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Enhancement of Quality of Service in MANETS through Weighted Fair

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Abstract

Video streaming in MANETs is most Challenging issue and it mainly affected by these factors like node mobility, dynamic change in topology, multi path shadowing and fading, collusion, interference and many more. The dynamic change in topology causes periodic connectivity which results in large packet loss. Therefore QoS is an important issue for the transmission of multimedia data in Mobile Ad-Hoc Network. Video streaming in real time requires special techniques that can overcome the losses of packets in the unreliable networks. Developments in mobile devices and wireless networking provide the technical platform for video streaming over mobile ad hoc networks (MANETs). Most of the research in MANET concentrates in QoS enhancement through routing mechanism. This paper concentrates on other QoS parameters of traffic characteristics like buffer size, bandwidth and stream synchronization. A model has been proposed which is reliable for data transmission that uses Weighted Fair Queue model that classifies the incoming packets based on the data rates and allocates bandwidth accordingly. H.264/AVC is used for encoding and decoding the video.

Keywords— *Mobile Ad Hoc Networks, Weighted Fair Queue, QoS, Mean Opinion Score, ffmpeg, MPEG4.*

I. INTRODUCTION

A mobile ad-hoc network is an autonomous system of mobile nodes connected by wireless links. The nodes are free to move randomly and organize themselves arbitrarily. Therefore, the network's wireless topology may change rapidly and unpredictably. Wireless communicational is becoming an important in our everyday lives. Among different kinds of available wireless networks, MANETs are expected to be widely deployed in future. A mobile node with low quality channel uses low data and adds more delay.

The demand for multimedia and information exchange over wireless mobile ad hoc networks is rapidly rising. Applications such as voice communication, video-on-demand, video conferencing, radio broadcasting and video gaming play dominant roles over such networks. Since in MANETs, the mobile nodes can move arbitrarily, the frequent changes in topology leads to variations in radio channel which may cause high error rates and power constraint of a node leads to frequent changes in connectivity. All these cause packets losses and degrade video quality. To provide an acceptable QoS for video transmission in MANETs, there should be effective rate control, error control, and routing mechanisms which can reduce packet loss. Many conventional distortion modeling techniques consider only a linear relationship between the packet loss and distortion. The levels of distortion may be based on position of packet loss in transmitted frames such as I (Intra-coded), P (Predicted) and B (Bi-Predictive) frames with Group of Picture (GOP)-level granularity.

The multimedia applications are keen for stable networks to make the QoS. The effective utilization of bandwidth for the multimedia applications with QoS is certainly one of the important challenges for the future generation of wireless networks. A higher amount of bandwidth is required for the transmission of sensitive applications.

D. Gozupek et al. proposed optimum rate allocation framework resources for improving QoS. QoE [3] is related to consumer's experience which needs feedback for evaluation in order to assess data quality. P. Goudarzi used an optimization network to decrease the loss-induced distortion which is generated from packet loss. D. Collange and J. L. Costeux prposed a method to find the correlation between QoS and QoE.

II. RELATED WORK

Many conventional modeling techniques consider packet loss and distortion from bandwidth allocation strategy which are based on some specific network constraints in MANET. Their main objective is improving QoE using a common metric for optimization. P. Goudrazi proposed an optimal rate framework based on which the overall distortion of the video sources can be minimized. S. Khan et al. used a linear mapping where mapping is done between MOS and Peak Signal to Noise Ratio (PSNR) in which distortion is linearly related to packet loss. E. Setton and B. Girod showed that the total distortion of decoded video is the distortion caused by the video encoder and late arrival during transmission.

C. Chellamuthu and P. Sankar have done a testing in a scenario in which data is transmitted from source node to destination node. A model has been developed that allocates the resources for the sensitive applications using WFQ [9]. P. L. U. Choi et al. developed an accurate model which can capture the exact effect of the packet loss in network on video rate performance distortion performance with group of picture level granularity in MANETs. The resulting data is decoded and the video quality analysis is done using metric (MOS). These optimal rates can be used a rate feedback of H. Before you begin to format your paper, first write and save the/AVC encoder. QoS varies depending upon the mobility of nodes. The quality of streamed data offer much better performance on network links that are not congested. To maintain a high image quality minimum bandwidth should be maintained.

Priority Queuing is the oldest technique of all the queuing techniques. Traffic is prioritized with a priority-list, applied to an interface with a priority-group command. The traffic goes into one of our queues: high, medium, normal or low priority. When the router is ready to transmit a packet, it searches the high queue for a packet. If the packet is present it will be sent. If not, the medium queue is checked. If there is a packet it will be sent. If not, the normal and low priority queues are checked. This process will be repeated for the forthcoming packets. If there is enough traffic in the high queue, the other queues may get starved and they never get serviced.

An overview of proposed model is discussed in Section III. Module description has been done in Section IV. Performance evaluation is in Section V. Remarks and Conclusion is in Section VI.

III. PROPOSED SYSTEM

This paper presents a resource allocation method for the sensitive applications using WFQ model in MANET. The WFQ model provides dynamic fair queues by dividing the resources of the traffic based on the weights of the packets. It is a flow-based algorithm that simultaneously schedules interactive traffic to the front of a queue to reduce the response time. The WFQ model uses the traffic priority management system. It dynamically sorts the traffic into messages that make conversation.

A. Prposed Model

The drawback of priority queuing can be overcome by WFQ which can prevent high-bandwidth traffic from overwhelming the resources of a network, a phenomenon which can cause partial or complete failure of low-bandwidth communication during periods of high traffic in poorly managed networks.

It breaks up the train of the packets within a conversation to ensure that the bandwidth is shared fairly among all the users. The packets are transmitted in a timely fashion. First, the model serves the queue with the lower weight. It works as a work-conserving queue model and provides QoS for end-users. The model reserves a minimum guaranteed band-width for each class. The remaining bandwidth will be divided proportionally to the reserved clients. The high priority classes can be defined as having strict priority.

Weighted Fair Queuing is a method of automatically smoothing out the flow of data in packet switched communication networks by sorting packets to minimize the average latency. In WFQ, the priority given to network traffic is inversely proportional to the signal bandwidth. Thus, narrowband signals are passed along first, and broadband signals are buffered. The resource sharing is done according to assigned weights. Each flow receives an equal allocation of network bandwidth, hence the term fair. A weight is applied to queues to give some queues higher priority. For example, one queue may get half the available bandwidth and other queue will get the remaining bandwidth. If high-priority q queues are not in use, lower-priority traffic uses its queues. This prevents high bandwidth traffic from grabbing an unfair share of resources.

B. Proposed Architecture

The block diagram consists of a codec named H.264. Using this code the video contents are compressed and sent at the transmission side. It is tested in a scenario in which data is transmitted from a source to a destination node.

The resulting video which is received at the destination is checked to calculate the loss and the amount of distortion varies according to the loss. WFQ model is added to minimize distortion by managing buffer based on weight. Finally the data are transmitted to the receiving side.

IV. MODULE DESCRIPTION

A. Square Grid Implementation

Nodes are placed on a square grid topology. The number of nodes has to be a square of an integer. Wireless nodes are distributed in a square grid scenario. It is assumed that the nodes are Omni-directional and have the capability of moving within the area of the square grid. Each path related to the source contains wireless links from source to destination.

The total transmission rate associated with the particular link consists of two components: one is the traffic rate allocated to that flow of source and the other component is associated with the time-varying links' cross (background) traffic. Based on the assumption of strong multipath fading caused by the nodes mobility, Bit Error Rate (BER) of the link in the path of the video source may be calculated. The transmission power depends on the geometry of the nodes' distribution and also based on the number of nodes.

B. Transmitter (Source)

1)Video Encoding

The free software tool 'ffmpeg'[11] in the MPEG4 format is chosen as video code for encoding the video, as it widely used in the real world and provides better documented methods for comparing video qualities. It is a collection of free software that can record, convert and stream digital audio and video frames. The video file which is to be transmitted is compressed into a set of frames of I-frames, B-frames and P-frames with GOP level granularity. GOP specifies the order in which the frames are arranged. It begins with I-frames followed by P-frames several and the remaining gap with B-frames.

2)Loss Distortion Model

The distortion caused by video encoder and transmission channel is used for distortion modeling. Distortion consists of two components namely, video source distortion and loss distortion. Here, the distortion which is caused as a sequence of network-related packet loss is concentrated more rather than source-related distortion. There are several mathematical models to find out the average loss distortion at the frame level. A simplified version of Group of Picture (GOP) distortion without intra prediction or with constrained inter prediction is used and is compared for performance after WFQ is implemented. It can be done by using the 'fffmpeg' tool.

3)WFQ Implemenation

Weighted Fair Queuing is a queuing algorithm that combines fair queuing and preferential weighting. The fairness aspect of WFQ functions similarly to round-robin queuing, with queue serviced in a continuously repeating sequence from top to bottom, and then starting at the top again. The weighting aspect of WFQ applies a "weight" to a queue that indicates the importance of the queue in relation to available resources.

The weight is used to ensure that more important queues get serviced more often than other less important queues. With WFQ, queues are first sorted in order of their increasing weighted value. Then, each queue is serviced in order of its weighted proportion to the available resources.

C. Receiver (Destination)

1) Decoding the video

At the receiving end the video is decoded using 'ffmpeg' tool in the MPEG$ format and the received video is sent for video quality analysis.

2) Transmitter (Source)

Analyzing the video quality is an important factor as it may vary according to the parameters like psnr, packet drop, delay, etc. There are so many techniques for video quality analysis such as QoE (Quality of Experience), MOS (Mean opinion Source) measures, etc. Here MOS is used. By giving rating to the received video quality, their quality can be analyzed. For example, if the MOS value is greater than 4 then the video is said to be very good quality. If it is less then it is of poor quality. Higher the value, higher will be the user satisfaction and hence the quality will be enhanced.

V.EXPERIMENTS AND SIMULATION RESULTS

A cif-yuv video file, with 8 seconds play time, 30 frames per second is encoded using H.264 encoder (using 'ffmpeg' tool) to generate video trace file of 250 frames. A trace file 'st' is created using the mp4 trace utility for the video file and is attached to a 'tcl' program which simulate the given network topology. When the tck file is made to run, it converts the 'st' file to a video data file and writes it with 'dat' extension. It also generates a sender dump (sd) and a receiver dump (rd0 files.

Similarly three types of cif-yuv files are encoded, namely, akiyo, waterfall, and flower files with 30 frames per second. The program is run using priority queue and the streamed data are transmitted from node 0 to node 8. The corresponding trace (tr) file is taken for performance measurement by plotting latency, jitter and psnr.

	PSNR before Compression			PSNR after Decompression		
psnr	akiyo	flower	waterfall	akiyo	flower	waterfall
30 Frames	32.30	23.58	27.54	29.69	19.30	27.55
45 Frames	32.39	23.52	27.55	32.39	16.83	27.56

Another set of three cif-yuv files are encoded with 45 frames per second and their trace file is taken for plotting the graph of latency, psnr and jitter whose values are compared with that of the graph obtained by using 30 frames.

The performance of video streaming are evaluated using the utilities of NS2, ffmpeg, matlab and excel. The performance parameters latency, jitter and psnr are evaluated at the destination node where the video is received.

A) Latency

Latency also called response time is the time between the start and completion of an event and is considered to be one of the important parameter for video quality. It might take a long time for each packet to reach its destination because, it might lead to destination.

B) Jitter

It is the measure of variability over time of packet latency across network. The variation in delay is known as jitter and it can affect the quality of video streaming. It can also be stated as variation of packet inter-arrival time.

That is, the difference between when a packet is expected and when it is actually received. Packets from the source will reach the destination with different delays. A packets' delay varies with its position in the queues of the routers along the path between source and destination and this position can vary unpredictably.

C) Frames vs Psnr

PSNR (Peak Signal to Noise Ratio) is commonly used as a measure of quality of recognition of loss compression codec.

Signal is considered to be the original data. Noise is the error introduced by compression. The graph shows the psnr value of both original video which is being transmitted and psnr value after compression in which erroneous data is being induced.

This psnr vs. frame graph shows the range of reconstruction of original video of 30 frames for one of the cif-yuv file among the three types which is taken for compression.

V. CONCLUSION

In this paper, the analysis has been done on two different set of frames and compared their performance using fair queuing. A new model, WFQ for video streaming is proposed which uses a flow based algorithm for improving the qulity of service parameters like stream synchronization and buffer management.

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