ISSN(Online): 2320-9801
ISSN (Print): 2320-9798

International Journal of Innovative Research in Computer and Communication Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 3, Issue 10, October 2015

An Investigative Study on Performance Metrics that Affects Video Conferencing over Wireless Network

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ABSTRACT: Videos conferencing is a method of group discussion that finds applications in real time situation for the purpose of achieving a certain objectives. In this paper we investigate the performance metrics that affect video conferencing over wireless network with a view of addressing some crucial issues involved. These metrics has a great effect of degrading videos quality, audio and lip synchronization. We attempt to proffer solution to these problems in order to achieve efficient and reliable group conversation over the internet

KEYWORDS: Packet Delay; Packet rejection; Packet Loss; Video conferencing; Multicast

I. INTRODUCTION

Delivering real-time video over the Internet is becoming the fasted and convenience way of communication because it saves a lot of time and resources. Many organization and schools adopted the use of this technology and it is becoming the pace of the day. Video conferencing over wireless network is present in three different multimedia applications: video broadcasting, video on-demand, and video conferencing. The main challenges towards reliable video conferencing over wireless networks are limited availability of bandwidth for transmission, packet losses, delay, congestion and rejection of packets that occur in the links [1].

Video conferencing is the process of communicating to a large or small group of people in different locations over the internet**.** Video-conferencing enabled meetings happen in either point-to-point or with multi-point.In point-to-point, one person or group is connected to another [2].In multi-point video conferencing, more locations are connected together, where all participants can see and hear each other, as well as see any content being shared during the meeting.

Figure 1: Video conferencing Concept

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In this situation, digital information streams of voice, video and content are processed by a central, independent software program [3]. As shown in Figure 1, Videoconferencing is used for various purposes: in electronic learning (courses, instructions, etc.) for connecting with guests and experts in various fields of expertise, for professional activities and social happening [4].

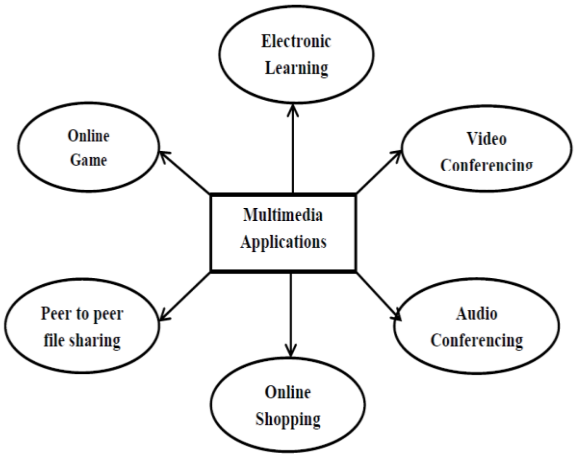

Fig 2: Application of Multicast [5].

Figure 2 above shows various application of multicast. Amongst these multimedia applications, video conferencing usually has large number of participants and it is frequently used for group discussions such as meeting, conferences, Lectures and so on.
Two main protocols have emerged in video conferencing namely, SIP (Session Initiation Protocol) and H.323. H.264 is the new industry standard for encoding and decoding of visual information and it was purposely developed to provide high-quality video at lower bandwidth over a various networks and systems.Scalable Video Coding

(SVC) is an extension to the H.264 protocol which is established to facilitate the enablement of video conferencing on a wider range of devices, such as tablets and mobile phones.
Meanwhile, due to the real-time interaction between users, the quality of a video and audio during video conferencing is more sensitive to packet losses and delays than one-way video streaming. It is therefore of great interest for end users and application developers to assess the quality of video calls under different network conditions[5].As shown in Figure 2, Video conferencing is one of the key application of multicast that is receiving great attention in recent times because of it the most reliable means of group communication over the internet.

II. RELATED WORK

Considerably amount of works has been carried out on the impact of performance metrics on video conferencing over wireless networks. In the work of [7], they studiedthe rate control and video quality of Skype video call by measuring it behavior on a controlled network testbed. This was achieved by varying packet loss rate, propagation delay and available network bandwidth in order to observe how Skype adjust its sending rate, video rate and frame rate. The study reveals that Skype is robust against mild packet loss and propagation delay and can efficiently utilize the bandwidth. The authors in [6]adopted the use of video coder decoder (Codec) software implementation recommended by UIT-T recommendation H.261 which was developed for integrated service digital network (ISDN) for internet environment. They proposed an error control scheme and an output rate control scheme that adopts the image coding process based on network condition. The new video conferencing software for internet application required lesser bandwidth and is compatible with the hardware codecs.[4]presented video conferencing system that meet performance constraints for use in consumers homes by improving the existing home technology such as video chat aimed at providing high quality audiovisual and low end to end delay. They focused on the development of the audiovisual

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composition engines by validating the audio visual composition within the acceptable limits of the delay. The system was compared with the existing video chat such as Skype and there is a qualitative difference in quality. The proposed scheme provides dynamic audiovisual composition and allows integrated of audio and video communication in other multimedia application such as online gaming system which is not possible with Gtalk, Skype and iChat. Also, [3]proposed an algorithm that achieved satisfactory encoding and decoding times at relatively low cost by utilizing vast computational capability of the modern. They adopt a method that is very suitable for video conferencing that demand real time performance. They employ the use of scalable video coding algorithms based on the contourlet transform that incorporates both lossy and lossless methods as well as variable bit rate en coding scheme to achieve compression.
In literature, [13]improved the performance of NCA algorithm for minimizing the cost of bandwidth during multicast within the channel. The results obtained were latter interpreted using Shannon-Hartley channel capacity theorem. The study reveals that as more key performance metrics of loss, delay and rejection of packets are collectively considered, the cost of bandwidth decrease. When interpreting the result using Shannon-Hartley channel capacity theorem, the cost of bandwidth within the channel was under estimated.
[2]Studied the performance of video chat application under congestion. They investigated that congestion causes changes in bandwidth which significantly affect video and audio quality. A testbed was built in order to analyse the behaviours of Skype, window live messenger, eyebeam and X-Lite as they react to changes in available bandwidth due to congestion. The study reveals that Skype adapts gradually to changes in bandwidth because it also monitors RTT and jitter on top of packet loss than Window live messenger, Eyebeam and X-lite.

The problem that prompted this research right from the onset is to determine the effects delay, loss, congestion and rejection of packets during video conferencing which are some of the great impairment that hinders smooth transmission of packets during multimedia applications over wireless network. This metrics degrades video quality, audio quality and lip synchronization during video conferencing. There is need to carefully study and investigate their effects on video conferencing in order to proffer solution for cost efficient multicast.
The reminder of this paper is organized as follows. Section one deals with the introduction, section two covers key performance metrics that hinders smooth video conferencing, the summary of the findings is presented in section three and section four concludes the paper.

III. PERFORMANCE METRICS THAT HINDERS EFFICIENT VIDEO CONFERENCING

There are quite a number of performance metrics that affect various multimedia applications over wireless networks. This factors leads to higher bandwidth consumption over the internet and it discourages many multimedia applications such as video conferencing, video streaming, and electronic shopping. These performance metrics includes:

A. *Packet loss:* Videoconferencing over the wireless network faces many challenges. Packet loss occurs when one or more packets of data travelling across a computer network fail to reach their destination. Packet loss is typically caused by network congestion. Packet loss is measured as a percentage of packets lost with respect to packets sent. Packet loss reduces the throughput of the network, severely degrade video quality and increase latency due to additional time needed for retransmission [8]. A single occurrence of packet loss can introduce error in a reference frames which can propagate to its succeeding frames and get amplified as more packet are lost making the compressed data very sensitive to transmission error. Packet loss due to transmission error often leads to serious video quality degradation, which not only degrades the quality of current frame, but also leads to error propagation to subsequent frames due to the motion-compensated prediction technique used in standard video codecs. It is measure in decibel or percentage of the messages sent. To overcome the problem of packet loss, Forward Error Correction is used by adding extra packets known as redundant packet or parity packet.

B. *Packet Rejection*: Packet rejection or dropping occurs when queues are not efficiently managed and packets begin to experience delay in such a way that the amount of buffered packets is greater than the queue length [9]. Packets that are delayed too long are rejected in order to meet the timing requirement. Packet rejection also occurs when packets are corrupted on transit and when the buffer occupancy is above the threshold value, the arriving low priority packet will be rejected [10]. Most numbers of rejected or dropped packets existed in literatures are expressed in percentage of the packets sent [11]. To avoid packet rejection, complete buffer occupancy should be avoided.

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C. *Packet delay:* The delay of a packet is defined as the time it takes for the packet to reach all its destination after it arrives at its source [12]. Similarly, one of the principle elements contributing to delay is queuing within the network [7]. To avoid packet delay, delay constraint can be adopted where a reasonable value of delay tolerant is chosen to allow maximum flow of packets during multicast, to ensure efficient packet delivery and to take care of large variations in end to end delay [13]. This also takes care of other delays such nodal processing delay, which deals with the physical characteristics of the path a packets traversed, delay on the process of transmission which is a function of link capacity, delay along the route, and queuing delay which depend on the traffic load along the path. Delay constraint is, however, one of the most important requirements in real-time applications. A video packet arriving later than its presentation time will become useless for a client, making packet scheduling important in retransmission-based error control for wireless video conferencing [14].

D. *Packet congestion:* Congestion has been the major reason for packet delay during multicasting and several packets are delayed in the network [15]. This occurs when the incoming packets at the node approaches or exceeds that of the outgoing packet. In such a scenario, the length of the queue at the node grows indefinitely and hence, the end to end solution for congestion control becomes very complex .The longer the path a packet has to travel, the higher its possibility of becoming lost due to collision or congestion, especially under high traffic [16]. Congestion control regulates the source rates to avoid overwhelming link capacity [17].

E. *Delay jitter:* jitter is the delay variation and is introduce by variable delay of packet over the network. This occurs because of router internal queues behavior or congestion. This parameter can seriously affect the quality of audio and video during video conferencing. To take care of delay jitter, packets should be collected and hold for long until the slowest packet arrives in time, rearranging them to flow in the correct sequence. Jitter buffer can be observing during video conferencing and it can be avoided so that continuous sending of packet during video conferencing can be possible [18].

IV. Summary of Findings

The study reveals that loss, delay, congestion and rejection of packets of packets have the following effects during on the quality of video and audio during video conferencing:

a. Packet loss introduces error in a reference frame which can be propagated to its succeeding frames and get amplified. It reduces the throughput of the network and increases latency due to additional time needed for retransmission of packet which incur additional cost of bandwidth.
b. Packet rejection occurs when queues are not effectively serviced in such a way that the amount of buffered packets is greater than the queue length. As a result of that it freezes video and audio at the receivers’ end during live video conferencing or streaming
c. Packet congestion results in voice quality degradation.
d. Packet delay causes lip synchronization between the video and audio during video conferencing as a result of the delay experience due to congestion.

IV. CONCLUSION AND FUTURE WORK

The key performance metrics that affect video conferencing over wireless network has being studied with the view of proffering situation to some of the problems investigated. This impairments has a severe effect on the quality of the signal been transmitted over the internet. Sometimes the video content freezes or is lost totally. Poor audio qualities are also encounter due to congestion and delay of packets which usually result to lip synchronization.A step toward addressing this problem will go a long way to encourage group communication over the internet. Future work should carry out detailed investigative study on all the performance metricsthat affects the quality of video and audio signal during video conferencing.

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ISSN (Print): 2320-9798

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