A Survey on Quality of Real Time Streaming in Wireless Cellular Network

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Abstract— *Real time streaming over a wireless cellular network is highly adventure and prominent field to multimedia mobile users. Real time streaming involves watching TV on mobile, stream on demand video clips and place video telephony calls using multimedia capable mobile devices. A Cyclic Redundancy Code (CRC), when used properly, can be an effective and relatively inexpensive method to detect data corruption across communication channels. A large amount of data may transmit on real time on cellular network. While transmitting an input data over a cellular network, it is not guarantee to recover it at the receiver side because of input packet drops and congestion and therefore mobile users find difficulties during the streaming congestion has to be remove as it is the main cause for packet loss and delay that degrade the quality of service in networks. But improving the quality of real time streaming in wireless cellular network is still a huge challenge. This paper presents a brief review survey on various researches on Congestion Controls in Streaming over wireless cellular networks.*

Keywords—Congestion Control, Cyclic Redundancy Check (CRC), stochastic model, Quality of Service, video streaming.

I. INTRODUCTION

Wireless cellular network have fixed infrastructure where each mobile access the networks and communicating to other mobiles through the base station. It means that base station is fixed infrastructure. Nowadays wireless cellular network greatly offer possibility to watch different TV program on mobile devices, which is off course an example of real time content streaming. And hence there is a great chance to increase such type of traffic demand and decrease the quality of real time streaming in future. There are number of user call for streaming over a wireless cellular networks server but because of limited service capacity there may be possibility that server will not be able to serve some users because of congestion occurs [1]. The factor influencing the quality of real time streaming are limited service capacity, packet loss, bandwidth, outage time and packet delay. Packet loss causes decrease in the quality of real time streaming over cellular network. Real time streaming requires special and effective techniques to reduce the packet loss and increase the service capacity of the server in unreliable network. The Data Link layer in OSI model is responsible for error control. Error control is defined as the combination error correction and error detection. When the data is transmitted over the network, it is the responsibility of Data Link layer to check for the error between the routers and to correct them. An Error is any unwanted change which reduces the usefulness of original data. There are two types of error-

A. Single bit error

When only one bit of data unit gets corrupted during the transmission known as single bit error. Its detection and correction is easy. They occur in parallel data transmission where n different cables are used to transmit n different data bits e.g.:

Transmitted data---1 1 1 0 1 1 0 1 (1 is changed to 0)
Received data------1 0 1 0 1 1 0 1

B. Burst error

When multiple bits of data unit get corrupted during the transmission known as burst error. It is also known as the multiple bit error .Its detection and correction is difficult. They occur in serial data transmission where all the n bits are transmitted using a single transmission media e.g.: In this, multiple bit are changed-

Transmitted data---1 1 1 0 1 1 0 1
Received data------ 1 0 1 0 0 1 1 1

II. QUALITY OF SERVICE

Wireless cellular network have been largely demanding nowadays and hence to improve and maintaining the quality of service of such network is challenging for network operators. QoS is the capability of a cellular network to provide better service to selected network traffic over various networking technology [2][3].The primary objective of QoS is to reduce packet loss, reduce delay, and improve throughput and server capacity of the system. Depending upon the handling of network traffic different applications have different requirement. Those requirements are expressed using the following QoS related parameters.

- Packet loss-input average packet loss
- Delay-duration for packet reached to receiver

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- Throughput –number of packets per second
- Increasing service capacity

III. CONGESTION CONTROL PROTOCOL

3.1 TCP and UDP

Transmission control protocol (TCP) [4], is the widely used transport protocol is not suitable for streaming application in mobile ad hoc wireless network. This is because of the TCP infers a missing packet as a sign of network congestion which is not always true for mobile ad hoc networks. Packet loss over wireless link is greater than the wired links due to increased mobility of nodes, channel bit errors, medium congestion and path failures. Generally the packet losses may be because of congestion or wireless link errors. Though packet losses if not due to congestion TCP protocol invokes only congestion control algorithm and halves its transmission rate. Such type of change in transmission rate without any need affects the quality of the streaming applications. TCP gives reliable service and flow control mechanism but the TCP's retransmission scheme may not be needed for loss tolerant multimedia streaming applications. User datagram protocol (UDP) is used in multimedia streaming but the efficiency of congestion control mechanism is poor. Therefore the performance of quality of congested network may degrade. Thus basic TCP and UDP without any enhancement cannot be used to achieve good QoS in multimedia streaming applications.

3.2 SCTP

Stream control transmission protocol is a general purpose transport layer protocol [5] [6]. It provide reliable service and flow control mechanisms like TCP and also UDP, it supports unreliable transmission which is called SCTP. It can make a distinction between the levels of reliability provided to messages. The main feature of SCTP is providing multi-streaming and multi-homing services for a single connection. Hence SCTP is selected for distribution of multimedia data over heterogeneous wireless networks. SCTP based single path transfer and concurrent multipath transfer solutions improve the quality levels perceived by the end user during multimedia delivery over wireless networks. SCTP focuses on the design of a toolset to find the performance of multimedia over SCTP.

3.3 SSVP

Scalable streaming video protocol SSVP [7] is an end-to-end protocol with the TCP friendliness feature. It works as a payload on UDP. SSVP use the AIMD mechanism similar to TCP and control the sending rate by tuning the inter packet gap IPG. SSVP use AIMD and IPG considering care in smoothness and achieves outstanding performance in transmitting real time video under several network variations. In case of cloudy network conditions a layered adaptation mechanism is applied using the receiver buffering capability. This mechanism adapts the video quality to the long term variation in the available bandwidth. Change of layers may however provide an adverse effect on the end user perceived video quality. But depending on the receiver buffer status and bandwidth availability SSVP sends a refinement layer avoids changes in unnecessary layer. In this way SSVP improves the viewer perceived video quality. SSVP can work better in transmitting a multimedia data even under limited bandwidth constraints.

3.4 DCCP

Datagram congestion control protocol (DCCP) is an unreliable transport protocol similar to UDP and incorporate end-to-end congestion control. It implements a congestion controlled, unreliable flow of datagram for multimedia streaming. Unlike SCTP it is enhanced to support partial reliability PR-DCCP. A data dropped option makes one endpoint to declare that a packet was dropped either because of congestion or due to any wireless link errors. This facilitates research into more appropriate rate control responses for the non network congestion losses. Existing congestion control algorithms as in TCP and DCCP are naturally data oriented. They may be suitable for video transmission as the characteristics of video data are not explicitly considered.

TABLE-I
COMPARISION OF SCTP WITH UDP OVER MANET

Performance matrices	SCTP Result	UDP Result
Throughput (kbps)	260.672	219.184
Delay (ms)	10525.1	20413.3

TABLE-II
COMPARISION OF SCTP WITH TCP IN MULTISTREAM OPERATION

Parameters	2 Multimedia files		4 Multimedia files	
	TCP	SCTP	TCP	SCTP
Average Transmission	6.20	3.10	18.02	6.90
Times (s)	1.55	0.86	1.61	1.07

IV. CRC (CYCLIC REDUNDANCY CHECK)

It is one of the most powerful error detection technique based on the principle of binary division. In this, a sequence of bits append to the data unit is known as the CRC [13] which has two important qualities-

1. It must be one less than the number of divisor bits.

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2. After appending these bits data unit must be exactly divisible by divisor.

There are two important steps of CRC technique-

4.1 CRC generator

It is the process of calculating the CRC bits at sender side using a generating function such that the data unit and generating function are operated using bit-wise XOR operation where generating function is of n+1 bits and CRC is of n bits. The data bits to be operated are appended with the n 0's which together referred as data unit to be transmitted to receiver end.

4.2 CRC checker

It is the process of calculating the CRC bits at receiver side to check whether they are equivalent to 0 or not. It also uses the same bit-wise XOR operation on data unit received from sender side and generating function. If all the CRC bits are 0 then data unit is accepted otherwise it is discarded.

V. VIDEO STREAMING

5.1 Cross-layer Design

Cross-layer design shares knowledge between all layers to achieve highest possible adaptively [8] [9]. This method is basically design to control number of collision independent of the number of active stations for data transmission. It also control data traffic parameter based on data traffic load. This techniques use transmission priority so that the video with higher priority data can be delivers first. The throughput is improved by calculating partial checksum is calculated for this sliced data by the information stored in the NAL header. The sliced data which is generated by VCL is the input of ASIC. The advantage of this scheme is that when there is limited bandwidth the transmission order is dependent on the priority of slice. When every packet has equal transmission priority, the packet loss and delay time cause video fragmentation. In high bit error ratio environment, the partial checksum can reduce error video packet loss so that the throughput can be improved. The multimedia network ASIC is designed in Verilog hardware description language and synthesized by synopsis using CCU 0.09um CMOS single poly eight metal standard cell they can deal with 10Gbps fast Ethernet traffic. Besides ASIC support NAL so that it is more suitable for H.264/AVC streaming video delivery library. The proposed multimedia network ASIC operates faster when compared to other networks.

5.2. Multipath Routing Schemes Multipath routing [10] can improve QoS by giving accumulation of bandwidth and delay, route load balancing and fault tolerance to reduce the effect of network failure onto affected video quality, it is important that the path are disjoint. In case the multipath routing protocol offers multiple path with sufficient path diversity, it is less probable that a link failure affecting one of the path simultaneously affect one of the other paths. This is generally advantageous in real time streaming, where the playback buffer is limited and the video coder no longer can rely only on time diversity. Hamid Gharavi proposed two multichannel routing protocols. The first multichannel routing protocol is depends on single path routing. In this routing protocol the intra-path interference in a collision avoidance network or carrier sense multiple access networks is suppressed. The above routing protocol is developed by using link partitioning scheme where the neighbour nodes operate at different non overlapping frequency bands. A technique named systematic channel assignment for this approach shows that the partitioning scheme can substantially amplifies the throughput performance of a multi-hop link. The second multichannel routing protocol is used for transmission of real time traffic; this approach can be affected unfortunately from co-channel interference due to concurrent transmission of packet via multiple routes. So a dual path routing protocol is developed which guarantees a different frequency band for each path thus eliminating inter-path interference. This protocol reduces the possibility of losing all the routes at the same time.

VI. STOCHASTIC MODELLING ANALYSIS

This model is an analytical approach for the evaluation of quality of real time service understands by the users in wireless cellular networks [11]. This model includes stationary probabilities of traffic demand process. It considers the traffic demand with different radio condition of the calls and parameters of given wireless cellular network. It also gives expressions for various important performance characteristic of this model like the mean time spent in outage and mean number of outage incident for a streaming call in function of its radio conditions. The benefits of this model is to use in analyze the real time streaming in a typical cell of a 3GPP Long Term Evaluation (LTE) cellular network considering orthogonal intra cell user with peak bit rate nearer to the theoretical Shannon's bound in the additive white Gaussian noise (AWGN) channel with extra cell interference as noise. This model consists of Markovian multiclass process of call arrivals and their independent arbitrarily distributed streaming times. Such call is served by server whose service capacity is limited. Because of such limited service capacity the server will not be able to serve some class of user because of congestion in server and hence these classes of users denied the service till the next call arrival or departure, when the condition is re-evaluated. This stochastic model allows for very general service capacity. This model is an extension of the classical loss model where the service denial is not definitive for a given call but only temporal having the form of outage periods.

VII. QoS IN OFDMA CELLULAR NETWORK

QoS in OFDMA cellular network use the analytical method for evaluation of the QoS perceived by the users in the downlink of OFDMA wireless cellular networks which serve streaming and elastic traffic. This method describes the resource allocation problem such as power and bandwidth and characterized its feasibility by some reference feasibility condition (FS). This method is based on multi-rate Erlang loss model blocking probabilities can be evaluated by means of Kaufman-Robert algorithm or by Erlang formula. This method also evaluates analytically the QoS of elastic users that is mean throughput and delay by using multiclass processor sharing model. To obtain numerical values, this method considers the most popular hexagonal network model, in which the BSs are placed on a regular hexagonal grid. Consider R be the radius of the disc whose area is equal to the hexagonal cell served by each BS and call R be the cell radius. To neglect the border effect it considers the network which is "wrapped around" that is deployed on torus consisting of 4*4=16 cells. To obtain a discrete model, each cell is decomposed into 5 equally thick rings around the BS. Users arrive uniformly to the network and don't move during their calls. It consider a propagation loss L(r) = (Kr)η; with η = 3.38 and K= 8667 where r is the distance between the transmitter and the receiver. The system bandwidth equals W= 5MHz.BSs have Omni-directional antennas having a gain 9dBi and no loss. The BS maximal total power equals 43dBm; thus ˜P=43+9=52dBm when we account for antenna gain and loss. the common channel power ˆP is the fraction ε=0.12 of ˜P and the ambient noise power WN0= -103dBm. In this three values of the cell radius R=0.525.3 or 5km and streaming voice calls at 12.2kbps are consider. The corresponding SNR in real channel is typically -16Db. Such SNR corresponds to a bit rate $W\log_2$ (1+SNR) =180Kbps on AWGN channel.

VIII. CONCLUSION AND FUTURE WORK

In this survey paper several research and invention on quality of real time streaming over wireless cellular network have been studied. We have explained some congestion control protocol used for enhancing the quality of real time streaming over cellular network which suggest how the packet loss as well as reducing throughput occurs because of congestion. This paper also review some video streaming techniques and resource allocation techniques like stochastic modelling and analysis methods for evaluation of quality of service in OFDMA wireless cellular to improve the quality of streaming over wireless cellular network. a number of techniques used for error detection at data link layer among which CRC provides desirable efficiency. It provides good performance in terms of accuracy and security compared to other techniques. Our future work is to developed efficient techniques to overcome such type of challenge and improve quality of service using CRC in real time streaming over cellular networks.

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