
User manual

(RTMP Server)

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Declaration

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Chapter 1 Introduction

Happytime rtmp server is a simple, lightweight, high-performance, and stable stream server.

It can be used to stream local media files, living screen, application windows, camera, microphone, live video/audio content to adobe flash player clients over RTMP protocol.

It developed based on C/C++, the code is stable and reliable, cross-platform porting is simple and convenient, and the code is clear and concise. The server is written to be lightweight and easy to understand, while having good performance, very low latency, video opened immediately.

Happytime rtmp server supports linux, windows, macos, ios, android, embeded linux platforms, supports cross-compiler, can be easily ported to other platforms.

Chapter 2 Key features

RTMP live streaming

Support variety of audio and video files

Support push video from camera and living screen

Support push video from application window

Support push audio from audio device

Support RTMP pusher

Support video codec H264, H265

Support audio codec AAC, G711A, G711U

Support automatic transcoding function

Support RTSP/SRT stream to RTMP stream

Support RTSP/RTMP/SRT relay

Support for configuring audio and video output parameters

Small size, suitable for embedded development

Code structure clear, easy to use

Chapter 3 Configuration

3.1 Configuration Templates

```
<?xml version="1.0" encoding="utf-8"?>
<config>
  <serverip></serverip>
  <serverport>1935</serverport>
  <loop_nums>-1</loop_nums>
  <log_enable>1</log_enable>
  <log_level>1</log_level>

  <application>
    <name>myapp</name>

    <output>
      <url>screenlive</url>
      <video>
        <codec>H264</codec>
        <width></width>
        <height></height>
        <framerate></framerate>
        <bitrate></bitrate>
      </video>
      <audio>
        <codec>AAC</codec>
        <samplerate>8000</samplerate>
        <channels>1</channels>
        <bitrate></bitrate>
      </audio>
    </output>

    <output>
      <url></url>
      <video>
        <codec>H264</codec>
        <width></width>
        <height></height>
        <framerate></framerate>
```

```
        <bitrate></bitrate>
    </video>
    <audio>
        <codec>AAC</codec>
        <samplerate></samplerate>
        <channels></channels>
        <bitrate></bitrate>
    </audio>
</output>

<proxy>
    <suffix>proxy</suffix>
    <url></url>
    <user></user>
    <pass></pass>
    <transfer>TCP</transfer>
    <ondemand>0</ondemand>
    <output>
        <video>
            <codec>H264</codec>
            <width></width>
            <height></height>
            <framerate></framerate>
            <bitrate></bitrate>
        </video>
        <audio>
            <codec>AAC</codec>
            <samplerate></samplerate>
            <channels></channels>
            <bitrate></bitrate>
        </audio>
    </output>
</proxy>

</application>
</config>
```

3.2 Configuring Node Description

3.2.1 System parameters

<serverip>

Specify the IP address of the RTMP server, if not specified, the rtmp server will listen to all network interfaces.

<serverport>

Specify the RTMP server service port, the default is 1935.

<loop_nums>

When streaming media files, specify the number of loop playback,-1 means infinite loop.

<log_enable>

Whether enable the log function,0-disable,1-enable

<log_level>

The log level:

TRACE	0
DEBUG	1
INFO	2
WARN	3
ERROR	4
FATAL	5

3.2.2 application node

<application> : it can configure multiple nodes

<name>: Application Name

The rtmp stream address is rtmp://[serverip]:[serverport]/[application-name]/FILENAME

The [application-name] is name tag value.

3.2.2.1 Output node

<output> : Specify the audio and video output parameters, it can configure multiple nodes

<url>

Match URL address, it can be filename, or file extension name, or special suffix. Such as:

screenlive : match live screen stream

videodevice : match camera video stream

*.mp4 : match all mp4 media file

sample.flv : match sample.flv file

If not config this node, it will match all url as the audio/video default output parameters.

The match order from top to bottom, therefore the default output configuration should be placed in the last.

<video> : Specify the video output parameters

<codec>

Specify the video stream codec, it can specify the following value:

H264 : output H264 video stream

H265: output H265 video stream (Non-RTMP standard, custom extension function)

<width>

Specify the output video width, If 0 use the original video width (live screen stream use the screen width, camera stream use the default width)

<height>

Specify the output video height, If 0 use the original video height (live screen stream use the screen height, camera stream use the default height)

<framerate>

Specify the output video framerate, If 0 use the original video framerate (live screen use the default value 15, camera stream use the default value 25)

<bitrate>

Specify the output video bit rate, if 0, automatically calculate the output bit rate, the unit is kb/s.

Note: This parameter is valid only if encoding is required (eg screenlive, videodevice) or if transcoding is required.

<audio> : Specify the audio output parameters

<codec>

Specify the audio stream codec, it can specify the following value:

G711A: output G711 a-law audio stream

G711U: output G711 mu-law audio stream

AAC: output AAC audio stream

<samplerate>

Specify the audio sample rate, it can specify the following values:

8000, 11025, 12000, 16000, 22050, 24000, 32000, 44100, 48000

If 0 use the original audio sample rate (audio device stream use the default

value 8000)

Note : G711A and G711U only support 8000 samplerate

<channels>

Specify the audio channel number, 1 is mono, 2 is stereo

If 0 use the original audio channel number (audio device stream use the default value 2)

<bitrate>

Specify the output video bit rate, if 0, automatically calculate the output bit rate, the unit is kb/s.

Note: This parameter is valid only if encoding is required (eg screenlive, videodevice) or if transcoding is required.

3.2.2.2 Proxy node

<proxy> : Specify the rtmp proxy parameters, it can configure multiple nodes

<suffix>

Specify the rtmp stream suffix, you can play the proxy stream from:

rtmp://[serverip]:[serverport]/[application-name]/[suffix]

<url>

The original rtsp/rtmp/srt stream address or http mjpeg stream address.

<user> <pass>

Specify the original rtsp/rtmp/srt stream or http mjpeg stream address login user and password information

<transfer>

Specify the rtsp client transfer protocol:

TCP: rtsp client uses RTP over TCP

UDP: rtsp client uses RTP over UDP

MULTICAST: rtsp client uses RTP multicast

<ondemand>

Connect on demand, 1-Connect when needed, 0-Always keep connected

<output>

Specify the stream output parameter. If the parameter does not appear, use the parameters of the original RTSP/RTMP/SRT stream. If it appears and the configured parameters are inconsistent with the parameters of the original RTSP/RTMP/SRT stream, then the transcode output is performed.

The child nodes under this node are consistent with the meaning of the <output> node.

Chapter 4 Run RTMP Server

The rtmp server is a console application.

Windows: to run the server, simply type "rtmpserver".

Linux: to run the rtmp server, type "./start.sh", on linux platform, rtmp server run as daemon by default.

rtmp server supports the following command line options:

`-c config` specify the configuration file

`-c` option specifies the configuration file, if not specified, the default configuration config.xml is used.

`-l [device|videodevice|audiodevice>window]`

`-l device` list available video and audio capture device

`-l videodevice` list available video capture device

`-l audiodevice` list available audio capture device

`-l window` list available application window

Below is sample output of `-l device`:

rtmpserver -l device

Available video capture device :

index : 0, name : FaceTime HD Camera (Built-in)

Available audio capture device :

index : 0, name : Headset Microphone (Apple Audio Device)

index : 1, name : Internal Digital Microphone (Apple Audio Device)

Note : The demo version supports up to 4 concurrent streams.

The release version supports up to 100 concurrent streams.

Chapter 5 Multiple capture devices support

1. If your system have multiple audio capture device, you can use `rtmp://[serverip]:[serverport]/[application-name]/audiodeviceN`, the N to specify the audio capture device index, start from 0, such as:

rtmp://192.168.0.100/myapp/audiodevice ; stream audio from the first audio device
rtmp://192.168.0.100/myapp/audiodevice1 ; stream audio from the second audio device

2. If your system have multiple video capture device, you can use `rtmp://[serverip]:[serverport]/[application-name]/videodeviceN`, the N to specify the video capture device index, start from 0, such as:

rtmp://192.168.0.100/myapp/videodevice ; stream video from the first video device
rtmp://192.168.0.100/myapp/videodevice1 ; stream video from the second video device

3. If your system have multiple monitors, you can use `rtmp://[serverip]:[serverport]/[application-name]/screenliveN`, the N to specify the monitor index, start from 0, such as:

rtmp://192.168.0.100/myapp/screenlive ; stream living screen from the first monitor
rtmp://192.168.0.100/myapp/screenlive1 ; stream living screen the second monitor

The audio index or video index represents which device can execute ***rtmpserver -l device*** to view.

videodevice or audiodevice can also specify the device name, such as `rtmp://[serverip]:[serverport]/[application-name]/videodevice=testvideo`

Execute the ***rtmpserver -l device*** command to get the device name.

Note that there can be no spaces in the device name, if the device name contains spaces, you need to use %20 instead of spaces.

If the device name is “FaceTime HD Camera (Built-in)”, the rtmp stream address is:

rtmp://[serverip]:[serverport]/[application-name]/videodevice=FaceTime%20HD%20Camera%20(Built-in)

Chapter 6 Capture application window

The rtmp server supports capturing application windows, you can use the following command to list valid application windows:

```
rtmpserver -l window
```

Below is a sample output of the command

Available window name :

```
C:\Windows\system32\cmd.exe - RtmpServer.exe -l window
user manual.doc - WPS Office
Rtmp-server Project - Source Insight - [Main.cpp]
RtmpServer
```

You can use the following url to capture the specified application window:

```
rtmp://[serverip]:[serverport]/[application-name]/window=[window
title]
```

Note : window title case insensitive

Such as :

```
rtmp://[serverip]:[serverport]/[application-name]/window=rtmpserv
er
```

Note that there can be no spaces in the window title, if the window title contains spaces, you need to use %20 instead of spaces. Such as:

```
rtmp://[serverip]:[serverport]/[application-name]/window=user%20m
anual.doc%20-%20WPS Office
```