User manual

（RTSP Server）

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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

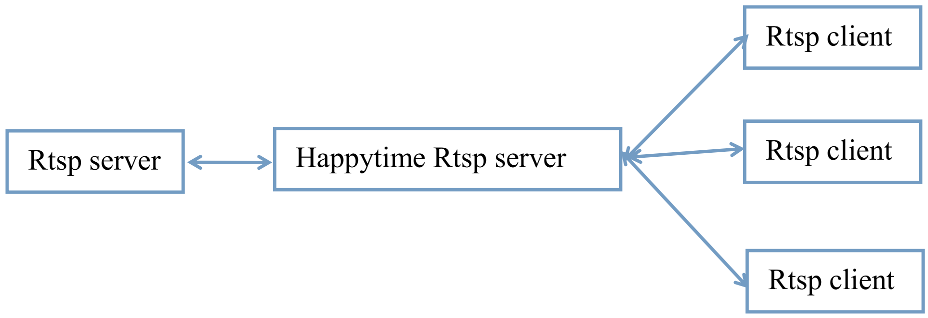
It can stream G711, G722, G726, AAC, OPUS audio stream

It support rtsp over http function

It support rtp multicast function

Support for data pusher function

Support 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 Configuration

3.1 Configuration Templates

```xml
<?xml version="1.0" encoding="utf-8"?>
<config>
    <serverip></serverip>
    <serverip></serverip>
    <serverport>554</serverport>
    <loop_nums>1</loop_nums>
    <multicast>0</multicast>
    <rtsp_over_http>1</rtsp_over_http>
    <http_port>8080</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>
        </video>
        <audio>
            <codec>G711U</codec>
            <samplerate>8000</samplerate>
            <channels>1</channels>
        </audio>
    </output>
    <output>
```

```
        <url></url>
        <video>
            <codec>H264</codec>
            <width></width>
            <height></height>
            <framerate></framerate>
        </video>
        <audio>
            <codec>G711U</codec>
            <samplerate></samplerate>
            <channels></channels>
        </audio>
    </output>
    <proxy>
        <suffix>proxy</suffix>
        <url></url>
        <user></user>
        <pass></pass>
    </proxy>
    <pusher>
        <suffix>pusher</suffix>
        <video>
            <codec>H264</codec>
            <width>1280</width>
            <height>720</height>
            <framerate>25</framerate>
        </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>
```

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

3.2 Configuring Node Description

3.2.1 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.

<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

3.2.2 *User* node

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

<username>

The login username

<password>

The login password

3.2.3 *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)

<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.

3.2.4 *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 stream address

<user> <pass>

Specify the original rtsp stream login user and password information

3.2.5 *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 : output H264 video stream

H265 : output H265 video stream

JPEG: output MJPEG video stream

<audio> : Specify the input audio data parameters

<codec>

Specify the audio 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

OPUS: output OPUS audio stream

<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.

<vport>: Specify the video data receiving port

<aport>: Specify the audio data receiving port

3.2.6 *Backchannel* node

<backchannel> : specify the audio back channel parameters

<codec>

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

G711A: output G711 a-law audio stream

G711U: output G711 mu-law audio stream

G722: output G726 audio stream

G726: output G726 audio stream

OPUS: output 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 4 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

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

Chapter 5 Run RTSP Server

The server is a console application. To run the server, simply type "rtspserver".

Note : The demo version has the following limitations

Maximum support two simultaneous client connections.

Only support one proxy rtsp stream

Only support one data pusher stream

Chapter 6 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