# A SIMULATION STUDY OF A LIMITED SENSING RANDOM ACCESS ALGORITHM FOR A LOCAL AREA NETWORK WITH VOICE USERS

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Abstract

The purpose of this paper is to evaluate the performance of a limited sensing random access algorithm in a local area network with voice users. Random access algorithms have proven to be very efficient in local area network environments with data users. However, in contrast to data packets, voice packets cannot experience long delays, because of the requirement that a voice "data stream" must be played out at the receiver. If a voice packet exceeds its established maximum delay, it is discarded. The simulation study finds the number of voice users that the network can support, provided that the packet loss rate, which can be tolerated by a customer does not exceed a certain threshold. Finally, a comparison is made with the simulation results of this algorithm and other commonly used protocols.

## 1. Introduction.

Local area networks (LANs) have been used extensively in the past few years for data communications ([1], [2],[3]). Musser et al demonstrated in [4] the technical feasibility of utilizing a LAN as a multidrop subscriber loop for a PABX (private automatic branch exchange). Their objective was to replace the multiple twisted pairs being pulled from the PABX with a single coaxial cable. Subscriber terminals (i.e. voice users) can simply tap into the cable. The technical feasibility of the "LAN arrangement" has been demonstrated in [4], by showing the ability of the Ethernet ([6]) and the GBRAM ([7]) protocols to support a population of voice users in the above described local area network setting.

It is worth noting that the "LAN arrangement" proposed by Musser, in addition to its economical advantages, (see [4] for more details), can support a variety of users (i.e. data voice or other). Since future communication networks are expected to handle a variety of traffic types, and an enormous effort is currently undertaken to incorporate voice and data on the existing telephone network, Musser's "LAN arrangement" is a step towards the right direction.

This paper examines the performance of a limited sensing random access algorithm for the "LAN arrangement" proposed by Musser in [4]. This random access algorithm (RAA) was introduced by Merakos et al in [3], and it was proved to be very efficient in a LAN environment with data users for both the slotted ([3]) and the uslotted channels ([5]). The RAA in [3] and [5] has a number of advantages. First, it is a limited sensing RAA, which implies that a voice user does not have to sense the channel unless it has a packet to transmit. Secondly, it has been proven to be very effective in a LAN environment with data users for both slotted and unslotted channels (see [3] and [5]). Thirdly, it is a stable algorithm for the infinite population user model (see [3] and [5]). Fourthly, it has last-come-first-serve characteristics, which is desirable in LANs with voice users, because voice packets cannot experience long delays. Finally, as we will see in section 5 it compares favorably against the Ethernet and the GBRAM protocols.

## 2. The Model.

The model assumes that the two ends of a voice circuit generate R bits/s of traffic into the system. Voice packets of constant size L bits are assembled at regular intervals and sent to the voice user buffer, which has a capacity of one packet. A packet from an active voice user will be generated at every F=L/R seconds (we do not distinguish between talkspurts and silence periods). A packet with transmission delay longer than F seconds results in a packet loss.

The model assumes that the capacity of the cable is C bits/s. Hence, a packet will require a slot



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length of $Q=L/C$ seconds for its transmission. The length of the cable is equal to d Km. The end to end propagation delay (i.e. the time that it takes for a packet to traverse the cable from one end of the cable to the other) is denoted by $\alpha$ and is equal to $d/v$, where v is the speed of light.

To facilitate our simulation we take $\alpha$ to be our unit of time (i.e. $\alpha=1$). The maximum packet lifetime F is equal to T units of time ($F=T\cdot\alpha$), and the packet transmission time is equal to P units of time ($Q=P\cdot\alpha$). Without loss of generality, P and T are assumed to be integers.

During the simulation, the system generates N packets every T units of time (i.e. we assume that N voice users are continuously active), and these packets are uniformly distributed over the interval of length T. The same packet generation model was also adopted in [4].

To simplify the simulation we make the following assumption.

A.1) The channel is divided into slots; the length of a slot is equal to the end-to-end propagation delay $\alpha$ (note that $\alpha=1$). Voice users are allowed to initiate their packet transmissions only at the beginnings of slots.

The model considers limited channel sensing and ternary feedback. That is it assumes that the voice users sense the channel only while they have a packet to transmit, and they can determine which one of the following events occurs.

a) no packet transmission (idle)

b) single packet transmission (success).

c) two or more packet transmissions (collision)

In the case of a collision, let $\beta$ denote the fraction of each packet (in units of time) that gets transmitted during the collision before the transmitting users abort their transmissions by detecting the interference. For local area networks using a cable, it is commonly assumed that $1\leqslant\beta\leqslant P$. This model assumes that $\beta=1$.

## 3. The algorithm.

The execution of the algorithm governing the accessing of the channel is divided into a series of steps. Algorithm steps start at the beginning of slots. Let $t_i$ ($i=0,1,2..$) denote the instants at which cosecutive algorithm steps begin. An algorithm step can be idle, success, or collision if no packet transmission, a single packet transmission or two or more packet transmissions occur during the algorithm step respectively. The duration of an idle, success or collision algorithm step is equal to 1, P+1, 2 respectively.

Let us now define.

Definition: A packet in the system is called "legitimate" if its delay is smaller than the maximum packet lifetime T.

Each "legitimate" packet has a counter, which assumes nonnegative values. Let $r_i$ denote the counter value of an arbitrary "legitimate" packet at algorithm instant $t_i$. The following operational rules are defined:

I) At instant $t_i$, all "legitimate" packets with $r_i=0$ are transmitted.

II) All users with "legitimate" packets, sense the channel and act as follows:

1) If a successful transmission occured at step i, then the "legitimate" packet with $r_i=0$ leaves the system. All "legitimate" packets with $r_i=r$ ;$r\geqslant1$ increment their counters by m-1 at instant $t_{i+1}$, and set $r_{i+1}=r_i+m-1$,where $m\geqslant1$ is an integer parameter.

2) If a collision occured at step i, then every "legitimate" packet with $r_i=0$, independetly of the others, sets its counter value to m-1+J, where J is an integer random variable uniformly distributed on {1,2,...n}, and n is an integer parameter such that $n\geqslant2$. Each of the "legitimate" packets with $r_i=r$ ; $r\geqslant1$ increments its counter by m+n-1. Thus, $r_{i+1}=r_i+m+n-1$.

3) If algorithm step i is idle, then all "legitimate" packets with counter values $r_i\geqslant1$ decrement their counter values by one (i.e. $r_{i+1}=r_i-1$).

III) When a new "legitimate" packet arrives during a slot at a user site, the user senses the channel at the beginning of the next slot. If the channel is idle, the packet sets its counter value to 0, and therefore, attempts transmission at the same instant. If the channel is sensed busy, the user waits until the channel is sensed idle for the first time (at the beginning of some slot), and only then the user sets the counter value of its packet to M, where M is an integer random variable uniformly distributed on {0,1,..m-1}.

The integers m,n are design parameters to be optimized. The general operation of the algorithm is perhaps better illustrated by introducing the concept of the stack ([8]), which is an abstract storage device consisting of an infinite number of cells, labelled 0,1,2,... At each algorithm instant, $t_i$, the kth cell of the stack contains the packets with $r_i=k$ ; $k \geq 0$. In figure 1, by using the concept of the stack, an idle, a successful, and a collision step are shown.

We name the above described algorithm LSAVU (Limited Sensing Algorithm for Voice Users).

## 4. Performance Measures- Simulation Results.

The most important performance measure of the effectiveness of LSAVU is the packet loss rate (averaged over all active voice circuits) versus the number of active voice circuits. The packet loss rate is defined to be the percentage of voice packets discarded by LSAVU.

For the simulation results the model of chapter 2 is adopted. The values of $\alpha$, $\beta$, P, T are needed to perform the simulations. In the model of chapter 2 we defined $\alpha$ to be our unit of time (i.e. $\alpha=1$) and we took $\beta=\alpha=1$. For the values of P and T the following cases are considered.

Case 1: P=231, T=3600
Case 2: P=116, T=1800
Case 3: P=58, T=900
Case 4: P=29, T=450
Case 5: P=24, T=3600
Case 6: P=12, T=1800
Case 7: P=6, T=900
Case 8: P=3, T=450

In cases 1 to 8, the length, d, of the cable is taken to be equal to 1.0 Km. Furthermore, in cases 1 to 8 each voice circuit generates R=64 Kbits/s of traffic into the system. In cases 1 to 4, the capacity of the cable is C=1.0 Mbps; in cases 5 to 8, C=10.0 Mbps. In cases 1 and 5 the packet length is L=768 bits (96 bytes); in cases 2 and 6, L=384 bits (48 bytes); in cases 3 and 7, L=192 bits (24 bytes); and in cases 4 and 8 L=96 bits (12 bytes). The same d,R,C and L values were also adopted in [4].

It was determined from experimentation that 10,000 voice packets were sufficient to produce reliable simulation results. Different values of m and n were also checked and it was found that for all cases (1-8) the optimum values were

$$m^*=1$$
$$n^*=3$$

The optimum values m* and n* were the ones, which produced the smaller packet loss rate for LSAVU. The LSAVU algorithm with m=m* and n=n* is denoted by LSAVUopt.

In figure 2 the packet loss rate versus the number of active voice circuits curves, corresponding to LSAVUopt algorithm and a 1.0 Mbps cable, are drawn (cases 1-4). Similar curves are drawn in figure 3 for the 10.0 Mbps cable (i.e. cases 5-8). Table 1 shows the number of voice circuits supported by LSAVUopt at a packet loss rate of 2% for a 1.0 Mbps and a 10.0 Mbps cable. It is worth noting that at a packet loss rate of 2%, LSAVUopt can support 15 circuits on a 1.0 Mbps cable, and 134 circuits on a 10.0Mbps cable, when the packet lenght is equal to 768 bits.

## 5. Comments and conclusions.

The simulation results showed that values of m and n near the optimum values m*=1 and n*=3 did not affect the performance of LSAVU. The simulation results also showed that for a cable of constant capacity, LSAVUopt performed better for the largest packet size. This a common characteristic of random access contention schemes in a LAN. Finally, the simulation results show that LSAVUopt operates near 0% packet loss rate up to a point and then there is a sharp increase in the packet loss rate. Therefore, the cutoff for the number of voice circuits supported is very abrupt. It is worth noting that the number of circuits that LSAVUopt supports, such that the maximum individual (per voice circuit) packet loss rate is smaller than 2%, is almost identical to the number of voice circuits that LSAVU supports, such that the packet loss rate (averaged over all voice circuits) is smaller than 2% . In only one case did the maximum individual packet loss rate lower the number of voice circuits that LSAVUopt supports; case 4 where the number of voice circuits that LSAVUopt supports was reduced from 14 to 13.

To make fair comparisons among LSAVUopt, Ethernet and GBRAM protocols we included in our model (as in [4]) a jam time of 4.8$\mu$s, a 9.6$\mu$s transmit/receive turnaround time and 48 bits (6 bytes) of control overhead in each packet. In figure 4, the packet loss rate versus the number of active voice circuits curves, corresponding to Ethernet, GBRAM and LSAVUopt (under the above additional modelling assumptions) for the 10.0Mbps cable and the L=812 bits packet length, are shown. Table 2 shows the number of voice circuits supported by Ethernet, GBRAM and LSAVUopt at a 2% packet loss rate (C=10.0 Mbps and L=812 bits). The comparison between LSAVUopt and

Ethernet favors LSAVUopt, since LSAVUopt supports more voice circuits. At a packet loss rate of 2% with a 10.0 Mbps cable and a packet length of 812 bits LSAVUopt supports 23% more circuits than Ethernet. Comparisons between LSAVUopt and GBRAM are more difficult, because they belong to two different classes of protocols. Although GBRAM supports 8% more voice circuits than LSAVUopt at 2% packet loss rate when C=10.0 Mbps and L=812 bits, one has to be careful when GBRAM is used to integrate voice and data on the cable. Ethernet and LSAVUopt are efficient for large number of bursty data users, while the performance of GBRAM deteriorates as the number of users increases. Even with a population of voice users GBRAM has to deal with the complicated procedure of signing on previously inactive users, which become active, and signing-off previously active users, which become inactive (see section 7 in [7]).

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Table 1

Number of circuits supported by LSAVUopt at 2% packet loss rate

| | | | | |
|---|---|---|---|---|
| Packet Length (in bits) | 768 | 384 | 192 | 96 |
| Circuits (1 Mbps cable) | 15 | 15 | 14 | 14 |
| Circuits (10 Mbps cable) | 134 | 119 | 99 | 72 |

Table 2

Number of Circuits supported by Ethernet, GBRAM and LSAVUopt at 2% packet loss rate, with cable capacity of 10 Mbps and packet length of 812 bits(102 bytes)

| | Ethernet | GBRAM | LSAVUopt |
|---|---|---|---|
| Circuits | 94 | 125 | 116 |

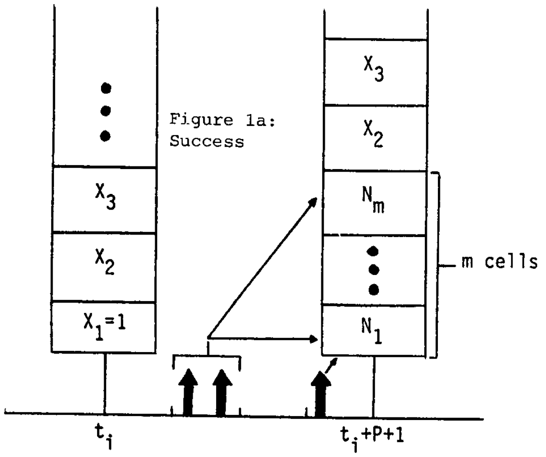

Figure 1a: Success

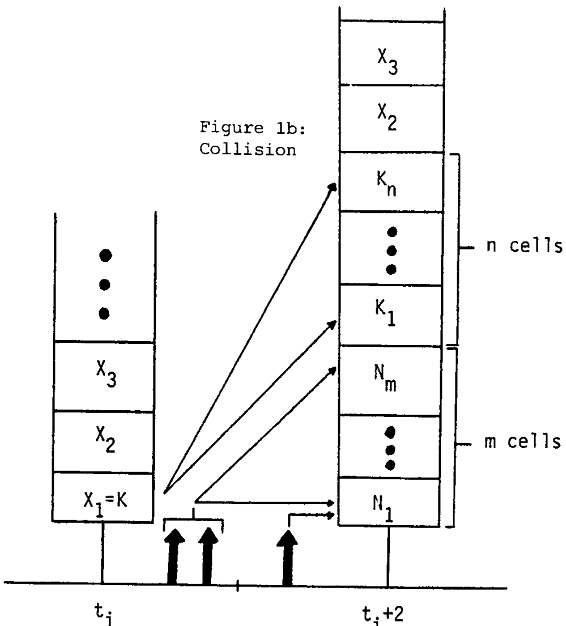

Figure 1b: Collision

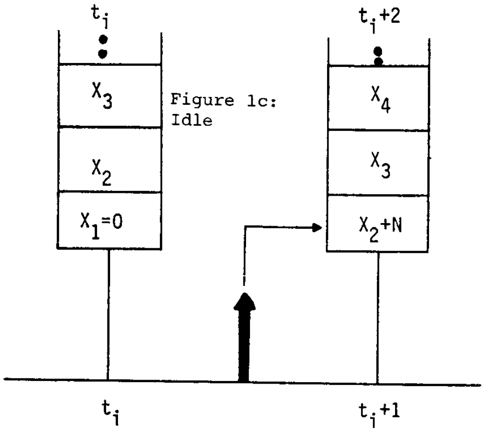

Figure 1c: Idle

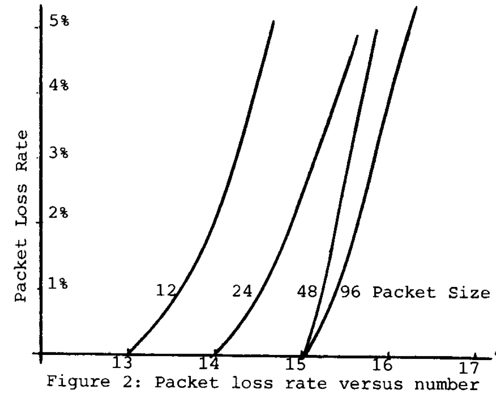

Figure 2: Packet loss rate versus number of Circuits parametrized on the packet length for 1 Mbps cable

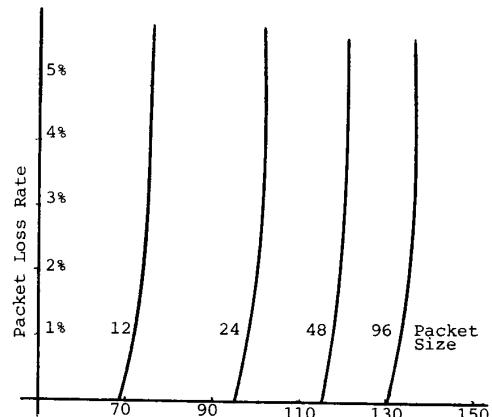

Figure 3: Packet loss rate versus number of circuits parametrized on the packet length for a 10 Mbps cable

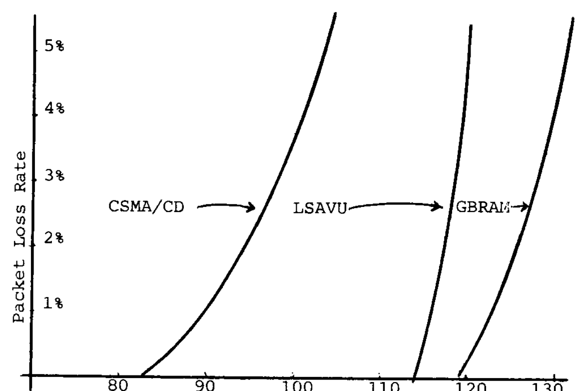

Figure 4: Comparison of packet loss for LSAVU versus CSMA/CD and GBram on a 10 Mbps cable with a packet length of 812 bits (102 bytes)