CS5390 – F15

Socket Programming: Video Streaming with RTSP and RTP

In this project you will implement a streaming video server and client that communicate using the Real-Time Streaming Protocol (RTSP) and send data using the Real-time Transfer Protocol (RTP). Your task is to implement the RTSP protocol in the client and implement the RTP packetization in the server.

We will provide you code that implements the RTSP protocol in the server, the RTP de-packetization in the client, and takes care of displaying the transmitted video.

1. Code

Client, ClientLauncher

The ClientLauncher starts the Client and the user interface which you use to send RTSP commands and which is used to display the video. In the Client class, you will need to implement the actions that are taken when the buttons are pressed. You do not need to modify the ClientLauncher module.

ServerWorker, Server

These two modules implement the server which responds to the RTSP requests and streams back the video. The RTSP interaction is already implemented and the ServerWorker calls methods from the RtpPacket class to packetize the video data. You do not need to modify these modules.

RtpPacket

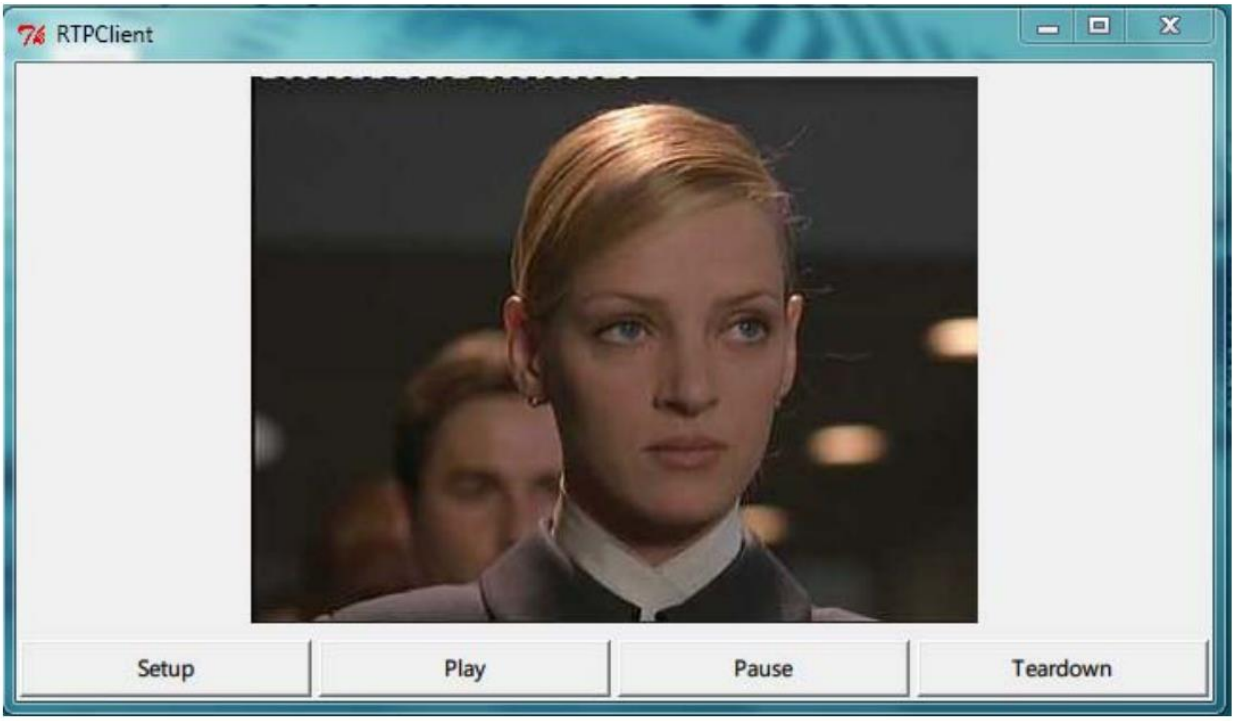
This class is used to handle the RTP packets. It has separate methods for handling the received packets at the client side and you do not need to modify them. The Client also de-packetizes (decodes) the data and you do not need to modify this method. You will need to complete the implementation of video data RTPpacketization (which is used by the server).

VideoStream

This class is used to read video data from the file on disk. You do not need to modify this class.

2. Running the code

After completing the code, you can run it as follows:
First, start the server with the command
python Server.py server_port
where server_port is the port your server listens to for incoming RTSP connections. The standard RTSP port is 554, but you will need to choose a port number greater than 1024.
Then, start the client with the command
python ClientLauncher.py server_host server_port RTP_port video_file
where server_host is the name of the machine where the server is running, server_port is the port where the server is listening on, RTP_port is the port where the RTP packets are received, and video_file is the name of the video file you want to request (we have provided one example file movie.Mjpeg). The file format is described in Appendix section.
The client opens a connection to the server and pops up a window like this:



You can send RTSP commands to the server by pressing the buttons. A normal RTSP interaction goes as follows:

1. The client sends SETUP. This command is used to set up the session and transport parameters.
2. The client sends PLAY. This command starts the playback.
3. The client may send PAUSE if it wants to pause during playback.
4. The client sends TEARDOWN. This command terminates the session and closes the connection.

The server always replies to all the messages that the client sends. The code 200 means that the request was successful while the codes 404 and 500 represent FILE_NOT_FOUND error and connection error respectively. In this lab, you do not need to implement any other reply codes. For more information about RTSP, please see RFC 2326.

3. The Client

Your first task is to implement the RTSP protocol on the client side. To do this, you need to complete the functions that are called when the user clicks on the buttons on the user interface. You will need to implement the actions for the following request types.
When the client starts, it also opens the RTSP socket to the server. Use this socket for sending all RTSP requests.

SETUP
• Send SETUP request to the server. You will need to insert the Transport header in which you specify the port for the RTP data socket you just created.
• Read the server's response and parse the Session header (from the response) to get the RTSP session ID.
• Create a datagram socket for receiving RTP data and set the timeout on the socket to 0.5 seconds.

PLAY
• Send PLAY request. You must insert the Session header and use the session ID returned in the SETUP response. You must not put the Transport header in this request.
• Read the server's response.

PAUSE
• Send PAUSE request. You must insert the Session header and use the session ID returned in the SETUP response. You must not put the Transport header in this request.
• Read the server's response.

TEARDOWN
• Send TEARDOWN request. You must insert the Session header and use the session ID returned in the SETUP response. You must not put the Transport header in this request.
• Read the server's response.

Note: You must insert the CSeq header in every request you send. The value of the CSeq header is a number which starts at 1 and is incremented by one for each request you send.

Example
Here is a sample interaction between the client and server. The client's requests are marked with C: and server's replies with S:. In this lab both the client and the server do not use sophisticated parsing methods, and they expect the header fields to be in the order you see below.

C: SETUP movie.Mjpeg RTSP/1.0
C: CSeq: 1
C: Transport: RTP/UDP; client_port= 25000

S: RTSP/1.0 200 OK
S: CSeq: 1
S: Session: 123456

C: PLAY movie.Mjpeg RTSP/1.0
C: CSeq: 2
C: Session: 123456

S: RTSP/1.0 200 OK
S: CSeq: 2
S: Session: 123456

C: PAUSE movie.Mjpeg RTSP/1.0
C: CSeq: 3
C: Session: 123456

S: RTSP/1.0 200 OK
S: CSeq: 3
S: Session: 123456

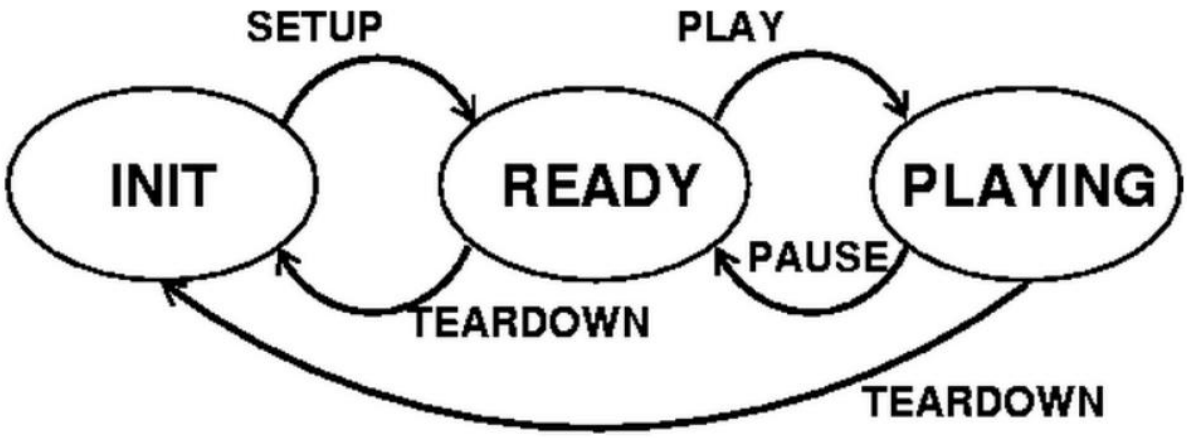
C: PLAY movie.Mjpeg RTSP/1.0
C: CSeq: 4
C: Session: 123456

S: RTSP/1.0 200 OK
S: CSeq: 4
S: Session: 123456

C: TEARDOWN movie.Mjpeg RTSP/1.0
C: CSeq: 5
C: Session: 123456

S: RTSP/1.0 200 OK
S: CSeq: 5
S: Session: 123456

Client State
One of the key differences between HTTP and RTSP is that in RTSP each session has a state. In this lab you will need to keep the client's state up-to-date. Client changes state when it receives a reply from the server according to the following state diagram.



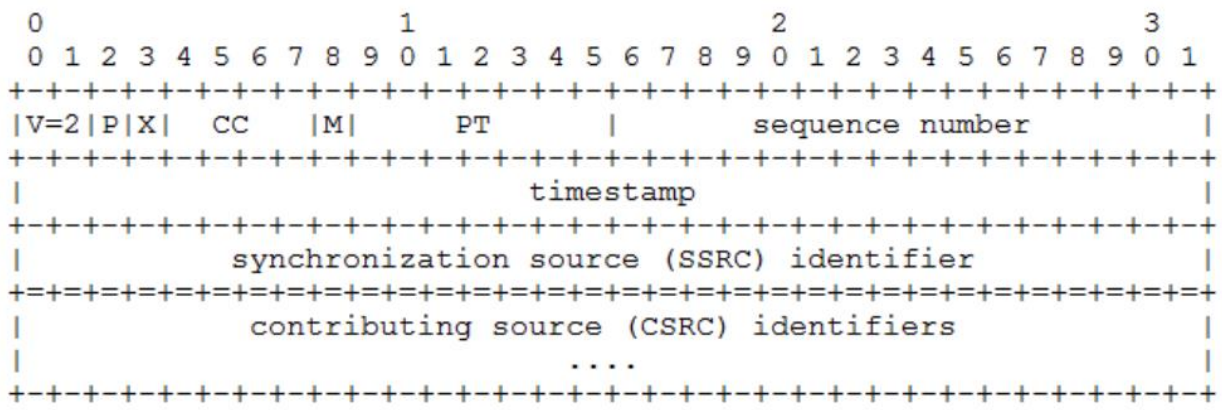
4. The Server

On the server side, you will need to implement the packetization of the video data into RTP packets. You will need to create the packet, set the fields in the packet header and copy the payload (i.e., one video frame) into the packet.

When the server receives the PLAY-request from the client, the server reads one video frame from the file and creates an RtpPacket-object which is the RTP-encapsulation of the video frame. It then sends the frame to the client over UDP every 50 milliseconds.

For the encapsulation, the server calls the encode function of the RtpPacket class. Your task is to write this function. You will need to do the following: (the letters in parenthesis refer to the fields in the RTP packet format below).

- Set the RTP-version field (V). You must set this to 2.
- Set padding (P), extension (X), number of contributing sources (CC), and marker (M) fields. These are all set to zero in this lab.
- Set payload type field (PT). In this lab we use MJPEG and the type for that is 26.
- Set the sequence number. The server gives this the sequence number as the frameNbr argument to the encode function.
- Set the timestamp using the Python's time module.
- Set the source identifier (SSRC). This field identifies the server. You can pick any integer value you like.
- Because we have no other contributing sources (field CC == 0), the CSRC-field does not exist. The length of the packet header is therefore 12 bytes, or the first three lines from the diagram below.



You must fill in the header fields in the header bytearray of the RtpPacket class. You will also need to copy the payload (given as argument data) to the RtpPacket's payload data field.

The above diagram is in the network byte order (also known as big-endian). Python uses the same byte order, so you do not need to transform your packet header into the network byte order.
For more details on RTP, please see RFC 1889.

Twiddling the Bits
Here are some examples on how to set and check individual bits or groups of bits. Note that in the RTP packet header format smaller bit-numbers refer to higher order bits, that is, bit number 0 of a byte is 2^7 and bit number 7 is 1 (or 2^0). In the examples below, the bit numbers refer to the numbers in the above diagram.

Because the header-field of the RtpPacket class is of type bytearray, you will need to set the header one byte at a time, that is, in groups of 8 bits. The first byte has bits 0-7, the second byte has bits 8-15, and so on.

To set bit number n in variable mybyte of type byte:
mybyte = mybyte | 1 << (7 - n)

To set bits n and n + 1 to the value of foo in variable mybyte:
mybyte = mybyte | foo << (7 - n)
Note that foo must have a value that can be expressed with 2 bits, that is, 0, 1, 2, or 3.

To copy a 16-bit integer foo into 2 bytes, b1 and b2:
b1 = (foo >> 8) & 0xFF
b2 = foo & 0xFF

After this, b1 will have the 8 high-order bits of foo and b2 will have the 8 low-order bits of foo.
You can copy a 32-bit integer into 4 bytes in a similar way.

Bit Example
Suppose we want to fill in the first byte of the RTP packet header with the following values:
V = 2
P = 0
X = 0
CC = 3

In binary this would be represented as
1 0 | 0 | 0 | 0 0 1 1
V=2 P X CC = 3
2^7 . . . . . 2^0

5. Required Extensions

In addition to the above, you are required to implement the following.

Statistics Analysis

Calculate statistics about the session. Statistics include RTP packet loss rate, video date rate (in bits of bytes per second), jitter and any other interesting statistics you can think of. To that end, you need to analyze the code and insert the necessary hooks to collect the data. If the client and server run on the same host, you will need to implement a process emulating the network impairments with settable RTP packet loss ratio, throughput and jitter.

Three-Button User Interface

The user interface on the RTPClient has 4 buttons for the 4 actions. If you compare this to a standard media player, such as RealPlayer or Windows Media Player, you can see that they have only 3 buttons for the same actions: PLAY, PAUSE, and STOP (roughly corresponding to TEARDOWN). There is no SETUP button available to the user. Given that SETUP is mandatory in an RTSP-interaction, how would you implement that in a media player? When does the client send the SETUP? Come up with a solution and implement it. Also, is it appropriate to send TEARDOWN when the user clicks on the STOP button?

6. Research Topics

Write a research paper approximately seven to ten pages long about one of these topics. Make sure, you credit any source you use in your document. At least five references required.

In the first phase, you are required to submit for approval a proposed outline listing the specific items in your paper before you work on the detailed paper to be submitted in the final team report. Below are the topics, along with some suggested items for the outline. You may include other items in your proposed outline, but they will have to be approved.

- Content Delivery Networks. Suggested menu of possible items
  - Describe the needs that necessitated having these networks.
  - Describe common topology and schemes of servers.
  - Benefits of using CDNs.
  - Comparison of major companies' solutions for CDN.
- Netflix. Suggested menu of possible items
  - Describe the architecture of Netflix.
  - Explain how Netflix's streaming platform works?
  - Give some insights about Dynamic Adaptive Streaming over HTTP (DASH).
  - CDN selection algorithm by Netflix.
- FaceTime (this is about conversational video, not video streaming). Suggested menu of possible items
  - Architecture
  - Background overview of SIP and RTP
  - How FaceTime uses SIP and RTP and SRTP
  - NAT traversal in FaceTime
  - Impact on traditional cellular networks' business models

7. What to Turn In

1. A proposed outline of the research paper for approval by date D1

2. A team report by date D2

a) Including the complete code for the four-button user interface and the three-button user interface. The complete code includes the pieces you modified or wrote and the pieces you did not have to modify.
b) Providing a background summary of RTP and RTSP and how they are utilized in streaming
c) Describing your hardware setup and configuration
d) Including the design document of your code
e) Providing screenshots showing the user interface, three-button and four-button
f) Providing results of the statistics analysis about the session: RTP packet loss rate, etc.
g) Describing what issues, if any, the team encountered during the project, how the team overcame the issues and what the team learned from the project. You can also provide suggestions on how the projects in Computer Networks could be improved in the future.
h) Including a video clip to demo the code running
i) Including the research paper

3. Individual reports, one for each team member by date D2

a) If you, as an individual team member, have anything specific to add to 1.g) in the team report, please do it in your individual report. Describe what issues, if any, you, as an individual team member, encountered during the project, how you overcame the issues and what you learned from the project (this is not necessarily just about the topic, could be related to teamwork, etc.). You can also provide suggestions on how the projects in Computer Networks could be improved in the future. This complements the team report with any individual viewpoint not included in the team report.
b) Describing what each team member (including yourself) did and contributed to the project

4. Powerpoint slides for the in-class presentation by date D2. The slides is a summary of the key points in the team report. The presentation should also include a demo of your code running. The demo could be a live demo or be based on the video clip. Aim for a 25-30 minute presentation including demo, to leave time for Q&A.

8. Appendix

1. Lab's proprietary MJPEG (Motion JPEG) format

In this lab, the server streams a video which has been encoded into a proprietary MJPEG file format. This format stores the video as concatenated JPEG-encoded images, with each image being preceded by a 5-Byte header which indicates the bit size of the image. The server parses the bitstream of the MJPEG file to extract the JPEG images on the fly. The server sends the images to the client at periodic intervals. The client then displays the individual JPEG images as they arrive from the server.