ENHANCED QUALITY OF SERVICE FOR THE MULTIMEDIA CONFERENCING SYSTEM USING RTCP

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LIST OF ABBREVIATIONS

RTP	Real-Time transport Protocol
RTCP	Real-Time Control Protocol
MCS	Multimedia Conferencing System
UDP	User Datagram Protocol
TCP	Transmission Control Protocol
VCs	Video Conferencing Systems
QoS	Quality of Service
NAT	Network Address Translation
MLIC	Multi LAN Internet Protocol Converter
CM	Communication Module
SOM	Session Operations Manager
MM	Multicast Manager
TM	Tunnel Manager
SR	Sender Reports
RR	Receiver Reports
SDES	Source Description
AG	Aggregator
VOIP	Video over Internet Protocol
NS	Network Simulator

PENINGKATAN KUALITI PERKHIDMATAN SISTEM PERSIDANGAN MULTIMEDIA MENGGUNAKAN RTCP

ABSTRAK

Dalam era yang berorientasikan teknologi ini, peningkatan permintaan aplikasi multimedia melalui internet meningkat dengan cepat. Aplikasi ini berkembang begitu juga dengan teknologi rangkaian untuk memberikan penyelesaian yang baik dan cekap kepada para pengguna. Persidangan video multimedia adalah penting untuk kehidupan kita baik dalam bidang akademik, industri mahupun di rumah-rumah. Menyediakan penyelesaian persidangan video multimedia yang cekap dan baik menjadi satu cabaran bagi para ahli dan penyelidik industri. Salah satu pusat di Universiti Sains Malaysia yang memberikan sistem persidangan video multipoint untuk multipoint bagi menyediakan penghantaran multimedia yang berkualiti baik serta dapat menyambungkan pelangan-pelangan dari serata dunia. Dalam kajian ini, kami mencadangkan penggunaan protokol RTCP dengan sistem MCS yang menggunakan protokol komunikasi IP / UDP / RTP , untuk menyediakan sistem yang mampu diharapkan. Dengan menggunakan RTCP, kita akan mampu untuk memperbaiki sistem yang sedia ada dan kualiti layanan (QoS), sehingga jumlah pakej hilang ketika transmisi multimedia akan dikurangkan. Kami menghasilkan topologi yang sama dengan topologi MCS yang mempunyai masa-nyata untuk menghasilkan persidangan video point ke multipoint. Pada rangkaian simulator NS-2 dan kemudiannya membandingkan sistem yang sedia ada dengan sistem MCS-RTCP yang telah ditingkatkan. Kami mengesahkan kajian kami dengan melakukan penilaian kuantitatif dengan bantuan graf yang diplot bagi menunjukkan penambahbaikan sistem MCS dengan RTCP dapat mengurangkan kehilangan

bungkusan ketika persidangan multimedia. Penyelidikan kami dapat dilanjut kerana sangat berpadanan untuk persidangan video yang berdefinisi tinggi di mana setiap perincian dari bingkai adalah penting dan dapat mengurangkan jumlah kehilangan pakej yang diperlukan. Akhirnya, kami cukup yakin bahawa kajian kami akan menghasilkan masa-nyata dan dapat ditambah kepenggunaannya dan boleh diharapkan untuk MCS yang sedia ada atau sistem persidangan video yang lain.

ENHANCED QUALITY OF SERVICE FOR THE MULTIMEDIA CONFERENCING SYSTEM USING RTCP

ABSTRACT

In this technology oriented era, the growing demand of multimedia applications over the Internet increases with an alarming rate. The application is evolving so does the networking technologies in order to provide reliable and efficient solutions to the end users. Multimedia video conferencing is an important part of our life whether in the academic area, industry or even at homes. Providing efficient and reliable multimedia video conferencing solution becomes a challenge for the industry experts and researchers. One of the centers of Universiti Sains Malaysia provided a multipoint to multipoint video conferencing system that provides good quality multimedia transmission and with clients connected from any part of the world. This study intends to explore the possibility of using RTCP protocol with existing MCS system that uses IP/UDP/RTP communication protocol, in order to provide reliability. By using RTCP, we will be able to improve the existing system and Quality of Service (QoS), therefore the amount of packet loss during the multimedia transmission will be reduced. The same topology was created that projects the real time MCS topology in order to generate the same point to multipoint video conferencing in the network simulator NS-2 then compared the existing system with the enhanced MCS-RTCP system. The present study was justified by doing quantitative evaluation with the help of plotted graph that shows the enhanced MCS with RTCP system reduces the packet loss during the multimedia conferencing. This research work can be extended further as it is highly suitable for the high definition video conferencing where every detail of frames is important and reduces number of

packet loss required. Finally, this study has potentials to be much similar in the real time and is highly scalable and reliable for existing MCS or other video conferencing systems.

CHAPTER 1

INTRODUCTION

1.1 Introduction

Over the last few years, the utilization of audiovisual conferencing solutions has increased due to the availability of the Internet to the public and with the enormous scientific expansions and diversity of resources of learning have increased the need to find modern communication means which combines features of efficiency, time saving and convenient, and facilitate the exchange of information, and in fact, videoconferencing solution was one of the most successful solutions.

We can identify the videoconferencing system as a set of interactive communication technologies which helps many locations be connected via audio and video transmission simultaneously. The videoconferencing system use communications of audio and video to facilitate people from different locations to conduct meetings which are akin to a real meeting. In this way, individuals and entities can save time and travelling costs attributed to joining conferences and meetings in different locations (Mcdonald and Nelson, 2008).

There are two types of audiovisual conferencing: one-to-one meetings and multi-point meetings. In both types, files and documents can be shared by members from different sites. Over the years, several video conferencing solutions have been presented either as commercial or free software products. In general, video conferencing systems have been adopted in many fields of our life, and have led to rapid developments and advancement in several significant areas like medical, social, business and education, distance learning in particular. It gives students ample chances of being in contact with their teachers who enlighten those never-ending-curious minds with answers for their queries and questions. Furthermore, teachers and lecturers will be able to reach remote students or students who live in isolated places (Prokkola and Hanski, 2005). Nowadays, students of video conferencing systems (VCs) are able to explore, analyze and share information, ideas, and communicate with their colleagues in different sites. Here are some examples of how video conferencing systems benefit the people in the campuses:

1. Collaboration with visiting lecturers to give lectures to other institutions.

2. Researchers collaborate with other colleagues in different schools and scientific institutions without wasting time and money due to high cost travels.

For that reason, the necessity for improving such service becomes highly significant. Many solutions are presented to enhance the video conferencing products family, either by providing clearer voice, better video quality, adding new features or combination of them. In terms of computer networks, the term "Quality of Service (QoS)" plays an important role in determining the acceptable range of bandwidth utilization, latency (delay), and error rate. Hence, to provide an acceptable service for a multimedia conferencing system, QoS feature is much required. Several methods have been proposed on how to enhance QoS feature for given software or applications (Vogel et al., 1995). In normal point-to-point conferencing system, the (RTP) Real-Time transport protocol can be used for transmitting videos and audios, and we have a particular protocol called Real-Time Control Protocol (RTCP) that runs together with the Real-Time transport Protocol

(RTP) which is used to provide out-of-band control information for an RTP flow, and used to ensure QoS for the multimedia communication over the internet.

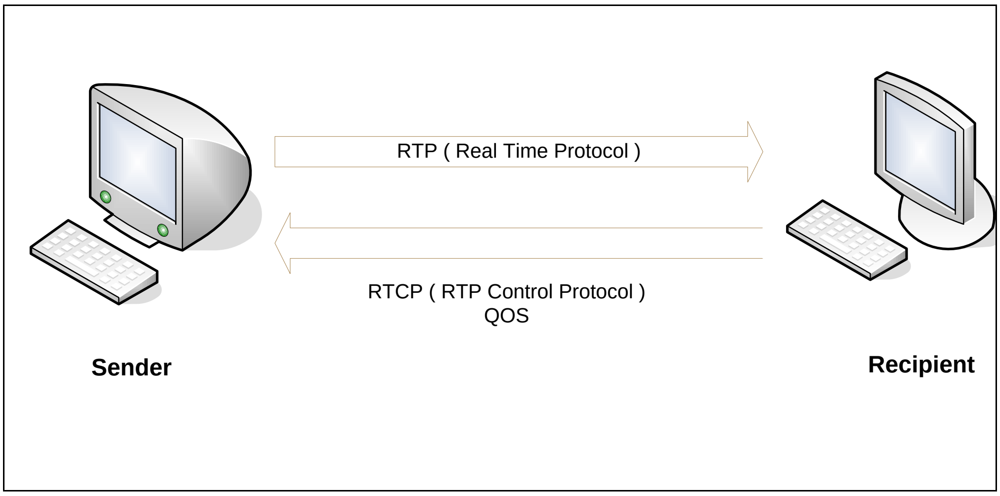

Figure 1.1: Normal point-to-point conferencing systems

1.2 Background

One of the widely used video conferencing systems these days is the Multimedia Conferencing System or (MCS) (Kolhar et al., 2008), which presents a revolutionary Multipoint-to-Multipoint conferencing system (Figure 1.2) and it allows conferencing with as many people as desired from anywhere around the world (Mlabs, 2005). Unfortunately, the current MCS does not provide Quality of Service (QoS) for video and audio transmission and it relies on Real-time Transport Protocol (RTP) for delivering video and audio content through the Internet, which adversely affects the efficiency of MCS and may lead to some problems during the communication between the MCS users, such as:

- *Cost of bandwidth:* There is a need to achieve acceptable bandwidth to transmit real-time video. In addition, the traditional routers typically do not actively participate in congestion control of excessive traffic which can cause congestion. This leads to degradation the throughput of real-time video.
- *Delay:* The congestion in the Internet could incur excessive delay as real-time requires timely delivery. Real-time video must be playing out continuously. Video packets arrive at the destination in time to be decoded and displayed. If the video packets do not arrive on time, the playback process will pause or considered lost which is annoying to human eyes (Krithivasan, 2006).
- Loss: The packet loss ratio could be very high during network congestion. Internet does not provide any loss guarantee, which leads to serious degradation of video quality.

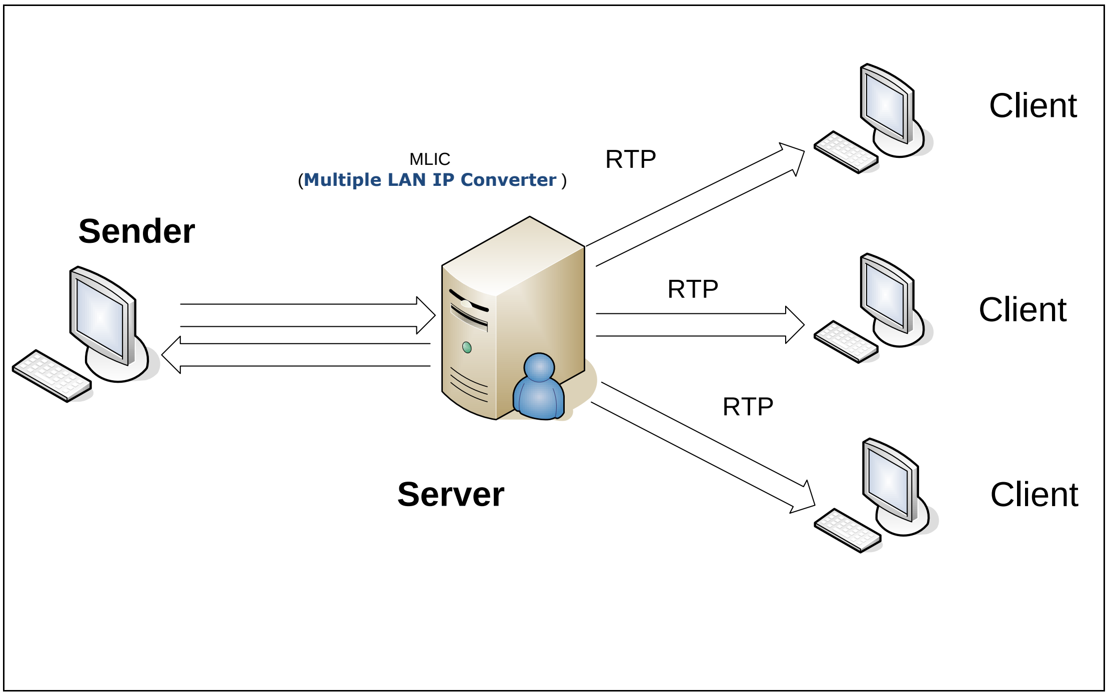

Figure 1.2: MCS architecture

1.3 Motivation and Justification

Due to the existing quality level of the multimedia communication elements of MCS, the demand on improving such elements becomes increasingly important to enable MCS' users have smoother communication with their friends, family members and also with their colleagues. Furthermore, relying on the multimedia conferencing applications worldwide has increased recently, due to the great benefits of these applications which may lead to the need of regular enhancement of such applications. In this study, the researcher attempts to answer the main question, is how to enhance video and audio Quality of Service for the Multimedia Conferencing System (MCS), and guarantee that the audio and video are clearly presented to all.

1.4 Problem Statement

As have been mentioned in the introduction section, MCS is one of the most popular applications that are used for multimedia communication over any IP-based networks (Kolhar et al., 2008). MCS has many advantages over so many other multimedia conferencing systems presented in Document Conferencing (DC), Application Conference (AC) and integrated Instant Messaging Services (IM) (Mlabs, 2005). But sometimes MCS may have the degrading problem in audio and video transmission such as; distorted and greenish picture (Figure 1.3), because there is no Real-Time Control Protocol (RTCP) mechanism implemented as additional feature to the transporter protocol which is represented by Real-Time transport Protocol (RTP).

Figure 1.3: Examples for degrading the quality in MCS

1.5 Scope and Limitations of research

This study focuses on adding and evaluating a suitable RTCP mechanism to MCS by analysing the communication architecture of MCS and by evaluating the current RTCP mechanisms. Hence, the scope of this study will be focusing on how to improve the quality of the audio and video communications exchange among MCS users by adopting a good RTCP mechanism. Therefore, it will not cover other applications, such as security services and confidentially.

Objectives of research

1- To study the possibility of implementing the RTP\RTCP with MLIC mechanism in order to provide better quality of service for the Multimedia Conferencing System (MCS).

2- To implement and evaluate the proposed RTCP mechanism in MCS.

1.6 Contribution of research

The main contribution of this research is represented by allowing MCS users to communicate smoothly by exchanging the multimedia elements with a higher quality that can be considered as a result of adding RTCP mechanism to MCS. Furthermore, this study will evaluate the existing RTCP mechanisms and choose the most suitable RTCP mechanism based on the architecture of MCS. By adding an RTCP mechanism to MCS, the audio and video quality will improve due to the built-in feature of RTCP such as reporting.

1.7 The organization of thesis

This thesis is categorized into five chapters. Chapter One begins by giving an introduction to the video conferencing and its importance, followed by presenting the motivation and problem statement followed by scope and main objectives for this thesis and ends with Contribution of thesis, Chapter Two offers background for the multimedia conferencing system, and it also covers the important transmission protocols RTP\RTCP, also it covers some related studies in the area of thesis, Chapter Three explains the proposed method in this thesis, Chapter Four consists of the findings of this research. It also contains the discussion and analysis of the study. Finally, Chapter Five discusses the conclusion of this study and presents some recommendations for future work.

CHAPTER 2

LITERATURE REVIEW

2.1 Introduction

In this chapter, the researcher makes every endeavor to shed light on the multimedia conferencing system (MCS) and explain the mechanism of the MCS, also reviews the two important transportation protocols that are used in MCS, the Real-Time transport protocol (RTP) which is used to send packets from recourses to participants, and RTCP real time control protocol which is an accompanying protocol for the Real-Time protocol which is used for feedback transmission in order to control the multimedia session behavior. The researcher will present some supposed modifications for RTCP like (S-RTCP) scalability RTCP (El-marakby and Hutchison, 1998). Finally, the next section provides a review and discussion for related work and previous researches in the QoS for video streaming, which has been carried out, and deemed useful in the present study.

2.2 Multimedia Conferencing System (MCS)

The MCS introduced by (Sureswaran, 1994) exhibit that multimedia conferencing system operates on a switching method principle for optimum low consumption of bandwidth with capacity of adopting a large number of users to participate in the conference. Furthermore a set of conference control options like the RSW control criteria controls multimedia conferences. The MCS presents a revolutionary Multipoint-to-Multipoint conferencing system with capabilities of accepting and accommodating requests from a vast number of users anywhere in the world, without affecting any other applications using the infrastructure of the network. MCS implements a customized and modified RTP (Real-Time Transport Protocol) for ensuring the proper sequencing of packets, and synchronization.

2.2.1 MCS Architecture

It's well known that video conferencing systems consume high bandwidth and system resources. On the other hand, MCS has been designed using a distributed architecture that divides the system into small components to process orders/requests from components. This greatly enhances multimedia conferencing tasks via the distributed processing mechanism. In addition, this distributed design can easily be adapted to work with any advanced network infrastructure for better transmission and delivery of data. The distributed architecture of MCS comprises of three main components that makes the MCS fully distributed system:

a) Server

The server component which is regarded as the main component of MCS, acts as the controller. Using predefined rules upon conditions and statuses, it manages the conferences. However, it's invisible to users, as the system runs transparently. The administrator has the ability to access this server to accomplish administrative tasks. Being the chairman, in each system only one server component allowing proper coordination of numerous sessions of multimedia conferences and control between clients wishing to obtain the right to speak. The server component listens to all incoming connections from the clients of MCS via TCP protocol which is well known for its reliable connections between server and the clients (Handley et al.,2003; Brosh et al.,2008).

b) Client.

The client components consist of individual PCs configured with suitable audio and video capturing or playback hardware. Through graphical user interfaces of clients, the users are able to interact with the conferencing system by logging in to create and execute required operations and logout through configured video and audio capturing or playback hardware. The client component comprises of six subcomponents (Figure 2.1):

i. Interface module
ii. Communication module
iii. Conference control module
iv. Compression and Decompression module
v. Video module
vi. Audio module

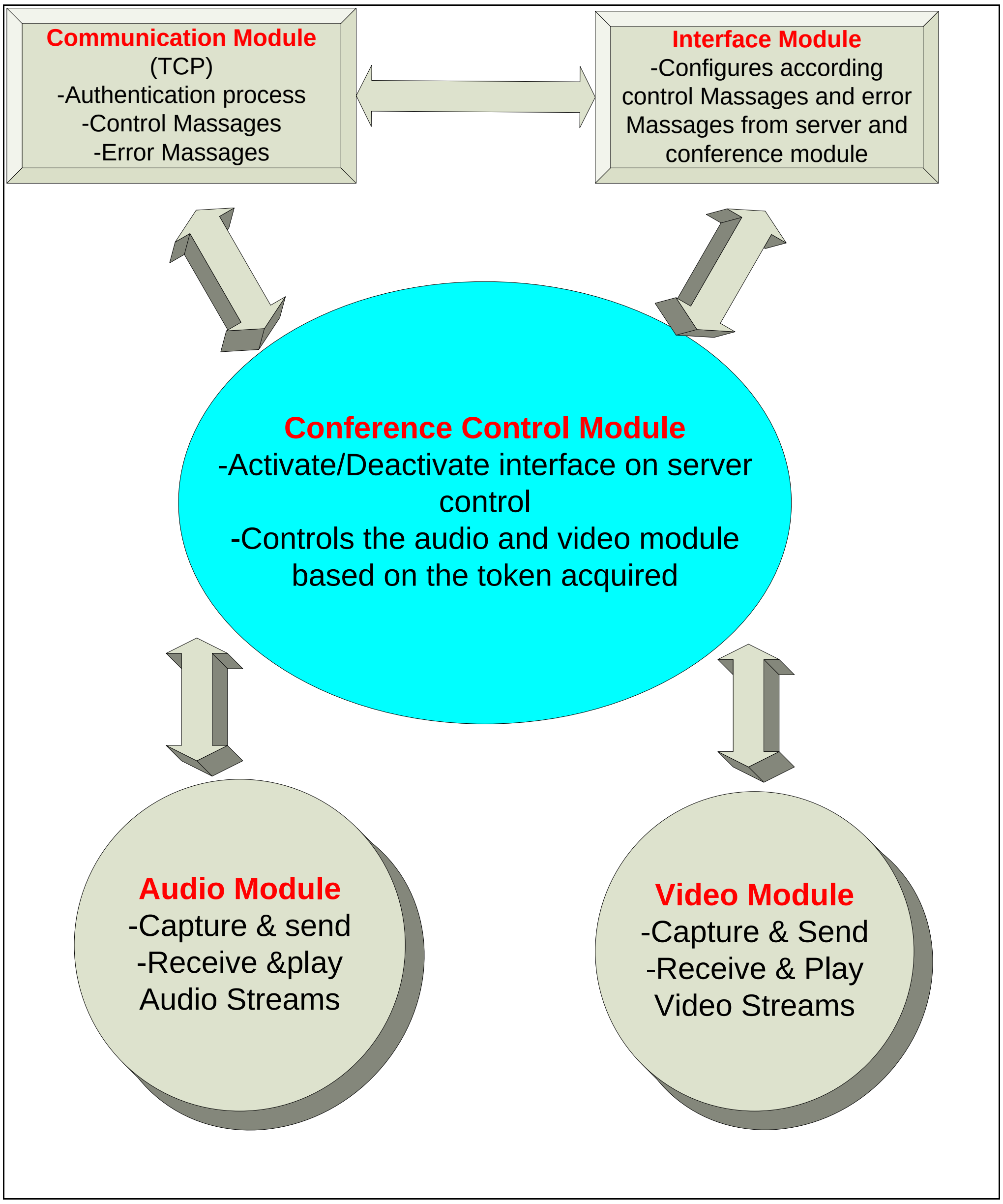

Figure 2.1: The MCS Client components

c) Data compression and data decompression

Due to the fact that multimedia data requires high processing power, and compression of the continuous stream will definitely take up even more resources, it requires this component to be implemented on a separate machine.

2.2.2 MCS System Description

MCS operates on a distributed architecture platform, consisting of different components, i.e. clients and servers, each on a well oriented TCP protocol suite. Both components work independently in a networked environment. Some special features installed within MCS are explained as below:

1) Security Mechanism.

   Only invited users can join the conference. However MCU (micro controller unit) allows any user to call in its IP address and interact with other conference members in a stealthy monitoring environment.

2) Real Time information updates.

   Tasks like latest statuses of the conference can be received by both the administrative and all other members interacting during that particular time and instances. These tasks include initiation and connection requests, termination of requests and connections of all participants at that very particular time.

3) Live updates or software patches.

   The system has the capability of updating itself automatically depending on how the administrator configured its related updating scheduled tasks.

4) Global Roaming and NAT Solution.

   According to the NAT capabilities for dynamic IP configuration, multimedia conferencing system frees the clients from the IP constraint and allows them to use the system from anywhere around the world.

5) Server-to-Server mechanism (S2S)

Utilize this mechanism by MCS resulting into unlimited number of participants in a single conference.

2.3 MLIC (Multi LAN Internet Protocol Converter)

MLIC function by taking the multicast stream, and then encapsulates it to unicast and tunnels it across to the LANs. The MLIC enhances the distributed multimedia conferencing environment architecture by encapsulating the functionality of multicast and transmission via non-multicast enabled network like the Internet. This functionality is provided by MLIC by creating tunnel between participants and multimedia conferences.

Generally, we have three types of communication:

1. Point-to-Point communication.

   This type is the simplest method to communicate on network.
2. Point-to-multipoint communication (one-to-many)
3. Multipoint-to-multipoint communication (many-to-many)

There are three methods to achieve point-to-multipoint or multipoint-to-multipoint communications from the network viewpoint (Table2.1):

Table 2.1: Kinds of transmission

Method of transmission	Describe
Broadcasting	(recommended) -All nodes in the network receive the transmission even if it not required. Hence the system resource is wasted.
Retransmission	(ineffective) -All the nodes must retransmit the same data to relevant receivers in the network one by one. As a result, the increase in number of receivers will also increase the consumption of bandwidth.
Multicasting	(recommended) - The best method for achieving Point to Point and multipoint to multipoint transmission without increasing the bandwidth consumption.

2.3.1 MLIC Architecture

In (Table 2.1) above, there are four components that work together in forming the MLIC architecture (Figure2.2):

a) CM (Communication Module)

The CM's function is to retrieve request from (MCS) server by using the sockets of TCP.

b) SOM (Session Operations Manager)

The job of SOM is handling the entire operations of MLIC server and sends configuration parameter to the tunnel manager (TM) and the multicast manager (MM).

c) MM (Multicast Manager)

The MM ensures the capturing and sending multicast information for the wide area conferencing. The MM tasks can divided into three sub modules; instructor, sender and receiver of multicast information

d) TM (Tunnel Manager)

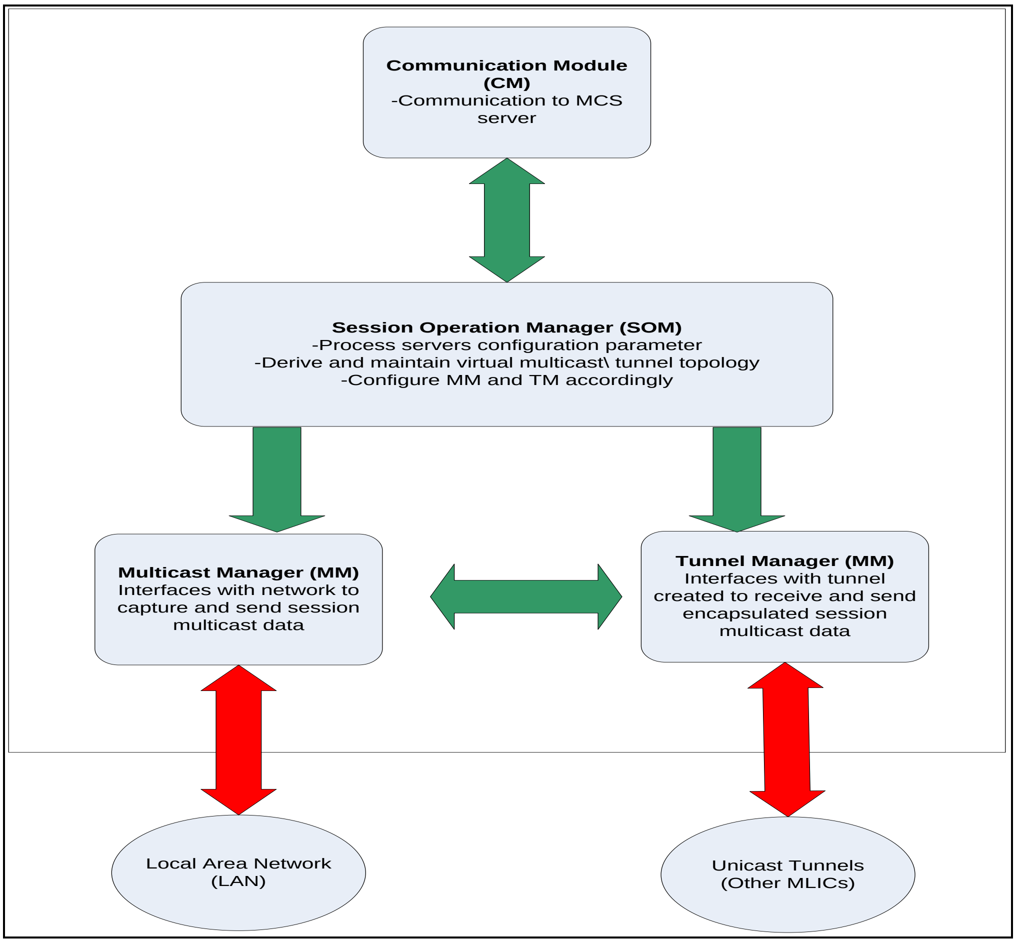

Figure 2.2: MLIC Architecture

2.3.2 Multiple LAN IP Converter (MLIC) Entity

The main function of MLIC is to enable multiple LANs interconnected by WANs to connect in a conference. When MCS transmits multicast packets, these packets are generally dropped by the WAN. In order to avoid the dropping of packets within WAN, MLIC converts these multicast packets into unicast packets to enable them to pass through these WAN routers to reach the other LANs and on the receiving end, another MLIC converts these unicast packets and retransmits them as multicast packets onto the LAN (Figure 2.3).

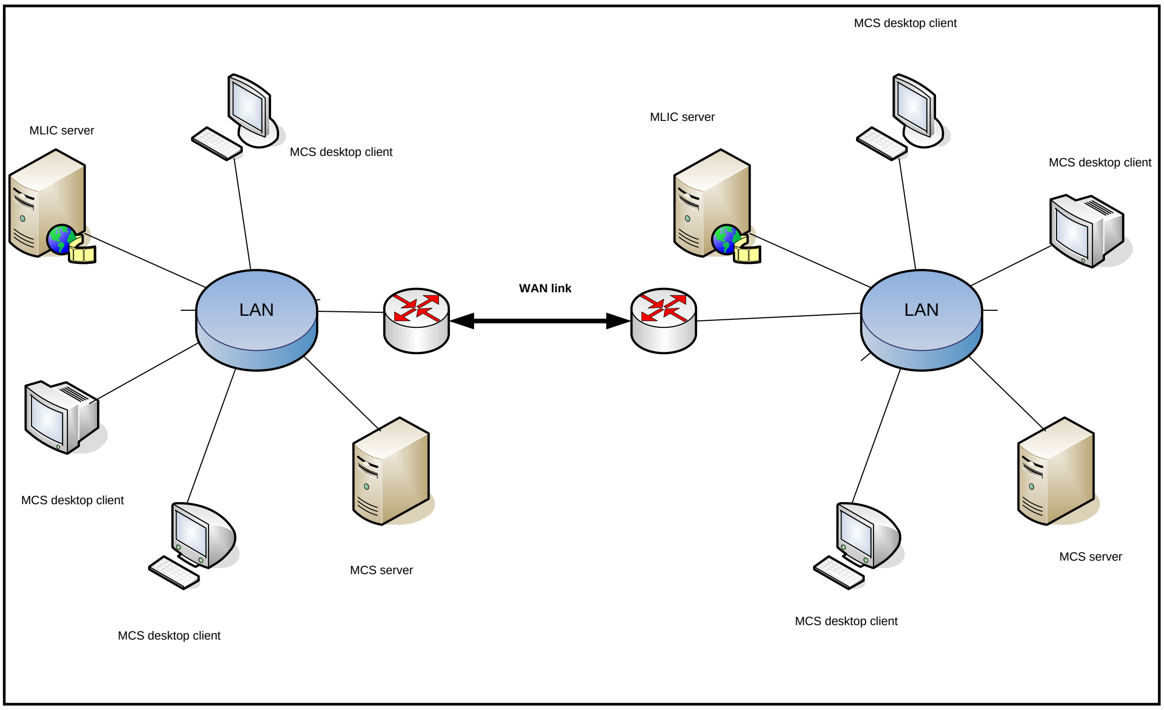

Figure 2.3: MCS in a Multi-LAN environment (With MLIC entity) (Saravanan, 1998)

In next section, the researcher will discuss the basic principle of Real-time transport protocol (RTP) and Real-Time Control Protocol (RTCP)

2.4 Overview of RTP implementation

RTP derived its name from Real-Time Transport Protocol, The key standard for media (audio/video) transport in IP network is the Real-Time Transport Protocol (RTP) (El-marakby and Enugula, 2006). RTP typically sits on a top of UDP/IP transport (Figure 2.4) and aims to provide services useful to enhance the transport of real-time media over IP network; these services include timing recovery, loss detection and correction, payload and source identification, reception quality feedback, media synchronization, and membership management. RTP itself doesn't guarantee timely delivery of packets, but it does provide mechanisms for the sending and receiving applications to support streaming data. RTP was originally designed for use in multicast conferences, using the lightweight session model. It provides a common media transport layer, independent of the signaling protocol and application (Cavusoglu et al., 2005).

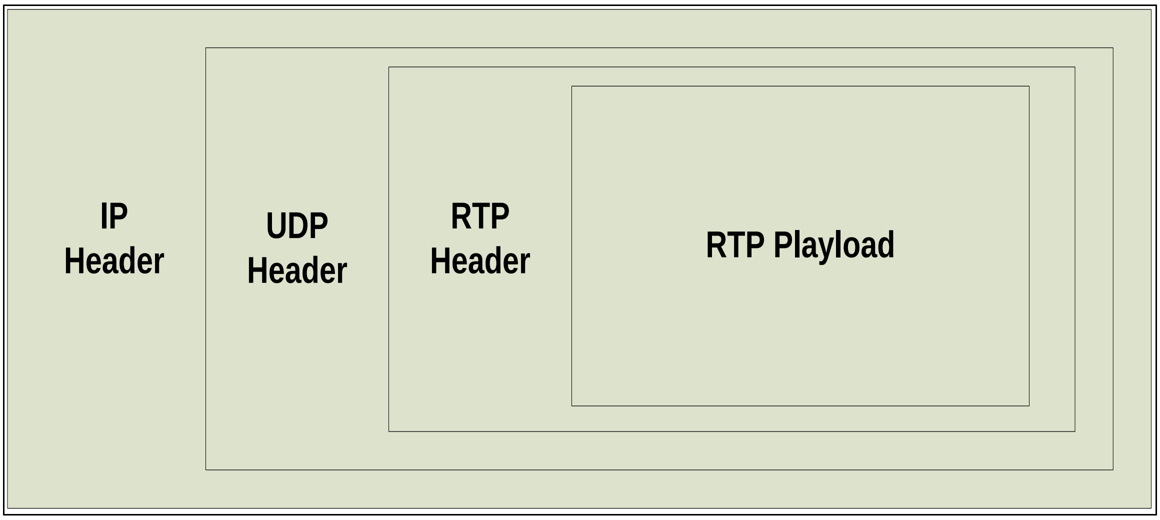

Figure 2.4: IP Packet Containing RTP Data

2.4.1 Behavior of RTP sender and receiver

The sender is responsible for capturing and transferring data (audio and video) for transmission, in addition to generating the RTP packets.

In the beginning, all the frames, which are compressed, will load into RTP packets, then ready to be sent. If the frame is large, they may be divided into several RTP packets, if they are small; several frames maybe be combined into a single RTP packet (Chow et al., 2007). Then to channel, coder maybe be used to correct the packets or to reorder packets before transmission instruction.

The sender also receives reception quality feedback from the participants and uses it to adapt its transmission of that information.

The receiver is functional for several things:

- Collect RTP packets from the network,
- Correct any losses.
- Decompress the media.
- Present the result to user.

2.4.2 The RTP packet format

V=2	P	X	CC	M	PT	Sequence Number
Time Stamp						
Synchronization Source (SSRC) Identifier						
Contributing Source (CSRC) Identifier(s)						
Payload Header (Optional Depending on the Codec)						
Payload						

Figure 2.5: RTP Header format

The following sections describe the octets in the RTP header shown in (Figure 2.5):

The fields in this first octet of the RTP header are described as follows:

■ Version (V): 2 bits - This field identifies the version of RTP. The Version field is set to a value of 2 in most RTP implementations to denote the RTP profile defined in RFC 3551.

■ Padding (P): 1 bit - this packet contains one or more additional padding octets at the end, which are not part of the payload. Some encryption algorithms with fixed block sizes might need padding to carry several RTP packets in a lower-layer protocol data unit.

■ Extension (X): 1 bit - if the extension bit is set, exactly one header extension must follow the fixed header.

■ Contributing Source (CSRC) count (CC): 4 bits - 4 bits-This contains the number of CSRC identifiers that follow the fixed header.

■ Marker (M): 1 bit- the interpretation of the marker is defined by the RTP profile in use. In order for significant events such as frame boundaries to be marked in the packet stream, The M bit comes into action. The M bit is helpful in video streams because it allows the endpoint to know that it has received the last packet of the frame so that it may display the full image. Without the M bit, the receiver would need to wait for one additional packet to detect a change to a new frame number.

2.4.3 RTP control protocol (RTCP)

RTP Control Protocol (RTCP) is the companion control protocol for RTP which used to provide out-of-band control information for an RTP flow as shown in the next figure and the reception quality feedback, participant identification, and the synchronization between media streams (Gharavi et al., 2004). RTCP runs alongside RTP and provides periodic reporting of this information (Baldo et al., 2004; Chesterfield, 2003). Although data packets are typically sent every few milliseconds, the control protocol operates on the scale of seconds. The information sent in RTCP is necessary for synchronization between media stream; i.e., for lip synchronization between (audio and video).

The RTCP's two main functions are:

a) It provides feedback on the quality of the media distribution. This function is performed by RTCP receiver and sender reports.
b) For each sender, RTCP maps RTP time stamps for each RTP stream to a common sender clock (Busse, 1996), which allows audio and video synchronization on the receivers.

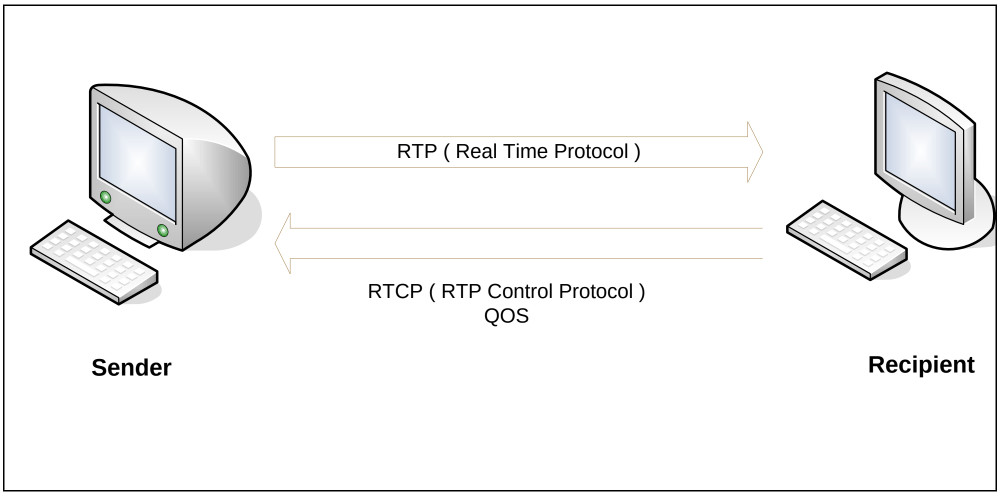

Figure 2.6: RTP and RTCP control protocol

The RTCP packets contain direct information for quality-of-service monitoring. The RTCP consumes about 5% of the total bandwidth (Chesterfield et al., 2003; 2007). The Sender Reports (SR) and Receiver Reports (RR) exchange information on packet losses, delay and delay jitter (Krithivasan and Iyer, 2006). This information may be used to implement a TCP like flow control mechanism upon UDP at the application level using adaptive encodings (Ponec et al., 2009). A network management tool may monitor the network load based on the RTCP packets without receiving the actual data or detect the faulty parts of the network. The RTCP packets carry also a transport-level identifier (called a canonical name) for RTP source, which is used to keep track of each participant. Source description packets may also contain other textual information (user's name, email address) about the source. Albeit the source of the RTP packets is already identified by the SSRC identifier, an application may use multiple RTP streams, which can be easily associated with this textual information.

2.4.4 RTCP packet formats

V=2	P	IC	PT	Length
Packet Type Specific Information				

Figure 2.7: RTCP packet format

Version (V): 2 bits—Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. The version used in most implementations is 2, corresponding to the RTP profile defined in RFC 3551.

■ Padding (P): 1 bit— this packet (RTCP) contains some additional padding octets at the end that are not part of the control information and it represents 1 bit. The padding octets that should be ignored are represented by the last octet of the padding normally taken into account.

■ Item count (IC): 5 bits—Some RTCP packet formats contain a list of items that are specific to the packet type. This field is used by the individual packet types to indicate the number of items included in this packet.

■ Packet type (PT): 8 bits-Identifies the RTCP packet type.

■ Length: 16 bits-Specifies the length of this RTCP packet, excluding this header. A value of 0 is valid and indicates that this packet contains just the fixed header, consisting of the first octet.

There are five types of RTCP control protocol:

Table2.2: Types of RTCP

RR	**Receiver report**
SR	**Sender report**
SDES	**Source description**
BYE	**Membership management**
APP	**Application-defined**

2.4.5 Format of sender report

The RTP senders (endpoints or conference server) provide information about their RTP streams through the SR packet type. SRs serve three functions:

- ■They provide information to synchronize multiple RTP streams.
- ■They provide overall statistics on the number of packets and bytes sent.
- ■They provide one half of a two-way handshake that allows endpoints to calculate the network round-trip time between the two endpoints.

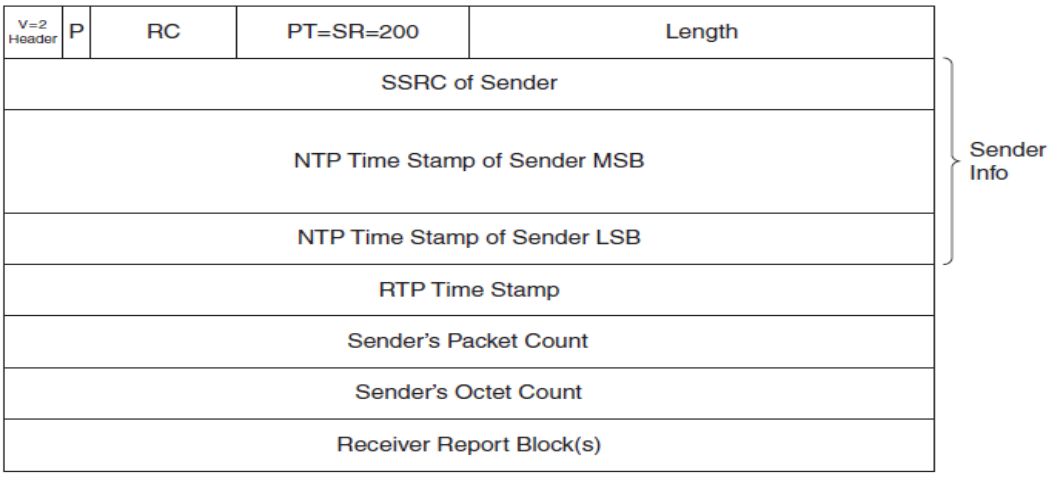

Figure 2.8: Sender report format

2.4.6 Format of receiver report

The RTP receivers (endpoints or conference server) provide periodic feedback on the quality of the received media through the RR packet type. An endpoint can use this information to dynamically adjust it's transmit rate based on network congestion. For example, if a video endpoint detects high network congestion as a result of packet loss, the endpoint may choose to send at a lower bit rate until the congestion clears.

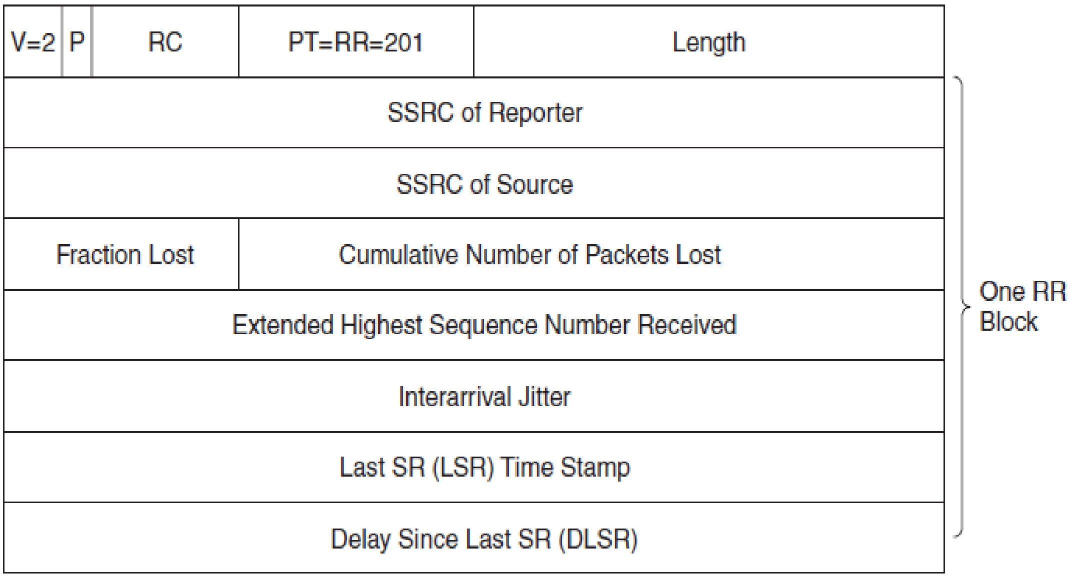

Figure 2.9: Format of receiver report format