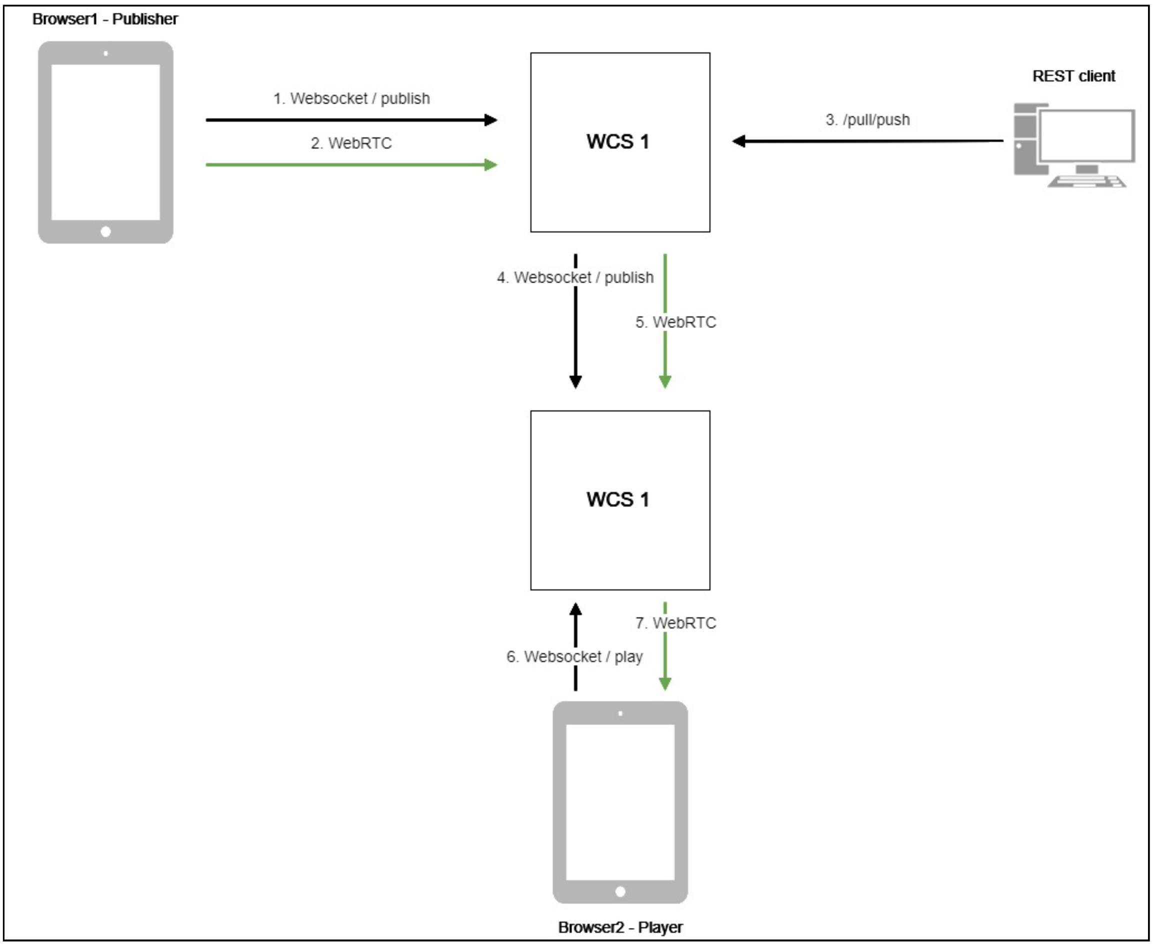
To another WCS server via WebRTC

Overview

WCS can rebroadcast a video stream via WebRTC to another WCS server on demand. Republishing of a WebRTC stream is managed by the REST API.

Operation flowchart



1. The browser connects to the WCS1 server via the Websocket protocol and sends the `publishStream` command.
2. The browser captures the microphone and the camera and send the WebRTC stream to the server.
3. The REST client sends to the WCS1 server the `/pull/push` query.
4. WCS1 publishes the stream to WCS2.
5. WCS2 receives the WebRTC stream from WCS1.

6. The second browser establishes a connection to the WCS2 server via Websocket and sends the `playStream` command.
7. The second browser receives the WebRTC stream and plays this stream on the page.

REST API

A REST query must be an HTTP/HTTPS POST query in the following form:

- HTTP: `http://test.flashphoner.com:8081/rest-api/pull/push`
- HTTPS: `https://test.flashphoner.com:8444/rest-api/pull/push`

Where:

- `streaming.flashphoner.com` - is the address of the WCS server
- `8081` - is the standard REST / HTTP port of the WCS server
- `8444` - is the standard HTTPS port
- `rest-api` - is the required prefix
- `/pull/push` - is the REST-method used

REST methods and responses

REST method	Request body	Response body	Response status	Description

REST method	Request body	Response body	Response status	Description
`/pull/push`	`{ "uri":"wss://demo.flashphoner.com:8443/websocket", "localStreamName": "testStream", "remoteStreamName": "testStream" }`		409 Conflict 500 Internal error	Broadcasts the WebRTC stream to the specified URL

REST method	Request body	Response body	Response status	Description

`/pull/find_all`		[
		{
		"localMediaSessionId":"da157e2b-2159-40c9-9560-ae1af8d4a0b5",
		"remoteMediaSessionId":null,
		"localStreamName":"testStream",
		"remoteStreamName":"testStream",
		"uri":"wss://demo.flashphoner.com:8443/websocket",
		"status":"NEW"
		}
		]
			404 Not found
			500 Internal error
				Find all pulled streams

REST method	Request body	Response body	Response status	Description
`/pull/terminate`	`{ "uri":"wss://demo.flashphoner.com:8443/websocket" }`		404 Not found 500 Internal error	Terminate a pulled stream

Parameters

Parameter	Description	Example
uri	URL of the WebRTC stream	`wss://demo.flashphoner.com:8443/websocket`
localMediaSessionId	Session identifier	`5a072377-73c1-4caf-abd3`
remoteMediaSessionId	Identifier of the session on the remote server	`12345678-abcd-dead-beaf`
localStreamName	Local name assigned to the captured stream. The stream can be fetched from the WCS server using this name	`testStream`
remoteStreamName	Name of the captured stream on the remore server	`testStream`
status	Current status of the stream	`NEW`

Configuration

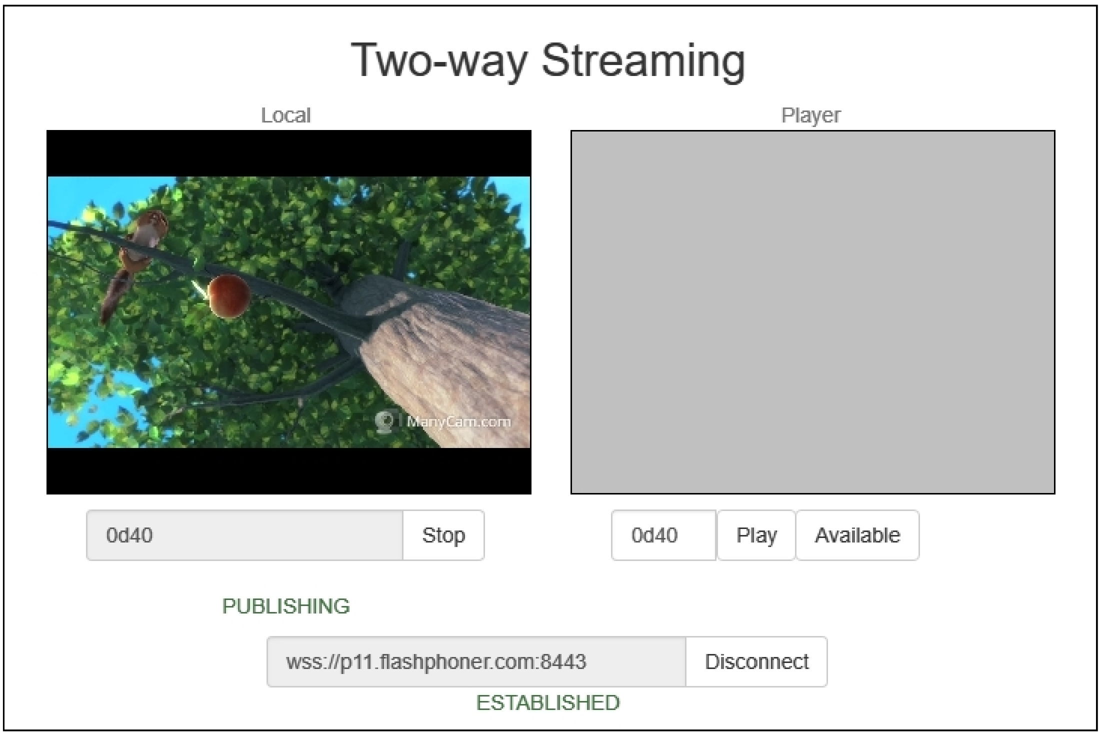
By default, WebRTC stream is pulled over unsecure Websocket connection, i.e. WCS server URL has to be set as `ws://demo.flashphoner.com:8080`. To use Secure Websocket, the parameter must be set in file flashphoner.properties

```
wcs_agent_ssl=true
```

This change has to be made on both WCS servers: the server that publishes the stream and the server the stream is pushed to.

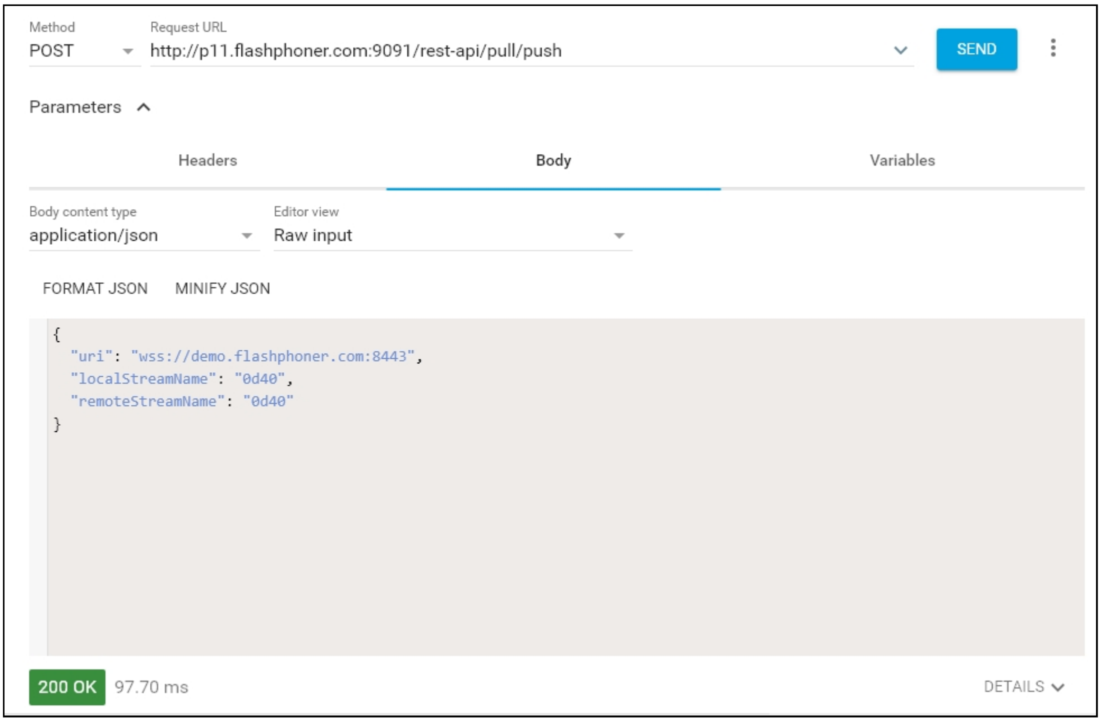
Quick manual on testing

1. For this test we use:
2. two WCS servers;
3. the Chrome browser and the REST client to send queries to the server;
4. the Two Way Streaming web application to publish the stream;
5. the Player web application to play the captured stream in a browser.
6. Open the Two Way Streaming web application, publish the stream on the server

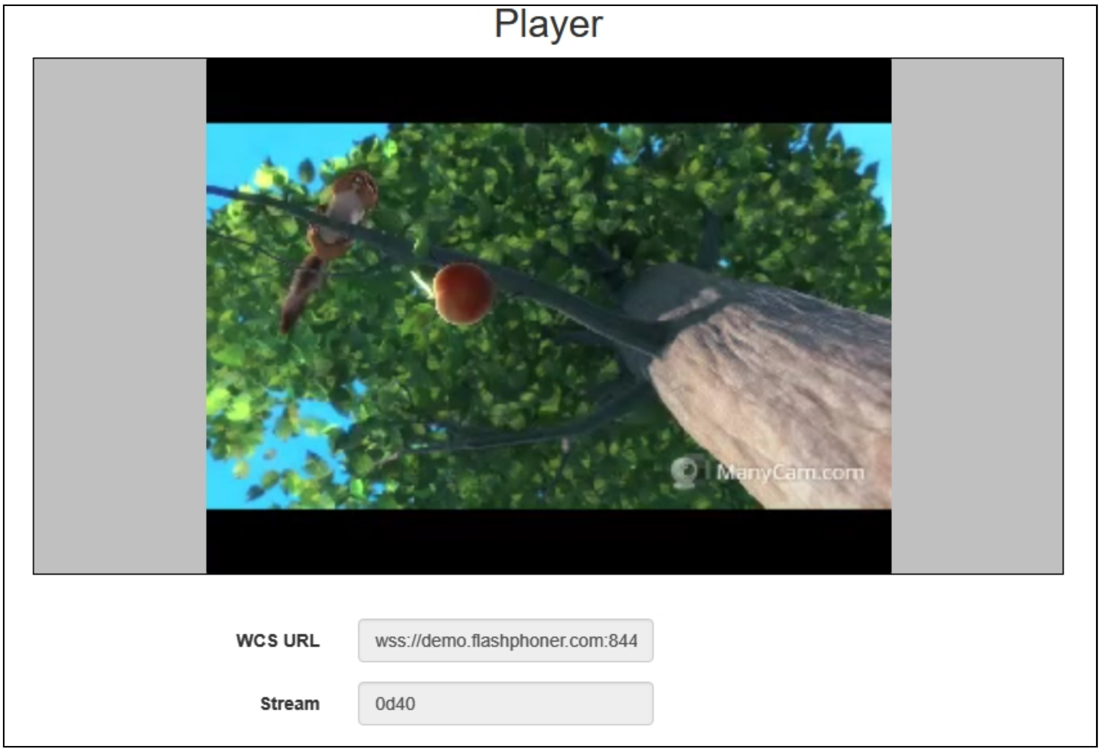


7. Open the REST client. Send the `/pull/push` query and specify in its parameters:
8. the URL of the WCS server the stream is captured from
9. the name of the stream published on the server

10. the local name of the stream



11. Open the Player web application and specify the local stream name in the `Stream` field, then click `Start`



Call flow

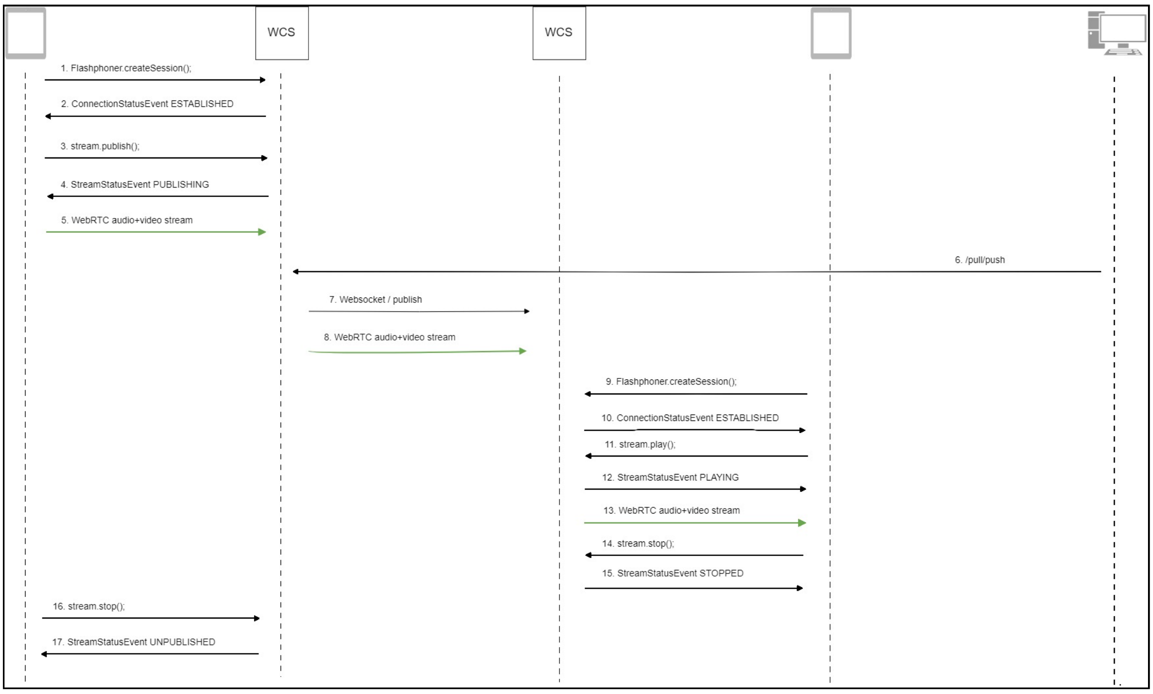
Below is the call flow when using the Two Way Streaming example to publish a stream on one WCS server and the Player example to play that stream on another WCS server.

two_way_streaming.html

two_way_streaming.js

player.html

player.js



1. Establishing a connection to the server
   `Flashphoner.createSession()` code

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED,
function (session) {
  setStatus("#connectStatus", session.status());
  onConnected(session);
}).on(SESSION_STATUS.DISCONNECTED, function () {
  setStatus("#connectStatus", SESSION_STATUS.DISCONNECTED);
  onDisconnected();
}).on(SESSION_STATUS.FAILED, function () {
  setStatus("#connectStatus", SESSION_STATUS.FAILED);
  onDisconnected();
});
```

2. Receiving from the server an event confirming successful connection
   `SESSION_STATUS.ESTABLISHED` code

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED,
function (session) {
```

```
    setStatus("#connectStatus", session.status());
    onConnected(session);
}).on(SESSION_STATUS.DISCONNECTED, function () {
    ...
}).on(SESSION_STATUS.FAILED, function () {
    ...
});
```

3. Publishing the stream
   `Stream.publish()` code

```
session.createStream({
    name: streamName,
    display: localVideo,
    cacheLocalResources: true,
    receiveVideo: false,
    receiveAudio: false
    ...
}).publish();
```

4. Receiving from the server an event confirming successful publishing of the stream
   `STREAM_STATUS.PUBLISHING` code

```
session.createStream({
    name: streamName,
    display: localVideo,
    cacheLocalResources: true,
    receiveVideo: false,
    receiveAudio: false
}).on(STREAM_STATUS.PUBLISHING, function (stream) {
    setStatus("#publishStatus", STREAM_STATUS.PUBLISHING);
    onPublishing(stream);
}).on(STREAM_STATUS.UNPUBLISHED, function () {
    ...
}).on(STREAM_STATUS.FAILED, function () {
    ...
}).publish();
```

5. Sending the audio-video stream via WebRTC to the server
6. Sending the `/pull/push` REST query to the server
7. Publishing the stream on the second server
8. Sending the audio-video stream via WebRTC to the second server
9. Establishing a connection to the second server
   `Flashphoner.createSession()` code

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED,
function(session){
    setStatus(session.status());
    //session connected, start playback
    playStream(session);
}).on(SESSION_STATUS.DISCONNECTED, function(){
```

```
  setStatus(SESSION_STATUS.DISCONNECTED);
  onStopped();
}).on(SESSION_STATUS.FAILED, function(){
  setStatus(SESSION_STATUS.FAILED);
  onStopped();
});
```

10. Receiving from the server an event confirming successful connection
`SESSION_STATUS.ESTABLISHED` code

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED,
function(session){
    setStatus(session.status());
    //session connected, start playback
    playStream(session);
}).on(SESSION_STATUS.DISCONNECTED, function(){
    ...
}).on(SESSION_STATUS.FAILED, function(){
    ...
});
```

11. Requesting to play the stream
`Stream.play()` code

```
stream = session.createStream(options).on(STREAM_STATUS.PENDING,
function(stream) {
    ...
});
stream.play();
```

12. Receiving from the server an event that confirms successful capturing and playing of the stream
`STREAM_STATUS.PLAYING` code

```
stream = session.createStream(options).on(STREAM_STATUS.PENDING,
function(stream) {
    ...
}).on(STREAM_STATUS.PLAYING, function(stream) {
    $("#preloader").show();
    setStatus(stream.status());
    onStarted(stream);
}).on(STREAM_STATUS.STOPPED, function() {
    ...
}).on(STREAM_STATUS.FAILED, function(stream) {
    ...
}).on(STREAM_STATUS.NOT_ENOUGH_BANDWIDTH, function(stream){
    ...
});
stream.play();
```

13. Sending the audio-video stream via WebRTC

14. Stopping the playback of the stream
    `Stream.stop()` code

```
function onStarted(stream) {
    $("#playBtn").text("Stop").off('click').click(function(){
        $(this).prop('disabled', true);
        stream.stop();
    }).prop('disabled', false);
    ...
}
```

15. Receiving from the server an event confirming the playback of the stream is stopped
    `STRTEAM_STATUS.STOPPED` code

```
stream = session.createStream(options).on(STREAM_STATUS.PENDING,
function(stream) {
    ...
}).on(STREAM_STATUS.PLAYING, function(stream) {
    ...
}).on(STREAM_STATUS.STOPPED, function() {
    setStatus(STREAM_STATUS.STOPPED);
    onStopped();
}).on(STREAM_STATUS.FAILED, function(stream) {
    ...
}).on(STREAM_STATUS.NOT_ENOUGH_BANDWIDTH, function(stream){
    ...
});
stream.play();
```

16. Stopping publishing the stream
    `Stream.stop()` code

```
function onPublishing(stream) {
    $("#publishBtn").text("Stop").off('click').click(function () {
        $(this).prop('disabled', true);
        stream.stop();
    }).prop('disabled', false);
    $("#publishInfo").text("");
}
```

17. Receiving from the server an event that confirms unpublishing of the stream
    `STREAM_STATUS.UNPUBLISHED` code

```
session.createStream({
    name: streamName,
    display: localVideo,
    cacheLocalResources: true,
    receiveVideo: false,
    receiveAudio: false
}).on(STREAM_STATUS.PUBLISHING, function (stream) {
    ...
}).on(STREAM_STATUS.UNPUBLISHED, function () {
    setStatus("#publishStatus", STREAM_STATUS.UNPUBLISHED);
```

```
    onUnpublished();
}).on(STREAM_STATUS.FAILED, function () {
    ...
}).publish();
```