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Optimization of Voice and Data in Home And Small Business Environment

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Abstract

Over the years, voice and data communications have been handled by different communication networks. One reason being that their application are separate and another reason being that the voice and data signal characteristics are different. Presently, with the advent of internet, it is possible to combine voice and data in a single network. In this paper, we will consider a method of effectively merging voice and data communication infrastructure together so that both of them can now function as one without incurring any unwanted distortions to the voice traffic as it is more sensitive. We implemented our method using MATLAB and provide some numerical results. From the results, it was shown that voice and data services can both be effectively managed on the same infrastructure while still maintaining a reasonably low level of delay in the network, thereby ensuring an effective communication link.

Keywords: Local Area network (LAN), voice packets, data packets, frames

1.0 Introduction

Optimization means making the best use of available resources. It can be defined as the perfect combination of available resources in order to maximize output while minimizing waste of resources. As the world dives deeper into the 21st century, the need for an improvement in technology can not be over emphasized and steps are constantly being taken to improve upon previous models.

In the past, networks have typically been designed to accommodate one specific type of traffic. There is the Public Switched Telephone Network (PSTN) for voice traffic; and Wide Area Networks (WANs), Local Area Networks (LANs), and the Internet for data traffic [1]. Consider a person speaking on the telephone and then sending an electronic message from a computer terminal. The two actions are handled by separate instruments and are perceived as serving separate purposes. Telephony permits immediate, personal, and interactive contact while data messages allow for more pre-meditated, formal, and non-interactive communication.

Interest in "integrating" voice and data communications was stimulated by deregulation of the U.S. telephone industry and international activities in planning the standards for the Integrated Services Digital Network (ISDN). Although ISDN will most likely be comprised of logically separate networks, it will provide subscribers with the functionality of a single, integrated network by offering a standardized, integrated user access to services [2].

Offering voice and data services in a single network promises several benefits. Users achieve convenience, flexibility, and economy. An integrated user interface allows different terminal equipment to be moved and plugged into any interface in the same way that different electrical appliances can use any standard electrical power outlet. Furthermore, services can be customize to individual needs without having to be concerned with the compatibility of different special-purpose networks. For network providers, integration promises benefits in efficiency and economy. Sharing facilities not only increases efficiency, but should also simplify network operations and maintenance. Reduced network costs should result in lower service prices to users.

2.0 Circuit Switching

Circuit switching is probably the most familiar switching since the public telephone system is the primary example of this approach. Circuit switching was developed during the early days of telephony when calls were connected by an operator at a manual switchboard [3]. Although the switching operation is now performed electronically, the principles of circuit switching are still much the same.

Circuit switching has a number of important features, namely that traffic is handled on a blocking basis, bandwidth is utilized efficiently only if the circuit is active fairly constantly since the channel is dedicated for the

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entire call [4,5]. Also circuit switching has several disadvantages for most data applications, such as bandwidth is utilized much less efficiently because most data is usually active only about 5 to 15 percent of the time, data bursts are often short and require only a brief connection, circuit switching works well for voice because a 4-kHz bandwidth is characteristic of all speech signals and circuit switching does not have the capability of node-to-node error- checking since the network is insensitive to the content of the transmitted information.

2.1 Packet Switching

Packet switching is based on the idea of message switching, or store-and-forward switching. Message switching resembles the method of mail delivery in the postal system. A message is formed by concatenating the data information with a header and an end-of-message flag, which is similar to putting a letter into an envelope. The header contains all the information necessary for routing the message through the network information such as source, destination, identity number, and checksum for error. The message is stored in a buffer at each switch which decodes the message header and determines the next node in the route. When the appropriate link becomes available, the message is forwarded to the buffer in the next switch. Usually, an acknowledgment is returned by the receiving node if the message is received without error; otherwise, the sender retransmits the message after some time [2,6].

Other networks and protocols have been proposed which are more suited for voice traffic. In particular, Asynchronous Transfer Mode (ATM) [7] switching networks have been championed as the solution to the single network problem. ATM networks provide different Quality of Service (QoS) levels and divide the channel into fixed length cells, which allow real-time traffic to acquire the channel at constant intervals, limiting the delay variance. Another solution synchronizes packetized voice and data, utilizing the master frame format of Time Division Multiplexing (TDM) to ensure real-time service delays. Also, the IEEE 802.1 Internetworking Task Group is currently working on the 802.1p standard which will enhance the Ethernet MAC layer to allow different traffic classes and implement priorities. However, all of these solutions require changes to existing Internet and Intranet networks, thus the need to analyze the ability of the existing LAN segments to carry voice traffic [8,9].

3.0 Network Model

In order to model a Local Area Network (LAN) [10], we begin with a collapsed 10Mbs backbone model, in which multiple workstations are connected through a single network hub. As in existing LANs, the model assigns no priorities to different traffic types or users. The model accounts for delays from the transport, network, data link, and physical layers of the network. The transport layer is assumed to be TCP. No particular MAC layer protocol is assumed; rather that each user has equal probability to be the next one served. The users consist of both data and voice sessions. A data session is an end-user terminal connected to the LAN. As in TCP, data packets are windowed and acknowledged, with one acknowledgment (ACK) per window. A data packet has size L_d, and there are *w* packets per window. Typical TCP values are shown in Table 1. A voice session is a telephone connected to the network, which transmits one voice packet every 30ms. Voice packets are not windowed and have size L_v, which is much smaller than the data packet size. Thus, the transmission time of a voice packet is substantially smaller than that of a data packet. For our purposes, transmission time is the amount of time it takes to transmit the packet plus an interpacket gap time (it is assumed that there is some small delay between packets).

Table 1: Model parameters and typical values

Description	Data Parameter	Typical Data Value	Voice Parameter	Typical Voice Value
Window size	w	1-6pkts	----	----
Packet Size	L_d	12kbits	L_v	512bits
Transmit time	t_{xd}	1.44ms	t_{xv}	0.06144
ACK Delay	t_d	25ms	----	----
Data sessions	N_d	30	N_v	250

3.1 Data Model

The model in which all users are data sessions is first developed and simulated to determine average data packet delay. It is assumed that data is sent one window at a time and one ACK is returned per window. Each user constantly transmits data, which means it can be in one of three states: with a window in the queue, transmitting a window, or waiting for acknowledgment. Thus, unless a user is transmitting or waiting for an ACK, it always has a single window waiting in its queue. In this model a user contends only once to transmit *w* packets, rather than contending *w* times. Each queue has an equal probability, on the average, of being the next one served (i.e. that session's window is transmitted), unless it is awaiting an ACK (which means its window was recently transmitted). This assumption follows the theory that the Ethernet is essentially fair over long periods of time, despite the capture effect. The wait time D_d of a given packet at the MAC layer is simulated using equation (1) and the simulation result is shown in Figure 1 for up to 30 data sessions.

$$D_d = \frac{1}{2}\left(N_d t_{xd} - t_d \pm \sqrt{(N_d t_{xd} - t_d)^2 + 4t_{xd} t_d}\right). \quad (1)$$

The results show that the average packet delay increases linearly with an increasing number of data sessions.

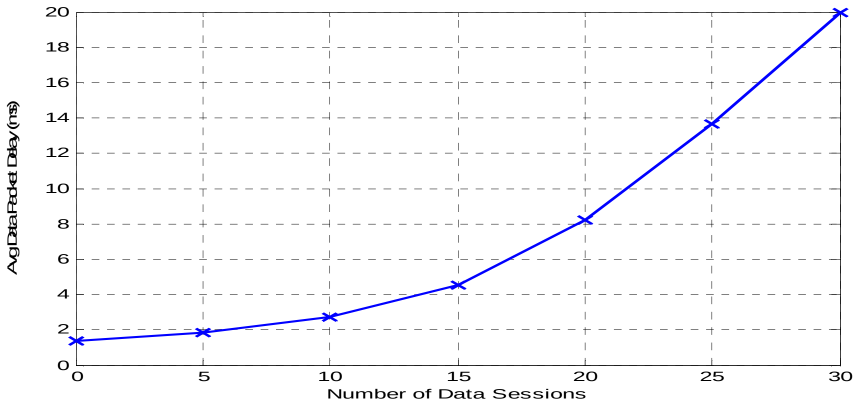

Figure 1: Average data packet delay for data model

3.2 Voice Model

We now consider a model in which all users are voice sessions. Each voice session is modeled as a queue which has a packet arrival every 30ms. The delay of a packet once it starts contending depends on its contention and transmission times. This model assumes an essentially First In First Out (FIFO) service order of packets and models delay as a combination of a constant propagation delay plus variable processing and queueing delays. Although packet arrivals are deterministic, the Markovian M/M/1 model can be used to find an upper bound for the mean delay of voice packets. With arrival rate $\lambda = 1/30$, the average voice packet delay is simulated using equation (2) and the result obtained is displayed in Figure 2 for up to 250 voice sessions.

$$D_v = \frac{\lambda (N_v t_{xv})^2}{1-(\lambda N_v t_{xv})^2} \quad (2)$$

The simulation model results shows that the average voice packet delay increases with increase in number of voice sessions.

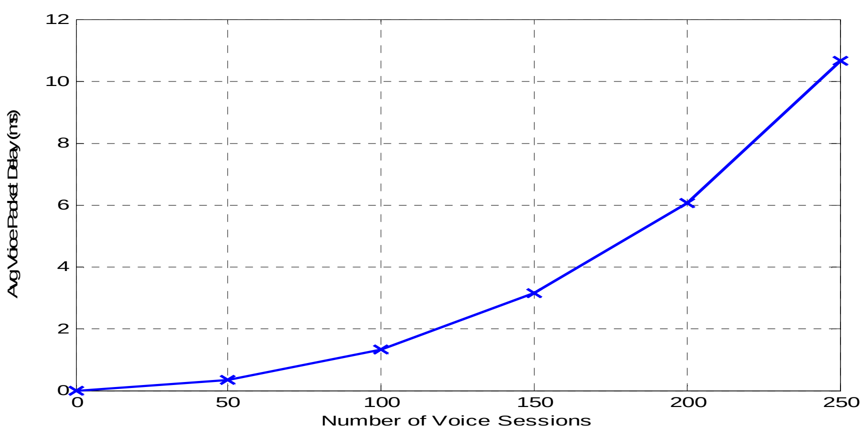

Figure 2: Average Voice Packet Delay for Voice Model

3.3 Integrated data and voice model

After considering the models for data and voice separately, we then combine them to represent a LAN with mixed voice and data traffic. Each traffic type has the same characteristics as in the first two cases, except that we simplify the data model by ignoring TCP acknowledgments. We assume there is always at least one packet waiting in each of the data queues (i.e. $P_d = 1$). Including acknowledgments would change the model by causing a user to not always be contending, so that $P_d < 1$. This is because the user will sometimes be waiting for an ACK, and thus will not have a packet waiting in the queue for transmission. Although transmission of the ACK would add traffic to the network (which contends with data packets), because ACK packets are small relative to data packets the traffic it adds is less than the traffic from a data packet. Actually the analysis results in a higher delay because the data packets are much larger than ACK packets.

3.3.1 Average Voice Packet Delay

We assume that each voice or data session with a packet waiting in its queue has equal probability of winning contention. Using a single voice user with the M/M/1 model to determine an upper bound on the average voice packet delay for a given voice session. The resulting average voice packet delay is given by

$$D_v = \frac{N_d P_d t_{xd}}{1-\lambda(N_v t_{xv} + N_d P_d t_{xd})} \quad (3)$$

and the simulation result is shown in Figure 3. As shown in Figure 3, with five data sessions (N_d=5) the average voice packet delay is less than 30ms for up to 200 voice sessions. However, with ten data sessions the average voice packet delay increases to higher than 30ms, which results in an unacceptable delay for voice traffic. Also, the number of voice sessions that can be accommodated within the same LAN is greatly reduced. For 250 voice sessions, the average voice packet delay was over 1500ms for 10 data sessions.

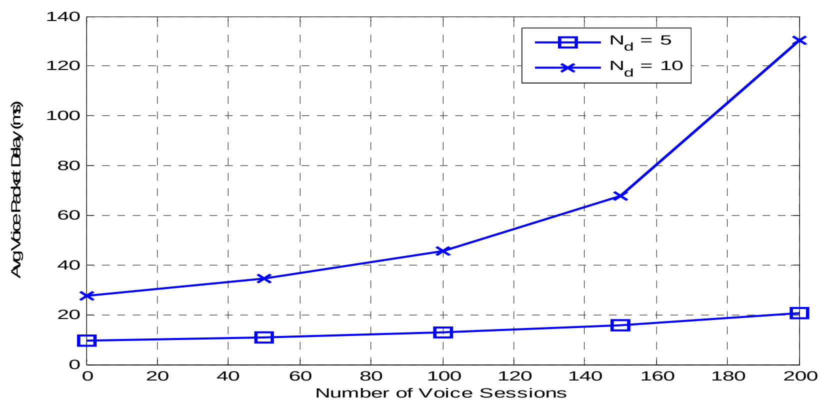

Figure 3: Average voice packet delay for 1 voice frame per packet

3.4 Sensitivity to Number of Sessions

The sensitivity of the average voice packet delay to the number of voice and data sessions is simulated using equation (4) and the result is shown in Figure 4.

$$N_d = \frac{D_v - D_v N_v \lambda t_{xv}}{P_d t_{xd} + D_v \lambda P_d t_{xd}} \quad (4)$$

Values for average voice packet delay from 10ms to 30ms are shown, average voice packet delays greater than 30ms are not considered because such delays would cause excessive distortion in the received voice and may cause the sender buffer to overflow.

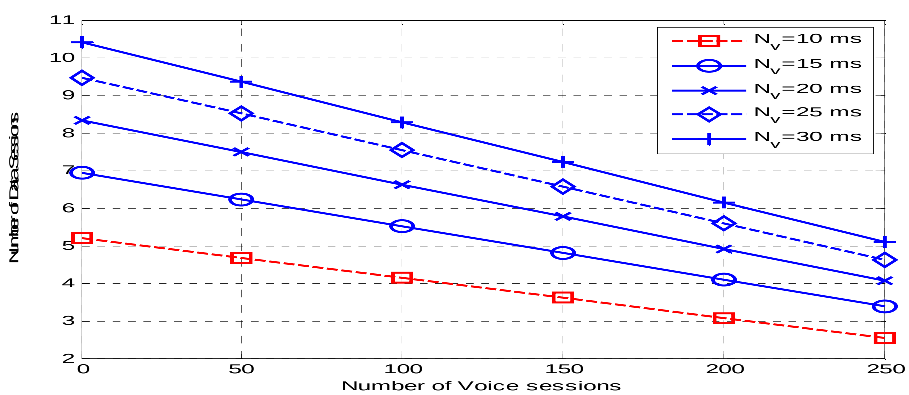

Figure 4: Sensitivity to number of data and voice sessions

It was observed that for up to 500 voice sessions, the number of data sessions reduced to about zero.

3.5 LAN Utilization

The model assumes that every data session always has a packet waiting for transmission (i.e. P_d=1). This corresponds to 100% LAN utilization for the data sessions. As in the data model, the percentage of time in which there is a packet waiting at the MAC (P_d) depends on many factors, including window size, network traffic, and ACK frequency. This value in the model can be altered according to utilization data for a given LAN and thus account for these factors. Figure 5 shows average voice packet delay for 50% LAN utilization (P_d = 0.5) for various values of

N_d (i.e. number of data sessions). It is observed that decreasing P_d by one half increases the number of data sessions by two for a given number of voice sessions, on the average, without increasing the average delay (compare Figure 5 with Figure 3). Increasing the number of voice sessions to 250 increases the average voice packet delay to over 1500ms.

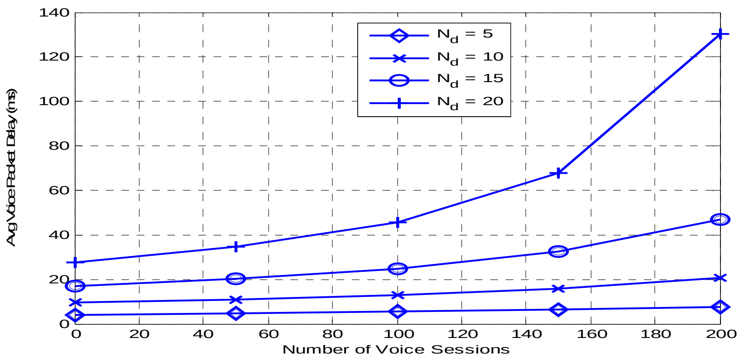

Figure 5: Average voice packet delay for 50% LAN utilization

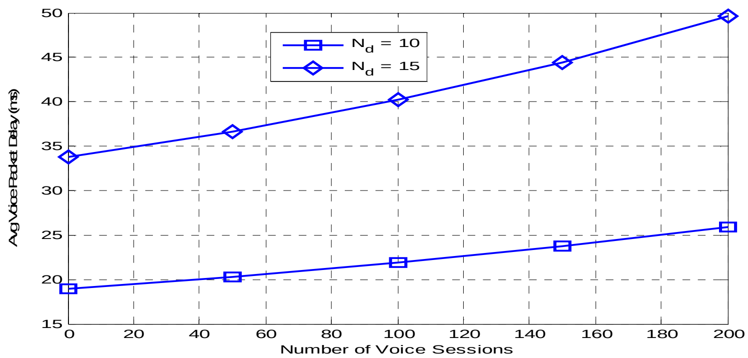

Figure 6: Average voice packet delay for 2 voice frames per packet

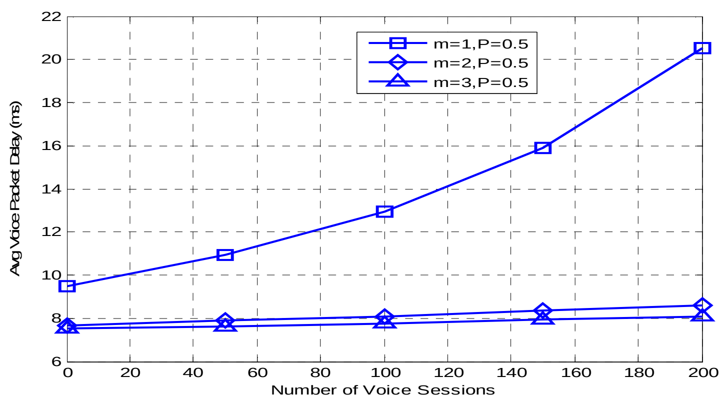

Figure 7: Different voice frames per packet for 50% LAN utilization

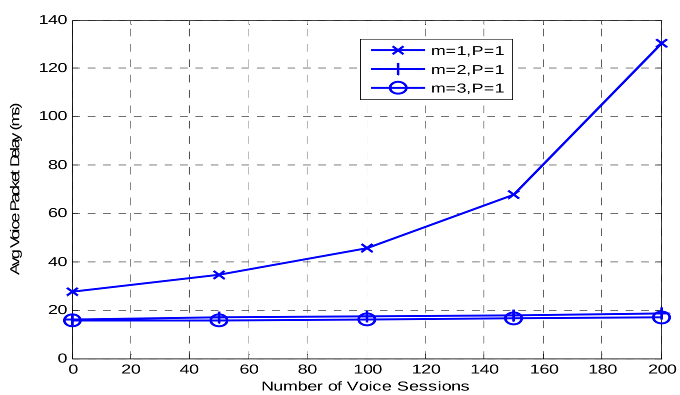

Figure 8: Different voice frames per packet for 100% LAN utilization

3.6 Multiple Voice Frames per Packet

We will check the effect of changing the number of voice frames per packet. Because a single voice frame is small, we can combine multiple frames in one packet before transmission over the LAN. If we combine m frames in one packet, the voice packet delay can be m times longer, since the receiver buffers m-1 frames and plays one every 30ms. This also decreases the amount of contention in the network, since the voice packet rate is m times smaller than the original rate (i.e. $\lambda = \frac{1}{m\times 30}$). The average voice packet delay also decreases, although the end-to-end delay for a given packet is larger (since the arrival time is m times larger).

Figure 6 shows the average voice packet delay against number of voice sessions for given values of N_d, when two voice frames are combined in a single packet. The results show that up to 15 data sessions can be combined with 200 voice sessions in this case, while with only one voice frame per packet (see Figure 3) only about 5 data sessions are allowed. Also, it is observed that increasing the data sessions to 30 is possible; however, the average voice packet delay increase to over 500ms. Combining three voice frames per packet further increases the number of data sessions to between 20 and 25 for 200 voice sessions. Thus, we can see that combining multiple voice frames per packet substantially increases the capacity of the LAN, where capacity is measured in number of sessions and is limited by the average voice packet delay.

It is observed that for different LAN utilization and voice frames, the average voice packet delay is drastically reduced for 50% LAN utilization as depicted in Figures 7 and 8, while the number of data sessions used was 10 in both cases. This technique helps in achieving effective optimization of the LAN.

4.0 Conclusion

In this paper, we have explored and developed a general model of a LAN where we have effectively merged data and voice communication infrastructure, while maintaining low level of delay in the network. Appropriate parameters were set in order to determine the number of data and voice sessions a LAN will support, using average voice packet delay as a constraint. Based on the results obtained, slightly increasing the number of data sessions substantially decreases the number of voice sessions supported, because adding a data session to a network increases the voice packet delay than an added extra voice session. Increasing the number of voice frames per packet decreases the average voice packet delay for 50% LAN utilization. The work in this paper differs from the one in [9], in that we did not consider any particular MAC layer protocol. However, in [9] the authors considered improving the performance of LAN that uses Carrier Sense Multiple Access with Collision Detection (CSMA/CD) protocol as an access technique to handle high traffic and collisions for either voice or data packets. This paper can serve as a basis for studying and understanding voice and data combination in local area network.

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