Evaluation of CATV Network for Multimedia Transport

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Abstract

The popularity of CATV network and the fact that it can support large bandwidth makes CATV network, an ideal choice to deliver broadband services, like video-conferencing, video-on-demand and internet access, to the Home. In this paper, we have simulated a CATV network based on the IEEE 802.14 draft standard MAC protocol and a FCFS scheduling policy. The simulations have been performed using real video conferencing traces. Given an acceptable delay bound and packet loss rate for video traffic, we have determined the maximum number of video conferencing sessions that can be supported in a single upstream channel. We have also studied the effects of bursty data traffic on video traffic.

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Extended Summary

1 Introduction

The popular cable TV (CATV) network infrastructure can be used to deliver broadband services, like video-conferencing, video-on-demand and Internet access, to the home. Recently, IEEE has released a draft standard of Media Access Control (MAC) protocol for interactive communication over cable TV network [4]. For video conferencing applications, the end-to-end delay must be between 100-200 ms [1]. The MAC protocol of CATV network can provide the bounded delay only by considering the packet as being lost if it experiences a delay in excess of the delay bound. The quality of the video displayed is degraded because of such packet losses. Thus, the challenge of a CATV network design is to provide bounded delay along with a reasonable quality of video.

Previously, Tobagi *et al* [1, 2, 3] have evaluated the performance of Ethernet and ATM for multimedia communications. Performance investigation of CATV network has been reported only very recently [7] using real Ethernet measurement as input traffic. To the best of our knowledge, the present paper is the first of its kind to evaluate the performance of CATV network using realistic video traces. In CATV network, performance bottleneck is primarily due to contention in the upstream path (*ie*, the path from the subscriber's end to headend). Hence, we concentrate only on the upstream path. We consider that the main source of video traffic in the upstream direction will be from video-conferencing sessions. Other video applications like video-on-demand will have only request data packets in the upstream direction.

2 Simulation Environment

Our simulator implements the MAC protocol of [4]. A station having a packet to transmit, sends its bandwidth request either in the contention slot or in its data slot by piggybacking. The number of Contention Slot (CS) and Data Slot (DS) contained in each upstream frame is assumed to be fixed. The incoming messages are packetized as ATM cells and then transmitted. We are not considering the packetization delay here. Our primary interest is in obtaining the queuing and transmission delays experienced by a video packet. The delay bound for video traffic is set to 100 ms [1]. We assume that a station generates either a video or a data traffic, but not both.

We model the data traffic as a Poisson source. The message size is uniformly distributed between 0 to 4800 bytes. The parameter λ of the Poisson process is set to 100. This corresponds to a mean interarrival time of 10 ms ($1/\lambda$). The video traffic is generated using frame size traces downloaded from [5]. The trace file used for this particular simulation belongs to a video conferencing session. These frame size trace have been extracted from MPEG-1 sequences encoded with the Berkeley MPEG-encoder adapted for motion-JPEG input. The video was captured in motion-JPEG format from a VCR with a SUN workstation. We have used the same trace file to generate the video traffic for all video stations for a given run of simulation. Random starting point has been used by each video station so as to decrease the correlation between the video traffic of different stations.

3 Simulation Results

3.1 Optimum CS/DS Ratio and its variation

An "Optimum" CS/DS ratio is defined to be that ratio at which the delay experienced by a packet for a given number of data and video traffic is minimum. The variation of the average video packet delay as a function of CS/DS ratio is shown in Figure 1. The results illustrate that there indeed exists an optimum CS/DS ratio, where the delay is minimum. The range of this region is quite broad when the number of data stations is less but decreases as the number of data stations increases.

The variation of the Optimum CS/DS ratio as a function of the number of video stations and data stations has also been studied. The simulation results show that when the number of video and data stations are less, the optimum CS/DS ratio is *independent of the number of video and data sources.* This optimum CS/DS ratio is about 0.1667.

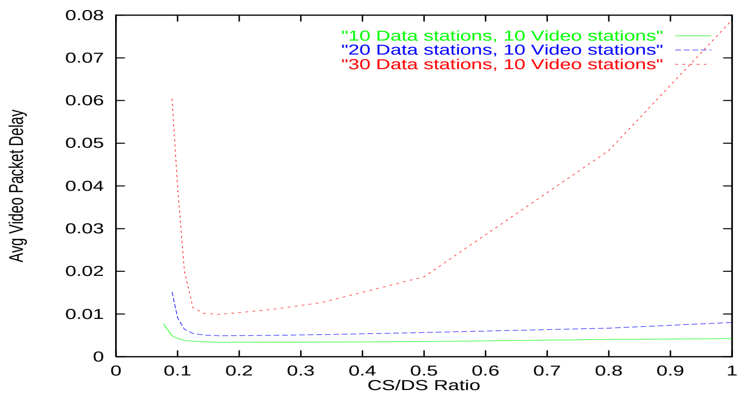

Figure 1: Average video packet delay Vs CS/DS

The detailed simulation results will be presented in the full paper.

3.2 Average Video Packet Delay and Video Packet Loss

The Best Average Packet Delay (BAPD) is defined as the average delay experienced by a packet when the optimum CS/DS ratio is used. To determine this delay, we perform our simulation using the optimum CS/DS ratio. The video packest experiencing delay greater than 100 ms are considered to be lost. We consider a maximum packet loss rate of 1% to be tolerable for acceptable quality of video as assumed in [3]. The variation of the best average packet delay as a function of both the number of video stations and the number data stations is shown in Figure 2.

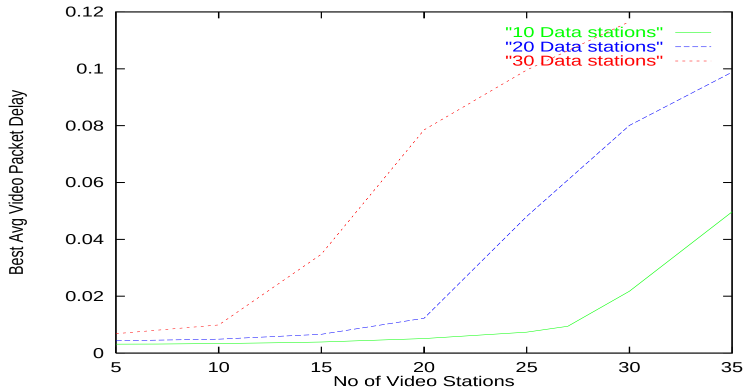

Figure 2: Best Average Video Packet Delay Vs Number of Video stations

The Best Video Packet Loss (BVPL) is defined in percentage as the ratio of the number of video packets lost to the total number of video packets generated by all video sources when the optimum CS/DS ratio is used. Figure 3 shows the variation of Video Packet Loss as a function of both the number of video sources and the number of data sources.

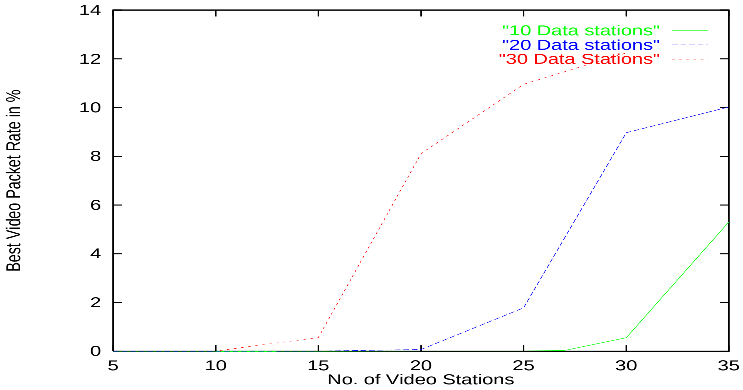

Figure 3: Best Video Packet Loss Vs Number of Video Stations

For a maximum packet loss rate of 1% needed for an acceptable video quality, we determine from Figure 3 that the CATV network can support about 15 video conferencing sessions, given a delay bound of 100 ms and 30 data sources.

4 Conclusions

We have evaluated the performance of CATV network carrying video and data (multimedia) traffic. Given the tolerable end-to-end delay bound of 100 ms and a maximum packet loss rate of 1%, it has been shown that the network can support about 15 video conferencing sessions along with 30 data sources. We feel that this number can be improved if the Headend scheduling policy is made to support priority classes and if the CS/DS ratio instead of being fixed is allocated dynamically by some algorithm, like the one proposed in [6]. This is currently under investigation. We are also investigating an appropriate characterization of the effect of packet losses on the video quality.

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