Analysis of MPEG-4 Traffic Decodable Frames with Packet Loss Rate

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Abstract— The paper presents a comprehensive repository of information required to implement network for multimedia traffic in NS-2. The paper is arranged in such a way that it builds up the framework required to be used in NS-2 for successful implementation of Fraction of decodable frames and packet loss rate for multimedia traffic. In the paper demonstrates a work place for conducive experimentation of video transmission in 802.11in MANETs. The first set of results deal with the performance parameter Fraction of decodable frames. The next set of results is for the performance parameter is packet loss rate. In the paper the performance of video transmission is tested and its ruggedness over a small wireless network and all the simulation steps can be used as groundwork for performing much other experimentation on the same topic in future. The paper presents complete analysis of the video transmission using MPEG_4 for two performance parameters via Fraction of decodable frames and packet loss rate in a very novel manner. These results can be used as benchmarks and rules of thumb for performing further research and implementing networks.

Keywords: MPEG-4 Traffic Decodable Frames, Packet Loss Rate, Mobile Ad-hoc Networks

I. INTRODUCTION

Mobile Ad-hoc Networks are infrastructure-less networks, is a co-operative engagement of a collection of mobile hosts without the required intervention of any centralized access point. Because of Wireless Ad hoc Network's ease in configuration and quick deployment, such networks are widely preferred for instant data transmission for many applications including multimedia. A lot of work on the improvement of routing of data in network for different classes has been done; latest being the multimedia traffic class. The available bandwidth for Wireless Ad hoc Networks fails to meet the requirements of multimedia video data transmission and results in packets loss, delay as well as decreases the quality of transmitted multimedia data. So, to overcome Wireless Ad hoc Networks limitations for multimedia video (video considered in MPEG 4 format) transmission, a variant of Network Coding that is Random Linear Network Coding with Multi Generation Mixing is employed by network nodes. Now a day, networks are becoming more and more audio-visual so it becomes very important to study the analysis of multimedia in Mobile Ad-hoc Networks. The research discusses the implementation of fragmentation on the multimedia traffic class, i.e. MPEG-4 in Mobile Ad-hoc Networks [8] [7] [13]. In recent years digital video is replaced analog video in many applications Examples in digital television, replacement of analog video cassettes by DVD to watch movies. The ever-increasing demand for video transmission motivates research on how to provide better-delivered video quality. The MPEG-4 standard provides key technologies that will enable much functionality with benefits such as improved quality and reliability. Previous studies use various real video traces to evaluate their proposed network mechanisms in simulation environment. Some researchers evaluate the different parameters like throughput, peak signal noise ratio (PSNR) etc. In today era, as in higher bandwidth network MTU (Maximum Transmission Unit) has increasing significantly. To study the effect of larger fragment size and thus smaller fragmentation on a video trace files becomes an important issue. In the proposed research work the multimedia traffic carried over the UDP is tested with different fragment sizes for the following parameters- fraction of decodable frames, Average end-to-end delay, Packet Loss rate, Throughput, Average PSNR and comparison between these results. MANET is a collection of mobile nodes without the required intervention of any centralized access point [5].It is a temporarily formed network which is created, operated and managed by nodes themselves. In MANETs nodes are wireless and battery powered [12]. A MANET can be considered as an autonomic system as they are self-configure, self –heal, self-organize, and self-protect. No fixed routers are available in these networks. Internet connectivity would benefit users from mobility offered by Mobile Ad hoc networks and connectivity provided by the Internet [9]. A MANET can be a standalone network or attached to a larger network, including the Internet. In these type of networks all nodes can freely communicate with every another node and nodes are independent to each other. Examples of a network are a P2P and multi-hop connected network. These networks have various advantages in terms of self-reconfiguration and adaptability to highly variable mobile characteristics like the transmission conditions [16]. The nodes in this network behave as routers which discover and maintain routes to other nodes in the network. Creation of routes depends only on nodes which forwards traffic on behalf of other nodes

II. RELATED WORK

The author describes the characteristics of mobile wireless networks and related these characteristics to the requirements of video transmission. The tests and simulations analyzed were designed to correlate objective video quality metrics with subjective video quality. In order to evaluate objective quality factors (bit-rate, link BW, propagation delay) Standard objective metrics such as PSNR were taken into consideration to specify effective subjective tests. The methodology SAMVIQ was used for subjective evaluations. From the simulation results we can conclude that the video quality depends on the percentage of lost frames as well as the end-to-and delay is related through PSNR values. When the percentage of lost frames will be higher the PSNR values will be lower. In this paper a parameter jitter is also considered that is an important factor that influences the video quality particularly if a video decoder does not provide buffering operation. They realize that the end to end delay does not play an essential role in the objective video quality which is a critical factor for real-time services and may influence the subjective video quality [15].The video quality is evaluated and attained in streaming MPEG-4 video over UMTS networks using an integrated tool environment. This tool environment basically consists an MPEG-4 encoder/decoder. A network simulator and video quality evaluation tools like evaluate trace program, PSNR program and MOS program. An encoded video frame is carried by UDP over the IP protocol. If the video frame size is larger than the UDP segment size of 520 bytes fragmentation will take place. This paper evaluated the perceived video quality in terms of MOS, cumulative jitter and video frame error rate. Two video clips with different features were considered. The Simulation results show that in RLC acknowledged mode the perceived video quality achieved using UMTS is better to unacknowledged mode. In unacknowledged mode there is no error recovery scheme as in the acknowledged mode case so the performance results are worst. The block error rates up to 30% can be tolerated the acknowledged mode before showing any sign of degradation in video quality due to the increase in cumulative jitter. A self-adaptive RLC acknowledged mode is proposed, based on the simulation results which is aware of MPEG-4 frames, playback buffer size and UMTS radio interface round-trip delay [11]. The packet loss effect is studied on MPEG video transmission quality in wireless networks. The distribution of packet losses in wireless network, and other packet loss issues was considered which would affect the video transmission quality. For simulation of wireless transmission losses, they adopted random uniform error model and Gilbert-Elliot (GE) error model. The random uniform model provides the distributed losses, and the GE model provides the burst losses. In multimedia communication, to set the play-out buffer at the receiving end is to reduce the jitter effect and smooth out the video played by the user. In this paper, they also evaluated the effect of the packet size on the video transmission quality. An evaluation metric, known as Decodable Frame Rate (Q), is adopted. They evaluate the effect of the size of the play-out buffer on the video transmission quality which causes the additional packet drops at the video receiver. They used different packet sizes for video transmission to evaluate the video transmission quality. The experiments are simulated on Network Simulator (NS) Version 2 pattern. The results show that the effect of burst packet losses on the video delivered quality is less than distributed packet losses in the same packet loss rate. The video quality of simulation is better than analytical model. When the play-out size increases the video quality decreased. In this experiment, they evaluate the effect of the transmission packet size on video quality. They set the maximum packet size as 500, 1000, and 1500 bytes. The packet error rate is set in the range of 0.02 to 0.2 with 0.02 intervals. The smaller maximum packet size results in the worse video transmission quality. If there was no recovery protection for video transmission, such as FEC, the video delivered quality of the larger packet size is better than smaller packet [6] The analysis of the three routing protocols AODV (Ad-hoc on demand distance vector), OLSR (Optimized link state routing) and ZRP (Zone routing protocol) is performed and compared by using proposed technologies (QualNet simulator) on the basis of performance metrics such as PDR (packet delivery ratio), end-to-end delay and throughput. Simultaneously, evaluation is done by varying network size and traffic load. The focus was concentrated on three performance metrics for two different levels of node density like low node density and high node density. When the network size & traffic load increases the effect was analyzed on these parameters Traffic source used was CBR (Constant Bit Rate). The packet size was 32 bytes and sending rate was set to 1 packet per second. The Two-Ray Propagation Model and IEEE 802.11 MAC layer protocol were used. The model chosen for conducting results was Random Waypoint Mobility Model. The simulation results shows that by increasing density of the network, the End-to-end delay for all the three protocols falls down. Also the throughput in a high density network improved for all the three routing protocols as compared to low density network. AODV seems to be a good performer up to network size of 200 nodes in both low and high dense networks. It has relatively higher packet delivery ratio and throughput, and low end-to-end delay [1]. The simulated RF packet delay and loss scenarios on MPEG-4 transmissions and described the detailed introduction of UMTS (Universal Mobile Telecommunication System) and also MPEG- 4 Standard. The Integrated Tool Environment was used in this paper, utilized to the Video Acquisition, encoding. In this paper researcher assumes that the packet transmission protocol used UDP (User Datagram Protocol). For a UDP packet a video frame is relatively bigger than the maximum allowable packet size or Maximum Transfer Unit (MTU). Researchers also points out that that a Frame size can be much larger than the MTU [10].

III. METHODOLOGY

A. General Research Methodology

In this proposed approach to MPEG analysis, the first step is to integrate Evalvid with NS-2. Next step involves the selection of the simulation environment/Scenario, as two scenarios have been proposed in this research work. After that the next step is to define the network parameters and evaluate the network performance on the basis of selected network performance parameters. Lastly, the comparative analysis is performed on the basis of the different values of the network parameter The methodology of this project lies around checking the fraction of decodable frames (Q) and the peak signal to noise ratio (PSNR) of the wireless network transmitting multimedia data (MPEG file) by varying the network parameter like bandwidth, fragment size, and channel error rate (CER) which can be easily varied within the networking. EvalVid is an open-source project, and supports trace file generation of MPEG-4 as well as H.263 and H.264 video. Using it together with the ns-2 interfacing perceived quality and objective measure like PSNR calculation can be obtained after network simulation. But still, this does not provide a solution for rate adaptive video investigation. The steps are illustrated in figure 3.1.

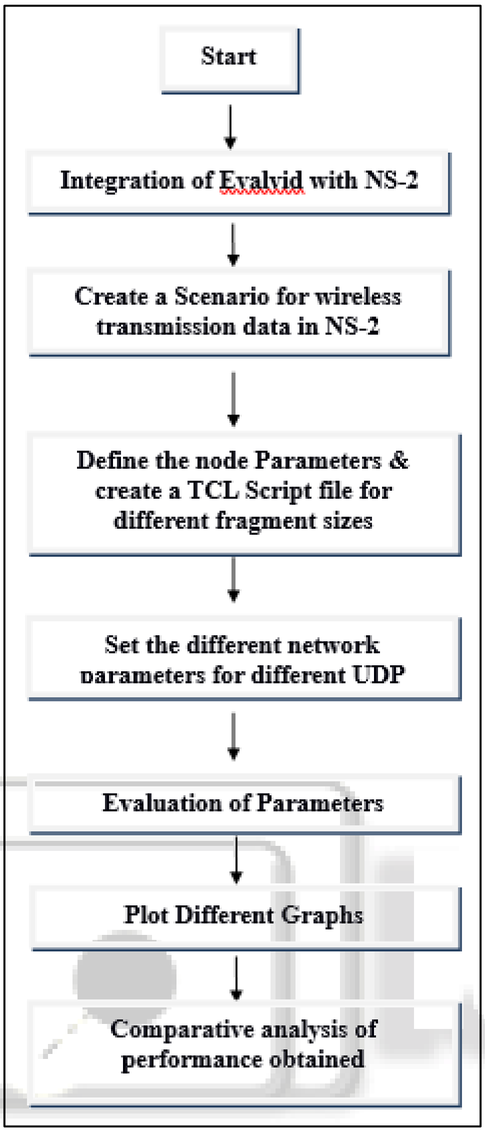

Fig. 3.1: Flow chart of research Methodology

B. Selection of Network and performance parameters

In this research work the analysis of MPEG traffic in mobile ad-hoc networks has been done on the basis of four network parameters. These four network parameters are varied and their effect on the MPEG performance is evaluated. The Three parameters that are used in this work are: Bandwidth, Delay, and Channel Error Rate

1) Bandwidth It decides how much amount of traffic can be sent over a wireless connection at specific time. Some connections have highest bandwidth like Ethernet connection in which bandwidth is 1000Mbps, while some connections have smallest bandwidth that act as a bottleneck for serving the packets from source to destination.
2) Delay Delay is an important parameter which needs to be minimizing in the case of videos that affect the quality of video. It decides how much time will take by packet to route from source to destination.
3) Channel error rate Different communication channels like Ethernet cables, coaxial cables, serial cables and fiber optical cables react differently to noise, fading, distortion, EMI and synchronization etc. Channel's inherent response to factors causing errors is known as Channel error rate. Specifying error in simulation is done through commands as the environment is virtual.
4) The most important network performance parameters that determine the performance of the network are: Packet loss rate, Throughput, Average end to end delay and Average PSNR.
5) Packet Loss Rate: Packet loss is the failure of one or more transmitted packets to arrive at their destination. In digital communication, Packet loss is distinguished as one of the three main error types, the other two being bit error and spurious packets caused due to noise[9] A problem of packet loss is much more complicated in Mobile Ad hoc Networks because wireless links are subject to transmission errors and the network topology changes dynamically[14].
6) Throughput: Throughput or network throughput is number of bits delivered successfully per second to the destination. It is the sum of bits received successfully by all destinations. This data may be delivered over a physical or logical link. The throughput is measured in bits per second (bps) [13]. Throughput = Total number of delivered packets / Total simulation time [4].

C. Proposed Model

1) UDP and TCP traffic classes along with packet loss rate, throughput and delay Scenario

Under this scenario, a network topology has been proposed in which both the UDP and TCP traffic classes are considered at the source nodes and at the sink node the UDP traffic class is evaluated on the basis of three parameters namely average/peak end-to-end delay, packet loss rate and throughput. The network has 2 source nodes (S1 and S2), one generating TCP traffic which is attached to FTP agent and the other generating UDP traffic connected to CBR (Constant bit rate) be transferred at the destination node (d). The three routers are set in between source nodes and destination node. All the three parameters bandwidth, error rate, fragment sizes are varied on the link between r1 and r2 by keeping delay constant as shown in Fig. 3.3. The destination node (d) is connected to two receiving nodes D1 and D2 in which sink agent that is used to record the information for TCP-based application at the receiver side attached to D1 and other is null agent that is used to record the information for UDP-based application at the receiver side attached to D2.

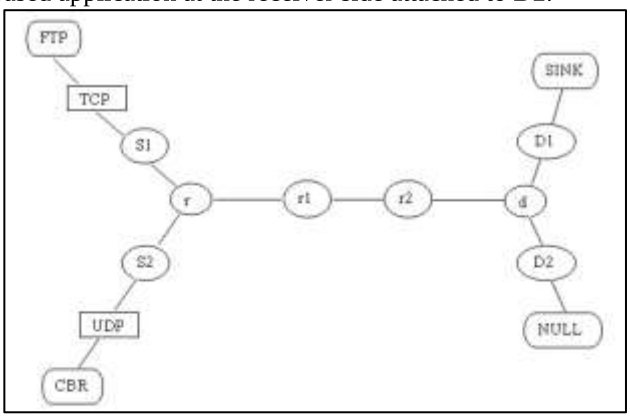

Fig. 3.3: Simulation Environment under First Scenario [2] [3]

The variations in four network parameters are considered in this scenario. In first case, the bandwidth of the main router to router link (r1-r2) is changed keeping the other network parameters unchanged. In the second case, the delay of the main router to router link (r1-r2) is changed keeping the other network parameters unchanged. In the third case, the error rate of the main link is changed and in the fourth case, the packet size is changed.

2) UDP traffic class along with average PSNR and Fraction of decodable frames scenario

Under this scenario, a network topology has been proposed in which only the UDP traffic class is considered at the source node and at the sink node the performance is evaluated on the basis of two parameters namely: average PSNR and fraction of decodable frames. The variations in two network parameters are considered in this scenario. Under this scenario, the Queue algorithm in the proposed network topology is changed four times keeping the fragment size unchanged. This process is repeated three times at the three different values of fragment size for comparative analysis.

The network has one source node (S1) for generating UDP traffic. The two routers (r1 and r2) are set in between source node (s1) and destination node (d1). The destination node (d1) is connected to null agent that is used to record the information for UDP-based application. The total size of the MPEG file is sent approximately 50Mb. The simulation setup is designed to analyze the performance metric-"fraction of decodable frames (Q)" and "PSNR". The data transfer takes place from node-1(source) to node-2(sink) in packets. The number of packets depends upon the frame size. If the frame size is small the number of packets is large i.e. fragmentation is more, if the frame size is large the number of packets will be small i.e. the fragmentation is less. The analysis reveals that upon increasing the number of frame size for constant channel error rate the fraction of decodable frames (Q) will be increases. We have tested the MPEG-4 Traffic on the following parameter:

Bandwidth	Fragment Size	Channel Rate Error
0.5 Mb	256 bytes	0.1
1.0 Mb	512 bytes	0.3
2.0 Mb	1024 bytes	0.6

Table 3.1: Different parameter for testing MPEG-4

IV. RESULT ANALYSIS

The simulations were conducted using NS2 and QoS assessment framework for video traffic Enabled by the new tool-set that combines EvalVid and NS2. To evaluate the quality of video stream, the fraction of decodable frames (Q) has to identify. The value of fraction of decodable frames (Q) lies between 0 and 1.0. The larger the value of Q, the better the video quality received by the end user. Where Q is defined as the fraction of decodable frame rate, which is the number of decodable frames over the total number of frames sent by a video source. As discussed earlier the standard MPEG encoders generate three distinct types of frames, namely I, P, and B frames. Due to the hierarchical structure of MPEG, I frames are more important than P frames, and in turn P frames are more important than B frames. Therefore, a frame is considered decodable if, and only if, all the fragmented packets of this frame and the other packets that this frame depends on are completely received. Thus, the decodable frame rate (Q) is defined as the number of decodable frames over the total number of frames sent by a video source. The larger the Q value, the better the video quality perceived by the end user.

A. Results for Packet loss rate

1) Total Packet loss:

Packet loss is the failure of one or more transmitted packets to arrive at their destination. In digital communication, Packet loss is distinguished as one of the three main error types, the other two being bit error and spurious packets caused due to noise. A problem of packet loss is much more complicated in Mobile Ad hoc Networks because wireless links are subject to transmission errors and the network topology changes dynamically.

$$PacketLoss\ \ Rate = \left(\frac{no.ofpacketst\ \ ransmitted\ \ - no.ofpacketsr\ \ eceived}{no.ofpacketst\ \ ransmitted} \right) * 100$$

B. Parameters for Fraction of Decodable Frames CER.

The fraction of decidable frames for different fragment sizes over varying channel error rate (CER)

Was computed and following observations were made.

1) The max. Value 0.998722 was recorded with fragment size 1024 over channel error rate of 0.1.
2) The min. value 0.113825 was observed with fragment size 256 over channel error rate of 0.1 as shown in table 4.1.

Fraction of Decodable Frames (Q)			
Fragment Size(kb)	256	512	1024
Channel Error Rate			
0.1	0.996544	0.998722	0.998722
0.3	0.849941	0.893575	0.903575
0.6	0.113825	0.194104	0.237004

Table 4.1: Fraction of Decodable Frames for various CER

1) Parameters for packet loss at fragment size 256 at various delay

The max packet was rate at fragment 256kb was observed when channel error rate was 0.6 and bandwidth was set to 0.3 while min packet rate was observed when channel error rate was 0.2 and bandwidth was 0.2 over varying delay as shown in table 4.2, table 4.3 and Table 4.4.

PACKET LOSS RATE-FRAG 256 at 10ms delay			
Error Rate B.W	0.2	0.4	0.6
0.3	49.90%	45.53%	61.80%
0.9	21.40%	39.40%	61.71%
1.5	19.60%	39.77%	61.71%

Table 4.2: Packet Loss Rate at Fragment size 256 at 10ms delays

PACKET LOSS RATE-FRAG 256 at 50ms delay			
Error Rate B.W	0.2	0.4	0.6
0.3	45.72%	42.28%	61.89%
0.9	19.23%	39.68%	61.80%
1.5	19.14%	39.59%	61.71%

Table 4.3: Packet Loss Rate at Fragment size 256 at 50ms delays

PACKET LOSS RATE-FRAG 256-100ms			
Error Rate	0.2	0.4	0.6

B.W			
0.3	43.86%	41.17%	61.71%
0.9	19.23%	39.77%	62.08%
1.5	19.60%	39.59%	61.89%

Table 4.4: Packet Loss Rate at Fragment size 256 at 100ms delays

2) Parameters for packet loss at fragment size 512 at various delays

The max packet was rate at fragment 512kb was observed when channel error rate was 0.6 and bandwidth was set to 0.3, 0.9 &1.5 and min. Packet loss rate at fragment 512kb was observed when channel error rate was 0.2 and bandwidth was set to 1.5 over varying delay condition as shown in table 4.5, 4.6 and 4.7.

PACKET LOSS RATE-FRAG 512-10ms			
channel Error Rate B.W	0.2	0.4	0.6
0.3	40.07%	42.30%	62.89%
0.9	18.92%	39.51%	62.89%
1.5	19.85%	38.40%	62.89%

Table 4.5: Packet Loss Rate at Fragment size 512 at 10ms delays

PACKET LOSS RATE-FRAG 512-50ms			
Error Rate B.W	0.2	0.4	0.6
0.3	42.67%	40.81%	62.89%
0.9	20.22%	38.96%	62.89%
1.5	20.03%	38.96%	62.89%

Table 4.6: Packet Loss Rate at Fragment size 512 at 50ms delays

PACKET LOSS RATE-FRAG 512-100ms			
Error Rate B.W	0.2	0.4	0.6
0.3	38.58%	41.18%	62.89%
0.9	21.89%	39.14%	62.89%
1.5	20.22%	39.14%	62.89%

Table.4.7: Packet Loss Rate at Fragment size 512 at 100ms delays

3) Parameters for packet loss at fragment size 1024 at various delays

The max packet was rate at fragment 1024kb was observed when channel error rate was 0.6 and bandwidth 0.3 under varying delay conditions and min. Packet loss rate at fragment 1024 kb was observed when channel error rate was 0.2 and bandwidth was set to 1.5 over varying delay condition as shown in table 4.8, 4.9 and 4.10.

PACKET LOSS RATE-FRAG 1024-10ms			
Error Rate B.W	0.2	0.4	0.6
0.3	38.04%	40.57%	63.04%
0.9	21.37%	41.30%	63.04%
1.5	20.65%	40.94%	63.04%

Table 4.8: Packet Loss Rate at Fragment size 1024 at 10ms delays

PACKET LOSS RATE-FRAG 1024-50ms			
Error Rate B.W	0.2	0.4	0.6
0.3	37.68%	41.66%	63.76%
0.9	21.73%	41.30%	63.40%
1.5	21.73%	42.75%	64.13%

Table 4.9: Packet Loss Rate at Fragment size 1024 at 50ms delays

PACKET LOSS RATE-FRAG 1024-100ms			
Error Rate B.W	0.2	0.4	0.6
0.3	37.31%	40.57%	63.76%
0.9	21.73%	42.75%	64.13%
1.5	21.30%	42.02%	64.49%

Table 4.10: Packet Loss Rate at Fragment size 1024 at 100ms delays

V. CONCLUSION

In this research work the performance analysis of MPEG-4 Traffic in Mobile Ad-hoc Networks has been done on the basis of five parameters. The results are prepared by considering different outcomes of the fraction of decodable frames (Q) of MPEG video on different parameters and on the basis of various performance metrics we have reached to conclusion that when the packet size increases at lower error rate (0.2 and 0.4) for all bandwidths there is a marginal decrease in the packet loss rate and marginal increase in the throughput but when the error rate is higher i.e. (0.6) there is a marginal increase in the packet loss rate and marginal decrease in the throughput. It has also been observed that at smaller fragment size the average packet end to end delay is lower and vice-versa. The peak (maximum) delay is lower at smaller fragment size and as the Packet size increases the peak (maximum) delay is higher. The delay should not be higher as much as possible to receive the packets at time to ensure the quality of service in multimedia applications. It can be said that good quality video stream is achieved while the fragment size was set to be 1024 compared to fragment sizes 512 and 256.Thus the paper work presents a novel technique of optimization of the traffic for UDP classes under various constraints of network parameters that may be useful in the analysis of study in multimedia routing in MANET.

VI. FUTURE SCOPE

In the future scope we can analyze the TCP Traffic using various routing protocols with different fragment sizes under various network parameters that may be useful to predicting the behaviour of quality of video in Mobile Ad-hoc Networks. We can compare the UDP and TCP traffic using various routing protocols with different fragment sizes. Different performance parameters can be evaluated under different queue management schemes with different fragment sizes. The study can be extended by comparison of the UDP and TCP traffic under different queue management schemes with different fragment sizes.

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