TO MY PARENTS

Packet-Switched Voice and its Application to Integrated Voice/Data Networks

by

Hassan HAGIRAHIM

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Department of Electrical Engineering
Imperial College of Science and Technology
(University of London)
London

ABSTRACT

This thesis is concerned with the performance evaluation of packetized voice systems (PVS) and the assessment of their performance capabilities when incorporated in integrated voice/data networks.

Two computer simulation approaches have been used to evaluate the performance characteristics of packetized voice systems. Results show that under certain conditions the voice packet arrival process can be approximated to a Poisson process.

Two criteria have been defined in order to ascertain the optimal packet length in a two node PVS. These criteria are the ratio of the actual number of channels to the nominal number of channels (which can be likened to the TASI advantage ratio of analogue time assignment systems) and the end-to-end delay. Results for a fully loaded system have been obtained and investigations have also been performed when the system operates under normal situations with varying traffic but to given average traffic intensities.

A PVS system where packets from channels can only wait one frame has been investigated and its performance characteristics have been evaluated. These results have been compared with the performance characteristics of a PVS system where packets can either be delayed indefinitely, or delayed by some fixed time. For the former PVS two packet organization techniques have been applied in an attempt to improve the TASI advantage performance of the system. This work has also been extended to assess the possibilities of multinode PVS.

Some computer simulation studies have been carried out to investigate the problems which exist in the integration of voice/data. These studies include the assessment of the circuit/packet Slotted-Envelope-Network (SENET) approach.

A new voice flow control mechanism has been suggested in order to improve the performance of the data traffic when the SENET system with movable boundary frame allocation strategy is employed.

Finally, two new multiplexing techniques have been investigated which can be used to handle the integration of both voice and data traffic for an environment which is entirely packet-switched. The first multiplexing scheme is based on employing the concept of SENET, while the second employs the first-come-first-served (FCFS) queueing basis. The performance of these integrated models have been evaluated using computer simulation approaches and the results obtained have been compared with results from analytical queueing models.

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CHAPTER 1

INTRODUCTION

1.1 Introduction

In general, there are three types of switching technology which can be used for establishing communication of digital traffic. These switching technologies are explained as follows:

(1) Circuit-switching

An end-to-end circuit is established for a pair of users. The end-to-end transmission facilities are dedicated to the users for the whole duration of the connection. The circuit is disconnected when either party goes off line, or when the caller goes off line.

(2) Message- switching

In this method a transmission channel is provided when the complete message has been assembled. The same channel is used for several input devices and delay may occur before the channel is available. The circuit is only assigned to a particular user for the means of transferring the message and is disconnected afterwards. A complete link may not exist between the caller and the called, there may be intermediate nodes where the message will be stored and later forwarded. A link can then be used by a number of users and can therefore be used efficiently. Delays are inevitable when more than one user requires the link at the same time. For digital traffic this concept assumes advanced electronic switches enabling the fast set up of a link in order to give short delays. If more than two switching nodes are involved a message may have to be stored at a node and later forwarded when a link is free.

(3) <u>Packet-switching</u>

This is a cross between circuit and message switching. Users send short messages across the network on a virtual circuit in the form of packets. The delays in the system are so short that the two end users feel that they have a dedicated connection between them. However the packets are being switched, delayed and forwarded as in a message system. The concept of packet-switching is based on the ability of modern digital facilities to transmit and switch short packets of information with very short delays. Long messages frequently have to be broken up into shorter packets. There are two protocol options which can be considered for this scheme and are explained below.

(a) Packet-virtual circuit (PVC): Here when a call is first set up a signalling message is propagated to the destination to establish a route for the call. The path set up involves setting appropriate pointers at tandem switching nodes which determine the outgoing link for each subsequent packet. Channel capacity is not reserved, neither is switch capacity dedicated. However packets follow the fixed path.

(b) Datagram: with this protocol, no initial route set up is employed. Each packet is transported to the destination independently of other packets of the same call. Packets can use different routes as appropriate and it is therefore possible for packets to get out of order.

Packet-switching and message-switching, as explained above, are suitable switching technologies for the efficient handling of bursty type traffic. The transmission of data between terminals and a computer is typically bursty.

When the traffic is bursty and contains long messages and long pauses message-switching is more efficient than packet-switching. For example, telemetry of data from weather transponders is a typical type of traffic which has the above characteristic. Packet-switching, however, is effective on bursty traffic which consists of short messages and short pauses; for example, the terminal to computer communication (or visa versa). If traffic is not bursty, then it should be handled by circuit-switching. For example,video and disk to disk transfers for computers.

Speech also has a bursty characteristic. The Brady's experimental analysis of speech, as reported in Ref [1], has shown that in a two way conversation each party, on average, is active not more than 50% of the time. This figure thus implies a peak to average ratio of 2:1. As a result of this, one could say that speech is a bursty traffic and thus could be transmitted advantageously either by message-switching, or packet-switching.

The time-assignment-speech-intepolation (TASI) [Ref.5] technique represents a type of message-switching on a multichannel link system. In this analogue, or digital, system the transmission is based on the interpolation of the active parts of the speech signals the talkspurts. This means that when a call is established, a link will be dedicated to the caller only during the active talkspurt periods of the calls. When the call is in a silent period (pause) the link may be taken away and assigned to another caller. When a talkspurt begins, a new link has to be assigned. This means that the initial syllable of the speech is often lost as it takes time to assign the link, or a link may not be available for a short time. The experience with this system has shown that if the channel capacity is large enough (i.e. around 40 circuits) then an interpolation factor, or a TASI advantage, of 2 can result. This means that the system can handle twice the number of subscribers as could have been handled by circuit-switching the same transmission system.

Packet-switching offers another advanced digital switching technology for the efficient handling of voice traffic. In this approach each talkspurt, after being detected, is divided into short messages which are called packets. The packets are transferred to a queue buffer and are serviced by the transmission link according to a first-come-first-served (FCFS) queueing protocol. One of the advantages of this switching technology is that a circuit, unlike in TASI, does not have to be reserved for the packet transmission of a particular burst. As was stated previously, in the TASI system the interpolation of talkspurts takes place by the disconnection of the non-active talkers and the insertion of the talkers which have recently become active.

Packet-switching is a more efficient switching system than the TASI system for handling voice traffic in a miltinode network. In a TASI system message signals have to go back and forth so that each end terminal keeps track of which circuit is assigned to each channel. Messages may have to be sent at the beginning and end of each talkspurt. In a multi-node system many signalling messages have to be sent through the network and these could quickly degrade the performance of the system. In a packet switched scheme the above difficulty would not be met. For example, if PVC packet-switching were to be used, fixed routes would be established at dial-up and the packets would follow the same route for the full duration of the conversation. In the datagram form no fixed routes are established and packets independently, and without the need of any signalling messages, are dynamically routed through the networks. As a result packet-switching is a more efficient method than TASI for handling bursty traffic in a network.

One of the aims of the project was to study the performance evaluation of packet-switching of voice traffic. The work described is mainly based on the study of a two node network model. The specific aims being to show how the performance of this system can be optimized with respect to end-to-end delay and how the optimum packet length can be evaluated. The study is further extended to investigate the relationship between the performance characteristics of the system and the fluctuations which are present due to the variations of arrivals and terminations of voice calls on an exchange.

The other main aim of this project was to study the application of the packet-switched voice in an integrated services digital network (ISDN). In the literature (i.e. Refs. [55]-[72]) there have been a great number of studies for integrating voice and data by using hybrid-switched technology. These schemes handle the voice traffic by circuit-switching and the data traffic by packetswitching. One of the disadvantages of this scheme is that transmission efficiency suffers as a direct consequence of the inefficiency in transmitting circuit-switched voice. Packet-switching of the voice traffic is one method of improving the efficiency.

In this thesis we shall describe two methods of multiplexing for the integration of voice and data so that both types of traffic are packet-switched. The performance of models based on these methods have been evaluated and will be compared with the results of the circuit/packet type of integration methods.

The results which are presented in this thesis were obtained mainly by the aid of computer simulation. Simulation models have been developed and are presented in full details.

1.2 Outline of the thesis

Chapter two of the thesis is dedicated to the review of the previous work on TASI, packetized speech communication system, and also integrated voice and data using different switching technologies. In section 2.6, a discussion is given on new investigations conducted in the area of packet-switched voice and integrated voice/data.

In chapter three the transmitter operation of the packetized voice-system (PVS) has been explained in detail (i.e. the packetization, queueing, and multiplexing operations of the system). Packets in this system experience stochastic queueing delays, for simplicity this system is referred to as the SQ-system. In sections 3.4 and 3.5 two different computer simulation models have been presented in order to evaluate the queueing performance of the SQ-system. In section 3.6, the results have been compared with the results of well known analytical queueing models. Some further investigations have also been undertaken in order to investigate when the packet arrival process can be approximated to a Poisson process.

Chapter four of the thesis deals with the end-to-end performance evaluation of the PVS. In section 4.2, the receiving operation of the system has been discussed. In section 4.3, two criteria have been presented and with the aid of computer simulations the optimal packet length for these two criteria have been obtained. In section 4.4 the effect of telephone traffic intensity upon the performance characteristics of the PVS have been investigated. A new criteria has been introduced from which the TASI advantage of the PVS becomes related to the blocking probability.

In chapter five a PVS system has been investigated where packets from the channels are only permitted to wait one frame duration, if they have not been processed by this time they are lost. Packets in this system do not experience variable or stochastic queueing delay. For simplicity this system is referred to as a non- stochastic queueing (NSQ) system. In section 5.2, the operation of this system has been explained. In section 5.3, the performance characteristics of the system have been evaluated by using a simple mathematical model and also simulation. In section 5.4, the performance characteristics of this system have been compared with the performance characteristics of the SQ-system. In section 5.5 two packet organization techniques have been applied to the NSQ system in order to increase the TASI advantage of the system. In section 5.6 an integrated voice/data model has been investigated which handles the voice traffic by the NSQ operation and always gives the priority of the packet transmission to the voice traffic. All the voice channels in this integrated scheme are assumed to be fully connected. In section 5.7 a new multinode PVS network model has been investigated the operation of which depends upon the NSQ packetized scheme. The performance characteristics of the network have been evaluated by using a simple mathematical model. In section 5.8, the effect of telephone traffic loading on the NSQ-system has been investigated. A mathematical formula has been derived to relate the TASI advantage performance of the system to the blocking probability.

Chapter 6 deals with the integration of voice/data using the circuit/packet slotted envelope network (SENET) approach [55]. In section 6.1, the operation of the SENET-system with fixed and movable boundary frame managements have been explained. In sections 6.3 to 6.6, the performance characteristics of the integrated SENET-systems have been obtained from computer simulations and related analytical queueing models. In section 6.7, an adaptive voice flow control technique has been suggested and incorporated into the SENET system in order to improve the data traffic performance when using the movable boundary operational techniques.

Chapter seven is concerned with the integration of voice/data in an environment which is entirely packet-switched. This chapter is divided into two main parts. In the first part, a new multiplexing model has been proposed and investigated which performs the integration of the two traffics; this is achieved by employing the concept of SENET multiplexing on both forms of traffic. The operation of this integrated system, has been explained in section 7.2. In section 7.3, the performance parameters of a totally packet-switched integrated SENET system have been evaluated when the packet-switched voice traffic is in the form of an NSQ system. In section 7.4, the operation of the packet-switched voice traffic is changed from NSQ to SQ and the performance parameters of the integrated SENET system have been evaluated. In section 7.5, in order to improve the performance of the data traffic handling, a boundary flow control mechanism has been applied to the SENET system with NSQ-packetized voice operation.

The second part of this chapter is devoted to the investigation of an integrated packet-switched voice and data scheme using a first-come-first-served (FCFS) multiplexing method. In section 7.6, the operation of the integrated system has been explained. In section 7.7, an analytical model has been presented which can be used to evaluate the performance parameters of a particular type of this integrated model. In section 7.8, a computer simulation model has been presented from which performance parameters for the general integrated system have been evaluated. Results have been obtained for the two cases where the voice packets are either discarded at the transmitter, or the receiver. In section 7.9 the two integrated systems (i.e. SENET and FCFS) are compared.

The summary and important conclusions, together with suggestions of several topics for future investigation have been presented in chapter eight.

CHAPTER 2

A REVIEW OF WORK ON TASI, PACKET-SWITCHED VOICE, AND INTEGRATED VOICE/DATA

2.1 Introduction

The aim in this chapter is to review previous work on the time-assignment-speech-interpolation (TASI) system, packetized speech communication systems, and also integrated voice and data systems which use different switching techniques.

Section 2.2 gives a brief review of speech characteristics. The review of the work on the TASI-system is given in section (2.3), a review of the packet-switching of voice communication systems is given in section 2.4, and review of integrated voice/data can be found in section 2.5. In section 2.6, new approaches in packet-switched voice and integrated voice/data have been suggested these, as yet are believed not to have been investigated by other researchers.

2.2 Speech analysis

Speech has been analysed experimentally by Brady in references [1] - [3]. In Ref [1] the aim of the experiments was to investigate the properties of speech which are concerned with determining the presence of speech on a telephone circuit. A speech detector was constructed to yield an output of spurts and gaps, corresponding to the presence or absence of speech energy above a threshold. A computer simulation programme was then used to correct this pattern for spurious noise operation and for gaps due to stop consonants, eventually yielding a pattern of talkspurts and pauses. Data reported in this reference includes the cummulative distribution functions of the spurts and gaps resulting from the detector as well as the distributions of talkspurts and pauses from the computer program.

In Ref [2] more extensive analysis has been undertaken. By using the same experimental procedures, distributions were obtained for ten events, namely; talkspurts, pauses, double talking, mutual silence, alternation silences, pauses in isolation, solidary talkspurts, interruptions, speech after interruptions, and speech before interruption. The same author in Ref [3] introduced a six-state Markovian model for generating the on-off speech pattern. The validity of the model was tested by comparing the results obtained from the computer simulation model with the results from real speech, as given in Refs [1,2]. Comparisons were also made for the cumulative distribution functions of the ten events mentioned above. The model yielded good fits to all events except 'speech before interruption". In Ref [3] it was also shown that the exponential distribution is a satisfactory empirical fit to talkspurts, but not to pauses.

Norwin and Murphy [4] also observed talkspurts and pauses in oscillograph recordings of telephone conversations made on a special circuit between New York and Chicago. The circuit had a round-trip delay of 600 m sec and was equipped with echo suppressors. Detailed observations were made from 4400 feet of graph paper, representing 51 calls with a total duration exceeding 13000 seconds. Data reported in this paper included the distributions of talkspurts, resumption times, and response times. However, the definition of the talkspurt in this paper was significantly different from that of Brady. As the talkspurts measured by Norwine and Murphy included pauses of short duration. This resulted in the Norwine and Murphy talkspurts being much longer than Brady's

2.3 Time assignment speech interpolation (TASI)

The experimental works of Norwine of Murphy [4], and Brady [1] - [3] both had a common conclusion. This is to say that the conversational speech is a bursty traffic because in a normal two-way conversation each talker uses the circuit for less than half of the time available. Bullington and Fraser in Ref [5] showed that the number of telephone circuits carried by a submarine cable could be doubled by using the pauses in speech to interpolate additional conversations. This form of switching technology is named as TASI, an abbreviation for Time Assignment Speech Interpolation. In this system when a caller whose call has been connected through the system begins to speak, the speech is detected and, if possible, a channel on the multiplexed cable is assigned to the talker for the duration of his talkspurt. Switching times are fast so the listner does not notice the switching. When the talkspurt is finished, the channel will be assigned to another talker. If no channel is available, the beginning of the talkspurt is "cut out" or "frozen out" until a channel becomes available. Listening tests by the same authors have shown that for satisfactory speech perception, the fraction of speech lost, or the cut-out fraction, should be kept below a threshold of 0.5%. The "TASI advantage" was defined as n/c, the ratio of the number of callers supported when the freezeout fraction is just maintained to the nominal number of circuits. The maximum potential TASI advantage was found to be 1/q where q is the fraction of time a typical talker is actually in talkspurt. The results reported in this paper are based on using the mathematical analysis of Cravis [6]. In the analysis, two specific formulas have been derived for the cut–out fraction. These were based on the assumed forms for the probability density function of talkspurt duration, being either constant or an exponentially distributed duration. Winestein in Ref [7], however, showed that the cut-out fraction does not depend upon the details of talkspurt duration distribution, but can be expressed in terms of the three system parameters:

(1) the number of speech sources,

(2) the number of channels,

(3) the probably that a source is issuing a talkspurt at a random time.

A new expression for the cut-out fraction was developed by Winestein and was presented in the same paper.

The concept of TASI, when used for digital transmission, is termed digital speech interpolation (DSI). Digital TASI has a number of advantages over analog TASI. For example, digital voice detectors perform better than their analogue counterparts. In addition, more precise switching of digital speech samples among the PCM channel slots of the time-division-multiplexed (TDM) frame is inherent to digital techniques. An all-digital approach is also compatible with the use of a digital channel assignment processor used for the recording of the channel assignment at any instant, and for the communication of this information to a companion digital channel assignment processor at the receiver. Campanella in Ref [8] investigated the performance of digital TASI in terms of the criterion that the percentage of speech-clips longer than 50 ms should be less than 2 percent. The analyses presented for the evaluation of this performance criteria are similar to those given by Cravis [6], and Bullington and Fraser [5].

An additional advantage of digital TASI is the ability to expand the number of transmission channels to avoid less freeze-out during instants of overload by reapproptioning the size of the digital transmission time slots. This technique, known as channel augmentation by bit reduction, can be invoked during overload conditions to avoid excessive competitive speech clipping. Campanella in the same paper (i.e. Ref [8]) investigated the bit reduction of 8-bit PCM words to 7-bit PCM words during the overload periods. He shows that when this bit reduction technique is used the probability of the clip duration exceeding 50 ms is significantly reduced at the cost of a 6-dB increase in quantization noise on all channels during the overload period.

Campanella in the same paper (i.e. Ref [8]) introduced another type of digital TASI technique. This technique is know as SPEC (speech prediction encoded communications) and it differs significantly from the digital TASI. One of its principal merits is total avoidance of freezeout. In this scheme, the PCM samples derived during each sample period from all the incoming channels are compared with the samples previously sent to the receiver they also stored in a memory at the transmitter. Any that differ by an amount equal to or less than some given number of quantizing steps, called the aperture, are discarded and are not sent to the receiver. These are referred to as "predictable" samples. The remaining "unpredictable" samples are transmitted to the receiver and replace the values formerly stored in memories at both the transmitter and receiver. At the receiver the unpredictable samples received in the SPEC frame replace previously stored samples in the receiver's memory. The most recent frame thus contains new samples on the channels that have been updated by the most recent SPEC frame and repetitions of the samples that have not been updated.

Some form of information has to accompany the frame to indicate which channels are renewed and which are to repeat previous information.

The performance of the above system has been obtained for conditions up to where the predictor removed 25 percent of the active PCM samples. This means that on average one sample out of four is predicted during a voice spurt. It was shown that this resulted in a signal-to-distortion (S/D) ratio decrease of only 0.5 dB. An analytical model was presented to evaluate the size of the aperture. This scheme results in a better TASI advantage than achievable by TASI alone.

Another type of digital TASI system which avoids freezeout has been investigated by McPherson et al [9]. The system is a combination of Adaptive Differential PCM (ADPCM) and time Assignment Speech Interpolation (TASI). This scheme is therefore called ADPCM/TASI.

The scheme employs a speech detector which, like TASI, provides a path from the transmitter to the receiver only during the time when speech is in a talkspurt mode. When a talkspurt initiates, the PCM samples are encoded into ADPCM. ADPCM transmits the difference between a sample value and a prediction of that value based on previously transmitted samples and so requires fewer bits per sample than PCM for the equivalent speech quality. Combined with the TASI and ADPCM is an overflow strategy which reduces the number of bits used to encode each sample when the system is heavily loaded. As the ADPCM/TASI system becomes heavily loaded (many users talking at the same time), the number of active channels is used to compute the number of bits L available for each speech channel. Knowing L, the ADPCM unit encodes the standard 8-bit PCM samples into L-bits ADPCM. All active channels share the available bits equally and therefore no freezeout will take place. This system was simulated and the signal-to-noise ratios were computed and audio tapes were subjectively evaluated. ADPCM/TASI was judged to be a good compromise between performance and complexity. In Ref [10], Agrawal et al described the design of the ADPCM/TASI by using a bit slice microprocessor structure. The system was considered to compress the output of two T1 24-channel PCM carrier terminals into a single T1 carrier line.

The concept of TASI, as described above requires a speech detector for detecting the talkspurts in real-time conversational speech. Miedema and Schachtman [11] designed such a speech detector for the practical analogue TASI scheme. This speech detector employs an operating time of 5 ms in order to eliminate noise, and a 240 ms hangover. This means that once the speech detector is operated, the circuit cannot be released until the hangover time has elapsed. The disadvantage of this speech detector is that the hangover time has a long duration and it lengthens the talkspurt. Yatsuzuka in Ref [12] introduces a highly sensitive speech detector with a shorter hangover time of 32 ms. This speech detector detects the presence of the speech signal by means of analyzing the short time energies, the zero crossing rate, and sign bit sequences of the input signal. Yatsuzuka in ref [13] used this speech detector to investigate the ADPCM/TASI concept, the operation of which is very similar to that investigated by the authors of Refs [9,10].

Another type of DSI system has been investigated and implemented by Nakhla and Black [14], [15]. Their scheme employed a 512 kbit buffer storage facility for storing up to 8.2 seconds of speech. This buffering facility thus permits the speech bursts to be stored rather than to be frozen out during the congestion periods. An upper limit of 1 second storage for any speech burst is placed by the software,this to ensure that the buffer memory capacity is allocated fairly. After one second of storage, a speech burst then is assumed to be clipped. One cause of speech impairment in this scheme is the variable delay. Subjective tests confirm that the tolerance of the ear to variable delays, or gap modulation, introduced by this system should not exceed 300 ms for less than 10 percent of time. A mathematical model based on queueing theory was developed for evaluating the statistics of delay, gap modulation, and freezeout fractions. The system was considered with three different configurations:

(1) 9 off-hook talkers over 6 channels (i.e. 9/6),

(2) 20 off-hook talkers over 11 channels (i.e. 20/11)

(3) 48 off-hook talkers over 24 channels (i.e. 48/24).

The performance of the system was obtained so that the gap modulation did not to exceed 300 ms more than 10 percent of the time. The results obtained from the analysis showed that the freezeout probability in each of the above cases was so low that it caused imperceptible speech impairments. Thus it was concluded that the dominant effect in the system is the variable delay introduced by the speech buffer.

In all the analyses and simulations presented in the above work on TASI or DSI [5] - [15], the number of talkers was fixed and was not allowed to fluctuate. Of course, in reality this assumption is not true for a real time communication network. The probability that all of the trunks will be busy depends upon the telephone traffic coming into the system. Kou et al [16] presented an analytical expression for evaluating the cut-out fractions as a function of the telephone traffic offered to the system. The expression derived is based on combining the Weinstein cut-out fraction [7] for the fully connected TASI and the Erlang loss formula. It was shown that the value of freezeout fractions evaluated for this case is considerably less than the value computed from the all channels busy assumption. Fischer in Ref [17] also investigated the effect of telephone traffic loading on the DSI system with a speech store facility. In this form, each talker provided with a buffer storage at the transmitting terminal and the talkspurts are buffered when there are no free channels. A mathematical model was developed and parameters such as the average number of talkspurts in the system, the average delay of a talkspurt in the buffer, and the complete delay distribution of a talkspurt have been evaluated.

2.4 Packetized-voice system (PVS)

With the introduction of packet-switched communication systems, interest has been developed in the TASI implication of packetized speech. Forgie [19] considered the network implication of packetized speech, i.e., the effects packetized speech has on overall voice quality, acceptability, and communicability when the voice conversations are transmitted in a packetswitched network. Packetization of the active parts of the voice conversation (the talkspurts) is in effect a switching concept where the philosophy of TASI can be implemented in the digital network, the packets generated by a talkspurt may be buffered when a channel is not available. Weinstein and Hofstetter conducted a simulation study [20] that considered the tradeoff between packet delay and the TASI advantage in that environment. They showed that the packetized system offers substantial improvements to the TASI advantage (number of talkers to nominal number of channels) even when the number of talkers are small, provided one allows the voice packets to experience some delay. The basic analysis tool used in [20] was a simulation model which was developed by Weinstein in Ref [7]. This simulation model has the form of a two dimensional Markov chain. One of the dimensions of this model is dedicated to the transition of talkspurt to silence and vice versa. These transitions are assumed to take place according to Poisson processes. The reasons for using such transit-time processes are based on the assumptions that the talkspurts and silent intervals have a negative exponential distribution . In the same paper (i.e. Ref [7]) it was shown that this approximation becomes valid if the number of talkers exceeds 25, it is less valid if the number of talkers is below 10. The other dimension of the simulation model corresponds to the packet queueing process. This two-dimensional simulation model proved to be attractive due to its efficient use of the computer simulation time.

Clarke and Turner [21] introduced another type of computer simulation for a packetized voice system. This simulation model utilized a speech activity model which employs the speech statistics as given by Norwine and Murphy [4]. In this simulation model a detailed structure of the speech and packetization operation has been performed in the simulation, it is therefore more costly upon computer simulation time than Weinstein's simulation model. This speech model, however, has been used in Ref [21] in order to evaluate the performance of a two node and different types of connected three node packetized voice systems.

2.4.1 Related queueing models for PVS

Much work has been conducted on conventional queueing systems and is discussed in many books [22-24]. However, all these queueing models are for the continuous arrival process. Minoli [26] introduced a discrete queueing model for the PVS. In this form, it was assumed that the packet arrival from n off hook speech terminals to be a discrete random process being distributed to a geometrical distribution. The length of the packets are assumed to be of fixed length, and the length of the queue buffer was assumed to be infinite. This queueing model was solved by Minoli, and an approximate expression derived for the average packet waiting time. Meisling [25] also solved this queueing model but he assumed the time to transmit a packet over the link to be an integer multiple of the interarrival period of all the packets onto the multiplexer switch, this is thus a limited scheme. The analysis of the Minoli model can be viewed as a completion to the analysis of Meisling for this type of queueing model.

In a later paper [27], Minoli used the expression of the average packet waiting time derived in [26] to derive an expression for optimizing the performance of a PVS. This optimization technique was based on the evaluation of the packet length which minimizes the end-to-end packet transmission delay. He showed that short packets (typically 300 bits) are best for low bit rate coding techniques (i.e. LPC). Longer packets (typically 700 bits) are shown to be best for high bit rate systems (i.e. conventional PCM). In a later paper [28], the same author has extended the analysis for finite buffer size. He examined the effects of packet size and transient behaviour, e.g. sudden instantaneous high link utilization. It was found that transient delays can take a long time to diminish, especially if infinite queue lengths are permitted. A better response to transient overhead, with some packet loss, is achieved with finite length buffering.

Kim in Ref [29], using a computer simulation technique, showed that when more than ten off-hook sources are multiplexed, the packet arrival process resembles that of a Poisson process. This conclusion, however, was achieved by comparing the results of average packet waiting times obtained from a computer simulation and an M/D/1 queueing model. However, in his comparisons he did look at average packet waiting time delays in excess of 1.66 ms.

Janakiraman et al [30] evaluated the queueing performance of the PVS by taking into account the statistical dependency of each source. In this form, the packet arrival from each off-hook terminal was modelled by a two-state Markov chain. A multi-server queueing model was used and the queueing performance of the PVS was evaluated in terms of average packet waiting time and queue buffer overflow probability. In a later paper by the same authors [31], the validity of this analytical model was investigated. This scheme was performed by obtaining the characteristics of delay versus TASI advantage from the analytical model and comparing them to the simulation results of Weinstein and Hofstetter [20]. The comparisons of the two sets of results have shown that higher TASI advantages resulted from the analytical queueing model of [30]. The reason for this lack of agreement, as stated in Ref [31], is due to higher link utilization indicted by the multiserver analytical queueing model of [30]. A multiserver system operates at the higher link utilization rate for the same mean queueing time as does its equivalent single server system. Since the link utilization is directly proportional to the TASI advantage, a higher link utilization implies a higher TASI advantage.

Pack [32] investigated the performance of a queueing system which consists of a multiplexer and a finite queue buffer. The packet arrival from the multiplexer to the queue was assumed to be a Poisson process and the service time was assumed to be constant, however the service was carried out in batches. Pack introduced an analytical queueing approach for the evaluation of the buffer overflow probability.

Dor [33], has derived expressions and algorithms for evaluating the buffer overflow probability in an M/D/1 queueing model.

Other modellings and performance evaluations involving queueing analysis for different communication networks have been reviewed with Kobayashi and Konheim [34]

2.4.2 Receiver buffering

The need for a packet reassembly strategy at the receiver is due to the stochastic behaviour of the network queueing delays. This effect causes the packets to be received with variable and incorrect gaps, so degrading the output of the speech. Barberis and Pazzglia [35] introduced three different reassembly strategies which can be adopted at the packet voice receiver (PVR) buffer. The simplest of the above approaches is the null timing information (NTI) approach. In this strategy the first packet of a talkspurt is delayed at the receiver by an amount independent of the delay the packet has already experienced. In the second approach the network transit time of the packets are estimated and the first packet of a talkspurt is delayed if the estimated transit time is less than a given threshold. This approach is called the incomplete timing information (ITI) strategy. The third approach is to include a few bits of information in the packet header indicating the time at which it was sent. Each packet can then be buffered until its total delay is equal to a given delay threshold and then played out. This approach is called the complete timing information (CTI) strategy.

In the same paper expressions were derived for the probability distribution functions of the overall packet transmission delay in each of the above reassembly approaches. These expressions were then used to evaluate the optimum design of the packet voice receiver buffer in each of the above reassembly approaches. This goal was achieved by evaluating a buffer length at the receiver which minimizes the overall packet delay variances, however, the mean overall delay had to be maintained below a given prescribed value. The analysis carried out in Barberis et al's paper was based on two important assumptions: (1) the pdf of the packet waiting time at the transmitter should be known; and (2), the queueing delay experienced by packets belonging to a single talkspurt should be independently and identically distributed. This latter assumption is only true for a multi-hop network with dynamic routing (or datagram) policy.

In a later paper [36], Barberis extends the analysis of [35] and developed an expression for the pdf of the packet waiting time at the PVR. By using this tool, an expression was developed to evaluate the packet-loss probability at the PVR for each of the three reassembly strategies.

In Ref [37], Weinstein and Forgie reported the experiments which were conducted with a packet speech system on 10 hop-path through the ARPA network. The packet length in this experiment was set at 134 ms fixed length. The CTI reassembly strategy was used for the packet reconstitution. It was shown that a delay of 500 ms at the receiver is required in order to have less than 1% packet losses in the system.

Further details on the nature of speech signals and on speech coding for digital networks are given by Gold [38]. This paper describes some speech related delay experiments which were carried out employing the ARPA networks. It was found that strict error and flow control procedures reduced the throughput, so the end-to-end acknowledgement procedure was suppressed, thus giving better results. Four different packet reassembly strategies are described, the concepts of each are based on using the CTI strategy. In the first reassembly scheme the delay applied at the receiver was assumed to be large so that all the packets could be reassembled without any gaps. In the second scheme the receiving delay was not as large as in the first scheme, the packets which were not received by their play-out time were discarded. The third reassembly strategy was similar to the second with the difference that packets were not discarded. This method implies that delayed packets delay subsequent packets and the delay gradually increases during a talkspurt. However, during the silence periods the receiving delay is decreased to its nominal value. The fourth reassembly strategy was similar to the third scheme with the difference that the reduction of the delay at the receiver was performed according to the instantaneous network loading.

Naylor and Kleinrock [39] have reported an adaptive destination buffering scheme for the NTI reassembly strategy. This scheme uses delay information, measured for the previous talkspurt, in order to compute destination buffering information.

Montgomery [40] suggests 5 different methods for packet voice reassembly and synchronization. These techniques have been discussed as follows:

1) <u>Blind delay</u>.

This being exactly the same as the NTI reassembly strategy.

2) <u>Roundtrip measurement</u>

This scheme is based on having synchronized clocks at the transmitters and receivers in order to measure the roundtrip delays in the communication paths between the transmitter and the receiver, and assumes that the delay to be equally distributed between both directions. The measurement is made by sending a packet containing a local clock reading from the transmitter. When the packet arrives at the PVR, it is immediately sent back to the transmitter. When it arrives back at the transmitter, the transmitter calculates the roundtrip delay by subtracting the clock value in the packet from the current time. The roundtrip delay is then sent to the PVR, and all subsequent packets sent by the transmitter to the PVR contain timing information relative to the first packet sent by the transmitter.

3) <u>Absolute timing</u>

This technique is exactly the same as the CTI strategy. This means that the transmitter and receiver are both synchronized to an absolute time reference.

4) <u>Added variable delay</u>

The operation of this scheme is the same as the CTI strategy, only with the difference that the network is not synchronized. There is only one clock at the transmitter and it inserts the delay-stamp of the packets when they exit from the transmitter node.

5) Adaptive strategies

This technique can be applied to all of the above four strategies and it results in an improvement in the delay measurements. This is based on changing the target play-out time (or the delay applied on the first packet of the talkspurts) during the silent periods.

2.4.3 Protocol considerations

Some comparisons between circuit and packet switched voice systems are made by Caviello in Ref [41,42]. He outlines some general network and performance definitions and identifies sources of delays in networks. Some results are given for a particular set of packet network parameters and these are compared with circuit switching. Some alternative packet header organisation schemes are described.

1) Current base design

This protocol represents the current design used for pure packet switching (e.g., datagram mode). Each packet in this mode is individually and independently transmitted and routed in the network. The packet header in this form contains flags, addresses, and cyclic redundancy check (CRC) bits.

2) Voice oriented design

In this approach, the CRC and the sending address are eliminated. This results in the total packet size header being reduced by one half.

3) <u>Fixed route/abbreviated header</u>

At the dial up, a route is initially established through the network and all the subsequent packets during that call traverse the fixed route. Forgie and Nemeth [43] used this type of protocol for investigating an integrated voice/data system. They referred to this strategy as the packet virtual circuit (PVC). This protocol is similar to that used in the x.25 standard protocol for data transmission [44,45].

4) <u>Concatenated header</u>

Several packets from different sources are put into a "super-packet", with activity indicators in the header to show which talkers have contributed to the super-packet.

2.4.4 Subjective effect of delay and packet losses

Klemmer and Riesz in [46], [47] conducted a number of statistical tests to evaluate the subjective effect of transmission delay upon telephone communications. These tests have shown that the participant's difficulty in carrying out the conversation is inversely proportional to the end-to-end (ETE) delay. However, it has been shown that it is hard to determine universal boundaries between satisfactory delay and unsatisfactory (too large) delay. However, these statistical tests have shown that very few people observe any quality degradation when the ETE delay is less than 300 ms. For delays between 0.3 and 1.5 seconds, the users notice the difference and change their speaking behaviour; longer delays are noticed between talkers and cause overlapping talk. When faced with longer delays,most talkers had difficulties in conducting a normal conversation.

Jayant and Christensen [48] discuss packet losses and coders (including variable rate and embedded techniques). They describe two interpolation methods for use when packets are lost (i.e. through error or discarding). The first is to substitute a packet full of zeros at the receiver. The second is to have odd and even samples in successive pairs of packets (i.e. all the odd samples in the first packet of the pair, and the even samples in the second). In this scheme, if one packet is lost, an approximation to its samples can be obtained by interpolation of the samples of the other packet of the pair. If both packets of a pair are lost, then interpolation by zero is used. This method, with its pair of packets, does lead to longer delays in packet pair generation and reconstitution. As with zero interpolation the odd-even interpolation method can only be used with waveform coding techniques, e.g. conventional PCM. The signal-to-noise ratio was measured for the two coding methods and different packet lengths for each of the two interpolation schemes and subjective tests were carried out. Odd-even interpolation was found to give better results at up to 5% packet loss.

Musser et al [49] conducted a set of subjective tests to evaluate the effect of packet losses in a local area network with the CSMA/CD protocol. They used two methods for replacing the packet losses at the receiver; (1) the zero stuffing technique as used by Jayant and Christensen [47], and (2) the recirculation technique, where the complete previous packet is replayed in order to replace the missing packet at the receiver. The examinations have shown that upto 1% dropped packets can be tolerated with the first method while with the second method the level of loss can be raised upto 2%.

Forgie [19] using formal testing with 16 kb/s speech recommended that packets should be less than 50 ms in length and lost packets should not exceed 1 percent in order to avoid significant loss of user acceptability.

Tucker and Flood [50] used another type of approach for evaluating the maximum percentage of packet losses. Their approach consist of a real time simulation of 21 talkers in conversation. It was assumed that 20 of the talkers come from the collected simulated data (i.e. non real) and the 21st was the real talker in real life conversation. The packet length chosen was assumed to be 8 ms and the transmitter buffer was assumed to be of finite length to ensure that the packet losses occur only at the transmitter. Their examinations have shown that even packet losses of order of 10% could be tolerated by the listners and the speech quality was considered fair by 90% of the listners.

2.4.5 Performance evaluation of PVS

Suda et al [51] evaluated the performance of a fully connected PVS system by using a computer simulation technique. In their approach it was assumed that the transmitter buffer was infinite. The optimal packet length was found according by Minoli's criteria [27]. At the receiver, the optimal buffer length, or the optimal delay applied to the packets of talkspurts, was found so that the packet losses were just equal to 1%. They also compared the silence fluctuations in the three strategies of Barberis and Pazzglia [35]. It was found that the CTI and ITI was superior to NTI.

Weinstein [7] suggested that the formula which he derived for the freezeout fractions of the conversional TASI, can also be applied to a PVS where packets in the transmitter buffer can only wait one frame. The present author, in Ref [52(a)] using a computer simulation technique, showed that the above is true. By introducing a de-multiplexing stage in the model it was shown that this type of PVS system can easily be incorporated into a multinode PVS. It was also shown that with a 7 node network, the TASI advantage of the system could be as high as 1.42. This work is also reported in chapter 5 of this thesis.

In Ref[52(b)] the present author investigated the performance of a two node PVS by using a computer simulation technique. The system was assumed to employ CTI reassembly strategy at the receiver. Two characteristics were used from which the optimal packet lengths were obtained. These characteristics were (1) the TASI advantage versus packet length and (2) the ETE delay versus packet length for the condition where the packet losses were set at 1%. The investigations were further extended to evaluate the effect of telephone traffic loading on the system. This work is also reported in chapter 4 of this thesis.

Tucker and Flood [50] optimized the performance of a one-hop PVS by minimizing the difficulty which occurs due to packet losses and the ETE delay. They assumed that the probability of conversation difficulty due to the delay rises linearly between 0 and 1200 ms. A subjective test was also performed and an expression was found for the probability of difficulty due to packet losses. Assuming the further assumption that the effect of packet losses and delay to be independent, an expression was developed for the probability of difficulty composed of delay and percentage of packet losses. This expression was used to optimize the performance of the system.

Liu et al in Ref [53] by extending the simulation model of Clarke and Turner [21], investigated the characteristics of packet losses versus packet length for a two node PVS with a maximum of 250 ms queueing delay. The extension of the simulation model was based on introducing the telephone traffic loading on the system. This scheme was performed by generating on-off patterns of conversations on each of the channels. The length of the duration of each conversation was assumed to be constant and to be equal to 2 minutes. The simulation results in this paper show that for packet lengths greater than 40 samples and having mean packet losses less than 0.5%, the system can handle 30 subscribers with 64 kb/s encoder outputs over a transmission link equivalent to 12 channel PCM system (i.e. a TASI advantage of 2.5).

2.5 Integration of voice/data

Harrington [54] investigated different forms of integrating voice and data by using circuit-switching concepts. These are as follows:

1) <u>Traditional circuit switching</u>.

In this form, a complete end-to-end circuit is established for each pair of voice and data users and is dedicated for the full duration of a given call.

2) Fast circuit switching.

In this system, voice and bulk data applications would use the traditional circuit switching method, while interactive data would be fast-circuit switched. This means that for the interactive data, a circuit would be established for every message and would be held for the duration of the transaction, and then would be disconnected, (e.g. Datagrams)

3) Enhanced circuit switching.

In this method, the voice traffic would use the traditional circuit switching supplemented with TASI, while the interactive data traffic would be handled with adaptive data multiplexing (ADM) techniques. The operation of ADM is the same as TASI, but with the difference that the freezeout fractions should be set to very small values (i.e. better than 10^{-7}). Weinstein's expression for the freezeout fraction [7] has been used to evaluate the performance of this integrated system. It has been shown that the channel efficiency in this type of integrated system is superior to the others.

Coviello and Vena [55] introduced the integration of voice and data by using hybrid-switching concepts. In this form, the communication between transmitter and receiver is assumed to be based on sending fixed frames. Each frame is divided into two classes. Class I is dedicated to the voice traffic and is circuit-switched, while the class II is dedicated to the data traffic and is packet-switched. This type of integrated scheme is also known as slotted envelope network (SENET). Coviello and Vena also presented two types of strategy for sharing the transmission capacity in this type of integrated system. These strategies are (1) fixed boundary; this is where the capacity of the channel is divided by a boundary, and (2) movable boundary: this is where the data traffic dynamically utilizes the idle free channel capacity normally dedicated to the voice traffic.

Fischer and Harris [57] have developed analytical models for the evaluation of the performance parameters of the SENET. The performance parameters include the blocking probability for the voice traffic, and the average link delay for the data traffic. The results obtained from these analytical models have indicated that the movable boundary strategy results in the achievement of much a better performance for the data traffic than for the fixed boundary.

Gitman et al [58,59] showed that for when the SENET frame length is of small duration (i.e. order of milliseconds), the blocking probability of the circuit-switched voice traffic can be approximated by the Erlang-loss formula. They also derived an expression for the average link delay for the data traffic, basing it on the M/G/1 queueing model. By using these analytical tools, some design methodologies have been developed to evaluate the performance parameters of a multinode SENET system. The design was biased to achieve minimum cost of network resources (i.e. nodes, links, capacities) which satisfy the average end-to-end delay for packet-switched data traffic, and average end-to-end loss probability for circuit-switched voice traffic.

The problem of optimum capacity allocation in a multinode SENET network has been dealt with by Avellaneda et al [60]. The problem addressed was to find the optimum division of capacity for each of the links so as to minimize the probability of blocking of a voice call with the constraint upon the average delay of a data message. The tools used for evaluation this problem were the Erlang-loss formula, to give blocking probability; and the M/M/1 queueing model to give the average data packet link-delay. The results have shown that considerable improvement in the performance is achieved when the capacity is allocated on a link-by-link basis.

Ross et al [61] have further elaborated on the SENET approach. They have considered many network details, including control of the network flow and signalling between the switching nodes. Architectural alternatives have been suggested for building a SENET switch which requires high-speech processing and appreciable nodal buffering needed for the many frames being assembled, disassembled, or transmitted at any given time. The M/M/N queueing model was used for the evaluation of the data packet link delay in the system.

Ross and Mowafi [62] investigated different approaches for the integration of voice and data by using the hybrid switching concept as follows:

1) SENET with fixed and movable boundaries

2) Enhance hybrid.

The operation of this scheme is similar to the SENET with movable boundary, however, with the difference that the circuit-switched voice traffic is supplemented with TASI.

3) Flexible hybrid.

In this approach the voice traffic is packet-switched and therefore the integrated system becomes a totally packeted-switched type.

The M/G/1 queueing model has been used and the characteristic of delay versus information throughput has been obtained as the performance criteria. The results have shown that the flexible hybrid has considerably better performance than the others.

Maglaris and Schwartz [64] have investigated the integration of voice and data by using a hybrid-switched time varying frame. Analysis of the performance of the system was carried out by using an M/G/N queueing technique. The comparison of the results with the fixed frame scheme has indicated the superiority of the variable frame technique, in terms of more efficient bandwidth utilization.

Weinstein et al in Ref [65] evaluated the performance of data traffic in the movable boundary SENET by using a computer simulation technique. The results obtained from the simulation have been compared with the corresponding results of Fischer and Harris [57]. These comparisons have indicated that under heavy traffic conditions the average data packet delays obtained for the simulation are significantly higher than the corresponding analytical results of Fischer and Harris. They found that these large data delays occur during periods of time when the voice traffic load through the miltiplexer exceeds its statistical average. In order to reduce this effect, two flow control mechanisms were investigated. In the first flow control method the bit rates of the voice callers who enter the system during the peaks of voice channels occupancies were cut down. It was found that the data performance was not satisfactory when using this type of flow control technique. In the second flow control method, the above voice flow control mechanism was combined with a data flow control mechanism. It was shown that this mechanism improved considerably the data traffic performance. The same authors in Ref [66] introduced a two-dimensional Markov chain model in order to evaluate the queueing performance of data traffic in the movable boundary SENET system. However, the numerical solution of this model entails significant computational difficulties. Williams and Leon-Garcia [67], however, presented several matrix techniques that circumvented the numerical difficulties of the above method. The same authors in Ref [68] and also Gaver and Lehoczky [69] presented another type of analytical model which can be used to evaluate the data traffic performance. This method is known as the fluid flow approximation.

Maglaris and Schwartz [70] have studied the optimal capacity allocation of the frame to each type of traffic in the movable boundary SENET. Two criteria were presented which involve the blocking probability for the voice traffic and the average queueing delay for the data traffic. The first optimising criterion considered, is that of minimizing the average packet queueing delay with the blocking probability of the voice calls constrained to be no more than a specified acceptable level. The second optimization criterion minimizes the weighted sum of the average packet queue size and the probability of blocking. These optimization problems have been formulated as a Markov decision process.

The other approaches to analytical evaluation of the data traffic performance in the movable boundary SENET were carried out by the authors of Refs [71],[72]. For example, Janakiraman et al [71] used a three-dimensional Markov chain model and also extended their analysis to the variable frame length movable boundary SENET. The validity of the analytical model was justified by comparing the obtained results with the simulation results of Weinstein et al [65]. Sriram et al [72] used the discrete-time Markov chain analysis approach. They also extended their analysis for the case where each voice channel contained a speech detector so that channels are only dedicated to the voice traffic when the connected talkers are in their talkspurts. As a result, more capacity becomes available for the data traffic, and therefore the data traffic performance will improve. This scheme is also known as the movable boundary SENET with speech activity detection (SAD). Bially et al [73] introduced a simulation model for evaluating the data traffic performance in the movable boundary SENET with SAD. The results obtained from this simulation model showed that the performacne of the data traffic is improved considerably over that of movable boundary SENET without SAD.

Fischer [74] developed a mathematical analysis for a single channel integrated voice/data system. In this model the voice calls are assumed to have priority over the data packets, that is the data packets are transmitted only when there are no voice calls present in the system, or the conversation is in a silent period.

Forgie and Nemeth [43] have investigated a single link integrated voice/data network with a totally packet-switched operation scheme. In this model the packets of the two traffics are stored in two separate queue buffers. The transmission priority is always given to the voice packets and the data packets are transmitted only when the voice queue buffer is empty. The voice queue buffer is finite; and in case of overflow, half of the voice packets in the queue are discarded. The talkers are also assumed to be always in their conversational modes. By aid of a computer simulation technique, the first and the second moments of the data packet service times have been obtained and an M/G/1 queueing model has been used to evaluate the data traffic performance in terms of average data packet waiting times. Arthurs and Stuck [76] have investigated an integrated voice/ data model the operation of which is similar to the integrated model of Forgie and Nemeth [43] mentioned above. However, they included the effect of telephone traffic fluctuations and they also assume that the maximum voice packet waiting time does not exceed one frame-length.

Mowafi and Kelly [77] have investigated the use of three different protocols on an integrated voice/data system the operation of which is similar to that presented by Forgie and Nemeth [43]. These protocols are (1) PVC, (2) PAR, and (3) MIXD. In the PVC, or the packet virtual circuit approach, both voice and data packets are routed along fixed paths, while in the PAR, or packet adapting routing approach, both voice and data packets are independently routed to their destination. In the MIXD approach, voice packets are routed over a fixed path while data packets are independently routed to their destinations. The results presented in this paper have shown that the MIXD protocol gives superior throughput delay performance characteristics.

Ueda et al in Refs [78],[79] have described another integrated service digital network system which is based on using packetized voice/data technology and has been developed to integrate voice, data, and facsimilie services. This integrated network consists of three kinds of nodes, i.e., an intelligent service unit (IDSU), a packet processing terminal (PPT), and a packet routing terminal (PRT). The IDSU is located at the customer's premises and interfaces between a digital subscriber line and various equipments, such as telephone, facsimile, and data terminals. Assembly and disassembly of packets are performed at the IDSU. The PPT is a packet multiplexer and switcher, which controls the signalling and information packets and supervises the status of the accommodated IDSU. The PRT is a high speed packet switcher. This scheme is called the packetized service integration network (PSI-NET).

2.6 Discussions and suggestions for the work to be done

As has been mentioned previously, one of the specific aims of this thesis is to study the integration of voice and data in an environment in which both traffic streams are packet-switched. From the review of the literature it can be noted that there has been a great number of studies on the integration of voice and data by using the circuit-packet SENET approach (i.e. Refs [55] - [72]). In the SENET approach the voice traffic is circuit-switched while the data traffic is packet-switched. From the review of the literature it can be concluded that only a few researchers have investigated the integration of voice and data where both traffic streams are packet-switched. (i.e. Refs. [43], [62], [76] and [77]). In the following pages we discuss the short comings of the work of the above authors and we will present new schemes which can be used to improve the performance of the integrated systems.

First we discuss the work of the Forgie and Nemeth [43]. We notice the following points:

(1) In their model the priority of packet transmission is always given to the voice traffic. This policy results in the efficient handling of the voice traffic, but the data traffic handling will be poor.

(2) In their analysis and simulations they assume that the voice channels were continually connected. This is an unrealistic situation and it will only be the case if the voice traffic intensity applied from the exchange is infinitely high.

(3) In their analysis they assumed that the data packets were from an independent random process and arrive according to a Poisson process. By using this, they obtained the queueing performance of the data traffic using an M/G/1 queueing model. The statistical moments of the data packet service times for this model were obtained from computer simulations. The disadvantage of this approach is that the M/G/1 is only valid for this scheme if the service times are an independent random process. It is unlikely that this is so. To explain this further, let us refer to the example given below:

Let us assume that there are two buffers, one which contains the voice packets and the other contains the data packets. We will also assume that there is a single transmission link available for servcing these two buffers. The transmission link only serves the data buffer if there are no packets waiting in the voice buffer. Thus only during certain clear periods is the data buffer served. As a result, one can conclude that the data packets service time in this queueing model is not an independent random process. Its value at a given time is dependent upon the status and content of the voice queue buffer (i.e. if the voice queue buffer is empty the data packet at the head of the queue will be served immediately, otherwise it has to wait until the content of the voice queue buffer is emptied). One undesirable effect which this can cause is that for periods when the channel is constantly serving the voice buffer there will be a large data queue build up in the data buffer. In the M/G/1 model, the service time is an independent random process, therefore these transients will not be predicted. Thus the results which the M/G/1 model gives will be significantly different from the real behaviour of the system. In chapter five (section 5.6) the author has conducted a simulation exercise to prove that the M/G/1 is not a valid model for this type of integrated system.

The other authors who have investigated totally packet-switched integrated voice/data system were Mowafi and Kelly [77]. They investigated the same type of integrated system as investigated by Forgie and Memeth [43], but they assumed that the voice packets arrive according to a Poisson process. By using this property they investigated the performance of the integrated system by using analytical approaches. However, in their analysis they too evaluated the performance of the data traffic by using the M/G/1 queueing model. As explained above, the M/G/1 model cannot be used to evaluate the data traffic performance in this type of integrated system.

Arthur and Stuck [76] also investigated an integrated model with an operation scheme similar to that of Forgie and Nemeth [43]. However, they assumed that the maximum voice packet waiting time does not exceed one frame-length which frees the loading of the system significantly. They also used the M/G/1 model to evaluate the performance of the data traffic. Their analysis was also extended to evaluate the effect of telephone traffic fluctuations on the system.

Ross and Mowafi, in Ref [62], investigated and compared two integrated voice/data models in an environment in which both types of traffic are packet-switched. Their first integrated model was the same as the one already investigated by Mowafi and Kelly [77]. The second integrated model had a similar operation to the former, however, the difference was that the voice and data packets in this system were transmitted in fixed frames. Thus in the latter model there is the possibility that a portion of the frames would be unused due to the frame formating scheme. It was shown that due to the unused capacity, the former integrated system had slightly better performance characteristics than the latter. In the same paper the M/G/1 model was used to evaluate performance characteristics of the data traffic.

It can be concluded that there are two main criticisms of the operation and analysis of the totally integrated packet-switched models as explained above. These criticisms are explained as follows:

The first criticism of all these integrated models is with their operation. In all these models it has been assumed that the priority for packet transmission is always given to the voice traffic. As was mentioned above, this protocol will cause the data traffic to have a poor performance. In chapter seven two new integrated models will be introduced that overcome this problem. The suggestion is that no priority be given to either of the two traffic streams. The operation of the first model employs the SENET concept. This means that the transmission policy is based on sending fixed frames employing a boundary allocation strategy. Some criteria will be presented as to how to set the boundary within the frames so that the capacity is divided appropriately between each of the two traffic streams.

The second suggested integrated model which has been studied in chapter seven employs the first-come-first served (FCFS) voice and data packet multiplexing approach. This means that at any instant the transmission link would transmit either voice or data packets according to which arrive first. Thus this multiplexing protocol allows no priority to either of the traffic streams for the packet transmission.

The second main criticism which can be levelled at the work of the previous authors is that they all used the M/G/1 queueing model to evaluate the data traffic performance. As was explained this model is not really valid for this type of integrated system. In chapter seven computer simulations have been used to evaluate the correct performance characteristics of the two integrated models. These simulation models are explained in full in that chapter.

The second main aim of this project has been to investigate the performance evaluation of the packet-switched voice traffic on its own. In chapter four characteristics are evaluated which have been used to obtain the optimum design of a two node packet-switched voice system. From the review of the literature it can be noted that little work has been done to investigate the effect of telephone traffic loading on packet systems. For example, Fischer in Ref [17] showed that in a DSI system with speech storage, the effect of telephone traffic fluctuations indicates that talkspurts experience less average queueing delay than calculated for the all channel busy situation. It can be concluded from these results, that an increase in the TASI advantage of the system can be permitted.

Kou et al in Ref [16] showed that in a conventional analogue TASI system, when the effects of telephone traffic loading were considered, the system experiences less average freezeout than when in all-channels busy situation.

The work of the both Fischer and Kou et al of Refs [16,17] are based on using average values. An important problem which has not yet been investigated, is whether the worst case situations are tolerable when traffic loading effects are used to increase the TASI advantage. With the aid of computer simulation techniques, this thesis sets out in chapter four (section 4.4) to investigate this problem for a two node packet-switched voice system. Liu et al in Ref [53] also investigates the performance of a PVS with the inclusion of the telephone traffic loading. They showed that the scheme can achieve a TASI advantage of 30/12 = 2.5. However, their scheme was a non-blocking system. This means that each of the multiplexing channels were connected to only one telephone. In such a system the probability of all the channels being connected simultaneously is extremely low. In this thesis we investigate the effects of telephone traffic loading with blocking probabilities ranging between 0.1% to 5%.

In Ref [7], Weinstein introduced a simple packet voice system where packets from the channels can only wait one frame and are lost if the delay is greater. In chapter five this system is studied more fully. New flow control mechanisms have been applied to this system to improve its TASI advantage performance. A new demultiplexing model has also been applied to the scheme which permits this scheme to be incorporated into a multinode PVS with ease. The applicability of this scheme when incorporated into an integrated voice/data network has also been investigated and is reported in chapter seven.

CHAPTER 3

COMPUTER SIMULATION TECHNIQUES FOR PACKETIZED VOICE COMMUNICATION SYSTEMS

3.1 Introduction

This chapter deals with the ways in which the performance of a packetized voice system (PVS) can be measured by using different types of computer simulation models. The PVS is assumed to utilize a large queueing buffer at the transmitter. Because of the stochastic behaviour of the queue, the system throughout this thesis is referred to as the stochastic queueing (SQ) system. In Section 3.2, the transmitter operation of this system is fully explained. Two simulation models are presented for this system which are named as the Microscopic type and the Markovian type. These simulation models are presented in full details in Sections 3.3 and 3.4 respectively. The results relating to the performance evaluation of the transmitter part of the PVS using these two simulation models have been obtained and are compared in Section 3.5.

In Section 3.6, queueing model approaches for this system, as suggested by other researchers, are presented. The results obtained from these queueing models are compared to the simulation results. It has been shown what the two sets of results do not always agree with each other. The reasons for this behaviour are discussed.

3.2 Transmitter Operation of the PVS

In this section the transmitter operation of the PVS is explained. Three aspects are specifically covered, (1) packetization, (2) multiplexing and (3) queueing and transmission operations.

3.2.1 Packetization operation

The block diagram of this process is given in fig. 3.1. In this form, the analogue speech signal is sampled and digitized at a uniform rate by the analogue to digital (A/D) encoder in the transmitting terminal for each speech circuit. The samples are then fed to a speech detector. The speech detector could be of the type used for the conventional TASI system as suggested and implemented by Miedema and Schachtman [11]. This implies that the speech detector utilizes a 5 ms operate time for eliminating the noise, - 40 dbm sensitivity and 240 ms hangover for bridging the short silences (i.e. gaps) in the conversational speech. The digital signal out of the speech detector then contains the significant utterences, i.e. 'talkspurts'. These active samples are then organized into constant length packets by the packetizer. Each packet, at this stage, has additional overhead bits added for address and other purposes (e.g., the destination address, flag, time stamping and sequence number), it is then forwarded to the transmission media via the statistical multiplexer and buffering facilities.

The packetization operation, as explained above, has been assumed to be performed for each talker separately, either at the transmitting terminal or at the exchange. The size of the overhead assigned for this scheme has been assumed to be as high as 100 bits; i.e. destination address (40 bits), CRC (32 bits), flag (8 bits), time stamping (8 bits) and sequence number (12 bits).

3.2.2 Multiplexing operation

The multiplexing operation, as depicted in fig. 3.2, is of the form of a synchronous time division statistical multiplexer, as described in [30], [32]. The operation of this type of multiplexing is a mixture of circuit and packet switching. The circuit-switching aspect is present because the synchronous multiplexer utilizes the Packet Virtual Circuit protocol. This implies that two one-way routes are established at dial-up and are retained for the duration of a conversation. The packet switching aspect is present because of the packetization of the speech samples and the use of the packet multiplexer. In brief, the multiplexer provides a means of time-sharing n channels onto a standard m-channel PCM transmission link, where here n is > m. The multiplexer uses a single data link to the transmitter buffer and regularly sends frames of information packets to the buffer. The duration of each multiplexer frame is assumed to be Δ_t seconds, which is equal to the packetization period. The multiplexer therefore requests a packet from each talker's pre-buffer every frame repetition interval and a packet is only transferred to the transmitter buffer if there is a complete packet waiting when its buffer is viewed. A packet will not appear at this buffer if the speech channel is in a silent period.

Pre-buffering is provided at each multiplexer input in order to handle the statistical fluctuations in the packet arrival process. As only one packet can be transmitted at a time from the multiplexer input to the transmitter buffer, two packet-lengths storage is allowed for in each pre-buffer and this prevents buffer overflow.

3.2.3 Queueing and transmission operations

The information contained in the transmitter buffer is sequentially fed to the transmission link, which has a constant transmission rate of c bits/sec. During high activity periods, when the number of packets in any frame produced by the talkers exceeds the number of packets that can be sent by the channel, the transmitter buffer occupancy starts to build up. This effect will cause the packets to experience variable queueing delays for their transmissions. The effect will be that the packets for each talker will be received with variable and incorrect gaps at the receiver. The ways employed to deal with this behaviour are dealt with in the next chapter.

During the low activity periods the transmitter buffer is empty most of the time as the transmission link is a high bit rate channel. Assuming the link to be a constant bit rate channel, then idle bits must be provided in order to keep the transmission link occupied. These idle bits have to be in the form of "idle-packets", each with a special identification header. In an integrated packet voice and data environment a data buffer could replace the idle period buffer and data packets could then be processed by the system. In this type of integrated system priorities are always given to the voice packets.

3.3 Microscopic Simulation Model

In this section the microscopic type of simulation model is presented. The simulation model is called Microscopic, as all stages of the system operations are considered microscopically in the simulation program. There are four processes involved in this model and each are separately discussed. The processes are namely: (1) speech sample generation, (2) packetization, (3) multiplexing and (4) queueing and transmission.

3.3.1 Speech sample generation

Conversational speech has been analysed by Brady [1] - [3] and can be represented by short bursts of activity (talkspurts) followed by comparitively long silences (pauses). However, within a talkspurt there are intervals of continuous activities (spurts) followed by short silences (gaps). This structure of talker activity model is shown in fig. 3.3

When considering packetized speech transmission with stochastic queueing behaviour the packets are subject to variable queueing delays. At the receiver, it is important to play-out speech with the gaps faithfully reconstructed. This is necessary as the short silences represent meaningful speech and any variations in these intervals could significantly degrade the speech quality (see Brady [1], Gruber [18] and Yatsuzuka [12]). The Brady's experimental investigations of speech [1] have indicated that the gaps have a range of duration between 10-250ms. The speech detector, as already suggested, has been given a hangover of 240 ms which therefore bridges the gaps. This forces the variable delay occurrences to be imposed on the relatively long silent periods (i.e. the gaps between phrases and sentences) where the variability is much less perceptible than if imposed on the short silent periods within words and syllables.

Taking the above aspects into account, the main objective of the simulation process at this stage is to generate random numbers to represent the talkspurts and the pauses for a set of talkers in conversation. To perform this task, one usually requires the probability distribution functions (pdf) of the talkspurt and pause intervals. Unfortunately, the statistics as given by Brady [1] are only for the cummulative distribution functions (CDF) of the talkspurt and pause intervals. However, the author has devised a simple scheme which makes it possible to draw random numbers from the CDF's as given by Brady (see Appendix A). By employing this method a set of random numbers which represent the talkspurts and the pauses for each talker are produced and they are stored into an array by the simulation program.

An important parameter which is used throughout this thesis for studying the performance of a PVS is the probability that a packet is issued at random by a talker in conversation. According to Weinstein [7], this parameter, as denoted by q, is approximated by the following relation:

$$q = \frac{L_T}{L_T+L_p} \qquad (3.1)$$

Where L_T and L_p represent respectively the average length of the talkspurt and pause intervals. Following Brady's figures for L_p and L_T (i.e. L_T = 1.34 sec, L_p = 1.67 sec) this makes $q = 1.34/(1/34+1.67) = 0.445$.

Another type of talker activity model has been recently suggested by Clarke and Turner [21]. The structure of this model is based on the statistical results of Norwine and Murphy [4] on two-way conversational speech, as depicted in fig. 3.4. In this model, as soon as a talker begins to speak, the talkspurt length is drawn from a lognormal distribution given by

$$p(x) = \frac{1}{x\sqrt{2\pi\sigma^2}} \exp\left(-\frac{(\log_e x-\mu)^2}{2\delta^2}\right) \qquad (3.2)$$

where μ and σ^2 are parameters of the distribution. Following the values of mean and mode from [4] (4.14 secs and 0.25 secs respectively) these give values of μ = 0.485 secs and σ = 1.871 secs. Also according to [4], the response times, or the times between the end of one talkspurt and the begining of the next, were approximated by a normal distribution with mean 0.41 secs and standard deviation of 0.584 secs. The normal distribution corresponds to negative values as well as positive values. A negative value of response time indicates that the second talker begins to speak while the first is still speaking, i.e. an interruption. A positive value corresponds to the more normal pauses observed by both speakers before the next begins to speak. From the examples in fig. 3.4, it is evident that the pause is equal to the addition of two response-time intervals drawn from a normal distribution plus a talkspurt drawn from a lognormal distribution. Accordingly, the packet generation probability for a single talker using this model becomes

$$
\begin{aligned}
q &= \frac{L_T}{L_T+L_T+2L_R} = \frac{L_T}{2(L_T+L_R)} \\
&= \frac{4.15}{2(4.15+.41)} = .455
\end{aligned}
\tag{3.3}
$$

where L_T and L_R respectively represent the average length of talkspurt and response time intervals.

Although the packet generation probabilities of the two models (Clarke & Turner's and Brady's) are very close, there is a significant difference between them as in the Clarke and Turner model, the average talkspurt length (i.e. L_T = 4.15 secs) is much higher than that of Brady's (i.e. L_T = 1.34 secs.). The reason for this is because the Norwine and Murphy definition of talkspurt is significantlly different from Brady's. The Norwine and Murphy's talkspurt includes pauses or silences greater than 250 ms other than the natural gaps which are common with both talkspurts (see also Brady [1], and Norwine and Murphy [4]). As it will be shown later, if the packet arrival from the speech terminals do not form a

Poisson process, then the queueing performance of the system will become highly dependent upon the statistical duration of the talkspurt and silence intervals. The comparisons of the results obtained from the two talker activity models are shown later.

3.3.2 Packetization Process

In the simulation program, the packetization of the speech samples are being performed by dividing the stored speech samples of the total n subscribers into equal periods of duration Δ_t seconds (see fig. 3.5). In each interval, if the speech sample of any talker is in the state of talkspurt, a packet of fixed length of duration Δ_t seconds is then generated at the end of the associated interval. The delay experienced per packet, or the packetization delay, D_z, at this stage is equal to $\Delta_t = p/f$ seconds. Where p is the packet length in bytes excluding overhead and f is the sampling rate of the speech signal. For f = 8000 samples/secs, D_Z in milliseconds becomes

$$D_Z = (p/f) * 1000 = p/8 \text{ ms} \qquad (3.4)$$

For simplicity throughout this thesis, the term 'packet length' is assumed to be the speech information part of the packet; which is either p/f secs, or p/8 ms as above.

3.3.3 Multiplexing process

In each frame interval, Δ_t secs, which is equal to the packetization period, the multiplexer spends an equal amount of time, duration γ_t seconds, on each talker's pre-buffer in order to transmit a packet to the transmitter buffer, that is if there is a packet available. If the total number of connected channels on the multiplexer is n, then γ_t is given by

$$\gamma_t = \frac{\Delta_t}{n} \quad \text{secs.} \tag{3.5}$$

Since $\Delta_t = D_Z = \frac{p}{f}$, then

$$\gamma_t = \frac{p}{fn} \quad . \quad \text{secs.} \tag{3.6}$$

We call this process, as shown in fig. 3.6, time division packet multiplexing (i.e. the frame is slotted into n time slots of duration γ_t secs each). This form of multiplexing forms a discrete packet arrival process where packets occur, or do not occur, in regular time slots. These time slots occur at $r\gamma_t$ seconds, where $r = 0,1,2,3,\ldots,(n-1)$.

3.3.4 Queueing and transmission process

As explained previously, the packet arrival process into the queueing system forms a discrete type of arrival with the interarrival times as a multiple integer of γ_t seconds. The ways in which the instantaneous queue length, $q(t)$, is varying with respect to time are explained as follows. Let us refer to fig. 3.7, which shows the fashion in which the packets arrive into the queue. Packet C_1 enters the system at time t_1. The queue length at this time is assumed to be zero, which makes the packet C_1 go straight into the service. This packet is of length P_1 bits given by

$$P_1 = pb + h \tag{3.7}$$

where h is the overhead in bits and b is the number of bits assigned per speech sample.

As the transmission link is of constant rate of c bits/sec, then during each slot interval (γ_t secs), the channel transmits δ bits as given by

$$\delta = c\gamma_t = \frac{cp}{fn} \tag{3.8}$$

Up to the time t_2, the unserviced part of the packet C_1 is given by

$$\begin{aligned} U &= P_1 - \delta \\ &= pb + h - \frac{pc}{fn} \quad . \end{aligned} \tag{3.9}$$

Thus at time t_2, the queue contains U bits which is the unserviced part of the packet C_1. The queueing delay, or the time required for transmitting U bits, is given by

$$\alpha_t = \frac{U}{c}$$

$$= \frac{pb+h}{c} - \frac{p}{fn} \qquad (3.10)$$

$$= \frac{pb+h}{c} - \gamma_t \; .$$

The packet service time, D_s, is also given by

$$D_s = \frac{p_1}{c} = \frac{pb+h}{c} \; . \qquad (3.11)$$

By substituting (3.11) into (3.10), α_t becomes

$$\alpha_t = D_s - \gamma_t \qquad (3.12)$$

At time t_2, the packet C_2 enters the queue. This packet finds U bits unfinished work in the queue and therefore it experiences α_t secs queueing delay before the packet goes into its service phase. By the same type of argument as above, it can be shown that at time t_3, the queue contains 2U bits unfinished work or a queueing delay of $2\alpha_t$ secs.

At time t_3, no packet enters the queue. Thus up to time t_4, the channel serves the queue constantly and reduces the amount of unfinished work in the queue. Since the interval $t_4-t_3=\gamma_t$ secs, then the queueing delay at time t_4 is $(2\alpha_t - \gamma_t)$ seconds. Thus as time progresses, this queueing process repeats itself. This is based on saying that; during each slot interval, γ_t secs, the presence of a packet will cause an extra amount of unfinished work of α_t secs and the absence of a packet will reduce the amount of unfinished work by γ_t secs.

The above procedure has been used in the simulation program. As is shown in fig. 3.8, each frame is partitioned into n-slots (where n denotes the total number of subscribers) each of size γ_t seconds. Each of the slots are checked whether it contains a packet or not. In the case of the presence of a packet, its instantaneous queueing delay plus its service time is first recorded and then α_t seconds is added to the amount of unfinished work in the queue. In the case of the absence of a packet in the given slot, γ_t secs of the amount of unfinished work is deducted. This process is then repeated a large number of times. Eventually averages for these figures are derived for plotting.

3.4 Markovian simulation model

This simulation model is based on the dynamic talker activity model which was first suggested by Weinstein [7]. The basic difference of this model and the Microscopic type is that in this model individual speaker identities are ignored and it deals instead with transitions from talkspurt to silence and vice versa over the entire speaker population. This approach allows the use of a simple birth-death Markov approximation for the aggregate process, and leads to a computationally efficient model that is valid, provided that the speaker population is sufficiently large.

This model, as shown in fig. 3.9, deals with n off-hook speakers in conversational mode, and switches alternately between talkspurt and silence states. The parameters

$1/\lambda$ = mean pause duration

$1/\mu$ = mean talkspurt duration

are defined for each talker. Let the random process $a(t)$ represent the number of talkers active (i.e. issuing talkspurts) at a given time. The behaviour of $a(t)$ will be characterized by step changes of ± 1 at random times. If n is large (i.e. $n \geq 25$ according to [7]), then the process representing transition events in $a(t)$ is approaching Poisson. This leads to a model for $a(t)$ as a birth-death process, as shown in fig. 3.9, with $n+1$ states representing the possible numbers of simultaneously active talkers. From state $a(t) = k$, there are k independent sources, each of average rate μ, for the transition to state $a(t) = k-1$, representing the event of one of the active talkers dropping into silence. Similarly there are $n-k$ rate λ sources for a new talkspurt, which would yield a transition to state $a(t) = k+1$. Thus the transition rates describing the birth-death process are

$$
\begin{aligned}
\lambda_k &= \text{rate of transition from state } k \text{ to state } k+1 \\
&= \lambda(n-k) \qquad 0 \leq k \leq n-1 \\
\mu_k &= \text{rate of transition from state } k \text{ to state } k-1 \\
&= \mu k \qquad 1 \leq k \leq n.
\end{aligned}
\tag{3.13}
$$

Following Weinstein [7], the steady-state probability that $a(t) = k$ is found to be a binomial distribution given by

$$P_k = \binom{n}{k} q^k (1-q)^{n-k} = b(n,k,q) \tag{3.14}$$

where as before

$$q = \frac{1/\mu}{1/\lambda+1/\mu} = \frac{\lambda/\mu}{1+\lambda/\mu} \tag{3.15}$$

is the ratio of the mean talkspurt duration to the sum of mean talkspurt duration and mean silence duration, and represents the probability that the individual talker is issuing a talkspurt at a random time.

The local behaviour of the Markov chain leads to a straightforward description of the time-varying behaviour of a(t). Assuming that a(t) = k at a given time, the holding time until the next transition will be exponentially distributed according to the pdf

$$f(t) = \frac{1}{T} e^{-t/T} \tag{3.16}$$

where

$$T = \frac{1}{k\mu+(n-k)\lambda} \tag{3.17}$$

After remaining in state k, the probabilities that the next transition will be upward (k → k+1) or downward (k → k-1) are determined according to

$$P[k \to k+1] = \frac{\lambda_k}{\lambda_k+\mu_k} = \frac{(n-k)\lambda}{(n-k)\lambda+\mu_k} \tag{3.18}$$
$$= 1 - P[k \to k-1].$$

In general, the way in which this model is used for simulating a packetized voice system are as follows:

As shown in fig. 3.10, we assume the system is started initially from the state k, i.e. a(t) = k where t = 0, k = 0. The time τ remaining in this state until the next transition is drawn via a random selection from the exponential distribution according to (3.16), (3.17). We next identify the next event of a(t) by using the relation (3.18), i.e. we determine the system will be in state k+1 or state k-1 after the subject interval elapses. The procedure, as above, is repeated for a large number of trials. During the course of this process, the time intervals are divided into intervals of duration Δ_t seconds, which is equal to the frame period of one reveloution of the multiplexer as in fig. 3.2. Since the number of packets, L, which are generated in each Δ_t seconds interval is equal to the total number of talkspurts existing in the associated interval, then

$$L = k + \eta \tag{3.19}$$

where k is the number of talkspurts at the beginning of a Δ_t seconds interval and η denotes the number of transitions from silence to talkspurt which have taken place during the associated interval (see the examples given in fig. 3.10).

During each of these frame intervals, the instantaneous number of packets in the queue is computed by

$$q(r\Delta_t) = \max(q[r-1)\Delta_t] + L - \frac{c\Delta_t}{p_1}, 0), \qquad (3.20)$$

where p_1, as given by (3.7), denotes the packet length in bits including the overhead. At the end of simulation period, the average number of packets in queue, $\bar{Q}_L$, is computed by

$$\bar{Q}_L = \frac{1}{M+1} \sum_{r=0}^{M} q(r\Delta_t) \qquad (3.21)$$

where $M\Delta_t$ denotes the simulation period. The average data packet waiting time in the queue, $\bar{Q}$, is then determined by the Little result [23] and is given by

$$\bar{Q} = \frac{\bar{Q}_L}{\theta}, \qquad (3.22)$$

where θ denotes the average packet arrival rate per second and is given by

$$\theta = \frac{nq}{\Delta_t}. \qquad (3.23)$$

3.5 Simulation results and comparisons

In this section simulation results relating to the comparisons of the Microscopic and Markovian simulation models are presented. The simulations have been carried out by considering a packetized-voice system which has an infinite buffering capability.

We have assumed the link to have a constant transmission bit rate (i.e. a 1.544 Mb/s T1-carrier PCM link). We have further assumed that there are 40 active connected channels and they are sampled at a constant rate and there are a fixed number of digits used to encode the samples (i.e., These are sampled at 8k samples/sec and each sample is encoded to give 8 data bits, so giving a net output of 64 kb/s per channel). In each simulation run the packet length was assumed to be of fixed length, with 100 bits overhead.

Two sets of curves were developed and are shown in figures 3.11 and 3.12. For each figure, the horizontal axis represents the packet length while the vertical axis denotes the summation of the average packet delay with its service time delay, also known as the average link delay. The simulation results presented in fig. 3.11 are for the comparisons of the two different speech models as suggested in the Microscopic simulation model (i.e. Brady's, Norwine and Murphy's statistics). It is evident that significantly higher queueing delays have resulted with the Norwine and Murphy's statistical model for speech sample generation. The simulation period for this scheme was set at 700 seconds of real time. The results obtained correspond to the last 600 seconds after ignoring the first 100 seconds in order to remove the initial condition effects. Each plotted point on these curves represents an average of five runs, where in each run a different random number seed was used.

As a result of the above comparisons and the arguments in Section 3.6.6, the author has employed the Brady's parameters in the future simulation exercises rather than the Norwine and Murphy parameters for speech sample generation.

Fig. 3.12 shows the comparisons of the results relating to the Microscopic and Markovian simulation models. It shows close agreement of the two sets of results. The simulation period for the Markovian type is set at 20,000 seconds of real time. The results obtained correspond to the last 19000 seconds of real time after ignoring the first 1000 seconds to remove the initial condition effects. Each plotted point on the fig. 3.12 represents an average of five runs, where in each run a different random number seed was used.

3.6 Related Queueing Models and Comparisons

In this section, the simulation results which have been obtained so far, are compared with the results obtained from two different queueing models. These queueing models are Minoli's discrete queueing model [26] - [28] and (2) the well known M/D/1 queueing model [23]. In order to compare the results from these models with the simulation results and to evaluate the validity of the queueing model representations, the link utilization of the system has to be understood.

3.6.1 Link utilization

This is denoted by ρ, also known as the traffic intensity, and is measured in Erlangs; it is defined as the product of the mean of the arrival rate (θ) and the mean of the service time, E(s), (see Meisling [25])

$$\rho = \theta \; E(s). \tag{3.24}$$

Following the equations (3.14) and (3.15), the average packet arrival rate per frame, E(r), is given by

$$E(r) = nq \tag{3.25}$$

As the frame is of constant duration Δ_t seconds, then the average packet arrival rate per second, θ, is given by

$$\theta = \frac{E(r)}{\Delta_t} = \frac{nqf}{p} \tag{3.26}$$

Since the channel is of constant bit rate (i.e. c bits/sec) and the packet length is fixed, then

$$E(s) = \frac{pb+h}{c} \tag{3.27}$$

By substituting equations (3.26) and (3.27) into equation (3.24), the link utilization, ρ, becomes

$$\rho = \frac{nqf(pb+h)}{pc} \tag{3.28}$$

3.6.2 Minoli's queueing model

As already discussed in Section 3.3.3, the packet arrival process into the queueing buffer forms a discrete type of arrival with the interarrival time being a multiple integer of γ_τ seconds, where γ_t denotes the size of the packet slot on the multiplexer (see fig. 3.6).

The Minoli's queueing model is also based on the same type of discrete packet arrival assumption as indicated above. However, another assumption was made by Minoli, the validity of which will be discussed later. This assumption is based on the way that the packets are generated in the system. At each time slot, of duration γ_t, a packet is assumed to join the queue with a fixed probability, q, the value of which can be derived by Eq.3.15. Another realization of this assumption is that the packet generation in the system can be viewed as being a sequence of Bernoulli trials. In this situation, the inter-packet arrival into the queue will then be geometrically distributed. Since the packet length is of fixed size and the channel rate is of constant bit rate, then the service time will become constant. The system can then be represented as a geometrical arrival, constant server queueing model. The solution for this queueing model, as given by Minoli in Ref [26], is based on the derivation of the average link delay, W, in the system (i.e. the average queueing delay and service time delay that a packet is expected to experience in this type of queueing system). This is given by

$$W = \frac{1.5 - \rho}{2(1-\rho)} D_s \qquad \text{if } \rho \geq 1/2 \tag{3.29}$$

$$= D_s \qquad \text{if } \rho < 1/2$$

where D_s and ρ are given by equations (3.27) and (3.28) respectively.

In order to test the correctness of the above result for the given queueing model, the author in the following section introduces a simplified simulation model based on the assumptions made in the Minoli's queueing model. The results obtained from this simulation model are then compared with analytical results obtained from equation (3.29).

3.6.3 Simulation model based on Minoli's queueing model

This simulation model, unlike the others, does not involve the use of the structure of talker activity model. The packet generation probability, q, is assumed to be fix and equal to .445, as has already been derived using equation (3.15). The simulation approach is based on iteration of the program for a large number of times. Each excecution of the loop corresponds to a period of γ_t seconds in the simulation program. During which the length of the queue is calculated. This is done by drawing a random number between 0-1 from a uniform distribution. If the drawn number is $\leq$ .445, then a packet is assumed to be generated and its waiting time is recorded. The content of the queueing delay is then incremented by α_t seconds as given by equation (3.10). For the case when the drawn number is >0.445, the content of the queueing delay is then decremented by γ_t seconds, as given by equation (3.5). This process is repeated for many time periods (i.e. one million packet generation events has been used here.)

3.6.4 M/D/1 queueing model

As shown previously, the inter-packet arrival in the Minoli's queueing model is geometrically distributed. Since the geometrical distribution, like the exponential distribution, enjoys the memoryless property (see Cooper [22] and Konheimer [34]); thus we can see that a similarity exists between this discrete queueing model and a Poisson arrival constant-single-server (i.e. M/D/1) queue. In order to compare the results of these two queueing models, the average link delay in the M/D/1 is formulated as follows:

From Kleinrock [23]; the average link delay, W, in an M/D/1 queueing model is given by

$$W = \frac{\rho D_s}{2(1-\rho)} + D_s \tag{3.30}$$

Where ρ denotes the link utilization and D_s represents the constant service time.

3.6.5 Results and comparisons

In this section the results relating to the queueing performance of the PVS are obtained from the suggested queueing models (i.e. Minoli and M/D/1) and are compared with the simulation results. It has been assumed that the system has the same fixed parameters as given in Section 3.5 (i.e. c = 1.544 Mb/s, n = 40, h = 100 bits, f = 8000 samples/sec., b = 8bits/sample and infinite buffering capability). The simulation results presented in this section were obtained as an average of five runs, where in each run a different random number seed was used.

Three sets of curves were developed and are shown in figures 3.13 - 3.15. Fig. 3.13 shows the comparisons of the Minoli's analytical results with the results obtained from the simulation model of Section 3.6.3. It is evident that the two sets of results have a close agreement with each other. This also proves the correctness of the analytical analysis of the Minoli for his given queueing model. Fig. 3.14 gives the full comparisons of the results obtained from the queueing models (i.e. Minoli and M/D/1) and the simulation models (Microscopic and Markovian). It can be observed that the results obtained from the Minoli's nearly approximated to the results of the M/D/1 model. However, it is evident that these analytical results do not have any agreement to that obtained from the real behaviour of the system under Microscopic or Markovian Simulation. As it can be observed, significantly higher queueing delays are encountered with the simulation results. The reason for this behaviour is discussed in the next section.

In a paper by Minoli [26], he suggested that in a packetized voice communication system with the presence of a transmitter buffering facility, the optimum packet length is obtained when it minimizes the total average packet transmission delay, as denoted here by $\bar{D}_{tr}$, and is given by

$$\bar{D}_{tr} = D_z + \bar{D}_q + D_s \tag{3.31}$$

where $\bar{D}_q$ denotes the average packet queueing delay. As the average link delay $W = \bar{D}_q + D_s$, then

$$\bar{D}_{tr} = D_z + W \tag{3.32}$$

Fig. 3.15 shows the results for this situation. As it shows, the optimal packet in the Minoli and the M/D/1 models have corresponded to 5.375 and 5.75 ms respectively. The simulation results, however, have shown that the optimal packet length is 15 ms for the Microscopic type and 16.25 ms for the Markovian type both of which are significantly longer than that given by the two queueing models.

The comparisons of the results given in figures 3.14 and 3.16 then indicate that the two analytical models for the PVS are over simplifications of the real packetization situation.

3.6.6 Discussions and further results

The simulations of the real behaviour of the PVS have shown that higher queueing delays would be experienced than those indicated by the two queueing models. The reason for this is simply because in the two queueing models the statistical dependency of the speech sources are ignored. In the sequel we shall explain why this statistical dependency will cause the queueing delay to increase.

Let's consider the Minoli's queueing model once again. The packet arrival in this model, as already shown, can be viewed as being a sequence of Bernoulli trials. An important conclusion which can result from this assumption is that the packet arrival will be a completely random process. For example consider two successive frames. The number of packets being generated in these two frames do not have any correlation and are independent of each other. Now let's consider the packet arrival in the simulation environment. In order to study this process, let us refer to the dynamic talker activity model of fig. 3.9. This model, shows that the number of packets which arrive into the queueing system can be highly correlated. For example, one effect of correlation is due to the local behaviour of the chain, which means that the system can be locked to any state of the chain for an interval which could be in the order of a number of frames. The second effect which gives rise to the correlation is the nature of the transition between the states, as is shown in fig. 3.9. The transitions, as depicted, are only to the neighbouring states.

Now let us assume the congestion states in the above chain model to be defined as the set of states in which the buffer occupancy of the queue increases per frame, provided that the system occupies these states for at least one frame interval. These states are from (J+1) to n, where J represents the total number of packets that the channel transmits within a frame interval. This is given by

$$J = \left\lfloor \frac{pc}{f(pb+h)} \right\rfloor \tag{3.33}$$

where $\lfloor x \rfloor$ denotes the greatest integer less than or equal to x. In the Minoli packet arrival model, since the frames are not correlated, the system occupies any of the congestion states for one frame interval. At the next frame the system can come out of the congestion states without any restriction. In the simulation, however, due to the effect of the above correlation the system can not come out of congestion but moves from one congestion state to another. On average, the system remains continuously in the congestion states for more time than in the Minoli's model, which implies a greater buffer build up.

Another effect of this time dependency in the frame arrival is that it makes the queueing behaviour of the system dependent upon the statistical nature of the talkspurt and silence lengths. As already observed experimentally the Norwine and Murphy's statistics for the talkspurt and silence intervals, which were longer than Brady's, gave comparatively higher queueing delays even though the packet generation probability (i.e. q) of the two models were close to each other. In order to study this behaviour further, another set of experiments were carried out. This time the Brady's statistics were considered. The packet generation probability was set at its fixed value (i.e. q = .445), and the average length of the talkspurt was varied from 1.67 seconds to 3.0 seconds. Accordingly in order to obtain the same q value, the average silence interval was varied from 1.67 seconds to 3.74 seconds. The Markovian simulation model was used for comparing the queueing results for the above situations. The results, as depicted in fig. 3.16, show that higher queueing delays result with the longer talkspurt and silence intervals than with the shorter ones. The reason is simply because the transitional rates of both silence to talkspurt, λ_K, and talkspurt to silence, μ_K, are inversely proportional to the mean silence duration, $1/\lambda$, and the mean talkspurt duration, $1/\mu$, respectively (see equation 3.13). Thus when these mean values increase, these transitional rates decrease; which in turn implies that the system will spend more time continuously in the congestion states and therefore the queue build up will be comparatively more than in the situation when these mean values are smaller.

It should however be noted that these transitional rates are also dependent upon the number of states. For example from equation 3.13, λ_K is proportional to $(n-k)$ and μ_K is proportional to k. Thus if the total number of subscribers (n) is increased, then these transitional rates will be increased, which results in a reduction of the frame correlation in the system. It can also be seen from equation 3.17, that if n is increased, then the system, on average, spends less time on the states of the chain, which this in turn results in a reduction of the time dependence correlation among the frames.

In order to investigate the above arguments, the number of subscribers (n) was increased from 40 to 140. In order to keep the link utilization, ρ, less than unity, the encoder output/channel was varied from 64 kb/s to 16 kb/s (i.e. each speech sample was assumed to be encoded to 2 bits instead of 8 bits)[1]. The Markovian simulation model was used to evaluate the same queueing performance parameters of the system (i.e. the characteristic of the average link delay versus packet length). The analytical solutions of the Minoli and M/D/1 models have also been used to evaluate the same characteristic. Fig. 3.17 shows the comparison of the results obtained for this situation. It is evident that the simulation results have a better agreement with the analytical results than in the previous case when the number of subscriber was 40. Another similar experiment was performed and this time the number of subscribers was increased to 250. This being achieved by reduncing the encoder output/channel to 8 kb/s. Fig. 3.18 shows that the analytical and simulation results in this case have an even better agreement. It can also be observed that as the number of subscribers are increased further, the two analytical results will have even better agreement. This is because as the number of subscribers is increased, the packet slot interval (i.e. $\ddot{Y}_t$ as given by equation 3.5) will be reduced to a shorter interval, which implies that the geometrical distribution tends more to become a negative exponential distribution (see Meisling [25]). Thus eventually the packet arrival process can be analytically approximated by a Poisson process.

---

1 Note the assumptions of lowering the encoder bit length are not practical but have been assumed to indicate the mathematical significances.

The conclusion which can be drawn from the comparison is that if the number of subscribers be sufficiently large (i.e. of order of hundreds), then the packet arrival process in the simulation can be approximated to a Poisson process. This conclusion will be used in Chapter 7 for evaluating the performance of an integrated voice/data system in a purely packetized environment.

Another major consideration is the optimization of packet delay. According to Minoli, the performance of the system in terms of delay is optimized when a packet length is chosen which minimizes the overall mean packet transmission delay. In practice, this criteria is only correct if the other effects of the system relating to the packet losses are ignored and specifically, when the extra delay requirements at the receiver are ignored. As it will be shown in the next chapter, there is a need for a delay at the receiver, its size being dependent upon the packet length. Further, the author in the next chapter introduces a stronger optimization technique based on the minimization of the end-to-end delay for a given maximum allowed packet-loss probability.

3.7 Conclusions

Two simulation models were developed for evaluating the performance of a packetized voice communication system. It was shown that the results obtained from these simulation models do not always agree with the results obtained from the related analytical models. The differences are due to the over simplifications which have had to be taken in the analytical models. It was however shown that if the number of subscribers are sufficiently large (i.e. order of hundreds), then these analytical models could be approximately applicable.

As a result of the above conclusions, in the next chapter the computer simulation techniques have been used to investigate the design aspects and performance evaluation of a two node PVS employing a T1-carrier link and 64 kb/s encoder output/channel.

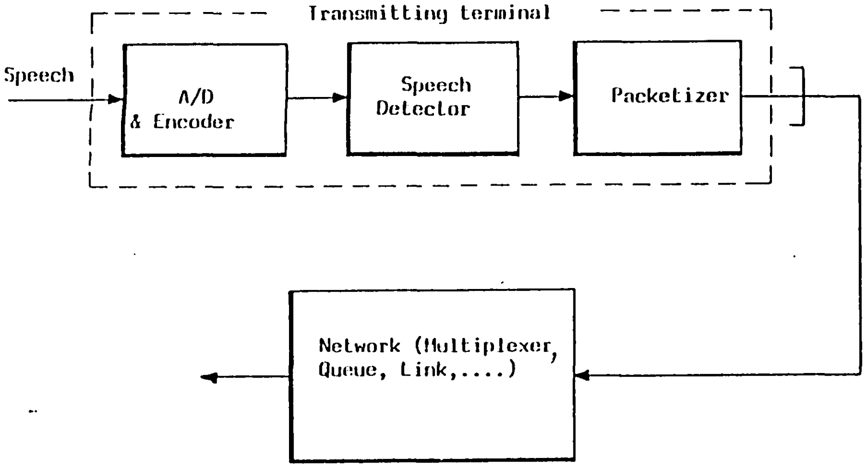


Fig. 3.1 The packetization process for each talker

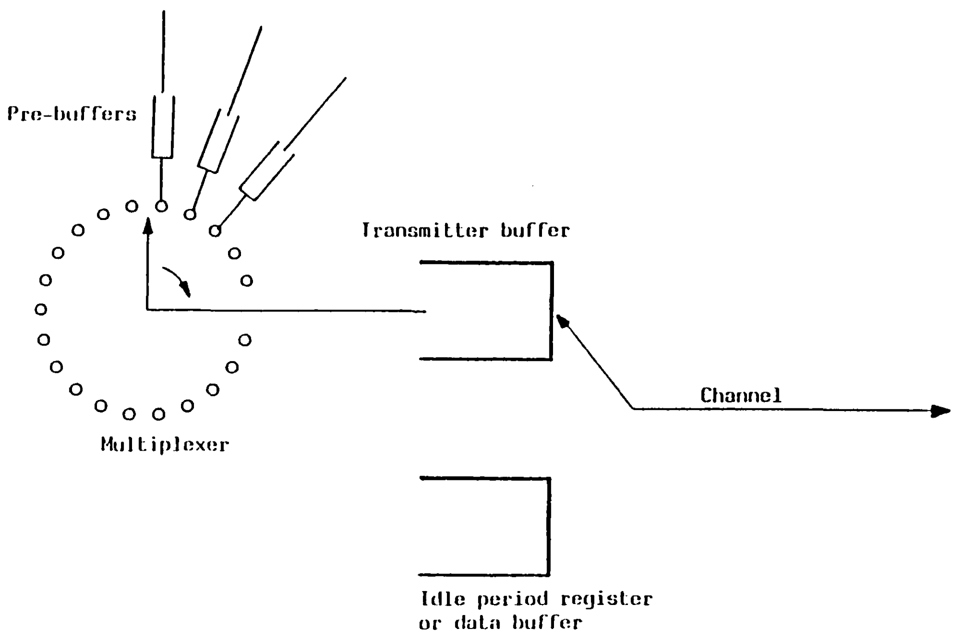


Fig. 3.2 Multiplexing, queueing and transmission aspects of the SQ-system

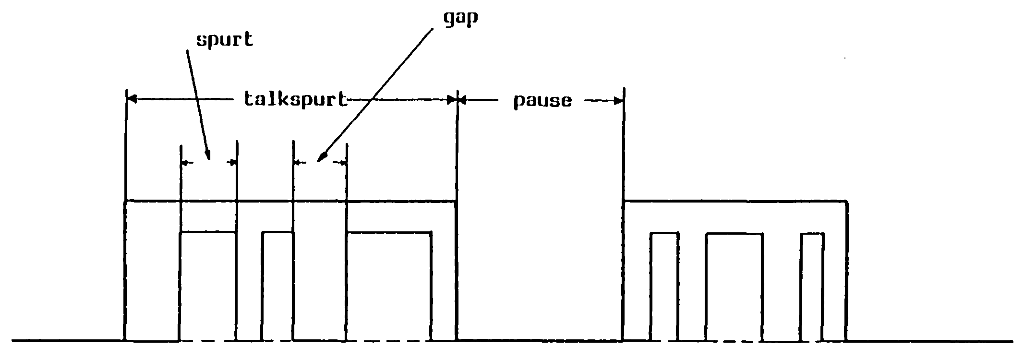


Fig. 3.3 The structure of talker activity model

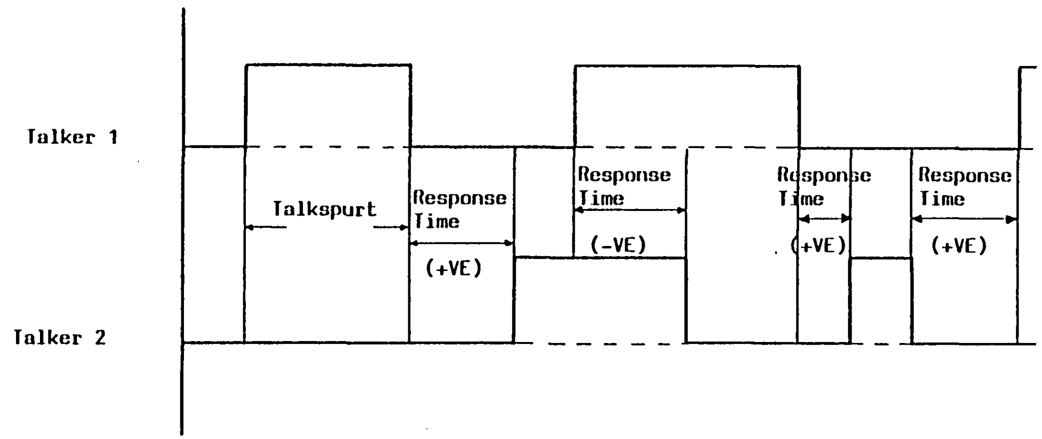


Fig. 3.4 Norwine and Murphy talker activity model

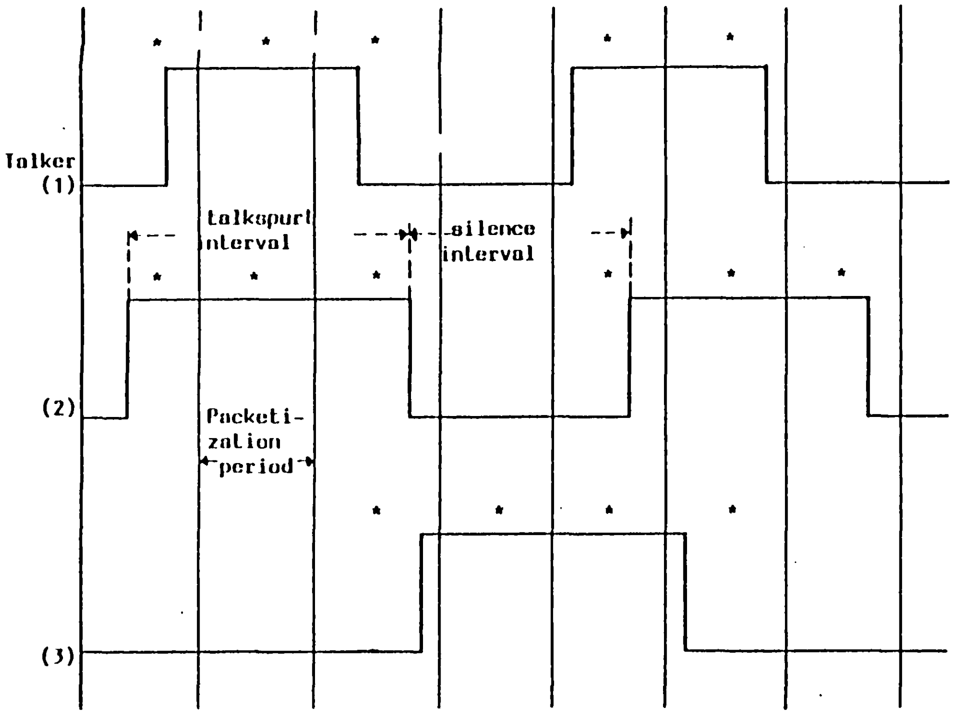


Fig. 3.5 The packetization technique in the simulation. The sign '*' denotes a packet from the associated talker is generated

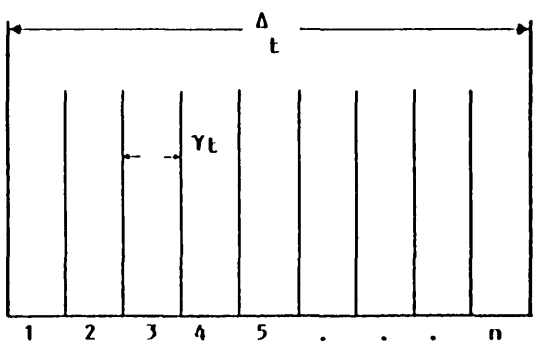


Fig. 3.6 The time division packet multiplexing

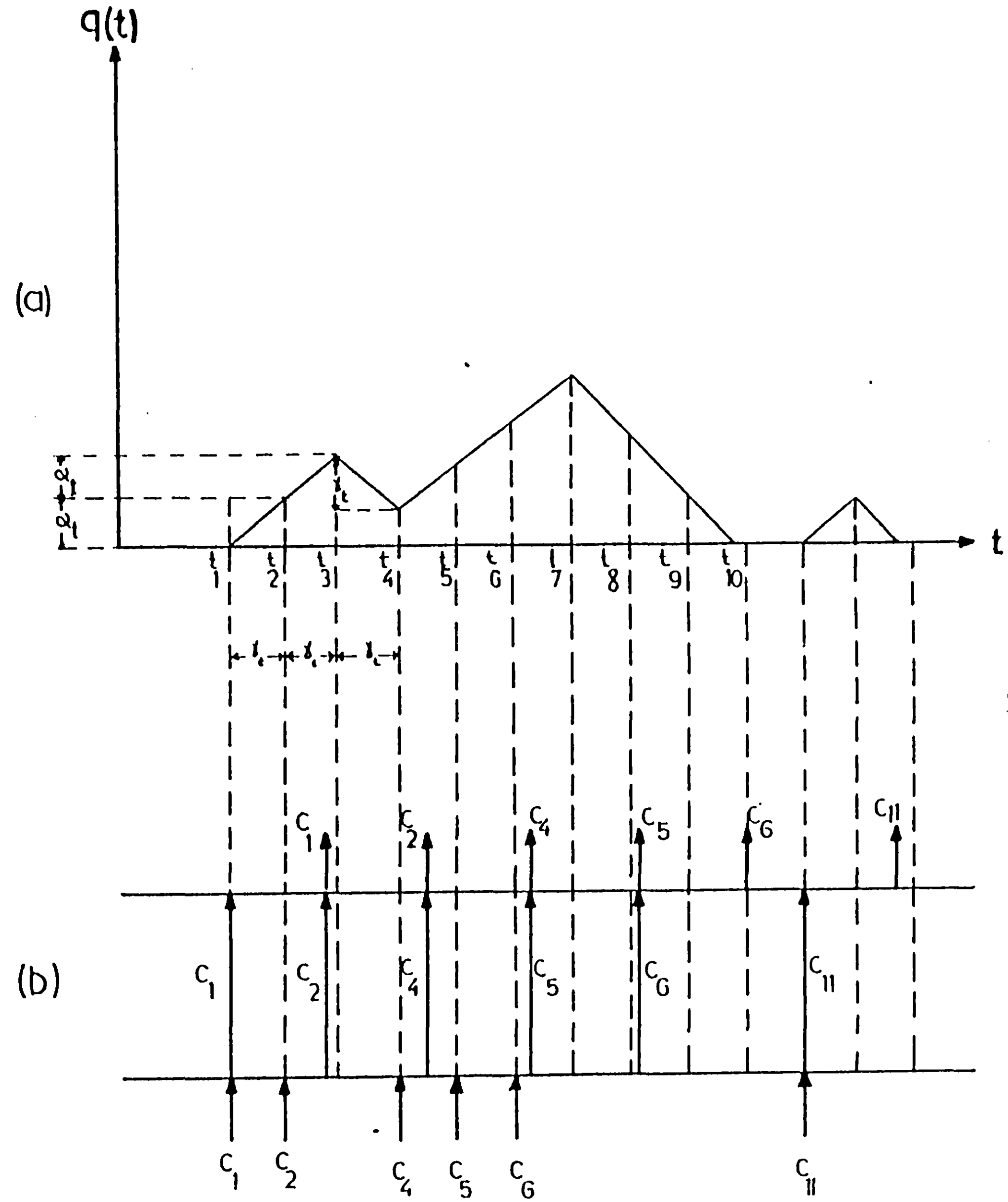


Fig. 3.7 (a) The unfinished work and (b) the packet arrival and departure history

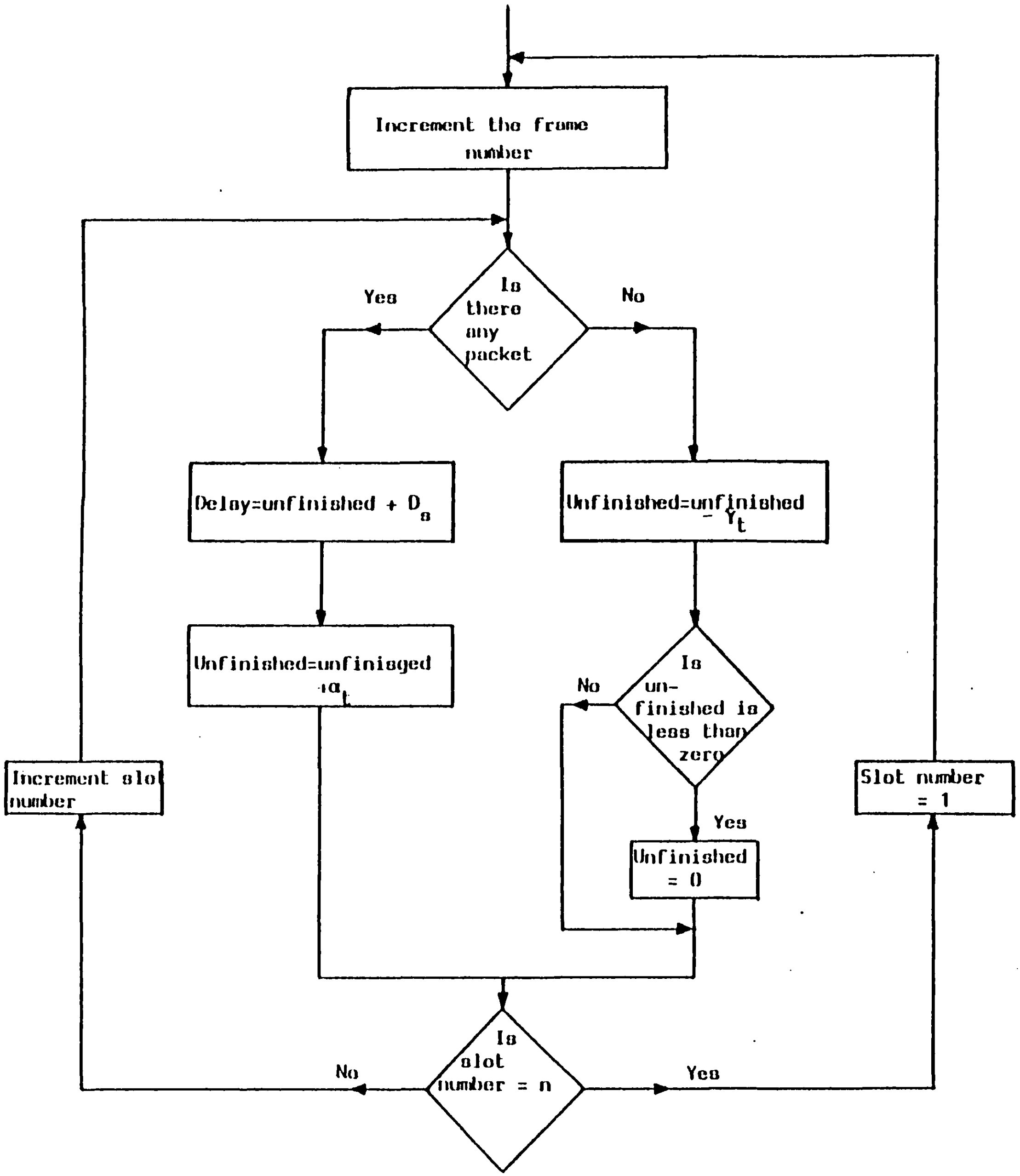


Fig. 3.8 The flow chart of the queueing and transmission process of the packets.

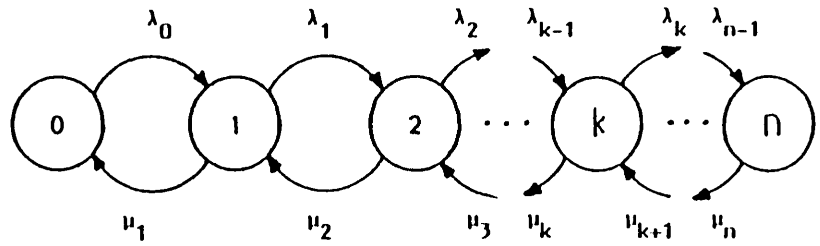


Fig. 3.9 Markovian talkspurt/silence model for the voice traffic

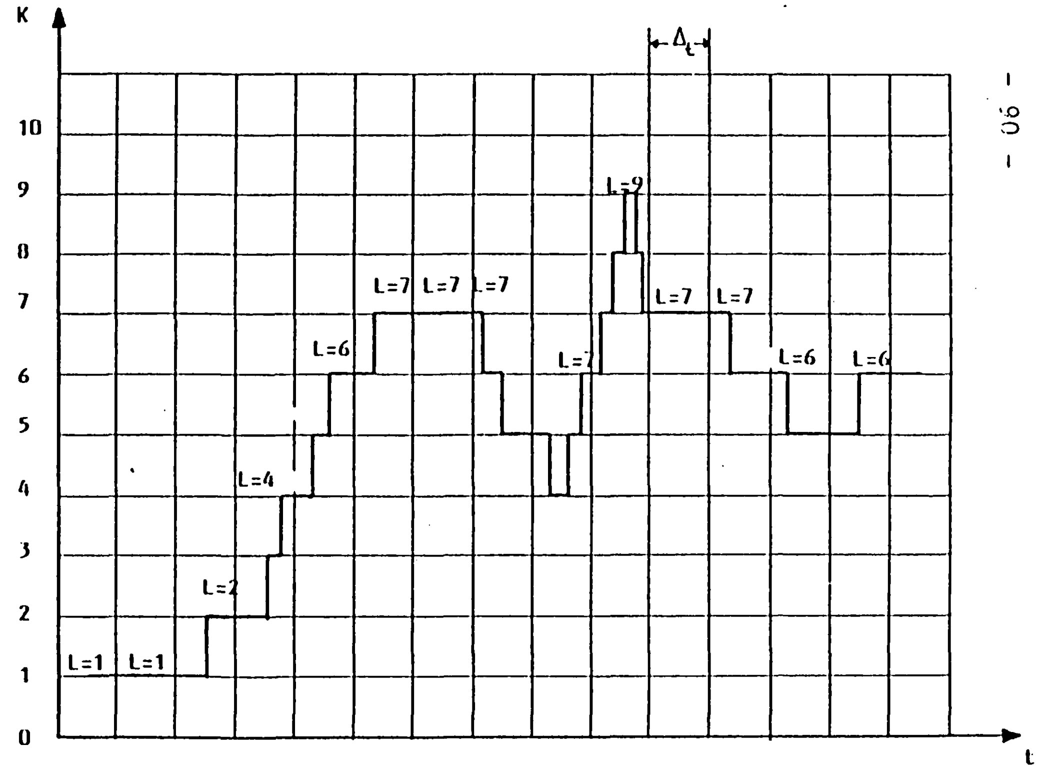


Fig. 3.10 An exemplary Markovian simulation process. L denotes the number of packets being generated within each interval

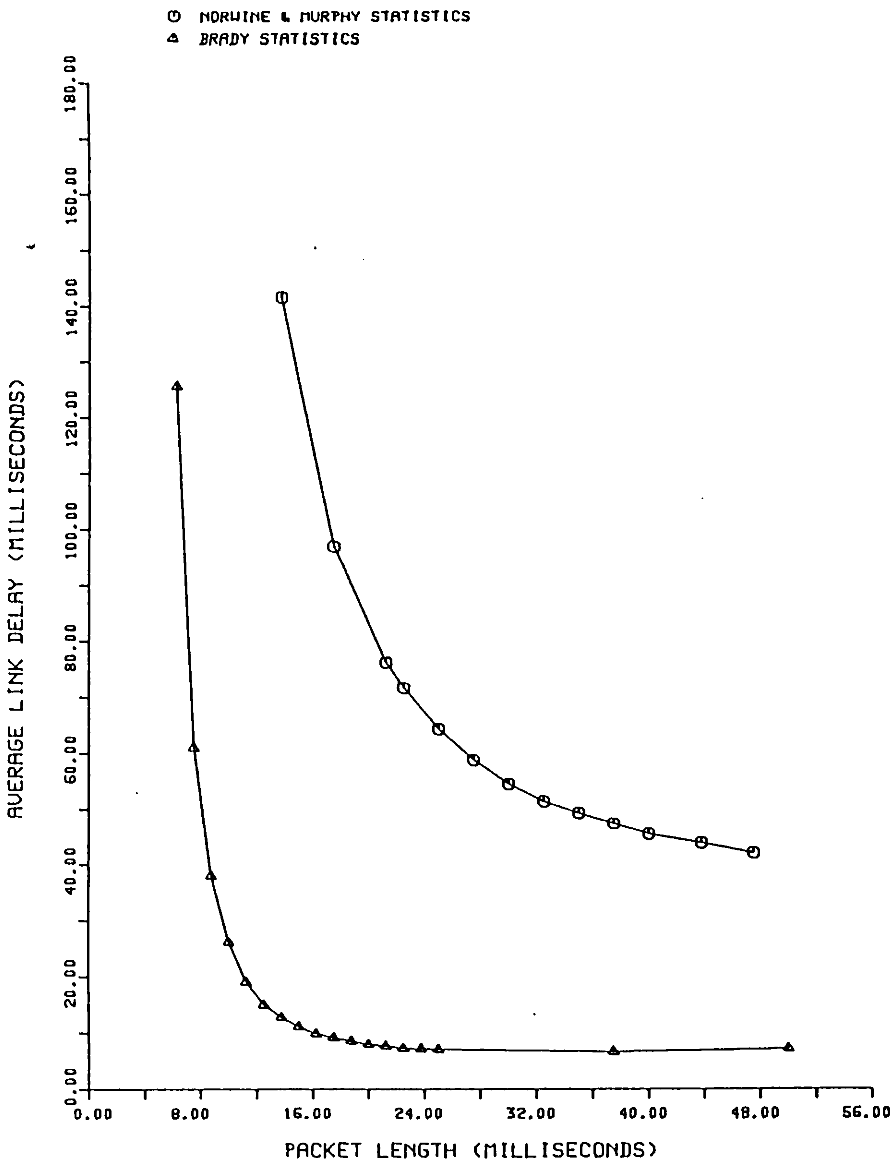


Fig. 3.11 The comparison of different speech sample generation models in the Microscopic Simulation

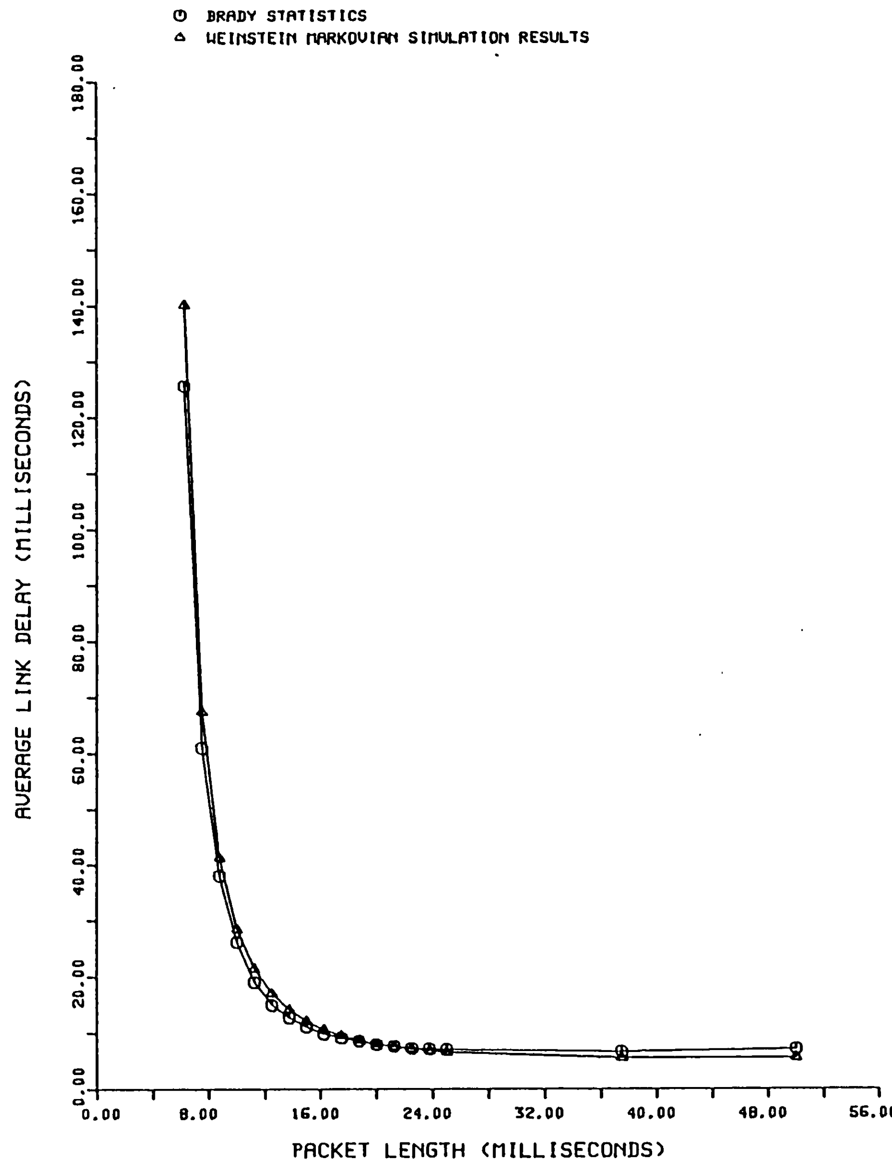


Fig. 3.12 The comparison of the results obtained from the Microscopic and Markovian Simulation models

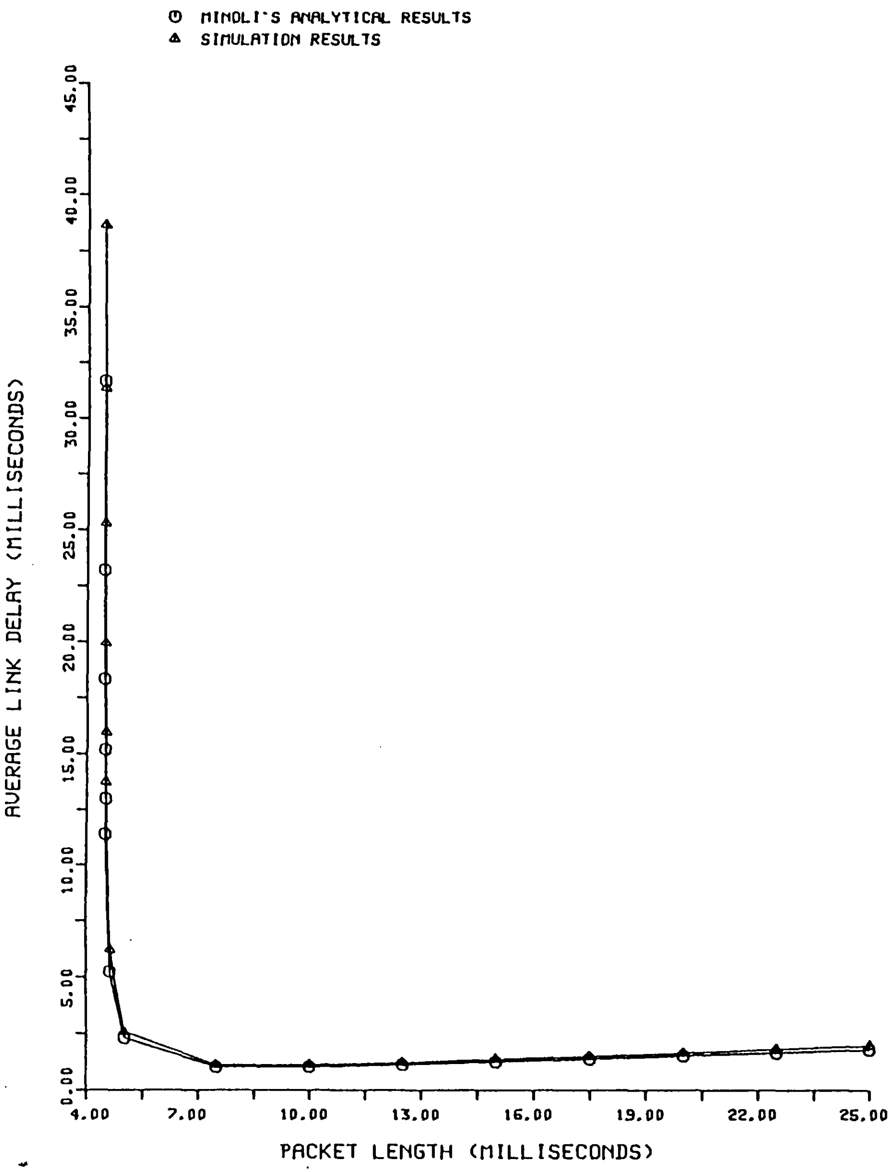


Fig. 3.13 The comparison of the results obtained from the Minoli's analytical model and the simulation model based on Minoli's assumptions

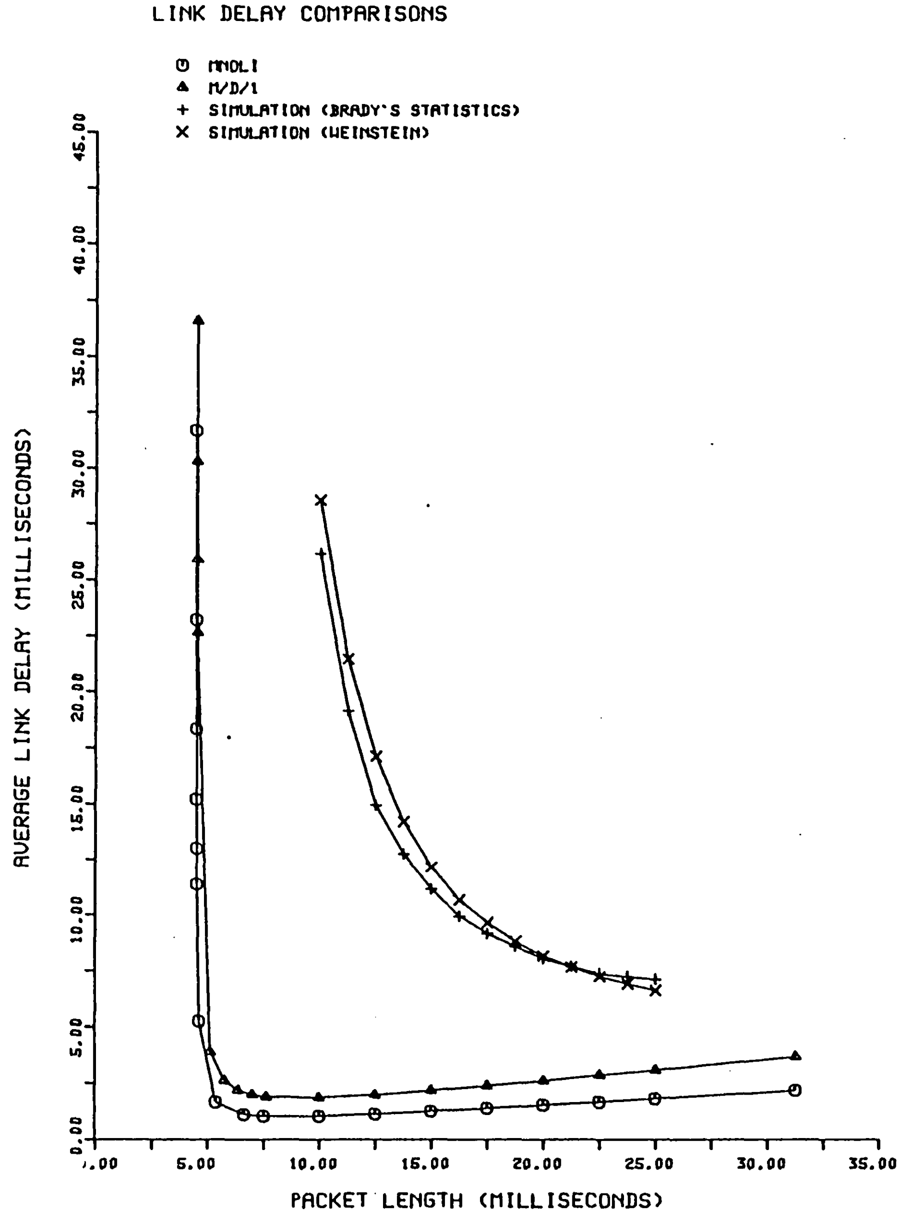


Fig. 3.14 The comparison of the results obtained from the queueing models and the simulation models

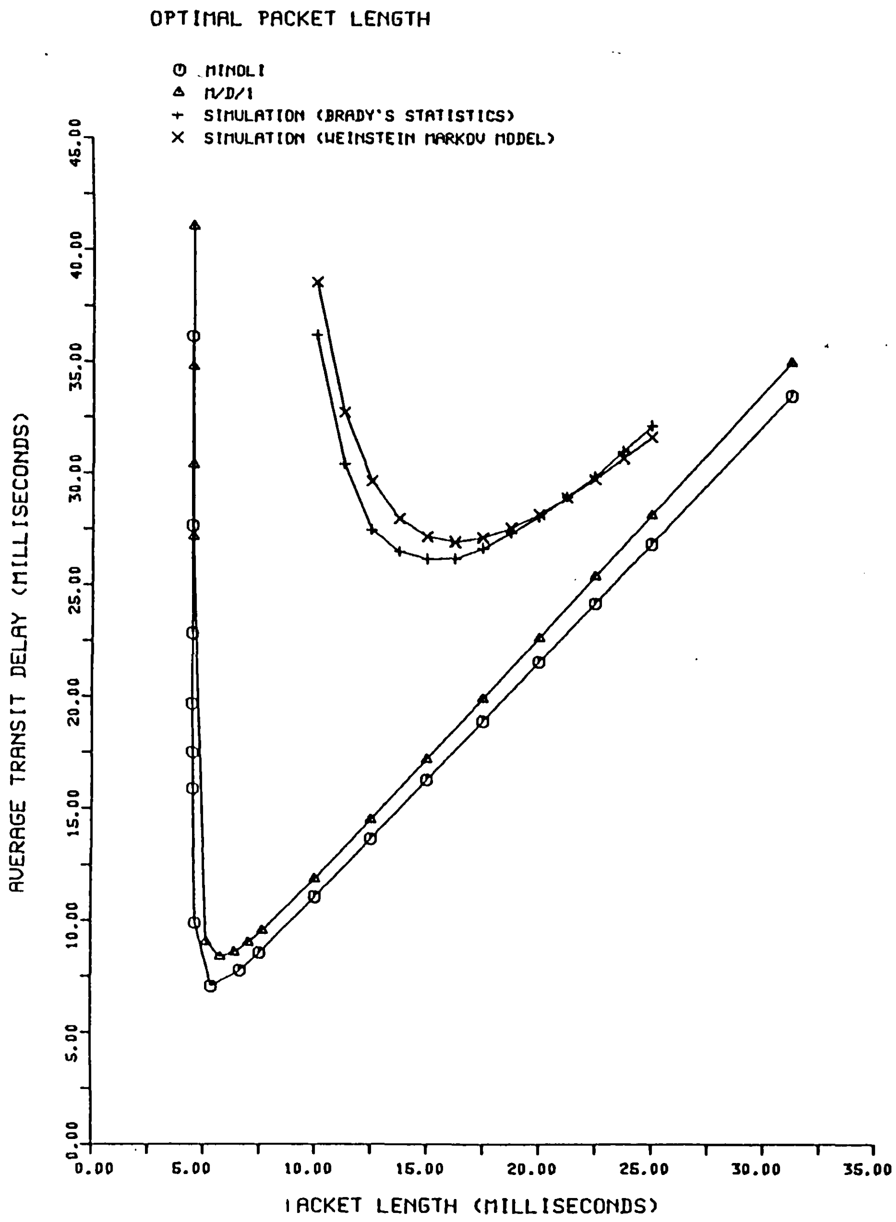


Fig. 3.15 The comparison of the results obtained from the queueing models and the simulation models

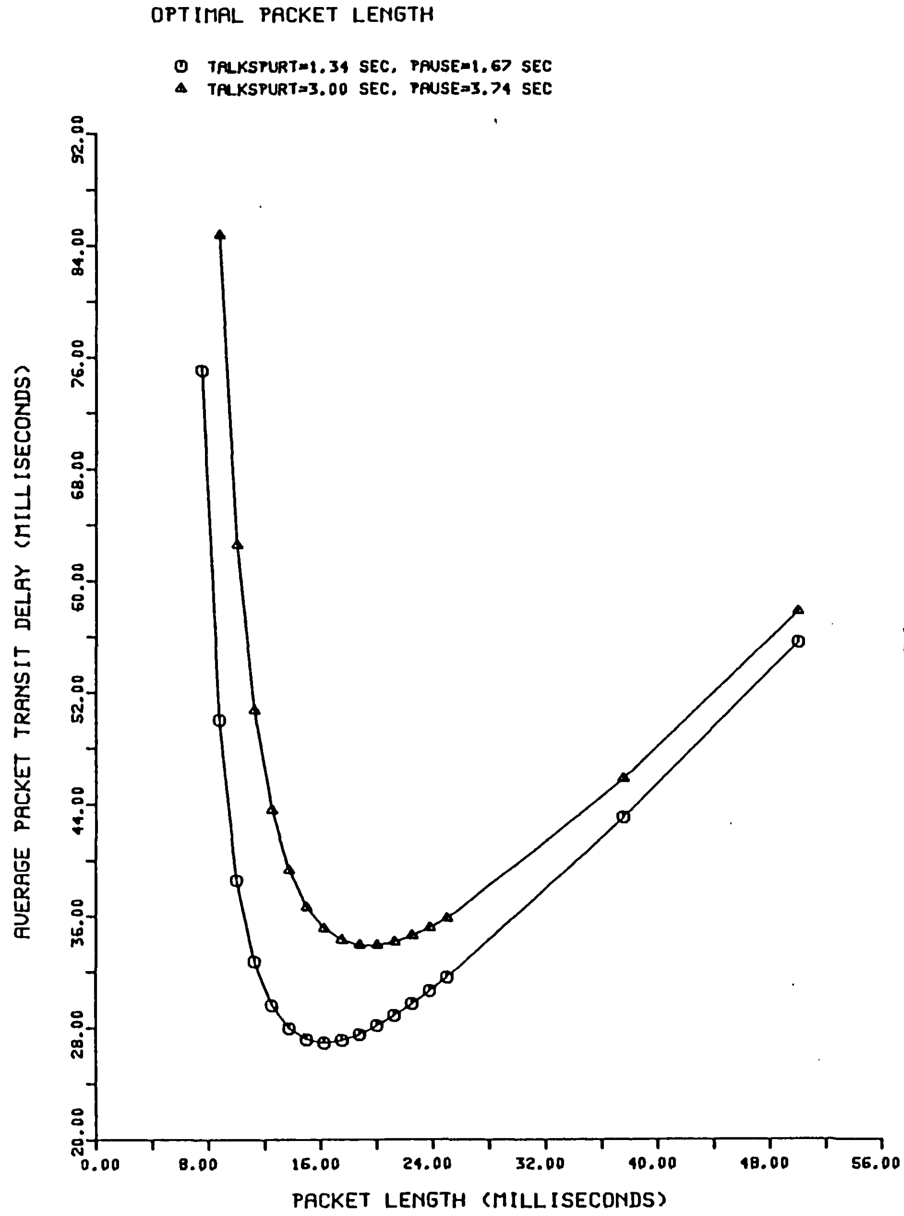


Fig. 3.16 The comparison of the results obtained for different talkspurt and silence intervals. The q value is set at .455

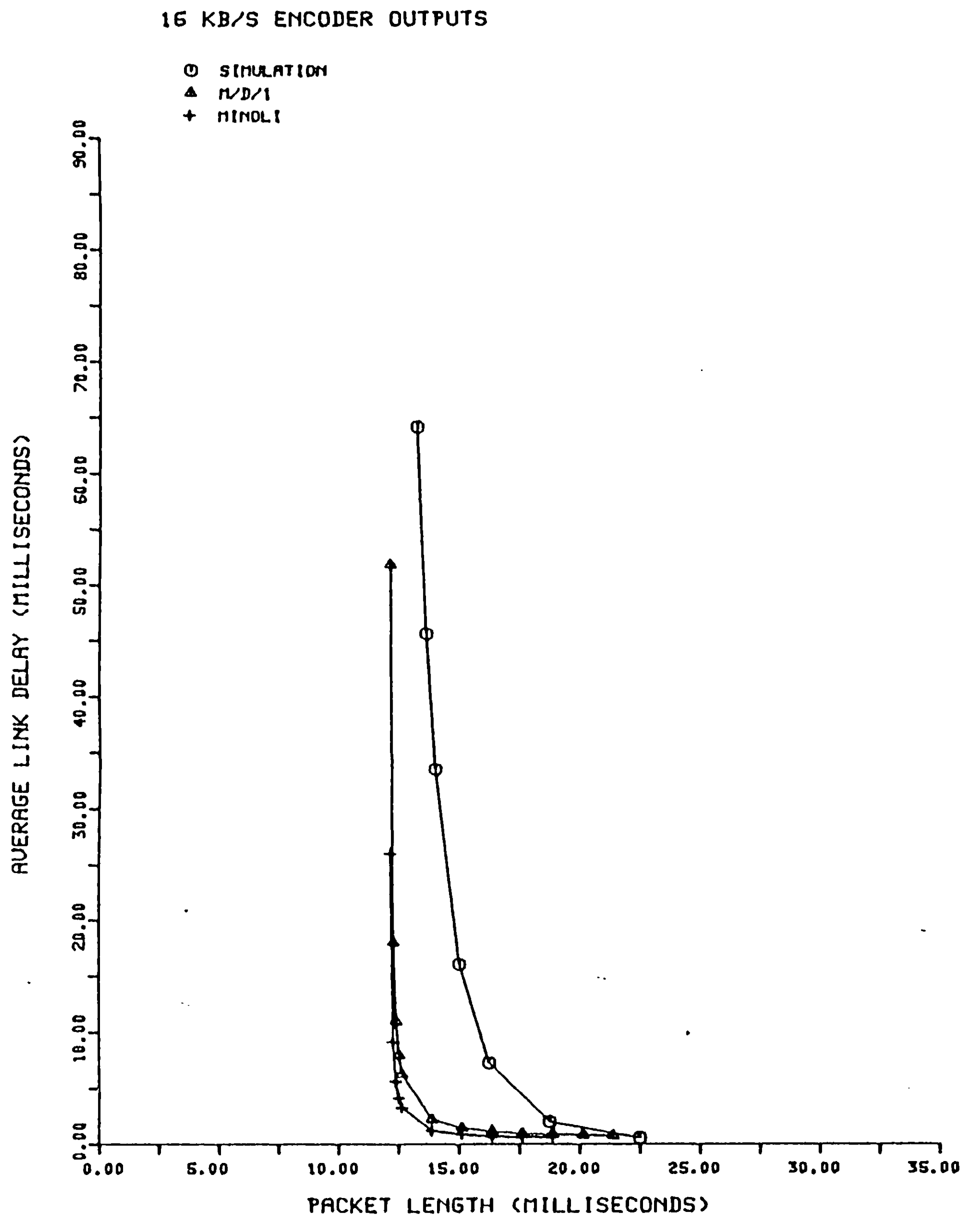


Fig. 3.17 The queueing performance of the system at 16 kb/s encoder output/channel. The number of subscribers is set at 140

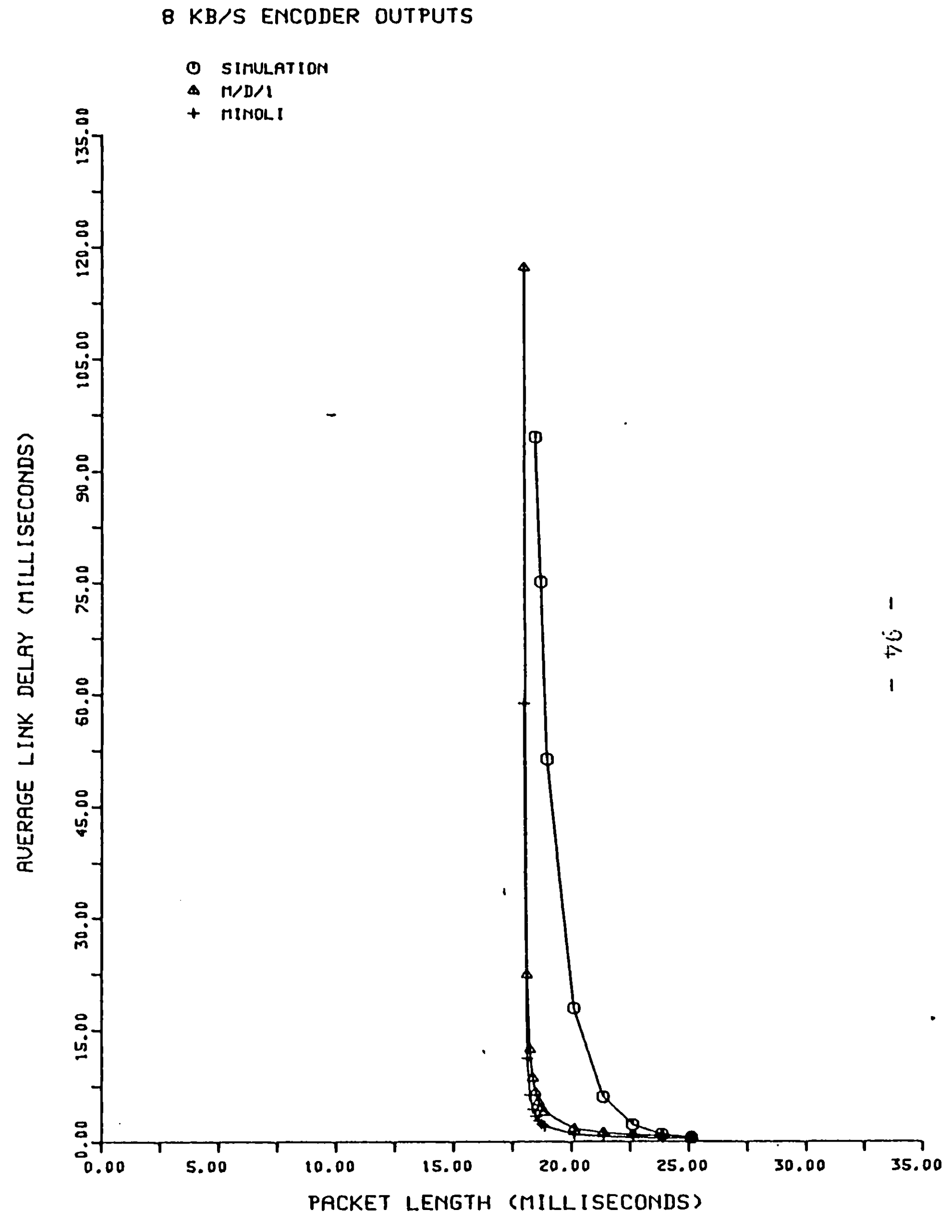


Fig. 3.18 The queueing performance of the system at 8 kb/s encoder output/channel. The number of subscribers is set at 250

CHAPTER 4

DESIGN AND PERFORMANCE EVALUATION OF A TWO NODE PACKETIZED VOICE SYSTEM

4.1 Introduction

In this chapter computer simulation techniques have been used to evaluate the performance characteristics of a two node packetized voice system. Two cases have specifically been covered:

(1) The fully occupied PVS.
In this case, all the input channels are occupied and the talkers are assumed to be in their conversational mode indefinitely. This is not a practical case. However, it does give the worst case design situation for this form of PVS. Results are reported in Section (4.3).

(2) The PVS taking into account telephone traffic loading
This is the practical case where the number of active channels has a probabilistic distribution dependent upon the traffic intensity A. In Section (4.4), the design and performance characteristics of a two node PVS system has been evaluated which has a fixed blocking probability. The results obtained for this case are compared with those for the fully occupied case.

4.2 System description

The block diagram and implementation diagram of a complete two node system are given in figs 4.1 and 4.2. The transmitter part of this system utilizes the same form of packet organization schemes as explained in the previous chapter.

Due to the stochastic behaviour of the queue at the transmitter, the packets for each talker are received at the receiver with variable gaps. These variations of the gaps would distort the speech and would cause significant perceptual degradation in the speech output. One of a number of strategies are required to smooth out inter-packet delays, so correcting the spacing of the packets.

To overcome this difficulty a number of researchers (see Refs [35] - [40]) have suggested different smoothing schemes, or reassembly strategies, which can be employed at the receiver. Each of these schemes incur buffering delays at the receiver. A cause of packet loss is also implicit with each of these schemes. This loss occurs when packets arriving at the packet voice receiver (PVR) are not received by their "play-out" time.

In Ref [35] three possible smoothing, or reasembly, strategies have been suggested. The operation of these three types of reassembly strategies for a two node PVS are explained as follows:

(1) <u>The Null Timing Information (NTI) strategy</u>.

The packet voice receiver delays the first packet of a talkspurt by a given amount of time, T (the control time), and plays out succeeding packets at the same uniform rate as they were generated. If a packet is not received by its play-out time, that packet is considered to be lost. This strategy does not require network synchronization and therefore is easy to implement; however, the end-to-end (ETE) packet delay (D_{ETE}) is variable for each burst of speech.

(2) <u>The Complete Timing Information (CTI) strategy</u>

If the link delay of a packet (i.e. queueing delay plus service time delay) is less than a given control time T, that packet is additionally delayed at the receiver by an amount equal to the control time T minus its link delay, it is then played out in proper sequence and with a fixed delay. A packet with the link delay greater than T is considered to be lost, even if it is the first one of a talkspurt. This strategy requires the terminals to be synchronized, and also requires timing information to be contained in the packet header. However, unlike the NTI strategy, this does keep the D_{ETE} packet delay constant at a value equal to $T+D_Z$; where D_Z is the packetization delay. The question of network synchronization in this strategy can be overcome by using a simple method of time stamping upon each packet as suggested by Montgomery [40]. In this scheme, there are clocks at the transmitter and the receiver which are synchronized to the same absolute time reference. In this case each packet, after being generated, carries an indication of its generation time. When the packet is received, the PVR substracts its receiving time from its generation time and uses this (which represents the link delay) to compute its play-out time.

(3) The NTI-CTI Mixed Strategy.

If the link delay of the first packet of the talkspurt is less than a given control time T, then that packet and successive ones are played out in the same way as that in the CTI strategy. If the link delay of the first packet is greater than T, the packet voice receiver plays out that packet immediately upon receiving it, and continues to play out successive packets at the same uniform rate as they were generated. Packets which are not received by their play-out times are considered to be lost. This strategy, as with the CTI strategy, requires the network to be synchronized. However, the D_{ETE} packet delay is variable.

Comparisons of these three strategies were carried-out by

Suda et al [51] using computer simulated speech traffic. They showed that the CTI and the NTI-CTI mix strategies were superior to the NTI strategy when considering the silence fluctuations, hence they predicted that the CTI and NTI-CTI methods would result in better speech perception.

The CTI strategy is considered to be the most suitable for use at the PVR buffer because of:

(1) Its better performance achievement in terms of silence fluctuations.

(2) This strategy results in the D_{ETE} being constant. This latter property, as will be shown later, makes it possible to control the D_{ETE} in the system. It also results in easier evaluation of the PVS performance.

The CTI strategy has been applied to both the PVS systems i.e where the transmitter buffer has either an infinite, or a limited queue buffer length. Fig. 4.3(a) portrays the timing diagram of this strategy applied to the transmitter queue buffer which has an infinite length. It shows that there are packet losses, and these occur when the instantaneous link delay of a packet, denoted by W(t), has exceeded the control time T at the receiver. Since the D_{ETE} is fixed and is equal to $T+D_Z$, then the control time T can be used at the receiver to control the D_{ETE} to any required value. This value is given by

$$T = D_{ETE} - D_Z \qquad (4.1)$$

where D_z in the above equation is the packetization delay.

Fig. 4.3(b) depicts the timing diagram of the CTI strategy applied to the finite queue buffer situation at the transmitter for a two node packetized voice system. In this form, the opposite loss situation results. Here there are no packet losses at the receiver, instead there are packet losses at the transmitter. This operation has been achieved by simply limiting the size of the transmitter buffer so that the maximum packet link delay, $\hat{W}(t)$, is made equal to the control time.

$$T = \hat{W}(t)$$

or

$$T = \hat{q}(t) + D_s. \qquad (4.2)$$

In the above equation $\hat{q}(t)$ denotes the maximum allowed queueing delay, and hence the size of the transmitter buffer required; D_s is the packet service time which is given by equation (3.11).

By substituting the value of T from the above equation into equation (4.1), the size of the transmitter buffer required, $\hat{q}(\ell)$, can be obtained. This is given by

$$\hat{q}(\ell) = c\hat{q}(t) = (D_{ETE} - D_z - D_s)c \qquad (4.3)$$

In order to compare the effects of packet losses experienced in the two systems, both systems have been simulated using the same system parameters as were used in the previous chapter (i.e. C=1.544 Mb/s, n=40, h=100 bits and 64 kb/s encoder output/channel).

Both the Microscopic and the Markovian simulation approaches were used for simulating the system. Fig. 4.4 depicts the characteristics of the mean percentage of packet losses versus packet length for the two systems with and without infinite queue buffering. For both systems, the control parameters using equations 4.1 and 4.3 were set so that the D_{ETE} was fixed at a constant 250 ms. The mean percentage of packet losses was measured by taking the ratio of the total packets lost to the total packets generated for the situation when all connected subscribers (40) were in active conversation. The results indicate that signficantly higher packet losses result with the infinite queue buffering scheme. This is because the finite queue buffer does not allow large numbers of packets to build up in the queue. Any packet whose queueing delay contributes to a D_{ETE} greater than 250 ms is rejected and so it does not cause an increase in the queue length.

Fig. 4.4 also shows that for the finite buffering situation similar results are achieved by the two simulation models. However, for the infinite queue buffering situation, a better approximation can be achieved by using the Markovian simulation. This is because in the infinite buffering scheme, there is a greater variation in the queue behaviour than in the finite buffering scheme. Thus the Markovian simulation technique, where the simulation period is long (i.e. 20,000 secs) would give a better approximation than the Microscopic simulation with the smaller simulation period (i.e. 700 secs). Hence in this thesis the rest of the work presented to assess the performance of the PVS will be based on use of the Markovian simulation model.

4.3 Design aspects of the system

In this section the evaluation of the performance characteristics of a two node system are investigated with the assumption that the system is fully occupied (i.e. always n talkers).

In general, the performance of the system using the CTI strategy is a function of various parameters, e.g. channel rate (c), number of subscribers (n), packet length (p), overhead (h), voice encoding rate (b), the percentage of packet losses (P_L), and the ETE delay (D_{ETE}). These are summarized in the following equation.

$$\text{Performance} = f(c,n,p,h,b,P_L,D_{ETE}). \qquad (4.4)$$

Since the computer simulation technique has been used for the performance evaluation of the system, it is expensive to propose a generalized design where all the above seven parameters are variables. This is one of the disadvantages of using a computer simulation approach. However, as previously, by making the following assumptions the performance of the system can be simplified (i.e. c = 1.544 Mb/s, h = 100 bits and b = 8 bits/sample or 64 kb/s encoder outputs per subscriber channel). Under these assumptions, the performance of the system now becomes a function of the following four parameters:

$$\text{Performance} = f(n,p,P_L,D_{ETE}) \ . \qquad (4.5)$$

The informal listening tests by Jayant and Christensen [48], and also the subjective tests by Musser et al [49] have shown that the maximum percentage of lost packets tolerated by the listeners in conversation could be as high as 1%, in these situations the packet absence would not be noticable. This, was based on not allowing the packet size to exceed 50 ms and the lost packets at the receiver being substituted either by zero values (i.e. a zero mean speech signal) or by the previous played out packet. The former case is called the zero-amplitude stuffing technique and the latter case is called the packet recirculation technique.

Taking into account the above findings; the mean percentage of packet losses, P_L, in equation (4.5) has been set at 1%. This assumption therefore makes a further simplification so that the performance of the system now becomes a function of three parameters, as follows:

$$\text{Performance} = f(n,p,\ D_{ETE})\ . \qquad (4.6)$$

By using the above assumptions, two forms of design criteria are presented which are based on having either D_{ETE} or n fixed.

4.3.1 When D_{ETE} is fixed

As shown earlier, it is possible to control the ETE packet delay in the CTI strategy to any constant value. This can be achieved either by setting the control time T, using equation (4.1) in the infinite transmitter queue buffering scheme; or by setting the length of the transmitter buffer, using equation (4.3) in the finite transmitter queue buffering scheme. Under these cases, the performance of the system will become a function of two parameters given by

$$\text{Performance} = f(n,p) \ , \qquad (4.6)$$

where the packet loss percentage is ≤ 1 and the ETE packet delay is set at a fixed constant value.

Fig. 4.5 gives the results obtained employing the arguments above, when D_{ETE} is constant (250 ms). This being approximately the maximum allowable ETE delay for a smooth conversational speech, according to [46], [47]. Two cases of infinite and limited transmitter queue buffering schemes have been considered. The horizontal axis of this figure is for variation of packet length and the vertical axis depicts the normalized value of the number of subscribers (i.e. n/24) when $P_L \leq 1\%$, also defined as the TASI advantage. As expected, a better TASI advantage is achieved with the finite buffering scheme. It is also evident that when the packet length is short, the TASI advantage is low and it rises rapidly for small increases in the packet length. This is due to the effect of the overhead on the short packets. For a packet length greater than 16 ms the TASI advantage becomes steady and maximized, this corresponds to the minimization of the effects of the overhead at this range.

Fig 4.6 shows the same type of characteristics obtained when the D_{ETE} is set at three times of packetization delay (i.e. $D_{ETE} = 3D_Z$). These particular characteristics, given in fig. 4.6, will be used in the next chapter for comparison with another type of packetized voice system.

4.3.2 When n is fixed

In this situation, the performance of the system becomes a function of two parameters given by

$$\text{Performance} = f(D_{ETE}, p). \tag{4.7}$$

The characteristic of D_{ETE} versus packet length p is required at this stage, so that the mean percentage of packet losses, P_L, at each point on the characteristic is kept just equal to 1%. This is done as follows:

Using the CTI strategy, the D_{ETE} is given by

$$D_{ETE} = D_Z + T. \tag{4.8}$$

In the above equation, D_Z is constant and represents the packetization delay. However, the control time T can be variable. We define the optimal control time here as the delay applied in the system so that the mean percentage of packet losses just equals 1%. Fig. 4.7 shows this situation for a two node PVS with finite queue buffering. The value of n is set at 40 and the packet length p is equal to 70 bytes or 8.75 ms. As is shown, the optimal control time corresponds to 25.7ms. This procedure was then repeated for different sets of packet lengths, and for each value of packet length its corresponding optimal control time was obtained so that it gave P_L just equal to 1%. These results, together with the results obtained for the packetization delays,are depicted in fig. 4.8. The summation of these two sets of results (i.e. optimal control times and packetization delays) then give the required characteristic of D_{ETE} versus packet length and are as depicted in fig. 4.8. From this characteristic it can be observed that there is a packet length which just minimizes the end-to-end delay. This corresponds to the packet length of 11.25ms and sets the D_{ETE} at 14.5 ms. Finally by using equation (4.3), the size of the transmitter buffer required for this situation is found to be 4149 bits ($\equiv$ 2.7 ms).

The same procedures have been carried out for the case when n = 45. The characteristic of D_{ETE} versus packet length for this situation is depicted in fig. 4.9. The optimal packet length corresponds to 33.75ms and this gives an end-to-end delay of 93.213ms. By using equation (4.3), the size of the transmitter buffer in this case is found to be 89552 bits ($\equiv$ 58 ms).

This optimization technique, and the results relating to it, as shown in figs. 4.8 and 4.9, differ from the technique suggested by Monoli [20] because:

(1) It is based on the characteristic of the total D_{ETE} versus packet length,

(2) the characteristic also contains the locus points of the mean percentage of packet losses set at 1%.

These combined assumptions are based on reported acceptable end-to-end delay maximum being 250ms and 1% loss of packets. Subjective tests will be necessary to determine the acceptability of these joint assumptions. Practical systems may need tighter values than have been used here.

4.4 Effect of telephone traffic loading

In all of the simulations of packet switched system presented so far, the number of participating talkers occupying channels has been kept fixed and not allowed to fluctuate (i.e. always n talkers). The intention was to simulate the worst case of system operation. The work has aimed at keeping the packet losses at less than 1% and total delays less than 250 ms.

In reality, there are fluctuations in the number of active calls unless the system is working at a very high traffic intensity and severe blocking is occurring. With lower loading, channels may be inactive, thus more time slots will be available than there would be in a fully occupied system. The aim of this section is to investigate the performance of the PVS when the advantages of the fluctuations in the number of voice calls is taken into account. This is likely to give a better average performance, or permit many more channels to be connected for the same performance criteria.

Generally, the offered telephone traffic load is defined as the average voice call arrival (λ_1) times the average voice call holding time ($1/\mu_1$). This is denoted by A and its unit is measured in Erlangs.

$$A = \lambda_1/\mu_1 \qquad \text{Erlangs} \tag{4.9}$$

Fig. 4.10 shows the block diagram of a PVS with the inclusion of voice call traffic via a switch. In this form, the offered load traffic, A, which arrives at the transmitting node is switched into n channels and fed into the PVS. When more than n talkers demand service, the system refuses service. The probability of this occurring is called the blocking probability, or grade of service (GOS). The blocking probability, P_B, is affected by the number of output channels (n) and the total offered traffic (A), and is evaluated by the Erlang loss formula.

$$P_B = \frac{\frac{A^n}{n!}}{\sum_{i=0}^{n} \frac{A^i}{i!}} \tag{4.10}$$

The other useful information which the blocking probability gives, is that it represents the fraction of time the system meets the condition when all the n channels are fully occupied. For example, if the blocking probability is 2%, this means that for only 2% of time is the system fully occupied, (i.e. all n channels occupied). For the other 98% of time there are free channels available (i.e. less than n channels being occupied). In order to evaluate the performance characteristics of a PVS which carries a telephone traffic load (A) and with a fixed blocking probability (P_B), a computer simulation model has to be introduced. This model is described in the next subsection.

4.4.1 Computer simulation model

This model has been developed by combining the previously described Markovian simulation model for the PVS, with a voice traffic simulation model. These combined models are dipicted in fig. 4.11. The voice traffic simulation model, as shown in fig. 4.11(a), is based on having a Poisson call arrival process of rate λ_1 and an exponential call holding time with mean $(1/\mu_1)$. This model determines the voice traffic variations which arise from call initiation and termination and allows up to a maximum of n calls to be active at any one time with blocking. The way in which a simulator has been extracted from these models is as follows:

Let us assume that $n_v(t)$ represents the number of active calls in progress at a given time. Assuming that $n_v(t)=k$ at a particular time, then $n_v(t)$ is held constant at k for a time τ drawn from an exponential distribution with mean τ_k, determined as

$$\tau_k = 1/(\lambda + k\mu) \quad k = 0,1,\dots,n-1$$
$$\tau_n = 1/n\mu. \qquad (4.11)$$

After a time τ, $n_v(t)$ is increased to k+1 with probability

$$P_{up}(k) = \lambda/(\lambda + k\mu) \qquad k = 0,1,\ldots,n-1$$
$$P_{up}(n) = 0 \tag{4.12}$$

or decreased to k-1 with probability $1 - P_{up}(k)$. This process is repeated as often as desired to generate sample functions of $n_v(t)$.

The PVS Markovian simulation model, as already explained in the previous chapter, is conjoined with the above voice traffic model, with necessary modifications as shown in fig. 4.11(b). That is, the total number of states of the chain which represents the maximum number of talkspurts and the transition rates from talkspurt to silence, or visa versa varies with the number of active calls in the system ($n_v(t)$) at a particular time.

4.4.2 Results and comparison

By using the above computer simulation model, tests have been carried out in order to ascertain the effects of telephone traffic loading upon the TASI advantage performance characteristic of the PVS. In the simulations the mean call duration, $(1/\mu_1)$, has been set at 100 seconds. Only the PVS with the finite buffering scheme has been considered and the end-to-end delay was set at 250 ms constant delay. The blocking probability, P_B, has been assumed to be 2%. The other system parameters such as c,h and b were assumed to be the same as their previous values. Under these assumptions, five experiments have been carried out and are reported as follows:

Experiment 1.

In this experiment the characteristic of the TASI advantage versus packet length has been obtained when the transmitter buffer overflow probability, or the overall percentage of packet losses, P_L, was less than or equal, to 1%. Since it has been assumed that the blocking probability is always equal to 2%, then this implies that for each value of n(or TASI advantage = n/24) a voice traffic intensity (A) can be found which meets the 2% blocking condition. In order to evaluate such voice traffic intensities, the equation (4.10) has been used and is rewritten as follows:

$$P_B = \frac{\frac{A^n}{n!}}{\sum_{i=0}^{n} \frac{A^i}{i!}} = .02 \qquad (4.13)$$

By using the False-position method (see Ref [81]) for each value of n, the corresponding value of A has been found from the above equation which sets the blocking probability to 2%. Then by fixing the mean call duration ($1/\mu_1$) to 100 seconds, the call arrival rate (λ_1) has been evaluated by equation 4.9. These parameters were set for runs of the simulation program. The program was run to simulate 60,000 seconds of real time. The mean percentage of packet losses, P_L, have been evaluated from the ratio of total packets lost to the total packets transmitted. If P_L was found to be greater than 1%, then the value of n was reduced and the same procedures as above were repeated until the condition where P_L is just less than or equal to 1% was met. Fig. 4.12 shows the characteristic of the TASI advantage (= n/24) versus packet length which has been obtained for this situation. This characteristic represents the locus of the points where the mean percentage of packet losses is $\leq$ 1% and the blocking probability is equal to 2% (i.e. 2% of the calls which arrive at the exchange are blocked and are lost). The offered voice traffic intensity, A, which is required in order to give this performance is observed in fig. 4.13. Fig. 4.14 and table 4.1 both show comparisons of the above results with the results which have already been obtained for the fully connected system (i.e. the results of fig. 4.5). It is evident that the PVS's which have been investigated in this Section (i.e. finite transmitter buffer and CTI type receiver with realistic voice traffic loading) have a considerably better TASI advantage performance than that predicted by the fully loaded situation.

Table 4.1

Packet length	n Fully connected	n Traffic loading with 2% blocking	TASI Fully connected	TASI Traffic loading with 2% blocking
6.25	40	46	1.6667	1.9167
10.00	43	49	1.7917	2.0417
13.75	45	51	1.8850	2.125
17.5	46	52	1.9167	2.1667
21.25	47	53	1.9583	2.2083
25.00	47	54	1.9583	2.25
50.00	47	54	1.9583	2.25

Experiment 2:

The result of the previous experiment have been based on the assumption that the transmitter buffer overflow probability is just less than, or equal to, 1%. In this experiment we investigate the cummulative distribution functions of packet losses for all callers when the traffic has been increased to take advantage of the improved TASI advantage but still ensuring that the average freezeout is just less than 1%. This task has been carried out by evaluating the percentage number of lost packets at simulation intervals of 100 sec. The cumulative distribution functions (CDF) of the percentage of freezeout have been obtained for a set of different packet lengths and significant points have been listed in table 4.2. Fig. 4.15 depicts these results for the 10 ms packet length. It can be observed that 76.5% of the calls in this case would experience less than, or equal to, 1% packet loss.

Table 4.2

Packet length ms		6.25	10	13.75	17.5	21.25
% packet loss	P_L	CDF	CDF	CDF	CDF	CDF
	1%	.758	.765	.756	.768	.746
	2%	.865	.892	.883	.894	.859
	3%	.949	.963	.949	.956	.944
	4%	.983	.986	.984	.985	.976
	5%	.993	.995	.996	.995	.991
	6%	1.00	.998	1.00	.998	.998
	7%	1.00	1.00	1.00	1.00	1.00

Table 4.2 also shows that the CDF results obtained for the different packet lengths are in reasonable agreement with each other. The conclusion that can be drawn from the two above experiments is that using the 1% overall buffer overflow assumption as a reasonable speech quality criteria, the number of connections to the system can be increased, so effectively giving an increased TASI advantage. However with the extra systems connected and keeping to the 1% average buffer overflow the percentage of calls experiencing greater than 1% will be increased - in the examples of packet lengths given this is as high as 24% of calls.

Experiment 3.

In this experiment, the characteristic of TASI advantage versus packet length has been obtained for the worst case freezeout condition when the traffic loading gives a 2% blocking probability. This means that the freezeout measure, P_L, has been set equal to the ratio of total number of packets lost to the total number of packets transmitted but only for the periods that the channels are all occupied (i.e. all channels are in conversational mode). This ratio was set at $\leq$ 1%, and the resultant characteristic has been obtained and is depicted in fig. 4.16, this can be compared with the characteristic of fully connected system. It is evident that the two characteristics are nearly identical. This comparison therefore shows that the system will experience the same degree of freezeout as is obtained from the evaluation of the fully connected system. Thus the effect of fluctuations in the number of voice calls does not alter the worst case of system operation under freezeout. The reason for this is because of the differences in the call fluctuations and talkspurt-to-silence (or vice versa) fluctuations. The number of calls active in the system varies slowly taking seconds to change, while the talkspurts vary rapidly, even over milliseconds (see equations 3.17 and 4.11). For each connection state, when the number of calls is constant for some seconds, there could be hundreds of talkspurt intervals (and hence many more packets) so 'steady-state' conditions soon occur for that particular connection state. this is why the TASI advantage results, which were obtained for the worst case freezeout, can be approximated by the always fully connected case.

Experiment 4.

The conclusion which can be drawn from the three experiments described above is that the fluctuations in the number of voice calls on the PVS (e.g. the free slots) should not be used for increasing the TASI advantage of the system. The reason, as found in experiment 3, is that the variations due to call length are much slower changing than the variations of the talkspurt durations. However, the experiments of 1 and 2 have shown that the TASI advantage can be increased substantially, but at the cost of letting 24% of the calls have greater than the 1% freezeout, so causing the received speech to be more noisy. From a network designer's view point this high percentage of noisy call connections may be too high. In this experiment we evaluate the characteristics of the TASI advantage versus packet length for channels with a lower percentage of noisy call connections, namely less than $\leq$ 5% and

$\leq$ 10% of channels have more than 1% freezeout. This criteria implies that at least 95% and 90% respectively of the time call connections are good. In order to evaluate these characteristics the buffer overflow assumptions as the performance criteria has been ignored and instead the proportion of good call connections has been taken as the criteria.

The good call connections have been assumed to be the calls whose percentage of freezeout do not exceed 1%. Fig. 4.17 shows the comparison of the characteristics of the TASI advantage versus the packet length which have been obtained using the above performance criteria for the two cases of having at least 90% and 95% of good quality call connections. It can be observed that as the requirement for more good call connections increases the TASI advantage of the system decreases and it approaches the characteristic of the fully connected PVS. The conclusion which can be drawn from this experiment is that even with the minimum of 95% of the good call connections, or by allowing $\leq$ 5% of the calls to experience packet losses greater than 1%, the system permits a better TASI advantage to be achieved compared with figures obtained for the fully connected PVS.

Employing the above criteria the TASI advantage of the system can now become a function of the blocking probability. This is investigated in the next experiment.

Experiment 5.

In this experiment we show that the use of the criteria of experiment 4 enables the TASI advantage of the system to be made a function of the blocking probability. Three sets of blocking probabilities have been chosen, namely; 5%, 2%, and 0.5%. The characteristic of the TASI advantage versus packet length for each of the above blocking probabilities has been obtain for the condition that at least 95% of the calls experience packet losses less than or equal to, 1%. The results, as depicted in fig. 4.18, show that as the blocking probability is reduced, so the TASI advantage can be improved.

4.5 Conclusions

In this chapter two criteria were used for the evaluation of the performance characteristics of a two node packetized voice system which utilizes the CTI reassembly strategy at the receiver. The first criteria was based on the evaluation of the TASI advantage versus the packet length for a fixed D_{ETE}. It was shown that under this criteria the optimum packet length corresponds to a packet length which maximizes the TASI advantage of the system. The second criteria was based on the characteristic of D_{ETE} versus the packet length for a given value of the total number of subscribers. It was shown that under this criteria the optimum packet length corresponds to packet length which minimizes the D_{ETE} delay. A computer simulation model was presented to evaluate and compare the performance characteristics of the PVS with the fully loaded and average telephone traffic load at the exchange. It was shown that the PVS, when subjected to average telephone traffic load will give a better TASI advantage performance only if one allows the calls during the busy periods to experience packet losses greater than the 1% permissible value. A further criteria was presented to evaluate the TASI advantage of the System for the condition that some of the calls during the busy periods experience packet losses greater than 1%. It was shown that by using this criteria, the TASI advantage of the system becomes a function of blocking probability.

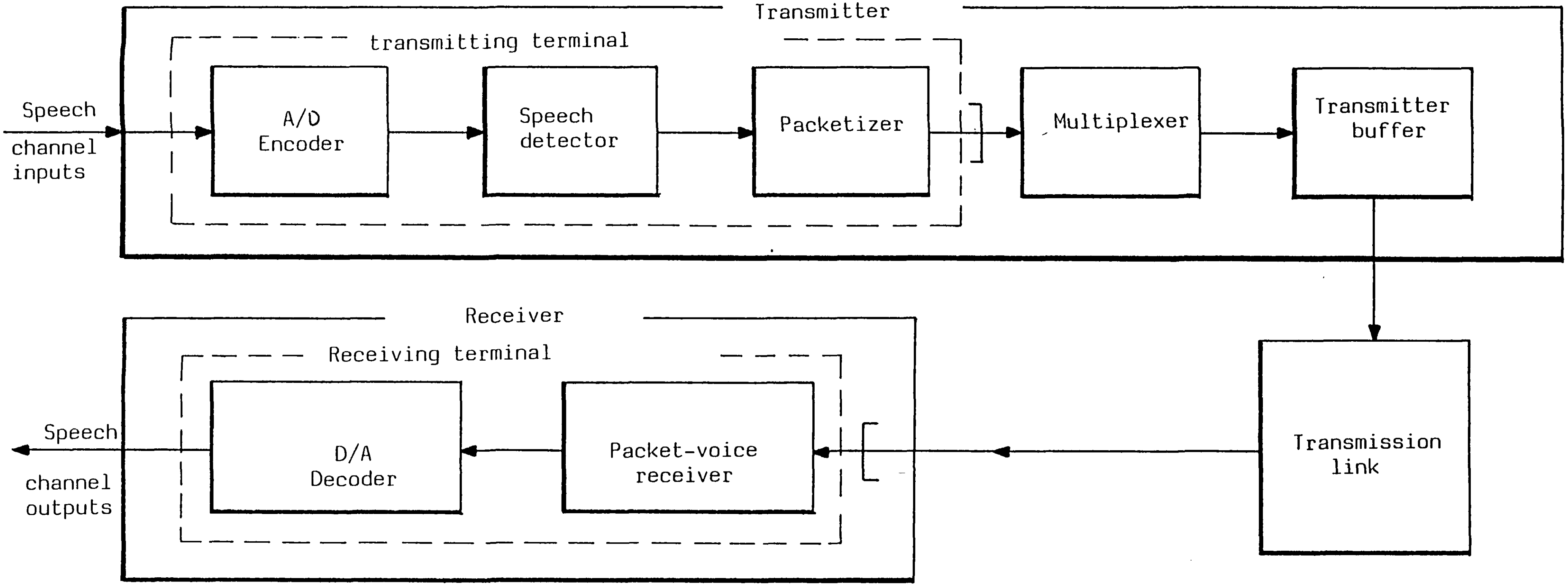


Fig. 4.1 The block diagram of a two node packetized voice system

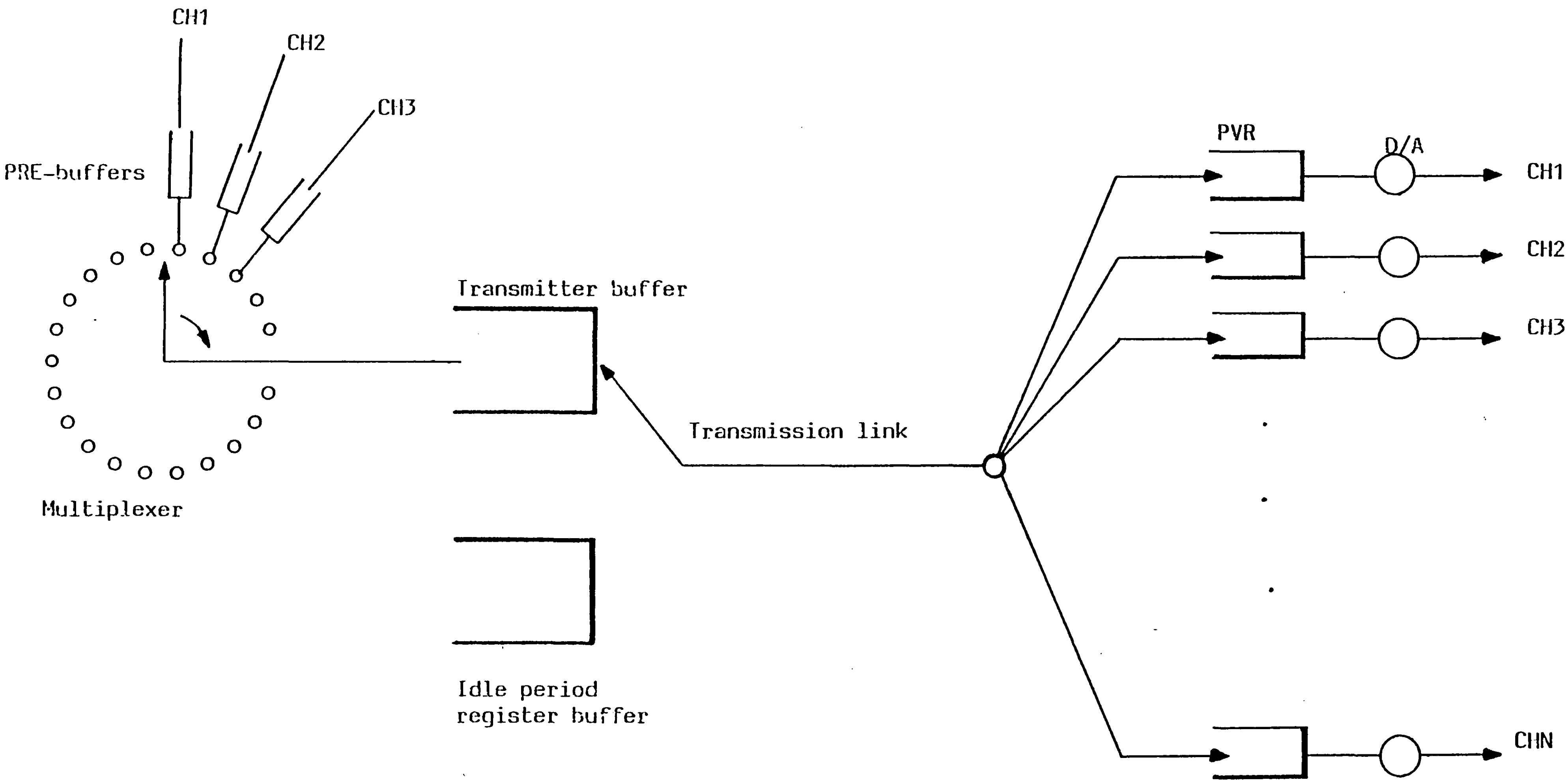
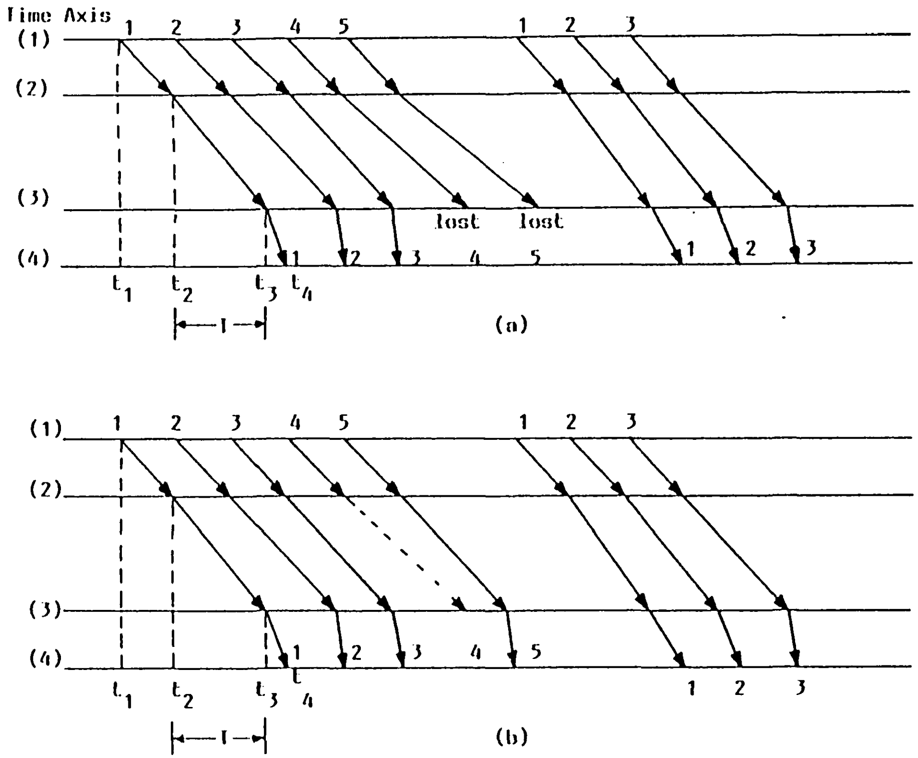


Fig. 4.2 The implementation diagram of a two node packetized voice system



1..... Begininning time of packetization
2 ... Beginning time of voice packet transmission
3 ... Packet arrival time, time at the packet voice receiver
4 ... Beginning time of packet play-out

t_2-t_1 ... Packetization delay

t_3-t_1 ... Packet link delay

t_3-t_1 ... Packet transmission delay

t_4-t_1 ... ETE packet delay

t_4-t_2 ... Control time delay

Fig. 4.3 Packetized voice CTI reassembly strategy. (a) Infinite transmitter queue buffer length (b) Finite transmitter queue buffer length.

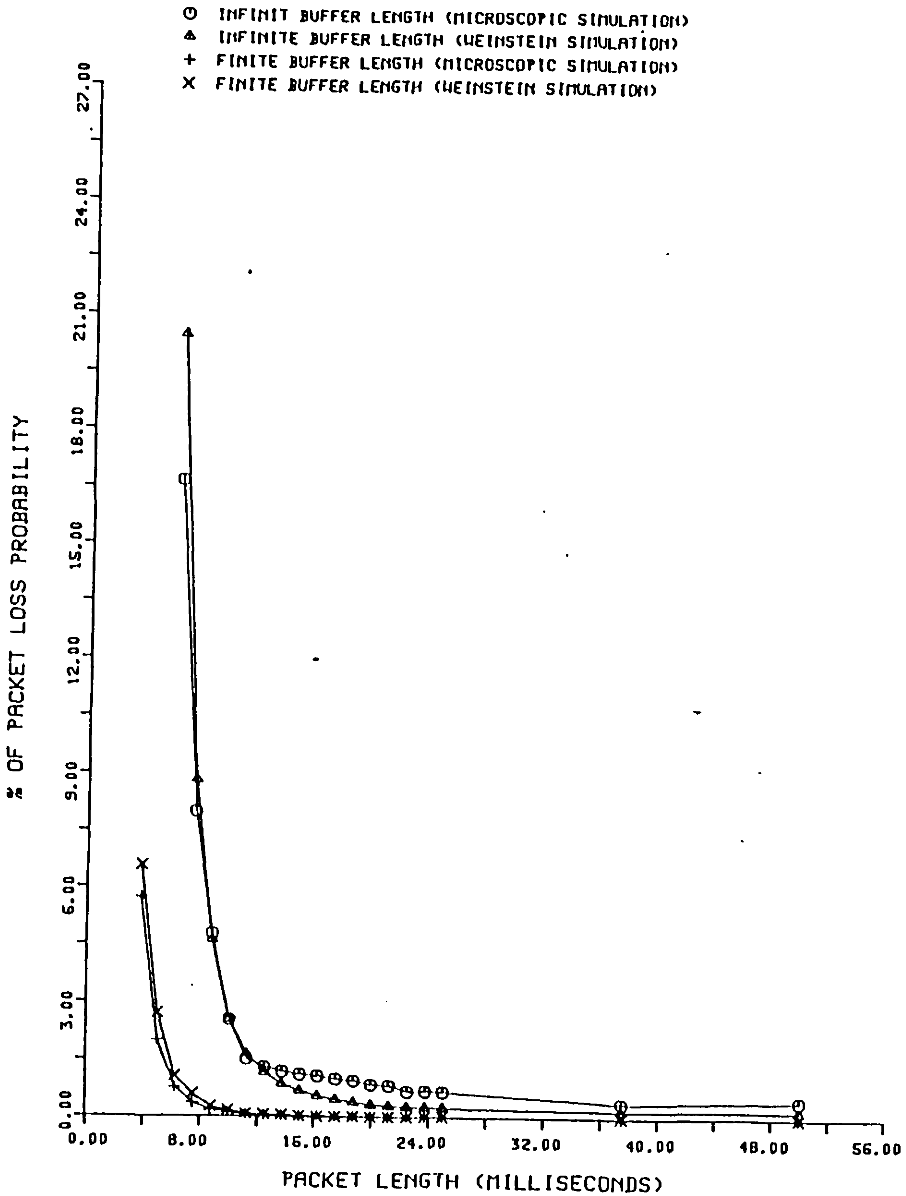


Fig 4.4 The characteristics of the % of packet loss probability versus packet length for the two PVS with and without infinite queue buffer. The total number of subscribers is equal to 40.

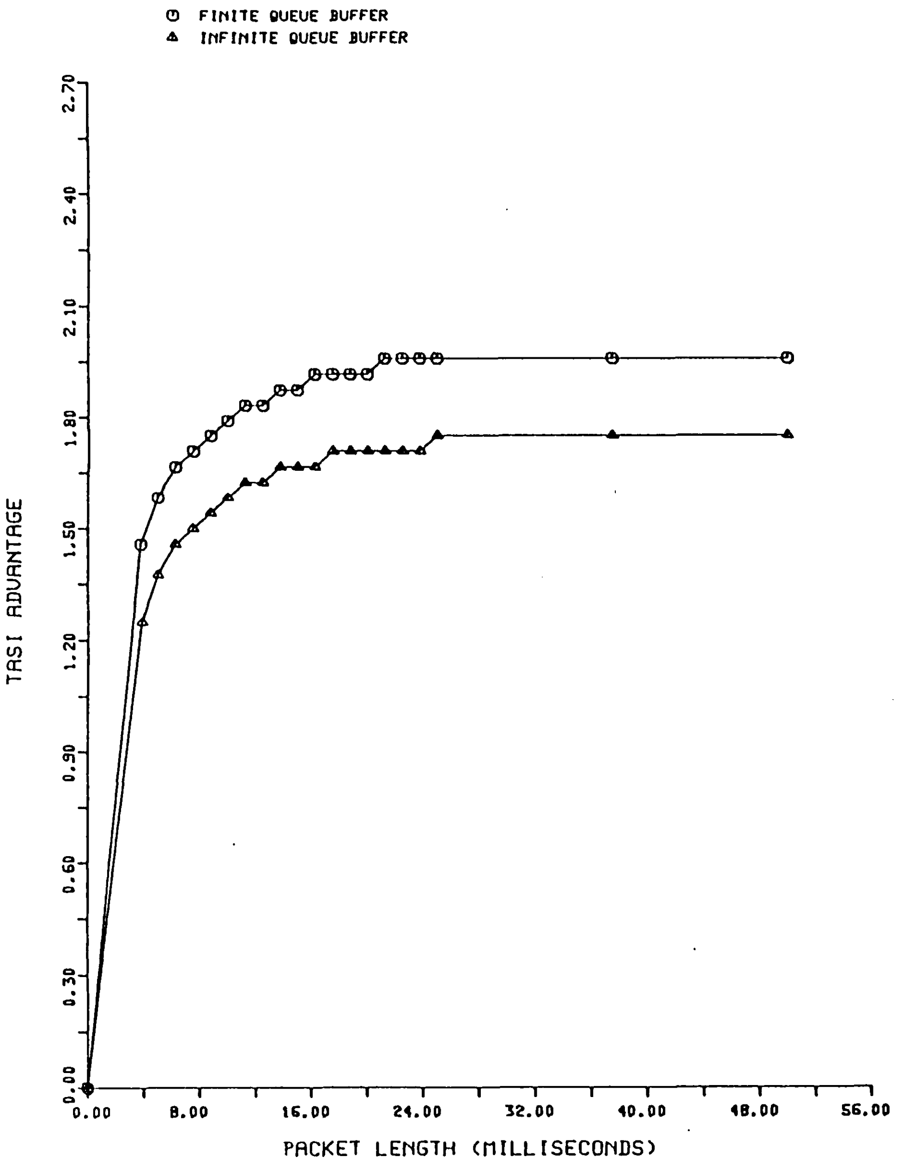


Fig 4.5 The characteristics of TASI advantage versus packet length for the two PVS with and without infinite queue buffer. The ETE packet delay is set at 250 ms

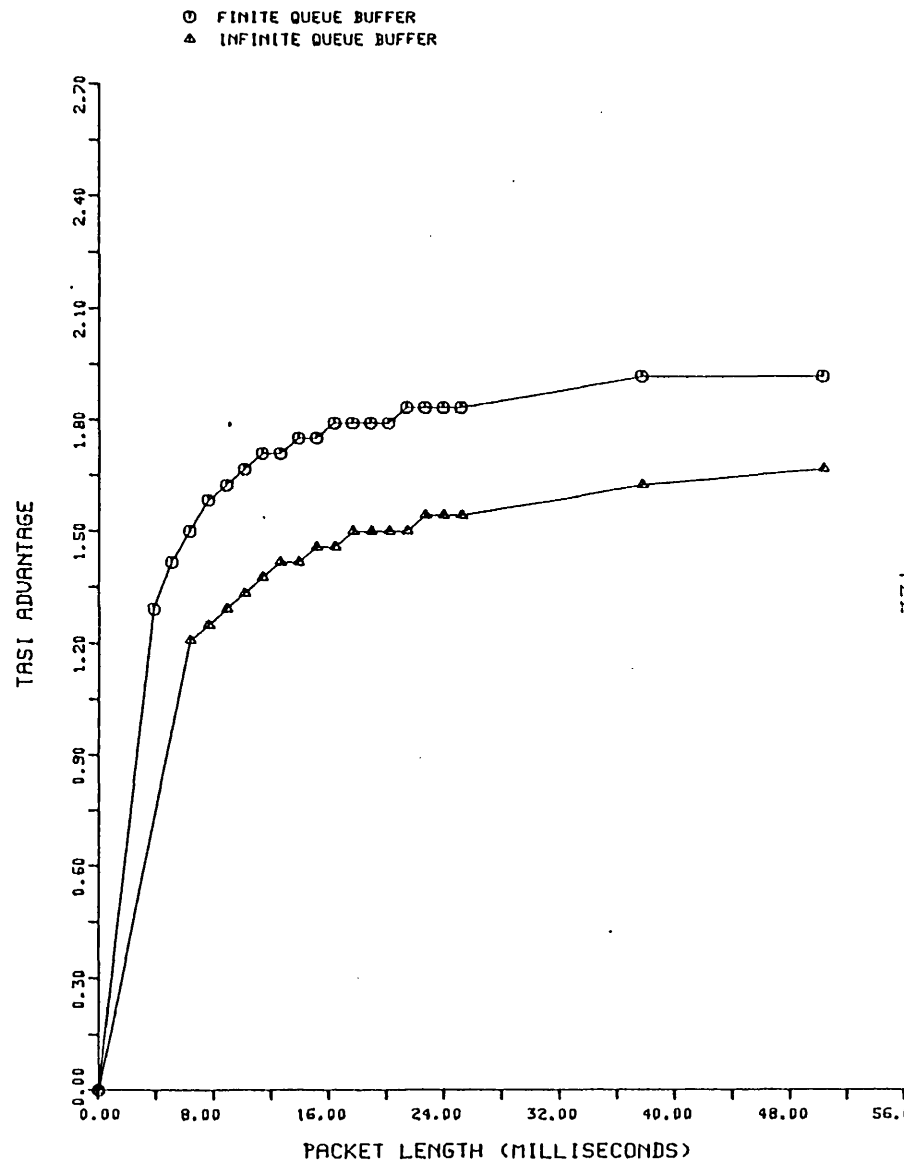


Fig. 4.6 The characteristics of TASI advantage versus packet length for the two PVS with and without infinite queue buffer. The ETE packet delay is set at three times of the packetization delay

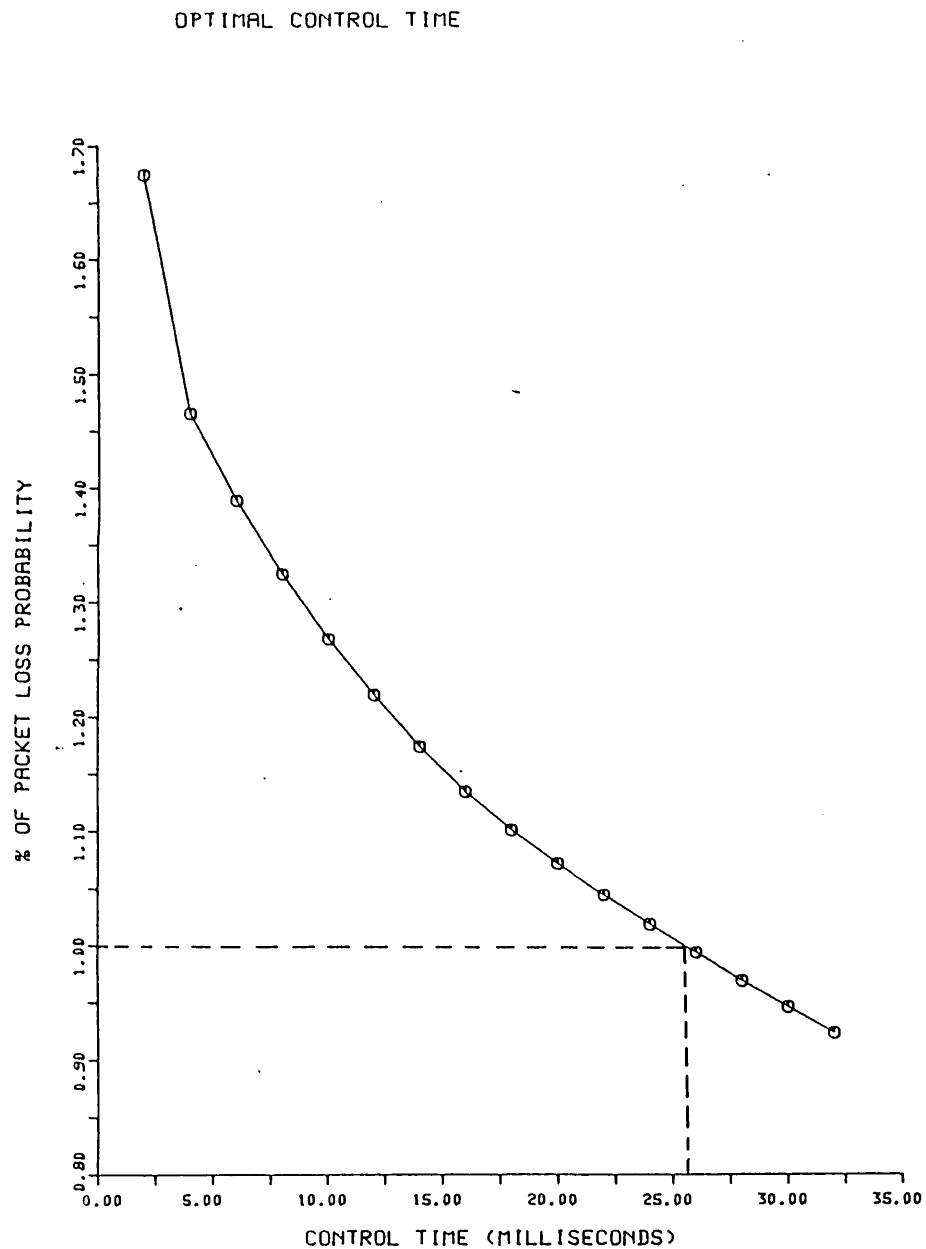


Fig. 4.7 The characteristic of % of packet loss probability as a function of control time. The total number of subscriber is 40 and the packet length is 8.75 ms

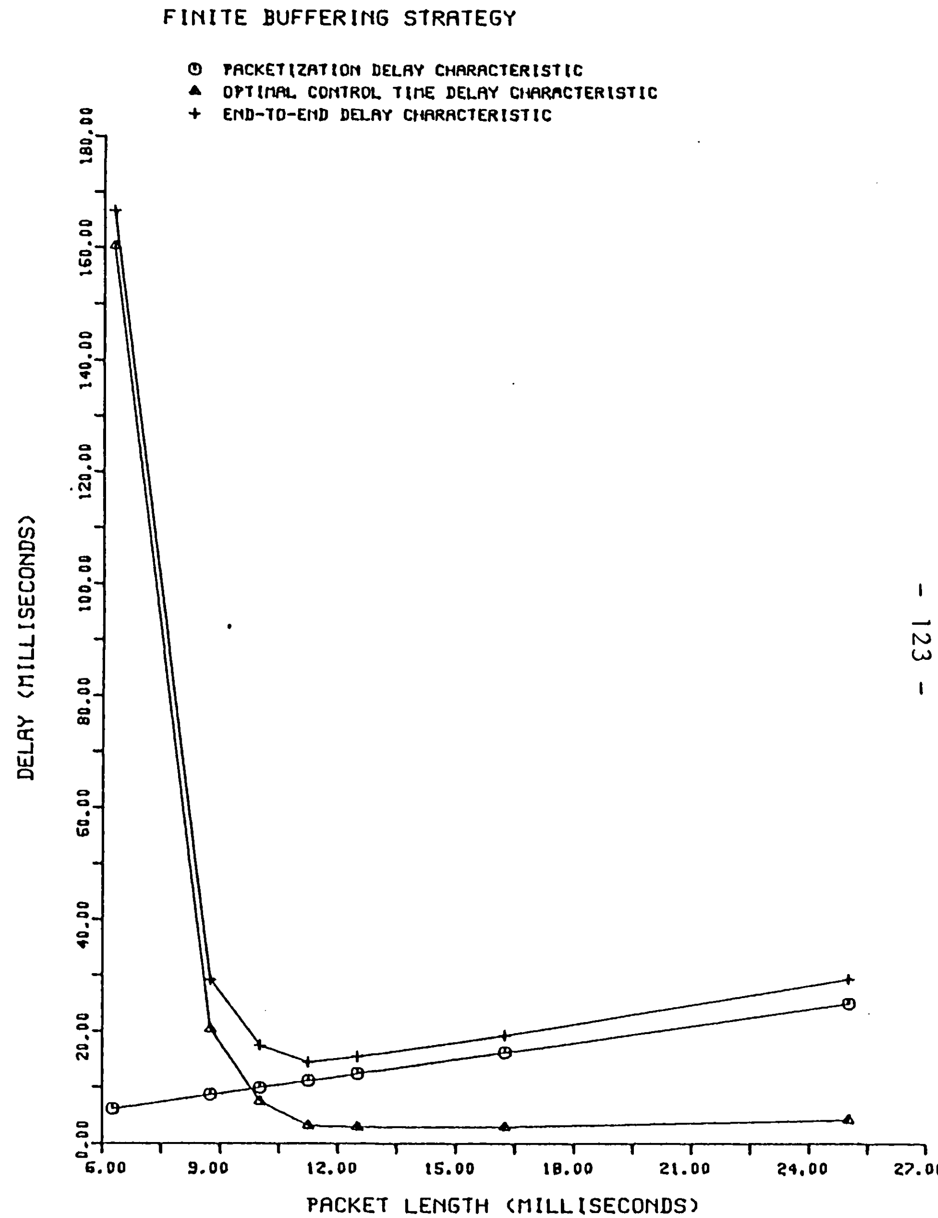


Fig. 4.8 The characteristics related to evaluation of optimal packet length for 40 subscribers

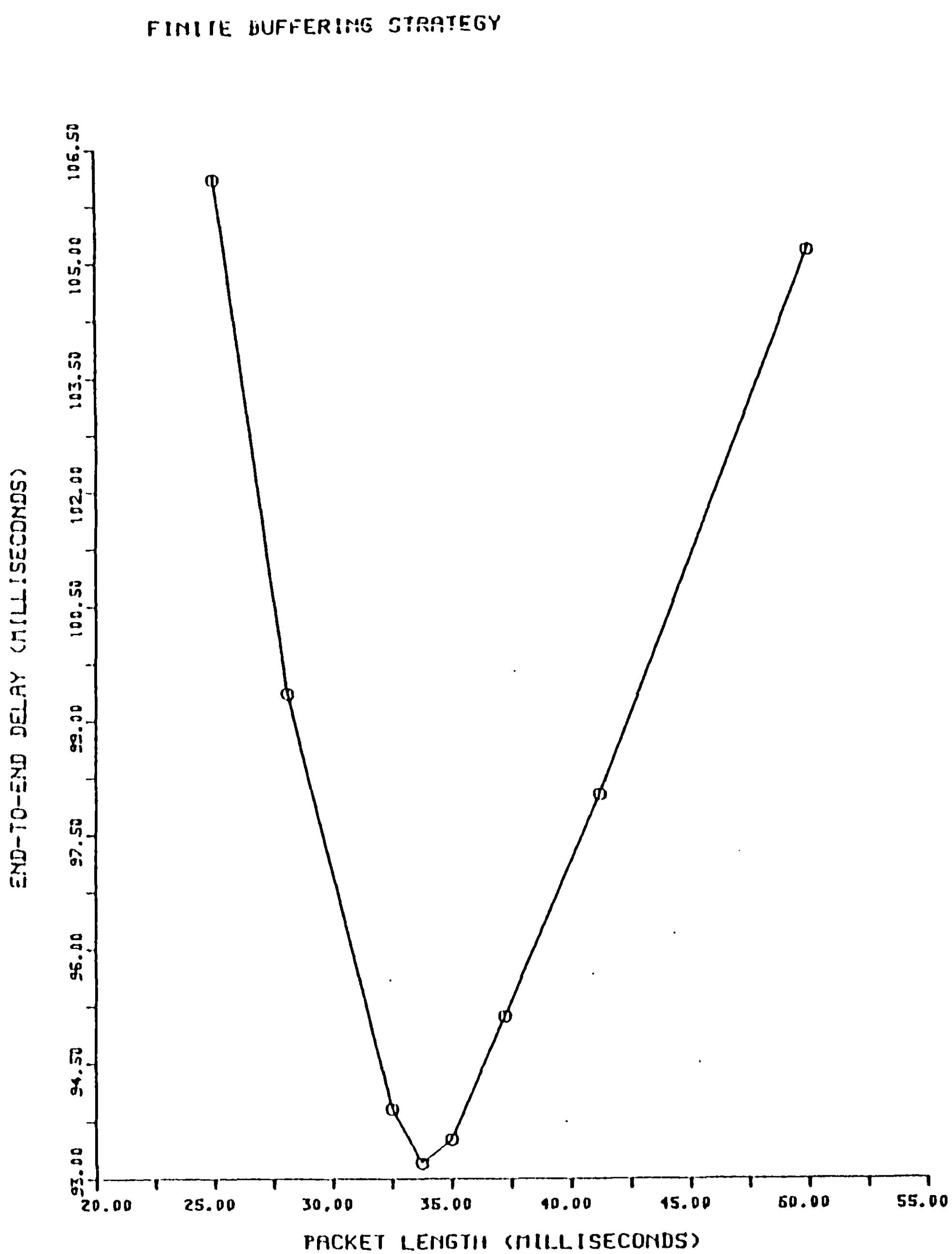


Fig. 4.9 The characteristic related to evaluation of optimal packet length for 45 number of subscribers

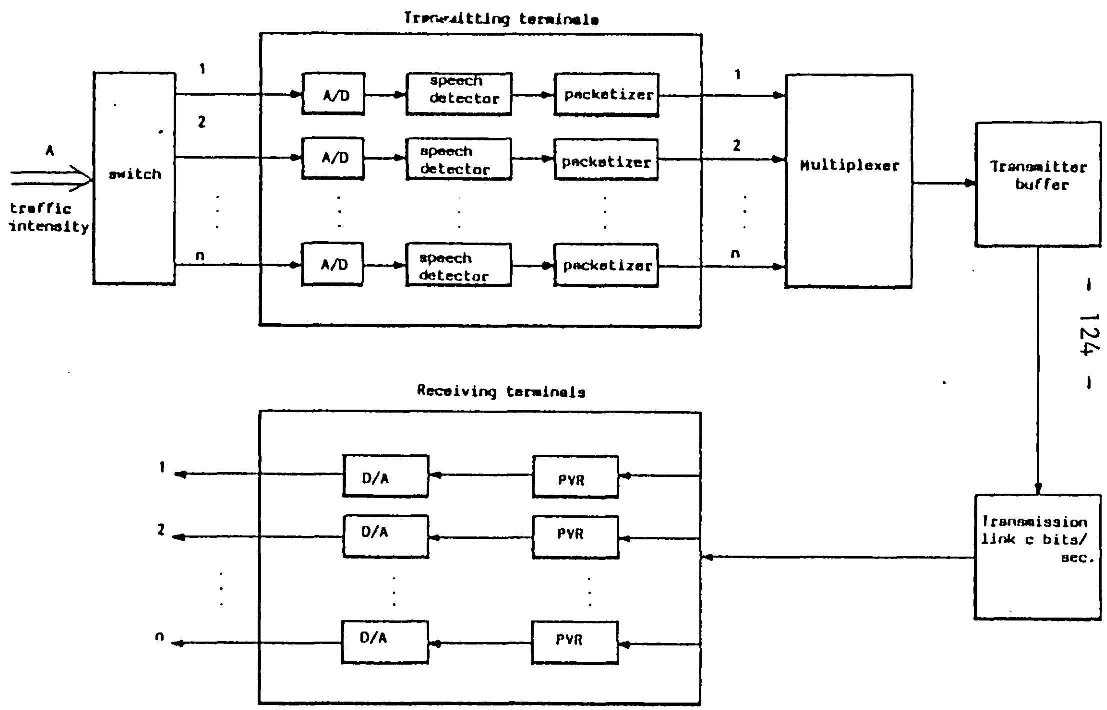
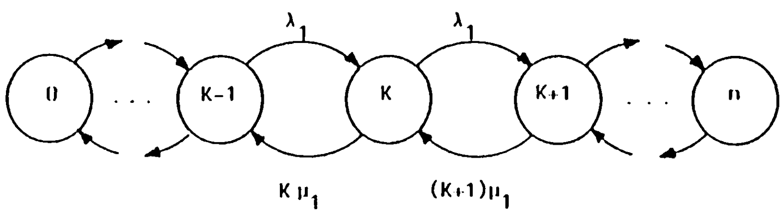


Fig. 4.10 The block diagram of a two node packetized voice system with the introduction of the telephone traffic loading

Voice traffice simulation model

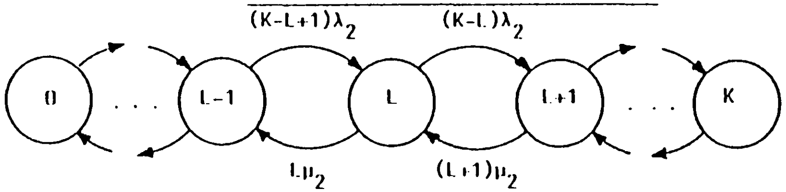


λ_1 = Call arrival rate

μ_1^{-1} = Mean call duration

(a)

Markovian simulation model for PVS



λ_2^{-1} = Mean silence duration

μ_2^{-1} = Mean talkspurt duration

(b)

Fig. 4.11 The computer simulation model for the PVS with the introduction of the telephone traffice loading

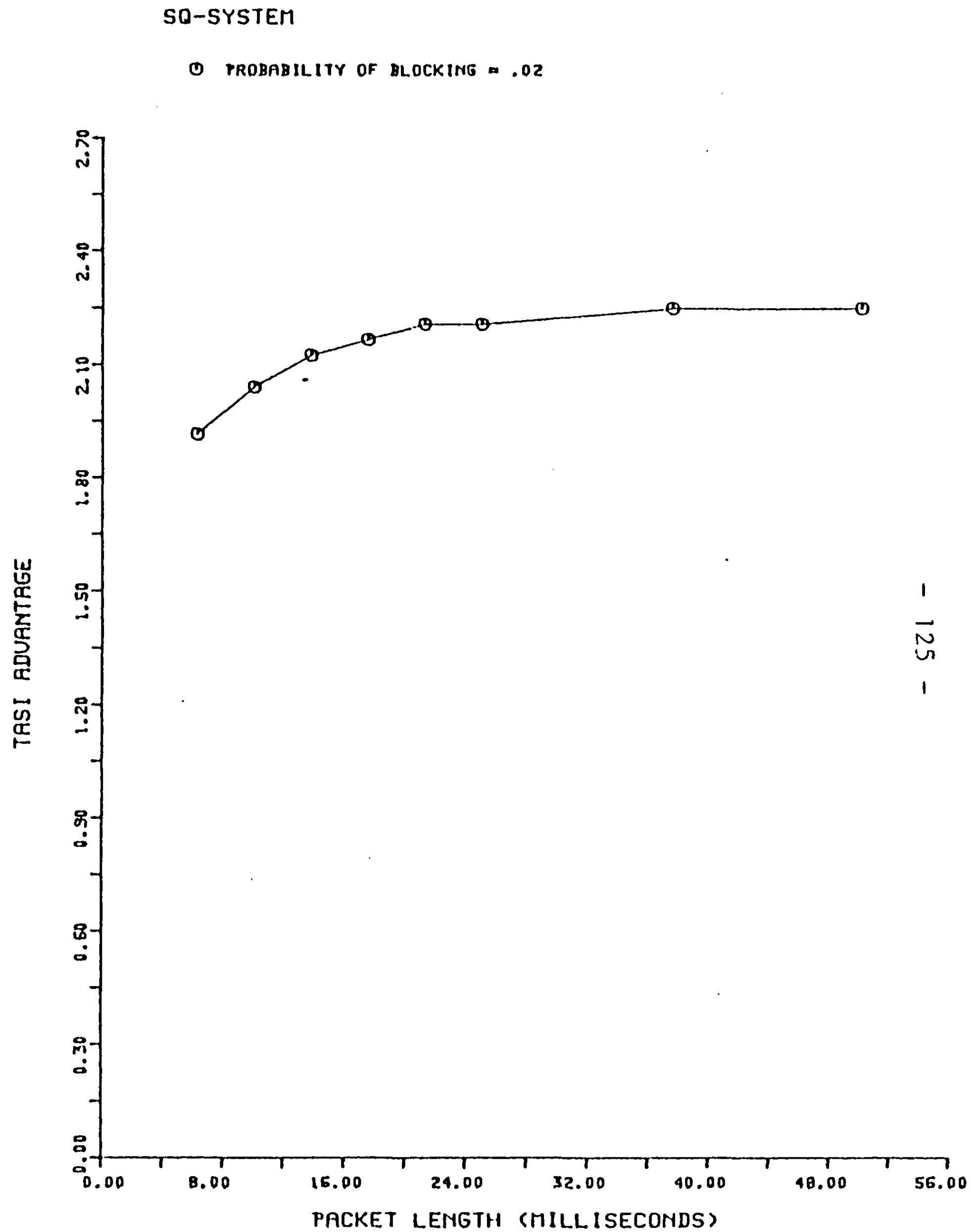


Fig. 4.12 The TASI advantage as a function of packet length. The blocking probability is set at .02.

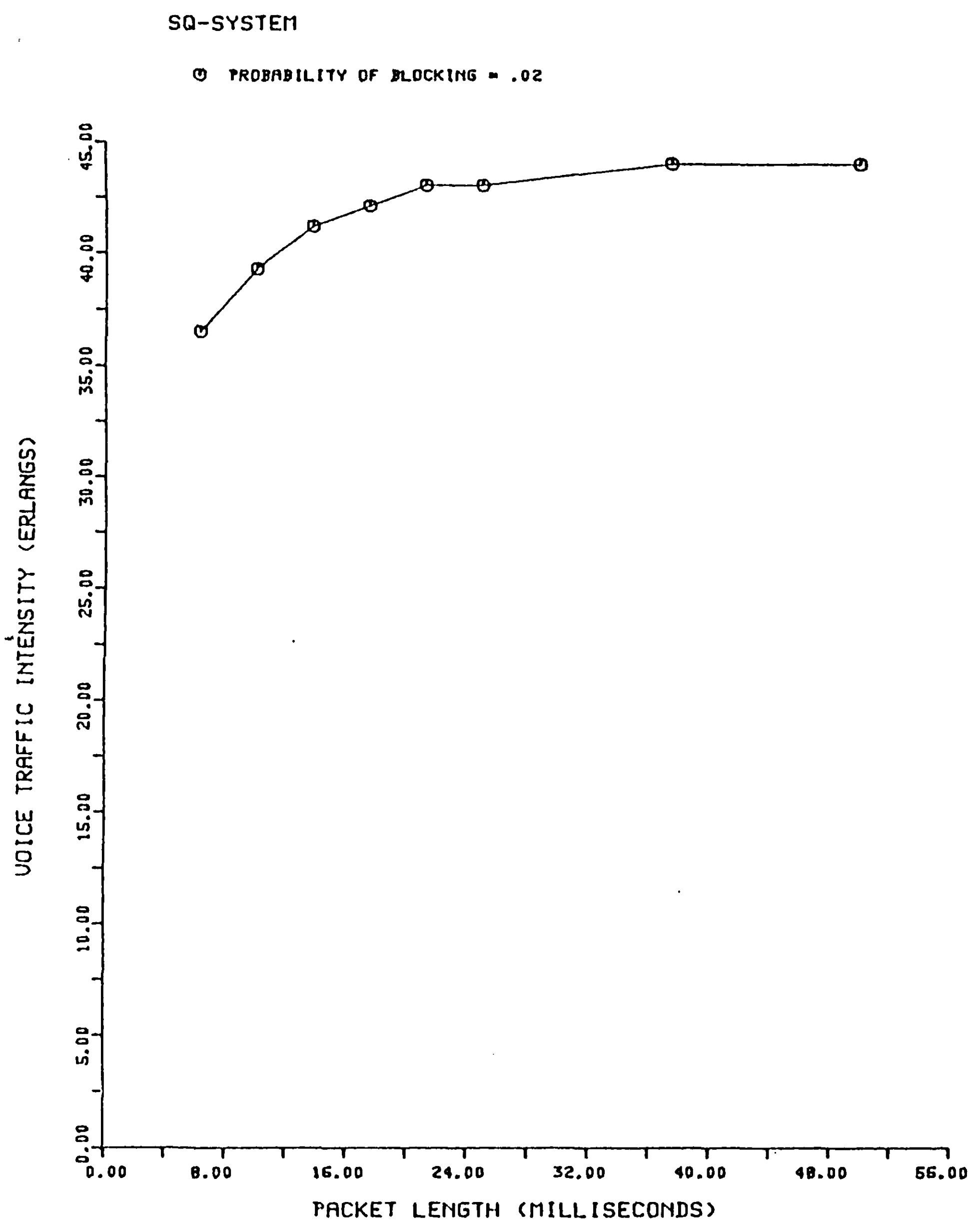


Fig. 4.13 The offered load voice traffic intensity as a function of packet length. The blocking probability is set at .02

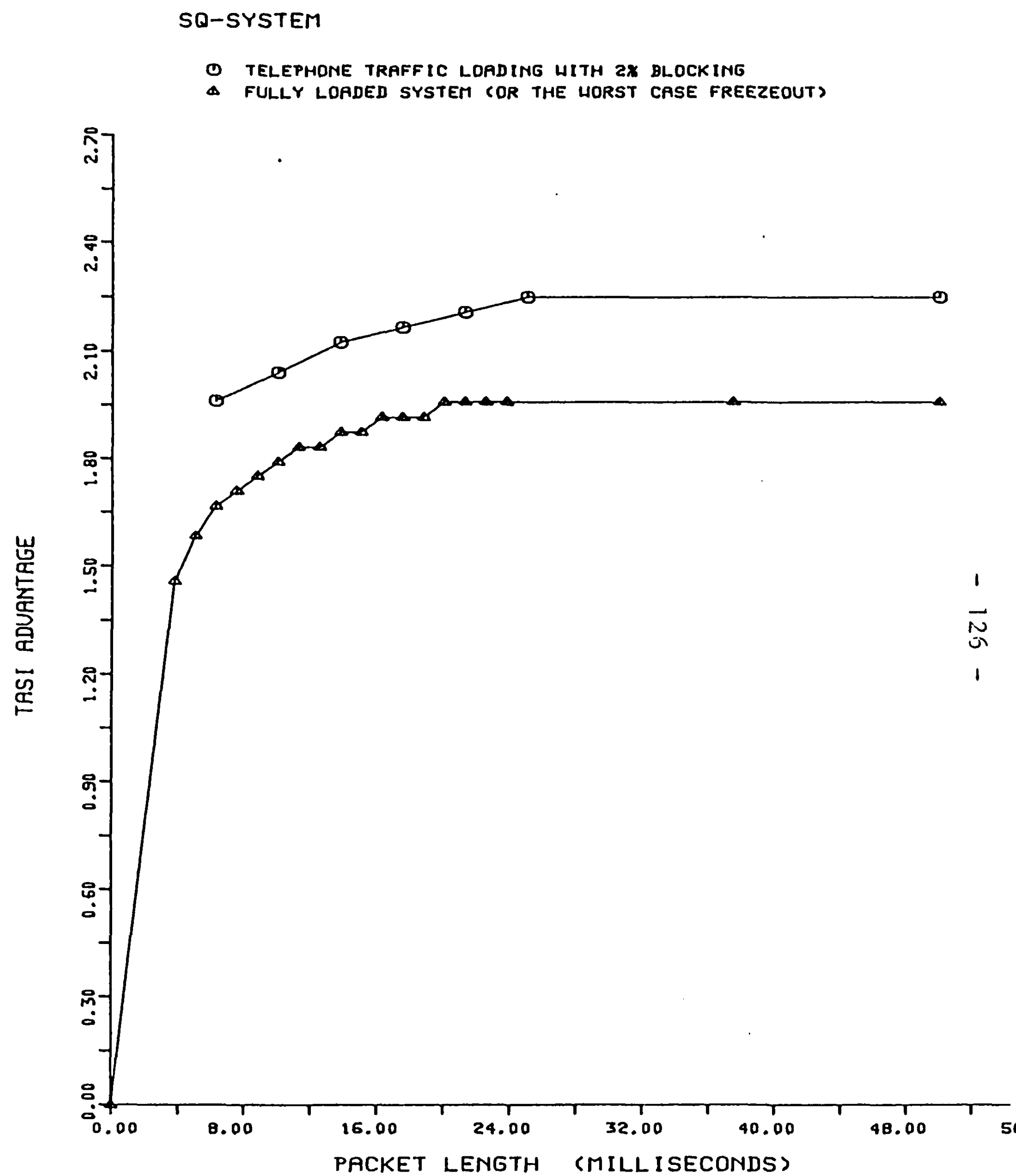


Fig.4.14 The comparison of the performance of the PVS with and without the presence of the telephone traffic load

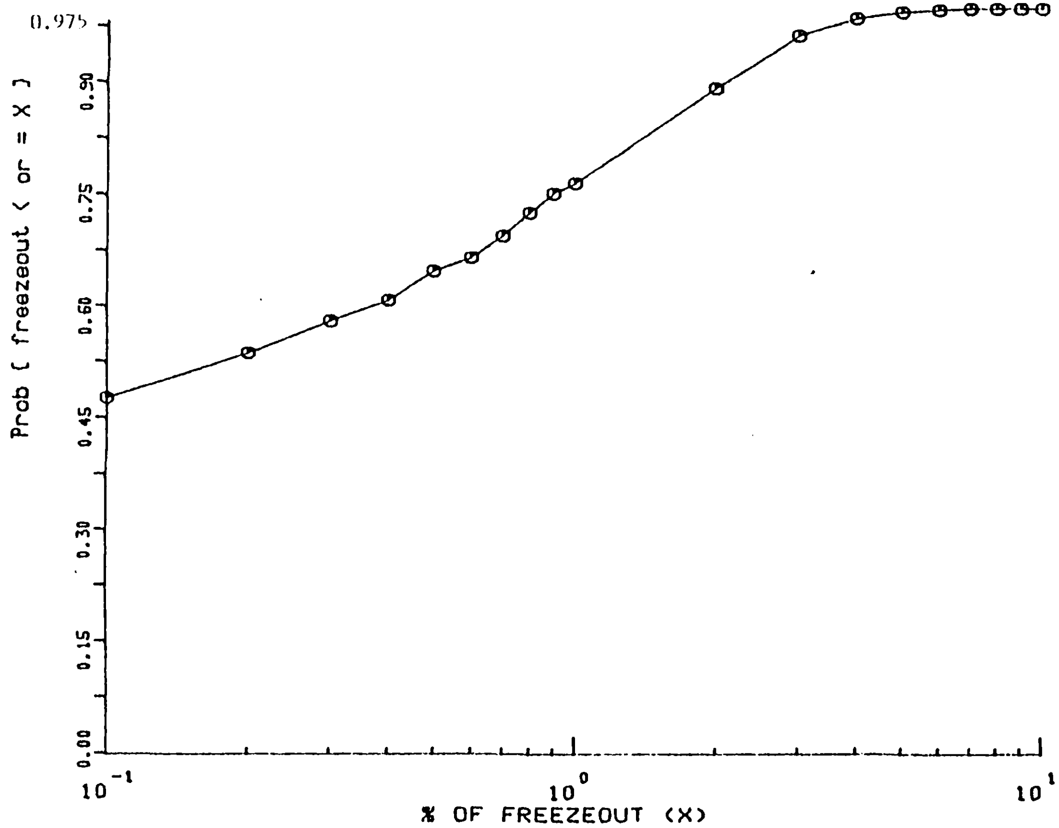


Fig. 4.15 The cummulative distribution function of the calls experiencing different pecentage of freezeout or packet losses

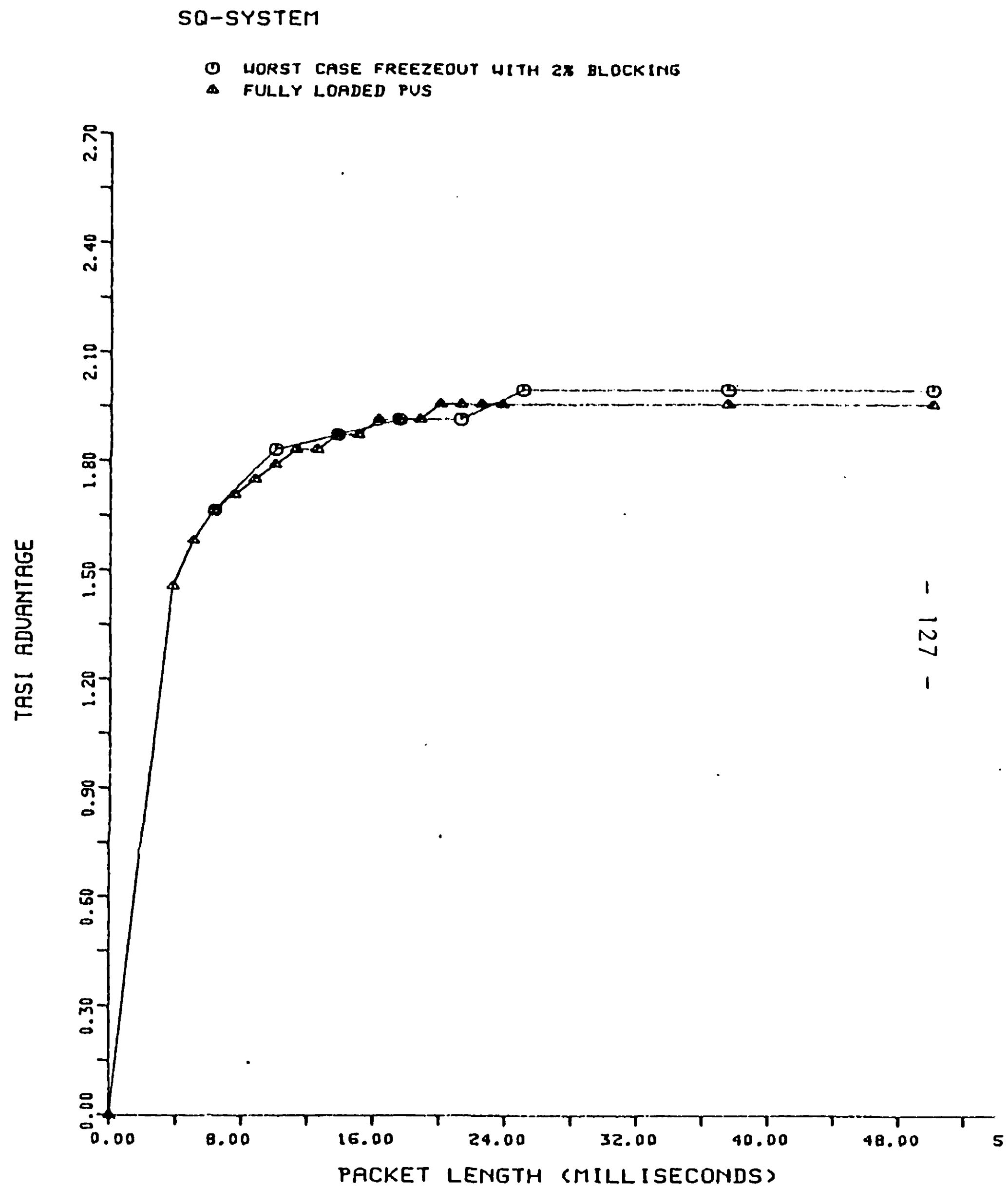


Fig. 4.16 The comparison of the worst case freezeout with 2% blocking probability and all the channel busy situation.

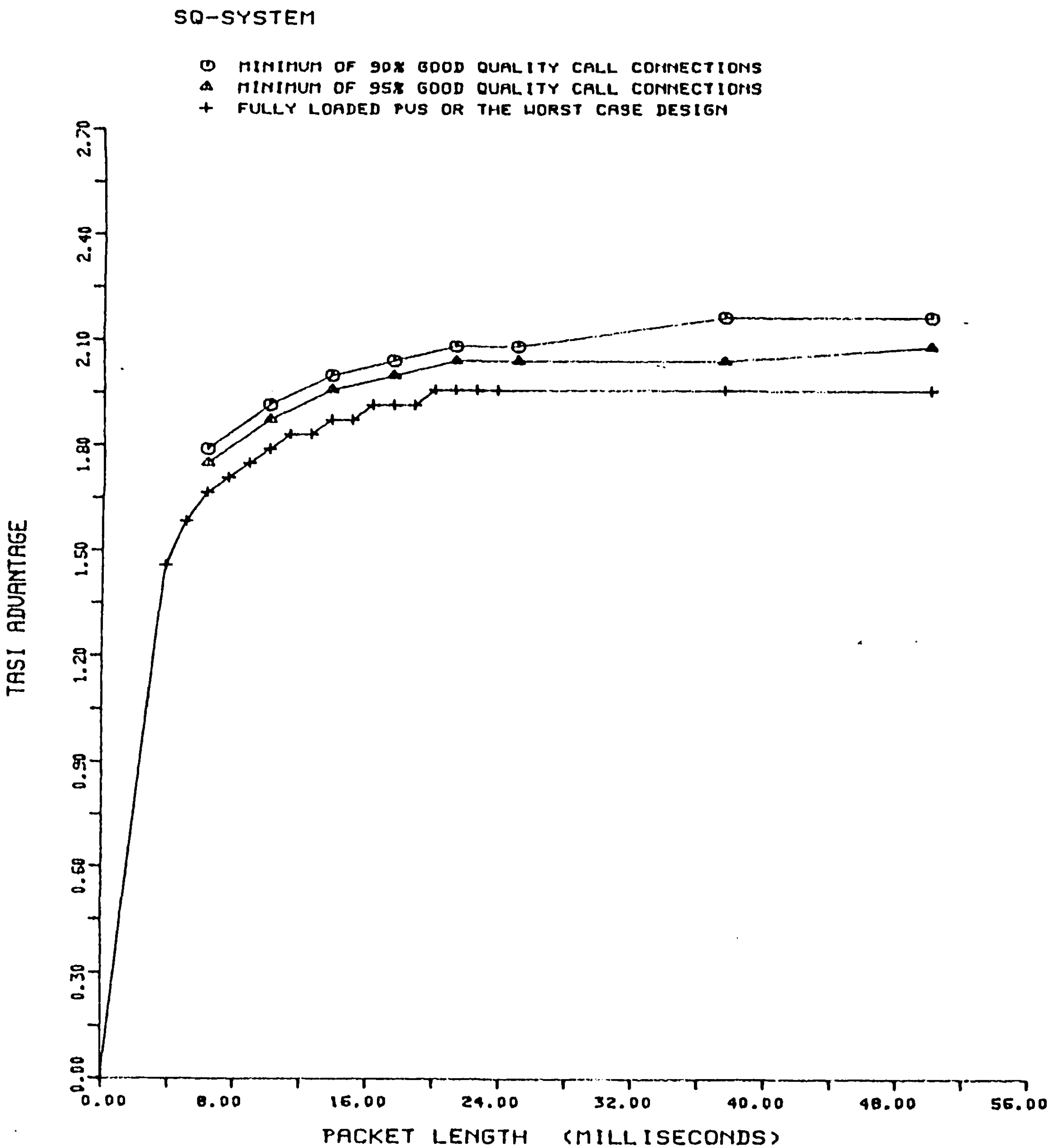


Fig. 4.17 TASI advantage as a function of packet length for different sets of good call connections

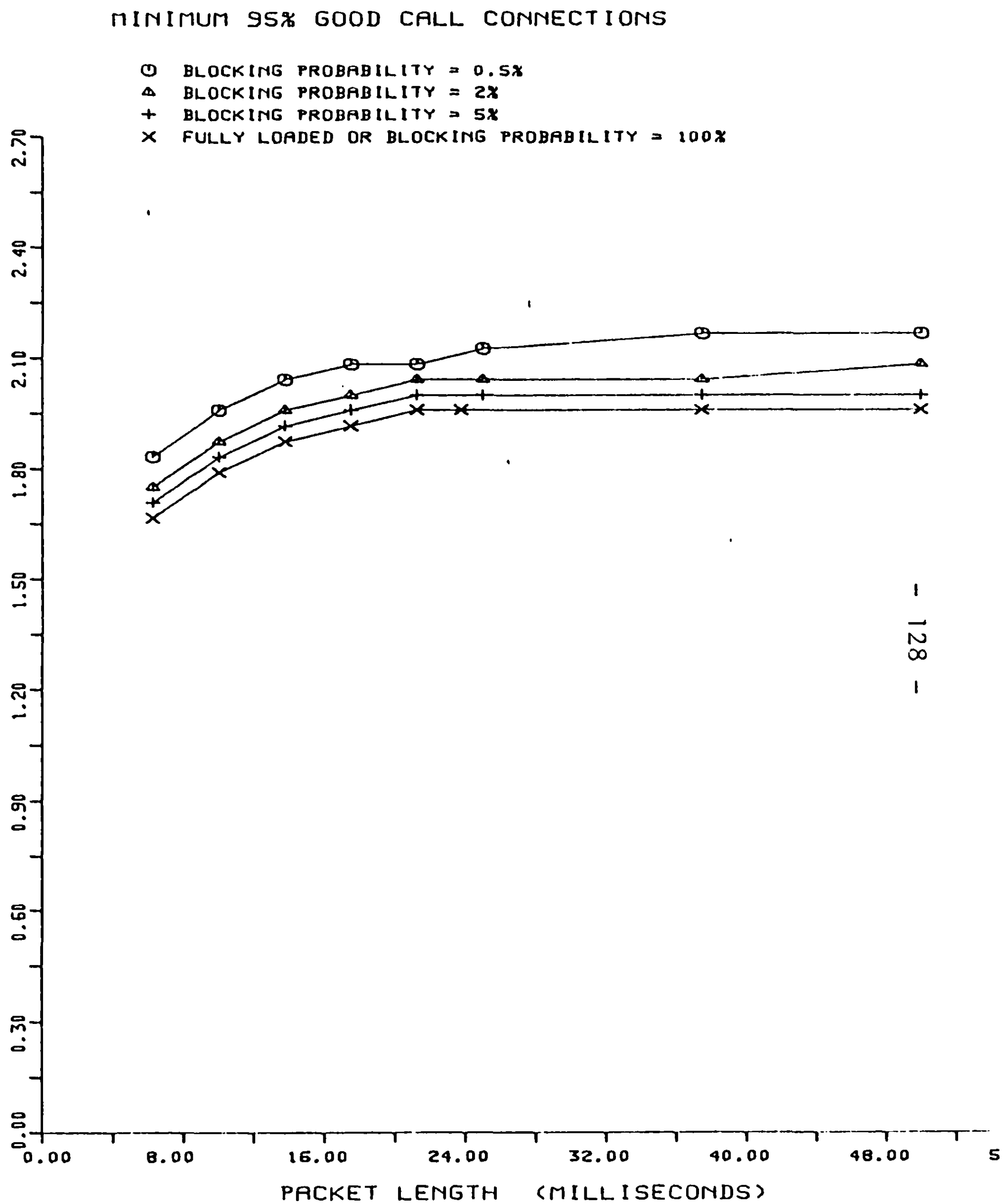


Fig. 4.18 TASI advantage as a function of packet length for different sets of blocking probabilities

CHAPTER 5

PACKET SWITCHED-VOICE WITH NON-STOCHASTIC QUEUEING (NSQ) OPERATION

5.1 Introduction

In this chapter a different type of packetized voice system has been investigated. The system uses a pure loss type of strategy for the multiplexing and demultiplexing of packets, this being a method which does not employ stochastic queueing operations at the transmitter. We call this the 'non-stochastic queueing' (NSQ) system. In Section 5.2 the operation and modelling of this system are explained. In Section 5.3 an analytical model has been developed for the system from which the performance characteristics are easily obtained. The comparisons of the computer simulation results of the system and those previously described for the SQ-system are presented in Section 5.4. In Section 5.5, two packet organization techniques have been applied to this scheme in order to improve the TASI advantage of the system and also the speech performance. In Section 5.6, an integrated voice/data model has been studied which handles the voice traffic by this type of packetized voice transmission. In Section 5.7, a multinode NSQ-packetized voice network model is presented and its performance characteristics obtained. The effect of realistic telephone traffic loading on the system has been dealt with and formulated in Section 5.8.

5.2 The Operation and Modelling of the System

As shown in figs 5.1 and 5.2 the packetization and multiplexing operations of this scheme are similar to those for the SQ-system. The differences between the two systems are due to the ways in which the packets are transmitted and received through the network. The packet transmission operation of this scheme limits the delay a packet can suffer, this is due to the use of a limited length buffer. This scheme ensures that a packet does not stay in the queue for a time longer than Δ_t seconds. For example, if the number of packets (K) in any frame produced by the connected channels is less than the number of packets(J) that can be sent by the channels, then the transmission frame will consist of (K) packets followed by (J-K) fill-in packets (or data packets). If the number of packets (K) in any frame exceeds (J), then (K-J) of the packets, chosen from among the (K) active talkers, are discarded in such a way that the packet loss probabilities are uniformly distributed amongst all the talkers. A delay of one frame length, Δ_t seconds, has to be incurred at the transmitter in order to organize the packets into a regular frame pattern, so ensuring order and synchronisation.

Upon the reception of a frame at the receiver, a full frame of speech information (active spurts, plus silent periods which were not transmitted) are reconstructed at the PVR and are then demultiplexed to the receiving terminals. In order to maintain the synchronization between the transmitter and the receiver, the demultiplexer's repetition interval has also to be set to Δ_t seconds.

5.3 Performance Criteria of the NSQ-System

In this section, a mathematical model is developed to measure the performance of the NSQ system in terms of the end-to-end (ETE) packet delay and the probability of the packet being lost.

5.3.1 Delay criteria

In practice there are three major sources of ETE delay, or network delay, D_{net}, associated with the NSQ system, namely:

(1) Packetization delay D_z.
This is the constant delay required to form a packet at the input and is equal to P/8 milliseconds (where P is the packet length in bytes, excluding overhead).

(2) Frame construction delay at the transmitter D_t.
This is the constant delay required at the transmitter for the construction of a frame of length Δ_t seconds. The delay experienced by each packet is therefore equal to Δ_t seconds, or P/8 milliseconds.

(3) Full frame reconstruction delay at the receiver D_r.
For the full frame reconstruction the packets arriving at the receiver have to experience a delay of Δ_t seconds, or P/8 milliseconds.

The equation for the network delay, D_{net}, in terms of D_z, D_t and D_r can be written as follows:

$$D_{net}' = D_z + D_t + D_r$$
$$= 3\Delta_t = 3P/8 \qquad (5.1)$$

This equation shows that the total delay per talker in the system is constant and its length is equal to three times the packetization delay.

5.3.2 The packet loss probability criteria

In Chapter 3, it was stated that the steady state probability of the multiplexer randomly transmitting r packets out of n possible packets can be approximated by a binomial p.d.f. (for proof see Weinstein [7]). We denote this relationship by B(n,r,q), where

$$B(n,r,q) = \sum_{r}^{n} q^r (1-q)^{n-r}, \qquad (5.2)$$

where q denotes the probability that a talker will issue a packet at random. According to equation (3.15) this value is equal to .445. Employing the above assumption, Weinstein [7] developed an analytical expression for the packet loss probability for this system. The derivation of this formula is given here as follows:
For a long period of system operation, i.e. the duration of $M\Delta_t$ seconds, where M is a large integer, the total number of packets offered by the n talkers will be:

$$\text{Packets offered} = \sum_{r=0}^{n} rMB(n,r,q)$$
$$= Mnq$$

Packet losses only occur when the system occupies one of the congested states between J+1 to n. The number of packets discarded will therefore be:

$$\text{Packets discarded} = \sum_{r=J+1}^{n} (r-J)MB(n,r,q), \tag{5.4}$$

and the resulting probability of packet loss is the ratio of (5.4) to (5.3), or

$$P_L = \frac{1}{nq} \sum_{r=J+1}^{n} (r-J)B(n,r,q); \tag{5.5}$$

where J represents the total number of packets that can be transmitted by the channel per frame and is given by:

$$J = \left\lfloor \frac{pc}{f(Pb+h)} \right\rfloor . \tag{5.6}$$

The equations (5.5) and (5.6) are used to compute the packet loss probability in the NSQ-system for various values of packet length. The results obtained for this situation are shown in fig. 5.3, together with the simulation results for the NSQ system. Close resemblance of the two sets of results is evident.

The same equations have also been used to compute the TASI advantage of the NSQ-system for various values of packet length. For each value of P, J is first calculated from equation (5.6), and then n is varied until P_L, from equation (5.5), is adjusted until it is just below or equal to 1%. The results obtained in this situation are shown in fig. 5.4, together with the simulation results of the NSQ system. Close resemblance of the two sets of results is also evident in this situation. Thus these sets of comparisons indicate that the Weinstein's packet loss probability expression is valid for this system. There is also the evidence of step changes in the characteristics of fig. 5.4. These are due to the step changes of the integer J caused by variations in P in equation (5.6). Fig. 5.5 shows the variations of J with packet length and is similar to the variations of the TASI advantage characteristic with packet length shown in fig. 5.4. Fig. 5.6 shows the results of further investigations for the comparison of the simulation and analytical results. It portrays the characteristics of the maximum number of interpolated subscribers versus channel rate normalized in the number of circuit-switched slots. These results are for 10 ms packets (or P=80 bytes), h=100 bits and an encoder output per channel of 64 kb/s. Close resemblance of the two sets of results is also evident at this stage.

The results from the simple mathematical model, tally closely with the simulation results, and so the mathematical model has been used to study the performance characteristics of the NSQ-system. The use of the mathematical model eases the assessment of the NSQ system whereas the SQ system can only be assessed using computer simulations. Simulation tends to be time consuming and expensive in computer time. Other detailed performance characteristics can be obtained using the mathematical model. For example, fig. 5.7, shows the characteristics of the TASI advantage versus packet length for different encoder bit lengths. Fig. 5.8 shows the variation of the TASI advantage versus packet length for different values of overheads used in a NSQ-system which employs 64 kb/s encoder outputs.

5.4 Comparisons of SQ and NSQ systems

As shown in the previous section, the ETE packet delay in the NSQ-system is constant and is equal to three times that of the packet length set in the system. In the previous chapter, it was shown that when the CTI strategy is employed in the SQ-system, the ETE packet delay in the system becomes controllable. Comparisons of the two systems have been carried-out here by setting the ETE packet delay of the two systems equal, (i.e. with ETE delay set at three times the packet length chosen for transmission). The simulation results of the two systems are depicted in figs. 5.9 and 5.10.

Fig. 5.9 shows the computer simulation results for the percentage packet loss probability versus packet length for the SQ and NSQ systems. It shows that the NSQ system has a better packet loss performance compared with the SQ-system with infinite transmitter buffer length and has a packet loss performance which is not quite so good as the SQ system which has a finite buffer length. These results are for the two packetized voice systems which have common system parameters of C=1.544 Mb/s, n=40, h=100 bits, f=8000 samples/sec and an encoder output per channel of 64 Kb/s.

Fig.5.10 shows the computer simulation results for the TASI advantage versus packet length for the SQ and NSQ systems. It shows that the NSQ-system has a better TASI advantage performance as compared with the SQ-system with infinite transmitter buffer length, but is not as good as the SQ-system with finite transmitter buffer length. The characteristics of the SQ-systems in this figure were derived in the previous chapter and were given in fig. 4.6.

In Ref [52], the present author investigated a similar type of comparison of the SQ and the NSQ systems. However, the SQ-system considered, was assumed to utilize the NTI reassembly strategy at the receiver with an arbitrary 50 ms control time. The length of the transmitter buffer was also limited in the SQ-system so that the maximum ETE packet delay in the system was limited to 250 ms. The complete operation of the SQ-system under this scheme resulted in having two types of packet losses (i.e. packet losses at both the transmitter and the receiver). The computer simulation results using this scheme have shown that better TASI advantage performance was obtained with the NSQ system (see appendix C).

5.5 Packet Organization Techniques for Improving the TASI Advantage of the NSQ-System

In this section two different strategies are investigated for improving the TASI advantage of the NSQ-system. The simple mathematical model which was introduced in the preceding section, has also been used here to evaluate the performance characteristics of these techniques.

5.5.1 Odd-even sample-interpolation techniques

In this scheme (as suggested by Jayant and Christensen [48]), adjacent coded speech samples are partitioned into two packets, an odd-sample and an even-sample packet, each with P coded speech samples. With statistically independent packet losses, and with a loss probability $P_L \leq 1\%$, a lost odd (or even) packet is very likely to be followed (or preceded) by an even (or odd) counterpart. The samples of the lost packet can then be estimated by a nearest neighbourhood interpolation technique; an interpolation method has been suggested in [48]. This interpolation, of course, necessitates an increased coding and decoding delay (2P/8 milliseconds instead of P/8 milliseconds) at the transmitter and also at the receiver. As well, an additional increase in the overheads is required due to the need to send extra parameters for packet interpolation purposes. The same authors [48] have shown that the increase in the overhead could be as high as 16 bits (i.e. h=116 bits used here instead of 100 bits). The informal listing tests conducted by the same authors [48] have indicated that the maximum percentage of the lost packets allowed at the transmitter under this scheme should not exceed 5%.

With the view presented above, this form of transmission scheme has been used in the NSQ-system. Fig. 5.11 shows the characteristics of TASI advantage versus packet length for this modified scheme and are compared with those previously obtained for the NSQ-system. The percentage packet loss probability for the NSQ-system with odd-even sample interpolation was set at 5%. It can be observed that a significantly better TASI advantage performance can be achieved using this odd-even scheme.

The equation of the network delay for this scheme can be written as

$$\begin{aligned} D_{net} &= D_z + D_t + D_r \\ &= 2P/8 + P/8 + 2P/8 \qquad (5.7) \\ &= 5P/8 \end{aligned}$$

which shows that the ETE packet delay for this scheme is longer than the normal NSQ-system which was 3P/8 ms from equation (5.1). Fig. 5.12 shows these comparisons.

For the odd-even sample interpolation scheme there is a possibility both the odd and even packets are occasionally lost. When this happens, it is assumed that zero-amplitude stuffing (or mean zero speech signal stuffing) is used for the entire block as would be the case in the Normal NSQ System.

5.5.2 Adaptive voice flow control technique

This is a packet voice flow control technique based on embedded speech coding which was first suggested by Bially, et al [75]. In this scheme, as indicated in fig. 5.13, each speech terminal is able to encode the speech at a variety of bit rates, so producing packets with variable length. At any particular time each terminal sends packets with the highest available bit rate. If the packet is received, the result is a high quality, high rate voice output. If the packet is not received, then the receiving terminal informs the sending terminal and the encoder bit rate is reduced so that shorter packets are sent. This still gives a reasonably effective bit rate and speech quality. The net effect is that received bit rates (and therefore the speech quality) will vary according to network conditions. The concept is shown in fig. 5.13 for the case of four embedded encoding rates. Network switching nodes are permitted to discard voice packets with longer packet sizes.

The authors of Ref [75], used the above technique in a three node SQ packetized voice network. They also used the Weinstein Markovian simulation model to evaluate and assess the performance of the networks. In this section the present author uses this type of voice flow control mechanism to increase the TASI advantage of a two node NSQ packetized voice system. The scheme employs an NSQ-system with a coder assigned per channel operating at one of the three encoding rates of 64 kb/s, 32 kb/s or 16 kb/s. It has been assumed that the system employing this scheme always operates with the highest possible encoding rates (for the highest possible speech quantity) and the coders switch from the higher bit rates to the lower bit rates only when necessary to avoid the occurance of packet losses in the system (i.e a non-loss type of NSQ-packetized voice system is considered). The switching thresholds (or the conditions required for changing the bit rates) for the scheme can be easily adjusted in this system. The bit size used is related to the available space and the number of packets to be included in the frame. These switching thresholds are shown in fig. 5.14 and they correspond to states J_1 , J_2 and J_3 respectively for the 64, 32 and 16 kb/s encoder outputs being assigned for each talker. These threshholds can be found by using equation (5.6) as follows:

$$J_1 = \left\lfloor \frac{pc}{f(Pb_1+h)} \right\rfloor$$

$$J_2 = \left\lfloor \frac{pc}{f(Pb_2+h)} \right\rfloor \qquad\qquad (5.8)$$

$$J_3 = \left\lfloor \frac{pc}{f(Pb_3+h)} \right\rfloor \qquad b_1 = 8,\quad b_2 = 4,\quad b_3 = 2$$

where b_1, b_2 and b_3 each denote the number of bits encoded per speech sample. In the following, the performance of this system has been evaluated by using a simple mathematical model. The performance parameters for this scheme include 'the TASI advantage' and 'the probability of usage of the encoder outputs assigned for each speech terminal'. Since the scheme is a non-loss type of PVS, then its TASI advantage can easily be obtained. For example; for the system as in fig. 5.14, the maximum number of talkers which the system can support without any packet losses is equal to J_3. This number indicates the condition at which all the talkers are operating with 16 kb/s encoder outputs. If we assume that the system employs a 24-channel PCM link, then using equation (5.8), the TASI advantage of the system becomes

$$\text{TASI} = J_3/24 = \left\lfloor \frac{pc}{f(Pb_3+h)} \right\rfloor /24 \qquad (5.9)$$

This gives the dependency of the TASI advantage of the system upon the packet length P. By using equation (5.9), the TASI advantages of the system have been computed for various sets of packet lengths. The results are as given in fig. 5.15 which shows that significantly higher TASI advantages are achievable with this scheme as compared with the previous NSQ-system. It should however be noted that the overhead per packet in this scheme is assumed to be longer than the normal NSQ-system due to the need to send extra information in order to indicate each packet's decoding rate at the receiver. This extra information is assumed to be 8 bits (i.e the resulting overhead has been considered to be 108 bits instead of 100 bits).

The results, as shown in fig. 5.15, show results for the 'worst case' system operation. This corresponds to the condition where all the talkers are assumed to be active in their talk– spurt phase and are issuing packets. This results in the full operation of the system with their lowest assigned bit rates (and therefore resulting in lowest achieved speech qualities). This probability as depicted in fig. 5.14, corresponds to the rightmost tail of the binomial packet arrival distribution. The probability of this worst case condition is expected to be very low. The average speech quality has therefore been defined here to be the average bit rates that one expects the talkers to be operating at, averaged over a long period of operation. This can be obtained by finding the steady state probabilities of each of the encoding bit rates which are denoted here by P_{64}, P_{32}, P_{16}. For example, in order to evaluate P_{64} (the Probability that only 64 kb/s encoder outputs are active), the following arguments are given:

In a long period of system operation of duration $M\Delta_t$ seconds, the total number of speech packets offered by the total J_3 connected talkers in conversations will be:

$$\text{Packets Offered} = \sum_{r=0}^{J_3} rMB(J_3,r,q)$$

$$= MJ_3q \qquad (5.10)$$

The coders are either fully operating with the 64 kb/s if the system occupies any of the states from 0 to J_1, or are partially operating with the 64 kb/s if the system occupies any of the states from J_1 to J_2 (see fig. 5.14). The total number of speech packets which have been offered with 64 kb/s encoder outputs will then be

$$\begin{array}{l}\text{Packets Offered} \\ \text{with 64 kb/s}\end{array} = \sum_{r=0}^{J_1} rMB(J_3,r,q) + \sum_{r=J+1}^{J_2} M(r-m_{32}(r))B(J_3,r,q) \qquad \ldots \qquad (5.11)$$

and the resulting probability of usage of 64 kb/s encoder outputs is therefore the ratio of (5.11) to (5.10), which is

$$P_{64} = \frac{1}{J_3q}\left\{\sum_{r=0}^{J_1} rB(J_3,r,q) + \sum_{r=J+1}^{J_2} (r-m_{32}(r))B(J_3,r,q)\right\} \qquad \ldots \qquad (5.12)$$

In equation (5.12) above, all the terms are known except $m_{32}(r)$. This term denotes the number of active channels that have to be switched from 64 to 32 kb/s in order to preserve the system without any packet losses. If we assume P_1 and P_2 to be the length of the speech packets, including their overheads in bits, and e to be the length of channel's frame in bits, then

$$(r-m_{32}(r))P_1 + m_{32}(r)P_2 \leq e \qquad (5.13)$$

Solving for $m_{32}(r)$, the following relation will result

$$m_{32}(t) \geq \frac{rP_1-e}{P_1-P_2} \tag{5.14}$$

Since we require the maximum number of channels to be operating at the 64 kb/s rate and the minimum number to be working at 32 kb/s then

$$m_{32}(r) = \left\lceil \frac{rP_1-e}{P_1-P_2} \right\rceil \tag{5.15}$$

where $\lceil x \rceil$ denotes the largest integer greater than or equal to x. The requirement of such a notation has been justified by some examples given in Appendix B.

By extending the approach and using the same type of arguments, it can be shown that

$$P_{32} = \frac{1}{J_3 q} \left\{ \sum_{r=J_1+1}^{J_2} m_{32}(r)B(J_3,r,q) + \sum_{r=J_2+1}^{J_3} (r-m_{16}(r))B(J_3,r,q) \right\}$$

and (5.16)

$$P_{16} = \frac{1}{J_3 q} \left\{ \sum_{r=J_2+1}^{J_3} m_{16}(r)B(J_3,r,q) \right\} \tag{5.17}$$

where

$$m_{16}(r) = \left\lceil \frac{rP_2-e}{P_3-P_2} \right\rceil \tag{5.18}$$

P_3 is the length of speech packet in bits including the overhead issued by the 16 kb/s encoder outputs. The resulting net average bit rates achieved from the coders, is then given by

$$B_r = (b_1P_{64} + b_2P_{32} + b_3P_{16})f \tag{5.19}$$

where $b_1 = 8$, $b_2 = 4$ and $b_3 = 2$ bits/sample, and f is the sampling rate of speech signal.

Figs. 5.16 to 5.18 show the characteristics of P_{64}, P_{32} and P_{16} as a function of packet length of duration P/f seconds. These characteristics have been obtained by using equations (5.13)-(5.18).

The Markovian simulation technique has been used to test the analytical model. In this simulation, for each frame the total number of packets which are required to be operating with the appropriate bit rate have been evaluated by using equations 5.15 and 5.18. At the end of simulation (i.e. 20,000 seconds of real time), the probabilities have been obtained from the ratio of the total packets being operated with a given encoding output rate to the total of the packets being generated from all the connected channels during the simulation period. The results are given in figs 5.16 to 5.18. As can be observed, each of these simulation results are in close agreement with the associated analytical results.

The characteristic of TASI advantage as a function of packet length for this scheme, as given in fig. 5.15, shows that for longer packet lengths, higher TASI advantages result. Fig. 5.19, however, shows the opposite situation relating to the speech quality. As can be observed for longer packet duration, there are on average less bits assigned per speech sample and therefore the quality of speech is degraded. For example, for the packet duration of 16 ms, the TASI advantage of the system, as can be observed from fig. 5.15, is 2.833. While under this condition the system is expected to be operating at an average of 43.27 kb/s as shown by fig. 5.19. The characteristic of fig. 5.19 has been obtained by first evaluating P_{64}, P_{32} and P_{16} using the equations (5.12) - (5.18) and then by using equation (5.19). The results for this figure have also been compared with computer simulation results obtained for this system, this shows that the two results have a close agreement.

The results, as given above, obtained here for this system correspond to the 'worst case design' based on the assumption that all the connected channels are in their conversational modes. In practice, this assumption is not likely and the occurance of the above event will happen with a very low probability. The system, therefore, is expected to be normally operating with a higher average bit rates and thus give better speech quality. This problem will be dealt with in Section 5.8.

5.5.3 Comparisons of the two schemes

Fig 5.20 shows the comparisons of the two packet organization techniques with respect to the characteristics of TASI advantage as a function of packet length. The results show the superiority of the adaptive voice flow control technique.

5.6 Integrating Voice and Data using the NSQ Scheme

In this section a two node, totally packetized integrated voice and data model has been studied. The operation of the system is according to the following criteria:

(1) The offered load voice traffic, A, is assumed to be infinite.

(2) The voice traffic is assumed to be serviced according to a loss basis. This means that if a voice call arrives and finds all the circuit channels busy, then the call is rejected. The above two assumptions suggest that the circuit channels dedicated to the voice traffic at the transmitter are always fully loaded and therefore there are no statistical fluctuations in the number of voice calls in the system.

(3) The transmission switching scheme for the voice calls through the network is assumed to use the NSQ type of packetized voice transmission. This assumption then suggests that the transmission policy between transmitter and receiver is based on sending and receiving fixed frames each of duration Δ_t seconds.

(4) The priority is always given to the voice traffic so that the maximum interpolation factor (or the TASI advantage) is achieved for the voice traffic. The mean percentage of voice packet lost is set to be below, or equal to 1%.

(5) The data traffic is assumed to be serviced according to a delay basis, and the data packets are assumed to be fixed and to arrive according to a Poisson process at a rate θ packets/

sec. The data packets are queued into an infinite buffer and then are transmitted through the network according to a first-come-first-served (FCFS) basis.

According to the above assumptions, the whole operation of the system could be characterized by transmitting frames of fixed duration Δ_t second through the network. As the priority is given to the voice packets, then the data packets are sent through the network only if there are available packet slots for them (i.e. data packets are sent only during the voice silent periods). In the following, we investigate the queueing performance of the data packets in this type of integrated system.

5.6.1 Simulation model

The simulation model for the integrated system has been developed by extending the Markovian simulation model, as already constructed for the packet voice traffic. This was done by setting up a queueing model for the data traffic in the simulation program. The queueing process is shown in fig. 5.21. In this model the number of data packets in the queue are computed at the beginning of each frame. For example, as in fig. 5.21, the number of data packets at the beginning of the i-th frame is assumed to be $n_d^{(i)}$. The total number of data packets sent by the channel up to the beginning of the (i+1)th frame is given by

$$n_{-} = \left\lfloor \frac{e - n_v(t) P_1}{P_2} \right\rfloor \qquad (5.20)$$

where e is the frame length in bits and is equal to $C\Delta_t$, $n_v(t)$ is the number of voice packets in the given frame and P_1 and P_2 are the sizes of the voice and data packets respectively. As the data packets are assumed to arrive in a Poisson process with rate θ packet/s, then number, n_+, of data packets arriving during the frame interval of duration Δ_t seconds is drawn from a Poisson distribution with mean $\theta\Delta_t$. Thus the number of packets in the queue at the (i+1)th-frame is

$$n_d^{(i+1)} = \max(n_d^{(i)} + n_+ - n_-, 0) \qquad (5.21)$$

The process is repeated for a large number of frames and the average number of packets in the queue, $\bar{n}_d$, is computed. The average waiting time in the queue for a data packet is then determined by Little's result [23] and is given by

$$\bar{q}_d = \frac{\bar{n}_d}{\theta} \qquad (5.22)$$

5.6.2 Approximated queueing model

In order to evaluate the queueing performance of the data packets analytically and to compare the results with the simulation results, Gitman's M/G/1 analytical queueing approach [58], [59] has been used. This queueing model is depicted in fig. 5.22 and is based on a single server queue where capacity is partially available for store-and-forward packet switching. Using this model, the steady state average data packet link delay, W_d, is given by

$$W_d = \frac{P_2(2c-\theta P_2)}{2c(c\sigma-\theta P_2)} \qquad (5.23)$$

where P_2 is the size of the data packet in bits, c is the rate of transmission link in bits/sec, θ is the average data packet arrival rate per second and σ is the probability that the channel is available for the data packets.

In the above equation all the terms are known except σ. The derivation of this term is as follows:
In a long run of system operation, the average portion of channel per frame used by the voice packets is equal to nq voice packets. This is due to the Binomial probability distribution of the voice packet arrivals per frame, as discussed previously. For a frame length of Δ_t seconds duration, the portion of the frame on average, U_d, available for the data packets in bits is given by

$$U_d = c\Delta_t - nqP_1 \qquad (5.24)$$

where P_1 is the voice packet length in bits and is equal to

$$P_1 = Pb + h \qquad (5.25)$$

Since the end of the frame may be unused and no data packet could be used to fill it in, the average portion of the frame allocated to the data packets in bits, I_d, becomes

$$I_d = \left\lfloor \frac{U_d}{P_2} \right\rfloor P_2 \qquad (5.26)$$

where $\lfloor x \rfloor$, as previously, denotes the greatest integer less than or equal to x.

Thus the average probability that the channel is available for the data packets becomes

$$\sigma = \frac{I_d}{c\Delta_t} \qquad (5.27)$$

where $c\Delta_t$ denotes the frame length in bits.

5.6.3 Results and comparisons

The integrated system is assumed to have the following fixed parameters:

(1) T_1-carrier transmission link (24 channel)

(2) 64 kb/s encoder output per voice channel

(3) 100 bits packet overhead

(4) 10 ms frame length. This sets the voice packet length, including the overhead, to 740 bits and the TASI advantage to 1.541 (from fig. 5.4) or 37 in total simulationed subscribers.

(5) P_2, the data packet size, is also assumed to be 740 bits, i.e. equal to the voice packet size (i.e. $P_1 = P_2 = 740$ bits)

Fig. 5.23 shows the results of the queueing performance of the data packets obtained from both analytical and simulation models. The analytical characteristic shows that a reasonably good use of the free channel capacity is achieved and the rate of the data packets could be approximately pushed up to an average of 4 packets/ frame. However, the simulation behaviour of the system (with the simulation period set at 20000 seconds) shows that the data packets fail to utilize the full free channel capacity which is available for them. The reason for this discrepency is due to the existance of the time dependence frame correlations which have been ignored in the analytical model. This effect will give rise to a significant buffer build up during the periods when the number of voice packets arriving per frame exceeds J packets (where J is the total number of speech packets sent by the channel in a frame). As previously discussed, due to the nature of state transitions in the talkspurt/silence Markov chain model (see fig. 3.9) and also the local behaviour of the chain, it is possible that the system remains occasionally in the set of states where J is exceeded for a comparatively long period, i.e. for large number of frames. During these periods no data packets can be serviced at all by the channel. The data packets which arrive with rate θ packets/sec will then flood the buffer queue. This results in significantly higher queueing delays than the theoretical model would predict.

The integrated system which has been studied here was based on the assumption that the voice call traffic arrival to the transmitting node was very high (i.e. infinite) so that all the voice channel were constantly busy. In the next two chapters, various other types of integrated systems have been studied and the voice-call traffic has not been made so unrealistically high, this results in statistical fluctuations in the number of active voice calls at the transmitting node and hence more data traffic can flow..

A similar type of investigation could have been carried out in the integrated system using the SQ-system rather than the NSQ-system. However, in the SQ-system the data traffic performance would have been expected to be even worse than for the above scheme. The reason being in the SQ situation there would be another factor which would increase the waiting time of the data packets. This factor is the buffer occupancies of the SQ-system (i.e. the data packets would be transmitted only during the periods where there are no voice packets waiting in the queue buffer). Forgie and Nemeth (in Ref[43]) investgated this type of integrated voice and data, they evaluated the queueing performance of the data traffic by using an M/G/1 queueing model. As shown for the NSQ simulations above, it can be concluded that the M/G/1 queueing model is not a valid model for the evaluation of the queueing performance of the data traffic in this type integrated voice/data system.

5.7 Miltinode Networks using the NSQ Technique

Another advantage of using the NSQ system is that the performance characteristics of multinode networks can easily be evaluated when using this form of packetized transmission.

In general the networks can be of any type of connected topology. The routing for the calls after dial up is assumed to be fixed. The communications between the nodes are based on transmitting and receiving information packets in fixed frames of duration, Δ_t seconds, which are equal to the packetization period.

In appendix (C), it has been shown that for this type of network both the end to end packet delay and the packet loss probability are dependent upon the number of nodes which have to be transversed. An example is given in appendix (C), this assumes the links are of a 1.544 Mb/s T1-carrier form. For this configuration, the analytical results indicate that the TASI advantage of the network becomes a function of the maximum number of traversed nodes. For example, it has been shown that for a call to traverse up to a maximum of 7 nodes, the TASI advantage of the network could be as high as 1.4. This value then implies that all the nodes in the network could be set up to support up to a maximum of 34 simultaneous calls. Thus at dial up, the routing which is assigned to a call should be such that it will firstly allow a call to be traverse no more than 7 nodes, and secondly not to cause the number of simultaneous call connections per node to be greater than 34.

The results indicate that by allowing fewer nodes to be transversed better TASI advantages could be achieved for the network.

5.8 The Effect of Telephone Traffic Loading on the NSQ-system

The work which has been reported so far on the NSQ-system, has been based on the 'worst-case' assumption, that is, when all the channels are connected and are in conversational mode. We consider the fully loaded situation as the number of simultaneous users change very slowly compared with the rapid fluctuations due to the conversation itself. Hence when the system happens to be in the fully connected situation it is likely to stay in that state for some second, the operation for this period will be as determined for the 'always-fully-loaded' situations. In chapter 4, it was shown that it is possible to increase the TASI advantage of a SQ-system in the presence of the voice traffic fluctuations, only if one allows some of the calls during the busy periods to experience packet losses greater than 1%. It was also shown that under this criteria, the potential TASI advantage for the system becomes dependent upon the blocking probability. The same criteria has also been applied here for the NSQ-system, however here, we have been able to use mathematical models for the evaluation of the TASI advantage performance. The mathematical model used has been based on an extension of the Weinstein packet loss probability expression used for the NSQ-system. The same threshold for the good call connections, (95% with < 1% freezeout), as used for the SQ-system, has also been applied to the NSQ-system. That is, to say, it has been assumed that for at least 95% of the time the calls should experience packet losses equal to, or less than, 1%.

5.8.1 The mathematical expression

In order to evaluate the TASI advantage of the NSQ-system under the above assumptions and criteria, we refer to the Markovian voice call initiation and termination model, as depicted in fig. 5.24. As the number of call connections increase, the probability of packet losses in the system increases. This is because the transmission link is of constant bit rate and it can only transmit up to J packets per frame. By using the Weinstein packet loss probability expression (i.e. equation 5.5), a number of call connections, k, can be found so that the steady state packet loss probability, or the freezeout fractions just exceed 1%. The problem now is to evaluate the total number of output channels, n, so as to set the probability of meeting the states between k to n to be less than, or equal to 5%. This also implies that at least 95% of the time the calls meet the condition that the packet loss probability is less than or equal to 1%. The equation below presents the above arguments.

$$\sum_{i=k}^{n} P_i \leq 5\% \tag{5.28}$$

where P_i represents the probability that the number of call connections is i. This is derived from the Erlang distribution and is given by

$$P_i = \frac{\dfrac{A^i}{i!}}{\displaystyle\sum_{r=0}^{n} \frac{A^r}{r!}} \tag{5.29}$$

5.8.2 Results and Comparisons

A similar approach to that carried out for the SQ-system, as described in the previous chapter, has been used here to evaluate the TASI advantage performance characteristic of the NSQ-system for a given blocking probability. This means that by using the False-position method, for each value of n, the voice traffic intensity, A, has been obtained so that the given blocking probability is met. Fig. 5.25 shows the characteristics of TASI advantage versus packet length obtained for the NSQ-system for different sets of blocking probabilities. These characteristics have been evaluated by using the criteria which have been set in the previous sub-section. That is, at least 95% of time the calls experience packet losses less than, or equal to, 1%. The results clearly show that as long as the blocking probability does not exceed 5%, then the TASI advantage of the system becomes dependent upon the blocking probability. It is also higher than the TASI advantage which has been obtained using the 'all channel busy' assumption. For example; as is shown in fig. 5.25, when the blocking probability is $\leq$ 5%, the TASI advantage is inversely related to the blocking probability. That is, as the blocking probability is reduced the TASI advantage increases, and visa versa. It is also shown that as the blocking probability exceeds 5%, the TASI advantage characteristic becomes asymptotic to the TASI advantage characteristics obtained for the fully loaded channel situation. The reason for this is that when we have a blocking probability of more than 5% most of the channels will be connected, and so the number experiencing more than the permitted 1% packet losses tends to exceed the 5% limit. The overall packet loss probability in this situation should therefore be kept below 1% in order that equation (5.28) can be satisfied.

Since the TASI advantage is directly proportional to the number of output channels from the exchange, n (i.e. TASI advantage=n/24), then the above results reveals that the system within the 5% blocking probability range can have a greater number of output channels than that for the 'always-fully-loaded' system. This conclusion in turn implies that the system within this 5% blocking probability range can handle more telephone traffic, A, than that when employing the all channel busy assumption.

Fig. 5.26 shows further results obtained using the above criteria when the packet length has been fixed at 16 ms. This figure shows the characteristic of TASI advantage versus the nominal number of channels. Results have been obtained for different sets of blocking probabilities. These curves also indicate the dependence of the TASI advantage upon the blocking probability.

5.8.3 Adaptive voice flow control technique with the telephone traffic loading

Since in this type of NSQ-system no packet losses should take place, then the TASI advantage of this system will not be affected by the offered voice traffic intensity and is given by equation (5.9). However, the probabilities of usage of each of the encoding output rates for each subscriber channel will be affected by the offered telephone traffic. By using the quasi-static approximation method, these new probabilities (i.e. P_{64}, P_{32}, and P_{16}) have been formulated here as follows:

$$P_{64} = \sum_{i=1}^{n} P_i \, P_{64}(i) \quad . \qquad (5.30)$$

Using equation (5.12), $P_{64}(i)$ is given by

$$P_{64}(i) = \frac{1}{iq} \left\{ \sum_{r=0}^{J_1} rB(i,r,q) + \sum_{r=J_1+1}^{J_2} (r-m_{32}(r))B(i,r,q) \right\},$$

and

$$P_{32} = \sum_{i=J_1+1}^{n} P_i \, P_{32}(i) \qquad (5.32)$$

From equation (5.16), $P_{32}(i)$ is given by

$$P_{32}(i) = \frac{1}{iq}\left\{\sum_{r=J_1+1}^{J_2} m_{32}(r)B(i,r,q) + \sum_{r=J_2+1}^{i} (r-m_{16}(r))B(i,r,q)\right\} \tag{5.33}$$

and

$$P_{16} = \sum_{i=J_2+1}^{n} P_i\ P_{16}(i) \tag{5.34}$$

From equation (5.17), $P_{16}(i)$ is given by

$$P_{16}(i) = \frac{1}{iq}\left\{\sum_{r=J_2+1}^{i} m_{16}(r)B(i,r,q)\right\} \tag{5.35}$$

By using equations (5.30) - (5.35), these probabilities (i.e. P_{64}, P_{32}, P_{16}) have been evaluated as a function of packet length for a 2% blocking probability. This blocking probability has been obtained by first evaluating the total number of subscriber channels, n, for each value of packet length. This is given by equation 5.8, which is

$$n = J_3 = \left\lfloor \frac{pc}{f(pb_3+h)} \right\rfloor , \tag{5.36}$$

then by using the false-position method , the offered voice traffic load, A, has been obtained in order to set the blocking probability to .02.

The characteristics of these probabilities (i.e. P_{64}, P_{32}, P_{16}), as a function of packet length, are depicted in figs 5.27-5.29 together with the characteristics for the previous situation where these probabilities were obtained when the system is fully loaded. From these characteristics, it can be observed that the effect of the voice traffic loading in this situation has caused the probability of usage of 64 kb/s, the best speech quality, to be much increased with subsequent significant decreases in the other probabilities. Thus this effect, as expected, results in increasing the quality of the speech compared with the worst-case-situation where the subscriber channels are fully active all the time.

By using equation (5.19), the average resulting individual channel bit rates have been obtained for this situation and the results are as given in fig. 5.30 together with the results of the fully loaded system. It is evident that the system, operates with considerably higher average bit rates.

5.9 Conclusions

In this chapter the performance characteristics of a pure loss type of packetized voice system, named as the NSQ-system, has been investigated. Although this system was first introduced by Weinstein in Ref [7] further work has been carried out as listed below:

(1) It was shown by means of simulation studies that the Weinstein packet loss probability expression for this system is valid.

(2) It was shown that this scheme has a lower TASI advantage performance than the SQ-system with finite buffering at the transmitter, while it has a better performance than the SQ-system with infinite buffering at the transmitter.

(3) A demultiplexing method was introduced for this scheme so that

the system could easily be adopted into a multinode network model with fixed routing protocols. The Weinstein packet loss probability expression was extended to evaluate the TASI advantage for the network.

(4) Two packet organization techniques were used for increasing the TASI advantage of this system. The first technique was the odd-even sample-interpolation scheme. Weinstein's packet loss probability expression was used to evaluate the TASI advantage performance of this scheme. The second technique was the adaptive voice flow control technique with no packet losses. New formulas have been derived to evaluate the TASI advantage and speech quality performance of the scheme.

(5) Weinstein's packet loss probability expression was extended and a new expression has been derived for the evaluation of the TASI advantage performance of the system with the inclusion of telephone traffic loading. The key criteria for this was that less than of 5% of the calls experienced packet losses greater than 1%. It was shown that the achievable TASI advantage for this system employing this criteria can become a function of the blocking probability and can be higher than the TASI advantage of the 'all channel busy' case, provided that the blocking probability is kept within a 5% range.

(6) Using this scheme, an integrated voice/data model was investigated, this model is similar to that used by Forgie and Nemeth [43]. The aim of this study was to show that the M/G/1 model was not a proper and valid model for the evaluation of the data traffic performance as has been suggested by Forgie and Nemeth in Ref [43].

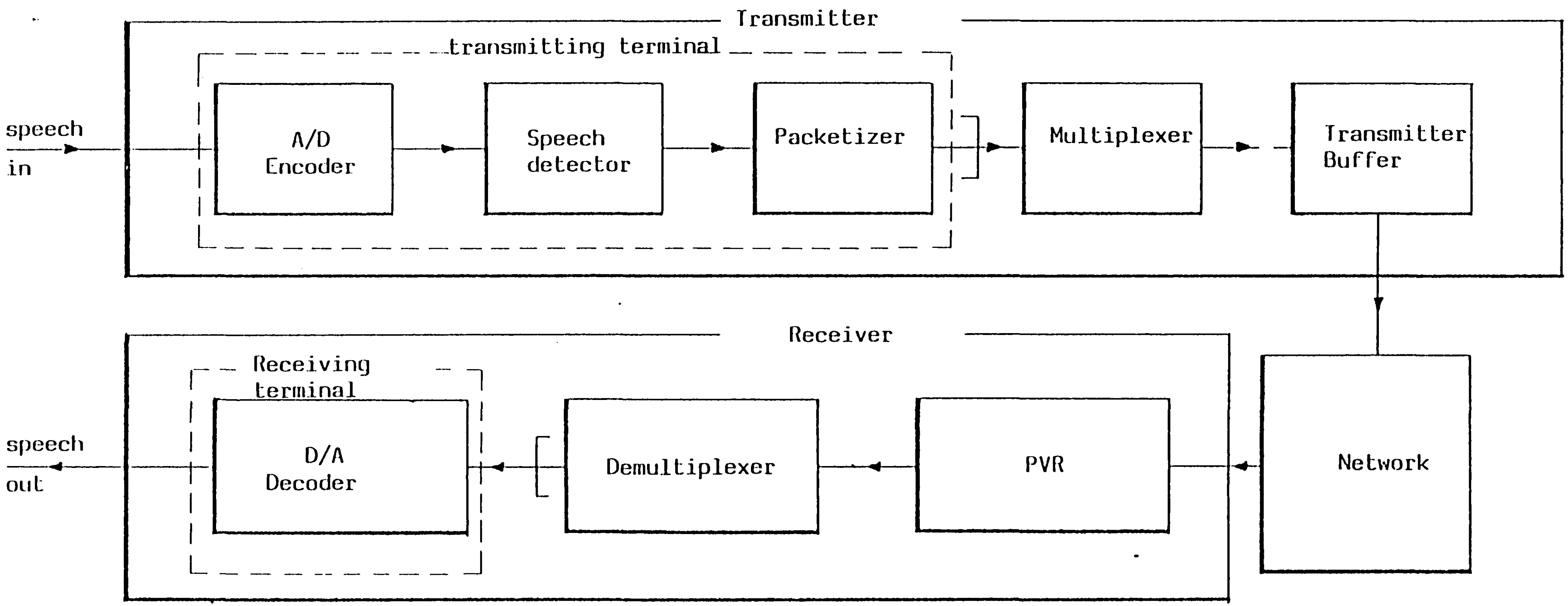


Fig. 5.1 The NSQ packet-voice system block diagram

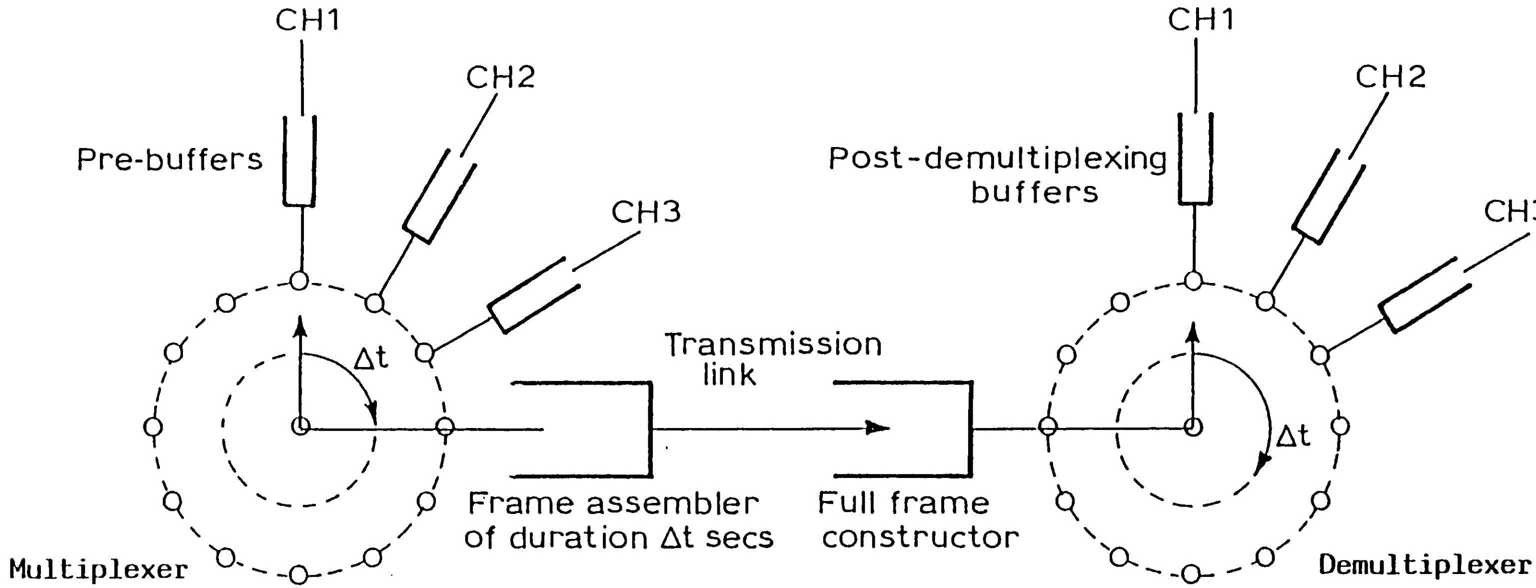


Fig.5.2 *The multiplexing, demultiplexing and bufferings of a two node NSO system*

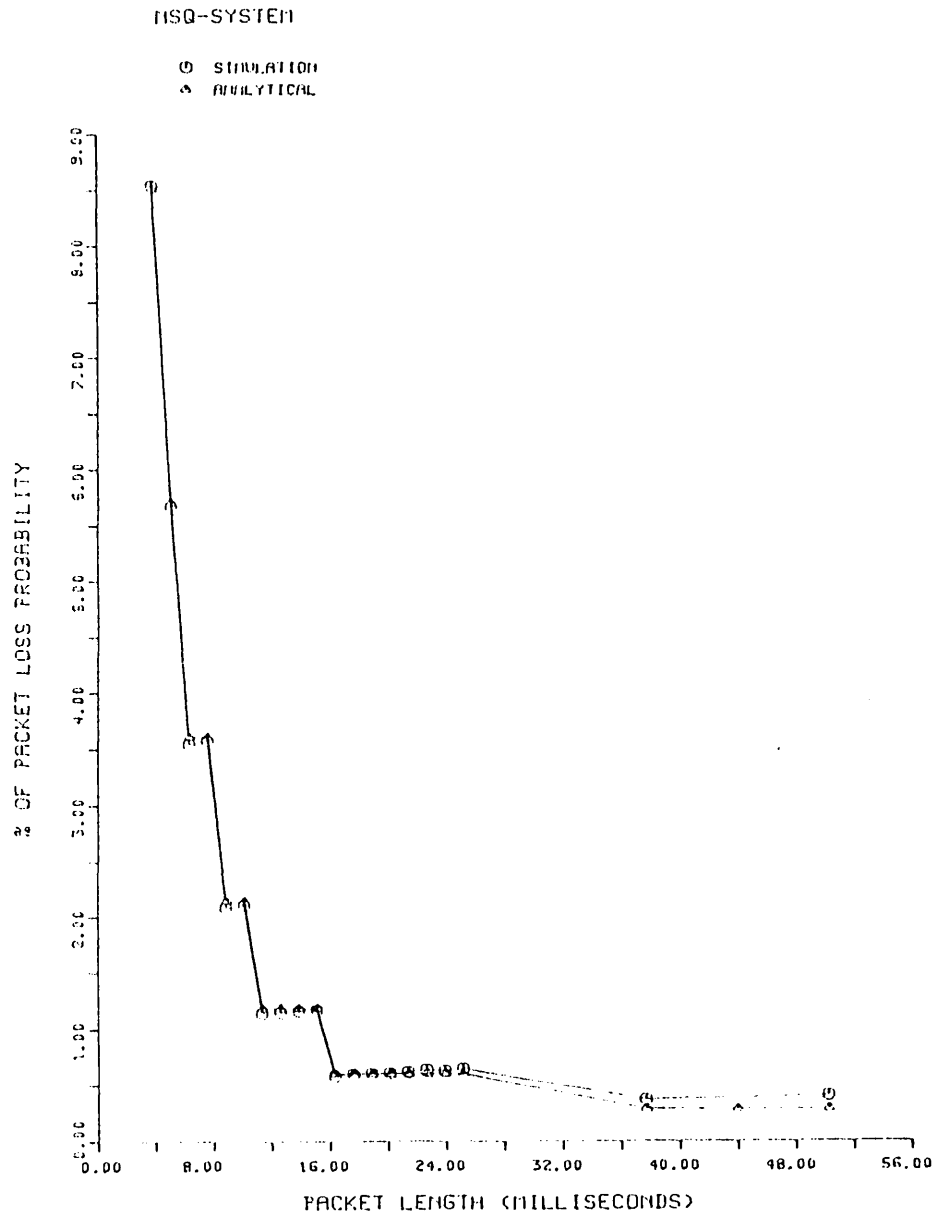


Fig. 5.3 The % of packet loss probability as a function of packet length (analytical and simulation comparisons)

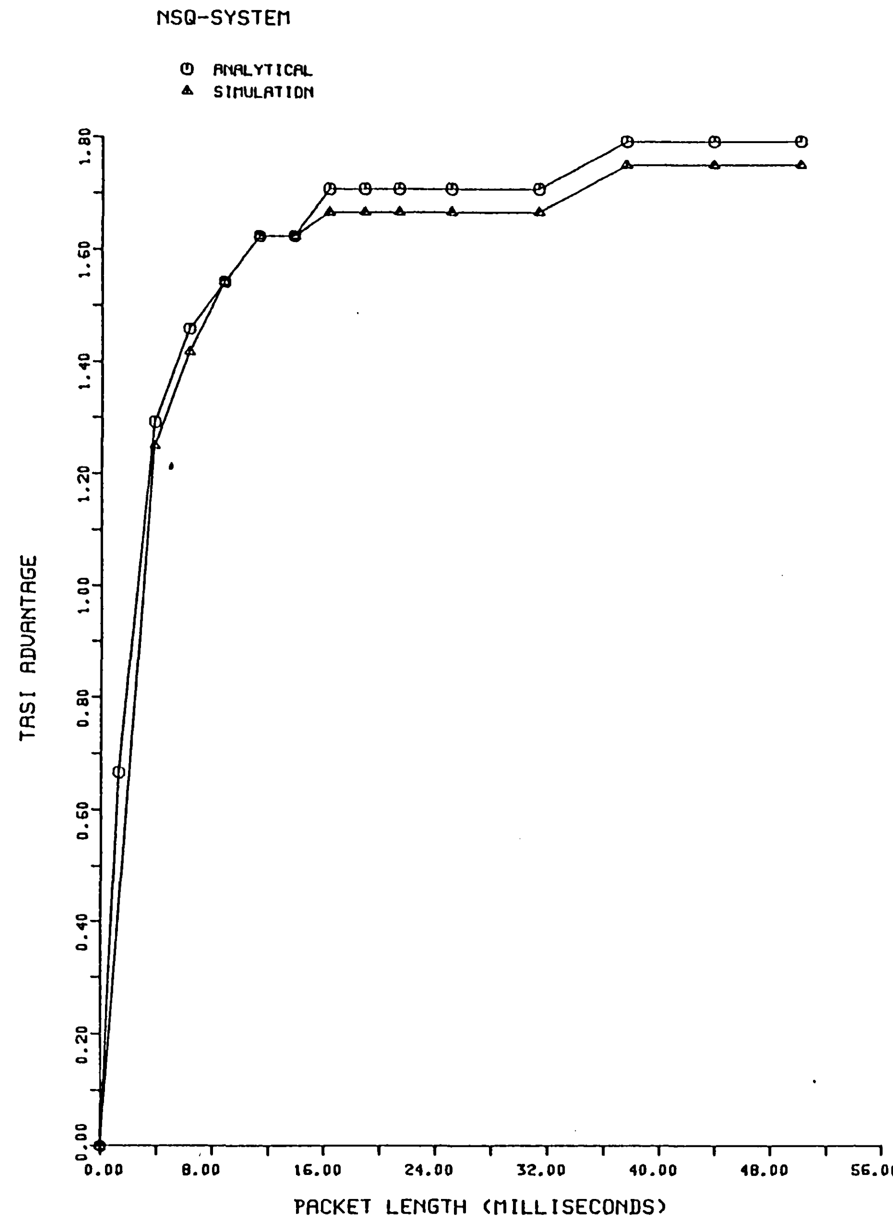


Fig. 5.4 TASI advnatage as a function of packet length (analytical and simulation comparisons)

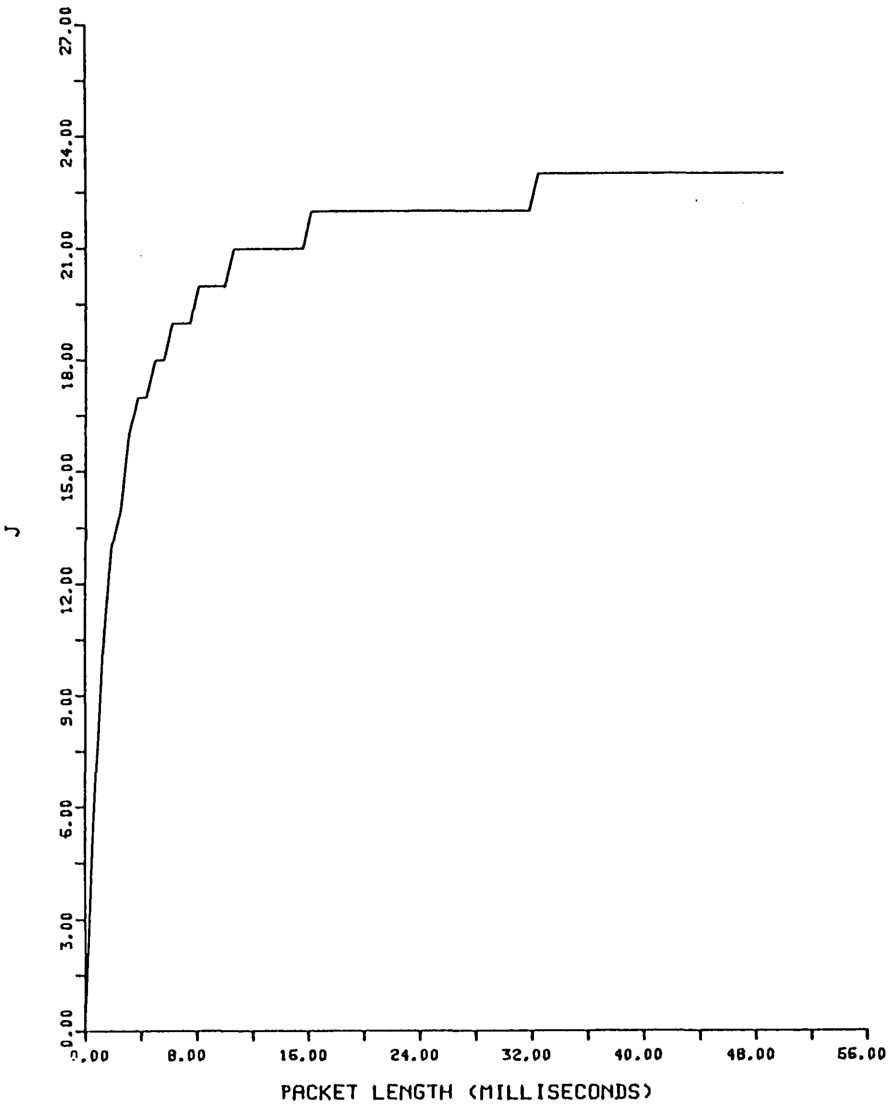


Fig. 5.5 The characteristic of J versus packet length

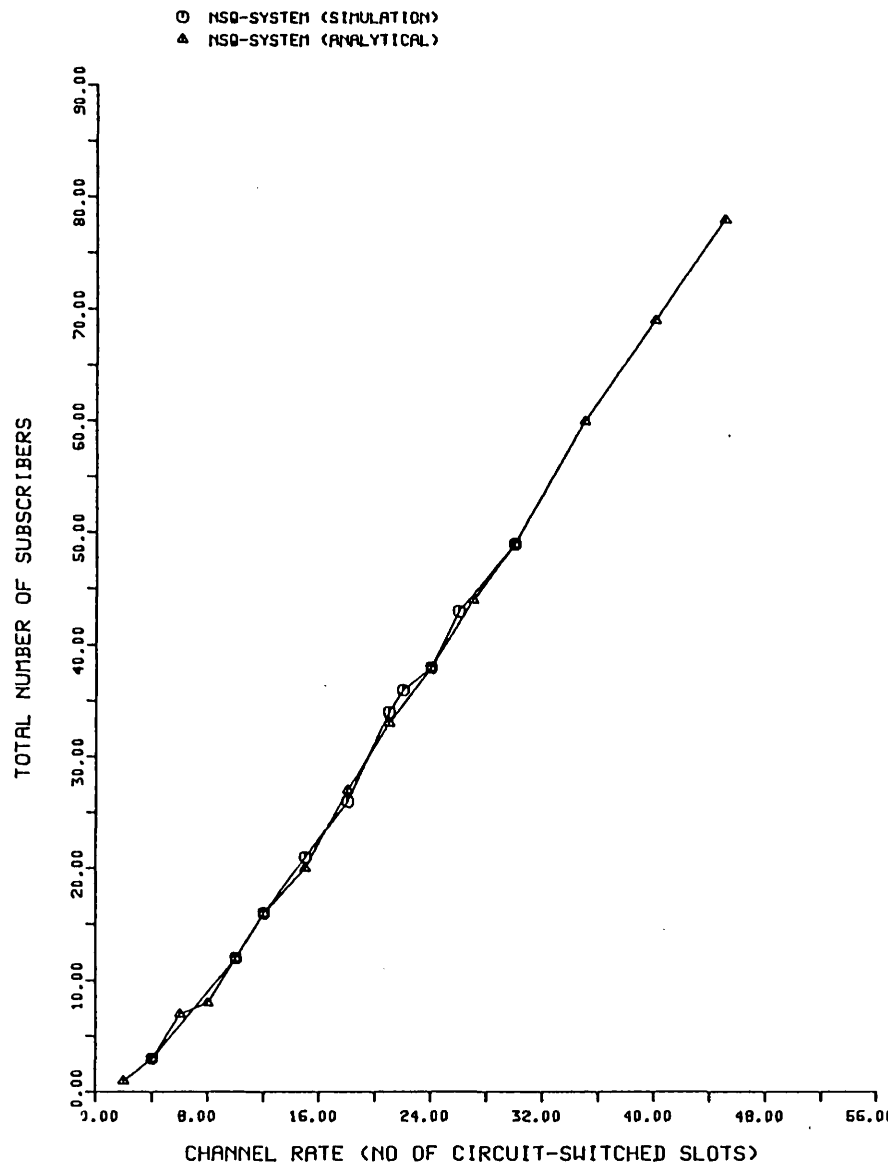


Fig. 5.6 Number of subscribers as a function of channel rate (analytical and simulation comparisons). The % of mean packet losses is ≤ 1.

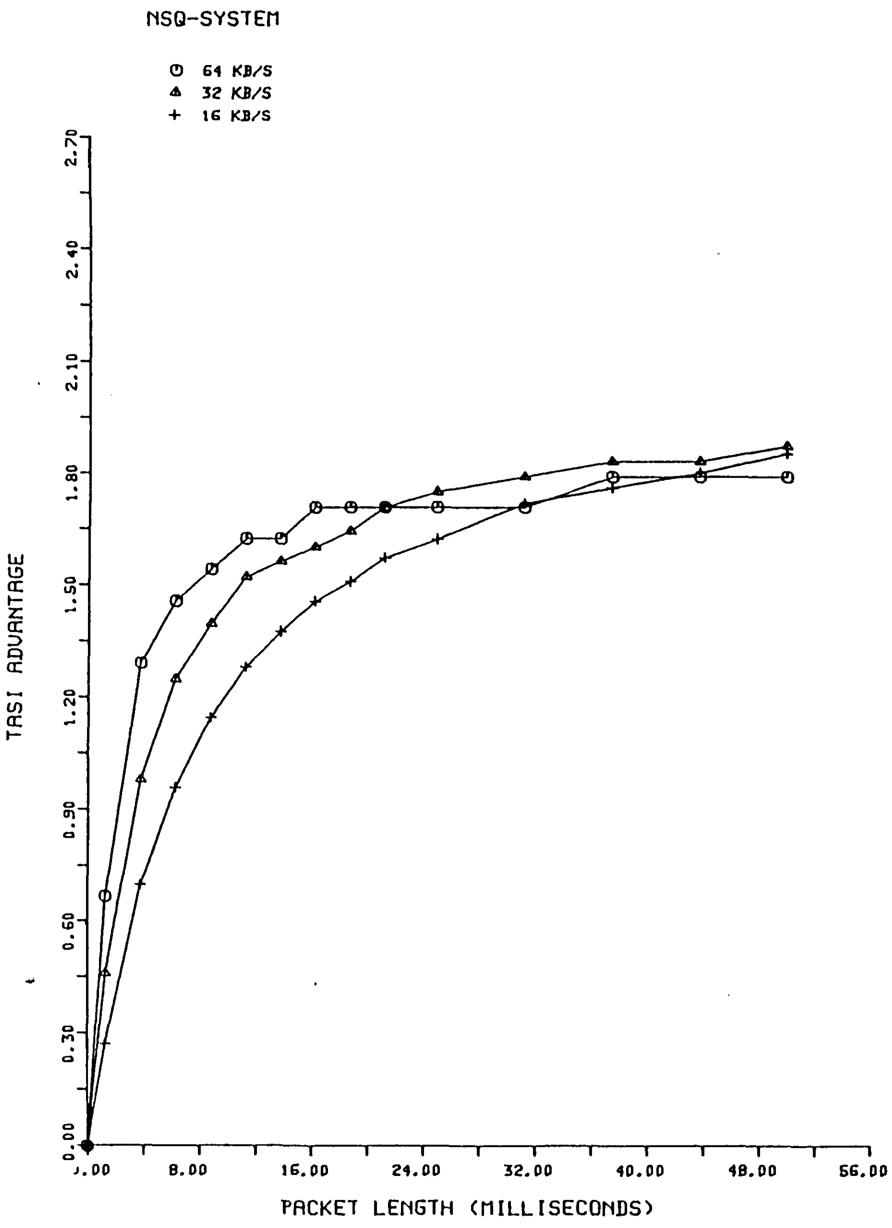


Fig. 5.7 Effect of varying the encoder output/channel on the performance of the system

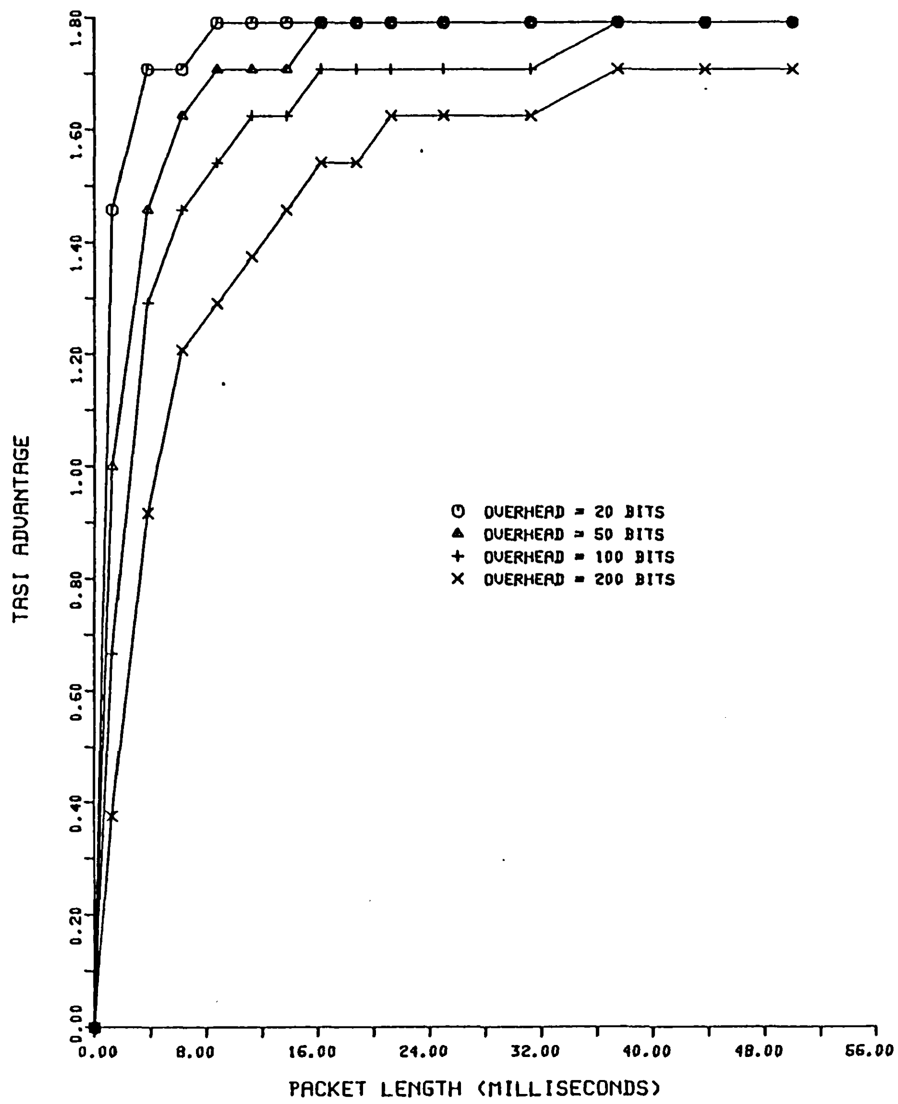


Fig. 5.8 Effect of varying the overhead on the performance of the system.

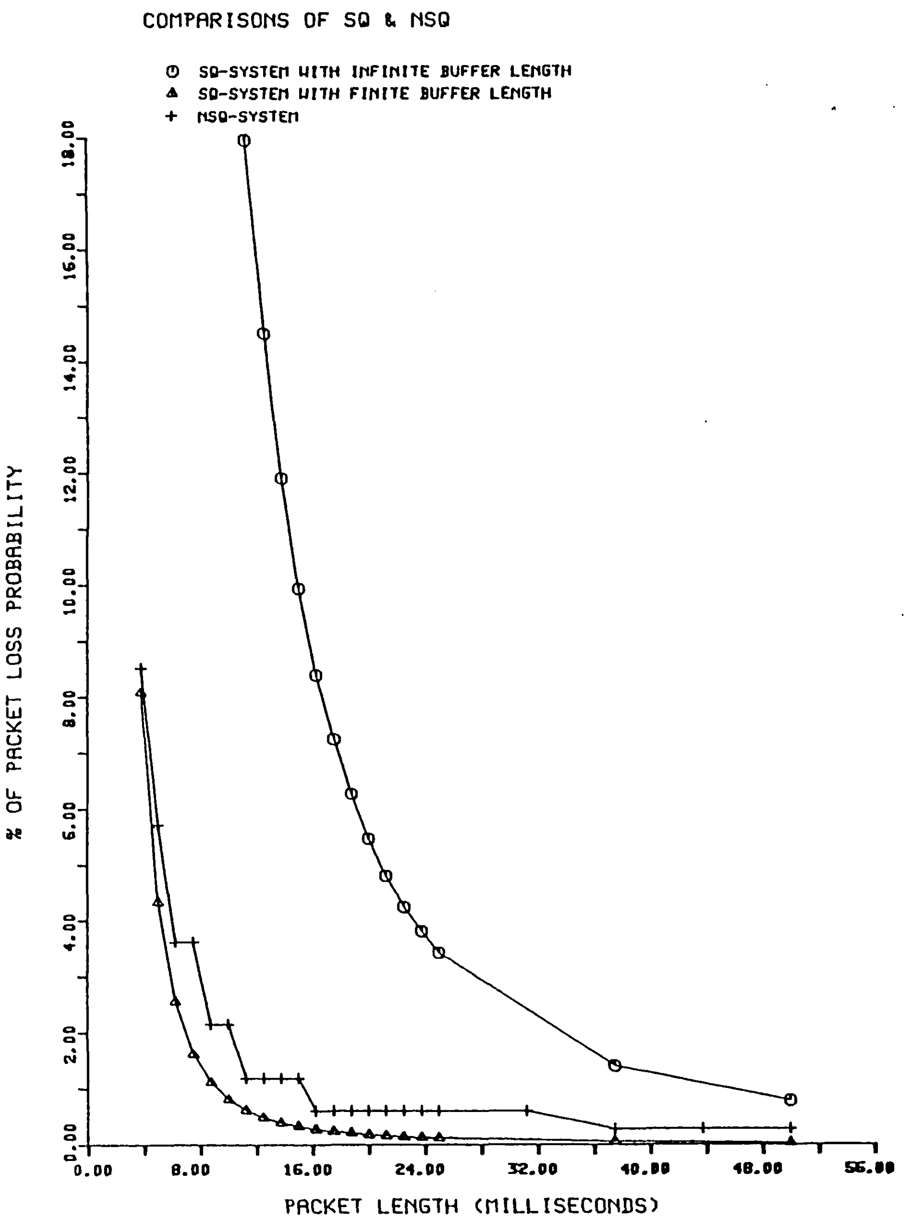


Fig. 5.9 The characteristics of % of mean packet losses versus packet length for the NSQ-system and SQ-systems with and without infinite buffer length. The end-to-end delay is set at three times the packetization delay.

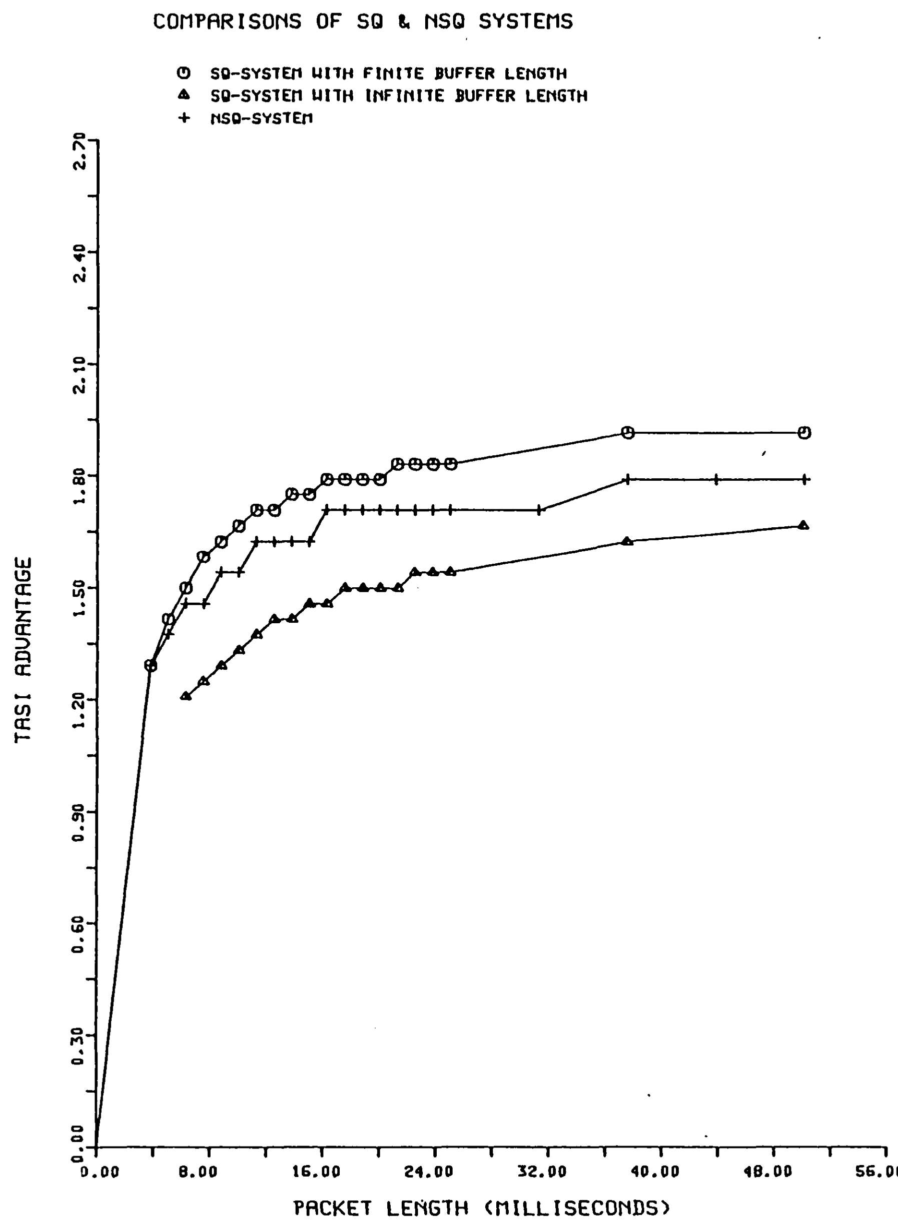


Fig. 5.10 The characteristics of the TASI asvantage versus packet length for the NSQ-system and SQ-system with and without infinite buffer length. The end-to-end delay is set at three times the packetization delay.

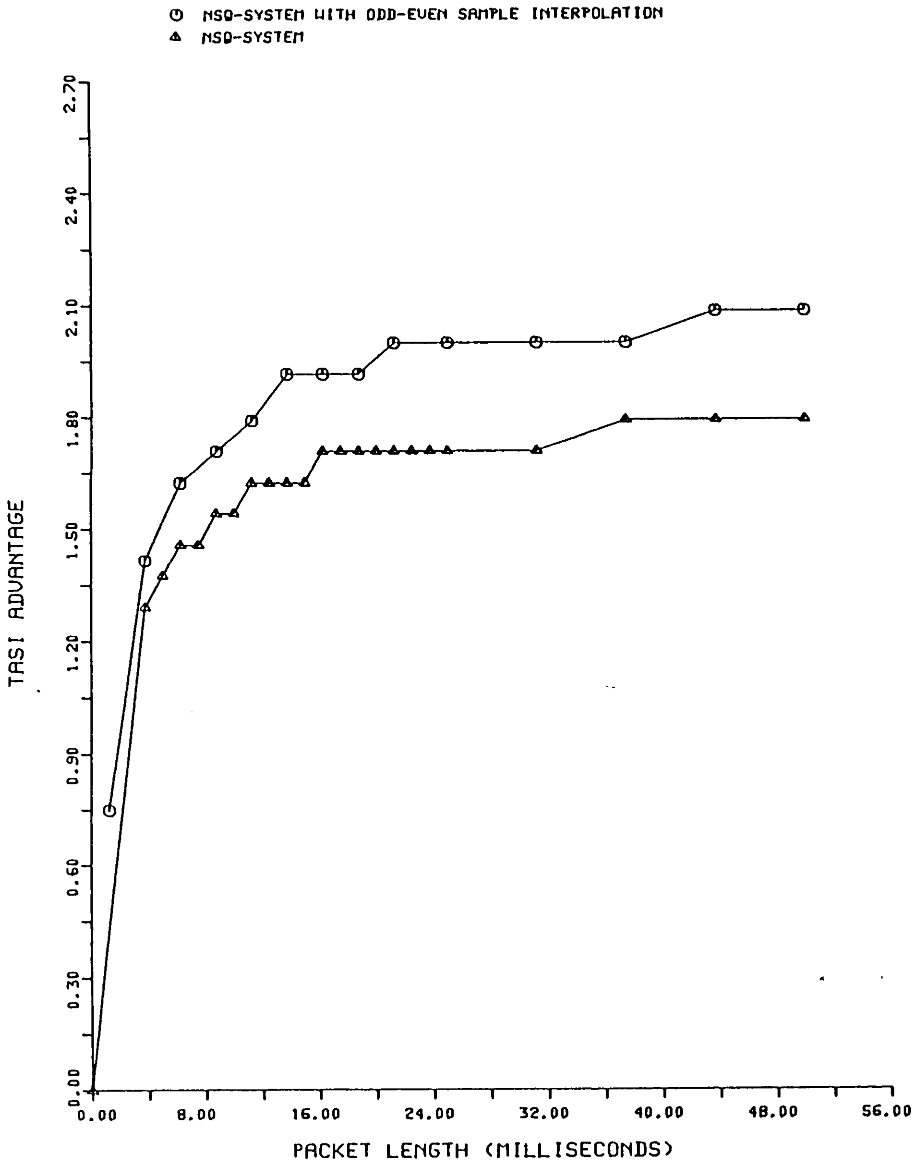


Fig. 5.11 The characteristics of the TASI advantage versus packet length for the NSQ-system and the NSQ-system with odd-even sample-interpolation technique

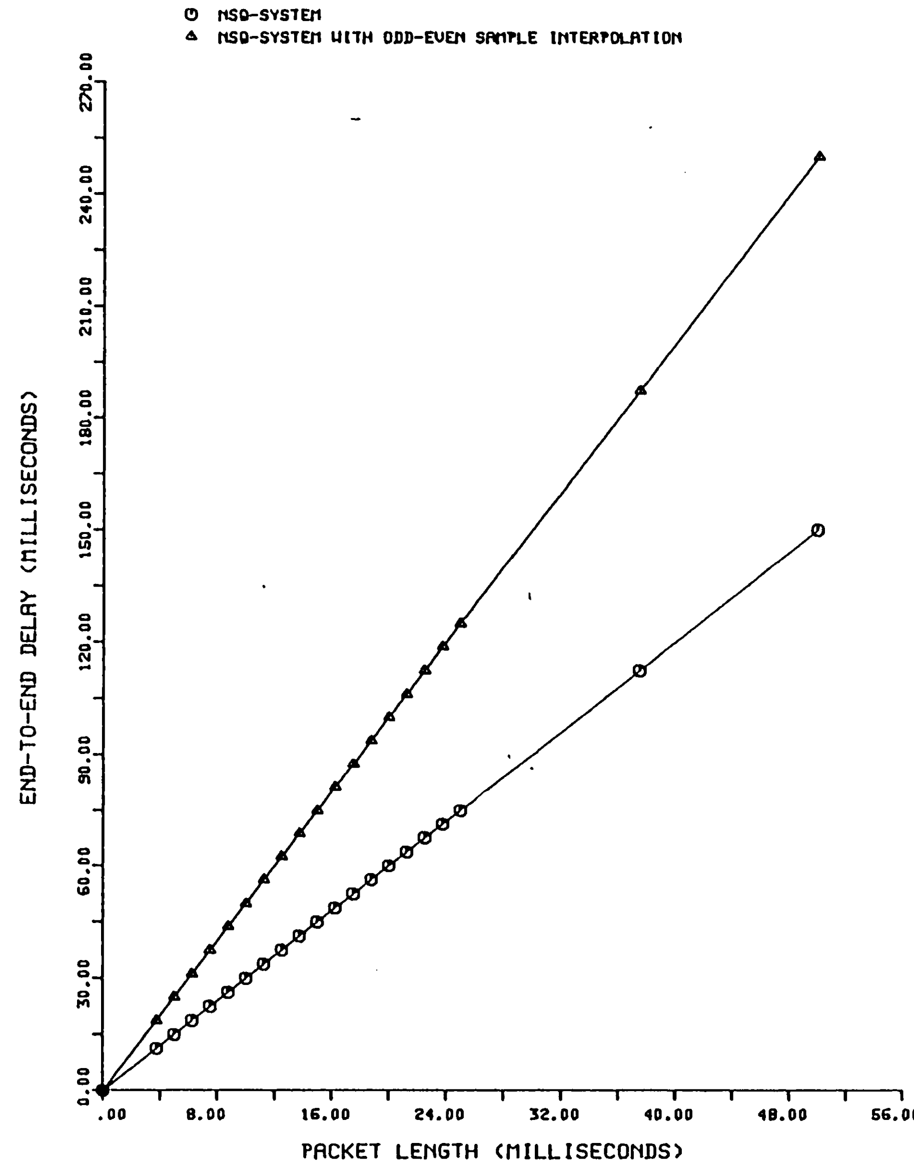


Fig. 5.12 The end-to-end delay as a function of packet length for the NSQ-system and the NSQ-system with odd-even sample-interpolation technique.

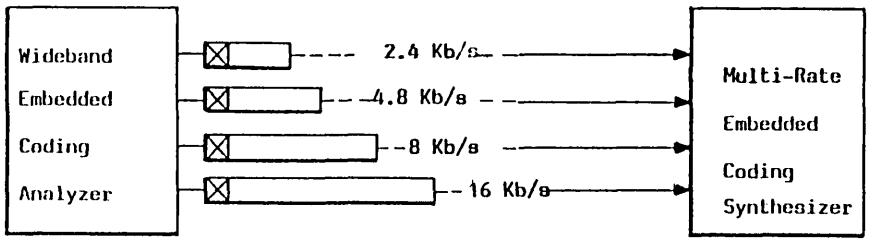


Fig 5.13 Packetized embedded coding concept

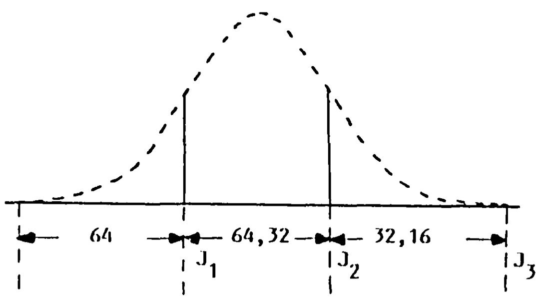


Fig. 5.14 The binomial pdf representing the probability of number of arrival packets per frame

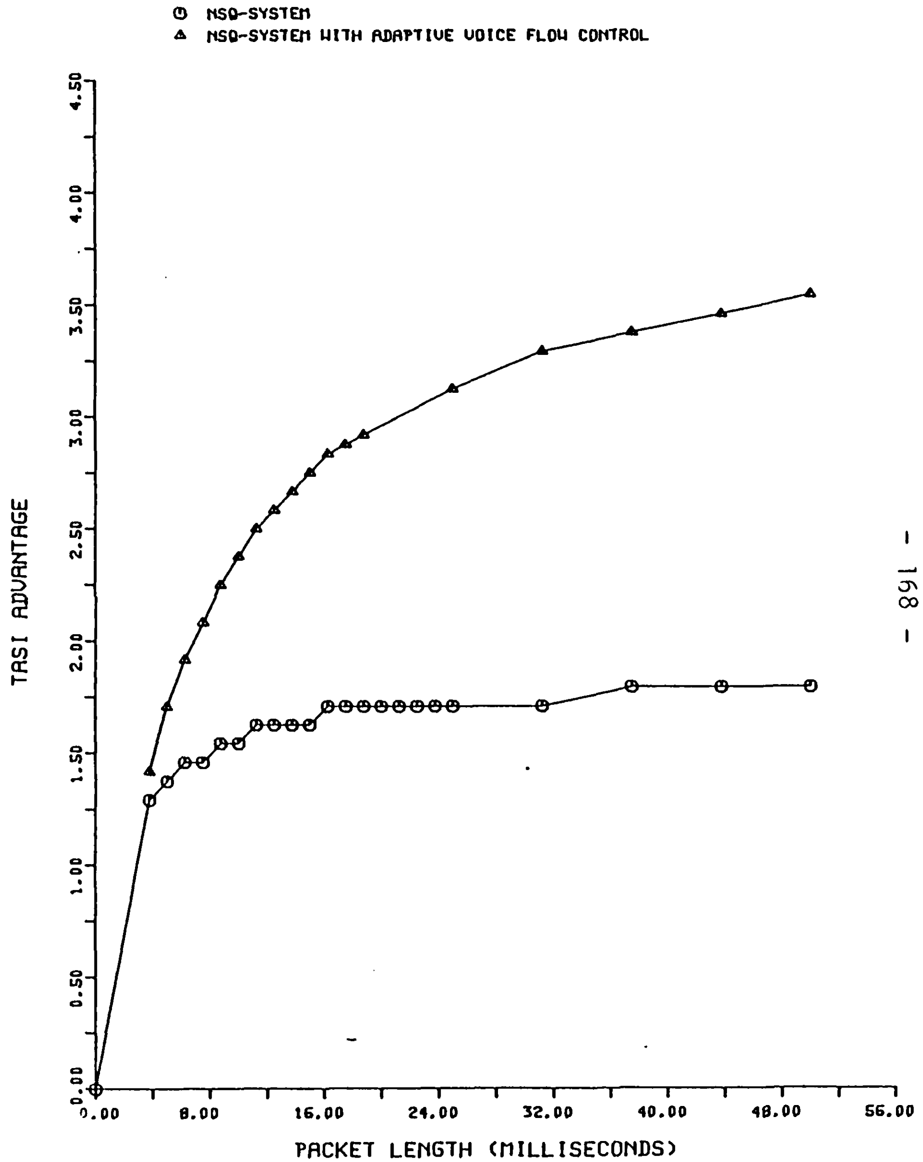


Fig. 5.15 The characteristics of the TASI advantage versus packet length for the NSQ-system and NSQ-system with adaptive voice flow control technique

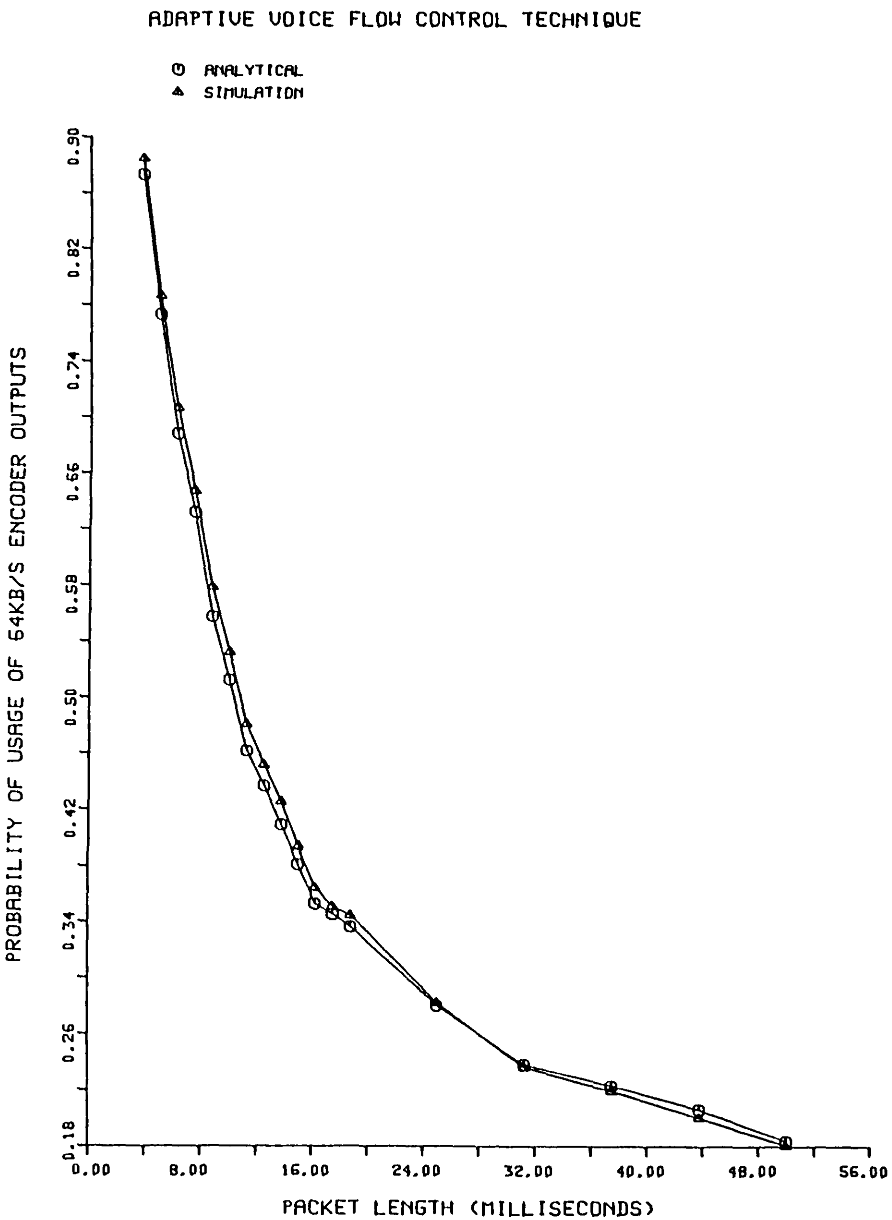


Fig. 5.16 The steady state probability of usage of 64 kb/s encoder output/chanel as a function of packet length.

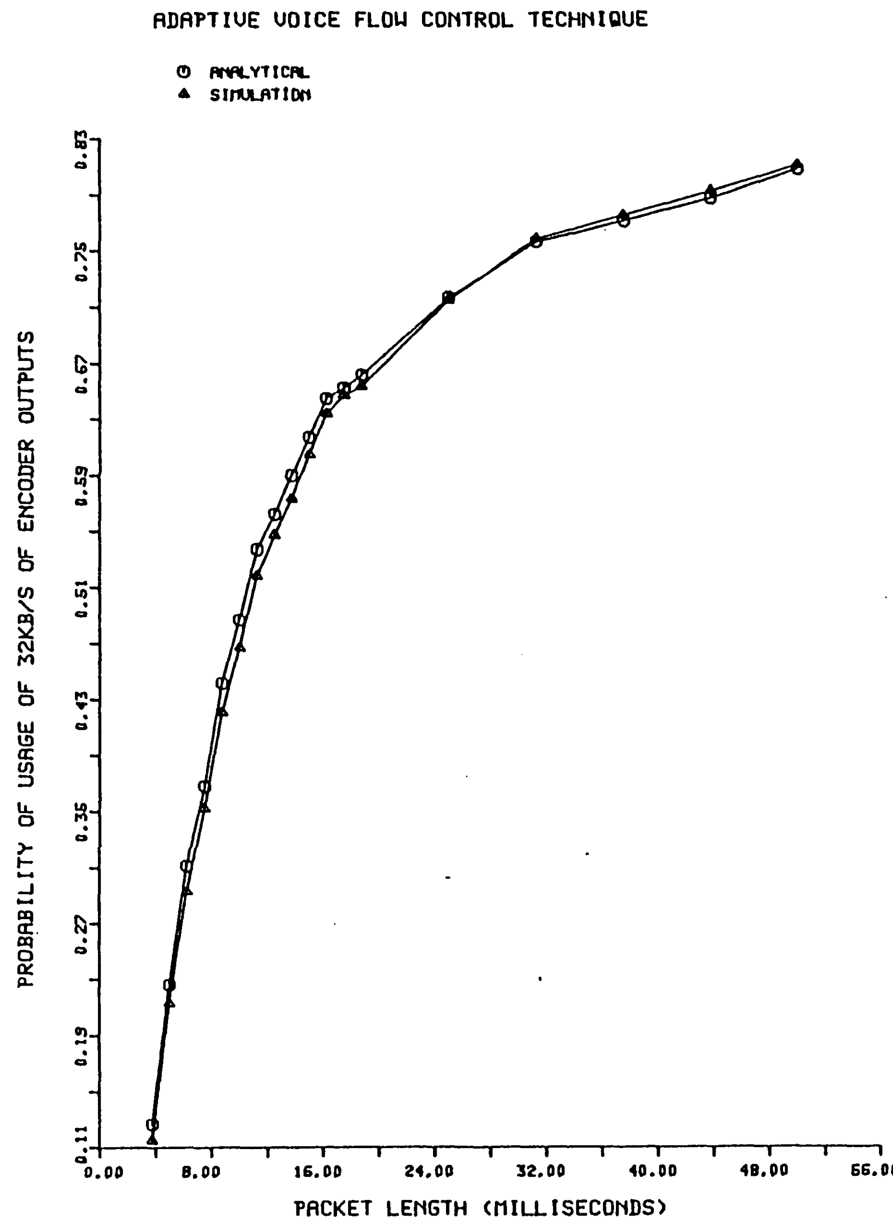


Fig. 5.17 The steady state probability of usage of 32 kb/s encoder output/channel as a function of packet length

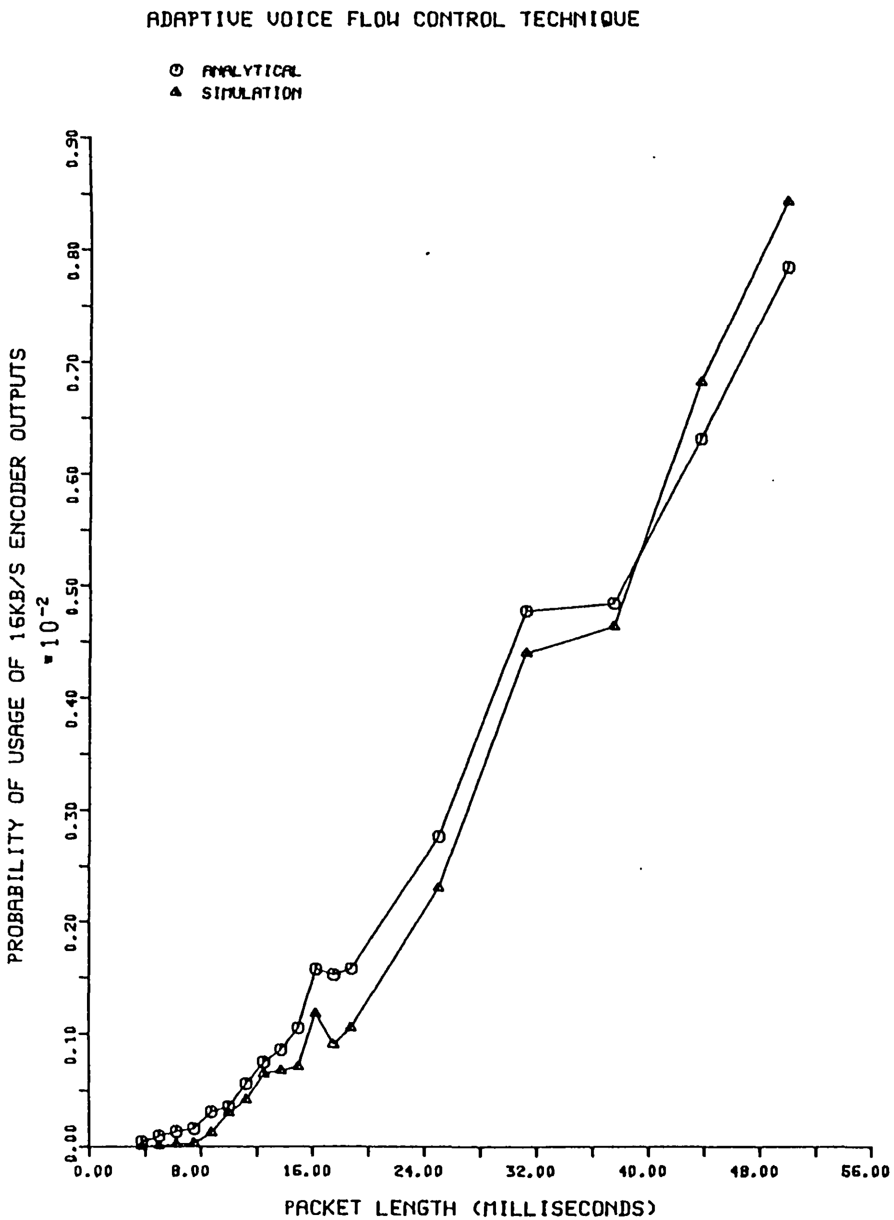


Fig. 5.18 The steady state probability of usage of 16 kb/s encoder output/channel as a function of packet length

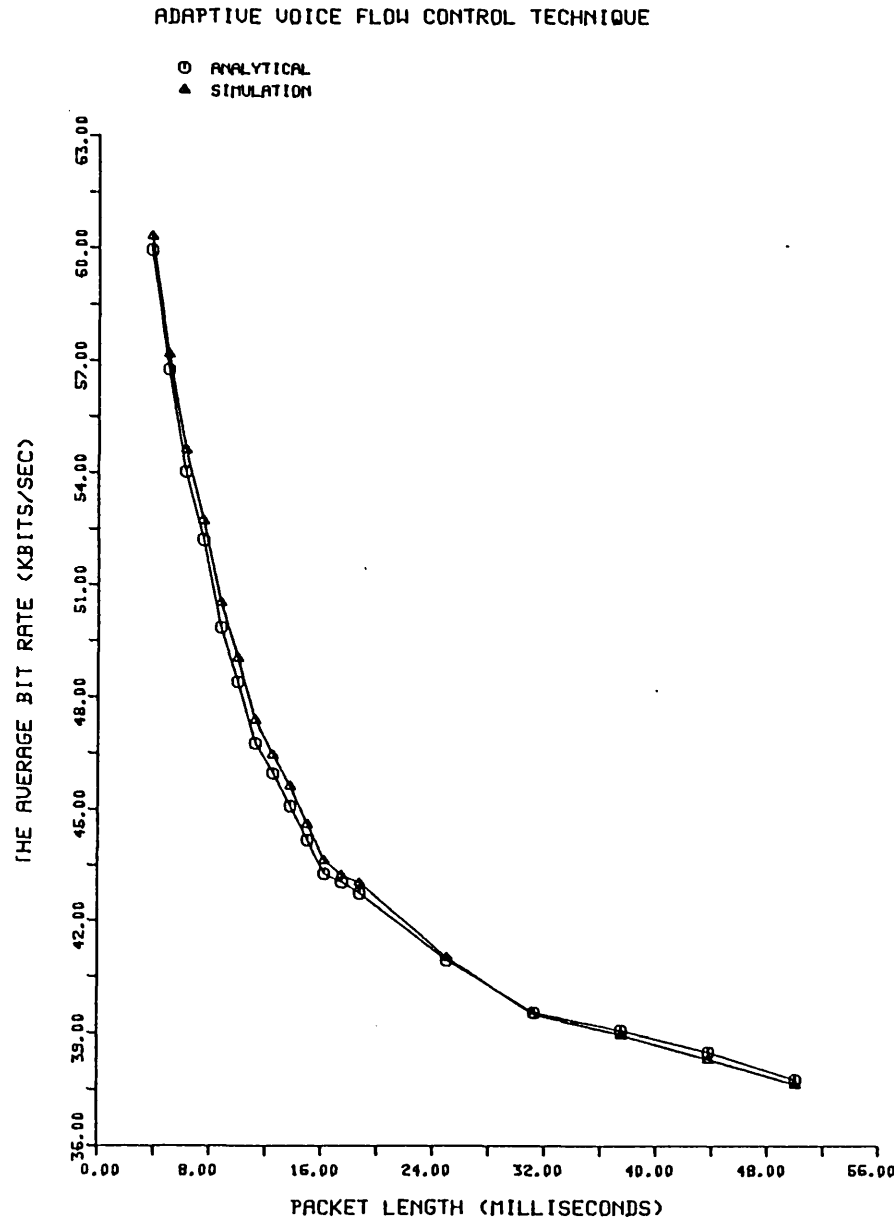


Fig. 5.19 The resulting average encoder output/channel as a function of packet length

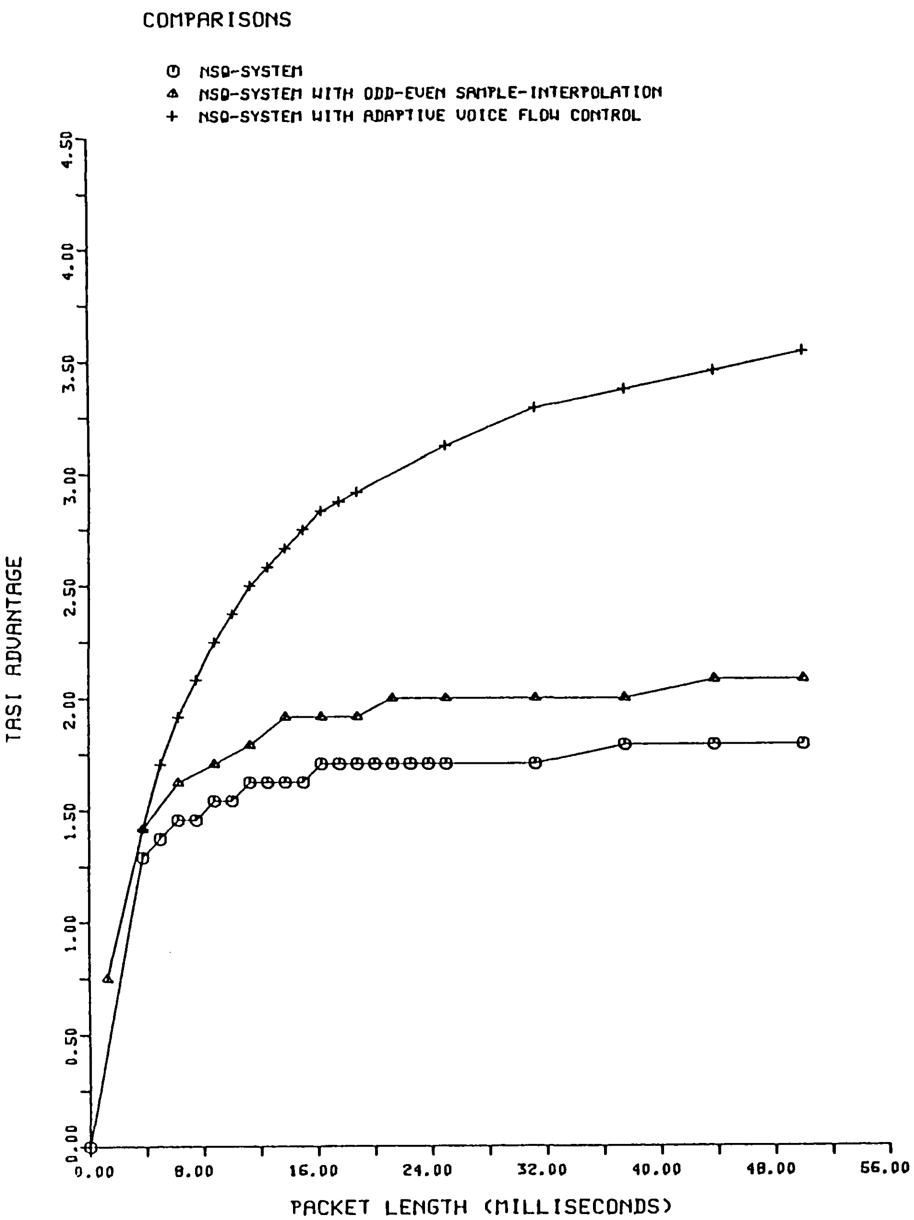


Fig. 5.20 The comparisons of the two packet organization techniques for increasing the TASI advantage of the NSQ-system

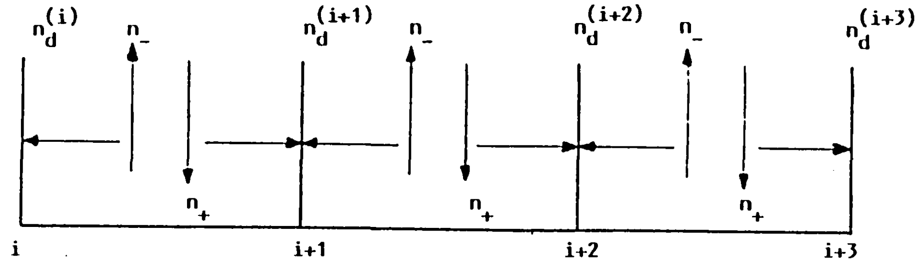
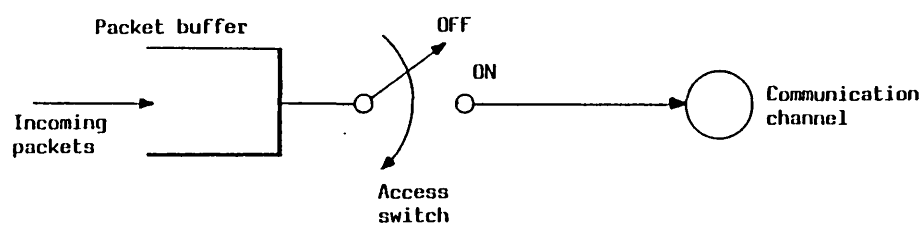


Fig. 5.21 The simulation process for the data traffic model



Switch is "OFF" for the packet-switched voice traffic
Switch is "ON" for the packet-switched data traffic

Probability switch is "ON" = σ
Probability switch is "OFF" = 1 - σ

Fig. 5.22 Single server queueing model for the integrated system

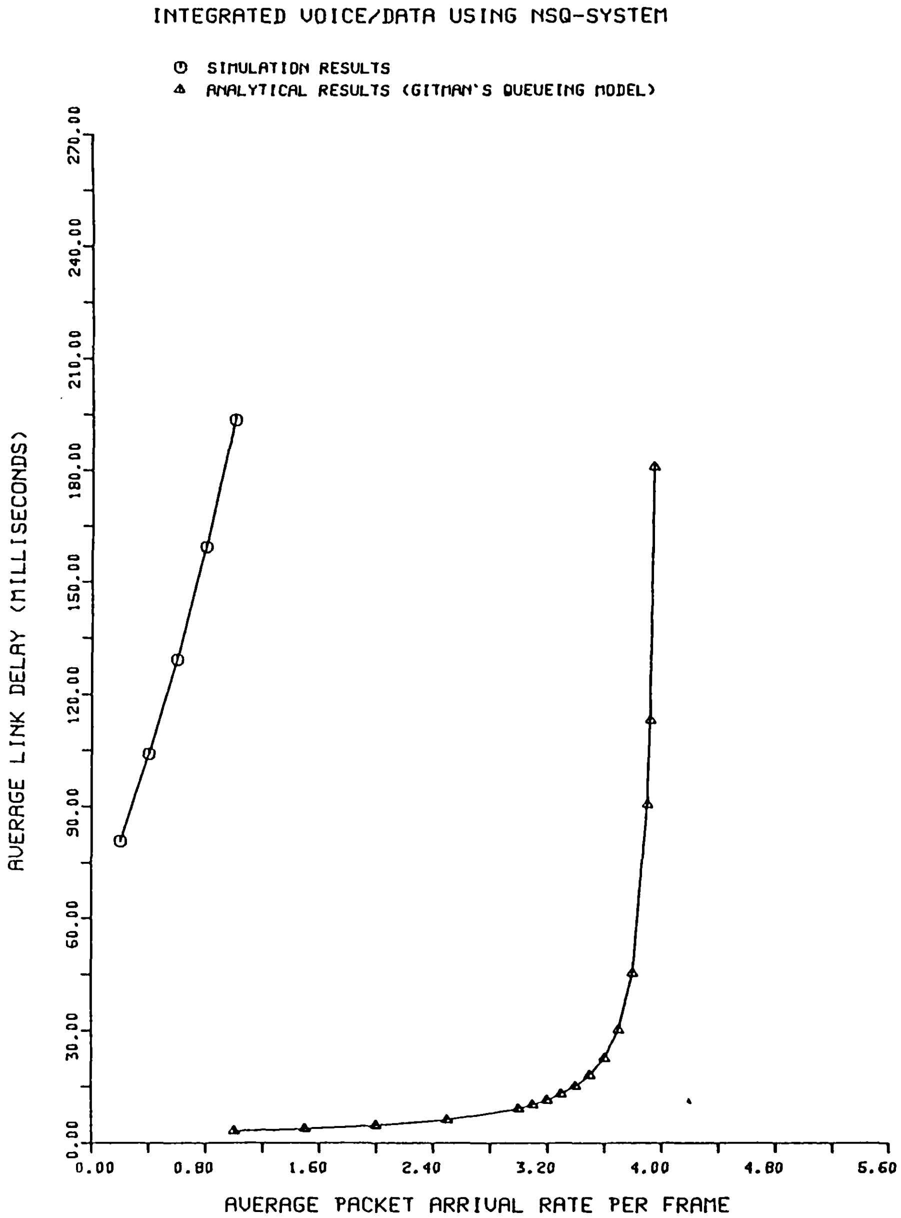
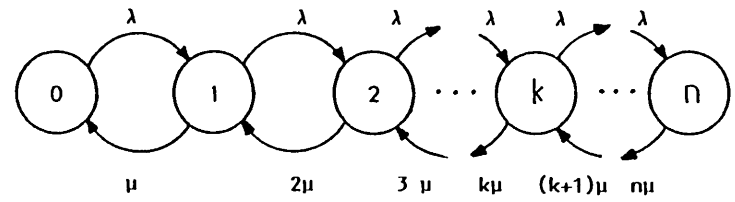


Fig. 5.23 The average data packet link delay as a function of average data packet arrival per frame



ig. 5.24 The voice traffic model for call initiation and termination

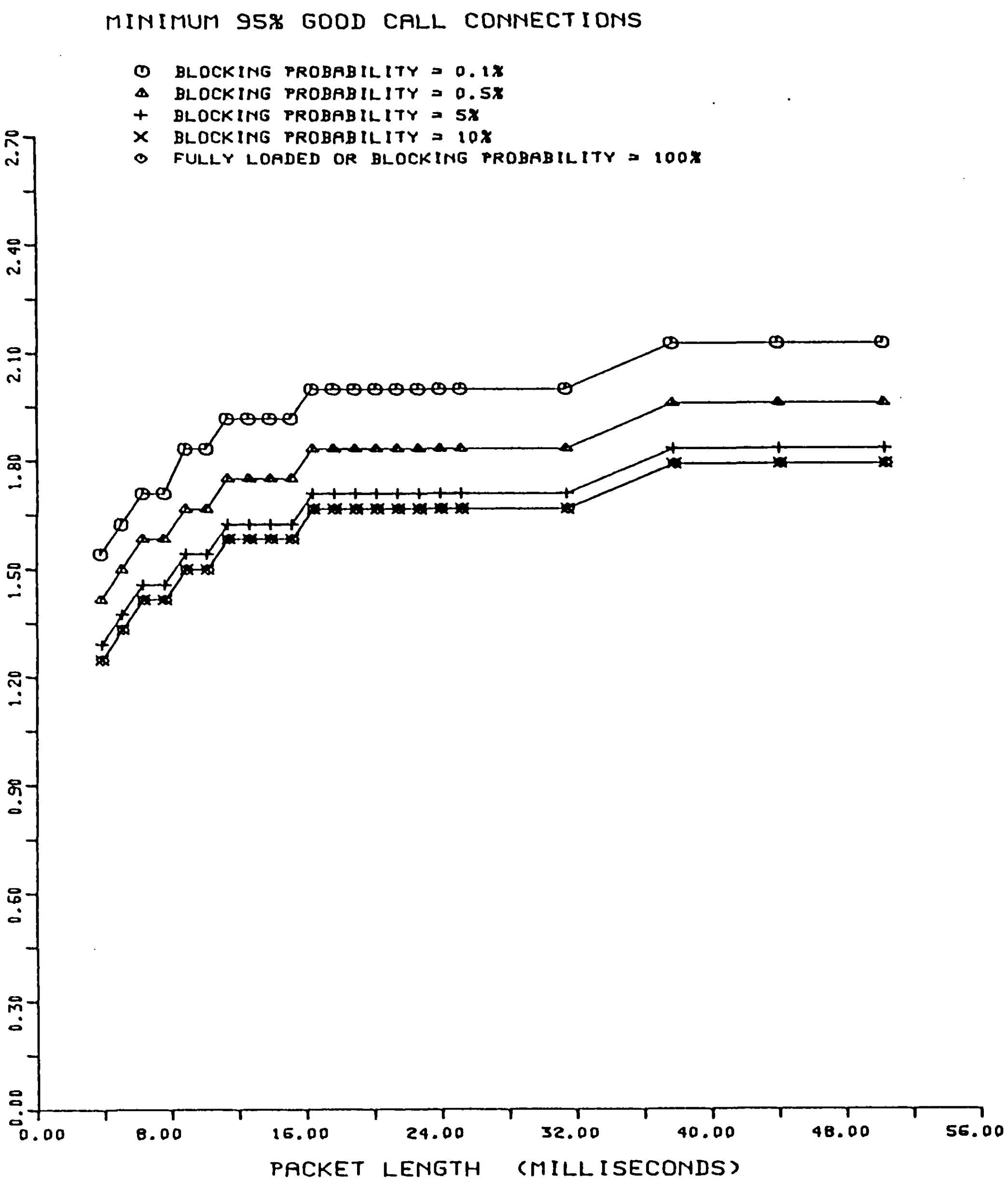


Fig. 5.25 TASI advantage as a function of packet length for different sets of blocking probability

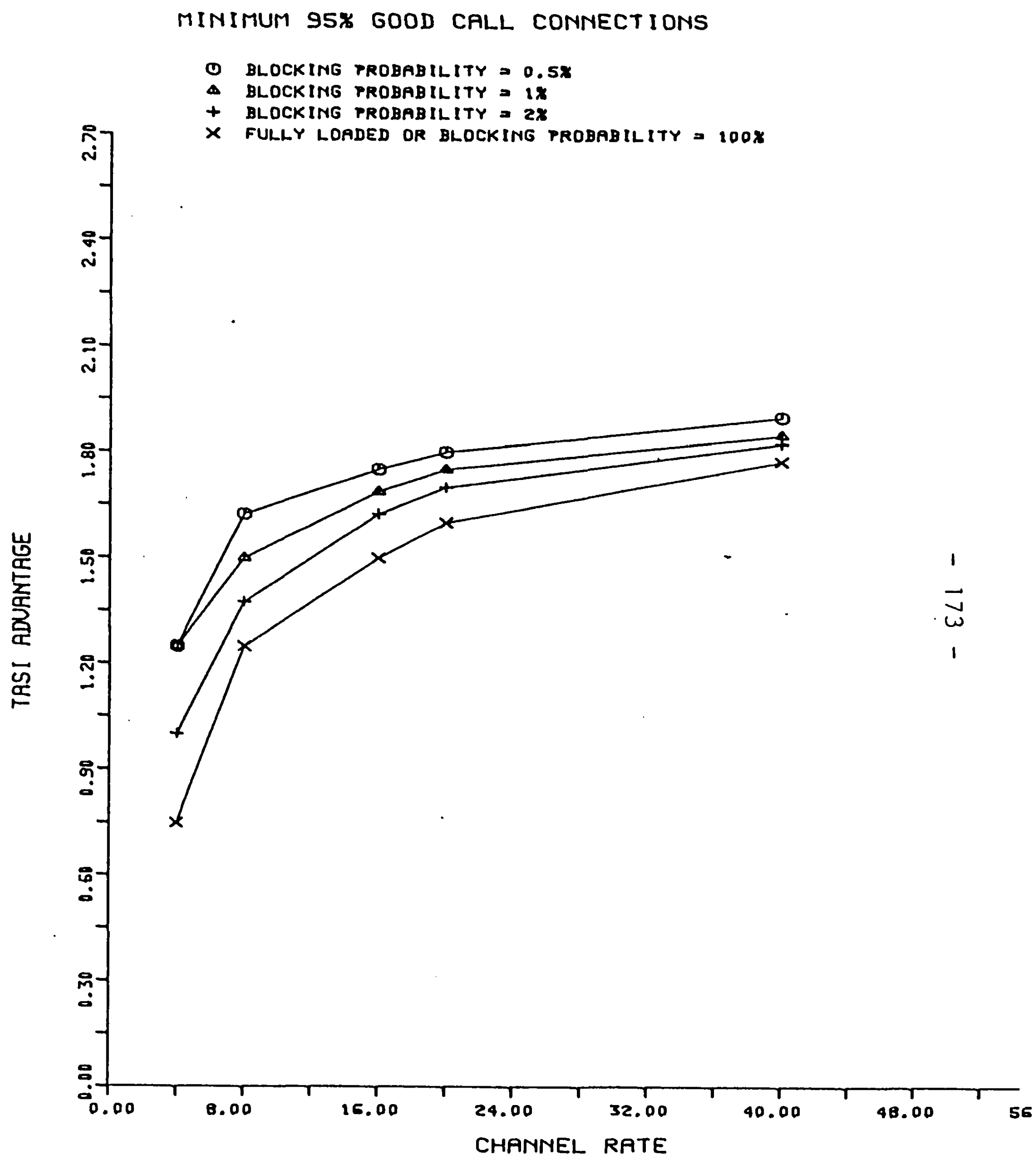


Fig. 5.26 TASI advantage as a function of channel rate for different sets of blocking probability. The packet length is set at 16 ms.

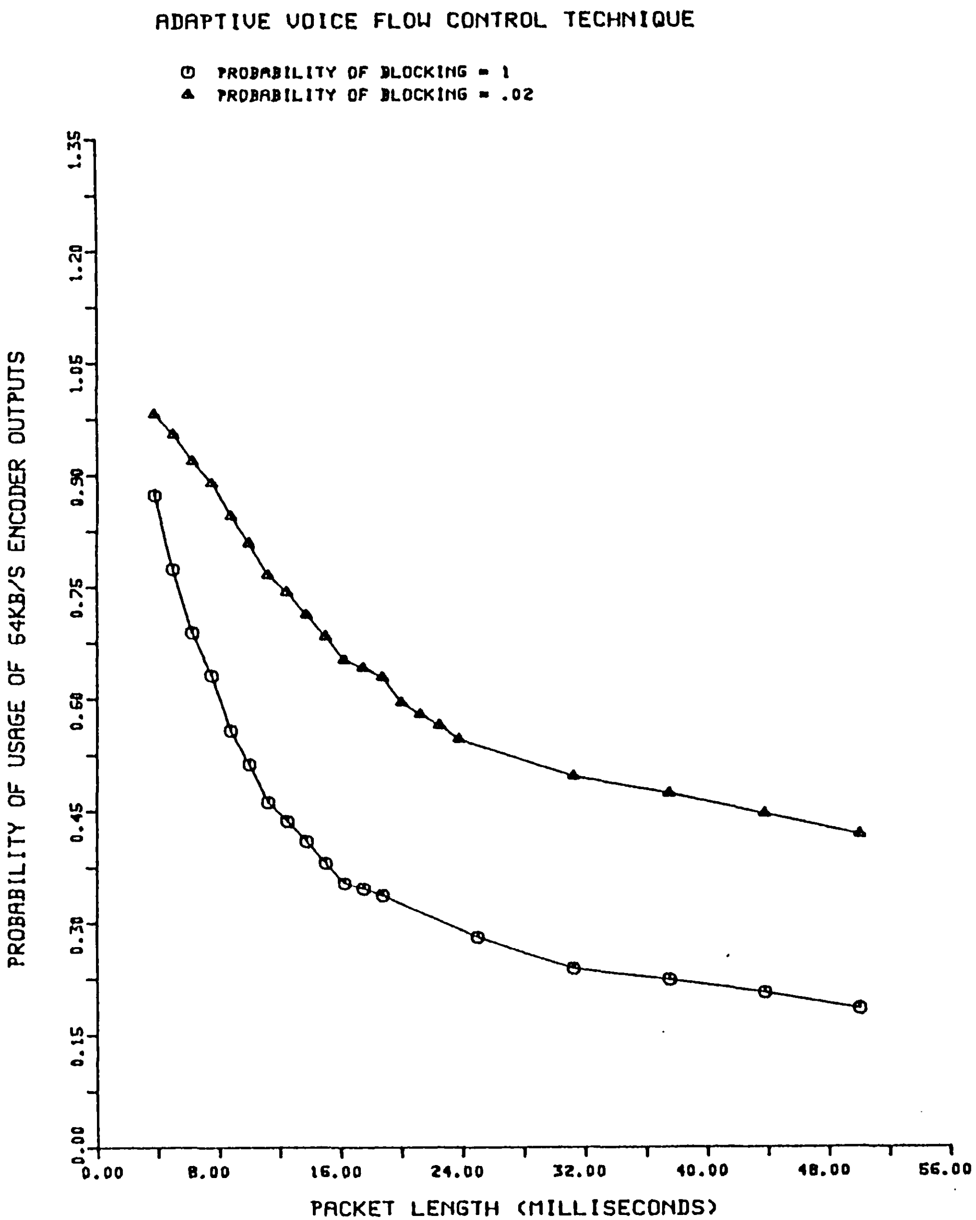


Fig. 5.27 The characteristics of the steady state probability of usage of 64 kb/s encoder output/channel as a function of packet length for probability of blockings of 2% and 100%. The 100% blocking probability represents the fully loaded PVS

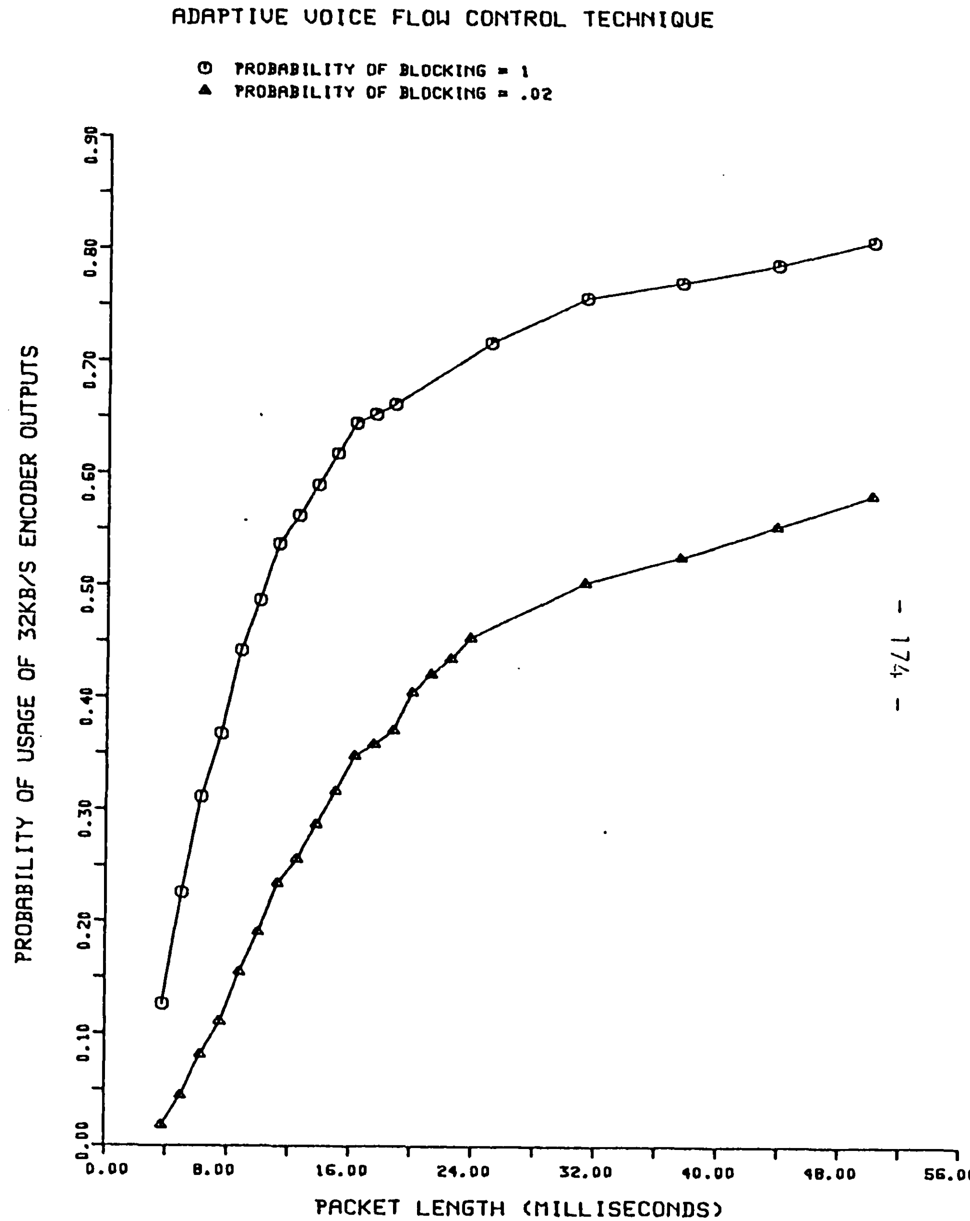


Fig. 5.28 The characteristics of the steady state probability of usage of 32 kb/s encoder output/channel as a function of packet length for the probability of blockings of 100% and 2%. The 100% blocking probability represents the fully loaded PVS

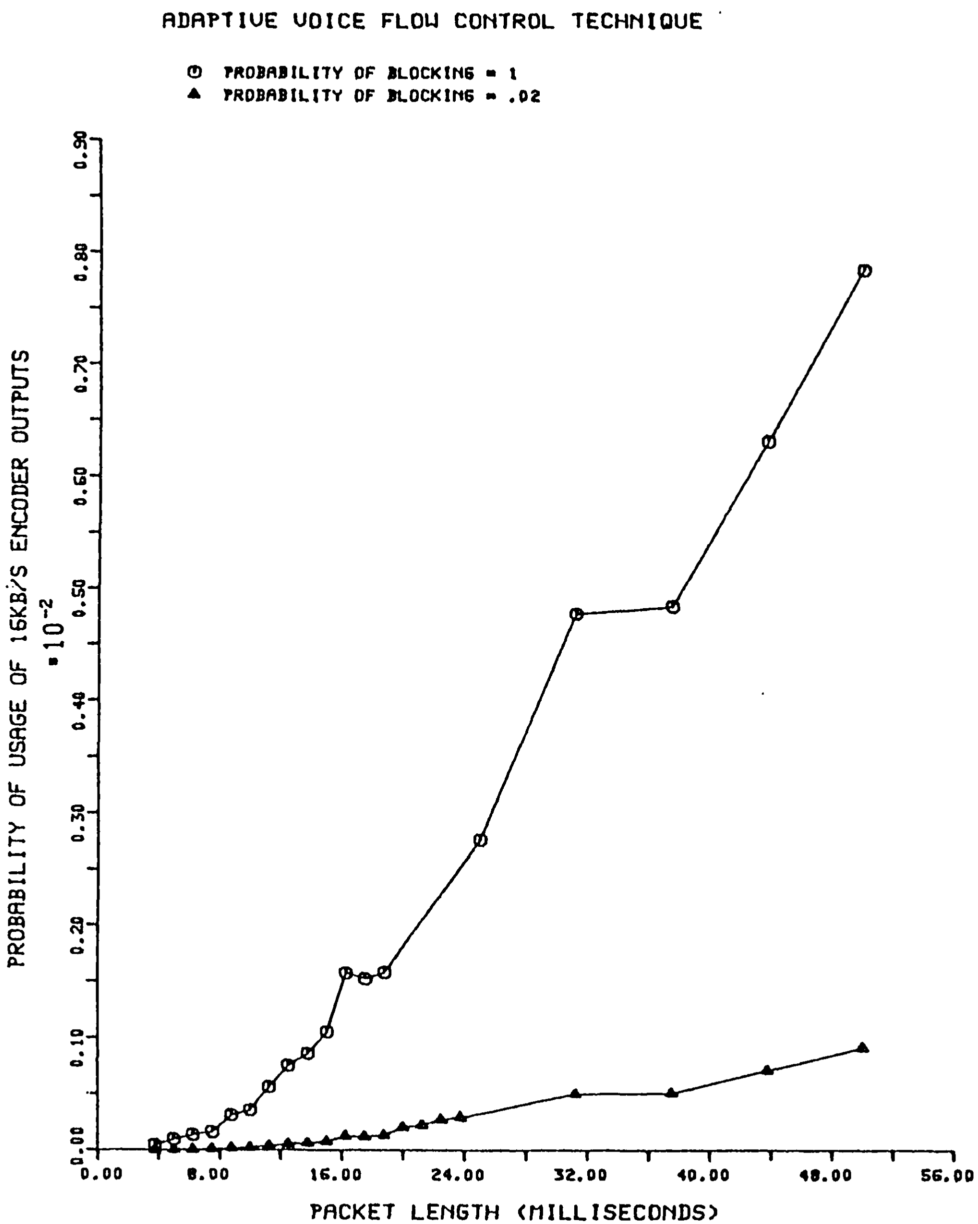


Fig. 5.29 The characteristics of the steady state probability of usage of 16 kb/s encoder output/channel as a function of packet length for probability of blockings of 100% and 2%. The 100% blocking probability represents the fully connected PVS.

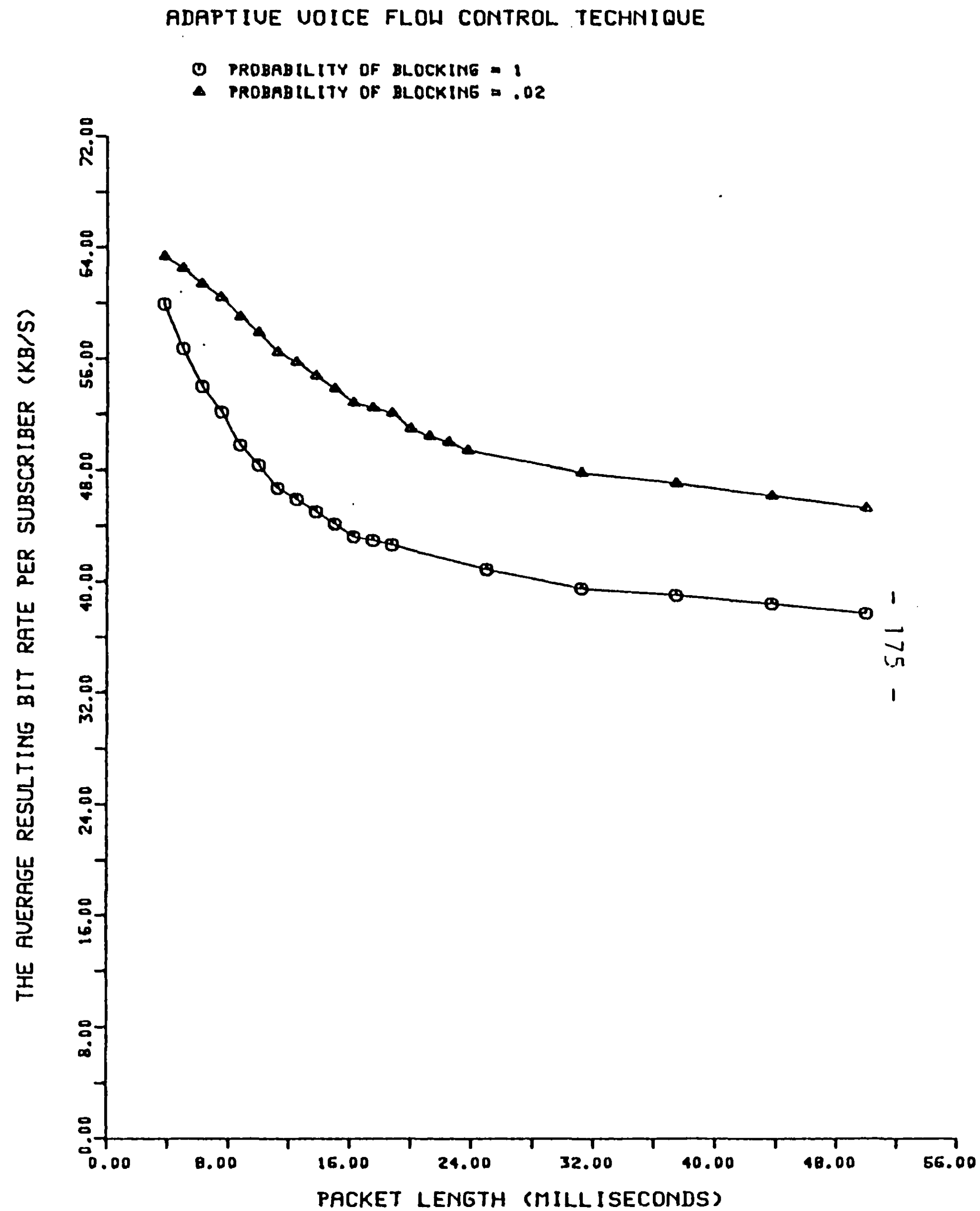


Fig. 5.30 The characteristics of the average resulting bit rate per subscriber channel as a function of packet length for probability of blocking of 100% and 2%. The 100% blocking probability represents the fully connected PVS

CHAPTER 6

INTEGRATED VOICE/DATA IN THE CIRCUIT-PACKET SENET-SYSTEM

6.1 Introduction

This chapter is concerned with investigating the performance of both voice and data traffic when employing the slotted enveloped network (SENET) approach. This is a structural multiplexing facility which is used for mixing both voice and data traffic in an integrated telecommunication system. This structure was first presented by Coviello and Vena [55]. In brief, the scheme utilizes a hybrid-switched master frame format of a time division statistical multiplexing facility. A certain portion of the frame is allocated to the voice calls which are circuit-switched. Data traffic is assigned to the remaining frame capacity and are packet-switched. In Section 6.2, the operation of this system with two boundary frame managements (i.e. fixed and movable) are explained. In Sections 6.3 to 6.6, computer simulation results are given and are compared with results obtained from appropriate analytical models. The performance parameters of the SENET link include 'the probability of blocking' for the circuit-switched voice traffic, and 'the average queueing delay' for the packet-switched data traffic. The simulation results have shown that when employing the movable boundary frame management the performance of the data traffic is not as good as the results obtained from the queueing model would predict. The results of previous workers has been reported and verified as an introduction to new work which follows. In section 6.7, an adaptive voice flow control technique has been suggested which improves the performance of the data traffic in the movable boundary SENET system. It is shown that the performance of both types of traffic in this scheme can be evaluated using analytical approaches.

6.2 The operation of the SENET system

In this approach (see fig. 6.1) the voice traffic is circuit-switched while the data traffic is packet-switched. The communication channel linking two nodes, utilizes a master-frame[1] format strategy. The frame partitions the link capacity into two regions
(i) a circuit-switched synchronous traffic region and
(ii) packet-switched asynchronous traffic region. Circuit-switched speech traffic is managed on blocked-call lost basis and packet-switched data traffic is managed on a delay basis. The boundary may be "fixed" so that no dynamic sharing between the two modes can take place, or it can be "movable", where data packets can seize currently idle circuit slots during a particular frame (but not the reverse).

An incoming circuit-switched call request, which arrives at an arbitrary instant in the current frame, is queued until the start of the next SENET frame. It is then ascertained whether a vacant circuit-switched slot is available for assignment to the call. If one exists, the call processor reserves the available slot and the call is allowed to use it in subsequent frames. If no slot is currently vacant, the call is "blocked" and is rejected from the system as is common to any normal trunk switching system. The system requires a buffering facility for each voice channel in

---

[1] This is multiple of the standard PCM frame.

order to assemble the channel bits so that groups of sample bits are transmitted in a block during the SENET frame. The scheme assumes that normal channel signalling is handled by common channel signalling in the data portion. The incoming data packets are buffered and wait in a queue indefinitely until a slot becomes available for transmission. The data packet-queueing delay could therefore be extended over several frames.

6.3 System description and computer simulation modellings

The SENET systems, both fixed and movable boundary, have been considered here to be working over a T1-carrier transmission link. The frame duration has been assumed to be 10 ms which implies a frame length of 15440 bits. 30 slots are reserved for circuit-switched voice traffic which are operated with 32 kb/s encoder outputs[2], and the remaining slots of the frame are allocated to the packet-switched data traffic. It has been assumed that the data packets are each of 1000 bits long fixed size, including their overheads. Voice traffic operates on a loss basis, while the data traffic is buffered in an infinite buffer and is serviced on a first-come-first-served basis.

The remaining part of this section deals with the presentation of the simulation models for the SENET system as originally carried out by Weinstein [65]. The following terms and values are used in this work.

---

[2]This assumes that speech encoding is viable at 4 bits/sample giving a 32kb/s system. We could have kept to the 8 bit/sample (i.e. a 64 kb/s system) but the number of channels would have to be halved.

c = Transmission channel rate = 1.544 Mb/s,

Δ = frame duration = 0.01 sec.,

e = frame length = $c\Delta$ = 15440 bits,

s = number of circuits switched slots reserved for speech traffic = 30

f = sampling rate of speech signal = 8000 samples/sec.,

b = average number of bits assigned per speech sample. For the 32 kb/s encoder per channel this is assumed to be 4 bits,

z = number of circuit-switched bits assigned per channel per frame
= Δfb = 320 bits,

P_2 = data packet size in bits (including overhead) = 1000 bits

N = number of data packet slots per frame, reserved for the data packets,

$$= \left\lfloor \frac{e-s-z}{P_2} \right\rfloor .$$

where $\lfloor x \rfloor$ denotes the greatest integer less than or equal to x.

6.3.1 Voice traffic simulation model

Voice traffic is modelled by a Poisson call arrival process of rate λ, and exponentially distributed call holding times with mean $(1/\mu)$. Let $n_v(t)$ denote the number of calls in progress at a given time. A simulation of the variation of $n_v(t)$ with time, based on a Markovian single-server loss-system model has been implemented as follows. Assume that $n_v(t) = k$ at a particular starting time. Then $n_v(t)$ is held at k for a time τ drawn from an exponential distribution with mean holding time, τ_k, determined as

$$\tau_k = 1/(\lambda+k\mu) \qquad k = 0,1,\dots,s-1$$
$$\tau_s = \frac{1}{s\mu} \quad . \tag{6.1}$$

After a time τ, $n_v(t)$ is increased to k+1 with probability

$$P_{up}(k) = \lambda/(\lambda+k\mu) \qquad k = 0,1,\dots,s-1$$
$$P_{up}(s) = 0 \tag{6.2}$$

or decreased to k-1 with probability $1-P_{up}(k)$. This process is repeated as often as desired to generate sample functions of $n_v(t)$.

6.3.2 Data traffic simulation model

In this Markovian simulation model, the number of packets in the queue is calculated at the beginning of each frame. For example; in fig. 6.2, it has been assumed that the number of packets in the queue at the beginning of the i^{th}-frame to be $n_d^{(i)}$. For the fixed boundary version, during each frame period, the maximum number of data packets which can be sent out during this time is given by

$$n_- = \left\lfloor \frac{e-sz}{P_1} \right\rfloor \tag{6.3}$$

while for the movable boundary scheme

$$n_- = \left\lfloor \frac{e - (s-n_v(t))z}{P_1} \right\rfloor \tag{6.4}$$

Data packets are assumed to arrive in a poisson process with rate θ packets/s. The number n_+, of data packets arriving during the frame interval is drawn from a Poisson distribution with mean $\theta\Delta$. Then the number of packets in the queue at the i+1th-frame is

$$n_d^{(i+1)} = \max(n_d^{(i)} + n_+ - n_-, 0) \tag{6.5}$$

The process is repeated for a large number of frames, and the average number of packets in the queue, $\bar{n}_d$, is computed. The average data packet waiting time, $\bar{w}_d$ in the queue is then determined by Little's results [23], which is

$$\bar{W}_d = \frac{\bar{n}_d}{\theta} . \tag{6.6}$$

6.4 Analytical model for the voice traffic performances

The SENET-link performance for the voice traffic is determined by the blocking probability of voice calls. We assume that the voice traffic is allowed to queue for a maximum of one frame period and no longer. That is,an arriving voice customer waits in a buffer until the time instant of the beginning of the next frame. If the number of free voice channels is greater than the number of voice customers ahead of him in the buffer, he receives service. If not, he is lost and leaves the system without service.

Fisher and Harris [57] developed an exact analytical expression for the fraction of lost calls, or the blocking probability for the voice traffic in the SENET-system. However, Gitman et al in Refs [58] and [59]; have shown that for a short frame duration (i.e. Δ

= 10ms), the blocking probability for the circuit-switched voice traffic can be closely approximated by the well-known Erlang loss formula equation given by

$$P_B = \frac{\frac{A^s}{s!}}{\sum_{i=0}^{s} \frac{A^i}{i!}} \tag{6.7}$$

where $A = \lambda/\mu$ is the offered circuit-switched voice traffic load in Erlangs.

6.4.1 Results and comparisons

The average holding time of the voice calls was set at 100 seconds (i.e. $1/\mu$ = 100 sec). The offered circuit-switched voice traffic, A, was varied by changing the interarrival time λ and for each value of A. The probability of blocking P_B was obtained using equation (6.7), as above. Using the voice traffic simulation model (as given in Section 6.2.1), a computer simulation program was written and the same procedures as described above were undertaken. The length of the simulation period was set at 50,000 seconds of real time. The program was run 5 times and the final result quoted corresponds to the averages of these 5 runs.

Fig. 6.3 shows that the analytical and simulation results have a very close agreement with each other.

In the remaining sections of this chapter, the performance characteristics of the data traffic will be studied by setting the offered voice traffic load fixed at 22 Erlangs or $1/\mu$ = 100 sec and λ = .22. This implies from equation (6.7) a blocking probability of .0205, (i.e. 2%)

6.5 Analytical model for the evaluation of the Data traffic performance in the SENET system with fixed boundary

This is a straight packet-switched type system with a Poisson packet arrival process and with a fixed throughput for the data packet transmission. Since the data packet lengths are of fixed length, then an M/D/1 queueing model is used here to determine the queueing performance of the data traffic. This approach neglects the fact that packets are generated in bursts, here their arrivals are viewed as being dispensed evenly throughout the frame. By making this assumption the effect of the frame structure on the packet queueing process is eliminated and a Markovian packet arrival constant server queueing model is appropriate and is shown in fig. 6.4.

From Kleinrock [23] (pp.405), the average waiting time in an M/D/1 queueing is given by

$$\bar{q} = \frac{\rho_d \bar{x}}{2(1-\rho_d)} \qquad (6.8)$$

where $\bar{x}$ is the average packet service time and ρ_d denotes the data traffic link utilization and is given by

$$\rho_d = \frac{\theta\Delta}{n_f} \qquad (6.9)$$

in which $\theta\Delta$ represents the average number of packet arrivals per frame interval n_f denotes the total number of packets transmitted by the channel per frame in the fixed boundary SENET system which is given by

$$n_f = \left\lfloor \frac{e-sz}{P_2} \right\rfloor \tag{6.10}$$

As before, $\lfloor x \rfloor$ represents the greatest integer less than or equal to x. The requirement of the symbol '$\lfloor \; \rfloor$' is due to the constant frame format operation of the SENET. The region of the frame which is allocated for the data packets can be a non-multiple integer of the packet length. This may result in leaving a part of the frame 'unused' as its length is less than a complete packet. As a result of this, we obtain the probability, σ_f, that the channel is available for the data packets in the fixed boundary scheme as follows:

$$\sigma_f = \frac{n_f P_2}{e} \tag{6.11}$$

or

$$n_f = \frac{e\sigma_f}{P_2} \tag{6.12}$$

and by substituting n_f from (6.12) into (6.9)

$$\rho_d = \frac{\theta\Delta P_2}{e\sigma_f} \tag{6.13}$$

or by using the relation $c = \frac{e}{\Delta}$,

$$\rho_d = \frac{\theta P_2}{c\delta_f} \quad . \qquad (6.14)$$

The service time is also constant and is given by

$$\bar{x} = \frac{P_2}{\delta_f c} \quad . \qquad (6.15)$$

By substituting ρ_d and $\bar{x}$ from equations (6.14) and (6.15) into (6.8), the average data packet waiting time for this system, $\bar{q}_f$, becomes

$$\bar{q}_f = \frac{\theta P_2^2}{2c\delta_f(c\delta_f - \theta P_2)} \quad . \qquad (6.16)$$

6.5.1 Results and comparisons

A computer simulation program has been written for the evaluation of data traffic performance characteristics (i.e. to find the average queueing delay as a function of average packet arrival rate per frame). The length of the simulation run was set at equivalent to 60,000 secs of real time. The final results for each plotted point represents an average of 5 runs each starting from a different random starting point. The results of the first of 10,000 secs of each run were ignored in order to remove the effects of the initial bias conditions and to enable the queueing behaviour of the system to reach its steady state conditions.

The results for the analytical case have been derived by using equations (6.8-6.16). Fig. 6.5 gives the results obtained from the two models. It is evident that the two sets of results have a good agreement with each other. Thus it can be said that the M/D/1 queueing model is a valid model for the approximation of the queueing performance of the packet-switched data traffic in the SENET system with fixed boundary.

6.6 Data traffic performance in the SENET with movable boundary

This strategy is used in order to improve the data traffic performance in the system as compared to the fixed boundary frame allocation scheme. Its aim is to take advantage of slots not used by the speech. The first analytical model, for this scheme was presented by Kummerle in [56] and then completed by Fisher and Harris in [57]. Results obtained from Fisher and Harris, as quoted in [57], have indicated that significant savings can be obtained in terms of channel requirements when using an integrated movable-boundary system, as compared to the fixed-boundary system handling the same voice traffic. However, due to the complexity and computational difficulties which are associated with this model, another comparitively simple model based on the M/G/1 queueing approximation was used by Gitman et al [58], [59] and also by Mowafi et al [62]. These gave similar results to those already obtained by Fisher and Harris [57]. However,Weinstein in Ref [65], by means of computer simulation modelling as presented in Section 6.3, showed that during the peak loading of the voice channels data packet delays build up, and this effect causes degradation in the performance of the data traffic. The simulation results, as presented in Ref [65], have indicated that the saving in the channel is significantly less than those quoted by Fisher and Harris in [57]. Due to this result, many other researchers have attempted to develop new analytical approaches to ascertain the performance of the data traffic in the movable boundary operation [66] - [72]. However, all these methods have proved to have significant computational difficulty associated with them.

In this section of the thesis the author has stated the nature of the problem which causes a degradation in the performance of the data traffic in the movable boundary operation. This has been approached by simulating the movable boundary system and then comparing simulation results with the expected results obtained from an M/G/1 queueing model.

6.6.1 M/G/1 queueing model

As stated earlier, in the movable scheme the nominal boundary between the circuit and packet regions of the frame do not move. The data packets can however use the free circuit-slots available due to the statistical variations in the voice traffic. One method which can be used to move the boundary is to use the variable frame multiplexing system, as implemented by the Codex Co operation [63]. Under this scheme, as also used in Ref [64], dedicated voice time slots which are not carrying traffic are skipped, as a result time slots are continuously being left at the end of the voice part of the frame . Fixed-frame time-division multiplexing (TDM) circuit-switching is thus replaced by a statistically varying frame TDM scheme. Such a scheme provides a saving in the channel capacity and eases the implementation of the integrated system which employs the movable boundary scheme. However, this will require additional hardware and software. Throughout our simulation and analysis of the movable-boundary scheme we have assumed that this variable TDM frame technique is provided for the circuit-switched voice traffic. Unlike the fixed boundary scheme, the channel capacity available for this scheme is not constant, it varies according to the variations in the activity of the voice channels. The effect therefore causes the packet capacity to vary and so varies the service time.

As the packet arrival process is assumed to be a poisson process, an M/G/1 queueing model can be used to approximate the queueing performance of the data traffic. Its analysis is given as follows:

From Kleinrock [23] (pp.190), the approximate expression for the packet waiting time in an M/G/1 queueing model is given by

$$\bar{q}_m = \frac{\theta \overline{x^2}}{2(1-\rho_d)} \tag{6.17}$$

where $\bar{x}$ and $\overline{x^2}$ denote the first and second moments of the service time. Also by using the relation $\rho_d = \theta\bar{x}$, $\bar{q}_m$ can be rewritten as

$$\bar{q}_m = \frac{\theta x^2}{2(1-\theta\bar{x})} \tag{6.18}$$

The average packet waiting time can therefore be obtained if the first moment, $\bar{x}$, and second moment, $\overline{x^2}$, of the packet service time is obtainable. These two terms have been obtained by the following:

$$\bar{x} = \sum_{k=0,}^{s} \frac{p_2}{c\delta_k} P_k \qquad (6.19)$$

$$\overline{x^2} = \sum_{k=0}^{s} \left(\frac{p_2}{c\delta_k}\right)^2 P_k \qquad (6.20$$

where δ_k represents the probability that the channel is available for the data packets with a k voice channel occupancy. P_k represents the probability that the voice channel occupancy is k and is given by the Erlang distribution [58].

$$\sigma_k = \frac{\left[\frac{e-kz}{P_2}\right] p_2}{e} \qquad (6.21)$$

$$P_k = \frac{\frac{A^k}{k!}}{\sum_{i=0}^{s} \frac{A^i}{i!}} \qquad (6.22)$$

6.6.2 Further queueing approach

Gitman et al in Ref [58], [59] derived an expression for the average link delay for the data traffic in the SENET with movable boundary. This is given by the following expression

$$W_m = \frac{p_1(2c - \theta p_1)}{2c(c\delta_m - \theta p_1)} \qquad (6.23)$$

where δ_m is the probability that the channel is available for the data packets, and is obtained from the expression.

$$\delta_m = \frac{\left\lfloor \frac{e^{-E(v)z}}{p_2} \right\rfloor p_2}{e} \qquad (6.24)$$

where E(v) denotes the average number of voice channels occupied, and according to Ref [58], is given by

$$E(v) = A(1 - P_B) \qquad (6.25)$$

6.6.3 Results and comparisons

Using the fixed system parameters (i.e. c = 1.544 Mb/s, 32kb/s coders, s = 30, p_2 = 1000 bits, A = 22 Erlangs, and 10 ms frame length) the SENET system with movable boundary scheme has been simulated employing the simulation models given in Section 6.3. The length of the simulation program was set at 60,000 seconds. The same procedures as were previously carried-out were used, i.e. the program was repeated 5 times and the average taken. The results for the first 10,000 seconds were ignored. The characteristics of the average packet waiting time $\bar{q}_m$ versus the average packet arrival rate per frame, $\theta\Delta$, were determined. Results for the analytical cases have been obtained by using equations (6.17 - 6.22) for the M/G/1 model and equations (6.23 - 6.25) for the Gitman's expression. Fig. 6.6. depicts the characteristics of $\bar{q}_m$ versus $\theta\Delta$ obtained from the two analytical models. It is evident that the two sets of results have a close agreement with each other. Thus, this verifies the validity of Gitman's expression.

Fig. 6.7 shows the comparisons of the simulation and analytical results obtained. The graphs clearly indicate that the simulations do not indicate that the system can achieve the improvements which the use of the M/G/1 analytical model would predict. This is due to the build up of the data queue when the instantaneous voice traffic is high and also the fact that not all unused time slots are useable for packet transmission.

Fig. 6.8 shows the results for the fixed and movable boundary schemes. The graphs show that less improvement has resulted with the employment of the movable boundary, as was expected by calculations using the M/G/1 model. For example; for an average packet waiting time of 40ms, $\theta\Delta$ in the fixed boundary is 4.9 packets/frame. While in the movable boundary for the same average queueing delay, it is expected that $\theta\Delta$ could be increased to 7.9 packets/frame using the M/G/1 queueing model. The simulation results, however, show that for the same average queueing delay, $\theta\Delta$ can only be increased to 5.9, i.e. an improvement of $\frac{5.9-4.9}{7.9-4.9} \times 100 \simeq 33\%$ only. This implies that the data packets can only use 33% of the free available slots.

6.6.4 Discussions

The reasons why the data traffic in the movable boundary scheme fail to achieve the full utilization of the free circuit-slots have been discussed widely in the literature (see Weinstain [65], Gaver et al [69], and Leon-Garcia et al [68]). In brief, the main reason is due to the way the system operates in modes for comparatively large numbers of frames. This effect causes the packet service time to vary slowly in the system and to cause an undesirable large buffer occupancy during peaks of the voice channel occupancy. This is now discussed in more detail by reference to fig. 6.9 which represents the Markovian voice call arrival model as used in the simulation program. Consider the fixed boundary SENET system. In this scheme the data traffic always operates in state S in the voice call model (i.e. single mode operation). The channel availability for the data packets is fixed and with the Poisson packet arrival process of average rate θ packets/sec, an M/D/1 queueing model can be used to approximate the queueing performance of the data traffic. In the movable boundary, however, the channel availability for the data packets is no longer fixed. In the voice call model the system can at any instant be in one of the states between 0 to S with the probability determined by the Erlang distribution given by equation (6.20). The periods of time that the system remains in any of these states is determined by an exponential distribution with a mean τ_k, as determined by equation (6.1). With μ = 100 sec, λ = .22 and using equation (6.1), it can be shown that τ_k is of the order of seconds (i.e. 2 sec < τ_k < 4.6 sec). This suggests that there is a probability that the system remains in each of these states for comparatively large number of consecutive frames. Now let's assume that the instantaneous voice channel occupancy is low (i.e. states near '0'). This corresponds to the probability that more of the channels are available for the data packets as compared with the fixed boundary system and this leads to a reduction in the packet service times. As a result there is a possibility of increasing $\theta\Delta$ higher than that indicated as being appropriate for the fixed boundary system (i.e. it can be higher than 4.9). However this increase has a sever drawback due to the times when the instantaneous voice channel occupancy is high (i.e. in states near S). During these periods there will be fewer channels available for data packet transmission and therefore results in higher packet service times. This therefore increases the probability that the data packet link utilization, ρ_d, will exceed unity, i.e. the queue size will build up rapidly. This behaviour is undesirable because under these conditions (i.e. $\rho_d \geq 1$), the system could stay a long time in this mode and would result in an enormous buffer build up (see also Weinstein [65]), this consequently gives a rise to the average data packet waiting time. This effect therefore does not allow the data packets to utilize efficiently the available free slots.

It should, however, be noted that this type of queueing behaviour was also experienced in the SQ-packetized voice system. The main difference is that in the SQ-system the packet service time is always constant.

6.7 Adaptive voice flow control technique in the SENET with movable boundary operation

From the simulation work carried out by the author in the previous section and the work reported by others [65] - [71] it can be concluded that there are two basic disadvantages associated with the movable boundary SENET system. These are:

(1) The performance of the data traffic cannot fully utilize the spare voice channel capacity.

(2) The difficulty in performing an exact analytical analysis for the evaluation of the data traffic performance.

The reason for these, as concluded in the previous section, was due to the way in which the packet service time slowly varies during the peaks of the voice channel occupancy. In this section the author presents another movable boundary SENET system, This system utilizes an adaptive voice flow control technique similar to that used in the NSQ-packetized voice system. The combined system represents a new movable boundary scheme where its boundary, as denoted by B_k, is less than the previous boundary which was set at S(i.e. $B_k < S$). It is shown that this scheme can overcome the two basic disadvantages which are associated with the previous movable-boundary SENET system but at the cost of a degradation in the speech quality.

6.7.1 System description

This scheme utilizes the same form of movable boundary system as previously. The adaptive voice flow control technique is applied replacing the 32 kb/s coders with new coders which are operated with two different encoding outputs namely, 32 kb/s and 16 kb/s. The threshold or the new boundary B_k is assumed to be assigned to one of the S states of the voice call model as in fig. 6.9 so that $B_k < S$. Under this scheme, the system is assumed to operate with the 32 kb/s encoder outputs whenever the instantaneous voice channel occupancy in the system is less than B_k calls. When the voice channel occupancy exceeds B_k calls, then some of the channels are switched to their 16 kb/s outputs. The number switched will be sufficient to allow the extra channels to be fitted into the nominal frame length (B_k). Other channels will continue to work with their encoders at encoders 32 kb/s.

This scheme can therefore be viewed to represent a new movable boundary SENET system so that the boundary is shifted from S to B_k. The net effect is that the undesirable variations of the packet service time during the states B_k to S will be removed and becomes more constant and to a value which is dependent upon B_k. Employing the two output encoding rates used here, it can be shown that for each new voice call arrival during the periods when there are more then B_k channels, there is a need for two channels to be switched from 32 kb/s to 16 kb/s. For example, consider the case when there are B_k voice calls in the system and another call requires service. If this call were to be assigned to operate at 16 kb/s, then with the frame length of 10 ms, there would be 160 extra bits required in the system as compared to the case when there are B_k calls in the system. There is therefore the need to switch one of the other active channels from 32 bits to 16 bits in order to accommodate the new channel. The operation of this scheme requires the periodic sending of control information from the transmitter to the receiver to indicate the decoding rates of each of the circuit channels. This problem is overcome by assigning S bits in the beginning of each frame (i.e. 30 bits for system considerations). Each of these S bits carries the digital information of '1' or '0', where for example '1' denotes the activity of a given subscriber with 32 kb/s and '0' denotes the activity with 16 kb/s. By sending this information, the decoding rate of each subscriber at the receiver can then be pre-determined during each frame interval. It has also been assumed that during the switching periods, the switching from the higher bit rate to the lower bit rate among the active talkers to be up dated in turns in each frame so that the degradation of speech quality during these periods is spread uniformly among the talkers.

In the remaining part of this section, analytical and simulation models are presented for determining the performance parameters of both voice and data traffic in this type of integrated system.

6.7.2 Voice traffic performance evaluation

In this scheme, the performance of the voice traffic is a function of two parameters, namely: (a) the probability of blocking for the circuit-switched voice traffic P_B, and (b) the voice quality performance.

The probability of blocking P_B, as shown previously, can be obtained by using the Erlang loss formula and is given by equation (6.7). The quality performance for the voice traffic is obtained by evaluating the steady state probabilities of usage of the lower and the higher encoder outputs which are assigned for each subscriber. The derivation of these two terms is given as follows:

The fraction of time that exactly r calls are active is determined from the Erlang distribution as given in equation (6.22). Out of these r active calls, a fraction of them (i.e. m_r) will be required to operate with the lower bit rate whenever r > B_k, where B_k denotes the new boundary. Thus over a long period of system operation, the ratio of the total number of active channels which have operated with the lower encoder outputs to the total active channels which have operated with both coders (or the probability of usage of the lower bit rate, P_{lower}) is given by

$$P_{lower} = \frac{\sum_{r=B_k+1}^{s} m_r P_r}{\sum_{r=0}^{s} m_r P_r} \tag{6.26}$$

$$= \frac{\sum_{r=B_k+1}^{s} m_r P_r}{A(1-P_B)}$$

In the above equation, all the terms are known except m_r. This term is found from the following equation.

$$(r-m_r)z_1 + m_r z_2 \leq B_k z_1 \tag{6.27}$$

where z_1 and z_2 represent the assigned slots per frame in bits for the higher and the lower encoder outputs repectively (i.e. z_1 = 320 bits, z_2 = 160 bits for the system under consideration). From the above equation m_r can now be expressed as

$$m_r \geq \frac{(r-B_k)z_1}{z_1 - z_2} \tag{6.28}$$

Due to the minimum requirement of the lower encoder outputs, and by using the same type of examples as given in appendix B, it can be shown that

$$m_r = \left\lceil \frac{(r-B_k)z_1}{z_1 - z_2} \right\rceil \qquad (6.29)$$

where $\lceil x \rceil$ denotes the greatest integer greater than or equal to x. By using equations (6.26) and (6.29), the probability of usage of the lower encoder outputs, P_{lower}, has been computed for different values of the new boundary B_k. A computer simulation program has also been developed for this scheme to test the scheme and to compare results obtained with the analytical model. The simulation is based on the previously described voice traffic simulation model. In this model, during each frame interval the total number of voice channel occupancies, k, is first computed. Then by using equation (6.29), the number of channels which have operated with the lower encoder outputs during each frame are computed. At the end of the simulation, the probability of usage of the lower encoder outputs is computed by dividing the total number of channels in all the frames which have operated with the lower encoder outputs by the total number of channels in all the frames which have operated with either encoder outputs.

Fig. 6.10 shows that the results obtained from the analytical and simulation models are in close agreement. The probability of usage of the higher-encoder outputs can also be obtained by $1-P_{lower}$. For example, for the case when the new boundary is set at 22 circuit-switched slots, the P_{lower} from 6.10 corresponds to 0.13 and therefore the probability of usage of the higher encoder outputs, P_{higher}, becomes 0.87. Thus for over a long time period of system operation one expects the encoders to operate at 32 kb/s for 83% of the time and to operate at 16 kb/s for 13% of time.

6.7.3 Data traffic performance evaluation

Figure 6.10 shows that the probability of usage of the lower encoder will increase, and therefore the speech quality will be degraded, as the boundary B_k is decreased. However, as B_k decreases the channel available for the data packets will increase and thus improve the data traffic handling. A computer simulation program has been written so that the queueing performance of the data traffic could be investigated when the boundary is at the average statistical voice channel occupancy, E(v), as obtained from equation (6.25). For A = 22 Erlangs and P_B = .0205, it can be shown from equation (6.25) that E(v), the new boundary, corresponds approximately to 22 circuit slots.

Fig. 6.11 shows the comparisons of the simulation results obtained from all the three integrated SENET systems (i.e. fixed boundary, movable boundary, and the movable boundary with adaptive voice flow control technique with the boundary set at E(v)). The superiority of the movable boundary with voice flow control technique ensues in a better utilization of the channel capacity for the data packets as can be seen from these results.

Fig. 6.12 shows that the queueing performance of the data traffic in this new movable boundary system can be approximated by a fixed boundary SENET system with the boundary set at E(v), (i.e. single mode operation). The results of the new fixed boundary were obtained by using an M/D/1 queueing model, which was previously shown to be is an applicable model for approximating the queueing performance of the single-mode, fixed boundary, SENET system(see Section 6.5).

Thus this scheme, when adopted in the SENET, can improve the data traffic performance. The data traffic performance can be readily obtained from an M/D/1 queueing model when the boundary is set at somewhere equal to or less than the average statistical voice channel occupancy.

The disadvantages associated with this technique are:

(1) it is more costly to implement due to the requirement of voice coders that have to be able to operate with different encoding rates,

(2) the voice quality will be degraded, the degree of degradation is dependant upon the position of the new boundary.

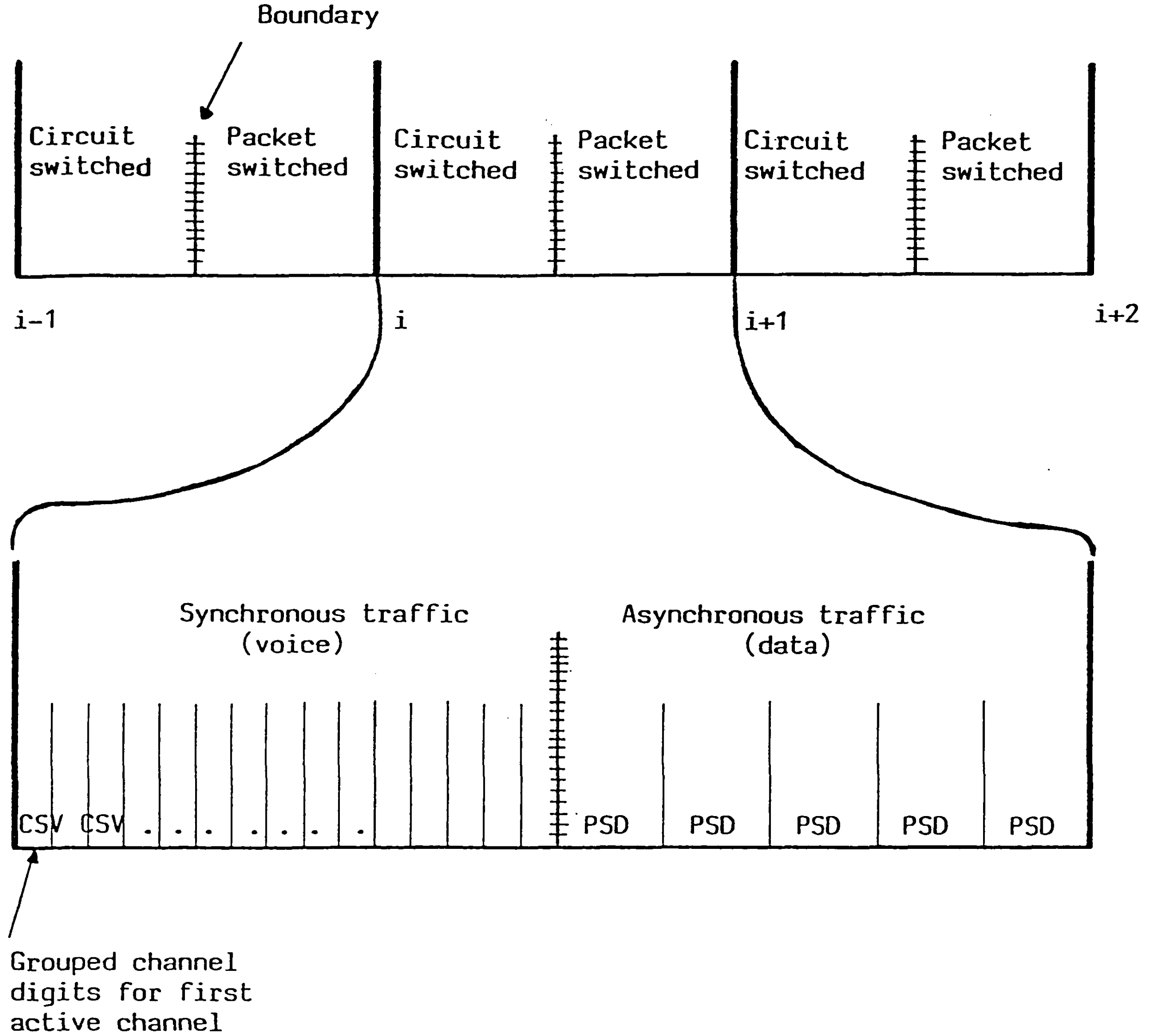
6.8 Conclusions

Computer simulation techniques have been carried out to evaluate the performance of the SENET integrated circuit and packet link. The following conclusions have been drawn:

(1) The queueing delay performance of the packet-switched data traffic in the fixed boundary frame management system can be approximated by an M/D/1 queueing model.

(2) The queueing performance of the data traffic in the movable boundary SENET-system is not as good as the results predicted from the M/G/1 queueing model would predict.

(3) When the movable boundary SENET is conjoined with the adaptive voice flow control technique, the queueing performance of the data traffic can be approximated by an M/D/1 queueing model.



CSV - Circuit Switched Voice Slot
PSD - Packet Switched Data Slot

Fig. 6.1 Exemplary SENET concept for integrating voice and data

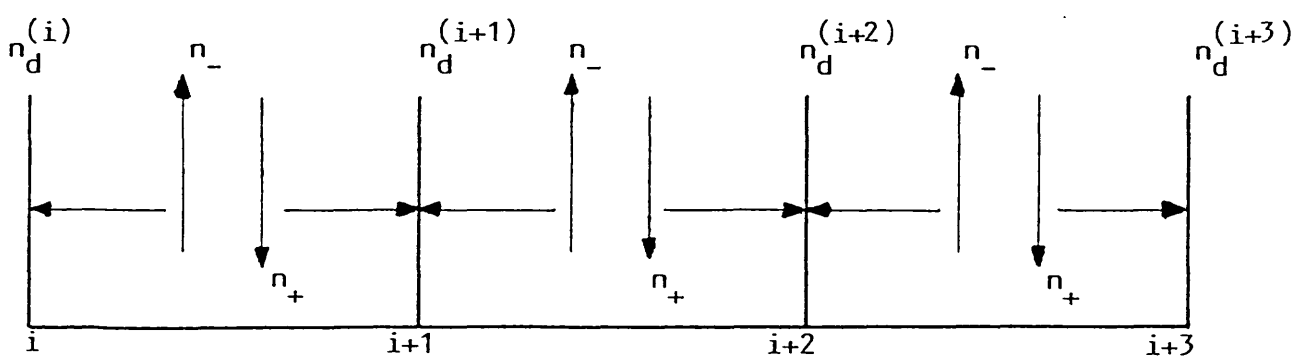


Fig. 6.2 The simulation process for the data traffic model

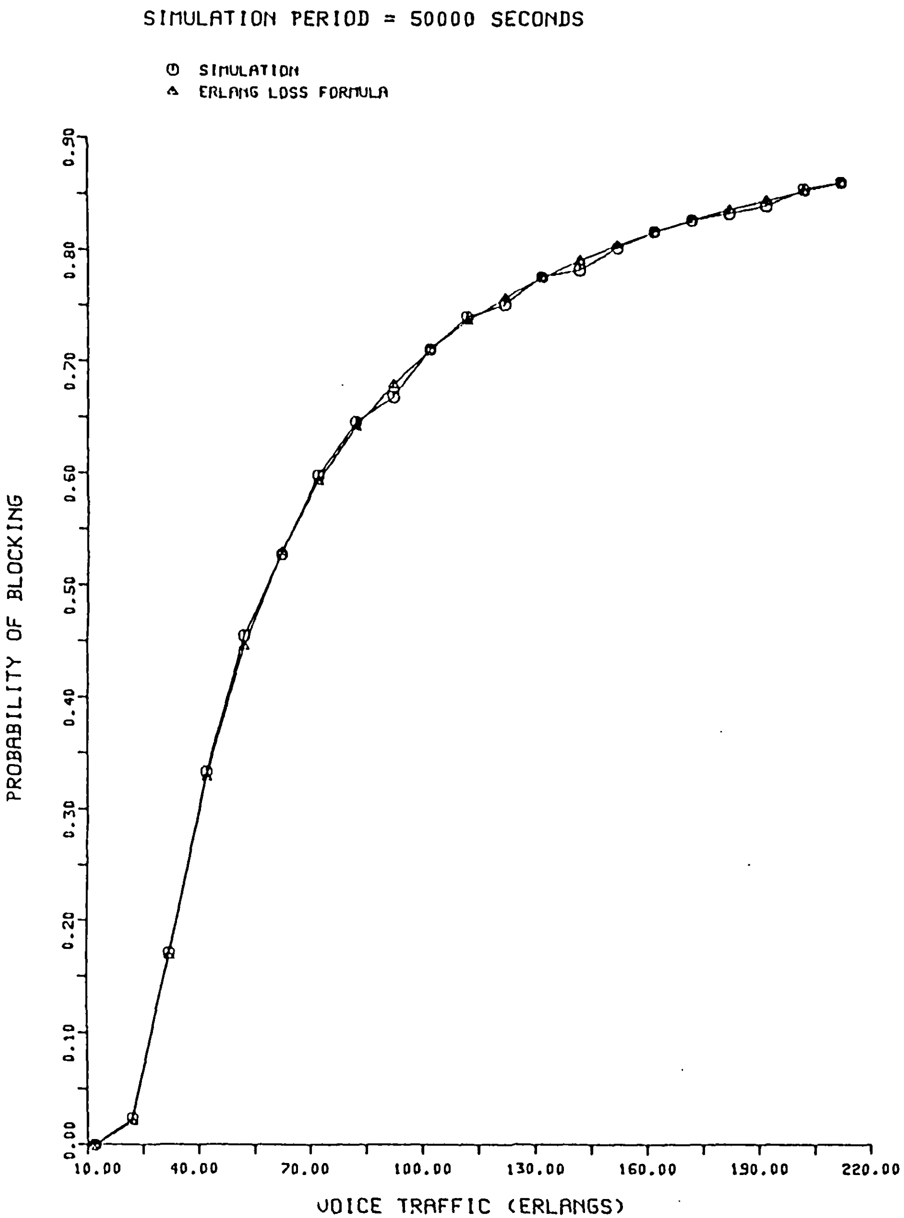


Fig. 6.3 Blocking probability as a function of offered voice traffic intensity

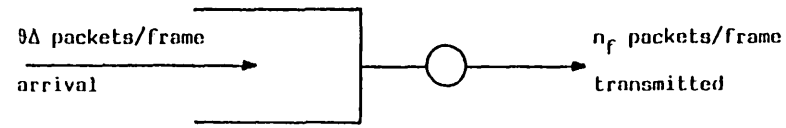


Fig. 6.4 Queueing model for the SENET system

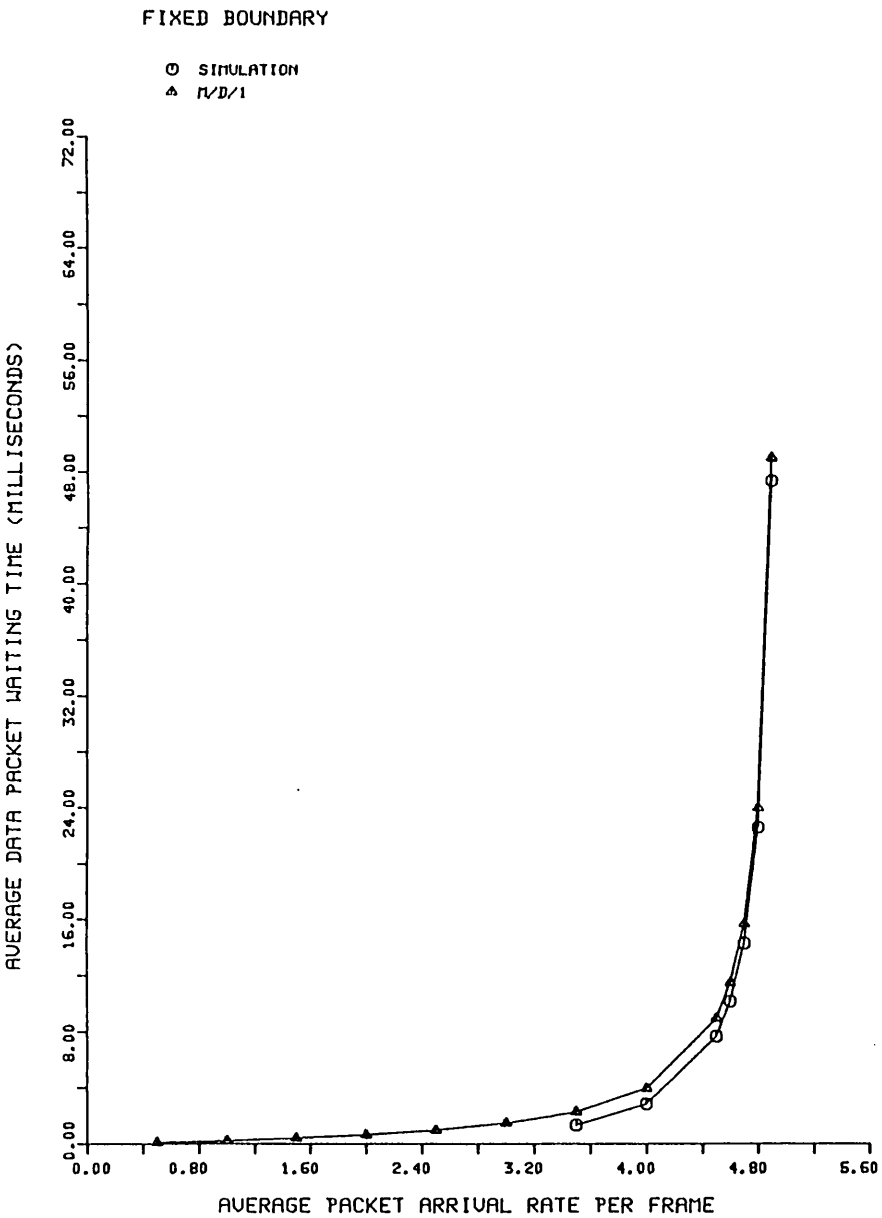


Fig. 6.5 The performance characteristic of the data traffic in SENET with fixed boundary operation

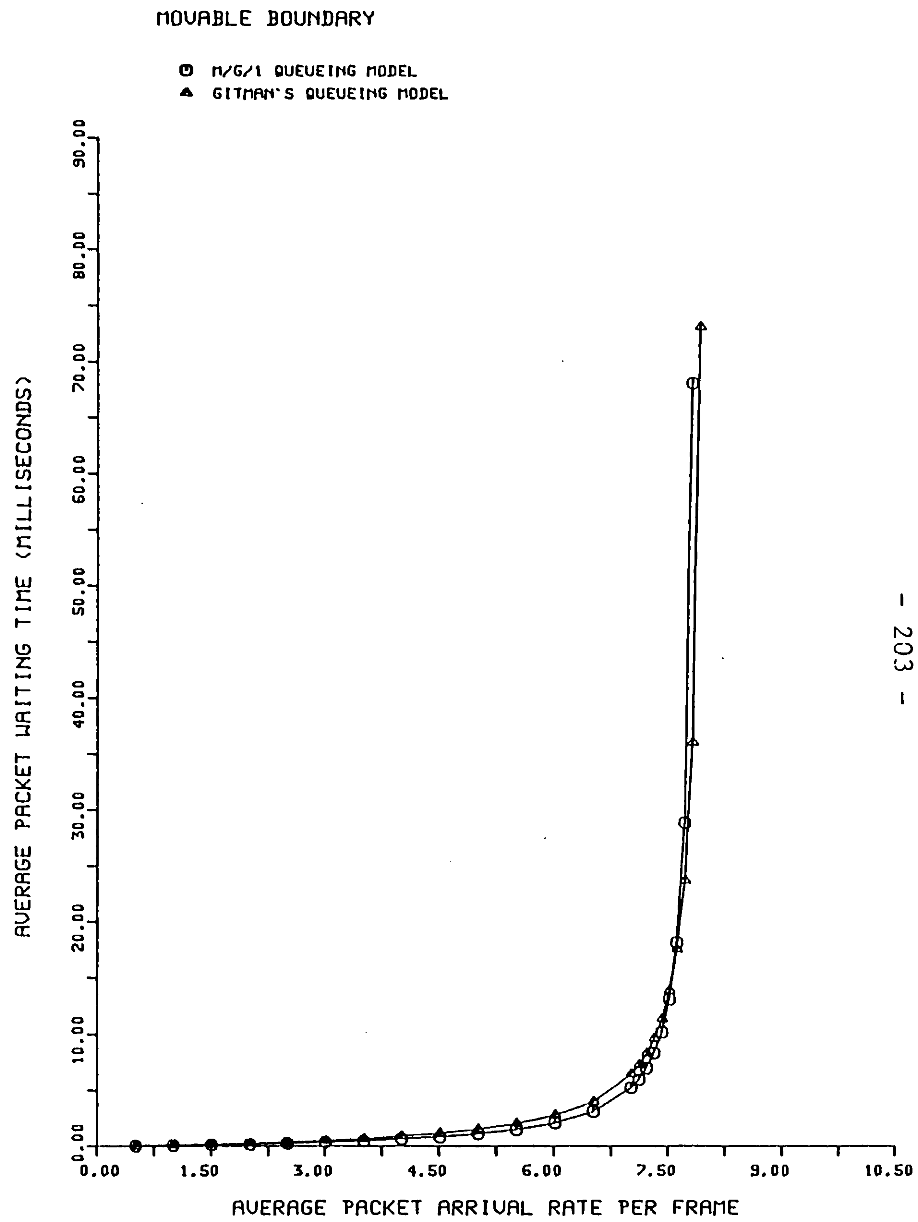


Fig. 6.6 The comparison of the analytical models for evaluating the performance characteristic of the data traffic in SENET with movable bounary

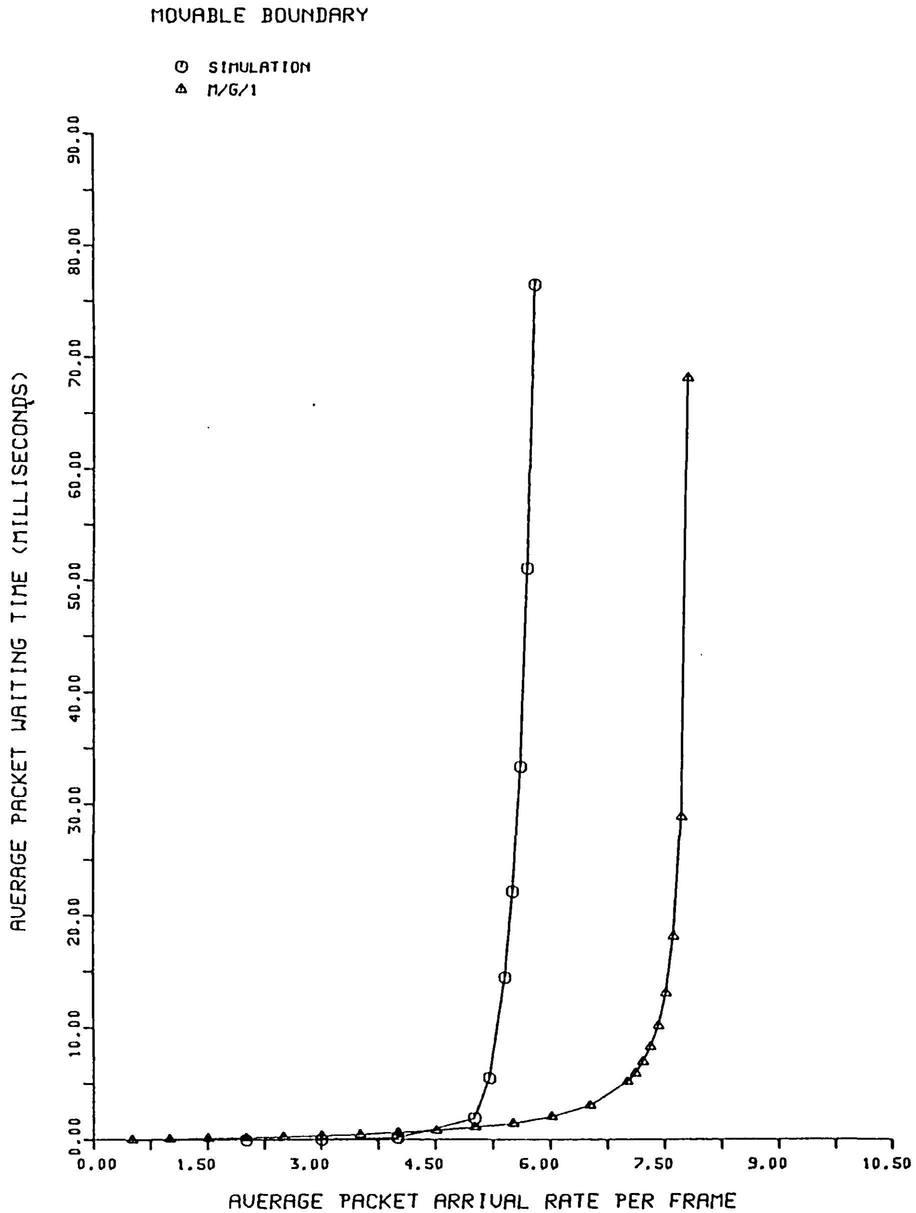


Fig. 6.7 The comparison of the analytical and simulation models for evaluating the performance characteristic of the data traffic in SENET with movable boundary

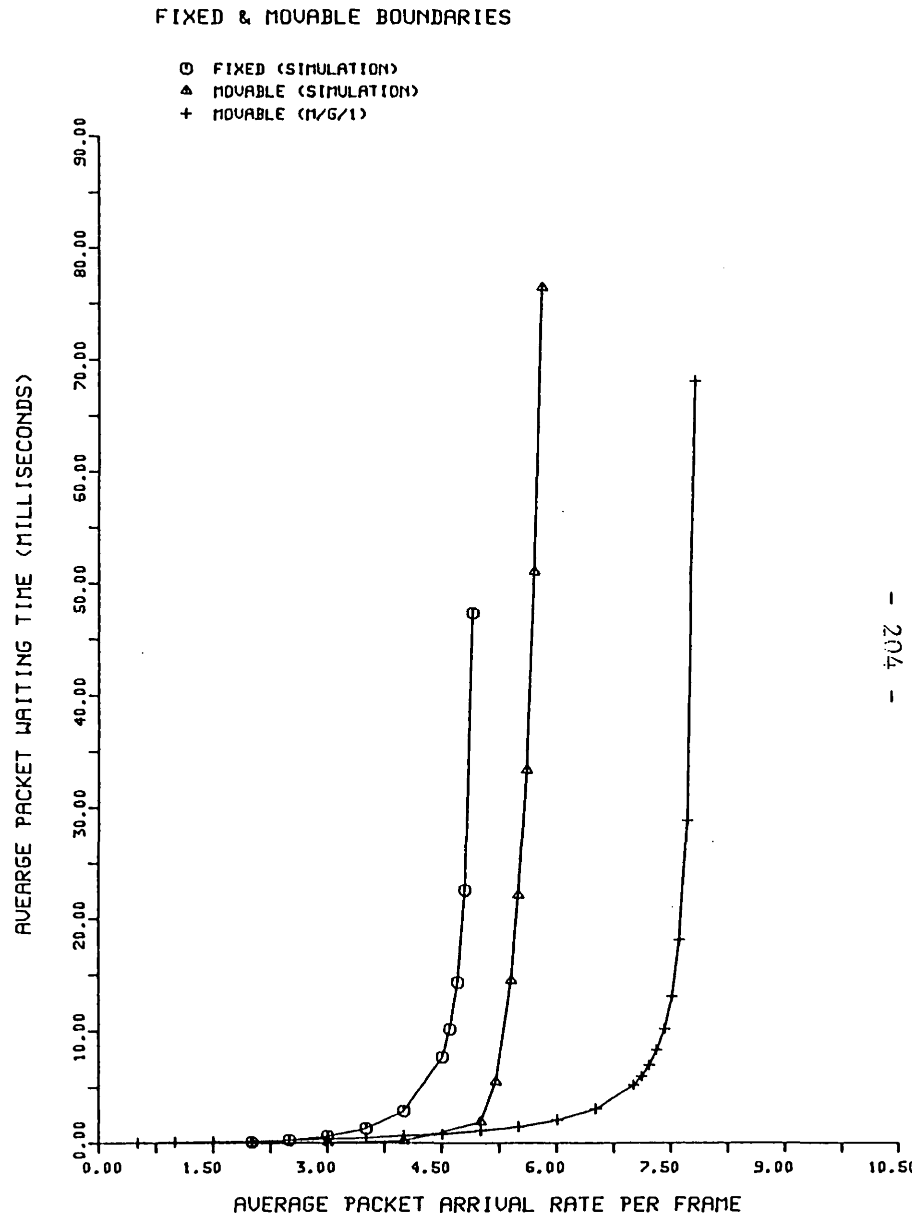


Fig. 6.8 The comparison of the data traffic performance in SENET with fixed and movable boundaries

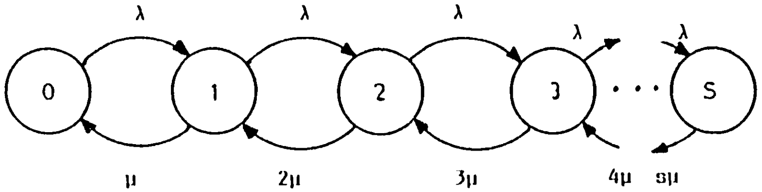


Fig. 6.9 The Markovian voice call arrival model.

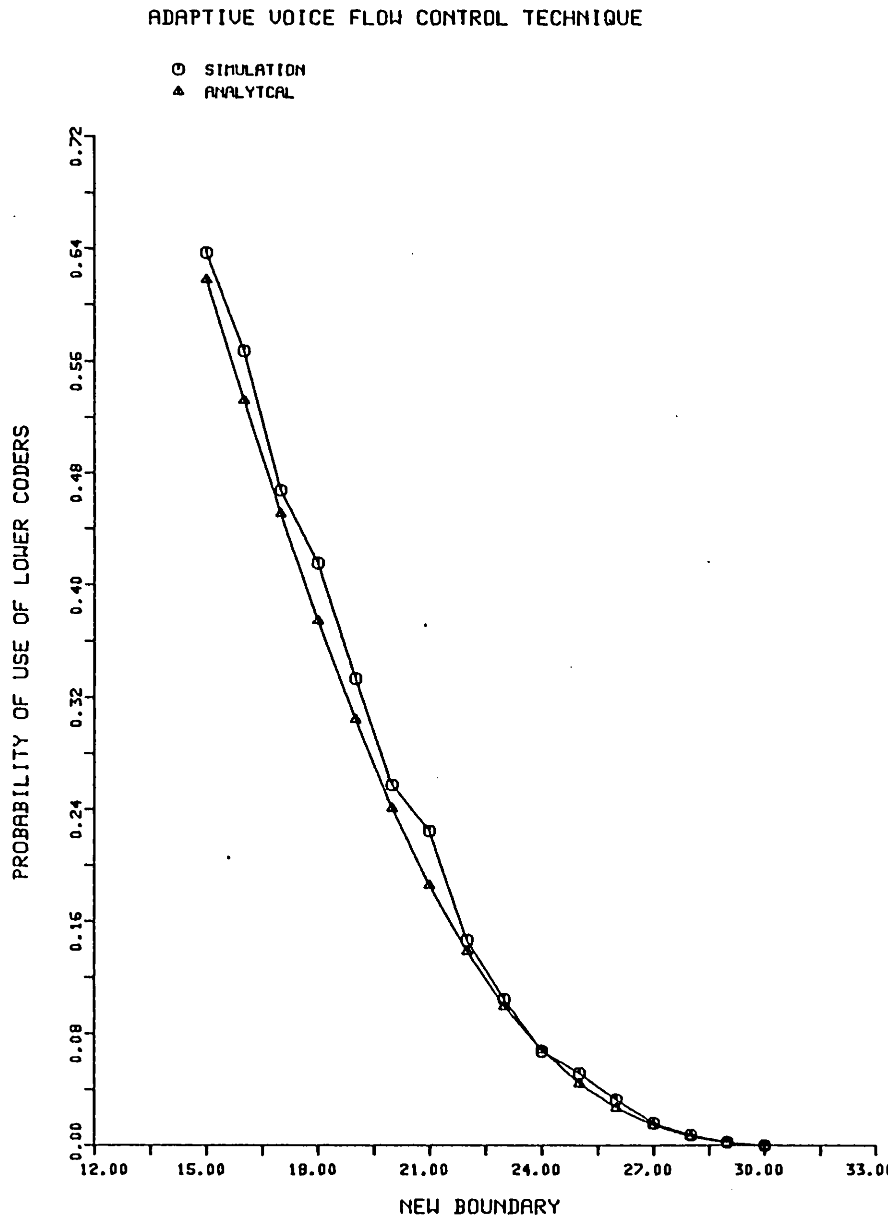


Fig. 6.10 The voice traffic performance evaluation in the movable boundary SENET conjoined with the adaptive voice flow control technique

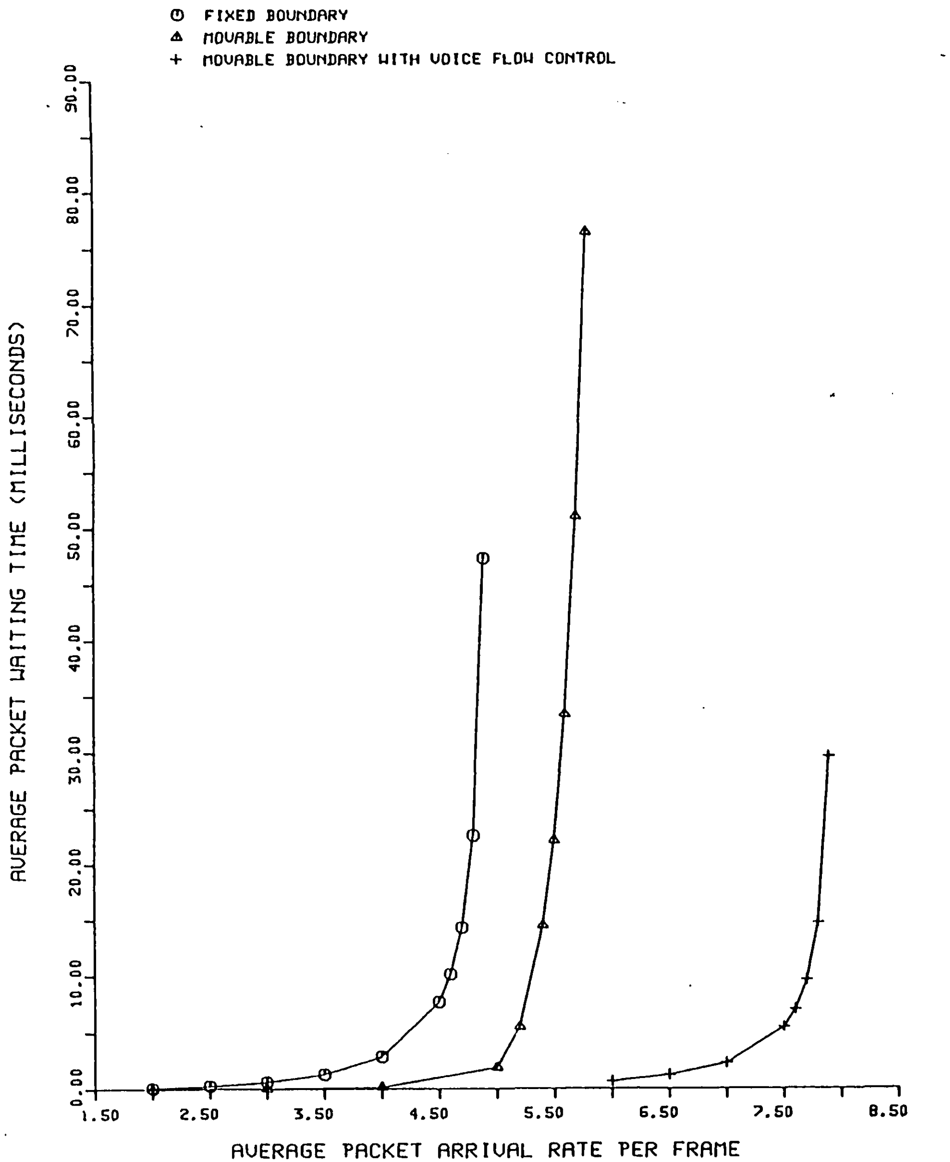


Fig. 6.11 The comparison of the data traffic performance in the SENET with fixed boundary, movable boundary, and movable boundary with adaptive voice flow control technique

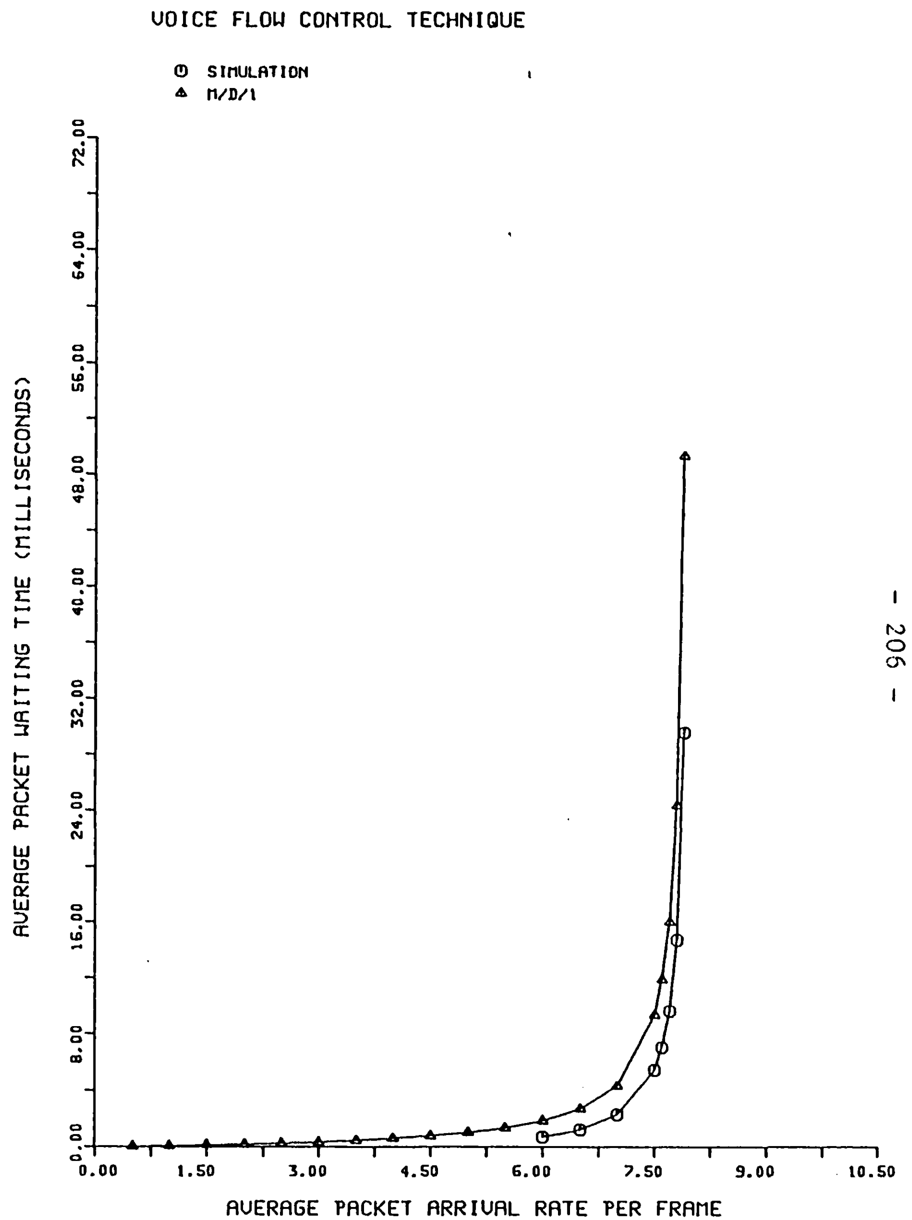


Fig. 6.12 The data traffic performance in SENET with adaptive voice flow control technique (simulation and analytical comparisons)

CHAPTER 7

INTEGRATING VOICE/DATA IN ENTIRELY PACKET-SWITCHED ENVIRONMENTS

7.1 Introduction

This chapter is concerned with the investigation of integrated voice/data in entirely packetized environments. Two separate new schemes have been considered, namely: (a) a SENET approach, and (b) a first-come-first-served (FCFS) queueing basis or mixed voice/data packet approach.

The SENET approach employs a capacity allocation scheme. This means that both the voice and data traffic require a channel reservation algorithm. The second approach is a new technique for integrating voice/data and does not need a capacity allocation scheme. Its operation is based on mixing the packets of two traffic streams according to a first-come-first-served queueing principle. This integrated approach is less complex and is easier to implement than the SENET approach.

Two different voice packetization schemes are possible with SENET i.e. the SQ and the NSQ system. Both have been investigated and their performances compared. Computer simulation models have been developed in order to evaluate the performance of the above SENET systems and the results obtained from the computer simulations have been compared with the results derived from approximate analytical queueing models. This work is reported in sections 7.2 to 7.4. In section 7.5, a flow control mechanism has been used to investigate whether the performance of the integrated SENET can be improved.

The investigation of the new integrated scheme (i.e. the FCFS or MIXED) is reported in sections 7.6 to 7.8. In section 7.9 the two integrated systems are compared together.

7.2 Integrated voice/data using the SENET approach

This scheme is as shown in fig. 7.1. In a similar manner to the previous circuit/packet SENET system, the communication between the transmitter and the receiver is based on sending and receiving frames of fixed duration Δ_t seconds. The frame is divided into two regions in order to allocate the channel capacity to each of the two traffic sources; this is depicted in fig. 7.2. The first region is allocated to the voice traffic and is partitioned into J voice packet slots, each slot is of size P_1 bits. The remaining frame capacity is allocated to the data traffic and is partitioned into L data packet slots, each slot being of size P_2 bits. Since the two traffic streams are packet-switched, it is possible for each of them to utilize idle slots assigned to the other. This means that the boundary within the frame can move forward and backward when there are free packet slots available. This operation is not possible in the circuit/packet SENET-system. In the circuit/packet form the data traffic can use the free voice channel slots but the reverse operation cannot take place due to the circuit-switched voice operation.

7.2.1. Design criteria

The scheme, as presented above, employs a boundary which must be assigned within each frame. This boundary allocation problem therefore becomes the key criterion for the design of this type of integrated system. This problem has been approached by choosing a boundary which satisfies the worst case system operation for the voice traffic. That is to say, J voice-packet slots should be assigned within the frame to the voice traffic in order to set the worst case voice-packet losses at just below, or equal to, 1%. It should be noted that the worst case voice-packet losses corresponds to the condition where all the voice channels are in conversation.

In the next two sections the integrated SENET system is investigated and the results are compared together for the two cases, i.e. for where the voice-traffic is either handled with the NSQ, or the SQ packetized-voice operation scheme. In order to facilitate the comparison of these results obtained from the entirely packetized integrated systems with the results which were obtained for the previous circuit/packet integrated SENET systems the same fixed system parameters have been used. These are as follows:

c =	1.544 Mb/s	transmission rate
n =	30	voice user populations
A =	22 Erlangs	voice traffic intensity
P_B =	.0205	probability of blocking
Δ_t =	10ms	frame length
h =	100 bits	voice packet overhead size
b =	4 bits/sample, or 32 kb/s encoder output/channel	
P_2 =	1000 bits	data packet size

7.3 SENET with NSQ-packetized voice operation

This scheme employs the NSQ-packetized system for handling the voice traffic. This means that the channels's frame period is equal to the packetization period of the voice-traffic as depicted in fig. 7.3. As shown in fig. 7.3, the frame is divided by a boundary into two regions. The first region is allocated to the voice traffic and is partitioned into J voice packet slots, where the size of each slot is equal to P_1 bits. P_1 is given by

$$P_1 = Pb + h \tag{7.1}$$

where b denotes the number of bits assigned per speech sample, h is the voice packet overhead and P is the number of voice samples in each packet. P is given by

$$P = f\Delta_t \tag{7.2}$$

where f is the sampling rate of speech signal in samples/sec, and Δ_t denotes the packetization period and is equal to the duration of one channel frame.

The second region is allocated to the data traffic and is partitioned into L data-packet slots where each slot is of size P_2 bits. The number of slots can be found from the following relationship

$$L = \left\lfloor \frac{e-JP_1}{P_2} \right\rfloor \tag{7.3}$$

where e is the channel's frame size in bits and is given by

$$e = c\Delta_t \ , \tag{7.4}$$

c is the rate of the transmission link in bits/sec.

The same synchronization delay procedure, as used for the NSQ-system, has been applied here. That is to say one frame delay is introduced at the transmitting node for the construction of a frame i.e. a duration Δ_t seconds. Thus in a framing period, if the number of voice packets (M) generated from the n-speech terminals is less than or equal to J, then the whole of the M packets are transmitted and the rest of the capacity of the voice region (if any) is given to the transmission of the data-packets. If the number of generated voice packets, M, is greater than J, then at this stage J voice packets are transmitted and M-J voice-packets are discarded, this is so that L data packets can be transmitted within the frame. If the number of data packets waiting for transmission within the frame is less than L data packets, then some, or all, of the (M-J) voice packets may use the free data- packet slots. This will result in the voice-packet losses being reduced in number.

As in the previous cases the data packets, are assumed to arrive according to a Poisson process and an infinite data queue buffer is assumed to be provided for them.

In the remaining part of this section we evaluate the performance characteristics of this type of integrated SENET system by using computer simulations and approximate analytical queueing models. The performance parameters used to assess the scheme are (i) the blocking probability, (ii) the worst-case-freezeout for the voice traffic, and (iii) the mean packet waiting time delay for the data traffic. We first present the allocation scheme which proportions the frame for this type of SENET-system.

7.3.1 Boundary allocation scheme

The location of the boundary has been obtained by using the Weinstein packet loss probability expression in order to find the value of J such that the worst case freezeout is just below, or equal to, 1%. This expression is given by equation (5.5) and is rewritten below,

$$P_L = \frac{1}{nq} \sum_{r=J+1}^{n} (r-J)\ B(n,r,q), \qquad (7.5)$$

where n denotes the total number of subscribers. Fig. 7.4 shows the characteristic of P_L versus J for the case when n = 30. It can be observed that in this case the value of J(the number of voice packet slots) for P_L to be just below, or equal to, 1% voice-packet loss corresponds to 17 packet slots. Thus by allocating 17 voice-packet slots to the voice traffic, the boundary within the frame is determined and the rest of the frame capacity is allocated to the data traffic. Similarly, this capacity is partitioned into L data-packet slots each being of size P_2 bits. For Δ_t = 10 ms and P_2 = 1000 bits, equation 7.3, yields the number of data slots,

$$L = \left\lfloor \frac{15440-17\times 420}{1000} \right\rfloor = \left\lfloor \frac{8300}{1000} \right\rfloor = 8 \qquad (7.6).$$

Thus by using the above capacity allocation scheme, the performance parameters for the voice-traffic have been evaluated and now the problem remains to evaluate the performance parameter for the data-traffic, this is the average data-packet waiting time. This problem is dealt with in the following section.

7.3.2 Data-traffic simulation model

This simulation model is simply developed by combining together at the SENET simulation model, as described in the previous chapter (section 6.3), and the Markovian simulation model for the packet-switched voice traffic. In this combined model, the voice-channel occupancy, K, during each frame interval is first computed by using the voice- traffic simulation model, as given in section 6.3.1. Then by using the Markovian talkspurt- silence model, the total number of voice packets, M, generated within each frame interval is evaluated. If M>J (where J denotes the boundary), then J voice- packets are transmitted and M-J voice packets are set aside to be discarded. The remaining frame capacity is then allocated for the data packet transmission. In this situation, the total number of data packets, n_{-},which can be accomodated is given by

$$n_{-} = L = \left\lfloor \frac{e-JP_1}{P_2} \right\rfloor \qquad (7.7)$$

However, if the total number of data packets waiting in the data queue buffer is less than L, then the (M-J) voice packets which are waiting to be discarded could use these free data- packet slots.

Alternatively, if $M \leq J$, then M voice packets will be transmitted and the remaining frame capacity will be allocated to the data packets and this is given by

$$n_{-} = \left\lfloor \frac{e-MP_1}{P_2} \right\rfloor \qquad (7.8)$$

As with the previous cases, it is assumed that the data packets arrive according to a Poisson process. By using equation (6.5), the number of data packets in the queue, n_+, is computed during each frame interval. The above process is then repeated for a large number of frames, and the average number of data- packets in the queue is computed. Then by using the Little's result, the average data- packet waiting time in the queue is evaluated.

7.3.3 Analytical queueing models for the data traffic

Similar to the previous circuit/packet movable boundary SENET system, the resulting data- packet service times are variable and are dependent upon the number of voice packets in a given frame. As the data packet arrival is a Poisson process, and the service times are variable these two properties indicate that the queueing behaviour of the data traffic could perhaps be analysed by the use of an M/G/1 queueing model. However, according to our previous experience in the circuit/packet movable boundary SENET system, we know that due to the quasi periodic effect of the voice channel occupanices the M/G/1 model is not a precise model. As was shown, this periodic effect causes much higher queueing delays to take place than predicted by an M/G/1 queueing model and therefore results in only 33% use of the free voice-channel capacity.

Taking into account the above views, the queueing performance of the system obtained from the simulation has been compared with the two analytical queueing models. The first model uses an M/G/1 queueing model. This being an idealistic model it represents the queueing behaviour of the data traffic ignoring the effect of the time dependence from correlation upon the data-packet service intervals. The second model is the single-mode fixed-boundary SENET system which was shown previously to be an M/D/1 model. This model has been used here in order to show how much the data-traffic will benefit from the movable boundary scheme.

(a) The M/G/1 model

The average data packet waiting time, $\bar{q}_{NSQ}$, is given by

$$\bar{q}_{NSQ} = \frac{\theta \overline{x^2}}{2(1-\theta\bar{x})} \tag{7.9}$$

where θ is the average data-packet arrival rate per second, and $\bar{x}$ and $\overline{x^2}$ denote the first and the second moments of the data packet service times, respectively,

$$\bar{x} = \sum_{k=0}^{n} \bar{x}_k P_k \tag{7.10}$$

and

$$\overline{x^2} = \sum_{k=0}^{n} \overline{x_k^2} \; P_k \tag{7.11}$$

where $\bar{x}_k$ and $\overline{x^2_k}$ are the first and the second moments of data packet service times respectively, given that the voice channel occupancy is k. P_k, as previously, represents the probability that the voice channel occupancy is k, and is given by the Erlang distribution,

$$\bar{x}_k = \sum_{r=0}^{k} x_r \; B(k,r,q) \tag{7.12}$$

$$\overline{x^2_k} = \sum_{r=0}^{k} x^2_r \; B(k,r,q) \tag{7.13}$$

where x_r denotes the data packet service time when the number of voice packets in a frame is r. This is given by

$$x_r = \frac{P_2}{c\delta_r} \tag{7.14}$$

where δ_r is the probability of the channel availability for the data traffic with r voice packets being occupied in a given frame. This is given by

$$\delta_r = \frac{\left\lfloor \frac{e-rP_1}{P_2} \right\rfloor P_2}{e} \qquad \text{if } r \leq J \tag{7.15}$$

and

$$\delta_r = \frac{\left\lfloor \frac{e-JP_1}{P_2} \right\rfloor P_2}{e} \qquad \text{if } r > J \;. \tag{7.16}$$

(b) The M/D/1 model

This is a straight forward single mode fixed boundary SENET system with the boundary set at J voice packet slots. The average data packet waiting time, denoted by $\bar{q}_{NSQ}$, is given by equation (6.16) which can be rewritten below

$$\bar{q}_{NSQ} = \frac{\theta P_2^2}{2c\delta_f(c\delta_f - \theta P_2)} \;, \tag{7.17}$$

where δ_f denotes the probability of channel availability for the data packet and is given by

$$\delta_f = \frac{\left\lfloor \dfrac{e - JP_1}{P_2} \right\rfloor P_2}{e} \tag{7.18}$$

7.3.4 Results and comparisons

It was shown that with the fixed system parameters (i.e. $c = 1.544$ Mb/s, $n = 30$, $\Delta_t = 10$ ms, $b = 4$ bits/sample and $P_2 = 1000$ bits), the capacity which has to be assigned to the voice traffic is equal to 17 voice-packet slots, with each slot being of size 420 bits. The capacity which has then to be assigned to the data traffic is 8 data-packet slots, where each of the data-packet slots is 1000 bits. Employing the above fixed-system parameters and setting the location of the boundary within each frame, the system has been simulated using the simulation model of section 7.3.2. The length of the simulation program run was set at 60,000 seconds of real time and statistics were compiled only after the system had run for 10,000 seconds. The program was run five times and in each run a different initial condition was set at the commencement of the simulation run. The simulation results for the average of five runs have been obtained and together with the corresponding results obtained from the analytical queueing models (i.e. M/D/1 and M/G/1) are dipicted in fig. 7.5. It can be observed that the same effect, as experienced previously in the circuit/packet movable boundary SENET system, has resulted here. That is to say, the results predicted by the M/G/1 model were not attained by the computer simulation. The same reasons hold for this difference as found previously; this is because the data packet service intervals are correlated from frame to frame. As a result of this effect the data traffic has to be limited and hence the free voice-packet slots which are available during the periods of low voice channel occupancies will be under utilized.

7.4 SENST with SQ-packetized voice operation

This scheme is shown in fig. 7.6. In this form, the voice-traffic utilizes the finite queue buffering scheme at the transmitter and the CTI-reassembly strategy at the receiver. The voice-packetization period is also assumed to be equal to the channel's framing period (i.e. $D_z = \Delta_t$). This also means that each revolution of the multiplexer takes place in Δ_t seconds. However, in a similar manner to the previous SENET system with NSQ operation, an additional Δ_t seconds delay has been added to the voice-traffic delay at the transmitter for the purpose of correct frame construction. This delay has been achieved by introducing an additional buffer, denoted by B1 in fig. 7.6, and is used for the assembly of each frame. After each frame has been assembled in Buffer B1 the frame is transferred to the queue buffer B2 at an equivalent rate equal of c bits/sec, this being equal to the transmission link bit rate. However, the transfer has to be in bursts and is not a continuous flow. The system has been assumed to use the same form of hybrid-switched, master-frame format strategy as was used for the SENET with NSQ operation. This means that the communication between the transmitter and the receiver is based on sending and receiving fixed frames each of duration Δ_t seconds. Each frame is divided into two regions by means of a boundary. The first region is allocated to the voice-traffic and is partitioned into J voice-packet slots, each slot being of size P_1 bits. The second region is dedicated to the data-traffic and is partitioned into L-data packet slots, each of size P_2 bits. Synchronization is ensured between the buffers as the data in B1 is assumed to be loaded into B2 at the commencement of each frame period.

During low activity periods, when the queue buffer B2 is low or empty, buffer B1 provides the channel with speech packets via buffer B2 with little or no extra delay except for the assembly delay. During these periods of low activity if the number of packets generated from the speech terminals, M, is less than J, then M voice packets are transmitted and the rest of the frame can be dedicated to data traffic (i.e. by means of a movable boundary operation similar to that of the circuit/packet SENET system). If the number of packets generated from the speech terminals, M, is greater than J, then J voice packets are transmitted in the given frame and M-J voice packets are retained in the queue buffer B2. At this stage if the number of data packets,n, waiting in the data-queue buffer are less than L, then n_d data packets will be transmitted and $L-n_d$ of the free data-packet slots can then be dedicated to the voice traffic. This means therefore that the boundary moves forward and backwards within the frame and speech packets can occur at the beginning and the end of a frame interval.

7.4.1 The boundary allocation

In a similar manner to the previous case, this scheme also requires the capacity of the transmission link to be allocated separately to each of the two traffic flows. This means that a particular boundary should be assigned within the frame. This boundary will provide the number of voice packet slots J necessary to hold the worst-case freezeout at 1%, or slightly less. The capacity allocation problem has been evaluated by means of computer simulations. The simulation model has been developed by modifying the Markovian simulation model for the PVS, as given in chapter 3 section

3.4. For assessment of the model the total number of voice channels, n, has been fixed at 30. By using the Markovian talkspurt/silence model, the total number of voice packets, M, generated during each frame interval have been computed. If, during any frame interval, $M > J$ (where J denotes the maximum number of voice packets transmitted per frame), then J voice packets are transmitted within the given frame interval and M-J voice packets are stored in the voice queue buffer B2. If $M \leq J$, and if the voice queue buffer B2 is empty, only M voice packets are transmitted within the frame and the remaining (J-M) voice-packet slots are dedicated to the data traffic. If the voice-queue buffer is not empty, then up to a maximum of J voice packets are transmitted, provided that the queue buffer B2 contains $\geq$ J-M voice packets. The above queueing-behaviour process has been used in the simulation program and can be summarized by the following equation

$$Q(i+1) = \max(Q(i) + M-J,0) \qquad (7.19)$$

where Q(i) denotes the length of the voice-queue buffer in number of voice packets at the i^{th}-frame.

It should be noted that the above queueing behaviour for the voice traffic represents a discrete process. This is because at the beginning of each frame, the number of packets in the queue is an integer. This SQ-system queueing behaviour considered here is therefore different from that considered previously. It has been assumed that the queue buffer B2 for the SQ-system is of finite length in order to limit the end-to-end (ETE) packet delay to 250 ms. The system is also assumed to employ the CTI reassembly strategy at the receiver for the voice traffic, and it is assumed that the ETE delay is set to a constant delay of 250 ms. Thus the length of the queue buffer B2, denoted by D_q, becomes

$$D_q = .25 - D_z - D_{sy} - D_s \text{ secs.} \tag{7.20}$$

where D_z is the packetized delay and is equal to Δ_t seconds, D_{sy} is the delay applied at the buffer B1 for synchronization and is equal to the packetization delay, and D_s is the voice packet service time delay and is equal to $\frac{P_1}{c}$ secs. By substituting these parameters into equation (7.20), the buffer length D_q, in seconds, becomes

$$D_q = .25 - \Delta_t - \Delta_t - \frac{P_1}{c}$$

$$= .23 - \frac{P_1}{c} \tag{7.21}$$

for 10 ms frames.

Alternatively, the length of the queue buffer B2, in number of voice-packets, Q_N, becomes

$$Q_N = \frac{CD_q}{P_1} = \frac{c(.23 - \frac{P_1}{c})}{P_1}$$

$$= \frac{.23c - P_1}{P_1} \tag{7.22}$$

Thus the voice queue buffer length, Q_N, in number of voice packets is set in the simulation program. Results have been obtained using our standard set of test parameters (i.e. c = 1.5444 Mb/s, n = 30, A = 22 Erlangs, Δ_t = 10 ms, and b = 4 bits/sample). The program was run for several values of J and in each run the percentage of voice-packet losses was measured. The simulation period was set at 60,000 seconds of real time and the results of the first 10,000 seconds were discarded. The results obtained from this simulation are depicted in fig. 7.7. This figure shows the worst-case voice-packet loss as a function of the number of voice-packet slots (J). It can be observed that in order to have just less than 1% voice-packet losses, 15 voice-packet slots are required. Thus this SENET-system will produce an improvement in the data-traffic performance as compared with the previous SENET with NSQ-voice operation. This is because in the SENET with NSQ-voice operation 17 voice-packet slots were required in order to meet the 1% worst-case freezeout condition. By using equation (7.3), the number of data-packet slots which will be reserved for the data traffic can be evaluated as

$$L = \left\lfloor \frac{15440-15\times 420}{1000} \right\rfloor = \left\lfloor \frac{9140}{1000} \right\rfloor = 9 \qquad (7.23)$$

which means that at least 9 data-packet slots will be allocated to the data traffic within each frame.

7.4.2 Data traffic simulation model

This simulation model has been constructed by using the simulation model presented for the SQ-system, as given in the previous section, and combining it with the voice-traffic simulation model together with the data-traffic simulation model which were given in section 6.3.1 and 6.3.2 respectively. The voice-traffic intensity was set at 22 Erlangs, this gives a blocking probability of .0205. The boundary was set within the frame and the data packets were generated at the rate of θ packets/sec. The average data-packet waiting time in this scheme has been evaluated by using a procedure similar to the one carried out in the previous chapter, section 6.3.2. The number of data packets in the queue for each frame interval has been evaluated by using the following relationship

$$n_d^{(i+1)} = \max(n_d^{(i)} + n_+ - n_-,\ 0), \qquad (7.24)$$

where $n_d^{(i+1)}$ and $n_d^{(i)}$ are the number of data packets in the queue at the commencement of the (i)th and the commencement $(i+1)^{th}$ frame respectively; n_+ is the number of data packets which arrive in the i^{th} frame and n_- is the total number of data packets sent in the i^{th} frame. This is given by:

$$n_- = \left\lfloor \frac{e - MP_1}{P_2} \right\rfloor \quad ; \quad 0 \leq M \leq J, \qquad (7.25)$$

The above equation is for the periods when the voice-queue buffer is empty and the number of voice-packets which arrive during the frame intervals are less than or equal to J. Thus during these periods the data packets use the free voice packet slots (i.e. there is a movable boundary operation). During the rest of system operation, the voice-queue buffer will grow and thus the maximum number of data-packets which will be sent per frame is given by

$$n_- = L = \left\lfloor \frac{e - JP_1}{P_2} \right\rfloor \qquad (7.26)$$

During these periods, however, if there are any free data packet slots available, then these free capacities will be dedicated to the voice-packets waiting in the voice queue buffer.

The above process is represented for a large number of frames and the average number of data packets in the queue has been evaluated. Then by using Little's result, the average data-packet waiting time has been evaluated.

7.4.3 Analytical queueing models for the data traffic

As with the previous movable boundary SENET-systems, the data packet-service intervals will be variable in this type of SENET-system and are dependent upon the number of voice packets transmitted within a given frame. Using similar arguments to those used earlier, the fact service intervals are variable, and taking the packets to arrive according to a Poisson process, we can approximately describe the queueing behaviour of the data traffic as an M/G/1 queueing process. As shown previously, in order to solve for the average data waiting time in the M/G/1 model the first and second moments of the data-packet service- times are required. These moments, however, are difficult to evaluate analytically in this situation. However, computer simulation, can be used to evaluate these moments. Our previous experience with the SENET-system employing movable boundary operations indicate that the quasi-periodic voice-channel occupancies imply that the M/G/1 queue is not a precise representation of the queueing behaviour for the data traffic. For these reasons no attempt has been made here to solve the above M/G/1 queueing model. However, the queueing behaviour of a single-mode, fixed boundary SENET-system has been solved. This model, which is an M/D/1 model, has been used to evaluate the benefits which data traffic experience when the movable boundary scheme is employed. The solutions for this model are given by equations (7.17) and (7.18).

7.4.4 Results and comparisons

By setting the capacity reservation of each of the two traffic streams and using our fixed system parmeters, the integrated SENET with SQ-packetized voice operation has been simulated. From the results the characteristic of the average data-packet waiting-time versus the average data-packet arrival-rate has been obtained. The length of the simulation period was again set at 60,000 seconds of real time and the results of the first 10,000 seconds were discarded. The program was repeated five times and in each run a different initial random number seed was used. The results of the average of five runs have been obtained and are given in Fig. 7.8, together with the previous computer-simulation characteristics obtained from all the other SENET systems (i.e. fixed boundary, movable boundary, movable boundary with voice flow control technique, and movable boundary with NSQ-packetized voice operation). It can be observed that the data-traffic performance in the two SENET-systems with packetized-voice operation (i.e. NSQ and SQ) are much superior to the others. As was expected, when packetized-voice schemes are adopted, the integrated system can handle considerably more data than that for circuit/packet SENET systems. The results also indicate that the data traffic has marginally better performance for the SENET with SQ than the SENET with NSQ packetized-voice operation. The reason is simply because with the SQ-SENET more of the transmission-link time is available for the data traffic than with the NSQ-SENET.

The single mode M/D/1 analytical queueing model has been solved for the SQ-SENET system and the characteristics have been obtained for J = 15 voice-packet slots. The results, together with the simulation results, of the SQ-SENET system are depicted in fig. 7.9. It can be observed that the simulation results indicate a better queueing performance than obtained for the M/D/1 model. However, the improvement is not as large as the improvement with the NSQ-SENET. (see fig. 7.5). This comparison therefore indicates that on average, the data traffic in the NSQ SENET utilizes more of the free voice-packet slots than in the SQ SENET. The reason for this is due to the voice- queue buffer build-up in the SQ-SENET system. This effect causes the data-packet's service intervals to be more correlated during the peaks of the voice-channel occupancies. Thus this correlation effect causes more data packets to build up, and therefore reduces the data- traffic performance.

Fig. 7.10 shows the computer simulation comparison of the data-traffic performance in both SQ and NSQ SENET systems when the voice and data packets are of equal size (i.e. $p_1 = p_2 = 420$ bits). Here the data-traffic performance in the SQ-SENET is slightly better than for the NSQ-SENET. The important conclusion which can therefore be drawn from the above results is that the performance of the data traffic in the SQ-SENET is substantially better than for the circuit/packet SENET systems, but is only slightly better than for the NSQ-SENET system.

7.5 Flow control Mechanism in SENET with totally packetized operation

The work which has been presented so far on the SENET with totally packetized operation, has been based on establishing the boundary that meets the worst case freezeout objective for the voice traffic. That is to say the boundary has been adjusted within the frame so that when all the voice channels are connected, the resultant voice- packet losses are below, or equal to 1%. Another way of adjusting the boundary is to let it be set according to the number of call connections so that in each case the resultant voice-packet losses are just below, or equal to 1%. The advantage of using this scheme is that as the instantaneous number of call connections reduces, less of the transmission link capacity will be needed to set the voice packet losses to the required limits. This flow control operation will therefore improve the data-traffic performance.

In this section, we introduce this type of flow control mechanism to the SENET with NSQ packetized-voice operation. By using equation (7.5), for each value of n(n denotes the instantaneous number of call connections), a value of J voice-packet slots has been evaluated and the results are listed in table 7.1. This table simply shows that as the number of call connections reduces, the number of packet slots required will be reduced. This table has been used in the simulation program for the SENET with NSQ packetized-voice operation the model for which has been explained in section 7.3.2. The necessary modifications required in the simulation program have been made in such a way as to allow the boundary to be moved in accordance with table 7.1.

The data-traffic performance characteristic has been obtained from the simulations and is depicted in fig. 7.11. This characteristic is for the condition that both the voice and data packets are of the same size, i.e. 420 bits. From this figure, it can be observed that little improvement in the data traffic performance has been achieved when this scheme is adopted. The simulation results for this scheme, similar to the previous cases, have been obtained by setting the simulation period to 60,000 seconds and ignoring the results of the first 10,000 seconds. Each plotted point also represents an average of 5 runs, where each run is performed with different initial conditions.

Table 7.1

Number of call connections	Boundary J, in number of voice-packet slots
0	0
1	1
2	2
3	3
4	4
5	4
6	5
7	6
8	6
9	7
10	7
11	8
12	8
13	9
14	9
15	10
16	10
17	11
18	11
19	12
20	12
21	13
22	13
23	14
24	14
25	15
26	15
27	16
28	16
29	17
30	17

7.6 Integrating voice/data by using first-come-first-served approach

This type of integrated system is different from those which have been studied so far. The scheme, as presented here, no longer needs the capacity allocation that is basic to the SENET approach.

Instead, in this form the packets of the two traffic sources are mixed together by using the concept of first-come-first-served (FCFS) queueing. The scheme is shown in fig. 7.12. In this form, the voice traffic employs the same packetization and multiplexing techniques that were used for the PVS. This means that the voice packets, after being generated, are fed to the voice queue buffer, B1, by the multiplexer. Similarly the data packets are assumed to arrive into an infinite data-queue-buffer, denoted by B2, and as with the previous cases it has been assumed that the data packets arrive according to a Poisson process with a rate of θ packet/secs. The transmission link is assumed to serve the packets of each traffic source according to a FCFS protocol. This means that at any particular time the channel serves either the data or the voice buffer according to the generation time of the packets which are at the head of the two queues. Simply, the packet which has been generated first, be it either a voice packet or a data packet, will be served first. In order to perform this FCFS operation a synchronization method is required. Synchronization can be achieved simply by providing two synchronised clocks for the two traffic streams, the clocks synchronized to the same time reference. Each packet needs to have a time stamp placed into its header.

In the transmission of packets arising this FCFS queueing scheme the packets of both the voice and the data traffic are mixed together. The system therefore can be viewed as to having one queue buffer instead of two. This is shown in fig. 7.12. As is shown, the integrated system can be replaced by a single queue buffer, denoted by B, and the input to this buffer is the mixed voice and data packets.

As a result of this mixed operation, both traffic streams will experience the same type of queueing delays. This is an important assumption which has been used here for the evaluation of the performance of this integrated system.

Although this integrated system can be easily implemented, it does not seem to have been investigated by other researchers.

7.6.1 Performance criteria of the integrated system

As stated above, the two traffic streams will experience the same type of queueing delays. However, since the speech is a real-time traffic, and should not experience large end-to-end delays, then it has been assumed that the system has to be designed to meet the requirements imposed upon the voice traffic. That is to say, the system should be designed so as to operate within the worst-case freezeout for the voice traffic of better than 1%. As with the previous cases, it has been assumed that the voice traffic employs the CTI-reassembly strategy at the receiver. It has also been assumed that the use of this strategy causes the voice traffic to have a constant ETE delay of 250 ms. This delay threshold is approximately the worst-case delay for smooth conversational telephony.

For the above situation , if the number of speech terminals in this system is fixed, then the maximum data-packet arrival-rate should be so controlled so as not to force the worst-case freezeout of the speech terminals (i.e. the condition where all the speech terminals are in conversation) to exceed 1%. Hence the main performance characteristic of this system is a graph of the number of speech terminals permitted versus the data packet arrival rate, for the condition when the worst-case voice-packet losses are below or equal to 1%.

In the next section, by using a simple analytical queueing model, the performance characteristic indicated above has been evaluated for a special case. As will be shown, the investigated case employs a number of crude assumptions which may not be true in a practical system. Thus in a later section, computer simulation models are presented which can be used to evaluate the true performance characteristics of this type of integrated system.

7.7 An analytical model for a special type of FCFS integrated voice/data system

In this section an analytical model is presented which evaluates the performance of an FCFS system based upon the following assumptions:

(1) The speech terminals are assumed to be continuously connected. This assumption is made in order to encounter the worst-case packet losses for the voice traffic.

(2) The packet arrival from both speech and data terminals are assumed to be Poisson processes.
This approximation is known to be good for the data traffic. For the voice traffic however, as was shown previously, this is only approximately valid when the number of speech terminals is fairly large (i.e. of order of 100, accordingly by figs. 3.17 and 3.18). The transmission link is assumed

to be a T1-carrier, and it is assumed that the encoders operate at 16 kb/s [this is in order to give a large number of speech terminals and to limit the simulation time]. The importance of this assumption is that as the two Poisson process are mixed together then the resulting process will also be a Poisson process (see Cooper [22]). The rate of the new Poisson process is equal to the sum of the rates of the two input processes.

(3) The data packet length is assumed to be variable so that when it is mixed with the voice packets, the resulting packet length distribution can be approximated by a negative exponential with the mean equal to the voice-packet length.

Employing the above assumptions, the queueing behaviour of the two mixed processes becomes approximated by a single M/M/1 queueing model. The analysis for evaluating the performance of this integrated system based on the M/M/1 queueing model is as follows: Let us assume the average rate of the voice and data-packet arrivals per second to be denoted by θ_v and θ_d respectively. Then according to assumption (2), the average packet arrival to the single queueing system (i.e. buffer B) is given by

$$\theta = \theta_v + \theta_d \qquad (7.27)$$

From Kleinrock [23] (vol. 1 pp 401), the CDF of the waiting time in an M/M/1 queueing model is given by

$$W(y) = P[\omega \leq y] = 1 - \rho e^{-\mu(1-\rho)y} \qquad (7.28)$$

where W(y) denotes the probability that the waiting time is less than, or equal to y; μ^{-1} is the average packet service time; and ρ is the packet-switched link utilization, and is given by

$$\rho = \frac{\theta}{\mu} = \frac{\theta_v + \theta_d}{\mu} \tag{7.29}$$

By using equation (3.23), the average voice-packet arrival rate, θ_v, is obtained and is given by

$$\theta_v = \frac{n_v q}{\Delta_t} \tag{7.30}$$

where n_v is the number of speech terminals, q is the voice packet generation probability and is equal to .445, and Δ_t is the voice packetization period, which is assumed to be 10 ms.

By employing the third assumption, the average packet service time, μ^{-1}, becomes equal to

$$\mu^{-1} = \frac{p_1}{c} = \frac{pb+h}{c} \tag{7.31}$$

By substituting (7.31) and (7.30) into (7.21), it can be shown that

$$\theta_d = \mu\rho - \frac{qn_v}{\Delta_t} \tag{7.32}$$

The above relationship shows that for a constant channel utilization ρ, the average data packet arrival rate, θ_d, varies linearly with the number of speech terminals, n_v. However, in order to obtain this characteristic the link utilization, ρ, should be determined. This term can be obtained by using the 250 ms constant ETE delay and the less than 1% packet loss requirements for the voice traffic. That is to say, the following relationship should be satisfied

$$W(y) = .99$$

where

$$y = .25 - \Delta_t - \mu^{-1} \qquad (7.33)$$
$$= .24 - \mu^{-1}.$$

By substituting (7.33) into (7.28), the following relation will result.

$$\rho e^{-\mu(1-\rho)(.24-\mu^{-1})} = .01 \qquad (7.34)$$

In the above equation all the terms are known except ρ. The Newton-Raphson method (see Ref [81]) has therefore been used for the evaluation of ρ from the equation 7.34. The consequence criterion employed for solving the equation 7.34 is

$$\rho_2 = \rho_1 - \frac{\rho_1 e^{-\mu(1-\rho_1)(.24-\mu^{-1})} - .01}{[1+\mu(.24-\mu^{-1})\rho_1]e^{-\mu(1-\rho_1)(.24-\mu^{-1})}} \qquad (7.35)$$

$$|\rho_2 - \rho_1| \leq 10^{-9} \qquad (7.36)$$

By evaluating ρ from the above equations (7.35 and 7.36)and using equation(7.32), the characteristic of n_v versus θ_d has been obtained and is depicted in fig. 7.13. In this figure the vertical axis represents the number of speech terminals and the horizontal axis represents the average data-packet arrival-rate per frame (i.e. $\theta_d\Delta_t$).

If the packet generation probability of the data terminals is known, then the horizontal axis of this figure can be easily converted to the number of data terminals, n_d, according to the following relationship.

$$n_d = \frac{\theta_d \cdot \Delta_t}{q_d} \tag{7.37}$$

where q_d represents the packet generation probability for the data terminals.

As shown previously, in this type of integrated system the average data-packet waiting-time is the same as the average voice-packet waiting time and according to the M/M/1 queueing model, this is given by

$$\bar{W} = \frac{1/\mu}{1-\rho} \tag{7.38}$$

Since the characteristic of figure 7.13 has been obtained for a constant ρ, then for this characteristic the average data packet waiting time, according to equation (7.38), will be constant and is given by

$$\bar{W} = \frac{(260/1544000)\text{x}1000}{1 - .99676} \cong 52 \text{ ms} \tag{7.39}$$

There is a substantial difference between this type of integrated system and the previous SENET type of integrated approach. In the SENET approach, the data-packet waiting-time increases as the data-arrival rate increases, as with all conventional queueing systems. High data traffic does not however cause deterioration in the flow of voice traffic as adequate slots are permanently provided. However, in the FCFS system the maximum data traffic has to be limited in order for voice traffic to be kept within design specifications.

Unfortunately, the above M/M/1 analytical approach consists of a number of unrealistic assumptions. For example, the third assumption is the most unrealistic as the two packet distributions are unlikely to form one negative exponential distribution. If this is to be the only unrealistic case, then the queueing model of the system should be changed from the M/M/1 to an M/G/1. However, the CDF of the waiting time for the M/G/1 model is difficult to evaluate analytically. This assumption is not the only unrealistic one. The second assumption cannot always be true. For example, in the evaluation of the characteristic of fig. 7.13, as the number of speech terminals are reduced, the packet arrival pattern will no longer be a Poisson process.

Due to the severe limitations discussed above, for the rest of this chapter computer simulation techniques have been used to evaluate the true performance characteristics of this type of integrated packet-switched voice/data system.

7.8 Performance evaluation of the FCFS integrated voice/data system using computer simulations

The simulation model for this scheme has been developed by combining the Poisson data packet arrival simulation model and the PVS simulation model for which all the speech terminals are in conversational mode. These simulation models have been presented in detail in chapter 6 and chapter 3 respectively.

With this type of integrated system the packets of the two traffic sources are mixed together, so it has been assumed that one queue Buffer at the transmitter can represent the queueing situation. (i.e. the buffer B in fig. 7.12). During each frame interval, i.e. Δ_t seconds, the packets which have been generated from both the traffic streams are mixed together and they are stored in the single queue buffer B. For simplicity it is assumed that the packets of the two streams are of constant length (i.e. $P_1 = P_2 = Pb+h$). Thus during any two successive frames; e.g., i^{th} frame, the total number of packets in the queue buffer B is given by

$$Q^{(i+1)} = \max(Q^{(i)} + K_v + K_d - L, 0) \tag{7.40}$$

$Q^{(i)}$ is the total number of packets in the queue at the beginning of the i^{th} frame, K_v is the number of voice packets arriving during the i^{th} frame, K_d is the number of data packets arriving during the i^{th} frame, and L is the number of packets transmitted by the channel during the i^{th} frame. Hence the number in the buffer at the start of the (i+1) frame is given by the following relationship

$$L = \frac{c\Delta_t}{Pb+h} \tag{7.41}$$

where P is the number of speech samples in a packet, b is the number of bits assigned per speech sample, and h is the size of the voice packet overhead in bits.

In this simulation, the performance characteristic of the integrated system (i.e. the characteristic of the number of speech terminals versus the average data-packet arrival-rate per frame), was obtained. The following assumptions have been made.

(1) The listed parameters have been assumed to be fixed throughout the simulations:

$c = 1.544$ Mb/s,

$\Delta_t = 10$ ms,

$b = 8$ bit/sample or 64 kb/s/subscriber channel,

$h = 100$ bits

$P_L \leq 1\%$,

$P_1 = P_2 = 740$ bits.

(2) The data packet arrival rate, θ_d, has been set at a particular value.

(3) By means of prediction, the number of speech terminals, n_v, have been chosen for the above data-packet arrival-rate, θ_d, so that the percentage of packets exceeding 250 ms is just above 1%.

(4) The buffer length threshold, y, has been obtained so that the ETE voice-packet delay becomes equal to 250 ms. This threshold, expressed in numbers of packets, is

$$y = \frac{240C - (Pb+h)}{1000(Pb+h)} \tag{7.42}$$

(5) The simulation program has been run for 60,000 seconds of real time with the results of first 10,000 seconds being ignored. During the course of this simulation process, a record was kept of (a) the total number of packets generated by the two traffic streams, x_{tot}; and (b) the total number of packets for which the queue length exceeded y-packets, x_{250}. The voice-packet loss probability, P_L, is obtained from the ratio of x_{250}/x_{tot}. If P_L was found to be greater than 1%, then the number of speech terminals was reduced by 1 and steps (3 to 5) above, were repeated and a new value for P_L obtained. If P_L becomes $\leq$ 1%, then the data packet arrival rate, θ_d, was changed and the procedures as 2 to 5 were repeated as above.

Fig. 7.14 shows the characteristic of the number of speech terminals, n_v, versus the average data packet arrival rate obtained from the simulations.

The average data-packet waiting time has been obtained by employing the same procedure as was undertaken previously for the SENET-system. Using equation (7.40), the average queue length has been obtained from by the following relationship

$$\bar{Q} = \frac{\sum_{i=0}^{XL} Q^{(i)}}{XL} \qquad (7.42)$$

where $Q^{(i)}$ is the queue length at the i^{th} frame, and XL denotes the total number of frames in the simulation run. Hence by employing Little's result the average data packet waiting time can be obtained using the relation

$$\bar{W} = \frac{\bar{Q}}{(\theta_d+\theta_v)} \quad ; \qquad (7.43)$$

where θ_v is the average voice packet arrival rate per second, as given by equation (7.30).

Fig. 7.15 shows the characteristic of the average data packet waiting time versus the average data-packet arrival-rate per frame. The number of speech terminals for this characteristic is as given by figure 7.14. The voice-packet losses for this characteristic are $\leq$ 1%. As was expected, it can be seen that the average data-packet waiting time is nearly constant throughout for a range of average data-packet arrival-rates.

Figures 7.16 and 7.17 show the comparison of the simulation results with the results obtained from the M/M/1 queueing model, as formulated in the previous section. Both curves are for a coder output rate of 64 kb/s. It can be seen that there is little agreement between the two sets of results. As was expected, the M/M/1 queueing model has been found to be an inadequate approach to the performance evaluation of this integrated model.

In this integrated system model , the buffers B1 and B2 in fig. 7.12 were of infinite size and voice packets were not discarded at the transmitter. It was therefore assumed that the voice packets whose end-to-end delays exceed 250 ms had to be discarded at the receiver. The disadvantage of this method of operation is that the voice packets which have to be discarded at the receiver load the queue buffer at the transmitter and waste resources of the transmission link. It would be better if these voice packets were discarded at the transmitter. However, unlike in the simple PVS system, the finite-buffering strategy cannot be used to discard the voice packets at the transmitter. This is because the queueing delay that a voice packet experiences in this integrated system is a function of the content of the two queue buffers in the system. However, the maximum voice-queue buffer which the system needs is predictable and is given by equation (7.42). This is determined for the case where the data-queue buffer is empty and therefore the transmission link can serve the voice queue buffer continually. Thus, any packets that overflow this buffer length will necessarily have delays exceeding 250 ms. When the buffer is not empty, the system should not let the voice buffer fill up because the packets at the end of this buffer will have delays exceeding 250 ms. The system therefore requires a more complex scheme for discarding the voice packets. This can be achieved by the use of a flag which is set when the packet enters the voice-queue buffer. The system sets this flag according to the status of the two queue buffers. Thus whenever the total number of voice and data packets in the two queue buffers exceeds y-packets (y is given by equation (7.42)), then the flag is activated. Hence when any voice packet arrives at the transmitter when the flag is activated, it is rejected by the system, and is hence not transmitted.

The computer simulation model, as previously presented for this integrated system, has been modified to incorporate the above system operation. The results obtained from this program, together with the previous results (where the voice packets are discarded at the receiver), are depicted in fig. 7.18. This figure shows the characteristics of the number of speech terminals versus the average data packet arrival rate per frame for the two cases. It is evident that the integrated system has a better performance when the system operates with the voice packet losses taking place at the transmitter.

That is to say, the system can handle more speech and data terminals with finite voice-queue buffering. The comparisons of the average data-packet waiting time for the two integrated systems are shown in fig. 7.19. It can be observed that there are substantially larger queueing delays for the data traffic when the voice packets are discarded at the transmitter. This being due to the larger number of voice channels which can be permitted to be attached to a system as a result of the improved TASI figures.

7.9 Comparison of the SENET and the FCFS

The aim in this section is to compare the performance characteristics of the two integrated systems. We assume the two systems have the following fixed system parameters.

transmission rate $c = 1.544$ Mb/s,

number of voice users $n = 30$,

probability of blocking $P_B = 2\%$

voice digitization rate $B = 32$ Kb/s,

frame length $\Delta_t = 10$ ms,

voice packet overhead $h = 100$ bits,

voice packet size $P_1 = 420$ bits,

data packet size $P_2 = P_1 = 420$ bits

percentage of voice packets lost $P_L \leq 1\%$.

The performance characteristics of the SENET schemes with the above system parameters have already been obtained, and are as shown in fig. 7.10. Since it is intended to compare the two integrated systems the comparison should be on equal grounds. As already explained, in the SENET approach it is possible that some parts of the frames, or the channel capacity, to be wasted due to the partitioning of the master frames into voice and data packet slots. It can be shown that in the SENET systems with the above system parameters there are 320 bits wasted in each frame. Thus this implies that the effective channel capacity is more like 1.512 Mb/s instead of 1.544/Mb/s. As in the FCFS this channel wastage does not take place, the channel capacity for the FCFS has been assumed to be 1.512 Mb/s .

Fig. 7.20 shows the characteristic of the percentage of voice packet loss versus the data packet arrival rate for the FCFS system with packet losses taking place only at the transmitter. The voice traffic for this characteristic has been assumed to be fully loaded. It can be seen that as the data-arrival rate increases, so the voice-packet loss increases rapidly. It can also be observed that the data-packet arrival rate is 21.9 data packets/frame when the voice packet loss is 1%. Thus this represents the maximum data-packet arrival-rate that the FCFS can handle.

The voice-traffic loading simulation model has now been incorporated into the FCFS simulation program so as to include the arrival and transmission of the voice calls into the FCFS system. The voice intensity has been set at 22 Erlangs, which gives a 2% blocking probability for the voice traffic. The characteristic of average data-packet waiting time versus data-packet arrival rate per frame has been obtained from simulations and is depicted in fig. 7.21. Also in the figure the characteristics of the SENET with SQ and NSQ packetized voice operations have been included. It can be observed that when the data arrival-rate is below, or equal to 21.9 data packets/frame, the average data-packet delay for the FCFS is nearly equal to the average data-packet delays of the SENET systems. For example when the data arrival rate is 21.9, the average delay in the FCFS is nearly 7 ms while for the SENET system with SQ-packetized voice operation it is nearly 9 ms. It should be noted that beyond the 21.9 data arrival rate the characteristic of the FCFS should be ignored. This is because fig. 7.20 indicates that when the data arrival-rate exceeds this limit, the voice-packet losses exceeds the 1% permissible value and so the speech traffic would be corrupted by the packet losses. In the SENET system the increase in the data arrival-rate does not have such an undesirable interaction with the voice packet losses. Thus it can be concluded that the SENET approach is a more flexible method for integrating voice/data than the FCFS.

An important observation which can be made from figure 7.21 concerns the SENET system, when the arrival-rate exceeds 21.9 the data-packet queueing delay rises rapidly. This means that if the data arrival-rate exceeds this limit the data-queueing delay also rises rapidly with the arrival rate, and can become very large (see also table 7.2). For example when the data arrival-rate is 22.8, the average data-queueing delay in the SQ SENET approaches 40 ms. This implies that at the expense of approximately 40-7 = 33 ms delay the data handling performance of the SQ SENET becomes approximately

$\frac{22.8 - 21.9}{21.9} \times 100 \cong 4\%$ better than the data-handling performance

of the FCFS. Thus it can be concluded from this comparison that the performance of the SENET approach for integrated voice/data in an environment which is totally packet switched is only slightly better than the performance of the FCFS. However, the FCFS is a simpler scheme in design and implementation than the SENET approach.

Table 7.2

θ_d	NSQ SENET $\bar{W}_d$ (ms)	SQ SENET $\bar{W}_d$ (ms)	FCFS $\bar{W}_d$ (ms)
21	3.73	1.70	3.08
21.3	5.35	3.07	4.16
21.6	7.78	5.40	5.55
21.9	11.36	9.02	7.37
22.2	17.12	14.86	-
22.5	26.31	24.40	-
22.8	42.28	40.98	-

7.10 Conclusions

To investigate the integration of voice and data in entirely packet-switched environments, two new multiplexing modelling schemes have been studied in this chapter. The first scheme was based on using the SENET approach, while the second scheme was based on employing the FCFS packet-switching approach. In each of these two cases, computer simulation models have been developed in order to evaluate their performance characteristics.

The SENET type of approach has been investigated with the two different voice packetization operations, namely the NSQ and the SQ-types. It was shown that the data-traffic queueing performance in the SENET with SQ-packetized operation is nearly the same as the data-traffic performance of the SENET with NSQ-packetized voice operation.

It was also shown that the data-traffic performance in this type of SENET-system cannot be approximated by an M/G/1 queueing model. The reason for this was found to be due to the correlation of the data-packet service times. A boundary flow control mechanism was studied to investigate whether the data-traffic performance could be improved. The results obtained however from the computer simulations show that little improvement can be achieved by using this type of flow control scheme.

The second new integrated technique investigated was the FCFS packet-switched system. This system employed a completely different technique to the SENET approach. The main advantages of using the FCFS technique over the SENET are:

(a) The scheme can be implemented more easily than the SENET.

(b) Unlike the SENET, in this scheme no part of the channel will be wasted. In the SENET this wastage of channel capacity occurs whenever the frame length is not a multiple integer of the data-packet size.

The main disadvantage of the FCFS scheme is that the data-traffic rate cannot be permitted to vary independently of the voice traffic. Data traffic must be constrained in order to maintain the worst-case voice-packet loss criteria below 1%. It was also shown that the performance of the SENET is slightly superior to that of the FCFS.

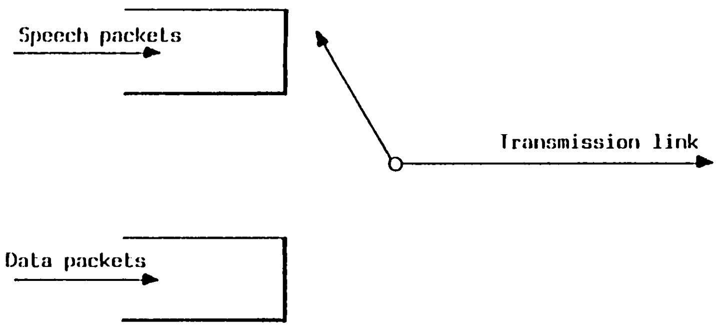
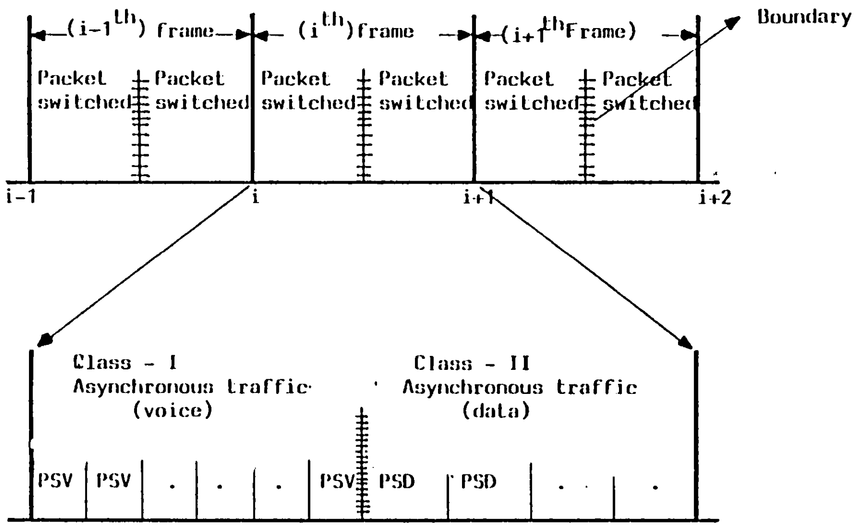


Fig. 7.1 Integrated voice/data in purely pacetized fashion by using the slotted-envelope network concept



PSV - packet switched voice slot (P_1 bits)
PSD - Packet switched data slot (P_2 bits)

Fig. 7.2 Exemplary SENET concept for integrating voice and data in purely packet-switched environment

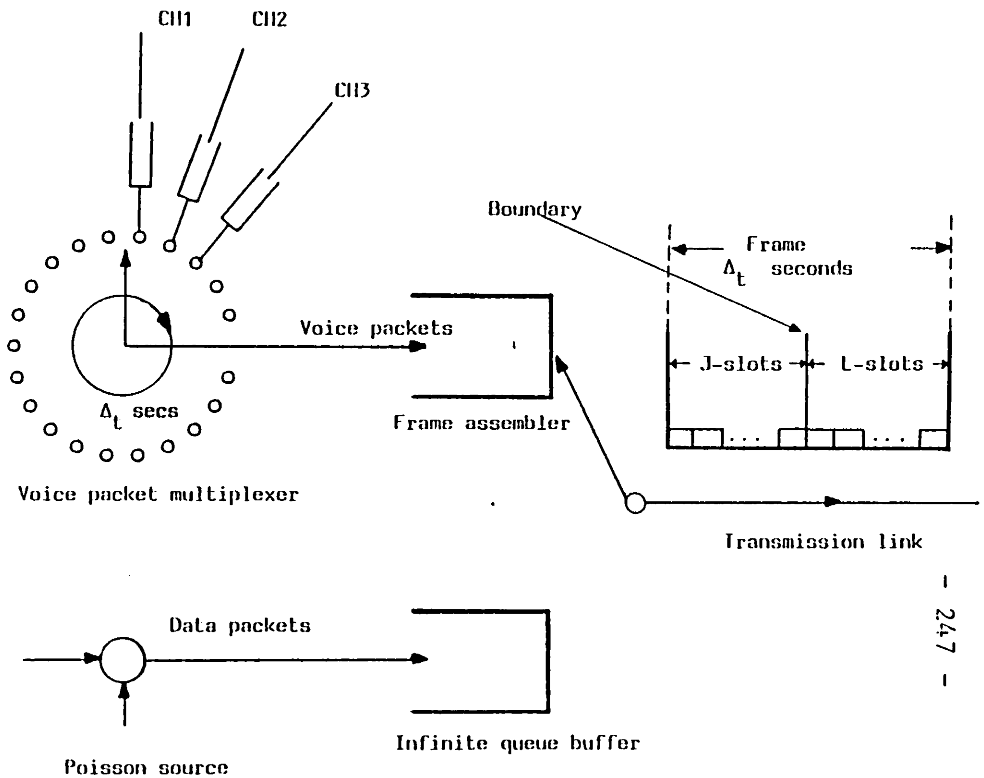


Fig. 7.3 Exemplary SENET system for intefrating voice/data by using NSQ packetized voice scheme

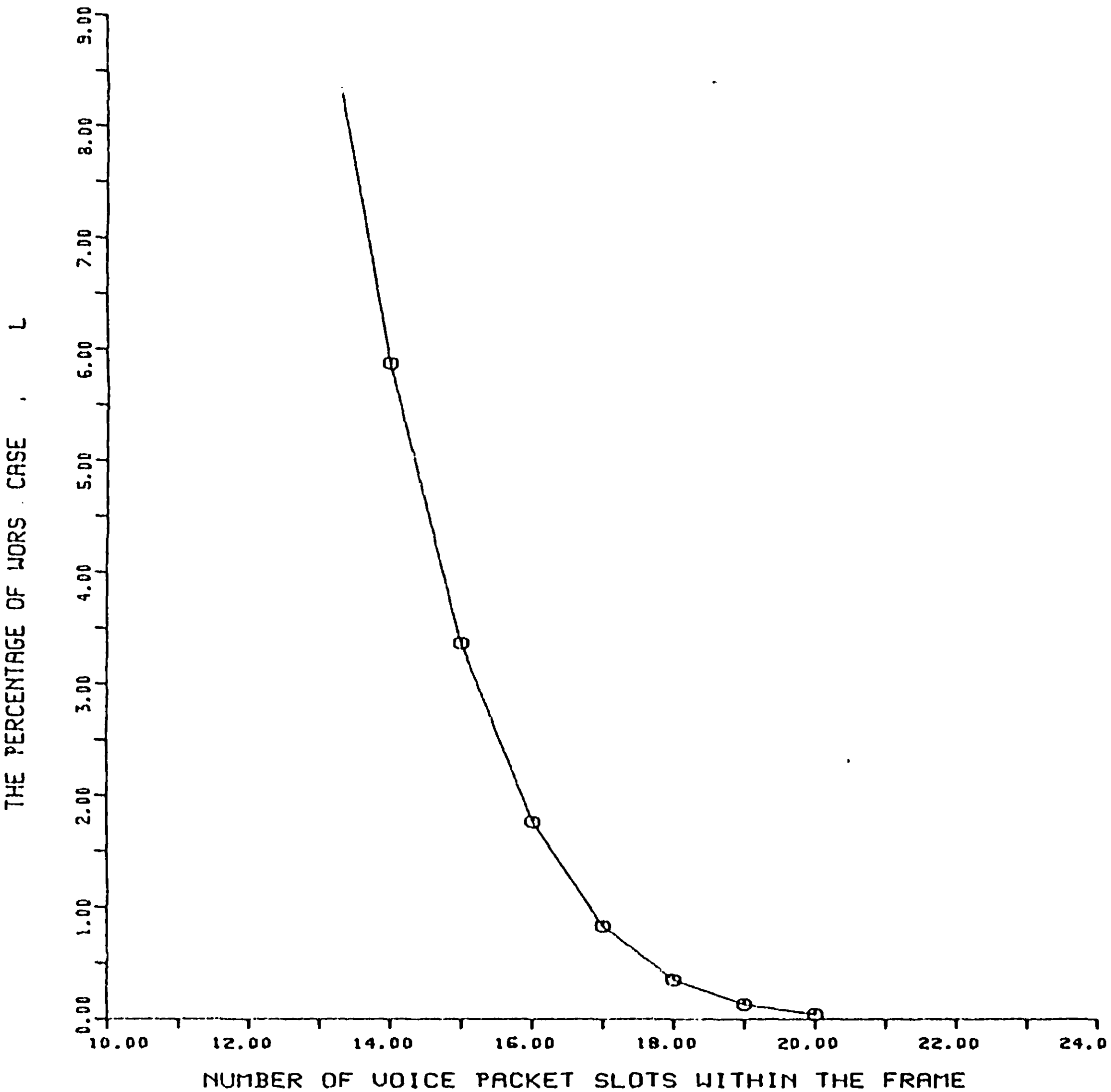


Fig. 7.4 Percentage of voice packet losses as a function of number of packet slots per frame in the NSQ-packetized voice system

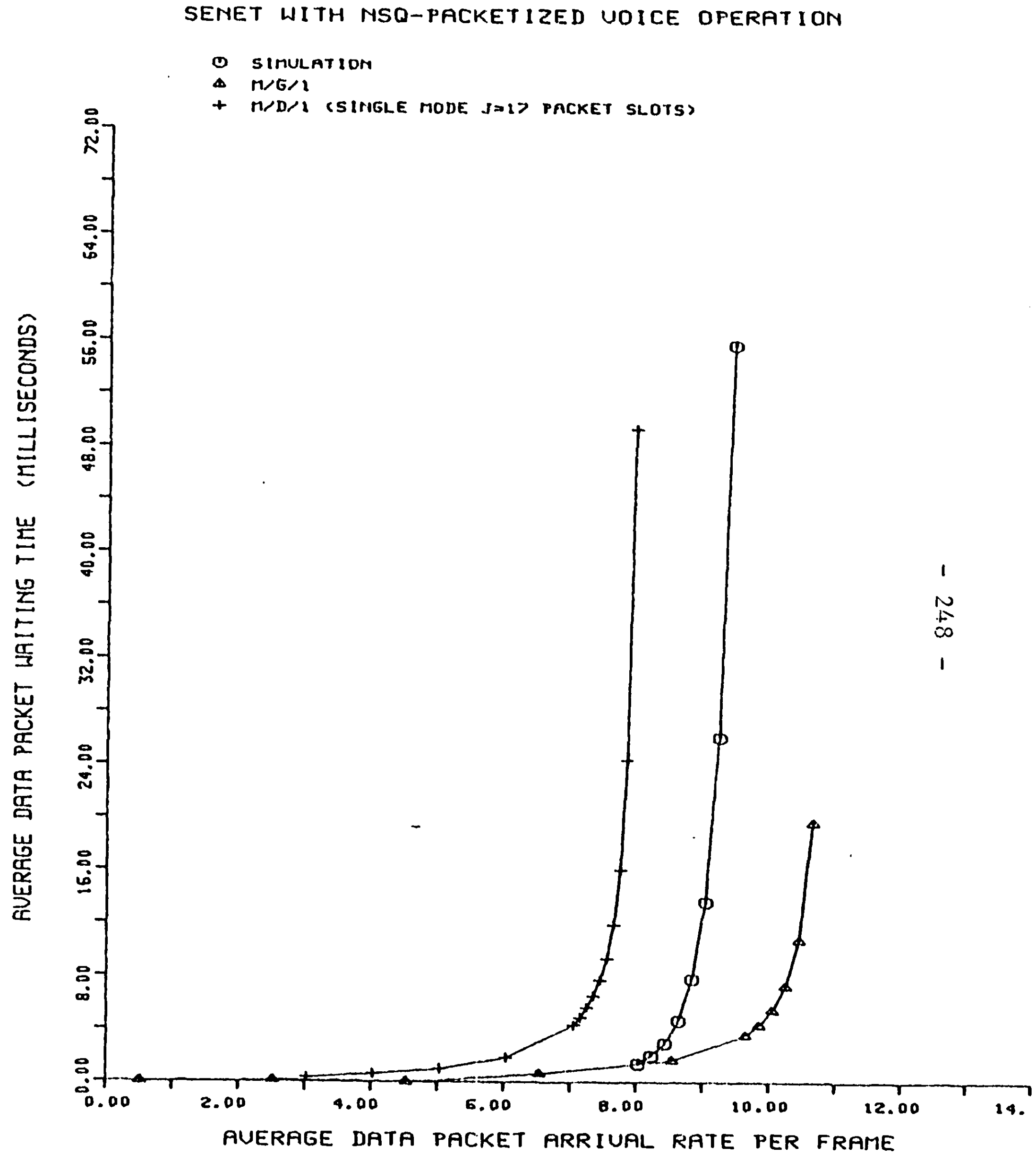


Fig. 7.5 The comparison of analytical and simulation results of data traffic performance in SENET with NSQ-packetized voice operation

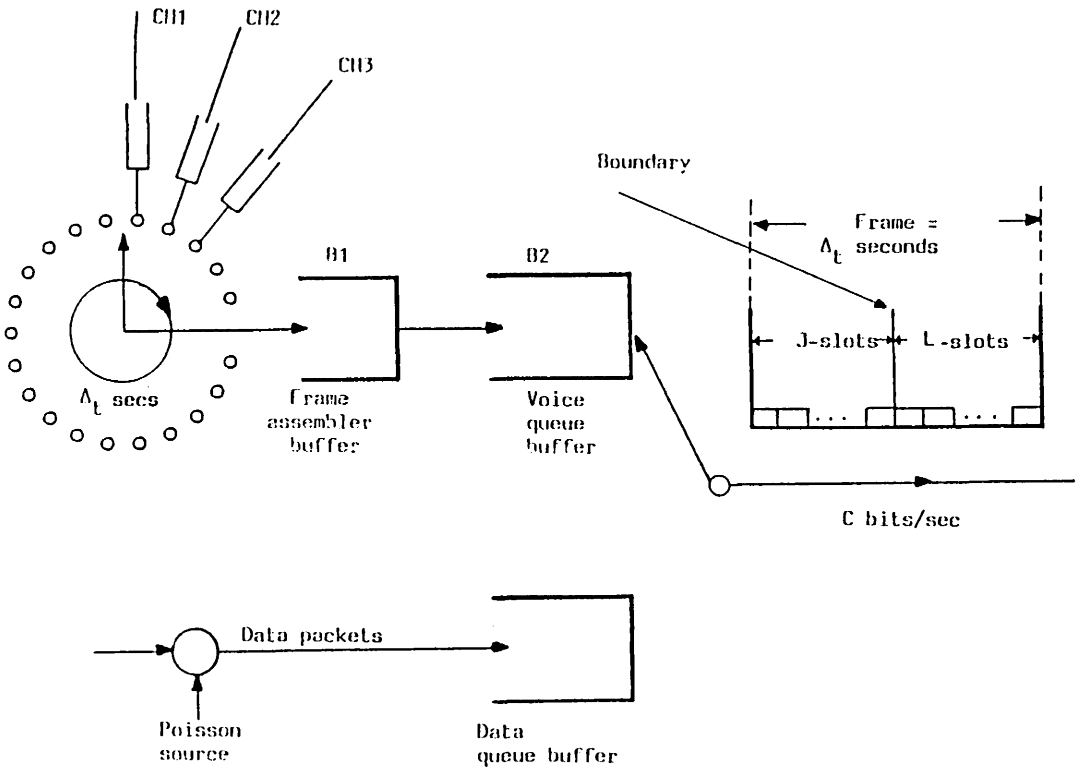


Fig. 7.6 SENET system for integrating voice/data using SQ-packetized voice operation

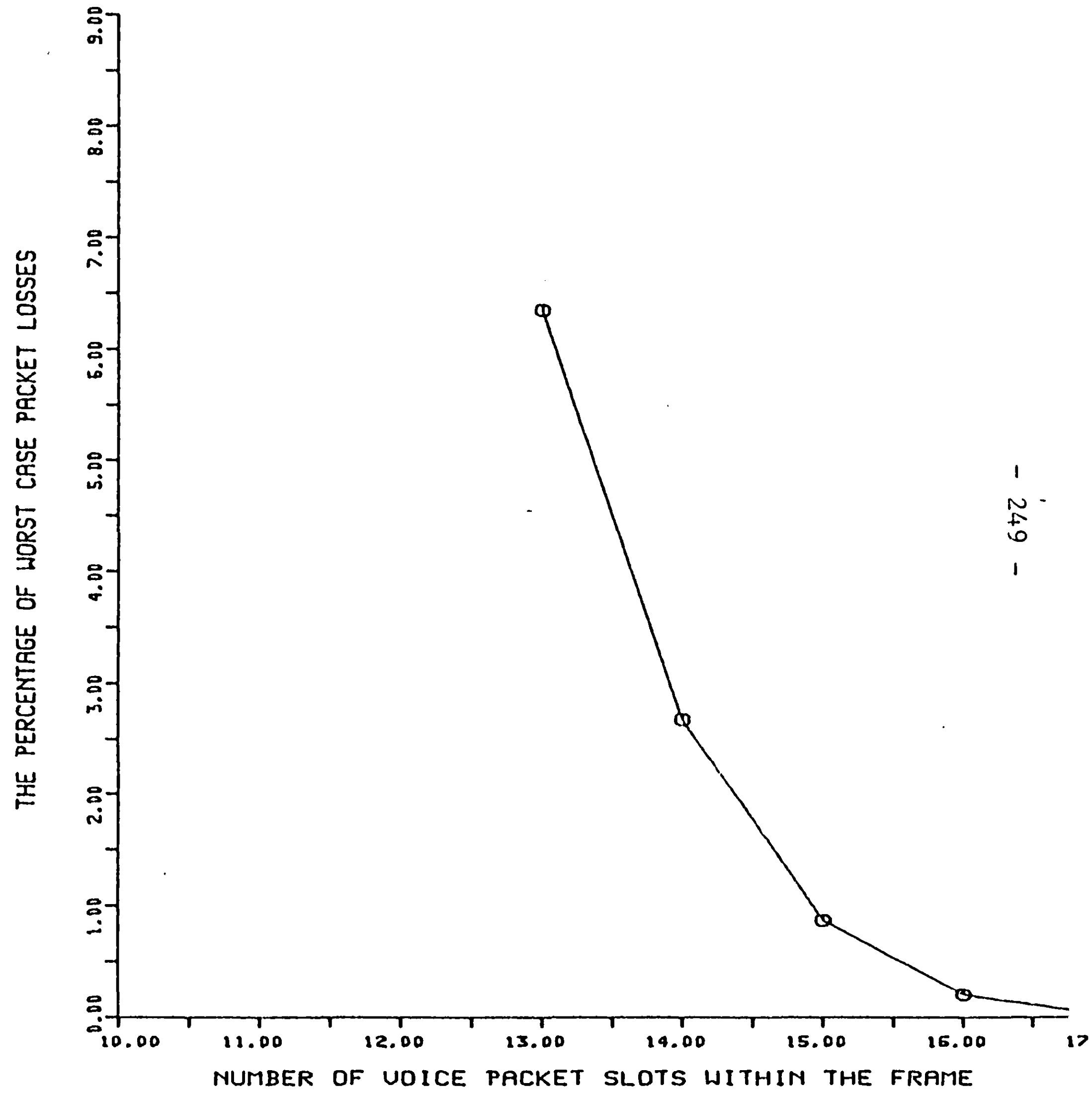


Fig. 7.7 Percentage of voice packet losses as a function of the number of packet slots per frame in the SQ-packetized voice system

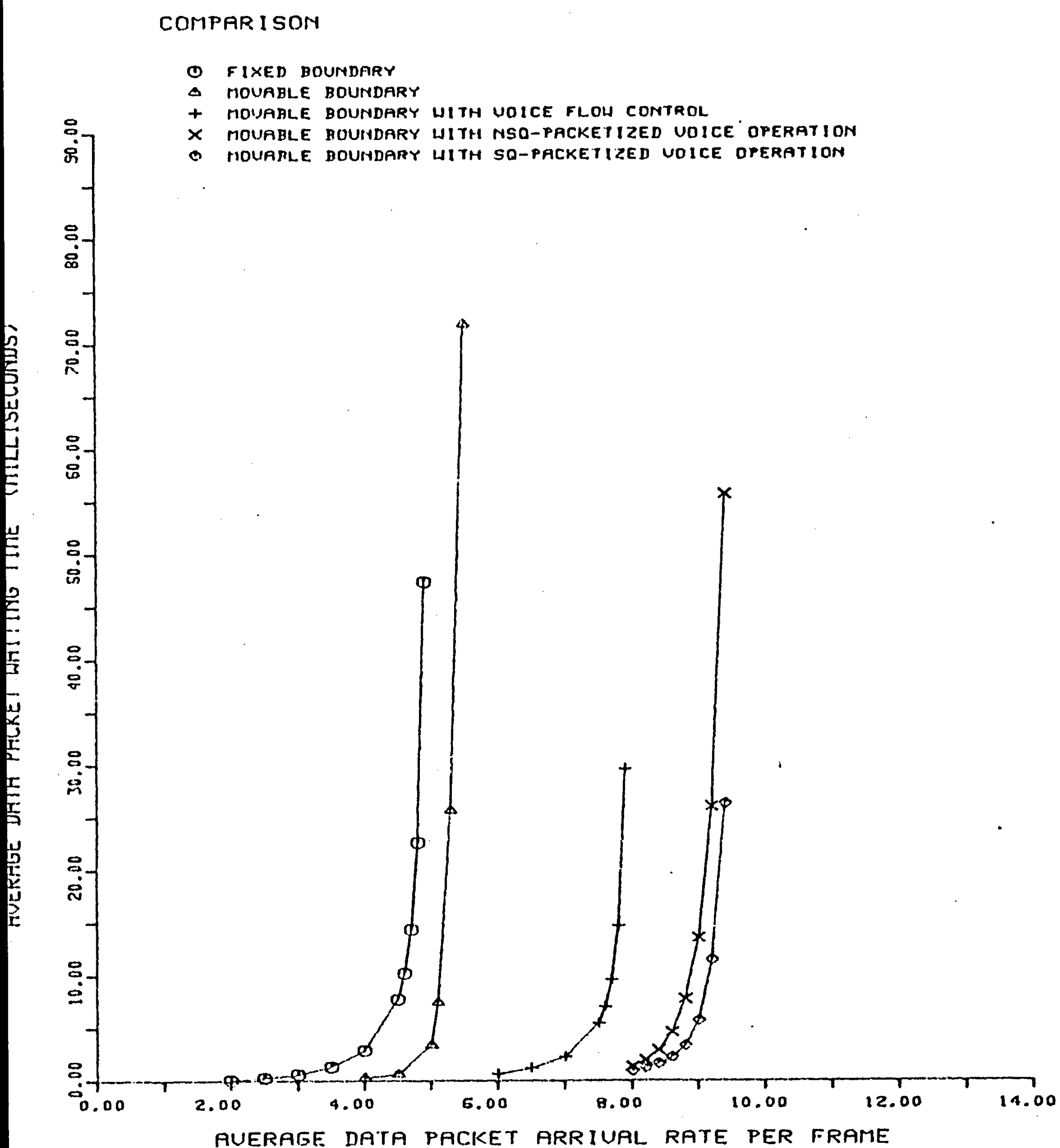


Fig. 7.8 The comparison of the data traffic performance in the various type of SENET-systems

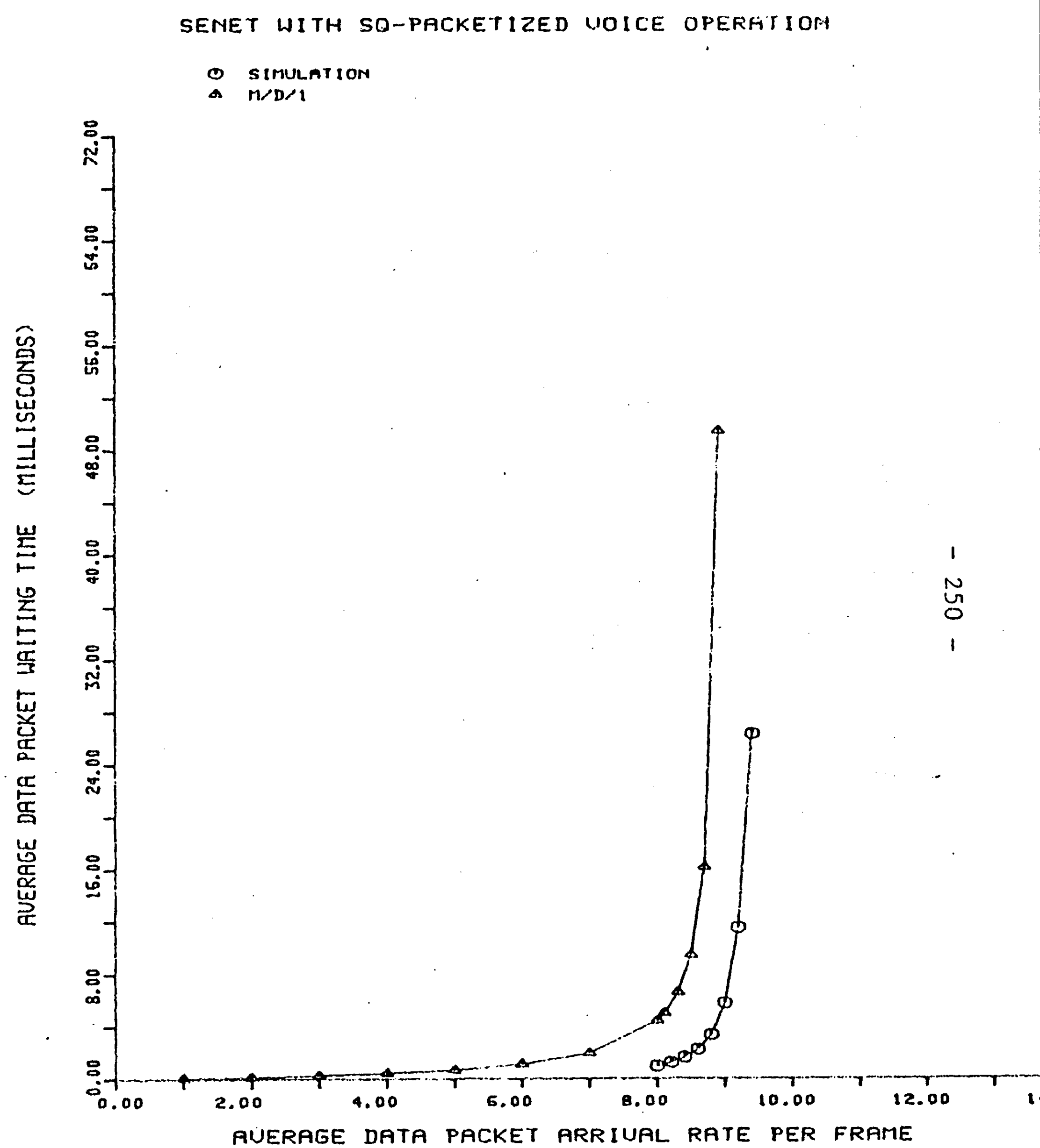


Fig. 7.9 The comparison of analytical and simulation results of data traffic performance in SENET with SQ-packetized voice operation

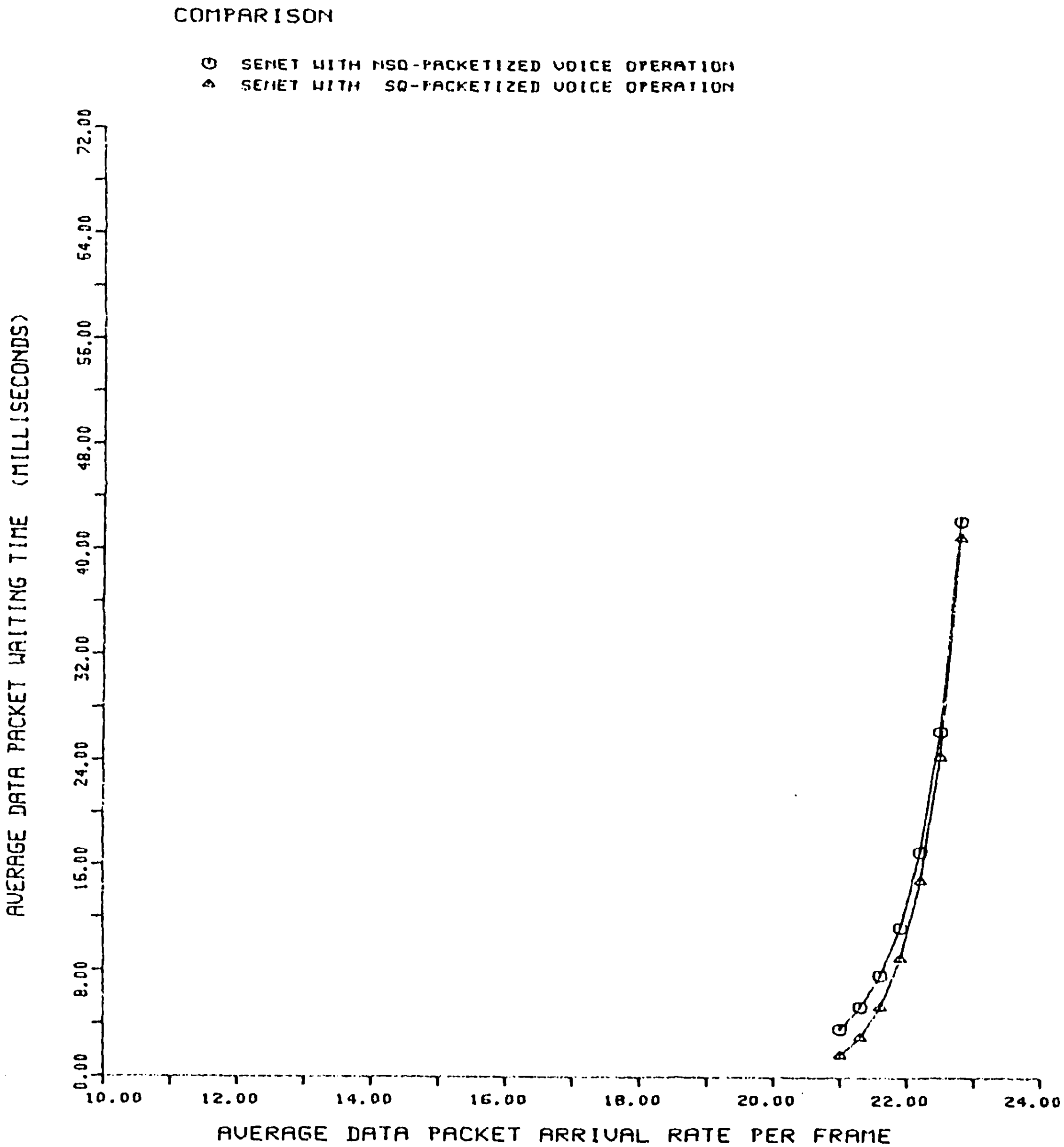


Fig. 7.10 The comparison of data traffic performance in the SENET with SQ and NSQ packetized voice operation. The voice and data packet sizes are equal.

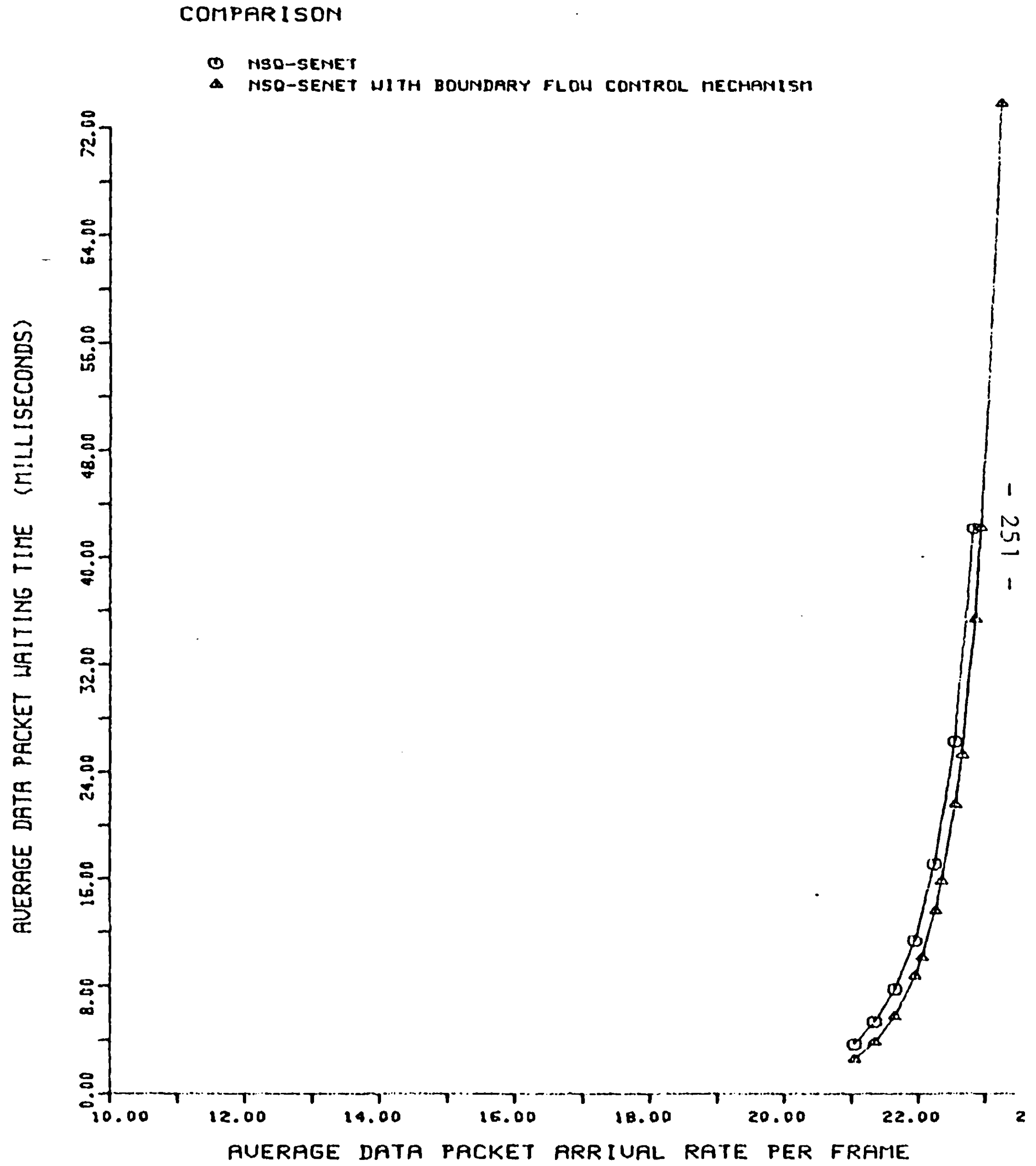


Fig. 7.11 The comparison of data traffic performance in the NSQ SENET and NSQ SENET with boundary flow control mechanism.

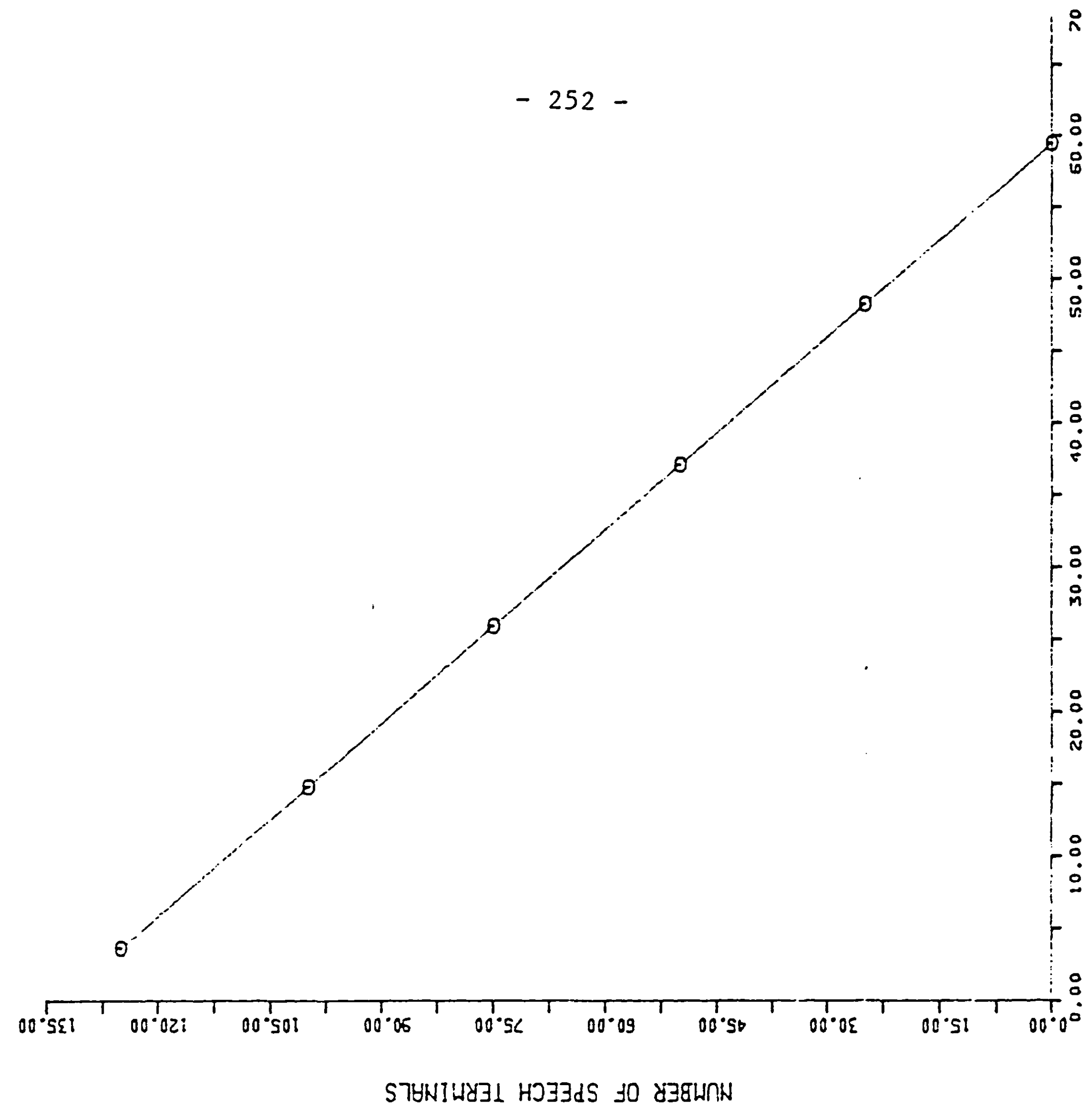


Fig. 7.13 The characteristic of number of speech terminals versus average data packet arrival per frame obtained from the M/M/1 queueing model. The encoder bit rate of each voice channel is set at 16 kb/s.

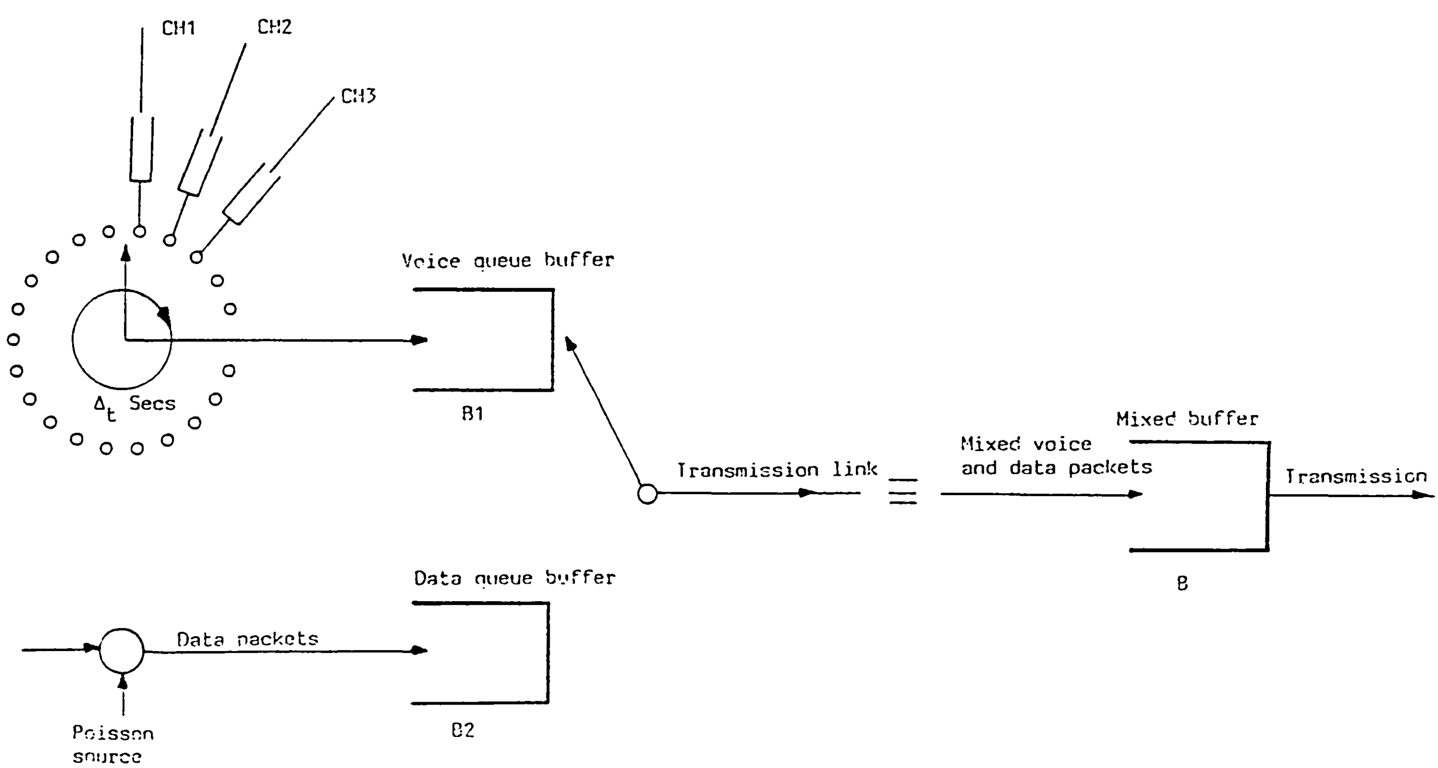


Fig. 7.12 The integrated voice/data in the FCFS purely packet-switched environment.

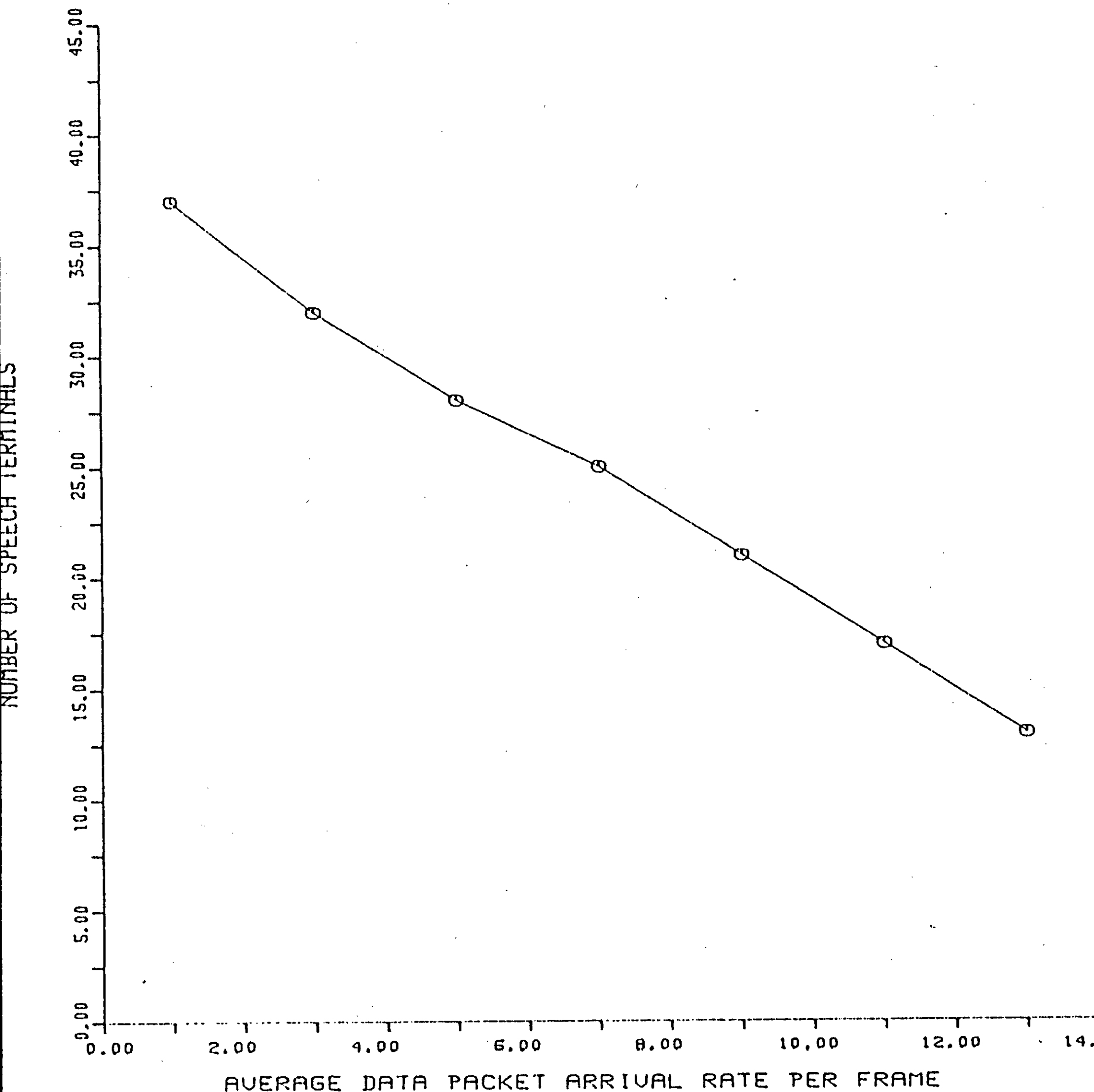


Fig. 7.14 The characteristic of number of speecg terminals versus average data packet arrival per frame obtained from the computer simulation. The encoding bit rate of each voice channel is set at 64 kb/s

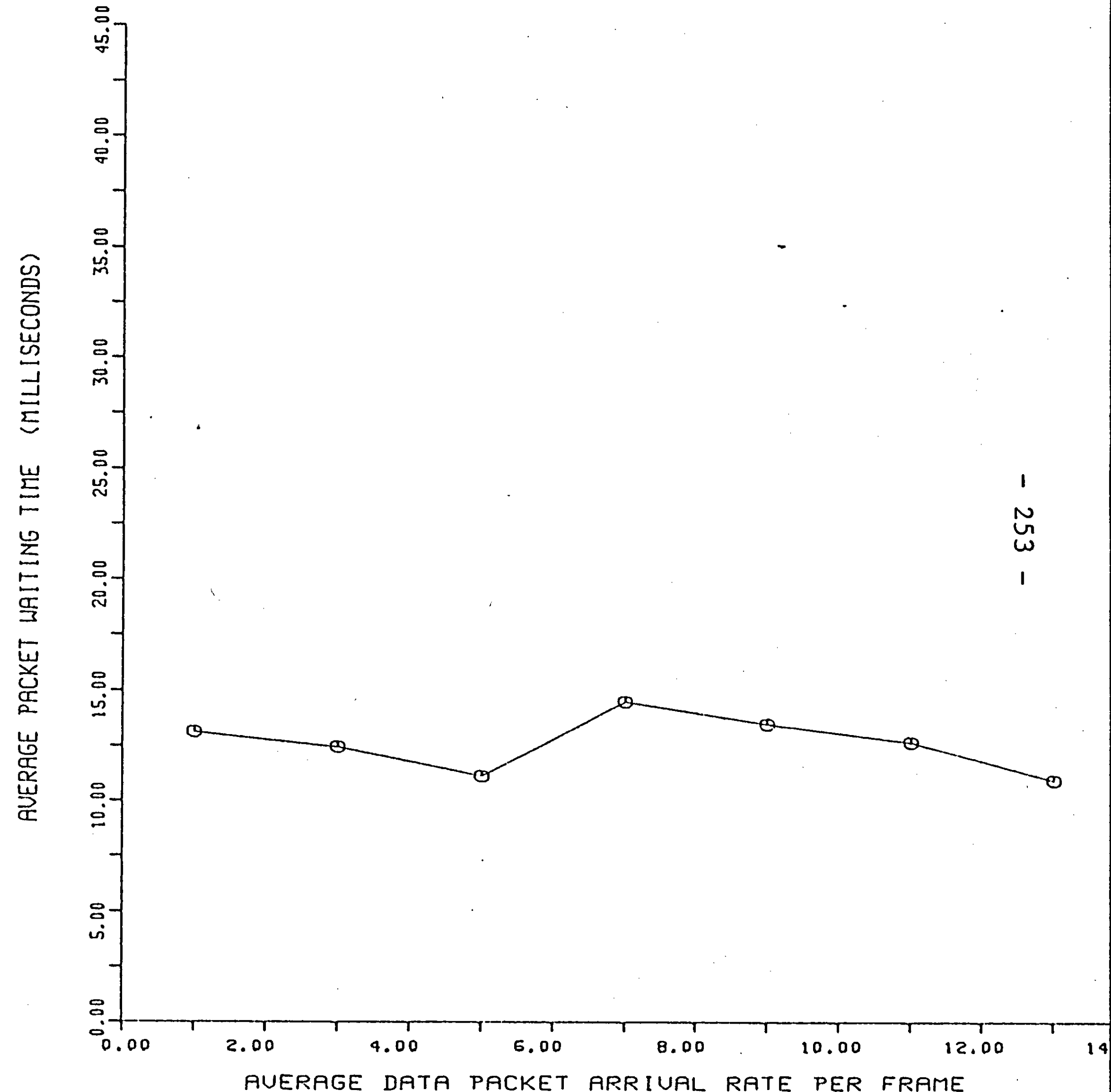


Fig. 7.15 The charqcteristic of average packet waiting time versus average data packet arrival per frame obtained from the computer simulation

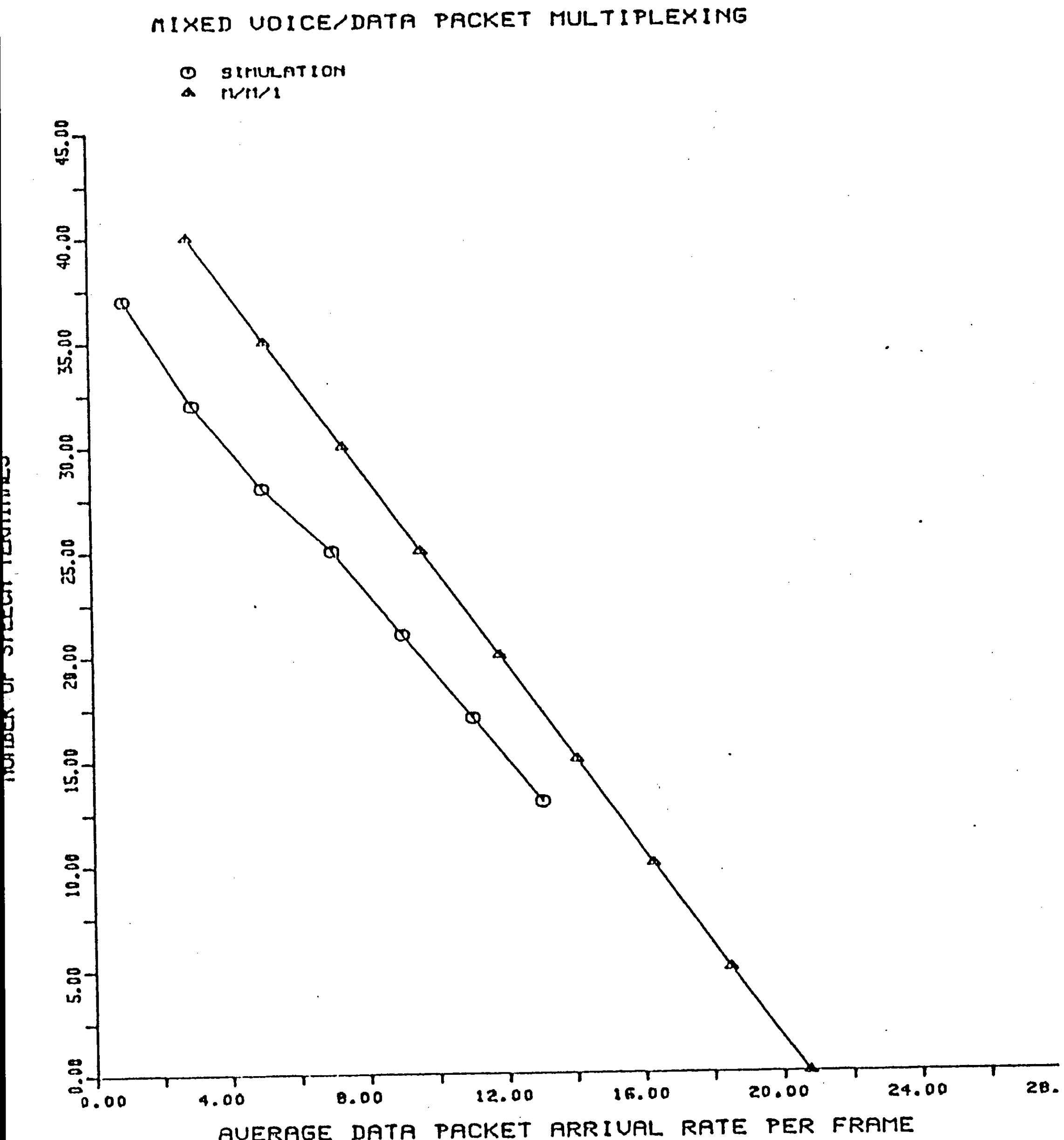


Fig. 7.16 The comparison of the characteristics of the number of speech terminals versus data packet arrival rate per frame obtained from two methods: (a) computer simulation, (b) M/M/1 queueing model. The encoding bit rate of each voice channel is set at 64 Kb/s.

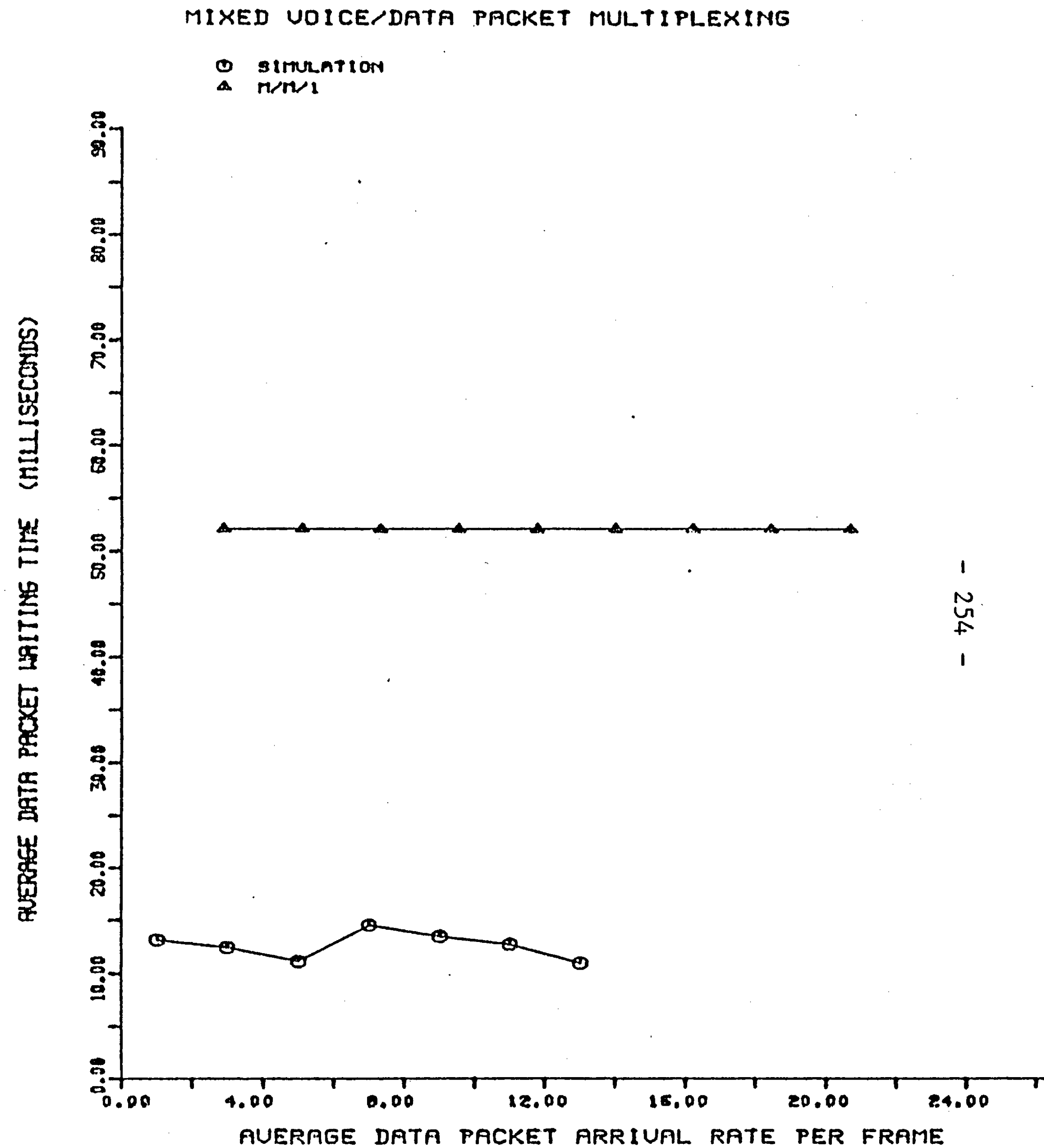


Fig. 7.17 The comparison of the characteristics of the average data packet waiting time versus average data pacet arrival rate per frame obtained from two methods: (a) computer simulation, (b) M/M/1 queueing model. The encoding bit rate of each voice channel is set at 64 kb/s

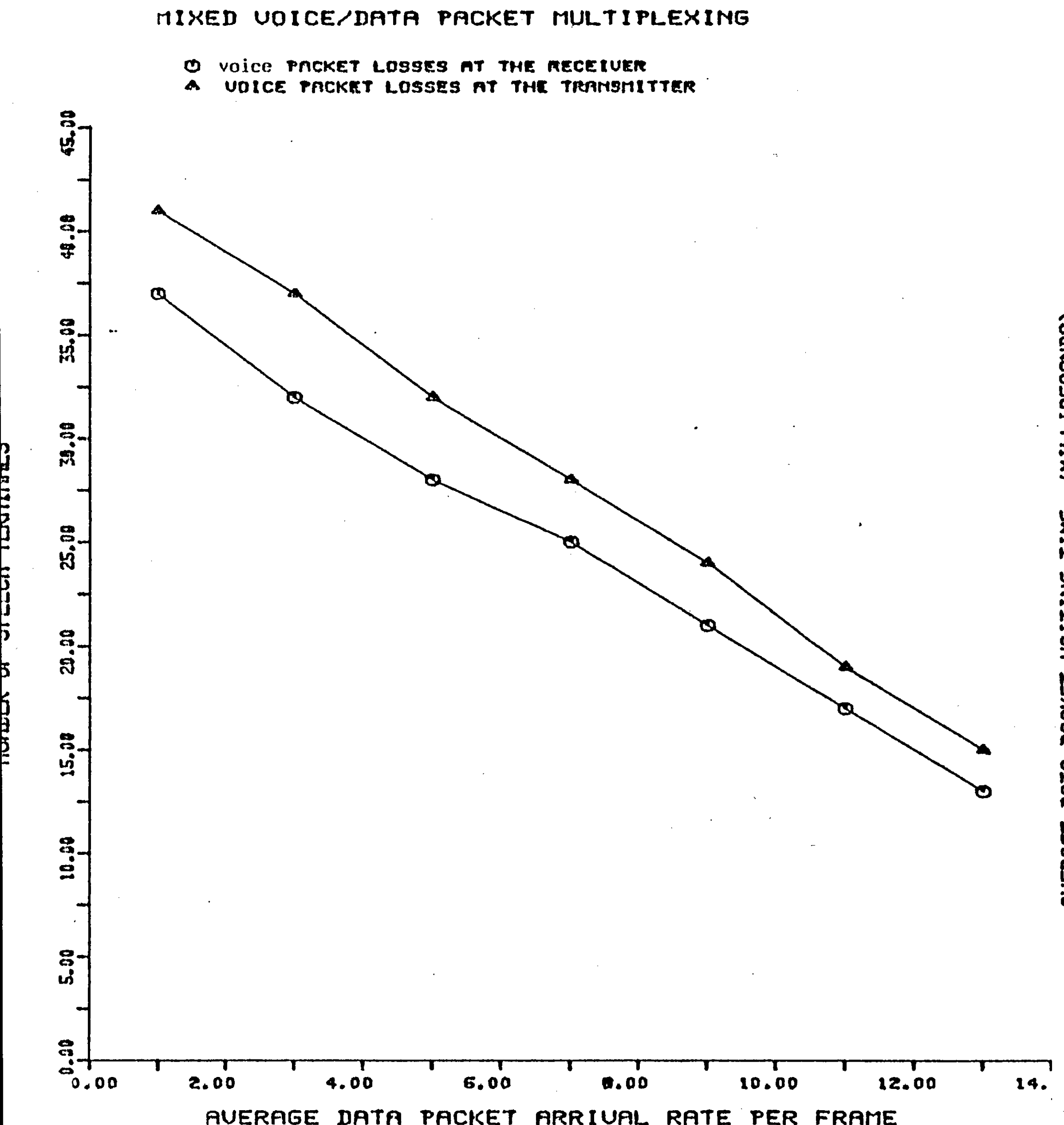


Fig. 7.18 The comparison of the characteristics of number of speech terminals vesus average data pacet arrival rate per frame for two cases: (a) voice paket losses at the recever, (b) voice packet losses at the transmitter. The encoding bit rate of each voice channel is set at 64 kb/s

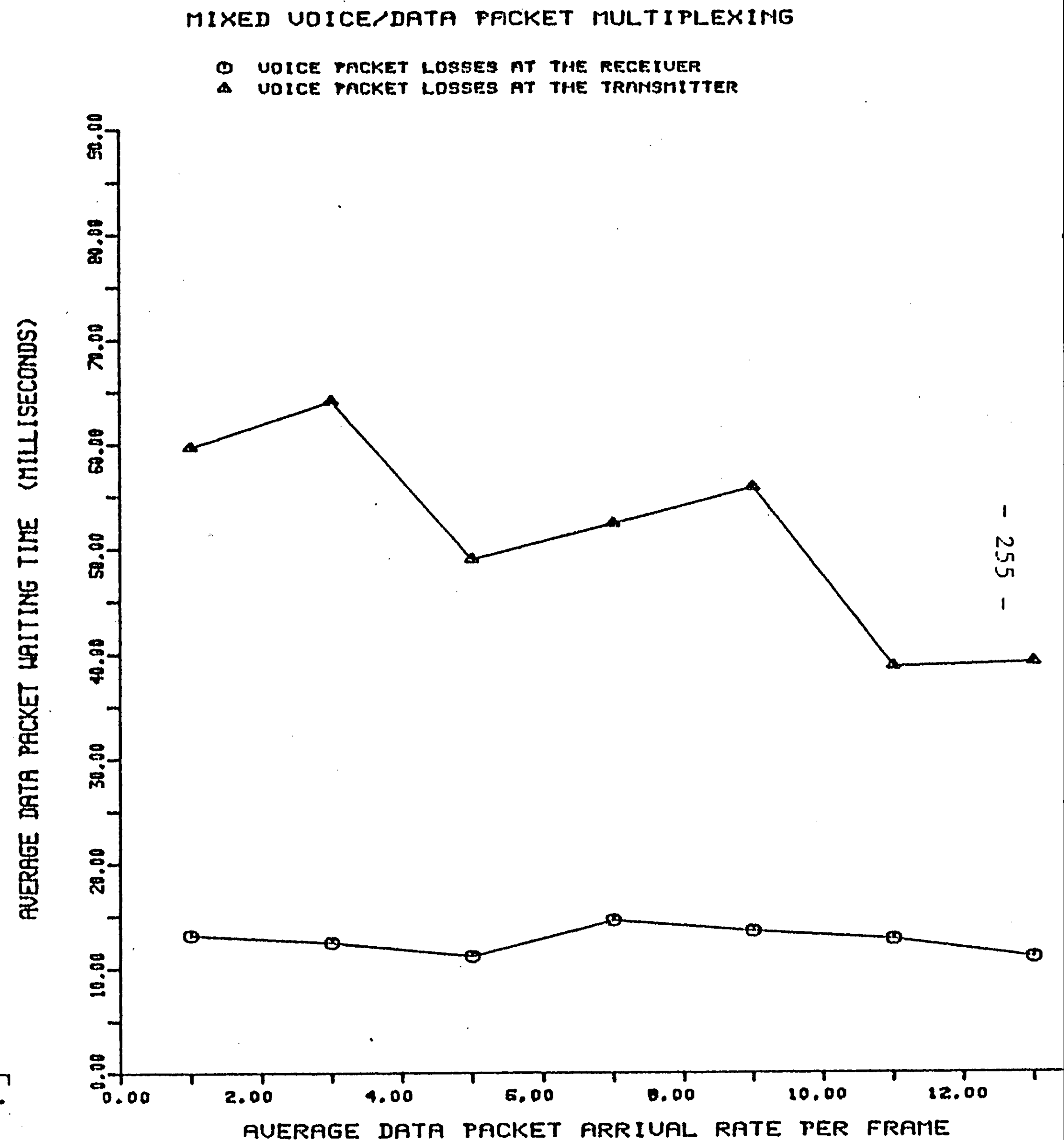


Fig. 7.19 The comparison of the characteristics of average packet waiting time versus average data packet arrival rate per frame for two cases: (a) voice pacet losses at the receiver, (b) voice packet losses at the transmitter

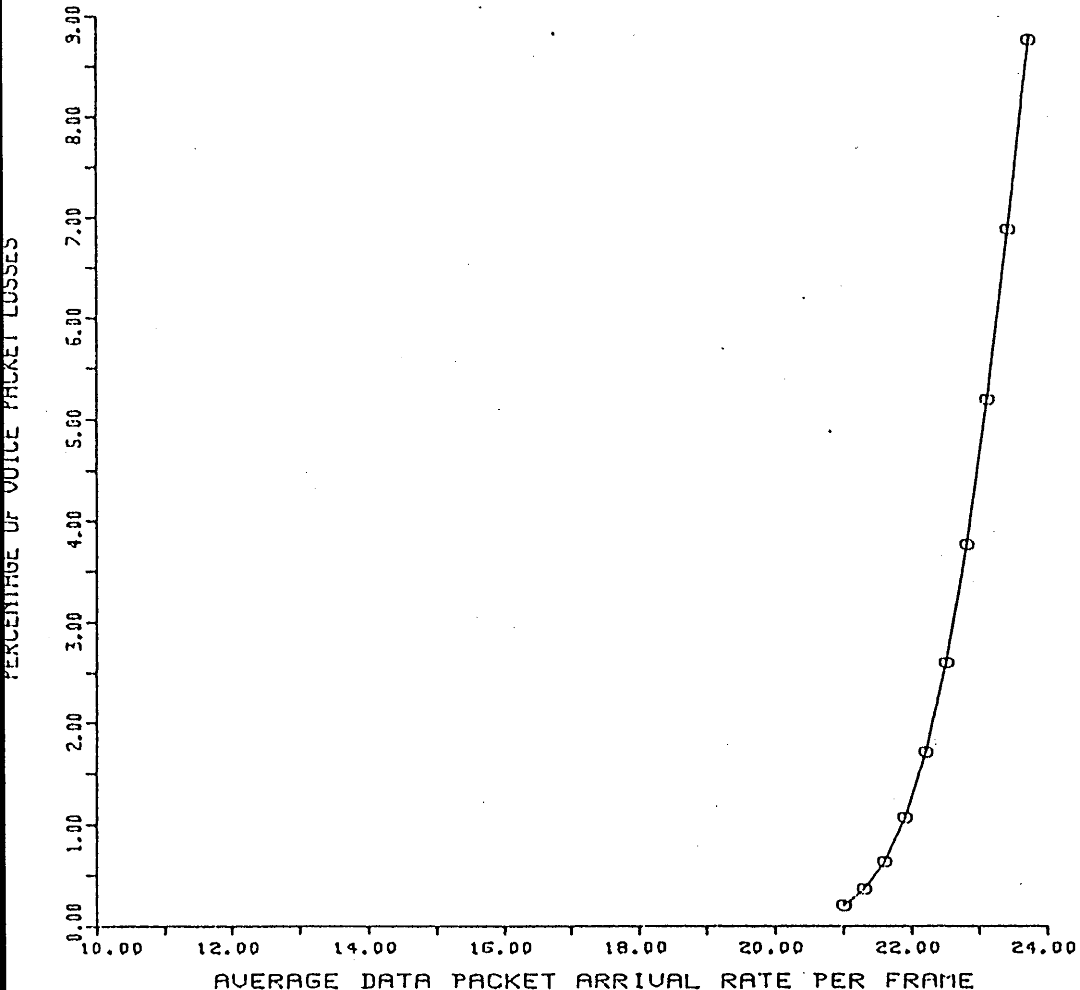


Fig. 7.20 Percentage of voice packet loss as a function of average data packet arrival rate per frame for the FCFS integrated system

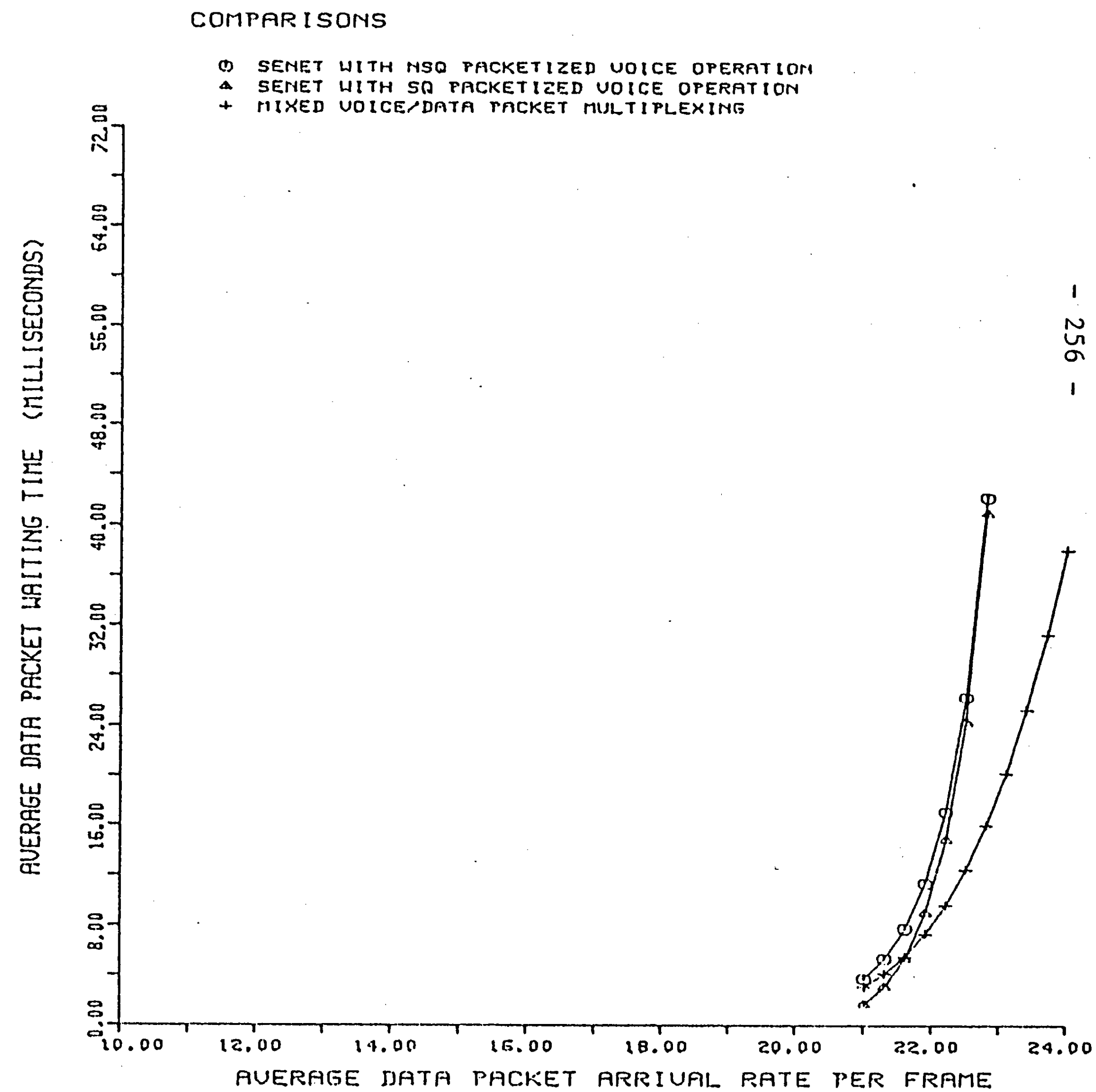


Fig. 7.21 Characteristics of the average data packet waiting time versus average data packet arrival rate performance for the two integrated systems

CHAPTER 8

SUMMARY, CONCLUSIONS AND SUGGESTIONS FOR FURTHER WORK

8.1 Introduction

In this chapter it is intended to summarize the content of the previous chapters of this thesis. The results, the conclusions drawn from the results, and the comparisons of the results concerning the work done on packet-switched voice and its application to integrated voice/data will be discussed. In addition, several topics for future investigation will be suggested.

The summary of the work carried out by the author is presented in section 8.2, with general conclusions in section 8.3. In section 8.4 suggestions for further work will be discussed.

8.2 Summary

The aims of the work reported in this thesis was to investigate two major problems. The first problem was to investigate the evaluation of the performance characteristics of a packet-voice system (PVS). The second problem was concerned with the performance evaluation of an integrated voice/data system in an environment where both traffic streams are packet-switched.

In chapter one a brief introduction was given to three types of switching techniques, namely; circuit-switching, message-switching, and packet-switching. It was stated that the conventional Time Assignment Speech Interpolation system is similar to a message-switching type of approach for speech transmission. Two reasons were given why packet-switching of the voice traffic is more advantageous than the conventional TASI systems. These were (1) packet-switching is less complex and is easier to implement than TASI, and (2) it handles the voice traffic more efficiently than the conventional TASI when considering a multinode network situation.

In chapter 2, some previous work (from the literature) on TASI, packet-switched voice, and different switching techniques for integrating voice/data were reviewed. In section 2.6 a discussion was given which highlights the topics which have not yet been investigated by other researchers in the area of packet-switched voice and integrated voice/data.

In chapter 3, the transmission operation of a packetized-voice system (PVS) was fully explained. This included a detailed study of the operations of packetization, multiplexing, queueing and packet transmission. As in this system the packets experience variable, or stochastic queueing delay, then for simplicity this type of PVS was referred to as a stochastic queueing (SQ) system. In sections 3.3 and 3.4 two simulation techniques were used to investigate the queueing performance of this system. These were (1) the Microscopic technique, and (2) the Markovian technique. The Microscopic simulation technique is based on the microscopic consideration of all stages of system operation and they are included in the simulation program. This scheme utilized a speech model which employs speech statistics as given by Brady [1] - [3].

The second simulation technique was based upon the dynamic talker activity model, suggested by Weinstein [7], this is based upon the use of a simple birth-death Markov approximation for the aggregate process and leads to a computationally efficient model. The packet generation probabilities for this model were also based on the speech statistics obtained by Brady. Results were obtained from the two simulation models and then were compared together. It was found that the two sets of results had a close agreement with each other. Thus it was concluded that either of the two simulation models could be used to evaluate the queueing performance of the SQ-system.

In section 3.6, two analytical queueing model approaches were tried for the evaluation of the queueing performance of the SQ-system. The first of these was Minoli's discrete queueing model [26] - [28]. This was based on the assumption that the inter-packet arrival is geometrically distributed (i.e. a discrete Poisson process). The second model was an M/D/1 queueing model which is based on the assumption that the interpacket arrivals are exponentially distributed (i.e. a continuous Poisson packet arrival process). Results were obtained from these two analytical models and were compared together. It was found that the two sets of results had a close agreement with each other. Thus it was concluded that the discrete packet arrival process, if assumed to be Poisson, can be approximated by a continuous process.

The results of the analytical approaches were compared with the simulation results for the condition where the total number of subscribers was equal to 40. It was found that the two sets of results do not agree with each other. It was found that significantly higher queueing delays were encountered for the simulation results. It was concluded that the reason for this lack of agreement was due to the statistical dependency of the talkspurts. It was found that this dependency causes a correlation to exist between the packet-arrival intervals and therefore causes the queue to build up in busy intervals. This blockage is not taken into account in the two analytical queueing models. It is thus concluded that the packet arrival for this situation (i.e. n = 40) does not result form of a true Poisson process. However, further tests were performed to discover when it is possible of the packet arrival to be regarded as a Poisson, or a completely a random process. To test this, it was necessary to get rid of the correlations which existed between the frames. The only possible way of doing this was to increase the number of subscribers. This was achieved by means of reducing the encoding bit rates,and the number of subscribers were increased from 40 to 140. It was found that the simulation and analytical results were then in much better agreement than in the previous case. The number of subscribers were then increased further to 250 by further dropping the bit rates. It was found that the agreement of the analytical and simulation results become even closer. Thus it was concluded that when the number of subscribers is large (i.e. of order of hundreds), then the packet-arrival process can be approximated by a Poisson process.

In chapter 4, the performance evaluations of a two node SQ-system have been investigated. The system was assumed to employ a T1-carrier link and 64 kb/s encoder outputs/channel. As the maximum number of subscribers which this system can handle would be less than 50, computer simulation techniques were used for performance evaluation. The CTI-reassembly strategy was assumed to be used for smoothing, or reassembly of packets at the receiver. The subscribers were assumed to be continuously connected, as this represents the worst-case loading condition on the system. Two assumptions were made in order to carry out the design. The first assumption was to do with the setting of a threshold for the frequency of packet losses. This threshold was assumed to be just below, or equal to 1%, this being based on the recommendations of the previous authors (i.e. Forgie [19], Musser et. al. [49], and Jayant and Christensen [48]). The second assumption was concerned with the difficulty in conversation which occurs due to delay (in a two-way conversation). According to Klemer [46], when the end-to-end delay in a two way conversation exceeds 300 ms the participants notice difficulties in performing the conversation. The author throughout this thesis has used 250 ms as an acceptable threshold for a smooth conversation.

By taking these two assumptions into account, the author presented two criteria against which to evaluate the performance of the system, and these have been reported in section 4.3. The first criteria was based on the evaluation of the characteristic of the TASI advantage versus the packet length for a constant ETE delay. Both infinite and limited transmitter queue buffering schemes were considered. It was shown that when the packet length is short (i.e. below

16 ms), then the achievable TASI advantage rises rapidly. As the packet length becomes equal to, or exceeds 16 ms, then the achievable TASI advantage of the system becomes more steady and is maximized. Thus it was concluded in this situation that the 16 ms packet length represents an optimal packet length.

The second criteria used for evaluating the performance of the system was the characteristic of ETE delay versus packet-length. This characteristic was obtained when the total number of subscribers was fixed and the percentage of packet losses set at 1%. It was shown that in this characteristic the ETE delay is minimized for a given value of packet length. This packet length therefore corresponded to an optimal packet length for this situation.

In section 4.4, further work was carried out to investigate the effect of the telephone traffic intensity upon the performance of the SQ-system. A computer simulation model was developed which determined the arrival and the termination of the voice calls. This model was combined with the Markovian simulation and five experiments were performed employing a finite buffering strategy at the transmitter.

In the first experiment the overall percentage of buffer overflow was set at below or equal to 1%, then the characteristic of the TASI advantage versus the packet length was obtained with a 2% traffic blocking probability. It was found that in this situation significantly better TASI advantages were obtainable, as compared with the previous case when all the channels were fully loaded.

In the second experiment, the transient behaviour of the effect of packet losses for experiment-one were studied. This was achieved by measuring the percentage of packet losses in equal 100 seconds intervals, so enabling the determination of how the calls were connected on average. The results obtained showed that for nearly 24% of time the calls would experience packet losses greater than 1% and therefore these would not be good connections. It was thus concluded that the 1% overall buffer overflow limit allows a significant increase in the TASI advantage, but at the cost of allowing 24% of the calls to experience packet losses greater than 1%.

In experiment-three the characteristic of the TASI advantage versus packet length with 2% blocking probability was obtained by keeping the worst-case freezeout of the system below or equal to 1%. The worst-case freezeout was obtained by taking the ratio of the total packets lost to the total packets transmitted only during the periods when all the channels were occupied. This characteristic was compared with the characteristic of the fully connected PVS. Close agreement of the two sets of results were obtained. It was therefore concluded that the fully connected PVS represents the worst-case system operation when all the channels become busy and the fluctuation of the number of voice calls does not alter the queueing performance of the worst-case system operation.

In the fourth experiment the threshold of the good call connections were set to several given values, the characteristics of the TASI advantage versus packet length were then obtained. The results showed that as the requirement for a greater number good call connections increases, so the achievable TASI advantage of the system reduces and it approaches the TASI advantage of the fully connected PVS.

The conclusion which can be drawn from the four experiments is that the fluctuations of the number of voice calls can only be used to increase the TASI advantage of the system if one allows some of the calls during the peaks of the traffic connection loading to experience packet losses greater than 1%. As a result, in experiment-five the author presents a new criteria which causes the TASI advantage of the SQ-system to be dependent upon the blocking probability of the voice calls. This criteria is established so that 95% of calls were considered to be good-call connections taken for intervals of 100 seconds duration. This criteria means that at least 95% of time the calls experience packet losses less than or equal to 1%. In this situation it was shown that if the blocking probability is below 5%, then the TASI advantage of the system becomes dependent upon the blocking probability.

In chapter 5, a pure packet loss PVS was investigated. As the packets in the system do not experience variable or stochastic queueing delay, then for simplicity this system is referred to as a "non-stochastic queueing (NSQ) system". In section 5.2, the operation of this system was explained. In section 5.3 mathematical models were developed for this system and were used to evaluate the percentage of packet losses and the ETE delay. The mathematical model used to evaluate the percentage of packet losses was based on Weinstein's formula as given in Ref [7]. The validity of this mathematical model was justified by comparing results with computer simulation results.

The results obtained from the computer simulation tallied closely with the results obtained from the mathematical model. The ETE delay for this system was shown to be constant and to be equal to three times the packetization delay.

In section 5.4, the TASI advantage characteristic of this NSQ system was compared with the TASI advantage characteristics of the SQ-system with and without having an infinite transmitter buffer. It was found that the NSQ-system had a better TASI advantage performance than SQ-system with infinite queue buffer at the transmitter. It had a lower TASI advantage performance than the SQ-system with limited queue buffer at the transmitter.

In section 5.5 two strategies were presented which can be used to improve the TASI advantage of this system. The first one was based on using the odd-even sample-interpolation technique [48]. Using this scheme, the percentage of lost packets could be increased by up to 5%. Employing this figure results in an increase in the TASI advantage for the system without degradation of the speech quality.

The second strategy used an adaptive voice flow control technique. In this scheme each speech source was assumed to employ a set of coders, each with different encoding rates (or one encoder with different output rates). The sources assumed to operate normally at the highest available encoding rate. However, when congestion starts to occur the coders are switched from the higher bit rates to their lower bit rates in order that no packets are lost. Analytical models were developed for the evaluation of the obtainable TASI advantage and the steady-state operational probabilities of the encoders.

In section 5.6 an integrated voice/data model was investigated in which the voice traffic was handled using the NSQ-packetized technique. It was assumed that the voice calls were continuously connected and active and therefore there were no statistical fluctuations in the number of voice calls in the system. Priority for packet transmission was given to the voice-packet traffic. Computer simulation techniques were used and the performance characteristic of the data traffic was obtained in terms of the average data-packet waiting time versus average data-packet arrival-rate. An M/G/1 queueing model was also used to evaluate the performance of the data traffic analytically. By comparing the results obtained from the M/G/1 model with the results of the computer simulation, it was found that the M/G/1 model is not an appropriate tool to determine the performance of the data traffic. The reason for this was found to be due to the existence of the time dependence of the frame-to-frame conditions which affect the data-packet service times. This dependency has not been incorporated in the M/G/1 queueing model. The dependency effect causes the data in the real life traffic to experience much higher queueing delays than are experienced in the M/G/1 model and in the modified simulation model.

In section 5.7 a multinode network which handles the voice traffic by using the NSQ-packetized operation was presented. The operation of this scheme is based on sending and receiving fixed frames of duration equal to the packetization period. A demultiplexing technique, as used for the NSQ-system and described in section 5.2, was used between the nodes in order to maintain the synchronization and to allow for the easy operation of the network. Mathematical formulas were derived to evaluate the performance of the network in terms of percentage of packets lost and the ETE delay. The Weinstein formula [7] was extended in order to develop a new formula for evaluating the percentage of packets lost in this network. The characteristic of TASI advantage versus packet length for the network was obtained with respect to the maximum number of nodes which a call can transverse in the system. It was shown that the network can handle up to 7 nodes and still a reasonable TASI advantage can be achieved.

In section 5.8, the effect of telephone traffic loading on the NSQ-system was investigated. The criteria which was developed in the fifth experiment of chapter 4 was used to investigate the effect of the blocking probability upon the achievable TASI advantage performance of the NSQ-system. The scheme was evaluated for when at most 5% of the calls were allowed to experience losses greater than 1%. The Weinstein formula [7] was extended and also a new formula was derived to evaluate this situation. The results showed that if the blocking probability of the system is kept below 5%, then the TASI advantage of the system becomes related to the blocking probability.

New formulas were also derived in order to evaluate the effect of blocking probability upon the steady state operational probabilities of the encoders in the NSQ-system when using adaptive voice flow control techniques.

In chapter 6, integration of voice/data by using the circuit/packet SENET approach was investigated. In section 6.2, the operation of the SENET scheme with two boundary frame-management schemes (i.e. fixed and movable) were explained. The performance parameters for this integrated system were defined as the blocking probability for the voice traffic and the average data-packet waiting time for the data traffic. In section 6.3 computer simulation models were presented and used to evaluate these performance parameters. In section 6.4, results obtained from the voice-traffic computer simulation runs showed that the voice-traffic performance parameters can be approximated using the Erlang loss-formula. In section 6.5, the data-traffic performance parameter of the SENET with fixed boundary frame management was obtained from the computer simulations. It was shown that the data-traffic performance in this case can be approximated by an M/D/1 queueing model.

In section 6.6, the data-traffic performance in the SENET with movable boundary frame management was modelled analytically by an M/G/1 queueing model. However, the comparisons of the results obtained from the computer simulations and the M/G/1 model did not agree. The reason was found to be due to the existence of time dependence frame-to-frame correlations which adversely effect the data- packet service times. This dependency effect causes significant degradation in the data traffic performance, which the M/G/1 model cannot predict.

In section 6.7, an adaptive voice flow control technique was applied to the movable-boundary SENET system in an attempt to improve the data traffic performance at the cost of degrading slightly the speech quality. The voice flow control mechanism applied reduced the encoding bit-rates of some of the subscribers during peaks of the voice-channel occupancies. The net effect of using this technique was to reduce the size of the reserved channel capacity to the voice traffic per frame, or to shift the boundary to the voice-traffic region within the frame. The technique was applied to the SENET system and was simulated for the situation where the new boundary was set at the average statistical voice-channel occupancy. The results showed a significant improvement in the data-traffic performance. It was also shown that the data-traffic performance in this case can be approximated with an M/D/1 queueing model. The subjective effect upon the quality of the speech has not been investigated but the percentage of time the encoders had to be working in their different modes was investigated and is given in section 6.7.

In chapter 7 the integration of voice/data in the environment where both traffic streams are packet-switched was investigated. Two new multiplexing schemes were introduced to perform this type of integration. The first scheme employs the concept of SENET multiplexing, and the second scheme employs a simple FCFS queueing approach.

The SENET multiplexing approach was explained in section 7.2. It was shown that this scheme employs a boundary to separate the allocation of the channel capacity to each of the traffic streams. A criteria was established to evaluate the nominal position of the boundary within the frame. This was set at such a position as to reserve a sufficient number of packet slots for the voice traffic such as to meet the 1% packet losses at the full loading of the voice traffic. The boundary was assumed to be totally dynamic. This means that the both traffic streams were able to utilize their free packet slots. This form of multiplexing was investigated with the two packetized voice systems (i.e. NSQ and SQ).

In section 7.3, the SENET with NSQ-packetized voice operation was investigated. The position of the nominal boundary within the frame was evaluated by using the Weinstein formula [7]. A computer simulation model was then presented and the data-traffic performance characteristic was evaluated. An M/G/1 queueing model was also formulated to obtain the data traffic performance characteristic analytically. The results obtained from these two models were compared together. There did not appear to be a close agreement between the two sets of results. Thus it was concluded that here the same effect of the time dependence correlation among the frames existed as in the circuit/packet SENET.

In section 7.4, the SENET with SQ-packetized voice operation was investigated. In this system the nominal position of the boundary was found by using a computer simulation. It was found that in this scheme less of the channel capacity was needed to handle the voice traffic. It was then expected that the data-traffic performance in this scheme would be better than the data-traffic performance in the NSQ-SENET. The data-traffic performance characteristic was obtained from a computer simulation and the results were compared with the results of the NSQ-SENET. The comparison of the results, however, showed a small improvement in the data-traffic performance in the SQ SENET over that obtained for the NSQ-SENET. It was therefore concluded that the data-packet service time in the SQ SENET is more correlated than in the NSQ-SENET. The reason of the greater time dependence correlation in the SQ-SENET is due to the existence of the voice-queue buffer employed in the SQ-SENET.

The work, as presented so far, was based on setting the nominal boundary within the frame so that the 1% level of voice packet losses would be met for the condition when all the voice channels were connected. In section 7.5 a boundary flow-control mechanism was presented which was based on setting the boundary according to the number of call connections. In each case the resultant voice-packet losses were set to be just below, or equal to 1%. This flow-control mechanism was applied to the SENET with NSQ packetized voice operation. The computer simulation results obtained for the scheme, however, showed that little improvement was achieved in the data-traffic performance.

In section 7.6 the performance characteristic of the integrated voice/data by using the FCFS multiplexing approach were investigated. The performance characteristics in this integrated model were defined as 'the number of speech terminals versus average data packet arrival rate', and 'the average data packet waiting time versus average data packet arrival rate'. In section 7.7 some operating assumptions were made and these characteristics were obtained from an M/M/1 queueing model. It was shown that some of the assumptions in this model are unrealistic. As a result, in section 7.8 a computer simulation model was presented to obtain the true performance characteristic of this integrated model. Two cases were considered. In the first case, the late voice packets were assumed to be discard at the receiver, and in the second case the late voice packets were assumed to be discarded at the transmitter. The results obtained from the computer simulation runs showed that when the voice packets were discarded at the transmitter; the FCFS integrated model had a better performance characteristic for the number of permitted speech terminals versus the average data-packet arrival-rate. However it had a worse performance when considering the average data-packet delay.

In section 7.9 the performance of the two integrated systems (i.e. SENET and FCFS) were compared together. The results showed that the performance of the SENET in terms of data handling can be slightly better than for the FCFS.

8.3 Conclusions

In this thesis two main problems have been investigated. The first problem was the performance evaluation of the PVS, this work is reported in Chapters 3 to 5. The second problem was the assessment of the performance capabilities of the PVS when incorporated into integrated voice/data networks, this work is reported in chapters 6 to 7. The aim here is to present the most significant conclusions of the work which has been carried out by the author in chapters 3 to 7.

From the work of chapter 3, the main conclusion which can be drawn is that the packets which arrive from a small number of talkers are not uncorrelated from frame-to-frame. Thus computer simulation techniques have to be used for studying the performance evaluation of the PVS. However, it was concluded that if the number of subscribers is large (i.e. of order of hundreds), then the packet arrival process can be approximated by a Poisson process and hence the effect of packet correlation will be minimal.

In chapter 4, two main conclusions have been drawn, these are explained as follows:

(1) It was shown that the performance of a two node PVS which employs a CTI-reassembly strategy at the receiver can be evaluated using two characteristics. These characteristics are (a) the permissible TASI advantage versus packet length, and (b) the ETE delay versus packet length. The optimal packet length can be determined by consideration of both these characteristics.

(2) It was shown that taking into account the effect of telephone traffic an improvement in TASI advantage of the PVS, can be permitted. This is provided some of the call connections during the busy periods are allowed to experience packet losses greater than 1%. When employing this criteria, the permitted TASI advantage of PVS will be related to the blocking probability.

Weinstein originally proposed and analysed the NSQ-system; however in chapter 5 the author has carried out the following original work on this system.

(a) Two packet organization techniques have been applied to this system in order to increase the TASI advantage. These were (i) the odd-even sample-interpolation technique, and (ii) the adaptive voice flow control technique. These investigations were carried out using novel analytical approaches.

(b) A demultiplexing scheme was proposed for this system in order to enable it to be incorporated into a multinode network PVS. Mathematical formulas were used to evaluate the performance characteristics of the network.

(c) New formulas were developed to relate the performance characteristic of a two node NSQ-system to the voice blocking probability. This was based on the assumption that some of the calls could experience packet losses greater than 1% during the busy period.

In chapter 6, the circuit/packet SENET-system was investigated and two conclusions may be drawn. These are as follows:

(1) It was shown that with the movable boundary SENET, the service times of the data packets are correlated from frame-to-frame. It was shown that this correlation causes a significant degradation in the data traffic performance.

(2) It was shown that a proposed adaptive voice flow control technique can considerably improve the data performance in the movable boundary SENET, but at the cost of dropping the speech quality. It was also shown that with this scheme the data traffic performance may be obtained by using a simple M/D/1 queueing model.

The main conclusions drawn from the work reported in chapter 7 are as follows:

(1) The performance of the data traffic in the totally integrated packet-switched SENET system is considerably better than its performance in the circuit/packet system.

(2) It was shown that the service time of data packets are still time correlated from fame-to-frame in this totally integrated packet-switched SENET.

(3) The performance of the data traffic in the SENET with SQ packetized voice operation is only marginally better than the performance of the data traffic in the SENET with NSQ packetized voice operation.

As the NSQ SENET system is (i) less costly to implement, (ii) is easier to implement and (iii) has a better ETE delay performance than the SQ SENET system; then the NSQ-SENET would be the better scheme to employ.

(4) It was shown that the proposed boundary flow control mechanism did not significantly improve the performance of the data traffic in the integrated SENET with totally integrated packetized operation.

(5) The FCFS multiplexing approach for integrating voice/data in the totally integrated packet-switched environment has been investigated. This scheme is a less costly method and is easier to implement than the SENET systems. However, it was

shown that the SENET system can have a slightly better performance than the FCFS system.

A major problem with the FCFS system concerns the form of the data. The assessment has been carried out using average data loading, but in practice this can be very variable and could change its characteristics from time to time depending upon the devices feeding into the system. Due to these latter aspects it is practically impossible to control the data patterns upon which satisfactory working of the FCFS system would depend.

8.4 Suggestions for further work

A fairly wide range of investigations have been carried out during the course of the work reported in this thesis. Some topics which were not investigated and which are suggested by the results obtained are now presented.

In the PVS two further problems can be investigated. The first problem is how to derive an analytical approach which can be used to evaluate the percentage of packet losses and the average packet waiting time in the SQ-system. The basic analytical model which could be used for this task is the Markovian simulation model, as used in this thesis. This is a two dimensional model and it will give the correlation of the packets from frame to frame. A two dimensional Markovian solution could be used to evaluate both the average packet waiting time and the percentage of buffer overflow probability.

The second problem in the PVS worth investigating is the subjective effect of the correlation of the packet losses. The work done in this thesis assumes that the packet losses are from an independent random process (i.e. without correlation or dependency) and the maximum percentage loss has been kept below 1%, this being the figure recommended by the authors of Ref [19], [48], [49]. In chapter 3, it was shown that the packet arrival could be correlated for small number of channels. This effect of packet arrival correlation can introduce an additional correlation into the packet loss mechanism. Under supervision of Dr G.J. Hawkins and the present author an attempt was undertaken by Mr Izzat [82] to investigate the subjective effects of the correlation of packet losses. The model used an infinite transmitter buffer SQ-system multiplexing 40 users onto a 24-channel T1-carrier link. A 50-second sample of the instantaneous queueing delays from each of the packets of a channel were derived from the microscopic simulations reported earlier in this thesis. These were to delay packets derived from actual 2-second recording; the NTI strategy was employed at the receiver. Informal listening tests were conducted to assess the acceptability of the talkspurt subjected to the presence of packet losses. The listening tests, in general, indicated that the packet losses of up to 4 to 5 per cent could be acceptable. For each value of the control time the total number of times out of 25 that the talkspurt was acceptable was registered. This number was then divided by 25 to obtain the probability that a talkspurt would be acceptable for the given control time situation. The control time was then varied and the characteristic of the acceptability of the talkspurt versus the control time was obtained. The characteristic showed that as the control time approached 20 ms the talkspurt acceptability rose rapidly. As the control time increased above 20 ms the curve flattened off at about 90% acceptability. Indications are that greater acceptability ensues as the control time is increased. Only a few tests have been conducted and much further work is required.

The simulation and subjective testing work, described above, was performed using an infinite buffer at the transmitter. As was shown earlier in the thesis, it is better for the packet losses to take place at the transmitter. Thus it would be worthwhile investigating the subjective effects of correlation of packet losses in the SQ-system having finite buffering at the transmitter employing various other control strategies.

Finally analytical approaches need to be developed which are capable of evaluating the performance characteristics of the totally integrated packet-switched voice/data system. Solvable multi-dimensional Markov chain models, similar to the simulation models used in this thesis, need to be developed in order to evaluate these problems analytically. However they are very complex and to date suitable solutions have not been found.

APPENDIX A

In this appendix we show the technique which has been used for drawing random numbers from the cummulative distribution functions of the talkspurt and pause intervals as given by Brady in Ref [1] (PP 17). The following procedures have been carried out:

(1) The vertical axises of the Brady's CDF for the talkspurt and the pause intervals are divided into 20 segments. As the total height of each vertical axis corresponds to probability one, then each segment has the maximum probability toleration of 1/20 = 0.05.

(2) For each segment; the maximum length (X_{max}), the minimum length (X_{min}) and the difference length ($X_{max} - X_{min}$) of the talkspurt and pause intervals on the horizontal axis are evaluated. The results are then summarized accordingly in tables A.1 and A.2.

(3) In order to draw random numbers from each distribution; a random number, as denoted by r, is first drawn from a uniform distribution ranged from 0 to 1. The segment which this random number belongs to is evaluated according to the following relation.

$$I = \lceil 20r \rceil \qquad (A.1)$$

where r represents the random number drawn from the uniform distribution, X denotes the greatest integer greater than or equal to X and I represents the segment number ranged from 1 to 20.

After finding the segment the random number with the probability r is belonged to, then the length of the sample is found by

$$X = X_{min}(I) + (r - 20/I)d(I) \qquad (A.2)$$

where

$$d(I) = X_{max}(I) - X_{min}(I) \qquad (A.3)$$

Some tests were performed in order to see the validity of this technique. A computer simulation program was written which was based on following the above procedures (1-3). In each case (i.e. talkspurt or pause simulation), 50,000 samples were drawn and the corresponding CDF was obtained. The results as depicted in fig. A.1 shows that good agreements are achieved by comparing these results with the Brady's.

TABLE (A.1)

TALKSPURT

I	Probability r	X_{min} milliseconds	X_{max} milliseconds	$d=X_{max}-X_{min}$
1	0 -0.05	10	16	6
2	0.05-0.1	16	60	44
3	0.1 -0.15	60	150	90
4	0.15-0.2	150	200	50
5	0.2 -0.25	200	250	50
6	0.25-0.3	250	320	70
7	0.3 -0.35	320	400	80
8	0.35-0.4	400	500	100
9	0.4 -0.45	500	600	100
10	0.45-0.5	600	790	190
11	0.5 -0.55	790	990	200
12	0.55-0.6	990	1200	210
13	0.6 -0.65	1200	1400	200
14	0.65-0.7	1400	1600	200
15	0.7 -0.75	1600	2000	400
16	0.75-0.8	2000	2200	200
17	0.8 -0.85	2200	2800	600
18	0.85-0.9	2800	3200	400
19	0.9 -0.95	3200	4200	1000
20	0.95-1.0	4200	7200	3000

TABLE (A.2)

PAUSE

I	Probability r	X_{min} milliseconds	X_{max} milliseconds	$d=X_{max}-X_{min}$
1	0 -0.05	200	225	25
2	0.05-0.1	225	250	25
3	0.1 -0.15	250	280	30
4	0.15-0.2	280	320	40
5	0.2 -0.25	320	360	40
6	0.25-0.3	360	400	40
7	0.3 -0.35	400	460	60
8	0.35-0.4	460	530	70
9	0.4 -0.45	530	600	70
10	0.45-0.5	600	700	100
11	0.5 -0.55	700	830	130
12	0.55-0.6	830	1000	170
13	0.6 -0.65	1000	1200	200
14	0.65-0.7	1200	1400	200
15	0.7 -0.75	1400	1700	300
16	0.75-0.8	1700	2100	400
17	0.8 -0.85	2100	2800	700
18	0.85-0.9	2800	4000	1200
19	0.9 -0.95	4000	6000	2000
20	0.95-1.00	6000	15000	9000

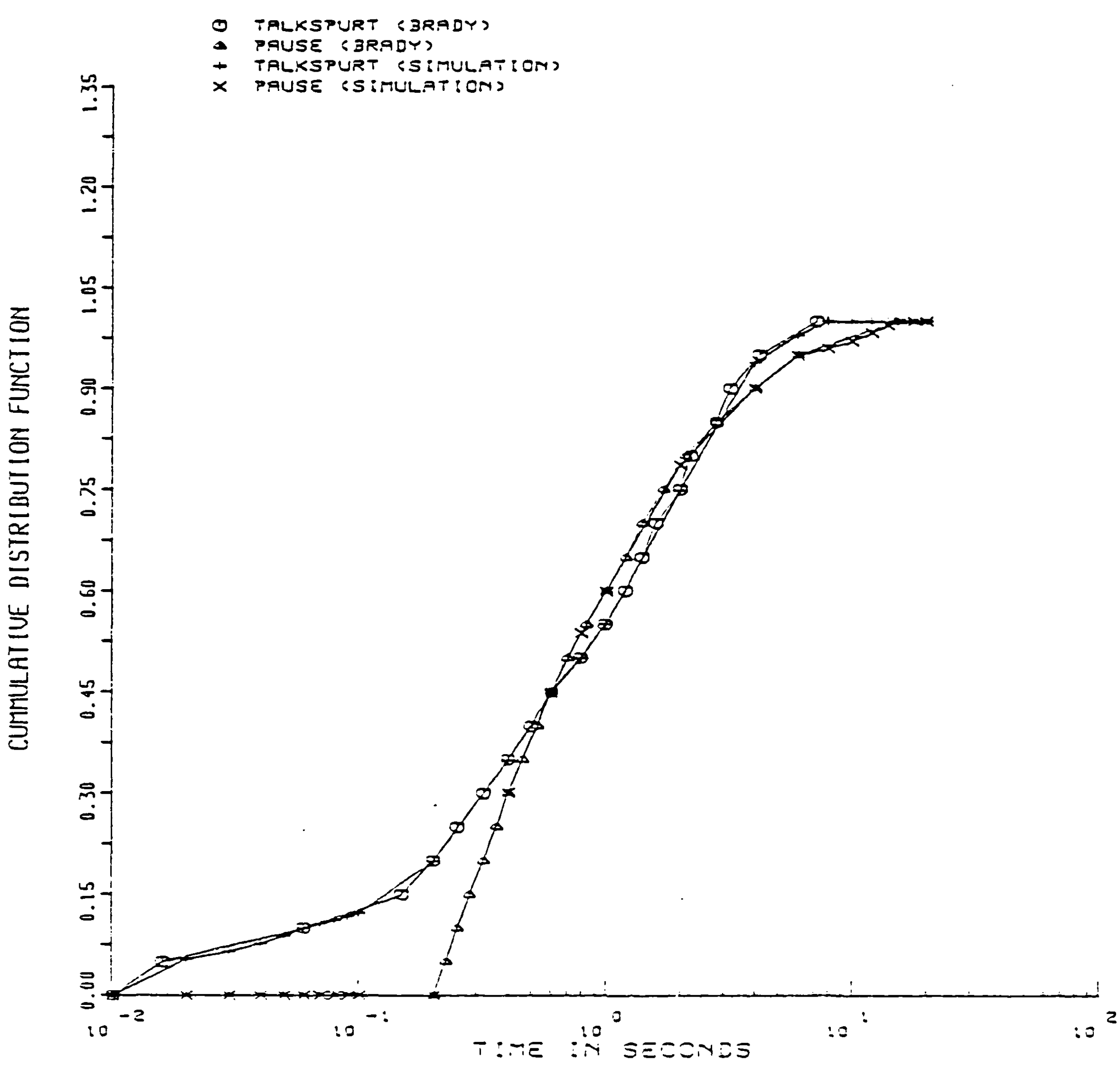


Fig. A.1 Talkspurt and Pause distributions

APPENDIX B

Some examples are given here in order to illustrate the requirement of the notation " $\lceil x \rceil$ " in equation (5.15) which has been rewritten below.

$$m_{32}(r) = \left\lceil \frac{rP_1 - e}{P_1 - P_2} \right\rceil \qquad (5.15)$$

In this equation $m_{32}(r)$ represents the number of active channels that are required to be switched from 64 kb/s to 32 kb/s in order to let the system work without packet losses. For a 1.544 Mb/s transmission link and with 10 milliseconds frame length; e, P_1 and P_2 are simply equal to

$$e = 15440 \text{ bits}$$

$$P_1 = 740 \text{ bits}$$

$$P_2 = 420 \text{ bits.}$$

J_1 or the maximum number of connected channels that are operating at 64 kb/s without any switching requirements, is then given by

$$J_1 = \left\lfloor \frac{e}{P_1} \right\rfloor = \left\lfloor \frac{15440}{740} \right\rfloor = 20.$$

Therefore, the switching from 64 kb/s to 32 kb/s will be required if the instantaneous number of active channels, r, exceed 20. The examples given below consider the situations when r = 21, 22 and 23.

Following the equation (5.15) above, $m_{32}(r)$ for these cases will be

$$m_{32}(21) = \left\lceil \frac{100}{320} \right\rceil = 1 \qquad (B.1)$$

$$m_{32}(22) = \left\lceil \frac{840}{320} \right\rceil = 3 \qquad (B.2)$$

$$m_{32}(23) = \left\lceil \frac{1580}{320} \right\rceil = 5 \qquad (B.3)$$

The validity of each of the above results are now discussed as follows:

Consider the case when r = 20. For this case, in each frame 20 packets, with the length of 740 bits each, are transmitted by the channel. The remaining frame bits left vacant because the frame length is not an integer multiple of the packet length can be evaluated as follows:

$$\varepsilon = e - 20P_1$$

$$= 15440 - 20\text{x}740 = 640 \text{ bits.}$$

Now consider the case when another channel becomes active (i.e. r = 21). If all channels in this case operate at 64 kb/s, then an addition of 740 bits are required in order to transmit all the 21 packets in the frame. As only 640 additional bits are available, then switching to 32 kb/s is required for one of the channels. In this case, only one of the active channels needs to be switched to 32 kb/s. The equation (B.1) above also indicates this condition.

For the case when r = 22, if we consider two of the active channels to be switched to 32 kb/s, then an addition of 2 x 420 = 880 bits will be required. As only 640 bits are available, then another channel will have to switch to the lower rate. The equation (B.2) above, indicates that for this condition 3 channels need to be switched.

By the same type of arguments it can be shown that when r = 23, there will be the requirement of 5 channels to be switched from 64 kb/s to 32 kb/s as given by the equation (B.3) above.

APPENDIX C

Analysis of a slotted-frame, packetized speech communication system.

H. Hagirahim and G. J. Hawkins

SUMMARY

In this paper investigations on two types of packetized-voice system are reported. The packet transmission for the first type employs a form of a stochastic queueing (SQ) operation while the second type employs a non-stochastic queueing (NSQ) operation. Computer simulation models of both types have been set up.
The results of the simulations show that a better TASI advantage can be obtained with a non-stochastic transmission system.

An analytical model for the non-stochastic method has been developed, the results of which tally closely with the results obtained by computer simulation.

Results from an analysis of a multinode network based on the NSQ system are also given.

1. Introduction

The development of packet-switching concepts for voice communications offers many advantages. For example in the efficient use of the channel and in the flexibility of the network to traffic fluctuations. Comparisons between a packet-voice system (PVS) and a circuit-switched system (CSS) have been fully discussed in reference [1]. One way in which a PVS can be implemented is shown in fig. 1.
This type of packetized voice system has been studied by many researchers [1] - [9]. In this form, the analogue speech at the exchange is digitized at a uniform rate by the A/D encoder in the transmitting terminal. The samples are then organized into constant length packets by the packetizer. The speech detector, which can be a simple integrator, judges each packet as to whether it contains an active part of the voice signal or not.
All packets which contain speech, even short bursts, are transmitted through the network on a time sharing, store and forward basis by the multiplexer. Total non-speech packets are not transmitted.

The multiplexer's operation, as depicted in fig. 2, is in the form of a synchronous time division statistical multiplexer, which has been described in references [4], [9] and [13]. The operation of this type of multiplexing is a mixture of circuit and packet switching. The circuit-switching aspect is present because the synchronous multiplexer utilizes the switched virtual circuit (SVC) standard protocol [14]. The packet-switched aspect is present because of the packetization of speech samples and the use of the packet multiplexer. The multiplexer uses a single data link to the transmitter buffer and regularly sends frames of information packets to the buffer. The duration of each multiplexer frame is assumed to be Δt seconds, which is equal to the packetization period. The multiplexer therefore requests a packet from each talker's pre-buffer every frame repetition interval and a packet is transferred to the transmitter buffer only if there is a packet waiting when the pre-buffer is viewed. A packet will not appear at the pre-buffer if the speech channel is in a completely silent period.
The information contained in the transmitter buffer is sequentially fed to the transmission link, which has a constant transmission rate of c bits/sec. During high activity periods, when the number of packets in any frame produced by the talkers exceeds the number of packets that

can be sent by the channel, the transmitter buffer occupancy starts to build up. This causes the packets for each talker to be received with increasing or variable delay at the receiver. In order to restore the fidelity of the speech, received packets need to be regularised. A number of schemes or strategies for the regularisation of the samples have been suggested for use at the Packet-Voice Receiver (PVR) [see refs. 6-8]. Each of these schemes incur different buffering delays at the receiver. It is evident that at this stage there can be a loss of packets and this will occur when packets arriving at the PVR are not received by their proper 'play-out' time [6].

Reference [15] shows that for acceptable conversational speech, the maximum end-to-end delay should be less than 250 milliseconds. One method according to [9] by which this limitation can be imposed on the system is to set the size of the transmitter buffer, D_q, by the following relation:

$$D_q = 250 - \frac{P}{8} - T \quad m \quad \text{seconds} \qquad (1)$$

Where P is the packet length in bytes excluding the overhead and T is the control time, or the maximum delay applied at the receiver. For example in the 'null timing information' (NIT) [6] strategy, T represents the delay applied to the first packet belonging to the talkspurt from which subsequent packets have originated.

As the size of the transmitter buffer is finite, another cause of packet loss is due to transmitter buffer overflow.

2. The NSQ System

In this paper the system, as explained above, will be referred to as the (SQ) stochastic queueing system. We now look at a pure loss type of strategy for multiplexing and demultiplexing packets, this being a method which does not employ stochastic queueing operations at the transmitter. We call this the non stochastic queueing (NSQ) system. The operation of this scheme, is as shown in fig. 3. It is similar to the SENET approach for integrating voice and data [10], and is based on sending and receiving information packets in fixed frames of duration Δt seconds. For example, if the number of packets (K) in any frame produced by the connected channels is less than the number of packets (J) that can be sent by the channel, then the frame which is constructed for transmission will consist of (K) packets followed by (J-K) fill-in packets (or data packets). If the number of packets (K) in any frame exceeds (J), then (K-J) of these packets, chosen from among the (K) active talkers, are discarded in such a way that the packet loss probabilities are uniformly distributed amongst all the talkers.

A delay of one frame length, Δt seconds, has to be incurred at the transmitter in order to organise the packets into a regular frame pattern, so ensuring order and synchronisation.

Upon the reception of a frame at the receiver, a full frame of speech information (active spurts plus silent periods which were not transmitted) are reconstructed at the PVR and are then demultiplexed to the receiving terminals. A delay of one frame length, Δt seconds, is also required at the PVR for the frame reconstruction. In order to maintain the synchronization between the transmitter and the receiver, the demultiplexer's repetition interval has to be set to Δt seconds.

Fig. 4 shows the computer simulation results for the TASI advantage versus packet length for the SQ and NSQ systems. The simulation models were developed employing the statistics for the talkspurt and response time intervals as used by Clarke and Turner [16]. The NTI reassembly strategy was assumed to be used at the PVR for the SQ system due to its ease of implementation. The maximum end-to-end packet delay was set at 250 milliseconds and the average freezeout fractions per talker for acceptable speech quality was assumed to be 1%, as recommended in references [11] and [12].

The better TASI advantage performance of the NSQ system, as compared with the SQ system, is due to the fact that in the SQ system there are possibilities of losses at the receiver caused by the packet reassembly strategy. These receiver losses are in addition to the losses due to buffer overflow which are common to both systems.

Another advantage of the NSQ system lies in that it is not a stochastic system. This permits the formulation of an exact detailed analysis model.

3. Performance criteria of the NSQ system

In this section, a mathematical model to measure the performance of the NSQ system in terms of the end-to-end delay and the probability of packet loss is presented.

3.1 Delay Criteria

In general there are three major sources of end-to-end delay, or network delay, D_{net}, associated with the NSQ system, namely;

(1) Packetization delay D_z.

This is the constant delay required to form a packet at the input and is equal to P/8 milliseconds (where P is the packet length in bytes, excluding overheads).

(2) Frame Construction delay at the transmitter D_t

This is the constant delay required at the transmitter for the correct construction of a frame of length Δt seconds. The delay experienced by each packet is therefore equal to Δt seconds or P/8 milliseconds.

(3) Full frame reconstruction delay at the receiver D_r

At this stage, for the full frame reconstruction, the packets arriving at the receiver would have to experience a delay of Δt seconds, or P/8 milliseconds.

The equation for the network delay, D_{net}, in terms of D_z, D_t and D_r, can be written as follows:

$$D_{net} = D_z + D_t + D_r = 3P/8 \qquad (2)$$

This equation shows that the total delay per talker in the system is constant and its length is proportional to the packet length.

3.3 The Packet loss probability criteria

If we assume that at any given time, the probability that a talker will produce a packet is q, where q is fixed (this has been found by Clarke and Turner [16] to be around .45) and that all talkers can be considered to be independent, then the number of packets transmitted from the multiplexer within a given frame has a binomial distribution. The packet loss probability, Prob(L), can then be shown to be

$$P_L = \frac{1}{Nq} \sum_{r=J}^{N} A(N,r,q)(r-J) \qquad (3)$$

$$J = \frac{PC}{f(PB+H)} \qquad (4)$$

where

$A(N,r,q)$ = Prob(of r Packet arrivals within a frame of size N-packets)

N = total number of connected channels,

P = Packet length in bytes, excluding overhead,

H = overhead in bits,

C = channel rate in bits/sec,

f = sampling rate of speech signals.

B = Number of bits per speech signal.

Simulation results for this situation are shown in fig. 5 together with calculated values from the mathematical model. Close resemblance of the two sets of results is evident.

4. Multinode network using the NSQ technique.

This section investigates the parameters and characteristics of multinode networks. In the network model (fig. 6), we assume the voice traffic to be handled in packetized form using a fixed-path protocol [17]. One-way routes are established at dial-up and are retained for the duration of a conversation. The flow of traffic is assumed to be from left to right. At the 1st-node, the packet delay per user is of the magnitude of 2 Δt seconds (i.e. Δt seconds delay for packetization and Δt seconds delay for the frame construction). At all the other intermediate nodes (i.e. 2,3,....n-1), the delay assigned per user packet is $2\Delta t$ seconds, which corresponds to Δt seconds delay for frame sychronization between the nodes and another Δt seconds delay for the new frame construction. At the destination node (i.e. node n), a delay of Δt seconds is assigned for the full frame reconstruction procedure and for demultiplexing to the receiving terminals.

The equation of the network delay, D_{net}, can then be written as

$$D_{net} = (2n-1)\ \Delta t \qquad (5)$$

where n is the total number of nodes.

For the simulation exercises the links are all assumed to be of the same fixed rate (i.e. 1.544 Mb/s, T1- carrier PCM links). This suggests that the packet loss probabilities between the nodes are all equal. As the probability of packet loss at node n is zero, then the average freezeout fraction of the flow of traffic per node is limited by

$$(P_L)_{\substack{per\\node}} = \frac{(P_L)_{end\ to\ end}}{n-1} \qquad (6)$$

$$= \frac{1}{n-1}\ \%$$

By using equations 3,4,5 and 6 above, the TASI advantages of the system, in terms of packet length, have been computed for different numbers of tandom nodes, the maximum allowable end-to-end packet delay per talker is assumed to be less than 250 milliseconds and the average freezeout fractions per talker is assumed be be less than 1%. From fig. 7, for packet lengths less than 20 milliseconds and with a reasonable achievement in the TASI advantage, up to 7 nodes can be traversed. For example, for a 16 milliseconds packet length with 7 tandom nodes with 208 milliseconds constant end-to-end delay per talker, a TASI advantage of 1.4 can be achieved.

5. Conclusions

In this paper we have investigatedand compared two types of packetized-voice systems, one with the stochastic queueing behaviour (SQ) and the other with a non-stochastic queue (NSQ) behaviour at their transmitters. It has been shown that with the NSQ system higher TASI advantages are possible. A mathematical model was introduced for the NSQ system which can be used to evaluate the performance characteristics of the system. It has also been shown that a multinode network model can be implemented using the NSQ system.

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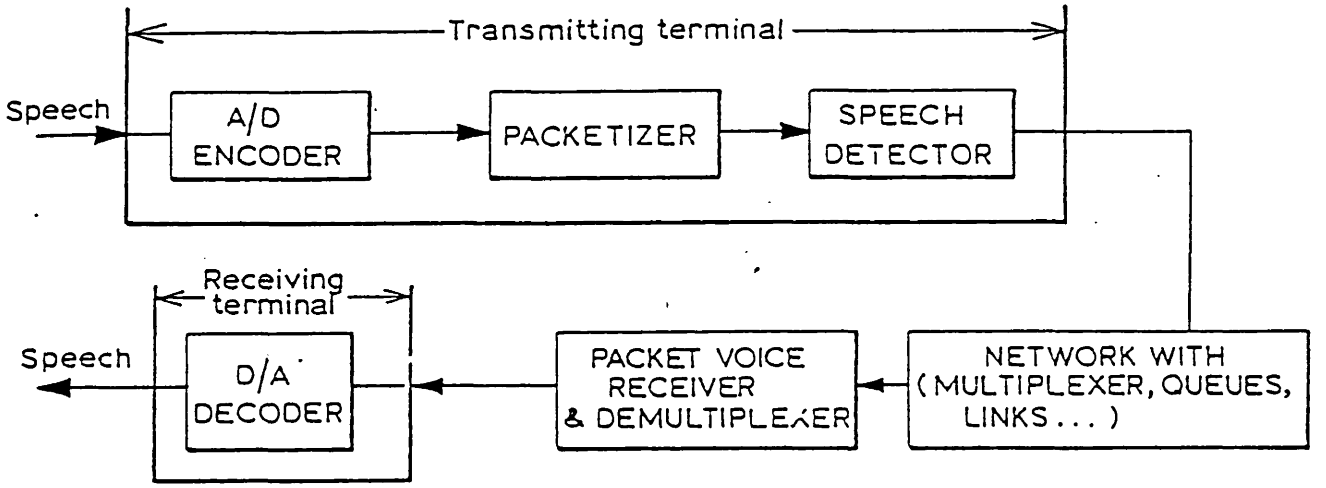


Fig. 1 The SQ pocket-voice system block diagram

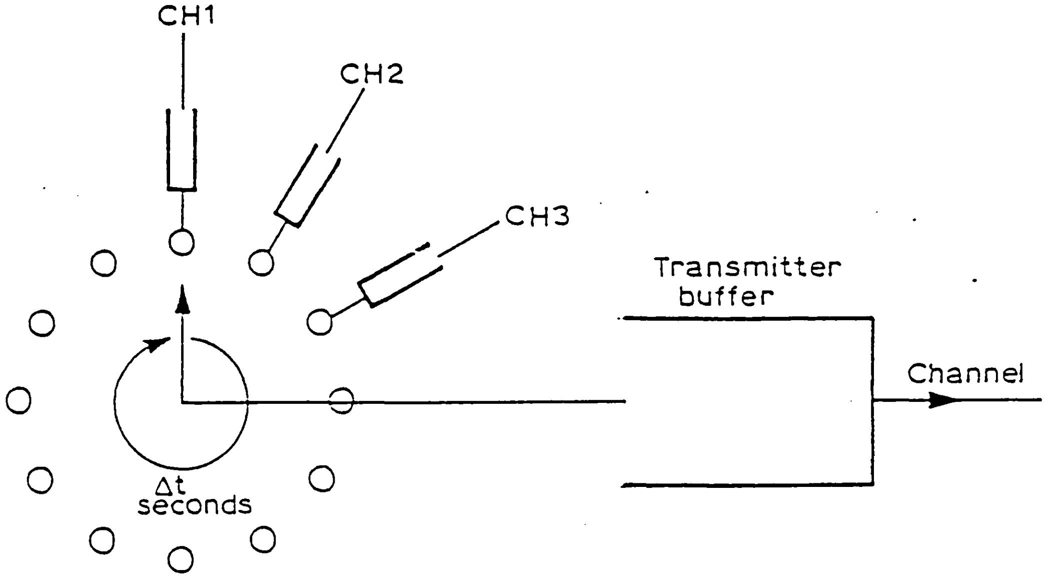


Fig. 2 The multiplexing operation of a pocket-voice system

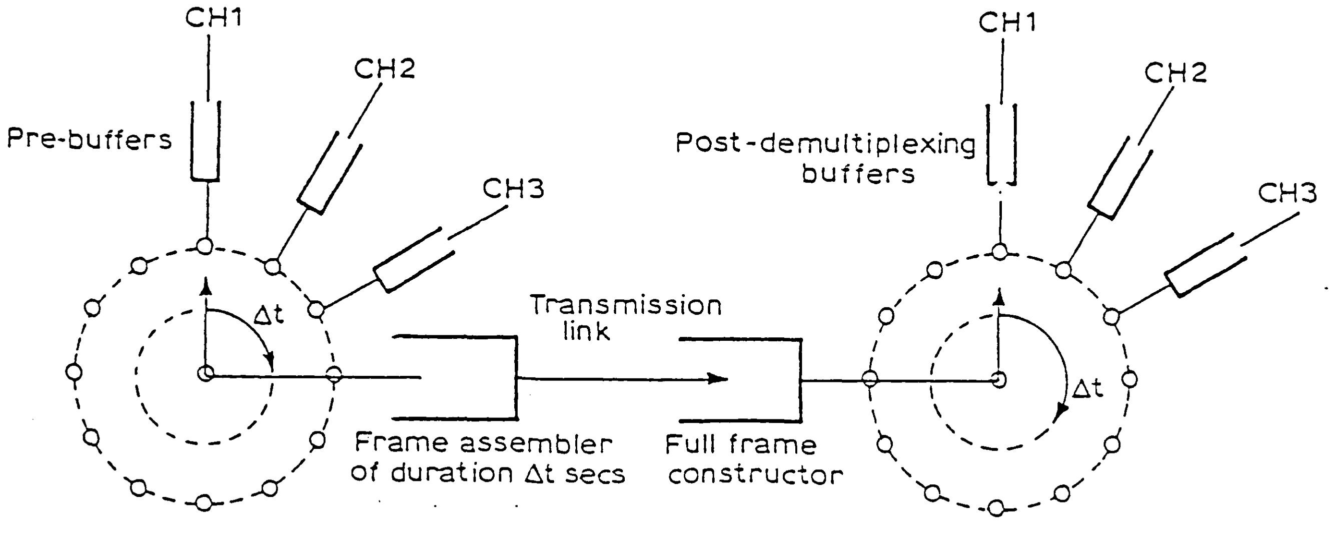


Fig. 3 The multiplexing, demultiplexing and bufferings of a two node NSQ system

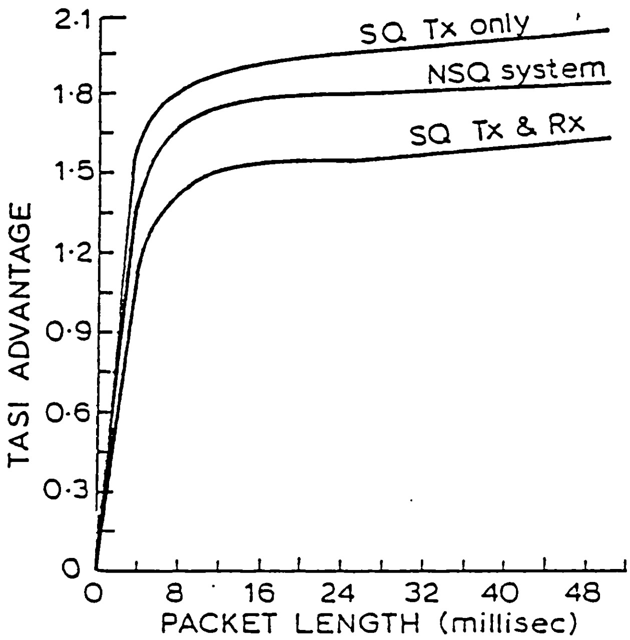


Fig. 4 The comparison of SQ and NSQ systems

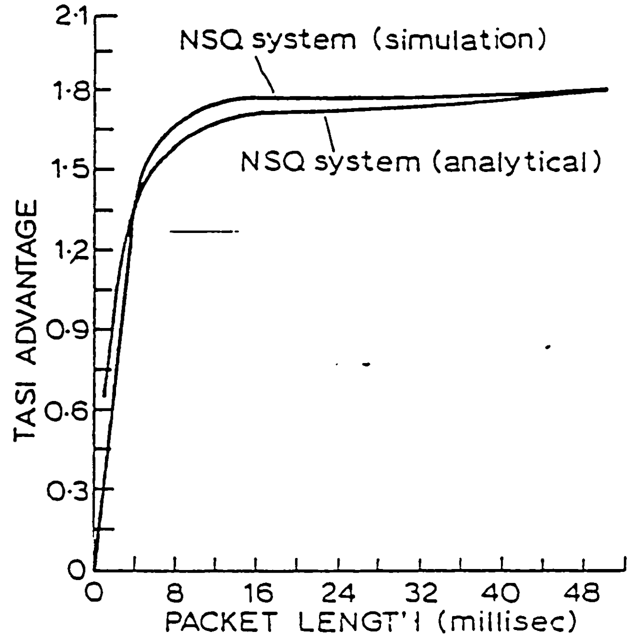


Fig. 5 The comparison of simulation and analytical results for the NSQ system

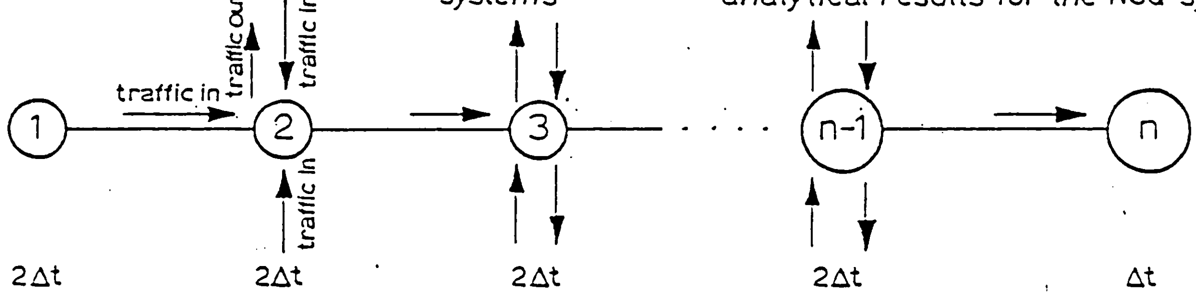
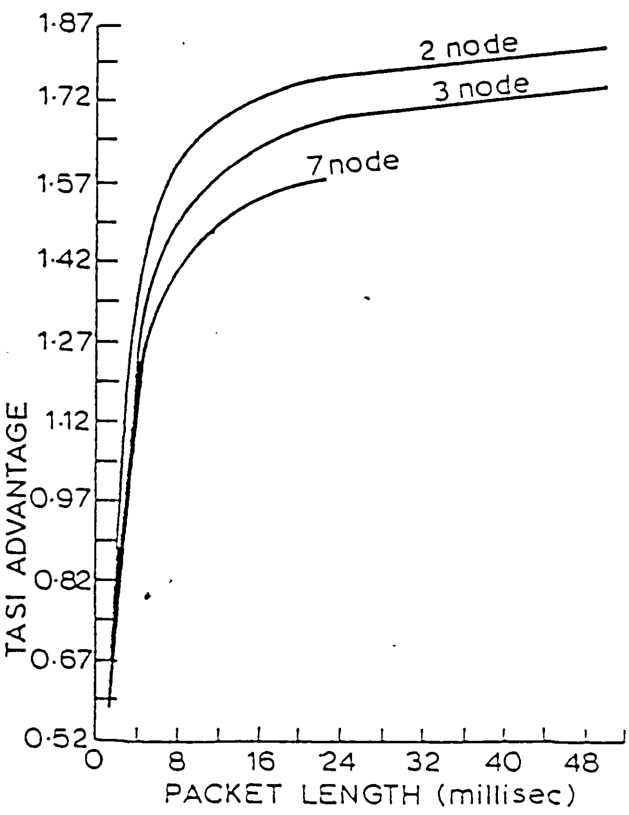


Fig. 6 The network model for the pocket-voice system



Notes for Figures 4 , 5 and 7
These results are for a transmission channel rate of 1.544 Mbps; packet overhead of 100 bits; sampling rate of speech signal 8 kbps and 64 kbps encoder outputs.

Fig. 7 TASI advantage versus pocket length for different number of connected nodes

APPENDIX D

LIST OF PROGRAMS

```
C    *********************************************
C    *                                           *
C    * THIS PROGRAM EVALUATES THE CHARACTERISTIC *
C    * OF TASI ADVANTAGE VERSUS PACKET LENGTH BY *
C    * USING THE MICROSCOPIC SIMULATION  MODEL.  *
C    *                                           *
C    *********************************************
C
 PROGRAM SIM1(INPUT,OUTPUT,TAS50,APPB1,DIST,CAL,SOR1,SD,
+ TAPE5=INPUT,TAPE6=OUTPUT,TAPE3=TAS50,TAPE19=DIST,TAPE17=CAL,
+ TAPE4=APPB1,TAPE33=SOR1,TAPE34=SD)
 COMMON IPLEN,NSUB
 NSUB=35
 DO 12,IPLEN=30,190,10
 CALL DVFT
 12 CONTINUE
 DO 13,IPLEN=200,400,100
 CALL DVFT
 13 CONTINUE
   STOP
   END
C-------------------------------------------------------
C
 SUBROUTINE DVFT
 COMMON IPLEN,NSUB
C
C
DIMENSION IPACK(100),IA(100),IB(100),ISUM(100),I(100)
DIMENSION IDUM(100),IMESS(48,800)
DIMENSION IBSUM(100),TALK1(20),TALK2(20),PAUSE1(20),PAUSE2(20)
C IFL=FRAME LENGTH
C NPACK=NO OF PACKETS IN A FRAME LENGTH
C IPLEN=PACKET LENGTH
   INTEGER IPLEN
 REAL A,B
 REAL X2,XX1,XX2
 REAL G05CAF
133 CALL G05CBF(0)
 DATA TALK1/10.,16.,60.,150.,200.,250.,320.,400.,500.,600.,790.,
+           990.,1200.,1400.,1600.,2000.,2200.,2800.,3200.,4200./
DATA TALK2/6.,44.,90.,50.,50.,70.,80.,100.,100.,190.,200.,220.,
+           200.,200.,400.,200.,600.,400.,1000.,3000./
DATA PAUSE1/200.,225.,250.,280.,320.,360.,400.,460.,530.,600.,
+    700.,830.,1000.,1200.,1400.,1700.,2100.,2800.,4000.,6000./
DATA PAUSE2/25.,25.,30.,40.,40.,40.,60.,70.,70.,100.,130.,170.,
+           200.,200.,300.,400.,700.,1200.,4000.,9000./
 MB=8
 IC=1544000
 IO=100
 ISAM=8000
 PI=FLOAT(IPLEN*MB+IO)
 SERVICE=PI*1000./FLOAT(IC)
 PACKZD=FLOAT(IPLEN)/8.
 F1=(FLOAT(MB*IPLEN+IO))/FLOAT(IC)
 F2=(FLOAT(IPLEN))/(FLOAT(NSUB*ISAM))
 PDELAY=(F1-F2)*1000.
 CDELAY=F2*1000.
 DELAY=0.
 XLOSS=0.
 XMAX=50.
 XNOM=0.
 DO 100,IS=1,100
     IPACK(IS)=0
     IA(IS)=0
     IB(IS)=0
     IDUM(IS)=0
     ISUM(IS)=0
     I(IS)=0
     IBSUM(IS)=0
 100 CONTINUE
     NSIM=800*ISAM/IPLEN
     NST=100*ISAM/IPLEN
C         NST=1
      INK=NSIM*IPLEN
      DO 150,IS=1,NSUB
      DO 175,IK=1,800
      IMESS(IS,IK)=0
  175 CONTINUE
 150 CONTINUE
   AC=0.
      DO 195,IS=1,NSUB
    IK=0
75  G=G05CAF(G)
   IF(G.EQ.0.)THEN
   GOTO 75
   ENDIF
   IG=G*20.+.999999
   G=(G-(FLOAT(IG-1)/20.))*20.
   B=TALK1(IG)+TALK2(IG)*G
   B=B/1000.
      IBX=B*ISAM
      ISUM(IS)=ISUM(IS)+IBX
      IF(ISUM(IS).GE.NST*IPLEN)THEN
      IBSUM(IS)=IBSUM(IS)+IBX
      ENDIF
      IK=IK+1
      IMESS(IS,IK)=IBX
95  G=G05CAF(G)
   IF(G.EQ.0.)THEN
   GOTO 95
   ENDIF
   IG=G*20.+.999999
```

```
    G=(G-(FLOAT(IG-1)/20.))*20.
    A=PAUSE1(IG)+PAUSE2(IG)*G
    A=A/1000.
       IK=IK+1
       IMESS(IS,IK)=A*ISAM
       ISUM(IS)=ISUM(IS)+IMESS(IS,IK)
        IF(ISUM(IS).LT.INK)THEN
       GOTO 75
       ENDIF
      IW=(IBSUM(IS)*100)/((NSIM-NST)*IPLEN)
01220C         WRITE(6,*)IS,IW,IK
        AC=AC+IW
  195 CONTINUE
01250C      WRITE(6,*)AC/FLOAT(NSUB)
01260C      Q=AC/(100.*FLOAT(NSUB))
      N=0
 200    N=N+1
      DO 300,IS=1,NSUB
      IF (IA(IS).GT.0)THEN
      IF(IA(IS).GT.IPLEN)THEN
      IPACK(IS)=0
      IA(IS)=IA(IS)-IPLEN
      IDUM(IS)=0
      ELSE
      IF(IA(IS).EQ.IPLEN)THEN
      IPACK(IS)=0
      ELSE
      IPACK(IS)=IPLEN
      ENDIF
      IDUM(IS)=IPLEN-IA(IS)
      IA(IS)=0
      ENDIF
      ELSE
      IF(IB(IS).GT.0)THEN
      IPACK(IS)=IPLEN
      IDUM(IS)=0
      IB(IS)=IB(IS)-IPLEN
      ELSE
      I(IS)=I(IS)+1
      IB(IS)=IMESS(IS,I(IS))-IDUM(IS)
      IF(IB(IS).LT.0)THEN
      IB(IS)=IPLEN
      ENDIF
      IPACK(IS)=IPLEN
      IDUM(IS)=0
      IB(IS)=IB(IS)-IPLEN
      ENDIF
      IF(IB(IS).LE.0)THEN
      I(IS)=I(IS)+1
      IA(IS)=IMESS(IS,I(IS))+IB(IS)
      IF(IA(IS).LT.0)THEN
        IA(IS)=IPLEN
        ENDIF
        IB(IS)=0
        ENDIF
        ENDIF
      IF(IPACK(IS).GT.0)THEN
      IF(N.GT.NST)THEN
      XNOM=XNOM+1.
      ENDIF
      DELAY=DELAY+PDELAY
      IF(DELAY.GT.XMAX)THEN
      IF(N.GT.NST)THEN
      XLOSS=XLOSS+1.
      ENDIF
      DELAY=DELAY-CDELAY-PDELAY
      ENDIF
      ELSE
      DELAY=DELAY-CDELAY
      IF(DELAY.LT.0.)THEN
      DELAY=0.
      ENDIF
      ENDIF
      300 CONTINUE
      IF(N.LT.NSIM)THEN
      GOTO 200
      ENDIF
      XLOSS=XLOSS/XNOM
      XLOSS=XLOSS*100.
      IF(XLOSS.LE.1.)THEN
      NSUB=NSUB+1
      GOTO 133
      ELSE
      NSUB=NSUB-1
      TASI=FLOAT(NSUB)/24.
      ENDIF
      WRITE(6,*)FLOAT(IPLEN)/8.,TASI,NSUB
      WRITE(3,*)FLOAT(IPLEN)/8.,TASI,NSUB
      NSUB=NSUB-1
      END
```

```
C     ********************************************
C     *                                          *
C     * THIS PROGRAM EVALUATES THE CHARACTERISTIC *
C     * OF PACKET LOSSES VERSUS PACKET LENGTH BY  *
C     * USING THE MARKOVIAN SIMULATION MODELLING. *
C     *                                          *
C     ********************************************
C
      PROGRAM MARKOV(OUTPUT,Z1,YS2,YS3,YS4,TAPE3=Z1,TAPE12=YS2,
     + TAPE13=YS3,TAPE14=YS4,TAPE6=OUTPUT)
      COMMON IPLEN
      DO 12,IPLEN=50,190,10
      CALL MARK
   12 CONTINUE
      DO 13,IPLEN=200,400,100
      CALL MARK
   13 CONTINUE
      STOP
      END
C
C
      SUBROUTINE MARK
      COMMON IPLEN
      REAL G05DBF
      REAL G05CAF
      CALL G05CBF(0)
      B=1.34
      A=1.67
      Q=B/(B+A)
      NSUB=40
      A=1./A
      B=1./B
      SUM=0.
      G=0.
      IFLAG=0
      TRANS=0.
      Z=Q*FLOAT(NSUB)
      F=FLOAT(IPLEN)/8.
      ZT=250.
      C=1544000.
      E=F*C/1000.
      P1=8.*FLOAT(IPLEN)+100.
      SERVICE=P1*1000./C
      XMAX=ZT-F-SERVICE
      XMAX=(XMAX*C)/(1000.*P1)
      PT=E/P1
      PQ=0.
      XLOSS=0.
      SUMQ=0.
      NST=1000
      NSIM=20000
      SUMP=0.
      XL=0.
      K=0
  100 A1=A*FLOAT(NSUB-K)
      B1=B*FLOAT(K)
      T=1./(A1+B1)
      X=G05DBF(T)
      GX=X*1000.
      GX=GX+G
      IF(GX.GT.F)THEN
      M=GX/F
      G=GX-FLOAT(M)*F
      DO 200,J=1,M,1
      PQ=PQ+FLOAT(K)-PT
      IF(IFLAG.EQ.1)THEN
      PQ=PQ+1.
      KFLAG=1
      IFLAG=0
      ENDIF
      IF(PQ.LT.0.)THEN
      PQ=0.
      ENDIF
      IF(SUM.GE.FLOAT(NST))THEN
      XL=XL+1.
      SUMQ=SUMQ+PQ
      SUMP=SUMP+FLOAT(K)
      IF(KFLAG.EQ.1)THEN
      SUMP=SUMP+1.
      KFLAG=0
      ENDIF
      IF(PQ.GT.XMAX)THEN
      XLOSS=XLOSS+FLOAT(K)
      ENDIF
      ENDIF
  200 CONTINUE
      ELSE
      G=GX
      ENDIF
      PUP=A1/(A1+B1)
      Y=G05CAF(Y)
      IF(PUP.GE.Y)THEN
      K=K+1
      ELSE
      IF(G.GT.0.)THEN
      IFLAG=1
      ENDIF
      K=K-1
      IF(K.LT.0)THEN
      K=0
      ENDIF
      IF(K.GT.NSUB)THEN
```

```
K=NSUB-1
ENDIF
ENDIF
SUM=SUM+X
TRANS=TRANS+1.
IF(SUM.LT.FLOAT(NSIM))THEN
GOTO 100
ENDIF
XLOSS=XLOSS/SUMP
XLOSS=XLOSS*100.
Z1=F
Z2=XLOSS
Z3=SUMQ/XL
Z=Z*1000./F
Z3=Z3/Z
Z3=Z3*1000.
Z3=Z3+SERVICE
Z4=Z3+Z1
WRITE(6,*)Z1,Z2,Z3,Z4
WRITE(3,*)Z1,Z2,Z3,Z4
END
```

```
00100C  *********************************************
00110C  *                                           *
00120C  * THIS PROGRAM EVALUATES THE CHARACTERISTIC *
00130C  * OF TASI ADVANTAGE VERSUS PACKET LENGTH *
00140C  * WITH 2% BLOCKING PROBABILITY. THE MODEL *
00150C  * USED IS A MARKOVIAN SIMULATION APPROACH *
00160C  *                                           *
00170C  *********************************************
00180C
 PROGRAM MARKOV(OUTPUT,SQT4,YS2,YS3,YS4,TAPE3=SQT4,TAPE12=YS2,
00200+ TAPE13=YS3,TAPE14=YS4,TAPE6=OUTPUT)
COMMON IPLEN,NSUB
NSUB=39
DO 12,IPLEN=50,200,30
CALL MARK
12 CONTINUE
DO 13,IPLEN=300,400,100
CALL MARK
13 CONTINUE
STOP
END
00310C
00320C
 SUBROUTINE MARK
COMMON IPLEN,NSUB
DIMENSION W(200)
REAL G05DBF
REAL G05CAF
DO 15,IZ=1,200
W(IZ)=0.
15 CONTINUE
 133 B=1.34
 A=1.67
 A=1./A
 B=1./B
SUM=0.
G=0.
IFLAG=0
F=FLOAT(IPLEN)/8.
ZT=250.
C=1544000.
E=F*C/1000.
P1=FLOAT(IPLEN*8)+100.
SERVICE=P1*1000./C
XMAX=ZT-F-SERVICE
XMAX=(XMAX*C)/(1000.*P1)
PT=E/P1
IP=E/P1
PQ=0.
XLOSSN=0.
NST=1
```

```
NSIM=60000
SUMPN=0.
XL=0.
K=10
IFLAG=0
M=5
D=.01
R0=100.
R1=1.
S=1.
SM0=0.
SM1=0.
DO 20,I=0,NSUB,1
S=S*FLOAT(I)
IF(S.EQ.0.)THEN
S=1.
ENDIF
S0=(R0**I)/S
S1=(R1**I)/S
SM0=SM0+S0
SM1=SM1+S1
20 CONTINUE
FR0=(S0/SM0)-0.02
FR1=(S1/SM1)-0.02
CH=FR0*FR1
IF(CH.GE.0.)THEN
WRITE(6,*)"ERROR"
ENDIF
33 R2=R0-((R0-R1)*FR0/(FR0-FR1))
S=1.
SM2=0.
DO 30,I=0,NSUB,1
S=S*FLOAT(I)
IF(S.EQ.0.)THEN
S=1.
ENDIF
S2=(R2**I)/S
SM2=SM2+S2
30 CONTINUE
FR2=(S2/SM2)-0.02
XH=FR2*FR0
IF(XH.GT.0.)THEN
DX=ABS(R2-R0)
R0=R2
FR0=FR2
ELSE
DX=ABS(R2-R1)
R1=R2
FR1=FR2
ENDIF
IF(DX.GT.0.0000001)THEN
GOTO 33
ENDIF
R=R2*D
W(NSUB)=R
300 IF(M.EQ.NSUB)THEN
T1=1./(FLOAT(NSUB)*D)
PUP1=0.
ELSE
T1=1./(R+FLOAT(M)*D)
PUP1=R/(R+FLOAT(M)*D)
ENDIF
X1=G05DBF(T1)
SUMX2=0.
100 A1=A*FLOAT(M-K)
B1=B*FLOAT(K)
IF(IFLAG.EQ.0)THEN
L=K
ENDIF
T=1./(A1+B1)
X=G05DBF(T)
SUMX2=SUMX2+X
GX=X*1000.
GX=GX+G
IF(GX.GT.F)THEN
IFLAG=0
IM=GX/F
G=GX-FLOAT(IM)*F
DO 200,J=1,IM,1
PQ=PQ+FLOAT(L)-PT
IF(PQ.LT.0.)THEN
PQ=0.
ENDIF
IF(SUM.GE.FLOAT(NST))THEN
XL=XL+1.
IF(M.GE.IP)THEN
SUMPN=SUMPN+FLOAT(L)
ENDIF
IF(PQ.GT.XMAX)THEN
IF(M.GE.IP)THEN
XLOSSN=XLOSSN+PQ-XMAX
ENDIF
PQ=XMAX
ENDIF
ENDIF
200 CONTINUE
ELSE
G=GX
IFLAG=1
ENDIF
PUP=A1/(A1+B1)
Y=G05CAF(Y)
```

```
IF(PUP.GE.Y)THEN
K=K+1
IF(IFLAG.EQ.1)THEN
L=L+1
IF(L.GT.M)THEN
L=M-1
ENDIF
ENDIF
ELSE
K=K-1
IF(K.LT.0)THEN
K=0
ENDIF
ENDIF
IF(SUMX2.LT.X1)THEN
GOTO 100
ENDIF
Y=G05CAF(Y)
IF(PUP1.GE.Y)THEN
M=M+1
ELSE
M=M-1
IF(M.LT.0)THEN
M=0
ENDIF
ENDIF
SUM=SUM+X1
IF(SUM.LT.FLOAT(NSIM))THEN
GOTO 300
ENDIF
XLOSSN=XLOSSN/SUMPN
XLOSSN=XLOSSN*100.
IF(XLOSSN.LE.1.)THEN
NSUB=NSUB+1
GOTO 133
ELSE
NSUB=NSUB-1
R=W(NSUB)
TASI=FLOAT(NSUB)/24.
ENDIF
Z1=F
Z2=TASI
Z3=R/D
WRITE(6,*)Z1,NSUB,Z2,Z3
WRITE(3,*)Z1,NSUB,Z2,Z3
NSUB=NSUB-2
END
```

```
00100C ********************************************
00110C *                                          *
00120C * THIS PROGRAM EVALUTES THE CHARACTERISTIC *
00130C * OF TASI ADVANTAGE VERSUS PACKET LENGTH *
00140C *        FOR THE NSQ-SYSTEM              *
00150C *                                          *
00160C ********************************************
00170C
 PROGRAM BINOM(TINFIN,OUTPUT,TAPE3=TINFIN,TAPE6=OUTPUT)
COMMON IPLEN,NSUB
NSUB=10
DO 12 IPLEN=30,190,10
CALL MATHAN
12 CONTINUE
DO 13,IPLEN=200,400,50
CALL MATHAN
13 CONTINUE
STOP
END
00290C ==============================================
00300C
   SUBROUTINE MATHAN
00320C
00330C ==============================================
00340C
COMMON IPLEN,NSUB
DIMENSION XI(100),XJ(100)
133 LSUB=NSUB+1
ISAM=8000
MB=8
IO=100
IC=1544000
ITHRESH=(IPLEN*IC)/(ISAM*(IPLEN*MB+IO))
P=.445
  XI(1)=1.
XI(2)=1.
XJ(1)=0.
XJ(2)=0.
DO 100,K=3,100
XI(K)=0.
XJ(K)=0.
100 CONTINUE
DO 200,N=2,NSUB
DO 300,M=1,100
IF(M.EQ.1)THEN
XJ(M)=1.
ELSE
XJ(M)=XI(M)+XI(M-1)
ENDIF
300 CONTINUE
DO 600,L=1,LSUB
```

```
XI(L)=XJ(L)
600 CONTINUE
200 CONTINUE
SUM=0.
DO 700,IK=ITHRESH,NSUB
R1=P**(IK)
R2=(1.-P)**(NSUB-IK)
R=(R1*R2*XI(IK+1))*(FLOAT(IK-ITHRESH))
SUM=SUM+R
700 CONTINUE
SUM=SUM/(P*FLOAT(NSUB))
SUM=SUM*100.
IF(SUM.LE.1.)THEN
NSUB=NSUB+1
GOTO 133
ELSE
NSUB=NSUB-1
TASI=FLOAT(NSUB)/24.
ENDIF
WRITE(6,*)FLOAT(IPLEN)/8.,NSUB,TASI
WRITE(3,*)FLOAT(IPLEN)/8.,NSUB,TASI
NSUB=NSUB-5
END
```

```
00100C  ********************************************
00110C  *                                      *
00120C  * THIS PROGRAM EVALUATES THE PERFORMANCE *
00130C  * CHARACTERISTIC OF THE DATA TRAFFIC IN *
00140C  * THE CIRCUIT/PACKET SLOTTED ENVELOPE *
00150C  * NETWORK (SENET) INTEGRATED SYSTEM *
00160C  *                                      *
00170C  ********************************************
00180C
 PROGRAM SIM(XMM1,OUTPUT,TAPE3=XMM1,TAPE6=OUTPUT)
COMMON ITITA
DO 12,ITITA=200,500,100
CALL PROG
12 CONTINUE
DO 14,ITITA=520,540,20
CALL PROG
14 CONTINUE
DO 13,ITITA=550,600,10
CALL PROG
13 CONTINUE
STOP
END
00320C
00330C
 SUBROUTINE PROG
 COMMON ITITA
REAL R(500)
INTEGER G05ECF
INTEGER G05EYF
CALL G05CCF
 S=30.
00410C S=NO OF CIRCUIT-SWITCHED SLOTS
 C=1544000.
 F=10.
00440C F=FRAME LENGTH IN MILLISECONDS
 V=4.
00460C V=NO OF BITS ASSIGNED TO EACH VOICE SAMPLE
 P=1000.
00480C P=DATA PACKET LENGT IN BITS
 G=8000.
00500C G=SAMPLING RATE OF SPEECH SIGNAL
 E=F*C/1000.
00520C E=FRAME LENGTH IN BITS
 Z=F*V*G/1000.
00540C Z=SIZE OF THE CIRCUIT-SLOTS PER FRAME IN BITS
PUP=0.
A=.22
B=.01
K=15
SUM=0.
SUMQ2=0.
```

```
 QUEUE2=0.
 L=0
 LOST=0
 IA=0
 TRANS=0.
 100 IF(FLOAT(K).EQ.S)THEN
 T=1./(S*B)
 PUP=0
 ELSE
 T=1./(A+FLOAT(K)*B)
 PUP=A/(A+FLOAT(K)*B)
 ENDIF
 X=G05DBF(T)
 N=(E-FLOAT(K)*Z)/P
00750C N=THE NO OF DATA PACKETS SENT PER-FRAME
 IX=X*1000./F
 T1=FLOAT(ITITA)*F/1000.
00780C T1=MEAN PACKET ARRIVAL DURING THE FRAME INTERVAL F
 IFAIL=0
 CALL G05ECF(T1,R,500,IFAIL)
 DO 200,J=1,IX
 IY=G05EYF(R,500)
 QUEUE2=QUEUE2+FLOAT(IY-N)
 IF(QUEUE2.LT.0.)THEN
 QUEUE2=0.
 ENDIF
 IF(SUM.GE.10000.)THEN
 SUMQ2=SUMQ2+QUEUE2
 L=L+1
 ENDIF
 200 CONTINUE
 Y=G05CAF(Y)
 IF(PUP.GE.Y)THEN
 K=K+1
 IA=IA+1
 IF(FLOAT(K).EQ.S)THEN
 LOST=LOST+1
 ENDIF
 ELSE
 K=K-1
 IF(K.LT.0)THEN
 K=0
 ENDIF
 ENDIF
 TRANS=TRANS+1.
 SUM=SUM+X
 IF(SUM.LT.60000)THEN
 GOTO 100
 ENDIF
 PB=FLOAT(LOST)/FLOAT(IA)
01110C PB=PROBABILITY OF BLOCKING
 Q2=SUMQ2/FLOAT(L)
01130C Q2=AVERAGE QUEUE LENGTH IN NO OF PAVKETS
01140C TQ1=AVERAGE QUEUEING DELAY IN MILLISECONDS
 TQ1=(Q2/FLOAT(ITITA))*1000.
 ROW1=A/B
 AV=(1.-PB)*ROW1
 U=(E-AV*Z)/P
01190C U=AVERAGE PACKETS SENT PER FRAME BY THE CHANNEL
 ROW=T1/U
 WRITE(6,*)T1,Q2,TQ1,PB,ROW
 WRITE(3,*)T1,Q2,TQ1,PB,ROW
 END
```

```
00100C  **********************************************
00110C  *                                            *
00120C  * THIS PROGRAM OBTAINES THE PERFORMANE OF THE *
00130C  * DATA TRAFFIC IN THE SENET INTEGRATED SYSTEM *
00140C  * WITH  NSQ-PACKETIZED  VOICE  OPERATION     *
00150C  *                                            *
00160C  **********************************************
00170C
 PROGRAM MARKOV(OUTPUT,NSQBUSY,YS2,YS3,YS4,TAPE3=NSQBUSY,TAPE12=YS2,
00190+ TAPE13=YS3,TAPE14=YS4,TAPE6=OUTPUT)
COMMON ITITA
DO 13,ITITA=800,980,20
CALL MARK
13 CONTINUE
STOP
END
00260C
00270C
 SUBROUTINE MARK
COMMON ITITA
REAL RX(100)
REAL G05DBF
REAL G05CAF
INTEGER G05ECF
INTEGER G05EYF
CALL G05CBF(0)
B=1.34
A=1.67
NSUB=30
E=15440.
IPLEN=80
A=1./A
B=1./B
SUM=0.
G=0.
IFLAG=0
F=FLOAT(IPLEN)/8.
MP=17.
P1=FLOAT(IPLEN*4)+100.
P2=1000.
TX=FLOAT(ITITA)*F/1000.
SUMQ2=0.
QUEUE2=0.
NST=10000
NSIM=60000
SUMP=0.
XL=0.
K=10
IFLAG=0
M=5
R=.22
D=.01
300 IF(M.EQ.NSUB)THEN
T1=1./(FLOAT(NSUB)*D)
PUP1=0.
ELSE
T1=1./(R+FLOAT(M)*D)
PUP1=R/(R+FLOAT(M)*D)
ENDIF
X1=G05DBF(T1)
SUMX2=0.
100 A1=A*FLOAT(M-K)
B1=B*FLOAT(K)
IF(IFLAG.EQ.0)THEN
L=K
ENDIF
T=1./(A1+B1)
X=G05DBF(T)
SUMX2=SUMX2+X
GX=X*1000.
GX=GX+G
IF(GX.GT.F)THEN
IFLAG=0
IM=GX/F
G=GX-FLOAT(IM)*F
IFAIL=0
CALL G05ECF(TX,RX,100,IFAIL)
DO 200,J=1,IM,1
IY=G05EYF(RX,100)
IF(L.GT.MP)THEN
N=(E-17.*P1)/P2
ELSE
N=(E-FLOAT(L)*P1)/P2
ENDIF
QUEUE2=QUEUE2+FLOAT(IY-N)
IF(QUEUE2.LT.0.)THEN
QUEUE2=0.
ENDIF
IF(SUM.GE.FLOAT(NST))THEN
XL=XL+1.
SUMQ2=SUMQ2+QUEUE2
ENDIF
200 CONTINUE
ELSE
G=GX
IFLAG=1
ENDIF
PUP=A1/(A1+B1)
Y=G05CAF(Y)
IF(PUP.GE.Y)THEN
K=K+1
IF(IFLAG.EQ.1)THEN
```

```
L=L+1
IF(L.GT.M)THEN
L=M-1
ENDIF
ENDIF
ELSE
K=K-1
IF(K.LT.0)THEN
K=0
ENDIF
ENDIF
IF(SUMX2.LT.X1)THEN
GOTO 100
ENDIF
Y=G05CAF(Y)
IF(PUP1.GE.Y)THEN
M=M+1
ELSE
M=M-1
IF(M.LT.0)THEN
M=0
ENDIF
ENDIF
SUM=SUM+X1
IF(SUM.LT.FLOAT(NSIM))THEN
GOTO 300
ENDIF
Q2=SUMQ2/XL
TQ1=(Q2/FLOAT(ITITA))*1000.
WRITE(6,*)TX,Q2,TQ1
WRITE(3,*)TX,Q2,TQ1
END
```

```
00100C ************************************************
00110C *                                              *
00120C * THIS PROGRAM EVALUATES THE CHARACTESITIC OF  *
00130C * NUMBER OF SPEECH TERMINALS VERSUS AVERAGE    *
00140C * DATA PACKET ARRIVAL PER FRAME IN THE FIRST-  *
00150C * COME-FIRST-SERVED (FCFS) INTEGRATED SYSTEM   *
00160C *                                              *
00170C ************************************************
00180C
 PROGRAM MARKOV(OUTPUT,N11,YS2,YS3,YS4,TAPE3=N11,TAPE12=YS2,
00200+ TAPE13=YS3,TAPE14=YS4,TAPE6=OUTPUT)
COMMON ITITA,NSUB
NSUB=48
DO 12,ITITA=100,1500,200
CALL MARK
12 CONTINUE
STOP
END
00280C
00290C
 SUBROUTINE MARK
COMMON ITITA,NSUB
DIMENSION RF(200)
REAL G05DBF
REAL G05CAF
INTEGER G05ECF
INTEGER G05EYF
133 CALL G05CBF(0)
 B=1.34
 A=1.67
 A=1./A
 B=1./B
SUMQ=0.
G=0.
IFLAG=0
IPLEN=80
F=FLOAT(IPLEN)/8.
TF=FLOAT(ITITA)*F/1000.
ZT=250.
C=1544000.
E=F*C/1000.
P1=FLOAT(IPLEN*8)+100.
SERVICE=P1*1000./C
XMAX=ZT-F-SERVICE
XMAX=(XMAX*C)/(1000.*P1)
PT=E/P1
PQ=0.
SUMX2=0.
XLOSS=0.
NST=10000
NSIM=60000
```

```
SUMP=0.
XL=0.
K=0
IFLAG=0
100 A1=A*FLOAT(NSUB-K)
B1=B*FLOAT(K)
IF(IFLAG.EQ.0)THEN
L=K
ENDIF
T=1./(A1+B1)
X=G05DBF(T)
SUMX2=SUMX2+X
GX=X*1000.
GX=GX+G
IF(GX.GT.F)THEN
IFLAG=0
IM=GX/F
G=GX-FLOAT(IM)*F
IFAIL=0
CALL G05ECF(TF,RF,200,IFAIL)
DO 200,J=1,IM,1
IY=G05EYF(RF,200)
PQD=PQ+FLOAT(IY)-PT
IF(PQD.GT.XMAX)THEN
XLOSS=XLOSS+FLOAT(L)
PQ=PQD
ELSE
PQ=PQD+FLOAT(L)
ENDIF
IF(PQ.LT.0.)THEN
PQ=0.
ENDIF
XL=XL+1.
SUMP=SUMP+FLOAT(L)
SUMQ=SUMQ+PQ
200 CONTINUE
ELSE
G=GX
IFLAG=1
ENDIF
PUP=A1/(A1+B1)
Y=G05CAF(Y)
IF(PUP.GE.Y)THEN
K=K+1
IF(IFLAG.EQ.1)THEN
L=L+1
IF(L.GT.NSUB)THEN
L=NSUB-1
ENDIF
ENDIF
ELSE
K=K-1
IF(K.LT.0)THEN
K=0
ENDIF
ENDIF
IF(SUMX2.LT.FLOAT(NSIM))THEN
GOTO 100
ENDIF
XLOSS=XLOSS/SUMP
XLOSS=XLOSS*100.
WRITE(6,*)ITITA,NSUB,XLOSS
IF(XLOSS.GT.1.)THEN
NSUB=NSUB-1
GOTO 133
ENDIF
CAL=FLOAT(NSUB)*.445*100.
TITA=CAL+FLOAT(ITITA)
Q2=SUMQ/XL
TQ=(Q2/TITA)*1000.
WRITE(6,*)TF,NSUB,TQ
WRITE(3,*)TF,NSUB,TQ
NSUB=NSUB-3
END
```

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