Republishing to other RTMP server

Overview

Web Call Server may convert a WebRTC audio and video stream to RTMP and send it to the specified RTMP server on demand. This way you can run a broadcasting from a web page to Facebook, YouTube Live, Wowza, Azure Media Services and other live video services.

Republishing of an RTMP stream can be made using REST queries or JavaScript API.

Supported platforms and browsers

	Chrome	Firefox	Safari	Edge
Windows	✅	✅	❌	✅
Mac OS	✅	✅	✅	✅
Android	✅	✅	❌	✅
iOS	✅	✅	✅	✅

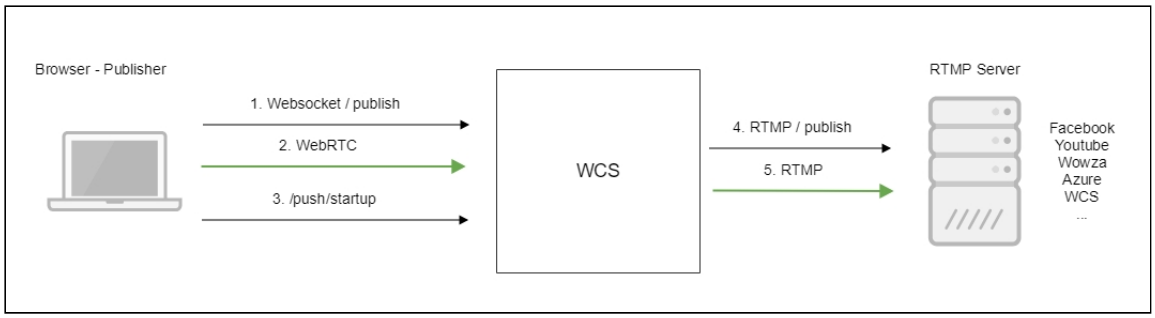
Supported codecs

- Video: H.264
- Audio: AAC, G.711, Speex 16

RTMP server authentication

It is supported. Specify the name and password in the URL of the server, for example `rtmp://name:password@server:1935/live`

Operation flowchart



1. The browser connects to the server via the WebSocket protocol and sends the `publishStream` command.
2. The browser captures the microphone and the camera and sends the WebRTC stream to the server.
3. The REST client sends the `/push/startup` query from the browser.
4. The WCS server publishes the RTMP stream on the RTMP server at the URL specified in the query.
5. The WCS server sends the RTMP stream.

REST API

Republishing a video stream to another server can be performed using REST queries.

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

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

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
- `/push/startup` - 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

`/push/startup`	<pre>{
	
	"streamName": "name",
	
	"rtmpUrl": "rtmp://localhost:1935/live",
	
	"rtmpTransponderFullUrl": false,
	
	"rtmpFlashVersion": "LNX 76.219.189.0",
	
	"options": {}
	}</pre>
		<pre>{
		
		"mediaSessionId": "eume87rjk3df1i9u14elffga6t",
		
		"streamName": "rtmp_name",
		
		"rtmpUrl": "rtmp://localhost:1935/live",
		
		"width": 320,
		
		"height": 240,
		
		"muted": false,
		
		"soundEnabled": false,
		
		"options": {}
		}</pre>
			400 Bad request 409 Conflict 500 Internal error	Create a transponder that subscribes to the given stream and sends media traffic to the specified `rtmpUrl`. The name of the stream specified in the query can be the name of an already published stream or the name reserved when the SIP call was created (to send media traffic received from SIP). If a transponder for the given stream and rtmpUrl already exists, 409 Conflict is returned. If `rtmpUrl` is not set, or is set incorrectly and cannot be resolved by DNS, 400 Bad request is returned.

REST method	Request body	Response body	Response status	Description

`/push/find`	<pre>{
	
	"streamName":
	"name",
	
	"rtmpUrl":
	"rtmp://
	localhost:1935/live",
	}</pre>
		<pre>[
		{
		
		"mediaSessionId":
		"eume87rjk3df1i9u14elffga6t",
		
		"streamName":
		"rtmp_name",
		
		"rtmpUrl":
		"rtmp://
		localhost:1935/live",
		
		"width":
		320,
		
		"height":
		240,
		
		"muted":
		false,
		
		"soundEnabled":
		false,
		
		"options":
		{}
		}
		]</pre>
			404 Not found
			500 Internal error
				Find transponders by a filter

REST method	Request body	Response body	Response status	Description

`/push/find_all`		<pre>[
		    {
		"mediaSessionId": "eume87rjk3df1i9u14elffga6t",
		"streamName": "rtmp_name",
		"rtmpUrl": "rtmp://localhost:1935/live",
		"width": 320,
		"height": 240,
		"muted": false,
		"soundEnabled": false,
		"options": {}
		    }
		]</pre>
			404 Not found
			500 Internal error
				Find all transponders

REST method	Request body	Response body	Response status	Description

`/push/terminate`	{
	"mediaSessionId": "eume87rjk3df1i9u14elffga6t"
	}
			404 Not found
			500 Internal error
				Terminate the transponder

`/push/mute`	{
	"mediaSessionId": "eume87rjk3df1i9u14elffga6t"
	}
			404 Not found
			500 Internal error
				Turn off audio

`/push/unmute`	{
	"mediaSessionId": "eume87rjk3df1i9u14elffga6t"
	}
			404 Not found
			500 Internal error
				Turn on audio

REST method	Request body	Response body	Response status	Description

`/push/sound_on`	<pre>{
	"mediaSessionId": "eume87rjk3df1i9u14elffga6t"
	"soundFile": "test.wav"
	"loop": true
	}</pre>
			404 Not found
			500 Internal error
				Insert audio from a RIFF WAV file located in the `/usr/local/FlashphonerWebCallServer/media/` directory on the WCS server

`/push/sound_off`	<pre>{
	"mediaSessionId": "eume87rjk3df1i9u14elffga6t"
	}</pre>
			404 Not found
			500 Internal error
				Stop inserting audio from the file

Parameters

Parameter	Description	Example
streamName	Name of the republished stream	`streamName`
rtmpUrl	URL of the server the stream is republished to	`rtmp://localhost:1935/live`
rtmpFlashVersion	RTMP subscriber Flash version	`LNX 76.219.189.0`

Parameter	Description	Example
options	Transponder options	`{"action": "mute"}`
mediaSessionId	Unique identifier of the transponder	`eume87rjk3df1i9u14elffga6t`
width	Image width	`320`
height	Image height	`240`
bitrate	Video bitrate, kbps	`500`
keyFrameInterval	Video keyframe interval	`60`
fps	Video framerate	`30`
muted	Is sound muted	`true`
soundEnabled	Is sound enabled	`true`
soundFile	Sound file	`test.wav`
loop	Loop playback	`false`
rtmpTransponderFullUrl	Take stream name to publish to RTMP server from RTMP URL	`false`

The `options` parameter can be used to turn off audio or insert audio from a file when creating a transponder.

Example,

```
"options": {"action": "mute"}
"options": {"action": "sound_on", "soundFile": "sound.wav", "loop": true}
```

Stream transcoding while republishing

Since build 5.2.560, if picture width and height are not set in `/push/startup` query parameters

```
{
    "streamName": "name",
    "rtmpUrl": "rtmp://localhost:1935/live"
}
```

or they are set to 0

```
{
    "streamName": "name",
```

```
    "rtmpUrl": "rtmp://localhost:1935/live",
    "width": 0,
    "height": 0
}
```

transcoding will not be enabled for stream republishing.

If picture height is set explicitly (for example, if destination server does not accept streams below 720p)

```
{
    "streamName": "name",
    "rtmpUrl": "rtmp://localhost:1935/live",
    "width": 1280,
    "height": 720
}
```

the stream will be transcoded and pushed to destination server in defined resolution.

Specified width is applied only if picture aspect ratio preserving is disabled, and height is also specified. If only `width` parameter is passed - without `height` - it is not applied, and the stream is not transcoded.

Since build 5.2.785, there are two more parameters enabling transcoding: `keyFrameInterval` and `fps`. Since build 5.2.1043 `bitrate` parameter is added which also enables stream transcoding while republishing.

Therefore, stream will be transcoded while republishing with any of the following parameters:

```
{
    "streamName": "name",
    "rtmpUrl": "rtmp://localhost:1935/live",
    "height": 240,
    "keyFrameInterval": 60,
    "fps": 30,
    "bitrate": 500
}
```

Set stream name to publish to RTMP server

By default, a stream will be published to RTMP server with the same name as it is publishing on WCS, and the prefix `rtmp_`, for example `rtmp_test`. This behaviour can be changed by the following parameters

```
rtmp_transponder_full_url=true
rtmp_transponder_stream_name_prefix=
```

But, these settings are applyed to all the republishings, and require server restart. That's why since build 5.2.860 the `/push/startup` query parameter is added to allow to define full RTMP

URL, including stream name on RTMP server, regardless of server settings

```
POST /rest-api/push/startup HTTP/1.1
Host: localhost:8081
Content-Type: application/json

{
 "streamName":"stream1",
 "rtmpUrl":"rtmp://rtmp.flashphoner.com:1935/live/test",
 "rtmpTransponderFullUrl":true
}
```

In this case, the stream will be published to RTMP server with the name defined in RTMP URL even with default WCS settings.

JavaScript API

Using Web SDK you can republish a stream to an RTMP server upon creation, similar to the SIP as stream function. Usage example for this method is available in the WebRTC as RTMP web application.

webrtc-as-rtmp-republishing.html

webrtc-as-rtmp-republishing.js

When a stream is created, the method `session.createStream()` receives the parameter `rtmpUrl` that specifies the URL of the RTMP server that accepts the broadcast. The name of the stream is specified in compliance with rules of the RTMP server.

code:

```
function startStreaming(session) {
    var streamName = field("streamName");
    var rtmpUrl = field("rtmpUrl");
    session.createStream({
        name: streamName,
        display: localVideo,
        cacheLocalResources: true,
        receiveVideo: false,
        receiveAudio: false,
        rtmpUrl: rtmpUrl
        ...
    }).publish();
}
```

Republishing of the stream starts directly after it is successfully published on the WCS server.

Server configuration

When WCS creates an RTMP transponder it automatically adds a prefix to the republished stream as set in the flashphoner.properties file:

```
rtmp_transponder_stream_name_prefix=rtmp_
```

If the server the stream is republished to has certain requirements to the name (Facebook, YouTube), this line must be commented out.

The option

```
rtmp_transponder_full_url=true
```

turns on a possibility to pass some request parameters to RTMP server.

A network interface to bind RTMP client for republishing may be set with the following parameter

```
rtmp_publisher_ip=127.0.0.1
```

In this case, RTMP will be republished to localhost only.

Parameters passing in server URL

It is possible to pass some parameters to server. to which a stream should be republished. Parameters to pass are specified in server URL, e.g.

```
rtmp://myrtmpserver.com:1935/app_name/?user=user1&pass=pass1
```

or, if a stream supposed to be published to a specified instance of RTMP server application

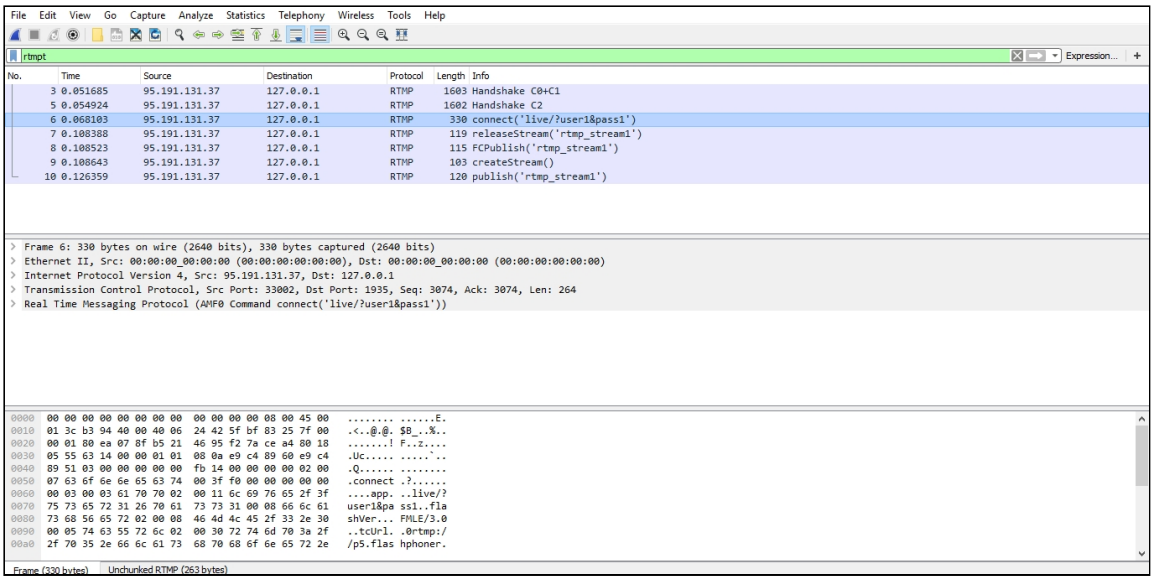
```
rtmp://myrtmpserver.com:1935/app_name/app_instance/?user=user1&pass=pass1
```

Where

- `myrtmpserver.com` is the RTMP server name
- `app_name` is the application on the RTMP server name
- `app_instance` is the instance name of the RTMP server application

Stream name is set in REST query `/push/startup` parameter `streamName` or in corresponding stream creation option.

This is the example on RTMP connection establishing with query parameters passing



Stream name passing in server URL

In some cases, a stream publishing name should be passed in the server URL. To do this, the following option must be set in flashphoner.properties file

```
rtmp_transponder_full_url=true
```

Then, the URL to publish should be set in REST query `/push/startup` parameter `rtmpUrl` or in corresponding stream creation option like this:

```
rtmp://myrtmpserver.com:1935/app_name/stream_name
```

or, to publish to another application instance

```
rtmp://myrtmpserver.com:1935/app_name/app_instance/stream_name
```

In this case, `streamName` parameter or REST query `/push/startup` or corresponding stream creation option is ignored.

Automatic republishing to a specified RTMP server

WCS server can automatically republish all the published streams to a specified RTMP server. To activate this feature, set the following options in flashphoner.properties file:

```
rtmp_push_auto_start=true
rtmp_push_auto_start_url=rtmp://rtmp.server.com:1935/
```

where `rtmp.server.com` is RTMP server name to republish all streams from WCS.

Warning

This feature is supposed to be used for debug only, not in production.

Since build 5.2.1110 it is possible to set authentication parameters

```
rtmp_push_auto_start_url=rtmp://user:password@rtmp.server.com:1935/live
```

or

```
rtmp_push_auto_start_url=rtmp://rtmp.server.com:1935/live?
username=user&password=pwd
```

Parameters will be passed in RTMP connect `command`.

Known limits

Only one RTMP URL can be used for automatic republishing.

Automatic reconnection when channel is closed

When RTMP stream is published to another RTMP server, connection to this server may be interrupted and channel may be closed for some reasons (destination server restart, network problems etc). In this case automatic reconnection and RTMP stream republishing can be enabled with the following parameter in flashphoner.properties file:

```
rtmp_push_restore=true
```

Reconnection attempts maxumum count and interval between attempts in milliseconds should also be set

```
rtmp_push_restore_attempts=3
rtmp_push_restore_interval_ms=5000
```

In this case, 3 attempts will be made to reconnect to RTMP server with 5 seconds interval. After that, reconnection stops.

RTMP outgoing stream buffering

Since build 5.2.700 outgoing RTMP stream can be buffered. This icreases translation latency, but allows to play the stream more smooth from destination RTMP server. Bufferization is enabled with the following parameter

```
rtmp_out_buffer_enabled=true
```

The following bufferization parameters can be tuned

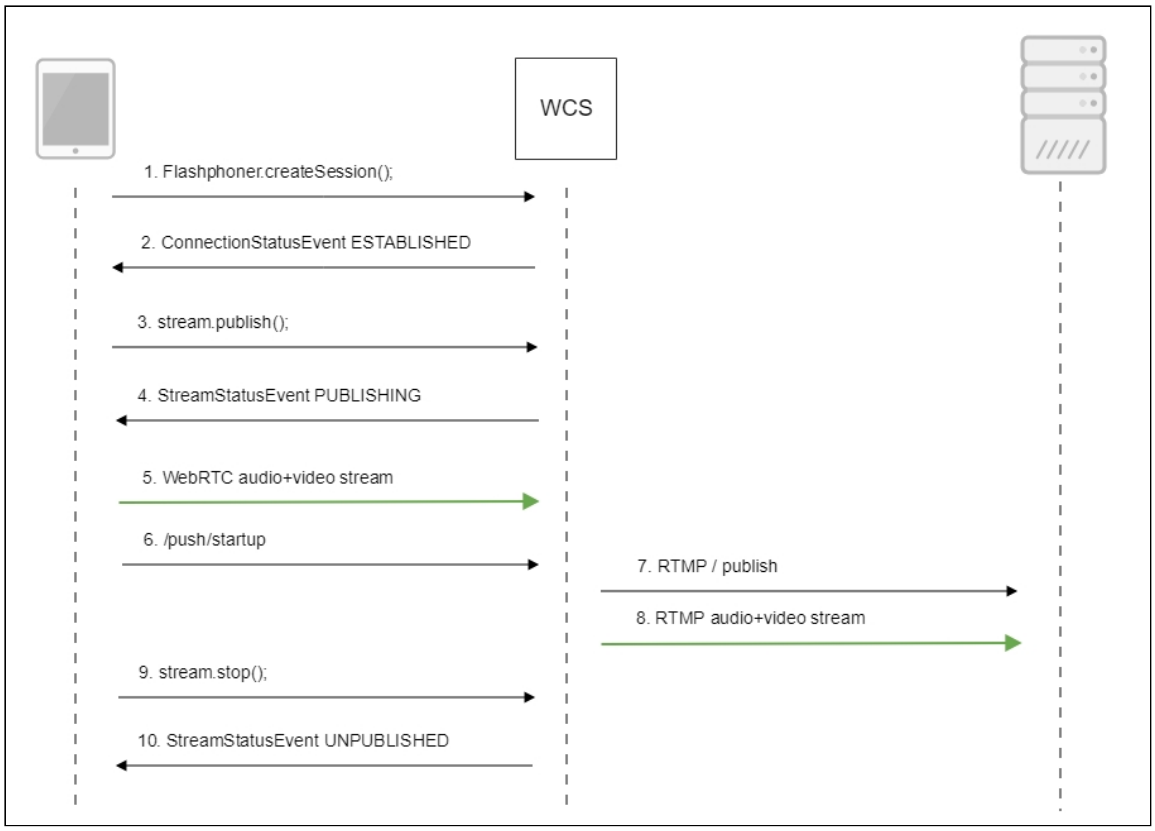
Parameter	Default value	Description
`rtmp_out_buffer_start_size`	300	Stream buffer start size, ms
`rtmp_out_buffer_initial_size`	2000	Stream buffer initial size, ms
`rtmp_out_buffer_polling_time`	50	Buffer polling timeout, ms
`rtmp_out_buffer_max_bufferings_allowed`	-1	Maximum stream bufferings allowed, unlimited by default

Call flow

Below is the call flow when using the Two Way Streaming example to publish a stream and the REST client to send the `/push/startup` query:

two_way_streaming.html

two_way_streaming.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 and 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
6. Sending the `/push/startup` query

```
http://demo.flashphoner.com:8081/rest-api/push/startup
{
   "streamName": "testStream",
   "rtmpUrl": "rtmp://demo.flashphoner.com:1935/live/testStream"
}
```

7. Establishing a connection via RTMP with the specified server, publishing the stream
8. Sending the audio-video stream via RTMP
9. 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("");
}
```

10. Receiving from the server an event confirming 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();
```

Known issues

1. When stream is republished to RTMP server and is played from this server in JWPlayer, stream picture aspect ration can be distorted

Symptoms

Playing stream aspect ratio in JWPlayer differs from published one

Solution

Enable metadata sending while stream republishing as RTMP

```
rtmp_transponder_send_metadata=true
```

2. Republishing may fail if RTMP destination server requires specific Flash version

Symptoms

RTMP handshake fails, the channel is closed with RTMP error in WCS server log

Solution

Specify RTMP subscriber Flash version, either using `rtmp_flash_ver_subscriber` setting in flashphoner.properties, or `rtmpFlashVersion` parameter in republishing REST request

For example, for republishing to Periscope:

```
rtmp_flash_ver_subscriber = LNX 76.219.189.0
```

3. RTMP destination server may require specific stream parameters: bitrate, keyframe interval, or framerate

Symptoms

Server displays warnings about not corresponding to the recommended settings

Solution

Set specific constraints to the source stream (e.g., for audio bitrate) and specify required parameters in republishing REST request (`keyFrameInterval` and `fps`)

4. When republishing streams with big frame size, data packets to send may not fit to socket buffer

Symptoms

Artifacts occur while playing republished RTMP stream via good channel

Solution

Enable RTMP packets buffering with the parameter

```
rtmp.server_buffer_enabled=true
```