Multimedia Integration of Variable Bit Rate Sources Over the FDDI Network.

by

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to my family .....

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ABSTRACT

Multimedia is the future of today's communication field. Multimedia communications involve the transmission of different sources, such as video, voice, data and graphics, on the same network. This thesis focuses on the performance study of the Fibre Distributed Data Interface (FDDI) network when multimedia sources are integrated. In the first part of the study, Variable Bit Rate (VBR) video and data traffic are studied on FDDI. The second part of the study is the integration of VBR video and voice sources. In order to improve the multiplexing gain, the bit dropping algorithm is used on voice sources. It is shown that due to the dynamic bandwidth transfer property of FDDI, multimedia sources can be integrated without affecting the quality of service required by various media. Performance measures such as delay, variance and efficiency are calculated for each component of multimedia. Also using the bit dropping algorithm on voice sources, the delay and the probability of loss of voice packets decreases while the quality of voice is much higher than required.

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Acronyms

ADPCM	Adaptive Differential Pulse Code Modulation
ASC	Accredited Standards Committee
BW	Bandwidth
BWN	Broadsite Wideband Network
BWmax	Maximum Bandwidth
CSMA/CD	Carrier Sense Media Access/Collision Detection
DAS	Dual Attachment Station
DPCM	Differential Pulse Code Modulation
DQDB	Double Queue Double Bus
Dmean	Mean Delay
Dprop	Propagation Delay
Dqueue	Queuing Delay
Dtotal	Total Delay
Dtrans	Transmission Delay
FC	Frame Check
FCS	Frame Check Sequence
FDDI	Fibre Distributed Data Interface
LLC	Logical Link Control
MAC	Media Access
MaxPacket_Length	Maximum Packet Length
Mbps	Mega Bits Per Second
N_b	Number of bits Per Sample
N_p	Number of pixels Per Frame
PDUs	Protocol Data Units
PHY	Physical
PMD	Physical Medium Dependent
QNAP2	Queuing Network Analysis Program
QOS	Quality of Service
RAR	Release After Rotation

SA	Source Address
SAS	Single Attachment Station
SFD	Starting Frame Delimiter
SMT	Station Management
S_{max}	Maximum Synchronous Bandwidth
Si	Time to transmit Synchronous traffic on i^{th} station
THT	Token Holding Time
TRT	Token Rotation Time
TTRT	Target Token Rotation Time
T_Pr	Asynchronous Traffic Priority Threshold
VBR	Variable Bit Rate
Kbps	Kilo Bits Per Second
packet_t	Packet Transmission Time
t_pack	Packetization time
t_smax	Transmission Time for whole Synchronous Traffic
t_{asy}	Time to Transmit Asynchronous Packet
token_p	Token Propagation Time
token_t	Token Transmission Time
$t_{overhead}$	Time to Transmit Overhead Data
tpack_max	Maximum Packetization Time

Chapter 1

Introduction

1.1 Background

In the developing advanced information society, digital video is expected to become a major traffic component on high speed integrated networks including video based services ranging from video telephony to full motion video. An important requirement of such future networks is the capability to provide variable transmission rates. This capability makes it possible to provide constant quality with variable bit rate (VBR) coding. Uncompressed video signal requires very high bandwidth (hundreds of Mbps). Therefore in limited bandwidth communication networks, only a few video sources will fill the network capacity. However, different video compression schemes are currently used to compress the video signal so that more video communications can be supported on high speed networks. Since compressed video is more sensitive to packet loss than uncompressed video, it is suggested to keep the packet loss for video as low as possible.

Multimedia traffic components have different quality of service (QOS) requirements. Video and voice traffic are sensitive to delay but can accept some packet loss while data traffic cannot accept any packet loss. To meet these QOS requirements of multimedia components, network systems have to be able to allocate bandwidth dynamically. This property is available in high speed local area networks such as the Fibre Distributed Data Interface (FDDI) [1,2]. This thesis deals with the performance study of FDDI when multimedia traffic is present. The multimedia traffic considered in this thesis has only two components - video/data or video/voice. In this thesis, the main constraint is the delay to the video packet and the performance study is carried out up to the delay limit of 250 ms for the delay to the video packet [3].

It is also assumed that sufficient buffers are available for video packets so that video packets are never lost. Figure 1.1 shows the topology of FDDI with each station having two queues for different traffic. Two queues are considered because FDDI supports synchronous as well as asynchronous traffic.

1.2 Thesis Motivation

This thesis concentrates on the performance study of FDDI when VBR sources are integrated over FDDI. Performance of FDDI is examined with traffic characteristics such as mean delay, buffer size, and variance of mean delay. At the first stage only VBR video is considered. Parameters of FDDI are set such that maximum numbers of VBR video stations are supported. The next stage includes the integration of VBR video and data (file transfer + control information). The last part deals with the integration of VBR video and voice sources. As voice source is also delay sensitive, the use of bit dropping algorithm is explored on voice sources when voice is treated as asynchronous traffic on FDDI. It is shown that by using bit dropping algorithm, the performance of FDDI improves i.e. the probability of loss voice packets decreases and the delay to video packet decreases.

1.3 Research Methodology

The thesis objective is achieved by organizing the research as follows:

- FDDI is considered as a high speed LAN for supporting multimedia communications. Each station is assumed to comprise buffer management schemes for each component of multimedia traffic. Different traffic packets are stored in two queues. Synchronous traffic packets are stored in one queue and asynchronous packets in a second buffer queue. In other words , synchronous queue will have only synchronous traffic .

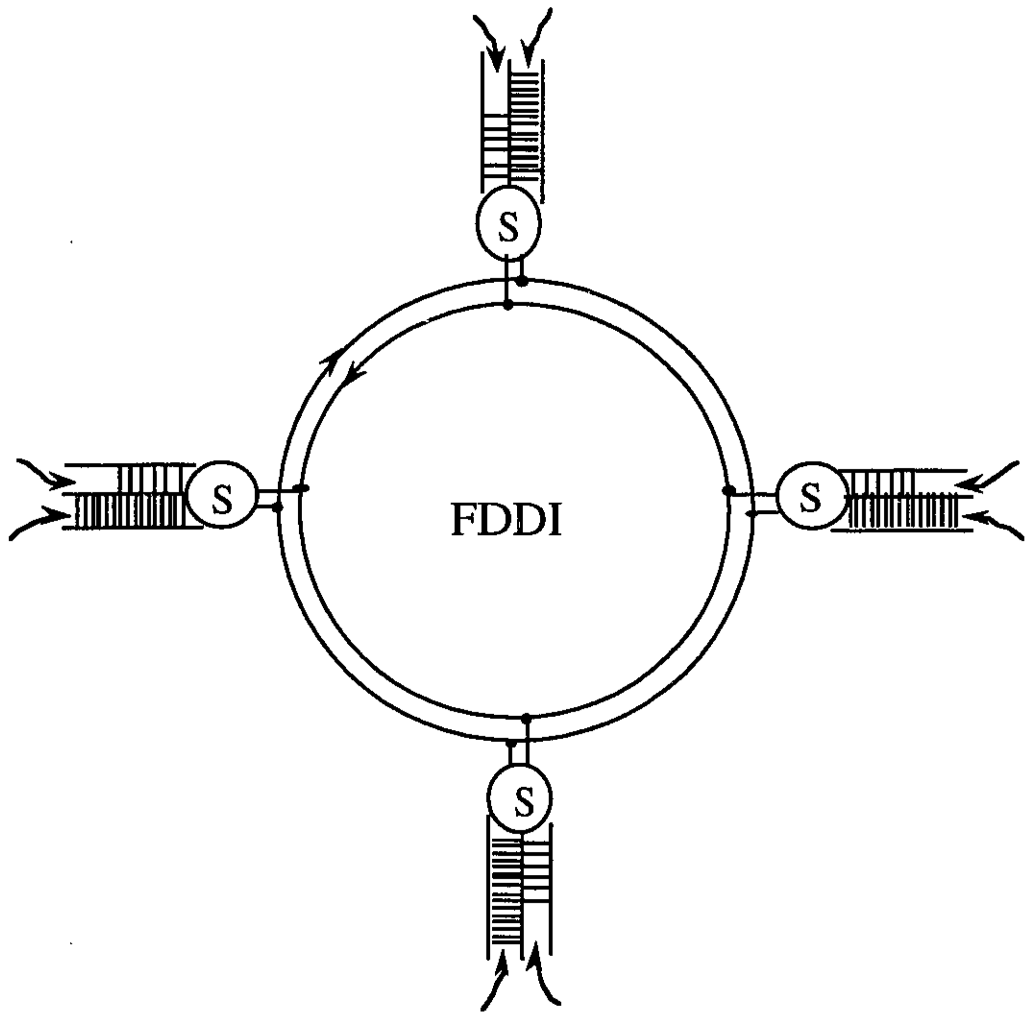


Figure 1.1 High Speed LAN Supporting Multimedia Services

- The integrated traffic consist of the following types of traffic:
  1. VBR Video: The VBR video signal is packetized and transmitted as synchronous traffic or traffic with priority. Complete details of VBR video modeling are explained in chapter 4.

2. Voice Source: Each voice source is modeled as a two state Markov chain with exponentially distributed ON and OFF periods. Packets are generated only in the ON period and silence detection is employed using time stamp.
3. Data traffic: The data traffic is a mixture of file transfer application and control information.

- In the first part of the simulations, only VBR video is investigated on FDDI for the case when there is no waiting delay to video packets and when some permissible delay is acceptable to video.
- The second part of the study looks at the performance study of the integration of VBR video and asynchronous data (file transfer and control information).
- The third part of the study considers the integration of VBR video and voice sources over FDDI. The effect of bit dropping algorithm on voice sources is explored.
- The last part of the study is an extension of the results of the previous part. Video and voice are integrated on FDDI, a limit of the number of multimedia stations is reached when delay to video packet is more than 250 ms. However, more voice stations can be added to the existing multimedia network. This part studies the performance of FDDI when more voice stations are added to the network so that delay to video is less than 250 ms.

1.4 Thesis Organization

This thesis will present a detailed performance study of FDDI supporting multimedia communications. The study will focus on VBR sources so that uniform quality of the picture can be maintained at the receiver end. The key part of the performance study is the delay encountered by the video packets. Video is considered as either synchronous traffic or priority traffic (when the delay to video is permissible). The effect of high bandwidth requirement by video on other multimedia traffic is also explored.

In order to evaluate the performance of FDDI supporting multimedia communications, simulation models have been written in QNAP2 (Queuing Network Analysis Program) [4] . This program provides different methods to evaluate the results with a confidence level of 95%. Most of the results in this thesis have been evaluated with confidence interval of 5% or less. Some results, for heavily loaded conditions , were obtained with confidence intervals of 10%. For such cases to get better results than 10% confidence interval, very long simulation time (2 days) was required.

The remainder of the thesis is composed of five chapters. Chapter 2 describes the details of the FDDI standard and token ring at MAC. The difference between token ring and FDDI protocol is clearly discussed.

Chapter 3 focuses on the simulation model of the FDDI. There are three simulation models. First , we consider only VBR video and no other traffic. The second model is written for FDDI supporting VBR video and asynchronous data (file transfer and control information). The third model integrates video and voice source on FDDI and also explores the effect of bit dropping algorithm on voice sources.

Chapter 4 gives the complete performance study of FDDI when (1) only VBR video is present (2) when VBR video and file transfer are integrated over FDDI. It is concluded that due to the variable length of video packets, simulations are required to set TTRT so that the FDDI timed token protocol is valid. The interesting property of FDDI known as dynamic transfer of bandwidth from asynchronous to synchronous traffic is also shown.

Chapter 5 presents the performance study of FDDI when VBR video and voice are integrated. Voice is treated as asynchronous traffic. To improve the performance the use of the bit dropping algorithm is investigated on voice. It is found that bit dropping on voice in case of congestion improves the performance of FDDI.

Chapter 6 is the conclusion of thesis with some suggestions for further work

1.5 Thesis Contributions

Contributions of this thesis are as follows:

1. Performance study of VBR video on FDDI when some delay is permissible for video and also when perfect synchronous service is provided to video, i.e., no waiting delay to VBR video.
2. Performance analysis of integration of VBR video and data (file transfer and control information) is depicted in terms of delay to video packet and delay to data packet. It is also shown that when multimedia stations increase, bandwidth transfers dynamically from data to video.
3. Performance analysis of integration of VBR video and voice source over FDDI. Effect of bit dropping algorithm over system performance is explored. It is also explored that with maximum limit of video stations on FDDI supporting multimedia (video and voice), some bandwidth is still available to add more voice stations. Further study found that many more voice stations can be multiplexed to the existing multimedia network with permissible delay to video packet less than 250 ms.

Chapter 2

FDDI and Related Standards

2.1 Introduction

Among the existing high speed local area networks, FDDI and DQDB are recognized as the premier high speed networks. These networks are expected to support multimedia communications. This chapter gives the detail of the FDDI standard which is the high speed integrated network under study in this thesis. Details of Media Access Control (MAC) and other issues of FDDI are explained. Also, the token ring (IEEE 802.5) and FDDI are compared.

2.2 FDDI Protocol Suite

As the IEEE 802.3 [5], IEEE 802.4 [6], and IEEE 802.5 [7]standards, the FDDI standard specifies both the media access layer and the physical layer. Figure 2.1 shows the architecture of the FDDI standard. This standard uses the IEEE 802.2 Logical Link Control (LLC) . There are four main components of the standard:

- Medium access control (MAC)
- Physical protocol
- Physical medium dependent layer
- Station management protocol

The MAC layer is specified in terms of MAC services and protocol. The MAC service specification defines, in functional terms, the service provided by FDDI to LLC or any other higher layer. The interface includes facilities for transmitting and receiving protocol data units (PDUs).

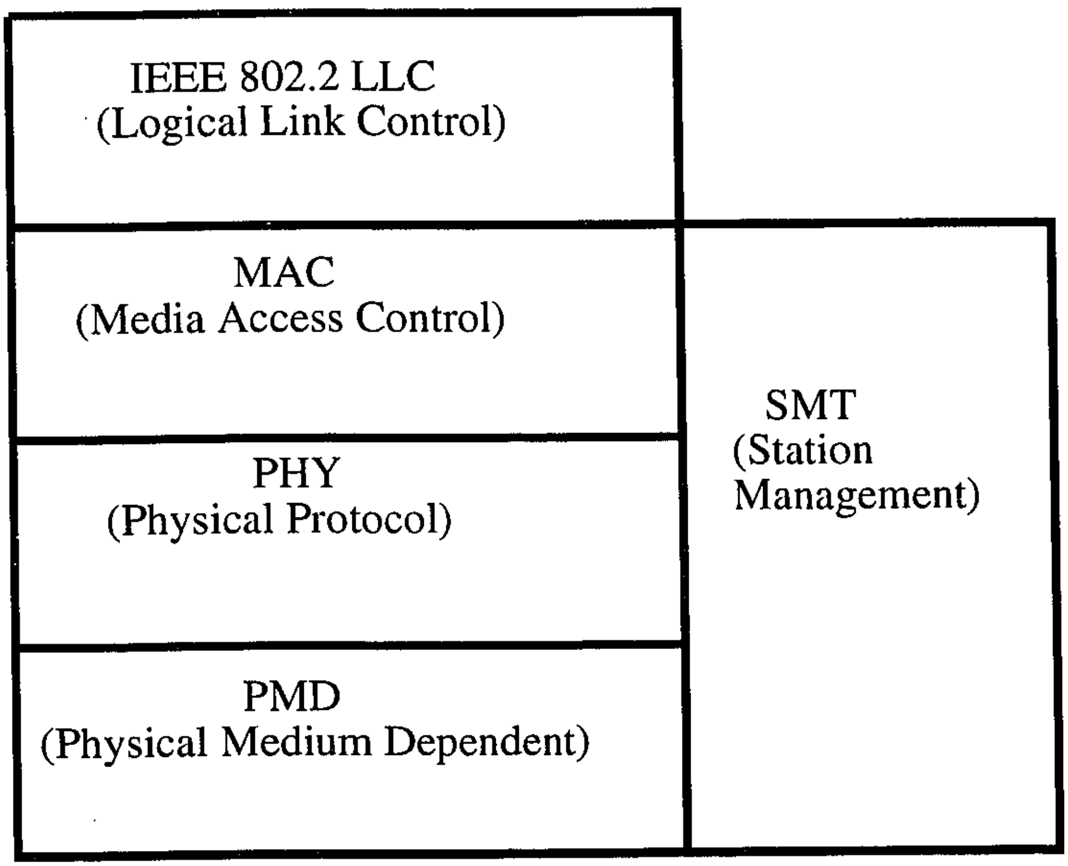


Figure 2.1 Layered Architecture of FDDI

The service specification hides the detail of the MAC and physical layers from the MAC users. The MAC protocol is the heart of FDDI standard. The specification defines the packet structure and the medium arbitration mechanism.

The physical protocol (PHY) is the medium independent portion of the physical layer. This includes a specification of the service interface with the MAC layer. The interface specification of the service defines facilities for passing a pair of serial bit streams between MAC and PHY. It also gives the detail of encoding digital data for transmission. The physical medium dependent (PMD) sub-layer of the physical layer defines and characterizes the fibre optic driver and receivers. It also gives details of the attachment of stations to the ring , cabling and connectors. The full details of PHY and PMD are given in the FDDI standard [1] [2].

The station management protocol is the network management protocol. This protocol controls all the processes underway in the different layers of the FDDI such that a station can work cooperatively on a ring. Details of SMT can also be found in the FDDI standard [1] [2].

2.3 Packet Structure

Before discussing the operation of FDDI media access protocol , it is necessary to explain the packet structure. The standard expresses the structure in terms of symbols exchanged between MAC entities. Each symbol corresponds to 4 bits. This assignment was chosen because, at the physical layer, data are transmitted in chunks of four bits. Figure 2.2 shows the packet structure and details of different fields of packet. The overall packet format consists of the following fields:

- Preamble (PA): Synchronizes the packet with each station's clock. There are 16 symbols (64 bits) used for PA. It may have more than 16 symbols depending upon the synchronization requirement of the station if the station is not the originator of the packet

(a) Packet Format

←SFS→		←FCS Coverage→					←EFS→	
PA	SD	PC	DA	SA	INFORM.	PCS	ED	FS

SFS = Start of Frame sequence
PA = Preamble (16 or more symbols)
SD = Starting Delimiter(2 symbols)
PC = Packet Control(2 symbols)
DA = Destination Address(4 or 12 symbols)
SA = Source address(4 or 12 symbols)

INFO = information(0 or more symbols)
PCS = Packet check sequence(8 symbols)
EFS = End of Frame Sequence
ED = Ending Delimiter(1 symbol)
FS = Frame status(3 or more symbol)

(b) Token Format:

PA	SD	FC	ED

(c) Starting Delimiter (SD):

J	K

J = J symbol
K= K symbol

(d) Packet Control

CLFF	ZZZZ

C = Class bit
L = Address bit length
FF = Format bits
ZZZZ = Control bits

(e) Token Ending delimiter(ED)

T	T

T = Terminate symbol

(f) Packet Ending delimiter(ED)

T

T = Terminate symbol

(g) Frame status (FS)

E	A	C				
R/S	R/S	R/S	R/S	R/S	......	T

T = Terminate symbol
R = False(reset)
S = true (set)

Figure 2.2 Packet Format in FDDI Standard

- Starting Delimiter (SD): Indicates the start of the packet. It consists of signaling pattern that are always distinguishable from data. SD is coded as two symbols: JK, where J and K are non data symbols.
- Packet Control (FC): FC has the bit format CLFFZZZ, where C indicates whether it is a synchronous or asynchronous packet, L indicates the use of 4 or 12 symbol packet address, and FF indicates whether this is a LLC frame or a MAC control frame.
- Destination Address (DA): This specifies the station(s) for which the packet is intended. It may be a unique or Multicast address. DA has a 4 to 12 symbol address.
- Source Address (SA): Specifies the address of the sending station.
- Information: This field contains LLC data or data related to control information.
- Packet Check Sequence (FCS): A 32 bit cycle redundancy check.
- Ending Delimiter (ED): ED contains non data symbols to indicate the end of a packet. The delimiter is 4 bits in token format and 8 bits in other packets.
- Packet Status (FS): FS field of the returning frame contains one of the three indicators E if error detected , A if address recognized and C if frame copied. Each indicator is represented by a symbol, where R represents false and S represents true. The FS field may contain additional trailing control indicators whose use is implementer defined. If there is an odd number of additional symbols, FS field ends with a T symbol.
- Address Fields: There are two formats for addressing the FDDI packets. The first bit is always set to zero in the source field address. In the destination address first the bit is set to zero to indicate individual address and set to one to indicate a group address. A group address of all 1's is a broadcast address for all active stations on LAN. Further details on addressing fields can be found in the FDDI standard [1] [2].

2.4 Basic Operation of the FDDI Timed Token Protocol

Figure 2.3 gives an example of the ring operation . After station A has seized the token, it transmits the packet P1 and immediately transmits a new token . P1 is addressed to station, which copies it as it circulates . The frame eventually returns to A, which deletes it . Meanwhile, station B seizes the token issued by A and transmits packet P2 followed by the token. This action could be repeated any number of times so that at any one time, there could be multiple frames circulating around the ring. Each station is responsible for deleting its own frames based on the source address field.

A station which wishes to transmit waits until a token packet goes by, as indicated by a FC field with FF bits set to 00 and ZZZZ bits set to 0000. The station then seizes the token by absorbing the remainder of the token from the ring before the entire FC field is repeated. After the captured token is completely received, the station may start transmitting packets. First synchronous packets are transmitted irrespective of whether the token is late or in time and then asynchronous packets are transmitted until Token Holding Time (THT) expires.

Other stations listen to the ring and repeat passing packets. Each station introduces into the ring approximately 1 bit delay as the time to examine, copy or change a bit as necessary. Each station can check passing bits for errors and can set the E indicator if an error is detected. If a station detects its own address, it sets the A indicator; it may also copy the frame, setting the C indicator. The resulting conditions for the originating station are:

- station nonexistent / nonactive
- station exists but packet not copied
- packet copied

The status indicators (E,A,C) in the ending delimiter are examined to determine the result of the transmission. If there is an error MAC does not re-transmit but LLC takes care of it.

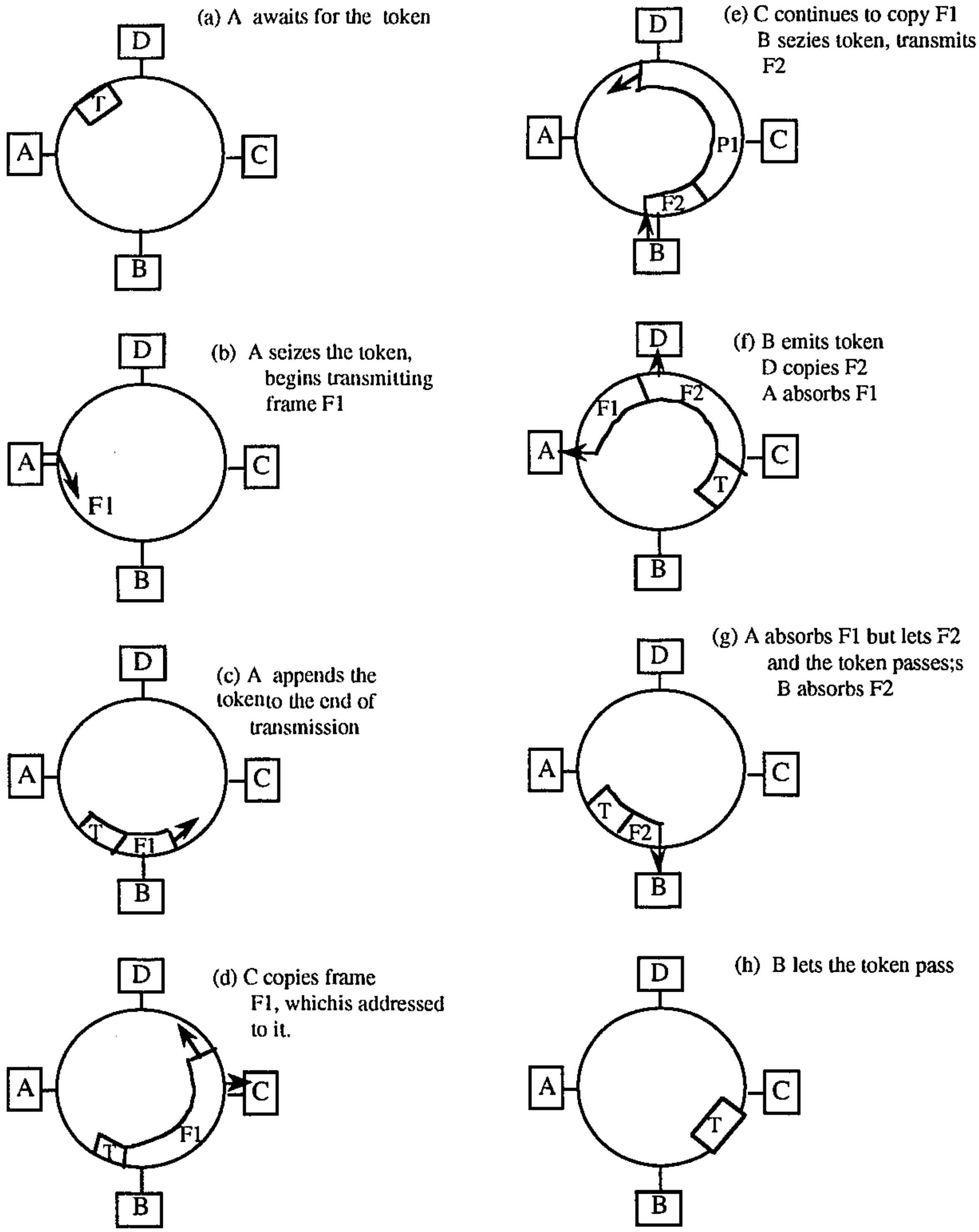


Figure 2.3 FDDI Token Ring Operation

The status indicators (E,A,C) in the ending delimiter are examined to determine the result of the transmission. If there is an error MAC does not re-transmit but LLC takes care of it.

2.4.1 Bandwidth Allocation

The priority scheme used in 802.5 will not work in FDDI, as a station will often issue a token before its own transmitted packet returns. Hence, the use of a reservation field is not effective. Furthermore, FDDI is intended to provide more control over bandwidth allocation to meet the requirements of a high speed network. Specifically, FDDI's bandwidth allocation scheme has two requirements:

- Support for a mixture of stream and busty traffic
- Support for multipacket dialogue

To accommodate the first requirement, FDDI defines two types of traffic: synchronous and asynchronous. The scheme works as following. A Target Token Rotation Time (TTRT) which is the minimum among the response times required by all stations, is stored in each station. Some or all the stations may be provided a synchronous bandwidth allocation (S_i), which may vary among stations. The bandwidth allocation must be such that TTRT satisfies the following equation:

$$t_i + prop\_t + trans\_p + overrun <= TTRT \qquad (2.1)$$

where

t_i = time to transmit synchronous traffic on station i

prop_t = propagation time for one complete circuit of the ring

trans_p = time required to transmit a token

overrun = time required to transmit one maximum length asynchronous packet

Whenever a station receives a token it measures the time since it last received a token, which is counted in a token rotation timer (TRT). This value is saved in the token holding time (THT). TRT is reset to zero and begins counting again. The station can transmit according to the following rules:

- It will transmit synchronous packets for time s. In this thesis FDDI BW has been allocated such as to transmit only one video packet corresponding to synchronous transmission of that station.
- After transmitting synchronous packets or if there were no synchronous packets to transmit, THT is enabled and begins to run from its set value. The station may transmit asynchronous data only as long as THT < TTRT.

It has been proved that for FDDI timed token protocol, the maximum token rotation time can't be more than 2*TTRT [8].This property of FDDI protocol is used to set the TTRT of the network. In this thesis, VBR video is considered as synchronous traffic. To provide pure synchronous services to video TTRT is set half of the packetization time of video packet so that maximum token rotation time is never more than packetization time of video packet.

Asynchronous traffic can be further divided into eight levels of priority. Each station has a set of eight levels of thresholds, T_Pr(1),......,T_Pr(8), such that T_Pr(i)= maximum time that a token can take to circulate and still permit priority i packets to be transmitted. The maximum value of T_Pr(i) must be no longer than TTRT. As we have only one type of asynchronous packets , therefore all the asynchronous packets have the same priority.

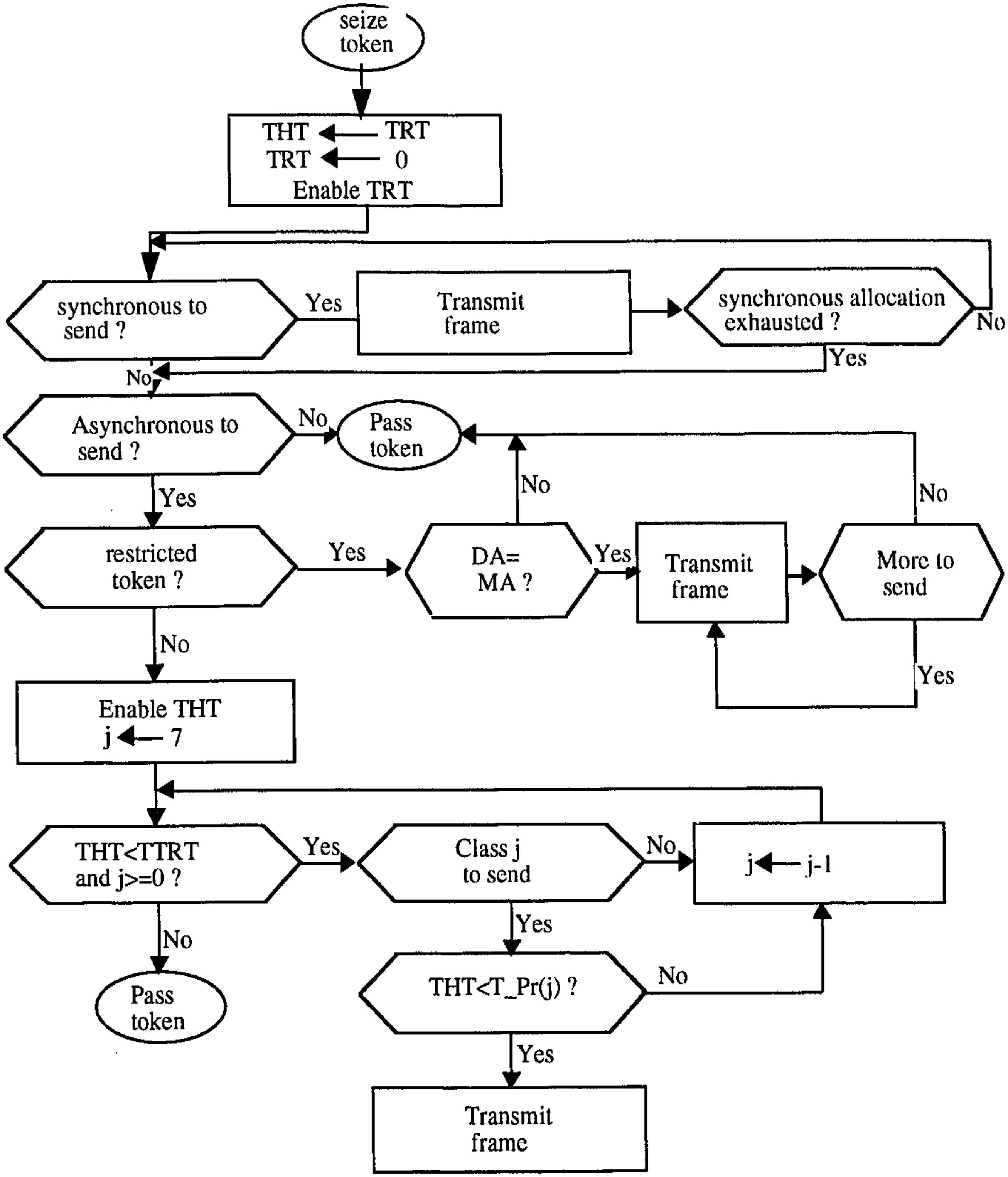


Figure 2.4 Bandwidth Allocation Scheme

In addition, the FDDI mechanism provides a mechanism which satisfies the requirement for dedicated multipacket traffic. When a station wishes to enter an extended dialogue, it may gain control of all the unallocated bandwidth by using a restricted token. The station catches a non restricted token, transmits the first packet of the dialogue to the destination station and then issues a restricted token. The two stations may then exchange data packets and restricted tokens for an extended period, during which no other station may transmit asynchronous packets. The standard specifies that restricted transmissions must not violate the TTRT limitation, but it does not require the use of THT during this mode [1] [2]. In this thesis non-restricted mode is considered for asynchronous traffic.

Figure 2.4 depicts the bandwidth allocation scheme and Figure 2.5 summarizes the relation among various types of bandwidth allocation schemes.

FDDI ring monitoring processes for error correction can be found in the FDDI standard[1] [2].

2.4.2 Priorities

According to the FDDI standard, there are eight levels of priorities for asynchronous traffic [1][2]. There are two modes of token release for asynchronous traffic : restricted token mode and non-restricted token mode. Since our study supports only one type of asynchronous traffic, the traffic is given the highest priority level. As no extended dialogue is required between two stations for asynchronous transmission, the non-restricted mode is chosen for asynchronous traffic. The FDDI model considered in this thesis is restricted to the functions of the source station media access layer (MAC) to the destination station MAC layer. Upper layer functions are not considered.

2.5 FDDI Station and Connection Types

A FDDI network uses a double counter - rotating ring topology [9] , or a "dual ring of trees" with stations characterized by their attachment types and number of MACs. A dual attachment station (DAS) has two physical attachments, one to each ring of the dual trunk. ring, and one or two MACs (single or dual MAC). Higher end stations, servers and routers, will generally be DAS stations. Stations that have only one physical attachment (Single Attachment Stations: SAS) are connected to one of the dual rings by stations called concentrators. Lower end desktop workstations and personal computers are likely to be FDDI SAS stations. In Figure 2.6 nodes 1 and 2 are DASs and node 3 is a concentrator providing attachment to the rings of nodes 5 and 6, which are SASs and node 4, even though it is a DAS must perform as a SAS as its attachment by concentrator is affected. The operation of the second PHY of node 4 is allowed on another ring, but there can not be any interconnection of two rings at the level visible to MAC or PHY.

There are two possible ways to use FDDI:

- For back-end applications, computers which are directly connected to FDDI networks take advantages of the high throughput offered by FDDI in order to transfer data at high-speed over the network. These applications are expected to be heavily used in the next years .
- For back-bone applications, computers which are connected to low-speed LANs such as Ethernet and Token Ring communicate using FDDI as the backbone for connection to LANs. Examples of interconnections of LANs by FDDI are at INRIA France one of the big computer companies using 20 Ethernet networks connected through FDDI and at Aachen University in Germany which has a MAN of 21 FDDI LANs connected through optical fibre [10].

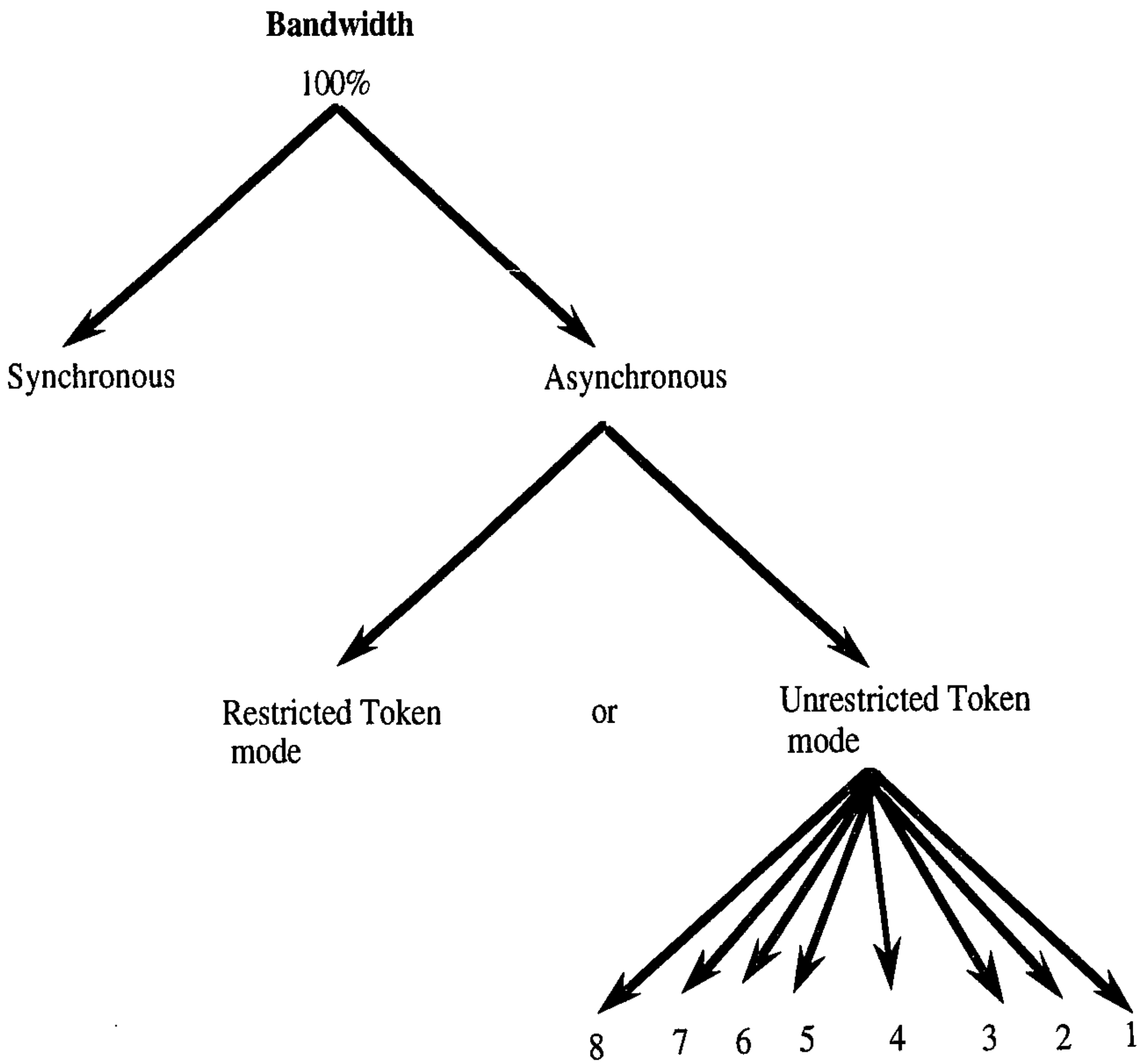


Figure 2.5 Synchronous vs. Asynchronous

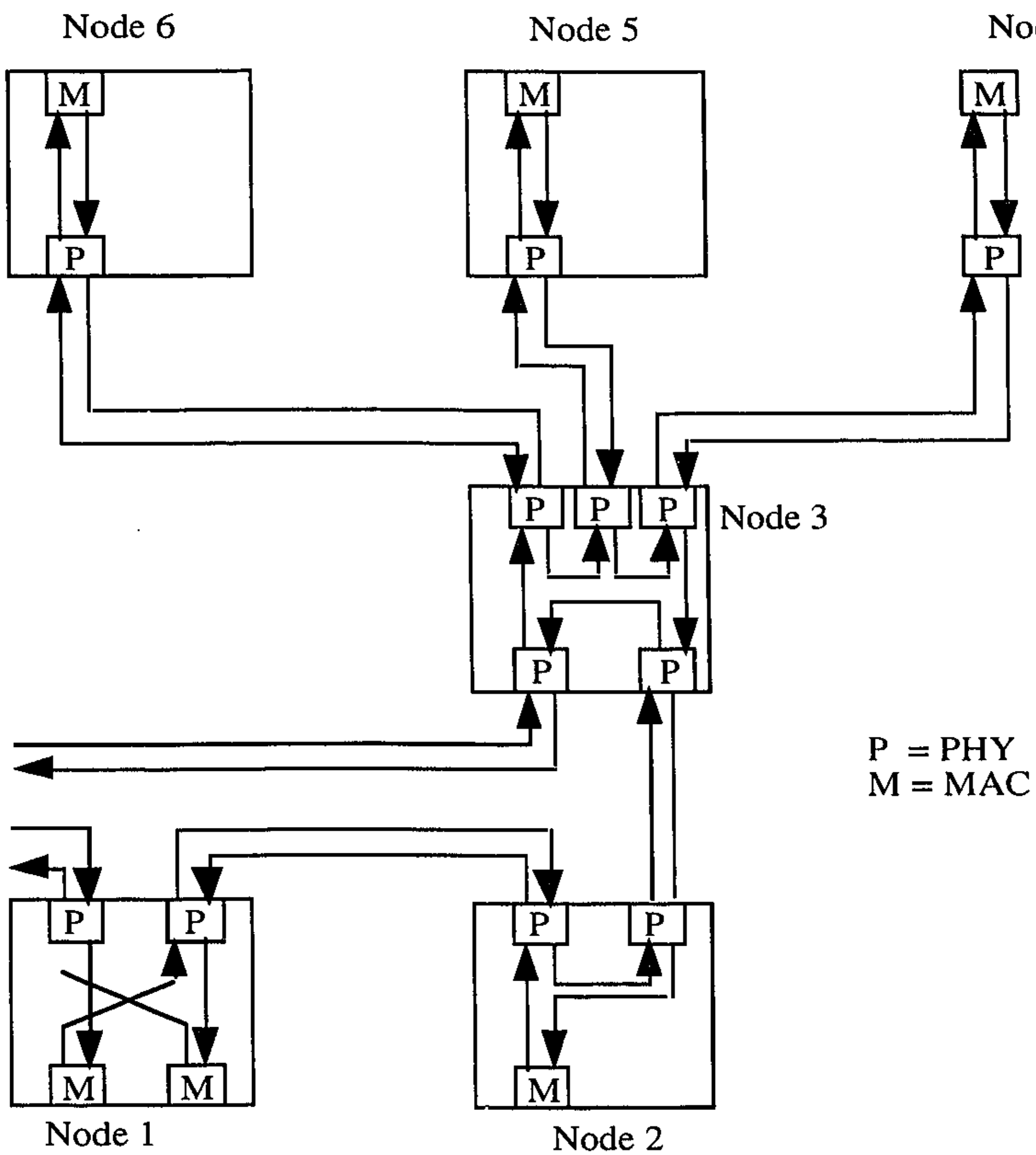


Figure 2.6 FDDI Station Types and Connections

2.6 Comparison of FDDI with Token Ring

FDDI uses an extended version of the IEEE 802.5 Release After Reception (RAT) token ring protocol. In the token ring, a small token frame (packet) circulates around the ring. A station wishing to transmit must wait until it detects a free token passing by. It then seizes the token as soon as the usable token is recognized, transforms the free token into a busy one by simply setting a special bit in the token, called the token bit, to one.

It then starts transmitting one or more packets by attaching packet to the busy token called the SFD of frame until either its queue is empty or a special timer called token holding time (THT) expires. This period is fixed for each station at initialization time . Since there is no other token on the ring, other stations wishing to transmit must wait. The packet on the ring will make a round trip and be purged by the transmitting station. The transmitting station inserts a new token on the ring when its THT timer expires. Such an operation, guaranteeing the existence of only one token on the ring, is called single-token operation.

From the above token operation some differences can be observed between the FDDI and the token ring. First, the FDDI station does not set the token bit of the received token. The station just catches the token into it and start transmitting packets. Second, in FDDI, a station releases a new token as soon as the station completed the transmission, even if it has not begun to receive its own transmission. This difference is only true for Release After Reception (RAR) token ring. The other difference is the transmission of asynchronous packets. The token ring does not differentiate between synchronous and asynchronous packets at the time of transmission. It just transmits its packets for time THT. But in FDDI , the transmission of asynchronous packets depends upon the actual token rotation time. If THT is positive then the station can transmit asynchronous packets, otherwise not.

Chapter 3

Modeling of FDDI

3.1 Introduction

This chapter is divided into two parts. The first part gives an overview of the previous work done on FDDI and the second part describes the simulation model of FDDI for multimedia communications. We will use this basic model for the different scenarios studied in this thesis. The first scenario simulates only VBR video traffic on FDDI. The second one integrates asynchronous traffic with VBR video (File Transfer). The third version simulates the VBR video integrated with voice sources without any clipping on voice sources. Finally the fourth scenario includes the clipping on voice sources with VBR video and voice sources both present on FDDI.

3.2 Previous Work

From the local area network point of view, FDDI employs an optical fibre medium technology. FDDI was developed by the Accredited Standards Committee (ASC) X3T9 - charted to develop computer input/output (I/O) interface standards in 1983 [9]. FDDI uses the packet format of IEEE 802 and an extended version of IEEE 802.5 token ring protocol.

The priority scheme of IEEE 802.4 Token Bus, which was selected for FDDI standard, has been studied by Danthine [11]. Danthine calculated performance parameters such as mean delay to asynchronous traffic with different priorities levels. Performance analysis of FDDI 100 Mbps optical token ring was conducted by A. Schill [12], T. Welzel [13], R.O. LaMaire [14] and P. Martini [15]. Schill investigated the behavior of timed token protocol with respect to its controlling timers and the ring topology. He also evaluated the effectiveness and sensitivity of the FDDI priority mechanism. All of his studies were based on simulations. LaMaire discussed the importance of delay analysis in the FDDI network and developed an approximation method for delay investigation. Green studied the performance of analysis of FDDI [16]. He calculated the fundamental parameters of FDDI for network design . Th. Welzel and S. Rudloff studied the analysis of FDDI and token ring in a mixed traffic environment [17]. They considered typical file transfer and control data as asynchronous traffic on the network. They studied FDDI and Broadsite Wideband Network (BWN) in a backbone environment. They studied the operation of both protocols with respect to several aspects e.g. percentage of token use or number of packets transmitted per token capture. They found that both media access protocols work well for number of stations equal or less than 25 in the asynchronous mixed traffic environment. At heavy load FDDI performed better due to multiple service capability and reduced overhead especially for large value of TTRT.

The throughput performance of FDDI has been analyzed by Dykeman and Bux [18]. They did not considered synchronous traffic for their study. They focused on the effects of the protocol parameters and the physical ring characteristics on the throughput behavior of asynchronous priority class. First they developed an equation for the total throughput of FDDI when only one asynchronous priority level is used. They found that the maximum throughput remains high as long as the TTRT is large with respect to the ring latency. In the second part of their study, they provided a procedure to calculate estimates for the throughput of multiple priority levels over a range of arrival rates. They also proposed a procedure useful to network manager, to set the parameters of FDDI for a given performance objective defined either in terms of peak throughput or the required throughput of each asynchronous priority level. P. Anura and N. Priya studied the performance of FDDI for multiple classes of traffic [19]. They proposed an analytical model to evaluate the throughput of FDDI in the presence of multiple classes of traffic. The main advantage of this model compared to Dykeman's study is to calculate the throughput of the different priority levels. The simulation study conducted by Pang and Tobagi [20] , was the performance study of timed token bus (802.4) which used a protocol similar to that of FDDI. Another study with respect to the traffic characteristics was done by N. Wainwright and A. Myles [21]. They studied different network sizes under various traffic conditions and compared the delay characteristics of the FDDI and the IEEE 802.6 MAN network-access mechanisms. The authors concluded that the maximum throughput of both networks is adversely affected by the presence of short packets and this effect is much more dominating in the IEEE 802.6 network. The actual measurement and experiments of a FDDI-installation was done by P. David, T. Measer and O. Spaniol at the Aachen University of Technology [10]. They commented that while most research contributions focus on high load in FDDI or DQDB and fairness problems under heavy load, realistic situations call for more attention to lower and intermediate load ranges. A comparison study of MAC protocols of CSMA/CD, Token Bus, Token Ring and Slotted Ring for use with high speed LAN as FDDI is given by Danthine [22]. He found that under heavy load performance of CSMA/CD is very poor compared to Token Ring.

Raj Jain [23] studied the effect of parameters of FDDI network's performance. The metrics used by as measurements are: productivity and responsiveness. Productivity for FDDI is measured by its throughput while responsiveness is measured by the response time and access delay. Jain assumed "bursty Poisson" arrival for his simulation model . He found that the Target Token Rotation Time (TTRT) is the key parameter to optimize the performance of FDDI when only bursty asynchronous traffic is present. He also found that other parameters that effect the performance of FDDI are length of cable, total number of stations, number of active stations and frame size.

Watson [24] investigated packetized voice on FDDI. He treated voice as synchronous traffic. He considered asynchronous traffic in such a way that the network is always heavily loaded. Watson did not use silence detection for voice communications. Watson first presented an analysis of maximum number of voice conversations that can be supported on FDDI. In a simulation study he showed that a much greater number of calls can be supported by accepting some packet loss.

Although FDDI has been considered as high speed network, no work has been done related to multimedia communications on FDDI. This thesis studies multimedia communications on FDDI. The following section describes the simulation model for supporting multimedia communications on FDDI.

3.3 Simulation Model of FDDI

There are several studies analyzing the performance of token rings. A concise summary of those efforts can be found in [25]. Although FDDI belongs to this category of token rings, the peculiarities of the timed token protocol pose significant difficulties in performance evaluation. In fact, the timed characteristics of protocol creates some dependencies among transmissions of various stations and these dependencies complicate the analysis of protocol performance.

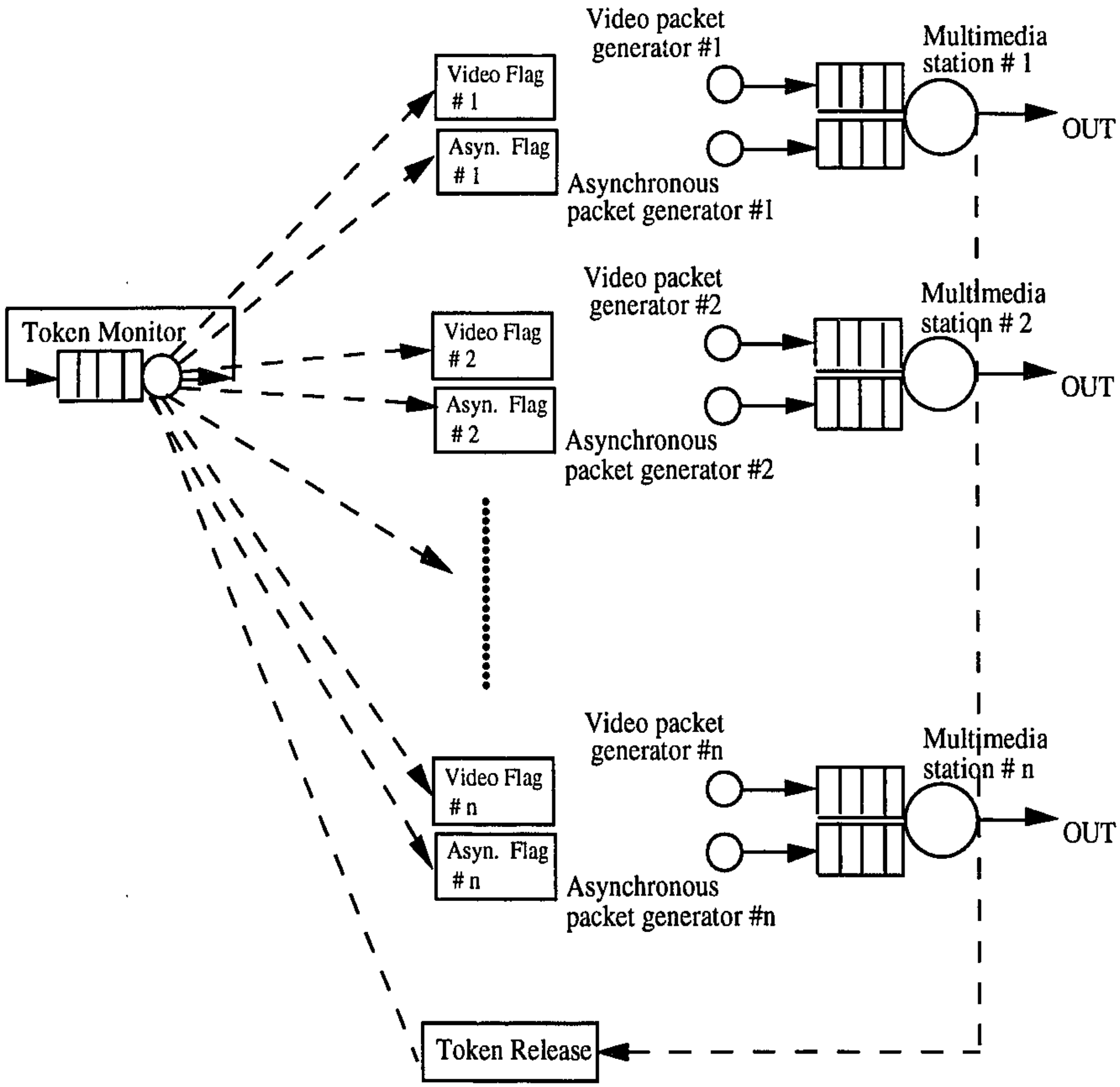
Ulm [26] tried to analyze a similar timed protocol and to give some performance figures. He also investigated the dependence of some performance measures on various design parameters. There are other simulation studies such as the one used by A. Sastry which only simulated asynchronous traffic on FDDI [27].

The following is a simulation model for FDDI supporting multimedia services. The simulation program was built using the QNAP2 (Queuing Network Analysis Program). This tool provides powerful features for modeling and evaluation of queuing networks. Different processes can be synchronized in QNAP2 by flags. Packets are routed through a system of stations consisting of servers and related queues according to the predefined transition rules and specific processing times.

The complete QNAP2 model of FDDI is shown in figure 3.1. Each FDDI station supports synchronous as well as asynchronous traffic. To store synchronous and asynchronous packets separately, each station is modeled as having two queues. Figure 3.1 consists of the following parts:

- Video Packet Generators: serves as input to video queue at each multimedia station
- Asynchronous Packet Generators : sends the packets to asynchronous traffic queue
- Stations are modeled as having two queues: The FDDI protocol is implemented in the two queues forming one FDDI station. Video packet are stored in the synchronous queue and asynchronous packet are stored in asynchronous queue.
- The Token Monitor Station : controls the circulation of the token.

The token monitor station is a fictious station that serves as the medium access control. In FDDI, each station is given permission for transmission in a round robin fashion. This, in our model, is achieved by setting flags. There are two flags for each station, one for the video queue and other for the asynchronous traffic queue. Upon polling the token monitor first checks the video queue. If there are video packets, then the flag for the video queue is set and the token holding time for asynchronous traffic is calculated. Only one video packet (although there may be more than one video packet) is served. After serving one video packet, the global flag of token monitor station is set. If the asynchronous queue of the station has asynchronous packets to transmit then the flag associated to the asynchronous queue is set and asynchronous packets are served depending upon the value of token holding time. The token holding time is calculated in the video queue for each FDDI station. If there were no video packets then the token holding time is calculated when the flag of the asynchronous queue is set. After the THT



No token propagation and transmission delay from video queue to asynchronous queue

Token propagation and transmission delay from one station to next station

Figure 3.1 The Queuing Model of FDDI Supporting Multimedia Traffic

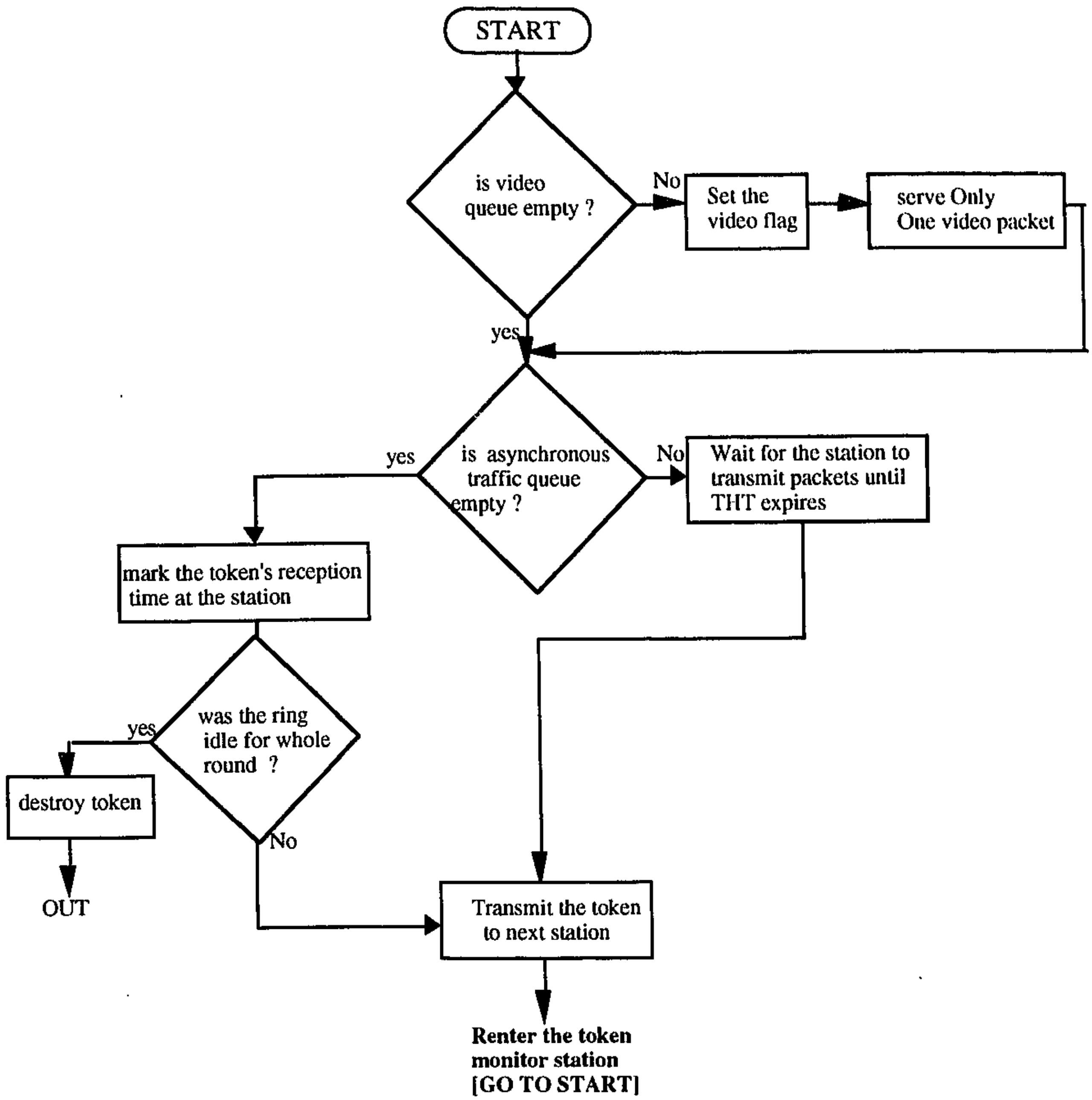


Figure 3.2 Logic Diagram of Token Monitor Station

of the asynchronous queue expires, no more asynchronous packets are transmitted and the token moves to next station. The token propagation and transmission time is considered from a station to the next station .

Finally, there is another function incorporated in our token monitor service routine, having been adopted from Schill [12]. Whenever the ring is sensed idle (there are no packets in any of the two queues at each station) for more than a complete token circulation, the token is destroyed. This is done to limit the actual simulation time. There will be a new token generated whenever a packet will be generated in one of the stations in order to have proper operation of the ring. For this reason , the first newly generated customer will also generate a token and then the token monitor station will begin its polling from a station selected at random. Figure 3.2 gives the logic of token monitor station.

Table 3.1 Parameters of simulation

Station to station distance	200 meters
Propagation Time	5.085 ms/Km
Station Latency	600 ns
Synchronous Traffic	One Video packet
Video packet overhead	224 bits
Asynchronous packet overhead	160 bits
Token length	88 bits
Asynchronous packet length (File Transfer+control)	426 bytes
Video packet length (average) (2 ms)	8000 bits

From the model discussed above , we derived three simulation models. First , we investigate only VBR video on FDDI. In this case , video queue will have packets and there will be no packets in the asynchronous queues. The second model will have video and data traffic. The third model considers video and voice where voice packets are stored in the asynchronous queue. In the fourth model, for multimedia traffic of video/voice , the bit dropping algorithm was also investigated on voice. In this case when the number of packets in a asynchronous queue increases beyond a limit, some bits from the transmitted packet are chopped.

Table 3.1 gives the FDDI simulation parameters. The length of video packet is variable. TTRT is set according to the quality of service requirements and is explained in the following chapters.

Chapter 4

Multimedia Integration of Variable Bit RateVideo and Data Over FDDI Networks

4.1 Introduction

Recent developments in the area of digital video permit us to predict the wide use of computer based pictorial information. Standards organizations are currently carrying out efforts to define video coding standards [28]. A wide variety of coding schemes are being suggested to support many different applications. Target applications, such as videophone, videoconferencing and full-motion video are mainly being considered .

Once the development of specialized video processors and computer platforms supporting video gets underway, the main challenge will be the design of wide area distributed multimedia systems. These systems will integrate various types of media such as text, voice, image and video. The development of such distributed systems will require the use of communication services capable of coping with the diverse communication needs of the various media. Therefore, one of the first tasks towards the design of this kind of systems is the assessment of communication services which may be provided by the high speed communication networks, and particularly those to be provided by the integrated services networks. Among the recent developments in this area, the ISO MAN standard FDDI (Fibre Distributed Data Interface) has a special place [1][2].

In this chapter, we study the integration of variable bit rate (VBR) video and data over FDDI networks. This study aims to assess the suitability of FDDI for supporting multimedia (video/data) applications.

4.2 Variable Bit Rate Video Source Modeling

This section focuses on interframe coding with conditional replenishment which has been experimentally used to code video phone scenes [29]. This coding scheme encodes the significant difference of pixels between successive frames of a moving sequence. The bit rate of such coding increases during periods of high activity and decreases during periods of low activity.

4.2.1 Why VBR Coding

There is always a complex tradeoff between the minimum achievable coding rate, R, and the distortion of the decoded pictures [30]. The bit rate (in bits/pixel) of a source determines the maximum compression achievable for lossless (D=0) coding. For moving images the rate will be time varying depending upon the instantaneous activity of the scene. Figure 4.1 shows the variable bit rate coding requirement to have uniform quality of the picture at the receiver. If some distortion, D_0, can be tolerated, the coding bit rate can be decreased, but it will remain time varying for constant quality video output as shown by the segment AB in Figure 4.1. Similarly if the coding bit rate is constant, equal to R_0, the video quality will be varying, worse for high activity scenes as shown by the segment CD of Figure 4.1.

4.2.2 Interframe Coding with Conditional Replenishment

Interframe coding uses frame to frame redundancy. It uses temporal correlations that exist at a delay of exactly one frame period. The first frame of the scene is fully transmitted and is stored at both the transmitter and the receiver ends.

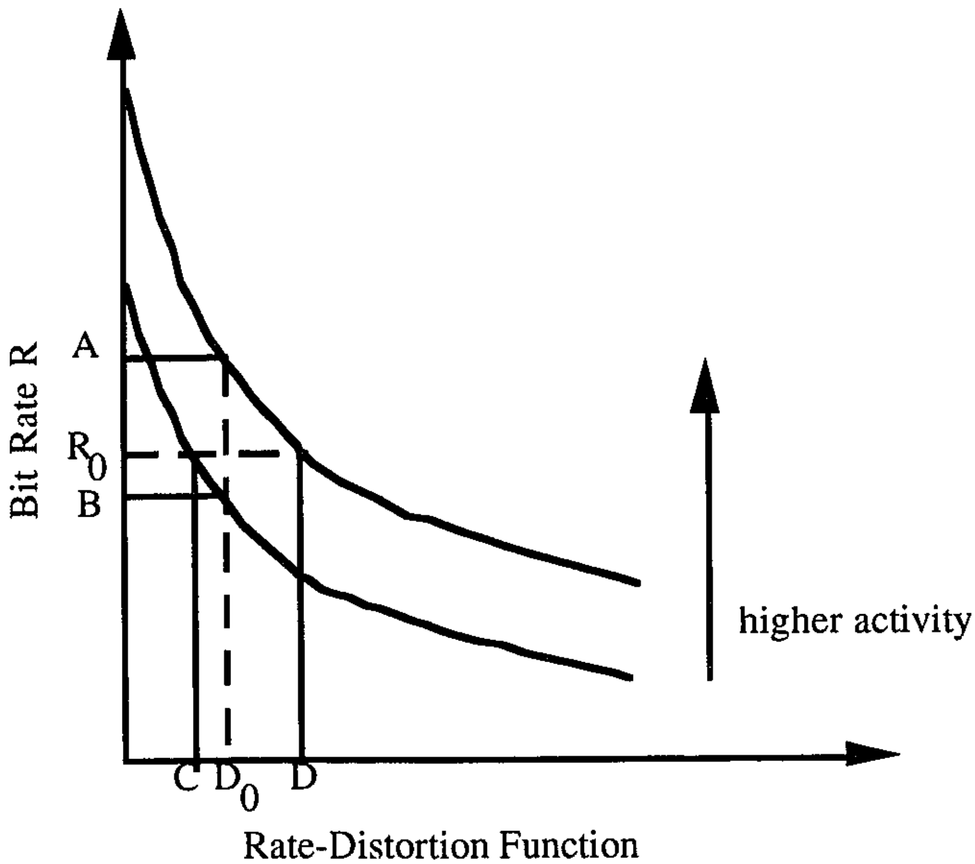


Figure 4.1 Bit Rate vs. Quality of Picture (Distortion Function)

The next frame is predicted from the original stored frame. Therefore, for a frame size of N x N, a storage size of N^2 pixels is required. The information for the next frame is the difference between the stored frame and predicted one. Usually this information is motion information and change in pixels. The motion information tells how much movement has occurred in any given block of the original frame. At the receiver end, the next frame is generated by using the difference signal. The simplest interframe coder for moving images is a frame-to-frame DPCM coder as shown in Figure 4.2.

The difference signal is the difference between pixel, x_A ,to be coded and the corresponding quantized pixel, y_{AF} ,that is one frame or N^2 pixels away:

$$d = x_A - y_{AF} \qquad (4.1)$$

Conditional Replenishment Coders: In these interframe coders, transmission capacity is used only to replenish those pixels which have changed significantly since the previous frame. Reconstruction values, y_A ,for unchanged pixels, for which $|x_A - y_{AF}|$ is less than threshold, are simply set equal to corresponding y_{AF}. The coding of changed pixels is based on quantizing and transmitting a difference signal, using the interframe coder given by Equation 4.1. Typical results with such techniques have been the encoding of low activity videophone using the average bit rate R in the range of 1 bit/pixel and even in the range of 0.1 to 0.05 bits per pixel for images where some blurring is acceptable.

Conditional replenishment systems need to transmit addressing information to tell the DPCM receiver which of the (N x N) pixels in a frame are active and thus in need of replenishment. Therefore, the amount of the side information increases with image activity . However, the amount of side information can be reduced by *cluster coding* in which changed pixels tend to occur in clusters or runs. Therefore economics in side information results by signaling only the beginning and ending addresses of each cluster or equivalently the beginning address and cluster length.

4.2.3 Experimental Data Using Interframe Coding From Video-Phone Scene

In order to investigate the performance of a multiplexer for VBR video sources, Maglaris collected VBR data from video phone scene [29]. This data was collected from a videophone scene without any abrupt motion using interframe coding with conditional replenishment scheme as explained earlier. The duration of the scene was 10 seconds (300 frames). The measured data was instantaneous bit rate $\lambda(t)$ in bits per pixel.

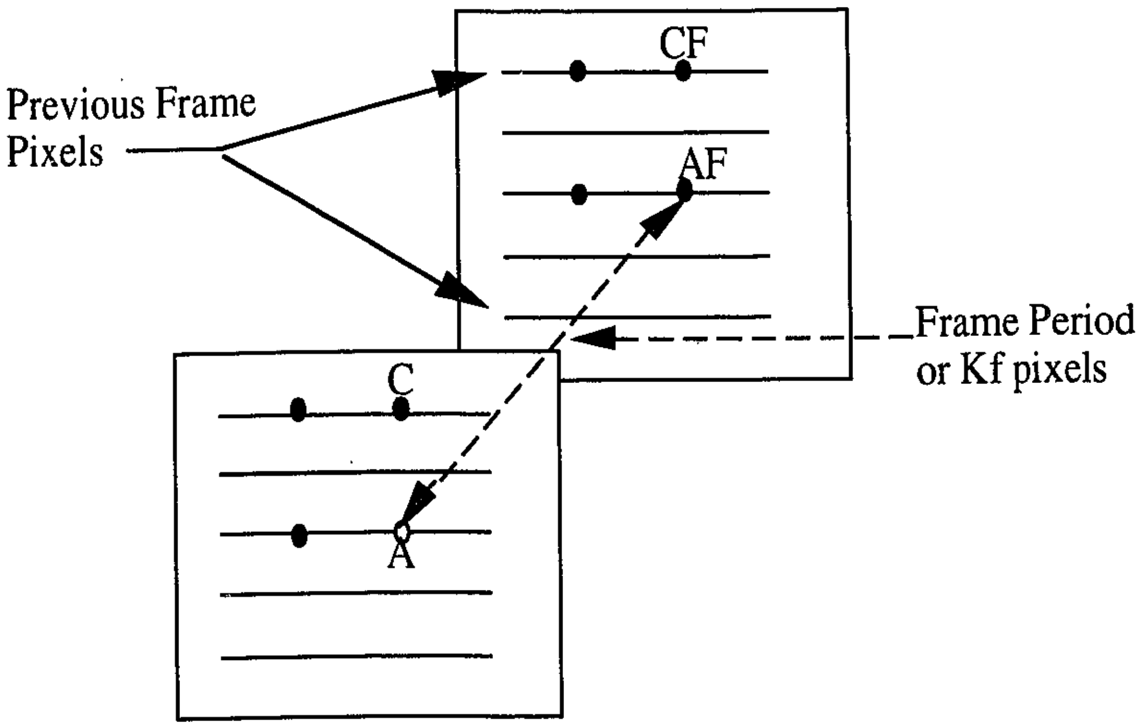


Figure 4.2 Pixel Notation for Frame to Frame Differential Coding

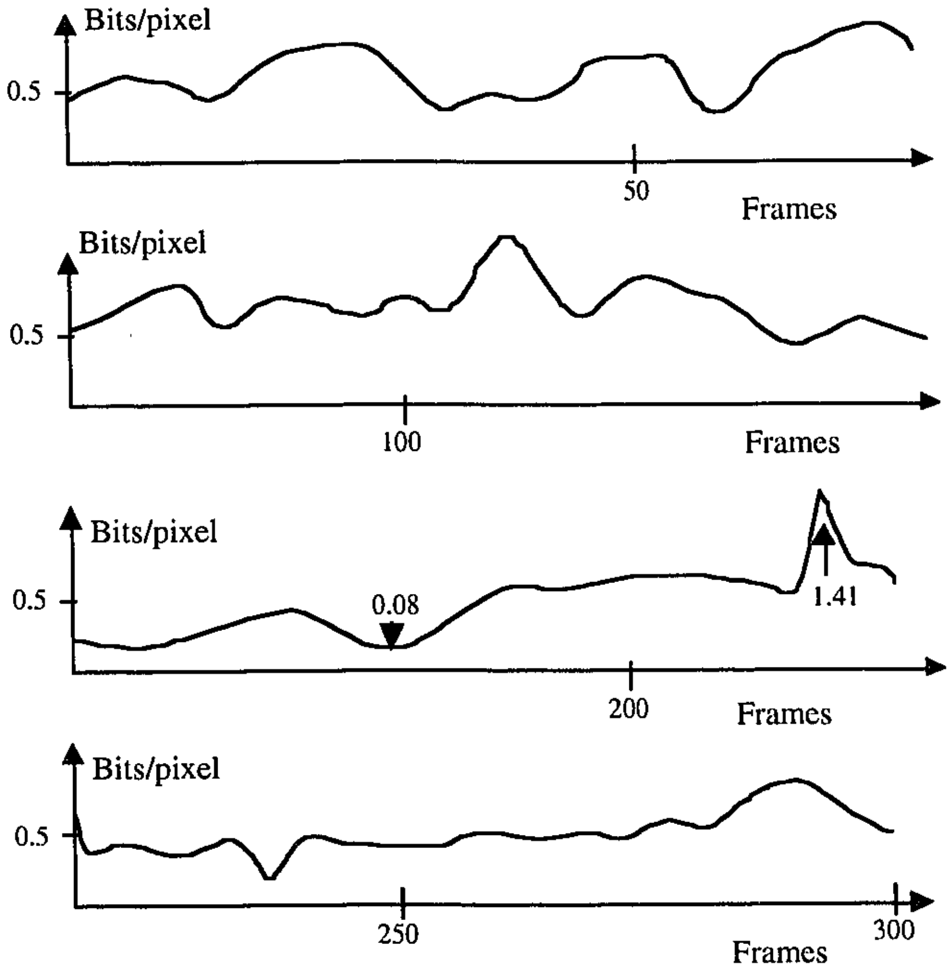


Figure 4.3 Coding Bit Rate of the Captured Sequence (bits/pixel)

Figure 4.3 shows the bit rate (averaged for each frame) for all 300 frames (10 sec) of the test sequence. This data is used in this thesis for modeling variable bit rate video.
The key parameter in evaluating the bit rate is the number of pixels, N_p ,which combined in a group give rise to a coded bit stream of length, N_b ,into the buffer. The average bit rate is then equal to N_b/N_p bits/pixel. A large N_p leads to more averaging in the buffer, but it needs a larger buffer and introduces longer delays. A small N_p values preserves the variations of individual source rates. On the other hand, a too small value of N_p will introduce quasi-periodic variations with the period equal to one frame duration. This can be avoided by using prebuffering techniques which introduce a delay of one frame. Therefore the number of pixels chosen for calculating the bit rate are the number of pixels in one frame. This choice makes $\lambda(t)$ dependent only on the varying activity of the frame sequence. The average value, μ ,of $\lambda(t)$ over 300 frames is 0.52 bits per pixel and standard deviation,σ, is 0.23 bits/pixel. The maximum value of bit rate is 1.41 bits/pixel and minimum is 0.08 bits/pixel.

Figure 4.4 shows the probability distribution function of bit rate $\lambda(t)$, which can be approximated by a truncated Gaussian distiibution. It is also noticeable that bit rate can not be negative and it is not exactly symmetric around the average value μ.

The autocovariance function $C(\tau) = E\{\lambda(t)\ \lambda(t+\tau)\} - \mu^2$ of the captured scene is found to be matching to an exponential form given by $C'(\tau)= \sigma^2\ e^{-a\tau}$ with $a=3.9\ sec^{-1}$. There is a slight difference between the autocovariance function of the experimental data and exponential fit due to rounding by the computer. From this experimental data, two models were derived for the bit rate of the video source. The models have the same mean value and autocovariance function as that of the experimental data. We are presenting the model suitable for the simulation which is used in this thesis [29].

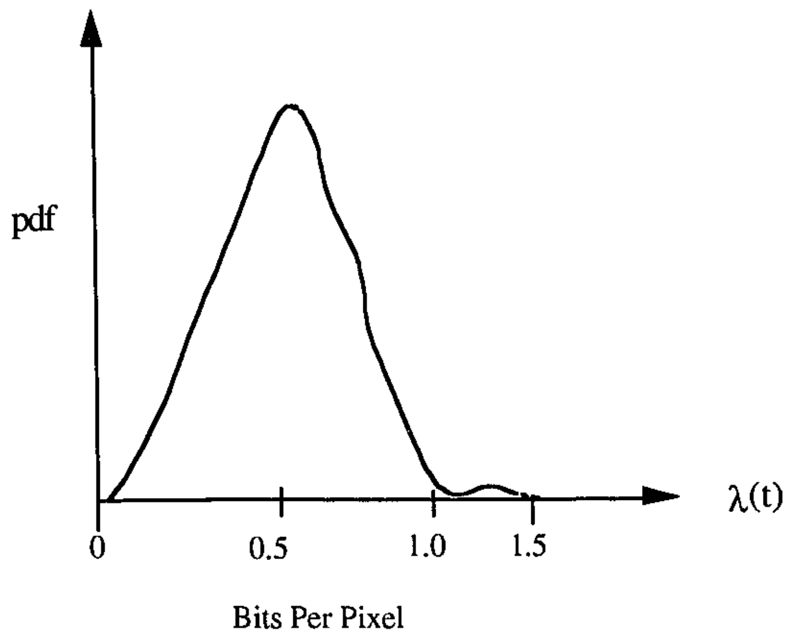


Figure 4.4 Probability Distribution of Bits Per Pixel

4.2.4 Mathematical Model for Variable Bit Rate of a Video Source

Video source bit rate, λ ,can be modeled as a continuous state - discrete time stochastic process. Let $\lambda(n)$ represents the bit rate of a single source during the n^{th} frame and is given by a first order auto regressive Markov process defined by the following recursive relation:

$$\lambda(n) = a\ \lambda(n-1) + b\ w(n) \qquad (4.2)$$

where $\lambda(n-1)$ is the bit rate of the previous frame, a and b are constants and $w(n)$ is a sequence of Gaussian random variables. Assume that $w(n)$ has a mean value η and variance 1. Further assume that $|a| < 1$; thus, the process achieves steady state with large n.

For the process described by Equation 4.2, according to probability theory [31], the average value and autocovariance function, are given by:

$$E[\lambda] = \frac{b}{a}\eta \tag{4.3}$$

$$C(n) = \frac{b^2}{1-a^2}a^n \tag{4.4}$$

The autocovariance function is exponential and can fit the experimental data. Therefore, there is a reasonable match of experimental data with the auto regressive model given by Equation 4.1. From the experimental data for a video phone scene of 10 seconds, the average value and autocovariance function of coded bit rate are:

$$E[\lambda] = 0.52 \quad \text{(bits/pixel)} \tag{4.5}$$

$$C(n) = 0.0536\, e^{-0.13n} \quad \text{(bits/pixel)}^2 \tag{4.6}$$

From Equations 4.2, 4.3 ,4.4 and 4.5 we obtain a = 0.8781, b = 0.1108 and η = 0.572. In the following, we use these results for modeling the VBR video sources.

4.3 Asynchronous Traffic Modeling

In this study we consider a scenario comprising file transfer and interactive traffic components for asynchronous traffic as used by Th.Welzel [17]. He assumed a performance requirement of 2 Mbps average throughput at each station. It was also assumed that with such values LAN is heavily loaded and throughput is split between file transfer and interactive traffic. Welzel also assumed that 90% of the load is offered by file transfer. Here for asynchronous traffic, we use all the same values and assumptions as used by Th. Welzel.

File transfer appears to be the most important application in high speed networks. But in practice before transmitting a file, some control information (interactive traffic) needs to be transmitted. Therefore in our study, we have considered asynchronous traffic which is a mixture of file transfer and control information. File transfer can be modeled as a bursty arrival process. During a burst, packets are transmitted one after another followed by silence period after the transmission of the last packet of the burst. Consecutive files are separated by a relatively long inter gap. The interarrival time between two files is assumed to be exponentially distributed with the mean value according to an average throughput of 1.8 Mbps. To transmit packets between two stations some control information is also required, such as handshaking, which we refer to as interactive traffic. The interarrival times of interactive traffic is also assumed to be exponentially distributed with the mean value chosen to satisfy 200 Kbps data rate. Combined asynchronous traffic model is shown in Figure 4.5. The mean length of the information field of a packet is 426.4 bytes and interarrival time is exponentially distributed with a mean value of 1.476 ms to have average data access rate of 2.0 Mbps for asynchronous traffic.

4.4 Integrated Services Over FDDI

Figure 4.6 shows the system configuration under study. The system consists of an FDDI network interconnecting a set of multimedia (data/video) stations.

Every station uses a Token Holding Time (THT) to decide when it can release the token. The THT of a station is the difference between the Target Token Rotation Time (TTRT) and the (actual) Token Rotation Time (TRT). The TTRT is initially set for the whole network depending upon the communication requirements of the stations. Once in operation, a station can transmit asynchronous data if its THT is positive. A station is allowed to overrun the time for which it can hold the token if the THT expires while the station is in the process of transmitting an asynchronous packet [1][2].

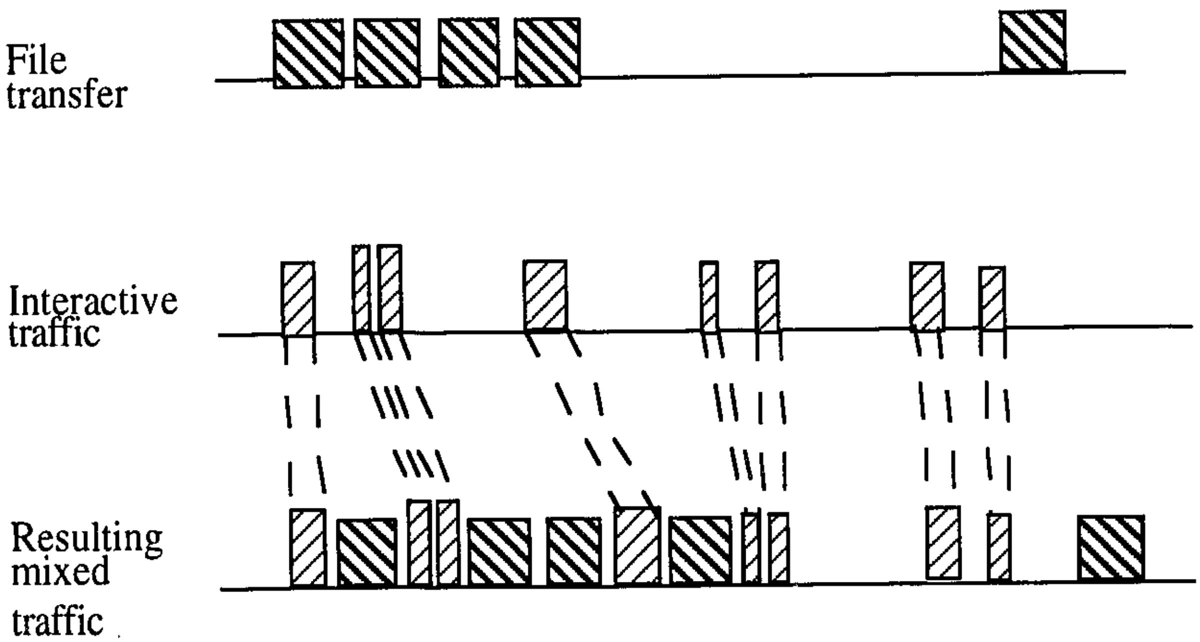


Figure 4.5 Asynchronous Traffic Arrival Process

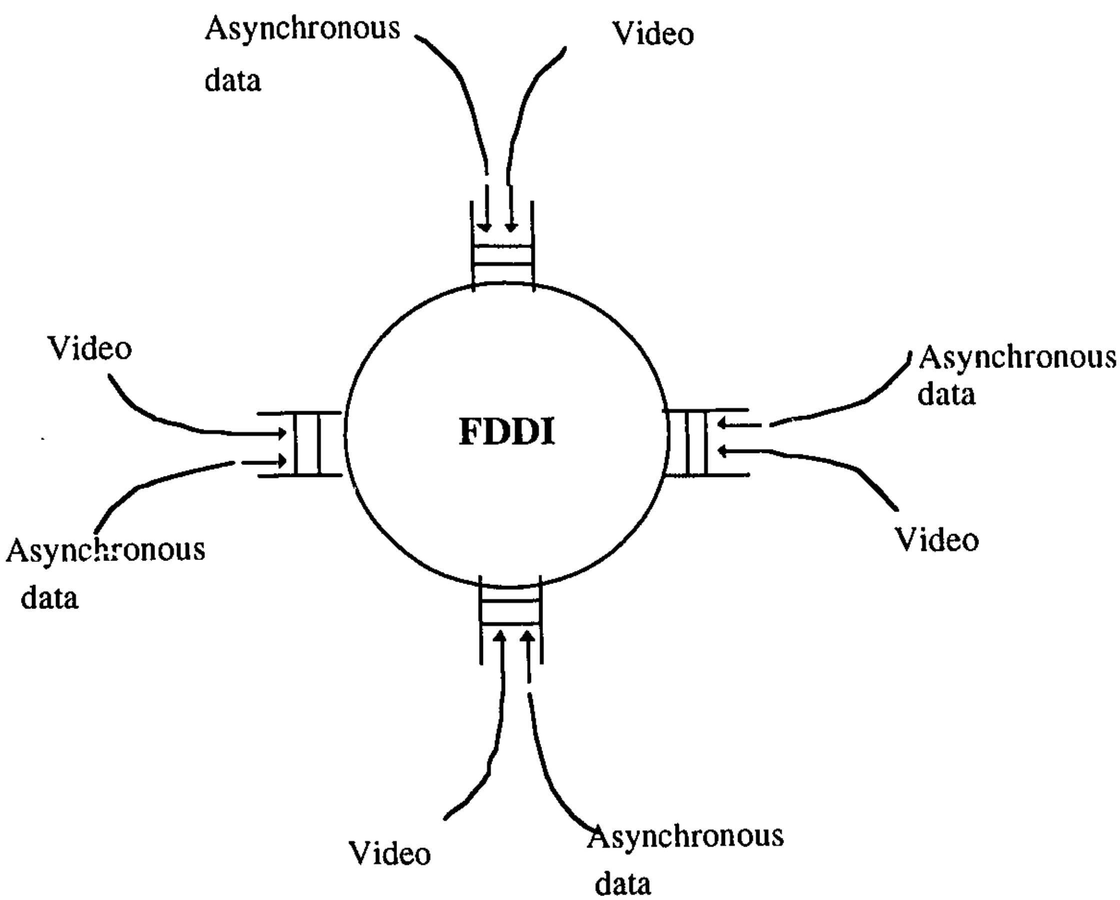


Figure 4.6 FDDI with Integrated Services

4.5 Performance Evaluation

4.5.1 Assumptions

As stated earlier we assume the use of an interframe coding scheme with conditional replenishment for the video signal. The amount of information generated by a video station is 30 frames per second. Each frame consists of approximately 250000 pixels. The amount of data per frame is therefore equal to 250000 times the number of bits per pixel (λ). The amount of data in a frame is variable depending upon the number of bits per pixel (λ).

With respect to the packet length used by FDDI, the packetization time should be set such that the maximum packet length, including the header bits, does not exceed 36000 bits. The maximum packetization time, tpack_max, is the value where the length of packet does not exceed the limit and is given by:

$$tpack\_\max = \frac{MaxPacket\_Length}{\lambda * Ns} \qquad (4.7)$$

where MaxPacket_Length is 36000 bits, N_S is the number of pixels collected per second and λ is peak bit rate (bits/pixel). For λ equal to 1.41 and N_S of 7500000 , tpack_max comes out to be 3.4 ms.

We assumed packetization times of 2.0 and 3.3 ms . The respective values of the number of samples gathered are 15000 and 25000. The maximum packet length with these packtization times is less than the maximum packet length suggested in the FDDI standard [1,2]. The maximum number of overhead bits (header and trailer),224, is considered for each packet as suggested in the FDDI standard. It is further assumed that the token length is 88 bits and every station is 200 meters from its neighboring station. From these latter values, the token propagation, token_p, and transmission time, token_t, are 1.017 µsec and 0.88 µsec respectively.

The amount of bandwidth provided by FDDI for synchronous communication can be simply expressed by:

$$s = \frac{\mathrm{TTRT} - \mathrm{token\_p} - \mathrm{token\_t} - \mathrm{overrun}}{\mathrm{TTRT}} 100\,\mathrm{Mbps} \qquad (4.8)$$

where overrun accounts for the time exceeding the overall THT, i.e. the addition of all the time exceeded by all the stations in a complete token rotation. This may result from the fact that a station may exceed its THT while transmitting in asynchronous mode.

4.5.2 VBR Video Communications

In our study the performance index of interest is the delay to the video packet. The delay to the video packet consists of queuing delay and service delay. This delay is considered from the point where the packet is generated to the point where it is delivered to its destination. The following equation gives the mean delay to video packet in terms of the different delay components.

$$D_{mean} = D_{queue} + D_{trans} \qquad (4.9)$$

where D_{queue} is the queuing delay, D_{trans} is the transmission delay .

The first part of the study deals with the evaluation of the performance of FDDI with only VBR video present without any queuing delay to video packet i.e. providing synchronous service to video. For this purpose, the maximum token rotation time should never exceed the packetization time of the video packet. According to FDDI standard [1,2], maximum token rotation time can go upto 2*TTRT in overrun mode due to the presence of asynchronous traffic. As for the first part there is no asynchronous traffic present, therefore the maximum TRT will be equal to TTRT. As result we set our TTRT at the packetization time of the video packet so that the number of packets waiting to be served will be bounded to one. Following table 4.1 gives the results when there is no queueing delay for the video packet.

From table 4.1 it is clear that with higher value of packetization time we can support more VBR video stations. This can be explained from equation 4.8. From equation 4.8 we see that higher the TTRT more is the synchronous bandwidth available.

But we can increase TTRT upto 1.65 ms (t_pack = 3.3) such that maximum packet length does not exceed the maximum FDDI packet length.

Table 4.1 Video Communication Results

Packetization time of video (ms)	TTRT (ms)	Maximum video packet length	Maximum TRT (ms)	VBR video stations
2 ms	2.0 ms	21150+224	1.98 ms	12
3.3 ms	3.3 ms	35000+224	2.987 ms	16

The next part of the study deals with the evaluation of FDDI when some permissible delay to video is accepted. As we found in previous study that for greater t_pack we can have more video stations, therefore for this part of study we assume t_pack of 3.3 ms. The system under study for this part is shown in figure 4.7. In this figure, x is the number of stations which can be added to the existing system such that the delay to video packet is in permissible limits. As some permissible delay to video is acceptable, therefore TTRT can be more than the packetization time of the video packet. For the system shown in figure 4.7 , the following equation is true.

$$2 \times TTRT = (N + x) trans\_t$$

or

$$x = \frac{2 \times TTRT}{trans\_t} - N \qquad (4.10)$$

where N=16 is the number of stations that can be supported with pure synchronous service to video and trans_t is the transmission time of one video packet. For average and maximum video packet length , the respective values of trans_t are 0.125 ms and 0.352 ms . TTRT was set at 3.66 which is more than the packetization time of the video packet. Putting all these values in equation 4.10, the respective values of x comes out to be 4 and 41.

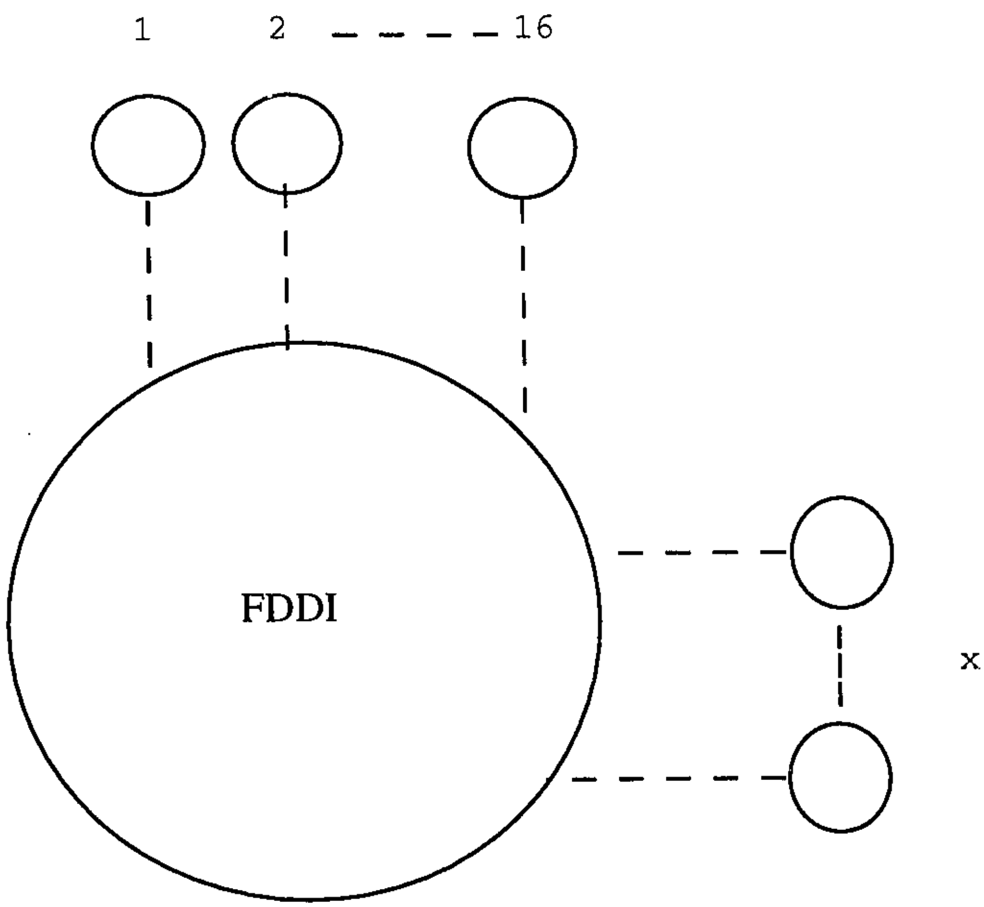


Figure 4.7 System Configuration When Some Delay to Video is Acceptable

But from the simulations, it was found that with TTRT of 3.66 ms, 23 VBR video stations can be supported within the maximum permissible delay of 250 ms .

The difference between simulation and calculated results is due to variable bit rate requirement of video sources. In this case every packet is generated every 3.3 ms and maximum token rotation time can be 2*3.66 =7.3 ms. Therefore the service to video is no longer synchronous and video packets have to wait in the video queue. In this case whenever the token arrives at a station , only one video packet is served although there could be more than one video packet waiting.

The maximum bandwidth required by each station is given by :

$$BW\_\max = \lambda_{\max} * N_S \tag{4.11}$$

The maximum number of bits per pixel is 1.41. On the average, each station will get token every 3.3 ms and therefore, the maximum bandwidth given to each station will be 25000*1.41/3.3 =10.68 Mbps. Therefore the number of stations supported with pure synchronous service to video is 100/10.68 =9. Note that in this calculation we assume that bandwidth required by token transmission and propagation is negligible.

Figure 4.8 shows the mean delay to video packet as a function of the number of video sources on the network. Mean delay to video packet is negligible for up to 20 video stations and increases exponentially as the number of video stations increases beyond 20. Delay to video packet starts building up more than 3.3 ms at 20 video stations which is very far from the results of Equation 4.11. The reason of this difference is due to variable bit rate characterstics of the video sources. The average bits per pixel is 0.52 and has standard deviation of 0.23. In equation 4.11 we have used the maximum bandwidth value to calculate the number of stations with pure synchronous service to video. It is clear that 23 video stations can be supported with mean delay of 99 ms to each video packet. When we increased the number of stations to 24, the mean delay to video packet shoots up to 2 seconds which is much greater than the permissible limit.

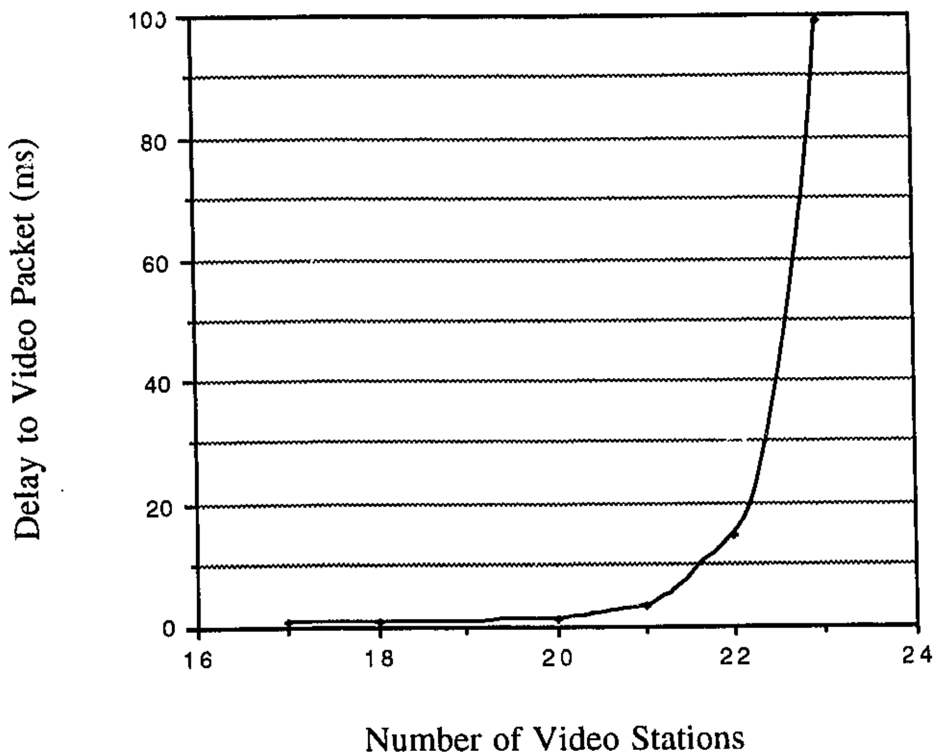


Figure 4.8 Mean Delay to Video Packet vs. Number of Video Stations

Figure 4.9 gives the variance of delay to video packet . As seen form the figure, the variance is negligible for upto 20 video stations.

From Figures 4.8 and 4.9 , it looks like video can have pure synchronous service for upto 20 video stations i.e. mean delay to video is less than 3.3 ms which is the packetization time of video packet. But however, there are some moments when the maximum TRT is greater than 3.3 ms (see Figure 4.10). Therefore, pure synchronous service to video is only available for upto 16 video stations. Although the probability of having maximum TRT more than 3.3 ms is very small.

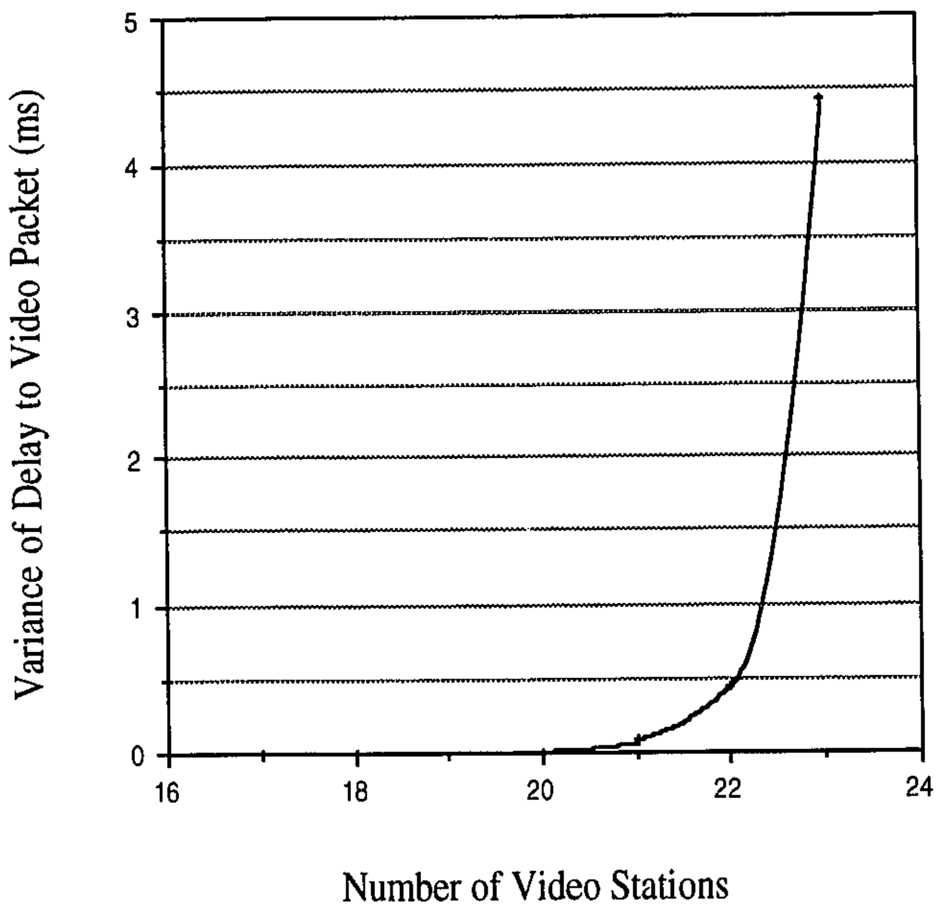


Figure 4.9 Variance of Mean Delay to Video Packet vs. Number of Video Stations

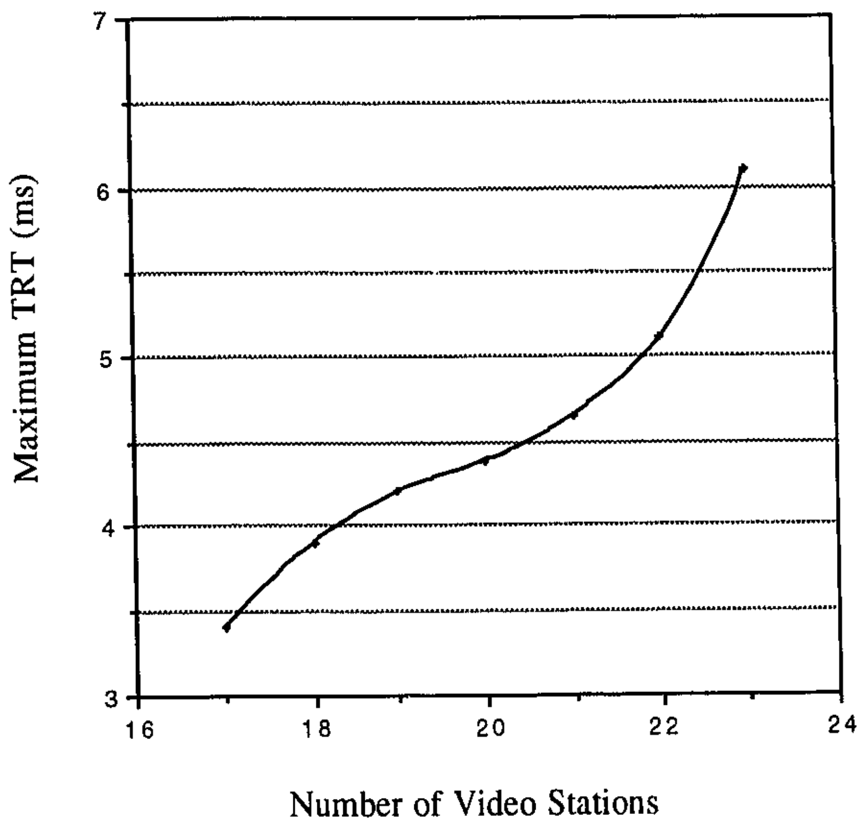


Figure 4.10 Maximum TRT vs. Number of Video Stations

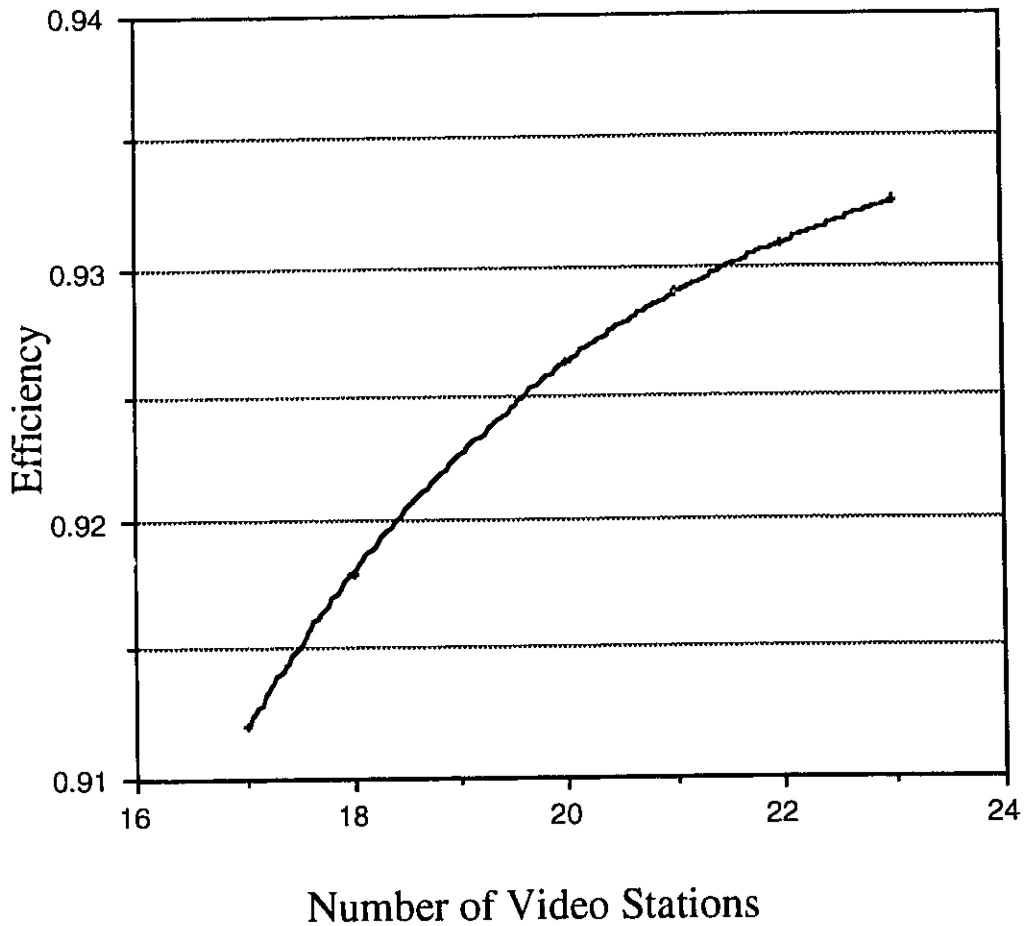


Figure 4.11 Efficiency vs. Number of Video Stations

Figure 4.11 shows the efficiency of the network as the number of video stations is increased. The efficiency of the network is defined as the percentage of bandwidth used for transmitting useful data i.e. the information bits in the video packet.

This efficiency is also called effective throughput and is given by equation 4.12 as following

efficiency = packet_t/(packet_t + token_t + token_p + overhead) (4.12) where packet_t is the time used to transmit useful data of packet, token_t is time to transmit token, token_p is the propagation time and overhead accounts for the time needed to transmit overhead of each packet.

The efficiency of the network is directly proportional to the information bits transmitted. In one video packet, the amount of the overhead bits is very less compared to the useful data. Therefore as the number of stations increase, the number of video packets transmitted on network increases.For 23 stations the efficiency of the network is 93.5%.

4.5.3 Multimedia (Video/data) Communications

In this section we look at the performance of FDDI when both VBR video and data (file transfer) sources are present. First, we looked into the performance of FDDI with no delay to video. For this purpose, setting the TTRT at half of the packetization time of the video packet, the number of packets waiting to be served will be bounded to one. This is because according to the FDDI standard[1,2], the maximum TRT can not be more than two times the TTRT. The following table 4.2 gives the complete simulation results for two different packetization times.

Table 4.2 MultimediaCommunication (video/data)Results for Different Packetization times of Video

Packetization time of video (ms)	TTRT (ms)	Maximum video packet length	Maximum TRT (ms)	Multimedia stations
2 ms	1.0 ms	21150+224	1.972 ms	8
3.3 ms	1.65 ms	35000+224	2.991 ms	11

From Table 4.2 it is clear that due to high bandwidth requirements, the number of multimedia stations supported are 8 for packetization time of 2 ms and 11 for t_pack of 3.3

ms. The reason for choosing packetization times equal to 2 ms and 3.3 ms is to have length of video packet less than maximum packet length of 36000 bits as defined in FDDI standard.

Figure 4.12 shows the maximum TRT vs. the number of multimedia stations for different video packetization times. When the maximum TRT is greater than the packetization time of video, the service provided to video is no longer synchronous. Therefore, it is clear that for packetization time of 2 and 3.3 ms, the maximum number of multimedia stations supported without any queuing delay to video are 8 and 11 respectively.

The next part of the study is to see the behavior of FDDI when data traffic is integrated with video and some permissible delay can be accepted to video. From previous simulations, it was found that 23 VBR video sources can be supported with a permissible delay of 99 *ms* to each video packet. As each data source needs less bandwidth as compared to VBR video, the number of multimedia stations chosen was 23. From table 4.2 it is clear that for higher values of TTRT more multimedia stations can be supported. Therefore for this part of our study the packetization time of video packet is set at 3.3 ms and TTRT was set at 3.69 *ms* according to Equation 4.13 as following:

$$TTRT = S_{max} + \text{Asynchronous} + \text{overhead} \qquad (4.13)$$

Where S_{max} represents the maximum synchronous bandwidth allocated to video. Asynchronous is the bandwidth allocated to data traffic and overhead. The number of pixels in one video packet are 25000. The average number of bits per sample is 0.52. However, the maximum number of bits per sample can be as high as 1.41.

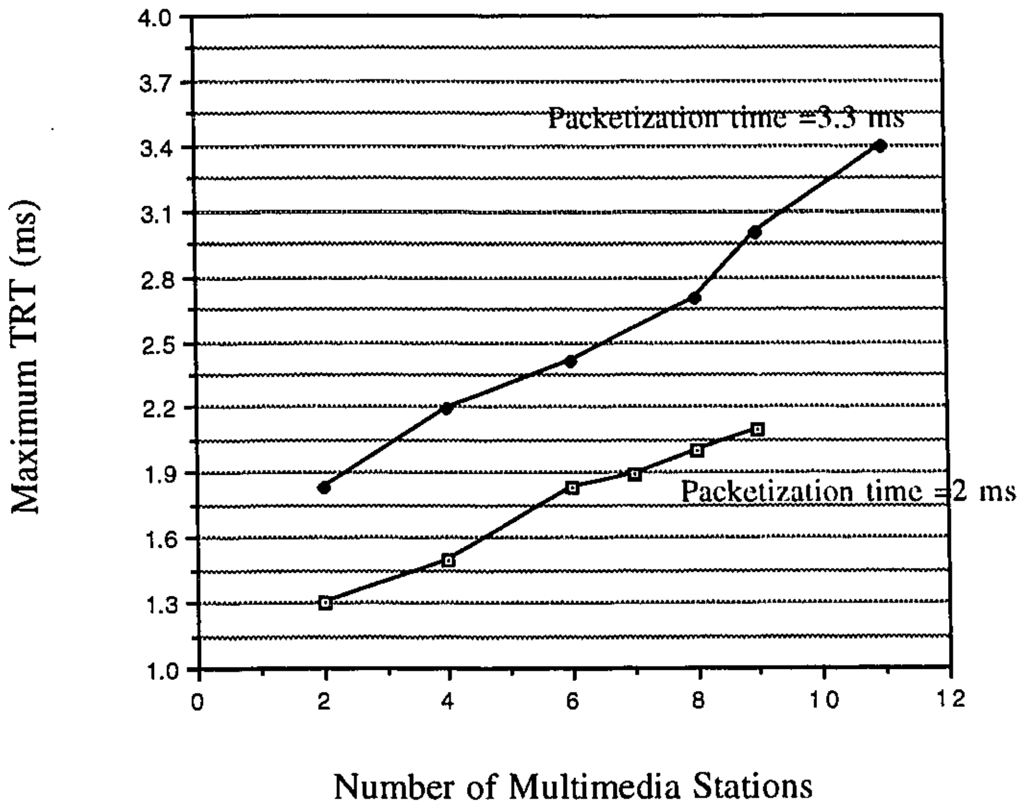


Figure 4.12 Maximum TRT vs. Multimedia Stations

For setting our TTRT, we used 0.7 bits/sample as average. The time required to transmit one average video packet is 0.17 *ms*. Therefore, in this case S_{max} comes out to be 3.68 *ms* for 23 stations. The minimum asynchronous available bandwidth is equal to bandwidth used to transmit one data packet in overrun mode of FDDI. The maximum time required to transmit a data packet is 4.54 μsec. The over head for 22 stations is 65 μsec. Using all these parameters TTRT comes out to be 3.680007 *ms* .

With packetization time of 3.3 ms and TTRT of 3.69 *ms*, the following performance results were obtained .

Figure 4.13 shows the efficiency of the network as the number of multimedia stations increases. The reason for the increase in efficiency can be explained as following. The amount of overhead data in one asynchronous (data) packet is large compared to the information part of the packet. On the other hand, the amount of overhead in one video packet is very small compared to the information bits. As the number of multimedia stations increases, due to dynamic bandwidth transfer property of FDDI, the number of video packets transmitted on the network increases and asynchronous packets decreases.

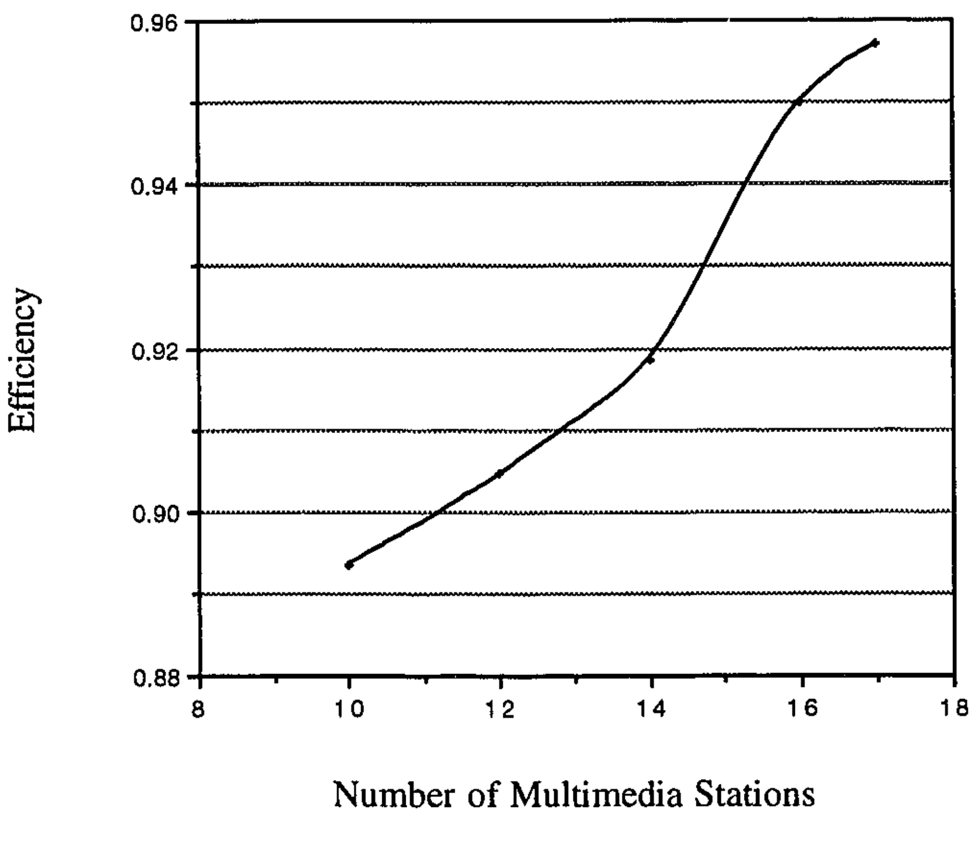


Figure 4.13 Efficiency vs. Number of Multimedia Stations

Therefore, the total amount of information data transmitted over the network increases resulting in an increase in network efficiency. Upto 14 multimedia stations the total bandwidth required by video stations is less than the available bandwidth and therefore the efficiency increases linearly. After 14 stations, the video sources start sucking asynchronous bandwidth resulting in less number of data packet transmission. As a result total transmitted useful information is much more than overhead and increase in efficiency exponential. At 23 multimedia stations the network starts saturating because there is no more bandwidth available to transfer from asynchronous to video traffic.

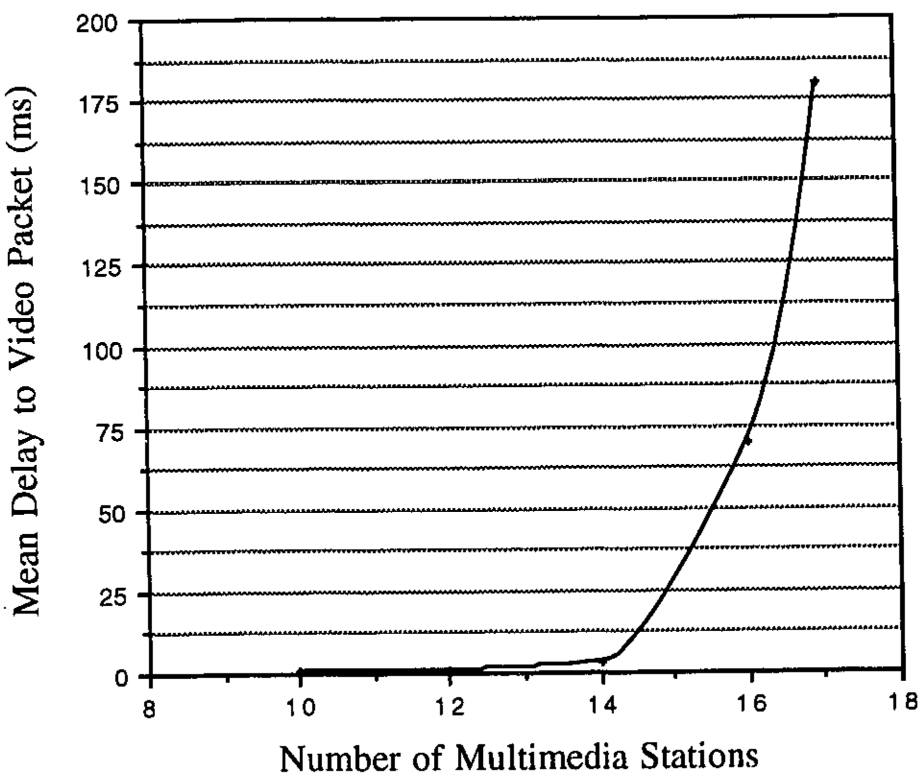


Figure 4.14 Mean Delay to Video Packet vs. Multimedia Stations

Figure 4.14 shows the delay experienced by the video packets as a function of the number of multimedia stations. From our simulations, we found that the maximum number of multimedia stations is bounded to 17 with a delay of 180 ms. The bound is determined by the fact that for a greater number of stations TRT exceeds the permissible bound of 2 x TTRT. The delay to video packet is negligible for up to 14 stations.

Figure 4.15 shows the delay faced by asynch onous data packets (file). As the number of multimedia stations increases, the delay to asynchronous packet increases. This is because the transmission of asynchronous packets is decided by THT. As the number of multimedia stations increases, the total bandwidth required for video services increases causing an increase in the delay to asynchronous packets. For 17 multimedia stations on FDDI network, the delay to asynchronous packets is 33 ms.

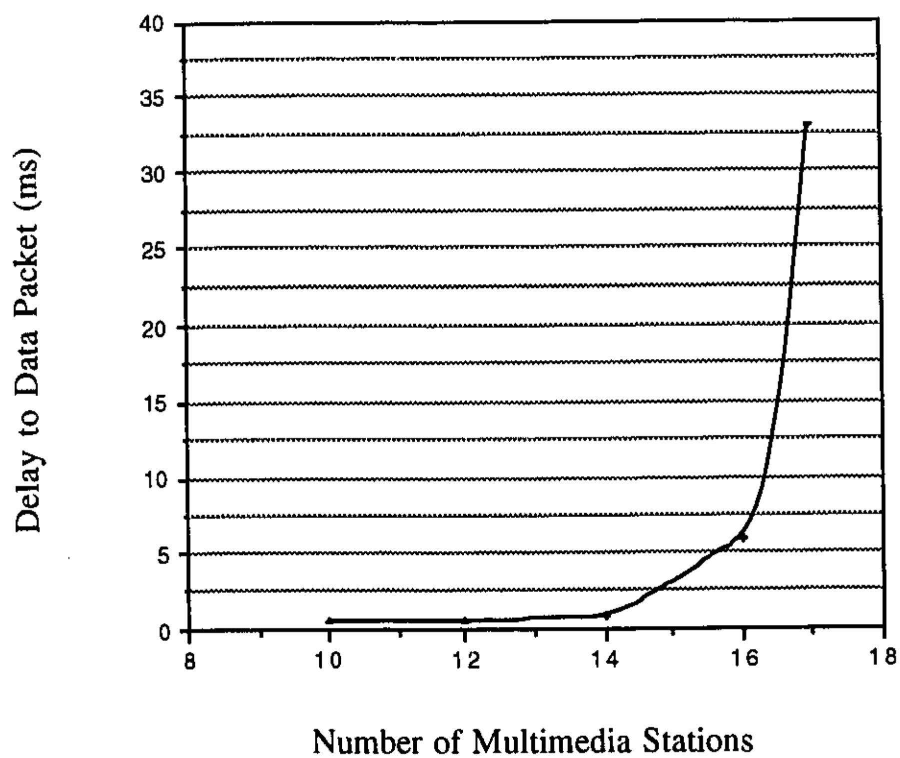


Figure 4.15 Delay to Data Packet vs. Multimedia Stations

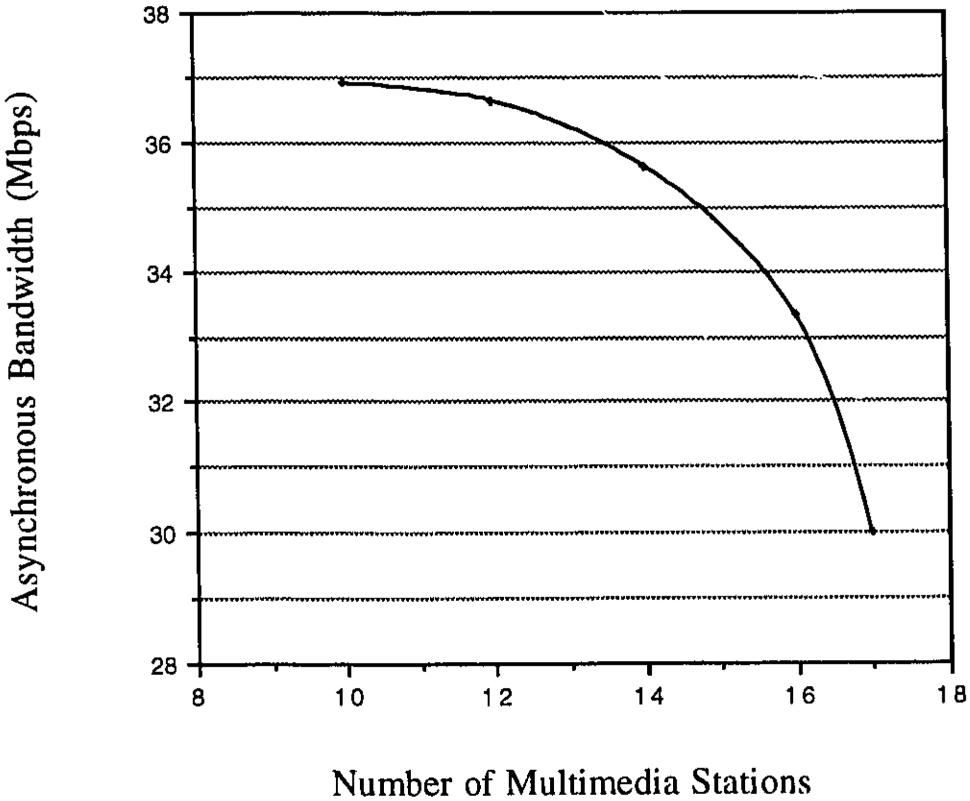


Figure 4.16 Asynchronous Bandwidth vs. Multimedia Stations

Finally, Figure 4.16 shows an interesting property of FDDI ,namely the dynamic transfer of bandwidth from asynchronous traffic to video. This is done by the timed token protocol of FDDI. When there are only 2 multimedia stations and the remaining 14 stations are just supporting asynchronous traffic, the bandwidth allocated to file traffic is 37 Mbps. As the load of multimedia stations increase, the bandwidth allocated to file traffic reduces to 30 Mbps.

4.6 Conclusion

This study analyzes the integration of multimedia applications, consisting of VBR video and data. First only VBR video is investigated on FDDI. As there was only one kind of traffic (synchronous), TTRT was set to the packetization time of the video packet. It was found that 12 and 16 VBR video stations can be supported with pure synchronous service to video for packetization times of 2.0 and 3.3 ms respectively. This concluded that with higher values of packetization time, more VBR video stations can be supported on FDDI. The performance of FDDI is also invesigated when some permissible delay to video is acceptable. The number of multimedia stations supported with maximum delay of 99 ms was found to be 23.

The second part of the study was to see the effects when data traffic was integrated with VBR video. The number of multimedia stations with pure synchronous service were reduced to 8 and 11. The performance of the FDDI was studied when some permissible delay to video is acceptable and multimedia traffic is present.. The number of stations drop to 17 and the maximum delay to video is 180 ms. The dynamic bandwidth transfer from data to video packet is also depicted .

Chapter 5

Study of Multimedia Variable Bit Rate Video and Voice Sources Over FDDI Networks.

5.1 Introduction

Significant effort is currently being devoted to the development of packet oriented integrated services over high speed networks [32][33]. The packetization of voice and video enables us to carry them together on an integrated packet network. The advantages of integrated networks are many, e.g., the efficient sharing of transmission and switching facilities, potential evolution towards fully integrated network which would provide video, voice and data services.

In this chapter, we study the integration of Variable Bit Rate (VBR) video and voice sources over FDDI networks. This study aims to assess the communication needs for supporting video and voice applications. A performance analysis of the integrated communication services to be provided by FDDI is presented. Voice sources are handled as asynchronous traffic. Major portion of the bandwidth is allocated to video sources. Due to this, the delay to voice increases.

From[34][35] it has been shown that by using the bit dropping algorithm on voice, the delay and probability of blocking of voice packets decreases significantly. In this chapter, we first consider the integration of video and voice sources without any bit dropping of voice packets. The second part of study deals with the integration of video and voice when bit drooping algorithm on voice sources is on.

5.2 Voice Source Modeling

A single voice source can be modeled as exponentially distributed talk-spurt and silent periods with means of 352 *ms* and 650 *ms* respectively [35]. The model for a single voice source is shown in figure 5.1.

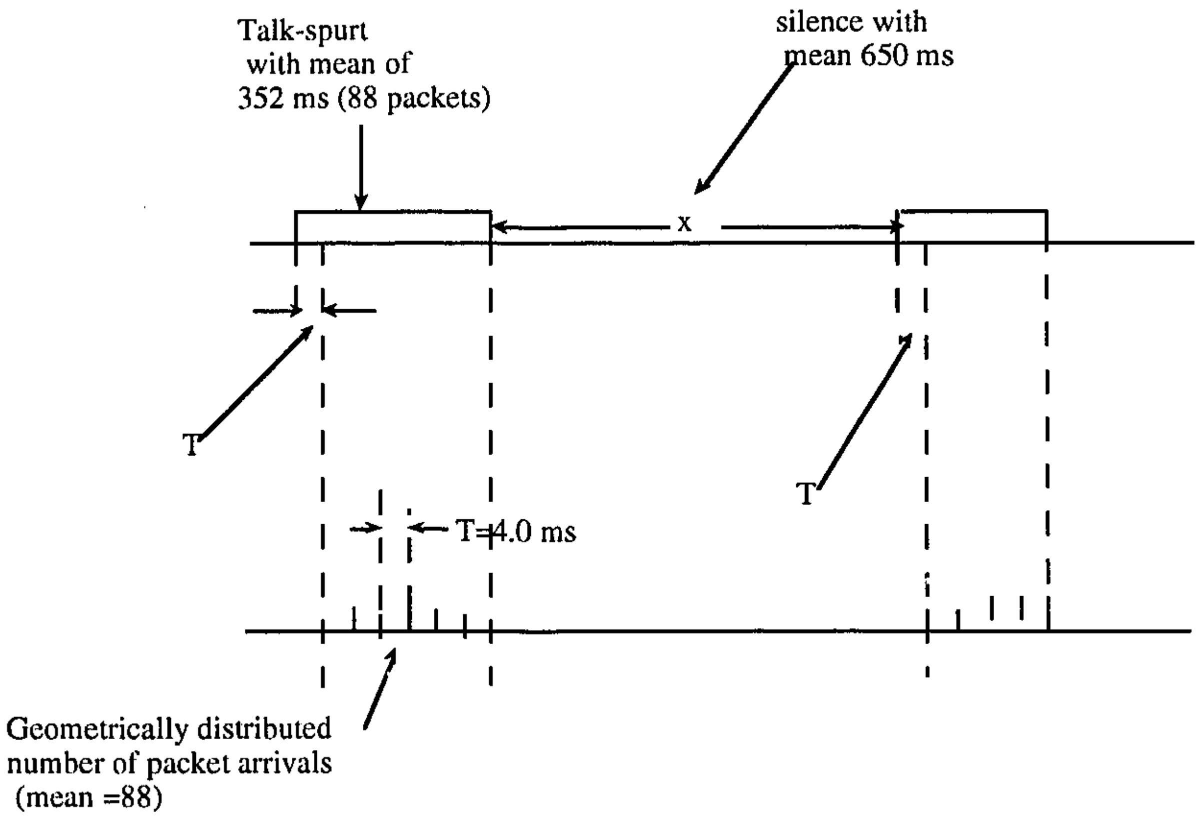


Figure 5.1 Packet Arrival Process From Single Voice Source

Each voice source is sampled at rate of 8 KHz and encoded using an embedded ADPCM scheme at a rate of 32 Kbps. Packetization time for voice is 4 ms. The reason for choosing 4 ms as the packetization time of voice is to reduce the idleness of the network.

A video packet is coming every 2 ms and the length of the video packet is variable. Therefore, there could be some instances when bandwidth required by video sources is very low. In this case the THT of stations will be positive and some asynchronous packets can be transmitted. If we set the packetization time of voice very high, say 12 ms, then for those instances when THT is positive either there will be no voice packets or one voice packet. If for that cycle total synchronous bandwidth required is very low then there will be no video packets in the next station and hence, bandwidth is just used in moving token from one station to the next station. Therefore, due to large voice packetization time the network will waste bandwidth for overhead transmission. For 4ms packetization time the percentage of the bandwidth used for useful data is 128/288=44%, which is quite less. But for packetization time of 12 ms, there will be no voice packet for 12 ms and the network is completely idle. Also for inter connection of FDDI and ATM, three voice packets can be collected (16 Bytes) to form one ATM cell. For 4 *ms* time interval, 32 samples are collected and organized into a packet as shown in Figure 5.2 . All the least significant bits from 32 samples are contained in block #1 of the packet, the next significant bits are in block #2 and the most significant bits are contained in blocks #3 and #4 . The maximum length of each packet is (128+224) bits. No packets are generated during the silent period. It is also assumed that silence detection of voice source is employed and packets are generated during talk-spurt only, as shown in Figure 5.1. The restoration of the silent period is automatically accomplished at the receiver as part of the play out strategy that uses time stamp information in the packet header.

5.2.1 Bit dropping algorithm

Bit dropping has been suggested as a means of reducing the delay offered to voice packets. Sriram used this bit dropping algorithm for voice data multiplexer[35]. He showed that the performance of the multiplexer improves considerably with bit dropping on voice packets. For our study we are using the same sort of algorithm and the parameters of this algorithm are as shown in Table 5.1.

Table 5.1 Bit Dropping Scheme on Voice Packets

Congestion state	*Congestion control action*	*Transmitted voice packet size*
$0 \leq S \leq K1$	Drop Nothing	128 bits
$K1 < S \leq K2$	Drop Block 1	96 bits
$S > K2$	Drop Block 1&2	64 bits

Bit dropping is done on each outgoing packet at each server just prior to the transmission. Let S denote the current number of voice packets in the voice queue. We consider K1, K2 and N as the first threshold, second threshold and size of buffer for congestion of voice packets. When S is smaller than the first threshold, no bit dropping is done. When the size of the voice queue, S, exceeds the first threshold K1, but is smaller than the second threshold, K2, then block #1, containing least significant bits, is dropped as shown in Figure 5.2. When S exceeds K2, both blocks #1 and #2 are dropped, thus reducing the information in the packet down to the most significant bits. The transmitted voice packet size is shown in Table 5.1. A packet is lost if the queue is full upon arrival.

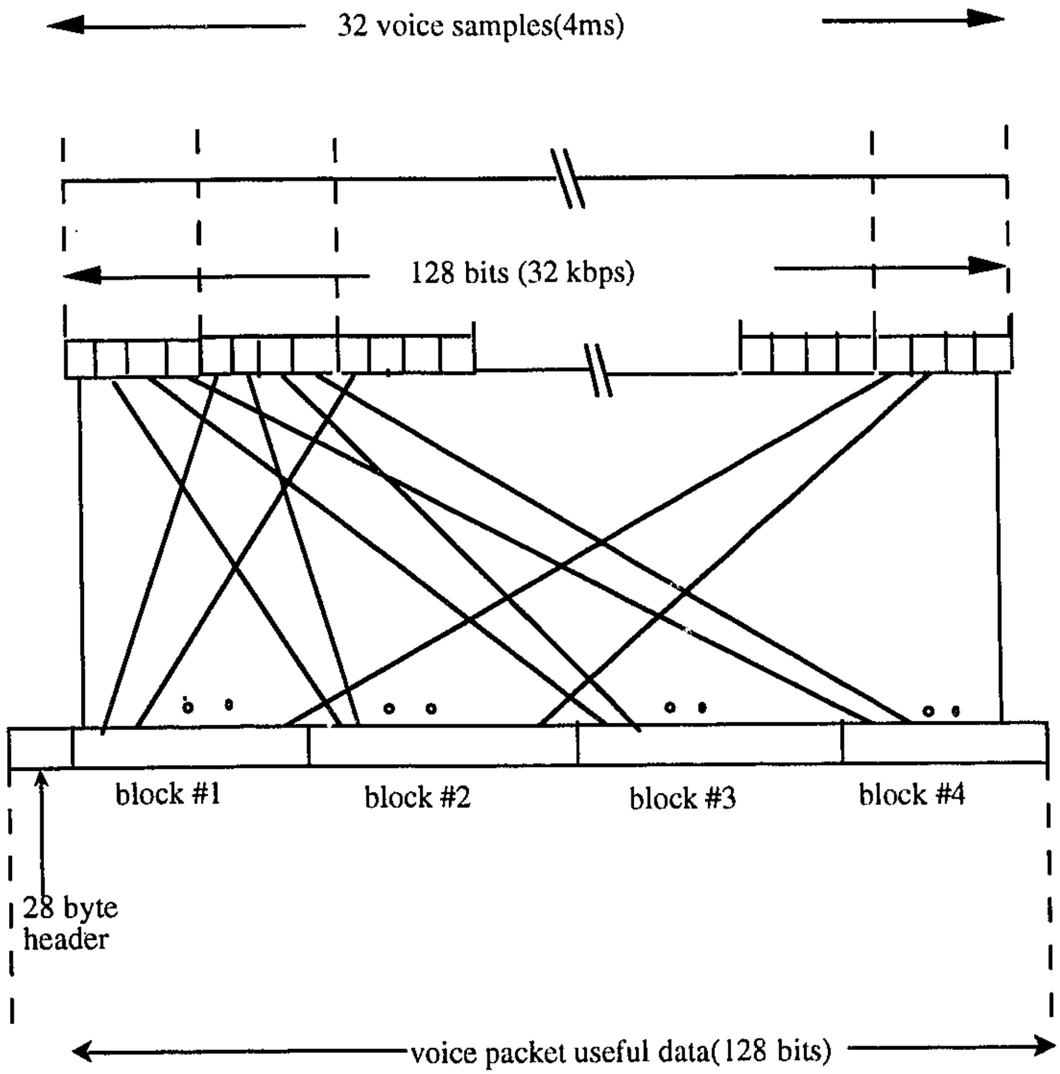


Figure 5.2 Voice Packet Structure

5.3 Integrated Video and Voice Services Over FDDI

Figure 5.3 shows the system configuration under study. The system consists of an FDDI network interconnecting a set of multimedia (video/voice) stations. The MAC protocol used is similar to the IEEE 802.5 token ring .

However, extension of the protocol provides the means to support synchronous and asynchronous traffic. Voice is treated as asynchronous traffic but the buffer size for voice has been set very small so as to limit the delay to voice to an acceptable level i.e. less than 2 seconds [3]. Also, bit dropping algorithm reduces the delay to voice. Fixed amount of bandwidth is allocated to the video traffic while the voice bandwidth is allocated dynamically. Whenever the token comes to a station, it transmits only one video packet and can transmit voice packets depending upon whether THT is positive or not. The TTRT is initially set for the whole network depending on the station's requirements.

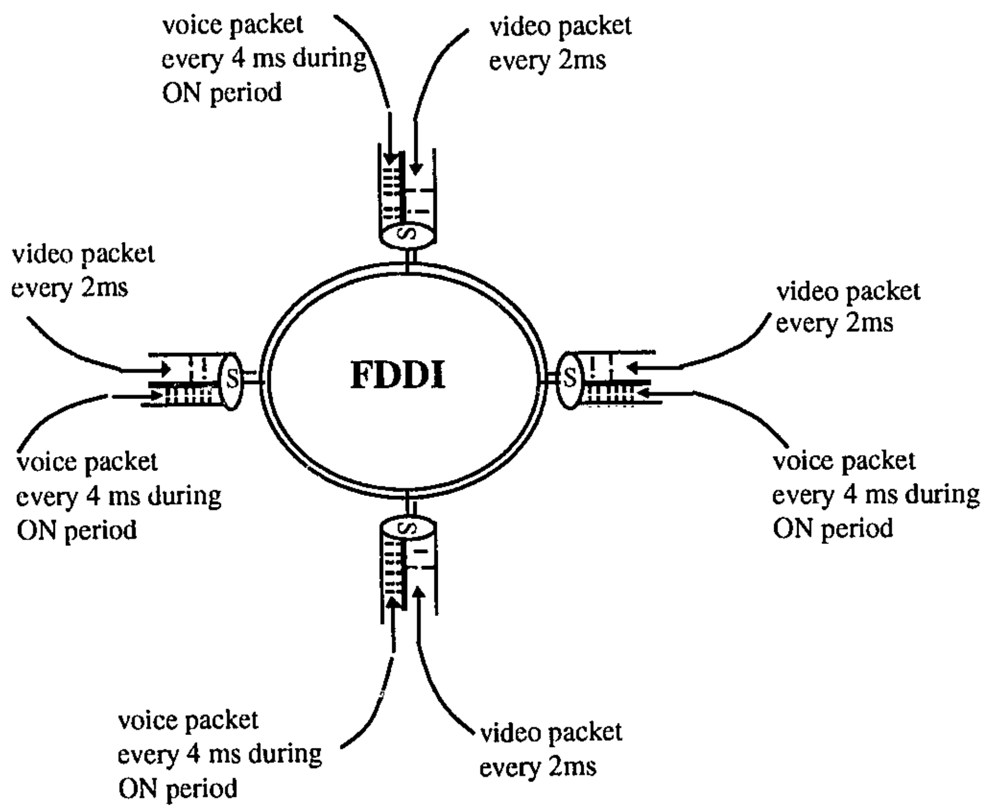


Figure 5.3 FDDI Supporting Video and Voice

5.4 Simulation Parameters

This section deals with the parameters for video and voice sources. For video source, we consider 30 frames per second. Each frame consist of approximately 250,000 pixels. The amount of data per frame is variable depending upon the activity in the scene and is equal to 250,000 $*\lambda$. The packetization times for video are 2.0 and 3.3 ms. The respective number of samples in one packet are 15000 and 25000. The amount of data in packet is variable depending upon the number of bits per pixel (λ). According to the FDDI standard the maximum number of overhead bits per packet is 224. For the chosen packetization times the video packet length is less than the FDDI maximum packet length, i.e., 36000. It is further assumed that token length is 88 bits and every station is 200 meters from its neighboring station. With theses values, the token propagation time (*token_p*) and token transmission time (*token_t*) are 1.017 μs and 0.88 μs, respectively.

5.5 Results and discussion

The results are divided into three parts. The first part gives the performance results, when all FDDI stations support multimedia (video/voice). The second part gives the performance results with respect to quality of voice and the last part gives the results when only voice stations are added to the existing multimedia network.

5.5 .1 Multimedia Communications

In our study, the main constraint is the quality of service (QOS) to video traffic. The QOS is measured in terms of offered delay and packet loss probability. It is assumed that very large buffer is available and no video packets are lost. For TV quality picture the delay to video should be less than or equal to 250 ms [3]. The delay to the video packet consists of a queuing delay and a service delay. This delay is considered from the moment when the packet is generated to the point where it is delivered to its destination

The following equation gives the mean delay to the video packet in terms of above delay components.

$$D_{mean} = D_{queue} + D_{trans} \qquad ....(5.1)$$

where D_{queue} is the queuing delay and D_{trans} is the transmission delay .

In chapter 4, we saw that for a video packetization times of 2 *ms* and 3.3 ms, 12 and 16 VBR video stations can be supported without any queuing delay to the video packet. It has been also shown that 22 VBR video stations can be supported with a mean delay of 99 *ms* with a packetization time of 3.3 ms. As voice traffic requires less bandwidth compared to video, we choose the above results to start our study on the integration of voice and VBR video sources.

Maximum synchronous bandwidth available to video is given by the following equation:

$$S_{max} = \frac{TTRT - t_{asy} - t_{overhead}}{TTRT} * 100Mbps \qquad ...(5.2)$$

S_{max} can be increased by increasing TTRT. The higher is the TTRT, the higher will be the bandwidth available for video. However, TTRT can not be more than half of the packetization time for pure synchronous services to video. Therefore, in the first part of the study, TTRT has been set to 1 ms and 1.65 which are half of the packetization times of the video packet. In this way, the number of video packets waiting are bounded to one. In other words D_{queue} is zero and only delay offered to video is due to transmission and propagation of packets.

5.5.1.1 Multimedia Communications with Pure Synchronous Service to Video

Figure 5.4 shows the variation of maximum token rotation time as the number of multimedia stations increases. It is clear from the graph that for a packetization time of 2.0 ms, when the number of multimedia stations increases to 12, the maximum token rotation time increases beyond 2.0 ms and hence the service available to video is no longer synchronous.

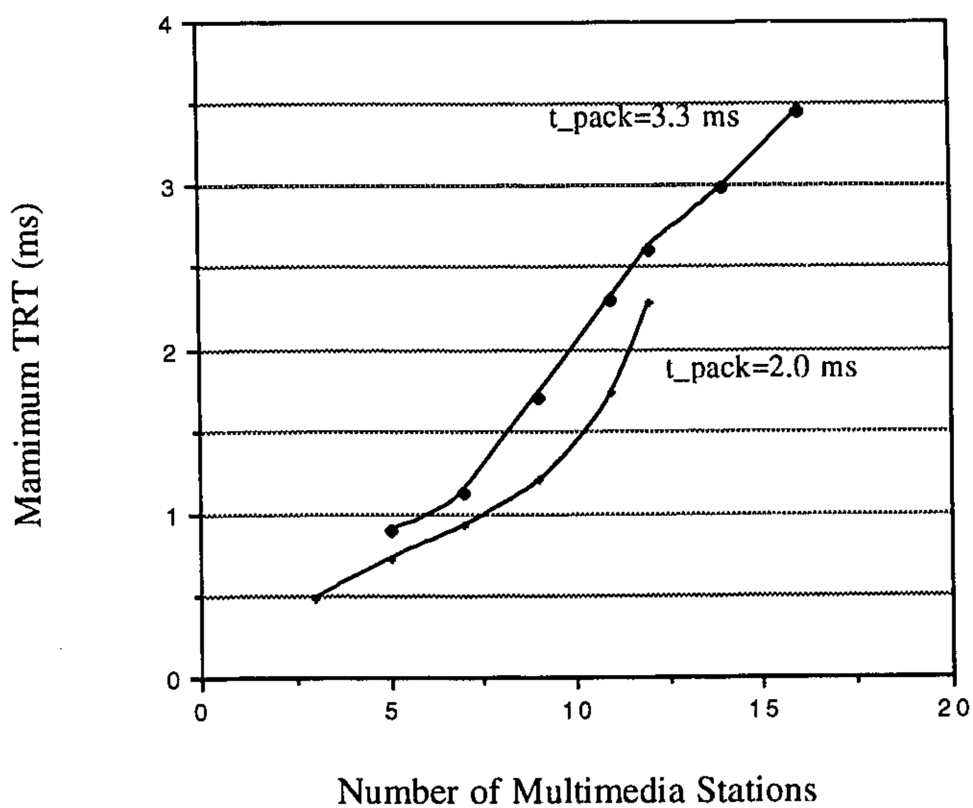


Figure 5.4 Maximum Token Rotation time vs. Number of Multimedia Stations

Similarly, for a packetization time of 3.3 ms, when the number multimedia stations increases to 16, the maximum TRT is 3.45 ms i.e., greater than the packetization time of video. Hence only 11 and 15 multimedia stations can have pure synchronous service over FDDI for packetization times of 2.0 ms and 3.3 ms respectively.

For both packetization times, the maximum number of voice packets was 2. Therefore there is no use of to see the effect of bit dropping. The reason for just 2 voice packets is due to VBR video traffic. When activity in the coded scene is small, the stations have THT positive and arriving voice packet never finds more than 2 packets in voice queue.

5.5.1.2 Multimedia Communications When Some Delay to Video is Acceptable

The next part of the study focused on the integration of video and voice sources so that the delay to video and voice sources is in acceptable range. In this case, voice sources are treated as asynchronous traffic and voice packets have to wait in a separate buffer (voice queue).

From our first results in Chapter 4, it was found that 23 VBR video sources can be supported with a permissible delay of 99 *ms* to each video packet. As each voice source needs less bandwidth (32 Kbps) as compared to VBR video, the number of multimedia stations chosen were 23. TTRT was set at 3.68 *ms* according to equation 5.3 as following:

$$TTRT = t\_s_{max} + t_{asy} + t_{overhead} \qquad (5.3)$$

Where t_s_{max} represents the transmission time for total synchronous bandwidth allocated to video in the network and t_{asy} is the time required to transmit one maximum length voice packet and overhead is the time used for the transmission and the propagation of the token. The number of pixels in one video packet is 25000. The average number of bits per sample are 0.52. However, the maximum number of bits per sample can be as high as 1.41 . For setting our TTRT, we used a value of 0.7 bits/sample which is the average value of bits per pixel plus its standard deviation (0.23). For 0.7 bits/sample, the time required to transmit one average video packet is 0.16 *ms*. Therefore t_s_{max} comes out to be 3.68 *ms* for 23 stations . Minimum asynchronous bandwidth available is used to transmit one voice packet in the overrun mode of the FDDI. The maximum time required to transmit a voice packet is 2.88 μsec. The over head for 23 stations is $69\mu sec$. Using all these parameters TTRT comes out to be 3.6800072 *ms* . With these parameters, it was found from simulations that 23 multimedia stations can be supported with D_{total} of 108 *ms* to each video packet. The effect of integrating voice source with VBR video increases the delay to the video packet from 99 *ms* to 108 *ms* .

The last part of the study involves seeing the bit dropping effect over the system performance . In the bit dropping algorithm ,the first threshold is K1=1, the second threshold is K2=2 and a buffer size for voice packets as N=3. The choice of N so small is to bound the delay experienced by voice packets to a minimum value. Also threshold values of K1 and K2 are set low so as to have minimum delay for voice in the asynchronous mode of voice.

Figure 5.4 shows the efficiency of the network as a function of the number of multimedia stations. The efficiency of network is defined as the percentage of bandwidth used for transmitting useful transmission, i.e., the information bits in the video packet. Sometimes this efficiency is also called effective throughput and is given by:

$$\text{efficiency} = \text{packet\_t}/(\text{packet\_t} + \text{token\_t} + \text{token\_p} + \text{overhead}) \qquad (5.4)$$

where packet_t is the time used to transmit useful data of packet, token_t is time to transmit token, token_p is the propagation time and overhead accounts for the time needed to transmit overhead of each packet.

As the number of multimedia stations increases, bandwidth is shifted from voice to video and the total number of transmitted video packets increases and hence efficiency increases. The efficiency wouldn't have been as high if we had fixed the number of multimedia stations at 18 and increased just the voice sources. The efficiency of the network increases as the number multimedia stations increases. The reason for this increase in efficiency is that the amount of overhead in one video packet (224 bits) is much less as compared to the useful information. On the other hand, the amount of overhead in a voice packet is quite comparable to the amount of useful information.

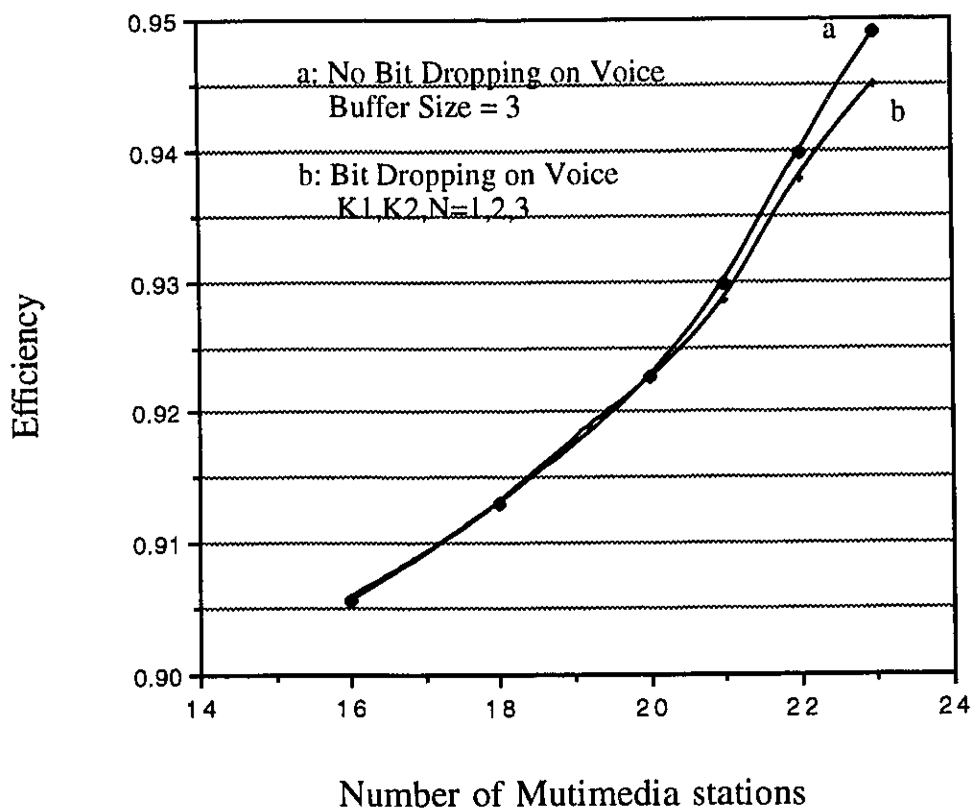


Figure 5.5 Efficiency vs. Number of Multimedia Stations

The efficiency can be further improved if the overhead bits for video packet are reduced from 224 to 160 bits. The header of the video packet can be reduced to 160 if the source and destination address are 16 bit long instead of the standard IEEE 48 bit. The effect of the bit dropping algorithm becomes effective when the number of multimedia stations increases to 21.

For more than 20 multimedia stations, a lesser amount of voice information is transmitted in bit dropping mode and the efficiency is less as compared to the regular mode.

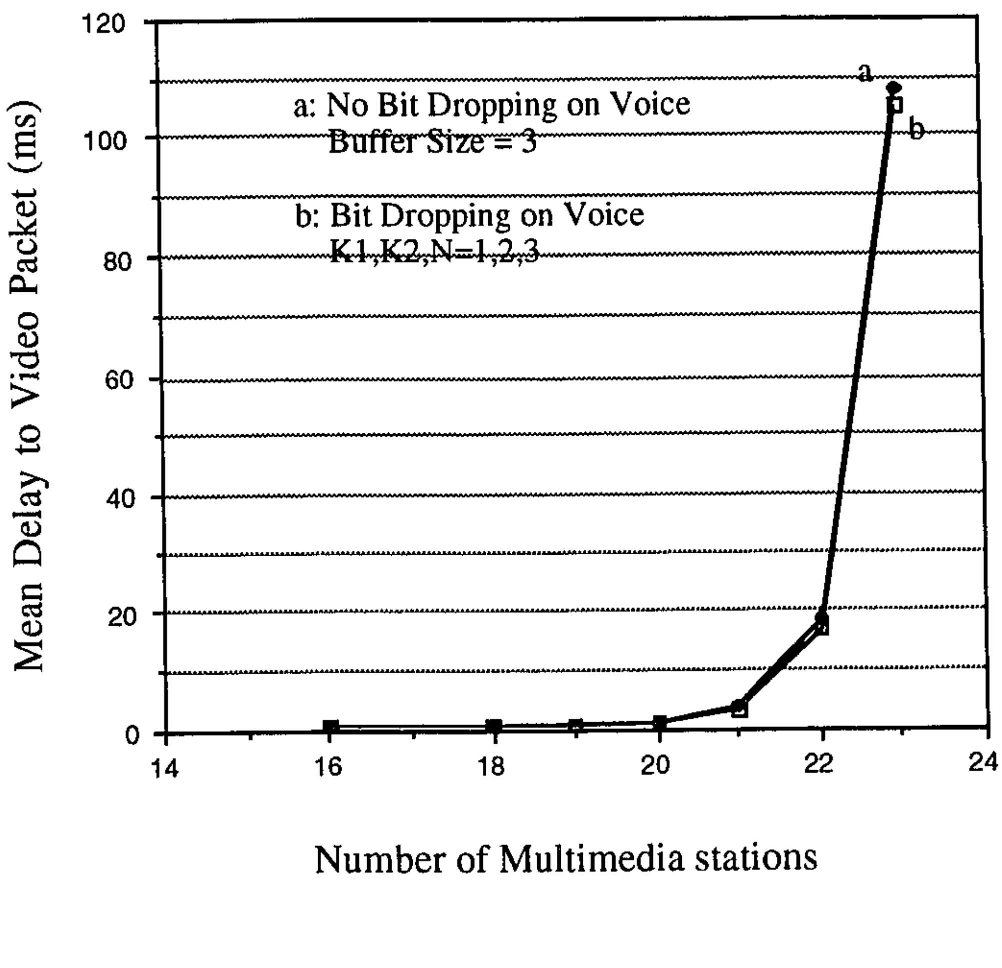


Figure 5.6 Delay to Video Packet vs. Number of Multimedia Stations

Figure 5.6 shows the mean delay to the video packet (*ms*) as the number of multimedia stations is increased from 16 to 23. It is clear that 23 multimedia stations(VBR video/voice) can be supported with total delay of 108 *ms* which is quite low when compared to the permissible delay of 250 *ms*. When the number of multimedia stations is increased to 24, the delay to the video packet increases to 3 sec which is much greater than the permissible value. The effect of bit dropping algorithm is clear from this figure. The total delay to the video packet reduces from 108 *ms* to 105 *ms*. The effect of the bit dropping algorithm comes into the picture only when the number of multimedia stations increases to 21. The total delay to the video packet is negligible (0.9 *ms).*

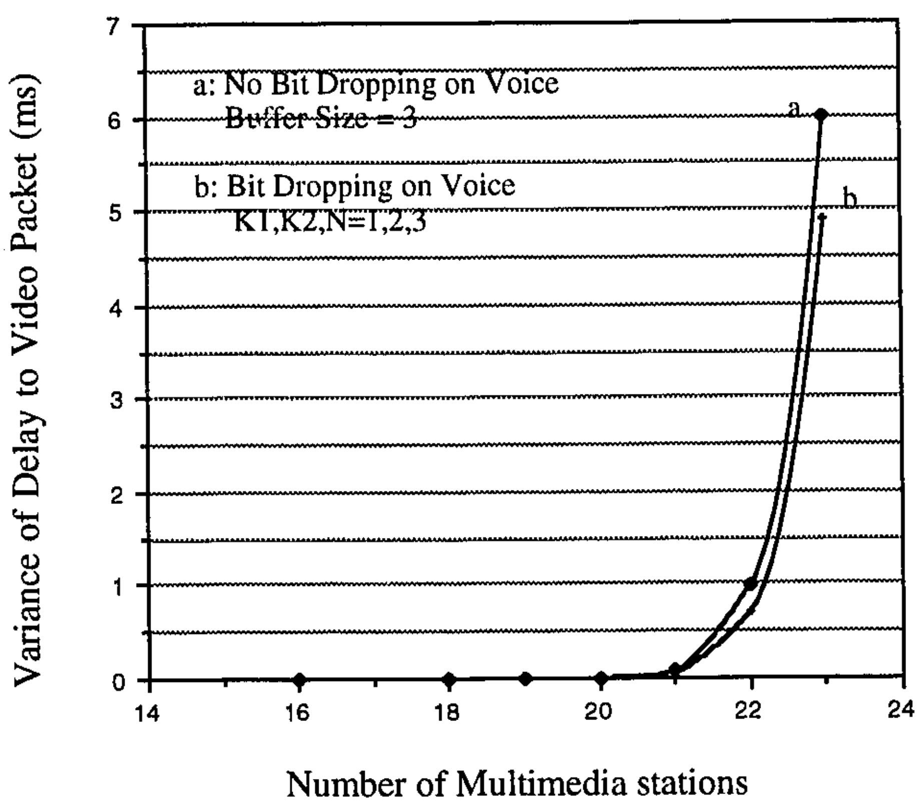


Figure 5.7 Variance of Delay to Video Packet vs. Number of Multimedia Stations

Figure 5.7 gives the variance of the delay to the video packet and is negligible until 19 multimedia stations. This figure also shows that the variance reduces with bit dropping effect.

From figures 5.6 and 5.7, it looks like video can be provided with pure synchronous service for upto 20 multimedia stations i.e., the mean delay to video is less than the packetization time of the video packet. But this is not true. There are some moments when the maximum TRT is greater than 2 ms and 3.3 ms(see Figure 5.8). Therefore, the service to video is only synchronous for upto 11 and 15 multimedia stations.

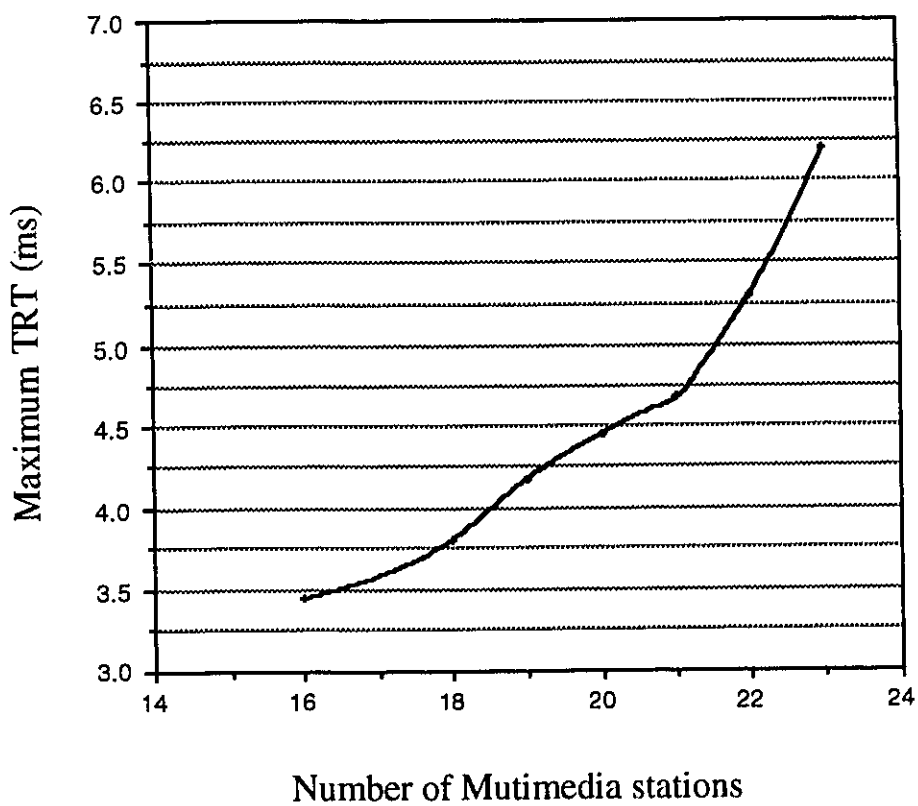


Figure 5.8 Maximum TRT vs. Multimedia Stations

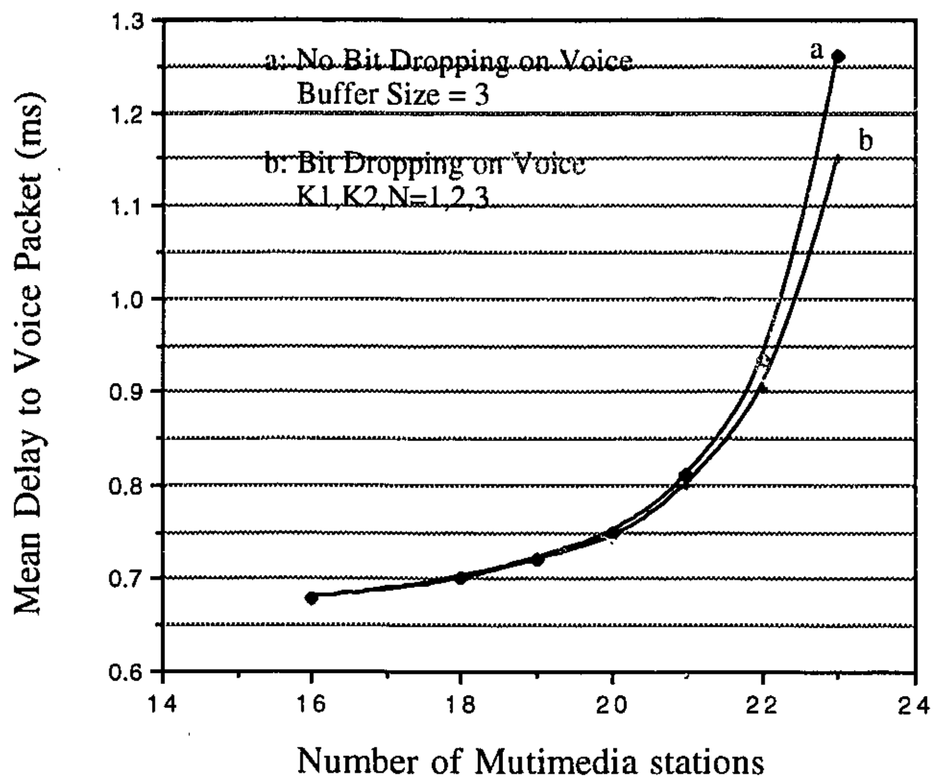


Figure 5.9 Delay to Voice Packet vs. Multimedia Stations

Figure 5.9 shows the delay offered to the voice packet as a function of the number of multimedia stations. The delay to the voice packet increases exponentially as the number of multimedia stations increases. With 23 multimedia stations, the maximum delay to the voice packet is 1.26 *ms* . Figure 5.9 also shows the effect of the bit dropping algorithm over the delay to voice. The delay decreases slightly by using bit dropping .

Figure 5.10 shows the bandwidth allocated to voice communications and the transmission of the token as the number of multimedia stations increases. There is a dynamic bandwidth shift from voice to video which is an interesting property of the FDDI. The bandwidth allocated to asynchronous traffic reduces from 4.8 Mbps to 3.1 Mbps . With bit dropping, less bandwidth is available to voice which is effective in reducing the delay to the video packet.

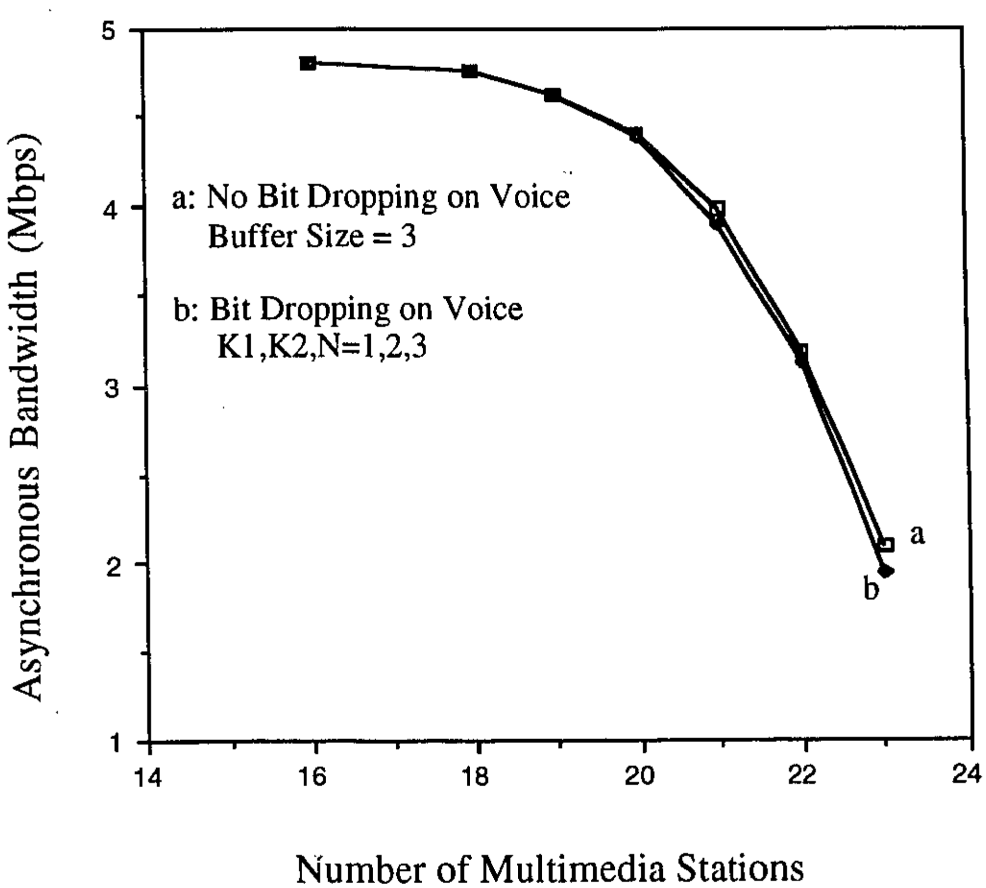


Figure 5.10 Asynchronous Bandwidth Vs. Multimedia Stations

5.5.1.2.1 Results and Quality of voice

Figure 5.11 shows the probability of loss of the voice packets as the number of multimedia stations increases. The buffer size has been set at 3. For upto 20 multimedia stations there is no packet loss . The probability of loss increases to 2.1E-04 when the number of multimedia stations increased to 23. The bit dropping algorithm decreases the probability of loss from 2.1E-04 to 1.78E-04 for 22 multimedia stations.

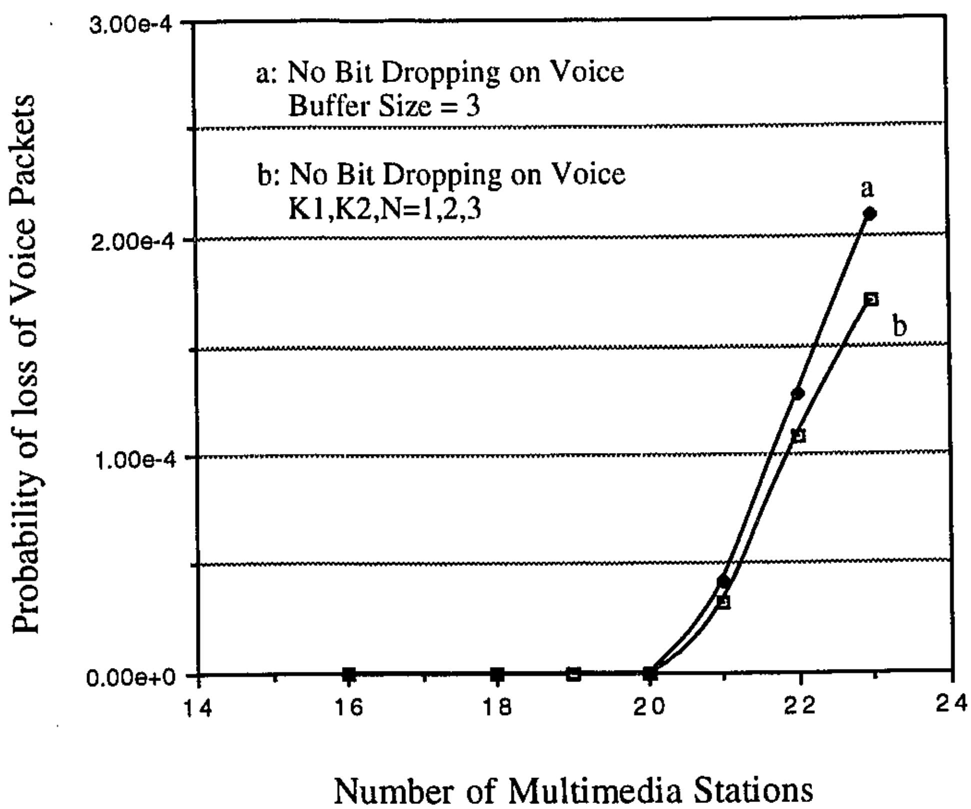


Figure 5.11 Probability of Loss of Voice Packets vs. Multimedia Stations

Figure 5.12 shows the probability of loss of voice packets as the buffer size for voice is increased. The bit dropping algorithm with different thresholds is shown in Figure 5.11. As the buffer size increases to 10, the bit effect dropping becomes negligible. The probability of loss to the voice packets decreases from 2.1E-04 to 1.7E-05 as the buffer size is increased to 10. For a buffer size of 10, the probability of loss of voice packets is almost the same for both cases.

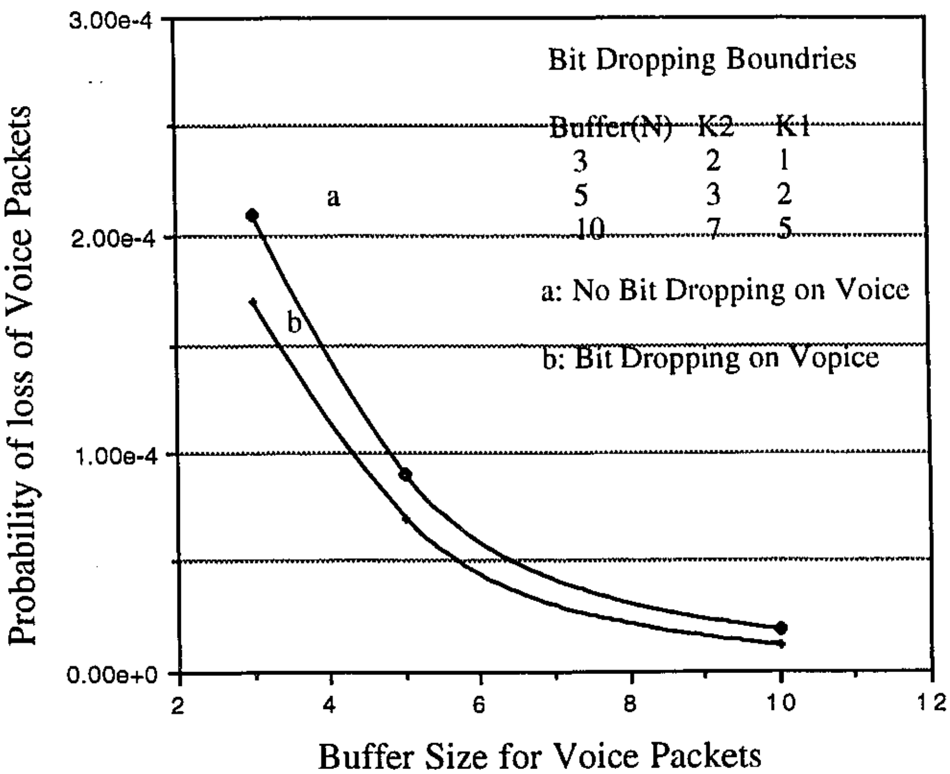


Figure 5.12 Probability of Loss of Voice Packets vs. Buffer Size

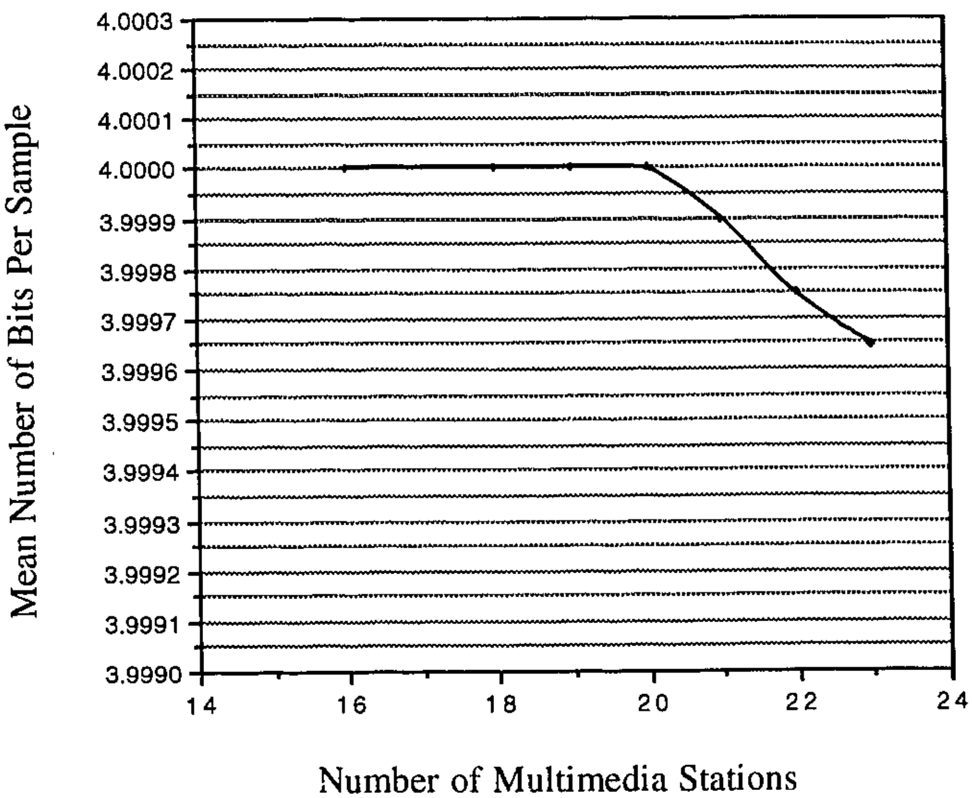


Figure 5.13 Mean Number of Bits Per Sample vs. Multimedia Stations

Figure 5.13 shows the mean number of bits used per sample for voice. The number of bits drops from 4 to 3.999 when the number of multimedia stations is increased from 16 to 23. For voice quality a mean bits per sample value of 3.8 and a packet loss fraction of 1.0E-03 is acceptable[36]. Therefore, 23 multimedia stations can be supported on the FDDI with good quality to the voice sources. From these results, it is also clear that better quality, in terms of fraction of voice packets lost, can be achieved with bit dropping.

5.5.1.3 Results When Voice Stations are Added to Existing Multimedia Network

From Figure 5.6 it is clear that the maximum delay to the video packet is approximately 108 ms when the number of multimedia stations is 23. When the number of multimedia stations is increased to 24, the delay to the video packet goes beyond the permissible limit. But 108 ms delay to video is small compared to the 250 ms. Therefore, the next part of the study was to analyze the effect of adding more voice sources to system of Figure 5.6 so that delay to video packet remains below than 250 ms.

As from the previous section it is clear that bit dropping on voice improves the performance, therefore bit dropping on voice is considered on the new system.

The next system considered has 23 multimedia(video + voice) stations with buffer size of 5 for voice packets. The values of first and second threshold, K1 and K2, are 2 and 3 respectively. The reason for choosing such low values of K1 and K2 are to have the bit dropping as effective as possible and to offer minimum delay to voice packets. Load on the network is increased in terms of adding more voice stations e.g. 30 voice stations on the network means 23 multimedia and 7 more voice stations, 52 voice stations means 23 multimedia stations and 29 more voice stations.

Figure 5.14 shows that as voice sources increase from 23 to 60, the delay to the video packet increases from 108 ms to 277 ms, which means that 34 more voice stations can be added to the existing FDDI network supporting 23 multimedia stations.

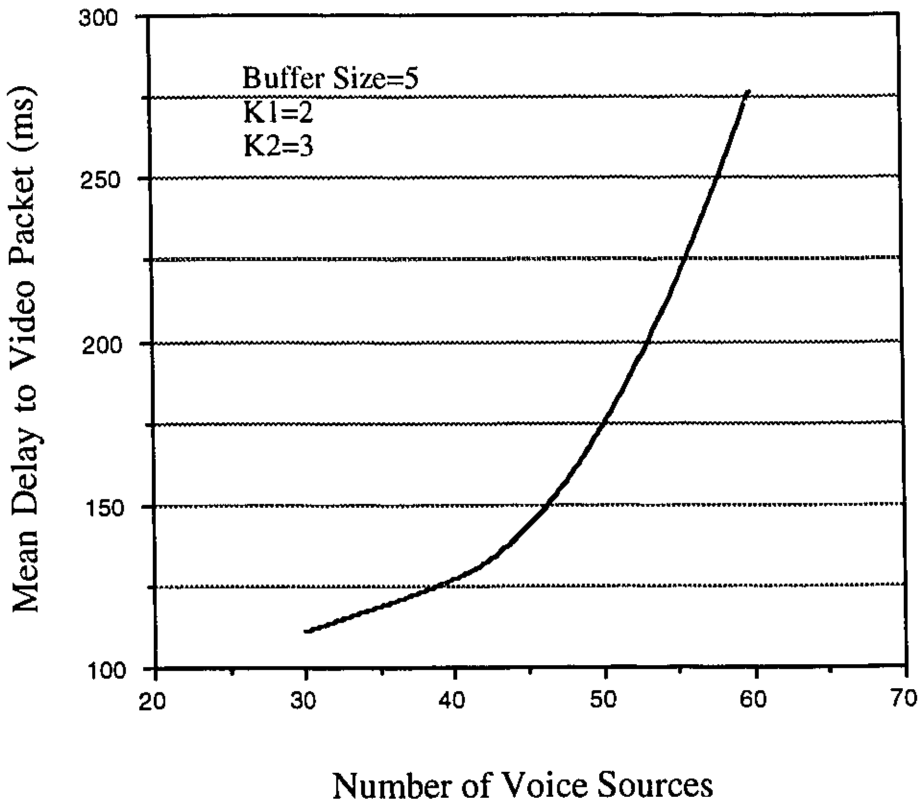


Figure 5.14 Delay to Video Packet with 22 Multimedia Stations on System and Increasing Number of Voice Sources Beyond 22.

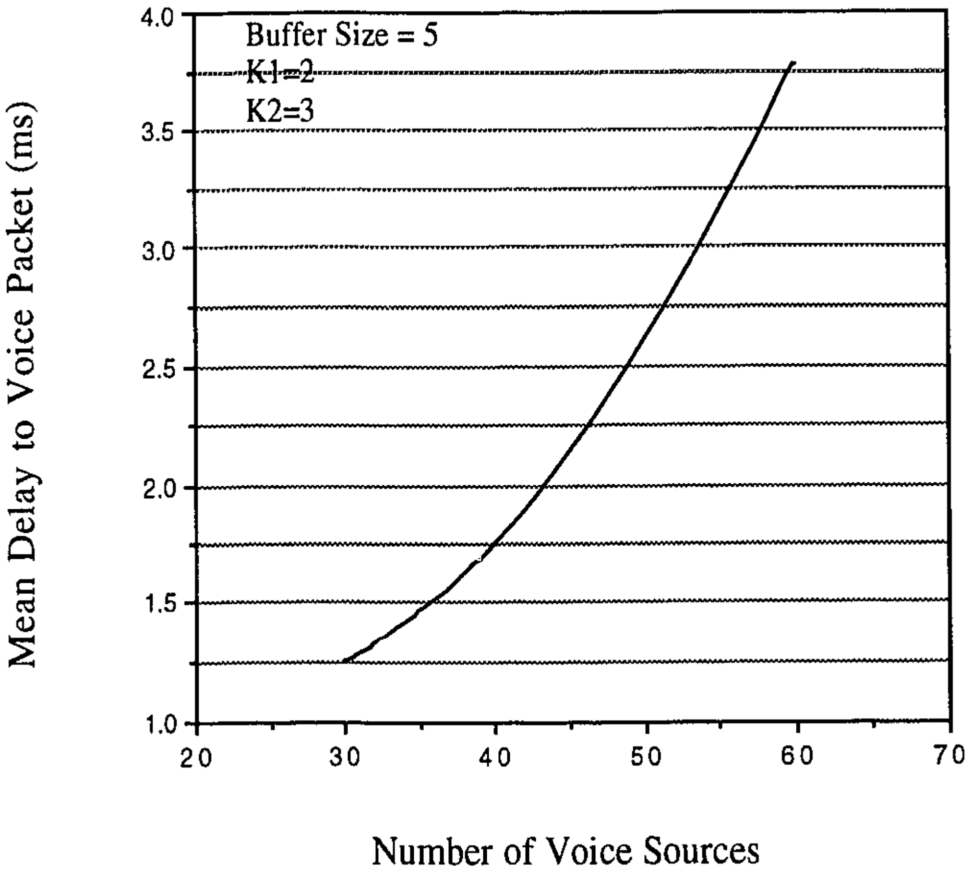


Figure 5.15 Delay to Voice Packet with 22 Multimedia Stations on System and Increasing Number of Voice Sources Beyond 22.

Figure 5.15 shows the delay offered to voice packets as number of voice sources are increased from 23 to 60. The maximum delay to voice packet is 3.7 ms which is much lower than permissible limit as 2 seconds[3].

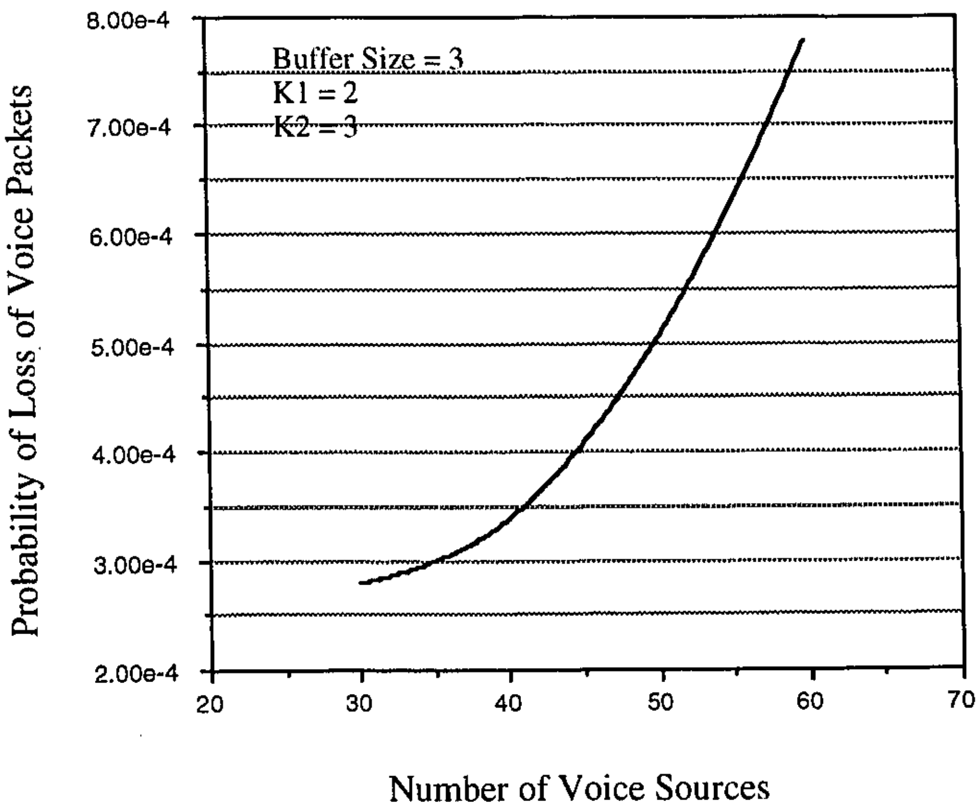


Figure 5.16 Probability of loss of Voice Packet with 22 Multimedia Stations and increasing Voice Sources Beyond 22.

Figure 5.16 shows the probability of loss of the voice packets as the number of voice sources is increased beyond 23 upto 60. FDDI with 23 multimedia stations(video + voice) is already investigated and that is why voice sources were increased beyond 23 onwards. Probability of loss for voice is 8.7E-04 .

5.6 Conclusion

In this chapter we have analyzed the performance study of VBR video and voice. We look at the performance of integrated VBR video and voice over FDDI. First the behavior of FDDI is studied when there is no queuing delay to the video packet. As the voice source is treated as asynchronous traffic, bit dropping algorithm is studied on voice to reduce the delay to the voice packet. For TTRT of 1.0 ms and 1.65 ms ,i t was found that 11 and 15 multimedia (video/voice) can be supported without any delay to video. For a permissible delay (0.25 *sec*) acceptable to video, with TTRT of 3.68 *ms*, 23 multimedia stations can be supported on the FDDI network with delay of 108 *ms* . Bit dropping algorithm on voice packets improves the performance of FDDI. The probability of loss of voice packet and delay to video packets is reduced in bit dropping mode. It is also concluded that bit dropping algorithm becomes ineffective when the voice buffer size increases to 10.

Further limit of adding more voice stations on network already supporting 23 multimedia (video/voice) is investigated with video packet having delay less than 250 *ms*. It is concluded that 34 more voice stations can be added to the previous network with delay of 246 *ms* to video packet.

Chapter 6

Conclusions

Multimedia transmission on high speed networks is the technology of the future. The requirement for multimedia transmission is dynamic bandwidth allocation. FDDI is recognized as a high speed local area network. Performance study of the integration of VBR video and other traffic on FDDI shows that VBR video requires a significant amount of bandwidth and therefore FDDI can support only limited number of multimedia stations. However, with improvements such as bit dropping on voice, the delay in to video packet is reduced and the probability of loss of voice packets is also reduced.

6.1 Guidelines to Set TTRT for Multimedia Traffic

TTRT is the key parameter of FDDI performance. This is the parameter which makes FDDI to support synchronous as well as asynchronous traffic. For multimedia VBR sources on FDDI, the following rules should be observed to set the TTRT.

• Maximum TRT can be as long as the two times TTRT. Therefore, a station supporting synchronous and asynchronous traffic may not see the token for a time period equal to two times TTRT. Hence for a pure synchronous service to synchronous traffic, station supporting synchronous traffic should request a TTRT value of one half the packetization time of the packetization time of synchronous time. For example in this thesis VBR video is considered as synchronous traffic with packetization time of 2 ms and therefore TTRT is set to 1 ms. If some permissible delay to synchronous traffic can be acceptable, then TTRT can be increased to support more multimedia stations.

• TTRT allows at least one maximum size asynchronous traffic packet along with synchronous traffic. Therefore TTRT should validate the following Equation:

TTRT= S_{max} + Asynchronous + overhead ....(6.1)

Where S_{max} is the total synchronous bandwidth required on the network, asynchronous is bandwidth required to transmit one asynchronous packet and overhead is ring latency with token transmission also. For variable bit rate sources in synchronous mode of FDDI, synchronous bandwidth allocated to each station is one synchronous traffic packet. In other words the synchronous bandwidth allocated to each station is the average packet length of VBR source.

• For multimedia sources on FDDI network, TTRT should be set such that maximum bandwidth is taken by synchronous traffic. This way we can have maximum possible multimedia stations on the network. Maximum synchronous bandwidth S_{max} can be formulated as following:

$$S_{max} = \frac{TTRT - t_{asy} - t_{overhead}}{TTRT} * 100Mbps \qquad ...(6.2)$$

Where t_{asy} is the time is the time required to transmit one asynchronous packet and $t_{overhead}$ is time required for token transmission and propagation.

From Equation 6.2, the maximum synchronous allocated bandwidth can be increased by increasing TTRT. But for multimedia VBR sources we can increase only such that maximum TRT is never greater than twice the TTRT. Maximum packetization time as calculated in chapter 4 is the value for which the length of synchronous packet is never greater than FDDI standard packet length. Therefore we can increase our TTRT to half of maximum packetization time of synchronous traffic.

6.2 Multimedia Communications

Each scenario is studied from two points of view : one when service to video is pure synchronous and the other when service is synchronous within some permissible delay limits. These conclusions are valid for those video sources for which the coded bit rate varies between 0.08 to 1.41. Major conclusions for such traffic are as following:

1. Firstly, only VBR video was investigated. TTRT was set to 2.0 and 3.3 ms and packetization times of video was 2.0 ms and 3.3 ms. TTRT was set equal to the packetization time because there is no asynchronous traffic present and token can be late up to 2*TTRT due to the presence of asynchronous traffic with synchronous traffic. It was found that 12 and 16 video stations can be supported by FDDI for a pure synchronous service. Secondly TTRT was increased to 3.66 ms. The investigation also showed that 23 VBR video stations can be supported by FDDI within a permissible delay of 250 ms.

2. VBR video was integrated with data traffic. For a packetization time of 2 ms and 3.3 ms per video packet, 8 to 15 multimedia stations (video and data) can be supported without any waiting delay to the video packet. The research further revealed that with a permissible delay of 250 ms to video packet, 17 multimedia stations can be supported for a packetization time of 3.3 ms. The property of dynamic transfer of bandwidth from data to video traffic is depicted as video traffic increases on the network.

3. The third part studied the performance of FDDI when VBR video was integrated with voice. For video packetization time of 3.3 ms and voice packetization time of 4.0 ms, it was found that 15 multimedia stations can be supported on FDDI. Bit dropping algorithms were also studied when 15 multimedia stations were present. However, since the number of voice packets never exceed the limit, the bit dropping effect is negligible.

4. The last part of the study looked at video and voice on FDDI when the permissible delay was acceptable for video and voice. It was found that 23 multimedia stations can be supported with a permissible range of delay to the video packet. The effect of bit dropping on voice when congestion occurs for voice traffic was also studied. It was found that the

probability of the loss of voice packets improves with an increase in the efficiency of the network.

5. From the results which showed that 23 multimedia stations can be supported with a delay of 108 ms, it was clear that some more voice stations could be added to the network with a maximum delay of less than 250 ms. It was found that a maximum of 34 voice stations could be added to the existing network.

6.3 Further Work

1. In chapter 4, when video and data were integrated the results showed the 17 multimedia stations can be supported with a delay of 108 ms while the maximum permissible delay is 250 ms. Therefore, the study could be extended to see the effect of adding only more data traffic such that the delay to video is less than 250 ms.
2. Investigation of clipped voice with data traffic on FDDI is not studied in this thesis. Further studies of this could prove to be interesting.

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