PRIORITY CSMA SCHEMES FOR INTEGRATED VOICE AND DATA TRANSMISSION

by

KAI-SANG CHING
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Department of Electrical Engineering

The University of British Columbia
Vancouver, Canada

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Abstract

Priority schemes employing the inherent properties of carrier-sense multiple-access (CSMA) schemes are investigated and then applied to the integrated transmission of voice and data. A priority scheme composed of 1-persistent and non-persistent CSMA protocols is proposed. The throughput and delay characteristics of this protocol are evaluated by mathematical analysis and simulation, respectively. The approach of throughput analysis is further extended to another more general case, p-persistent CSMA with two persistency factors, the throughput performance of which had not been analyzed before. Simulations are carried out to study the delay characteristics of this protocol. After careful consideration of the features of the priority schemes studied, two protocols are proposed for integrated voice and data transmission. While their ultimate purpose is for integrated services, they have different application. One of them is applied to local area network; the other is suitable for packet radio network. The distinctive features of the former are simplicity and flexibility. The latter is different from other studies in that collision detection is not required, and that it has small mean and variance of voice packet delay. Performance characteristics of both of these protocols are examined by simulations under various system parameter values.
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Chapter 1

Introduction

1.1 Background

With the advent of computers, which are ubiquitous nowadays, data traffic has grown tremendously. In order to handle data traffic efficiently, packet transmission networks have been developed [1]. ARPANET, ALOHA, and ETHERNET networks are typical examples. These packet-switched networks have established themselves as an attractive option of data transmission.

Traditionally voice and data transmission have been handled by different communication networks. But like data, voice traffic has also been increasing steadily [2]. With voice and data traffic growing at this alarming rate, it is inefficient to handle such data using disjoint networks. Thus, packet voice communication has been receiving considerable attention and basic issues associated with packet voice communication are well-elucidated in [3]. Reduction from noise and crosstalk is one of the benefits offered by packet voice. Another more important advantage is that packet voice is compatible with data packet with the consequence that voice and
data can be multiplexed together and transmitted in a single network [4]. Combining voice and data in a single network promises several advantages [5]:

- Cost reduction; transmission and switching facilities can be shared by both types of packets; and cost of maintenance, components, and production are guaranteed to be lower.

- Speech alternates between talkspurt and silence periods; data packets can make use of the silence periods which may result in higher utilization.

- More sophisticated services can be provided to users.

However, integrating voice and data is not a simple task. Because of their completely different characteristics, combining them creates problems. Data traffic is usually characterized as bursty; that is, there is a large peak-to-average data rate. Unlike data, voice packets are generated periodically while in talkspurts. Besides, voice packets demand small average and variance in delay, whereas data packets do not have such stringent demands. To fulfill these requirements, priority multiaccess protocols are necessary to prioritize voice and data packets.

Among the various random access priority protocols available, carrier sense multiple access (CSMA) is a very popular one [6,7,8]. There are two main reasons for this. First, it has been shown that compared to other random access schemes, CSMA performs quite efficiently if the propagation delay is short relative to the packet transmission time [9]. Second, an improved version, known as CSMA with collision detection (CSMA/CD), has been successfully applied to local computer networks, typified by the Ethernet [10].
All of these protocols [6,7,8] employ a time interval called reservation period. This reservation period ensures the higher priority messages having complete access to the channel over the lower priority ones. But this reservation period is an additional cost because users have to transmit this reservation period before sending its own packets. A $p$-persistent CSMA/CD priority resolution scheme that does not employ a reservation period has been proposed in [17]. In this scheme, the higher the priority of a packet, the higher the probability of transmission. In other words, each priority class has its own probability of transmission.

Since the characteristics of voice and data packets are dissimilar, ordinary priority schemes may not be suitable for integrated voice and data transmission. Protocols, therefore, must be well-designed. Probably due to the omnipresence of Ethernet and the advantages of collision detection, many good protocols (e.g. [11,12,13]) have been proposed for such application in local area networks (LANs). Nevertheless, these protocols are so specifically devised to suit the characteristics of voice and data packets that they become rather complex. Implementing a complex algorithm, sometimes, may be very costly. In view of this, a protocol which is simple, robust, and has an acceptable performance is undoubtedly desirable.

Strangely enough, while there are abundant protocols for integrated applications in LANs, exceedingly few protocols have been proposed for integrated voice and data transmission in packet radio network. As a matter of fact, some published works dealt with voice transmission only [14,15]. Since quick collision detection may not be available in a packet radio network, the performance of a protocol can be easily degraded if collision occurs frequently. As a result, a protocol for integrated services should be designed in such a way that frequent collisions between voice and
data packets should be avoided. One simple way of achieving this goal is to take advantage of the characteristics of voice and data traffic.

Some of the related works and motivations have been briefly reviewed in this part of the introduction. The objectives and organization of this thesis are described below.

1.2 Objectives and Organization of the Thesis

There are three variants of CSMA protocols, namely, non-persistent, 1-persistent, and p-persistent. Each of these protocols has its own inherent properties. With the proper combination, a priority scheme can be implemented.

In chapter 2, we suggest that 1-persistent and non-persistent is a good combination, and that it can be implemented as a non-preemptive priority scheme. It is different from other works in that, like the p-persistent priority scheme proposed in [17], it does not use reservation period. The throughput expressions are derived, one for each type. Although throughput expressions of a similar system\(^1\) has been analyzed in [16], the approach adopted in chapter 2 is much simpler. Most importantly, the approach can be easily extended to a more general case, p-persistent CSMA.

Unlike the p-persistent CSMA, there has been no delay-throughput characteristics of the p-persistent CSMA with different probabilities of transmission available in the literature. Hence, the analytic approach of chapter 2 is applied to this case.

\(^1\)The system in [16] consists of non-persistent, 1-persistent, and virtual-time CSMA.
CHAPTER 1.

The analysis shown in chapter 3 is the first piece of work done on the throughput performance of this protocol. Although only two transmission probabilities are involved in the analysis, the case of more than two can also be included. To better understand the throughput performance, the cases of infinite and finite users are considered too. Simulations are performed to observe the delay characteristics.

Observations from analytic and simulation results in chapter 3 show that, under data traffic, the protocol of $p$-persistent CSMA/CD with two transmission probabilities is well behaved. Hence, the application of this protocol to the integration of voice and data transmission in LANs is first proposed in chapter 4. Compared to other works, the distinctive features of this protocol are simplicity and flexibility. Besides, simulation results indicate that it is a viable scheme.

Since not many works have been done in the area of integrated voice and data transmission in packet radio network, in chapter 5 we propose a scheme [28] with an acceptable level of performance. One of the interesting properties of our protocol is that no collision detection is required. Furthermore, the mean voice packet delay, in general, is within two voice packets transmission time, and the variance of delay is low. These properties are important if good voice quality is desired.

Chapter 6 concludes the thesis. It summarizes the research results presented in previous chapters.
Chapter 2

Network with 1-persistent and non-persistent CSMA Protocols

This chapter is concerned with the throughput and delay characteristics of a network made up of two groups of users. The first group uses the 1-persistent CSMA protocol to access the channel while the second group applies the non-persistent CSMA protocol. Throughput expressions are derived. These will be used to investigate the throughput characteristics of each group. Problems involved in evaluating the resultant throughput are discussed. Besides, simulations are performed to observe the delay characteristics. Results indicate that a simple non-preemptive priority scheme can be implemented as follows: a higher priority group transmits in 1-persistent mode and the lower priority group transmits in non-persistent mode.
2.1 Review of Protocols

Operation of the 1-persistent and non-persistent in the network is assumed to be slotted. A slot is of $\tau$ seconds long, where $\tau$ is the end-to-end propagation delay. Besides, all users are forced to start transmission at the beginning of a slot.

Slotted non-persistent CSMA is considered first. When a user is ready the channel is sensed and the following steps are carried out:

1. If the channel is idle, the packet is transmitted; or
2. If the channel is busy, the ready user reschedules the packet according to the retransmission algorithm; and then, go to step 1.

The slotted 1-persistent protocol is as follows:

1. If the channel is idle, the packet is transmitted; or
2. If the channel is busy, the ready user persists sensing the channel and transmits the packet as soon as it is sensed idle.

2.2 Throughput Analysis

For simplicity, the users who transmit in 1-persistent mode are called class 1 and those who transmit in non-persistent mode are labeled class 2. Hence, the throughput of class 1 is denoted by $S_1$ and that of class 2 by $S_2$. It is assumed that the packet arrival process and channel traffic follow the Poisson distribution and that
the arrival processes of class 1 and class 2 are independent. Also, the channel traffic \( G \) comprises the traffic generated by class 1, \( G_1 \) and by class 2, \( G_2 \). Thus \( G = G_1 + G_2 \). Throughout the derivations, \( a \) represents the normalized end-to-end propagation delay and \( TP \), the transmission period which equals \( 1 + a \). Using the renewal theory arguments\(^1\) the average throughput is given by
\[
S = \frac{\bar{U}}{B + I}
\]  
where \( \bar{U} \) is average time during a cycle that the channel is used without conflicts, \( B \) the mean busy period, and \( I \) the mean idle period. Accordingly, \( S_1 = \bar{U}_1/(B_1 + I_1) \) and \( S_2 = \bar{U}_2/(B_2 + I_2) \).

2.2.1 Derivation of \( S_1 \)

Since the number of slots in an idle period and a busy period are geometrically distributed with mean \( 1/(1 - e^{-aG_1}e^{-aG_2}) \) and \( 1/(e^{-(G_1+aG)}) \), respectively, we have
\[
I_1 = \frac{a}{1 - e^{-aG}}
\]  
and
\[
B_1 = \frac{1 + a}{e^{-(G_1+aG)}}
\]
To find \( \bar{U}_1 \), one can compute the probability of success over the first \( TP \) and the probability of success over any other \( TP \). The mean number of \( TP \) is \( 1/(e^{-(G_1+aG)}) \); therefore,
\[
\bar{U}_1 = \frac{aG_1e^{-aG}}{1 - e^{-aG}} + [e^{(G_1+aG)} - 1] \frac{G_1(1 + a)e^{-(G_1+aG)}}{1 - e^{-(G_1+aG)}}
\]

\(^1\) A Markov chain approach can also be used. In [16], this method is applied to compute the throughput of a system with three different CSMA protocols.
Hence,

\[ S_1 = \frac{G_1(1 + a - e^{-aG})e^{-(G_1 + aG)}}{(1 + a)(1 - e^{-aG}) + ae^{-(G_1 + aG)}} \]  

(2.5)

2.2.2 Derivation of \( S_2 \)

If \( \tilde{I} \) denotes the number of idle slots in an idle period, then

\[ Pr[\tilde{I} = k] = e^{-(G_1 + aG)}(e^{-aG})^{k-1}(1 - e^{-aG}) \quad k \geq 1 \]  

(2.6)

Thus,

\[ \bar{I}_2 = \frac{ae^{-(G_1 + aG)}}{1 - e^{-aG}} \]  

(2.7)

In the non-persistent mode, \( \overline{B}_2 \) is simply a transmission period. Therefore, \( \overline{B}_2 = 1 + a \) and the average successful transmission is

\[ \overline{U}_2 = \frac{aG_2e^{-aG}}{1 - e^{-aG}} \]  

(2.8)

Putting \( \overline{U}_2, \overline{B}_2 \) and \( \bar{I}_2 \) together, we obtain

\[ S_2 = \frac{aG_2e^{-aG}}{(1 + a)(1 - e^{-aG}) + ae^{-(G_1 + aG)}} \]  

(2.9)

2.3 Throughput Characteristics

In the previous section, two expressions for \( S_1 \) and \( S_2 \) are derived in terms of the normalized propagation delay, \( a \), channel traffics of both classes, \( G_1 \) and \( G_2 \). In order to investigate the throughput characteristics of both classes, the approach in [16] is followed: introduce one more parameter, \( \alpha \), which is defined as follows.

\[ G_1 = \alpha G; \quad G_2 = (1 - \alpha)G; \quad 0 \leq \alpha \leq 1 \]  

(2.10)
The throughput characteristics of class 1 and class 2 for \( \alpha = 0.3 \) and 0.6 are shown in Figures 2.1 and 2.2. The value of \( \alpha \) is kept at 0.01, and is the same for all the results presented in this chapter. Each figure is composed of parts (a) and (b). Part (a) shows the throughput of class 1 and class 2 for a specific value of \( \alpha \). Part (b) compares the resultant throughput \( (S_1 + S_2) \) of this heterogeneous network with that of a network using either 1-persistent or non-persistent protocol. The resultant throughput is depicted by a dotted line which terminates at the maximum of \( S_1 \) or \( S_2 \), whichever occurs first. The reason for this will be explained in the next section.

As shown in Figure 2.1, when \( \alpha = 0.3 \), both classes have their maximum throughput occurred at the light traffic regions, which is similar to the 1-persistent. But as \( \alpha \) increases to 0.6, class 2 performs very badly and the maximum throughput is about 10%. The reason is that when class 1 traffic is dominant, the chance for class 2 users sensing a busy channel increases, which forces the ready users to become backlogged, thereby driving \( S_2 \) down. Observations from other \( \alpha \)'s show that when \( \alpha = 0.23 \), the maximum of \( S_1 \) and \( S_2 \) are very close, and that the resultant throughput cannot be greater than the case of the non-persistent protocol.

### 2.4 Delay Characteristics

The delay performance is investigated by simulation. To be confident of the simulator, it is validated by comparing the \((S,G)\) relationship obtained from simulation with those obtained from analytic expressions of \( S_1 \) and \( S_2 \); they match very well.

The normalized mean packet delay of both classes for \( \alpha = 0.3 \) and 0.6 are shown in Figure 2.3. It is a well known fact that the packet delay becomes enormously
Figure 2.1: Throughput characteristics of $S_1$ and $S_2$ ($\alpha=0.3$)
Figure 2.2: Throughput characteristics of $S_1$ and $S_2$ ($\alpha=0.6$)
(a) Packet delay versus throughput for $\alpha = 0.3$

(b) Packet delay versus throughput for $\alpha = 0.6$

Figure 2.3: Packet delay of class 1 and class 2 versus throughput for constant $\alpha$
large in the vicinity of maximum throughput. There is no exception in the case of two classes operating in the same network, packet delay approaches infinity as maximum of $S_1$ or $S_2$ is reached. Since these maximum points occur at different values of $G$, problems arise when adding $S_1$ and $S_2$. When $G_1$ and $G_2$ increase in such a way that $\alpha$ is maintained constant, users in one group may still enjoy the low delay, but those in the other group suffer from an infinitely large delay. Simulated results shown in Figure 2.3 support this observation. In consequence, the resultant throughput cannot be added arbitrarily. In fact it is dependent upon when the maximum throughput of each class occurs. That is why the dotted line terminated in part (b) of Figures 2.1 and 2.2.

Thus far the delay-throughput performance has been shown for constant value of $\alpha$. The case of various $\alpha$ is shown in Figures 2.4 and 2.5. Figure 2.4 describes the trade-off of packet delay and throughput when the input rate of the other class is fixed at 0.1, 0.2, and 0.3. As shown, the class 1 delay is low whereas the class 2 delay increases sharply. This is largely due to the inherent properties of 1-persistent and non-persistent protocols. The case of collision detection is shown in Figure 2.5. Collision recovery time is assumed to be two slots. Compared with Figure 2.4 (the case with no collision detection) one can notice the delay and throughput are greatly improved.

A CSMA/CD priority scheme using non-persistent and 1-persistent protocols has been proposed in [6]. However, their scheme requires an initialization signal (a form of reservation period) before a packet is transmitted. But one can see from Figures 2.4 and 2.5 that the class 1 packet delay is lower than that of class 2. In case of collision detection, the throughput-delay performance is even better. In other
Figure 2.4: Simulated delay-throughput performance (without collision detection)
CHAPTER 2.

Figure 2.5: Simulated delay-throughput performance (with collision detection)
words, the presence of class 2 users has little effect on the users of class 1, except when $S_1 + S_2$ approaches maximum throughput. This nice feature, therefore, can be used as a non-preemptive priority scheme. This scheme is simple because users have to follow the 1-persistent and non-persistent protocols. No further actions or reservation periods are required from the users.

2.5 Summary

A network with two groups of users, one using the 1-persistent, the other the non-persistent CSMA protocol, is considered in this chapter. Two expressions of throughput (without collision detection), one for each group, is derived. It is found that the resultant throughput is dependent on where the maximum throughput of each group occurs. In general, this throughput cannot perform better than the case if both groups apply the non-persistent protocol. However, one can notice, from the simulated delay-throughput performance, that a simple non-preemptive priority scheme can be implemented by simply assigning the higher priority a 1-persistent mode and the lower priority a non-persistent mode.
Chapter 3

Performance Evaluation of \( p \)-persistent CSMA with Two Persistency Factors

Recently, a priority resolution scheme using \( p \)-persistent CSMA/CD was proposed in [17]. In this scheme, the higher the priority of a packet, the higher its \( p \)-persistent probability for transmission. For example, if there are two classes of packets when the channel is sensed idle, the higher priority packets transmit with probability \( p_1 \) (persistency factor), while the lower ones transmit with probability \( p_2 \), where \( p_1 > p_2 \). In [17], the protocol was evaluated in terms of non-transmitting slots.\(^1\) However, the throughput-delay performance is also of great interest. To the best of the author’s knowledge, there has been no such performance analysis in the literature to date. Therefore, this chapter provides a throughput analysis of \( p \)-persistent CSMA (without collision detection) with two persistent factors. The

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\(^1\)Most of the priority schemes are implemented by reservations. As a result, time slots are wasted when there are no high priority packets. The protocol proposed in [17] employed probabilities rather than reservation to resolve conflicts; therefore, no time slots are wasted when there are no high priority packets.
approach adopted in the analysis can be extended to the case of any number of persistent factors, although the derivations will be very cumbersome. Both the infinite users case and finite users case are considered. As the delay analysis is of great difficulty and complexity, it is evaluated by simulation.

3.1 Review of Protocol

The slotted $p$-persistent CSMA is considered here. The time is quantized into slots of $\tau$ seconds, where $\tau$ is the end-to-end propagation delay. Besides, all users are forced to start transmission at the beginning of a slot. If a user is ready, it proceeds as follows:

1. If the channel is sensed idle, it transmits with probability $p$; with probability $(1 - p)$, it defers the transmission to the next slot.

2. At this new slot, if this user still senses an idle channel, step (1) is repeated; otherwise, this user reschedules its packet according to the retransmission algorithm.

3. If the channel is sensed busy, it persists sensing the channel until it is idle, and then it will go to step (1).
3.2 Throughput Analysis: infinite users

The analysis provided here focuses on two groups of an infinite number of users but the approach can be extended to any number of groups. For discussion purposes, users in the high priority group are named class 1, and those in the low priority group class 2. The persistency factors assigned to class 1 and class 2 are $p_1$ and $p_2$, respectively. As in the previous chapter, $S_1$ and $G_1$ denote the throughput and channel traffic of class 1; likewise, $S_2$ and $G_2$ denote those of class 2. The total channel traffic is $G_1 + G_2 = G$. Packet arrival processes of both classes and channel traffic are assumed Poisson. Besides, the arrival processes of class 1 and class 2 are assumed independent. The time axis is normalized to the packet transmission time, where $a$ denotes the normalized propagation delay.

The derivation of the throughput of $p$-persistent CSMA with two persistent factors follows the approach in [9] in which the $p$-persistent CSMA was analyzed. Again, from the renewal theory, $S_1$ and $S_2$ are given by $U_1/(B + I)$ and $U_2/(B + I)$, respectively. The definitions of $U_1$, $U_2$, $B$, and $I$ are the same as those in the previous chapter. Note that $B$ and $I$ do not need superscripts 1 and 2, because they are identical for both class 1 and class 2.

3.2.1 Derivation of $S_1$

Figure 3.1 shows an example of the busy and idle period of the $p$-persistent protocol, together with some notations. If $N_1'$ and $N_2'$ are the number of class 1 and class 2 packets present at the last slot of the previous idle period, and $N_1$ and $N_2$
$TP$ : transmission period

$IRTD$ : initial random transmission delay

$a$ : normalized end-to-end propagation delay

$B$ : busy period

$I$ : idle period

Figure 3.1: Channel state of p-persistent CSMA: busy period and idle period
are the number of class 1 and class 2 packets accumulated at the end of a transmission period (TP), then according to the Poisson arrivals, and assuming the arrival processes for both classes are independent,

\[ \pi'_{n_1,n_2} = \Pr[N'_1 = n_1, N'_2 = n_2] \]
\[ = \frac{1}{1 - e^{-n_1 \cdot \nu_1} e^{-n_2 \cdot \nu_2}} \left( \frac{g_1^{n_1}}{n_1!} e^{-\nu_1} \right) \left( \frac{g_2^{n_2}}{n_2!} e^{-\nu_2} \right), \quad n_1 + n_2 > 0 \] (3.1)

where \( g_1 = aG_1 \), \( g_2 = aG_2 \), and,

\[ \pi_{n_1,n_2} = \Pr[N_1 = n_1, N_2 = n_2] \]
\[ = \frac{[(1 + a)G_1]^{n_1}}{n_1!} \exp\{- (1 + a)G_1\} \]
\[ \cdot \frac{[(1 + a)G_2]^{n_2}}{n_2!} \exp\{- (1 + a)G_2\}, \quad n_1 + n_2 \geq 0 \] (3.2)

The average length of the IRTD between two consecutive TP's can be determined as follows. Let \( q_1 = 1 - p_1 \), \( q_2 = 1 - p_2 \), and \( t_{n_1,n_2} \) be the number of slots elapsed until some class 1 or class 2 packet is transmitted, given that \( N_1 = n_1, N_2 = n_2 \). Then,

\[ \Pr[t_{n_1,n_2} > k] = q_1^{(k+1)n_1} \exp\left( g_1 \left( \frac{q_1(1 - q_1^k)}{p_1} \right) - k \right) \]
\[ \cdot q_2^{(k+1)n_2} \exp\left( g_2 \left( \frac{q_2(1 - q_2^k)}{p_2} \right) - k \right) \] (3.3)
Therefore, for $k > 0$,

$$
\Pr[t_{n_1,n_2} = k] = g_1^{kn_1}q_2^{kn_2} \left[1 - q_1^{n_1}q_2^{n_2} \exp\{-g_1(1 - q_1^k) - g_2(1 - q_2^k)\}\right] \\
\cdot \exp\left[g_1\left(\frac{q_1(1 - q_1^{k-1})}{p_1} - (k - 1)\right)\right] \\
+ g_2\left(\frac{q_2(1 - q_2^{k-1})}{p_2} - (k - 1)\right) \\
$$

(3.4)

and for $k = 0$,

$$
\Pr[t_{n_1,n_2} = 0] = 1 - g_1^{n_1}q_2^{n_2} \\
$$

(3.5)

Removing the condition on $N_1$ and $N_2$, the average IRTD is given by

$$
\bar{t} = \sum_{n_1+n_2 > 0} \tau_{n_1,n_2} \frac{\pi_{n_1,n_2}}{1 - \pi_{0,0}} \\
$$

(3.6)

where,

$$
\tau_{n_1,n_2} = \sum_{k=0}^{\infty} \Pr[t_{n_1,n_2} > k] \\
$$

(3.7)

The computation of the probability of success over a transmission period (TP) is split into two parts: the first TP; and, then any other TP's. Let $TP_i$ denotes the $i^{th}$ TP in the busy period, $P_s^i(n_1,n_2)$, the probability of success for class 1 packets over $TP_i$, and $L_{n_1,n_2}$, the number of class 1 and class 2 packets present when $TP_i$ starts.
First consider the case $i \neq 1$. Given $N_1 = n_1$ and $N_2 = n_2$, 

$$
\Pr[L_{n_1,n_2} = (l_1,l_2)] = \sum_{k=1}^{\infty} \frac{(kg_1)^{l_1-n_1}}{(l_1-n_1)!} \frac{(kg_2)^{l_2-n_2}}{(l_2-n_2)!} e^{-kg_1} \cdot e^{-kg_2} \cdot \Pr \{ t_{n_1,n_2} = k \} 
+ (1 - q_1^{n_1} q_2^{n_2}) \delta_{l_1,n_1} \delta_{l_2,n_2}, \quad (l_1, l_2) \geq (n_1, n_2) \tag{3.8}
$$

where $\delta_{i,j}$ is the impulse function. Hence, the probability of success for class 1 packets over $TP_i$, $i \neq 1$ is given by

$$
P_{s,i}^1(n_1, n_2) = \sum_{l_1=n_1}^{\infty} \sum_{l_2=n_2}^{\infty} \frac{l_1! p_1 q_1^{l_1-1}}{1 - q_1^{n_1} q_2^{n_2}} q_2^{l_2} \cdot \Pr[L_{n_1,n_2} = (l_1,l_2)] \tag{3.9}
$$

Unconditioning $N_1$ and $N_2$, we obtain

$$
P_{s,i}^1 = \sum_{n_1 + n_2 > 0}^{\infty} P_{s,i}^1(n_1, n_2) \frac{\pi_{n_1,n_2}}{1 - \pi_{0,0}} \tag{3.10}
$$

Now consider the case $i = 1$. Let $t'_{n_1,n_2}$ be the initial IRTD, and $P_{s,1}^1(n_1, n_2)$, the probability of success for class 1 packets over $TP_1$. Although using different notations, these two quantities have the same distributions as those shown in equations (3.3) and (3.9), except that they are conditioned on $N'_1$ and $N'_2$. Thus removing these conditions yields,

$$
\bar{t}' = \sum_{n_1 + n_2 > 0}^{\infty} \bar{t}_{n_1,n_2} \pi'_{n_1,n_2} \tag{3.11}
$$
and,

\[ P_{s}^{1} = \sum_{n_{1}+n_{2}>0} P_{s}^{1}(n_{1}, n_{2}) \pi_{n_{1}, n_{2}} \]  \hspace{1cm} (3.12)

The task now is to express \( \overline{B} \) in terms of \( \bar{t}, \bar{t}', \) and \( \overline{U}_{1} \) in terms of \( P_{s}^{1}, P_{s}^{1}. \)

Notice that the number of TP's in a busy period is geometrically distributed with parameter \( \pi_{0,0}. \) Thus given that there are \( m \) TP’s, one can write

\[ \overline{B}_{m} = a\bar{t}' + (m - 1)a\bar{t} + m(1 + a) \]  \hspace{1cm} (3.13)

and,

\[ \overline{U}_{1m} = P_{s}^{1} + (m - 1)P_{s}^{1} \]  \hspace{1cm} (3.14)

The final expressions for \( \overline{B} \) and \( \overline{U}_{1} \) can be obtained by removing the condition on \( m. \) Hence,

\[ \overline{B} = \sum_{m=1}^{\infty} \overline{B}_{m} \pi_{0,0}(1 - \pi_{0,0})^{m-1} \]
\[ = a\bar{t}' + \frac{a\bar{t}(1 - \pi_{0,0}) + 1 + a}{\pi_{0,0}} \]  \hspace{1cm} (3.15)

and,

\[ \overline{U}_{1} = \sum_{m=1}^{\infty} \overline{U}_{1m} \pi_{0,0}(1 - \pi_{0,0})^{m-1} \]
\[ = P_{s}^{1} + \frac{1 - \pi_{0,0}}{\pi_{0,0}} P_{s}^{1} \]  \hspace{1cm} (3.16)
The average idle period is geometrically distributed with mean $1/(1 - e^{-\beta_1}e^{-\beta_2})$; therefore,

$$I = \frac{a}{1 - e^{-\beta_1}e^{-\beta_2}} \quad (3.17)$$

Putting equations (3.15), (3.16), and (3.17) into $\frac{U_i}{(B + I)}$, expression for $S_1$ is obtained.

### 3.2.2 Derivation of $S_2$

The expression for $S_2$ can be similarly written by noting that expressions for $B$ and $I$ are the same as those in $S_1$, and that the probability of success for class 2 packets over $TP_i$, $P^2_i(n_1, n_2)$, is given by

$$P^2_i(n_1, n_2) = \sum_{l_1=n_1}^{\infty} \sum_{l_2=n_2}^{\infty} \frac{l_2 p_1 q_1^{l_2-1}}{1 - q_1^{l_1} q_2^{l_2}} \cdot q_1^{l_1} \cdot \Pr[L_{n_1, n_2} = (l_1, l_2)] \quad (3.18)$$

The procedures for obtaining $S_2$ would be exactly the same as those for $S_1$.

### 3.3 Throughput Characteristics: infinite users

Recall that $S_1$ and $S_2$ are expressed in terms of $G_1$, $G_2$, $p_1$, $p_2$, and $a$. As in the previous chapter, another parameter $\alpha$ is required so that $G_1$ and $G_2$ are related to $\alpha$ and $G$ as follows.

$$G_1 = \alpha G, \quad G_2 = (1 - \alpha)G, \quad 0 \leq \alpha \leq 1 \quad (3.19)$$
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Strictly speaking, there are innumerable combinations of \( p_1, p_2, \) and \( \alpha \). Thus, it is unrealistic to show the effects of these parameters on \( S_1 \) and \( S_2 \). In view of this, three representative cases of different \( p_1 \)'s and \( p_2 \)'s are shown from Figures 3.2 to 3.4, in which the values of \( \alpha \) are fixed at 0.2, 0.5, and 0.8. Also, \( \alpha = 0.01 \) is used in this chapter. Observations from these figures lead to the following points:

1. When \( p_1 \) and \( p_2 \) are close, such as the case shown in Figure 3.2, the maximum of \( S_1 \) and \( S_2 \) occurs almost at the same value of \( G \). Furthermore, the shapes of \( S_1 \) and \( S_2 \) are very similar.

2. As observed from Figures 3.3 and 3.4, the peaks of \( S_1 \) and \( S_2 \) start shifting more when \( p_1 \) and \( p_2 \) are further apart. In other words, for a specific value of \( \alpha \), the peaks of \( S_1 \) and \( S_2 \) may occur at different values of \( G \). Consequently, as explained in the previous chapter, the resultant throughput cannot be summed arbitrarily. Also, the shapes of \( S_1 \) and \( S_2 \) are different, which is typified by a case of \( \alpha = 0.2 \) in Figure 3.4.

Another interesting thing to look at is the resultant throughput (\( S_1 + S_2 \)) of this protocol. Figure 3.5 shows the maximum throughput versus \( \alpha \) for different values of \( p_1 \) and \( p_2 \). The maximum throughput here is obtained by adding \( S_1 \) and \( S_2 \) up to the peak value of either \( S_1 \) or \( S_2 \), whichever occurs first. Another point which should be mentioned is that when \( \alpha = 0 \) or \( \alpha = 1 \) all users adopt \( p_2 \)-persistent (\( \alpha = 0 \)) or \( p_1 \)-persistent (\( \alpha = 1 \)) protocol. Figure 3.5 indicates that the maximum throughput decreases from the maximum point to the lowest point as \( \alpha \) varies from 0 to 1. Although four combinations of \( p_1 \) and \( p_2 \) are shown in Figure 3.5, a lot of cases have been tested and they all show this phenomenon. The conclusion here is
Figure 3.2: Throughput characteristics of $S_1$ and $S_2$, infinite users ($p_1 = 0.9$, $p_2 = 0.6$, $\alpha = 0.2, 0.5, 0.8$)
Figure 3.3: Throughput characteristics of $S_1$ and $S_2$, infinite users ($p_1 = 0.8$, $p_2 = 0.3$, $\alpha = 0.2, 0.5, 0.8$)
Figure 3.4: Throughput characteristics of $S_1$ and $S_2$, infinite users ($p_1 = 0.8$, $p_2 = 0.03$, $\alpha = 0.2, 0.5, 0.8$)
Figure 3.5: Maximum throughput of $p$-persistent CSMA with two persistency factors versus $\alpha$
that the maximum throughput of this protocol is upper-bounded by the maximum throughput of $p_2$-persistent and lower-bounded by that of $p_1$-persistent.

### 3.4 Throughput Analysis: finite users

This section is mainly motivated by two facts. The first one is the resultant throughput. As mentioned before, the resultant throughput cannot be arbitrarily added for a specific value of $\alpha$. The second motivation comes from [18]. The authors in [18] carried out a throughput analysis of $p$-persistent when there were a finite number of users, and pointed out that the throughput did not degrade to zero when $p$ was small. Motivated by these two results, one would like to investigate the performance of this protocol under the condition of finite users.

Expressions of $S_1$ and $S_2$ can be easily derived by following the approach in [18]. As shown in Appendix A, $S_1$ and $S_2$ are given by,

$$S_1 = \frac{p_1 M_1 \sum_{k=0}^{\infty} A \cdot B^{M_1-1} \cdot C^{M_2}}{1 + a + a \sum_{k=1}^{\infty} D^{M_1} \cdot E^{M_2}} \quad (3.20)$$

where,

$$A = (1 - p_1)^k - (1 - g_1)^{1+(1/a)} \left[ \frac{p_1 (1 - p_1)^k - g_1 (1 - g_1)^k}{p_1 - g_1} \right] \quad (3.21)$$
$B = (1 - p_1)^{k+1} - (1 - g_1)^{1+(1/\alpha)} p_1 \left( \frac{(1 - p_1)^{k+1} - (1 - g_1)^{k+1}}{p_1 - g_1} \right)$ \hfill (3.22)

$C = (1 - p_2)^{k+1} - (1 - g_2)^{1+(1/\alpha)} p_2 \left( \frac{(1 - p_2)^{k+1} - (1 - g_2)^{k+1}}{p_2 - g_2} \right)$ \hfill (3.23)

$D = (1 - p_1)^k - (1 - g_1)^{1+(1/\alpha)} p_1 \left( \frac{(1 - p_1)^k - (1 - g_1)^k}{p_1 - g_1} \right)$ \hfill (3.24)

$E = (1 - p_2)^k - (1 - g_2)^{1+(1/\alpha)} p_2 \left( \frac{(1 - p_2)^k - (1 - g_2)^k}{p_2 - g_2} \right)$ \hfill (3.25)

Also, $G_1 = \alpha G$, $G_2 = (1 - \alpha)G$, $0 \leq \alpha \leq 1$, $g_1 = aG_1/M_1$, and $g_2 = aG_2/M_2$. $M_1$ and $M_2$ are the number of class 1 and class 2 users, respectively. All other notations here are the same as those defined for the infinite users case. Similarly, one can write $S_2$ by changing all subscripts 1 in equation (3.20) to 2.

### 3.5 Throughput Characteristics: finite users

The throughput characteristics are shown in Figures 3.6 and 3.7, which include part (a) and part (b). Part (a) shows the throughput of each class and the resultant throughput is shown in part (b). Also depicted in part (b) are the throughputs of $p_1$- and $p_2$-persistent CSMA with $M = 10$. As shown in both figures, the throughputs
Figure 3.6: Throughput characteristics of $S_1$ and $S_2$, finite users ($\alpha = 0.1$)
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(a) $S_1$ and $S_2$ versus $G$

(b) Throughput comparison

Figure 3.7: Throughput characteristics of $S_1$ and $S_2$, finite users ($\alpha = 0.3$)
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of both classes do not drop to zero eventually. Part (b) of Figure 3.7 shows the maximum resultant throughput is bounded by the maximum throughputs of 0.05- and 0.3-persistent. However, part (b) of Figure 3.6 reveals the maximum resultant throughput can be slightly higher. Recall that $p_i$ is the transmission probability of class 1 users and this value, shown in both figures, is quite low; this may result in an undesirable large delay. If $p_i$ (or $p_j$) is raised to a higher value, such as shown in Figure 3.8, both $S_1$ and $S_2$ degrade to zero. It is observed from other results that the throughput of either class, in general, does not degrade to zero ultimately if both persistent factors are small. In summary, if the protocol proposed in [17] is applied to an environment, in which there are finite number of users and no collision detection is involved, one has to take the stability problem into account when evaluating the throughput performance.

3.6 Delay Characteristics

The complexity of the delay-throughput analysis involved in the $p$-persistent CSMA with priority can be seen in [8]. It was mentioned there that the procedure is computationally expensive to be of practical use if the system is large (e.g., if the number of users is greater than 5). In view of this, one would like to rely on simulation to study the trade-off of throughput and delay. Furthermore, the primary interest here is to examine the delay performance of $p$-persistent with two persistent factors, therefore a simulator assuming there are infinitely many users was written. The main reason for this assumption is that it usually requires less time to acquire equilibrium results. This simulator, like the one in chapter 2, is tested by comparing the $(S_1,G_1)$ and $(S_2,G_2)$ relationships obtained from simulation with
Figure 3.8: Throughput characteristics of $S_1$ and $S_2$, finite users ($\alpha = 0.5$)
those obtained from analytic expressions of $S_1$ and $S_2$.

Figures 3.9 and 3.10 show the normalized mean packet delay versus $S_1$ and $S_2$ without collision detection and with collision detection, respectively. In both figures, the packet delay of class 1 and class 2 are obtained by fixing the input rate of the other class at 0.1, 0.2, and 0.3. Also, to study the sensitivity of packet delays to $p_1$ and $p_2$, two pairs of persistent factors are compared. They are (0.8, 0.03) and (0.7, 0.1), where (*,*) represents ($p_1$, $p_2$). One can clearly observe from part(b) of both figures that packet delay for class 2 with persistent factor 0.03 is larger than that with 0.1. But there is hardly any difference in class 1 packet delays in part(a) of Figure 3.9, which is reasonable as 0.8 and 0.7 are very close. However, when there is collision detection, $S_1$ can be increased further and the effects coming from class 2 become more conspicuous. As shown in part (a) of Figure 3.10, the class 1 packet delay with persistent factor 0.8 is lower than that with 0.7, when the system approaches saturation. This is mainly because class 2 users with persistent factor 0.03 have little influence on class 1 users who are using 0.8.

3.7 Summary

A throughput-delay performance of p-persistent CSMA (without collision detection) with two persistency factors is examined. The protocol was originally proposed in [17]; however, no throughput-delay performance was available there. The objective of this chapter was to provide such information. The throughput of this protocol without collision detection is examined by deriving the throughput expression for each class. Both cases of infinite users and finite users are considered. It is observed
Figure 3.9: Simulated delay-throughput performance (without collision detection)
Figure 3.10: Simulated delay-throughput performance (with collision detection)
that when the number of users is infinite the maximum throughput of this protocol is upper-bounded by that of $p_2$-persistent and lower-bounded by that of $p_1$-persistent. The trade-off of throughput and delay is investigated by simulation. The results indicate that the class 1 packet delay is lower than that of class 2 and that the transmission probability of class 2 has a definite effect on class 1.
Chapter 4

Protocol for Integrated Voice and Data Transmission in LAN

There have been many studies on voice transmission or integrated voice and data transmission in the local area network (LAN). Results for CSMA/CD based channels have indicated the data load would have definite effects on voice transmission [19] [20]. To avoid the interference from data packets, protocols are specifically devised so as to achieve good performance for both voice and data packets. However, these protocols require either a complex algorithm [13] or a special packet format [12]. One would, certainly, notice the simplicity of the $p$-persistent CSMA/CD with two persistency factors. In addition, it has been shown in chapter 3 that its delay-throughput characteristics are well behaved, under data traffic. This chapter, therefore, proposes the application of this simple protocol for combined voice and data transmission. Simulation results are presented to evaluate its performance and comparisons with other works, which use CSMA/CD, are also discussed.
CHAPTER 4.

4.1 Channel Access Protocol

Both the voice and data packets acquire the channel by following the slotted $p$-persistent CSMA/CD protocol. Voice packets are transmitted with probability $p_v$ and data packets with probability $p_d$. As explained below, voice packets should avoid long delay and are of higher priority over data packets; therefore, $p_v > p_d$. In case of collisions the packets involved are retransmitted according to the binary exponential backoff algorithm. This will be elaborated below.

4.2 Characteristics and Requirements for Voice and Data Traffic

The traffic characteristics for voice and data are very dissimilar. Although voice traffic has been regarded as a continuous source, it has been shown [24] that a typical conversation has an on-off characteristic and can be modeled as having alternate "talkspurts" and "silences". The lengths of these talkspurts and silences are exponentially distributed. Also, to keep voice intelligible voice packets must be under a very stringent delay constraints. Furthermore, delay variance must be small [21]. Good voice quality can still be maintained if 2% of voice packets are discarded [21]. Recent results [22] also reveal that a higher percentage (about 5%) of the voice packets loss rate is acceptable without sacrificing the quality and intelligibility of speech. The data traffic, on the other hand, is bursty in nature. Data packets arrive sporadically and long delay and large variance in packet delay can be tolerated [21]. However, loss of data packets is generally considered unacceptable.
4.3 Simulation Model

In this simulation model there are \( n \) finite number of calls. Each of the \( n \) speakers alternates independently between intervals of talkspurt and silence. The number of calls in each simulation is maintained constant and the voice throughput, \( \rho_v \), can be adjusted by varying the value of \( n \). Note that variations in the number of speakers due to calls entering and leaving the system are not included in this study because the variations of talkspurt/silence occur much more frequently than the variations of \( n \).

The number of data sources, on the other hand, is assumed to be infinite. Collectively these form an independent Poisson source. Besides, the buffer size is assumed infinitely large; therefore, no data packet is lost. The data throughput, \( \rho_d \), can be varied by changing the data packet interarrival time, which has an exponential distribution.

In the simulation the durations of talkspurt and silence period are assumed to be exponentially distributed with mean \( \tau_t \) and \( \tau_s \), respectively. As a call is active less than 50% of the time [23], two speech activity factors,\(^1\) \( saf \), are used; they are 42% and 45%. Note that for \( saf = 42\% \) (or 45%), \( \tau_t = 1.0 \) (or 1.34), and \( r_s = 1.4 \) (or 1.67) [24] are used. It is assumed that the coding rate is 64 Kbps and that the channel transmission rate is 10 Mbps.

The effects of voice packet length on the performance of voice transmission will be studied; the packet length chosen to be 5.75 ms, 10 ms, and 28 ms. Assuming

\(^1\)speech activity factor, \( saf \), is given by \( \tau_t / (\tau_t + \tau_s) \).
there are 208 overhead bits [20] which include 64 synchronization bits, 112 bits of header and 32 bits CRC field, the voice packet length for frame size of 5.75 ms, 10 ms, and 28 ms are 576 bits, 848 bits and 2000 bits, respectively. To facilitate programming, the length of data packet is the same as that of voice packet.

The time axis in the simulator is assumed slotted; the size of a slot is the propagation delay normalized by the packet transmission time. Due to the propagation delay, the channel remains busy for a certain time after a successful transmission. This time is assumed to be 1 slot. If collision occurs, 3 slots are required to change from the busy state to the idle one. Regardless of the type of packet, the packets involved in a collision are rescheduled for a later retransmission by using the same algorithm: binary exponential backoff (BEB).

The BEB algorithm is as follows. If the first attempt fails, the $n^{th}$ retransmission is scheduled at the point $r$ slots from the time the channel is sensed idle again, where $r$ is chosen at random from the set $\{0, 1, 2, \ldots, 2^k - 1\}$ and $k = \min\{n, 8\}$. If at this new scheduling time the channel is sensed busy, the same procedure is repeated but $n$ is not incremented. Note that if a packet has more than eight collisions, the same interval will be used for selecting a retransmission time. Although both the data and voice packets adopt same BEB algorithm for retransmission, they stay in the system in a different manner. Voice packets remain in the system until the delay exceeds the voice frame size and thus are discarded. Unlike voice packets, data packets stay in the system until they are successfully transmitted.
4.4 Simulation Results

A computer simulation, written in C language, was developed to study the performance of both voice and data transmission using p-persistent CSMA/CD with two persistency factors. Simulation length of each run is about 5 min. The results are divided into two parts. The first part examines the performance of voice transmission; the second part studies the effects of data load on the voice transmission. Comparisons with other works are also discussed.

4.4.1 Voice Transmission

Figure 4.1 shows the voice packet loss rate (in %) as a function of number of calls for various frame sizes and values of $a$, the normalized propagation delay. For 5.75 ms and 10 ms frame sizes, $saf$ is 45%, whereas $saf$ is 42% for 28 ms frame size. Results are obtained without any data load; only voice packets are allowed in the system and they are transmitted with $p_v = 0.8$. Recall that voice packets are discarded if they are delayed to a time longer than the frame size.

As shown in Figure 4.1, the number of calls that can be accommodated in the system increases as the voice frame size increases. This is understandable as there are more slots in a larger frame. If 2% is the maximum allowable loss rate, the system can support approximately 130 and 170 calls for frame size of 5.75 ms and 10 ms, respectively. The effects of propagation delay can also be observed. For the 28 ms frame size, if the value of $a$ is doubled, the number of calls drops from 290 to 262, approximately.
Figure 4.1: Voice packet loss rate versus number of calls, n
4.4.2 Integrated Voice and Data Transmission

The primary objective here is to examine the effects of data load on the performance of voice transmission. In this simulation study, the number of calls \( n \) is fixed (i.e. \( p_v \) is a constant) and the data throughput \( (p_d) \) is varied gradually until the system approaches saturation. The parameters in the simulation are \( p_v = 0.8, p_d = 0.03, saf = 42\%, a = 0.0125, \) and voice frame size is 28 ms. Also, the delay of voice packet (or data packet) has been normalized to the voice packet (or data packet) transmission time.

The mean delay of voice packet and data packet versus data throughput when \( n = 50, 100, 150, \) and 200 is described in Figure 4.2. As can be observed, the delay of voice packet is well below that of data packet. When \( n \) is small, voice packet delay increases almost linearly whereas data delay increases exponentially. Note that there is a limitation on the level of data throughput for a specific \( n \). This can be observed from the delay of data packets; this becomes very large at higher data throughput.

The loss of voice packets resulted from the combined voice and data transmission is described in Figure 4.3. For \( n = 50, 100, 150 \) and various values of \( p_d \), very few voice packets are discarded. Loss of voice packets becomes appreciable when \( n = 200 \) and increase sharply as \( p_d \) is further increased. This phenomenon, to a large extent, is attributed to the parameters \( p_v \) and \( p_d \). Recall that \( p_v = 0.8 \) and \( p_d = 0.03 \) are used in the simulation. As values of \( p_d \) is very small compared with that of \( p_v \), the chance for both voice and data packets to transmit at the same slot is very rare. This reduces significantly the frequency of collisions between voice packet
Figure 4.2: Normalized mean delay versus data throughput ($\rho_d$)
Figure 4.3: Voice packet loss rate versus data throughput ($\rho_d$)
and data packet. Loss of voice packets, therefore, is mainly due to collision among voice packets. When $n$ is large, for example 200, chance of collisions increases. More collisions require more reschedules and voice packets are further delayed. As a result, voice packets which have been delayed to a time longer than the voice frame, are discarded.

Figure 4.4 shows the standard deviation of delay for both voice and data packets. As shown in the figure, voice packet has a low standard deviation of delay. It increases only at large $n$. Data packet, however, has a large standard deviation of delay. This large value should not pose a problem as data packet, as mentioned above, can tolerate high delay variance. All these results indicate that the protocol can be used for combined voice and data transmission. Moreover, two basic requirements for voice traffic, namely small delay and low delay variance, have been achieved.

### 4.4.3 Comparisons and Discussions

It would be interesting to compare these with other schemes that use CSMA/CD. One should notice that because of different values of parameters and model assumptions, a strict comparison is inappropriate. For example, some schemes do not discard voice packet if the delay exceeds the voice frame size. Besides, some works reported in the literature do not describe how a conversation is modeled. In the following, it is assumed that two calls represent one conversation.

Some measurements of voice packet transmission in the Ethernet has been reported in [25]; however, the results obtained there were under different parameters.
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\[ \rho_v = 0.15 \text{ (50 calls) } \]
\[ \rho_v = 0.30 \text{ (100 calls) } \]
\[ \rho_v = 0.45 \text{ (150 calls) } \]
\[ \rho_v = 0.60 \text{ (200 calls) } \]

Figure 4.4: Standard deviation of delay versus data throughput (\(\rho_d\))
Models and parameters that are similar to those used in this study are found in [20] and [13].

In [20], simulation results of voice transmission with frame size of 5.75ms are obtained in the Ethernet, in which 1-persistent CSMA/CD is adopted. Results from [20] show that about 50 to 55 conversations (100 to 110 calls) can be supported, assuming 2% of voice packets are discarded. It has been shown in the previous section that with 5.75ms frame size the system can support about 130 calls. This suggests an observation. Compared with the 1-persistent, 0.8-persistent CSMA/CD can resolve conflicts better. This results in higher utilization (at the expense of higher voice packet delay but not large enough to increase the voice packet loss rate). A complicated protocol 2 is proposed in [13]. With 6 ms (slightly larger than 5.75 ms) frame size, that scheme can support about 75 conversations (150 calls), which is higher than 130 calls reported here. However, considering the simplicity of 0.8-persistent CSMA/CD and the complexity of the protocol in [13], it may be justified to have a lower number of calls.

Since different frame sizes (5.75 ms and 6 ms) are used for combined voice and data transmission in [20] and [13], comparison with their results is not appropriate. Nevertheless, observing the throughput, mean, and standard deviation of delay of voice and data packets, in addition to the simplicity of the protocol, it is reasonable to state that p-persistent CSMA/CD, with two persistency factors, is a viable scheme for integrated voice and data transmission in LAN.

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2Each user, in this scheme, is equipped with two clocks. One of the clocks, if permitted to run, runs \( \eta \) times faster than the other, where \( \eta \) is a system parameter. Transmission of voice packets is activated by an algorithm.
Obviously, the parameters $p_v$ and $p_d$ play a crucial part in the protocol. Throughout the simulation $p_v = 0.8$ and $p_d = 0.03$ are used. One may, therefore, wonder if these are the optimal values. More simulations can be carried out to determine the answer but these simulations are too time consuming and expensive, especially at the transmission rate of 10 Mbps. Consequently, these attempts are avoided.

4.5 Summary

In this chapter, the application of $p$-persistent CSMA/CD protocol in the integrated environment is proposed. Two types of traffic are studied, namely voice and data. Priority is achieved by assigning a higher persistency factor to the voice traffic, and a lower one to the data traffic. Performance of this scheme is evaluated through simulation. Evaluations are based on the voice packet loss rate, delay, variance of delay, and number of calls that can be supported. Results indicated that both the requirements of voice traffic and data traffic are satisfied.
Chapter 5

Integrated Voice and Data Transmission in Packet Radio Network

As briefly outlined in chapter 1, combining voice with data for transmission offers many advantages. One of them is cost savings. Because of these potential benefits, we propose a scheme for voice only or combined voice and data transmission which is applicable to the packet radio network. The proposed scheme does not require collision detection, attains an acceptable level of throughput if about 1% of voice packets are allowed to be discarded, limits the delay of voice packet to the voice frame size, and, most importantly, satisfies the stringent transmission requirements of voice packets which demand small mean and variance in delay. The proposed scheme is a variation of movable slot TDM [12] and \( p \)-persistent CSMA [9]. It takes advantages of the periodicity of voice packet generations while a voice user is in talkspurt, the bursty nature of data packets, and the random transmission of \( p \)-persistent CSMA. The performance of this scheme is evaluated by simulation.
5.1 Description of Protocol

In this section we first describe the access protocol for voice and data users and then explain the characteristics of the protocol. The operation of the protocol is assumed slotted. This requires all users to start transmissions only at the beginning of a slot. The size of a slot is equal to the end-to-end propagation delay.

During a talkspurt, voice packets are generated at a fixed period of time ($T$). For discussion purposes, we call the first packet of a talkspurt the “first voice packet” and subsequent packets other than the first one the “subsequent voice packet”. We use the term “voice packet” to refer both of them. We assume that voice packet size is fixed and that data packet size is not greater than that of a voice packet.

If a voice user has a ready first voice packet it proceeds as follows.

1. If the channel is sensed idle it transmits the packet with probability $p_v$ or defers the transmission to the next slot with probability $(1 - p_v)$. If this first voice packet is successfully transmitted at time $t$, the subsequent voice packet is scheduled for transmission at time $t + T$. However, if this first voice packet involves in a collision, or its delay exceeds the packet generation time ($T$), it is discarded. The subsequent voice packet immediately following the discarded one is regarded as the first voice packet and the algorithm is repeated.

2. If the channel is sensed busy it keeps on sensing the channel until the channel is idle. At this idle time, it further defers the transmission for one slot and senses the channel again. Dependent upon the state of the channel, step (1) or (2) is repeated.
The following steps are carried out for the subsequent voice packets.

1. If the channel is sensed idle, the subsequent voice packet is transmitted.

2. If the channel is sensed busy, the subsequent voice packet is transmitted as soon as the channel is idle; that is, the transmission time of this subsequent voice packet has been shifted to a new time \( t' \). If it suffers a collision, it is discarded. Regardless of the outcome of this packet, the next subsequent voice packet is scheduled at time \( t' + T \).

A ready data user operates as follows after the channel is sensed.

1. If the channel is idle, it transmits the data packet with probability \( p_d \), or with probability \( 1 - p_d \), the user defers the transmission to the next slot.

2. If the channel is busy, it senses the channel one slot after the channel becomes idle and repeats the algorithm.

3. If the data packet suffers a collision, it is retransmitted according to a retransmission algorithm (discussed in section 3).

We should note that the above access protocols for first voice packet and data packet are the variations of \( p \)-persistent CSMA and subsequent voice packet follows the \( 1 \)-persistent CSMA. The idea of shifting transmission time of subsequent voice packets is adopted from [12].

In the following we describe some characteristics of the proposed scheme. Due to the shifting of transmission time of subsequent voice packets, they would not
collide with one another. Also, the maximum time-shift is one packet transmission
time because at the new transmission time the subsequent voice packet is either
successfully transmitted or collided with other voice packets. The transmission of
the first voice packet, as we propose in the scheme, is probabilistic; it may still collide
with other voice packets if they transmit at the same slot with a first voice packet.
Should collisions occur, voice packets continue colliding on successive transmissions
as they are generated periodically. However, the proposed scheme stipulates that
if the first voice packet is discarded, the next subsequent voice packet is treated
as a first voice packet, acquiring the channel with probability $p_v$. This strategy,
therefore, reduces the chance of collision significantly. As will be seen from the
simulation results, the performance of this protocol is dependent on a judicious
choice of $p_v$. With the proposed scheme, a considerable number of voice packets
that are discarded are at the front parts of the talkspurts, which may cause speech
clipping. However, it has been observed from a listening test that this clipping does
not degrade the voice quality if the percentage of voice packets that are discarded
is small [15]. Note that when a subsequent voice packet is waiting for transmission,
voice bits are arriving which cannot be placed in the speech bits of voice packet.
Therefore, an overflow area is required for carrying these extra bits. As a result,
the format of voice packet is composed of header bits, speech bits, and overflow
bits, whereas the data packet consists of only header bits and data bits. When data
traffic is combined with voice, the former would inevitably affect the latter. To
mitigate this interference, we require $p_d < p_v$. 
5.2 Simulation Model

The approaches of simulation are very similar to those used in chapter 4. In the simulation model, there are $n$ finite number of calls. Each of the $n$ speakers alternates independently between intervals of talkspurt and silence. The number of calls, in each simulation, is maintained constant and the voice throughput, $\rho_v$, can be adjusted by varying the value of $n$. The number of data sources, on the other hand, is assumed to be infinite. These collectively form an independent Poisson source. Besides, the buffer size is assumed infinitely large; therefore, no data packet is lost. The data throughput, $\rho_d$, can be varied by changing the data packet interarrival time, which has an exponential distribution.

The durations of talkspurt and silence period are assumed to be exponentially distributed with mean 1.0 sec and 1.4 sec, respectively. Thus, the speech activity factor is 42% [23]. If the first voice packet is delayed, to a time longer than the voice frame size, it is assumed lost.

We first study the performance of voice transmission under the proposed scheme. In particular, we are interested in the voice packet loss rate versus the voice throughput with different $\rho_v$'s. Then we examine the effects of data load on the voice traffic.

Since the channel rate, voice packet length, coding rate, and $\rho_v$ are important to the system performance, several values are used in this simulation study. We use the notation $(tr, vf, cr, pr)$ to describe them, where $tr$, $vf$, $cr$, and $pr$ represent channel transmission rate in Kbps, voice frame size in msec, coding rate in Kbps, and $\rho_v$, respectively.
In the simulation of voice transmission we use \((10^3, 28, 32, p_v)\): the first set, \((32, 100, 2.4, p_v)\): the second set, and \((16, 20, 2.4, p_v)\): the third set, where \(p_v\) is varied to optimize the performance. We assume there are 172 header bits for the first set and 20 header bits for the second and third sets. Recall that a voice packet comprises header bits, speech bits, and overflow bits. To compute the required overflow bits in each set of parameters, we solve the following equation

\[
\frac{cr}{tr} (H + S + O) = O
\]  

(5.1)

where \(cr\) is the coding rate, \(tr\) the channel transmission rate, \(H\) the header bits, \(S\) the speech bits, and \(O\) the overflow bits. Therefore, \(O\) equals 36, 22, and 12 bits for the first, second, and third set, respectively. Also, the normalized propagation delay \((a)\) used in the simulation is 0.0125 for the first set and 0.01 is used for the remaining sets. If \(c\) denotes the number of slots available in a frame, then \(c=25.0\) slots for 28 msec, 11.2 slots for 100 msec, and 4.0 slots for 20 msec.

To evaluate the effects of data load on voice traffic, we choose the second set of parameters, and in the simulation we assume the size of data packet is the same as that of voice packet. If a data packet becomes involved in a collision, it is retransmitted as follows. If the first attempt fails, the \(n^{th}\) retransmission is scheduled at the point \(r\) slots from the time the channel is sensed idle again, where \(r\) is chosen at random from the set \(\{0, J, 2J, \ldots, (2^k - 1)J\}\), where \(k=\min\{n, 16\}\), and \(J\) is a constant which increases with the data throughput to ensure stable operation of the system. All of the afore-mentioned parameters do not correspond to a particular system; nonetheless, they provide an indication of how the proposed scheme operates.
5.3 Simulation Results and Discussions

The results presented here are obtained from a simulator written in C language. The value of each point in our figures is the average of three simulation runs. Each run requires at least 100,000 voice packets and data packets in the system.

5.3.1 Voice Transmission

We first examine the performance of voice transmission. The parameters chosen are \((10^3, 28, 32, p_v)\). Figure 5.1 not only shows the voice packet loss rate (in %) versus voice throughput \((\rho_v)\), it also reveals the important role played by \(p_v\). As can be observed from the figure, when \(\rho_v\) is small the loss rates for various values of \(p_v\) are no different from others. As \(\rho_v\) increases from 0.3 to 0.75, loss rate for \(p_v=0.3\) is the highest, while the loss rates for other \(p_v\)'s are still very close. However, as \(\rho_v\) is further increased from 0.75 to 0.87, \(p_v=0.04\) has the lowest discard rate, followed by \(p_v=0.1\) and \(p_v=0.01\). Such phenomenon is no surprise to us. Recall that \(c=25.0\) in 28 msec frame size; therefore, the number of calls supported in the system is large. With \(n\) becoming larger, \(p_v=0.3\) is obviously not able to resolve conflicts, resulting in many collisions which lead to higher loss rate. In the medium load (0.3 to 0.75), other \(p_v\)'s perform equally well. However, as \(\rho_v\) is in the high range (0.75 to 0.87), \(p_v=0.01\) has a higher discard rate than that of \(p_v=0.1\), which is quite contrary to one's expectation because the lower the values of \(p_v\), the better the chance of resolving conflicts. But notice that at such a high voice throughput, the voice packets are already very tight-packed, with very few idle slots left in between. If the first voice packet is transmitted with low probability, such as 0.01, it misses
Figure 5.1: Voice packet loss rate versus voice throughput for different $p_v$'s
these slots very easily. Once the chance for transmission is not available, it has to wait for an idle channel, which incurs more delay. Consequently, voice packets are discarded as delay is larger than the frame size. We should note that if 1% is the maximum allowable loss rate, the achievable voice throughput is over 80% with $p_v=0.04$.

To better evaluate the performance of the proposed scheme, we assume all $n$ voice users operate in TASI mode [23], which is an ideal scheduling, and compare the loss rate obtained from simulation with the analytic results of freeze-out fraction [26], which is given by

$$\phi = \frac{1}{Np} \sum_{k=c'+1}^{N} (k - c') \binom{N}{k} p^k(1-p)^{N-k}$$  \hspace{1cm} (5.2)

where $N$ is the number of voice paths, $p$ is the speech activity factor, and $c'$ the number of channels. Figure 5.2 depicts the comparisons, together with two more sets of parameters. Since TASI represents an ideal scheduling, the loss rate of TASI provides a lower bound. We can see from the figure that as $p_v$ is small, the simulated results are quite close to those of TASI. However, as it increases, simulated results deviate further from ideal ones. To keep voice intelligible, loss rate must be low, therefore our scheme performs quite efficiently. One point is worth mentioning. When the number of slots is small, for example $c=4$, the voice throughput is less than 60% at the 1% loss rate.

5.3.2 Integrated Voice and Data Transmission

The primary objective here is to examine the effects of data load on the performance of voice traffic by utilizing the proposed scheme. In this simulation study the number
Figure 5.2: Comparison of voice packet loss rate of TASI with that of the proposed scheme
of calls \(n\) is fixed (i.e., \(p_v\) is a constant) and the data throughput \(\rho_d\) is increased gradually. Throughout this subsection, the parameters used are \((32, 100, 2.4, 0.1)\). We used \(p_d=0.01\) to reduce the interference from data load.

Figure 5.3 shows the voice packet loss rate versus data throughput \(\rho_d\) for a fixed number of calls. Recall from Figure 5.2 that when there is no data load, the \(p_v\) is about 70% at the 1% limit. The system throughput \((p_v + \rho_d)\) in Figure 5.3 is no less than 70% at the same loss rate. Hence, the data packets, under these parameters, can make efficient use of silence intervals of voice calls by accessing the channel with 0.01-persistent CSMA.

We next investigate the mean delay and standard deviation of delay for first voice packets, subsequent voice packets, and data packets. The delay of a packet is defined as the difference between the packet arrival time and the time that the packet is successfully transmitted. To facilitate presentation of the results, the delay has been normalized to the packet transmission time. We can see from Figure 5.4 that both the first voice packet delay and its standard deviation increase almost linearly when \(\rho_d\) is small and increase rapidly as \(\rho_d\) becomes larger. The reason is that when there are more data packets in the system the first voice packets have to wait longer for an idle channel before they can be transmitted. The delay of subsequent voice packets and its standard deviation are in sharp contrast with those of the first voice packets. Figure 5.5 shows the delay increases linearly, while the standard deviation decreases at high data throughput. Also, the waiting time (i.e., the amount of time that has been shifted) is well below 1.0, which is reasonable because the maximum amount of time that can be shifted is one packet transmission time. Due to this upper limit, the standard deviation drops slightly at high channel utilization. As to
Figure 5.3: Voice packet loss rate versus data throughput for different $\rho_v$'s
(a) Mean first voice packet delay

(b) Standard deviation of first voice packet delay

Figure 5.4: Mean and standard deviation of first voice packet delay
Figure 5.5: Mean and standard deviation of subsequent voice packet delay
data packets, both delay and standard deviation increase exponentially, which can be observed from Figure 5.6.

5.4 Comparisons with Reservation Aloha

The scheme, as proposed in this chapter, requires carrier sensing. Under bursty data traffic, it is a well-known fact that CSMA can attain a higher throughput than the slotted Aloha one. Nevertheless, it has been shown [27] that a variant of slotted Aloha, called reservation Aloha (r-aloha) scheme, is able to achieve a throughput beyond the $1/e$ limit if there is more than one packet in a message. Recall that voice packets arrive periodically while in talkspurts. In general there are many packets in a talkspurt. One may, therefore, surmise that the r-aloha scheme may be suitable for packet voice transmission or even combined voice and data transmission. A comparison of the performance of this alternative for integrated services with that of the proposed scheme is the prime objective of this section.

The following describes the protocols for voice and data users. In r-aloha, the channel time is assumed slotted and slots are organized into frames with $c$ slots per frame. If voice user $i$ has a successful transmission in slot $k$ (acquiring the channel with slotted Aloha mode), this slot $k$ in the next frame is reserved for user $i$ as long as it still has packets to transmit. If it transmits its last packet, the slot $k$ in the next frame is available to other users. If the first attempt of user $i$ fails, the packet is retransmitted in the next unreserved slot with probability $p_{rt}$. Data users transmit their packets with slotted Aloha mode and data message is assumed to be composed of one packet. Retransmission algorithm of data packets is the same as
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Figure 5.6: Mean and standard deviation of data packet delay
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that used in the proposed scheme.

Following the previous notations, parameters chosen for the simulation are: $tr=720$ Kbps, $vf=16$ msec, $cr=32$ Kbps, and $p_r=p_{rt}=0.1$. Assuming there are 64 header bits, the overflow bits required for the proposed scheme is 27. Hence the number of slots available per frame is 19 and 20 for the proposed and the r-aloha schemes, respectively. In the simulation, the normalized propagation delay is 0.01 and speech activity factor of 42% is used. Also, voice packets are discarded if delay exceeds 32 msec.

When both schemes are applied to voice transmission, the voice packet loss rate versus the number of calls is shown in Figure 5.7, together with the freeze-out fraction of TASI (equation 5.2). Figure 5.7 indicates the voice packets loss rates for both schemes are very similar. In comparing the results, one should observe the number of slots available per frame for each scheme. Due to the overflow bits, the proposed scheme has 19 slots per frame, which is one less than that in the r-aloha.

Although voice packet loss rates for both schemes are almost the same, the way that voice packets are discarded is quite different. From the description of the r-aloha scheme, one notices that voice packets are lost only at the beginning of talkspurts. However, due to the probabilistic transmission of first voice packet in the proposed scheme, the first voice packet may collide with either the first voice packets or the subsequent voice packets (both from other users). Therefore, loss of voice packets may occur at the beginning of talkspurts as well as other parts of talkspurts. To acquire a deeper understanding of how the voice packets are lost in the proposed scheme, voice packets are split into two components: first voice packet
Figure 5.7: Voice packet loss rate versus number of calls without data load
and subsequent voice packet. Their loss rates are plotted in Figure 5.8. As can be observed, loss of subsequent voice packets is insignificant compared to that of first voice packets. In other words, the chance of collision between these two types is quite rare and most of the loss occurs at the front parts of talkspurts.

If a large number of voice packets are discarded at the front parts of talkspurts, speech may be severely clipped and voice quality degraded. It is interesting to note that packet voice transmission using the r-aloha has been examined by simulation recently [15]. They performed a listening test and concluded that at the 1% loss rate voice quality is still good and intelligible. Hence, it is very useful to compare the patterns of lost voice packets in both schemes. Figures 5.9 to 5.12 show the frequency (that is the number of times it occurs) versus the number of consecutive packets that are discarded. The frequency values are represented by thick black vertical lines. The symbol circle in these figures represents the number as located occurs only once. (Other thinner vertical lines are for the purpose of marking the x-axis only.) Also depicted in these figures are the number of calls (n), packets that are lost or successfully transmitted, and lost rate in a simulation run. Comparison of the lost patterns in both schemes can be done by fixing the value n. We can observe from these figures that lost patterns are very similar. When n = 30, the loss of consecutive packets is small. As n increases to 37 (the loss rate is about 1%), the shapes in Figures 5.11 and 5.12 are quite spread out, and consecutive lost packets can go beyond 30, however, this frequency is small. Since a listening test has been carried out in [15] and the lost patterns in both schemes are similar, it is reasonable to say that the criterion of 1% loss rate in the proposed scheme is acceptable.
Figure 5.8: Components of voice packet loss rate versus number of calls
proposed scheme: \( n = 30 \)

loss of voice packets = 537

successful voice packets = 47726

loss rate = 0.112 %

Figure 5.9: Frequency versus loss of consecutive packets in the proposed scheme, \( n = 30 \)
r-aloha scheme: \( n = 30 \)

- loss of voice packets = 927
- successful voice packets = 141791
- loss rate = 0.065 %

Figure 5.10: Frequency versus loss of consecutive packets in the r-aloha scheme, \( n = 30 \)
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proposed scheme: $n = 37$

- loss of voice packets = 5717
- successful voice packets = 575056
- loss rate = 0.984%

Figure 5.11: Frequency versus loss of consecutive packets in the proposed scheme, $n = 37$
r-aloha scheme: $n = 37$

loss of voice packets = 14821
successful voice packets = 1725678
loss rate = 0.852%

Figure 5.12: Frequency versus loss of consecutive packets in the r-aloha scheme, $n = 37$
The performances of both schemes vary significantly when data load is mixed with voice traffic. The normalized delay of voice and data packets versus data throughput ($\rho_d$) are shown in Figures 5.13 and 5.14. Note that “voice packets” implies both first voice and subsequent voice packets and that number of calls, as shown, are fixed at 10 and 20 calls. Observations from both figures show the mean voice packet delay of the proposed scheme is maintained within two packets transmission time. On the other hand, delay of voice packet in the r-aloha increases rapidly. The data packet delay of the proposed scheme is also well below that of the r-aloha one. The dramatic increase in delay in the r-aloha scheme is mainly due to the influence from data load and also the protocols adopted by both voice and data users. Recall that data users simply use the slotted Aloha mode and, with data traffic only, the upper bound of $\rho_d$ is $1/e(\approx 0.36)$. Since each voice user reserves a slot after a successful transmission, these reserved slots are off limits to data users who have packets to send. As soon as an unreserved slot is available, data packets which have waited for transmission and those new voice and data packets would all transmit into this slot. Consequently, collisions occur which result in a long delay for both types of packets. Another disadvantage of r-aloha scheme is the small value of $\rho_d$. If there are even 10 calls in the system, data packets cannot take advantage of the remaining 10 slots available in every frame. In the proposed scheme, however, $\rho_d$ can go as high as 0.5 without saturating the system. Although the proposed scheme can achieve a higher system throughput, it does have disadvantages. The simulation model assumes each terminal is in line of sight with others; that is, no hidden terminals problem exits in the system. The performance of the proposed scheme will be certainly deteriorated if this assumption is relaxed.
Figure 5.13: Normalized voice packet delay versus data throughput ($\rho_d$)
Figure 5.14: Normalized data packet delay versus data throughput ($\rho_d$)
5.5 Summary

A multiaccess scheme for packetized voice and data transmission in a packet radio network is proposed. The performance of the proposed scheme is evaluated by simulation. In this simulation study, three different sets of parameters are selected for voice transmission, whereas a particular set of parameters is used for integrated voice and data transmission. At the 1% voice packet loss rate, results reveal that for voice transmission, under some system parameters, the channel throughput ($\rho_v$) is above 80% and that for combined voice and data transmission, under the chosen system parameters, the channel throughput ($\rho_v + \rho_d$) is higher than the case of voice only transmission. Another option for integration of voice and data traffic is also considered, namely, the r-aloha. Owing to the limited throughput of the slotted Aloha, the system throughput of the r-aloha is small and delays for both voice and data packets are large, compared to those of the proposed scheme.
Chapter 6

Conclusion

The research results of the thesis are composed of two areas. The first part is concerned with the throughput and delay characteristics of some combinations of CSMA protocols, namely non-persistent with 1-persistent and $p$-persistent with various probabilities of transmission. Protocols suitable for integrated voice and data transmission in local area network and packet radio network are proposed in the second part.

In chapter 2, the delay-throughput performance of a system using 1-persistent and non-persistent CSMA protocols is examined. We derive the throughput expressions of each protocol by using a simple renewal theory argument. Observing the simulated delay characteristics, we find the resultant throughput cannot be added arbitrarily. However, due to the inherent properties of these two protocols, it is noticed from the results that a non-preemptive priority scheme can be implemented by assigning 1-persistent CSMA to the higher priority and non-persistent CSMA to the lower priority.
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A more general case of CSMA scheme, p-persistent, is considered in chapter 3. The protocol of p-persistent CSMA/CD with various transmission probabilities used as a priority resolution scheme is proposed in [17]. Since no delay-throughput performance is given in [17], we extend the analysis of chapter 2 to this protocol, and derive the throughput expressions, and simulate the delay characteristics. To limit mathematical complexity, only two transmission probabilities ($p_1$ and $p_2$) are considered in the throughput analysis but the approach can be extended to the case of various transmission probabilities. The analysis includes both cases of infinite and finite users. The resultant throughput is found to be bounded by the throughput of $p_1$-persistent and that of $p_2$-persistent CSMA protocols. It is also observed from the delay characteristics that a good priority scheme is dependent on a judicious choice of $p_1$ and $p_2$.

Intrigued by the flexibility and simplicity of the protocol examined in chapter 3, we apply this p-persistent CSMA/CD protocols with two persistency factors to the integrated voice and data transmission in a local area network (LAN). In the protocol, we stipulate the transmission probability of voice packets be greater than that used by data packets and voice packets are dropped if delay exceeds the voice frame size. The performance measures are the voice packet loss rate, mean, and variance of data and voice packets, as well as the system throughput. The simulated results suggest that this protocol is a viable scheme for integrated services applications because both the traffic requirements of voice and data are satisfied. In addition, compared to other proposed protocols published in the literature, it is less structured in the sense that users do not need to follow a complex algorithm and thus the scheme can be implemented easily.
Although there are plenty of protocols proposed for integrated services in LAN, there are very few suggested for the packet radio network. In view of this, we propose a protocol which can be applied to a packet radio network. The proposed protocol actually resulted from the one considered in chapter 4 and other published works. However, it turns out to perform well. Comparison of the voice packet loss rate of the proposed scheme with that of the ideal case suggests this observation. Besides, it also possesses two interesting properties. Small mean and variance of voice packet delay is one of these; the other is that no collision detection is required.
Appendix A

This appendix outlines the derivations of the throughput expressions of p-persistent CSMA (without collision detection) with two persistent factors, assuming finite number of users. The number of users are assumed to be divided into two classes, class 1 and class 2. Also, the number of users in class 1 and class 2 are $M_1$ and $M_2$, respectively.

The throughput analysis of p-persistent CSMA with one persistent factor is available in [18]. The approaches shown in [18] can be easily extended to the case of two persistent factors, mainly because of the independent arrival processes assumption. In fact, many of the intermediate results, during the derivations, can be obtained by simply adding similar terms, and distinguishing them with subscripts 1 and 2. Hence, some of the intermediate steps will not be shown here. Because many notations used here have the same definitions as those in the case of infinite users, only those that are new will be defined.

The arrival process, in this case, is geometrically distributed. Packets in class 1 and 2 arrive in each slot with probability $g_1$ and $g_2$, respectively. As in the case of infinite users, channel state is divided into busy period ($B$) and idle period ($I$).
However, each busy period, in this case, is further divided into \( J \) sub-busy periods and each sub-busy period consists of a transmission delay \( R^{(j)} \) and a transmission time \( T^{(j)} \) (also see Figure A.1). The following describes how \( S_1 \) is obtained.

From Figure A.1, one can conclude that

\[
B^{(j)} = R^{(j)} + 1 + a \quad j = 1, 2, \cdots
\]  

(A.1)

and,

\[
B = \sum_{j=1}^{J} B^{(j)}
\]

(A.2)

If \( U_1^{(j)} \) denotes the successful transmission time of class 1 packets in the \( j^{th} \) sub-busy period, then

\[
U_1 = \sum_{j=1}^{J} U_1^{(j)}
\]

(A.3)

Note that \( J \) is geometrically distributed with parameter

\[
(1 - g_1)^{(1+(1/a))M_1} (1 - g_2)^{(1+(1/a))M_2}
\]

(A.4)

and that \( B^{(j)} \) and \( U_1^{(j)} \), \( j = 2, 3, \cdots, J \) are identically and independently distributed. For \( j = 1 \), they are also independent from others but have different distributions. Since \( J \) is independent of \( B^{(j)} \) and \( U_1^{(j)} \), then

\[
\bar{B} = E[B^{(1)}] + (J - 1)E[B^{(2)}]
\]

(A.5)

and,

\[
\bar{U}_1 = E[U_1^{(1)}] + (J - 1)E[U_1^{(2)}]
\]

(A.6)
\( a \): normalized end-to-end propagation delay

\( R^{(j)} \): retransmission delay of the \( j^{th} \) sub-busy period

\( T^{(j)} \): transmission period of the \( j^{th} \) sub-busy period

\( B^{(j)} \): the \( j^{th} \) sub-busy period

Figure A.1: Components in the \( j^{th} \) sub-busy period
where,

\[ J = \frac{1}{(1-g_1)(1+(1/a))^{M_1} (1-g_2)(1+(1/a))^{M_2}} \]  \hspace{1cm} \text{(A.7)}

As in the infinite users case, \( I \) is geometrically distributed with mean

\[ I = \frac{a}{1 - (1 - g_1)^{M_1} (1 - g_2)^{M_2}} \]  \hspace{1cm} \text{(A.8)}

Since \( S_1 \) is given by \( \overline{U_1}/(\overline{B} + I) \), the task now is to find \( \overline{U_1} \) and \( \overline{B} \). If \( \pi_{n_1,n_2}(x) \) denotes the probability that there are \( n_1 \) and \( n_2 \) arrivals out of \( M_1 \) and \( M_2 \) users in \( x \) slots, then

\[
\pi_{n_1,n_2}(x) = \frac{1}{1 - (1 - g_1)^x(1 - g_2)^x} \cdot \binom{M_1}{n_1} [1 - (1 - g_1)^x]^{n_1} \cdot (1 - g_1)^{x(M_1-n_1)} \cdot \binom{M_2}{n_2} [1 - (1 - g_2)^x]^{n_2} \cdot (1 - g_2)^{x(M_2-n_2)}
\]

\[ n_1 = 0, 1, \ldots, M_1; \quad n_2 = 0, 1, \ldots, M_2; \quad n_1 + n_2 > 0 \]  \hspace{1cm} \text{(A.9)}

Now, let \( N_0^{(j)} \) be the number of class 1 and class 2 arrivals at the starting time of \( B^{(j)} \). Therefore,

\[
\Pr[N_0^{(j)} = (n_1, n_2)] = \begin{cases} 
\pi_{n_1,n_2}(1) & j = 1 \\
\pi_{n_1,n_2}(1 + (1/a)) & j = 2, 3, \ldots
\end{cases}
\]  \hspace{1cm} \text{(A.10)}
Following the approach in [3], and after some algebraic manipulations, one can write the following.

\[
E \left[ R^{(j)} \mid N_0^{(j)} = (n_1, n_2) \right] = a \sum_{k=1}^{\infty} (1 - p_1)^{kn_1} \left[ \frac{p_1(1 - g_1)^k - g_1(1 - p_1)^k}{p_1 - g_1} \right]^{M_1-n_1} \\
\cdot (1 - p_2)^{kn_2} \left[ \frac{p_2(1 - g_2)^k - g_2(1 - p_2)^k}{p_2 - g_2} \right]^{M_2-n_2} \tag{A.11}
\]

\[
E \left[ U_1^{(j)} \mid N_0^{(j)} = (n_1, n_2) \right] = n_1 p_1 \sum_{k=0}^{\infty} (1 - p_1)^{(k+1)n_1} (1 - p_2)^{(k+1)n_2} \\
\cdot \left[ \frac{p_1(1 - g_1)^{k+1} - g_1(1 - p_1)^{k+1}}{p_1 - g_1} \right]^{M_1-n_1} \\
\cdot \left[ \frac{p_2(1 - g_2)^{k+1} - g_2(1 - p_2)^{k+1}}{p_2 - g_2} \right]^{M_2-n_2} \\
+ (M_1 - n_1) g_1 p_1 \sum_{k=1}^{\infty} (1 - p_1)^{(k+1)n_1} (1 - p_2)^{(k+1)n_2} \\
\cdot \left[ \frac{1 - g_1^k - (1 - p_1)^k}{p_1 - g_1} \right]^{M_1-n_1-1} \\
\cdot \left[ \frac{p_1(1 - g_1)^{k+1} - g_1(1 - p_1)^{k+1}}{p_1 - g_1} \right]^{M_1-n_1-1} \\
\cdot \left[ \frac{p_2(1 - g_2)^{k+1} - g_2(1 - p_2)^{k+1}}{p_2 - g_2} \right]^{M_2-n_2} \tag{A.12}
\]

Removing the conditions on \(n_1\) and \(n_2\), equation (A.11) then becomes

\[
E \left[ R^{(j)} \right] = \begin{cases} 
    r(1) & j = 1 \\
    r(1 + 1/\alpha) & j = 2, 3, \ldots 
\end{cases} \tag{A.13}
\]
APPENDIX A.

where,

\[ r(x) = \frac{a}{1 - (1 - g_1)^{M_1}(1 - g_2)^{M_2}} \]
\[ \cdot \sum_{k=1}^{\infty} \left\{ \left( (1 - p_1)^k - (1 - g_1)^z \right) p_1 \left[ \frac{(1 - p_1)^k - (1 - g_1)^k}{p_1 - g_1} \right]^{M_1} \right. \]
\[ \cdot \left( (1 - p_2)^k - (1 - g_2)^z \right) p_2 \left[ \frac{(1 - p_2)^k - (1 - g_2)^k}{p_2 - g_2} \right]^{M_2} \]
\[ - (1 - g_1)^{zM_1} \left[ \frac{p_1(1 - g_1)^k - g_1(1 - p_1)^k}{p_1 - g_1} \right]^{M_1} \]
\[ \cdot \left( 1 - g_2)^{zM_2} \left[ \frac{p_2(1 - g_2)^k - g_2(1 - p_2)^k}{p_2 - g_2} \right]^{M_2} \} \] \hspace{1cm} (A.14)

and equations (A.12) can be written as

\[ E[U_1^{(j)}] = \begin{cases} \ u_1(1) & j = 1 \\ \ u_1(1 + 1(1/a)) & j = 2,3, \ldots \end{cases} \] \hspace{1cm} (A.15)

where,

\[ u(x) = \frac{p_1M_1}{1 - (1 - g_1)^{zM_1}(1 - g_2)^{zM_2}} \]
\[ \cdot \sum_{k=0}^{\infty} \left[ (1 - p_1)^k - (1 - g_1)^z \cdot \frac{p_1(1 - p_1)^k - g_1(1 - g_1)^k}{p_1 - g_1} \right]^{M_1-1} \]
\[ \cdot \left[ (1 - p_1)^{k+1} - (1 - g_1)^z \cdot p_1 \cdot \frac{(1 - p_1)^{k+1} - (1 - g_1)^{k+1}}{p_1 - g_1} \right]^{M_1-1} \]
\[ \cdot \left[ (1 - p_2)^{k+1} - (1 - g_2)^z \cdot p_2 \cdot \frac{(1 - p_2)^{k+1} - (1 - g_2)^{k+1}}{p_2 - g_2} \right]^{M_2} \]
\[ - \frac{g_1p_1M_1(1 - g_1)^{zM_1}(1 - g_2)^{zM_2}}{1 - (1 - g_1)^{zM_1}(1 - g_2)^{zM_2}} \]
\[ \cdot \sum_{k=1}^{\infty} \left[ \frac{(1 - g_1)^k - (1 - p_1)^k}{p_1 - g_1} \right] \]
\[ \cdot \left[ \frac{p_1(1 - g_1)^{k+1} - g_1(1 - p_1)^{k+1}}{p_1 - g_1} \right]^{M_1-1} \]
\[ \cdot \left[ \frac{p_2(1 - g_2)^{k+1} - g_2(1 - p_2)^{k+1}}{p_2 - g_2} \right]^{M_2} \] \hspace{1cm} (A.16)
Using equations (A.1), (A.5), (A.7), (A.8), (A.13) and (A.14), one then get

\[
\bar{B} + \bar{I} = \frac{a}{(1 - g_1)^{(1+1/a)M_1}(1 - g_2)^{(1+1/a)M_2}} \\
\cdot \sum_{k=1}^{\infty} \left\{ \left( (1 - p_1)^k - (1 - g_1)^{(1+1/a)} \right) p_1 \left[ \frac{(1 - p_1)^k - (1 - g_1)^k}{p_1 - g_1} \right]^{M_1} \right\} \\
\cdot \left( (1 - p_2)^k - (1 - g_2)^{(1+1/a)} \right) p_2 \left[ \frac{(1 - p_2)^k - (1 - g_2)^k}{p_2 - g_2} \right]^{M_2} \\
+ \frac{1 + a}{(1 - g_1)^{(1+1/a)M_1}(1 - g_2)^{(1+1/a)M_2}}
\]  

(A.17)

Substituting equations (A.7) and (A.15) into (A.6), the mean successful transmission time of class 1 is given by

\[
\bar{U}_1 = \frac{p_1 M_1}{1 - (1 - g_1)^{(1+1/a)M_1}(1 - g_2)^{(1+1/a)M_2}} \\
\cdot \sum_{k=0}^{\infty} \left[ (1 - p_1)^k - (1 - g_1)^{(1+1/a)} \right] \cdot p_1 \left[ \frac{(1 - p_1)^k - (1 - g_1)^k}{p_1 - g_1} \right]^{M_1-1} \\
\cdot \left[ (1 - p_1)^{k+1} - (1 - g_1)^{(1+1/a)} \right] \cdot p_1 \left[ \frac{(1 - p_1)^{k+1} - (1 - g_1)^{k+1}}{p_1 - g_1} \right]^{M_1-1} \\
\cdot \left[ (1 - p_2)^{k+1} - (1 - g_2)^{(1+1/a)} \right] \cdot p_2 \left[ \frac{(1 - p_2)^{k+1} - (1 - g_2)^{k+1}}{p_2 - g_2} \right]^{M_2} 
\]  

(A.18)

The expressions of \( S_1 \) is obtained by putting equations (A.17) and (A.18) into \( \bar{U}_1/(\bar{B} + \bar{I}) \), which is shown as follows.

\[
S_1 = \frac{p_1 M_1 \sum_{k=0}^{\infty} A \cdot B^{M_1-1} \cdot C^{M_2}}{1 + a + a \sum_{k=1}^{\infty} D^{M_1} \cdot E^{M_2}}
\]  

(A.19)
where,

\[ A = (1 - p_1)^k - (1 - g_1)^{1+1/a} \left[ \frac{p_1(1 - p_1)^k - g_1(1 - g_1)^k}{p_1 - g_1} \right] \]  \hspace{1cm} (A.20)

\[ B = (1 - p_1)^{k+1} - (1 - g_1)^{1+1/a} p_1 \left[ \frac{(1 - p_1)^{k+1} - (1 - g_1)^{k+1}}{p_1 - g_1} \right] \]  \hspace{1cm} (A.21)

\[ C = (1 - p_2)^{k+1} - (1 - g_2)^{1+1/a} p_2 \left[ \frac{(1 - p_2)^{k+1} - (1 - g_2)^{k+1}}{p_2 - g_2} \right] \]  \hspace{1cm} (A.22)

\[ D = (1 - p_1)^k - (1 - g_1)^{1+1/a} p_1 \left[ \frac{(1 - p_1)^k - (1 - g_1)^k}{p_1 - g_1} \right] \]  \hspace{1cm} (A.23)

\[ E = (1 - p_2)^k - (1 - g_2)^{1+1/a} p_2 \left[ \frac{(1 - p_2)^k - (1 - g_2)^k}{p_2 - g_2} \right] \]  \hspace{1cm} (A.24)

The throughput of class 2 can be similarly derived. Its expression is the same as that of $S_1$, except that all subscripts 1 are replaced by 2.
Bibliography


