Analysis of Voice and Data Integration on the IEEE 802.5 Token Ring

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Beside of the application of Local Area Networks (LAN's) in data communications, the requirements for voice communication facilities in these systems become increasingly important. A number of voice support strategies in LAN's have recently been developed [5],[10]. In this paper the performance of a voice prioritization scheme for Token Ring LAN's according to the IEEE standard 802.5 is investigated. The approximate analysis is based on a non-exhaustive service polling system with finite capacity and non-zero switchover times. Stations are modelled by a subsystem consisting of two queues for voice and data, operating in a priority order. Results are presented to discuss the voice quality in presence of data traffic and in accordance with the priority operating mode. The approximation accuracy is validated by means of detailed computer simulations.

1. Voice and Data Integration

The increasing range of applications of Local Area Networks (LANs) reflect their importance in todays inhouse communication. The requirements in office scenarios to support both voice and data connections focus the point of view on LAN systems, which are capable of integrating these different traffic streams.

To support the synchronous traffic characteristic of voice communication the main objectives of packet-oriented LAN access mechanisms are to minimize the end-to-end voice transmission delay and to guarantee the required bandwidth for the voice connections.

Basically, the media access of Token Ring LAN's is an asynchronous access scheme, where data packets are transmitted over the ring as frames. The prioritization and reservation mechanism, defined for the IEEE 802.5 Token Ring LAN can be used to integrate voice communication facilities without any modification of the present standard.

Many modifications have been made, mainly adaptions of time division multiplex techniques [10, 7], to support the IEEE 802.5 Token Ring for voice communication. Analytic approaches can be found in [2, 3, 6]. In this paper we derive an approximate analysis of voice and data integration applying the access scheme described in the IEEE 802.5 standard. In Section 2 the offered traffic is charcterized and the Token Ring model is presented which deals with a modified polling system with limited-one service and non-zero switchover times. The analysis given in Section 3 is based on a cycle time analysis as discussed in Kühn [4], and takes into account finite buffer capacities, investigated by Tran-Gia and Raith [9]. Finally some results are presented in Section 4 which depict the system characteristics for voice and data traffic carried over the IEEE 802.5 Token Ring.

2. Station Modeling & Traffic Characteristics

For the analysis a model based on a non-exhaustive polling system with finite buffer capacities and non-zero switchover times is investigated (Figure 1). The Token Ring system is described by two types of stations: data stations, carrying only data packets, and so-called combined voice/ data stations, supporting voice and data packets. Both station types are mixed within the system, where the maximum number of all stations is denoted by g.

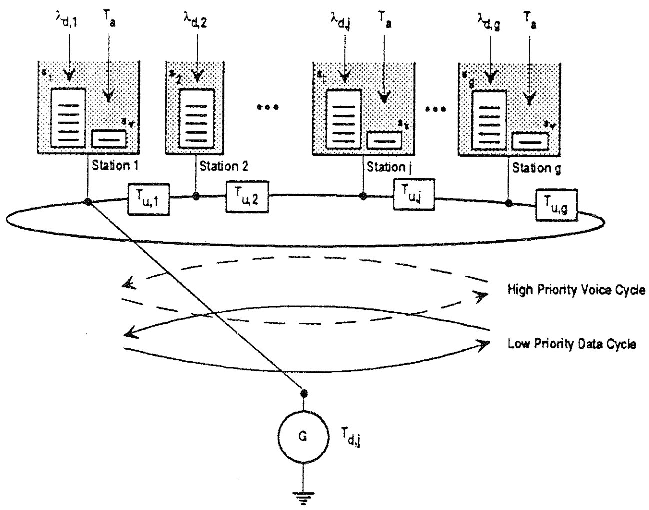

Figure 1: Model of the Token Ring System with two priorities and limited buffer capacities S_j for data packets and a single buffer for voice packets.

The system supports two priority classes; the high priority class is reserved for voice transmission, the lower priority class carries data packets. In normal operation mode a low priority token rotates on the ring. Each time a station receives a free token it has the permission to transmit exactly one packet. Voice packets are internally prioritized inside the combined voice/data stations. If a low priority data packet is passing a station with a voice packet waiting for transmission, then a reservation for the high priority token is made (see [8] for detailed description of the reservation mechanism). The high priority token can only be used by combined voice/data stations to transmit their voice packets. After exactly one high priority cycle, where all voice buffers are served, the token priority has to be switched back to the lower level to guarantee a minimum bitrate for data transmission.

While the arrival process of the data traffic is assumed to be Poissonian with station individual arrival rates $\lambda_{d,j}$, the transmission times for data packets may have general distributions and are described by the random variables $T_{d,j}$.

Since no silence suppression strategy is considered in this paper, the offered voice traffic to a station is characterized by the constant interarrival time T_a for voice packets. Furthermore, it is assumed that all voice packets arrive at the same instant at all combined voice/data stations. The ring latency between two neighbour stations and the transmission of the token is modelled by the generally distributed switchover time $T_{u,j}$.

3. Analysis

In this section the numerical algorithm for the analysis of the above described model will be presented. Based on the cycle time analysis [4] and using the technique of the embedded Markov chain the performance characteristics of the considered stations can be calculated [9].

3.1 Cycle Time Analysis

Assuming that all voice packets arrive at the same instant at all combined voice/data stations two cases have to be distinguished. First, the reservation is made within a low priority data packet; then, all pending voice packets are transmitted within the high priority cycle. If no reservation is made the first voice packet is transferred on the low priority level. Thus, the other pending voice packets have the possibility of reserving a high priority token. Therefore, the priority mechanism can be considered by inserting complete voice cycles. The length of this inserted cycle is described by the random variable T_h. The prioritization and reservation mechanism of the IEEE 802.5 MAC protocol leads to four different service sequences, which have to be distinguished. These service sequences depend on the station type and on whether a voice cycle was inserted due to a reservation or not. Figure 2 depicts the resulting cycle time segment of an arbitrary station j.

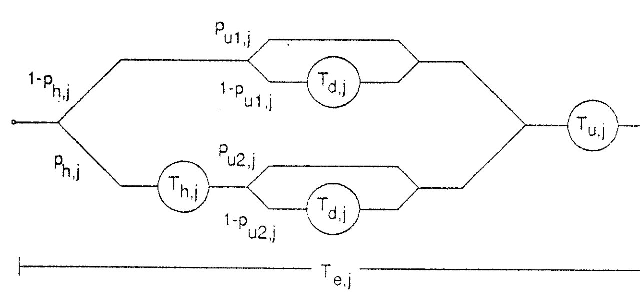

Figure 2: Cycle Time Segment of Station j

Let $\Phi_{e,j}$ be the Laplace-Stieltjes transform (LST) of the cycle time segment $T_{e,j}$ of station j. For ease of reading the station index j will be omitted in the following. Φ_e is given by:

$$\Phi_e = \Phi_u \{ (1-p_h)[p_{u1} + (1-p_{u1})\Phi_d] + \\ + p_h\Phi_h[p_{u2} + (1-p_{u2})\Phi_d] \}, \qquad (1)$$

where Φ_u, Φ_d, and Φ_h denote the LST of the switchover time, the service time and the high priority voice cycle time, respectively. p_h represents the probability of inserting a high priority voice cycle, p_{u1} reflects the probability that the considered data queue is empty and p_{u2} is the branching probability that no data packet is served after an inserted voice cycle according to Figure 2. They can be expressed by Eqns. (2 - 4).

$$p_h = \frac{E[T_e]}{T_a} \qquad (2)$$

$$p_{u1} = q_0 \qquad (3)$$

$$p_{u2} = 1 - (1-q'_0)p_d \qquad (4)$$

In Eqn. (3) and (4) q_0 and q'_0 denote the probability that the data queue is empty, when the station receives a low priority token and after an inserted voice cycle, respectively. The probability p_d reflects the influence of the priority and reservation mechanisms. It is obtained by distinguishing the cases whether the voice packets arrive during the token switchover or during the service time at a pure data station (Eqn. 5a) or combined voice/data station (Eqn. 5b). Furthermore, the case of voice overload has to be taken into account. After having taken all these cases under consideration p_d can be expressed by Eqn. (5):

$$p_d = 1 \qquad (5a)$$

$$p_d = [1-F_c(gT_a)] \Big[(1-p_h)(1-p_{u1}) \frac{E[T_d]}{E[T_u]+E[T_d]} + \\ + p_h(1-p_{u2}) \frac{E[T_d]}{E[T_u]+E[T_d]} \Big] \qquad (5b)$$

In the above Eqn. F_c denotes the distribution function of the cycle time T_c which is obtained by calculating the total probability from four conditional cycle times T'_c - T''''_c (see Section 3.2):

$$T_c = (1-p_h)\, p_{u1}\, T'_c + (1-p_h)\,(1-p_{u1})\, T''_c + \\ + p_h\, p_{u2}\, T'''_c + p_h\,(1-p_{u2})\, T''''_c \qquad (6)$$

3.2 Markov Chain State Probability

The system characteristics of the data queues can be obtained by the method of an embedded Markov chain. Let $L_j(t)$ be the number of packets waiting in data queue j just prior to the arrival of a low priority token. Then, the steady state probabilities of the Markov chain can be expressed by

$$q_{k,j} = \Pr\{ L_j(t) = k\}, \qquad k = 0..S_j. \qquad (7)$$

Now we introduce four conditional cycle times $T'_{c,j}$, $T''_{c,j}$, $T'''_{c,j}$, and $T''''_{c,j}$ corresponding to the four alternatives of Figure 2, i.e. each conditional cycle time represents the time interval between two successive low priority token arrivals at station j under the condition that in the preceding cycle neither a voice cycle is inserted nor a data packet is transmitted, no voice cycle is inserted but a data packet is transmitted, only a voice cycle is inserted, or a voice cycle is inserted and a data packet is transmitted at the considered station. The LSTs $\Phi'_{c,j}$, $\Phi''_{c,j}$, $\Phi'''_{c,j}$, and $\Phi''''_{c,j}$ of these four conditional cycle times are given by Eqns. (8 - 11)

$$\Phi'_c = \prod_{\substack{k=1\\k\neq j}}^{g} \Phi_{e,k} \qquad (8)$$

$$\Phi''_{c,j} = \prod_{\substack{k=1\\k\neq j}}^{g} \Phi_{e,k}\ \Phi_{d,j} \qquad (9)$$

$$\Phi'''_{c,j} = \prod_{\substack{k=1\\k\neq j}}^{g} \Phi_{e,k}\ \Phi_h \qquad (10)$$

$$\Phi''''_{c,j} = \prod_{\substack{k=1\\k\neq j}}^{g} \Phi_{e,k}\ \Phi_{d,j}\, \Phi_h, \qquad (11)$$

where Φ_h denotes the LST of the length of a voice cycle T_h.

With Eqns. (8 - 11) the transition probabilities of the Markov chain can be computed. Since the cycle segment length T_e depends on the state probabilities q_0 and q'_0, an iterative algorithm must be introduced in order to compute the steady state probabilities q_k (see [9] for detail). Since no data packets can be served during an inserted high priority cycle the state probabilities q'_k for the instant just after this inserted cycle can easily be derived from the q_k's by considering the arrivals during T_h.

With the introduction of the random variables $T_c^{r'} - T_c^{r''''}$ which denote the backward recurrency times of $T'_c - T''''_c$, repectively, the state probabilities can be calculated at arbitrary time instants.

3.3 Voice Buffer Characteristics

One of the most interesting performance characteristics in LAN's with voice/data integration is the calculation of the waiting time for voice packets. Therefore, in a first step we have to distinguish whether voice packets are lost or not. In the best case a voice packet can be served immediately after its arrival. This occurs when the preceding packet has already made a reservation and the station gets the token just at the instant the voice packet arrives. In the worst case a voice packet can at most wait the voice packet interarrival time T_a. If it was not transmitted after this time, it will be overwritten by the next packet due to the assumption of a single voice buffer at the stations. Thus, the mean waiting time for voice packets is given by $T_a/2$.

If no voice packets are overwritten, the mean waiting time consists of two parts, the residual service time of the previous station and the waiting time until the station gets a high priority token. The last has a mean of $T_h/2$. The residual service time T_{rj} depends on the considered station j. Its mean is given by

$$E[T_{rj}] = (1-p_h)\, q_0\, E[T^v_{uj}] + p_h\, p_{u2}\, E[T^v_{uj}] + \frac{E[T_{uj}]\, E[T^v_{uj}] + E[T_{dj}]\, (E[T_{uj}]+E[T^v_{dj}])}{E[T_{uj}]+E[T_{dj}]} \cdot [\, p_h\, (1-p_{u2}) + (1-p_h)(1-q_0)\,] \qquad (12)$$

where T^v_u and T^v_d denote the forward recurrency time of the switchover and service time, respectively. If a voice packet arrives no prediction can be made which station starts the voice cycle. But it is possible to calculate a mean value for all stations by weighting the station dependent residual service times. Finally we obtain for the mean waiting time of voice packets $E[T_{wv}]$

$$E[T_{wv}] = \frac{T_h}{2} + \sum_{j=1}^{g} \frac{E[T_{rj}]}{g} \qquad (13)$$

To decide whether the system works under voice overload, the mean of the cycle time may T_c with may be compared with gT_a. Since at every station only one voice cycle may be inserted, a maximum of g voice packets can be served per low priority cycle at each combined voice/data station. If $gT_a < E[T_c]$, then voice packets must have got lost.

4. Results

In this section numerical results, which are obtained by the analysis described above, will be presented and discussed for a token ring LAN with a fixed number of 100 stations including a variable number of combined voice/data stations. The transmission rate on the ring was assumed to be 4Mbps. The length of the info field of a data packet is negative exponentially distributed with a mean of 1000 bit. The packet overhead has a length of 144 bit. The offered data traffic is equally distributed over all stations. Voice packets have a constant length of 640 bit with a deterministic interarrival time T_a of 10ms. The switchover time is assumed to be constantly 0.75µs and is obtained from [1]. In order to validate the approximation, detailed computer simulations are provided. The simulation results will be depicted with their 95% confidence intervals.

Figure 3 depicts the throughput of data packets versus the number of combined voice/data stations. It can be seen, that the algorithm shows an excellent accuracy. The throughput is constant until the number of combined stations reaches a critical point. Then the data throughput

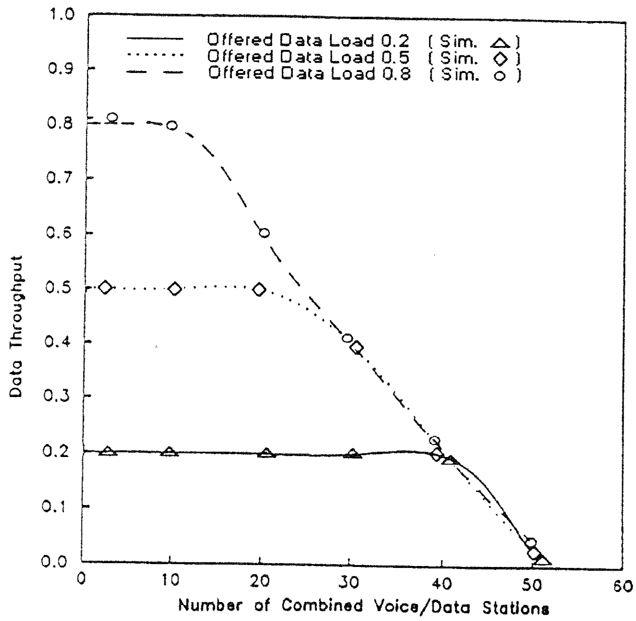

Figure 3: Data Troughput versus Number of Combined Stations

decreases rapidly. At this point the ring is totally utilized. If now the number of combined stations is increased (which means an increasing offered voice load) the low prioritized data traffic gets not enough bandwidth. The data packets are blocked, thus, their mean waiting time increases roughly (see Fig. 4). Then increasing the actual voice traffic, the mean data waiting time increases

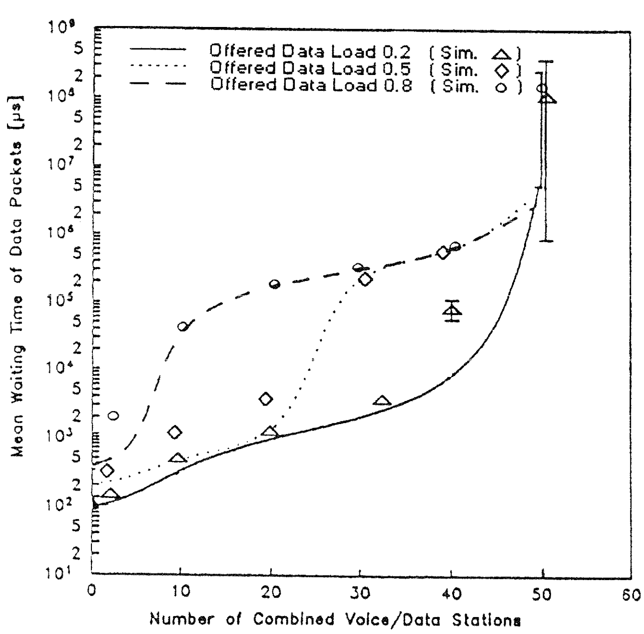

Figure 4: Mean Waiting Time of Data Packets versus Number of Combined Stations

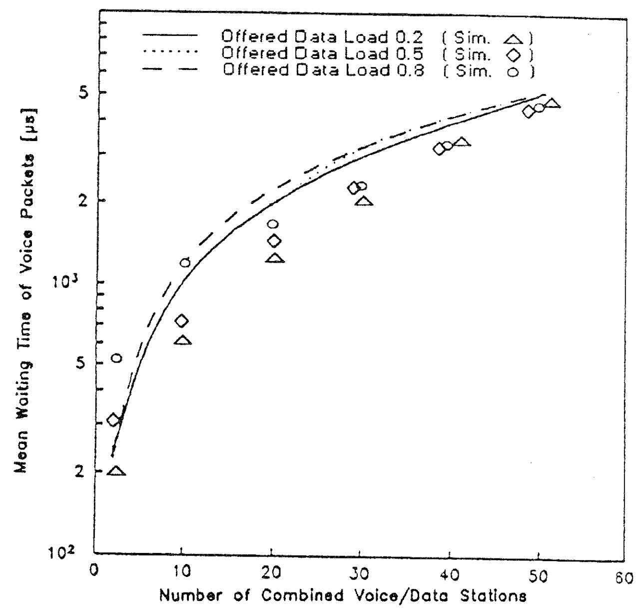

Figure 5: Mean Waiting Time of Voice Packets versus Number of Combined Stations

slightly due to buffer limitations of the data queues.

Figure 5 shows the mean waiting time of voice packets versus the number of combined stations. It can be seen, that it is nearly independent of the offered data load. The reason for this effect is the higher priority level for voice transmission, where only the lower prioritized data packets are blocked if the total system capacity reaches the overload threshold.

The accuracy is here not as good as in the previous figure but still the main tendencies are provided by the analysis.

5. Conclusion

An analysis for Token Passing LAN's according to the IEEE 802.5 standard with two priority classes is provided. The iterative solution, based on a conditional cycle time analysis and an embedded Markov chain approach for the finite data queue is developed. The results in terms of waiting times, throughput and loss probabilities are presented. Using a realistic Token Ring assumption the prioritization behaviour is discussed and the accuracy of the algorithm is validated by detailed computer simulations.

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