ANALYSIS AND PERFORMANCE EVALUATION FOR PACKETIZED VOICE / DATA INTEGRATED TRANSMITTED ON A LAN

By

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Abstract -Integrated packet-switched networks have potential for providing improved performance by dynamically sharing transmission bandwidths between voice and data users. This paper investigates the performance of a token passing ring network with packetized voice and data mixed traffic. Token passing ring local area networks are shown to effectively handle both voice and data traffic. The effects of system parameters (e.g. voice packet length, voice traffic intensity, data traffic intensity, number of voice calls, and service discipline) on network performance are discussed. The queueing delay of the system is obtained and from the results the minimum value of the queueing delay is determined.

1. Introduction

Voice and data have different characteristics and different fidelity requirements. Packetization of voice makes it possible to carry voice along data on a single integrated packet-switched network. The performance of a token passing ring network when it is subjected to a voice load in addition to a data load is presented. Past research has focused on analyzing its performance for data applications.

This paper evaluates the performance of a token passing ring network through extensive simulations utilizing, whenever possible, the IEEE 802.45 standard for token passing rings. It examines the dependencies of network performance on both voice and data traffic, and the ways in which system parameters may be altered to allow a network to yield an acceptable level of performance. The performance measures obtained include : the distribution of transmission delays for voice packets, the average transmission delay for voice packets, the number of voice users allowed on a network which satisfying the real-time constraints of speech, and the average transmission delay for data packets. A time division multiplexing protocol is considered in this paper for the accommodation of voice and data traffic in an integrated services digital network. The movable boundary TDM protocol has been adopted the voice / data integration, and advent studies in discrete-time have been on the generating function approach, [1], [2], [3]. The discrete-time, movable boundary TDM system considered in this paper is different from those studied in the past in at least two aspects. First, the data traffic is assumed to be correlated. Second, the voice traffic is not necessarily accommodated in a subframe of contiguous slots, but it may be spread over the entire frame according to any pattern.

Significant effort is currently being devoted to the development of packet oriented technologies for integrated multiplexing and switching of voice and data [4]. In this paper, we concentrate on integrated networks in which voice packets are also queued and therefore served synchronously. However, generally voice packets would be given priority over data so as to ensure a low delay for voice.

2. Simulation model

2.1 Network model

In our simulation model, two types of users are assumed on a token passing ring network : voice and data users. At each voice user, a continuous voice analog signal is digitized by a coder. For instance, a typical PCM encoder produces one 8 bit word every 125 MS. the generated samples are accumulated a packetizer. When the number of samples in the packetizer reaches the predetermined packet length, a header is attached and a voice packet is generated.

The generated voice packet is then examined by a speech activity detector to see if it contains some minimal level of speech activity. Silent packets are discarded. Nosilent packets are stored in the buffer in the order of their generation, and await transmission. The packet generation cycle is independent of the packet transmission process; thus the queue size at the buffer continues to grow while a packet is waiting for transmission since we assume infinite buffer capacity, there is no packet loss at the voice-source user.

Transmission control is based on a token, as in IEEE 802.5 standard, no priority is assumed standard, no priority is assumed for voice users over data users [5]. When a free token is passed to a user ready to send a packet, the user changes the token status to busy and appends a packet to the token. The packet circulates around the ring to the intended destination user and then released downstream to the next user with a packet available for transmission.

In general. we assume that only the head packet of a user buffer can be served by a token (i.e., the limited service discipline described in the IEEE 802.5 standard). The packets remaining in the buffer must wait for later visits of the free token.

The performance model for a token ring is shown in Fig. (1), which is a single-server queueing model with an many queues as stations attached to the ring.

The queues are serviced in a cyclical manner symbolized by the rotating switch which stands for the free token. The service time of a packet is given by $T_p = (L_h + L_p)/v$. The time needed for passing the free token from station i to station i +1 (propagation delay plus additional latency within station i for token handling, etc.) is modeled by a constant switch over delay. τ_i.

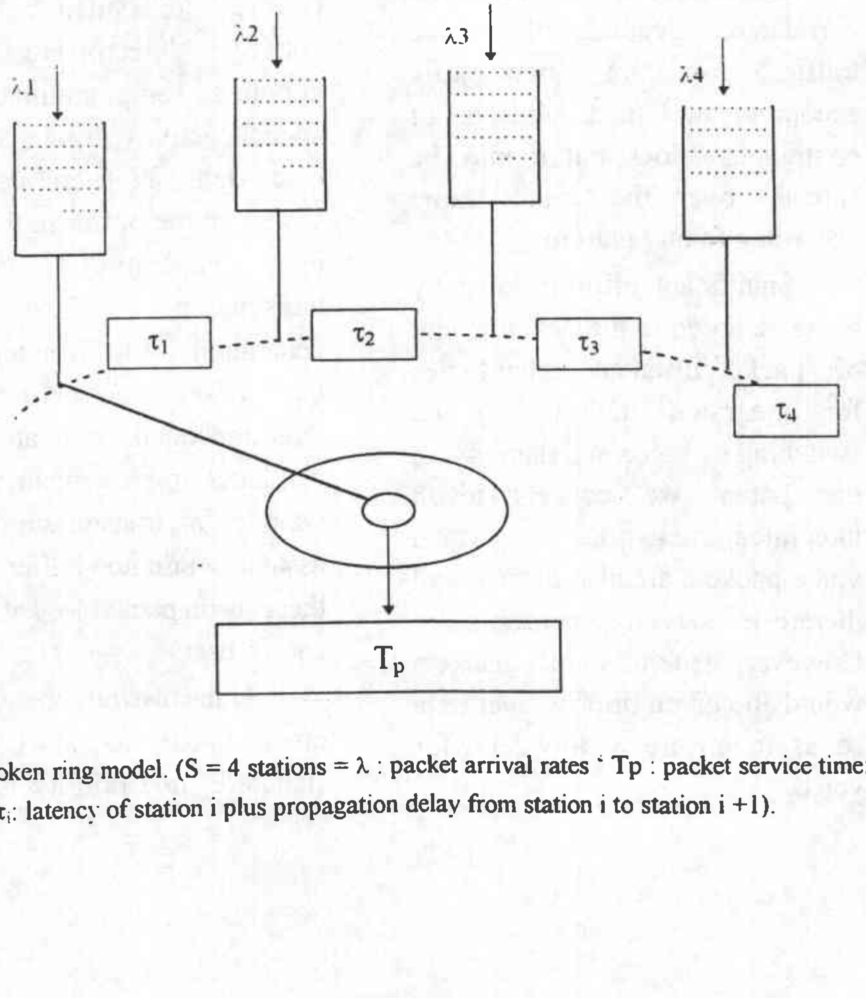

Fig. (1) : Token ring model. (S = 4 stations = λ : packet arrival rates ' Tp : packet service time: τ_i: latency of station i plus propagation delay from station i to station i +1).

2.2 Queueing model :

The data traffic is assumed to be correlated: it is modelcd as a Markov Modulated Generalized Bernoulli (MMGB) process [6]. According to this process, packet arrivals are governed by an underlying Markov chain (with some state space s). Transitions between states of the chain occur at slot boundaries. The packet arrival process is determined in terms of a probabilistic mapping a (.) : S {0,1,............., R}, where R, R <∞, is the maximum number of packets delivered per slot by the MMGB process.

The movable boundary policy is adopted, the relation $T_v \leq T$ is assumed to hold (no blocking of voice traffic) and voice packets are assumed to be transmitted over the first Ri (contiguous) slots of the i-th frame, at the beginning of which R_i voice sources are active.

The performance analysis of the voice queue is straight forward, [6]. The analytically challenging problem associated with the TDM system (with data queue and voice queue) is that of the evaluation of the queueing behavior of the data packets, which is affected by the activity in the voice queue. The interference from the voice queue on the data queue is modeled as an independent form anything associated with the data queue process $\{R_j\}_j \geq 0$.

→ The interference process $\{R_j\}_j \geq 0$ is defined by; $R_j = 1$ if $0 \leq j \bmod T \leq R_i - 1$, $i = [\, j/T]$, and $Rj = 0$ otherwise. The behavior of the data queue can be studied by considering the equivalent system without service interruptions (Fig. 2) and assuming HOL priority policy for the packets delivered by $\{Rj\}_j \geq 0$. Let d^d_q and d^v_q denote the mean packet delay of the data and voice packet respectively.

The mean packet delay indeed under the FIFO policy for the queueing system in Fig. (2), is denoted by ; D_F [7] :

$$D_F = \frac{\lambda_d\, d^v q + \lambda_v\, d^d_q}{\lambda_d + \lambda_v} \qquad (1)$$

Where : λ_d and λ_v are the input traffic rates to the data and voice queue.

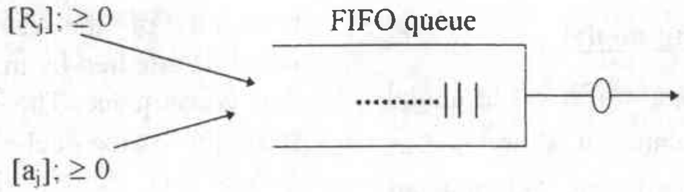

Fig. (2) : The eqwialent queueing system

3- Performance Measures

Since excessive delays can have seriously disruptive effects on human conversation, voice packets must be received at the destination user within a fixed amount of time after their generation at the source user. Those packets that do not arrive within this time bound are considered lost and are discarded upon their arrival at the destination user. A small number of lost packets has been shown to have little, if any, effect on human spach intelligibility.

3.1 Voice packet transmission delay

If the transmission delay (D_v) of a voice packet is defined as the time interval between the beginning of its digitization and the time it is played out at its destination, then D_v becomes.

$$D_v = T_p + D^v_q + T_v + R_{o,d} \quad (2)$$

Where Tp is the voice packet generation period

$$T_p = P_v/v = \frac{\text{Voice packet length excluding header bits}}{\text{Voice coding rate (bits/sec)}}$$

The queueing delay (d_q') is related to the rate of arrival packets on the queue and the capacity of the outgoing. In the network delay probability density function p (t) is assumed to be an exponential one [8],

$$\text{i.e } p(t) = \lambda e^{-\lambda t}$$

$$= \mu (1-\rho) e^{-\mu (1-\rho) t} \; L \geq 0$$

Where : μ is the transmission service time

ρ is traffic intensity

$1/\lambda$ is average network delay and standard deviation.

Clearly queueing delay, d_q^v, will be a direct function of the expression μ (1 - p) as seen from equation (3).

$$d_q^v \; \alpha \; \frac{1}{\mu (1-\rho)} \quad (4).$$

$$d_q^v = c_k \frac{1}{\mu (1-\rho)} \quad (5).$$

Where : C_k is constant related to the number of tandem links k.

We have previously specified each voice packet to be Tp second long; therefore the number of packets capable of being generated by a single voice circuit is equivalent to 1/Tp packets sec.

If we assume an average of N voice circuits active on a link and we further assume that since packet is only active about half the time. we have the relationship.

$$= N / 2\,Tp\ ;\ (\lambda/\mu = \rho) \quad (6)$$

$$\mu = \lambda/p = \frac{N}{2Tp}\ \frac{1}{\rho}$$

$$\therefore dq = C_k \quad \frac{2\,Tp.\,\rho}{N\,(1-\rho)}$$

$$= Ck\ \frac{2Tp}{N}\ [\frac{\rho}{1-\rho}] \quad (7)$$

The voice queueing delay is incurred while gaining access to the network, and in the actual transmission of the packet (T_v).

$$T_v = \frac{(P_v + H)}{C} \quad (8)$$

Where : C is the charnel speed (bits / s).

Ro,d (s) is the sum of the bit latency and propagation delay between the origin and destination users. Bit latency is the delay introduced at users to monitor and change the token bit pattern.

3.2. Data packet transmission delay

The transmission delay D_d of a data packet is defined in a similar manner : $D_d = d_q^{d} + T_d + R_{o,d}$ where d^d_q is the queueing delay of a data packet at the source user and Td is the transmission time of a data packet. Td is given by $Td = P_d / c$ where p_d is the data packet length, including header, in bits.

Data traffic intensity on the network is defined as follows :

$\rho_d = N_d\ \lambda_d P_d/c$ where N_d is the number of data users, and λ_d is the arrival rate of data packets from a data user.

The expected number of data packets waiting for service in the system is given by [9].

$$\frac{\rho_d(1+\rho_v)^2 + \rho_v\,\lambda_d\,\mu_v}{(1+\rho_v)\,(1-\rho_v\,\rho_d-\rho_d)} \qquad (9)$$

Where μ_v is the average holding tire then the average data packet queueing delay is given by : $d_q^d = E\,(Qd) / \lambda_d$ (10)

$$= \frac{\rho_d(1+\rho_v)^2 + \rho_v\,\lambda_d\,\mu_v}{(1+\rho_v)\,(1-\rho_v\,\rho_d-\rho_d)}$$

4- Numerical results

4.1 The effects of voice palat length on performance

In the network model used, the length of the network ring is assumed to be 1km and voice and data users are distributed uniformly around the ring. The values used for the token length (3 octets) the packet header length H (21 octets) and the channel speed C (1 M bits/s) are based on the IEEE standard 802.5 for token ring local area network [5]. We assume 1 bit latency at each user, and a propagation delay of 5 µs/km of cable.

From fig. (3) , it is clear that the shorter the voice packet length, the smaller the average network delay. This is due to the influence of the packetizaton delay on D_v. The relation between the voice packet queueing delay and the voice packet length is shown with two values for the number of voice users (N = 10, N = 5). Where the value of the voice packet queueing delay is larger with N = 10. The selection of voice packet length (P_v) to maximize the number of voice users (N) or a network depends on the performance criteria used.

4.2. The effects of voice and data traffic intensity on performance

Fig (4). Shows the voice packet queueing delay and the voice traffic intensity as a function of the number of voice calls on a network (N. = 15,10).

In the figure , the voice packet queueing delay increases with the voice traffic intensity with voice coding rate 64 kbps and constant voice packet length (P_v = 1000 bits) the value of d^v_q for N = 5 more than N = 10 with increasing of P_v.

Fig (5) Shows the effect of

data traffic intensity on the data packet queueing delay as a function of the voice traffic intensity (ρ_v = 0.1 and 0.5) form figure it is clear that d^d_q is decreased with small values of voice and data traffic intensity.

The relation between the voice intensity and the data packet queueing delay for different data traffic intensities is shown in fig (6).

4.3. <u>The system queueing delay of the integrated packet switching system.</u>

The analytically challenging problem associated with the TDM system with data queue and voice queue, is that of the evaluation of the queueing behavior of the data packets, which is affected by the activity in the voice queue (the queueing model of the system).

The mean packet delay induced under the FIFO policy for the queueing system D_F is given by eq. (1).

Fig. (7) Shows the variation of the system queueing delay with the data arrival rates. From figure it is shown that there is no effect for the small variation of the voice calls allowed on a network.

Fig. (8) Displays the variations of voice traffic with the system queuing delay for N = 10. In the figure, the minimum value of D_F can be obtained and for example the minimum value of the system queueing delay is 5.22 m. sec at ρ_v = 0.3 (with λ_d = 10).

5. Conclusions

Packetization of voice makes it possible to carry voice along data on a single integrated packet-switched network. The advantage of integrated packet voice/data networks are many, e.g., the efficient sharing of transmission and switching facilities, the capacity advantage due to statistical multiplexing, and the potential evolution toward a fully integrated network which would provide image and video services as well.

In this paper, simulation models were developed to evaluate the performance of packetized-voice/data transmission on a token passing ring network. The results suggest that system parameters (e.g. the voice packet length, voice and data traffic intensity) can be adjusted to allow token passing ring networks to support both voice and data traffic with acceptable performance.

We concentrate on delay as the performance parameter of interest.

The results show that the minimum value of the queueing delay of the integrated system is obtained at special voice traffic intensity values.

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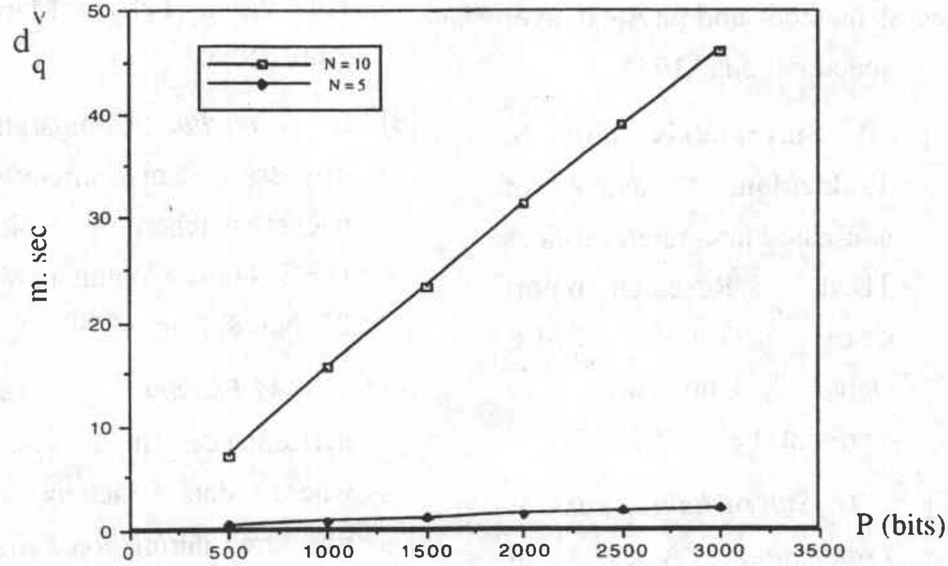

Fig. (3) The effect of the voice packet length on the voice packet queneing delay

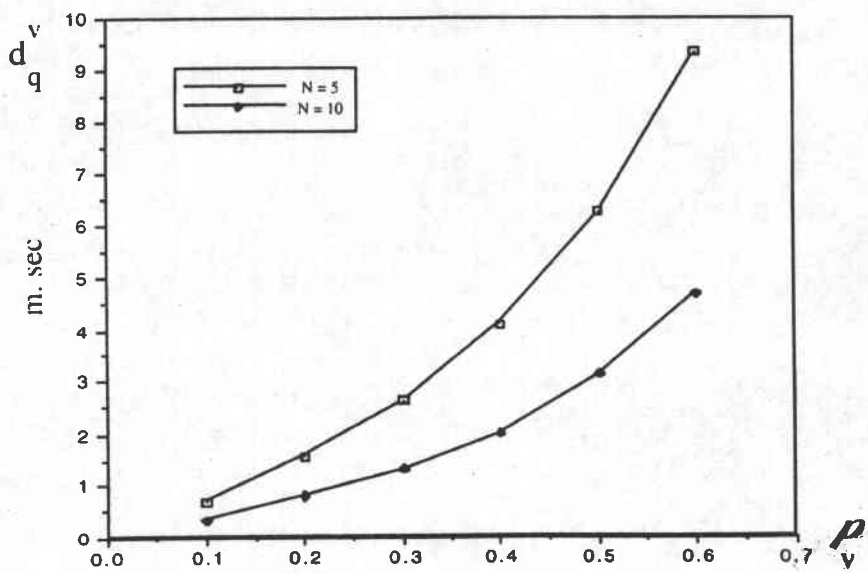

Fig. (4) The voice packet queneing delay versus the voice traffic intensity

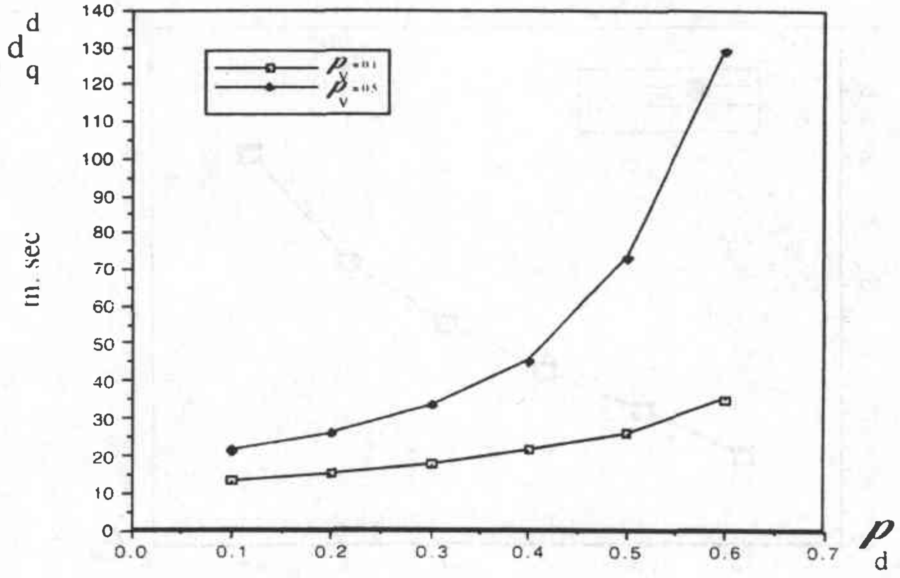

Fig. (5) Data packet queneing delay versus the data traffic intensity

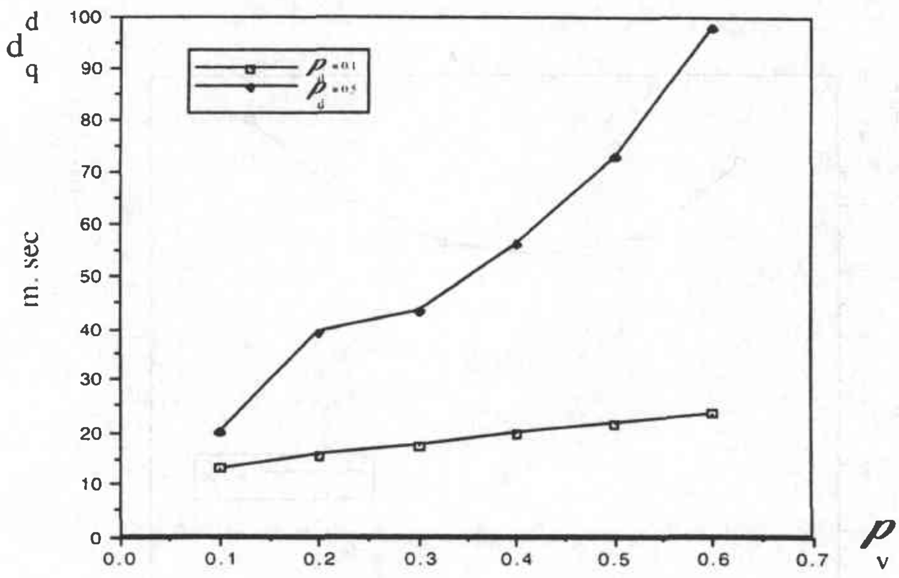

Fig. (6) The relation between the voice intensity and the data packet queneing delay for different data traffic intensities

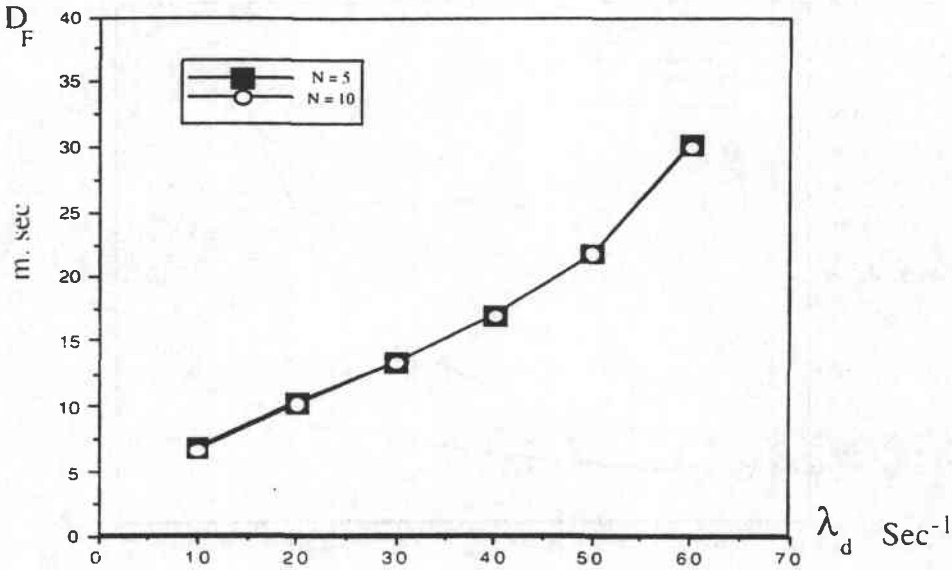

Fig. (7) The system queneing delay of the integrated packet switching system versus the data arrival rate.

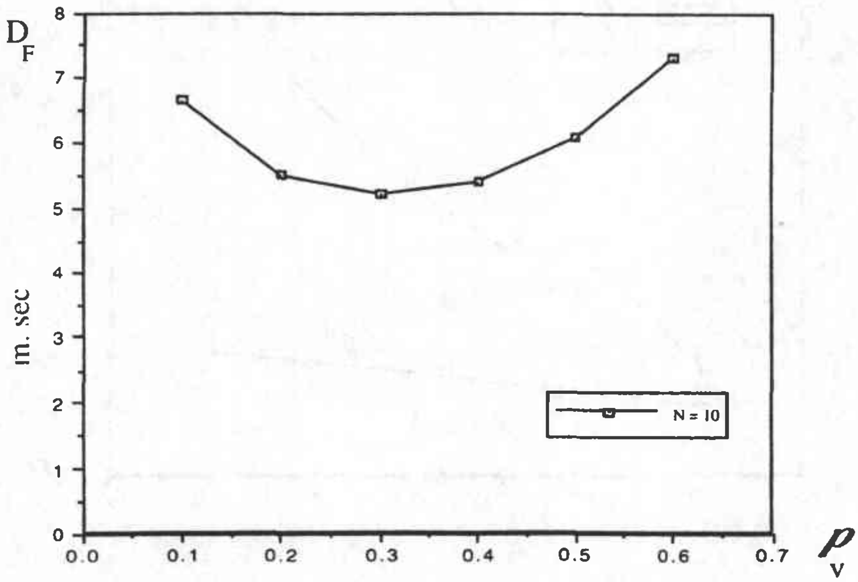

Fig. (8) The system queneing delay versus the voice traffic intensity