default size of H264 sps buffer

### **h264\_sps\_max\_dec\_frame\_buffering**

*Default:*

-1

*Type:*

Integer

*Need restart:* No

*Description:*

SPS VUI decoder buffer

### **h264\_sps\_rbsp\_scale**

*Default:*

1.5

*Type:*

Double

*Need restart:* No

*Description:*

Buffer scale for H264 SPS RBSP

### **h264\_strict\_kframe\_detect**

*Default:*

false

*Type:*

Boolean

*Need restart:* Yes

*Description:*

If true, set frame as keyframe only if contains SPS and PPS NAL units or IDR NAL

### **h265\_buffer\_nack\_list\_threshold**

*Default:*

30

*Type:*

Integer

*Need restart:* No

*Description:*

JitterBuffer will be reset upon reaching this number of NACK packets

### **h265\_buffer\_reset\_on\_flush\_indicator**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

Clear h265 buffer state upon flush indication

### **h265\_max\_rtp\_packet\_size**

*Default:*

1400

*Type:*

Integer

*Need restart:* No

*Description:*

Maximum size of H265 carrying packet

### **handler\_async\_disconnect**



*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable asynchronous disconnect handler

### **hangup\_incoming\_call\_state**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, do not send SIP messages to browser

### **hls.address**

*Default:*

`0.0.0.0`

*Type:*

`InetAddress[]`

*Need restart:* Yes

*Description:*

Listening address for HLS server

### **hls.http.port**

*Default:*

`8082`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

HLS server HTTP port

### **hls.https.port**

*Default:*

`8445`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

HLS server HTTPS port

### **hls\_abr\_auto\_start**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable HLS ABR autostart

### **hls\_abr\_enabled**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable ABR and master playlist for HLS

### **hls\_abr\_path\_template**

*Default:*

```
{streamName}{abrSuffix}/{streamName}{abrSuffix}.m3u8
```

*Type:*

```
String
```

*Need restart:* Yes

*Description:*

Template for HLS ABR streams path

### **hls\_abr\_stream\_name\_suffix**

*Default:*

```
-HLS-ABR-STREAM
```

*Type:*

```
String
```

*Need restart:* Yes

*Description:*

This is a suffix for HLS ABR stream names, client that wants to get ABR version instead of ordinary version should append this suffix to original stream name'

### **hls\_abr\_with\_cdn**

*Default:*

```
false
```

*Type:*

```
Boolean
```

*Need restart:* Yes

*Description:*

Use HLS ABR with CDN or use current node for transcoding

### **hls\_acao\_header\_domain\_mask**

*Default:*

```
true
```

*Type:*

```
Boolean
```

*Need restart:* No

*Description:*

Enable origin replacement in HLS Access-Control-Allow-Origin header

### **hls\_access\_control\_headers**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

HLS response headers

### **hls\_auth\_enabled**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable check auth tokens for hls

### **hls\_auth\_token\_cache**

*Default:*

`10`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Timeout for cache auth tokens in seconds

### **hls\_auto\_start**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable HLS autostart

### **hls\_debug\_dir**

*Default:*

`hls-debug`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Folder for debug HLS stream dumps

### **hls\_debug\_stream\_name\_suffix**

*Default:*

`-DEBUG`

*Type:*

`String`

*Need restart:* Yes

*Description:*

This is a suffix for recorded stream names, used only for debug purposes

### **hls\_delayed\_shutdown**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true waits some time before removing HLS provider

### **hls\_delta\_list\_size**

*Default:*

`6`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Number of segments in playlist delta

### **hls\_dir**

*Default:*

`hls`

*Type:*

`String`

*Need restart:* Yes

*Description:*

HLS base folder

### **hls\_disable\_cleanup**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Do not remove inactive hls files from hdd

### **hls\_enabled**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable HLS support

### **hls\_fps\_discontinuity**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If false, disable discontinuity tag on fps change

### **hls\_fps\_threshold**

*Default:*

10

*Type:*

Integer

*Need restart:* No

*Description:*

Value of the threshold in percent change in fps, at which the segment is marked as discontinuity when the setting hls\_fps\_discontinuity is enabled

### **hls\_fragmented\_mp4**

*Default:*

true

*Type:*

Boolean

*Need restart:* Yes

*Description:*

If true, enables FragmentedMP4 container for low-latency hls media files

### **hls\_hold\_segments\_before\_delete**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, hold segments on disk before delete

### **hls\_hold\_segments\_size**

*Default:*

5

Type:

Integer

Need restart: No

Description:

How many segments to hold, before delete. May be useful for high-latency HLS subscribers.

### **hls\_keep\_min\_segment\_duration**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true keep minimum duration of parent segments, so duration will be  $\geq$  hls\_time\_min

### **hls\_list\_size**

Default:

8

Type:

Integer

Need restart: No

Description:

Maximum number of segments in playlist

### **hls\_ll\_enabled**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, enable Low Latency HLS

### **hls\_manager\_provider\_timeout**

Default:

300



Type:

Long

Need restart: No

Description:

HLS manager provider timeout

### **hls\_manifest\_file**

Default:

index.m3u8

Type:

String

Need restart: Yes

Description:

HLS master playlist file name. Default is 'index.m3u8'

### **hls\_max\_size\_queue**

Default:

50

Type:

Integer

Need restart: No

Description:

Maximum size of buffer for hls media data

### **hls\_metrics\_log\_size**

Default:

50

Type:

Integer

Need restart: No

Description:

Number of HLS log lines in hls/find\_all response

### **hls\_min\_list\_size**

Default:

1

Type:

Integer

Need restart: No

Description:

Minimum number of segments in playlist (should be less than 11)

### **hls\_min\_size\_queue**

Default:

10

Type:

Integer

Need restart: No

Description:

Minimal size of buffer for hls media data

### **hls\_path\_template**

Default:

{streamName}/{streamName}.m3u8

Type:

String

Need restart: Yes

Description:

Template for HLS non ABR streams path

### **hls\_player\_height**

Default:

480

Type:

Integer

Need restart: No

Description:

HLS player height

### **hls\_player\_width**

Default:

640

Type:

Integer

Need restart: No

Description:

HLS player width

### **hls\_playlist\_delta\_enabled**

Default:

true

Type:

Boolean

Need restart: No

Description:

LL-HLS playlist delta

### **hls\_preloader\_dir**

Default:

hls/.preloader

Type:

String

Need restart: No

Description:

HLS preloader dir

### **hls\_preloader\_enabled**

Default:

true

Type:

Boolean

Need restart: No

Description:

If true, enables HLS preloader

### **hls\_preloader\_segment\_count**

Default:

5

Type:

Integer

Need restart: No

Description:

LL-HLS Preloader segment count

### **hls\_preloader\_time\_min**

Default:

2000

Type:

Long

Need restart: No

Description:

Minimal size of preloader's HLS segment in milliseconds

### **hls\_provider\_traffic\_waiting\_time**

Default:

6000

Type:

Integer

Need restart: No

Description:

Time in milliseconds that hls provider waits traffic

### **hls\_sdp\_filename**

Default:

hls.sdp

Type:

String

Need restart: Yes

Description:

Filename of HLS sdp

### **hls\_segment\_name\_suffix\_randomizer\_enabled**

Default:

false

Type:

Boolean

Need restart: No

Description:

HLS segment name suffix randomizer

### **hls\_server\_enabled**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, activate HLS server

### **hls\_session\_log\_level**

Default:

INFO

Type:

String

Need restart: No

Description:

Level (INFO, DEBUG, ERROR) for HLS session

### **hls\_static\_dir**

Default:

client2/examples/demo/streaming/hls\_static

Type:

String

Need restart: No

Description:

HLS static dir

### **hls\_static\_enabled**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, enables HLS static content

### **hls\_store\_segment\_in\_memory**

Default:

true

Type:

Boolean

Need restart: No

Description:

Store HLS segments in memory

### **hls\_subscriber\_active\_timeout**

Default:

2000

Type:

Long

Need restart: No

Description:

Timeout for active state for hls subscriber

### **hls\_test\_interval**

Default:

182000

Type:

Integer

Need restart: Yes

Description:

HLS test interval

### **hls\_test\_run\_for**

Default:

180

Type:

Integer

Need restart: Yes

Description:

HLS test duration in seconds

### **hls\_test\_start\_streams**

Default:

10

Type:

Integer

Need restart: Yes

Description:

HLS test streams count

### **hls\_test\_start\_writers**

Default:

10

Type:

Integer

Need restart: Yes

Description:

HLS test writers count

### **hls\_time**

Default:

4

Type:

Integer

Need restart: No

Description:

Size of one HLS segment in seconds

### **hls\_time\_min**

Default:

2000

Type:

Long

Need restart: No

Description:

Minimal size of one HLS segment in milliseconds

### hls\_version

Default:

9

Type:

Integer

Need restart: No

Description:

HLS version

### hls\_wrap

Default:

20

Type:

Integer

Need restart: No

Description:

Maximum number of ts-files. The option is necessary to prevent disc overflow

### http.address

Default:

0.0.0.0

Type:

InetAddress[]

Need restart: Yes

Description:

Listening address for HTTP server (statistics)

### http.port

Default:

8081



Type:

Integer

Need restart: Yes

Description:

WCS server HTTP port

### **http\_client\_connection\_read\_timeout**

Default:

2000

Type:

Integer

Need restart: No

Description:

HTTP client connection read timeout in milliseconds

### **http\_client\_connection\_timeout**

Default:

2000

Type:

Integer

Need restart: No

Description:

HTTP client connection timeout in milliseconds

### **http\_enable\_paths**

Default:

rest,action,admin,shared,client,client\_records,embed\_player,empty,health-check,zclient-invite,zclient-join,verify,rest-api-spec

Type:

String

Need restart: No

Description:

List of permitted access to the web interface

### **http\_enable\_root\_redirect**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable root redirect to /admin

### **https.address**

*Default:*

`0.0.0.0`

*Type:*

`InetAddress[]`

*Need restart:* Yes

*Description:*

Listening address for HTTPS server (statistics)

### **https.port**

*Default:*

`8444`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

WCS server HTTPS port

### **https\_server\_enabled**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, activate HTTPS server

### **ice\_add\_ipv6\_candidate**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, server will try to add IPv6 ICE candidates

### **ice\_authorize\_by\_address**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, send binding request instead of binding indication for consent freshness

### **ice\_keep\_alive\_enabled**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enables ICE keep-alive

### **ice\_keep\_alive\_timeout**

*Default:*

15

*Type:*

Integer

*Need restart:* No

*Description:*

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\_channel\_high\_water\_mark**

*Default:*

104857600

*Type:*

Integer

*Need restart:* Yes

*Description:*

High watermark for ICE tcp channels

### **ice\_tcp\_channel\_low\_water\_mark**

*Default:*

10485760

*Type:*

Integer

*Need restart:* Yes

*Description:*

Low watermark for ICE tcp channels

### **ice\_tcp\_receive\_buffer\_size**

*Default:*

1048576

*Type:*

Integer

*Need restart:* Yes

*Description:*

Receive buffer size for ice tcp channels

### **ice\_tcp\_send\_buffer\_size**

*Default:*

1048576

*Type:*

Integer

*Need restart:* Yes

*Description:*

Send buffer size for ice tcp channels

### **ice\_tcp\_transport**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, use tcp transport only

### **ice\_tcp\_transport\_force**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, use tcp transport regardless of client config

### **ice\_timeout**

*Default:*

15

*Type:*

Integer

*Need restart:* No

*Description:*

ICE keep-alive timeout in seconds. By default, ICE session will be terminated if no refresh packets from browser in 15 seconds

**ice\_transport\_new**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, use new udp transport

**ice\_udp\_channel\_high\_water\_mark**

*Default:*

`104857600`

*Type:*

`Integer`

*Need restart:* No

*Description:*

High watermark for ice udp channels

**ice\_udp\_channel\_low\_water\_mark**

*Default:*

`10485760`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Low watermark for ice udp channels

**ice\_udp\_receive\_buffer\_size**

*Default:*

1048576

*Type:*

Integer

*Need restart:* No

*Description:*

Receive buffer size for ice udp channels

### **ice\_udp\_send\_buffer\_size**

*Default:*

1048576

*Type:*

Integer

*Need restart:* No

*Description:*

Send buffer size for ice udp channels

### **ice\_udp\_transport\_new**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, use new udp transport

### **ignore\_incoming\_call\_if\_sip\_login\_port\_does\_not\_match\_request\_uri**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, terminate incoming call if the SIP port does not correspond to the user indicated in Request-URI

### **ignore\_incoming\_rtp**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Discard incoming rtp before decoding/decryption. For test purposes only

### **in\_jitter\_buffer\_enabled**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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

### **inbound\_video\_rate\_stat\_rels\_throttle**

*Default:*

`1800`

*Type:*

`Integer`

*Need restart:* No



*Description:*

Inbound video rate stats interval throttle for RELS. 0 - disabled

**inbound\_video\_rate\_stat\_send\_interval**

*Default:*

0

*Type:*

Integer

*Need restart:* No

*Description:*

Inbound video rate stat send interval in sec needed to calculate channel quality. 0 - disabled

**increase\_equals\_timestamp**

*Default:*

100

*Type:*

Integer

*Need restart:* No

*Description:*

Timestamps are equal within this interval in milliseconds

**inject\_wait\_keyframe\_ms**

*Default:*

1000

*Type:*

Long

*Need restart:* No

*Description:*

Time server should wait for the injected stream to produce keyframe. Once elapsed server will start to generate video stream with the watermark. Use -1 to turn it off.

**injector\_pli\_request\_interval**

*Default:*

1000

*Type:*

Long

*Need restart:* Yes

*Description:*

PLI interval at changeover inject to target stream

### **injector\_queue\_threshold**

*Default:*

1000

*Type:*

Long

*Need restart:* Yes

*Description:*

Minimum injector's queue size for overload identification

### **ip**

*Default:*

0.0.0.0

*Type:*

String

*Need restart:* Yes

*Description:*

External IPv4 address. This IP address will differ from specified with ip\_local option when WCS server is behind NAT

### **ip\_local**

*Default:*

0.0.0.0

*Type:*

String

*Need restart:* Yes

*Description:*

WCS server will create sockets and listen on this interface

### **ip\_v6**

*Default:*

1

Type:

String

Need restart: Yes

Description:

External IPv6 address. This IP address will differ from specified with ip\_local option when WCS server is behind NAT

### **jitter\_buffer\_always\_detect\_frame\_type**

Default:

false

Type:

Boolean

Need restart: No

Description:

Enables mandatory detecting frame type from rtp packets

### **jitter\_buffer\_attempt\_to\_correct\_broken\_timestamp**

Default:

false

Type:

Boolean

Need restart: No

Description:

If enabled, jitter buffer adding +1 to broken rtp timestamp

### **jitter\_buffer\_capacity**

Default:

0

Type:

Integer

Need restart: No

Description:

JitterBuffer will drop frames when value exceeded

### **jitter\_threshold**

*Default:*

50

*Type:*

Integer

*Need restart:* No

*Description:*

RTMFP. If jitter between UDP media packets is greater than the threshold, it will be reported

### **jni\_cache\_class**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, cache JNI Class object

### **jni\_debug\_enable**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, enables jni logs in stdout

### **keep\_alive.algorithm**

*Default:*

HIGH\_LEVEL

*Type:*

INTERNAL

NONE

HIGH\_LEVEL

*Need restart:* Yes

*Description:*

Keep-alive algorithm: INTERNAL, NONE, or HIGH\_LEVEL

### **keep\_alive.enabled**

*Default:*

websocket,rtmfp

*Type:*

String

*Need restart:* Yes

*Description:*

Enable keep-alive for the listed protocols

### **keep\_alive.peer\_interval**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

2000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Keep-alive peer interval (Not in use)

### **keep\_alive.probes**

*Default:*

10

*Type:*

Integer

*Need restart:* Yes

*Description:*

Number of unsuccessful 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**

*Default:*

5000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Interval in milliseconds between attempts to ping connected client (WebSocket, RTMP, RTMFP)

**keep\_alive\_streaming\_sessions\_enabled**

*Default:*

false

*Type:*

Boolean

*Need restart:* Yes

*Description:*

If true, server sends keep-alive REST requests to check if stream playback is allowed to continue / resume

**kill\_event\_scanner**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Debug option, for development only

**ll\_hls\_can\_skip\_segments\_for\_delta\_list**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`6`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Number of segments in playlist can skipped for delta playlist

### **ll\_hls\_create\_preloader**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, enables LL-HLS Preloader creator

### **ll\_hls\_custom\_preloader\_dir**

*Default:*

`custom-preloader`

*Type:*

`String`

*Need restart:* Yes

*Description:*

LL-HLS custom preloader base folder

### **ll\_hls\_max\_number\_of\_parent\_segments\_containing\_partials**

*Default:*

`5`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Max number of parent segments containing partials

### **ll\_hls\_part\_hold\_back\_count**

*Default:*

6

*Type:*

Integer

*Need restart:* No

*Description:*

PART-HOLD-BACK attribute value in Part Target Duration

### **ll\_hls\_partial\_time\_max**

*Default:*

400

*Type:*

Long

*Need restart:* No

*Description:*

Maximum size of one partial HLS segment in milliseconds

### **ll\_hls\_preloader\_segment\_duration**

*Default:*

400

*Type:*

Long

*Need restart:* No

*Description:*

Duration of preloader LL-HLS segment in milliseconds

### **load\_balancing\_acao\_header**

*Default:*

l

*Type:*

String



*Need restart:* Yes

*Description:*

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**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, activate loadbalancer

### **log\_metrics\_stats**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enables/Disables log metrics statistic

### **log\_metrics\_time\_buffer**

*Default:*

`10000`

*Type:*

`Long`

*Need restart:* Yes

*Description:*

Setting for time buffer(lesser value = faster result, higher value = slower and more accurate).  
min 1 second

### **mail.password**

*Default:*

`null`

Type:

String

Need restart: Yes

Description:

Password for the mail server

### **mail.username**

Default:

null

Type:

String

Need restart: Yes

Description:

Username for the mail server

### **mail.verification.ttl**

Default:

86400000

Type:

Long

Need restart: Yes

Description:

Verification transaction TTL

### **mail.verification.ttl.active**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

Remove invalid verification transactions

### **mail.verification.ttl.interval**

Default:

60000

Type:

Long

Need restart: Yes

Description:

Verification transaction cleanup interval

### **mail.verification.url**

Default:

null

Type:

String

Need restart: Yes

Description:

Base url for verification service, e.g. http://my.wcs.ip:8081/

### **manager\_http\_ports\_enabled**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, bind old manager http(s) ports 9091 and 8888

### **matroska\_unknown\_segment\_size**

Default:

false

Type:

Boolean

Need restart: No

Description:

Unknown segment atom size

### **max\_callid\_length**

Default:

32

Type:

Integer

Need restart: No

Description:

Maximum length of SIP callID. If the length of generated callID exceeds this value, it will be cut to this length

**max\_drop\_rate**

 **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

null

Type:

String

Need restart: No

Description:

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**

 **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

null

Type:

String

Need restart: No

Description:

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\_dir**

*Default:*

`media`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Media base folder

### **media\_port\_from**

*Default:*

`31001`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Beginning of media ports range for ICE, RTP, SRTP, RTCP

### **media\_port\_stress\_test\_iterations**

*Default:*

`1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Media port stress test iterations

### **media\_port\_stress\_test\_thread\_sleep**

*Default:*

`5`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Media port stress test thread sleeping interval

### **media\_port\_stress\_test\_threads**

*Default:*

5

*Type:*

Integer

*Need restart:* No

*Description:*

Media port stress test threads count

### **media\_port\_to**

*Default:*

32000

*Type:*

Integer

*Need restart:* Yes

*Description:*

End of media ports range for ICE, RTP, SRTP, RTCP

### **media\_ports\_auditor\_interval**

*Default:*

5000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Audit interval for busy and free ports, in milliseconds

### **media\_ports\_auditor\_max\_attempts**

*Default:*

3

*Type:*

Integer

*Need restart:* Yes

*Description:*

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 successful audits is reached

### **media\_processor\_input\_data\_stat\_window**

*Default:*

`30000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Window for gathering min and max incoming data arrival time, in ms

### **media\_transponder\_sdp\_filename**

*Default:*

`media_transponder.sdp`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Filename of transponder sdp

### **min\_drop\_rate**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Queue size will not be decreased lower than this minimum value.

Is used only if `out_jitter_buffer_enabled=true` or `in_jitter_buffer_enabled=true`

### **mixer\_activity\_timer\_cool\_off\_period**

*Default:*

`1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Mixer will be terminated after  $\{\text{mixer\_activity\_timer\_cool\_off\_period} * \text{mixer\_activity\_timer\_timeout}\}$  since last stream activity for the corresponding mixer

### **mixer\_activity\_timer\_timeout**

*Default:*

`-1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

If there are no streams added to mixer within this timeout in milliseconds, corresponding mixer will be terminated

### **mixer\_app\_name**

*Default:*

`defaultApp`



Type:

String

Need restart: No

Description:

AppName for mixer streams

### **mixer\_audio\_enabled**

Default:

true

Type:

Boolean

Need restart: No

Description:

When false, mixer stream has video-only

### **mixer\_audio\_only\_height**

Default:

360

Type:

Integer

Need restart: No

Description:

Height constraint for mixer audio only frame

### **mixer\_audio\_only\_width**

Default:

640

Type:

Integer

Need restart: No

Description:

Width constraint for mixer audio only frame

### **mixer\_audio\_opus\_float\_coding**

Default:

false

Type:

Boolean

Need restart: No

Description:

Use float optimisations for opus audio coding.

### **mixer\_audio\_silence\_threshold**

Default:

-50.0

Type:

Double

Need restart: No

Description:

Audio silence threshold in db

### **mixer\_audio\_threads**

Default:

4

Type:

Integer

Need restart: No

Description:

How many threads should multithreaded audio mixer use

### **mixer\_auto\_create\_delimiter**

Default:

#

Type:

String

Need restart: No

Description:

Mixer auto create stream/room delimiter

### **mixer\_auto\_start**

Default:

true

Type:

Boolean

Need restart: No

Description:

If true, enable mixer autostart

### **mixer\_autoscale\_desktop**

Default:

true

Type:

Boolean

Need restart: No

Description:

Separate screen share font size from other frames

### **mixer\_debug\_mode**

Default:

false

Type:

Boolean

Need restart: No

Description:

Turns on debug mode, this will output debug information directly onto mixers canvas'

### **mixer\_decode\_stream\_name**

Default:

false

Type:

Boolean

Need restart: No

Description:

Decode stream name to mixer's canvas

### **mixer\_desktop\_align**

Default:

TOP

Type:

TOP

BOTTOM

LEFT

RIGHT

CENTER

Need restart: No

Description:

Alignment of screen sharing stream

### **mixer\_display\_stream\_name**

Default:

false

Type:

Boolean

Need restart: No

Description:

Output stream name to mixer's canvas

### **mixer\_font\_size**

Default:

20

Type:

Integer

Need restart: No

Description:

Font size for stream name and debug info

### **mixer\_font\_size\_audio\_only**

Default:

40

Type:

Integer

Need restart: No

Description:

Font size for stream name and debug info for audio only streams

### **mixer\_frame\_background\_colour**

*Default:*

`0x2B2A2B`

*Type:*

`String`

*Need restart:* No

*Description:*

Hex value of frame background colour

### **mixer\_idle\_timeout**

*Default:*

`60000`

*Type:*

`Long`

*Need restart:* No

*Description:*

Mixer idle timeout in milliseconds

### **mixer\_in\_buffering\_ms**

*Default:*

`200`

*Type:*

`Integer`

*Need restart:* No

*Description:*

How much stream should be buffered before it gets into mix

### **mixer\_incoming\_time\_rate\_lower\_threshold**

*Default:*

`0.95`

*Type:*

`Double`

*Need restart:* No

*Description:*

Relation between incoming stream time and actual machine mixing time, 0.9 means that

incoming time rate can be 10% lower than actual stream playback rate

### **mixer\_incoming\_time\_rate\_upper\_threshold**

*Default:*

`1.05`

*Type:*

`Double`

*Need restart:* No

*Description:*

Relation between incoming stream time and actual machine mixing time, 1.2 means that incoming time rate can be 20% bigger than actual stream playback rate

### **mixer\_layout\_class**

*Default:*

`com.flashphoner.media.mixer.video.presentation.GridLayout`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Name of class for custom mixer layout

### **mixer\_layout\_dir**

*Default:*

`''`

*Type:*

`String`

*Need restart:* No

*Description:*

Directory name for custom mixer descriptors

### **mixer\_linear\_smoothing\_audio**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Smoothly fade-in and fade-out audio in mixer. 20ms of audio is lost during fade-in

### **mixer\_lossless\_video\_processor\_enabled**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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**

*Default:*

`200`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Max size that is allowed for mixers incoming buffer, after reaching this point processor will use own buffer instead'

### **mixer\_lossless\_video\_processor\_wait\_time\_ms**

*Default:*

`20`

*Type:*

`Integer`

*Need restart:* No

*Description:*

How long to wait before checking mixer's incoming buffer again in case it was full

### **mixer\_maintain\_streams\_delay\_while\_buffered**

*Default:*

`true`

Type:

Boolean

Need restart: No

Description:

If false, mixer does not drop audio video frames to maintain minimum set delay  
mixer\_in\_buffering\_ms

### **mixer\_mcu\_audio**

Default:

false

Type:

Boolean

Need restart: No

Description:

Enable mcu like audio mixing, each added stream will have dedicated audio mix available as a separate stream

### **mixer\_mcu\_multithreaded\_delivery**

Default:

false

Type:

Boolean

Need restart: No

Description:

Use separate threads for mcu streams inject into engine.

### **mixer\_mcu\_multithreaded\_mix**

Default:

false

Type:

Boolean

Need restart: No

Description:

Mix audio/video in separate threads.

### **mixer\_mcu\_video**



*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Works only with mcu audio, send video to each audio mcu stream. Video stays the same as in the root mixer.

### **mixer\_minimal\_font\_size**

*Default:*

`1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Minimal font size for stream name if autoscaling is on

### **mixer\_out\_buffer\_enabled**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable buffer for out mixer streams

### **mixer\_out\_buffer\_initial\_size**

*Default:*

`2000`

*Type:*

`Long`

*Need restart:* No

*Description:*

Initial size of output mixer buffer in milliseconds

### **mixer\_out\_buffer\_max\_bufferings\_allowed**

*Default:*

**-1**

*Type:*

**Integer**

*Need restart:* No

*Description:*

mixer\_out\_buffer\_max\_bufferings\_allowed

### **mixer\_out\_buffer\_overflow\_allowed\_deviation**

*Default:*

**1000**

*Type:*

**Long**

*Need restart:* No

*Description:*

max allowed difference between min(buffer) and max(buffer). if this constraint met over the rtmp\_in\_buffer\_overflow\_deviation\_window overflow state will be set leading to clock acceleration

### **mixer\_out\_buffer\_overflow\_deviation\_window**

*Default:*

**30000**

*Type:*

**Integer**

*Need restart:* No

*Description:*

window for gathering min and max buffer sizes over time, in ms

### **mixer\_out\_buffer\_overflow\_rate**

*Default:*

**0.15**

*Type:*

**Double**

*Need restart:* No

*Description:*

buffer clock acceleration rate. To calculate increase in output speed use  $(1 + \text{rate}) / \text{buffer} = 1/x$

### **mixer\_out\_buffer\_polling\_time**

*Default:*

`100`

*Type:*

`Long`

*Need restart:* No

*Description:*

Output mixer buffer polling time in milliseconds

### **mixer\_out\_buffer\_start\_size**

*Default:*

`150`

*Type:*

`Long`

*Need restart:* No

*Description:*

Start size of output mixer buffer in milliseconds

### **mixer\_prune\_streams**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

When true, prune mixer stream

### **mixer\_realtime**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Turns on realtime version of mixer

### **mixer\_separate\_buffering\_audio\_video**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, separate audio video buffering in mixer

### **mixer\_show\_separate\_audio\_frame**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Show audio frame for audio+video stream if added with hasVideo: false

### **mixer\_text\_align**

*Default:*

`BOTTOM_LEFT`

*Type:*

`TOP_LEFT`

`TOP_CENTER`

`TOP_RIGHT`

`CENTER`

`BOTTOM_LEFT`

`BOTTOM_CENTER`

`BOTTOM_RIGHT`

`EXTERNAL_TOP_CENTER`

`EXTERNAL_BOTTOM_CENTER`

*Need restart:* Yes

*Description:*

Text position relative to frame

### **mixer\_text\_autoscale**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable stream name autoscaling

### **mixer\_text\_background\_colour**

*Default:*

`0x2B2A2B`

*Type:*

`String`

*Need restart:* No

*Description:*

Hex value of stream names background colour

### **mixer\_text\_background\_opacity**

*Default:*

`100`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Opacity of text background percentage

### **mixer\_text\_bulk\_write**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Use bulk write with DirectByteBuffer for text

### **mixer\_text\_bulk\_write\_with\_buffer**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Use bulk write with DirectByteBuffer for text, cache whole text as a frame

### **mixer\_text\_colour**

*Default:*

`0xFFFFFFFF`

*Type:*

`String`

*Need restart:* No

*Description:*

Hex value of stream names colour

### **mixer\_text\_cut\_top**

*Default:*

`3`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Clip top part of the text

### **mixer\_text\_display\_room**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Display room name in participant stream names

### **mixer\_text\_font**

*Default:*

**Serif**

*Type:*

**String**

*Need restart:* No

*Description:*

Font of mixer text

### **mixer\_text\_outside\_frame**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

**NO**

*Type:*

**String**

*Need restart:* Yes

*Description:*

Text position relative to frame

### **mixer\_text\_outside\_frame\_padding**

*Default:*

**50**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

External padding for outside frame text

### **mixer\_text\_padding\_bottom**

*Default:*

5

*Type:*

Integer

*Need restart:* No

*Description:*

Padding for the bottom side of text in pixels

### **mixer\_text\_padding\_left**

*Default:*

5

*Type:*

Integer

*Need restart:* No

*Description:*

Padding for the left side of text in pixels

### **mixer\_text\_padding\_right**

*Default:*

4

*Type:*

Integer

*Need restart:* No

*Description:*

Padding for the right side of text in pixels

### **mixer\_text\_padding\_top**

*Default:*

5

*Type:*

Integer

*Need restart:* No

*Description:*

Padding for the top side of text in pixels

### **mixer\_thread\_priority**



*Default:*

`5`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Mixer thread priority, min 1 max 10

### **mixer\_type**

*Default:*

`NATIVE`

*Type:*

`JAVA`

`NATIVE`

`MULTI_THREADED_NATIVE`

*Need restart:* No

*Description:*

Mixer implementation, can be JAVA, NATIVE or MULTI\_THREADED\_NATIVE

### **mixer\_use\_sdp\_state**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable audio/video only stream detection via sdp state

### **mixer\_video\_background\_filename**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Mixer video background. Example: background.png

### **mixer\_video\_bitrate\_kbps**

*Default:*

`2000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Encoded video bitrate kbps

### **mixer\_video\_buffer\_length**

*Default:*

`1000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Video buffer length for decoded frames

### **mixer\_video\_desktop\_fullscreen**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Display desktop stream in fullscreen mode

### **mixer\_video\_desktop\_layout\_inline\_padding**

*Default:*

`10`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Padding between video streams in bottom row (under screen sharing stream)

### **mixer\_video\_desktop\_layout\_padding**

*Default:*

30

*Type:*

Integer

*Need restart:* No

*Description:*

Padding between top row (screen sharing stream) and bottom row (other streams)

### **mixer\_video\_enabled**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

When false, mixer stream has audio-only

### **mixer\_video\_fps**

*Default:*

30

*Type:*

Integer

*Need restart:* No

*Description:*

Fps constraint for mixer stream

### **mixer\_video\_grid\_layout\_middle\_padding**

*Default:*

10

*Type:*

Integer

*Need restart:* No

*Description:*

Padding between video streams in one row (when there is no screen sharing stream)

### **mixer\_video\_grid\_layout\_padding**

*Default:*

`30`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Padding between rows of video streams (when there is no screen sharing stream)

### **mixer\_video\_height**

*Default:*

`720`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Height constraint for mixer stream

### **mixer\_video\_layout\_desktop\_key\_word**

*Default:*

`desktop`

*Type:*

`String`

*Need restart:* No

*Description:*

Keyword for screen sharing streams

### **mixer\_video\_profile\_level**

*Default:*

`42c02a`

*Type:*

`String`

*Need restart:* No

*Description:*

Mixer video profile and level in hex. Example: 42c02a

### **mixer\_video\_quality**

*Default:*

24

*Type:*

Integer

*Need restart:* No

*Description:*

Encoded video quality (CRF)

### **mixer\_video\_stable\_fps\_threshold**

*Default:*

15

*Type:*

Integer

*Need restart:* No

*Description:*

Streams with fps lower then threshold won't trigger buffering of the stream if video buffer was exhausted

### **mixer\_video\_threads**

*Default:*

4

*Type:*

Integer

*Need restart:* No

*Description:*

How many threads should multithreaded video mixer use

### **mixer\_video\_width**

*Default:*

1280

*Type:*

Integer

*Need restart:* No

*Description:*

Width constraint for mixer stream

### **mixer\_voice\_activity**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable/disable voice activity frame

### **mixer\_voice\_activity\_colour**

*Default:*

`0x00CC66`

*Type:*

`String`

*Need restart:* No

*Description:*

Hex value of voice activity colour

### **mixer\_voice\_activity\_frame\_position\_inner**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Draw voice activity frame inside the frame. If false - draw around the frame

### **mixer\_voice\_activity\_frame\_thickness**

*Default:*

`6`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Thickness of voice activity frame

### **mixer\_voice\_activity\_switch\_delay**

*Default:*

`0`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Voice activity indicator switch off delay in milliseconds

### **mp4\_container\_moov\_first**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

When recording mp4 write moov atom first so recording can be played/downloaded progressively

### **mp4\_container\_moov\_first\_reserve\_space**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Turn on space reservation for moov atom to avoid additional filesystem copy

### **mp4\_container\_moov\_reserved\_space\_size**

*Default:*

`2048`

Type:

Integer

Need restart: No

Description:

When writing moov first how much space should be reserved for moov atom in kilobytes

### **mp4\_cutter\_dir**

Default:

records

Type:

String

Need restart: Yes

Description:

Folder to place MP4 fragments while playing recording files in browser

### **mp4\_cutter\_manager\_cache\_expire**

Default:

60000

Type:

Integer

Need restart: Yes

Description:

Mp4 cutter manager cache expire

### **mpeg1.gop\_size**

Default:

60

Type:

Integer

Need restart: No

Description:

GOP size or k-frame interval

### **mpeg1.qmax**

Default:

24



Type:

Integer

Need restart: No

Description:

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**

Default:

4

Type:

Integer

Need restart: No

Description:

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**

Default:

0

Type:

Integer

Need restart: No

Description:

Trellis quantization

### **mpegts\_agent\_sdp\_filename**

Default:

mpegts\_agent.sdp

Type:

String

Need restart: Yes

Description:

Filename of MPEG-TS publisher sdp

### **mpegts\_max\_pts\_diff**

*Default:*

`1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

if server receives a packet whose PTS differs by more than this value in seconds, sessions of subscribers will be terminated

### **mpegts\_stream\_timeout**

*Default:*

`90000`

*Type:*

`Long`

*Need restart:* No

*Description:*

MpegTS stream with no data will be terminated after this timeout in milliseconds

### **mpegts\_udp\_constant\_socket**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

if true, the server accepts packets only from the first client socket. Packets from other sockets will be ignored

### **mse\_sdp\_filename**

*Default:*

`mse.sdp`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Filename of MSE sdp

### **msrp\_port**

*Default:*

2855

*Type:*

Integer

*Need restart:* No

*Description:*

Port for receiving MSRP / TCP connections

### **multi\_record\_dir**

*Default:*

records

*Type:*

String

*Need restart:* Yes

*Description:*

MultiRecord base folder

### **multi\_recorder\_mkv\_fill\_gaps**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

Fill gaps in matroska tracks

### **multi\_recorder\_type**

*Default:*

MP4

*Type:*

MKV

MP4

*Need restart:* No

*Description:*

MP4 or MKV multiRecorder format

### **multipart\_message\_service\_uri**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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

### **multiple\_pull\_test\_server\_url**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Server url for mpt test

### **multiple\_pull\_test\_stream\_name**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Stream name for mpt test

### **multiple\_pull\_test\_subscribers**

*Default:*

`100`

Type:

Integer

Need restart: No

Description:

multiple\_pull\_test\_subscribers

### **native\_test\_aac**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, enable AAC native test

### **native\_test\_decoder**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, enable decoder native test

### **native\_test\_encoder**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, enable encoder native test

### **native\_test\_opus**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, enable Opus native test

### **native\_test\_resampler**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, enable native test resampler

### **native\_test\_run\_for**

Default:

180

Type:

Integer

Need restart: Yes

Description:

Native test duration

### **native\_test\_start\_threads**

Default:

10

Type:

Integer

Need restart: Yes

Description:

Native test threads count

### **native\_test\_thread\_interval**

Default:

200

Type:

Integer

Need restart: Yes

Description:

Native test interval

### **netty\_deadlock\_aware\_worker\_timeout**

Default:

10000

Type:

Integer

Need restart: No

Description:

Timeout to detect SSL connection with Netty deadlock

### **no\_media\_dump\_interval**

Default:

15000

Type:

Long

Need restart: No

Description:

Period in milliseconds, within which media traffic should be captured by tcpdump when client sends bug report with no\_media type

### **notification\_apns\_key\_id**

Default:

null

Type:

String

Need restart: No

Description:

Key Id for ios apns

### **notification\_apns\_key\_path**

*Default:*

`/usr/local/FlashphonerWebCallServer/conf/apns_auth_key.p8`

*Type:*

`String`

*Need restart:* No

*Description:*

Full path to p8 key file for ios apns

### **notification\_apns\_team\_id**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Team Id for ios apns

### **notify\_message\_call\_timeout**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Timeout in milliseconds to wait for client confirmation 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\_multiple\_record\_hook\_script**

*Default:*

`on_multiple_record_hook.sh`

*Type:*

`String`

*Need restart:* No



*Description:*

This option points to shell script located in /usr/local/FlashphonerWebCallServer/bin directory, which is started when multipleRecorder is terminated. By default, the script run offline\_mixer\_tool.sh script located in /usr/local/FlashphonerWebCallServer/tools with default offlineMixer config located in /usr/local/FlashphonerWebCallServer/conf.

**on\_record\_hook\_script**

*Default:*

`on_record_hook.sh`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

If true, then wait for a client response prior to responding with 200 OK to an OPTIONS request

### **opus.encoder.bitrate**

*Default:*

`-1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Target bitrate for Opus encoder, in bps

### **opus.encoder.complexity**

*Default:*

`-1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Target complexity for Opus encoder

### **opus\_formats**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Comma-separated list of Opus formats (name=value).

Example: maxaveragebitrate=20000.

These formats will be listed in SDP

### **order\_threads\_by\_seq**

*Default:*

`true`

Type:

`Boolean`

Need restart: No

Description:

If true, order incoming SIP messages by sequence number and wait if number is out of order

### **out\_jitter\_buffer\_enabled**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

`null`

Type:

`String`

Need restart: No

Description:

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**

Default:

`null`

Type:

`String`

Need restart: No

Description:

SIP port. If this parameter is set, it will redefine values that were transmitted during connection

### **outbound\_proxy**

Default:

`null`

Type:

`String`

*Need restart:* No

*Description:*

SIP outbound proxy. If this parameter is set, it will redefine values that were transmitted during connection

#### **outbound\_video\_rate\_stat\_rels\_throttle**

*Default:*

`1800`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Outbound video rate stats interval throttle for RELS. 0 - disabled

#### **outbound\_video\_rate\_stat\_send\_interval**

*Default:*

`0`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Outbound video rate stat send interval in sec needed to calculate channel quality. 0 - disabled

#### **parse\_system\_stats**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, gather system level statistics such as netstat, lsof, etc. The parsing may take a lot of time

#### **periodic\_fir\_request**

*Default:*

`false`

Type:

Boolean

Need restart: No

Description:

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**

Default:

5000

Type:

Integer

Need restart: No

Description:

Interval to send RTCP FIR in milliseconds

### **play\_stream\_force\_video\_orientation**

Default:

true

Type:

Boolean

Need restart: No

Description:

Force negotiation of 3gpp video orientation extension for play stream requests

### **port\_from**

Default:

30000

Type:

Integer

Need restart: No

Description:

Beginning of range of ports for SIP signaling

## **port\_to**

*Default:*

`31000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

End of range of ports for SIP signaling

## **preserve\_non\_mixed\_recorded\_files**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

RTMFP. If true, print statistics of streams in logs

## **print\_rtcp\_stats**

### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, print RTCP report on end of session

### **priority\_outside\_codecs**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, then outside (browser) codecs will be in first place

### **process\_remote\_sdp\_candidates**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, process candidates from SDP

### **profiles**

*Default:*

`640028`

*Type:*

`String`

*Need restart:* No

*Description:*

Comma-separated list of H.264 profiles. These profiles will be used in SDP for video calls

### **proxy\_propagate\_fir**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Propagate FIR requests through proxy

### **proxy\_use\_h264\_packetization\_mode\_1\_only**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, use H.264 packetization mode 1

### **ptime**

*Default:*

`20`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Packetization time. Use carefully

### **ptime\_corrector\_enabled**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enabling corrector by required packetization time

### **publication\_report\_format**



*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

RTMFP. Sets format for statistics.

Possible value: csv

### **pull\_streams**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Comma separated list of urls to pull from at server startup

### **queue\_ping\_period**

*Default:*

`2000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Queue ping interval in ms

### **queue\_stat\_log**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable queue statistics logging

### **queue\_transcoder\_core\_router\_uri**

*Default:*

`tcp://127.0.0.1:5555`

*Type:*

`String`

*Need restart:* No

*Description:*

Queue transcoder core router URI

### **queue\_transcoder\_receive\_timeout**

*Default:*

`500`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Queue transcoder receive timeout

### **queue\_transcoder\_shm\_path**

*Default:*

`/dev/shm/`

*Type:*

`String`

*Need restart:* No

*Description:*

Path to shared memory objects for queue transcoder

### **queue\_transcoder\_shm\_size**

*Default:*

`5`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Shared memory object size for queue transcoder

### **queue\_transcoder\_transmit\_timeout**

*Default:*

`500`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Queue transcoder transmit timeout

### **queue\_transcoder\_worker\_router\_uri**

*Default:*

`ipc:///tmp/flashphoner.pipe`

*Type:*

`String`

*Need restart:* No

*Description:*

Queue transcoder core router URI

### **record**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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\_buffer\_max\_size**

*Default:*

`100`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Record audio buffer size

### **record\_audio\_codec\_channels**

*Default:*

`2`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Codec channel count used for recording streams

### **record\_audio\_codec\_sample\_rate**

*Default:*

`44100`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Codec sample rate used for recording streams

### **record\_audio\_processor\_pcm**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, record audio on stream as PCM16. (Then record= option should point to a valid path, e.g. record=/tmp/)

### **record\_close\_scheduling\_period**

*Default:*

`20`

Type:

Integer

Need restart: Yes

Description:

Buffer check period for closing a record in milliseconds

### **record\_dir**

Default:

records

Type:

String

Need restart: Yes

Description:

Record base folder

### **record\_fdk\_aac\_bitrate\_mode**

Default:

5

Type:

Integer

Need restart: No

Description:

Record FDK bitrate mode. 0 - CBR, 1-5 - VBR

### **record\_filename\_template**

Default:

null

Type:

String

Need restart: No

Description:

Filename template for an audio call recording. Besides the default fields, {date} field can also be used

### **record\_flash\_published\_streams**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, record streams published from native Flash clients and RTMP live encoders such as Wirecast, FFmpeg, FMLE, etc.

### **record\_formats**

*Default:*

h264-mp4, vp8-webm

*Type:*

RecordFormats

*Need restart:* No

*Description:*

H264 and VP8 recorder type

### **record\_h264\_to\_ts**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If set, record to TS instead of mp4

### **record\_mixer\_streams**

*Default:*

false

Type:

`Boolean`

Need restart: No

Description:

When true, mixer streams are recorded

### **record\_response\_content\_disposition\_header\_value**

Default:

`null`

Type:

`String`

Need restart: No

Description:

/client/records/ path content-disposition header

### **record\_rotation**

Default:

`null`

Type:

`String`

Need restart: No

Description:

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\_rotation\_index\_enabled**

Default:

`true`

Type:

`Boolean`

Need restart: No

Description:

If true, rotation for stream recording files is enabled

### **record\_rtsp\_streams**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, record RTSP streams

### **record\_stop\_timeout**

*Default:*

`15`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Record stop timeout in seconds

### **record\_streams**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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

### **record\_tmp\_dir**

*Default:*

`records`

*Type:*

`String`

*Need restart:* Yes



*Description:*

Recording temporary files base folder

### **recording\_by\_user**

*Default:*

false

*Type:*

Boolean

*Need restart:* Yes

*Description:*

If true, call is recorded for the initiator of the call only

### **reg\_expires**

*Default:*

3600

*Type:*

Integer

*Need restart:* No

*Description:*

Value in seconds, which will be used in Expires header when SIP REGISTER request is sent

### **rels\_client\_type**

*Default:*

HTTP

*Type:*

JDBC

HTTP

*Need restart:* Yes

*Description:*

ClickHouse client implementation

### **rels\_consumer\_interval**

*Default:*

1000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Buffer consumer interval

### **rels\_database\_address**

*Default:*

`http://localhost:8123/wcs?user=wcs&password=wcs`

*Type:*

`String`

*Need restart:* No

*Description:*

Address of ClickHouse database that will be used for events storing

### **rels\_enabled**

*Default:*

`0`

*Type:*

`EnumSet`

*Need restart:* No

*Description:*

Enables logging of the listed event types to database

### **rels\_event\_buffer\_initial\_size**

*Default:*

`5000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Initial buffer size

### **rels\_event\_buffer\_interval**

*Default:*

`60000`

*Type:*

`Integer`

*Need restart:* No

*Description:*  
Buffer interval

### **rels\_event\_buffer\_max\_size**

*Default:*

40000

*Type:*

Integer

*Need restart:* No

*Description:*  
Max buffer size

### **rels\_media\_session\_events\_default\_frequency**

*Default:*

1000

*Type:*

Integer

*Need restart:* Yes

*Description:*  
Default frequency (ms) of RELS media session events if not specified by REST API request

### **rels\_test\_start\_threads**

*Default:*

2

*Type:*

Integer

*Need restart:* Yes

*Description:*  
RELS test threads count

### **rels\_test\_thread\_events**

*Default:*

2

*Type:*

Integer

*Need restart:* Yes

*Description:*

RELS test thread events

### **rels\_test\_thread\_interval**

*Default:*

`2`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

RELS test thread interval

### **remove\_ssrc\_attr**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, remove ssrc attribute

### **replace\_cached\_pool\_with\_default\_pool**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, replaces cached thread pool with default

### **resample\_video**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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\_credentials**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Rest-api response `access_control_allow_credentials` header

### **rest\_access\_control\_allow\_headers**

*Default:*

`content-type, x-requested-with`

*Type:*

`String`

*Need restart:* No

*Description:*

Rest-api response `access_control_allow_headers` header

### **rest\_access\_control\_allow\_methods**

*Default:*

`POST`

*Type:*

`String`

*Need restart:* No

*Description:*

Rest-api response access\_control\_allow\_methods header

### **rest\_access\_control\_allow\_origin**

*Default:*

`*`

*Type:*

`String`

*Need restart:* No

*Description:*

Rest-api response access\_control\_allow\_origin header

### **rest\_access\_control\_headers**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

REST response headers

### **rest\_client\_request\_retry\_count**

*Default:*

`3`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

How many times to retry sent request; 0 means no retries.

### **rest\_client\_request\_sent\_retry\_enabled**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

True if it's OK to retry non-idempotent requests that have been sent.

### **rest\_hook\_secret\_key**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Rest hook secret key

### **rest\_hook\_send\_hash**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Rest hook send hash

### **rest\_max\_connections**

*Default:*

`200`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Rest max connexions

### **rest\_request\_timeout**

*Default:*

`15`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Rest request timeout in seconds

### **rfc2833\_packets\_count**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Number of RTP packets for sending one DTMF

### **rmi.port**

*Default:*

`1098`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Internal RMI port for communications with WCS Manager

### **room\_idle\_timeout**

*Default:*

`60000`

*Type:*

`Long`

*Need restart:* Yes

*Description:*

Room idle timeout in milliseconds

### **rtc\_ice\_add\_local\_component**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No



*Description:*

If true, add local component for ICE candidates

### **rtc\_ice\_add\_local\_interface**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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\_pli\_request\_interval**

*Default:*

`1000`

Type:

Long

Need restart: No

Description:

Minimal waiting time to send PLI after receiving K-frame

### **rtcp\_sender\_interval**

Default:

0.1

Type:

Double

Need restart: No

Description:

Guard RTCP interval based on the specified fraction of RTCP bitrate

### **rtmfp.address**

Default:

0.0.0.0

Type:

InetAddress[]

Need restart: Yes

Description:

Listening address for RTMFP server

### **rtmfp.port**

Default:

1935

Type:

Integer

Need restart: Yes

Description:

RTMFP server port, UDP

### **rtmfp\_server\_enabled**

Default:

true

Type:

`Boolean`

Need restart: Yes

Description:

Enable/disable rtmpfp server

### **rtmp.address**

Default:

`0.0.0.0`

Type:

`InetAddress[]`

Need restart: Yes

Description:

Listening address for RTMP server

### **rtmp.port**

Default:

`1935`

Type:

`Integer`

Need restart: Yes

Description:

RTMP server port, TCP

### **rtmp.server\_buffer\_enabled**

Default:

`false`

Type:

`Boolean`

Need restart: No

Description:

Enable/disable buffering rtmp data on java's heap if socket buffer is full

### **rtmp.server\_channel\_high\_water\_mark**

Default:

`52428800`

Type:

Integer

Need restart: Yes

Description:

High watermark for connected rtmp channels

**rtmp.server\_channel\_low\_water\_mark**

Default:

5242880

Type:

Integer

Need restart: Yes

Description:

Low watermark for connected rtmp channels

**rtmp.server\_channel\_send\_buffer\_size**

Default:

1048576

Type:

Integer

Need restart: Yes

Description:

Send buffer size for rtmp channels

**rtmp.server\_read\_socket\_timeout**

Default:

0

Type:

Integer

Need restart: Yes

Description:

TCP socket write timeout for RTMP server, in seconds

**rtmp.server\_socket\_timeout**

Default:

0

Type:

Integer

Need restart: Yes

Description:

TCP socket write and read timeout for RTMP server for, in seconds

#### **rtmp.server\_write\_socket\_timeout**

Default:

0

Type:

Integer

Need restart: Yes

Description:

TCP socket write timeout for RTMP server, in seconds

#### **rtmp.use\_server\_socket\_timeout**

Default:

false

Type:

Boolean

Need restart: Yes

Description:

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**

Default:

1

Type:

Integer

Need restart: No

Description:

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**

*Default:*

`60000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

If there is no subscribers for an RTMP stream within this timeout in milliseconds, corresponding RTMP session will be terminated

### **rtmp\_agent\_sdp\_filename**

*Default:*

`rtmp_agent.sdp`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Filename of RTMP agent sdp

### **rtmp\_appkey\_source**

*Default:*

`default`

*Type:*

`String`

*Need restart:* No

*Description:*

RTMP appkey source: default/app

### **rtmp\_command\_amf3**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

rtmp\_command\_amf3

### **rtmp\_detect\_h264\_frame\_type**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, for H264 frames, frame type will be determined by the NAL units instead of RTMP control field

### **rtmp\_dump\_dir**

*Default:*

rtmp\_dump\_dir

*Type:*

String

*Need restart:* Yes

*Description:*

Rtmp dump base folder

### **rtmp\_dump\_incoming\_video**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, write incoming rtmp video (4 bytes length | flv packet)

### **rtmp\_dump\_republished\_video**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, write republished outgoing rtmp video (4 bytes length | flv packet)

### **rtmp\_flash\_ver\_publisher**

*Default:*

`FMLE/3.0`

*Type:*

`String`

*Need restart:* No

*Description:*

RTMP publisher Flash version

### **rtmp\_flash\_ver\_subscriber**

*Default:*

`LNX 9, 0, 124, 2`

*Type:*

`String`

*Need restart:* No

*Description:*

RTMP subscriber Flash version

### **rtmp\_in\_buffer\_clear\_threshold**

*Default:*

`30000`

*Type:*

`Long`

*Need restart:* No

*Description:*

once reached in overflow state buffer will clear up to overflow lower bound

### **rtmp\_in\_buffer\_enabled**

*Default:*

`false`

*Type:*

`Boolean`



*Need restart:* No

*Description:*

If true, enable buffer for incoming RTMP streams

### **rtmp\_in\_buffer\_increase\_rate**

*Default:*

0.25

*Type:*

Double

*Need restart:* No

*Description:*

buffer increase rate

### **rtmp\_in\_buffer\_initial\_size**

*Default:*

2000

*Type:*

Long

*Need restart:* No

*Description:*

Initial size of incoming RTMP buffer in milliseconds

### **rtmp\_in\_buffer\_input\_delay\_threshold**

*Default:*

0

*Type:*

Long

*Need restart:* No

*Description:*

Once stream delay reached the threshold buffer will passthrough stream without buffering, 0 - turned off

### **rtmp\_in\_buffer\_max\_bufferings\_allowed**

*Default:*

-1

Type:

Integer

Need restart: No

Description:

rtmp\_in\_buffer\_max\_bufferings\_allowed

**rtmp\_in\_buffer\_overflow\_allowed\_deviation**

Default:

1000

Type:

Long

Need restart: No

Description:

max allowed difference between min(buffer) and max(buffer). if this constraint met over the rtmp\_in\_buffer\_overflow\_deviation\_window overflow state will be set leading to clock acceleration

**rtmp\_in\_buffer\_overflow\_deviation\_window**

Default:

30000

Type:

Integer

Need restart: No

Description:

window for gathering min and max buffer sizes over time, in ms

**rtmp\_in\_buffer\_overflow\_lower\_bound**

Default:

1000

Type:

Long

Need restart: No

Description:

Lower bound for switching from overflow to hold state

**rtmp\_in\_buffer\_overflow\_rate**

*Default:*

0.15

*Type:*

Double

*Need restart:* No

*Description:*

buffer clock acceleration rate. To calculate increase in output speed use  $(1 + \text{rate}) / \text{buffer} = 1/x$

### **rtmp\_in\_buffer\_overflow\_state\_change\_delay**

*Default:*

10000

*Type:*

Long

*Need restart:* No

*Description:*

Enforces delay between two consecutive overflow states

### **rtmp\_in\_buffer\_polling\_time**

*Default:*

100

*Type:*

Long

*Need restart:* No

*Description:*

Incoming RTMP buffer polling time in milliseconds

### **rtmp\_in\_buffer\_start\_size**

*Default:*

300

*Type:*

Long

*Need restart:* No

*Description:*

Start size of incoming RTMP buffer in milliseconds

### **rtmp\_metadata\_to\_sdp\_state**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Translate publishers metadata into sdp state, this is used in conjunction with mixer\_use\_sdp\_state'

### **rtmp\_out\_buffer\_enabled**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable buffer for outgoing RTMP streams

### **rtmp\_out\_buffer\_initial\_size**

*Default:*

`2000`

*Type:*

`Long`

*Need restart:* No

*Description:*

Initial size of outgoing RTMP buffer in milliseconds

### **rtmp\_out\_buffer\_max\_bufferings\_allowed**

*Default:*

`-1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

rtmp\_out\_buffer\_max\_bufferings\_allowed

### **rtmp\_out\_buffer\_overflow\_allowed\_deviation**

*Default:*

1000

*Type:*

Long

*Need restart:* No

*Description:*

max allowed difference between min(buffer) and max(buffer). if this constraint met over the rtmp\_in\_buffer\_overflow\_deviation\_window overflow state will be set leading to clock acceleration

### **rtmp\_out\_buffer\_overflow\_deviation\_window**

*Default:*

30000

*Type:*

Integer

*Need restart:* No

*Description:*

window for gathering min and max buffer sizes over time, in ms

### **rtmp\_out\_buffer\_overflow\_rate**

*Default:*

0.15

*Type:*

Double

*Need restart:* No

*Description:*

buffer clock acceleration rate. To calculate increase in output speed use  $(1 + \text{rate}) / \text{buffer} = 1/x$

### **rtmp\_out\_buffer\_polling\_time**

*Default:*

50

Type:

Long

Need restart: No

Description:

Outgoing RTMP buffer polling time in milliseconds

### **rtmp\_out\_buffer\_start\_size**

Default:

300

Type:

Long

Need restart: No

Description:

Start size of outgoing RTMP buffer in milliseconds

### **rtmp\_output\_writer\_bufsize**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

0

Type:

Integer

Need restart: No

Description:

Buffer time for FFRtmpOutputWriter old outbound buffer for as-RTMP cases

### **rtmp\_port\_from**

Default:

33001

Type:

Integer

Need restart: No

*Description:*

First port in RTMP ports range, for RTMP republisher

### **rtmp\_port\_to**

*Default:*

34000

*Type:*

Integer

*Need restart:* No

*Description:*

Last port in RTMP ports range, for RTMP republisher

### **rtmp\_ports\_auditor\_interval**

*Default:*

10000

*Type:*

Integer

*Need restart:* No

*Description:*

Audit interval for RTMP ports, in milliseconds

### **rtmp\_ports\_auditor\_max\_attempts**

*Default:*

3

*Type:*

Integer

*Need restart:* No

*Description:*

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 successful audits is reached

### **rtmp\_publisher\_ip**

*Default:*

0

Type:

String

Need restart: Yes

Description:

IPv4 address for outgoing RTMP publishing

#### **rtmp\_publisher\_start\_time\_ts**

Default:

1000

Type:

Long

Need restart: No

Description:

RTMP publisher start time

#### **rtmp\_pull\_agent\_account\_for\_lost\_audio**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, enable RTMP pull agent account for lost audio

#### **rtmp\_pull\_allow\_to\_reuse\_uri**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, allow to multiple pulling with the same URI

#### **rtmp\_pull\_rtp\_activity\_detection**

Default:

true



Type:

Boolean

Need restart: No

Description:

If true, enable RTP activity detection while RTMP pulling

### **rtmp\_push\_auto\_start**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, enable RTMP push autostart for newly published streams

### **rtmp\_push\_auto\_start\_url**

Default:

null

Type:

String

Need restart: No

Description:

RTMP server address to auto start pushing to

### **rtmp\_push\_restore**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, then reconnect after connection reset by peer

### **rtmp\_push\_restore\_attempts**

Default:

3

Type:

Integer

Need restart: No

Description:

RTMP push reconnect attempts

**rtmp\_push\_restore\_interval\_ms**

Default:

5000

Type:

Integer

Need restart: No

Description:

RTMP push reconnect interval in ms

**rtmp\_receive\_buffer\_size\_predictor\_factory**

Default:

2053

Type:

Integer

Need restart: Yes

Description:

RTMP receive buffer size predictor factory in bytes

**rtmp\_send\_video\_first**

Default:

false

Type:

Boolean

Need restart: Yes

Description:

Send video first in RTMP

**rtmp\_server\_channel\_receive\_buffer\_size**

Default:

0

Type:

Integer

Need restart: Yes

Description:

RTMP receive buffer size in bytes

### **rtmp\_server\_enabled**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

Enable/disable rtmp server

### **rtmp\_transponder\_force\_kframe\_interval**

Default:

true

Type:

Boolean

Need restart: No

Description:

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**

Default:

false

Type:

Boolean

Need restart: No

Description:

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**

*Default:*

`60`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Forced k-frame interval in frames. See also `rtmp_transponder_force_kframe_interval=` setting.

### **rtmp\_transponder\_metadata**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

RTMP transponder metadata

### **rtmp\_transponder\_send\_metadata**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, RTMP transponder will send metadata

### **rtmp\_transponder\_stream\_name\_prefix**

*Default:*

`rtmp_`

*Type:*

`String`

*Need restart:* No

*Description:*

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

### **rtmp\_use\_stream\_params\_as\_connection**

*Default:*

false

*Type:*

Boolean

*Need restart:* Yes

*Description:*

Use stream params as connection

### **rtp\_activity\_audio**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, RTP activity check is enabled for audio.

### **rtp\_activity\_audio\_exclude**

*Default:*

*Type:*

String

*Need restart:* No

*Description:*

Disable RTP activity for audio stream with name matches this regex pattern

### **rtp\_activity\_detecting**

*Default:*

null

*Type:*

String

*Need restart:* No

*Description:*

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\_timeout**

*Default:*

60

*Type:*

Long

*Need restart:* No

*Description:*

RTP activity timer in seconds

### **rtp\_activity\_video**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, RTP activity check is enabled for video.

If false, this check is enabled for audio only

### **rtp\_activity\_video\_exclude**

*Default:*

||

*Type:*

String

*Need restart:* No

*Description:*

Disable RTP activity for video stream with name matches this regex pattern

### **rtp\_bundle**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable rtp bundle

### **rtp\_elapsed\_time\_threshold**

*Default:*

`10000`

*Type:*

`Long`

*Need restart:* No

*Description:*

RTP elapsed time threshold, in milliseconds

### **rtp\_generator\_start\_timeout**

*Default:*

`2000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Time in ms for enable generator if no rtp in call

### **rtp\_in\_buffer**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, use RTP in buffer

### **rtp\_in\_buffer\_initial\_size**

*Default:*

2000

*Type:*

Integer

*Need restart:* No

*Description:*

Initial size of incoming RTP buffer in milliseconds

### **rtp\_in\_buffer\_polling\_time**

*Default:*

100

*Type:*

Long

*Need restart:* No

*Description:*

Incoming RTP buffer polling time in milliseconds

### **rtp\_in\_reset\_marker**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, use RTP in reset marker

### **rtp\_paced\_sender**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, enable paced sender for WebRTC video session. EXPERIMENTAL

### **rtp\_paced\_sender\_capacity**



*Default:*

200000000

*Type:*

Long

*Need restart:* No

*Description:*

RTP paced sender capacity

### **rtp\_paced\_sender\_increase\_interval**

*Default:*

50

*Type:*

Integer

*Need restart:* No

*Description:*

Paced sender increase interval

### **rtp\_paced\_sender\_initial\_rate**

*Default:*

200000

*Type:*

Integer

*Need restart:* No

*Description:*

Paced sender initial rate

### **rtp\_paced\_sender\_k\_deviation**

*Default:*

0.02

*Type:*

Double

*Need restart:* No

*Description:*

Paced sender K deviation

### **rtp\_paced\_sender\_k\_down**

*Default:*

0.02

*Type:*

Double

*Need restart:* No

*Description:*

Paced sender K down

### **rtp\_paced\_sender\_k\_up**

*Default:*

0.04

*Type:*

Double

*Need restart:* No

*Description:*

Paced sender K up

### **rtp\_paced\_sender\_period**

*Default:*

1000

*Type:*

Long

*Need restart:* No

*Description:*

RTP paced sender period

### **rtp\_paced\_sender\_queue\_size**

*Default:*

2000

*Type:*

Integer

*Need restart:* No

*Description:*

Outgoing queue maximum size

### **rtp\_paced\_sender\_refill**

*Default:*

200000000

*Type:*

Long

*Need restart:* No

*Description:*

RTP paced sender refill

### **rtp\_packet\_cache\_size**

*Default:*

250

*Type:*

Integer

*Need restart:* No

*Description:*

Cache size for sent packets. This is used only on video sessions to provide response to NACK requests

### **rtp\_receive\_buffer\_predicator\_size**

*Default:*

1500

*Type:*

Integer

*Need restart:* No

*Description:*

DatagramSocket constructing: receiveBufferSizePredicatorFactory size

### **rtp\_receive\_buffer\_size**

*Default:*

65536

*Type:*

Integer

*Need restart:* No

*Description:*

Buffer size for incoming RTP and SRTP (WebRTC).

DatagramSocket constructing: receiveBufferSize

### **rtp\_send\_buffer\_size**

*Default:*

65536

*Type:*

Integer

*Need restart:* No

*Description:*

Buffer size for outgoing RTP and SRTP (WebRTC).  
DatagramSocket constructing: sendBufferSize

### **rtp\_session\_init\_always**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true init rtp session for all media providers

### **rtsp.address**

*Default:*

0.0.0.0

*Type:*

InetAddress[]

*Need restart:* Yes

*Description:*

Listening address for RTSP server

### **rtsp.port**

*Default:*

554

*Type:*

Integer

*Need restart:* Yes

*Description:*

RTSP server port

### **rtsp\_activity\_timer\_cool\_off\_period**

*Default:*

1

*Type:*

Integer

*Need restart:* No

*Description:*

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**

*Default:*

60000

*Type:*

Integer

*Need restart:* No

*Description:*

If there is no subscribers for an RTSP stream within this timeout in milliseconds, corresponding RTSP session will be terminated

### **rtsp\_auth\_cnonce**

*Default:*

1234567890

*Type:*

String

*Need restart:* Yes

*Description:*

RTSP server port

### **rtsp\_client\_address**

*Default:*

0.0.0.0

Type:

InetAddress

Need restart: Yes

Description:

RTSP client address

### **rtsp\_client\_strip\_audio\_codecs**

Default:

null

Type:

String

Need restart: No

Description:

Comma-separated list of audio codecs which will not be used for RTSP

### **rtsp\_fail\_on\_error\_track**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, RTSP pulling fails on error in any track

### **rtsp\_interleaved\_channels**

Default:

null

Type:

String

Need restart: No

Description:

Interleaved mode channels: audio channels;video channels. Default: dynamic channels

### **rtsp\_interleaved\_enable\_rtcp**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable replying to RTCP packets on the RTSP interleaved channel

### **rtsp\_interleaved\_mode**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, interleaved mode for RTSP (RTP over RTSP/TCP) is enabled

### **rtsp\_pcap\_server\_custom\_stream\_name**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

User custom stream name for all pcap sources with incrementing #N at the end

### **rtsp\_pcap\_server\_handler\_redirect\_url**

*Default:*

`null`

*Type:*

`String`

*Need restart:* Yes

*Description:*

Rtsp pcap server redirect URL

### **rtsp\_pcap\_server\_redirect\_method**

*Default:*

OPTIONS

*Type:*

String

*Need restart:* Yes

*Description:*

Rtsp pcap server redirect method: OPTIONS/DESCRIBE

### **rtsp\_port\_from**

*Default:*

32001

*Type:*

Integer

*Need restart:* No

*Description:*

First TCP port in the port range for RTSP pooling agent

### **rtsp\_port\_to**

*Default:*

33000

*Type:*

Integer

*Need restart:* No

*Description:*

Last TCP port in the port range for RTSP pooling agent

### **rtsp\_ports\_auditor\_interval**

*Default:*

10000

*Type:*

Integer



*Need restart:* No

*Description:*

Audit interval for RTSP ports, in milliseconds

### **rtsp\_ports\_auditor\_max\_attempts**

*Default:*

**3**

*Type:*

**Integer**

*Need restart:* No

*Description:*

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 successful audits is reached

### **rtsp\_refresh\_requests\_limit**

*Default:*

**5**

*Type:*

**Integer**

*Need restart:* No

*Description:*

Maximum number of non-answered GET\_PARAMETER refresh requests. Stop sending refresh requests if the limit has been reached

### **rtsp\_server\_auth\_enabled**

*Default:*

**false**

*Type:*

**Boolean**

*Need restart:* No

*Description:*

If true, enable RTSP server authentication

### **rtsp\_server\_enabled**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, activate RTSP server

### **rtsp\_server\_forse\_interleave**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, force interleaved mode for RTSP server and answer with interleaved mode SDP

### **rtsp\_server\_packetization\_mode**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

H.264 packetization mode for RTSP server. FU-A by default

### **rtsp\_server\_profile\_level\_id**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

H.264 profile-level-id for RTSP server

### **rtsp\_server\_sdp\_filename**

*Default:*

`rtsp_server.sdp`

*Type:*

`String`

*Need restart:* No

*Description:*

Filename of RTSP server sdp

### **rtsp\_user\_agent**

*Default:*

`l`

*Type:*

`String`

*Need restart:* No

*Description:*

User agent indicated in RTSP packets

### **rvg\_timer\_activity**

*Default:*

`500`

*Type:*

`Integer`

*Need restart:* No

*Description:*

RVG timer interval in milliseconds

### **rvg\_timer\_delay**

*Default:*

500

*Type:*

Integer

*Need restart:* No

*Description:*

RVG timer initial delay in milliseconds

### **scheduling\_service\_core\_threads**

*Default:*

5

*Type:*

Integer

*Need restart:* Yes

*Description:*

Core threads count for scheduling service

### **sdp\_origin\_username**

*Default:*

Flashphoner

*Type:*

String

*Need restart:* No

*Description:*

Sdp Origin value

### **send\_receive\_buffer\_size**

*Default:*

1600

*Type:*

Integer

*Need restart:* Yes

*Description:*

RTMFP buffer size in bytes

### **send\_receive\_on\_incoming\_re\_invite**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, send receive' on incoming re-INVITE'

### **session\_idle\_timeout**

*Default:*

`300000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

RTMFP server-side timeout in milliseconds if no UDP messages received over RTMFP/UDP session

### **sessions\_auditor\_interval**

*Default:*

`60000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Audit interval for pending media sessions

### **sessions\_auditor\_session\_timeout**

*Default:*

`60000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Audit timeout for pending media sessions

### **set\_sync\_time\_from\_ts**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Workaround for SIP audio only

### **sfu\_bridge\_enabled**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

With the bridge SFU will wrap tracks into streams

### **sfu\_mongo\_storage\_database**

*Default:*

`wcs_sfu`

*Type:*

`String`

*Need restart:* Yes

*Description:*

MongoDB database name

### **sfu\_mongo\_storage\_url**

*Default:*

`mongodb://localhost/?w=majority`

*Type:*

`String`

*Need restart:* Yes

*Description:*

MongoDB connection url

### **sfu\_periodic\_fir\_request**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, then every `sfu_periodic_fir_request_interval` seconds WCS server sends an RTCP Full Intra Request (FIR) message to the input stream source'

### **sfu\_periodic\_fir\_request\_interval**

*Default:*

`30000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Interval to send RTCP FIR in milliseconds to SFU participant

### **sfu\_proxy\_pli\_requests**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, then every PLI request from viewer will be forwarded to source

### **sfu\_storage\_type**

*Default:*

`YAML`

*Type:*

`YAML`

`MONGO`

*Need restart:* Yes

*Description:*

SFU storage type, YAML or MONGO

### **sip.pre\_init**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, use SIP pre-initiation

### **sip\_add\_contact\_id**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, record SIP as RTMP stream and SIP as stream

### **sip\_as\_rtmp\_java\_client**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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**

*Default:*

`live`

*Type:*

`String`



*Need restart:* No

*Description:*

Sets RTMP AMF stream type for as-RTMP cases

### **sip\_auditor\_dialog\_timeout**

*Default:*

10000

*Type:*

Integer

*Need restart:* No

*Description:*

SIP auditor dialog timeout

### **sip\_auditor\_transaction\_timeout**

*Default:*

50000

*Type:*

Integer

*Need restart:* No

*Description:*

SIP auditor transaction timeout

### **sip\_dns\_failover**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, enable DNS failover.

See also sip\_srv\_lookup= option

### **sip\_force\_rtcp\_feedback**

*Default:*

false

Type:

Boolean

Need restart: No

Description:

If true, force rtcp feedback to sip provider

### **sip\_force\_session\_expires**

Default:

1800

Type:

Integer

Need restart: No

Description:

Forced session expiration timeout in seconds. WCS server will send refresh request before the timeout is reached

### **sip\_force\_tcp**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, force TCP usage for SIP messaging

### **sip\_invite\_params\_to\_headers**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, place SIP INVITE parameters to headers

### **sip\_msg\_listener**

*Default:*

`com.flashphoner.sdk.sip.ChangeCallIdListener`

*Type:*

`String`

*Need restart:* No

*Description:*

Full name of Java class that implements interface ISipMessageListener

```
public interface ISipMessageListener {  
    void processMessage(SIPMessage sipMessage);  
}
```

### **sip\_ports\_auditor\_interval**

*Default:*

`10000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Audit interval for SIP ports, in milliseconds

### **sip\_ports\_auditor\_max\_attempts**

*Default:*

`3`

*Type:*

`Integer`

*Need restart:* No

*Description:*

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 successful audits is reached

### **sip\_record\_stream**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, record SIP as RTMP stream and SIP as stream

### **sip\_remove\_video\_sdp\_section\_instead\_of\_adding\_inactive\_with\_zero\_port**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, fully remove video part of SDP. If false, just set video part to inactive

### **sip\_sdp\_unsupported\_protocols**

*Default:*

String

*Type:*

UDP/DTLS/SCTP

*Need restart:* No

*Description:*

null

### **sip\_session\_expires\_header**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, use Expires header

### **sip\_single\_route\_only**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, then traffic is passed only to the streaming engine, and is not passed to the SIP caller

### **sip\_srv\_lookup**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable DNS SRV lookup.

See also sip\_dns\_failover= option

### **sip\_thread\_pool\_size**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Size of SIP thread pool

### **sip\_timer**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Value of timer T1 according to RFC 3261, in milliseconds

### **sip\_traffic\_class**

*Default:*

`null`

Type:

String

Need restart: No

Description:

QoS class for SIP traffic

### **sip\_use\_netty**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, use Netty

### **sip\_use\_reentrant\_listener**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, enable SIP reentrant listener

### **sip\_use\_tls**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, TLS used for SIP connections

### **sip\_user\_agent\_shutdown\_timeout**

Default:

5000

Type:

Integer

Need restart: No

Description:

Timeout for remove sip user agent for unregister in sip provider. Default is 5000 ms

### **snapshot\_auto\_dir**

Default:

/usr/local/FlashphonerWebCallServer/snapshots

Type:

String

Need restart: No

Description:

Snapshots dir

### **snapshot\_auto\_enabled**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, then enable snapshot auto cut

### **snapshot\_auto\_naming**

Default:

mediaSessionId

Type:

String

Need restart: No

Description:

Snapshot auto naming

### **snapshot\_auto\_rate**

Default:

60

Type:

Integer

Need restart: No

Description:

Snapshot rate. By default save every 60 frame

### **snapshot\_auto\_retention**

Default:

20

Type:

Integer

Need restart: No

Description:

Snapshot retention. By default keep last 20 frames

### **snapshot\_taking\_attempts**

Default:

30

Type:

Integer

Need restart: Yes

Description:

The number of attempts to take a snapshot. By default 30

### **snapshot\_taking\_interval\_ms**

Default:

3000

Type:

Integer

Need restart: Yes

Description:

Snapshot taking interval. By default 3000 milliseconds

### **speex\_g711\_speex\_transcoding**



 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

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**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

Speex encoding settings used in transcoding featuring the codec.

*Default:*

8 - Quality

false - VBR encoding

8 - Quality of VBR

4 - Algorithmic complexity

### **start\_test**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, tests listed in streaming\_tests= setting will be launched after WCS server startup

### **stats**

*Default:*

false

*Type:*

Boolean

*Need restart:* Yes

*Description:*

If true, enable sampling for streams. The sampling is used for charts

### **stats\_average\_calculation\_window**

*Default:*

30

*Type:*

Integer

*Need restart:* Yes

*Description:*

Window size for general average stats calculation

### **stats\_bitrate\_window**

*Default:*

1000

*Type:*

Integer

*Need restart:* No

*Description:*

Window size to collect bitrate statistics

### **stats\_fps\_window**

*Default:*

1000

*Type:*

Integer

*Need restart:* No

*Description:*

Window size to collect FPS statistics

### **stats\_incoming\_stream\_monitor\_deviation\_threshold**

*Default:*

500

*Type:*

Integer

*Need restart:* No

*Description:*

If deviation between audio and video is greater than the threshold in milliseconds, it will be logged

### **stats\_sampling\_frequency**

*Default:*

1000

*Type:*

Long

*Need restart:* Yes

*Description:*

Interval in milliseconds. Stream sampling will be taken with the specified frequency

### **stream\_idle\_bitrate\_monitoring**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Enable monitoring of published streams based on bitrate

### **stream\_idle\_bitrate\_monitoring\_threshold\_bps**

*Default:*

10000

*Type:*

Long

*Need restart:* No

*Description:*

Lowest bitrate possible for the active stream

### **stream\_idle\_bitrate\_monitoring\_window\_sec**

*Default:*

120

*Type:*

Integer

*Need restart:* No

*Description:*

Mean stream bitrate calculation window in seconds

### **stream\_record\_policy**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

1

*Type:*

String

*Need restart:* No

*Description:*

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**

*Default:*

1

*Type:*

String

*Need restart:* No

*Description:*

If set, name of recorded file will be built using the specified template.

Example: {streamName}-{startTime}-{sessionId}-{mediaSessionId}-{login}-{audioCodec}-

{videoCodec}-{duration}

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**

*Default:*

**1**

*Type:*

**String**

*Need restart:* No

*Description:*

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**

*Default:*

**1**

*Type:*

**Integer**

*Need restart:* No

*Description:*

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**

*Default:*

**1000**

*Type:*

**Long**

*Need restart:* No

*Description:*

StreamingCustomStreamStressTest / Period in milliseconds. Each period a new subscriber will be added

### **streaming\_custom\_stream\_stress\_test\_stream\_name**

*Default:*

**STRESS\_TEST\_STREAM**

*Type:*

**String**

*Need restart:* No

*Description:*

StreamingCustomStreamStressTest / Name of stream published on WCS server, which will be used for the test

**streaming\_custom\_stream\_stress\_test\_subscriber\_ttl\_sec**

*Default:*

**30**

*Type:*

**Long**

*Need restart:* No

*Description:*

StreamingCustomStreamStressTest / Lifetime of subscriber in seconds

**streaming\_distributor\_audio\_subgroup\_queue\_max\_waiting\_time**

*Default:*

**5000**

*Type:*

**Integer**

*Need restart:* No

*Description:*

Maximum time that subgroup thread will wait for frame arrival before executing next iteration

**streaming\_distributor\_audio\_subgroup\_queue\_size**

*Default:*

**300**

*Type:*

**Integer**

*Need restart:* No

*Description:*

Size of queue for distribution subgroup)

### **streaming\_distributor\_audio\_subgroup\_size**

*Default:*

500

*Type:*

Integer

*Need restart:* No

*Description:*

Video sessions per group

### **streaming\_distributor\_dump\_interval**

*Default:*

10

*Type:*

Integer

*Need restart:* Yes

*Description:*

Interval in minutes for getting distributor thread dumps

### **streaming\_distributor\_queue\_max\_waiting\_time**

*Default:*

5000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Maximum time that distributor thread will wait for frame arrival before executing next iteration

### **streaming\_distributor\_queue\_size**

*Default:*

300

*Type:*

Integer

*Need restart:* Yes

*Description:*

Size of queue. Processor will block distributor queue upon it reaching this size (i.e., no more space for new frames)

#### **streaming\_distributor\_queue\_size\_dump\_threshold**

*Default:*

0.95

*Type:*

Double

*Need restart:* No

*Description:*

Distributor queue size threshold for getting dump

#### **streaming\_distributor\_queue\_size\_log\_threshold**

*Default:*

10

*Type:*

Integer

*Need restart:* Yes

*Description:*

Threshold for logging distributor queue size

#### **streaming\_distributor\_subgroup\_enabled**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Enable video distribution subgroups)

#### **streaming\_distributor\_subgroup\_queue\_max\_waiting\_time**

*Default:*

5000



Type:

Integer

Need restart: No

Description:

Maximum time that subgroup thread will wait for frame arrival before executing next iteration

### **streaming\_distributor\_subgroup\_queue\_size**

Default:

300

Type:

Integer

Need restart: No

Description:

Size of queue for distribution subgroup)

### **streaming\_distributor\_subgroup\_size**

Default:

2

Type:

Integer

Need restart: No

Description:

Video sessions per group

### **streaming\_distributor\_video\_proxy\_pool\_enabled**

Default:

false

Type:

Boolean

Need restart: No

Description:

Use thread pool for video distribution, proxy only

### **streaming\_load\_test\_duration\_minutes**

*Default:*

5

*Type:*

Long

*Need restart:* No

*Description:*

StreamingLoadTest / Test duration in minutes

### **streaming\_load\_test\_encoding\_subscriber\_groups**

*Default:*

1

*Type:*

String

*Need restart:* No

*Description:*

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**

*Default:*

1

*Type:*

Integer

*Need restart:* No

*Description:*

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**

*Default:*

5000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Maximum time that processor thread will wait for frame arrival before executing next iteration

**streaming\_processor\_queue\_size**

*Default:*

300

*Type:*

Integer

*Need restart:* Yes

*Description:*

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**

*Default:*

■

*Type:*

String

*Need restart:* No

*Description:*

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**

*Default:*

10000

*Type:*

Long

*Need restart:* No

*Description:*

StreamKeepAliveEvent sending interval. See also streaming\_sessions\_keep\_alive\_app\_keys= option

**streaming\_stress\_test\_duration\_minutes**

*Default:*

5

*Type:*

Long

*Need restart:* No

*Description:*

StreamingStressTest / Test duration in minutes

### **streaming\_stress\_test\_encoding\_subscriber\_groups**

*Default:*

1

*Type:*

String

*Need restart:* No

*Description:*

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**

*Default:*

100

*Type:*

Integer

*Need restart:* Yes

*Description:*

Websocket connections to test

### **streaming\_stress\_test\_rate**

*Default:*

1000

*Type:*

Long

*Need restart:* No

*Description:*

StreamingStressTest / Period in milliseconds. Each period a new subscriber will be added

### **streaming\_stress\_test\_subscriber\_ttl\_sec**

*Default:*

30

*Type:*

Long

*Need restart:* No

*Description:*

StreamingStressTest / Lifetime of subscriber in seconds

### **streaming\_tests**

*Default:*

!

*Type:*

String

*Need restart:* No

*Description:*

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**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

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

### **streaming\_video\_decoder\_wait\_for\_distributors**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Stop decoding temporarily if one of the distributors fails to keep up with decoding

### **streaming\_video\_decoder\_wait\_for\_distributors\_max\_queue\_size**

*Default:*

`5`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Stop decoding when one of distributors queue reaches specified size (See streaming\_video\_decoder\_wait\_for\_distributors)

### **streaming\_video\_decoder\_wait\_for\_distributors\_timeout**

*Default:*

`33`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Specifies how long decoding should wait before another distributors queue check (See streaming\_video\_decoder\_wait\_for\_distributors)

### **streaming\_video\_decoder\_warmup**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Warmup video decoder with P frame after I frame regardless of decoding point availability

### **streaming\_video\_decoder\_warmup\_frames**

*Default:*

**5**

*Type:*

**Integer**

*Need restart:* No

*Description:*

How many P frames should be used for warmup

### **strict\_get\_callee\_policy**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

**false**

*Type:*

**Boolean**

*Need restart:* No

*Description:*

Not in use

### **stun\_freshness\_period**

*Default:*

**1500**

*Type:*

**Integer**

*Need restart:* No

*Description:*

STUN freshness period in milliseconds

### **stun\_freshness\_timeout**

*Default:*

`15000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

STUN freshness timeout in milliseconds

### **stun\_server**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`stun1.l.google.com:19302`

*Type:*

`String`

*Need restart:* No

*Description:*

STUN server, which is used for WebRTC ICE, if `enable_candidate_harvester=true`

### **stun\_socket\_buffer\_size**

*Default:*

`100`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Size of STUN socket buffer

### **stun\_socket\_queue\_timeout**

*Default:*

`1500`

*Type:*

`Integer`



*Need restart:* No

*Description:*

STUN socket queue timeout in milliseconds

### **stun\_stack\_default\_thread\_pool\_size**

*Default:*

`0`

*Type:*

`Integer`

*Need restart:* No

*Description:*

STUN default thread pool size

### **stun\_wait\_candidate\_timeout**

*Default:*

`1000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

STUN waiting candidate timeout for nominate in milliseconds

### **suppress\_audio**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, globally suppress audio on server. This feature is not available for Trial license

### **suppress\_dynamic\_logs**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, suppress dynamic logs update

**suppress\_dynamic\_logs\_to\_server\_log**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, suppress dynamic server logs update

**sync\_time\_force\_newest**

*Default:*

true

*Type:*

Boolean

*Need restart:* Yes

*Description:*

Force newest sync time after sync change, prevent new packet from getting sync time from past

**tcp\_relay\_packetization2**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

If true, enable TCP relay packetization for WSPlayer. Should be false when WSPlayer 1.0 is used

**tcp\_relay\_packetization\_time**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

**20**

*Type:*

**Integer**

*Need restart:* No

*Description:*

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**

*Default:*

**2000**

*Type:*

**Integer**

*Need restart:* No

*Description:*

RTCP packets generation interval for TCP relay in milliseconds. RTCP is used to carry stream synchronization

### **thread\_pool\_default\_core\_threads**

*Default:*

**4**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

Default core threads count in thread pool (equal to CPUs count)

### **thread\_pool\_default\_max\_threads**

*Default:*

**8**

Type:

Integer

Need restart: Yes

Description:

Maximum core threads count in thread pool

### **thread\_pool\_default\_queue\_size**

Default:

100

Type:

Integer

Need restart: Yes

Description:

Default thread pool queue size

### **thread\_pool\_default\_thread\_timeout\_sec**

Default:

300

Type:

Integer

Need restart: Yes

Description:

Default thread timeout, in seconds

### **thread\_pools\_config\_filename**

Default:

thread\_pools\_config.json

Type:

String

Need restart: Yes

Description:

Thread pools config filename

### **throughput\_test\_receivers\_qty**

Default:

1

Type:

Integer

Need restart: No

Description:

Throughput test receivers quantity

### throughput\_test\_sender\_dst

Default:

localhost

Type:

String

Need restart: No

Description:

Throughput test sender destination host

### throughput\_test\_senders\_qty

Default:

1

Type:

Integer

Need restart: No

Description:

Throughput test senders quantity

### timing\_shift

 **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

null

Type:

String

Need restart: No

*Description:*

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**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, trace usage of socket file descriptors for HLS, HTTP, RTSP, WebSockets and HTTP LB client

**transcoder\_agent\_activity\_timer\_cool\_off\_period**

*Default:*

`1`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Transcoder agent will be terminated after  $\{\text{rtsp\_activity\_timer\_cool\_off\_period} * \text{rtsp\_activity\_timer\_timeout}\}$  since last subscriber activity for the corresponding RTSP stream

**transcoder\_agent\_activity\_timer\_timeout**

*Default:*

`60000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

If there is no subscribers for an Transcoder agent stream within this timeout in milliseconds, corresponding RTSP session will be terminated

**transcoder\_agent\_key\_frame\_interval**

*Default:*

60

*Type:*

Integer

*Need restart:* No

*Description:*

Transcoder agent key frame interval

### **transcoder\_agent\_rtcp\_send\_interval**

*Default:*

2000

*Type:*

Long

*Need restart:* No

*Description:*

Interval in ms for send rtcp from transcoder agent

### **transcoder\_align\_encoders**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Align video encoders of the same video input by key frames

### **transcoding\_disabled**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Force transcoding disabling

### **turn.server\_channel\_receive\_buffer\_size**

*Default:*

1048576

*Type:*

Integer

*Need restart:* Yes

*Description:*

Receive buffer size for turn channels

#### **turn.server\_channel\_send\_buffer\_size**

*Default:*

1048576

*Type:*

Integer

*Need restart:* Yes

*Description:*

Send buffer size for turn channels

#### **turn\_ip**

*Default:*

0.0.0.0

*Type:*

InetAddress

*Need restart:* Yes

*Description:*

TURN external IP address

#### **turn\_ip\_local**

*Default:*

0.0.0.0

*Type:*

InetAddress

*Need restart:* Yes

*Description:*

TURN internal IP address

#### **turn\_life\_time**



*Default:*

600

*Type:*

Integer

*Need restart:* Yes

*Description:*

TURN Allocation life time

### **turn\_media\_port\_from**

*Default:*

36001

*Type:*

Integer

*Need restart:* Yes

*Description:*

Beginning of media ports range for turn

### **turn\_media\_port\_to**

*Default:*

37000

*Type:*

Integer

*Need restart:* Yes

*Description:*

End of media ports range for turn

### **turn\_media\_ports\_auditor\_interval**

*Default:*

5000

*Type:*

Integer

*Need restart:* Yes

*Description:*

Audit interval for busy and free ports, in milliseconds

### **turn\_media\_ports\_auditor\_max\_attempts**

*Default:*

**3**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

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 successful audits is reached

### **turn\_password**

*Default:*

**coM77EMrV7Cwhyan**

*Type:*

**String**

*Need restart:* Yes

*Description:*

TURN password

### **turn\_port**

*Default:*

**3478**

*Type:*

**Integer**

*Need restart:* Yes

*Description:*

TURN server port

### **unsupported\_messages**

*Default:*

**null**

*Type:*

**String**

*Need restart:* No

*Description:*

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**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, switch to the local codec according to content received from SIP side.  
If false, use Speex16

**use\_control\_destination\_from\_incoming\_rtcp**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, set RTCP destination by received RTCP packets

**use\_fdk\_aac**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, use the fdk-aac for encoding and decoding

### **use\_ip\_local\_in\_call\_id**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, use value of ip\_local= option when forming callID

### **use\_mp4\_h264\_aac**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, use H.264 + AAC in MP4 container

### **use\_new\_aac\_encoder**

#### **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, use the latest AAC encoder

### **use\_new\_injector**

*Default:*

`true`

Type:

`Boolean`

Need restart: Yes

Description:

If true, enable new injectors

#### **use\_new\_rtcp**

##### **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

`true`

Type:

`Boolean`

Need restart: No

Description:

If true, use the latest RTCP module

#### **use\_rtcp\_synch**

##### **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

`true`

Type:

`Boolean`

Need restart: No

Description:

If true, use RTCP synchronization for audio and video

#### **use\_rtmp\_java\_client**

Default:

`true`

Type:

Boolean

Need restart: No

Description:

If true, use the latest implementation of RTMP agent for republishing

### **use\_speex\_java\_impl**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

If true, use Java implementation for Speex codec

### **use\_strict\_jitter\_buffer**

Default:

false

Type:

Boolean

Need restart: No

Description:

Enables strict jitter buffer

### **use\_tcp\_for\_long\_sip\_messages**

Default:

false

Type:

Boolean

Need restart: No

Description:

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**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, then broadcast SIP response TRYING to client as a call status TRYING

### **user\_agent**

*Default:*

`Flashphoner/1.0`

*Type:*

`String`

*Need restart:* Yes

*Description:*

User-Agent header value

### **video\_bitstream\_normalizer\_consecutive\_ts\_errors\_threshold**

*Default:*

`90`

*Type:*

`Integer`

*Need restart:* No

*Description:*

How many consecutive timestamp errors normalizer can absorb before falling back to original stream timestamp.

### **video\_decoder\_max\_threads**

*Default:*

`2`

*Type:*

`Integer`

*Need restart:* No

*Description:*

How many threads will be used for decoding

### **video\_decoder\_second\_thread\_threshold**

*Default:*

`777000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Resolution threshold. Once it is reached, decoder should start using second thread.

Example: 800x600 = 480000, 1280x720=921600. So, by default all 720p streams will be decoded using two CPU threads

### **video\_distributor\_multi\_test**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable video distributor multi test

### **video\_enabled**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Not in use

### **video\_encoder\_h264\_gop**



*Default:*

60

*Type:*

Integer

*Need restart:* No

*Description:*

GOP size for H.264 encoder

### **video\_encoder\_h265\_preset**

*Default:*

ultrafast

*Type:*

String

*Need restart:* No

*Description:*

Preset value for H.265 encoder

### **video\_encoder\_max\_threads**

*Default:*

3

*Type:*

Integer

*Need restart:* No

*Description:*

How many threads will be used for encoding

### **video\_encoder\_second\_thread\_threshold**

*Default:*

777000

*Type:*

Integer

*Need restart:* No

*Description:*

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**

*Default:*

`900`

*Type:*

`Integer`

*Need restart:* No

*Description:*

GOP size for VP8 encoder

### **video\_encoding\_quality**

*Default:*

`30`

*Type:*

`Integer`

*Need restart:* No

*Description:*

See information on FFmpeg CRF

### **video\_filter\_enable\_fps**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Enable video filter

### **video\_filter\_enable\_rotate**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Enable video rotate filter

### **video\_filter\_fps**

*Default:*

30

*Type:*

Long

*Need restart:* Yes

*Description:*

Video filter output fps

### **video\_filter\_fps\_gap\_coefficient**

*Default:*

2.0

*Type:*

Double

*Need restart:* Yes

*Description:*

Video filter gap coefficient (max gap C x FPS)

### **video\_filter\_fps\_gop\_synchronization**

*Default:*

0

*Type:*

Integer

*Need restart:* No

*Description:*

Filters gop value used to provide synchronization point for encoders, use with TRANSCODER\_ALIGN\_ENCODERS'

### **video\_force\_sync\_timeout**

*Default:*

100

*Type:*

Integer

*Need restart:* No

*Description:*

Waiting for RTCP sync packet on this interval in ms, for video

### **video\_mixer\_output\_codec**

*Default:*

`h264`

*Type:*

`String`

*Need restart:* No

*Description:*

Video mixer output codec (multiple codecs not allowed)

### **video\_processor\_multi\_test**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable video processor multi test

### **video\_reliable**

*Default:*

`partial`

*Type:*

`on`

`partial`

`off`

*Need restart:* No

*Description:*

RTMFP, reliability for video

### **video\_stream\_mode\_udp**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Not in use

**video\_streamer\_generate\_seq**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Should be set to true for transfer of video calls. Otherwise, there may be no video after transfer

**video\_transcoder\_preserve\_aspect\_ratio**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

Try to preserve original aspect ratio of incoming video during transcoding

**video\_transcoder\_round\_ratio**

*Default:*

`0`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Rounding up or down when preserving aspect ratio

### **vod\_activity\_timer\_cool\_off\_period**

*Default:*

**1**

*Type:*

**Integer**

*Need restart:* No

*Description:*

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**

*Default:*

**false**

*Type:*

**Boolean**

*Need restart:* No

*Description:*

If true, loop streaming MP4 file as VoD. EXPERIMENTAL

### **vod\_mp4\_container\_isoparser\_heap\_datasource**

*Default:*

**true**

*Type:*

**Boolean**

*Need restart:* No

*Description:*

If true, use heap datasource

### **vod\_mp4\_container\_new**

*Default:*

**false**

Type:

Boolean

Need restart: No

Description:

Use new implementation of mp4 container for vod

#### **vod\_mp4\_container\_new\_buffer\_ms**

Default:

0

Type:

Integer

Need restart: No

Description:

New implementation of mp4 container will buffer specified time in milliseconds

#### **vod\_mp4\_test\_file**

Default:

null

Type:

String

Need restart: No

Description:

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**

##### **Warning**

Deprecated parameter. Will be deleted in future releases

Default:

true

Type:

Boolean

Need restart: No

*Description:*

If true, loop streaming MP4 file. Not in use, replaced by vod\_live\_loop=

### **vod\_mp4\_test\_stream\_name**

*Default:*

`null`

*Type:*

`String`

*Need restart:* No

*Description:*

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\_rtcp\_send\_interval**

*Default:*

`2000`

*Type:*

`Long`

*Need restart:* No

*Description:*

RTCP Send interval for VOD

### **vod\_sink\_ready\_checks**

*Default:*

`50`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Waiting for first packet on audio streamer. If no packets within the specified number of checks, then audio injection is aborted

### **vod\_sink\_wait\_synch\_time**

*Default:*

`true`



Type:

Boolean

Need restart: No

Description:

If false, not wait sync time for playing incoming traffic after audio sink

### **vod\_stream\_timeout**

Default:

30000

Type:

Integer

Need restart: No

Description:

VoD stream with no subscribers will be terminated after this timeout in milliseconds

### **vow\_wait\_for\_sync**

Default:

false

Type:

Boolean

Need restart: No

Description:

If true, session will wait for audio AND video before sending stream to client

### **vp8\_buffer\_nack\_list\_threshold**

Default:

200

Type:

Integer

Need restart: No

Description:

JitterBuffer will be reset upon reaching this number of NACK packets

### **vp8\_max\_rtp\_packet\_size**

Default:

1400

Type:

Integer

Need restart: Yes

Description:

Maximum size of VP8 carrying packet

### vp8\_new\_buffer

#### Warning

Deprecated parameter. Will be deleted in future releases

Default:

false

Type:

Boolean

Need restart: No

Description:

Not in use

### wcs\_activity\_timer\_cool\_off\_period

Default:

1

Type:

Integer

Need restart: No

Description:

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

Default:

60000

Type:

Integer

*Need restart:* No

*Description:*

If there is no activity within this timeout in milliseconds, corresponding WCS agent session will be terminated

#### **wcs\_agent\_force\_video\_orientation**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Force negotiation of 3gpp video orientation extension for wcs agent's

#### **wcs\_agent\_port\_from**

*Default:*

`34001`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Beginning of range of ports for WCS agent

#### **wcs\_agent\_port\_to**

*Default:*

`35000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

End of range of ports for WCS agent

#### **wcs\_agent\_ports\_auditor\_interval**

*Default:*

`10000`

Type:

Integer

Need restart: No

Description:

Audit interval for WCS agent ports, in milliseconds

#### **wcs\_agent\_ports\_auditor\_max\_attempts**

Default:

3

Type:

Integer

Need restart: No

Description:

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 successful audits is reached

#### **wcs\_agent\_session\_alive\_check\_interval**

Default:

30000

Type:

Integer

Need restart: No

Description:

Interval in milliseconds to check if WCS agent session is alive

#### **wcs\_agent\_session\_audit**

Default:

true

Type:

Boolean

Need restart: No

Description:

If true, enable WCS agent session audit

#### **wcs\_agent\_session\_connect\_timeout**

*Default:*

`10000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Connect timeout in milliseconds

### **wcs\_agent\_session\_timeout**

*Default:*

`30000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

WCS agent session timeout in milliseconds

### **wcs\_agent\_session\_use\_keep\_alive\_timeout**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, WCS agent session will use keep alive timeout

### **wcs\_agent\_ssl**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable SSL for pulling/re-publishing streams

### **wcs\_agent\_uri\_path**

*Default:*

`/websocket`

*Type:*

`String`

*Need restart:* Yes

*Description:*

WCSAgent ws request uri path

### **wcs\_sfu\_bridge\_enabled**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

With the WCS SFU bridge wcs will wrap streams into tracks

### **wcsoam\_batch\_timeout**

*Default:*

`500`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

WCS OAM receive timeout

### **wcsoam\_buffer\_size**

*Default:*

`20000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

WCS OAM buffer size in kB

### **wcsoam\_chunk\_size**

*Default:*

64

*Type:*

Integer

*Need restart:* Yes

*Description:*

WCS OAM send chunk size in kB

### **wcsoam\_hostname**

*Default:*

null

*Type:*

String

*Need restart:* Yes

*Description:*

WCS OAM server hostname

### **wcsoam\_ip**

*Default:*

null

*Type:*

String

*Need restart:* Yes

*Description:*

WCS OAM server IP address

### **wcsoam\_keepalive\_period**

*Default:*

3000

*Type:*

Integer

*Need restart:* Yes

*Description:*

WCS OAM keep alive period

### **wcsoam\_keepalive\_timeout**

*Default:*

8000

*Type:*

Integer

*Need restart:* Yes

*Description:*

WCS OAM keep alive timeout

### **wcsoam\_ping\_enabled**

*Default:*

true

*Type:*

Boolean

*Need restart:* No

*Description:*

WCS OAM server ping enable

### **wcsoam\_ping\_interval**

*Default:*

10000

*Type:*

Integer

*Need restart:* Yes

*Description:*

WCS OAM server ping interval in ms

### **wcsoam\_port**

*Default:*

7777

*Type:*

Integer

*Need restart:* Yes

*Description:*

WCS OAM server port

### **wcsoam\_reconnect\_interval**



*Default:*

5000

*Type:*

Integer

*Need restart:* Yes

*Description:*

WCS OAM reconnect interval in ms

### **wcsoam\_sha\_salt**

*Default:*

123

*Type:*

String

*Need restart:* Yes

*Description:*

WCS OAM server SHA salt

### **web\_start\_with\_demo\_user**

*Default:*

false

*Type:*

Boolean

*Need restart:* No

*Description:*

Enable demo user

### **web\_token\_life\_time**

*Default:*

3600000

*Type:*

Long

*Need restart:* No

*Description:*

Web token life time, default value 1 hour

### **webm\_java\_writer\_enable**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Replace ffmpeg webm writer with java implementation

### **webrtc\_aes\_crypto\_provider**

*Default:*

`BC`

*Type:*

`BC`

`JCE`

*Need restart:* No

*Description:*

Crypto provider for WebRTC

### **webrtc\_agent\_use\_webrtc**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, switch WebRTC push and pull to AVP profile

### **webrtc\_cc2**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, the latest congestion control CC2 is used

### **webrtc\_cc2\_bitrate\_overuse\_event**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable NBE event raising

### **webrtc\_cc2\_bitrate\_overuse\_event\_interval**

*Default:*

`5000`

*Type:*

`Long`

*Need restart:* No

*Description:*

NBE event will be raised periodically with this interval in milliseconds

### **webrtc\_cc2\_bitrate\_overuse\_event\_threshold**

*Default:*

`0.05`

*Type:*

`Double`

*Need restart:* No

*Description:*

NBE event will be raised when loss on stream being played reaches this value (5% by default)

### **webrtc\_cc2\_cc**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, react upon WebRTC playback endpoint (e.g. Chrome) requests, e.g. request the

publisher to decrease bitrate

### **webrtc\_cc2\_cc\_interval**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

500

*Type:*

Long

*Need restart:* No

*Description:*

Congestion control interval, not in use

### **webrtc\_cc2\_cc\_k\_noise**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

0.1

*Type:*

Double

*Need restart:* No

*Description:*

Congestion control noise value, not in use

### **webrtc\_cc2\_cc\_retransmit\_rate\_threshold**

*Default:*

0.15

*Type:*

Double

*Need restart:* No

*Description:*

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**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, enable tracking of retransmit bitrate across all media groups

**webrtc\_cc2\_enable\_burst\_grouping**

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public. CC2 estimation will account for packet burst

**webrtc\_cc2\_local\_congestion\_event\_interval**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

`2000`

*Type:*

`Long`

*Need restart:* No

*Description:*

Not in use, legacy code

## webrtc\_cc2\_local\_k\_threshold

### Warning

Deprecated parameter. Will be deleted in future releases

*Default:*

0.1

*Type:*

Double

*Need restart:* No

*Description:*

Not in use, legacy code

## webrtc\_cc2\_min\_remb\_bitrate\_bps

*Default:*

100000

*Type:*

Long

*Need restart:* No

*Description:*

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

*Default:*

1000

*Type:*

Long

*Need restart:* No

*Description:*

Window size for receiver state, in milliseconds. Default: 1000 - keep and account reports received in last second

## webrtc\_cc2\_twcc

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, enable TWCC reports. EXPERIMENTAL

### **webrtc\_cc\_bitrate\_window**

*Default:*

`1000`

*Type:*

`Integer`

*Need restart:* No

*Description:*

Time window in milliseconds. Bitrate estimator works on this time frame

### **webrtc\_cc\_initial\_avg\_noise**

*Default:*

`0.0`

*Type:*

`Double`

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_initial\_e\_0\_0**

*Default:*

`100.0`

*Type:*

`Double`

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_initial\_e\_0\_1**

*Default:*

0.0

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

#### **webrtc\_cc\_initial\_e\_1\_0**

*Default:*

0.0

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

#### **webrtc\_cc\_initial\_e\_1\_1**

*Default:*

0.1

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

#### **webrtc\_cc\_initial\_offset**

*Default:*

0.0

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

#### **webrtc\_cc\_initial\_process\_noise\_0**



*Default:*

1.0E-13

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_initial\_process\_noise\_1**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

0.001

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_initial\_slope**

*Default:*

0.015625

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_initial\_threshold**

 **Warning**

Deprecated parameter. Will be deleted in future releases

*Default:*

15.0

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_initial\_var\_noise**

*Default:*

50.0

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_k\_down**

*Default:*

1.8E-4

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_k\_up**

*Default:*

0.01

*Type:*

Double

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_max\_bitrate**

*Default:*

`10000000`

*Type:*

`Long`

*Need restart:* No

*Description:*

Maximum global bitrate for publishing WebRTC streams

### **webrtc\_cc\_min\_bitrate**

*Default:*

`30000`

*Type:*

`Long`

*Need restart:* No

*Description:*

Minimum global bitrate for publishing WebRTC streams

### **webrtc\_cc\_overusing\_threshold**

*Default:*

`10.0`

*Type:*

`Double`

*Need restart:* No

*Description:*

Internal bitrate estimation configuration, must not be exposed to public

### **webrtc\_cc\_use\_sync\_ts**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

If true, timestamp is used as synchronization source

### **webrtc\_pre\_init**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* Yes

*Description:*

If true, use WebRTC pre-initialization

### **webrtc\_pre\_init\_timeout**

*Default:*

`20000`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

Maximum allowed time for WebRTC preinit

### **webrtc\_sdes\_extensions**

*Default:*

`true`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Enable sdes rtp header extensions

### **webrtc\_sdp\_bandwidth\_bps**

*Default:*

`0`

*Type:*

`Long`

*Need restart:* No

*Description:*

b=AS/b=TIAS in publish sdp

### **webrtc\_sdp\_h264\_exclude\_profiles**

*Default:*

█

*Type:*

String

*Need restart:* No

*Description:*

List of H264 profiles which should be excluded in response on SDP negotiation.

42 - Baseline, 4d - Main, 64 - High

### **webrtc\_sdp\_max\_bitrate\_bps**

*Default:*

0

*Type:*

Long

*Need restart:* No

*Description:*

x-google-max-bitrate in publish sdp

### **webrtc\_sdp\_min\_bitrate\_bps**

*Default:*

0

*Type:*

Long

*Need restart:* No

*Description:*

x-google-min-bitrate in publish sdp

### **websocket\_uri\_path**

*Default:*

█

*Type:*

String

*Need restart:* Yes

*Description:*

WebSocket request uri path

## work\_around

### Warning

Deprecated parameter. Will be deleted in future releases

*Default:*

`false`

*Type:*

`Boolean`

*Need restart:* No

*Description:*

Not in use

## ws.address

*Default:*

`0.0.0.0`

*Type:*

`InetAddress[]`

*Need restart:* Yes

*Description:*

Listening address for WebSocket server

## ws.dashboard.port

*Default:*

`8086`

*Type:*

`Integer`

*Need restart:* Yes

*Description:*

WebSocket dashboard connection port

## ws.ip\_forward\_header

*Default:*

`X-Real-IP`

Type:

String

Need restart: No

Description:

Header for IP forwarding

### **ws.map\_custom\_headers**

Default:

false

Type:

Boolean

Need restart: Yes

Description:

If true, parse and inject custom HTTP headers to REST requests

### **ws.port**

Default:

8080

Type:

Integer

Need restart: Yes

Description:

WebSocket connection port

### **ws\_client\_id\_unique\_part**

Default:

true

Type:

Boolean

Need restart: No

Description:

Add unique part to ws client id

### **ws\_connections\_test\_run\_for**

Default:

1800

Type:

Integer

Need restart: Yes

Description:

Websocket connections test duration in seconds

### **ws\_connections\_test\_uri**

Default:

ws://192.168.88.100:8080

Type:

String

Need restart: Yes

Description:

Websocket connections test URI

### **ws\_read\_socket\_timeout**

Default:

true

Type:

Boolean

Need restart: Yes

Description:

Enable WebSocket read timeout

### **ws\_read\_socket\_timeout\_sec**

Default:

120

Type:

Integer

Need restart: Yes

Description:

WebSocket read timeout value (if enabled)

### **wss.address**

Default:

0.0.0.0



Type:

`InetAddress[]`

Need restart: Yes

Description:

Listening address for WebSocket SSL server

### **wss.cert.password**

Default:

`password`

Type:

`String`

Need restart: Yes

Description:

Key password to the SSL certificate in keystore

### **wss.dashboard.port**

Default:

`8446`

Type:

`Integer`

Need restart: Yes

Description:

Secure websocket dashboard connection port

### **wss.keystore.file**

Default:

`/usr/local/FlashphonerWebCallServer/conf/wss.jks`

Type:

`String`

Need restart: Yes

Description:

Keystore file containing SSL certificate for secure WebSocket connection

### **wss.keystore.password**

Default:

`password`

Type:

String

Need restart: Yes

Description:

SSL certificate keystore password

### **wss.port**

Default:

8443

Type:

Integer

Need restart: Yes

Description:

WebSocket SSL connection port

### **wss.ssl.cache\_size**

Default:

0

Type:

Integer

Need restart: Yes

Description:

SSL session objects cache size

### **wss.ssl.session\_timeout**

Default:

0

Type:

Integer

Need restart: Yes

Description:

Cached SSL session objects timeout, in seconds

### **zapp\_conf\_dir**

Default:

/usr/local/FlashphonerWebCallServer/conf/zclient/conf/

Type:

String

Need restart: Yes

Description:

Zapp configuration directory

### **zgc\_log\_parser\_enable**

Default:

false

Type:

Boolean

Need restart: Yes

Description:

Launch ZGC parser for stat output

### **zgc\_log\_parser\_path**

Default:

logs/gc-core-[0-9]{4}-[0-9]{2}-[0-9]{2}-[0-9]{2}-[0-9]{2}.log

Type:

String

Need restart: Yes

Description:

ZGC logs path to parse

### **zgc\_log\_time\_format**

Default:

yyyy-MM-dd'T'HH:mm:ss.SSSZ

Type:

String

Need restart: Yes

Description:

ZGC logs time format used to parse gc pause time