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Research Article

Optimized Multimedia Streaming and Congestion Control for WLAN-3G Networks

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Abstract: Congestion control of a variable bit-rate video stream over the wireless link is critical issue to assure the quality of the received video. If congestion occurs it leads to the buffer overflows which in turn results in packet loss in the networks. In order to overcome this issue, an Optimized Multimedia Streaming and Congestion Control technique for WLAN-3G Network is proposed. In this study, a two-tier integrated heterogeneous wireless network including WLAN and 3G cellular are considered. The streaming rate is estimated by using Fuzzy Logic congestion (FLC) controller. The Congestion level determination unit determines the congestion level and the rate of change of congestion level based on the inter packet gaps which are taken as input to the FLC. Based on the output of FLC, new streaming rate is estimated. The application adjusts to the new streaming rate in order to maintain the buffer level. Also, the threshold value for buffer level is optimized by using the Greedy Streaming to Threshold and Stop (GSTS) algorithm in order to reduce the streaming monetary cost.

Keywords: 3G networks, congestion control, fuzzy logic, multimedia, streaming, Wireless Local Area Networks (WLAN)

INTRODUCTION

Wireless LAN (WLAN): Recent advances in computer science and communication impose a need of communication among people with voice and video any time. The voice over IP (VoIP) has an increasing market and the next evolution of the market is video and voice over IP (V2IP). The WLAN make it possible to chat with computer or other terminal face to face. 802.11X is the widely used protocol on WLAN Hai-Tao and Gui-Quan (2011) IEEE 802.11 Wireless Local Area Network (WLAN) being in current market position for the (indoor) broadband wireless access networking defines the functionality of Medium Access Control (MAC) layer and physical (PHY) layer specifications for WLAN. The MAC has a mandatory part the Distributed Coordinated Function (DCF) which supports best-effort service without guaranteeing any QoS and having no service differentiation (Puthal *et al*., 2008) the increased IEEE 802.11 devices paved the way for the tremendous growth in the last ten years. In 1997, the first standard appeared defining the Wireless LAN MAC and Physical (PHY) Layers specifications which provide data transmission rates up to 1 Mbps and 2 Mbps over the 2.4 GHz range. Later, IEEE 802.11 g was approved supporting higher speeds up to 55 Mbps and working within the 2.4 GHz range. The fundamental medium access mechanisms of IEEE 802.11 protocol are called Distributed Coordination Function (DCF) and the optional access mechanism is Point Coordination Function (PCF). DCF is based on the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) protocol; however it is unable to provide the required performance for voice and video applications, because it is fundamentally developed for Best Effort services (Charfi *et al*., 2013) The configuration simplicity and the relatively cheaper network setup make accessing the Internet via WLANs to become more and more popular. WLAN has various applications anywhere in campuses, in offices and even at home by using mobile devices, notebooks, or PDAs. Among all applications, multimedia applications, like multimedia streaming, multimedia messaging, video telephony and video on-demand are the famous applications in most interesting and popular applications, as multimedia applications are easier to be accepted and are closer to human life. WLANs play a key role in home networks (Wei *et al*., 2004) WLAN has a serious safety problem of easily crack able password and QoS guarantee (Hai-Tao and Gui-Quan, 2011).

Multimedia streaming in WLAN: Applications like multimedia streaming over Internet include typical

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riding of the video over UDP. It has a challenge over different UDP and TCP flows on the network. Multimedia streaming over internet has been gaining attention for research (Dyahadray *et al*., 2008; Balan *et al*., 2003) video streaming is a network sensitive application. The video streaming solution needs to meet the following requirements along with ability to change bit rate:

- Efficient mechanism to deploy video data on network
- Provision for resilience against packet loss events common to Internet.
- Low computational complexity and memory requirements to guarantee minimal processing time (Dyahadray *et al*., 2008) Effective multimedia transmission has to face issues like
  - Significant data loss in the network
  - Jitter due to variation in congestion control scheme
  - Network Burst losses leading to loss of set of packets containing a single frame information making estimation techniques at the receiver ineffective
  - Synchronization loss between the encoder and decoder due to network losses
  - Loss of a significant amount of data with a loss of single packet that renders quality reconstruction almost impossible (Balan *et al*., 2003)

Congestion control in WLAN: Congestion control of a variable bit-rate video stream crossing the Internet is critical to ensure the quality of the received (Jammeh *et al*., 2008). Many researches have been done on developing and analyzing the various congestion control schemes and receiver side adaptation to reduce jitter for the multimedia traffic. If congestion occurs in TCP, it affects TCP throughput but inbuilt retransmission mechanism eventually guarantee reliable data delivery. Moreover in UDP, the packets are dropped and lost the information permanently. In such cases, most of the video codecs should depend on error concealment techniques to achieve adequate video quality at the receiver. However in extreme cases, a real-time streaming experience will be difficult to deliver to the end user. The data loss can be prevented by dynamically adjusting the bit rate of the stream relying on the current traffic level on the link. The contemporary techniques such as Bitstream Switching, Transcoding and Cross-layer Coding provide the mechanisms to adapt the bitrate in real Time (Dyahadray *et al*., 2008; Balan *et al*., 2003).

Objective of the work: A buffer management scheme is proposed (Hassan *et al*., 2010) to buffer the stream of packets in multimedia streaming in FIFO order for maximizing successfully delivered frames, i.e., goodput. A weighted priority algorithm is proposed based on which rank of the frames is calculated. This buffer management can reduce congestion. However in the buffer management scheme alone cannot reduce congestion. Congestion if occurs leads to buffer overflows resulting in Packet loss in networks. The streaming cost in 3G cellular network will be high. The streaming rate should be estimated based on incoming and outgoing packets apart from the buffer level.

In this study, an optimized multimedia streaming and congestion control technique for integrated heterogeneous wireless networks is proposed in order to reduce the streaming cost.

LITERATURE REVIEW

Dyahadray *et al*. (2008) proposed hysteresis-based layer switching scheme for providing efficient congestion control and immunity against network fluctuations. This congestion control system achieved robustness, flexibility and ease of implementation in addition to ensuring optimal user experience. In addition, finer control mechanism can be incorporated using FGS (Fine Grain Scalability) encoding scheme. FGS adds to the flexibility in bit rate adaptation as it offers more number of operating points for a given spatio-temporal layer.

Jammeh *et al*. (2008) incorporated fuzzy logic to a rate-adaptive transcoder. Fuzzy logic control of congestion was a sender-based system for unicast flows. The receiver sent back a feedback message denoting time-smoothed and normalized changes to packet inter-arrival time. These enable the sender to compute the network congestion level through pre-designed fuzzy models. The sender enables application of a control signal to the transponders' quantization level, as a reflection of the anticipated congestion. The packet stream dispersion are measured during busy router queues especially tight links representing the point of minimum available bandwidth on the network path thereby achieving congestion control without packet loss feedback.

Zahran and Sreenan (2010) proposed a generic practical framework that optimizes media streaming in heterogeneous systems by taking advantage of cost and resource characteristic diversity of the integrated access technologies and the buffering capability of streaming applications. The proposed optimization framework represents a means to compromise the tradeoff between different performance metrics including streaming monetary cost, signaling load and session quality. Additionally, it accommodates different design challenges including mobility randomness, limited processing capacity and handoff delay requirements.

Scalosub *et al*. (2010) presented a model capturing inter-packet dependencies and designed algorithms for packet discarding. A buffer management algorithm was proposed with specific guidelines. The scheme provided guarantees for the proposed algorithms for any traffic arrival pattern and without any stochastic or deterministic assumptions on the processes generating the traffic. However there is a tradeoff between fairness and throughput.

Hassan *et al*. (2010) proposed an adaptive and fair priority scheduler for video streaming over wireless links which chose to serve according to the instantaneous occupancy of the decoder buffers of the wireless clients as well as the quality of the channel as seen by these clients in addition to the sensitivity of scheduled video frames. The proposed scheme recorded the characteristics and strict requirements of multimedia applications with considerably low computational complexity. However, the performance degrades if the channel condition is totally ignored.

Vadakital and Gabbouj (2011) analyzed prediction structure of video as binary relations generating a prediction graph. Reachability concepts of graph theory were then applied to construct binary valued indicator functions. These indicator functions report the reception, decoding and play-out states of each access-unit in the video sequences. Zapping-delay is minimized for each individually, while the final zapping-delay does not exceed a known bound, is found. However, the proposal only partially solves the problem of bounded zapping-delay.

PROPOSED SOLUTION

Overview: In this proposal we propose to design an optimized multimedia streaming and congestion control technique for integrated heterogeneous wireless networks.

A two-tier integrated heterogeneous wireless networks comprising WLAN and 3G cellular are considered. Streaming monetary cost of media in WLANs is cheaper than the corresponding cost in the cellular network. The streamed media is precoded with a fixed data rate and the streaming rate is adjusted before leaving the cheaper network to reduce the signaling cost (Zahran and Sreenan, 2010). This can be done using the Fuzzy Logic Congestion controller (FLC).

The client-side timer unit monitors the incoming packet dispersion and broadcast this information to the Congestion Level Determination (CLD) unit. Then Congestion level determination unit monitors the outgoing packet stream, especially the packet sizes and combines this information with feedback from the client, to determine congestion level as calculated based on inter packet gaps (Jammeh *et al*., 2008). Then the congestion level, congestion-level rate of change and the measured buffer level are treated as input to the FLC. Then the new streaming rate is estimated from the output of FLC. The application readjusts the streaming rate to the estimated new stream rate, in order to maintain the buffer level until the mobile terminal exits the congested hot spot The threshold value for buffer level can be optimized using the Greedy Streaming to Threshold and Stop (GSTS) method of Zahran and Sreenan (2010).

Figure 1 represents the proposed block diagram. In the proposed approach, a two-tier integrated heterogeneous model is considered. To compute, the new streaming rate, a Fuzzy logic controller technique is used. Congestion level and rate of change of congestion level are taken as input and based on the output of the fuzzy logic new streaming rate is computed. Using the GSTS algorithm, threshold of the buffer level is optimized.

Two-tier integrated heterogeneous system model: In the proposed model, a two-tier integrated wireless system (Zahran and Sreenan, 2010) composed of

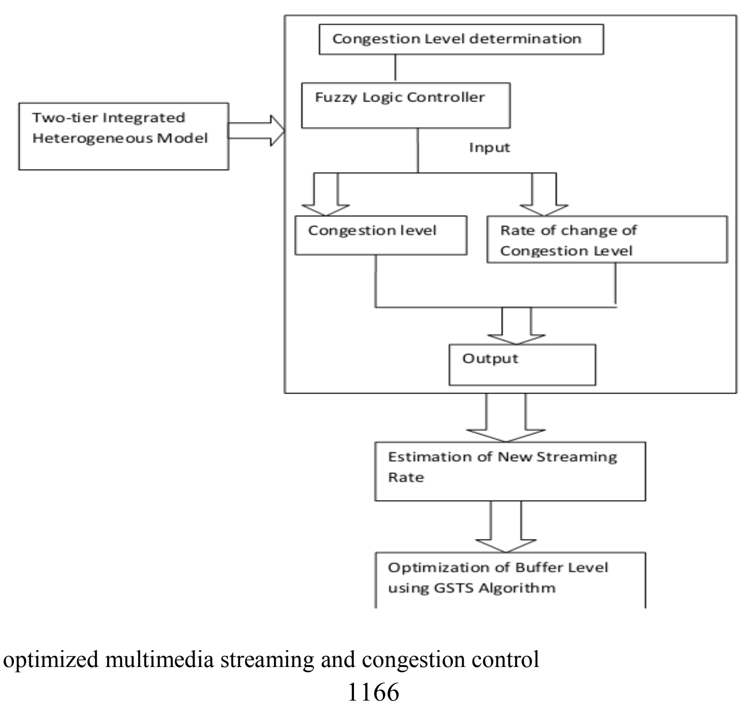

Fig. 1: Block diagram for optimized multimedia streaming and congestion control

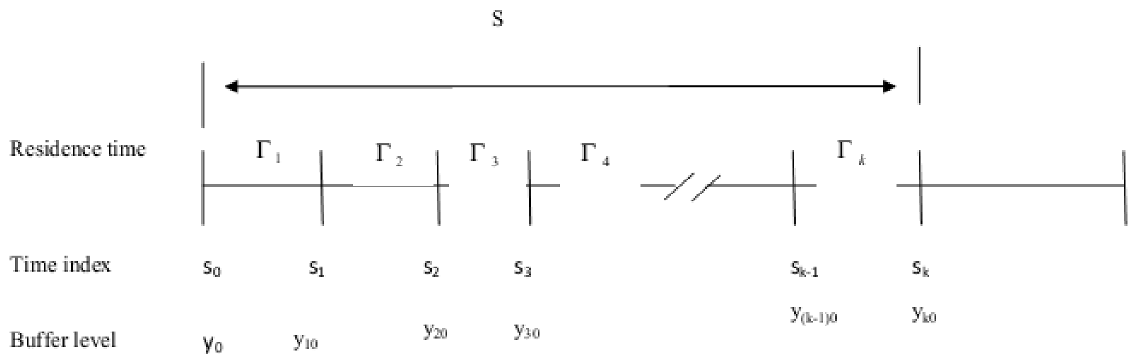

Fig. 2: Streaming session

networks N_n for $m \in \{u,v\}$ where u and v correspond to the technologies that provide universal and intermittent coverage respectively. Here, for example, the 3G cellular network and WLAN, respectively, represent the technologies with the universal and intermittent coverage in a 3G-WLAN integrated system. The presented system can be simplified to any two-tier integrated system. It is assumed that each network has a continuous non decreasing rate-dependent cost profile, represented as $Y_n(w_n)$ where w_n denotes the average data service rate in the network N_n. It is important to note that commonly adopted usage-based pricing strategies are equivalent to linear rate dependent strategies for continuous use of the resource, similar to case for media streaming. To demonstrate, if $Y_n(w) = aw$, the monetary specific period p equals $(aw)p = a(wp)$ in which wp denotes the size of downloaded data and a denotes the cost per data unit. It is assumed that the streaming monetary cost of considered media in WLAN is cheaper than the corresponding cost in the cellular network. The reverse case is unimportant as the user always maintains its association with universal network.

Figure 2 represents a typical state for streaming session in a two-tier heterogeneous network. Usually, session duration which is denoted as S, follows a generic heavy extremity distribution. Once, the session starts at S_o, the mobile terminal meet consequent transition between unique and technological zones at certain time s_i, till the session terminates at s_j. The time period spent by the user in different technological region, denoted as Γ_n, are supposed to have cumulative density function and generic probability denoted as $F_n(\Gamma_n)$ and $f_n(\Gamma_n)$ respectively. Moreover, $\overline{\Gamma_n}$ represents the residual time distribution of Γ_n.

The dynamics of the streaming application buffer level, $y(t)$ is managed based on the following differential equation:

$$y'(s) = w_n(s) - w_o \tag{1}$$

where, w_n denotes the server streaming rate during time period Γ_n and w_o denotes the average playout data rate. In general, w_n is non-negative and upper bounded by a maximum service rate of $w_{n\max}$ in network N_n, that means $w_n \in [0, w_{n\max}]$.

Here, the streamed media is precoded with a fixed data rate which means that in the proposed model assumes that the user selects a specific media quality for the complete session. Moreover, proactive stream management is considered in which the streaming rate is modified before leaving the cheaper network to lessen the signaling cost and assure seamless VHOs.

Here, the application proactively tunes the streaming rate in order to maintain the buffer level above the initial playout latency point, in case the rate of change of buffer level is in negative. The main objective of considering this assumption is that, it assures that at each and every point buffer has enough data for smooth playout of the media. That clearly means that the buffer overlooks the buffer fluctuation that may arise as a result of playout rate variability and channel degradation. As these fluctuations are of minor effect on the proposed optimization problem due to large time scale variation between these fluctuation and zone residence duration, hence the estimated playout and streaming rate are optimal values.

Congestion level detection model: This section describes about the estimation of congestion level (Jammeh *et al*., 2008) in the proposed method, the level of network is calculated by the application itself. The exact difference between reception from and transmission rate into the network for the same application is analytic of the network congestion level. In terminology of packets, this amount to finding packet dispersion, which is computed from Inter Packet Gaps (IPGs). The sending and receiving rates are computed from IPGs at the server and client respectively. The IPGs are then normalized based on the packet size.

At server side, the normalized IPG is the reciprocal of the bandwidth spent by the application in broadcasting individual packet into the network and a similar relationship employs at the client. Let the IPG of the packet be P_S and P_C for the packet entering the network and exiting the network respectively. P_C equal

P_S when the available network bandwidth is equal to or more than the transmitting rate of the packets. A congested network usually has a dispersive effect on the IPG that results in P_C greater than P_S. Hence, the difference between P_C and P_s is a graded evaluation of network congestion.

Normalizing the IPGs based on packet sizes make them dependent on the network congestion level. The normalized value of P_C and P_S enable computation of the network congestion level. Hence, the congestion level is determined without relying on packet loss, which is important for video. An average packet transfer time K^S_a is measured at the server and K^C_a is measured for the client. The averages are computed in a frame transmission duration which is used to determine the level of network congestion. The sending rate is altered, only when a feedback message arrives at the receiver. The sending rate alters itself only at the beginning of a video frame to assure consistent quality within a frame. A description of the network congestion-level measuring algorithm is described as below:

For two consecutive packet, of sizes E_n and E_{n-1}, received at times T_n and T_{n-1}, the transfer time, K_n is then defined as below:

$$K_n = \frac{T_n - T_{n-1}}{E_n} \tag{2}$$

where,
T_n = The arrival time of the current packet.
T_{n-1} = The arrival time of previous packet
E_n = The size of the current packet.

A set of transfer time, $\{K^S_w\}$ with cardinality sn, is computed at the server during a frame transmission time. A set of transfer times at the client, $\{K^C_w\}$ with cardinality cn, is also gathered over the same time period:

$$\{K^S_w\} = \{G_{s1}, G_{s2}, ...G_{sn}\} \tag{3}$$

$$\{K^C_w\} = \{G_{c1}, G_{c2}, ...G_{cn}\} \tag{4}$$

In general, $sn \neq cn$, as because of the possibility of packet loss.

The members of sets $\{K^S_w\}$ and $\{K^C_w\}$ are assigned respectively to H_S and H_C equally spaced bins. The range of the bins is found dynamically at the server and client to give:

$$H_S = \frac{K^{\max}_{sn} - K^{\min}_{sn}}{\delta z},\ H_C = \frac{K^{\max}_{cn} - K^{\min}_{cn}}{\delta z} \tag{5}$$

where, $K^{\max}_{sn}$ and $K^{\min}_{sn}$ are respectively the maximum and minimum of the $\{K^S_w\}$. $K^{\max}_{cn}$ and $K^{\min}_{cn}$ are maximum and minimum of the $\{K^C_w\}$.

Also, $\frac{1}{\delta z}$ represents pre-set resolution bitrate, which mainly depends on the maximum encoding rate, with number of bins in practice being constrained to 256. The objective of binning process is to reduce the effect of IPG compression, which may occur in the presence of cross traffic and as a consequence it may spread the measured transfer times. Based on potential cross traffic, δz is estimated to be equivalent to 1% of the encoding rate. In case, the resulting number is too large or small, then the available bandwidth mode might not be as accurately estimated, due to spread in measurements.

Based on the binned time intervals, a frequency weighted average transfer time is computed for the server and client respectively as K^S_a and K^C_a in (6):

$$K^S_a = \frac{\sum_{i=1}^{H_S} J^S_i \times L^S_i}{\sum_{i=1}^{N} L^S_i},\ K^C_a = \frac{\sum_{i=1}^{H_C} J^C_i \times L^C_i}{\sum_{i=1}^{N} L^C_i} \tag{6}$$

where, J^S_i and J^C_i represents the bin transfer values at the server and client respectively and L^S_i and L^C_i are their frequencies of occurrence.

A continuous exponentially weighted moving average sort out measurement noise from both values by updating K^S_{aj} and K^C_{aj}:

$$\begin{aligned} K^S_{aj} &\Leftarrow \beta K^S_a + (1-\beta) K^S_{aj} \\ K^C_{aj} &\Leftarrow \beta K^C_a + (1-\beta) K^C_{aj} \end{aligned} \tag{7}$$

where, initially at time zero K^S_{aj} and K^C_{aj} are set to initial values of K^S_a and K^C_a respectively and the exponential parameter $\beta \leq 1$ is normally set to 0.1, since higher values result in excessive fluctuation of the rate.

The transmission rate of the application into the network and the receiving rate of the client from the network can be calculated as B_S and B_C below:

$$B_S = \frac{1}{K^S_{aj}},\ B_C = \frac{1}{K^C_{aj}} \tag{8}$$

The difference between the two rates B_D is then calculated as below:

$$B_D = B_S - B_C \tag{9}$$

The network congestion level, C_N, is then consequently calculated as below:

$$C_N = \frac{B_D}{B_S} = 1 - \frac{K^S_{aj}}{K^C_{aj}} \tag{10}$$

Finally, δC_N is computed as simply the difference between the present and previous value of C_N. Both C_N

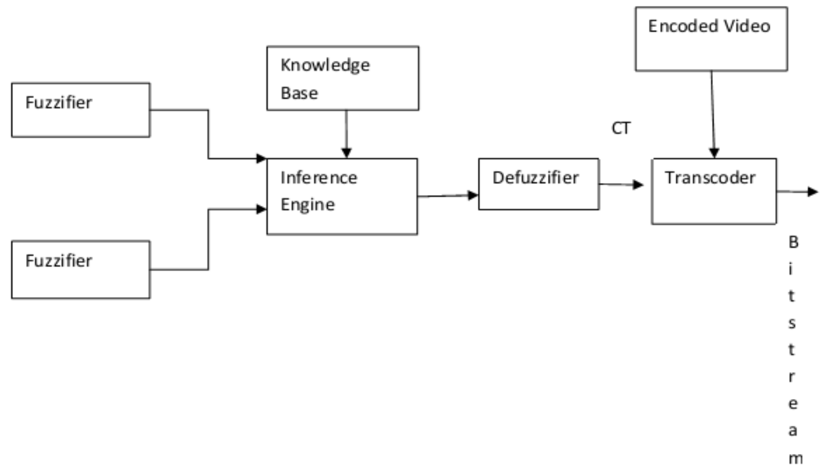

Fig. 3: Block diagram of fuzzy logic controller

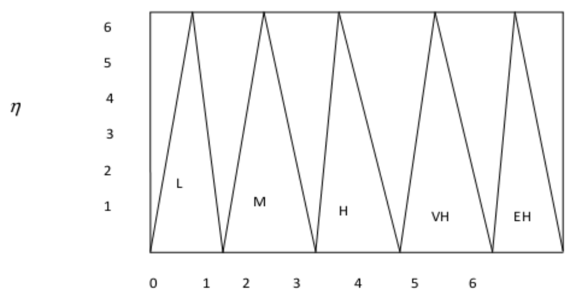

Fig. 4: Congestion level (C_N)

Table 1: C_N

Value	Meaning
L	Low
M	Medium
H	High
VH	Very high
EH	Extremely high

Table 2: δC_N

NVH	Negative very high
NM	Negative medium
NH	Negative high
NL	Negative low
Z	Zero
PL	Positive low
PM	Positive medium
PH	Positive high
PVH	Positive very high

and δC_N depends on packet dispersion and these are considered as inputs to the congestion controller.

Fuzzy Logic Congestion (FLC) controller: This section describes about the fuzzy logic congestion controller technique. Fuzzy logic follows a control process, similarly like a human expert is regulating the transmission rate. Multiple fuzzy membership functions form the uncertainty in that expert's view of the feedback, while an output rate decision is made accurate by the process of defuzzification, which transform uncertainty in the output to a crisp value, i.e., a precise control signal value.

Figure 3 represents a block diagram of Fuzzy Logic Controller. The technique is explained in the following step:

Step 1: Fuzzifiers convert the input C_N and δC_N into an appropriate linguistic variable.

Step 2: A knowledge base summarize expert knowledge of the application with the required control objective. It classifies the labels that help to specify a set of linguistic rules.

Step 3: The inference engine block is the brain power of the controller, with the ability of emulating the decision making process, by considering knowledge database and embedded rules for making suitable decision.

Step 4: The defuzzification transforms the inferred fuzzy control decision from the inference engine to a crisp value, which is then converted to a control signal, CT as shown in Fig. 3 to the transcoder, which gives an output in the form of re-compressed Bitstream.

Fuzzification: Fuzzification is the name given to the application of a membership function, η, to a data value to obtain its membership possibility, i.e., $\eta(x)$ generates the possibility of membership of the fuzzy subset for which η represents the membership function.

The input variables are fuzzified based on triangular-shaped membership functions, being the usual compromise between reduced computation times at the cost of a sharper transition from one state to another. Selecting the number of membership function is vital, as it determines the smoothness of bit-rate granularity. Also, the number of membership function is directly proportional to the computation time. The congestion level, the rate of change of congestion level and the control output are each divided into a set of overlapping triangular membership functions, with the overlap such that size of any one triangle reached the midpoint of the base of another.

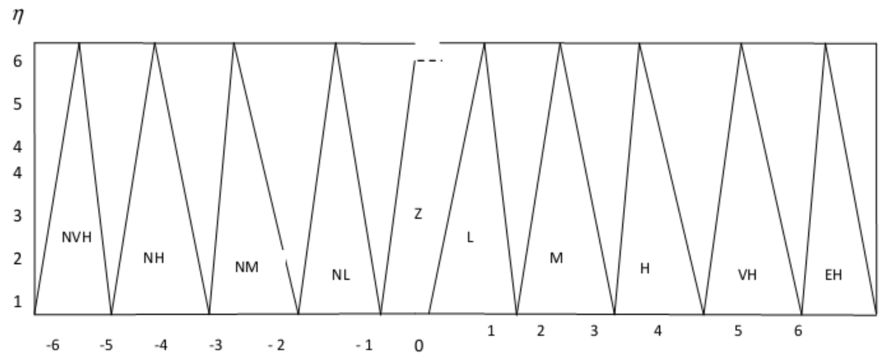

Fig. 5: Rate of change of congestion level (δC_N)

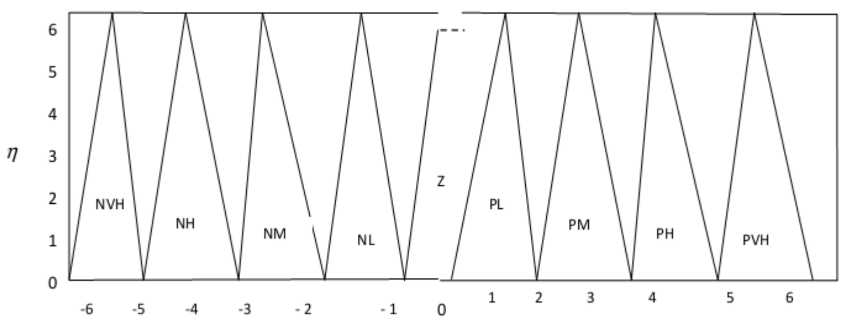

Fig. 6: Output membership function (W)

Table 3: All the inference rules of a complete set

$\delta C_N/C_N$	L	M	H	VH	EH
NVH	WPH	WPM	WPL	WZ	WNL
NH	WPM	WPL	WZ	WNL	WNM
NM	WPL	WZ	WZ	WNM	WNM
NL	WPL	WZ	WNL	WNM	WNH
Z	WZ	WNL	WNM	WNH	WNH
PL	WNL	WNL	WNM	WNH	WNH
PM	WNL	WNM	WNH	WNH	WNVH
PH	WNM	WNH	WNH	WNVH	WNVH
PVH	WNM	WNH	WNVH	WNVH	WNVH

Figure 4 to 6 represents the membership function for the input and output variables. For computational efficiency, triangulation functions are used to design membership function that gives optimal result.

In Table 1, linguistic variable for input C_N and in Table 2 δC_N is shown. The inference rules are given as below:

If C_N is L and δC_N is NVH then W is WPL
If C_N is M and δC_N is NVH then W is WZ

where, W is the defuzzified output from the controller. L, M and NVH are input membership function for associated fuzzy subsets and WPL and WZ are the output membership functions. A given pair of values of C_N and δC_N may each be potential members of two fuzzy subsets with the same or conflicting possibilities and hence, more than one rule may apply, as given in the examples, when C_N may be in the fuzzy subsets associated with L and M. The defuzzified output comprises of same linguistic variable as δ C_N, as given in Table 2, but abbreviated with a prefix of W, e.g., WZ(zero). All the inference rules of a complete set is given in Table 3.

The center of gravity technique is used for defuzzification. The following Eq. (11) maps the input to the output of the controller:

$$CT = \frac{\sum_{i=1}^{R} S_i G_i}{\sum_{i=1}^{R} G_i} \tag{11}$$

where, R represents the number of rules S_i represents the value of the output for rule *i* and G_i represents the conditional weight of the ith output membership function. In particular, S_i is the value at the middle of the range of data values that are potential members of the *ith* fuzzy subset. G_i is the area under the ith output membership function, concise by the minimum possibility (which may be zero) of membership of C_N and δC_N in the two membership function of the *ith* rule. As more than one rule may well apply, hence Eq. (11) is in the form of a weighted average of the values appearing from the different rules that are valid, with outputs of zero for those remaining fuzzy subsets for which the data value is not a member.

The control signal CT, as precise in Eq. (11), is normalized to the range (0, 1], subject to a minimum lower bound. For input bitrate B_{in}, the target output bitrate is $B_{out} = CT.B_{in}$ through multiplication. In steady state, to attain sending rate B_{out}, the quantization scale of the transcoder is directly proportional to CT and even to the dynamic range of available quantizers. Figure 7 represents the architecture of video streaming.

Optimization framework of buffer level: After obtaining the new streaming rate, which is estimated from the output of FLC, the application readjust the streaming rate to the obtained new stream rate to maintain the buffer level.

Figure 8 represents a typical optimization cycle which shows the buffer level variations in its two time stages, which represent the time spent by the user under the coverage of intermittent and universal network. In the Fig, σ_q represents the buffer level that relate to the Initial playout latency. In general, playout latency is employed to assure the presence of ample media in the application buffer to start media playout. This delay is calculated to avoid successive playout disruption due to buffer depletion caused by channel condition degradation. Therefore, the streaming strategies are designed in order to maintain the buffer level above this threshold. In addition, σ_q represent a buffering threshold whose value is optimized in the proposed framework. Also, y_0, y_1 and y_2 represent the buffer level at the initiation of the optimization cycle, on existing the cheap network and at the termination of the optimization cycle respectively.

The optimization cycle can be explained in the following steps:

First stage of the optimization cycle:

Step 1: First, the application stream the channel at a rate that is greater than the nominal stream rate, that means, $w_i > w_0$.

Step 2: This implies that, buffer level enhances at a rate of $w_i - w_0$ till the buffer level attain the buffering threshold.

Step 3: At this particular instance, the application re-modify the streaming rate to new obtained streaming rate from the output of FLC, that means, w_0, to manage the buffer level till mobile terminal exist the hotspot.

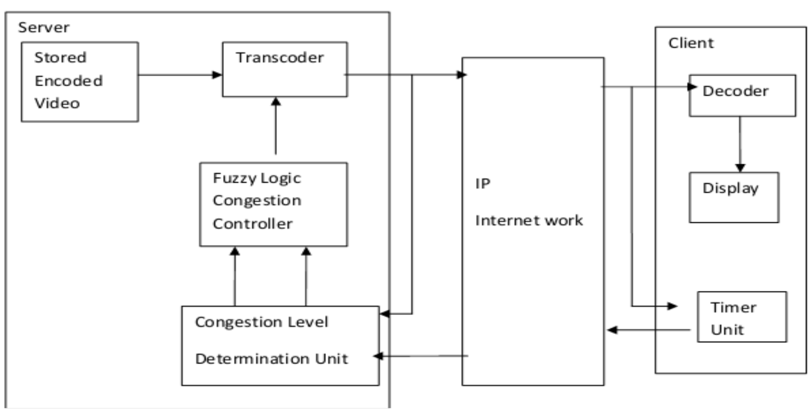

Fig. 7: Video streaming architecture

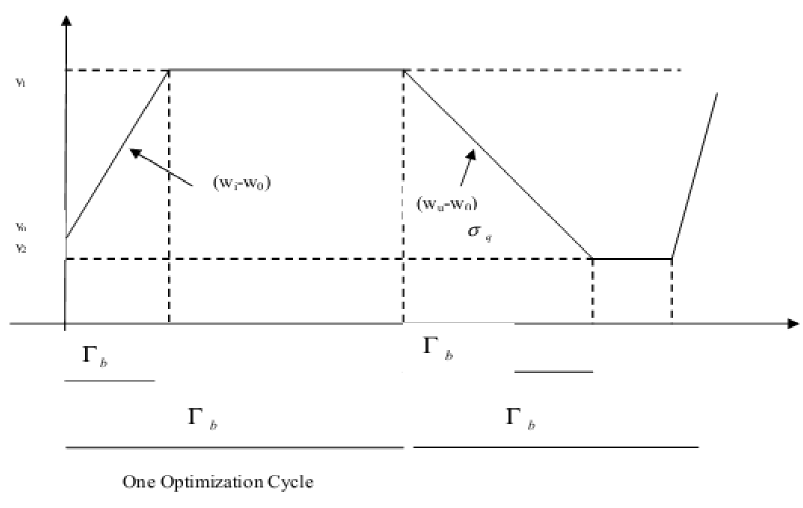

Fig. 8: Optimization cycle

Second stage of optimization cycle:

Step 4: The application in this stage depends on the buffered data by adjusting streaming rate w_u, such that $0 \leq w_u < w_0$.
Step 5: Hence, the buffer is used up at a rate of w_0 - w_u. As, the buffer level move towards the QoS threshold, σ_q in the expensive network, the application in the network proactively adjusts the streaming rate to the minimal stream rate.

Hence, the streaming strategy will not cause any playout disruption request is neglected by the 3G network.

Towards this aim, the model of streaming plan under the proposed framework includes a number of parameters such as:

- Buffering threshold σ_q
- Streaming rate w_i and w_u

Each plan allocates a subset of these parameters as static values and optimizes the target values of the parameters in the complementary subset. The proposed optimization framework is developed on the foundation of stochastic optimization technique which mainly includes two main steps:

- Developing deterministic form of the problem
- Solving the problem with well-known optimization tools.

Specifically, the streaming parameter can be optimized, once the user starts any session in a WLAN or when it is handed over to a WLAN during running time of the session. The initiation of the session in the following network includes:

- In case, the session initiates in the cheaper network, then the streaming policy parameter is optimized with the help of residual time distribution of cheaper network that means $\overline{\Gamma}_i$
- In case, the streaming session initiate in the residual network, then the application re-adjust the server streaming rate to the obtained nominal rate, i.e., w_0 in order to decrease the session cost and satisfy QoS constraint.
- Lastly, as the user roam across the two networks, the design parameters are computed online according to different operating condition such as service cost and buffer level.

Greedy Streaming to Threshold and Stop (GSTS) Algorithm: Here, the GSTS uses the static values for streaming rates w_i and w_u and after that optimizes the buffering threshold value. The algorithm can be explained in the following steps:

Step 1: In case of cheaper network, application adjust the streaming rate to the maximal available bandwidth, that means $w_i = w_{i\max}$
Step 2: In case of streaming rate, GSTS stop the streaming process, i.e., $w_u = 0$
Step 3: The buffering threshold value is optimized in order to decrease the session based on the optimization framework

The main objective behind considering the buffering threshold is to decrease the streaming monetary cost in the cheaper network after assuring an ample amount of streamed media in the application buffer.

Algorithm for OMSCC (Optimized Multimedia Streaming and Congestion CONTROL:

1. Development of two-tier heterogeneous model
2. //Estimation of new Streaming rate in a two-tier heterogeneous system//
3. Estimation of congestion level detection
4. //Fuzzy Logic Congestion Controller//
5. Determine fuzzy sets
6. Consider C_N and δC_N
7. Apply inference rule
8. If C_N is L and δC_N is NVH then W is WPL
9. If C_N is M and δC_N is NVH then W is WZ
10. Defuzzify to obtain the output
11. Based on the output, estimate new streaming rate
12. Application readjust to the new streaming rate to maintain buffer level
13. Threshold value of the buffer level is optimized using GSTS Algorithm
14. //GSTS Algorithm//
15. In cheaper network, application adjust the streaming rate to maximum available bandwidth $w_i = w_{i\max}$
16. In expensive network, GSTS stop streaming process, $w_u = 0$.

SIMULATION RESULTS

Simulation parameters: We use NS2 [ ] to simulate our proposed Optimized Multimedia Streaming and Congestion Control (OMSCC) protocol. We use the IEEE 802.11 for wireless LANs as the MAC layer protocol. It has the functionality to notify the network layer about link breakage. In our simulation, the packet sending rate is varied as 500, 750, 1000 and 1250 Kb, respectively. The area size is 600 m×600 m^2 region for 50 sec simulation time. In our simulation, the BufferSize is varied from 300, 350, 400 and 450 bytes,

Table 4: Simulation parameters

No. of nodes	5
Area	600×600
MAC	802.11
No. of wired nodes	6
No. of base nodes	5
Simulation time	50 sec
Traffic source	CBR and Exp
Rate	500, 750, 1000 and 1250 Kb/s
Packet size	512 bytes
Propagation	Two ray ground
Buffer size	300, 350, 400, and 450 Kb
Speed	5 m/s

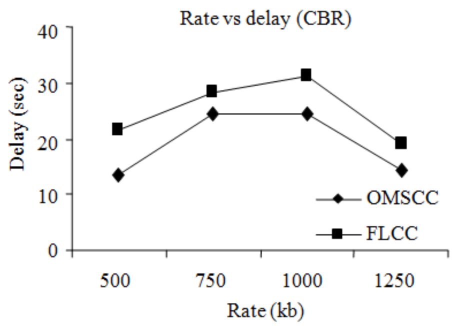

Fig. 9: Rate Vs delay

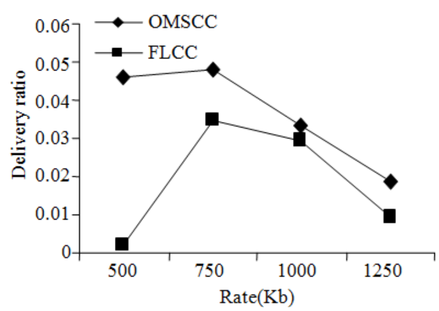

Fig. 10: Rate Vs delivery ratio

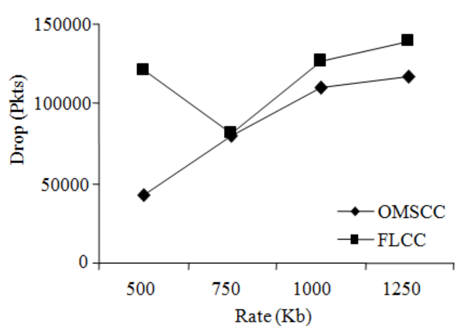

Fig. 11: Rate Vs drop

respectively. The simulated traffic is Constant Bit Rate (CBR) and Exponential (Exp).

Our simulation settings and parameters are summarized in Table 4.

Performance metrics: We evaluate performance of the new protocol mainly according to the following parameters. We compare the (FLCC) [ ] protocol with our proposed CMSCC protocol.

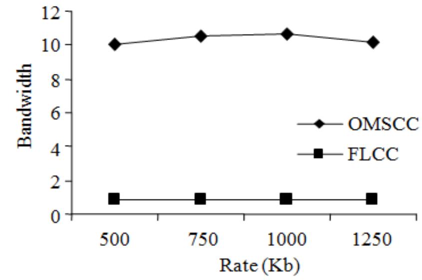

Fig. 12: Rate Vs bandwidth

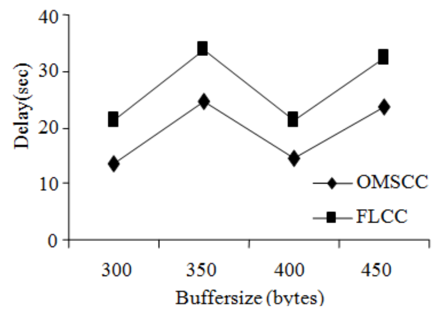

Fig. 13: Buffersize Vs delay

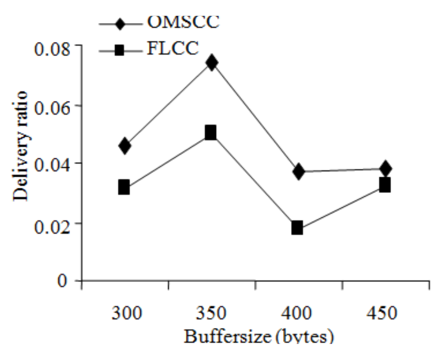

Fig. 14: Buffersize Vs delivery ratio

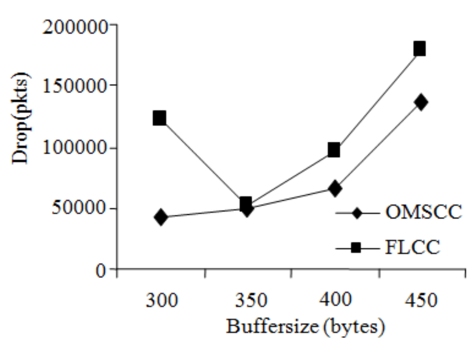

Fig. 15: Buffersize Vs drop

Average packet delivery ratio: It is the ratio of the number of packets received successfully and the total number of packets transmitted.

Average end-to-end delay: The end-to-end-delay is averaged over all surviving data packets from the sources to the destinations.

Received bandwidth: The received bandwidth is the amount of data that can be sent from the sources to the destination.

Packet drop: It is the number of packets dropped during the data transmission.

Results and analysis: The simulation results are presented in the next section.

Scenario 1: Inter handoff for CBR traffic:
Based on rate: In our first experiment we are varying the rate as 500, 750, 1000 and 1250 kb/s, respectively.

Figure 9 to 12 show the results of delay, delivery ratio, packet drop and bandwidth for the packet sending rate 500, 750, 1000 and 1250 in CMSCC and FLCC protocols. When comparing the performance of the two protocols, we infer that CMSCC outperforms FLCC by 24.2% in terms of delay, 47% in terms of delivery ratio, 24% in terms of drop and 92% in terms of bandwidth.

Based on buffer size: In our second experiment we are varying the Buffer Size as 300, 350, 400 and 450 bytes, respectively.

Figure 13 to 16 show the results of delay, delivery ratio, packet drop and bandwidth for the buffer size 300, 350, 400 and 450 bytes, respectively in CMSCC and FLCC protocols. When comparing the performance of the two protocols, we infer that CMSCC outperforms FLCC by 31% in terms of delay, 34% in terms of delivery ratio, 32% in terms of drop and 91.2% in terms of bandwidth.

Scenario 2: Inter handoff for exponential traffic:
Based on rate: In our third experiment we are varying the rate as 500, 750, 1000 and 1250 kb/s, respectively.

Figure 17 to 20 show the results of delay, delivery ratio, packet drop and bandwidth for the packet sending rate 500, 750, 1000 and 1250 in CMSCC and FLCC protocols. When comparing the performance of the two protocols, we infer that CMSCC outperforms FLCC by 16% in terms of delay, 34% in terms of delivery ratio, 38.2% in terms of drop and 91% in terms of bandwidth.

Based on Buffersize: In our fourth experiment we are varying the BufferSize as 300, 350, 400 and 450 bytes.

Figure 21 to 24 show the results of delay, delivery ratio, packet drop and bandwidth for the buffersize 300, 350, 400 and 450 in CMSCC and FLCC protocols. When comparing the performance of the two protocols, we infer that CMSCC outperforms FLCC by 31% in terms of delay, 29% in terms of delivery ratio, 46% in terms of drop and 91% in terms of bandwidth.

Scenario 3: Intra Handoff for CBR traffic:
Based on rate: In our fifth experiment we are varying the rate as 500, 750, 1000 and 1250 kb/s, respectively.

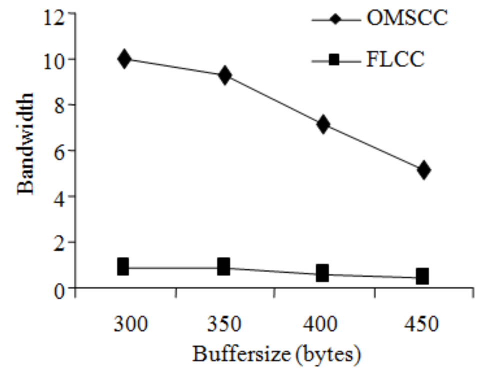

Fig. 16: Buffersize Vs bandwidth

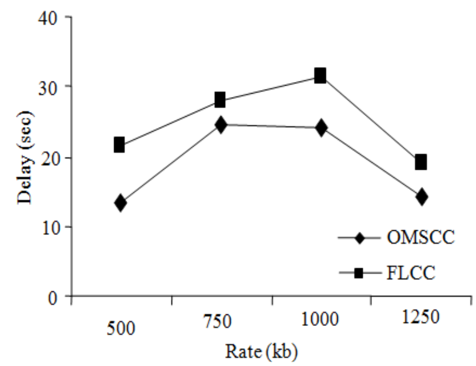

Fig. 17: Rate Vs delay

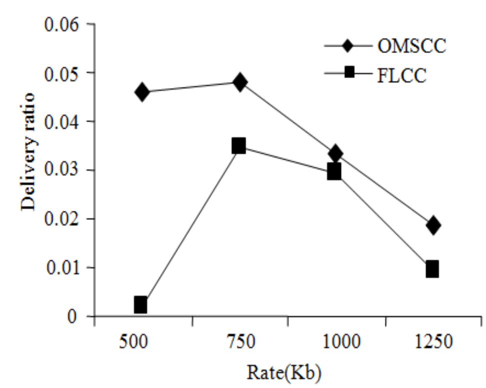

Fig. 18: Rate Vs delivery ratio

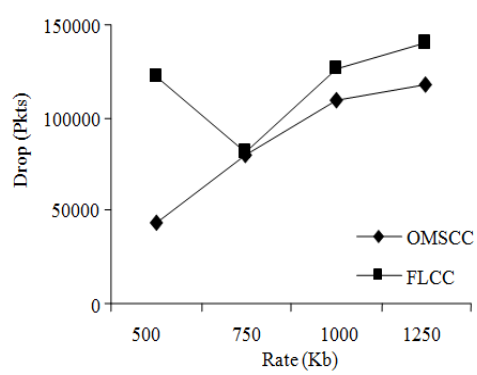

Fig. 19: Rate Vs drop

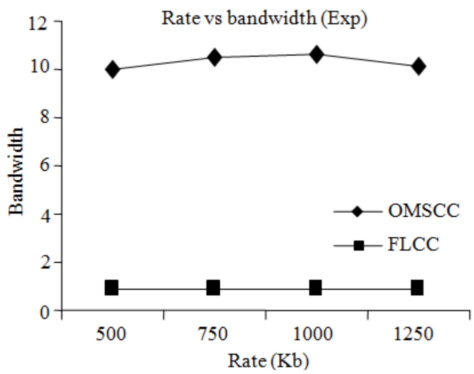

Fig. 20: Rate Vs bandwidth

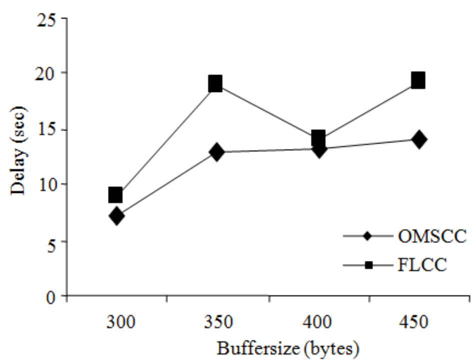

Fig. 21: Buffersize Vs delay

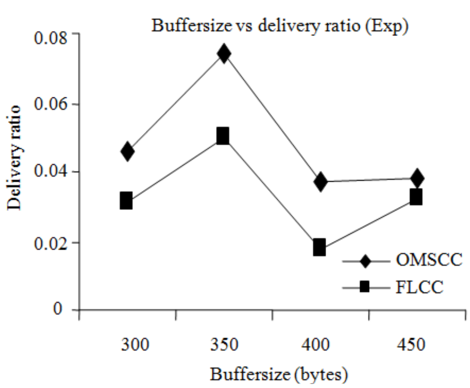

Fig. 22: Buffersize Vs delivery ratio

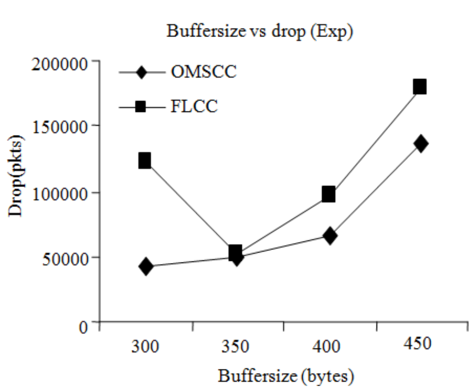

Fig. 23: Buffersize Vs drop

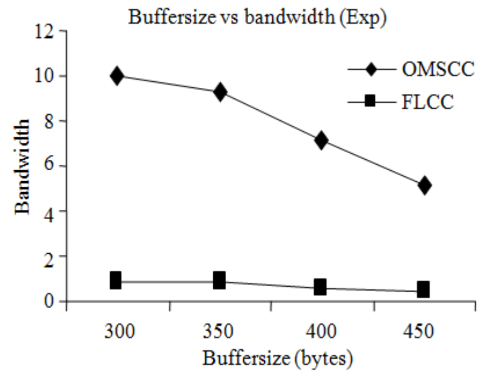

Fig. 24: Buffersize Vs bandwidth

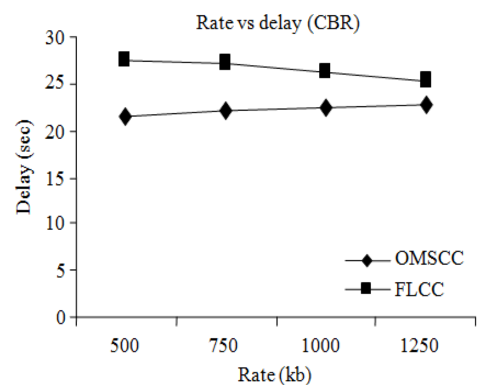

Fig. 25: Rate Vs delay

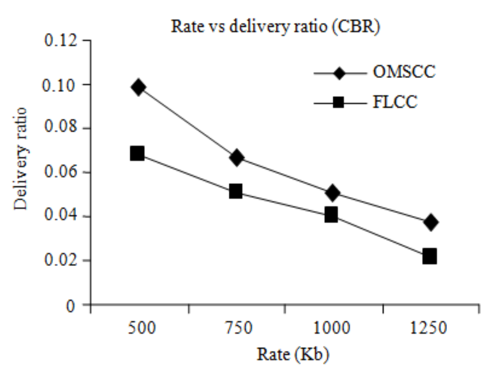

Fig. 26: Rate Vs delivery ratio

Figure 25 to 28 show the results of delay, delivery ratio, packet drop and bandwidth for the packet sending rate 500, 750, 1000 and 1250 in CMSCC and FLCC protocols. When comparing the performance of the two protocols, we infer that CMSCC outperforms FLCC by 16% in terms of delay, 30% in terms of delivery ratio, 8% in terms of drop and 90% in terms of bandwidth.

Based on BufferSize: In our sixth experiment we are varying the BufferSize as 300, 350, 400 and 450 bytes, respectively.

Figure 29 to 32 show the results of delay, delivery ratio, packet drop and bandwidth for the buffersize 300,

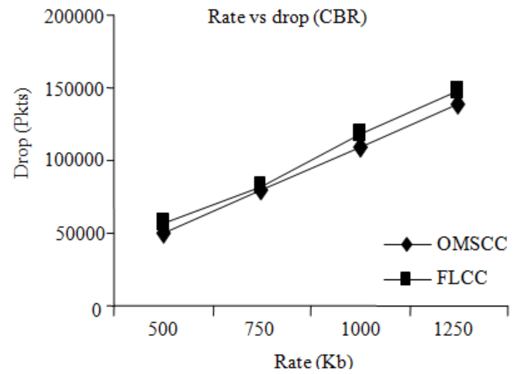

Fig. 27: Rate Vs drop

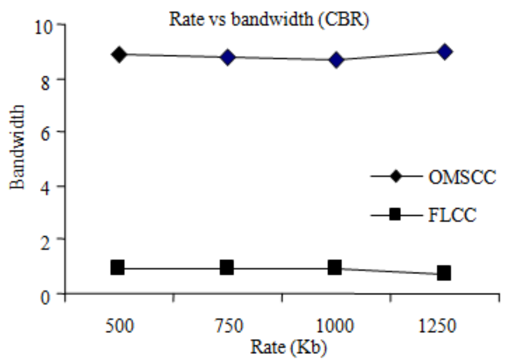

Fig. 28: Rate Vs bandwidth

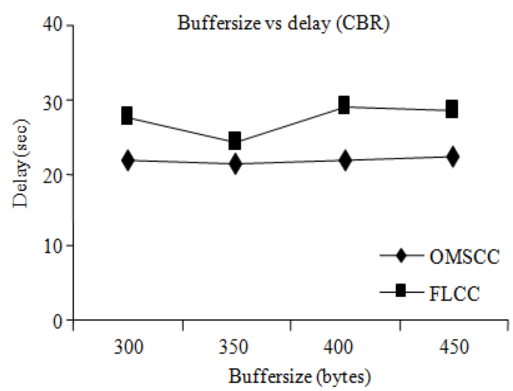

Fig. 29: Buffersize Vs delay

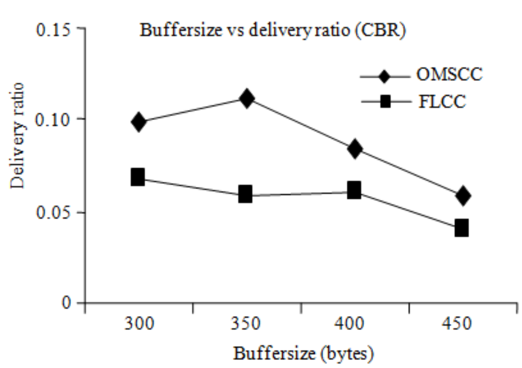

Fig. 30: Buffersize Vs delivery ratio

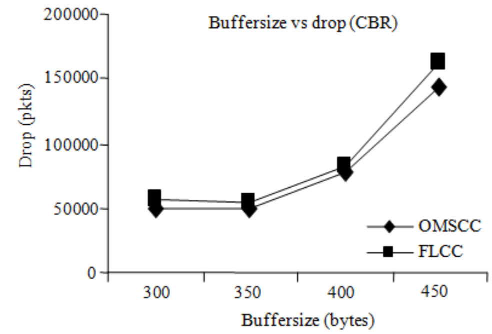

Fig. 31: Buffersize Vs drop

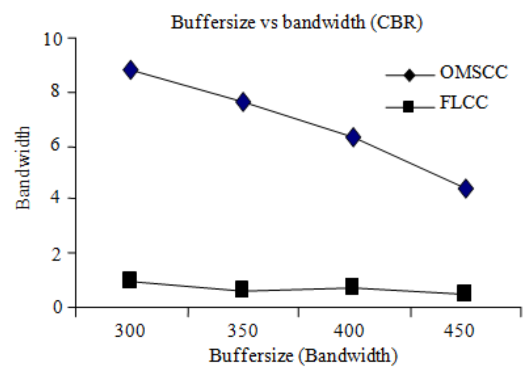

Fig. 32: Rate Vs bandwidth

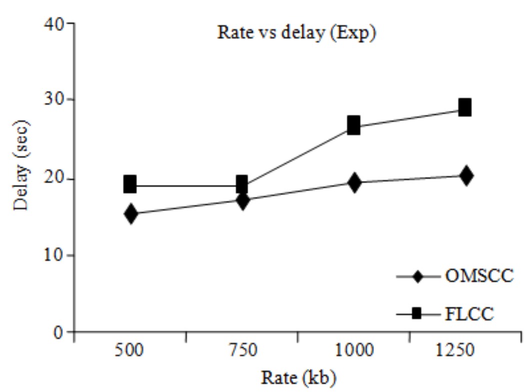

Fig. 33: Rate Vs delay

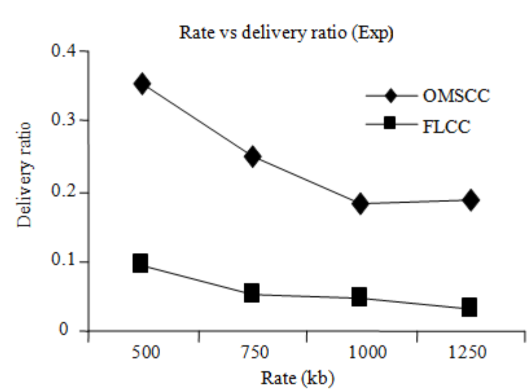

Fig. 34: Rate Vs delivery ratio

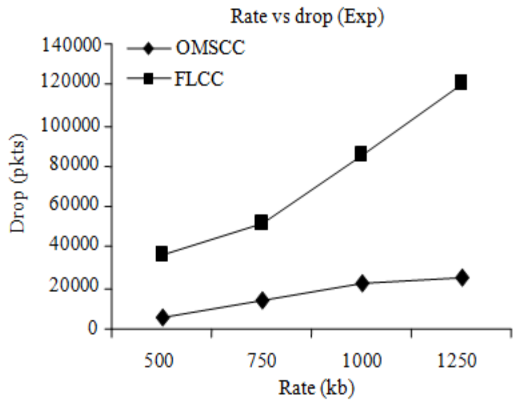

Fig. 35: Rate Vs drop

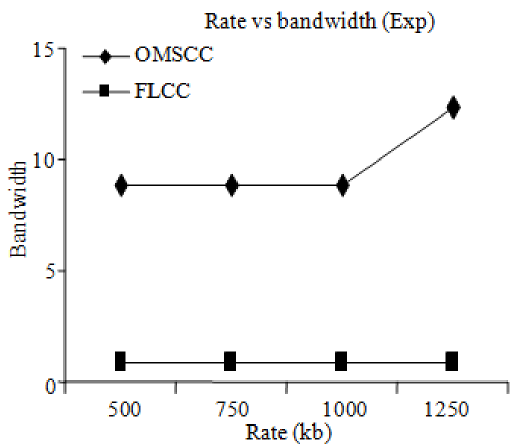

Fig. 36: Rate Vs bandwidth

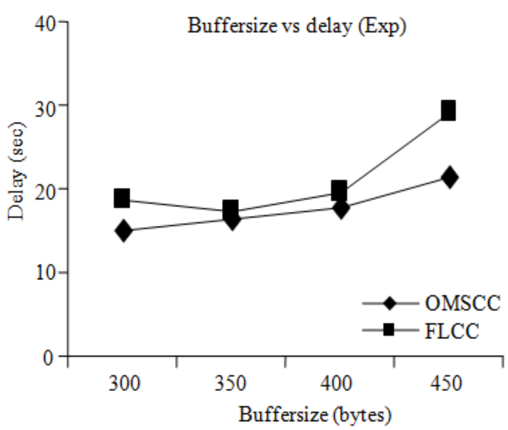

Fig. 37: Buffersize Vs delay

350, 400 and 450 in CMSCC and FLCC protocols. When comparing the performance of the two protocols, we infer that CMSCC outperforms FLCC by 20% in terms of delay, 35% in terms of delivery ratio, 10% in terms of drop and 90% in terms of bandwidth.

Scenario 4: Intra handoff communication for exponential traffic:

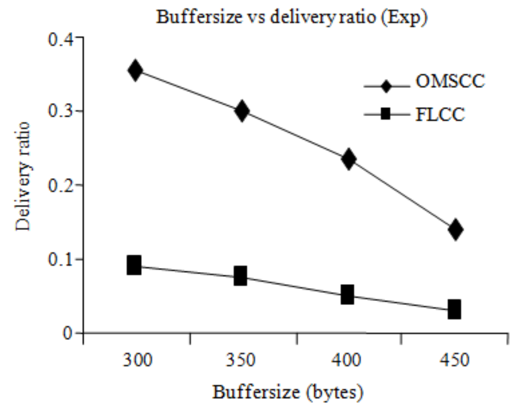

Fig. 38: Buffersize Vs delivery ratio

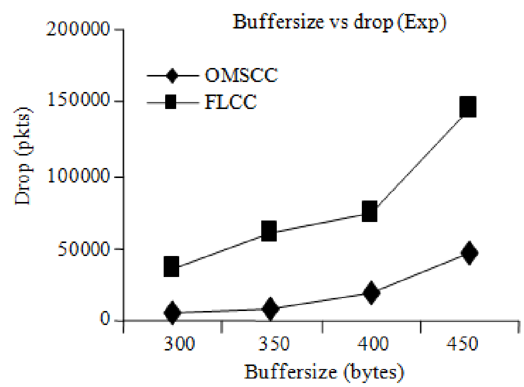

Fig. 39: Buffersize Vs drop

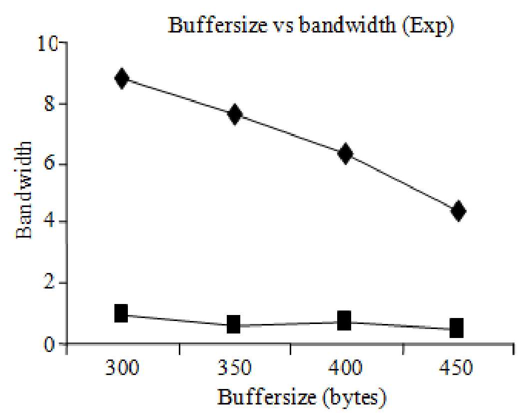

Fig. 40: Buffersize Vs bandwidth

Based on rate: In our seventh experiment we are varying the rate as 500, 750, 1000 and 1250 kb/s.

Figure 33 to 36 show the results of delay, delivery ratio, packet drop and bandwidth for the packet sending rate 500, 750, 1000 and 1250 in CMSCC and FLCC protocols. When comparing the performance of the two protocols, we infer that CMSCC outperforms FLCC by 22% in terms of delay, 78.2% in terms of delivery ratio, 78.2% in terms of drop and 91% in terms of bandwidth.

Based on BufferSize: In our eighth experiment we are varying the BufferSize as 300, 350, 400 and 450 bytes, respectively.

Figure 37 to 40 show the results of delay, delivery ratio, packet drop and bandwidth for the buffersize 300, 350, 400 and 450 in CMSCC and FLCC protocols. When comparing the performance of the two protocols, we infer that CMSCC outperforms FLCC by 15.2% in terms of delay, 77% in terms of delivery ratio, 79% in terms of drop and 90.4% in terms of bandwidth.

CONCLUSION

In this study, an Optimized Multimedia Streaming and Congestion Control technique for WLAN-3G Network is proposed. Here, a two-tier integrated heterogeneous wireless network including WLAN and 3G cellular are considered. The new streaming rate is estimated by using the FLC controller. Congestion level determination unit is used to estimate congestion level and the rate of change of congestion level which is considered as input to the FLC. Based on the output of FLC, streaming rate is estimated. The application readjusts the streaming rate to the new streaming rate in order to maintain the buffer level. Also, the threshold value for buffer level is optimized by using the Greedy Streaming to Threshold and Stop (GSTS) algorithm in order to reduce the streaming monetary cost.

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