Performance of VBR Packet Video Communications on an Ethernet LAN: A Trace-Driven Simulation Study

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Abstract

Provision of multimedia communication services on today's packet-switched network infrastructure is becoming increasingly feasible. However, there remains a lack of information regarding the performance of multimedia sources operating in bursty data traffic conditions. In this study, a videotelephony system deployed on the Ethernet LAN is simulated, employing high time-resolution LAN traces as the data traffic load. In comparison with Poisson traffic models, the trace-driven cases produce highly variable packet delays, and higher packet loss, thereby degrading video traffic performance. In order to compensate for these effects, a delay control scheme based on a timed packet dropping algorithm is examined. Simulations of the scheme indicate that improvements in real time loss rates of videotelphony sources can be achieved.

1 Introduction

The introduction of networked multimedia services is an important and current research issue. In the near future, a range of new services will be developed and integrated into the packet-switched network infrastructure; e.g. multimedia mail, image transfer and storage, multimedia document handling and motion video communication. While the exact application mix is difficult to specify, it is expected that real time video communications, including surveillance, telephony, conferencing and sequence retrieval, will have a significant impact [1, 2].

It is possible to identify a number of recent advances which promote the use of packet-switched networks for transportation of video communication services. Firstly, the advantages of variable bit rate (VBR) codecs, including low coding delay, constant image quality and bandwidth efficiency through statistical multiplexing may be exploited. Secondly, integrated multimedia workstations can provide audio and video I/O at the desktop. Finally, emerging broadband networks offer advanced support for multimedia communications in the wide area, including statistical multiplexing, dynamic bandwidth allocation, multiple-priority access, multicast support and transparent interconnection to ISDN networks [3].

Although deployment of high-speed ($\geq$ 100Mbit/s) LAN technologies such as FDDI and ATM will proceed in the medium term, we can expect that current LAN technologies will be utilized in the near term for packet transport of video applications [4, 5, 6]. Characterizing the performance of current networks carrying video communications traffic is therefore an important issue. This paper investigates packet transport of real time video communications traffic, characteristic of videotelephony applications, on the popular 10 Mbit/s Ethernet LAN.

Previous work has established that Ethernet is capable of supporting video communication traffic in the presence of Poisson data traffic [6, 7]. However, recent studies of high time-resolution LAN traffic have observed highly bursty traffic patterns which sustain high variability over timescales of milliseconds to hours [18, 19]. Conventional traffic models fail to capture these characteristics: it has been demonstrated that the traffic measurements exhibit a fractal-like behaviour [20]. To examine the effects of these traffic patterns on video communications performance, two novel mechanisms have been incorporated in our study. Firstly, data traffic is simulated in a trace-driven manner using recorded LAN traffic. Secondly, video traffic performance is evaluated over three minute intervals, a timescale considered comparable to the lifespan of a videotelephony call. Through application of these mechanisms, we hope to derive performance measures for videotelephony traffic operating in a realistic LAN environment.

The organization of the remainder of this paper is as follows. Section 2 details statistical models which emulate packet streams produced by videotelephony stations. In Section 3 we examine the data traffic records and describe equivalent Poisson models for comparison purposes. Section 4 presents the LAN simulation model and its validation with analytical results, followed in Section 5 by analysis of results. We then conclude with a summary of our findings.

2 Videotelephony Traffic Model

Packet transmission of audiovisual information requires coding and compression functions to reduce bandwidth requirements. These functions are performed by audio and video codecs, which exploit the statistical properties of each information stream to achieve the requisite compression ratios. The resulting compressed streams are multiplexed and encapsulated in communication protocols for transmission over the LAN.

The flow statistics of compressed videotelephony streams are dependent on a variety of factors, e.g. video scene content, compression schemes and protocol overheads. For the simulated system, audio and video codecs are based on ITU-T[1] Recommendations: H.261 video codec and G.721 audio codec. Following a description of each codec, we detail statistical models for characterization of the relevant streams.

The H.261 standard for video coding is expected to be widely applied for narrowband video communication services. It is intended for videotelephony and videoconferencing applications operating over constant bit rate (CBR) ISDN channels at px64 kbit/s, where p = 1,2,...,30 [8, 9]. The codec, shown in block diagram form in Figure 1, achieves constant

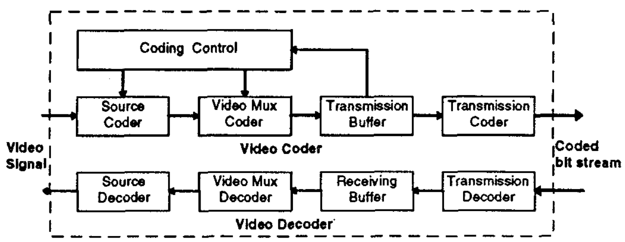

Figure 1: H.261 Video Codec

rate transmission through control of the inherently variable rate source coder via feedback from the transmission buffer. However, it is possible to operate the codec in a VBR mode through bypass of the control loop. In this mode, termed H.261-VBR, advantages of reduced transmission delay and constant image quality may be exploited [4]. Interconnection to ISDN channels may be accomplished through VBR-CBR transcoding at the LAN/ISDN interface.

The H.261 source coder employs a hybrid Discrete Cosine Transform (DCT) and Differential Pulse Code Modulation (DPCM) coding scheme with optional motion compensation. The coding algorithm is based on application of the DCT to fundamental image areas of 8x8 pels (Blocks). Two modes of operation are possible: intraframe mode and interframe mode. In intraframe mode, each Block is DCT transformed and lossy quantized, exploiting the spatial redundancy a frame. This mode is used for the first frame of a call and when scene changes occur; the standard specifies an intraframe must be sent at least every 132 frames to avoid DCT mismatch error. In interframe mode, the prediction error of successive image areas of 16x16 pels is compared with a threshold. If the threshold is exceeded, the error is DCT coded and quantized; in this case a higher compression ratio is obtained due to the temporal redundancy of video frame sequences. Quantization of DCT coefficients is accomplished by selection of a set of 31 linear laws; variation in the quantization step size Q_s may be used to control the mean codec bit rate.

Information produced by the source coder is progressively passed to the video multiplex coder, where it is organized according to a hierarchical structure. A Macro Block (MB) consists of four luminance Blocks and two colour-difference blocks, while a Group of Blocks (GOB) contains 33 MB. A complete frame consists of 12 GOBS for the Common Intermediate Format (CIF=352x288 pels) or 3 GOBs for Quarter CIF (QCIF=176x144 pels).

The audio coding standard G.721 [10] specifies a subjective speech quality identical to conventional 64 kbit/s digital telephony services. The codec utilizes adaptive prediction and DPCM (ADPCM) to achieve a coding rate of 4 bits per sample, 32 kbit/s. Since active speech periods, or talkspurts, typically comprise on average 30-40 % of a call period, packet audio systems may achieve further bandwidth efficiency by encoding only those periods.

A. Compressed Video Stream

A VBR compressed video stream may be assumed to consist of synchronous frame arrivals, where each frame consists of a burst of one or more packets. Burst inter-arrival periods are fixed at $1/R_v$, where R_v is the video frame rate, while packet inter-arrival periods depend on coding and interfacing overheads. Bursts tend to exhibit temporal correlation with some random variation over short lag intervals; an autoregressive (AR) process may be used to characterize this behaviour [11]:

$$x(n) = x'(n) + E[x(n)] \tag{1}$$

$$x'(n) = \sum_{m=1}^{M} a(m)x'(n-m) + e(n) \tag{2}$$

where $x(n)$ is the size of the nth frame, $E[\cdot]$ is the expectation operator, $a(m)$ are the model parameters, M is the model order and $e(n)$ is a (0, σ^2) Gaussian random process. Parameters for the first-order model were derived through encoding a luminance-only CIF sequence [12]. Table 1 shows the measurements for three step sizes Q_s. Due to the trade-off between image quality and bandwidth utilization, parameters for the simulation model were selected with Q_s = 7 and R_v = 15 frames/s. A comparison of the AR model and measured correlation statistics over the lag interval [0,1]s is shown in Figure 2. A fair approximation of the measured data is achieved in agreement with [11].

Table 1: Experimental Sequence Measurements

Parameter	Q_s		
	3	7	11
$E[x(n)]$ (kbits)	24.1	9.64	5.96
σ (kbits)	8.37	3.48	2.25
$a(1)$	0.772	0.761	0.752

Protocols for encapsulation of coded video information are assumed to employ UDP/IP transport with a specialized

[1] Telecommunication Standardization Sector of the ITU (formerly CCITT).

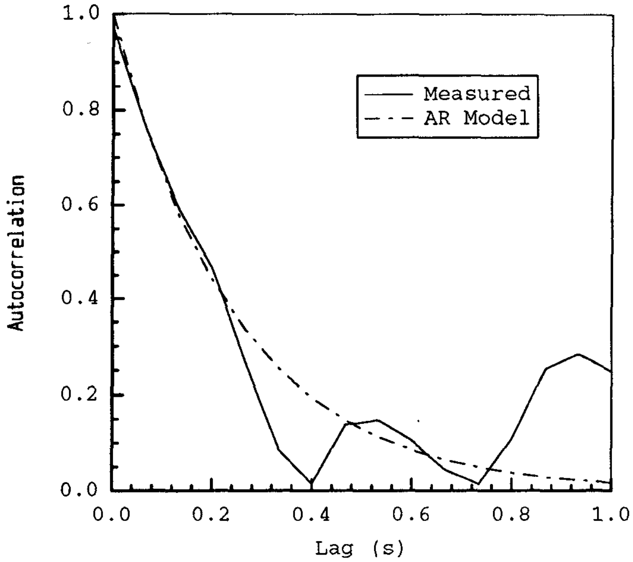

Figure 2: Source Model and Measured Autocorrelation.

video communication protocol at the presentation layer [13, 14]. For this stack, packet headers contribute an overhead of p_{oh} = 560 bits. With this overhead, around 70% of coded frames may be encapsulated in a maximum-length Ethernet packet (1526 bytes). Although better transport efficiency is obtained by encapsulating large packets, smaller packets are preferred as firstly, the subjective impact on image quality is reduced during packet loss, and secondly, reduced delay is achieved through progressive transmission of information as it is produced by the codec [15].

A base unit for packet video transmission is the GOB [7], where packets are encapsulated using a fixed number of GOBs[2]. As the number of GOBs per packet is increased, protocol overheads and resultant bandwidth requirements are reduced. Given the constraints on video quality, we consider a tradeoff at 3 GOBs per packet.

B. Compressed Audio Stream

Audio bursts consist of sequences of packets which transport the compressed speech of each talkspurt. Thus burst inter-arrival periods and burst size are dependent on the statistical characteristics of conversational speech. It has been shown [16, 17] that the distribution of talkspurt and silence durations cans be approximately modelled using two weighted geometric pdfs.

Speech packetization intervals are short, on the order of 16-30ms, in order to sustain real time communication. A fixed packetization interval of T_a=20ms is assumed, with a hang-over interval of 20ms. Based on the statistical model, the audio stream burst characteristics were derived as shown in Table 2.

[2]We assume GOB information is uniformly distributed across the 12 GOBs of a coded CIF frame.

Table 2: Audio Stream Statistics

Mean Talkspurt Duration E[d_t]	350 ms
Mean Silence Duration E[d_s]	793 ms
Talkspurt Rate R_t	52.5/min

Audio packets are encapsulated in an identical manner to video packets; based on the audio and video stream characteristics, we can calculate the mean bit rate r_{av} of the videotelephony stream:

$$r_{av} = 4R_v\,(\mathrm{E}[x(n)]/4 + p_{oh}) + \frac{R_t \cdot \mathrm{E}[d_t]}{T_a}\,(32{\times}10^3 T_a + p_{oh}) \tag{3}$$

which yields a mean rate of 196 kbit/s.

3 Data Traffic

A. Trace Records

The trace-driven simulations employ excerpts from a 30-minute LAN trace detailing 1 million packets, recorded on an Ethernet LAN at Bellcore's Morristown Research and Engineering Facility on 5 October 1989 during 11:00-11:30 a.m. This measurement is a fraction of an extensive set of recorded data used in a comprehensive study of LAN traffic characteristics comprising tens of millions of Ethernet packets [18]–[20].

The 30-minute trace was subdivided into ten 3-minute intervals. Over each interval, the average utilization varies from a minimum of 19.6% during 11:15-11:18, to a maximum of 44.6% during 11:21-11:24. As representation of the observed best- and worst-case traffic conditions, these periods were incorporated in our simulations. For identification these traces are designated as light traffic trace (LTT) and heavy traffic trace (HTT). Figure 3 plots network utilization of each trace averaged over 1s intervals. The variability

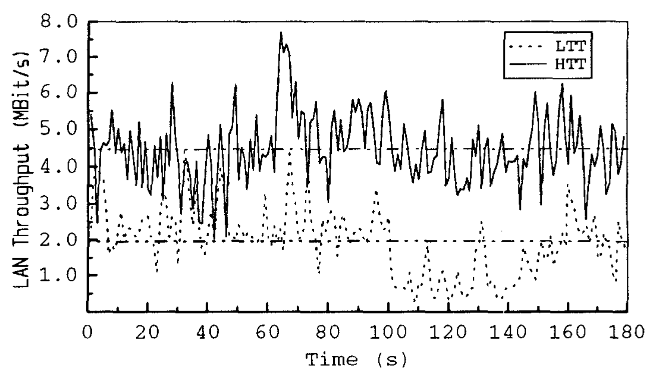

Figure 3: HTT and LTT Traffic

observed at this resolution tends to be exhibited both at shorter intervals, i.e. traffic "spikes" tend to be composed of shorter spikes, and at longer intervals, as suggested by the mean 3-minute utilizations.

The observed packet lengths are grouped around four ranges, as shown in Figure 4. Analysis of the extensive records indicates TCP/IP protocols dominate the higher lay-

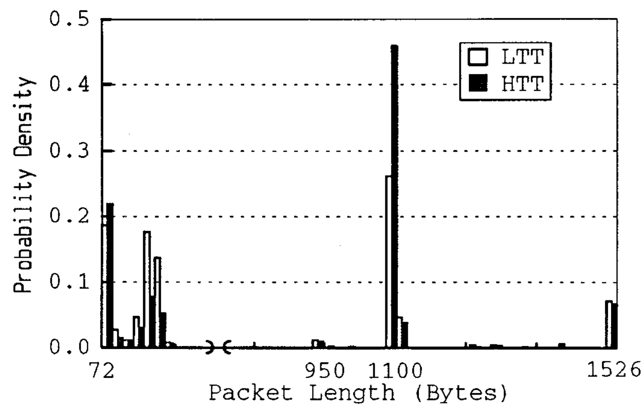

Figure 4: LTT and HTT Packet length pdfs

ers: shorter packet lengths are attributable to interactive applications (e.g. telnet, rlogin), while longer lengths are produced by bulk transfer applications (smtp, ftp) [21]. We note that HTT contains a significantly higher proportion of longer packets, indicating increased bulk transfer activity.

The distribution of packet interarrival periods are shown in Figure 5. The pdfs show approximately 89% of interarrival times for LTT and 96% for HTT. The corresponding means

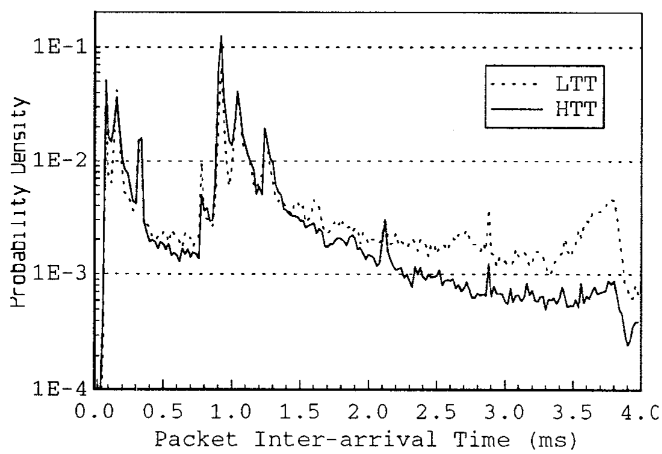

Figure 5: Packet Inter-Arrival Time pdfs

are 2.25 ms and 1.29 ms. Note that the peaks of the pdfs correspond with the major packet length transmission periods; the magnitude of these peaks are in turn correlated with the relative frequency of packet length. These results indicate firstly, that packet arrivals are non-Poisson[3], and secondly, that a high percentage of packets arrive in closely-spaced bursts.

B. Poisson Traffic Models

To quantify relative performance effects of trace data traffic on videotelephony traffic performance, we formulated equivalent traffic models which match the mean utilization levels of LTT and HTT using a Poisson arrival process. The corresponding models are termed light Poisson model (LPM) and heavy Poisson model (HPM). The packet arrival statistics are simulated according to a Poisson pdf:

$$p(t) = \lambda \exp(-\lambda t) \qquad (4)$$

where the mean inter-arrival period $1/\lambda$ is set to the measured trace statistic, and the length of each arriving packet is selected from an approximation to the measured pdfs. In this way, LAN utilization is maintained at the same level as the trace-driven cases over 3 minute intervals.

4 LAN Simulation Model

The LAN model consists of a number of video stations N_v, and data stations N_d connected to a linear broadcast bus. The relevant parameters of the model are given in Table 3. The selection of N_v is motivated by a recent experimental study [5] which indicates six medium-rate videotelphony stations can be supported on the Ethernet[4].

Table 3: LAN Simulation Parameters

Bus Length	500 m
Bus Propagation Delay	5 μs/km
Nominal Bit Rate	10 MBit/s
No. Data Stations N_d	50
No. Video Stations N_v	6
Inter-Station Spacing	Uniformly Distributed

All stations access the shared channel at the Media Access Control (MAC) sublayer through the well known CSMA/CD protocol. The simulation program [22] fully implements the protocol specification [23]. At each station, packets are generated and queued in transmission buffer, where they are forwarded to a single packet buffer representing the LAN interface. For audio and video packets, the following data is recorded as each packet is either successfully transmitted or aborted due to an excessive number of collisions:

- Packet type (audio or video)
- Sending station
- Time of transmission
- Transmission buffer queuing delay
- CSMA/CD transmission delay
- Number of collisions

Validation of our CSMA/CD implementation was undertaken by comparing the delay-throughput characteristics with theoretical analysis. Although the literature provides a number of analytical models (e.g. [24, 25]), the study of Chen and Li [26] is to our knowledge the only analytical

[3] A log-linear plot of a Poisson pdf should produce a linear characteristics.

[4] This system consists of six constant-rate video codecs operating at 192 kbit/s

model which takes into account the effects of the CSMA/CD binary exponential backoff algorithm. We compared analytical results with simulations for a network of 50 stations transmitting fixed packet lengths. The results for mean delay over three packet lengths of T bits are shown in Figure 6. The simulations compare favourably at T=1000 bits. For longer packets, the simulated case yields better performance at high load, with slightly higher delays at medium utilization. However, given the analytical simplifications, we consider the simulation model reasonably represents the performance characteristics of the CSMA/CD protocol.

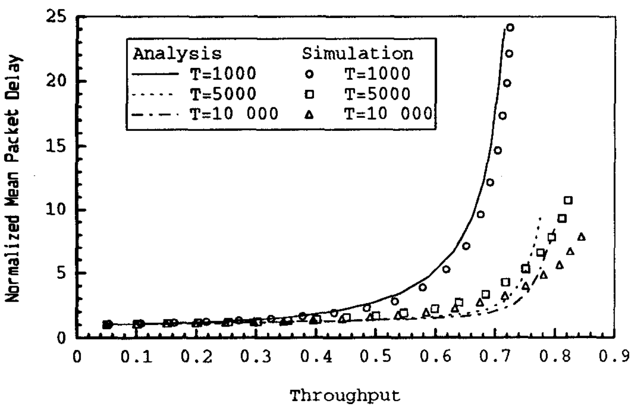

Figure 6: CSMA/CD Performance Comparison

Four simulation runs were executed according to the each of the data traffic models detailed in the previous section (LTT, HTT, LPM, and HPM) over a simulated time period of three minutes. Data stations generate packets in a trace-driven manner (LTT, HTT) or according to a Poisson traffic model (LPM, HPM), while videotelephony stations generate packets according to the appropriate audio and video stream models. No compensation for loss of video packets is undertaken by the video stations, i.e. degraded service persists until the next successful transmission of an intraframe.

5 Simulation Results

This Section details important results for videotelephony traffic for each simulation run. We are primarily concerned with measures of network utilization, packet delay and packet loss. Packet loss may be considered to consist of two components: packets aborted at the MAC sublayer due to an excessive number of collisions, and packets discarded at the receiving transport layer due to delays which exceed a known real-time delivery limit D_{max}.

A. Network Utilization

The utilization measured over each run are shown in Table 4. For the light and heavy traffic simulations, we note similar utilization levels are achieved over the 3 minute intervals. Videotelephony traffic contributes around 80,000 packets for a mean utilization level of 12%, as expected, with utilization increasing from 19.6% in the light traffic cases and 44.6% Mbit/s for the high traffic cases.

Table 4: Network Throughput Results

Simulation	Total packets	Transmitted bits	Utilization
LPM	161,517	563	31.3%
LTT	160,142	563	31.3%
HPM	220,608	1014	56.3%
HTT	221,453	1012	56.2%

B. Packet Delay and Packet Loss

Statistical measures of videotelephony packet delay are detailed in Table 5. Despite almost identical utilization levels, significant difference in performance is observed between the respective trace-driven and Poisson runs. In the cases of LTT and HTT, higher mean delay and delay variance is observed.

Videotelephony delay performance can be further characterized in the form of packet loss due to transmission delays exceeding D_{max}. This gives some idea of the visible image degradation which occurs at receiving stations. Figure 7 illustrates the recorded loss performance, where the packet loss rate is plotted against D_{max}. An exact delay bound

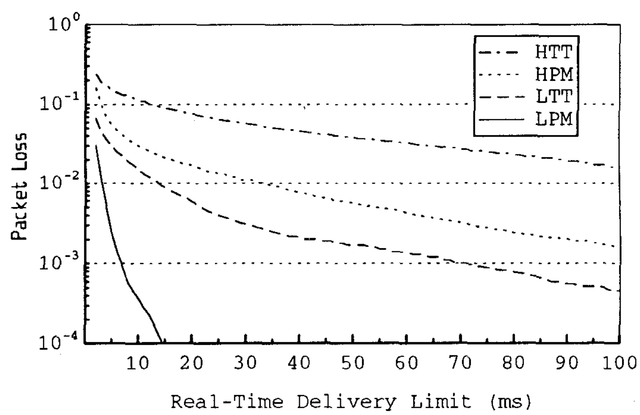

Figure 7: Videotelephony Packet Loss vs D_{max}

is difficult to quantify, as it is dependent on various system factors, e.g. video codec delays, protocol and interfacing overheads. An upper limit of 100 ms may be adequate for LAN applications, while a delay budget as low as 10 ms may be required for WAN operation [4]. However, over a wide range of delay limits it is apparent that the equivalent Poisson traffic models significantly underestimate loss rates of the trace-driven cases. For example, HTT yields a loss rate four times that of HPM at D_{max}=10 ms (12%), six times at 50 ms (4%) and ten times at 100 ms (1.6%). Additionally, we see a significant difference in performance due to the variation in traffic levels: a user placing a videotelephony call during 11:15-11:18 a.m. will experience better service quality than the same call placed during 11:21-11:24 a.m. Some insight into the reasons for the high trace-driven delay

Table 5: Videotelephony Packet Transmission Statistics

Simulation	Mean delay	Standard deviation	Maximum delay	Collision free packets	Aborted packets	Audio packets	Video packets
LPM	0.43 ms	0.57 ms	27.0 ms	86.0%	-	16,504	64,800
LTT	0.91 ms	4.78 ms	247.4 ms	84.3%	-	15,474	64,800
HPM	1.82 ms	7.85 ms	252.4 ms	60.5%	-	16,027	64,800
HTT	6.68 ms	24.7 ms	469.9 ms	60.8%	26	16,900	64,800

can be derived from the packet collision measurements of Table 5. A similar number of packets experience no collisions, while for the HTT traffic case some packets are aborted. A histogram of the collision distributions, shown in Figure 8 reveals that for LTT and HTT, many more of videotelephony packets incur a significant number of collisons. Not surprisingly, the high collision rates are concentrated around peak traffic periods. For example, 48% of packets with ≥ 10 collisions are distributed over 30 seconds of the HTT simulation; 22 of the 26 aborted packets occur during a period of eight seconds. The resulting transmission delays introduce long buffer queueing delays in successive video bursts, even though packets comprising these bursts may undergo minimal transmission delay. We observed a 'cascading' effect, where one or two packets delayed for a significant period is followed by a burst of packets as the transmission buffer is emptied.

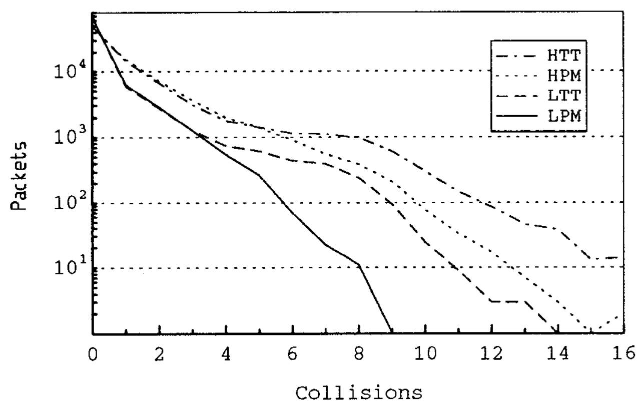

Figure 8: Videotelephony Packet Collision Histogram

C. Packet Delay Control

The preceding results motivate consideration of control mechanisms which may be employed by videotelephony stations to improve packet loss rates. In an internetworked environment, video packet loss may be detected through the use of explicit feedback messages or implicit timing controls [2]. However, in the LAN environment packet loss may be detected quickly via feedback from the network interface.

The proposed control scheme involves a simple modification to the CSMA/CD MAC protocol. The scheme attempts to exploit the last-come, first-served characteristics of CSMA/CD, i.e. that it is more likely for packets which have incurred no collisions to pre-empt packets experiencing higher numbers of collisions. The algorithm is implemented as follows. Firstly, it is assumed that D_{max} is known at the transmitter. On reception of an audio or video packet at the LAN interface, a millisecond timer is reset to zero and started. At each collision, the value of the timer is examined; if the packet has undergone a transmission delay $> D_{max}$, it is discarded. In this manner, packets which have exceeded D_{max} are removed at the sender, rather than consuming LAN bandwidth only to be discarded at the receiver. Additionally, we seek to reduce the queueing delays of succesive audio and video bursts.

The scheme was tested by simulation using HTT as the data traffic load. While we find that more packets are aborted (1838), the mean videotelephony packet delay is reduced to 1.17 ms, with a standard deviation of 3.45 ms. The resulting packet loss statistics are shown in Figure 9. At the limit of 10ms, we find that the loss is reduced from 12% in the uncontrolled case to 5% in the controlled case. Note that delays $> D_{max}$ are not eliminated, as random backoff periods can increase transmission delay above D_{max}, and thus buffer queueng delay can build. However, it is clear that the scheme reduces real-time packet loss. In practice, integration with the video codec will be required to map discarded image information to the relevant sections and provide intra-coded updates.

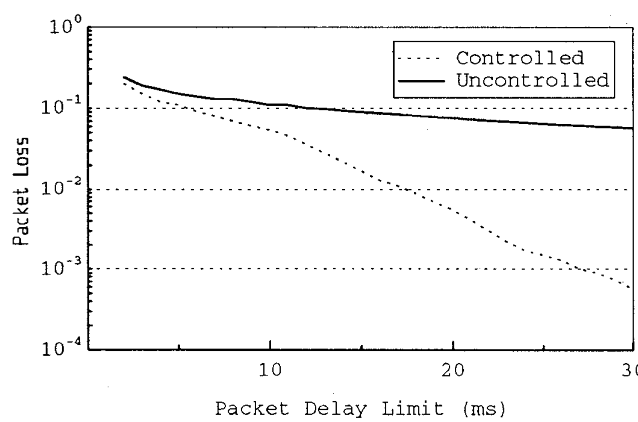

Figure 9: Packet Loss: Delay Control Scheme

6 Conclusion

A simulation study of videotelephony stations operating on the 10 Mbit/s Ethernet was undertaken. Through the use of high time-resolution traces, the effects of bursty LAN data traffic on video communications performance were evaluated and compared to equivalent Poisson models. The simulations indicate the highly variable LAN traces cause significantly higher video communications packet loss due to increased packet transmission delays during peak traffic periods. These are important results for characterizing performance of real-time traffic operating on the Ethernet LAN: since little "average" utilization can be discerned, it is difficult to specify expected performance levels, or plan for a given video traffic capacity.

Although the specific traces may not be characteristic of all LAN traffic, the implication of packet loss extending over several video frame periods may warrant development of specialized control algorithms for reduction of packet loss during heavy traffic periods. We have proposed and tested a simple algorithm which succeeds in reducing real-time packet loss. More advanced schemes may provide better performance, however cost and difficulty of implementation may prove to be limiting factors.

Acknowledgements

The authors would like to thank Will Leland, Henry Fowler, Daniel V. Wilson and Bellcore for their assistance in this study.

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