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DREAM – A Data Streaming Application Using RTP/RTSP in a Local Area Network

Shraddha Zingade[1], Rahul Joshi[2], Rishikesh Shendkar[3], Poorvi Arora[4]
UG SCholar[1, 2, 3, 4]
Department of Computer Engineering
Smt. Kashibai Navale College of Engineering, Pune, India
shraddhazingade29@gmail.com[1,] rahuljoshi6070@gmail.com[2], solidxrishi@gmail.com[3], arorapoorvi10@gmail.com[4]

Abstract:
The growing use of smartphones along with the available connections, say Wi-Fi or cellular networks enables people to visually report live events. The currently available solutions limit the configurability of such services by permitting video streaming through Internet. This project, DREAM, proposes architecture of a real-time video streaming service from any device to other devices connected in a network. The purpose of this project is to implement data streaming using client-server architecture over a local network. DREAM is an audio-visual live data-streaming platform that allows a user to broadcast audio-visual content from a server in a Local Area Network and clients to view the content live on their devices. The project is built upon open source code and open protocols to implement a set of software components that streams live audio-visual data. Additionally, this architecture requires minimal resource, is easily implementable and scalable.

Keywords: Live Streaming, Local Area Network, Real Time Protocol, Real Time Streaming Protocol.

I. INTRODUCTION

The idea of this project, DREAM, is inspired from the notion of making live streaming applications with an advancement of making it workable on a Local Area Network, enabling live streaming more efficiently. The main application focus of DREAM is to implement fast streaming of audiovisual data, thus eliminating the need to download them. The project develops a solution capable of streaming live data in a Local Area Network giving its users, the ability to share live videos and audios in classrooms, conferences and seminars along with the feature of pause and play to the sender.

II. REAL TIME STREAMING PROTOCOL

Real Time Streaming Protocol (RTSP) is an application layer protocol that allows the client to control real time media streaming on the server. RTSP allows the client to send requests to the server that controls the multimedia stream. RTSP does not handle the transmission of the data but only controls the streaming at the client end.

The client can SETUP, PLAY, PAUSE, DESCRIBE, RECORD, TEARDOWN from his user interface.

1. SETUP

This causes the server to allocate resources for a stream and starts an RTSP session. This manipulates state. This means, a SETUP request specifies how a single media stream must be transported. This should be done before a PLAY request is sent. Every media stream must be configured using SETUP before an aggregate play request is to be sent.

2. PLAY

This starts data transmission on a stream allocated by SETUP. In short, this manipulates state. A PLAY request causes one or all media streams to be played. Sending multiple PLAY requests can also stack it. Use can also specify a play range. If no range is specified, the stream is played from the beginning and plays till the end.

3. PAUSE

This temporarily halts transmission of a stream without freeing server resources. All in all, this manipulates the state. If the range parameter is omitted, the pause occurs indefinitely. It is to be noted that this is handled the same way as playing. The two actions might be implemented using a single button in the client's GUI, toggling between the two.

4. DESCRIBE

This causes the server to return a description of the protocol using the SDP (Session Description Protocol). It includes an RTSP URL and the type of reply data that can be handled. It includes the presentation description in SDP format. Furthermore, the presentation description lists media streams controlled with the aggregate URL. This means, there is one media stream each for audio and video.

5. RECORD

This starts a server recording an allocated stream. Record initiates recording a range of media data according to the description. The time stamp reflects start and end time. If no time range is specified, use the start or end time provided in the description of the presentation.

6. TEARDOWN

It frees resources associate with a stream so that the RTSP session ceases to exist. The request TEARDOWN can only be issued at the end of the session. This makes the server release its resources pertaining to the session, and new SETUP requests is sent in order to restart the media streams.

These features are enabled while using RTSP, whereas Real Time Transport Protocol acts as the data transport protocol.

III. REAL TIME TRANSPORT PROTOCOL

Real Time Transport Protocol (RTP) is a network layer protocol that defines a standardized packet format for delivering audiovisual data in the network. RTP is designed for real-time, end-to-end, transfer of data that provides facilities for jitter compensation and detection of out of sequence arrival of data, which is common during transmission on a network. RTP allows data transfer to multiple destinations using multicast. RTP can tolerate some packet loss and yet achieve prompt delivery of data. For example, loss of packet in an audio application may result in loss of a fraction of data, which might go unnoticed. For this purpose, majority of RTP implementation is done using User Datagram Protocol (UDP), since retransmission and acknowledgement are not a necessity.

The RTP components are:

a) **Sequence number**: detects packet loss.

b) **Payload identification:** adapts to a change in bandwidth.

c) **Frame indication:** signifies the beginning and end of frame.

d) **Source identification:** identifies the sender of the frame.

e) **Intra-media synchronization:** uses timestamps to detect delay jitter in a single stream.

IV. LOCAL AREA NETWORK

Wireless access points in the Local Area Network provide access to mobile devices like laptops or smartphones. Wireless Access Point provides an environment to achieve high-speed wireless transmission without using the Internet. It provides the clients a medium to access the server in the network. The access point has a range of 100-150 meters enough to cover a conference or a classroom.

In the proposed system, the packet transfer takes place over a LAN but with the exception of using Internet for communication purpose. All the devices being connected to the same router makes it easier to have a master-slave configuration between two or more devices.

V. PROPOSED SYSTEM

We have proposed a system 'DREAM' to stream multimedia files using RTSP/RTP protocol in a Local Area Network. The figure given below shows the proposed architecture for live streaming of multimedia using client-server architecture and RTSP/RTP protocol.

The multimedia file is sent from the RTSP server to the RTSP client over WiFi network using UDP. The RTSP server sends the file with the appropriate codec (Video format such as .mjpeg) to the RTSP Client. The RTSP Client receives the data and plays it through the in-built video player. The RTSP server has the functionality to pause, play or stop the file streaming at the client end using the corresponding command messages.

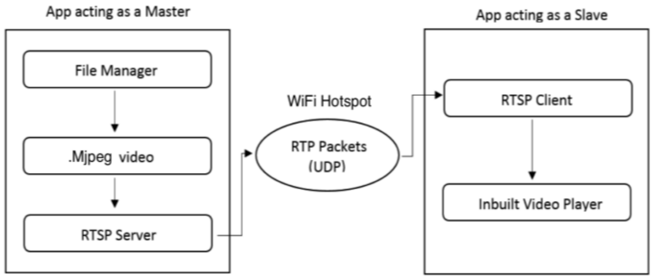

FIGURE 1
ARCHITECTURE OF PROPOSED SYSTEM USING RTSP/RTP []

The network has one server device acting as the master and a number of client devices acting as the slaves.

VI. PROCESS OF COMMUNICATION

RTP carries out the transmission of data, whereas RTSP carries out the control commands of the client. The figure given below describes the integration of RTSP and RTP together to carry out the communication over the LAN in an ideal case. (It only explains the working of two protocols together, not of the proposed system)

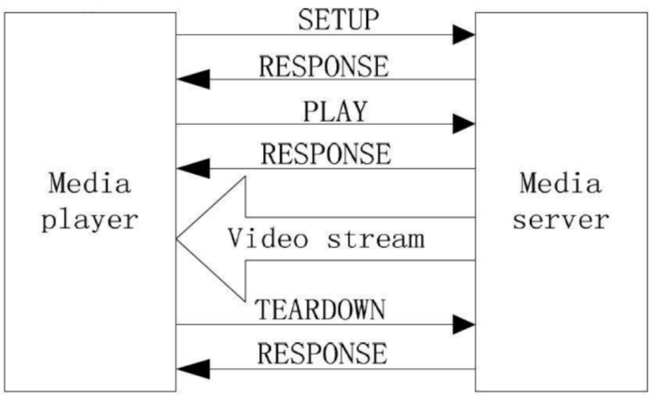

FIGURE 2
BASIC WORKING OF THE RTSP SYSTEM []

1) The RTSP client of the media player sends SETUP message to RTSP server to establish a connection between them.

2) Server allocates the resources required for streaming and starts an RTSP session.

3) Once the connection is established, the RTSP client requests for the multimedia data by sending the PLAY message.

4) The server starts data transmission on the allocated stream.

5) This is achieved by using RTP protocol to send the data packet to the media player.

6) When the media player receives the data, it is played on the client side instantly making way for the next data packet to play.

7) Once the media file is played and the client wants to exit the file, client sends the TEARDOWN message to disconnect with the media server.

8) The server responds by freeing the resources associated with the stream. Thus, the RTSP session ceases to exist on the server.

An additional function commonly used by the client is PAUSE. If the client sends the PAUSE message, the streaming temporarily halts but the resources aren't reallocated. The streaming resumes from the same point once the client sends the PLAY command.

Process of communication in the proposed system:

In the proposed system, the working is similar to the above Figure 2 with a few modifications. The figure given below gives us a detailed explanation of the working of the proposed system.

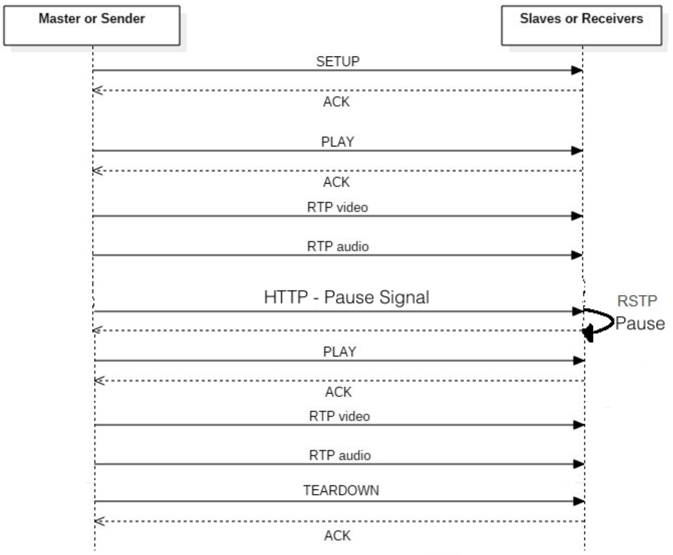

FIGURE 3 WORKING OF THE PROPOSED SYSTEM []

The working is exactly similar from step 1 to step 6 as that of the Figure 2 working. The only modification that takes place is that of the HTTP signal sent by the server to pause the streaming of data.

As mentioned before, the application of this proposed system is in classrooms, conferences, seminars or training centers. This makes it necessary to give the control of PAUSE to the sender i.e the server. To achieve this, we use HTTP signals over a simple socket connection.

1) The audio-visual file starts streaming on the multiple client devices using SETUP and PLAY.

2) Mid-way through the data file, the sender or the server feels it necessary to pause the streaming, thus activating the pause function given on his user interface.

3) The pause function at the RTSP Server end is functioned such that it sends an HTTP signal to the client using basic socket communication,

4) This HTTP signal received at the client end, in-turn activates the PAUSE function of the RTSP client thus successfully achieving the pause functionality during streaming.

5) The PLAY function is used to continue the streaming of the audio-visual file again through RTP Audio and Video.

6) The TEARDOWN command as mentioned previously, is used to disconnect the session between client and server.

VII. CONCLUSION

This project thus implements the live streaming of data using RTP and RTSP protocol in a Local Area Network. It uses .Mjpeg files for streaming to improve network adaptability. The output is a good quality live data streaming on the client's devices with minimal delay.

The server effectively realizes the real time transmission for playing the audio-visual file with RTSP protocol. The play, pause, stop and other functions of streaming media are also available at the client's end. This system is well suited for institutions and organizations to create their own media delivery system with ideal applications being Classrooms, Seminars and Conferences.

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