Server functions

Streaming video functions

IN - streams incoming to the server (publishers)
OUT - outgoing streams (spectators)

IN / OUT

WebRTC browser

Flash Player

MSE

WSPlayer

HLS

RTSP

Android app, WebRTC

iOS app, WebRTC

WebRTC Browser

Webcam

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Canvas

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Screen

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Flash Player

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RTMP encoder

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RTSP IP cam

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RTMP server

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WCS server

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SIP call

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Android app

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VOD

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Supported codecs

Audio	Hz	Video	Hz
Opus	48000	H.264	90000
Speex	16000	VP8	90000
G.711	8000		
AAC	48000		
AAC	44100		
G.729	8000		

Incoming streams operations

- Management of camera, microphone, bitrate, resolution etc.
- Mixing streams
- Taking a preview snapshot of a stream as PNG
- Recording streams
- Forced stopping of streams on the server
- Searching for current streams on the server

Video streams republishing functions

- To another RTMP server
- To another WCS server via WebRTC
- To a SIP call

Complex functions

- Working with chat rooms
- CDN 1.0, static
- CDN 2.0, dynamic
- CDN 2.1 with transcoding nodes

WebRTC-SIP gateway functions

From - the caller
To - the callee

From / To	WebRTC browser	Android App, WebRTC	iOS App, WebRTC	SIP
WebRTC browser	+	+	+	+
Android App, WebRTC	+	+	+	+
iOS App, WebRTC	+	+	+	+
SIP	+	+	+	+

Call management functions

- DTMF
- Hold
- Transfer
- Call recording

Supported codecs

Audio	Hz	Video	Hz
Opus Speex G.711 G.729	48000 16000 8000 8000	H.264 VP8	90000 90000

Additional capabilities

- Working through Firewall
- Load testing
- Load balancing