User manual

（RTSP Server）

Happytimesoft Technology Co., LTD

Declaration

Since the product version upgrade or other reasons, this manual will subsequently be updated. Unless otherwise agreed, this manual only as a guide, this manual all statements, information, recommendations do not constitute any express or implied warranties。

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Chapter 1 Introduction

Happytime RTSP Server is a complete RTSP server application. It can stream audio and video files in various formats.

It can also stream video from camera and live screen, stream audio from audio device.

It can stream H265, H264, MP4, MJPEG video stream and G711, G722, G726, AAC, OPUS audio stream.

These streams can be received/played by standards-compliant RTSP/RTP media clients.

It support rtsp proxy function.

It support audio back channel function.

It support rtsp over http function.

It support rtp multicast function.

Support for data pusher function.

Enjoying multimedia content from your computer can be a pleasant way for you to spend your free time. However, sometimes you might need to access it from various locations, such as a different computer or a handheld device, Happytime RTSP Server, that can help you achieve quick and efficient results.

Chapter 2 Key features

The server can transmit multiple streams concurrently

It can stream audio and video files in various formats

It can stream audio from audio device

It can stream video from camera and live screen

It can stream H265, H264, MP4, MJPEG video stream

It can stream G711, G722, G726, AAC, OPUS audio stream

It supports rtsp over http function

It supports rtsp over websocket function

It supports rtp multicast function

It supports data pusher function

It supports RTSP proxy function, as the following:

Support Audio Backchannel

Happytime rtsp server comply with ONVIF backchannel specification, the url is :

https://www.onvif.org/specs/stream/ONVIF-Streaming-Spec-v1706.pdf

5.3.2.1 Example 1: Server without backchannel support:

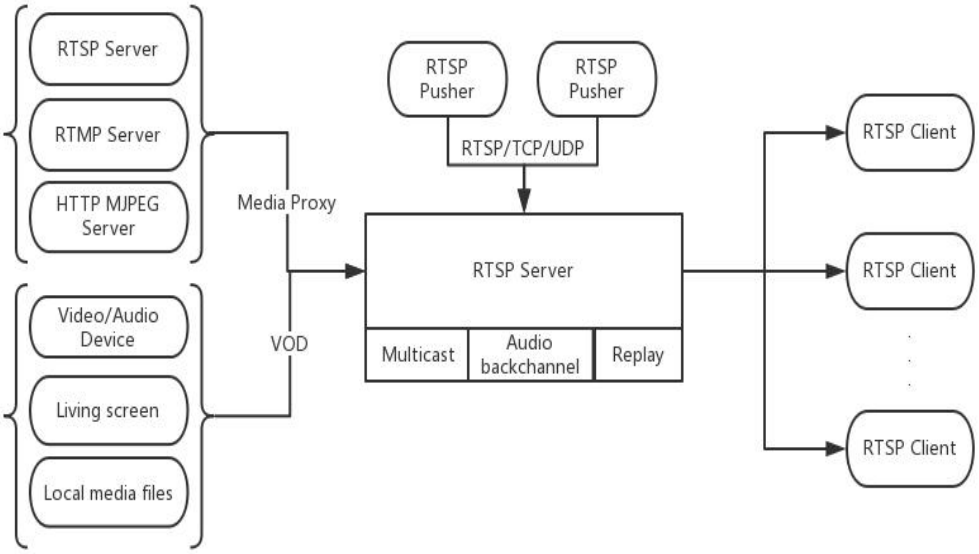
```
Client - Server:          DESCRIBE rtsp://192.168.0.1 RTSP/1.0
                          Cseq: 1
                          User-Agent: ONVIF Rtsp client
                          Accept: application/sdp
                          Require: www.onvif.org/ver20/backchannel

Server - Client:          RTSP/1.0 551 Option not supported
                          Cseq: 1
                          Unsupported: www.onvif.org/ver20/backchannel
```

5.3.2.2 Example 2: Server with Onvif backchannel support:

```
Client - Server:            DESCRIBE rtsp://192.168.0.1 RTSP/1.0
                            Cseq: 1
                            User-Agent: ONVIF Rtsp client
                            Accept: application/sdp
                            Require: www.onvif.org/ver20/backchannel

Server - Client:            RTSP/1.0 200 OK
                            Cseq: 1
                            Content-Type: application/sdp
                            Content-Length: xxx

                            v=0
                            o= 2890842807 IN IP4 192.168.0.1
                            s=RTSP Session with audiobackchannel
                            m=video 0 RTP/AVP 26
                            a=control:rtsp://192.168.0.1/video
                            a=recvonly
                            m=audio 0 RTP/AVP 0
                            a=control:rtsp://192.168.0.1/audio
                            a=recvonly
                            m=audio 0 RTP/AVP 0
                            a=control:rtsp://192.168.0.1/audioback
                            a=rtpmap:0 PCMU/8000
                            a=sendonly

Client - Server:            SETUP rtsp://192.168.0.1/video RTSP/1.0
                            Cseq: 2
                            Transport: RTP/AVP;unicast;client_port=4588-4589

Server - Client:    RTSP/1.0 200 OK
                            Cseq: 2
                            Session: 123124;timeout=60
                            Transport:RTP/AVP;unicast;client_port=4588-4589;
                            server_port=6256-6257

Client - Server:            SETUP rtsp://192.168.0.1/audio RTSP/1.0
                            Cseq: 3
                            Session: 123124
                            Transport: RTP/AVP;unicast;client_port=4578-4579

Server - Client:            RTSP/1.0 200 OK
                            Cseq: 3
                            Session: 123124;timeout=60
                            Transport:RTP/AVP;unicast;client_port=4578-4579;
                            server_port=6276-6277

Client - Server:            SETUP rtsp://192.168.0.1/audioback RTSP/1.0
                            Cseq: 4
                            Session: 123124
                            Transport: RTP/AVP;unicast;client_port=6296-6297
                            Require: www.onvif.org/ver20/backchannel

Server - Client:            RTSP/1.0 200 OK
                            Cseq: 4
                            Session: 123124;timeout=60
                            Transport:RTP/AVP;unicast;client_port=6296-6297;
                            server_port=2346-2347
```

```
Client - Server:        PLAY rtsp://192.168.0.1 RTSP/1.0
                        Cseq: 5
                        Session: 123124
                        Require: www.onvif.org/ver20/backchannel

Server - Client:        RTSP/1.0 200 OK
                        Cseq: 5
                        Session: 123124;timeout=60
```

Chapter 3 Function chart



Chapter 4 Configuration

4.1 Configuration Templates

```
<?xml version="1.0" encoding="utf-8"?>
<config>
    <serverip></serverip>
    <serverip></serverip>
    <serverport>554</serverport>
    <loop_nums>1</loop_nums>
    <multicast>0</multicast>
    <metadata>1</metadata>
    <rtsp_over_http>1</rtsp_over_http>
    <http_port>80</http_port>
    <need_auth>0</need_auth>
    <log_enable>1</log_enable>
    <log_level>1</log_level>

    <user>
        <username>admin</username>
        <password>123456</password>
    </user>
    <user>
        <username>user</username>
        <password>123456</password>
    </user>

    <output>
        <url>screenlive</url>
        <video>
            <codec>H264</codec>
            <width></width>
            <height></height>
            <framerate></framerate>
            <bitrate></bitrate>
        </video>
        <audio>
            <codec>G711U</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>G711U</codec>
            <samplerate></samplerate>
            <channels></channels>
            <bitrate></bitrate>
        </audio>
    </output>

    <proxy>
        <suffix>proxy</suffix>
        <url></url>
        <user></user>
        <pass></pass>
        <transfer>TCP</transfer>
        <output>
            <video>
                <codec>H264</codec>
                <width>640</width>
                <height>480</height>
                <framerate>25</framerate>
                <bitrate></bitrate>
            </video>
            <audio>
                <codec>AAC</codec>
```

```
                <samplerate>32000</samplerate>
                <channels>2</channels>
                <bitrate></bitrate>
            </audio>
        </output>
    </proxy>

    <pusher>
        <suffix>pusher</suffix>
        <video>
            <codec>H264</codec>
        </video>
        <audio>
            <codec>G711U</codec>
            <samplerate>8000</samplerate>
            <channels>1</channels>
        </audio>
        <transfer>
            <mode>UDP</mode>
            <ip></ip>
            <vport>50001</vport>
            <aport>50002</aport>
        </transfer>
        <output>
            <video>
                <codec>H264</codec>
                <width>640</width>
                <height>480</height>
                <framerate>25</framerate>
                <bitrate></bitrate>
            </video>
            <audio>
                <codec>AAC</codec>
                <samplerate>32000</samplerate>
                <channels>2</channels>
                <bitrate></bitrate>
            </audio>
        </output>
```

```
    </pusher>

    <backchannel>
        <codec>G711U</codec>
        <samplerate>8000</samplerate>
        <channels>1</channels>
    </backchannel>
</config>
```

4.2 Configuring Node Description

4.2.1 System parameters

<serverip>

Specify the IP address RTSP server bindings, if not specified, the RTSP server will bind to the default routing interface IP address.

Note: This node can configure multiple instances,meaning that the server can bind multiple IP addresses or domain names.

<serverport>

Specify the port RTSP server binding, the default is 554.

<loop_nums>

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

<multicast>

Whether to enable rtp multicast function, 0-disable, 1-enable.

<metadata>

Whether to enable the meta data stream, 0-disable, 1-enable.

<rtsp_over_http>

Whether to enable rtsp over http function, 0-disable,1-enable.

<http_port>

Specify the HTTP service port for rtsp over http function.

<need_auth>

Whether enable the user authentication function,0-disable,1-enable

<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

4.2.2 *User* node

<user> : Specify the login username password, it can configure multiple nodes

<username>

The login username

<password>

The login password

4.2.3 *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. 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

MP4: output MP4 video stream

JPEG: output MJPEG video stream

<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

G722: output G726 audio stream

G726: output G726 audio stream

AAC: output AAC audio stream

OPUS: output OPUS 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)

<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)

Note : G726 only support mono.

<bitrate>

Specify the output audio 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 (such as screenlive, videodevice) or if transcoding is required.

4.2.4 *Proxy* node

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

<suffix>

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

rtsp://youip/suffix

<url>

The original rtsp/rtmp/http mjpeg stream address.

<user> <pass>

Specify the original rtsp/rtmp/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 multicast

<output>

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

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

4.2.5 *Pusher* node

<pusher> : Specify the data pusher parameters, it can configure multiple nodes

<suffix>

Specify the rtsp stream suffix, you can play the pusher stream from:

rtsp://youip/suffix

<video> : Specify the the input video data parameters

<codec>

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

H264 : H264 video stream

H265 : H265 video stream

MP4: MP4 video stream

JPEG: MJPEG video stream

<audio> : Specify the input audio data parameters

<codec>

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

G711A: G711 a-law audio stream

G711U: G711 mu-law audio stream

G722: G726 audio stream

G726: G726 audio stream

OPUS: OPUS audio stream

AAC: AAC audio stream

<samplerate>

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

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

<channels>

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

Note : G726 only support mono.

<transfer>： Specify the data transfer parameters

<mode>：Specify the data transer protocol, it can specify the following value:

TCP: use TCP connection to transfer the data

UDP: use UDP connection to transfer the data

RTSP: use RTSP connection to transfer the data, it support FFMPEG rtsp pusher.

<ip>: Specified data receiving IP address, if there is no configuration, the default IP address is used (valid in TCP or UDP mode).

<vport>: Specify the video data receiving port (valid in TCP or UDP mode)

<aport>: Specify the audio data receiving port (valid in TCP or UDP mode)

<output>

Specify the stream output parameter. If the parameter does not appear, use the parameters of the original pusher stream. If it appears and the configured parameters are inconsistent with the parameters of the original pusher stream, then the transcode output

is performed.

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

4.2.6 *Backchannel* node

<backchannel> : specify the audio back channel parameters

<codec>

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

G711A: G711 a-law audio stream

G711U: G711 mu-law audio stream

G722: G726 audio stream

G726: G726 audio stream

OPUS: OPUS audio stream

<samplerate>

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

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

If 0 use the default value 8000

<channels>

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

If 0 use the default value 1

Note : G726 only support mono.

Chapter 5 Data pusher

Data pusher means that RTSP server receives external data sources and then sends them out as RTSP streams.

The data pusher support TCP, UDP and RTSP mode.

Audio and video data are packaged and sent in RTP format.

If it is TCP mode, you need to add 4 bytes in front of the RTP header, as the following:

```
typedef struct
{
    uint32    magic    : 8;
    uint32    channel  : 8;
    uint32    rtp_len  : 16;
} RILF;
```

magic: 0x24

channel: 0

ttp_len: the RTP load length, including RTP header,

You can download the examples of sending H264 data from the following link:

http://happytimesoft.com/downloads/happytime-rtsp-h264-data-pusher-example.zip

Note: If you use TCP or UDP mode data push, you need to add <pusher>tag in the rtsp server configuration file, specify the push audio and video parameters and push port, etc.

If you use RTSP mode to push data, no configuration is required. The url suffix of the pushed RTSP address can be any legal string.

If it is RTSP mode, it supports standard RTSP push stream, such as FFMPEG rtsp pusher.

FFMPEG rtsp over UDP:

ffmpeg -re -i test.mp4 -vcodec libx264 -acodec copy -preset ultrafast -f rtsp rtsp://yourip/pusher

FFMPEG rtsp over TCP:

ffmpeg -re -i test.mp4 -vcodec libx264 -acodec copy -preset ultrafast -f rtsp -rtsp_transport tcp rtsp://yourip/pusher

Examples of protocols pushed by RTSP are as follows:

C->S:

OPTIONS rtsp://192.168.3.27/mypusher RTSP/1.0

CSeq: 1

User-Agent: happytimesoft rtsp client

S->C:

RTSP/1.0 200 OK

Server: happytime rtsp server V5.0

CSeq: 1

Date: Tue, Sep 15 2020 00:45:17 GMT

Public: DESCRIBE, SETUP, PLAY, PAUSE, OPTIONS, TEARDOWN, GET_PARAMETER, SET_PARAMETER, ANNOUNCE, RECORD

C->S:

ANNOUNCE rtsp://192.168.3.27/mypusher RTSP/1.0

CSeq: 2

User-Agent: happytimesoft rtsp client

Content-type: application/sdp

Content-Length: 476

v=0

o=- 0 0 IN IP4 192.168.3.27

s=session

c=IN IP4 192.168.3.27

t=0 0

a=control:*

m=video 0 RTP/AVP 96

a=rtpmap:96 H264/90000

a=fmtp:96 packetization-mode=1;profile-level-id=000015;sprop-parameter-sets=Z2QAFaw07AoDmwEQAAADABAAAAMDCPFi04A=,aO+8sA==

a=control:realvideo

m=audio 0 RTP/AVP 97

a=rtpmap:97 MPEG4-GENERIC/8000/2

a=fmtp:97 streamtype=5;profile-level-id=1;mode=AAC-hbr;sizelength=13;indexlength=3;indexdeltalength=3;config=159056E500

a=control:realaudio

S->C:

RTSP/1.0 200 OK

Server: happytime rtsp server V5.0

CSeq: 2

Date: Tue, Sep 15 2020 00:45:17 GMT

C->S:

SETUP rtsp://192.168.3.27/mypusher/realvideo RTSP/1.0

CSeq: 3

Transport: RTP/AVP/TCP;unicast;interleaved=0-1

User-Agent: happytimesoft rtsp client

S->C:

RTSP/1.0 200 OK

Server: happytime rtsp server V5.0

CSeq: 3

Date: Tue, Sep 15 2020 00:45:17 GMT

Session: 41

Transport: RTP/AVP/TCP;unicast;interleaved=0-1

C->S:

SETUP rtsp://192.168.3.27/mypusher/realaudio RTSP/1.0

CSeq: 4

Session: 41

Transport: RTP/AVP/TCP;unicast;interleaved=2-3

User-Agent: happytimesoft rtsp client

S->C:

RTSP/1.0 200 OK

Server: happytime rtsp server V5.0

CSeq: 4

Date: Tue, Sep 15 2020 00:45:17 GMT

Session: 41

Transport: RTP/AVP/TCP;unicast;interleaved=2-3

C->S:

RECORD rtsp://192.168.3.27/mypusher RTSP/1.0

CSeq: 5

Session: 41

Range: npt=0.0-

User-Agent: happytimesoft rtsp client

S->C:

RTSP/1.0 200 OK

Server: happytime rtsp server V5.0

CSeq: 5

Date: Tue, Sep 15 2020 00:45:17 GMT

Session: 41

Chapter 6 RTSP over HTTP

The key of RTSP over HTTP is to allow RTSP packets to communicate via HTTP port.

We know that the standard port of RTSP is 554, but due to various security policy configurations such as firewalls, there may be restrictions when the client accesses port 554, which prevents the normal transmission of RTSP packets.

But the HTTP port (port 80) is generally open, so there is the idea of letting RTSP packets pass through port 80, namely RTSP over HTTP

The details of RTSP over HTTP are as follows:

First, the client opens two socket connect to the rtsp server HTTP ports. We call these two sockets "data socket" and "command socket".

Step 1. The client sends an HTTP GET command through the "data socket" to request an RTSP connection.

Step 2. The server responds to the HTTP GET command through the "data socket" and responds with success/failure.

Step 3. The client creates a "command socket" and sends an HTTP POST command through the "command socket" to establish an RTSP session.

At this point, the auxiliary function of HTTP is completed, and the server does not return the client's HTTP POST command. Next is the standard process of RTSP on the HTTP port, but it needs to be completed through two sockets. The "command socket" is only responsible for sending, and the "data socket" is only responsible for receiving.

Step 4. The client sends RTSP commands (BASE64 encoding) through the "command socket".

Step 5. The server responds to the RTSP command (in plain text) through the "data socket".

Step 6. Repeat Step4-Step5 until the client sends the RTSP PLAY command and the server responds to the RTSP PLAY command.Step 6. Repeat

Step4-Step5 until the client sends the RTSP PLAY command and the server responds to the RTSP PLAY command.

Step 7. The server transmits audio and video data to the client through the "data socket"

After the data exchange is complete...

Step 8. The client sends the RTSP TEARDOWN command (BASE64 encoding and) through the "command socket"

Step 9. The server responds to the RTSP TEARDOWN command (in plain text) through the "data socket".

Step 10. Close the two sockets.

Chapter 7 RTSP over Websocket

First establish an HTTP connection, and then upgrade to the websocket protocol, RTSP over websocket protocol upgrade process:

C-->S:
GET /websocket HTTP/1.1
Host: 192.168.3.27
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Key: KSO+hOFs1q5SkEnx8bvp6w==
Origin: http://192.168.3.27
Sec-WebSocket-Protocol: rtsp.onvif.org
Sec-WebSocket-Version: 13

S-->C:
HTTP/1.1 101 Switching Protocols
Upgrade: websocket
Connection: Upgrade
Sec-WebSocket-Accept: G/cEt4HtsYEnP0MnSVkKRk459gM=
Sec-WebSocket-Protocol: rtsp.onvif.org
Sec-WebSocket-Version: 13

After the protocol upgrade is successful, perform normal rtsp protocol exchange, and send and receive data through websocket connection.

Chapter 8 Run RTSP Server

The server is a console application.

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

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

Note : The demo version has the following limitations

Maximum support four concurrent sessions.

Chapter 9 Multiple capture devices support

1. If your system have multiple audio capture device, you can use rtsp://yourip:port/audiodevice*N, the N to specify the audio capture device index, start from 0, such as:*

rtsp://192.168.0.100/audiodevice ; stream audio from the first audio device

rtsp://192.168.0.100/audiodevice1 ; stream audio from the second audio device

2. If your system have multiple video capture device, you can use rtsp://yourip:port/videodevice*N, the N to specify the video capture device index, start from 0, such as:*

rtsp://192.168.0.100/videodevice ; stream video from the first video device

rtsp://192.168.0.100/videodevice1 ; stream video from the second video device

3. If your system have multiple monitors, you can use rtsp://yourip:port/screenlive*N, the N to specify the monitor index, start from 0, such as:*

rtsp://192.168.0.100/screenlive ; stream living screen from the first monitor

rtsp://192.168.0.100/screenlive1 ; stream living screen the second monitor