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	+	+	+	+	+	+	+	+
	Canvas	+	+	+	+	+	+	+	+
	Screen	+	+	+	+	+	+	+	+
Flash Player		+	+	+	+	+	+	+	+
RTMP encoder		+	+	+	+	+	+	+	+
RTSP IP cam		+	+	+	+	+	+	+	+
RTMP server		+	+	+	+	+	+	+	+
WCS server		+	+	+	+	+	+	+	+
SIP call		+	+	+	+	+	-	+	+
Android app		+	+	+	+	+	+	+	+
iOS app		+	+	+	+	+	+	+	+
VOD		+	+	+	+	+	+	+	+

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 transcoder 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	48000	H.264	90000
Speex	16000	VP8	90000
G.711	8000		
G.729	8000		

Additional capabilities

- Working through Firewall
- Load testing
- Load balancing