Stream capturing and publishing to the server

- From a web camera in a browser via WebRTC
- From the computer screen (screen sharing) in a browser via WebRTC
- From an HTML5 Canvas element (whiteboard) in a browser via WebRTC
- By means of Flash Player via RTMP
- Using RTMP encoder
    - Using Adobe FMLE
    - Using ffmpeg
    - Using OBS Studio
    - Using Wirecast
- From an IP camera via RTSP
- From another server via RTMP
- RTMP stream capturing by re-publishing from another RTMP server
- From another WCS server via WebRTC
- From an Android mobile app via WebRTC
- From an iOS mobile app via WebRTC
- Capturing VOD from a file
- RTP stream publishing via RTSP
- MPEG-TS RTP stream publishing
- WebRTC publishing via WHIP
- Automatic streams capture on server start
- RTSP-interleaved stream capture from dump file
- Managing camera and microphone
- Bitrate management when capturing WebRTC stream in browser
- Key frames management while capturing WebRTC in browser
- Published stream normalizing
- Jitter buffer and frames collection in stream published