WebRTC RTP bundle support

Overview

Since build 5.2.660, WCS supports sending and receiving audio and video tracks via the same connection while publishing or playing WebRTC stream. This allows to reduce media ports usage and to decrease server load. The feature is supported for both UDP and TCP transport.

RTP bundle is enabled by default, and will be used if client supports it. If some problem occurs while connection establishing, RTP bundle can be disabled with the following parameter in flashphoner.properties file

```
rtp_bundle=false
```

RTP bundle inside CDN

Since build 5.2.1759, RTP bundle works also for WebRTC sessions between CDN nodes, including a composite media session used to stream a couple of video qualities in HLS ABR case.

Known issues

1. Two busy media ports may be shown in server statistics

When RTP bundle is enabled, the server statistics page may show 2 busy ports per connection because media ports are reserved before SDP exchange is completed, and server detects how much ports client will use.