ISSN (Online) : 2277-5420

Evaluating the Performance of FQM Framework for Supporting Multimedia Applications in MANET

[1]Mohammed Saghir

[1]Computer Department, Hodeidah University,
Hodeidah, Yemen

Abstract

The different Quality of Service (QoS) requirements for multimedia applications should be taken into account while proposing QoS multicast frameworks for mobile ad hoc networks (MANETS). In addition, the application viewpoint should be taken into account while designing QoS multicast frameworks and the validation of new QoS multicast framework should be studied for different multimedia applications. In this paper, the performance of the QoS multicast framework (FQM) for supporting three different classes of multimedia applications is studied. The analysis of simulation results have shown that the performance of FQM for supporting the three classes of multimedia applications is quite the same. Although out of order packets and delay packets can be adapted and accepted in streaming stored applications, the enhanced in the performance of FQM for supporting this type of multimedia applications is not high. In addition, the results show that increasing the generated data packets as a result of decreasing payload size has a high effect on the performance of the FQM framework.

Keywords: *MANET, REAL-TIME, INTERACTIVE.*

1. Introduction

Wireless networking and multimedia applications are growing in importance rapidly. The motivation for supporting QoS multicasting in Mobile Ad hoc Networks (MANETs) is the fact that multimedia applications are becoming important for group communication. Among types of wireless networks, MANET provides flexible communication with low cost. All communications are done over wireless media without the help of wired base stations. The environment for MANETs is very volatile so connections can be dropped at any moment. Distant nodes communicate over multiple hops and therefore nodes must cooperate with each other to provide routing. The challenges in MANETs are attributed to mobility of intermediate nodes, absence of routing infrastructure, low bandwidth and computational capacity of mobile nodes.

While the bandwidth of wireless channels and the computational power of mobile devices are increased, real-time applications are expected to become available on wireless ad hoc networks in the near future. Real-time applications are fundamentally different from best-effort applications, since interactive real-time applications are delay and loss packet sensitive. The later real-time packets will be dropped while best-effort packets can be accepted. Therefore, the retransmission techniques are not generally applicable to real-time interactive applications, especially in multicast situations [1]. The multimedia applications can be classified into three general classes: The first class is real-time interactive audio/video which allows people to use audio/video to communicate with each other in real-time. The second class is one-to-many streaming of real-time audio and video which is similar to broadcast radio and television except that the transmission takes place on the internet. The third class is streaming stored audio and video where clients request on-demand compressed audio or video files which are stored on server [2].

Multicast is an efficient method to implement multipoint-to-multipoint communication. The multipoint-to-multipoint multimedia communication is required in most of real-time multicast applications as video conferencing, database management, distributed computation, real-time workgroup activities, emergency operations, disaster relief operations and tactical military networks [3]. In multicast streaming applications, the video server needs to transmit a single video stream for the multicast group, regardless of the number of clients that will view it [4]. Different multimedia applications have different QoS requirements resulting in real-time multimedia applications such as video and audio which are delay-sensitive but capable of tolerating a certain degree of errors. Because of this, the application viewpoint should be taken into account while designing QoS multicast protocols and their admission controls [3]. Furthermore, non-real time media such as web data is less delay-sensitive but requires reliable transmission [5].

This paper is structured as follows: Section 2 gives an overview on the previous work whereas Section 3 describes the QoS multicast framework FQM and defines the three main classes of multimedia applications. In Section 4, the simulation results of implementing FQM with the three main classes of multimedia applications are presented. Finally, Section 5 provides the conclusions of this study and gives some suggestions for future work.

2. Literature Review

The test-bed architecture (CASTADIVA) was used to study the performance of real-time videoconferencing in ad hoc networks through emulation. A video call is tested using OLSR protocol in different scenarios, varying the number of hops between the caller and the receiver. The results show that in an ad hoc network with a large number of hops, the quality of video calls suffers a significant degradation even in the absence of mobility [6].

The application–level QoS of live audio and video multicasting in a wireless ad hoc network is assessed in [7]. The study focuses on the quality of media synchronization as the major part of the application–level QoS. The master–slave destination scheme, the synchronization maestro scheme, and the distributed control scheme are used to study the quality of media synchronization. The study concludes that the master–slave destination scheme provides higher quality of application–level QoS and can be used as a first choice; however, the QoS of this scheme is sensitive to the location of the master destination. The synchronization maestro scheme can be the second best choice when appropriate destination for the master is difficult to be founded.

The mesh-evolving ad hoc QoS multicast (MAQM) routing protocol is developed to address the resource efficiency and QoS problems. In MAQM, the availability of resources for each node within its neighborhood is tracked and the QoS status is monitored continuously and announced periodically [8].

The Network Simulator (NS) does not directly support packet telephony and therefore, a complete system to implement packet telephony in a mobile ad hoc network is proposed. A new procedure was specified to perform simulation of packet telephony in network simulator. The overall system comprises of five layers including application, transport, network, data link and physical layers [9].

3. The FQM with the Three Classes of Multimedia Applications

In this section, an overview on the FQM QoS multicast framework is given and the three classes of multimedia applications are described.

3.1 The FQM QoS Multicast Framework

Multicast routing is more efficient in MANETs because it is inherently ready for multicast due to their broadcast nature that avoids duplicate transmission. Packets are only multiplexed when it is necessary to reach two or more destinations on disjoint paths. This advantage conserves bandwidth and network resources [10]. A cross-layer framework FQM is proposed to support QoS multicast applications for MANETs [11]. The FQM framework consists of five components. The first component of the framework is a new and efficient QoS multicast routing protocol (QMR) which is used to find and maintain the paths that meet the QoS requirements. The second component is a distributed admission control which used to prevent nodes from being overloaded by rejecting the request for new flows that will affect the ongoing flows. The third component is an efficient way to estimate the available bandwidth and provides the information of the available bandwidth for other QoS schemes. The fourth component is a source based admission control witch used to prevent new sources from affecting the ongoing sources if there is not enough available bandwidth for sending to the all members in the multicast group. The fifth component is a cross-layer design with many QoS scheme: classifier, shaper, dynamic rate control and priority queue. These schemes work together to support real-time applications.

The traffic is classified and processed based on its priority; therefore, control packets and real-time packets will bypass the shaper and sent directly to the interface queue at MAC layer. The best-effort packets should be regulated based on the dynamic rate control. In terms of queue priority, control packets and real-time packets have higher priority than best-effort packets. All components in the framework are cooperating to provide the required level of services.

3.2 The Real-Time Interactive Audio and Video

The real-time interactive audio and video allow people to communicate with each other in real-time. This class of application is also called many-to-many real-time applications such as video conferencing, database management, distributed computation and real-time workgroup activities. The audio and video applications are very sensitive for packet delay and jitter; late and out-of-order packets are discarded. During group meeting through video conferencing, users may need to open window for each user which add additional need for bandwidth in addition to the main speaker [2].

3.3 The One-to-Many Streaming of Audio and Video

The real-time non-interactive applications are similar to broadcast radio and television except that transmission on the internet. These applications allow user to receive radio or TV from any place in the world. This class of application is also called non-interactive real-time applications because a client cannot control a server's transmission schedule. The non-interactive application can be divided to non-interactive cognitive as remote lecturing situations where listeners may not have an opportunity to interact with the lecturer as a normal conversation and non-interactive social which include situations where only one person speak[15]. Requirements for packet delay and jitter are not as real-time interactive applications as audio conversation and video conferencing [2].

3.4 The Streaming Stored Audio and Video

The streaming stored audio and video applications where clients request on-demand compressed audio or video files which are stored on server. For audio, these files can contain audios of professor lectures and historical archival recording. For video, these files can contain videos of professor lectures, full-length movies, video archive of historical events and music video clips. At any time, a client can request an audio or video file from a server. After delay of few seconds the client begins to playback the audio or video file while it continues to receive the file from the server. Streaming is the fetcher of playing back audio or video while the file is being received. This class of multimedia also called audio and video on demand. The requirements for packet delay and jitter in streaming stored applications are not as real-time interactive applications or one-to-many streaming applications.

Streaming stored applications are not delay sensitive because video can take several seconds before playing and also are largely not jitter sensitive because jitter can be smoothed out by application buffering [2]. In addition, video streaming might contain valuable content, such as e-learning applications or multicast company meetings and in which case it requires service guarantees [12]. In conventional video streaming systems, audio and video are played back after buffering at the client side while data transmission rate is adjusted at the transmission side in response to the reception status on the receiver side [17].

4. Performance Evaluation

In this section, the performance of FQM framework for supporting different classes of multimedia applications is studied using GLOMOSIM [13]. Several scenarios are used for real-time interactive audio and video, one-to-many streaming audio and video and streaming stored audio and video applications. The experiments were conducted under general QoS requirements and basic characteristics for different multimedia applications instead of implementing specific codec and characteristics for specific multimedia applications. However, this simplistic assumption allows focusing experiments on the effect of constructing paths for different sessions using different numbers of sources and different interactive durations of each turn in a session. This is because QoS routing is the main issues for FQM framework.

This simulation was run using a MANET with fixed number of nodes moving over a rectangular 1000 m × 1000 m area for over 900 seconds of simulation time. Nodes in simulation moved according to the Random Waypoint mobility model provided by GLOMOSIM. Mobility speed was ranged from 0-20 m/s and the pause time was 0 s. Many sources are used for constructing different sessions which represent different multimedia applications. The radio transmission range was 250 M and the channel capacity was 2Mbit/s. Each data point in this simulation represents the average result of ten runs with different initial seeds. The analysis of simulation results focused on the comparing between the performances of FQM for supporting the three classes of multimedia applications. The study of the performance of FQM already has done in the previous study [11].

The performance of FQM for supporting the three classes of multimedia applications is studied through the following performance metrics:

- *Packet delivery ratio:* the average of the ratio between the number of data packets received and the number of data packets that should have been received at each destination. This metric indicates the reliability of the proposed framework.

- *The Control overhead:* the number of transmitted control packet (request, reply, acknowledgment) per data packet delivered. Control packets are counted at each hop. The available bandwidth in MANETs is limited so it is very sensitive to the control overhead.
- *Average latency:* the average end-to-end delivery delay is computed by subtracting packet generation time at the source node from the packet arrival time at each destination. The multimedia applications are very sensitive to the packet delay; if the packet takes long time to arrive at destinations, it will be useless.
- *Jitter:* the variation in the latency of received packets. It is determined by calculating the standard deviation of the latency [14]. This is an important metric for multimedia applications and should be kept to a minimum value.
- *Group Reliability:* the ratio of number of packets received at 95% of destination and number of packets should be received. This means that the packet is considered to be received only if it is received by 95% of the number of multicast group.
- *OUT of order Packets:* the average of the ratio between the number of data packets received and the number of data packets that received out of ordered at each destination. A packet is considered out of ordered if the sequence number is smaller than the sequence number of the previous packet received. This metric reflect the effect of out of order packets on the real-time applications that drop out of order data packets.

4.1 The Performance of FQM under Different Interactive Durations of Each Turn in a session

In this section, the effect of using different interactive durations of each turn in a session on the performance of FQM framework is studied. Different real-time interactive applications use different interactive durations. These applications include audio conversation, video conferencing and real-time workgroup activities.

The audio conversation and video conferencing may use short interactive durations than other real-time interactive applications. Actually, these applications often involve a conversational style of interaction. Furthermore, interactive applications vary in their interactivity with some such as group discussions are much more conversational whereas other interactive applications such as presentations or lectures mainly involving one speaker talking in monologue with few or no changes of speaker or instances of overlapping speech [15].

In many-to-many real-time interactive scenarios, different number of sources are used for constructing many-to-many real-time interactive session. The interactive durations of each turn in a session ranges from 10s to 160s, the payload size is 800 bytes and the traffic rate is 118kbps. Each source needs to constructs paths to the multicast group.

The interactive durations of each turn in a session changed to study the effect of long and short interactive durations on the performance of FQM while supporting these types of applications.

4.1.1 Packet Delivery Ratio (PDR)

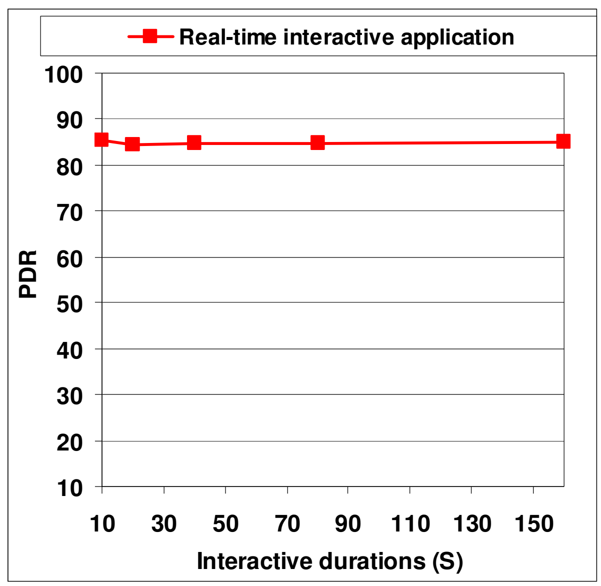

Fig.1 Performance of PDR vs. interactive durations of each turn in a session

The PDR as a function of different interactive durations of each turn in a session is given in Figure1. The PDR increased slightly when interactive durations increased.

Each source updates its forward nodes every 3s, so increasing interactive duration does not have a large effect on the PDR as shown in Figure 1. The slightly increase in PDR can be attributed to the paths stability.

4.1.2 Control Overhead (OH)

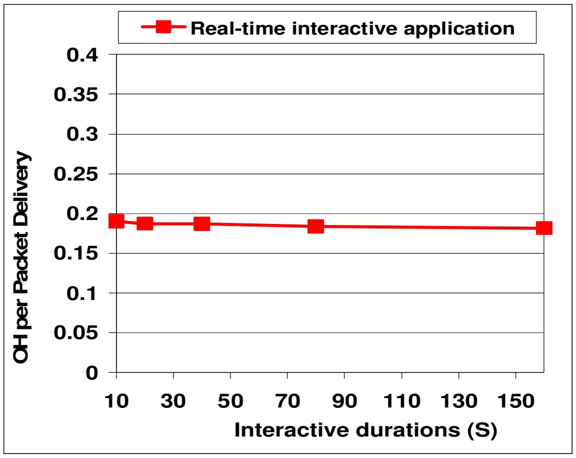

Fig.2 Performance of OH vs. interactive durations of each turn in a session

The OH as a function of different interactive durations of each turn in a session is given in Figure 2. Increasing interactive duration does not have a high effect on the control OH as shown in Figure 2. This is because each source updates its forward nodes every 3s and there is no additional overhead while increasing interactive duration.

4.1.3 Average Latency (AL)

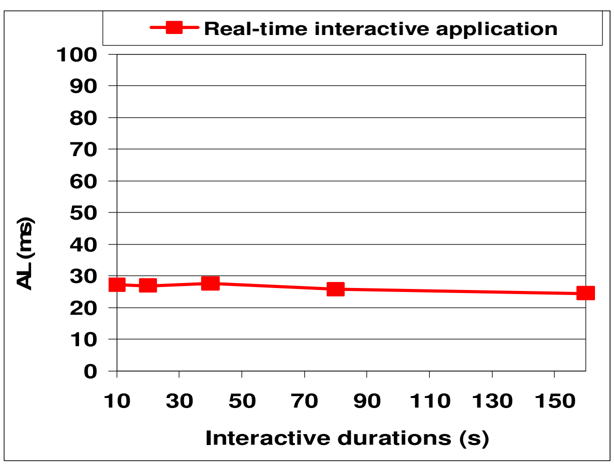

Fig.3 Performance of AL vs. interactive durations of each turn in a session

The AL as a function of different interactive durations for each turn in a session is given in Figure 3. The AL slightly decreased while interactive durations increased. This is because, constructing forward nodes for source at the beginning of each short interactive duration takes longer time more than updating forward nodes for the same source when interactive duration is long. When duration increases, time for constructing forward nodes decreases.

4.1.4 Jitter

Figure 4 reflects the jitter as a function of different interactive durations of each turn in a session. Increasing interactive durations has small effect on the jitter. For long interactive duration, some forward nodes will be congested and as a result some data packets will take long time to arrive at destinations while others arrive through different forward nodes with short time. The differences between packets delays increase resulting in an increase in the jitter.

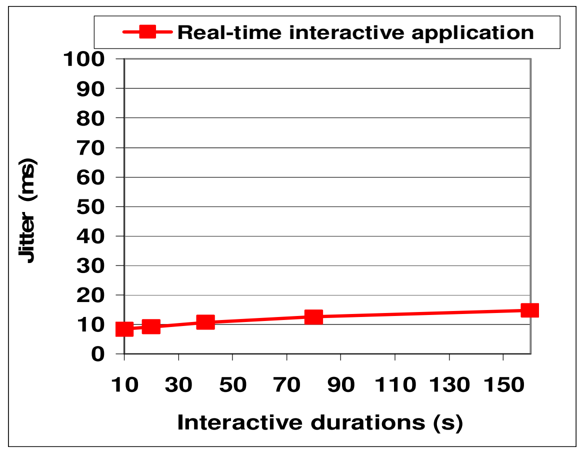

Fig.4 Performance of Jitter vs. session interactive durations of each turn in a session

4.1.5 Group Reliability (GR)

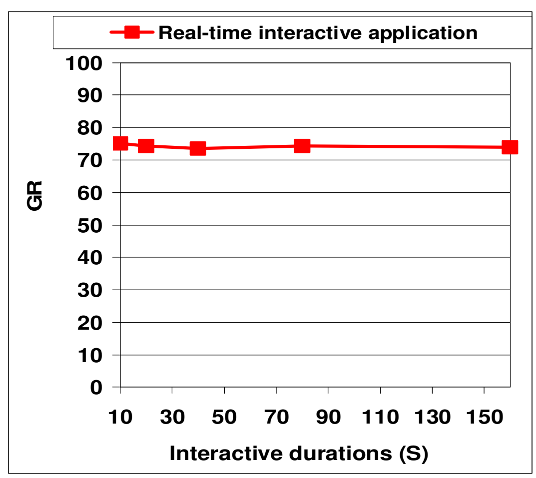

Fig.5 Performance of GR vs. interactive durations of each turn in a session

The GR as a function of different interactive durations of each turn in a session is given in Figure 5. Increasing interactive durations does not have a high effect on GR.

With long interactive duration, data packets may have high chance to arrive at destinations and as a result GR slightly increases.

4.2 The Performance of FQM under Different Form of Real-Time Interactive Applications

In this section, the effect of using different payload size for different form of real-time interactive applications audio or video on the performance of the FQM framework is studied. For audio conversation scenario, two sources were used to multicast 128kbps data traffic rate (64kbps for each source) with 200 byte payload size. In addition, for video conferencing scenario, one source was used to multicast 128kbps data traffic rate with 800 byte payload size. For the same data traffic rate, when payload size is small, the number of packets will be higher than number of packets when payload size is large. Multicast is usually related to real-time applications such as audio and video conferencing. This is reflected in the universal use of Constant Bit Rate traffic generators (CBR). However, the large packet size more closely reflects packet sizes used in video and not in audio [3]. In good channel conditions, a higher number of voice users can be supported by using large payload size [16].

4.2.1 Packet Delivery Ratio (PDR)

The PDR as a function of audio and video is given in Figure 6. The PDR for video application is higher than PDR for audio application. This is because the number of data packets for audio application is higher than the number of data packets for video application and as a result, the forward nodes for audio application are congested and some data packets are dropped. In addition, the dropped data packets while constructing new paths increases.

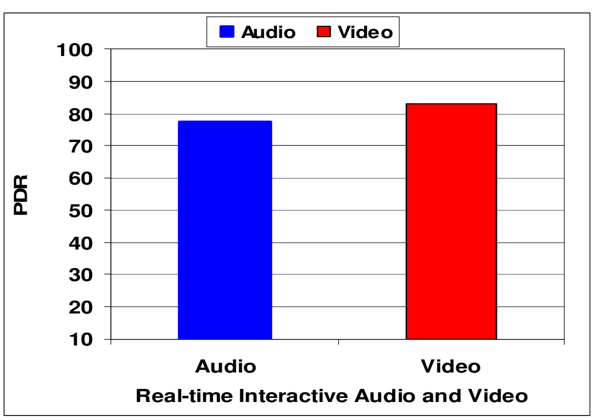

Fig.6 Performance of PDR vs. audio and video applications

4.2.2 Control Overhead (OH)

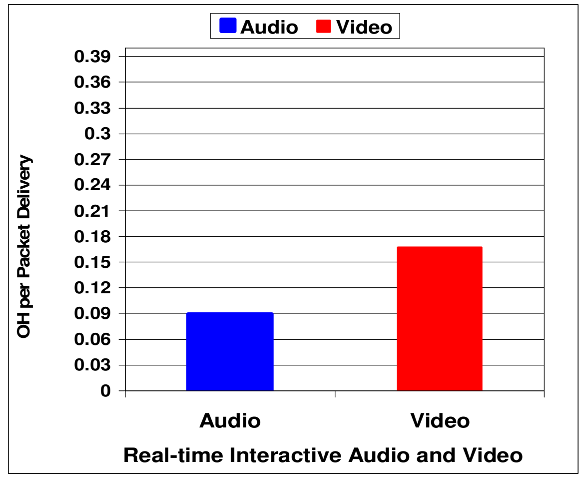

Fig.7 Performance of OH vs. audio and video applications

The OH as a function of audio and video applications is given in Figure 7 which shows that the average control overhead for video application is higher than average control overhead for audio application. This is because the number of data packets that generated for audio application is higher than the number of data packets that generated for video application. The number of control packet does not depend on payload size because each source updates its forward nodes every 3s.

4.2.3 Average Latency (AL)

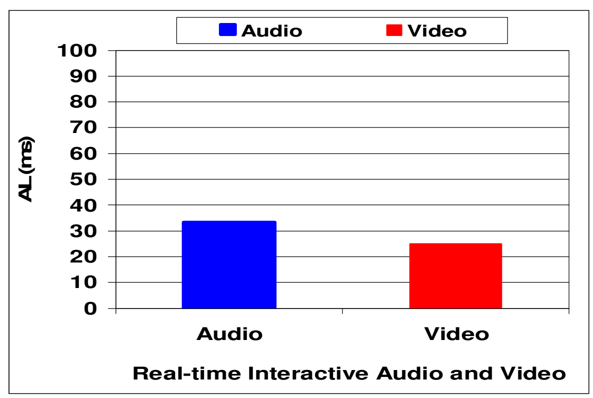

Fig.8 Performance of AL vs. audio and video applications

The AL as a function of audio and video applications is given in Figure 8. The AL for audio application is higher than AL for video application because the forward nodes in audio application are congested as a result of increasing the number of data packets which are generated for audio application as discussed in section 4.2.1. When forward nodes are congested, some data packets take long time to arrive at destinations and as a result the AL increases.

4.2.4 Jitter

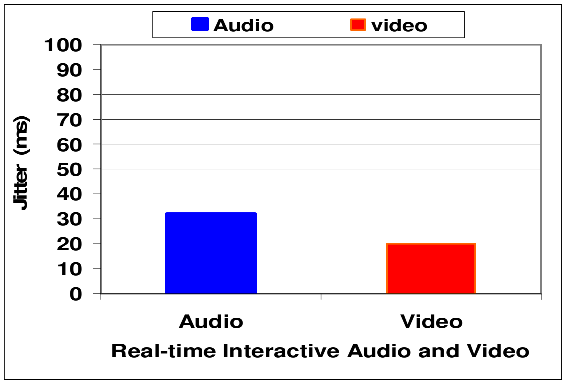

Fig.9 Performance of Jitter vs. audio and video applications

Figure 9 reflects the jitter as a function of audio and video applications. The figure reflects that jitter for audio application is higher than jitter for video application. This is because some forward nodes are congested and data packets that forwarded through these forward nodes will take long time to arrive at destinations as discussed in section 4.2.3. As a result of this, there is an increase in the differences between average latency for data packets and consequently there is an increase in jitter.

4.2.5 Group Reliability (GR)

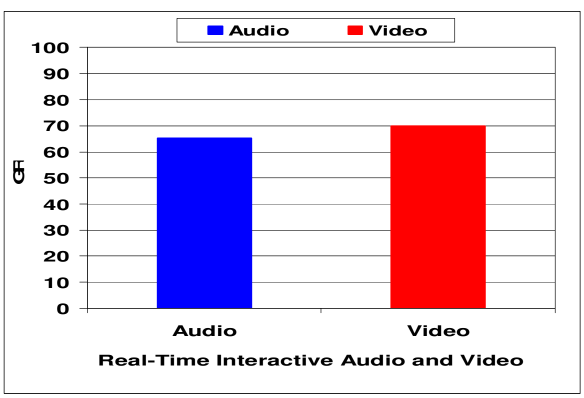

Fig.10 Performance of GR vs. audio and video applications

The GR as a function of audio and video applications is given in Figure 10. The figure shows that GR for video application is higher than GR for audio application. For video application, the forward nodes will be less congested as discussed in section 4.2.1 and as a result, data packets may have high chance to arrive at all destinations. Consequently, the GR for audio applications is less than the GR for video applications.

4.3 The Performance of FQM for Supporting Different Forms of Multimedia Applications

The multimedia applications have different QoS requirements and as a result, the performance of FQM for supporting the QoS requirements for each application should be studied. In this section, the performance of FQM for supporting three different forms of multimedia applications is studied. The three forms of multimedia applications are real-time interactive audio and video, one-to-many streaming of audio and video and streaming stored audio and video. The 256kbps is used with 800 bytes to reflect the effect of data traffic on the delay packets and out of order packets.

4.3.1 Packet Delivery Ratio (PDR)

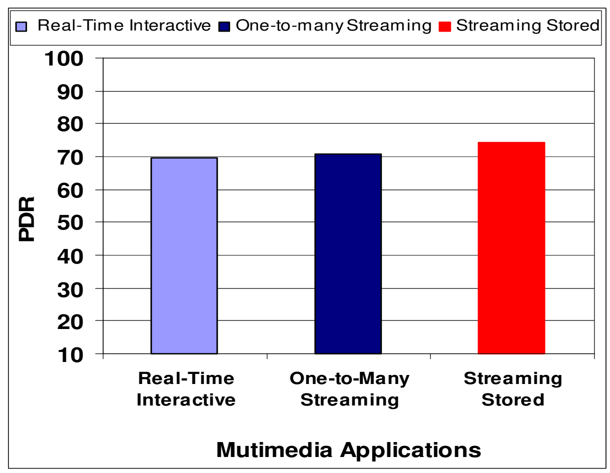

Fig.11 Performance of PDR vs. multimedia applications

The performance of PDR as a function of multimedia applications is given in Figure11. The PDR for real-time interactive applications and one-to-many streaming applications are quite same and less than PDR for streaming stored applications. The PDR for streaming stored applications is higher than PDR for real-time applications because the streaming stored applications are not sensitive for packet delay and out of order packets as real-time applications. The out of order data packets and delay packets can be accepted for streaming stored applications while dropped for real-time interactive applications.

4.3.2 Control Overhead (OH)

Figure 12 shows the Control OH as a function of multimedia applications. The average control overhead is quite same in the three multimedia applications and this is because FQM framework uses the same mechanism for updating forward nodes for the three multimedia applications. The small decrease in control OH for streaming stored applications is being because the PDR for streaming stored application is higher than PDR for one-to-many streaming application and real-time interactive application.

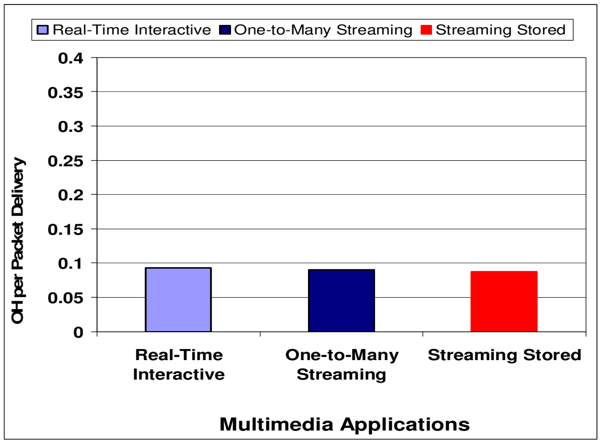

Fig.12 Performance of OH vs. multimedia applications

4.3.3 Average latency (AL)

The packet delay is not very sensitive for streaming stored applications as real-time interactive applications. Figure 13 shows the AL as a function of multimedia applications. The results show that AL for streaming stored application is higher than AL for real-time interactive applications and one-to-many application. Although AL is high, this AL value is still very low than limit of delay for data packets for streaming stored applications.

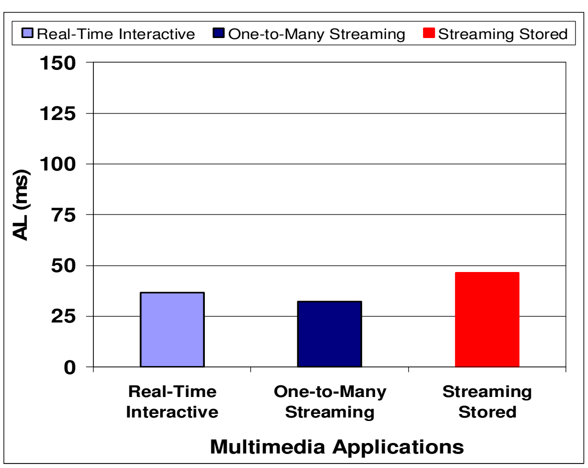

Fig.13 Performance of AL vs. multimedia applications

4.3.4 Jitter

Figure 14 gives an overview on the performance of jitter as a function of multimedia applications. The results show that the jitter of streaming stored applications is higher than jitter for real-time interactive applications and one-to-many streaming applications. This is because the AL for streaming stored applications is higher than the AL for real-time interactive applications and one-to-many applications as discussed in section 4.3.3 and some packets may take longer time than others to arrive at destinations. The jitter for streaming stored application can be adapted through buffering.

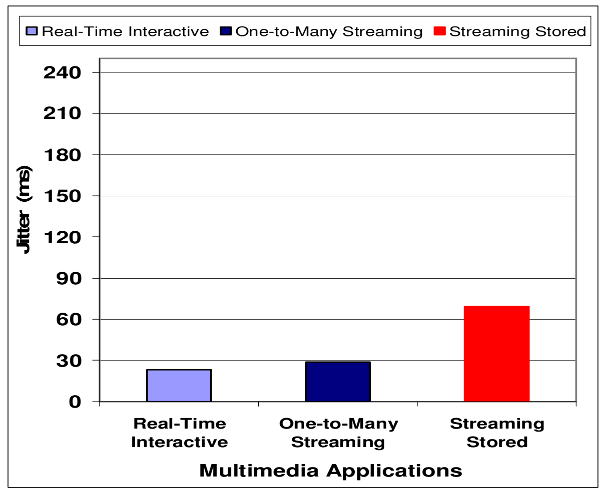

Fig. 14 Performance of jitter vs. multimedia applications

4.3.5 Group Reliability (GR)

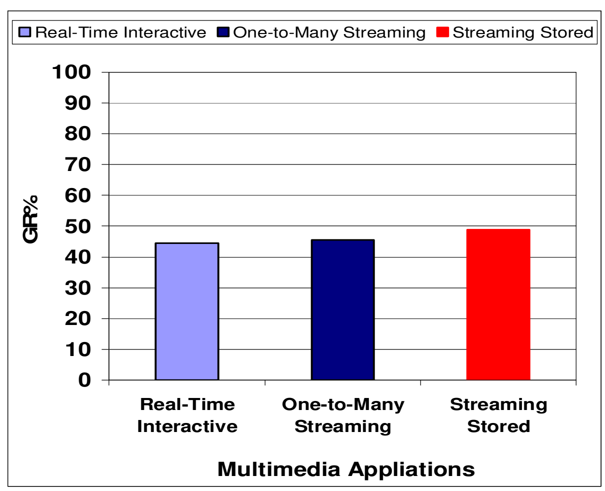

Fig. 15 Performance of GR vs. multimedia applications

The GR as a function of multimedia applications is given in Figure 15. The GR for streaming stored application is higher than GR for real-time interactive application and one-to-many application. This is because PDR in streaming stored application is higher than that in real-time interactive application and one-to-many application, as discussed in section 4.3.1.

4.3.6 Out of Order Packet (OUT)

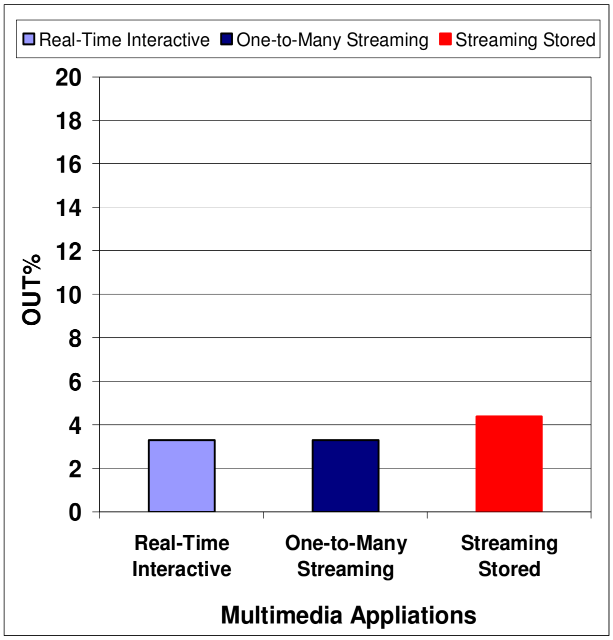

Fig. 16 Performance of OUT vs. multimedia applications

The OUT of order data packet as a function of multimedia applications is given in Figure 16. The out of order data packet for streaming stored application is not a big issue and can be adapted at destinations. The average of out of order packets for streaming stored applications is higher than the average of out of order packets for real-time interactive applications and one-to-many streaming applications. This is because streaming stored application accepts high delay packets at forward nodes while in real-time and one-to-many applications, high delay packets are dropped at forward nodes and this decrease the out of order packets that arrive at destinations.

5. Conclusion and Future Work

In this paper, the performance of FQM framework for supporting different classes of multimedia applications with different QoS requirements for each class is studied. The analysis of results shows that using long and short interactive duration of each turn in a session does not have high effect on the performance of the FQM framework. Using long interactive durations of each turn in a session can enhance the performance of FQM framework through decreasing average delay of packets. In addition, the results of experiments reflect that the performance of FQM for supporting multimedia applications with large payload size (800 byte) is better than supporting multimedia applications with small payload size (200 byte). Using large payload size enhances the performance of FQM framework through increases PDR and GR while decreases AL and jitter. Furthermore, the results show that increasing the generated data packets as a result of decreasing payload size has a high effect on the performance of the FQM framework. Finally, the analysis of results show that the performance of FQM framework for supporting real-time interactive applications, one-to-many applications and streaming stored applications are quite the same. Although out of order packets and delay packets can be adapted and accepted in streaming stored applications, the enhanced in the performance of FQM framework for supporting this type of applications especially in PDR and GR is not high. In future work, we intend to study the performance of the FQM QoS multicast framework under specific codec for each class of multimedia applications.

References

[1] W. Wei and A. Zakhor, “Multiple Tree Video Multicast over Wireless Ad Hoc Networks”, IEEE Transactions on Circuits and Systems for Video Technology, Vol. 17, No. 1, 2007, pp. 2-17.

[2] G. Aggelou,, Mobile ad hoc networks from wireless LANs to 4G Networks, USA: McGraw-Hill, 2005.

[3] V. Thi, L. Landmark and I. Kure, “A Survey of QoS Multicast in Ad Hoc Networks”, Future Internet, Vol. 2, No. 3, 2010, pp.388-416.

[4] E.D. Kanmani and N. Kasthuri, “QoS Enhancement using Efficient Routing Protocol for Video over Wireless Ad hoc Networks”, International Journal of Computer Applications, Vol. 7, No. 2, 2011, pp. 30-35.

[5] H. Jiang, W. Zhuang and X. Shen, “Cross-layer design for resource allocation in 3G wireless networks and beyond”, IEEE Communications Magazine, Vol. 43, No. 12, 2005, pp.120-126.

[6] J. Hortelano, J. Cano, C. M. Tavares and P. Manzoni, “Evaluating the Performance of Real Time Videoconferencing in Ad Hoc Networks Through Emulation”, in Workshop on Principles of Advanced and Distributed Simulation - PADS , 2008, pp. 119-126.

[7] T. Nunome and S. Tasaka, “Application–Level QoS Assessment of Continuous Media Multicasting in a Wireless Ad Hoc Network”, IEEE Communications Society, 2004, pp. 2047-2053.

[8] K. Bür and C. Erso, “Performance evaluation of a mesh-evolving quality-of-service-aware multicast routing protocol for mobile ad hoc networks”, Performance Evaluation, Vol. 66, 2009, pp. 701-721.

[9] P.K.Suri and S. Maan, "Simulation of Packet Telephony in Mobile Ad hoc Networks Using Network Simulator", International Journal of Advanced Computer Science and Applications (IJACSA), Vol. 2, No.1, 2011, pp. 87-92.

[10] M. Hasana and L. Hoda, "Multicast Routing in Mobile Ad Hoc Networks", Kluwer Academic Publishers, 2004.

[11] M. Saghir, T. C. Wan, and R. Budiarto, "A New Cross-Layer Framework for QoS Multicast Applications in Mobile Ad hoc Networks," International Journal of Computer Science and Network Security, (IJCSNS), Vol. 6, No.10, 2006, pp. 142-151.

[12] Tim Szigeti, Christina Hattingh, "Quality of Service Design Overview",Cisco Press, 2004, http://www.ciscopress.com/authors/bio.as http://pcl.cs.ucla.edu/projects/glomosim.

[14] K. Farkas, D. Budke, B. Plattner, O. Wellnitz, and L. Wolf, "QoS Extensions to Mobile Ad Hoc Routing Supporting Real-Time Applications", in the proceeding of the 4th ACS/IEEE International Conference on Computer Systems and Applications, 2006, pp.54-61.

[15] J. Mullin, L. Smallwoo, A. Watson and G. Wilson, "New techniques for assessing audio and video quality in real-time interactive communications", IHM-HCI, tutorial, 2001.

[16] D. P. Hole and F. A. Tobagi, "Capacity of an IEEE 802.11b wireless LAN supporting VoIP", in Proceedings of IEEE International Conference on Communications, 2004, pp. 196–201.

[17] H. Aoyama, A. Miyata, R. Ohgushi and H. Ishimaru, "latest technology for video-streaming gateway of M-stage V live-Assuring Video Quality and Usability", Technical Journal (NTTDoCoMo), Vol. 6 No. 4, 2005, pp. 13-18.

Mohammed Saghir received his B.S from Technology University, Iraq in 1998, M.Sc. from Al Al-Bayt University, Jordan in 2004; and his Ph.D from Universiti Sains Malaysia in 2008 in Computer Sciences. Currently he is working as a senior lecturer in Hodeidah University, Yemen. His current research interests include Mobile Ad hoc networks, QoS multicast routing in MANETs and WiMax.