PERFORMANCE OF A LINK CONTROL PROTOCOL FOR LOCAL WIRELESS MULTIMEDIA COMMUNICATIONS

Pallapa Venkataram, Anirban Roy* and **A. Chockalingam**†

Department of Electrical Communication Engineering, Indian Institute of Science, Bangalore 560012, INDIA.
e-mail: pallapa@ece.iisc.ernet.in
†Qualcomm, Inc., 6455 Lusk Boulevard, San Diego, CA 92121, U.S.A.
e-mail: achockal@qualcomm.com URL: http://cwcserv.ucsd.edu/~choks

Abstract— **The convergence of wireless communications and multimedia services throws open interesting research topics to address. The variety of media-specific QoS requirements and the maintainence of synchronization among different streams to be presented at the receiver in multimedia applications demand additional functionalities at the data link layer. In this paper, we present an approach for efficient link control and management of multimedia applications in next generation wireless networks. We advocate the use of multiple connections, one for each traffic component of the application, as an effective solution for transport of multimedia applications over wireless. The effectiveness of this scheme is guaranteed by providing a new link control technique that works on top of a PRMA-type access protocol and jointly manages different streams of a multimedia application. We show that the proposed scheme performs well, especially during periods in which the system traffic load is high, and reacts well to degradation of multimedia service quality, both in terms of QoS parameters and synchronization of its traffic components.**

I. INTRODUCTION

First generation wireless systems, which primarily provide analog voice service, are widely in use worldwide. Second generation systems support digital voice/data traffic; some of these systems are already deployed or undergoing deployment. Third generation wireless networks will ultimately carry multimedia traffic that are characterized by combination of different informaion streams of diverse nature (e.g., voice, video, image, data) [1]. Some of the salient features of multimedia applications are high speed and changing bit rates (periodic and bursty arrivals), several virtual connections over the same access, synchronization of different information streams, and various service/delivery requirements (QoS) [2]. It is clear that next generation wireless networks will be required to interface with much higher bandwidth fiber-based wired networks, possibly carrying B-ISDN/ATM type traffic, which poses interesting management and control issues like admission control, resource sharing and link contol [1],[3]. The objective is to provide "seamless wireless communication" to users, irrespective of channel impairments [4], [5] .

Multimedia applications have varying bandwidth requirements. A media access control (MAC) protocol capable of assigning multiple slots, based on demand and QoS requirements, is thus a natural choice. It is also desired that the protocol supports statistical multiplexing to improve channel utilization exploiting bursty nature of the traffic. In this paper, we propose a link control protocol which is a combination of a suitable MAC protocol and a priority based link management and control functionality to support multimedia streams over wireless. A reservation type access protocol capable of assigning multiple slots upon request is employed at the MAC layer. On the top of the MAC layer, a burst level admission control and link management function, based on a two-level priority mechanism, is designed to identify and associate together different components of the multimedia application and suitably handle them as parts of an integaretd flow. On a "call-basis," a static priority is specified by the user to each component of the multimedia application, which is used during link activation. A second level of priority (dynamic priority) is assigned by the system on a "spurt-basis" according to the traffic of all components of the application, and is used during bandwidth allocation and reservation queue management.

The rest of the paper is organized as follows. In Section 2, we describe the MAC protocol. In Section 3, the proposed priority based link control mechanism which resides on top of the MAC layer is described. Section 4 provides the system model and the simulation results of the system performance. Conclusions and topics for further investigation are given in Section 5.

II. MAC PROTOCOL

Existing access protocols like PRMA and its variants basically focus on the transport of voice/data, and are only partially interested in the problem of transporting real-time multimedia traffic components over the radio interface [6],[7].

For multimedia transport and management, the dynamic reservation multiple access (DRMA) protocol in [8] represents a good starting point, as it has been explicitly conceived with the aim of adapting traditional reservation multiple access to multimedia traffic needs, and sharing of buffer resources and channel bandwidth. Here, we propose a MAC

*A. Roy is presently with Lucent Technologies India Ltd., Bangalore, India.

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protocol similar to the DRMA protocol. The choice of maintaining a complete separation between Reservation slots (R-slots) and Information slots (I-slots), and dynamically adapting the percentage of reservation bandwidth within a frame to traffic conditions coupled with a suitable bandwidth allocation strategy, is prefered in order to meet the QoS requirements of multimedia (integrated video/voice) services.

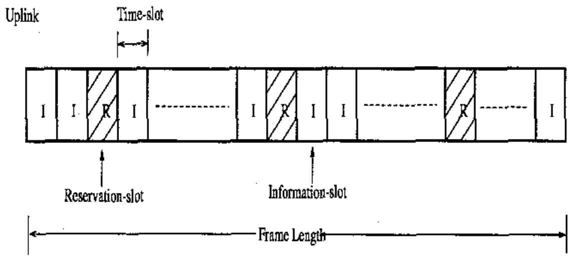

Fig. 1. Transmission Frame Format of the MAC Protocol

The transmission frame format of the proposed MAC protocol is shown in Figure 1. The frame format is mainly based on the structure of PRMA++, constituting a sequence of I-slots and R-slots. The total number of time slots in a frame is S. A maximum and minimum number of R-slots ($S_{r_{max}}$ and $S_{r_{min}}$) are defined. At each instant of time the actual number of activated (thus available for reservation) R-slots in the frame is S_r, such that $S_{r_{min}} \leq S_r \leq S_{r_{max}} < S$. All non-activated R-slots are used for transporting information, like I-slots. The positions for $S_{r_{max}}$ R-slots within the frame are fixed. They are scattered in a homogeneous manner along the frame, so that a new burst will not have to wait a long time to intercept an available R-slot. The model of the proposed access mechanism is sketched in Figure 2.

When a terminal has packets to transmit (i.e., start of a spurt is detected), it sends a reservation packet on the first available R-slot, with probability p_t (called permission probability as in S-ALOHA). If the reservation packet collides or is corrupted by channel impairments on the R-slot, a negative acknowledgment is sent from the base station on the downlink. Unsuccessful terminals retry to transmit their reservation packets with probability p_r (also known as retransmission probability), on the next free R-slot (generally $p_t \geq p_r$). What really makes our MAC protocol different from other reservation protocols, is the dynamic management of reservation bandwidth. That is, based on the traffic load variations (in terms of number of transmitting terminals) the base station is allowed to modify the number of active R-slots within the transmission frame.

It has been demonstrated in [8] that better performances compared to fixed number of active R-slots (i.e., fixed S_r) can be obtained by properly varying S_r. A reduction of S_r is preferred when traffic load increases and an increase in S_r is desired when the load decreases. The optimum number of active R-slots for a given channel load value has to be chosen according to a quality index. The proposed protocol is able to dynamically adapt itself, on a frame basis, when traffic load variations occur. Particularly, at the end

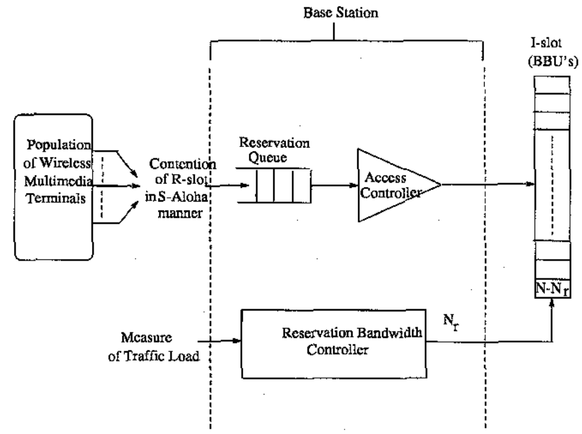

Fig. 2. Model of the Access Mechanism

of each frame, the base station runs an algorithm which ex amines the channel load and determines a suitable S_r valu This algorithm is computed by the Reservation Bandwidt Controller, as shown in Figure 2. An efficient and simpl technique to manage the variation of S_r by traffic load fluc tuations, is a "threshold-based strategy." According to thi strategy, the base station decides to change S_r when traffi load exceeds one of a set of fixed threshold values. Wit this approach, a persistent need to vary S_r in subsequer frames will arise when the load rapidly fluctuates around th thresold value, This phenomenon would create an undesire overload of signalling traffic on the downlink. In order t overcome this problem, an hysteresis control mechanism i introduced. Precisely, each threshold value is provided wit a *hysteresis margin*, as illustrated in Figure 3, which permit a change in S_r only when offered load shows a convincin trend of rising (when crossing L_r) or falling (when crossin L_f).

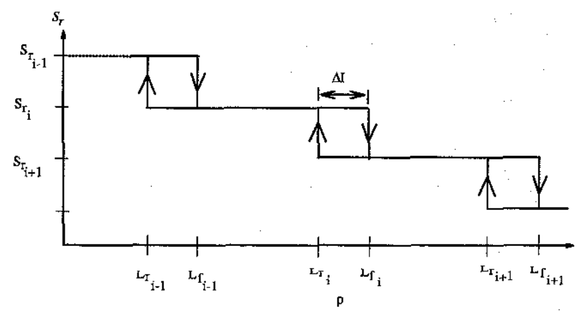

Fig. 3. Optimum number of active R-slots (S_r) versus traffic load (ρ)

III. Priority Based Link Control

The main components of the proposed link control mech anism which resides on top of the MAC protocol are a follows: a) bandwidth allocation, b) admission control us ing static priority, c) synchronization using dynamic priority, and d) scan and serve policy for radio resource management. See Figure 4.

A. Bandwidth Allocation

Several bandwidth allocation strategies, such as complete sharing (CS), complete partitioning (CP), and mutually restricted access (MRA), in broadband networks have been proposed and analyzed [9]. In these schemes, the transport capacity at network nodes is split into Basic Bandwidth Units (BBUs) that can be allocated to traffic connections. In the wireless context, each uplink frame can be seen as a channel in which each slot represents one BBU.

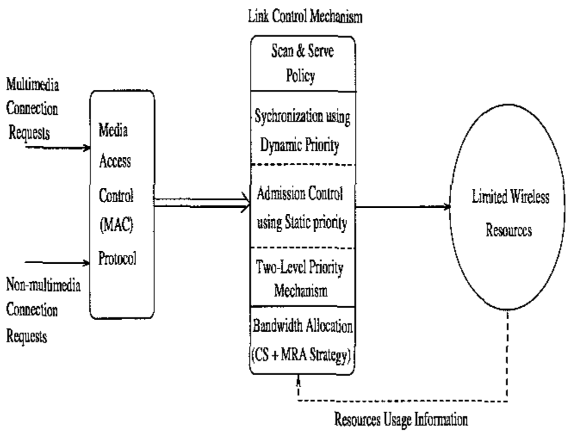

Fig. 4. Proposed Link Control Mechanism

Complete sharing (CS) of bandwidth without preemption for non-multimedia, and mutually restricted access (MRA) without overflow for multimedia services are promising strategies in a multimedia wireless environment [10]. In this study, we adopt CS for voice and MRA for video, in a wireless video-phone application. We enhance radio resource management by adopting the above bandwidth allocation strategies, and coupling them to new policies for reservation queue management at the base station.

B. Admission Control and Prioritized Access

We adopt an admission control policy based on a priority mechanism. On a "call-level", a *static priority* is defined by the wireless terminal for each component of its multimedia traffic, and it is used to adapt link activation procedures to the nature of multimedia traffic; it is also known as *call-level admission control.* The static priority is assigned on the basis of both qualitative (related to quality assessment based on subjective index) and quantitative (based on experienced activation frequency of each component) measures. Higher priority is assigned to those components which are considered more necessary than the others for the progression of the call, and thus handled better during the execution of control level procedures, like call set up. A second level of priority, namely, *dynamic priority*, is assigned on a "spurt-basis", according to the traffic activity of all the multimedia service components; it is also known as *burst-level admission control.* The dynamic priority is assigned to a spurt accessing the radio channel, according to the activity or inactivity of the other components within the same flow with a lower static priority. The joint use of these priorities also permits the system to enhance the degree of synchronization among the different traffic components of the same flow.

C. Synchronization using Dynamic Priority

A multimedia call is composed of separate connections which can be switched on/off at different instants of time during the progression of the call. Thus a multimedia stream can be seen as a sequence of blocks (frames) of units (slots) of different nature (voice, video, data), delay constraints and integrity requirements. The type of the units comprising a block change dynamically during the call. The concern is to aid units belonging to the same block to access the air interface 'at the same time.' In order to facilitate the acquisition of radio resources to a component delayed in the buffer when other units of the same component are being already transmitted over the air, we propose an approach that exploits the *dynamic priority* concept, as illustrated by the following example.

Consider wireless video-voice service. During call setup the terminal can assign a higher static priority to the voice component than the video component, as it is more acceptable to deliver voice without video instead of a video sequence without its associated voice. Let us say that a talk spurt attempts access later during the call and the request is queued in the RQ waiting for resources to be available. If the video component of the same service is already active, the base station increases the priority of voice in the RQ in order to enable voice to get faster allocation of resources so that the overall quality of the service can be maintained. Similar approach is also possible when more than two service components are involved.

D. Scan and Serve Policy

We can consider slot allocation procedure with two types of traffic – wideband traffic (requiring multi-slot allocation) and narrowband traffic (requiring single slot allocation) – as a multi-server queuing system making use of a FIFO queue with reservation requests, with S identical, parallel servers where requests are served to completion. An arriving request which cannot be served immediately is 'put on hold' in the queue for delayed service and can wait for a specified time and is dropped thereafter. The important property of this class of queues is that a given request can enter service only when all its requested number of servers are available. This implies that servers maybe idle while requests are waiting. This queuing system thus calls for a queue service policy that utilize the servers efficiently, while providing equitable access to different types of requests.

In the absence of any specific service order control, different traffic types access the S-server channel in their order of arrival or FIFO order. It thus appears, at first glance, that under the FIFO policy a 'large' request (wideband call) waiting at the head-of-the-line (HOL), may adversely block a large number of 'smaller' (narrowband call) requests who could get service sooner. This can be construed to mean that the FIFO policy is inefficient as far as the servers' utilization/throughput is concerned. However, we can observe that any attempt to allow narrowband requests to overtake wide-band requests will result in indefinitely postponing the wide-band requests from accessing the channel. Now, suppose we ignore this fairness issue and want to overcome the FIFO disadvantage of having a large request wait at the head-of-the-line while servers are unutilized. In the video-phone example, by making use of a FIFO scheme, a video reservation request which occupies the head of the queue could block following voice traffic reservation requests when fewer slots than needed are available. This can be achieved in two ways. One way would be to scan the queue of waiting requests and insert into service those sets of 'smaller' requests, if any, that best fit into the currently available portion of the channel. We refer to this policy as *scan and serve policy*. Another way would be to use 'preemptive priority'. Since preemptive policy dose not suit well in wireless multiple access situations, we prefer adopt the *scan and serve policy.*

IV. Simulation and Results

The performance of the proposed link control protocol has been evaluated through simulations for the wireless video-phone application. The following parameter values, applicable to video-voice service, are employed in the simulations. The average length of talk spurts $T_{on,voice} = 1$ sec, and the average length of silence gaps $T_{off,voice} = 1.35$ sec [6]. Voice activity rate, $\gamma_{voice} = \frac{T_{on,voice}}{T_{on,voice}+T_{off,voice}} = 0.4255$. For video traffic, the activity rate, $\gamma_{video} = \frac{T_{on,video}}{T_{on,video}+T_{off,video}}$, is taken to be 0.615 [12]. The frame length is 6 msec (T_f=6 msec). Each frame consists of 200 time slots ($S = 200$). The aggregate bit rate on the wireless link is taken to be 2 Mb/s. Voice at 8 Kb/s rate and low resolution video at 32 Kb/s rate are assumed to come from mono-media (non-multimedia, nm) as well as multimedia (mm) terminals for channel access. With these parameters, voice and video sources will generate one packet per frame and four packets per frame, respectively. In fact, we have chosen the frame duration to match the voice source rate. The time slots are of size 60 bits and packets are of size 48 bits. So, the extra 12 bits in each slot account for MAC layer protocol overhead. The slot and packet sizes chosen here are compatible with those suggested for ATM cell format, which suggests 48 Byte ATM cell payload or a suitable integer sub-multiple of a cell, e.g., 24,16,..,6 bytes, as the basic unit of data within the wireless network [13]. It is assumed that the uplink is error-free and pakcet failures are only due to collision. Further, instantaneous and error-free feedback is assumed on the dowlink.

A time-of-expiry (TOE) based queue, discipline where packets are dropped if their waiting time within the queue exceeds a maximum predefined delay value, is used. An upper bound of 32 msec (as in [6]) has been chosen as the acceptable access delay for voice (i.e., time spent from the instant a talk spurt is generated to its first packet transmission over the channel). Similarly, an upper bound of 300 msec (as in [14]) has been adopted as the access delay acceptable for video traffic. Also, in the contention process, if the reservation request encounters a collision, the request is repeated by the terminal until a maximum waiting time T_{max} is reached (as in [15], W_{max} is taken to be $3\times$maximur acceptable delay). At this point, the request is considere to be 'blocked', and the packets of the corresponding burs are assumed to be lost. In the ALOHA contention process the permission probability for transmitting a new reservatioı request is $p_t = 0.8$. Once the terminal encounters a col lision, it retransmits the request with retransmission proba bility $p_r = 0.4$ until success or blocked, whichever occur earlier. Acceptable average packet dropping probability o voice and video services assumed are 10^{-2} [6] and 10^{-} [14], respectively.

A. Quality Index

In order to compare the performance and express the qual ity improvement of the proposed protocol (we refer to thi as an *enhanced* protocol adopting priority based strategie for bandwidth allocation, admission control, synchroniza tion, scan and serve policy), we define the following Qualit Metric (Q) for the video-voice service.

$$Q = \phi \frac{1 - \xi' P_{voice,mm} - (1-\xi'')P_{voice,nm}}{\frac{1}{T_{on,voice}}[T_{on,voice} + \xi' D_{voice,mm} + (1-\xi'')D_{voice,nm}]}$$
$$+(1-\phi)\frac{1 - \xi' P_{video,mm} - (1-\xi'')P_{video,nm}}{\frac{1}{T_{on,video}}[T_{on,video} + \xi' D_{video,mm} + (1-\xi'')D_{video,nm}]}$$

where $\xi' = \frac{\psi}{\phi}$ and $\xi'' = \frac{\psi}{1-\phi}$. ϕ is the percentage voice load from both multimedia (mm) as well as non-multimedia (nm), and ψ is the percentage multimedia load. P_{voice} and D_{voice} are average packet drop probability and delay of voice packets. P_{video} and D_{video} are average packet drop probability and delay of video packets achieved in the system. Note that the suffix mm refers to multimedia traffic and nm refers to non-multimedia traffic.

B. Results

Results obtained from simulation runs over 100 minutes of real-time are highlighted in this subsection. A total offered load of 0.8 is considered. The effects of an increasing percentage of multimedia traffic has been evaluated and the observed delay and packet loss behavior is plotted in Figures 5 to 8. Our aim is to take a constant total load, change the composition of traffic (multimedia versus non-multimedia) in the system, and observe the effects of increasing multimedia component in the traffic. We have taken the voice and video percentage in the whole traffic (multimedia and non-multimedia) as $\phi = 56\%$ voice and 44% video. The number of R-slots in the frame considered for a constant total load of 0.8 is 10, which optimizes MAC protocol behavior [8].

We have plotted the performance of the proposed protocol (shown as the *enhanced protocol* in Figures 5 to 8) as well as the protocol without the enhancements (PRMA equivalent). Also, the performance plots for both multimedia and non-multimedia components of the traffic are shown. Figure 5 shows the delay performance for voice traffic and Figure 6 shows the packet loss performance for voice traffic. Similarly, Figure 7 and 8 shows the delay and packet loss performance for the video traffic.

It can be noted from the plots that the performance of the overall system performance remains almost flat for a per-

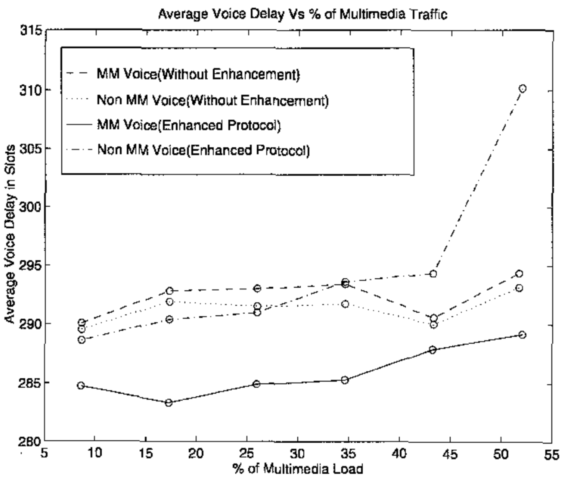

Fig. 5. Average *voice* delay versus percentage of multimedia traffic load

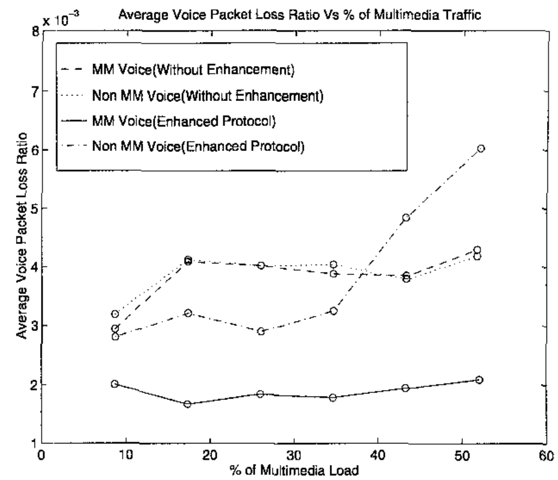

Fig. 6. Average *voice* packet loss ratio versus percentage of multimedia traffic load

centage of multimedia traffic upto 35% and then begins to degrade slowly. The enhanced protocol is seen to provide the best delay and packet loss performance for multimedia (mm) voice and video for all the percent multimedia loads considered. Thus the proposed protocol provides a clear improvement of the service quality expressed in terms of both average delay and packet loss probability of multimedia voice component (MM Voice in Figures 5 and 6) as well as multimedia video component (MM Video in Figures 7 and 8). The non-multimedia (nm) components, however, suffers little quality degradation at high percentages of multimedia load (NM Voice and NM Video in Figures 5 to 8).

Figure 9 shows the variation of the qulaity index Q (defined in Section 5) as a function of the percentage multimedia load. Here again, the proposed protocol is seen to result in significant improvement in service quality compared to the protocol without enhancement. Also, the enhanced link control protocol imparts more stable quality-of-service of the system and allows only a graceful degradation in the performance under increasing multimedia load. It clearly emerges from these curves that as the multimedia traffic load in the system increases, the proposed protocol adapts its operational strategy to the changed traffic conditions.

V. Conclusion

In this paper, we proposed an enhanced link control protocol for wireless multimedia services. The proposed link control protocol, designed for handling different traffic types, offers a 'good wireless link sharing solution' in a varying quality-of-service paradigm, in spite of distributed queuing and multiple access problems in a multiuser, single wireless link environment. Performance results related to delay, packet loss, and quality index demonstrate that the additional control functionalities provided on top of the media access control layer provide improved quality-of-service to multimedia traffic, without significantly perturbing the non-multimedia traffic performance. The performance of the protocol in the presence of wireless channel induced random and bursty packet errors is the topic of an on-going investigation. Other issues that could be investigated in future studies include modelling realistic VBR video sources, efficient random access techniques like splitting algorithm, mobility issues like call hand-over.

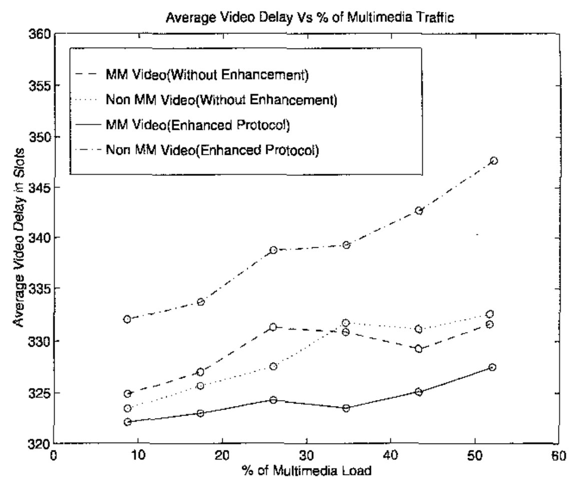

Fig. 7. Average *video* delay versus percentage of multimedia traffic load

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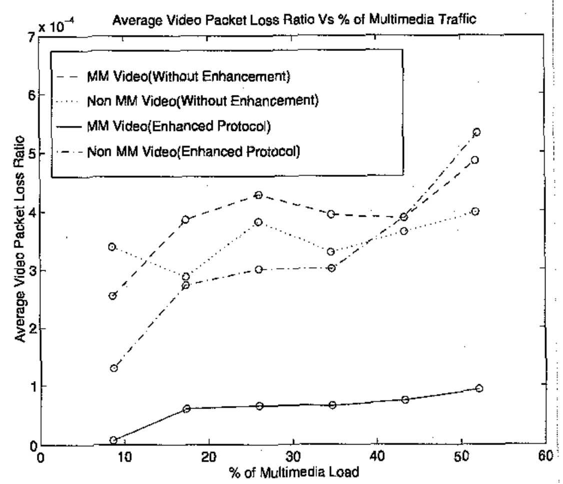

Fig. 8. Average *video* packet loss ratio versus percentage of multimedi traffic load

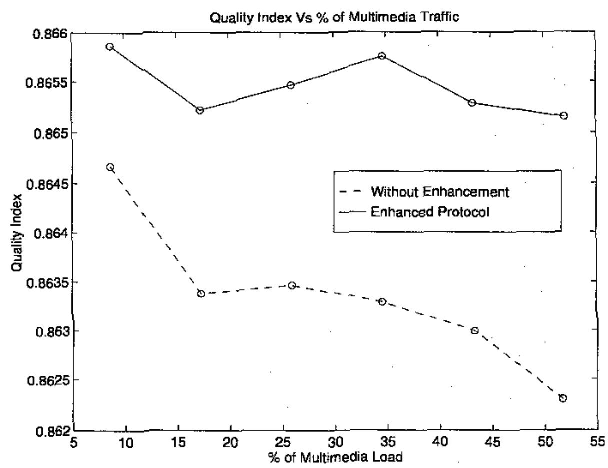

Fig. 9. Quality index (Q) versus percentage of multimedia traffic load

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