Setup and testing

First client launch and setup

1. On first launch, a license agreement will be displayed. Press OK





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3. Press **OK** in the next window

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1	If you change your mind, you can run the auto-configuration wizard ar again from the Tools menu.	ıy time
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4. Press Settings button. On Stream tab choose WebRTC Flashphoner Streaming Server in Stream Type drop-down list. Enter Secure Websocket server URL to Web Server Name field, for example wss://demo.flashphoner.com:8443, and set stream name to Server Room field, for example test_obs

R	General	Stream Type	WebRTC Flashphoner Streaming Server	
9	Stream	Madifying i Web Server Name	re bandweith manualy will not affect the quality of the stream as it is managed automatically by WebRITS. wss://demo.flashphoner.com:8443	
\bigcirc	Output	Server Room	test_obs	
	Audio			
Š	Video			
	Hotkeys			
	Advanced			

If necessary, set maximum video resolution on Video tab, the settings are the same as OBS Studio. Press Ok to save and apply settings.

5. Add media source, local file for example

	🗹 Local File			
Local File			Browse	
	🗹 Loop			
	Restart playback when source becomes active			
	🗹 Use hardware decoding when available			
	🗹 Hide source when playback ends			
	Close file when inactive			
Defaults		ОК	Cance	

6. To escape echo while stream publishing, mute all other sound sources excluding media source chosen

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Mic/Aux		-3.3 dB
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Martia Source		0.0 dB

Translation is ready to start.

Server setup

Enforce bitrate increasing

To prevent translation quality decreasing while Full HD stream is publishing, bitrate increasing should be enforced on server side with the following parameters in flashphoner.properties file

```
webrtc_sdp_min_bitrate_bps=3000000
webrtc_sdp_max_bitrate_bps=7000000
```

In this case publishing bitrate will be held between 3000 kbps and 7000 kbps.

Playback codec configuration

By default, for playing WebRTC video stream the same video codec is used with which it is published (VP8), and transcoding on server is not performed.

To use H.264 codec for playback, VP8 can be excluded on browser side using Web SDK stipCodecs option:

```
playStream = session.createStream({
    ...
    stripCodecs: "VP8"
}).on(STREAM_STATUS.PENDING, function (publishStream) {
    ...
});
playStream.play();
```

If H.264 is used for playback, VP8 to H.264 transcoding is performed on server.

WebRTC transport setup

By default, OBS WebRTC Client publish a stream using UDP transport. Enable TCP transport to prevent packet loss using the following WCS settings:

ice_tcp_transport=true

Quick testing

1. Press Start Streaming button in OBS WebRTC Client, allow Windows Firewall bypassing to application if necessary. Green square in bottom right conner shows that stream is published successfully



2. Open Player example page on server, set stream name test_obs in Stream field and press Start. The stream playback starts

	Player
WCS URL	wss://demo.flashphoner.com:844
Stream	test_obs
Volume	
Full Screen	*
	PLAYING Stop