International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395-0056
Volume: 04 Issue: 08 | Aug -2017 www.irjet.net p-ISSN: 2395-0072

Effective and Secure Scheme for Video Multicasting Using Real Time Transport Protocol (RTP)

Jitu G. Patil[1], Prof. Pyush singh[2]

[1]M-Tech Student, [2]Assistant Professor, Department of Computer Science Engineering
RKDF Institute of Science and Technology, Hoshangabad Road, Bhopal, Madhya Pradesh.

---***---

Abstract - This paper describes our IntraLAN video multicasting system that is specially designed for real time streaming of video over Local Area Networ (LAN). System allows users to make Video Transmission directly from their computer. The use of this software will greatly reduce the manual involvement and try to simplify the means of communication. Unlike other API's this software will develop the RTP Client that can be used in any organization and can be configured in the intranet of that organization. This software can extend its functionality to Distance Education, E-learning, Video Conferencing Meetings over the internet. From the user's point of view, the Video Multicasting can be used in any organization for communication and can be configured in the intranet of that organization.

Key Words: Video streaming, Advanced Encryption Standard (AES), Real Time-Transport Protocol (RTP), User Datagram Protocol (UDP), Secure Real Time-Transport Protocol (RTP).

1. INTRODUCTION

Today, the vigorous generation and transmission of real-time video is still an exciting problem. In our research, real-time video multicasting system is proposed because of our interest in unique immersive forecast and acquisition situations for telepresence. In particular, our video multicasting system is mixture of camera and nodes, which acquire images and perform image processing. The resulting information is streamed on a reconstruction node, which is computes the actual representation of the observed object is done. Camera and reconstruction nodes are at the same physical location and they all are connected in a local area network. The video data is then streamed to a rendering node in a real-world telepresence application, which runs at a remote location.

The APIs implemented through a standalone user interface. The RTP Client developed can be used in any organization for communication and can be configured in the intranet of that organization. A server needs to be fixed and the members of the organization can communicate with one another by simply registering and creating an account. Real Time Transmission (RTP) is a general term for a family of transmission technologies for delivery of Video communications over networks. As explained, rendering and reconstruction nodes need to share a common data structure, and, depending on the video streaming procedure, this data structure must satisfy different consistency requirements. The main contribution of this paer consists in proposing a communication framework for distributed real-time video reconstruction and rendering and in analyzing the transmission of the subsequent streams with respect to changing networking conditions. This software could provide some essential features that rarely present in any API's such as Video Conferencing, Audio transmission etc.

2. SCOPE AND MOTIVATION

To maintain an unwanted flow of data also Quality of service (QOS), it is to avoid packet delay in the broadcast data stream. Applying QOS and giving Multicast data packets priority over other packets. The Broadcast stream usually have some buffering built in so that losses smoothed out than the unicast communication. The data is transfer from one point-to-point communication or single directions. A service where data is delivery from a sender to some of all receiver groups are called broadcast communication. Main aim of this project is it eliminates redundant bits and gives the High quality of results, the more important gives the long distances communication by using LAN cable.

3. SYSTEM REQUIREMENT AND SPECIFICATION

This project or software is a RTP has Client/Server architecture based computer application that allows users to make Video conferencing directly from their computer. With this application, we can be made from the PC to PC, from networks to networks or between soft phone-enabled computers. The software for this application usually mimics the appearance of a real time conferencing and can take the form of either a standalone program with its own window or an embedded program in a Web application or other PC program. Conversations are conducted on a headset with a built-in microphone. The computer's sound card is used to provide audio input and output for the application. The main RTP entities are:

3.1 User Agents: The User Agent Client is the RTP Client that initiates RTP request to a server or through an intermediary such a Proxy, Redirect, and Location Sever. The RTP server that handles RTP UAC request and sends back corresponding response.

3.2 Software Requirements –

- Operating System: - Window XP/7/8.
- Platform :- JDK 1.6 ,
- JMF (Java Media Framework).
- Development Tool :- JCreator or MyEcllips
- Database: - Ms Access or MySQL.
- Programming Language: - Advanced Java.

- Designing software: Rational rose 98.

3.3 Hardware Requirement - Intel Pentium 4 or higher (Processor), 512 MB RAM for Windows XP/ 7 & 1GB RAM for Windows 8. Minimum 20 GB HDD. Web Camera. Speakers/Headset and Microphone LAN Card Switch for LAN Connectivity or/and Internet Any compatible Keyboard Any compatible Mouse and Keyboard.

3.4 Real Time Transmission Protocol:

RTP is described in RFC 1889 and RFC 2250. This transport protocol was developed specifically for streaming data across IP networks. RTP is the most important streaming standard. All media streams, regardless of their format and content, are encapsulated in RTP packets. RTP provides several data fields that are not present in TCP, in particular a Timestamp and a Sequence Number. RTP runs on UDP and uses its multiplexing and checksum functionalities. The Real-Time Transport Protocol (RTP) is an Internet protocol standard that specifies a way for programs to manage the real-time transmission of multimedia data over unicast or multicast network services. The Real-time Transport Protocol (RTP) is a network protocol for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications, television services and web-based push-to-talk features. It allows control of the media server so that the video stream is served at the correct speed. The media player is then able to reassemble the received RTP packets into the correct order and play them out at an appropriate speed. To transport real-time data on the network, a used protocol must fulfill the following constraints as fallows.

a) If the network is congested or a client is flooded by the arrivals of data the server must be able to reduce the multicast rate while respecting some temporal constraints.
b) Must allow the multiplexing and the multiplexing of different types of information while synchronizing them.
c) The goal of this protocol is to provide end-to-end real time data delivery, such as audio and video streams (interactive or not).
d) Resources are reserved, that allow the requested QOS to be delivered.
e) Clients may adapt to varying levels of bandwidth.
Compression must allow storage of many very heavy video streams from different sources.
f) Quality of compressed objects must remain acceptable to allow high resolutions.
g) Interleaving of video frames, particularly the one used by the televised frames must be managed.

4. SYSTEM IMPLIMENTATION AND TESTING:

In multicast data, communication the router system are used the IGMP. Broadcast is delivery of High quality video, Bandwidth Efficiency more than unicast communication, during the transmission of the data from server to client systems. Redundant bit are introduced leading to buffering in order to overcome this streaming is adopted, the network is performed the live programs like cricket, audio and video conferences and live radio programs etc.

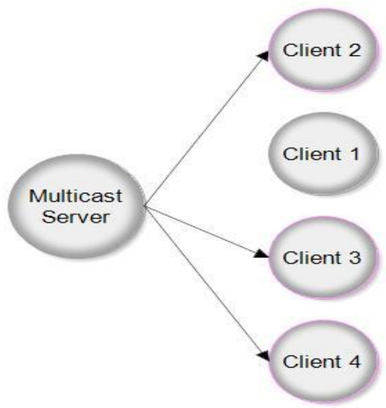

Figure 1: Multicast Communication

IP Broadcast point to group of all receiver points. IP Broadcast over ATM point to multipoint virtual circuits (VCs) as feature of dynamically. It creates the ATM point to multipoint is a switched virtual circuits gives the IP broadcast traffic more efficient. Components required are in hardware are Computer network, switches, Ethernet cable, optical cable, and software are Linux based on Ubuntu OS (version 14.04), Wireshark analyzer to analyses the real time results. VLC Player to streaming the video and audio of Multimedia files. Because of live program connection, oriented network used (RTP) Real time transport protocol like audio and video of live conferences. RTP is Monitor Transmission statistics and quality of service (QOS) and synchronization of multiple streams.

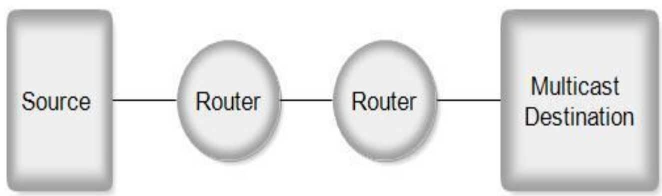

Figure 2: Block diagram of IP Broadcast group.

The above figure two shows Internet protocol Broadcast group of multimedia traffic. It consist of four network systems, first system is a server and two systems are act as a routers the last system has a client's system or broadcast group. Router is used to forward the packets from the server (sources) to client (destination) network systems. For long distance Communication, we have used LAN cable of 1Km for broadcast communication network. Multimedia is easy to watch and listen in the form of Text, audio, video. Activity flow diagram is shown in figure 3 and use case diagram in figure 4.

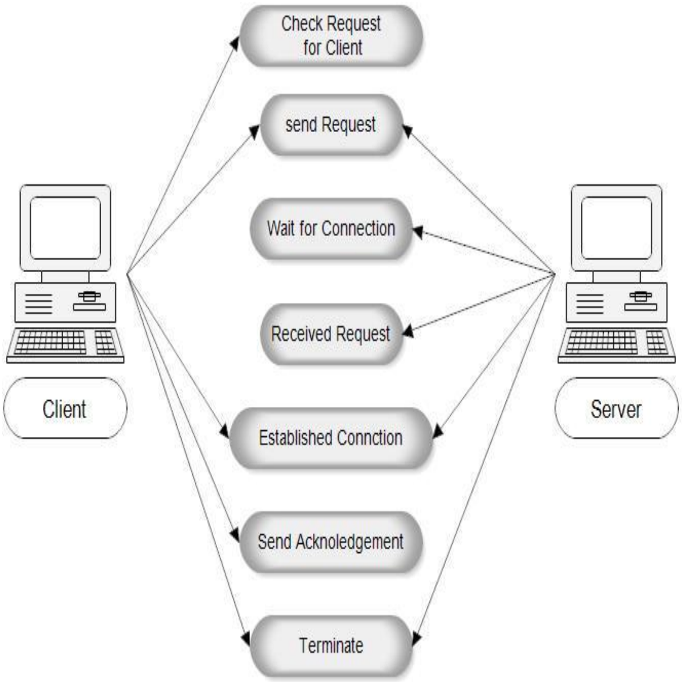

Figure 3: use case diagram for client and server.

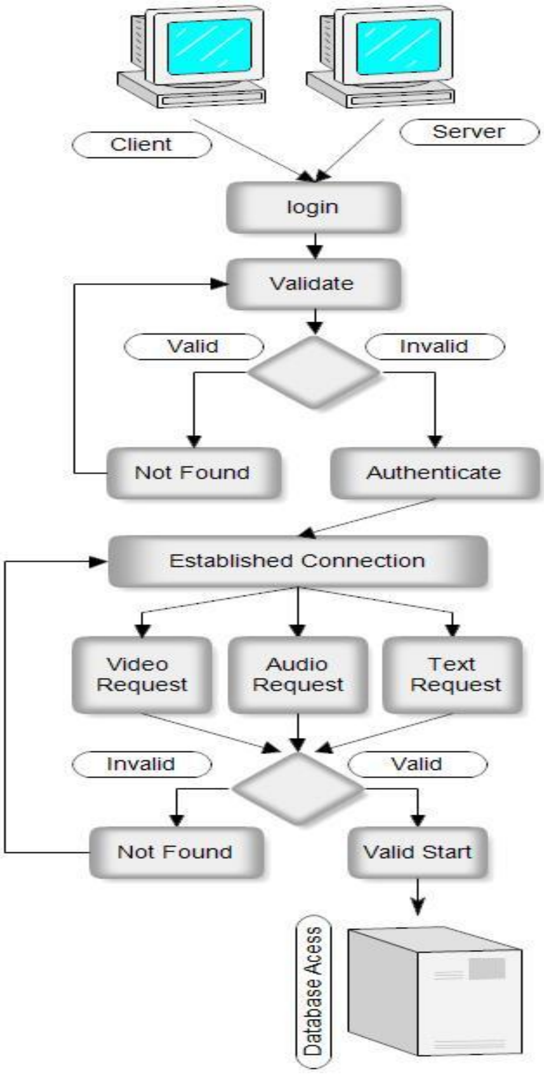

Figure 4: Client and Server Activity flow diagram

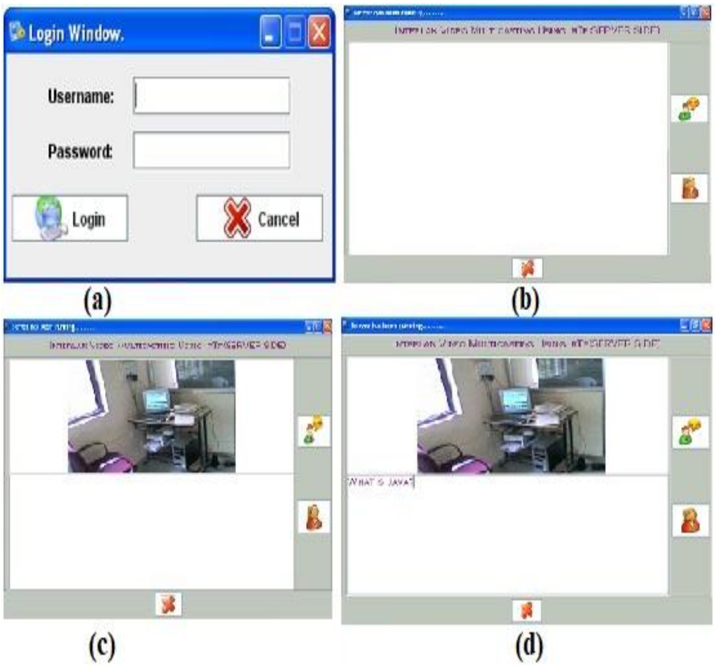

Figure 5: Implemented Server Side Images

Server Side: Figure 5 (a) shows Login Window When user want to be login he must enter the Username and Password and the click on login button to login themselves. Figure 5 (b) shows Server Side window After Login. After login in server first window to be displayed. Figure 5 (c) shows Server Side Window This time server would be started and start to transmitting video. Figure 5 (d) shows Server Side Window When Query Inserted After query sending from client side it can be displayed on server side.

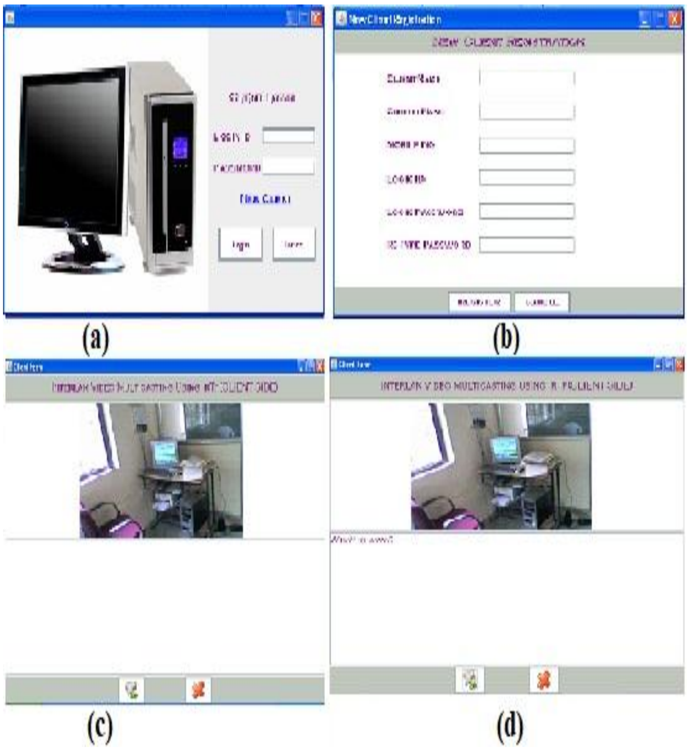

Figure 6: Implemented Client Side Images

Client Side: Figure 6 (a) shows Client Login window when user want to be login he must enter the Username and Password and the click on login button to login themselves, if the user enter the wrong information, he must give the message for his incorrect information. Figure 6 (b) shows New Client Registration Form. Figure 6 (c) shows Client Side Window with Video. Figure 6 (d) shows Client Side Window with Query.

Testing: Aim of testing process is to identify all defects in a software product. Testing is any activity aimed at evaluating the software for quality results it produces and the quality of results it can handle. Testing is an operation to detect the differences between the expected (required) result and the actual result. Testing a program consists of subjecting the program to a test inputs or test cases and observing if the program behaves as expected. If the program fails to behave as expected, then the condition under which failures noted for later debugging and correction. Our goal is to design a series of test cases that would have a high likelihood of finding errors. The software testing techniques provide a systematic guidance for designing tests that exercise the internal logic of software components and exercise the input & output domains of the program to uncover errors in program function, behavior and performance.

Detailed Test Cases:

Test ID	Name To Be Tested	Steps	Input	Outcome	Expected Output	Result
01	Username Field in Login	Enter Username	abc	Accept	Accept	Pass
	Username Field in Login	Enter Username	<blank>	Error Message	Error Message	Pass
	Username Field in Login	Enter Username	Anupam	Accept	Accept	Pass
	Username Field in Login	Enter Username	123	Reject	Reject	Pass
02	Password Field	Enter Password	abc	Accept	Accept	Pass
	Password Field	Enter Password	<blank>	Error Message	Error Message	Pass
	Password Field	Enter Password	message	Accept	Accept	Pass
03	Username Field in Account Creation	Enter Username	abc			Pass
04	Username Field in Account Creation	Enter Username	Anupam	Accept	Accept	Pass
	Username Field in Account Creation	Enter Username	<blank>	Error Message	Error Message	Pass
05	Password Field in Account Creation	Enter Password	abc	Accept	Accept	Pass
06	Re-enter Password Field in Account Creation	Confirm Password	abc	Accept	Accept	Pass
07	Password Field in Account Creation	Confirm Password	asd	Error Message	Error Message	Pass
	Re-enter Password Field in Account Creation	Enter Password	asd	Reject	Error Message	Pass

Test plan cases are first for a new user, check whether details entered as if e-mail id & password are valid. Second For a new user, check whether passwords and confirmed password fields have the same value. Third for an existing user, check whether username and password are valid and Software is tested from two different perspectives first Internal program logic is exercised using "White Box" test case design techniques second Software requirements are exercised using "Black Box" test case design techniques. In both cases, the intent is to find maximum number of errors with minimum effort and time, After registering the client, Client window would be displayed. These windows displayed the video captured and transmitted form server side. In addition, the audio should be listen on respective computers. At bottom side there is button for asking the query to server side whenever discussion is in progress. This query should be displayed on server side without disturbing server transmission. Server can give the answer for that query after ending discussion.

Test Case Summary:

Test Case ID	Description
01	To Check whether blank log in is allowed.
02	To Check whether blank password is allowed.
03	To Check if the username already exists.
04	To Check whether blank username is allowed and to ensure that a unique username is entered.
05	To accept a password from user.
06	To ensure that the user re-enters the same password and confirms it.
07	To ensure that the password entered by the user is at least 6 characters long.

5. CONCLUSIONS

RTP server applications transmit captured or stored median streams across the network. The main challenge in designing a video streaming application across the multimedia networks is how to deliver video streams to users with minimal replay jitters with video data security and efficient video data transmission. The media streams might be encoded in multiple media formats and sent out on several RTP sessions for conferencing with heterogeneous receivers. This paper describes a framework for video streaming services using RTP through the client-server network.

6. FURURE SCOPE:

Following are the future enhancements that can be done:

a) At College level this application can be work for teaching in multiple laboratory at single time.
b) For purpose of distance learning this application can be work for attending the lecture from remote place , if we can extends this application for internet also.
c) In future we can developed this application for conducting business meeting and conferences.
d) As build this application for Wireless network then it work for communicating with video from remote place.
e) After extending this application for internet making the new era of real time video communication.
f) On internet, our application would works for video chatting with relatives and friends.
g) Using this application can works for organization form remote place and do our works with prosperity and effectiveness.

ACKNOWLEDGEMENT

This work has been carried out as a part Master degree in Computer Science Engineering from RKDF Institute of Science and Technology, Hoshangabad Road, Bhopal, Madhya Pradesh. We would like to thanks our family, friends and specially Management, Principal, HOD and all teaching and non-teaching staff of Department for their support and guidelines.

REFERENCES

1) Abdemhim benslimane, "Real-Time Multimedia Services over Internet", IEEE, pp 253-261, 2000.

2) sony corporation "Streaming and Recording Capabilities", video communication system-technical documentation, vol. 1.1, August 2008.

3) Prof. Nitin. R. Talhar, Prof. Mrs. K. S. Thakare," Real-time and Object-based Video Streaming Techniques with Application to Communication System", International Symposium on Computing, Communication, and Control (ISCCC 2009) Proc .of CSIT, vol.1, pages 25-29, 2011.

4) Hatem BETTAHAR," Tutorial on Multicast Video Streaming Techniques", 3rd International Conference: Sciences of Electronic, Technologies of Information and Telecommunications March 27-31, 2005 – TUNISIA.

5) Eugene T. Lin, Christine I. Podilchuk, Ton Kalker, Edward J. Delp, "Streaming video and rate scalable compression: what are the challenges for watermarking", 2006.

6) Abhik Majumdar, Igor V. Kozintsev," Multicast and Unicast Real-Time Video Streaming Over Wireless LANs", IEEE transactions on circuits and systems for video technology, vol. 12, no. 6, june 2002.

7) Lei Chen, Chung- wei Lee "Multi-level secure video streaming over SRTP," Proceedings of the 43rd ACM Southwest Conference, Kennesaw, GA, USA, March 18-20, 2005.

8) I. Richardson, An overview of H.264 Advanced Video, [Online]. Available: http://www.vcodex.com,2007.

9) H. Schulzrinne, S. Casner, R. Frederick, and V. Jacob-son, “RTP: A Transport Protocol for Real-Time Ap-plications,” IETF RFC 3550, 2003.

10) M. Baugher, D. McGrew, M. Naslund, E. Carrara, and K. Norrman, “The Secure Real-time Transport Proto-col (SRTP),” RFC 3711, 2004.

11) C. Perkins, RTP: Audio and Video for the Internet, Ad-dison Wesley, 2003.

12) H. Schulzrinne, A. Rao, and R. Lanphier, “Real Time Streaming Protocol (RTSP),” IETF RFC 2326 (proposed standard), Apr. 1998. [Online]. Available: http://www.ietf.org/rfc/rfc2326.txt

13) J. Daemen and V. Rijmen, “AES Proposal: Rijndael,” February 2002.

14) William Stalling, “Cryptography and network Security” fourth Edition, 2006.