MILITARY TECHNICAL COLLEGE

CAIRO - EGYPT

A MODEL FOR VOICE / DATA INTEGRATION ON S- TDM

USING MOVABLE BOUNDARY,HYBRID SWITCHING TECHNIQUE

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Abstract

This paper assesses a model of voice / data integration on statistical time divsion multiplexing technique.

A comparison between fixed and movable boundary frame structure is introduced and the selection of a movable boundary than fixed boundary _due to the efficient sharing of voice and data transmission capacity in the frame period with higher channel utilization_ is discussed.

Hybrid switching system which combined circuit and packet switching techniques is provided in this model, where the hybrid switching frame period consists of both circuit switched voice traffic and packet switched data traffic

The performance of the model is evaluated and the results obtained are analyized , also the blocking probability of voice source and the waiting delay of data packet are measured .

The effect of number of slots(N 1,N 2) assigned to voice / data sources on the system performance is evaluated.

The effect of the offerd voice load on the waiting delay of data packets for the proposel integration model is also provided

1- Introduction

In recent year 's the modeling and analysis of voice and data integration on TDM systems have recived great attention [2]. generally , there are three ways to approch this problem simulation , exact dervation and approximate analysis [3]. section 2 present a discussion of the characteristics of statistical time division multiplex with some basic queueing concept used for voice/ data integration.

In section 3 a comparison between fixed and movable boundary frame structure for integrated voice / data is discussed from channel utilization, blocking probability, waiting delay and number of subscribers serviced point of view. A model of voice sources and data packets are described in section 4, section 5 present A summary of the paper with the important conclusion.

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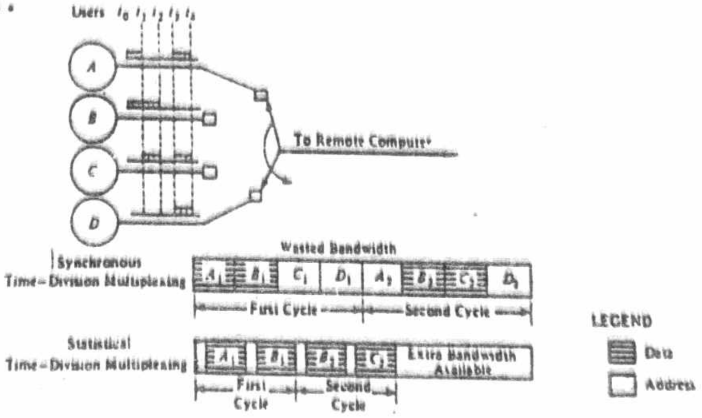
2- Statistical - TDM characteristics

In syncronous time division multiplex. Its generally the case that many of the time slots in a frame are wasted. A typical Application of a syncronous TDM involves linking a number of terminals to a shard computer port.

An alternative to syncronous TDM is statistical- TDM or asyncronous TDM. The statistical multiplexer exploits this common property of data transmission by dynamically allocating time slots on demand.

S - TDM has a number of I/O lines on one side and higher speed multiplexed line on the other, also, there are n I / O lines. but only k where k < n time slots available on the TDM frame.

Fig (1) contrasts statistical and syncronous TDM. Its shown that syncronous TDM takes sample from each user, whatever this user active or idle, while statistical takes sample from the active user only. It means A higher channel utilization and can service a higher number of user than syncronous TDM.



** Fig (1) syncronous TDM contrasted with S- TDM

2-1 The M/M/N/N queue system:

This system models the customers according to poisson arrivals with average rate and always find a trnsmission channel until the maximum number of N channel is occupied, then the customer arrival is blocked. [5] so, the blocking probability is estimated as :

$$Pn = \frac{\rho^N / N!}{\sum_{i=0}^{N} \rho^i / i!} \qquad \text{.....................} \quad (1)$$

for n = N then the blocking probability $P_b = \dfrac{\rho^N / N!}{\sum_{i=0}^{N} \rho^i / i!}$ (2)

2.2 Voice/ data integration on s-TDM concepts

Integrated circuit - packet switching system requires a certain control mechanisem in order to handle its traffic efficientely and smothely. This mechanism should implement to maxmize the system throughput while satisfied the required grade of service for each traffic type.

As in Fig (2) a master frame format of statistical time division multiplex facility, where certain portion of the frame is allocated for voice calls and the data traffic are allocated for the remaining frame capacity. This provide a hybrid switching techniques, voice calls will in effect be circuit switched by assigning syncronous slots while data traffic is statisically multiplexed by asyncronous packet distribution.

This technique provides a maximum flexibility in handling both traffic and efficient channel utilization with minimum blocking probability for voice call and minimum waiting delay for data packet as in [5], [6]. The genral model for voice/ data integrated systems assumes the users with different bandwidth (bit rate) requirements, different holding or service time, and different arrival rates for the circuit and packet switched traffic.

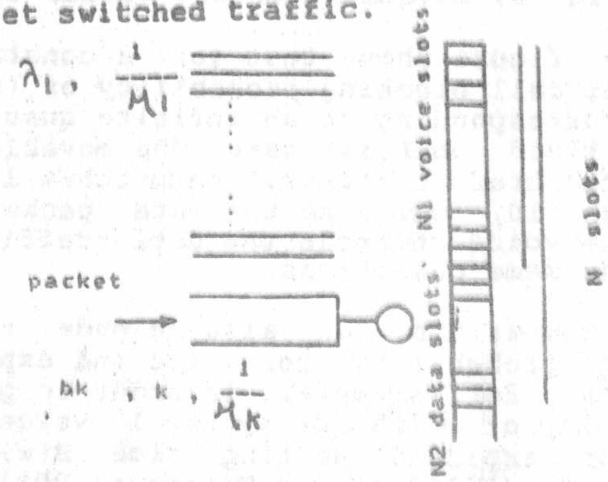


Fig (2) The integration in S-TDM system

3- A Comparison between fixed and movable boundary schemes:

The dicsussion of the movable boundary in recent papers have introduced a multiplexing structure for mixing voice and data traffic in integrated telecommunication system. This structure utilizes a master frame format of time division multiplexing facilities.

A certain portion of the frame is allocated for voice calls, and data traffic is assigned to the remaining frame capacity, to achieve a higher transmission utilization, data are allowed to use any residual voice capacity momentiarly available due to statistical variation in the voice traffic. the voice traffic is treated as loss system and data packet are bufferd, this technique is called movable boundary scheme.

After derivative an expression for the probability of lost voice call, and the expected waiting time for data packets [2]. The numerical results are given for a comparison of the fixed and movable boundary as in fig (3) .

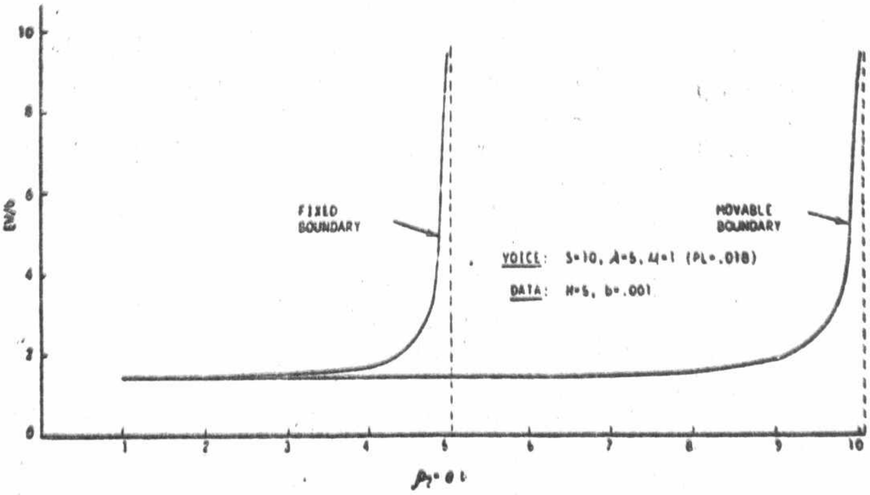


Fig (3) movable boundary versus fixed boundary

The figure shows that for a constant voice load with resultant call blocking probability of (.018) data intinsity of 0.1 corresponding to an infinite queue occures for $\gamma 2$ = 5 at the fixed boundary case. The movable case with the same system the traffic intensity approches 1.0 at the data load approches 10, which as the data packet are allowed to use the free voice channels the data traffic could be doubled under the same condations.

A comparison can also be made to satisfy a given blocking probability for voice and expected waiting time for data. for example, in order to gurantee a blocking probability of .0223 for system 16 voice channels are needed for the expected waiting time E(w) less than .0019, ($\lambda 2 = \theta$ = 8000 and b = Tframe = .001) nine data channel yield for the fixed boundary case while for the movable boundary case, only four channel are needed. The correspnoding channel utilization is 89%, but even if this were too high a large saving in the total number of channels can be achieved. [2] this summarized in table (1)

Table (1) A comparison between Fixed and movable boundary

Parameter	fixed boundary	movable boundary
Pb	.0223	.0223
E (w)	1.84	1.73
N	25	20
utilization	71%	89&

Another analysis for comparison between fixed and movable boundary is discussed in detail in [4]. which analyize the performance of an integrated switch with fixed and variable frame rate and movable voice/ data boundary. Its shown that a variable frame multiplexer with a min- max constraints on frame length can be designd to achieve higher channel utilization, and at the same time lower blocking probability and delay than the fixed frame counterpart.

4- A model of a movable boundary using hybrid switching on S-TDM

Assuming the link is based on Frame with duration T frame = b

The Frame is divided into N slots (channels), N1 of N is allocated to voice traffic on circuit switched basis, the remaining slots consists of N2 of N which assigned to data traffic on packet switched basis and the unused slots of voice traffic with considering voice slots size the same with data slot size with the help of the analysis done by (M.J fisher, harris) [2] so, the model can be considered as follow:

- our traffic is of class 1 voice calls and class 2 data message.
- call arrival of voice traffic have poisson arrival rate of $\lambda 1$
- holding time of call have exponential disterbution $1/\mu 1$
- number of server for voice traffic is N1 (No waiting room)
- date traffic have poisson arrival rate $\lambda 2$
- the service time of packet arrival with mean $1/\mu 2$
- the data traffic can use N 2 slots assigned for data packet and the free voice slots N * where $N^* = N2-\rho 1$ (1-pb)
- where B.... is the Blocking rate of voice call. $\rho 1$ is voice utilization, Fig (4) illustrate this queue model

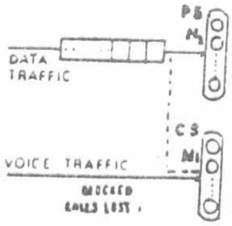


Fig (4) queue model

The analysis is done under the following assumptions:

(1) The system data queue are assumed to be infinite
(2) The frame width equals. T frame = b with N1 voice channels circuit switched, reigon, N2 data channel packet switched reigon as Fig (5)

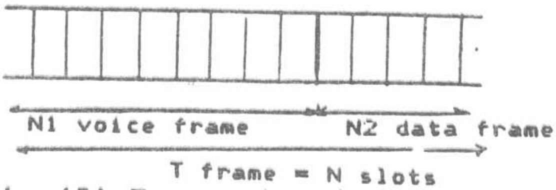


Fig (5) Frame structure

(3) The model is valied under steady state conditions which mean the rearrange periods of voice and data if error or collision is occure are neglected due to the small time of the end to end propagation delay in the order of several microsecond [5].
(4) The arrival rate and service time mean value are taken to data message (Packet) and also for voice calls.
(5) The data source can utilize the free voice slots in the frame.
 * one single data packet need one slots to be transmitted in one frame
(6) The incoming sources for voice and data entering the system is greater than the available slots for both traffic.

Modeling of voice system

The system state at time t is described by (i (t), j (t)) representing the number of two traffic types at time t. the free voice channel may be used by any data packets and the number of voice calls is not affected by any data packets using its channels because of the preemption allowed.
The voice calls are given by the erling- B formula for blocking probability, that represents the system having more than N1 active voice sources

$$P_B = \frac{\rho_1^{N_1}/N_1!}{\sum_{i=0}^{N_1} \rho_1^{i}/i!} \qquad \text{.................................} \quad (3)$$

at the frame has all N1 slots occupied with N1 active voice source So, new active source must wait until an occupied slot becomes free, this waiting time is limited with certain time out, and if its exeeds the limit, the call is consider to be blocked. due to [5] that illustrate A new protocol for integrated voice/ data which used priority system with respect to voice user in the queue that wait after the voice user slots becomes free. Also, heurstic means have shown that for certain voice source with address j the blocking probability is estimated as : [5]

$$PB_j = \left[\frac{\rho_1^{N1} / N1!}{\sum_{i=0}^{N1} \rho^i / i!} \right] * \left[\prod_{i=1}^{N1} \frac{(j - i)}{(G - i)} \right] \quad \text{at } j > N1 \quad ...(4)$$

where i is a counter, G.. total number of voice sources in the system and $\rho 1$ is the voice utilization = $\frac{\lambda 1}{\mu_1}$ where $\lambda 1$ is arrival rate of voice call and $\frac{1}{\mu_1}$ is the avarage call holding time, equation (4) represent the idealized case where the voice source is larger than the number of avialable voice slots in S - TDM.
So, due to the variation of voice utilization load we can sense by the blocking probability of voice calls with different voice channel allocations.

<u>Models of data sources:</u>

The system can be considerd to be an M/M/N1 for voice sources and an M/M/N2 + {N1 - ρ 1 (1-PB)} for date sources where N2 + {N1 - ρ 1 (1-PB)} is the number of free channels available for data sources. if data source does not find A free slot its enter to queue. the next state digram represent the case of this model.

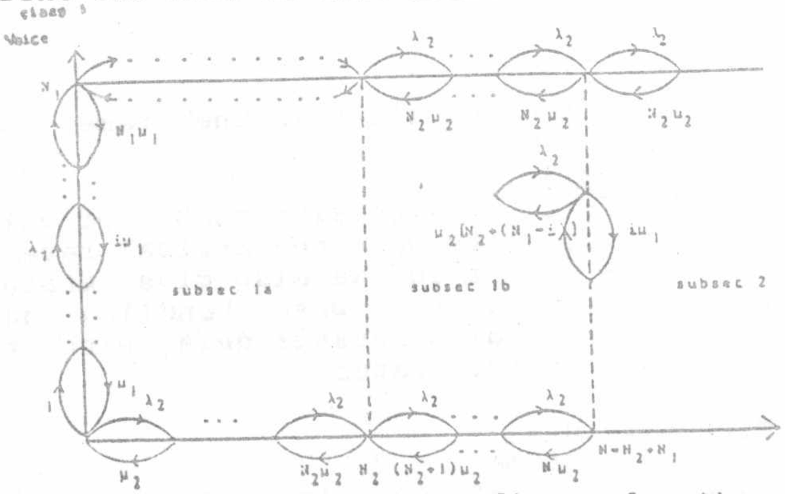


Fig (8) state transation digram for the system model the state transation diagram consists of two main parts 1,2 due to the moment genrating function, the balance equations of both parts are:

__Part 1__ For $0<i<N1, 0 < N2 + N1 - 1$

$$\lambda 2 + (1 - \delta\, iN1)\, \lambda 1 + i\,\mu 1 + \min\,(j,\ N2 + N1 - i)\,\mu 2\}\ Pij$$
$$= \lambda 2\ P\ i,\ j-1 + \lambda\ 1{*}p\ (i-1,j) + \min\,(j+1,\ N2 + N1-i){*}$$
$$\mu 2\ \ Pi,\ j+1 + (1 - \delta\, i\ N1)\ (i+1)\,\mu 1 + P(i + i),\ i \ \ldots \quad (5)$$

The min function indicate that the case of data traffic load i is not large enough.

__Part 2__ for $J > N2 + N1 \quad 0 < i < N1$

$$\lambda 2 + (1 - \delta\, iN1)\,\lambda 1 + i * \mu 1 + (N2 + N1 - i)\,\mu 2)\ pi,j$$
$$= \lambda 2\ \ Pi,j - 1 + \lambda 1\ P\ (i-1,j) + (N2 + N1 - i)\,\mu 2\ p(i+1),\ j$$
$$+ (1 - \delta\, iN1)\,\mu 1\ (i+1)\ P\ (i+1),\ j \quad \ldots\ldots\ldots\ldots\ldots\ldots\ (6)$$

the maximum available slots of data traffic = N2 + N1 -1

There are different methods to solve equations [5], [6] as in [3], [5]. by selecting the continous time approxmation method so, part (1.a) and part (1.b) for underload reigon where j < N2 and overload reigon where i > N2 respectively then the normalized waiting time for data messages are:

$$\mu_2 E(W) = \frac{1}{a}\left\{P(i)\right\} \frac{\rho_2^{N-i}}{\rho_2^{N-1}(N-i-\rho_2).(N-i-1)!\ \sum_{k=0}^{N-i-1} \rho_2^k/k!} \qquad (7)$$

Where $\rho 1$ is the voice utilization load in the system
$\rho 2$ is the data utilization load in the system
N1 is the number of available voice slots
N2 is the number of available data slots
i is the number of active voice sources.

$$a = N - \rho = N - (\rho_2 + \rho_1(1 - P_B))$$

$$P(i) = \frac{\rho^N/N!}{\sum_{i=0}^{N} \rho^i/i!}$$

N ..... is the total number of slots in the frame

__Performance evaluation:__

In this part, the effect of excessive number of voice sources on the blocking probability when the arrival rate of voice source is variable is evaluated ,we also give measure for the performance of the system when handling data sources, including the avarage data message delay with the variation of voice and data arrival rates.

__The system constraints:__

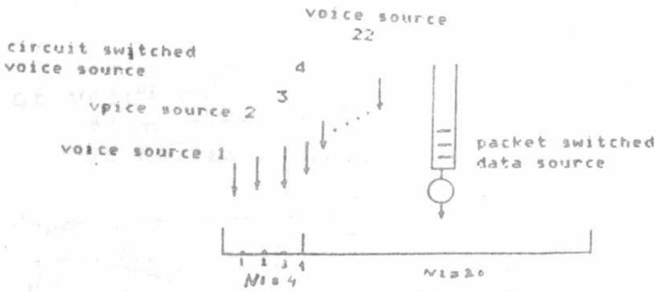
- Total number of slots per frame N = 24
- Number of incoming source M = 40 (22 voice, 18 data)

- T frame = frame width = 10 msec
- Serial basband per channel = 64 kb/sec
- Packet length = 240 bits
- avarage calls holding time $\frac{1}{\mu_1}$ = 100 sec
- avarage data message service time $\frac{1}{\mu_2}$ = 180 packet= 36msec.
- Voice utilization = ρ_1 with λ_1 (call arrival rate)
- data utilization = ρ_2 with λ_2 (message rate)
- Both voice and data under poisson arrival rate

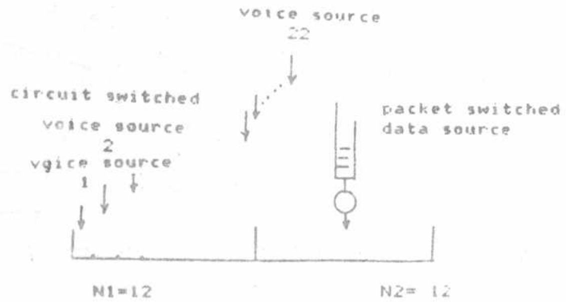
there are three different classes of voice and data slots available to analyize the Blocking probability of voice call and the waiting delay of data packet, as in table (2) and fig (7).

Table (2)

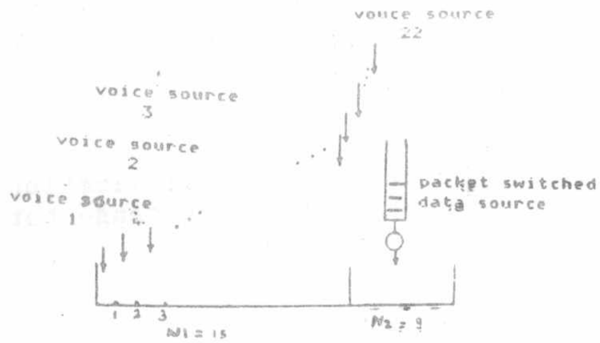
number of available slots	class 1	class 2	class 3
N1 (voice slots)	4	12	15
N2 (date slots)	20	12	9



(a) class (1)



(b) class (2)



(c) class (3)

Fig (7) different classes for available voice/ data slots.

The Blocking probability of voice calls:

The voice call Blocking probability is consider to measure the performance of voice sources in this system, the relation between blocking probability and voice utilization at different classes in table (2) is illustrated in fig (8-10) It is shown that the variation of voice utilization of the system as class (1) with 4 voice slots is available for 22 voice source is changed exponentialy due to the statistical variation of voice sources- assuming the sources from (1-4) is zero Blocking probability and number 5 is changed from 2.5705 E-06 for low load to 1.0804 E- 4 at high voice utilization.

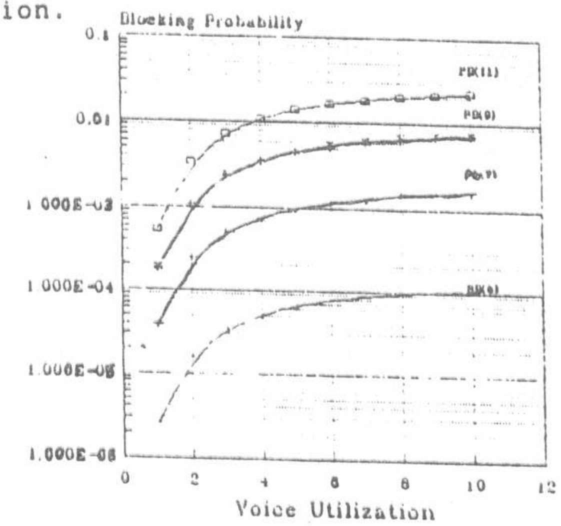


Fig (8) PB versus voice utilization at N1 = 4

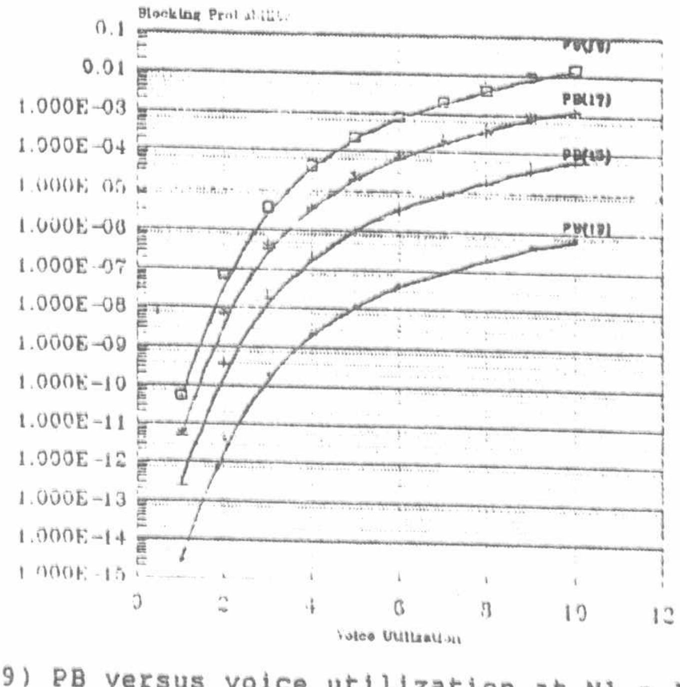


Fig (9) PB versus voice utilization at N1 = 12

So, Fig. (10) have the same response but the values of the results is close to resonable value with considering data sources are occupied the remaining 12 slots, Fig (10) are give an the best values of available voice slots.

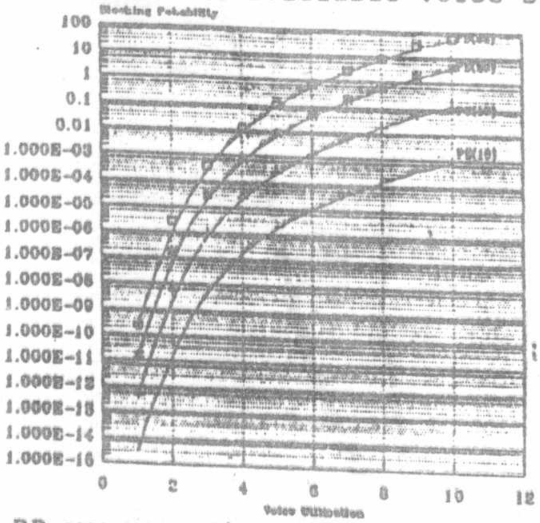


Fig (10) PB versus voice utilization at N1 = 15

<u>The data message waiting time:</u>

Due to equation (7) that shown the dependancy of voice utilization on the waiting delay. Assuming the classes in table (2), Then Fig (11-13) illustrate an estimation of the normalized waiting time versus data utilization with the variation of voice offered load.

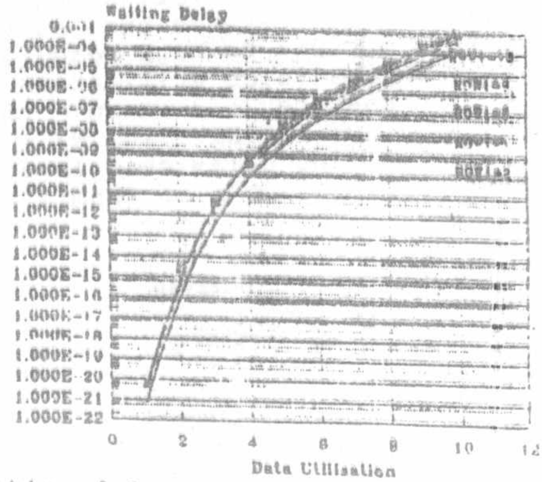


Fig (11) waiting delay versus data utilization at N1 = 4

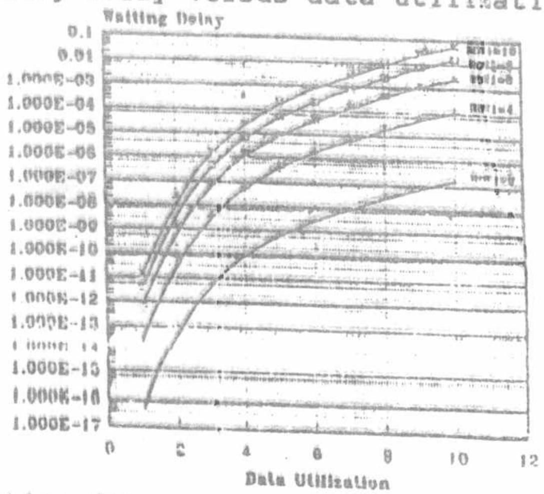


Fig (12) waiting delay versus data utilization at N1 = 12

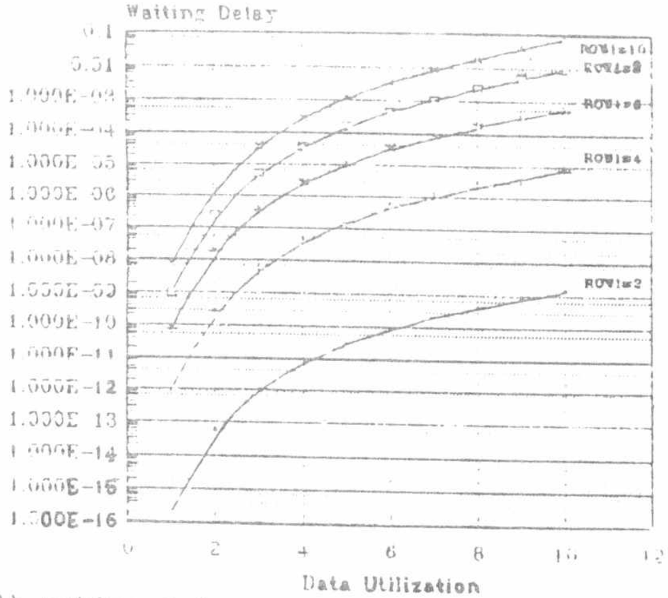


Fig (13) wating delay versus - data utilization at N1 = 15

There is an improvement in the value of waiting delay μ2E(W) When the number of available data slots increasing from (7.3 E-8ms to .759 ms) at N1 = 4, but at N1 = 12 the variation is (7.4E - 6 - .05 msec). this change are logically due to the number of available data slots is varied, with step change in voice utilizations, but to understand the dependancy of voice utilization on waiting delay we have to study the relation between them as the data utilization is changed from low load to high load.Fig (14-16) shows the waiting delay versus voice utilization so, the delay in the order of (.05 msec to 25.2 msec) the same variation but faster in the curves at N1 = 15, N2 = 9. at 2 changed from low to high the waiting delay changes from 8.2 E-6 ms at low voice load to 5.7 E - 3ms also from (.02 msec-32.4 msec) at high voice load.

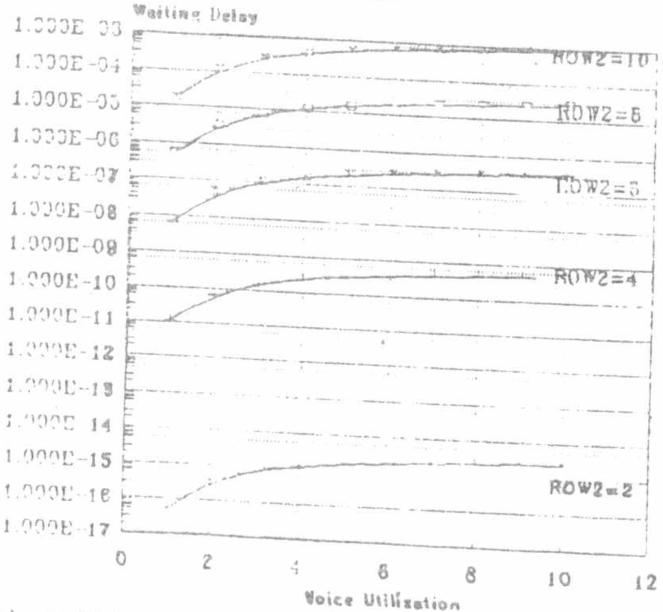


Fig (14) waiting delay versus voice utilization at N1 = 4

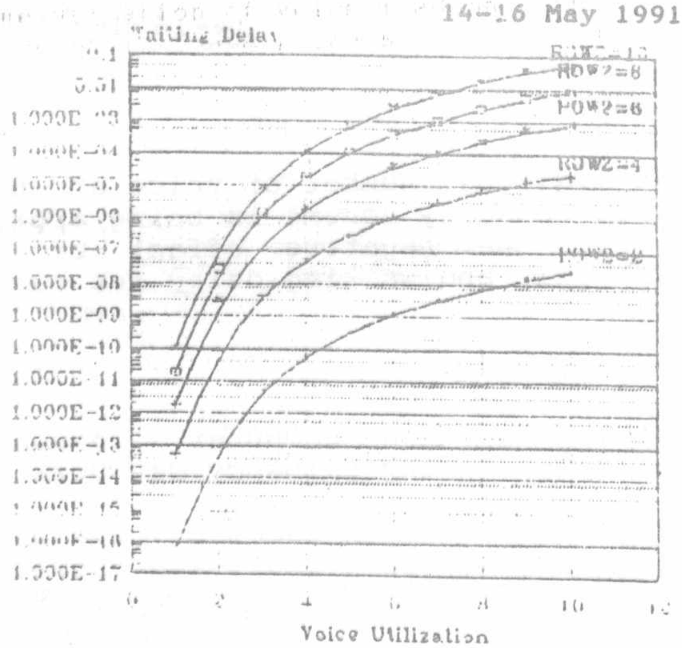


Fig (15) waiting delay versus voice utilization at N1 = 12

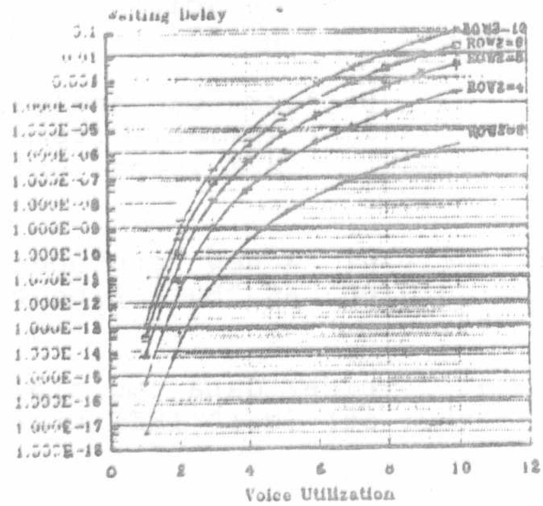


Fig (16) waiting delay versus voice utilization at N1 = 15

This results indicated that by increasing number of voice slots the waiting delay for the available data chanel is increased steadly with the effect of high data utilization every time the data utilization increased, the waiting delay is also increased with the dependancy of the dedicated voice slots.

Conclusions

A model of voice/ data integration using movable boundary scheme is presented. blocking probability of voice calls and waiting delay for data message is measured with the variaty of voice and data slots in S-TDM frame with considering A movable boundary frame structure .the results indicated that the blocking probability of voice call is increased with decreasing the available number of voice slots in the frame. The results also indecated that the waiting delay for data message is increased by decreasing the number of available slots with a variaty of data load.

The same variation of voice load have the same effect on waiting delay for data packets/ This results give an indication that the available number of voice and data slots in S-TDM with movable boundary scheme is being approximately be equal to avoid the negative effect of blocking probability of voice call and waiting delay for data packet.

Referances:

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