Movable Frame Hybrid MAC

A Multi-MAC Protocol for Wireless

Software Radios in Multi-rate Multimedia Applications

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ABSTRACT

Movable Frame Hybrid MAC – A Multi-MAC Protocol for Wireless Software Radios in Multi-rate Multimedia Applications

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We describe a new medium-access control (MAC) protocol for dynamic adjustment of the bandwidth requirement of multimedia applications. The purpose of this technique is handling multi-rate, multi-level traffic in an integrated wireless-access network (IWAN). A proposed mechanism divide total bandwidth in basic band and reservation band. Four conventional access techniques which are CDMA, CSMA, TDMA and FDMA are combined in the basic band. Reservation band choose these four techniques flexibly depending the traffic characteristics and quality of services (QoS) requirement. This Movable Frame Hybrid MAC (FMHMAC) is called software radios in third generation (3G) wireless network designing.

A comparative evaluation of this access technique is done by simulation procedure. Through simulations, the performances of the proposed access technique (e.g. call blocking probability, average delay and delay jitter) show that is both robust and suitable for the intended IWAN applications, this will results in high QoS guarantee for arbitrary traffic condition.

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List of Symbols and Abbreviations

N	number of total users in system
N_1	number of users in class 1
N_2	number of users in class 2
N_3	number of users in class 3
N_4	number of users in class 4
α	the probability of going from active to silent mode
β	the probability of going from silent to active mode
θ	traffic burstiness
δ	normalized traffic rate
ρ	traffic intensity
t_d	packet transmissions units
t_p	propagation time
n	number of bits in one codeword
k	number of parity bits in one FEC codeword
A	total traffic load
B	channel bandwidth
C	channel capacity
S	throughput

D	average packet delay
V^2	delay jitter
$Pr_{[blocking]}$	probability of call blocking
$Pr_{[overflow]}$	probability of buffer overflow
L	buffer size
Sn	number of total time slot in the observation
PG	processing gain
P_A	steady state probability of active period
P_S	steady state probability of silent period
K	number of users in CDMA channel
E	average packet size
T_F	frame length
$In_{()}$	in-progress state function
$O_{()}$	buffer overflow state function
$Bn_{()}$	blocked state function
$WT_{()}$	waiting time function
IWAN	integrated wireless access network
MAC	medium access control
QoS	quality of service
CAC	call access control
BER	bit error rate
SNR	signal to noise ratio
CDMA	code division multiple access

TDMA	time division multiple access
FDMA	frequency division multiple access
CSMA	carrier sense multiple access
BCDMA	basic CDMA band
RCDMA	reservation CDMA band
BTDMA	basic TDMA band
RTDMA	reservation TDMA band
BFDMA	basic FDMA band
RFDMA	reservation FDMA band
BCSMA	basic CSMA band
RCSMA	reservation CSMA band
MFHMAC	movable frame hybrid multiple access control
FFHMAC	fixed frame hybrid multiple access control
FEC	forward error control
WT	waiting time
MDT	maximum delay tolerance

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CHAPTER 1

INTRODUCTION

1.1 Introduction to Integrated Wireless Network

The aim of the third generation wireless network is to provide a single set of standards that can meet a wide range of wireless applications and provide universal access throughout the world. Third generation wireless networks will carry many types of information (voice data and video), will operate in varied regions (dense or sparsely populated regions), and will serve both stationary users and vehicular users travelling at high speed [8].

It is not surprising then that most industry analysts forecast that by 2004 voice telephone traffic will only count for less than 10% of total public network traffic. The rest, more than 90%, will be Internet and other data. In 1998, this picture was totally different. Voice telephone generated more than 60% of the total traffic [20].

The approach to accommodate multimedia, multi-rate traffic into one wireless system is represented by the proposal of an integrated wireless access network (IWAN) [23] whereby different services and systems are all integrated into one wireless channel [1]. The IWAN is intended to support service at different rates and grades of service while the main approach of the IWAN design is an efficient and robust medium access control protocol (MAC) that can integrate heterogeneous traffic types and meet their requirements for quality of service (QoS).

Many current mobile radio systems provide both voice and digital data communications, but they do so by handling each type of communication separately [10]. Recent literature has proposed various integrated systems [24, 25, 26]. Many publications have dealt with Call Admission Control (CAC) in IWAN [15, 16, 17, 18, 19, 20, 21]. All of them use the adaptive or hybrid mechanisms to obtain the QoS guarantee. In most cases, a packet scheduler is responsible for granting transmission to flows on the basis of their specific application QoS requirements while less delay and delay jitter requirements are considered.

With call admission control (CAC) IWAN carries heterogeneous traffic types simultaneously on the same channels. The channels are connection oriented and packet based so that customers can send at varying rates over time. Calls arrive and depart over time and the network can choose to accept or reject connection requests. The IWAN provides quality of service (QoS) guarantees at the packet level, e.g. packet loss probabilities and at the call level, e.g. call blocking probabilities. In turn, the network collects traffic load from customers for calls that it accepts into the networks.

As mentioned in [18] the admission control can be divided into two categories:

- Static CAC: The admission controller must decide which source combinations can be accepted into the IWAN and meet QoS constraints. This is based on the steady state behavior of the sources within the network. A connection request is accepted only when sufficient resources are available to establish the call at its required QoS and maintain the agreed QoS of existing calls. Meeting QoS requires a decision function that decides when adding a new call will violate QOS guarantees. Given the diverse nature of voice, video and data traffic and their often complex underlying statistics, finding good QoS decision functions has been the subject of intense research [27, 28, 29]. Recent results have emphasized that robust and efficient QOS decision functions require online optive methods [30].

- Dynamic CAC: a set of calls meet the QoS requirements. But yet the CAC policy may reject the new call. There might be several reasons for rejecting the connection. For example, rejection of a less valuable call, which is acceptable under QoS constraints, may make room for accepting a more valuable call. Thus, the network utility can be increased. On the other hand, calls may be accepted into the network even if the instantaneous QoS is violated. But when averaged over states the service quality is met.

In a packet communication environment voice and data have different QoS requirements, which will be described in the next section.

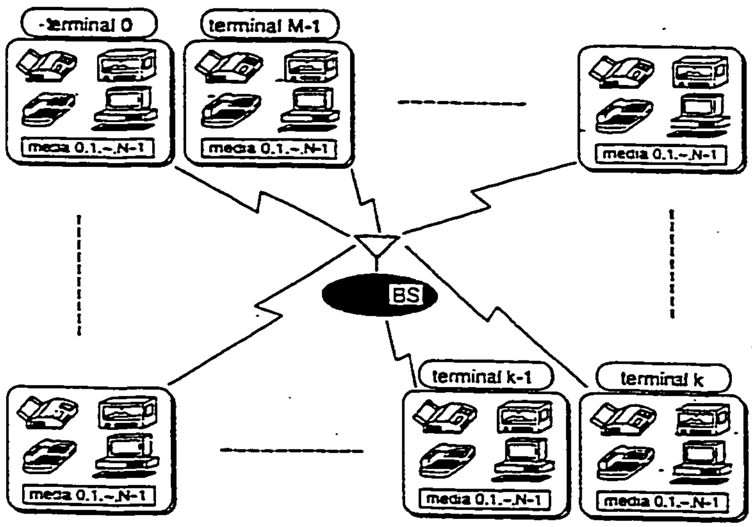


Fig.1. 1: System model of wireless multi-media communications.

1.2 QoS Requirements in IWAN

We consider service requirements in terms of call blocking probability, transmission delay, delay jitter and bit error rate (BER). Packets of voice can tolerate errors and even packet loss (a loss of 1.2% of the voice packets has an insignificant effect on the quality of service) [31], while the data packets are sensitive to loss and errors but can tolerate delays. The table below summarizes these requirements [23].

Service	**Maximum BER**	**Delay**
Speech (32, 16, 8kb/sec)	10^{-3}	Sensitive
Asynchronous data	10^{-9}	Insensitive
Facsimile	10^{-4}	Insensitive
Packet data	10^{-9}	Insensitive
Low resolution video (64, 128kb/sec)	10^{-5}	Sensitive

Table 1. 1 Required performance for telecommunication services

Inter-working with the wired telephone network will require that the transmission delays for voice services be minimized even if echo are employed. Thus the system must be designed to meet the required bit and packet error rates without requiring retransmission of damaged packets. This can be achieved by providing sufficient error connection capability and/or sufficient SNR. Bit error rates of 10^{-3} and packet error rates of 10^{-2} appear subjectively acceptable if the speech decoder employs the redundancy between successive packets to mitigate the errors in the reconstructed signal.

Asynchronous data transmission generally requires much lower error rates, as low as 10^{-9}. Fortunately, retransmission request protocols (ARQ) are available to ensure data integrity [31]. The penalty is error-detection overhead beyond the error-correction provided and delay to accommodate the retransmission attempts during poor channel conditions.

Facsimile transmissions may tolerate a higher error rate than asynchronous data, say 10^{-4}. The higher error rate allows a reduction in the number of retransmission requests and a corresponding increase in the throughput on the channel. We assume the presence of a protocol-converter near the base station converting signals between the land line protocol and the radio protocol. To minimize the interference exhibited to other users under high-traffic conditions, it may be desirable to reduce the wireless transmission rate if buffer overflows from the landline transmissions can be avoided.

Packet data transmissions are characterized by highly variable transmissions rate demands. Most of the time the mobile packet terminal is in standby mode ready to receive information; infrequently it transmits bursts of packets. Packet accuracy requirements are high, say 10^{-9}, and retransmission requests are invoked. Delay requirements for packet data are not severe, thus many packet terminals can be made to share a subchannel on a collision-detection basis.

The terminal may transmit periodic indications that it remains connected to ensure that the base remains synchronized. Alternatively, if the terminal is expected to remain in standby mode for long intervals, total overhead may be reduced by allowing for a rapid synchronization operation before information packets are transmitted. Limiting the short-time transmission demands of packet terminals appears desirable so as to avoid degrading the quality of services in progress on the other channels.

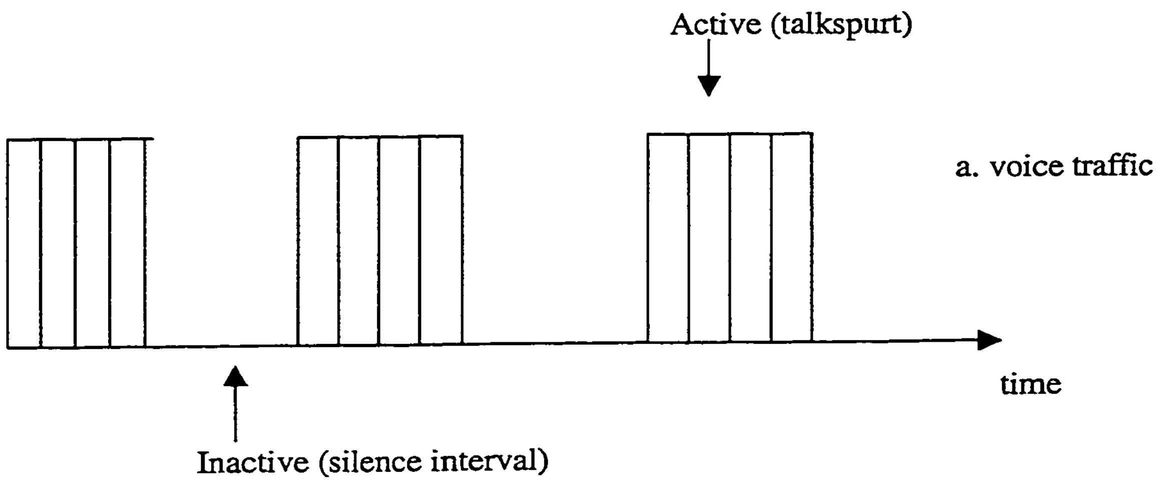
Low-resolution video transmission, such as may be useful for video conferencing, requires a transmission rate of at least 64 kb/s, preferably 128kb/s. One may be able to exploit the variable transmission rates required by a coder that processes only slowly moving imagery [32]. Bit error rates of 10^{-5} are desirable but somewhat higher error rates may be tolerable since the signal is to be presented to viewers in real-time delays due to ARQ protocols beyond the buffering delay are unacceptable.

1.3 Traffic Characteristics

In the preceding section we have stressed the fact of the different services requirements of different traffic. Given a number of ways of characterizing the traffic that might be deployed over a wireless network, it is then natural to use these models to study traffic access control.

Traffic models have played a significant role in the design and engineering of IWAN [1]. Poisson arrival and exponential call-holding time (call duration) statistics have served as excellent models for almost a century in carrying out both engineering and performance evaluation of circuit-switched voice telephony. Poisson arrival and the relatively simple packet-length models have been used extensively in studying the performance of packet-switched networks as well. It is not at all clear, however, that these older, well-established models will suffice in carrying out the design of the integrated wireless networks of the future. In fact, quite the contrary may be true. It appears that the integration of packetized voice, packet video and data traffic (whether brief bursts or much longer file transfers), each with its own multi-objective quality of service, requires the development of rather sophisticated traffic models to carry out accurate design and performance evaluation.

It is important to carefully characterize the traffic under study to ensure the models used lead to a useful network performance result. Generally speaking, all types of traffic is characterized by alternating, randomly varying periods of inactivity and activity. A bursty source is usually one in which the period of inactivity is much longer than the time during which it is active and transmitting packets. This can be shown in Fig. 1.2.



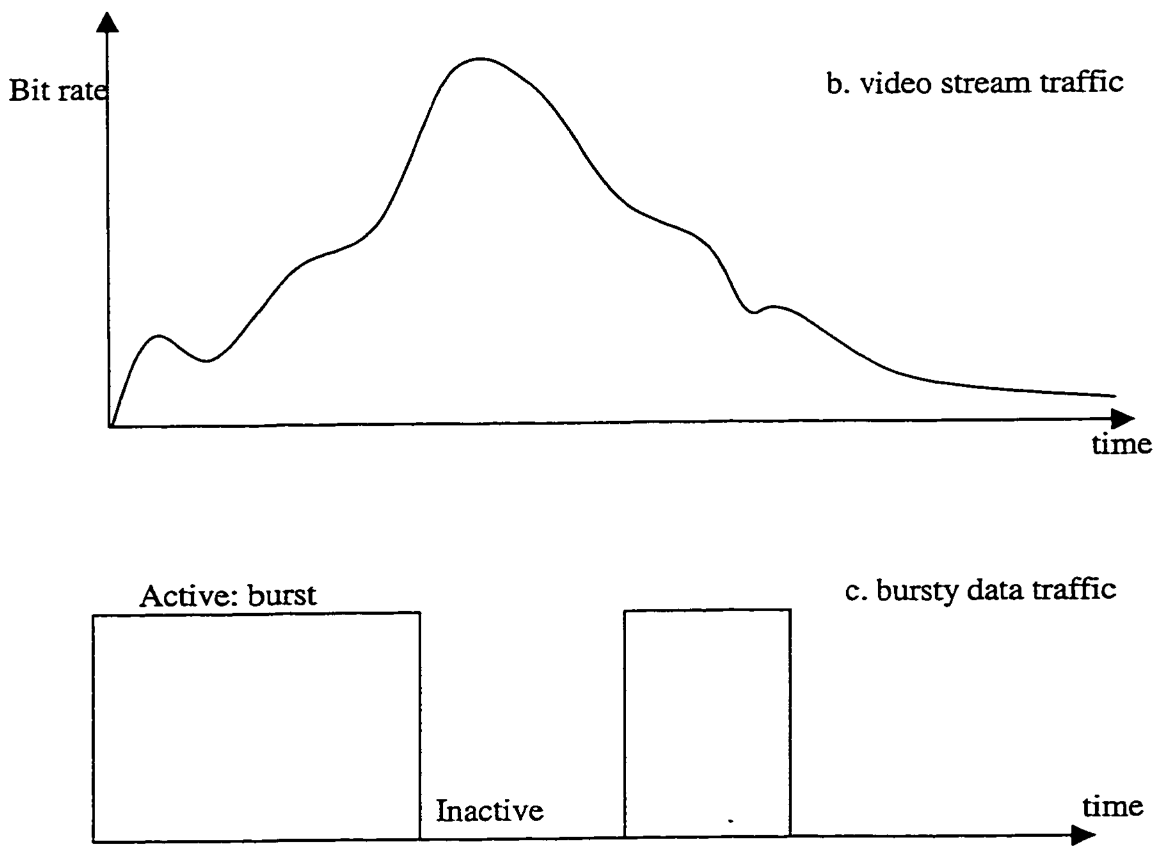


Fig. 1.2: Traffic mode characteristics.

From the user's point, the majority traffic in wireless LAN are voice traffic and data traffic. The differences between them are listed below.

Voice Traffic	**Data Traffic**
Isochronous	Asynchronous
Delay sensitive	Not delay sensitive

Minimize delay	Maximize throughput
Regular and deterministic	Irregular and bursty
Small packets	Large and small packets

Table 1.2: Voice and data traffic characteristics

It has been known for many years that both a voice and data source are well represented by a two-state process. An alternating sequence of active, or talk spurt intervals, followed by silence (inactive) intervals. To a reasonably good approximation, the states may be assumed to be exponentially distributed in length (The talk spurt duration is well-approximated by exponential distribution; the silent interval is less well-represented by this distribution. Added states have been proposed to improve the model, but this complicates the representation). This gives rise to the two-state birth death model. This process is shown in Fig. 1.3.

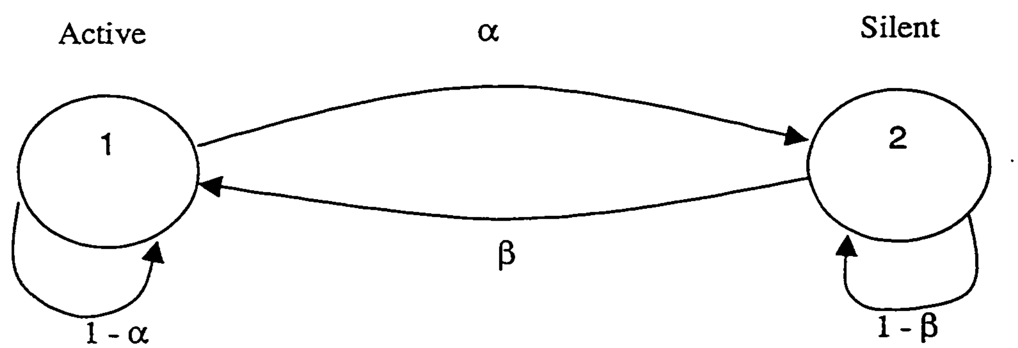


Fig. 1.3 State diagram of each user.

where:

α: the probability that a user changes its state from active to silent at the end of a packet.

β: the probability that a user changes its state from silent to active at the end of a packet.

To compute the parameters α and β, suppose P_A and P_S are the steady state probabilities in active and silence periods, respectively. Then:

$$\alpha P_A = \beta P_S \tag{1-1}$$

$$P_A + P_S = 1 \tag{1-2}$$

so:

$$P_A = \frac{\beta}{\alpha + \beta} \quad P_S = \frac{\alpha}{\alpha + \beta} \tag{1-3}$$

Usually we let P_A represent the burstiness of the traffic. When P_A increases (close to 1) we call the traffic stream traffic. When P_A decreases (close to 0) we call the traffic bursty traffic.

1.4 Thesis Outline

The thesis is organized as follows:

In Chapter 2, a brief review of MAC protocols is presented. Specific emphasis is placed on the CDMA, TDMA, FDMA and CSMA. The comparison of these MAC protocols on the different approaches to different traffic channels accesses are made based on this survey. We draw conclusions on which techniques have been chosen for wireless network.

In the first part of Chapter 3, a new hybrid medium access control technique amenable to software radio development is proposed, which is Movable Frame Hybrid Medium Access Control (MFHMAC). In contrast, a hybrid conventional medium access technique named Fixed Frame Hybrid Medium Access Control (FFHMAC) is also discussed. The performance of new techniques is evaluated based on various load and types conditions.

In the second part of Chapter 3, the effects of FEC coding in MFHMAC technique is evaluated, and a comparison of MFHMAC technique (with and without FEC coding) with FFHMAC are made.

Finally, some conclusions and future work are brought in Chapter 4.

CHAPTER 2

OVERVIEW OF MAC

2.1 Introduction to Multiple Access [8]

The medium access control for the radio interface of a wireless network is an important system component since it has to provide both efficient uses of the source, radio bandwidth and maintain QoS guarantee over the connections. Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA) and Code Decision Multiple Access (CDMA) are the three major access techniques used to share the available bandwidth in a wireless communication system. These techniques can be grouped as narrowband and wideband systems, depending upon how the available bandwidth is allocated to the users. The duplexing technique of a multiple access system is usually described along with the particular multiple access scheme.

In addition to FDMA, TDMA and CDMA, the packet radio access scheme is another multiple access technique that is mainly used in wireless network. Such that Carrier Sense Multiple Access (CSMA), Packet Reservation Multiple Access (PRMA) and Aloha, Slotted Aloha etc.

Basically, for a multiple access technique to be deemed acceptable, it must be capable of meeting the following criteria:

Must support various traffic types with vastly different quality of service guarantees. Typically, these techniques should provide low average packet delay and minimal real-time loss, all while maximizing the channel utility. This implies that the performance of these schemes should be measured with traffic models that are commensurate with the traffic type. Unfortunately, the performance of the majority of MAC protocols is based on Poisson type traffic, which is not a valid model for the multimedia traffic considered.

2.2 Frequency Division Multiple Access (FDMA)

Frequency Division Multiple Access (FDMA) assigns individual channels to individual users. It can be seen from Figure 2.1 that each user is allocated a unique frequency band or channel. These channels are assigned on demand to users who request service. During the period of the call, no other user can share the same frequency band. In FDD systems, the users are assigned a channel as a pair of frequencies, one frequency is used for the forward channel, while the other frequency is used for the reverse channel. The features of FDMA are as follows [8]:

- If an FDMA channel is not in use, then it sits idle and cannot be used by other users to increase or share capacity. It is essentially a wasted resource.
- After the assignment of a voice channel, the base station and the mobile transmit simultaneously and continuously.

- The bandwidths of FDMA channels are relatively narrow (30 kHz) as each channel supports only one circuit per carrier. That is, FDMA is usually implemented in narrowband systems.
- The symbol time is large compared to the average delay spread. This implies that the amount of intersymbol interference is low and, thus, little or no equalization is required in FDMA narrowband systems.
- The complexity of FDMA mobile systems is lower when compared to TDMA systems, though this is changing as digital signal processing methods improve for TDMA.
- Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronization and framing bits) as compared to TDMA.
- FDMA systems have higher cell site system costs as compared to TDMA systems, because of the single channel per carrier design, and the need to use costly bandpass filters to eliminate spurious radiation at the base station.
- The FDMA mobile unit uses duplexers since both the transmitter and the receiver operate at the same time. This results in an increase in the cost of FDMA subscriber units and base stations.
- FDMA requires tight RF filtering to minimize adjacent channel interference.

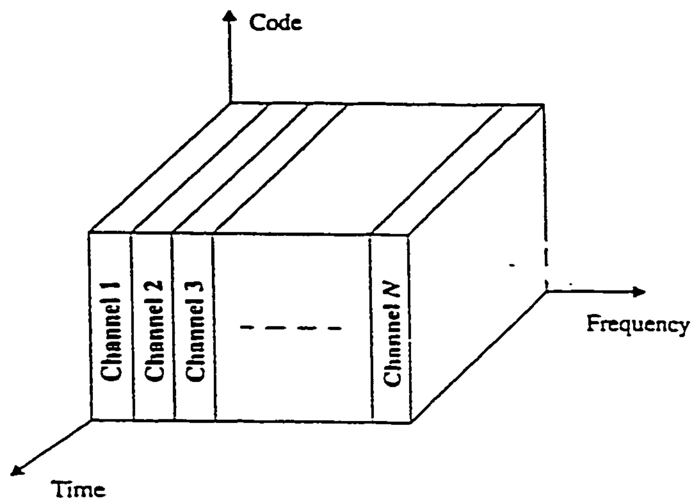


Fig. 2.1: FDMA where different channels are assigned different frequency bands.

2.3 Time Division Multiple Access (TDMA)

Time Division Multiple Access (TDMA) systems divide the radio spectrum into time slots, and in each slot only one user is allowed to either transmit or receive. It can be seen from Figure 2.2 that each user occupies a cyclically repeating time slot, so a channel may be thought of as particular time slot that reoccurs every frame, where N time slots comprise a frame. TDMA systems transmit data in a buffer-and-burst method, thus the transmission for any user is noncontiguous. This implies that, unlike in FDMA systems which accommodate analog FM, digital data and digital modulation must be used with TDMA. The transmission from various users is interlaced into a repeating frame structure. It can be seen that a frame consists of a number of slots. Each frame is made up of a preamble, an information message, and tail bits. In TDMA/TDD, half of the time slots in the frame information message would be used for the forward link channels and half would be used for reverse link channels. In TDMA/FDD systems, an identical or similar frame structure would be used solely for either forward or reverse transmission, but the carrier frequencies would be different for the forward and reverse links. In general, TDMA/FDD systems intentionally induce several time slots of delay between the forward and reverse time slots of a particular user, so that duplexers are not required in the subscriber unit.

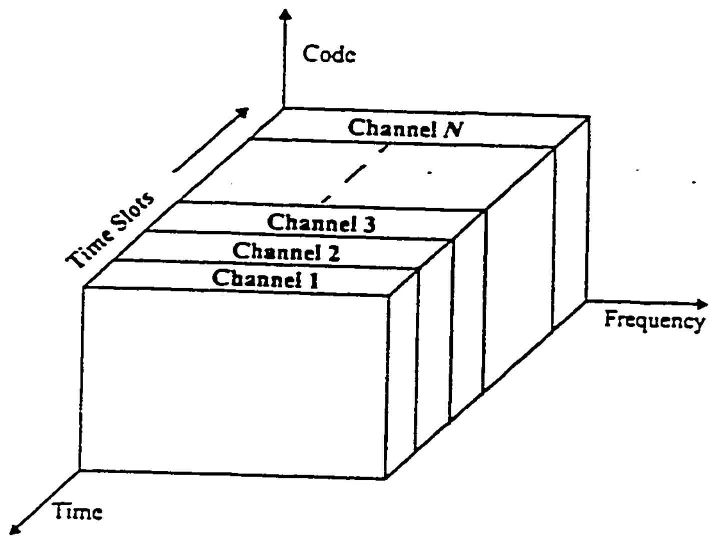


Fig. 2.2: TDMA scheme where each channel occupies a cyclically repeating time slot.

In a TDMA frame, the preamble contains the address and synchronization information that both the base station and the subscribers use to identify each other. Guard times are utilized to allow synchronization of the receivers between different slots and frames. Different TDMA wireless standards have different TDMA frame structures. The features of TDMA include the following [8]:

- TDMA shares a single carrier frequency with several users, where each user makes use of non-overlapping time slots. The number of time slots per frame depends on several factors, such as modulation technique, available bandwidth, etc.
- Data transmission for users of a TDMA system is not continuous, but occurs in bursts. This results in low battery consumption, since the subscriber transmitter can be turned off when not in use (which is most of the time).
- Because of discontinuous transmission in TDMA, the handoff process is much simpler for a subscriber unit, since it is able to listen for other base stations during idle time slots. An enhanced link control, such as that provided by Movable Assisted Handoff. (MAHO), can be carried out by a subscriber by listening on an idle slot in the TDMA frame.
- TDMA uses different time slots for transmission and reception, thus duplexers are not required. Even if FDD is used, a switch rather than a duplexer inside the subscriber units is all that is required to switch between transmitter and receiver using TDMA.
- Adaptive equalization is usually necessary in TDMA systems, since the transmission rates are generally very high as compared to FDMA channels.
- In TDMA, the guard time should be minimized. If the transmitted signal at the edges of a time slot are suppressed sharply in order to shorten the guard time,

the transmitted spectrum will expand and cause interference to adjacent channels.

- High synchronization overhead is required in TDMA systems because of burst transmissions. TDMA transmissions are slotted, and this requires the receivers to be synchronized for each data burst. In addition, guard slots are necessary to separate users, and this results in the TDMA systems having larger overheads as compared to FDMA.
- TDMA has an advantage in that it is possible to allocate different numbers of time slots per frame to different users. Thus bandwidth can be supplied on demand to different users by concentrating or reassigning time slots based on priority.

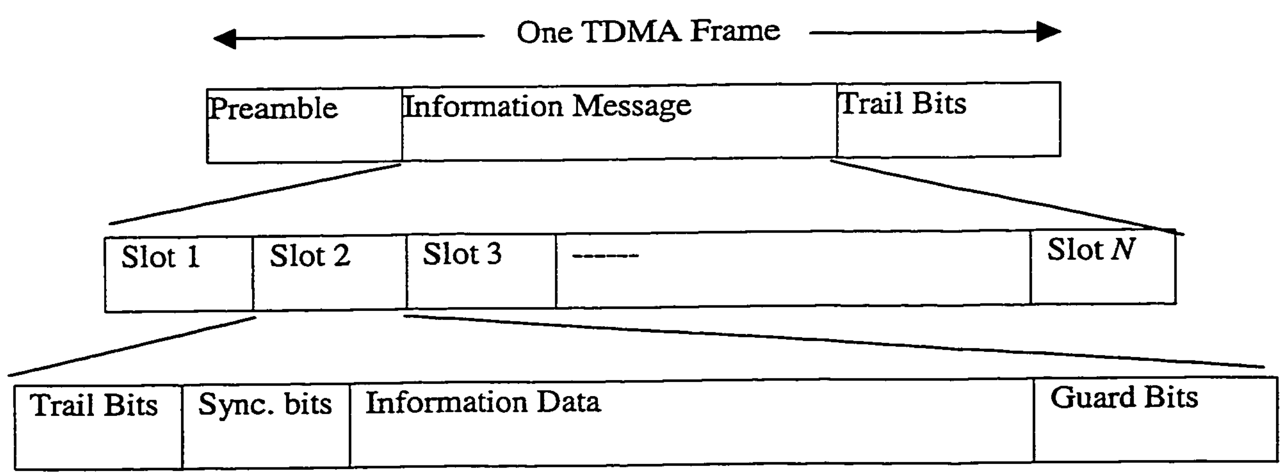


Fig. 2.3: TDMA frame structure.

2.4 Code Division Multiple Access (CDMA)

In *Code Division Multiple Access* (CDMA) systems, the narrowband message signal is multiplied by a very large bandwidth signal called the *spreading signal*. The spreading signal is a pseudo-noise code sequence that has a chip rate which is an order of magnitudes greater than the data rate of the message. All users in a CDMA system, as seen from Figure 2.4, use the same carrier frequency and may transmit simultaneously. Each user has its own pseudorandom codeword which is approximately orthogonal to all other codewords. The receiver performs a time correlation operation to detect only the specific desired codeword. All other codewords appear as noise due to decorrelation. For detection of the message signal, the receiver needs to know the codeword used by the transmitter. Each user operates independently with no knowledge of the other users.

In CDMA, the power of multiple users at a receiver determines the noise floor after decorrelation. If the power of each user within a cell is not controlled such that they do not appear equal at the base station receiver, then the *near-far problem* occurs.

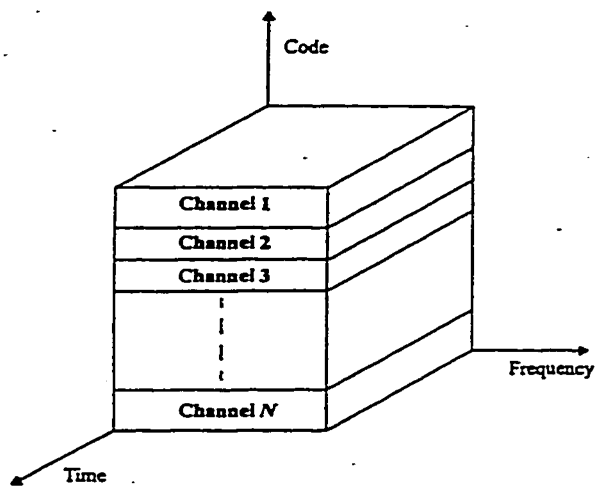


Fig. 2.4: CDMA in which each channel is assigned a unique PN code which is orthogonal to PN Codes used by other users.

The near-far problem occurs when many mobile users share the same channel. In general, the strongest received mobile signal will *capture* the demodulator at a base station. In CDMA, stronger received signal levels raise the noise floor at the base station demodulators for the weaker signals, thereby decreasing the probability that weaker signals will be received. To combat the near-far problem, *power control* is used inmost CDMA implementations. Power control is provided by each base station in a cellular system and assures that each mobile within the base station coverage area provides the same signal level to the base station receiver. This solves the problem of a nearby subscriber overpowering the base station receiver and drowning out the signals of far away subscribers. Power control is implemented at the base station by rapidly sampling the radio signal strength indicator (RSSI) levels of each mobile and then sending a power change command over the forward radio link. Despite the use of power control within each cell, out-of-cell mobiles provide interference which is not under the control of the receiving base station. The features of CDMA include the following [8]:

- Many users of a CDMA system share the same frequency. Either TDD or FDD may be used.
- Unlike TDMA or FDMA, CDMA has a soft capacity limit. Increasing the numbers of users in a CDMA system raises the noise floor in a linear manner. Thus, there is no absolute limit on the number of users in CDMA. Rather, the system performance gradually degrades for all users as the number of users is increased, and improves as the number of users is decreased.
- Multipath fading may be substantially reduced because the signal is spread over a large spectrum. If the spread spectrum bandwidth is greater than the coherence bandwidth of the channel, the inherent frequency diversity will mitigate the effects of small-scale fading.

- Channel data rates are very high in CDMA systems. Consequently, the symbol (chip) duration is very short and usually much less than the channel delay spread. Since PN sequences have low autocorrelation, mulitpath which is delayed by more than a chip will appear as noise. A RAKE receiver can be used to improve reception by collecting time delayed versions of the required signal.
- Since CDMA uses co-channel cells, it can use macroscopic spatial diversity to provide soft handoff. Soft handoff is performed by the MSC, which can simultaneously monitor a particular user from two or more base stations. The MSC may choose the best version of the signal at any time without switching frequencies.
- Self-jamming is a problem in CDMA system. Self-jamming arises from the fact that the spreading sequences of different users are not exactly orthogonal, hence in the despreading of a particular PN code, non-zero contributions to the receiver decision statistic for a desired user arise from the transmissions of other users in the system.
- The near-far problem occurs at a CDMA receiver if an undesired user has a high detected power as compared to the desired user.

2.5 Carrier Sense Multiple Access (CSMA) Protocols

ALOHA (Slotted ALOHA) and CSMA are common used in bursty packet radio networks [9]. ALOHA protocols do not listen to the channel before transmission, and therefore do not exploit information about the other users. By listening to the channel before engaging in transmission, greater efficiencies may be achieved. CSMA protocols are based on the fact that each terminal on the network is able to monitor the status of the channel before transmitting information. If the channel is idle (i.e., no carrier is detected), then the user is allowed to transmit a packet based on a particular algorithm which is common to all transmitters the network [8].

In CSMA protocols, *detection delay* and *propagation delay* are two important parameters. Detection delay is a function of the receiver hardware and is the time required for a terminal to sense whether or not the channel is idle. Propagation delay is a relative measure of how fast it takes for a packet to travel from a base station to a mobile terminal. With a small detection time, a terminal detects a free channel quite rapidly, and small propagation delay means that a packet is transmitted through the channel in a small interval of time relative to the packet duration.

Propagation delay is important since just after a user begins sending a packet, another user may be ready to send and may be sensing the channel at the same time. If the transmitting packet has not reached the user who is poised to send, the later user will sense an idle channel and will also send its packet, resulting in a collision between the two packets. Propagation delay impacts the performance of CSMA protocols. If t_p is the propagation time in seconds, R_b is the channel bit rate, and m is the expected number of bits in a data packet, then the propagation delay t_d (in packet transmissions units) can be expressed as

(2-1)

$$t_d = \frac{t_P R_b}{m} \tag{2-1}$$

There exist several variations of the CSMA strategy:

- **1-persistent CSMA** - The terminal listens to the channel and waits for transmission until it finds the channel idle. As soon as the channel is idle, the terminal transmits its message with probability one.
- **non-persistent CSMA** - In this type of CSMA strategy, after receiving a negative acknowledgement the terminal waits a random time before retransmission of the packet. This is popular for wireless LAN applications, where the packet transmission interval is much greater than the propagation delay to the farthermost user.
- **p-persistent CSMA** - *p*-persistent CSMA is applied to slotted channels. When a channel is found to be idle, the packet is transmitted in the first available slot with probability p or in the next slot with probability 1-p.
- **CSMA/CD** - In CSMA with collision detection (CD), a user monitors its transmission for collisions. If two or more terminals start a transmission at the same time, collision is detected, and the transmission is immediately aborted in midstream. This is handled by a user having both a transmitter and receiver which is able to support *listen-while-talk* operation. For a single radio channel,

this is done by interrupting the transmission in order to sense the channel. For duplex systems, a full duplex transceiver is used.

In Summary, the most suitable MAC protocols for more bursity traffic is CSMA and for more stream traffic is FDMA. Based on different types of network and different traffic, some MAC protocols are used in different networks, e.g. CSMA is used in Ethernet and TDMA with slotted Aloha is used in NEC. Table 2.1 below shows the multiple access techniques which should be used for different types of traffic conditions [8].

Type of Traffic	**Multiple Access Technique**
Bursty, short messages	Contention protocols
Bursty, long messages, large number of users	Reservation protocols
Bursty, long messages, small number of users	Reservation protocols with fixed TDMA reservation channel
Stream or deterministic (Voice)	FDMA, TDMA, CDMA

Table 2.1 Multiple Access Techniques for Different Traffic Types

CHAPTER 3

THE MOVABLE FRAME HYBRID MULTIPLE ACCESS SYSTEM

3.1 Objective and Motivation

The third generation of wireless communication will be strongly dominated by multimedia traffic. One approach to the design of such a system is represented by the proposal of an Integrated Wireless Access Network (IWAN) [1], whereby different services and systems are all integrated into one wireless system. IWAN presents several new challenges in its system design. One of the most serious challenges is the design of an efficient and robust Medium Access Control (MAC) protocol that can integrate heterogeneous traffic types and meet their requirements for Quality of Service (QoS) [1]. On the other hand, there has been a growing interest in designing flexible wireless networks where the modulation, FEC, access air interface, routing techniques etc change from time to time depending on traffic and QoS requirements. The software radio also leads the mobile unit made small and flexible to receive the down loaded software of all the applicable algorithms and program itself accordingly. This whole new area is called software radio [33,34,35].

The software radio figures prominently in plans for third generation (3G) wireless services architectures and products [33]. A software radio defines channel access and waveform synthesis (not just selection) in software. "Plug-and-Play" creates a variable architecture as modules are introduced into the environment and removed [34]. Software radios migrate the traditional hard-wired radio platforms to flexible software radio platforms that can support multiple modulation waveforms and multiple access standards [35]. These new ideas make it quite applicable that choose more than one MAC protocols in one system.

There are several types of wireless MAC protocols described in the literature. Many multi-access schemes currently deployed in mobile cellular networks employ a fixed frame structure. For instance, TDMA is a popular choice for several cellular systems such as GSM, IS-54, PDC, and for wireless personal communication systems such as DECT, WACS. However, to cater for packetized multimedia traffic, the selection of a suitable TDMA frame structure is a non-trivial task since it is unlikely that the exact mix of applications will either be known beforehand or remain stable. On the other hand, all conventional schemes like TDMA and CSMA restrict the user to a scheduled transmission scheme and require the users to wait until their next time slot to transmit (3). From this point, the general wireless protocols designed for voice networks use TDMA (GSM and DECT) and do not handle data efficiently, while those designed for data networks typically use some form of CSMA/CA (IEEE 802.11 and HIPERLAN) and do not handle voice traffic efficiently [2].

To overcome the limitation of current MAC technologies we have to investigate how to retain the advantage of both types of MAC protocols without increasing the overall complexity of the system while improving the quality of service for each class of traffic.

The purpose of this thesis is to provide a more efficient access technique named Movable Frame Hybrid Medium Access Control (MFHMAC) by combining existing TDMA, FDMA, CDMA and CSMA protocols properly to guarantee the QoS in each traffic. The proposed system is herterochronous in the sense that packet generation from any user can be any rate with any bursty level. Our strategy is to allocat different bandwidth to different traffic based on traffic rate and bursty level finally controls the desired quality of service in each traffic. In contrast, a Fixed Frame Hybrid Medium Access Control (FFHMAC) technique is also discussed in this thesis.

3.2 Description of FFHMAC Technique

Depending on the packet generated bursty level, we classify the traffic into two classes: bursty traffic and stream traffic. Based on the packet arrive rate, we also classify the traffic into two classes: high rate traffic and low rate traffic. Hence, there are a total four classes of traffic in the system. For instance, we defined:

Class 1 traffic: Bursty and low rate ($\theta<0.5$ and $R_1= \delta_L$).

Class 2 traffic: Bursty and high rate ($\theta<0.5$ and $R_2 = \delta_H$)

Class 3 traffic: Stream and low rate ($\theta>0.5$ and $R_3=\delta_L$).

Class 4 traffic: Stream and high rate ($\theta>0.5$ and $R_4 = \delta_H$).

Where θ represents the traffic level, R represents the packet arrive rate (bps) and δ_L and δ_H are normalized high rate and low rate traffic ratio, respectively. The traffic rate ratio is the ratio of the real traffic rate to standard packet rate.

We also assume all the users in each class are homogenous, this means each user has the same traffic descriptors and will require the same quality of service.

To accommodate heterogeneous class of traffic in one system, we propose a hybrid MAC protocol. This technique uses four access methods with four queuing buffers: Class 1 traffic uses CDMA to access the network and occupy 1/4 system bandwidth: Class 2 traffic uses CSMA to access the network and occupy another 1/4 bandwidth: Class 3 traffic uses TDMA to access the network and occupies the other 1/4 bandwidth: Class 4 traffic uses FDMA to access the network and also occupy 1/4 system bandwidth. Here, we say it is a fixed frame technique, meaning that as the traffic arrives from the source with different bursty levels and rates, it is buffered in four different finite-length buffers and then accesses the four corresponding bands by a fixed allocation. This can be shown in Fig.3. 1.

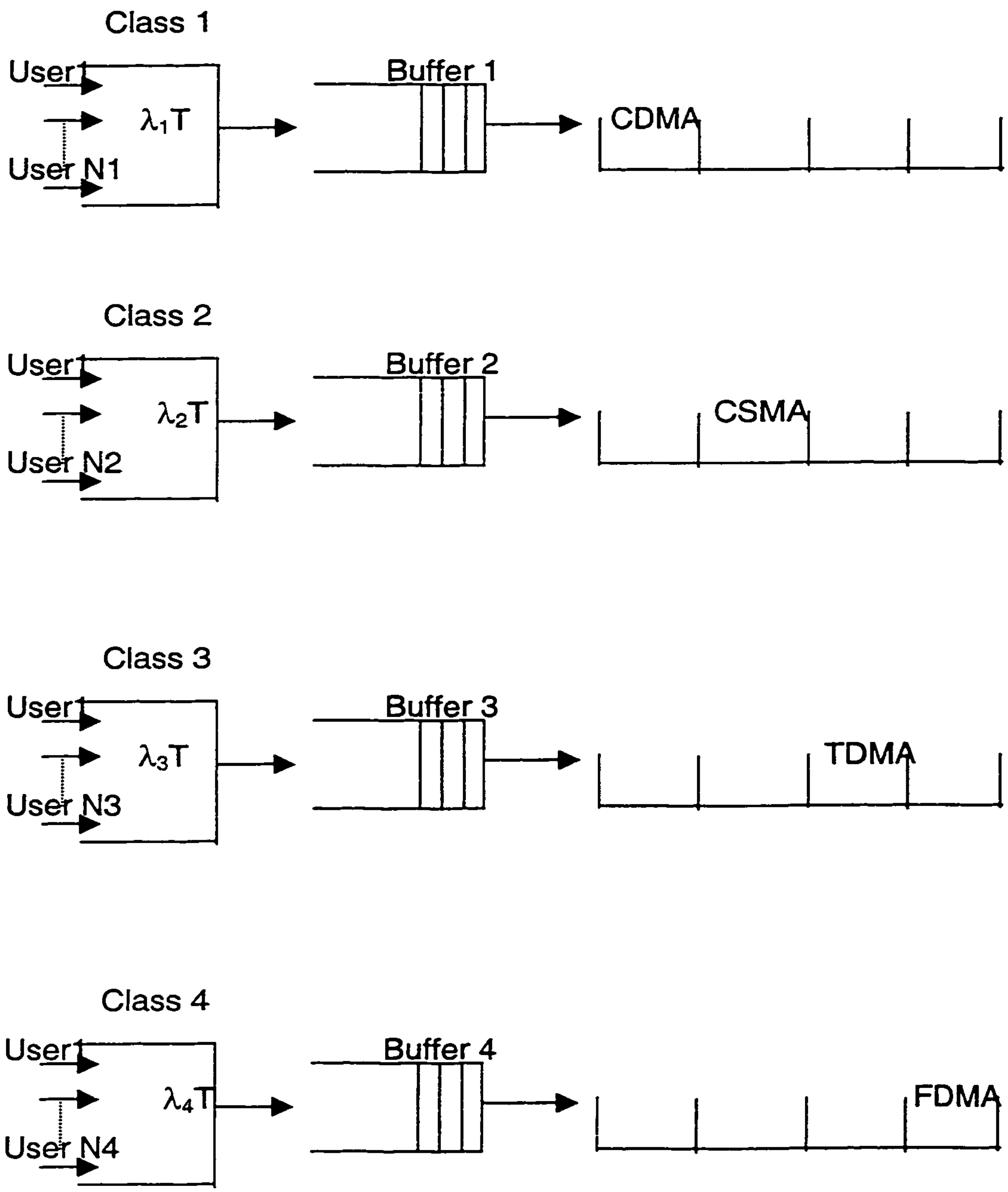


Fig. 3.1: FFHMAC system model. N1, N2, N3, N4 are numbers of users in each class respectively. λ_i (i=1,2,3,4) is the traffic intensity at the ith class user.

We assume that the mobile receives basic synchronization information from the base station and is able to transmit packet slot-synchronously. If there is any packet in the queue, the user attempts transmission at the beginning of the next slot.

Once a random user starts to generate the packet(s) to transmit, we define three parameters to characterize its traffic model, which is $(\theta_i, \delta_i, T_{d(I,j)})$. θ_i and δ_i, as mentioned above, and $T_{d(i,j)}$ is the call duration of the jth user in the ith class. Each can be assumed to have four possible states: "Silent," "In-progress," "Waiting" and "Blocked," based on the state of the buffer (full or empty) and the previous transmission. This can be shown in Fig. 3.2.

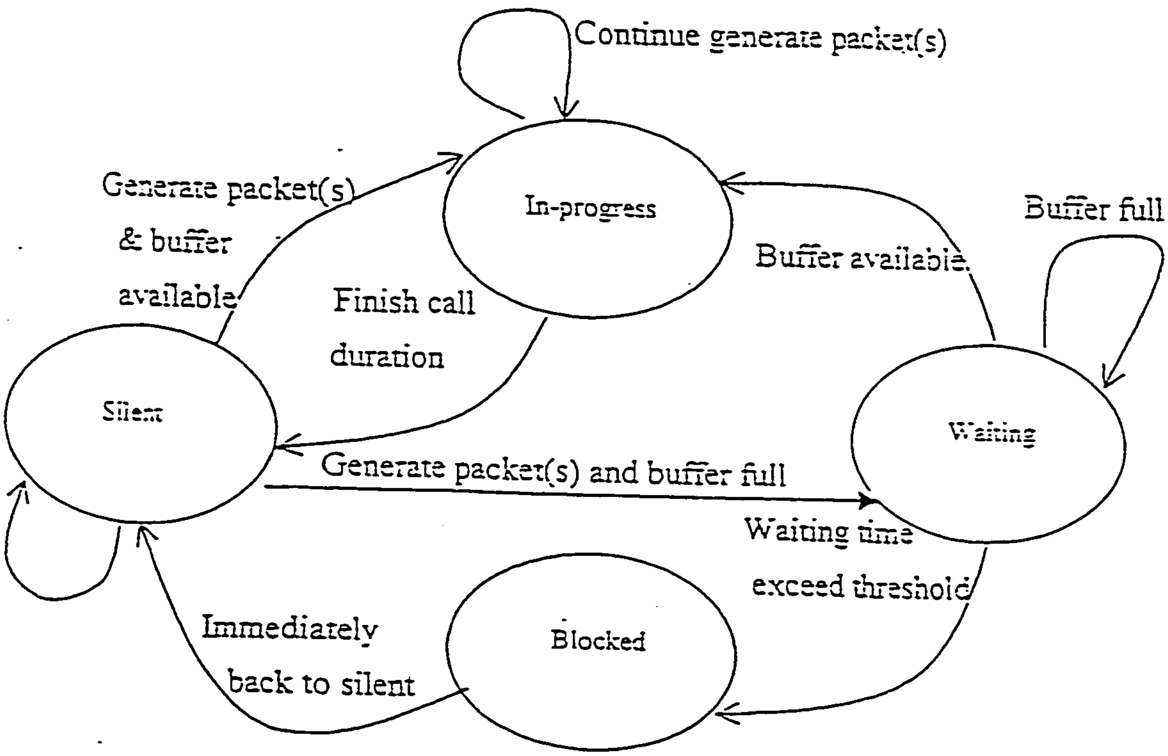


Fig. 3.2: The transition model of the FFHMAC system.

If there is no packet(s) needed to transmit, the user can be assumed to be in the "silent" state. If a "silent" user attempts to generate packet(s) and acquire the channel, the packet(s) is (are) queued in the transmission buffer, the user is in the "in-progress" state. If the user can not acquire the available channel (slots), then he will be in a "waiting" state. If a "silent" user still does not have a packet(s) to transmission, his next state will still be in "silent" state. When a user is currently in the "in-progress" state, after finishing his call duration, he will return to the "silent" state. During the transmission of the "in-progress" user, the system will keep the channel (slots) which the user already occupies, meaning this user will continue to be in the "in-progress" state before finishing his call. For a current in "waiting" state user, if his waiting time exceeds a certain threshold (maximum delay tolerance), this call will be clocked and this blocked user will immediately go back to the "silent" state. If the "waiting" state user acquires the available channel (slots), his next state will be in the "in-progress" state. Otherwise, the "waiting" user will still be in a "waiting" state.

If we consider the FFHMAC scheme as a combination of four conventional access techniques, which is CDMA/CSMA/TDMA/FDMA, then we can easily describe the access policies.

- For Class 1 traffic, we assign a CDMA channel for him and check the QoS satisfaction. If QoS is satisfied, CDMA band accepts this call; if QoS is not satisfied, this call has to wait for the next frame. At each time slot for each Class 1 user, repeat the above procedure until the call is accepted or blocked.
- For Class 2 traffic, we assign a CSMA channel, and check the QoS satisfaction. If the QoS is satisfied, the packet(s) can be transmitted by this

channel. If not, this call have to wait for next frame. At each time slot each Class 2 user who has the packet(s) needed to transmit will undergo the above procedure until the call will be accepted or the waiting time exceeds the maximum delay tolerance (MDT), then the call will be blocked.

• For Class 3 traffic, we assign the TDMA slot(s), if there exists available slots(s) in the current frame; if not, the packet(s) will have to wait for the next frame. At each time slot each Class 3 user who attempts to transmit his packet(s) will undergo the above procedure until the call will be accepted or the waiting time exceeds MDT, then the call will be blocked.

• For Class 4 traffic, we assign an FDMA channel, if there exists an available FDMA sub-channel; if not, the packet(s) will have to wait for the next frame. At each time slot each Class 4 user who has a packet(s) needed to transmit will undergo the above procedure until the packet(s) was (were) transmitted or the waiting time exceeds the MDT, then the call will be blocked.

• For all classes of traffic, when we allocate the channel (slot, sub-channel) for those, we must consider some priority rules based on each user's previous state:

(1) First serve the "in-progress" users if they still have packet(s) needed to transmit (didn't finish call duration). Keep the time slots (in TDMA frame) or a sub-channel (in FDMA frame) for him, until his call duration is finished.

(2) After the "in-progress" users, we first serve the users whose previous state was in "waiting." Unless the waiting time exceeds MDT.

(3) After the "waiting" users, we serve the new users whose previous state was "silent."

Finally, another important rule as mentioned in [1] when call generated packet(s) with rate $\delta_i(i=1,2,3,4)$, it will be assigned δ_i times bandwidth of the standard rate packet bandwidth in one frame.

3.3 Description of MFHMAC Technique

3.3.1 Introduction

It appears that the FFHMAC scheme is suitable for fairly distributed traffic load (four classes of traffic that are uniformly distributed), since each class of traffic independently corresponds with one fixed access technique. In order to adjust channel bandwidth allocation and improve the system performance by the concerned desired QoS, we introduce a Movable Frame Hybrid Medium Access (MFHMAC) technique that deals with a more complicated traffic load by setting a movable frame scheme.

As mentioned in 3.3, we also classify all traffic into four classes, but the difference is that we divide our total bandwidth into two parts: basic band and reservation band. The basic band also has four equal access bands: CDMA, CSMA, TDMA and FDMA. The reservation band can be used by any class of traffic, depending on the total system traffic load and the basic band state. This is shown in Fig. 3.3.

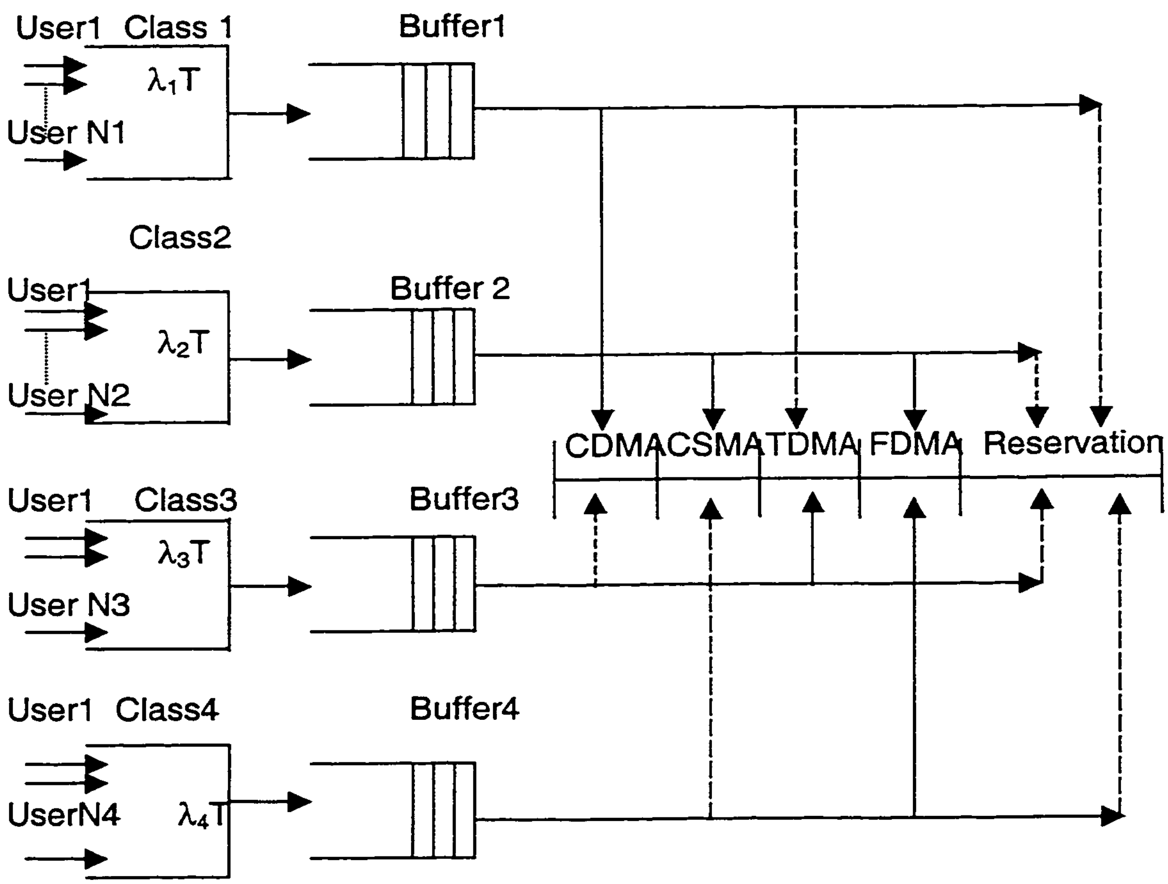


Fig. 3.3: System model of MFHMAC.

Initially, the reservation band is a free band. Based on the arriving traffic, we set the following access strategy:

(1) Users on basic CDMA, CSMA, TDMA and FDMA band will remain on an equally fixed bandwidth, even if they have very light traffic.

(2) As the traffic builds up, users will start to use some of the reservation band.

(3) Any user generate packet(s) at a certain time slot with probability θ_i.

(4) A certain call generate packet(s) from Class 1: We first try to accommodate it into basic CDMA band and check if the QoS satisfied. If not, we check if there exists a reservation CDMA band and try to put the packet(s) into this reservation CDMA band by checking the QoS satisfaction. If not, we try to accommodate the call into basic TDMA band if there exists available TDMA slots. If not, we then check if there exists a TDMA reservation band and try to put packet(s) into this reservation TDMA band if there exists available reservation TDMA slots. If not, we try to open a reservation CDMA band if there are some free bands in the reservation band, and put the packet(s) into this reservation CDMA band. If not, this call will have to wait for the next frame. This is shown in Fig 3.4.

(5) A certain call from Class 2 generate packet(s): We first try to accommodate it into the basic CSMA band and check if the QoS is satisfied. If not, we check if there exists a reservation CSMA band and try to put the packet(s) into this reservation CSMA band by checking the QoS satisfaction. If not, we try to put the packet(s) into basic FDMA band if there exists an available sub-channel. If not, we check if there already exists a reservation FDMA band and put the packet(s) into this reservation FDMA band if there exists some available sub-channels. If not, we try to open a reservation CSMA band if there exists some free bands in the reservation band, then put the packet(s) into this reservation CSMA band. If not, this call will have to wait for the next frame. This can be shown in Figure 3.5.

(6) A certain call from Class 3 generate pack3et(s): We first try to accommodate it into basic TDMA band if there exists an available TDMA slot(s). If not, we check if there exists a reservation TDMA band and has an available TDMA slot(s), then put the packet(s) in this reservation TDMA band. If not, we try to put it in basic CDMA band if the QoS satisfied. If not, we check if there exists reservation CDMA band and put the packet(s) in this reservation CDMA band. If not, we check if there exists free bands in the reservation band and open a reservation TDMA band, then put the packet(s) in this reservation TDMA band. If not, this call will have to wait for the next frame. This is shown in Fig. 3.6.

(7) A certain call from Class 4 generate packet(s): We first try to accommodate it in basic FDMA band if there exists FDMA sub-channel. If not, we check if there exists a reservation FDMA band and has sub-channel(s) available, then put the packet(s) in this reservation FDMA band. If not, we try to put packet(s) in basic CSMA band, if the the QoS are satisfied with this new call. If not, we check if there exists a reservation CSMA band and try to put packet(s) in this reservation CSMA band, if the QoS are satisfied with this new call. If not, we try to put the packet(s) in basic TDMA band if there exists available TDMA slot(s). If not, we check if there exists a reservation TDMA band and try to put the packet(s) in this reservation TDMA band if there exists available TDMA slot(s). If not, we try to put the packet(s) in basic CDMA band if the QoS are satisfied with new call. If not, we check if there exists a reservation CDMA band and try to put the packet(s) in this

reservation CDMA band if the QoS are satisfied with this new call. If not, we check if there exists some free bands in the reservation band and try to open a FDMA reservation band, then put the packet(s) in this reservation FDMA band. If not, this will have to wait for the next frame. This is shown in Figure 3.7.

(8) The priority rules are the same as FFHMAC.

(9) After one frame was transmitted, if there are no more packets in the reservation band, this reservation band will immediately go back to the pool (become a free band).

(10) For each technique used in the reservation band, only one reservation band is allowed.

(11) For the user who is in the "in-progress" state and uses the reservation band , he will still occupy this slot (for reservation TDMA and CSMA) or sub-channel (for reservation FDMA) during the next frame, until the call is finished.

(12) In each buffer, if any call's waiting time exceeds MDT, this call will be blocked.

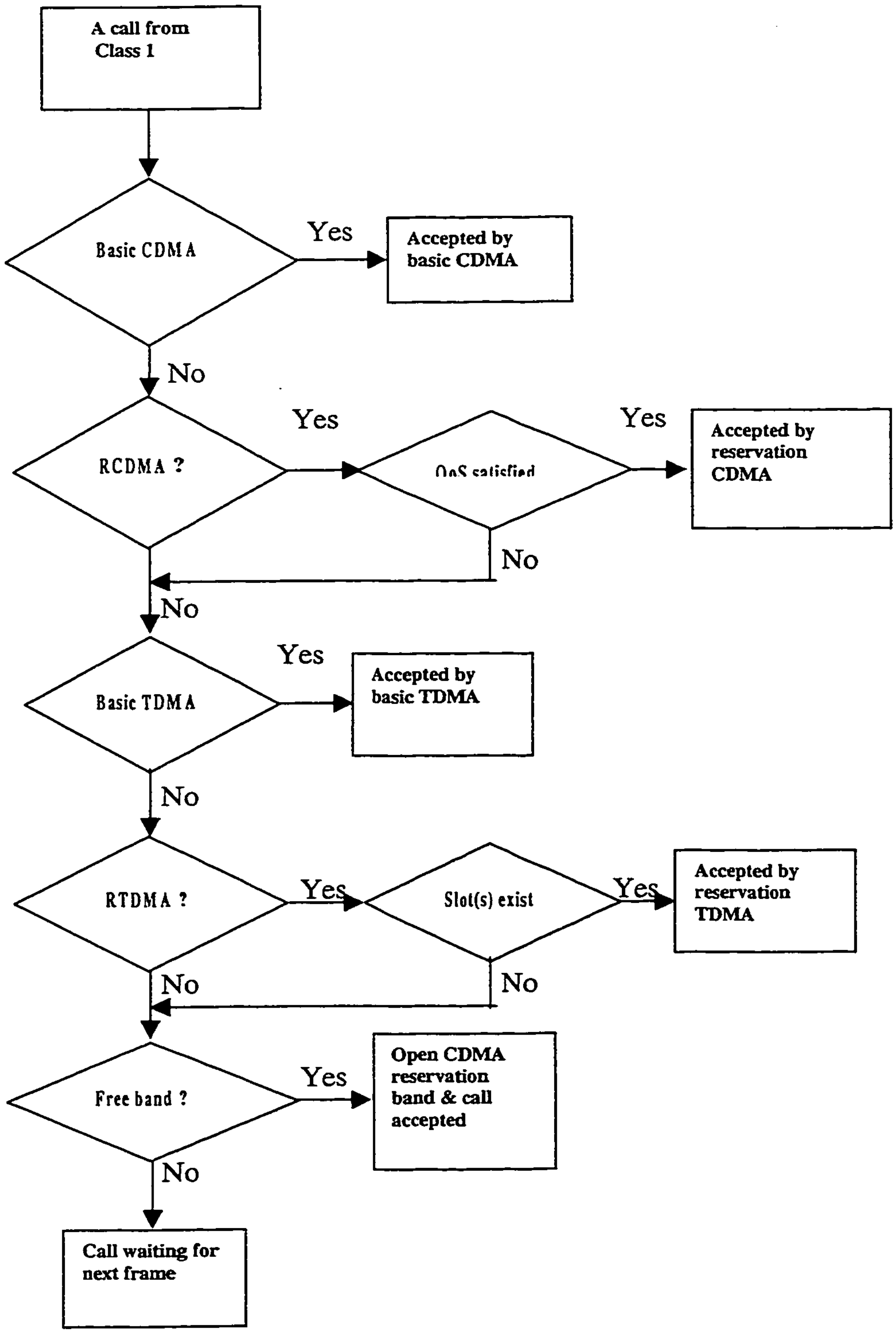


Fig.3.4: Block diagram of class 1 traffic in the MFHMAC scheme.

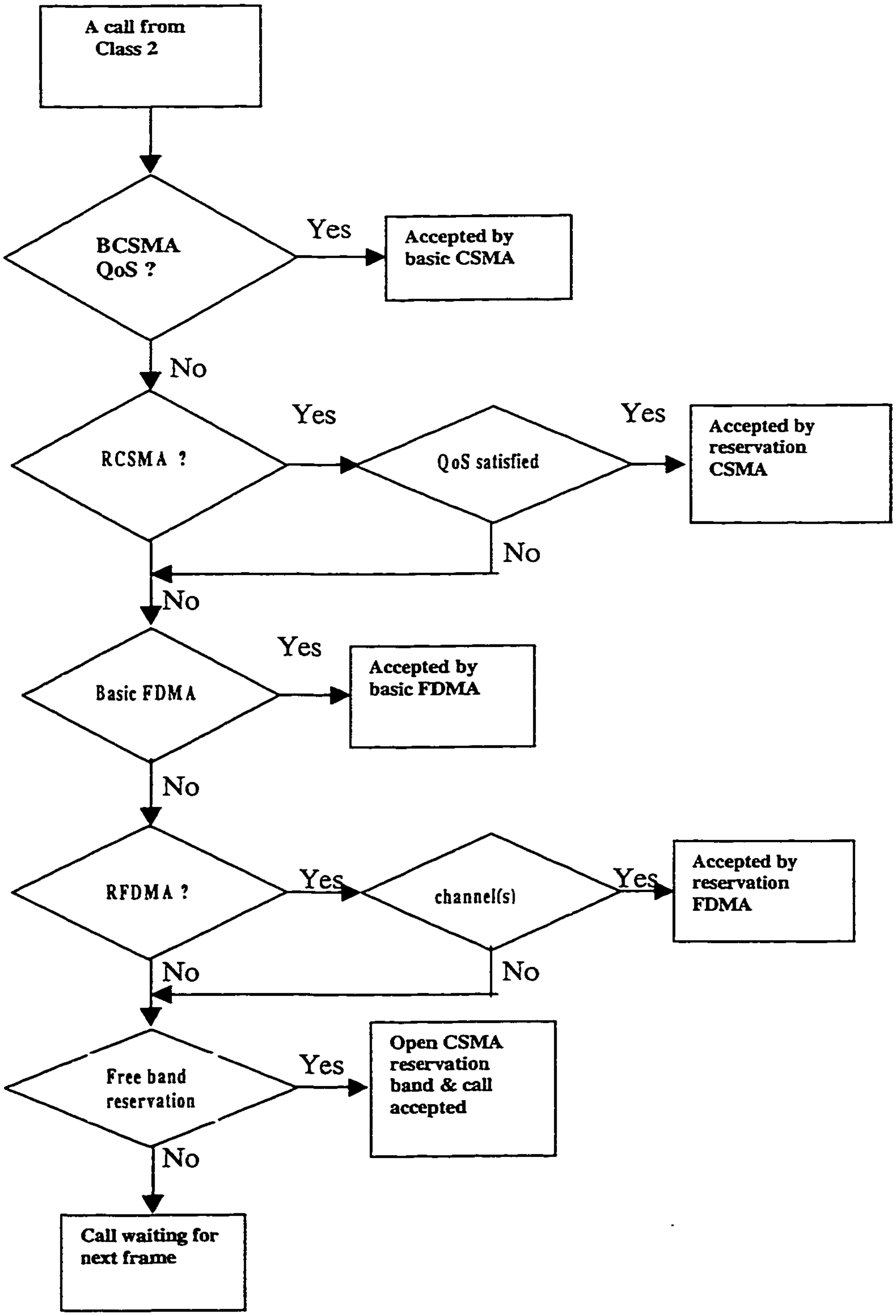


Fig.3.5: Block diagram of class 2 traffic in the MFHMAC scheme.

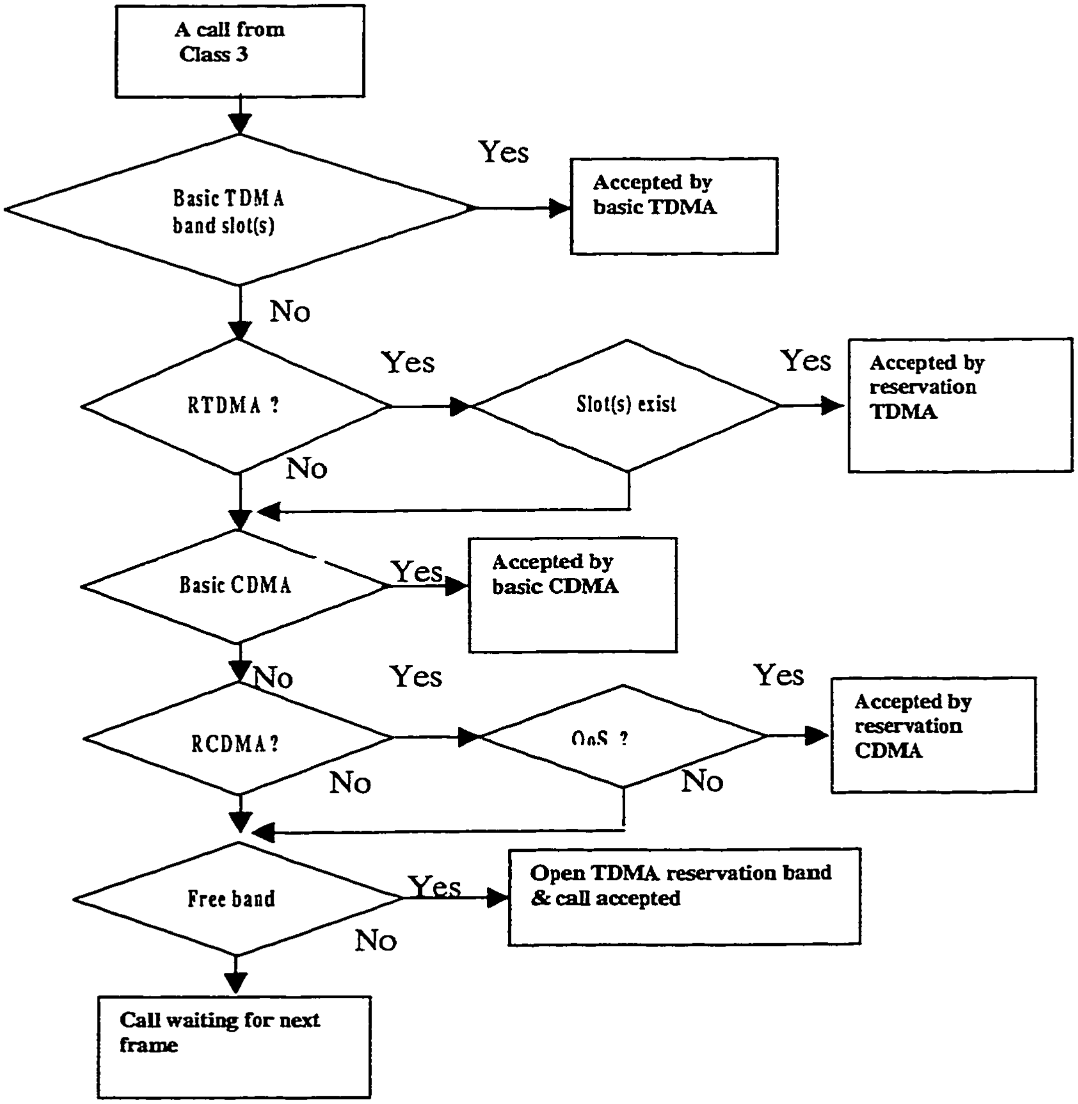


Fig.3.6: Block diagram of Class3 traffic in the MFHMAC system.

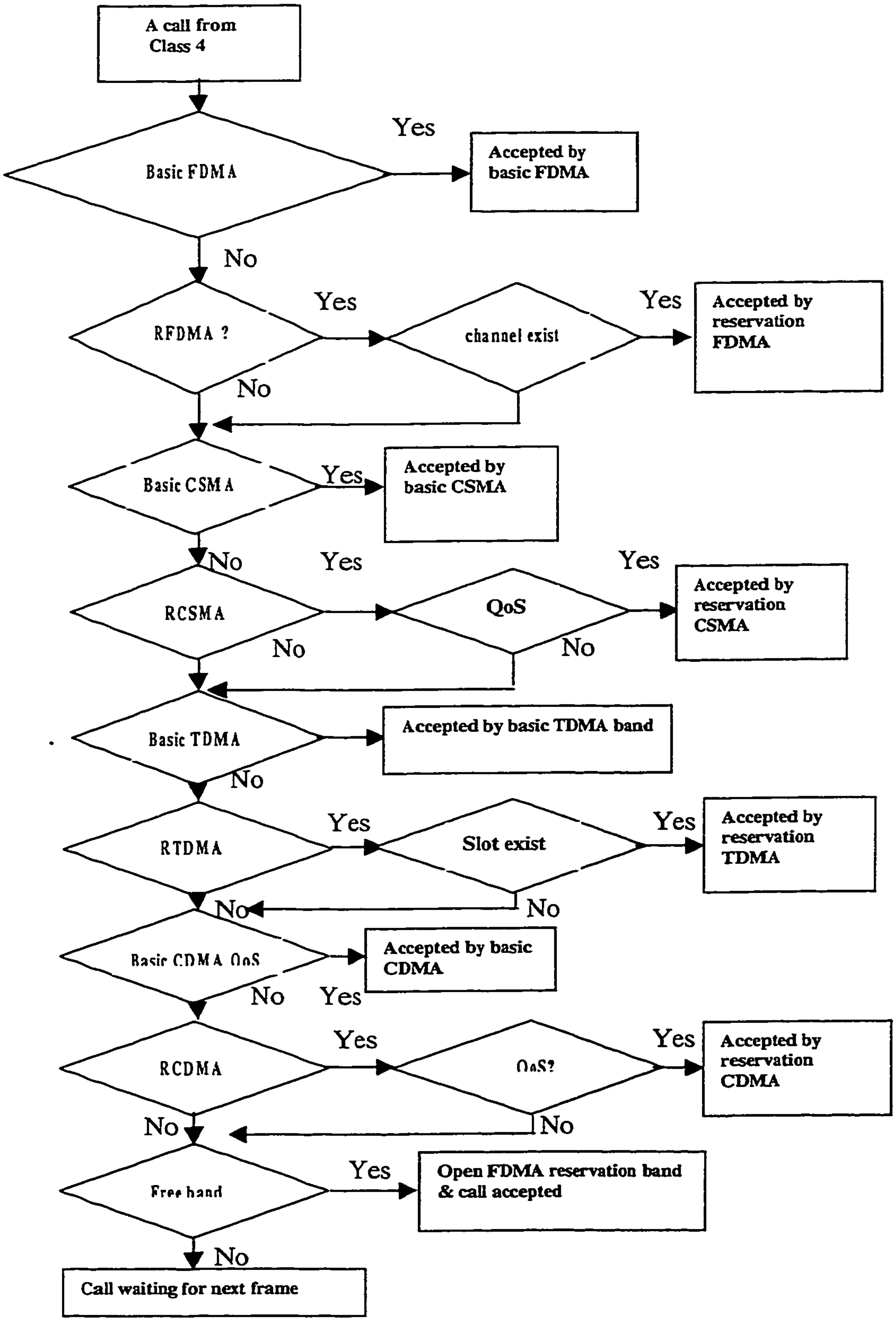


Fig.3.7 Block diagram of class 4 traffic in the MFHMAC scheme.

Next section, we will assess the QoS requirement in each access technique (band). As mentioned in Chapter 1, any access scheme can be evaluated through various criteria. The performance related to QoS have some important parameters:

(1) Average throughput

(2) Average call blocking probability.

(3) Average packet delay

(4) Delay variation (delay jitter)

(5) Probability of buffer overflowing.

In this thesis, we evaluate MFHMAC performance in view of these five criteria.

3.3.2 Average throughput analysis

The average throughput is defined as the ratio of the number of packets that are successfully transmitted in a very long interval to the maximum number of packets that could have been transmitted with continued transmission on the channel [9]. By simulation model, we can write the average throughput as:

In CDMA band:

$$S_{CDMA} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{4}\sum_{k=1}^{N_j} In_{(i,j,k)}\delta_{(i,jk)}}{C_{CDMA}\delta_{(r)}S_n} \qquad (3\text{-}1)$$

Where S_n is the number of time slots over a certain time interval, N_j is the number of users in jth class.

$$In_{(i,j,k)} = \begin{cases} 1 & \text{when kth user of jth class is "In - progress" at ith time slot} \\ 0 & \text{otherwise} \end{cases} \qquad (3\text{-}2)$$

and $\delta_{(i,j,k)}$ is the number of packets that kth user of jth class generated at ith slot

C_{CDMA} is the capacity of the CDMA band as defined in [6]:

$$C_{CDMA} = 1 + \frac{W/R}{\frac{E_b}{N_0}} - \frac{\eta}{S} \tag{3-3}$$

where W/R is processing gain, E_b / N_0 is SNR, η is background noise and S is the signal power.

In CSMA band:

$$S_{CSMA} = \frac{G_{(i)} \ell^{-2G_{(i)}}}{G_{(i)}(1+2\alpha) + \ell^{-2G_{(i)}}} \tag{3-4}$$

$where \quad G_{(i)} = \sum_{i=1}^{S_n} \sum_{j=1}^{4} \sum_{k=1}^{N_j} In_{(i,j,k)} \delta_{(i,j,k)}$ is the total traffic (including retransmission). (3-5)

and α is the normalized ratio of the propagation delay to the packet size

In TDMA band:

$$S_{TDMA} = \frac{\sum_{i=1}^{S_n} \sum_{j=1}^{4} \sum_{k=1}^{N_j} In_{(i,j,k)} \delta_{(i,j,k)}}{C_{TDMA} S_n} \tag{3-6}$$

Where C_{TDMA} is the capacity of the TDMA band (users/channel), which is the total slots in the TDMA channel.

In FDMA band:

$$S_{FDMA} = \frac{\sum_{i=1}^{S_n} \sum_{j=1}^{4} \sum_{k=1}^{N_j} In_{(i,j,k)} \delta_{(i,j,k)}}{C_{FDMA} S_n} \tag{3-7}$$

Where C_{FDMA} is the capacity of the FDMA band (users/channel), which is the total sub-channel of the FDMA channel.

3.3.3 Call blocking probability analysis

In the traditional trucking theory [8], the call blocking probability is defined by the Erlang B formula:

$$\Pr_{[blocking]} = \frac{\frac{A^C}{C!}}{\sum_{k=0}^{C} \frac{A^k}{k!}} \qquad (3\text{-}8)$$

Where C is number of available traffic channels and A is total traffic load.

And the call delayed probability by Erlang C:

$$\Pr_{[calldelayed]} = \frac{A^C}{A^C + C!(1-\frac{A}{C})\sum_{k=0}^{C} \frac{A^k}{k!}} \qquad (3\text{-}9)$$

So, the probability that any call is delayed in the queuing buffer for waiting time greater than t seconds is given by:

$$\Pr_{[wait>t\,/\,delay]} = e^{-\frac{C-A}{H}t} \qquad (3\text{-}10)$$

$$P_{[waiting>t]} = \Pr_{[call\ \mathrm{delayed}]} \Pr_{[waiting>t\ /\ delayed]}$$

$$= \left[\frac{\mathrm{A}^{\mathrm{C}}}{\mathrm{A}^{\mathrm{C}} + C!(1-\frac{A}{C})\sum_{k=0}^{C-1}\frac{A^k}{k!}} \right] e^{-\frac{C-A}{H}t} \tag{3-11}$$

In FFHMAC scheme, by setting t=MDT, we can easily get the average call blocking probability in each access band (CDMA/CSMA/TDMA/FDMA), However, in MFHMAC schemes, it's difficult to obtain the average call blocking probability through equation 3-11, since each frame is flexible (both in basic band and reservation band), parameters such as A and H can not be immediately determined. To simplify the evaluation of the call blocking probability, we use the simulation model to obtain the call blocking probability:

$$\Pr_{[call\ \mathrm{blocking}]} = \frac{\sum_{i=1}^{S_n}\left[B_{N_1(i)} + B_{N_2(i)} + B_{N_3(i)} + B_{N_4(i)}\right]}{(N_1 + N_2 + N_3 + N_4)S_n} \tag{3-12}$$

Where $B_{N1(i)}$, $B_{N2(i)}$, $B_{N3(i)}$, $B_{N4(i)}$ are the number of blocked users at ith slot in each class of users, and N_1, N_2, N_3 and N_4 are the number of users in each class respectively.

3.3.4 Delay analysis

We define the delay as the time from the instant that the caller generates packets to the instant when these are successfully received. The average packet delay is the ratio of the total delay of the packets in a very long interval to the number of the packets in the interval [9].

For Fixed Frame Access Technique (FFHMAC), the average delay in each band is independent and the equation is already derived in [4,5]:

$$D_{CDMA} = \frac{\rho T_F \overline{E^2}}{2(1-\rho)\overline{E}} + \overline{E}T_F \tag{3-13}$$

where $\rho = \lambda/\mu$ is the traffic intensity

T_F is the frame length

$\overline{E}$ and $\overline{E^2}$ are the average packet size

$$D_{TDMA} = \frac{\rho T_F \overline{E^2}}{2(1-\rho)\overline{E}} + (\overline{E} - \frac{1}{2})T_F + T_F / U \tag{3-14}$$

where U is the number of slots in one TDMA channel

$$D_{FDMA} = \frac{\rho T_F \overline{E^2}}{2(1-\rho)\overline{E}} + \overline{E}T_F \tag{3-15}$$

$$D_{CSMA} = \frac{NL\lfloor mS_{CSMA(N)} + 1 \rfloor}{mS_{CSMA(N)}} \tag{3-16}$$

where N is the number of users in the channel.

L is the buffer capacity.

m is the packet length in CSMA slots.

S_{CSMA} is the throughput of the CSMA channel, which is a function of N.

For Movable Frame Access Technique (MFHMAC), due to the reservation band's flexibility and each user's various occupation of each band, the theoretical model for average delay computation will become more complex. In this thesis, we evaluate the average delay through simulation.

$$D = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{4}\sum_{k=1}^{N_j}\delta_{(i,j,k)} WT_{(i,j,k)} n_{(i,j,k)}}{\sum_{i=1}^{S_n}\sum_{j=1}^{4}\sum_{k=1}^{N_j} n_{(i,j,k)}\delta_{(i,j,k)}} \tag{3-17}$$

where $n_{(i,j,k)} = \begin{cases} 1 & \text{if kth user of jth class in waiting state at ith slot.} \\ 0 & \text{otherwise.} \end{cases}$

and $\mathrm{WT}_{(i,j,k)} = \begin{cases} D_{(i,j,k)} \\ 0 \end{cases}$ is the waiting time of the kth user in jth class at ith slot.

3.3.5 Delay jitter

Delay jitter is the variance of the average delay. For the definition of the variance, we write:

$$\begin{aligned} V^2 &= \frac{1}{n-1}\sum_{i=1}^{n}(X_i - \overline{X})^2 \\ &= \frac{1}{n-1}\left[\sum_{i=1}^{n} X_i^2 - n\overline{X^2}\right] \end{aligned} \tag{3-18}$$

So the delay jitter in MFHMAC system can be written as:

$$V^2 = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{4}\sum_{k=1}^{N_j}\left[WT_{(i,j,k)} - \overline{D_{(j,k)}}\right]^2}{N_1 + N_2 + N_3 + N_4} \tag{3-19}$$

3.3.6 Probability of buffer overflow

In the FFHMAC scheme, the probability of buffer overflowing is defined as [8]:

$$\Pr_{[buffer\ overflow]} = \frac{\dfrac{A^L}{L!}}{\sum_{n=0}^{L} A^n \dfrac{1}{n!}} \tag{3-20}$$

where L is the buffer size

A is total offered load.

As mentioned above, we evaluate buffer overflow by a simulation procedure:

$$\Pr_{[buffer\ overflow]} = \frac{\sum_{i=1}^{S_n} O_{(i)}}{S_n} \tag{3-21}$$

$$\text{where } O_{(i)} = \begin{cases} 1 & \text{if buffer is overflowed at ith slot.} \\ 0 & \text{otherwise} \end{cases}$$

3.4 Simulation Procedure of FFMAC

As mentioned in 3.2, in this simulation we have four equal bandwidth access channels, and each one operates independently. The packets are generated from each randomly at any time slot. Following FFHMAC policy, all the packets are transmitted by the corresponding access channel or queued in the buffer. In the simulation, some parameters are listed in table 3-1:

	CDMA	CSMA	TDMA	FDMA
Bandwidth	1.875MHz	1.875MHz	1.875MHz	1.875MHz
QoS criteria	BER	Delay	Slot	Sub-channel
MDT	5s	5s	5s	5s
Packet size	2000	2000	2000	2000

Table 3-1: Parameters assumptions in the FFHMAC system simulations.

The Following assumptions are made in this simulation:

(1) Radio channels are ideal without fading, shadowing or multi-path (power control compensate for shadowing and fast fading is disregarded).

(2) Power control is perfect, and all packets arrive at the hub with same power.

(3) The standard packet arrive rate [7] is 5kbps and the packet size is assumed to be 2000bits/packet. So the packet unit time slot is 2000/5000=0.4s.

(4) The waiting time threshold assumed to be 5s=12 packet units.

Our QoS criteria setting and capacity rules are:

For class1 traffic, which corresponding the CDMA band, define the QoS satisfaction by concerning the average Bit Error Rate (BER) probability. The equation we can used is in [8]:

$$P_e = Q\left(\sqrt{\frac{1}{\frac{K-1}{3N}+\frac{N_0}{2T_b P_0}}}\right) \qquad (3\text{-}22)$$

$$\text{Re}\,write \qquad P_e = Q\left(\sqrt{\frac{2E_b}{\frac{2(K-1)}{3PG}+N_0}}\right) \tag{3-23}$$

Where $\quad \mathrm{PG} = \frac{\mathrm{T_b}}{\mathrm{T_c}} = \frac{W_{ss}}{R_s}$ is the processimg gain

K is the number of users in CDMA channel

$\frac{\mathrm{E_b}}{\mathrm{N_0}}$ is SNR.

For the digital packet, we define desired BER to be less than 10^{-3} [7].

For Class 2 traffic, which correspond with the CSMA channel, we define QoS by concerning the average packet transmission delay where for stream traffic the packet retransmission delay threshold is given in [7] 10^{-2}s.

For Class 3 traffic, which corresponds to the TDMA channel, we define the criteria of the call acceptance by the number of free time slots in one TDMA frame, where total slots is 1850/25=74.

For Class 4 traffic, which corresponds to the FDMA channel, the call acceptance criteria is defined by checking the number of free sub-channels in one frame. Where the total number of sub-channel in one FDMA frame are 1850/30=61.

From this procedure, we can evaluate the FFHMAC technique performance by computing the following parameters:

1) Call blocking probability:

   Use equation (3-11) and by setting:

T=MDT, C=C_{CDMA}, C_{CSMA}, C_{TDMA}, C_{FDMA}, and H=Td_1, Td_2, Td_3, Td_4 (which are capacities and average call duration in each band and each class) to obtain the call blocking probability of Class 1, Class 2, Class 3 and Class 4, respectively. Then the system average call blocking probability is

$$P_{[\text{veragecall blocking}]} = \frac{N_1 \Pr_1 + N_2 \Pr_2 + N_3 \Pr_3 + N_4 \Pr_4}{N_1 + N_2 + N_3 + N_4} \qquad (3\text{-}25)$$

where $\Pr_1, \Pr_2, \Pr_3, \Pr_4$ are the average call blocking probability in class1, class2, class3, and class4 repectively.

2) Average delay

Equations (3-13) to (3-16) are used in this simulation to obtain the average delay in Class 1, Class 3, Class 4 and Class 2. And the system average delay is obtained by:

$$\overline{D} = \frac{N_1 D_{CDMA} + N_2 D_{CSMA} + N_3 D_{TDMA} + N_4 D_{FDMA}}{N_1 + N_2 + N_3 + N_4} \qquad (3\text{-}26)$$

3) Delay jitter

Use equation (3-19) to obtain the delay jitter in each class and the system average delay jitter is

$$V^2 = \frac{N_1 V_{CDMA}^2 + N_2 V_{CSMA}^2 + N_3 V_{TDMA}^2 + N_4 V_{FDMA}^2}{N_1 + N_2 + N_3 + N_4} \qquad (3\text{-}27)$$

For the throughput and buffer flow in the FFHMAC system we didn't compute it in this simulation since it is simple and already evaluated in [8].

3.5 Simulation Procedures of MFHMAC Techniques

As mentioned in 3.3, we first divide the total radio band into two parts: basic band and reservation band. For instance, in our simulation scheme, assume we have a total band of 7.5 MHz (same as assumed in the FFHMAC simulation) and basic band/reservation band = 2/1. It means a total basic band = 5MHz and total reservation band = 2.5MHz. Then the basic bands are equally divided into four access band channels:

$$B_{CDMA} = B_{CSMA} = B_{TDMA} = B_{FDMA} = 1250KHz.$$

All other parameter settings are the same as the FFHMAC simulation. They are listed in Table 3-2.

	CDMA	CSMA	TDMA	FDMA	Reservation
Bandwidth	1.25 MHz	1.25 MHz	1.25 MHz	1.25 MHz	2.5 MHz
QoS criteria	BER	Delay	Slot	Sub-channel	
MDT	5s	5s	5s	5s	5s
Packet size	2000	2000	2000	2000	2000
Packet rate	Mixed	Mixed	Mixed	Mixed	Mixed

Table 3-2: Parameters setting in MFHMAC simulation.

Following the MFHMAC policy, the simulation procedure for Class 1-Class 4 traffic are shown in Fig. 3.8-Fig. 3.11.

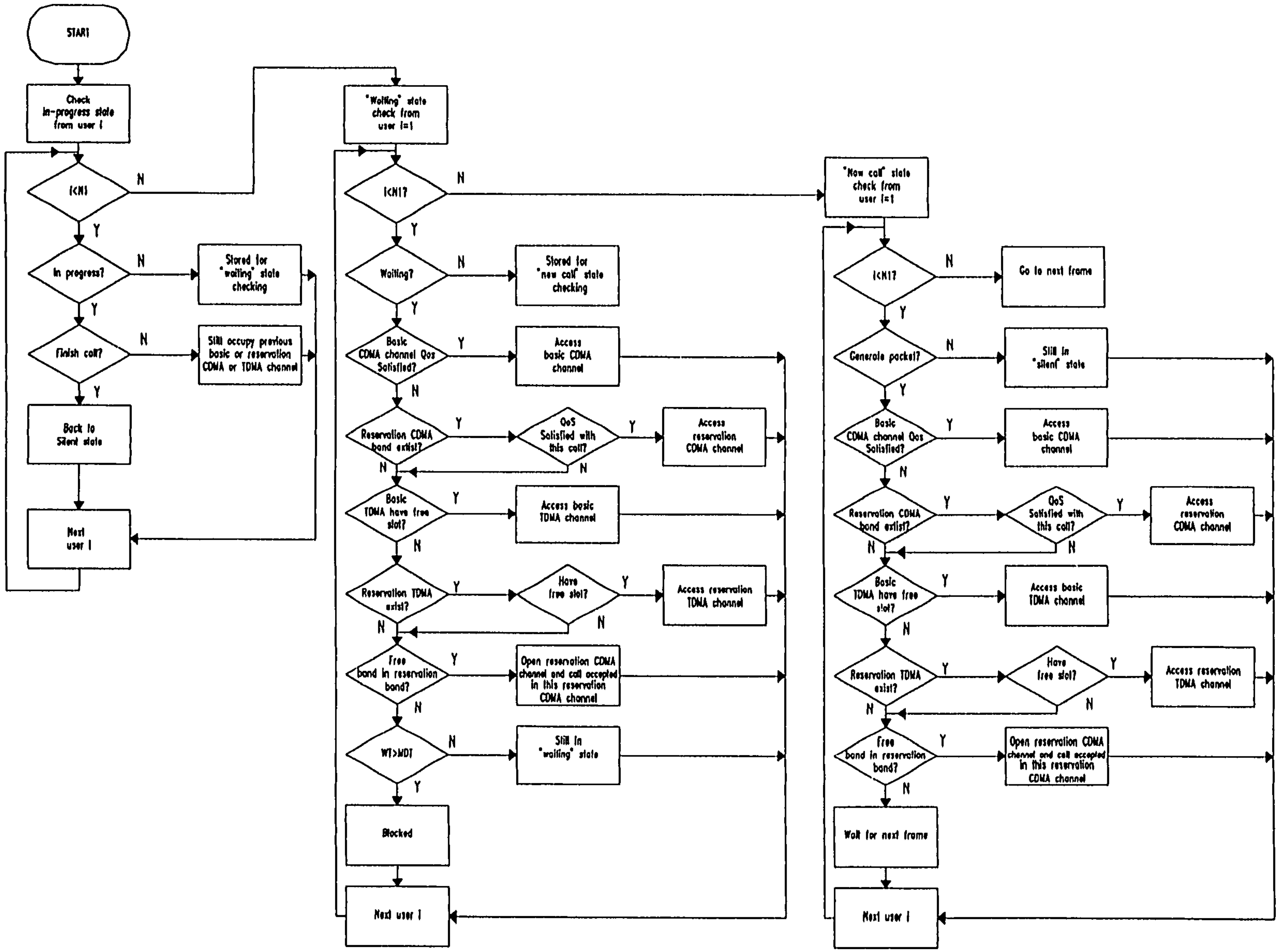


Fig.3. 7 Block diagram of MFH MAC simulation procedure with class 1 traffic

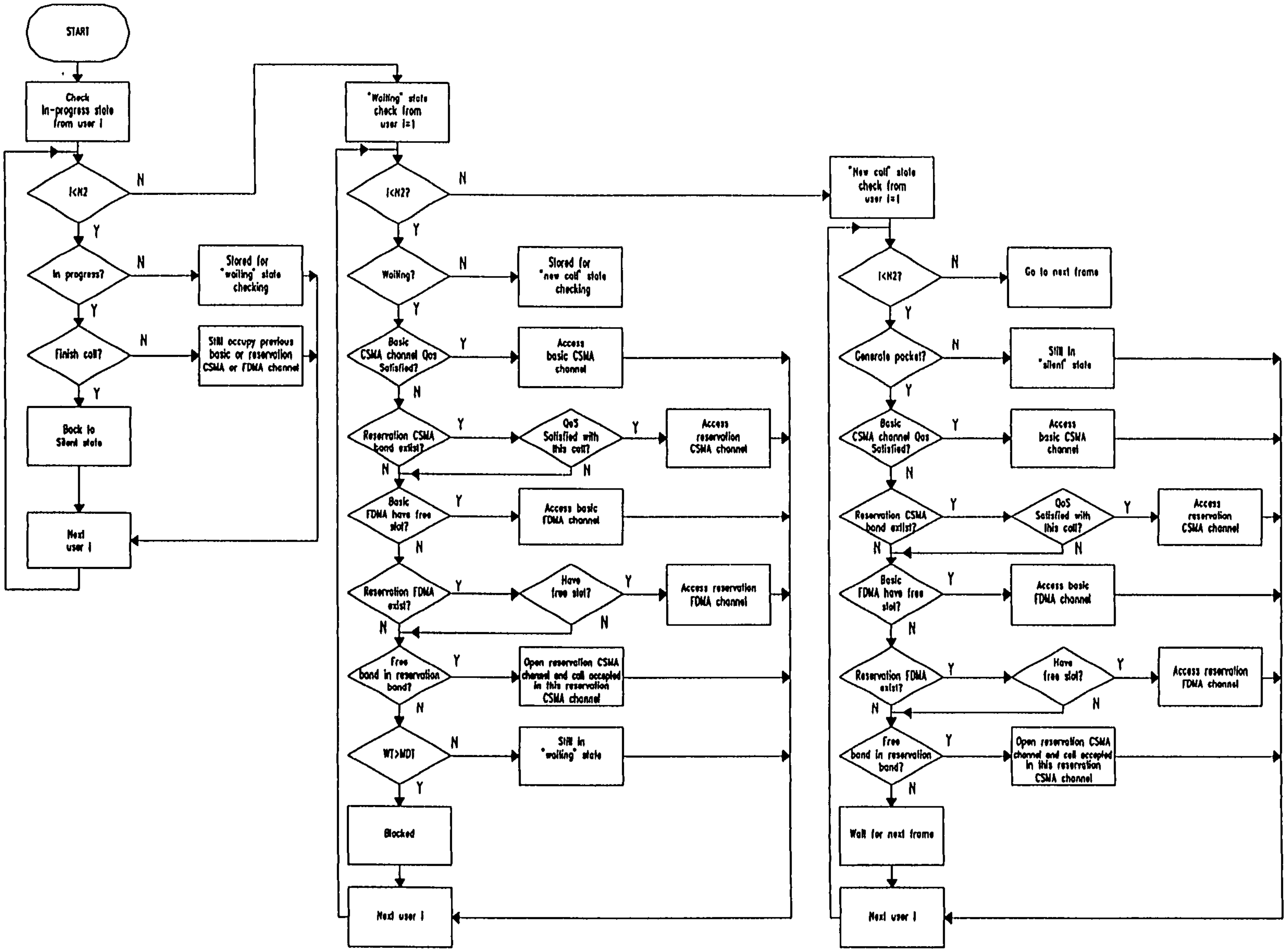


Fig.3. 8 Block diagram of MFH MAC simulation procedure with class 2 traffic

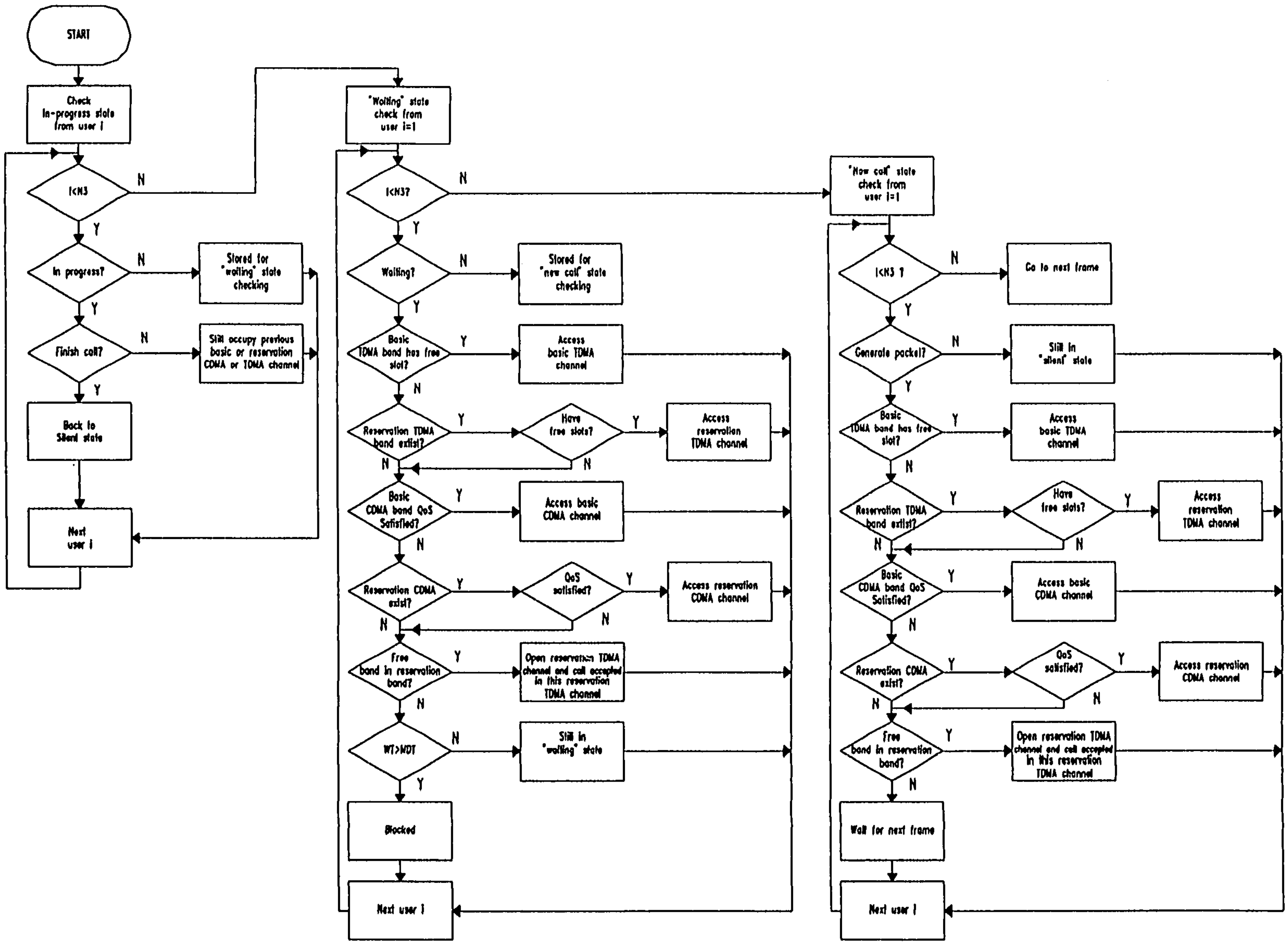


Fig.3. 9 Block diagram of MFH MAC simulation procedure with class 3 traffic

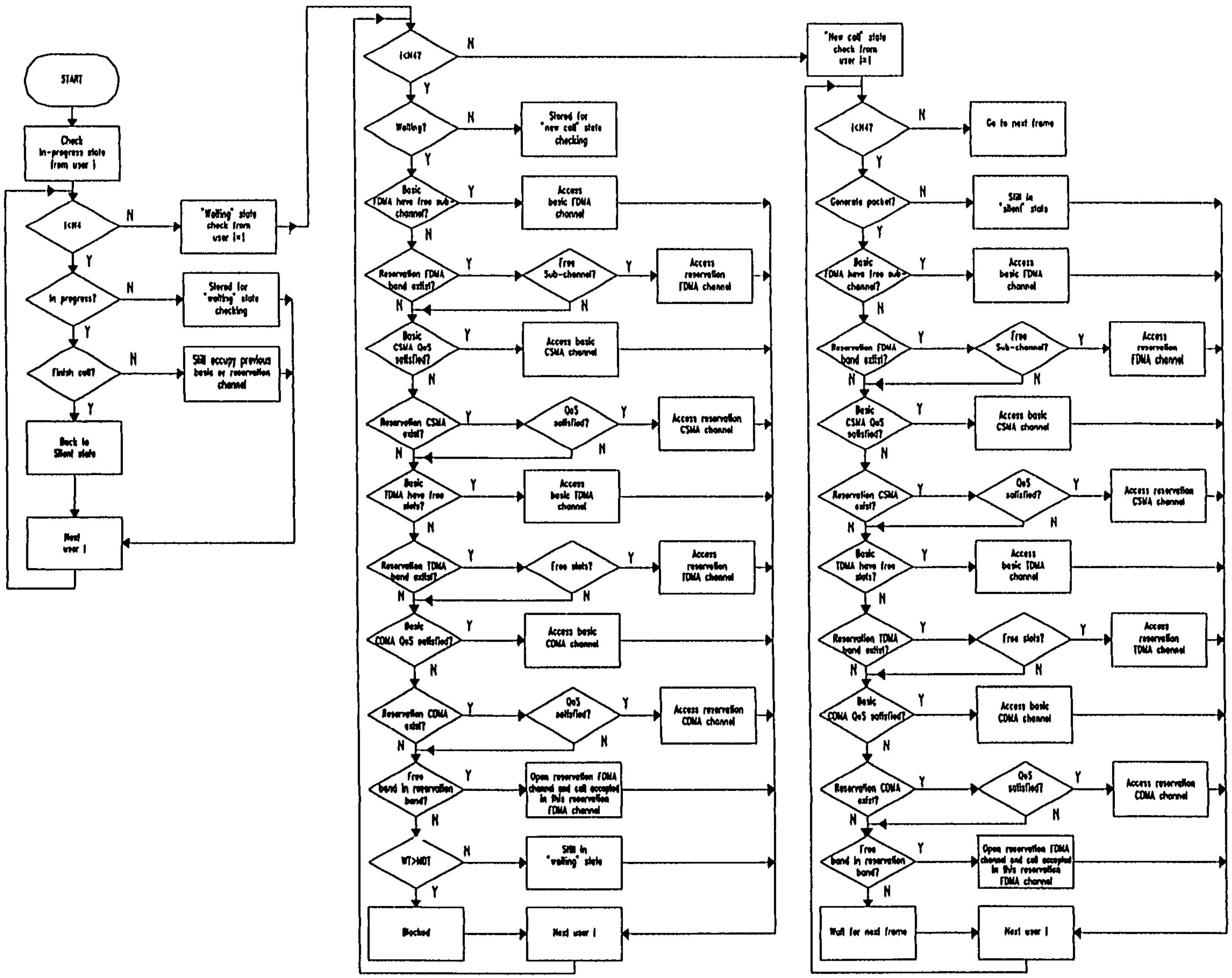


Fig. 3.10 Block diagram of MFH MAC simulation procedure with class 4 traffic

From Fig.3.8-Fig.3.11, we can see each type of traffic undergo three main steps following the priority rules. First part illustrates the "active" users' process. Second part illustrates the "waiting" users' process and third part illustrates new users' (previous state are "silent") process.

Over a long time interval Sn from the simulation we can evaluate the MFHMAC technique performance through the following calculation.

1. Call blocking probability

In Class 1:

$$\Pr_{[class1\,blocking]} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_1} Bn_{(i,j)}}{N_1 S_n} \qquad (3-28)$$

$$\text{where } Bn_{(i,j)} = \begin{cases} 1 & \text{when jth user is blocked at ith slot.} \\ 0 & \text{otherwise.} \end{cases}$$

In Class 2:

$$\Pr_{[class2\,blocking]} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_2} Bn_{(i,j)}}{N_2 S_n} \qquad (3-29)$$

In Class 3:

$$\Pr_{[class3\,blocking]} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_3} Bn_{(i,j)}}{N_3 S_n} \qquad (3-30)$$

In Class 4:

$$\Pr_{[class4\ blocking]} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_4} Bn_{(i,j)}}{N_4 S_n} \qquad (3\text{-}31)$$

And the average system blocking probability can be obtained by using equation (3-12).

2. Delay

In Class 1:

$$\overline{D_1} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_1} Wn_{(i,j)} WT_{(i,j)}}{N_1 S_n} \qquad (3\text{-}32)$$

where $Wn_{(i,j)} = \begin{cases} 1, & \textit{if} \text{ jth user is in "waiting" state at ith slot.} \\ 0, & \text{otherwise.} \end{cases}$

WT(i,j) is the waiting time of the jth user at the ith slot and 0<WT(i,j)<12.

In Class 2:

$$\overline{D_2} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_2} Wn_{(i,j)} WT_{(i,j)}}{N_2 S_n} \qquad (3\text{-}33)$$

In Class 3:

$$\overline{D_3} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_3} Wn_{(i,j)} WT_{(i,j)}}{N_3 S_n} \qquad (3\text{-}34)$$

In Class 4:

$$\overline{D_4} = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_4} Wn_{(i,j)} WT_{(i,j)}}{N_4 S_n} \qquad (3\text{-}35)$$

And the system average delay is obtained:

$$\overline{D}_{[average]} = \frac{N_1 D_1 + N_2 D_2 + N_3 D_3 + N_4 D_4}{N_1 + N_2 + N_3 + N_4} \tag{3-36}$$

3. Delay jitter

In Class 1:

$$V_1^2 = \frac{\sum_{i=1}^{S_n} \sum_{j=1}^{N_1} \left[WT_{(i,j)} - \overline{D_1} \right]^2}{N_1 S_n} \tag{3-37}$$

In Class 2:

$$V_2^2 = \frac{\sum_{i=1}^{S_n} \sum_{j=1}^{N_2} \left[WT_{(i,j)} - \overline{D_2} \right]^2}{N_2 S_n} \tag{3-38}$$

In Class 3:

$$V_3^2 = \frac{\sum_{i=1}^{S_n} \sum_{j=1}^{N_3} \left[WT_{(i,j)} - \overline{D_3} \right]^2}{N_3 S_n} \tag{3-39}$$

And in Class 4:

$$V_4^2 = \frac{\sum_{i=1}^{S_n} \sum_{j=1}^{N_4} \left[WT_{(i,j)} - \overline{D_4} \right]^2}{N_4 S_n} \tag{3-40}$$

The system average delay jitter is:

$$V^2_{[average]} = \frac{N_1V_1^2 + N_2V_2^2 + N_3V_3^2 + N_4V_4^2}{N_1 + N_2 + N_3 + N_4} \tag{3-41}$$

4. Throughput

In Class 1:

$$S_1 = \frac{\sum_{i=1}^{S_n}\sum_{j=1}^{N_1} In_{(i,j)}\delta_1}{[\alpha_{11}C_{CDMA} + \alpha_{12}C_{RCDMA} + \alpha_{13}C_{TDMA} + \alpha_{14}C_{RTDMA}]S_n} \tag{3-42}$$

$$where\ \alpha_{11} = \frac{\text{number of class1 users in basic CDMA}}{\text{number of total users in basic CDMA}}$$

$$\alpha_{12} = \frac{\text{number of class1 users in reservation CDMA}}{\text{number of total users in reservation CDMA}}$$

$$\alpha_{13} = \frac{\text{number of class1 users in basic TDMA}}{\text{number of total users in basic TDMA}}$$

$$\alpha_{14} = \frac{\text{number of class1 users in reservation TDMA}}{\text{number of total users in reservation TDMA}}$$

In Class 2:

$$S_2 = \alpha_{21}S_{CSMA} + \alpha_{22}S_{RCSMA} + \alpha_{23}S_{FDMA} + \alpha_{24}S_{RFDMA} \tag{3-43}$$

$$where\ \alpha_{21} = \frac{\text{number of class2 users in basic CSMA}}{\text{number of total users in basic CSMA}}$$

$$\alpha_{22} = \frac{\text{number of class2 users in reservation CSMA}}{\text{number of total users in reservation CSMA}}$$

$$\alpha_{23} = \frac{\text{number of class2 users in basic FDMA}}{\text{number of total users in basic FDMA}}$$

$$\alpha_{24} = \frac{\text{number of class2 users in reservation FDMA}}{\text{number of total users in reservation FDMA}}$$

In Class 3:

$$S_3 = \alpha_{31} S_{TDMA} + \alpha_{32} S_{RTDMA} + \alpha_{33} S_{CDMA} + \alpha_{34} S_{RCDMA} \qquad (3-44)$$

$$where\ \alpha_{31} = \frac{\text{number of class3 users in basic TDMA}}{\text{number of total users in basic TDMA}}$$

$$\alpha_{32} = \frac{\text{number of class3 users in reservation TDMA}}{\text{number of total users in reservation TDMA}}$$

$$\alpha_{33} = \frac{\text{number of class3 users in basic CDMA}}{\text{number of total users in basic CDMA}}$$

$$\alpha_{34} = \frac{\text{number of class3 users in reservation CDMA}}{\text{number of total users in reservation CDMA}}$$

In Class 4:

$$S_4 = \alpha_{41} S_{FDMA} + \alpha_{42} S_{RFDMA} + \alpha_{43} S_{CSMA} + \alpha_{44} S_{RCSMA}$$

$$+ \alpha_{45} S_{TDMA} + \alpha_{46} S_{RTDMA} + \alpha_{47} S_{CDMA} + \alpha_{48} S_{RCDMA} \qquad (3-45)$$

$$where\ \alpha_{41} = \frac{\text{number of class4 users in basic FDMA}}{\text{number of total users in basic FDMA}}$$

$$\alpha_{42} = \frac{\text{number of class4 users in reservation FDMA}}{\text{number of total users in reservation FDMA}}$$

$$\alpha_{43} = \frac{\text{number of class4 users in basic CSMA}}{\text{number of total users in basic CSMA}}$$

$$\alpha_{44} = \frac{\text{number of class4 users in reservation CSMA}}{\text{number of total users in reservation CSMA}}$$

$$\alpha_{45} = \frac{\text{number of class4 users in basic TDMA}}{\text{number of total users in basic TDMA}}$$

$$\alpha_{46} = \frac{\text{number of class4 users in reservation TDMA}}{\text{number of total users in reservation TDMA}}$$

$$\alpha_{47} = \frac{\text{number of class4 users in basic CDMA}}{\text{number of total users in basic CDMA}}$$

$$\alpha_{48} = \frac{\text{number of class4 users in reservation CDMA}}{\text{number of total users in reservation CDMA}}$$

The system average throughput is:

$$\overline{S} = \frac{N_1 S_1 + N_2 S_2 + N_3 S_3 + N_4 S_4}{N_1 + N_2 + N_3 + N_4} \qquad (3\text{-}46)$$

5. Probability of buffer overflowing.

Equation (3-21) can be used to obtain the probability of buffer overflowing through the simulation and the system average buffer overflowing probability can be written as:

$$\Pr_{[\text{buffer overflow}]} = \frac{N_1 \Pr_{[\text{class1 overflow}]} + N_2 \Pr_{[\text{class2 overflow}]} + N_3 \Pr_{[\text{class3 overflow}]} + N_4 \Pr_{[\text{class4 overflow}]}}{N_1 + N_2 + N_3 + N_4} \quad (3\text{-}47)$$

From the simulation result the above five parameters will be shown in 3.7

3.6 Utilization of FEC Coding in MFHMAC

As we have seen in 3.3, the CDMA band QoS components are defined by BER in equation (3-23). It is well known that channel forward error correcting (FEC) code can improve digital communication link performance by adding redundancy in the transmitted message. FEC code enables a limited number of errors to be detected and corrected without retransmission. FEC codes can be used effectively to improve the performance of a communications system when other means of improvements (such as increasing transmitter power or using a more sophisticated demodulation) are impractical [8].

However, using a linear block code for FEC coding with rate (k/n) will cause a reduction in the processing gain of the system. In other words, it expands the occupied bandwidth for a particular message data rate because the information data rate is reduced by adding parity check bits. This is compensated by the algebraic relation between transmitted channel symbols that enable error correction.

In general terms, by a reduction in the number of retransmission required after CDMA despreading some the data is protected against channel errors by FEC coding.

When FEC codes are deployed, the processing gain will decrease by a factor of k/n.

$$PG' = PG\,(k/n) = (T_b/T_c)(k/n)$$

From equation 3-1, channel symbol error rate will get worse, but after channel decoding BER gets better. There is a trade-off in choosing properly FEC coding by referring to n/k. Since with FEC code.

$$P_{be} = P_{se} = \frac{1}{n}\sum_{i=t+1}^{n} i\binom{n}{i}P_c^i(1-P_c)^{n-i} \qquad (3\text{-}48)$$

$$P_c = Q\left(\sqrt{\frac{2E_b}{\frac{2(K-1)}{3PG'}+N_0}}\right) \qquad (3\text{-}49)$$

Where: PG′ = PG(k/n), n is number of channel symbols per codeword.

Pc is channel symbol error rate.

t is the number of correctable errors per BCH codeword, t=(n-k)/2.

M is the number of bits per symbol, here assume M=2.

P_{se} is symbol error probability.

In this thesis we use BCH code (63,47). In MFHMC techniques on the four access channels and use the same procedure and equation to get all the performance parameters. The results are shown in the next section.

3.7 Simulation Results

Fig.3.12-Fig.3.36 show the MFHMAC performances. Fig.3.12 shows the call blocking probability vs. system total users. When the number of total users increase from 200 (each class has 50 users) to 400 (each class has 100 users), the average call blocking probability of the system will increase from 0.026 to 0.046. The class 4 users has hightest average call blocking probability and the class 2 users have lowest call blocking probability.

Fig.3.13 shows the call blocking probability vs. each class of users. When the number of users in class 1 increase from 0-100 (the number of all other three classes user fixed with 100), the call blocking probability of class 1 will increase from 0 to 0.018, and the probability of call blocking in class 2 (from 0.025 to 0.029), class 3 (from 0.034 to 0.042) and class 4 (from 0.068 to 0.069) increasing as well.

Fig.3.14 shows the probability of call blocking vs. arrived traffic rate. When the low rate traffic (class 1 and class 3) rate increase from 1 to 10 (normalized), the average call blocking probability of the system will increase from 0.032 to 0.052. When the high rate traffic (class 2 and class 4) rate increase from 4 to 12, the average probability of call blocking in the system will increase from 0.038 to 0.048.

Fig.3.15 shows call blocking probability vs. traffic burstiness. When the burstiness of bursty traffic (class 1 and class 2) increase from 0.1 to 0.5, the average call blocking probability increasing from 0.036 to 0.057. When the burstiness of stream traffic (class 3 and class 4) increase from 0.6 to 1.0, the average call blocking probability increasing from 0.034 to 0.04.

Fig.3.16 shows the probability of call blocking vs. call holding time (duration). When the average duration of calling time increase from 1 minute to 10 minutes, the average call blocking probability increasing from 0.037 to 0.056.

Fig.3.17 shows the average delay vs. system total users. When the number of total users increase from 200 (each class has 50 users) to 400 (each class has 100 users), the average delay of the system will increase from 3.4 (packet units) to 4.6 (packet units). The class 4 users have high-test packet delay and the class 2 users have lowest packet delay.

Fig.3.18 shows the average packet delay vs. each class of users. When the number of users in class 1 increase from 0-100 (the number of all other three classes user fixed with 100), the average delay of class 1 will increase from 0 to 2.1, and the packet delay in class 2 (from 2.3 to 2.4), class 3 (from 5.7 to 6.4) and class 4 (from 6.3 to 6.4) increasing as well.

Fig.3.19 shows the average packet delay vs. arrived traffic rate. When the low rate traffic (class 1 and class 3) rate increase from 1 to 10 (normalized), the average packet delay of the system will increase from 3.1 to 4.9. When the high rate traffic (class 2 and class 4) rate increase from 4 to 12, the average packet delay in the system will increase from 2.1 to 4.4..

Fig.3.20 shows packet delay vs. traffic burstiness. When the burstiness of bursty traffic (class 1 and class 2) increase from 0.1 to 0.5, the average packet delay increasing from 3.2 to 5.2. When the burstiness of stream traffic (class 3 and class 4) increase from 0.6 to 1.0, the average packet delay increasing from 4.1 to 6.0.

Fig.3.21 shows the average delay vs. call holding time (duration). When the average duration of calling time increase from 1 minute to 10 minutes, the average delay increasing from 3.4 to 5.4.

Fig.3.22 shows the average throughput vs. system total users. When the number of total users increase from 200 (each class has 50 users) to 400 (each class has 100 users), the average throughput of the system will increase from 0.33 to 0.91 and then decrease. The class 1 and class3 have high-test average throughput and the class 2 and class 4 have lower throughput.

Fig.3.23 shows the throughput vs. each class of users. When the number of users in class 1 increase from 0-100 (the number of all other three classes user fixed with 100), the throughput of class 1 will increase from 0 to 1.0, and the throughput in class 2 and class 4 almost no changing and class 3 increasing as well.

Fig.3.24 shows the throughput vs. arrived traffic rate. When the low rate traffic (class 1 and class 3) rate increase from 1 to 10 (normalized), the average throughput of the system will increase from 0.61 to 0.82. When the high rate traffic (class 2 and class 4) rate increase from 4 to 12, the average throughput in the system will increase from 0.61 to 0.90 then decrease.

Fig.3.25 shows throughput vs. traffic burstiness. When the burstiness of bursty traffic (class 1 and class 2) increase from 0.1 to 0.5, the average throughput increasing from 0.38 to 0.89 then decreasing . When the burstiness of stream traffic (class 3 and class 4) increase from 0.6 to 1.0, the average throughput increasing from 0.64 to 0.84 then decreasing.

Fig.3.26 shows the throughput vs. call holding time (duration). When the average duration of calling time increase from 1 minute to 10 minutes, the average throughput increasing from 0.73 to 0.86 then decreasing.

Fig.3.27 shows the delay jitter vs. system total users. When the number of total users increase from 200 (each class has 50 users) to 400 (each class has 100 users), the delay jitter of the system will increase from 4.2 to 10.8. The class 4 users has high-test delay jitter and the class 2 users have lowest delay jitter.

Fig.3.28 shows the delay jitter vs. each class of users. When the number of users in class 1 increase from 0-100 (the number of all other three classes user fixed with 100), the delay jitter of class 1 will increase from 0 to 0.18, and the delay jitter in class 2 (from 11.2 to 11.4), class 3 (from 7.1 to 14.8) and class 4 (from 14.7 to 14.8) increasing as well.

Fig.3.29 shows the jitter vs. arrived traffic rate. When the low rate traffic (class 1 and class 3) rate increase from 1 to 10 (normalized), the delay jitter of the system will increase from 6.8 to 15.1. When the high rate traffic (class 2 and class 4) rate increase from 4 to 12, the delay jitter in the system will increase from 2.3 to 11.8.

Fig.3.30 shows delay jitter vs. traffic burstiness. When the burstiness of bursty traffic (class 1 and class 2) increase from 0.1 to 0.5, the delay jitter increasing from 7.4 to 11.9. When the burstiness of stream traffic (class 3 and class 4) increase from 0.6 to 1.0, the delay jitter decreasing from 11.8 to 6.5.

Fig.3.31 shows the delay jitter vs. call holding time (duration). When the average duration of calling time increase from 1 minute to 10 minutes, the delay jitter increasing from 5.4 to 15.8.

Fig.3.32 shows the probability of buffer overflow vs. system total users. When the number of total users increase from 200 (each class has 50 users) to 400 (each class has 100 users), the probability of buffer overflow of the system will increase from 0.026 to 0.046. The class 4 users have high-test probability of buffer overflow and the class 2 users have lowest probability of buffer overflow.

Fig.3.33 shows the probability of buffer overflow vs. each class of users. When the number of users in class 1 increase from 0-100 (the number of all other three classes user fixed with 100), the probability of buffer overflow of class 1 will increase from 0 to 0.018, and the probability of call blocking in class 2 (from 0.025 to 0.029), class 3 (from 0.034 to 0.042) and class 4 (from 0.068 to 0.069) increasing as well.

Fig.3.34 shows the probability of buffer overflow vs. arrived traffic rate. When the low rate traffic (class 1 and class 3) rate increase from 1 to 10 (normalized), the probability of buffer overflow of the system will increase from 0.032 to 0.052. When the high rate traffic (class 2 and class 4) rate increase from 4 to 12, the average probability of buffer overflow in the system will increase from 0.038 to 0.048.

Fig.3.35 shows probability of buffer overflow vs. traffic burstiness. When the burstiness of bursty traffic (class 1 and class 2) increase from 0.1 to 0.5, the probability of buffer overflow increasing from 0.036 to 0.057. When the burstiness of stream traffic (class 3 and class 4) increase from 0.6 to 1.0, the probability of buffer overflow increasing from 0.034 to 0.04.

Fig.3.36 shows the probability of buffer overflow vs. call holding time (duration). When the average duration of calling time increase from 1 minute to 10 minutes, the probability of buffer overflow increasing from 0.037 to 0.056.

In Fig. 3.12- Fig. 3.16 the MFHMAC system blocking probabilities are shown. Fig. 3.12 and Fig. 3.13 show the effects of the users in each traffic class.

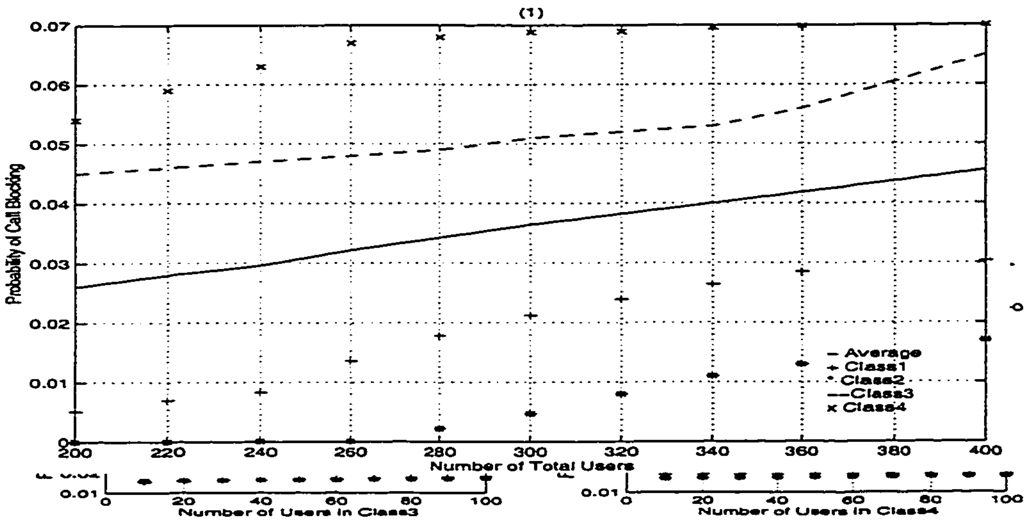


Fig. 3.12: Call blocking vs. system users where total users are equally distributed in four classes.

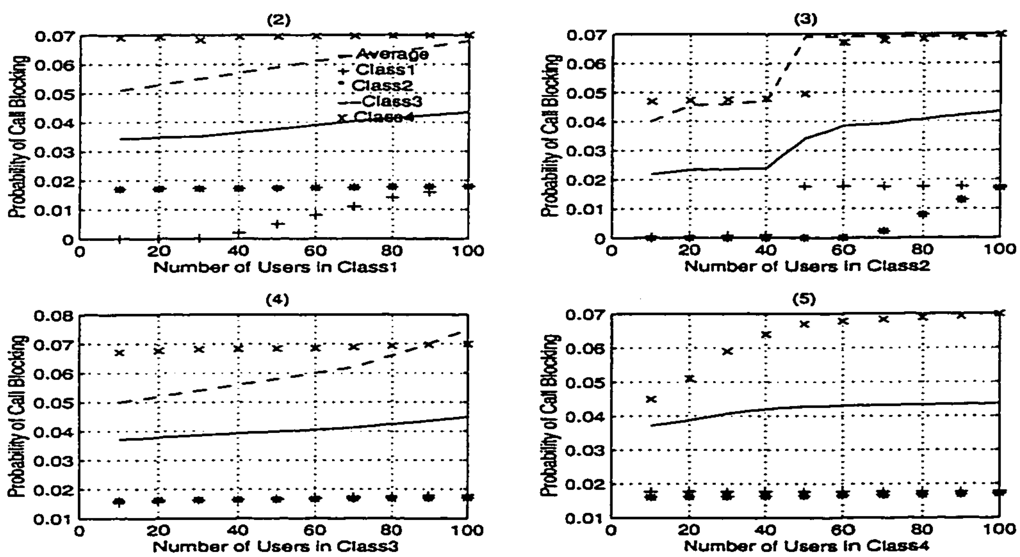


Fig. 3.13: Call blocking vs. each class users: (2) Class 1, (3) Class 2, (4) Class 3, (5) Class 4.

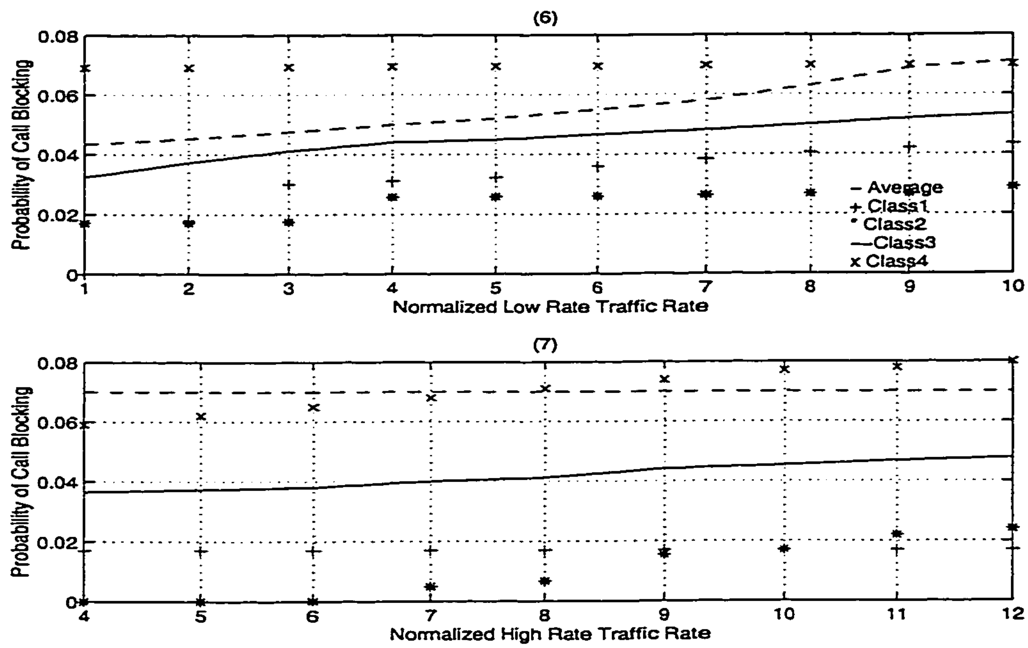


Fig. 3.14: Call blocking vs. traffic rate: (6) for low rate traffic, (7) for high rate traffic.

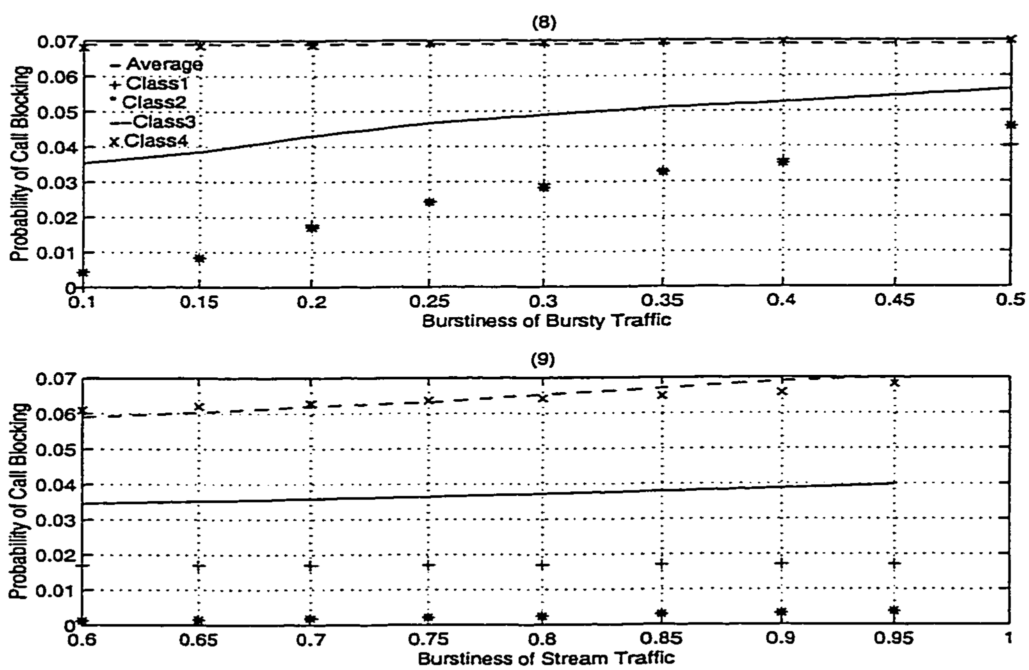


Fig. 3.15: Call blocking vs. traffic burstiness: (8) for bursty traffic, (9) for stream traffic.

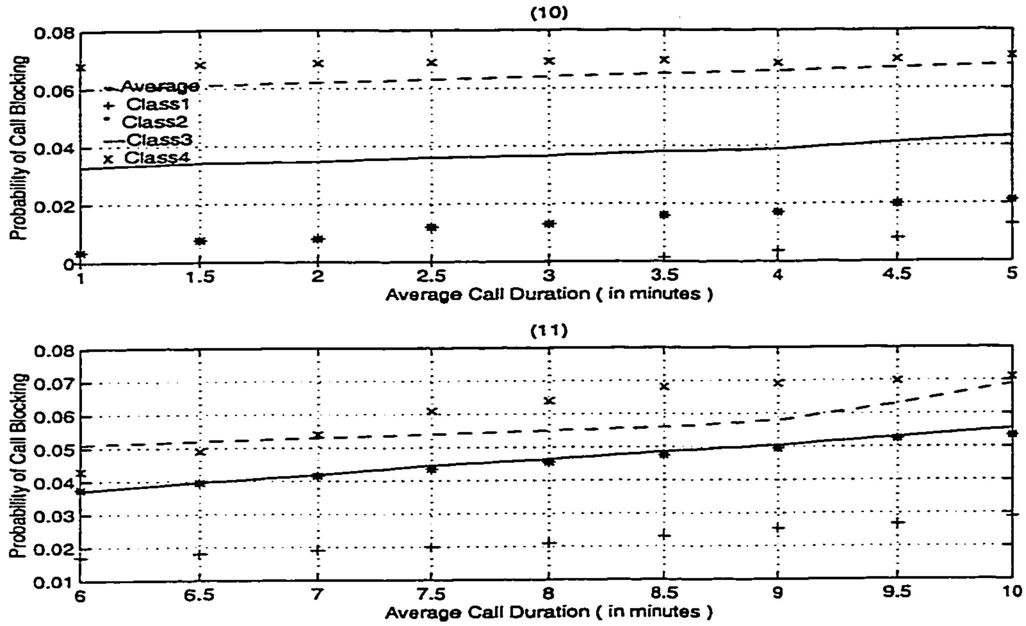


Fig. 3.16: Call blocking vs. call duration: (10) for short call (11) for long call.

From Fig.3.12, we noted that class 4 users have high-test call blocking probability due to the have traffic load (stream and high rate). Class 2 users have lowest call blocking probability due to the light traffic (bursty) and the first priority of service. Fig.3.13 showed that even only one class users increase, all other classes' call blocking are also increased. These are caused by the flexible adjustment bandwidth allocations. One important factor is that when traffic load increase, probability of call blocking didn't change too much. This means our techniques are robust for any type of traffic. However, from fig.3.14 we noted that when the low rate users' (class 1 and class 3) arriving packet rate increase, the probability of call blocking in high rate class (4) almost constant due to the high loading existence in basic and reservation FDMA band. The effects of burstiness θ and average call duration Td are showed very similarly in Fig.3.15 and Fig.3.16.

Fig. 3.17-Fig. 3.21 show the MFHMAC system average delay versus various traffic characters, load (number of users), burstiness, rate and call duration, and so on.

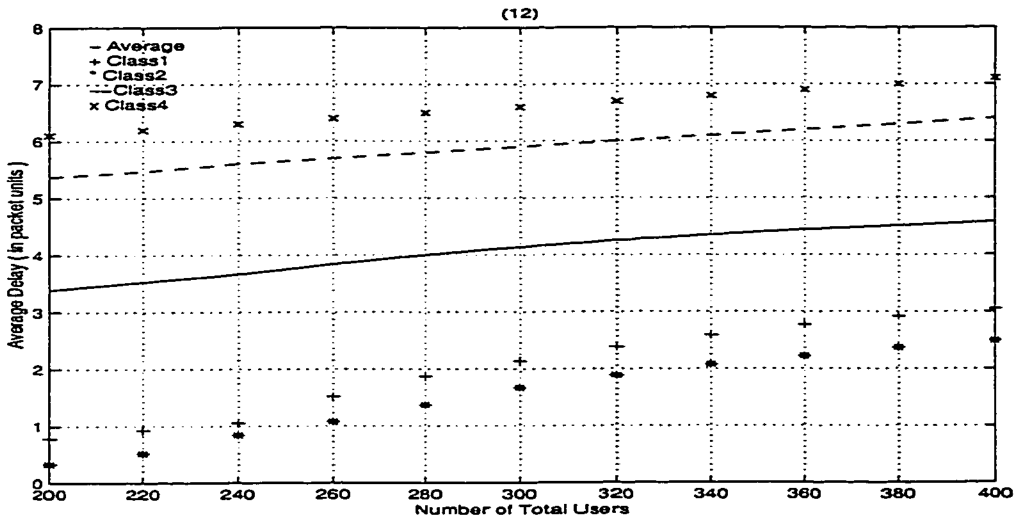


Fig.3.17: Delay Vs total users. Users are equally distributed in each class.

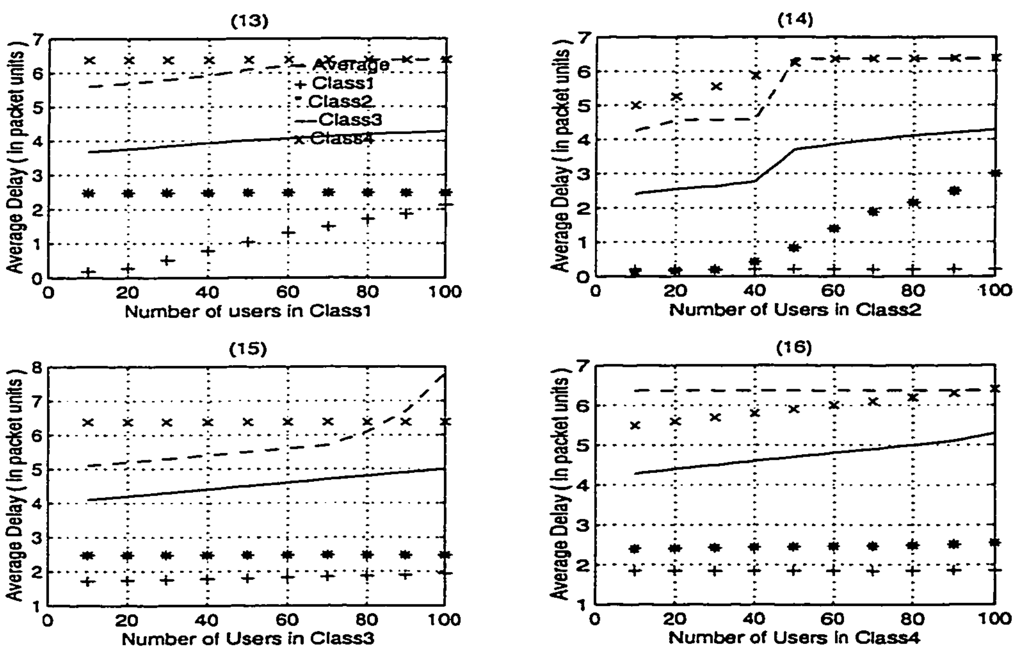


Fig.3.18. Delay vs. each class users where (13) for class 1, (14) for class 2, (15) for class 3 and (16) for class4.

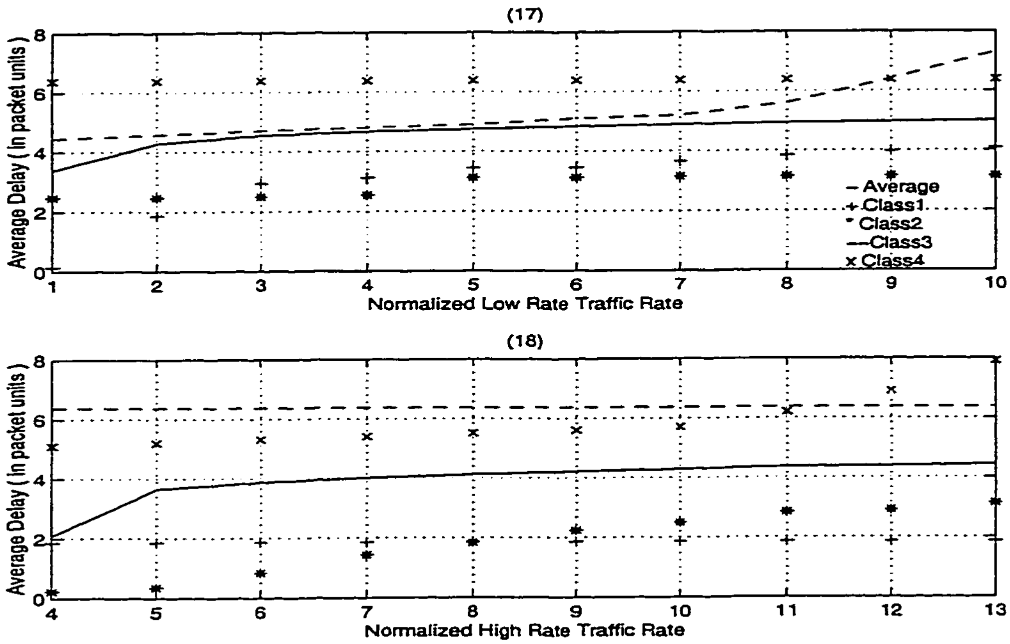


Fig.3.19: Delay Vs traffic rate. Where (17) for low rate traffic, (18) for high rate traffic.

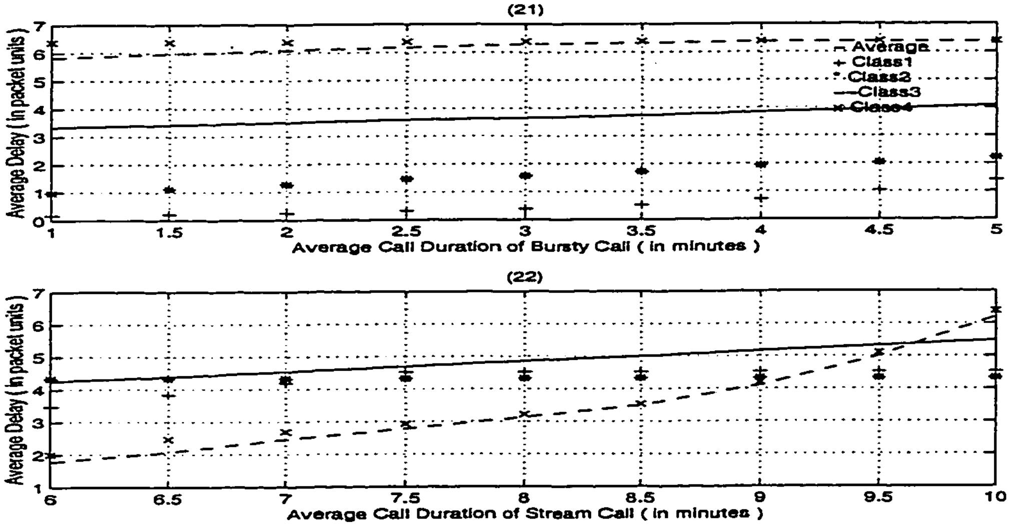


Fig.3.20. Delay vs. traffic burstiness where (19) for bursty traffic (θ_1 and θ_2) and (20) for stream traffic (θ_3 and θ_4).

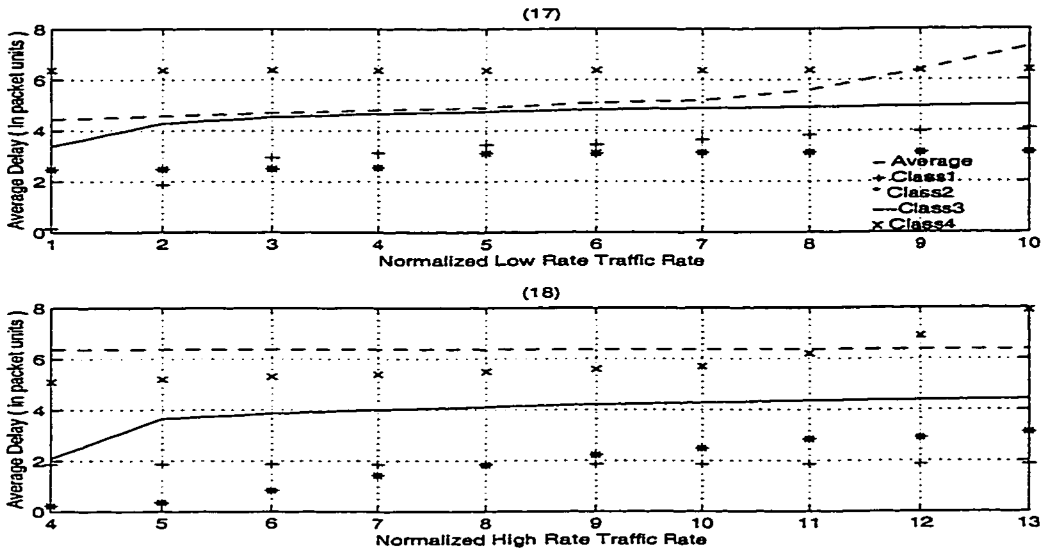


Fig. 3.21: Delay vs. call duration where: (21) for short calls (Td_1 and Td_2) and (22) for long calls (Td_3 and Td_4).

From Fig.3.17-Fig.3.21, we noted that delay in each class have same characteristic as call blocking scenarios. One class users increase, while all other classes' users fixed, average packet delay in each class is also increased as well (Fig.3.18). However, low rate users' (class 1 and class 3) packet arriving rate (δ_1 and δ_3) increase, high rate users' (class 2 and class 4) average delay are just changed a little (Fig.3.19), vice versa. The effects of traffic burstiness and call duration have the same characteristics as traffic rates (Fig.3.20 and Fig.3.21).

Fig. 3.22-Fig. 3.26 show the MFHMAC system average throughput vs. various traffic characters, load (number of users), traffic rate, traffic burstiness and call duration, and so on.

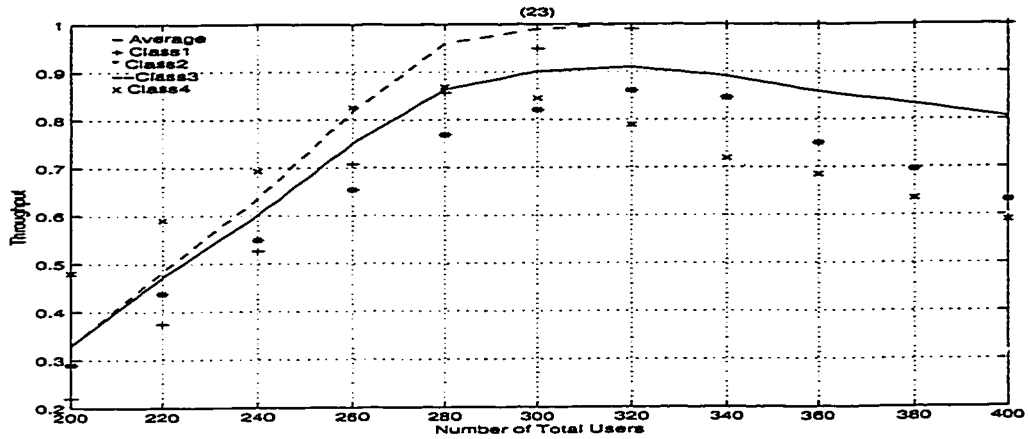


Fig. 3.22: Throughput vs. users where total users are equally distributed in four classes.

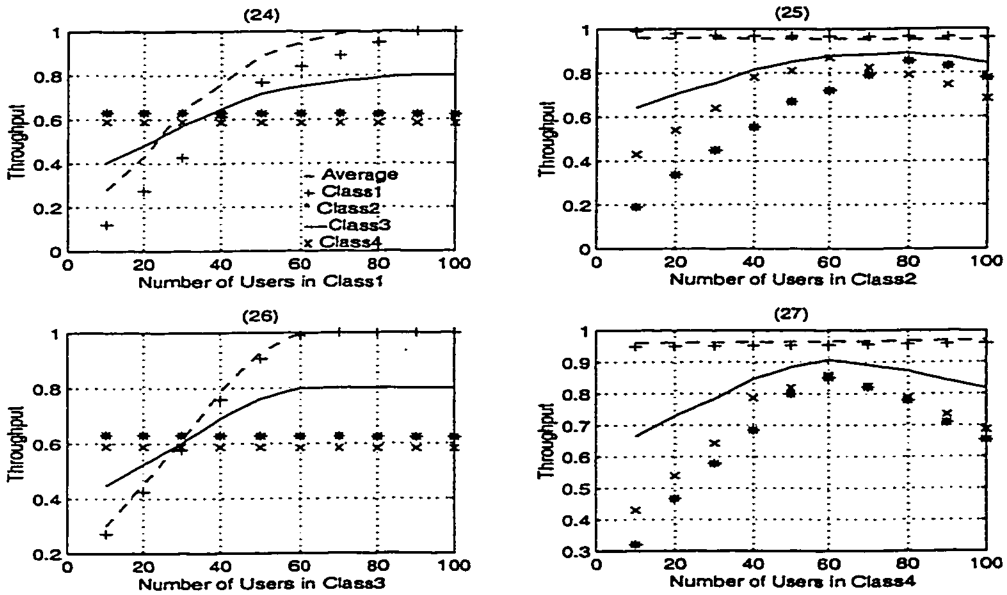


Fig. 3.23: Throughput vs. each class users: (24) for Class 1, (25) for Class 2, (26) for Class 3 and (27) for Class 4.

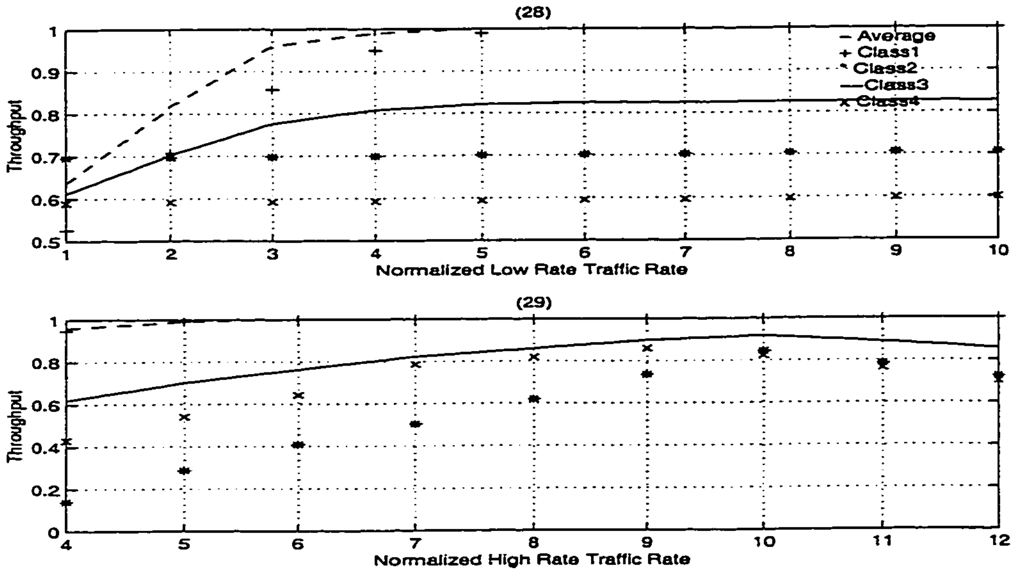


Fig. 3.24: Throughput vs. traffic rate where: (28) for low rate traffic (29) for high rate traffic.

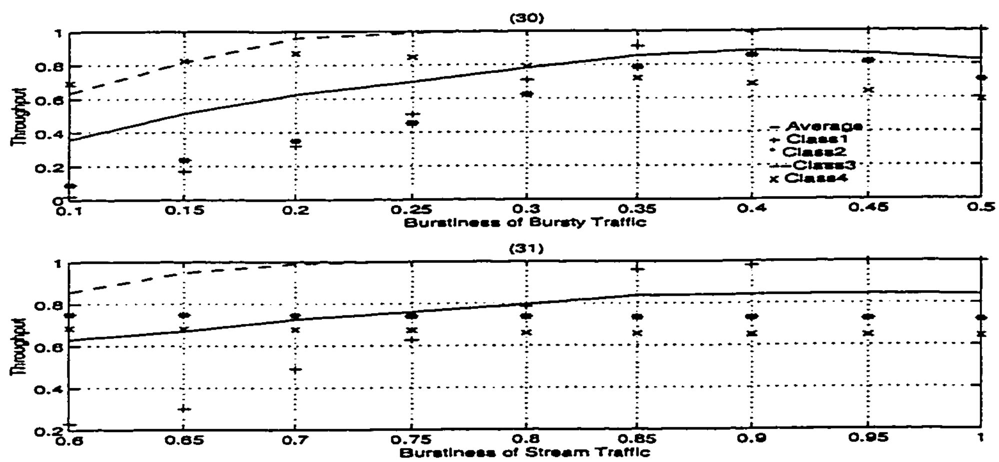


Fig. 3.25: Throughput vs. traffic burstiness where: (30) for bursty traffic and (31) for stream traffic.

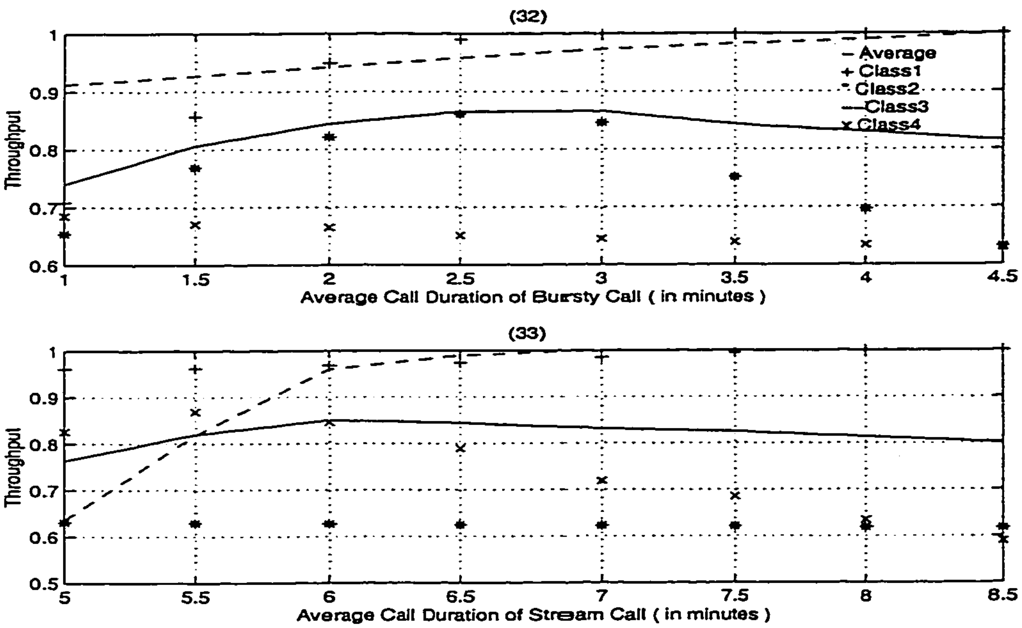


Fig. 3.26: Throughput vs. call duration where: (32) for short calls and (33) for long calls.

Interesting observation is that with the traffic load increasing the system throughput increase as well, but after reach up a certain value, the throughput will decrease (Fig.3.22), this was caused by the collision retransmissions from CSMA band. We noted that class 1 and class 3 users have higher throughput due to the high utilization of CDMA and TDMA techniques. Eventually, class 1 and class 3 users' throughput can reach 1. Class 2 and class 4 users may access radio channels by CSMA band, so, after reach maximum throughput, with the high collisions the throughput are decreasing. This also cause the average throughput in the whole system perform the same property. Other conditions which represent the traffic characteristics have the same effects as traffic load.

Fig. 3.27-Fig. 3.31 show the delay jitter for MFHMAC vs. various traffic characters, load (number of users), traffic rate, traffic bursty level and call duration, and so on.

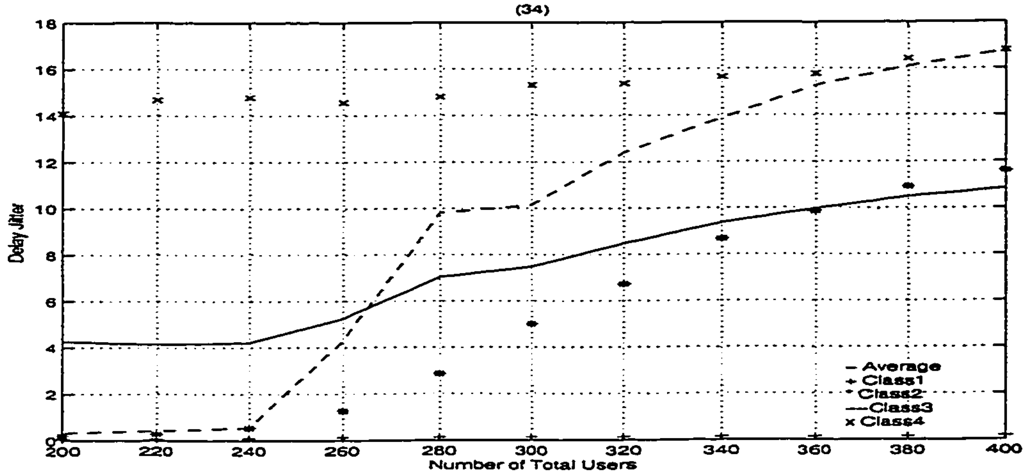


Fig. 3.27: Delay jitter vs. system users where the total users are equally distributed in each class.

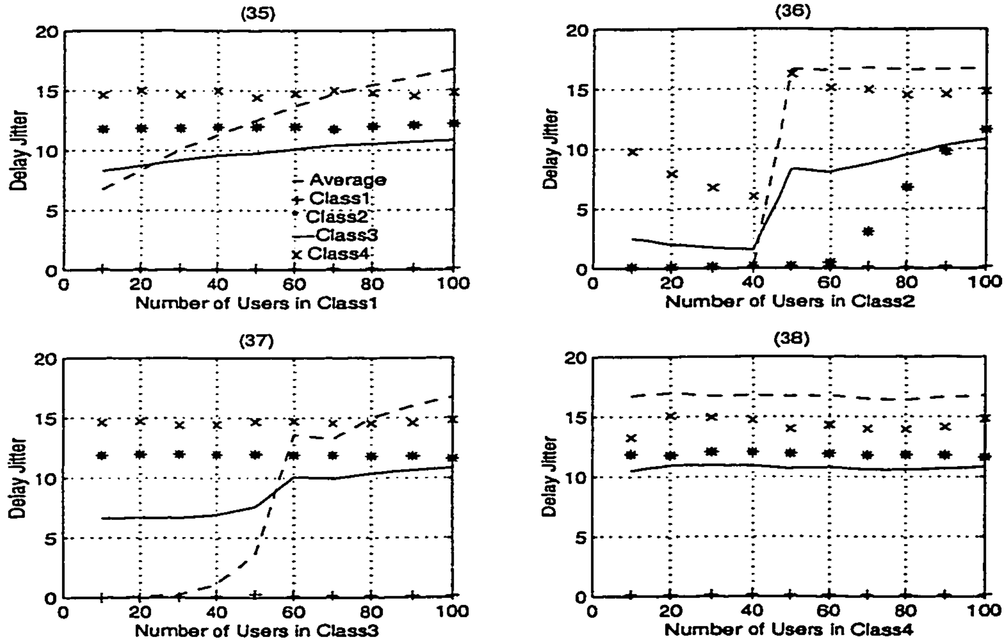


Fig. 3.28: Delay jitter vs. each class of users where: (35) for Class 1, (36) for Class 2, (37) for Class 3 and (38) for Class 4.

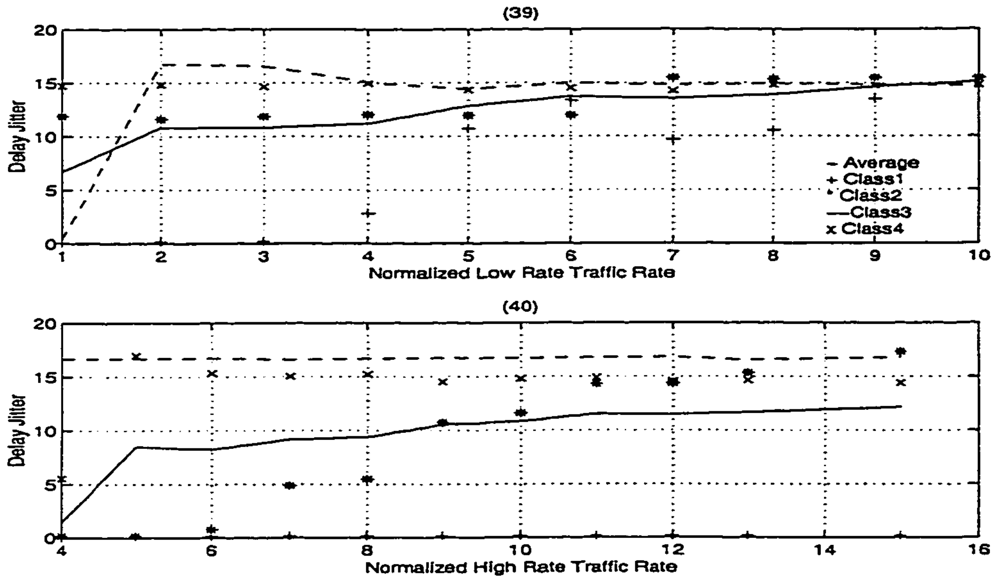


Fig. 3.29: Delay jitter vs. traffic rate where: (39) for low rate traffic and (40) for high rate traffic.

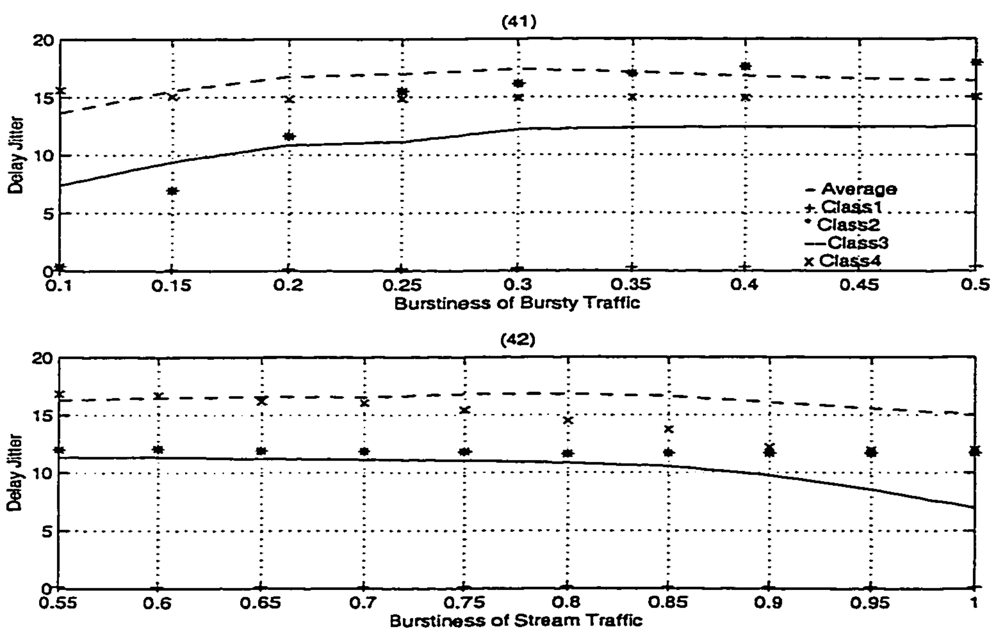


Fig. 3.30: Delay jitter vs. traffic burstiness where: (41) for bursty traffic and (42) for stream traffic.

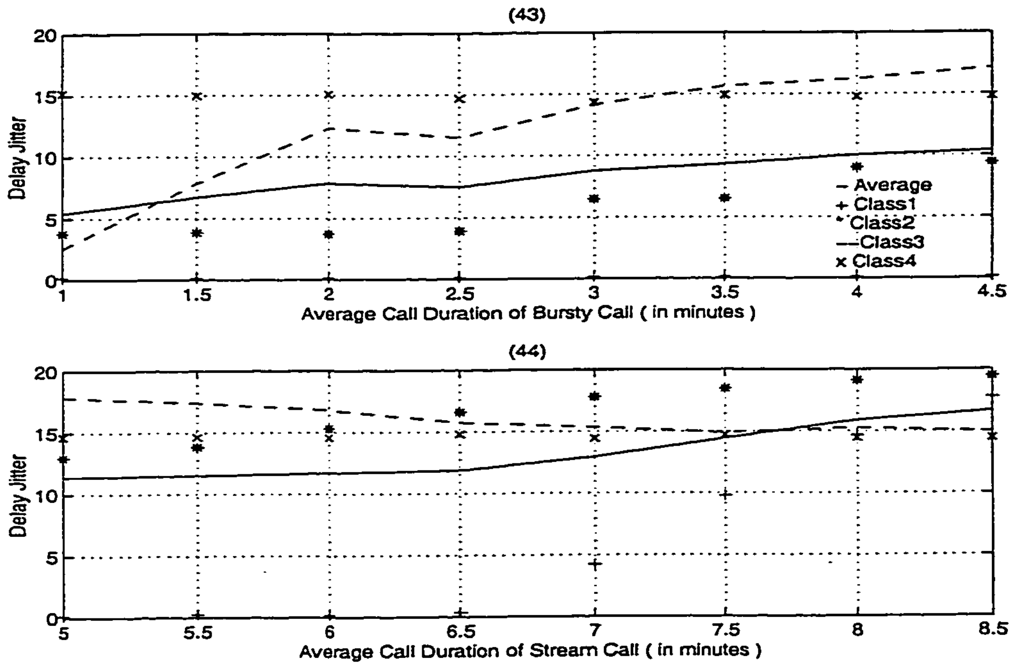


Fig. 3.31: Delay jitter vs. call duration where: (43) for short calls and (44) for long calls.

We noted from Fig.3.27 that the jump of delay jitter for a small changing is number of users. Class 4 users don't have much variations in delay due to every packet undergo longer delay. Class 2 and Class 3 users have some jump points due to the channel adjustment: with light traffic loading, class 2 and class 3 users with the high priority to access basic or reservation band without longer queuing. After the traffic build up, the class 2 and class 3 users are also undergo longer queuing delay even with high priority. Class 1 users always carrying light traffic, so class 1 users have very lower delay jitter. Other effects such as δ, θ and Td are same as the effects in the evaluation of call blocking, delay and throughput.

Fig. 3.32-Fig. 3.36 show the MFHMAC system buffer overflow probability vs. various traffic characters, load (number of users), traffic rate, traffic bursty level and call duration, and so on.

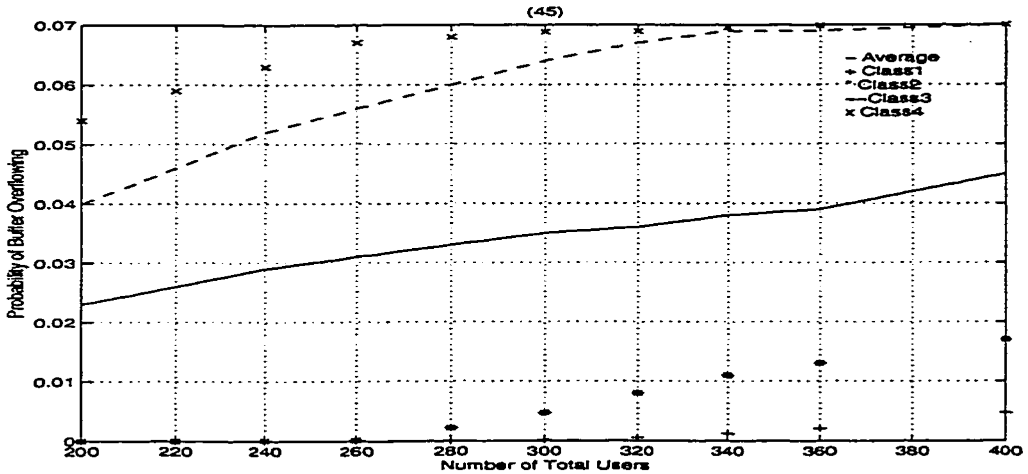


Fig.3.32. Buffer overflow vs. system users, where total users are equally distributed in each class.

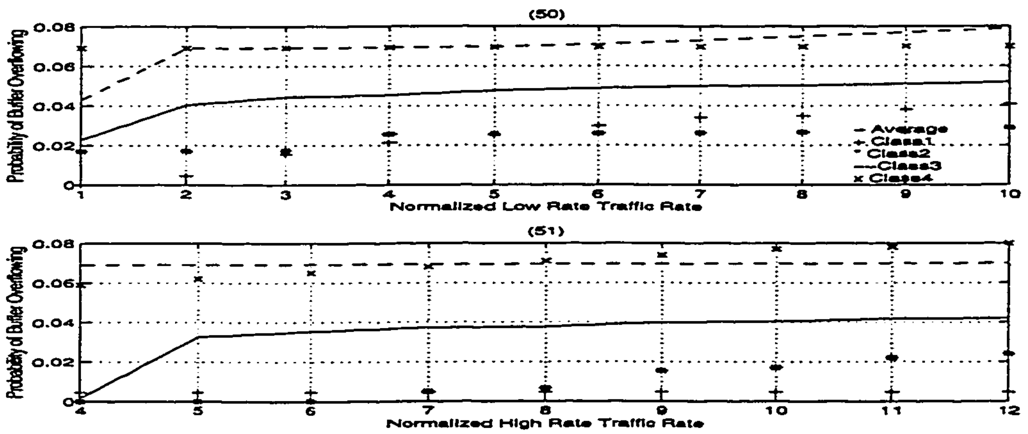


Fig.3.33. Buffer overflow vs. each class of users, where (46) for class 1, (47) for class 2, (48) for class 3 and (49) for class 4.

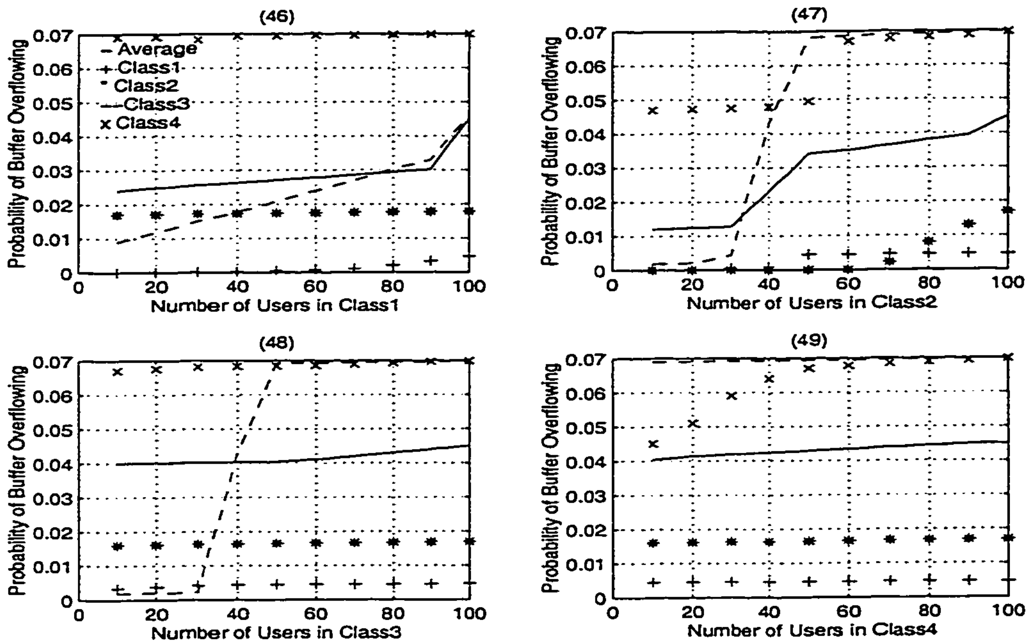


Fig. 3.34: Buffer overflow vs. traffic rate where: (50) for low rate traffic (51) for high rate traffic.

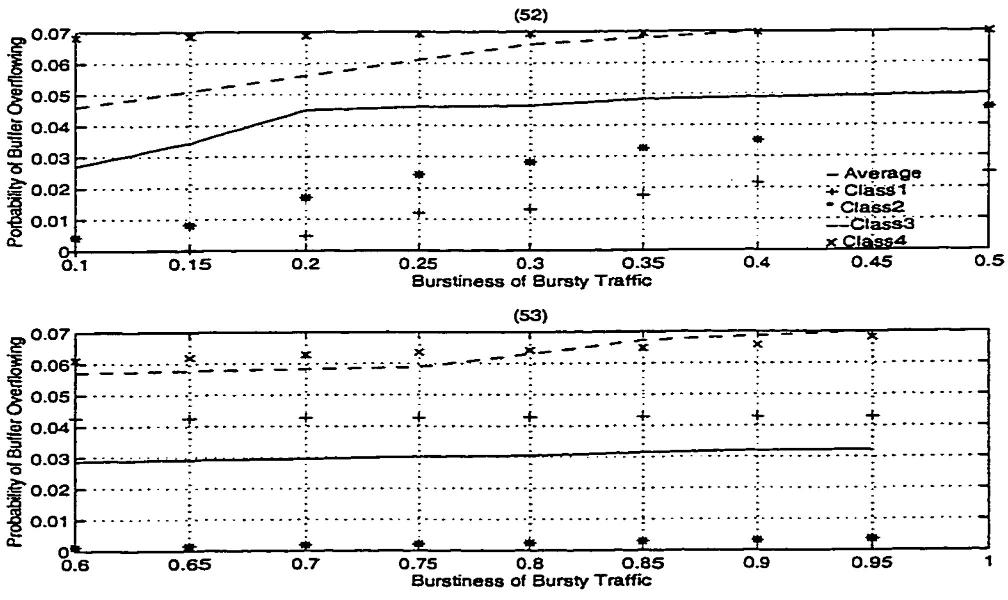


Fig. 3.35: Buffer overflow vs. traffic burstiness where: (52) for bursty traffic and (53) for stream traffic.

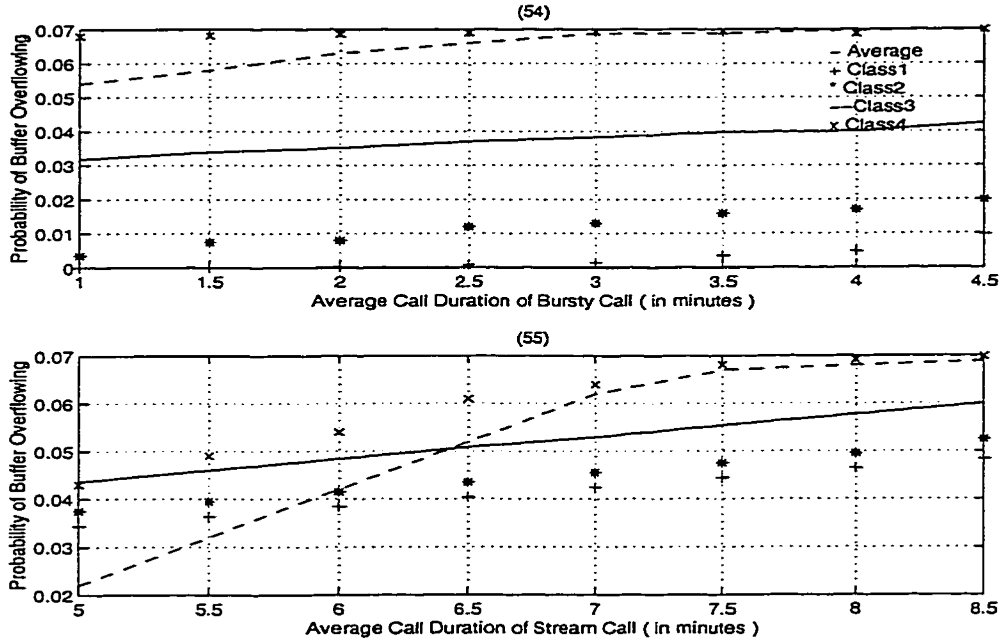


Fig. 3.36: Buffer overflow vs. call duration where: (54) for short calls and (55) for long calls.

From Fig.32-Fig.3.36, we noted that the probability of buffer overflow in each class almost perform same as the probability of call blocking in each class. This due to the use of finite buffer in each class.

When the FEC code used in an MFHMAC system, the simulation results are also shown in this section. Fig. 3.37-Fig. 3.41 show the call blocking versus various traffic characters, load (number of users), traffic rate, traffic bursty level and call duration in an MFHMAC system.

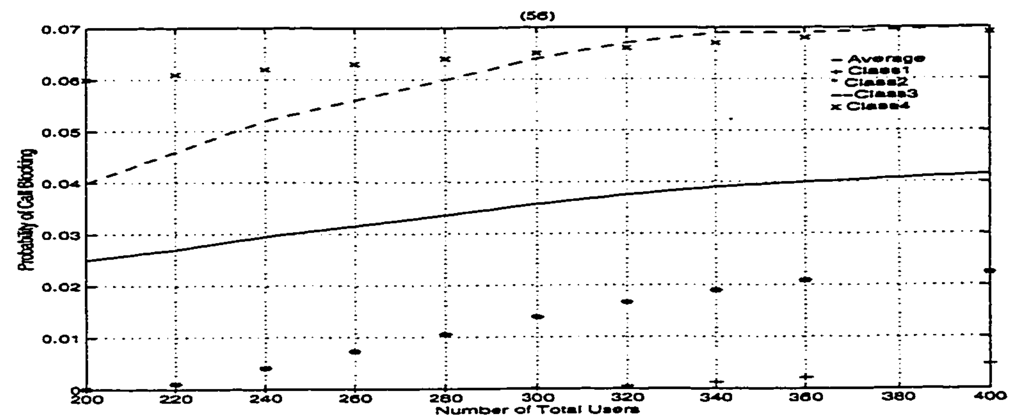


Fig. 3.37: Call blocking vs. system users in MFHMA with FEC coding where total users are equally distributed in each class.

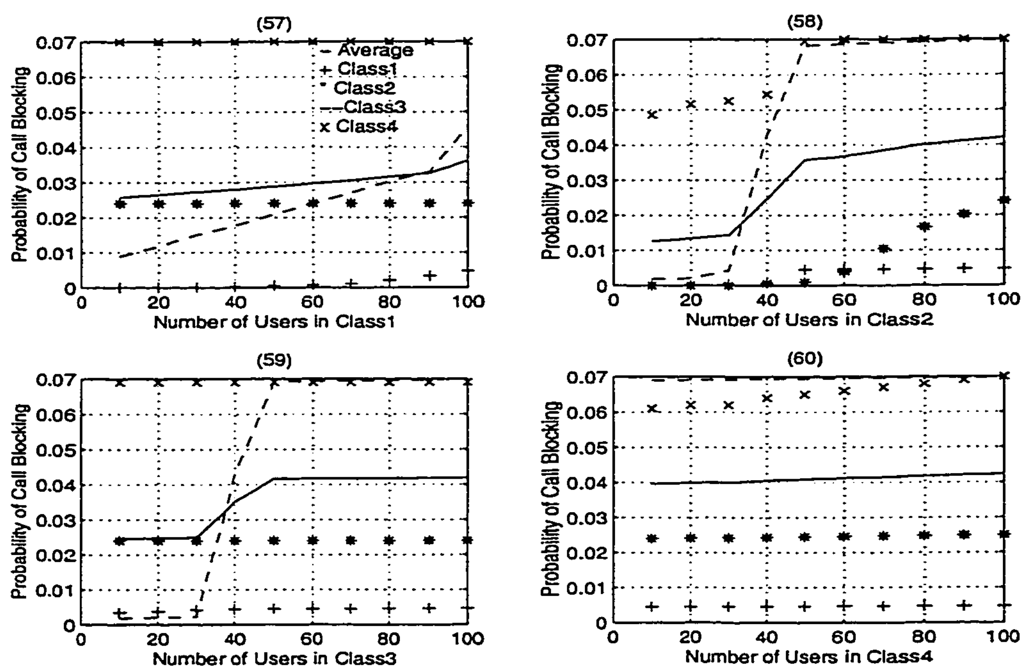


Fig. 3.38: Call blocking vs. each class of users in MFHMAC with FEC coding where: (57) for Class 1, (58) for Class 2, (59) for Class 3 and (60) for Class 4.

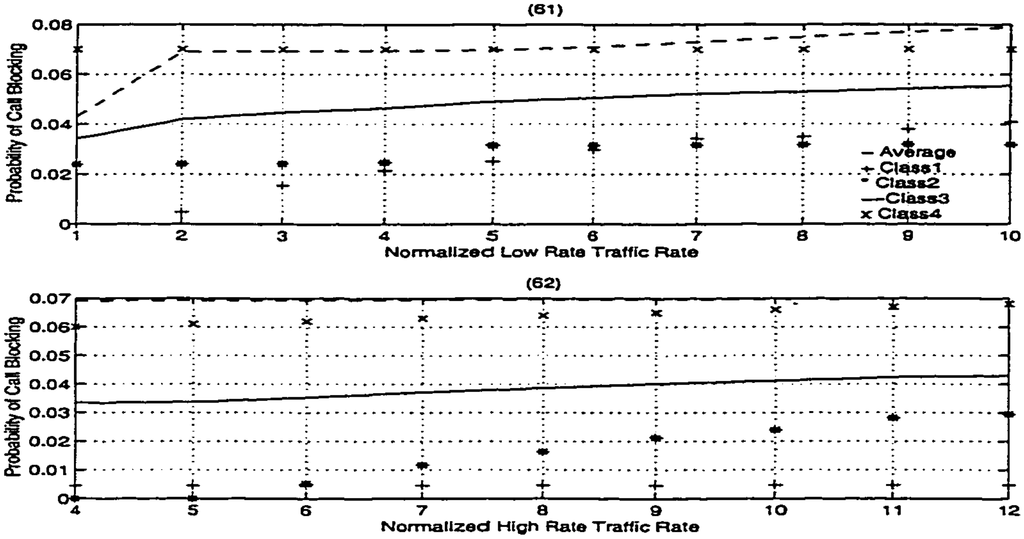


Fig. 3.39: Call blocking vs. traffic rate in MFHMAC with FEC coding where: (61) for low rate traffic and (62) for high rate traffic.

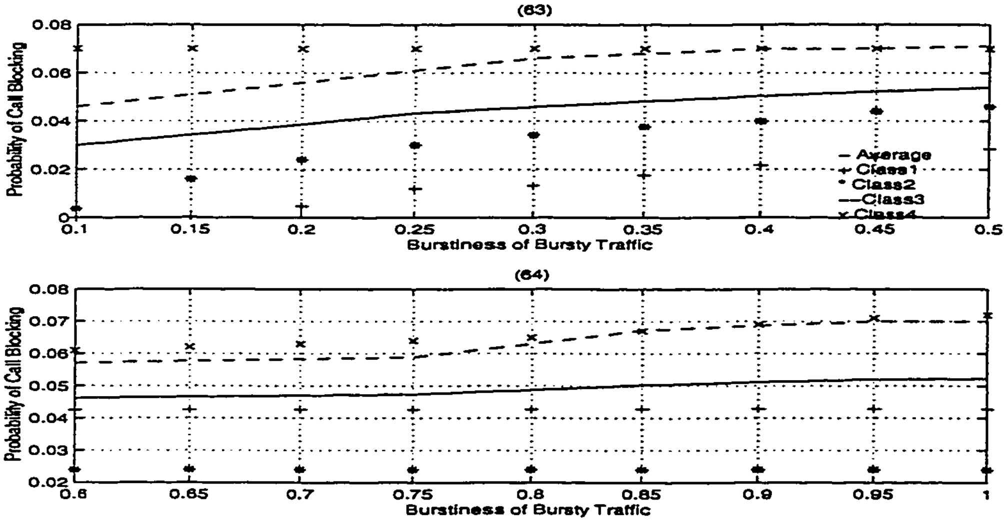


Fig.3.40. Call blocking vs. traffic burstiness in MFHMAC with FEC coding, where (63) for bursty traffic and (64) for stream traffic.

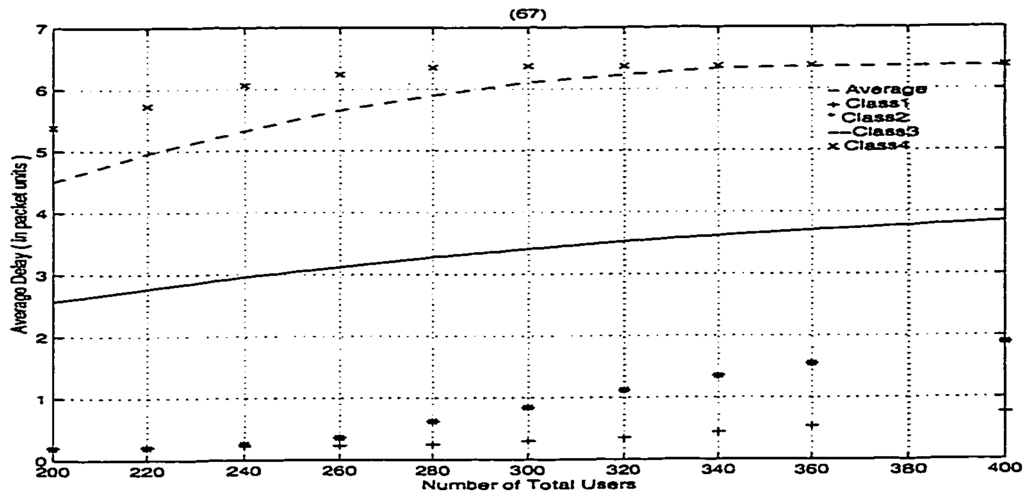


Fig. 3.41: Call blocking vs. duration in MFHMA with FEC coding where: (65) for short calls and (66) for long calls.

The simulation results of average delay in an MFHMAC system with FEC coding are shown in Fig. 3.42-Fig. 3.46.

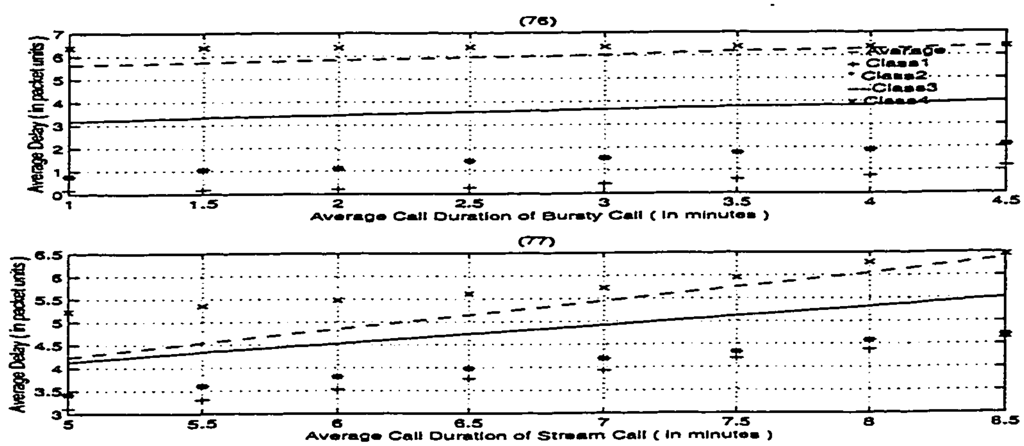


Fig.3.42. Delay vs. system users, where the total users are equally distributed in four classes.

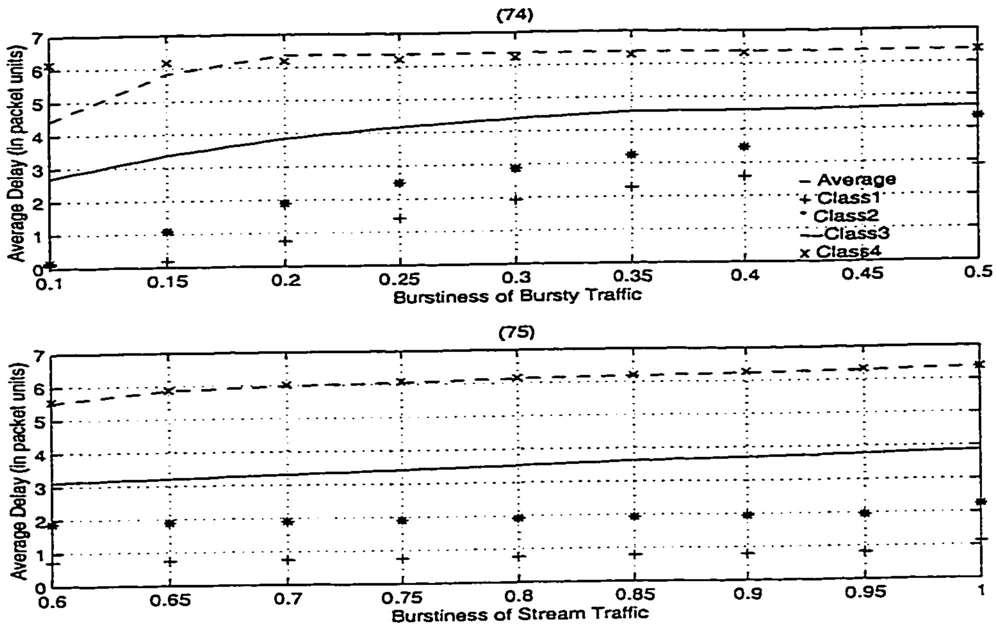


Fig. 3.43: Delay vs. each class of users where: (68) for Class 1, (69) for Class 2, (70) for Class 3 and (71) for Class 4

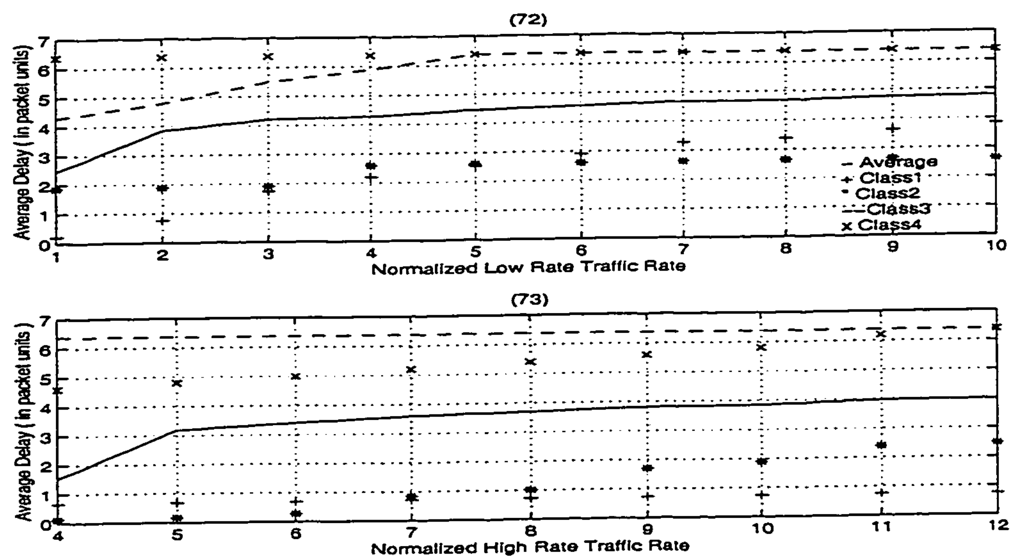


Fig. 3.44: Delay vs. traffic rate where: (72) for low rate traffic and (73) for high rate traffic.

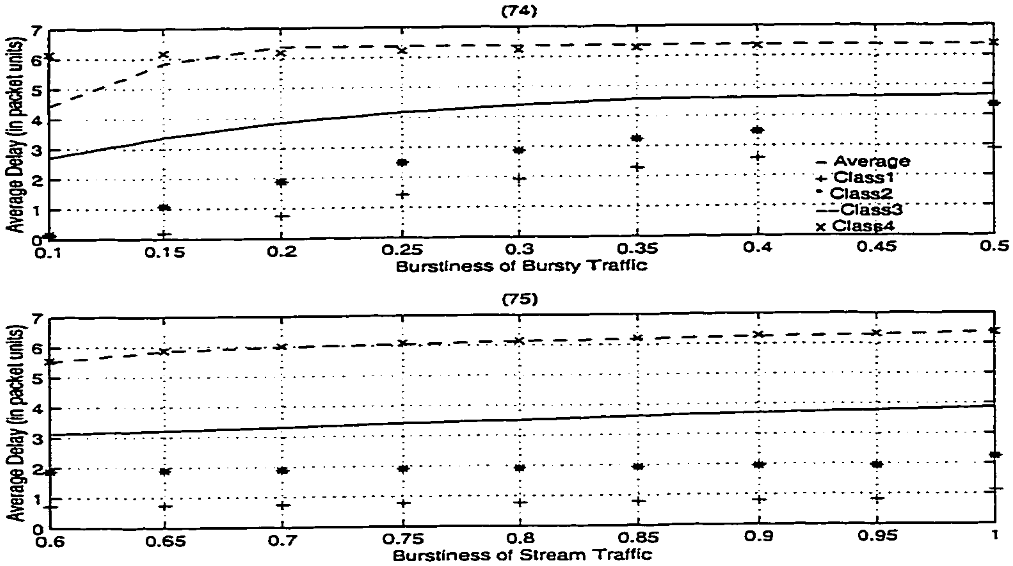


Fig. 3.45: Delay vs. traffic burstiness where: (T4) for bursty traffic and (75) for stream traffic.

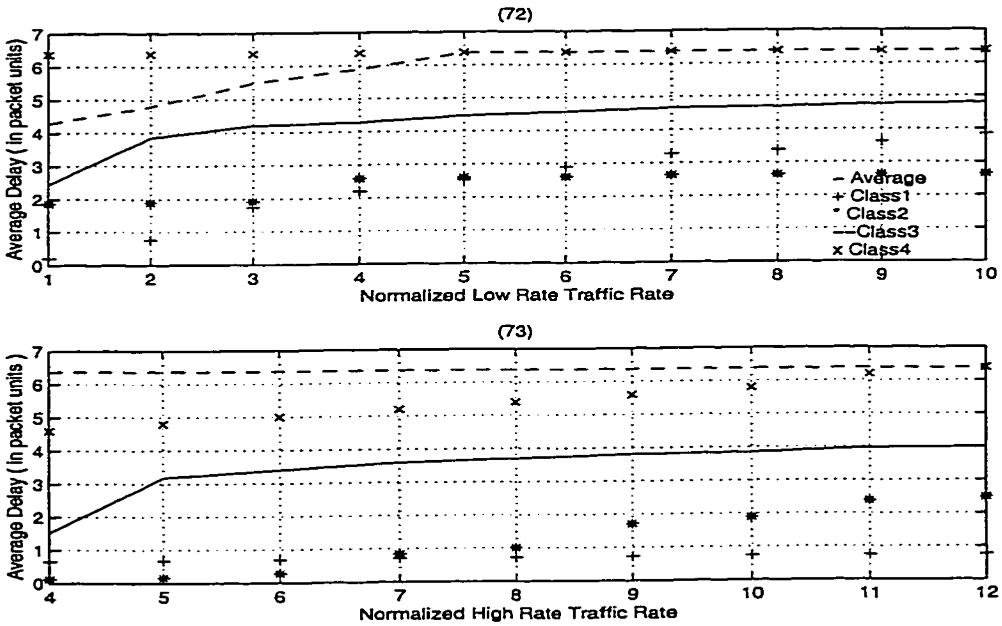


Fig. 3.46: Delay vs. call duration where: (76) for short calls and (77) for long calls.

The simulation results of delay jitter in an MFHMAC system with FEC coding are shown in Fig. 3.47-Fig. 3.51.

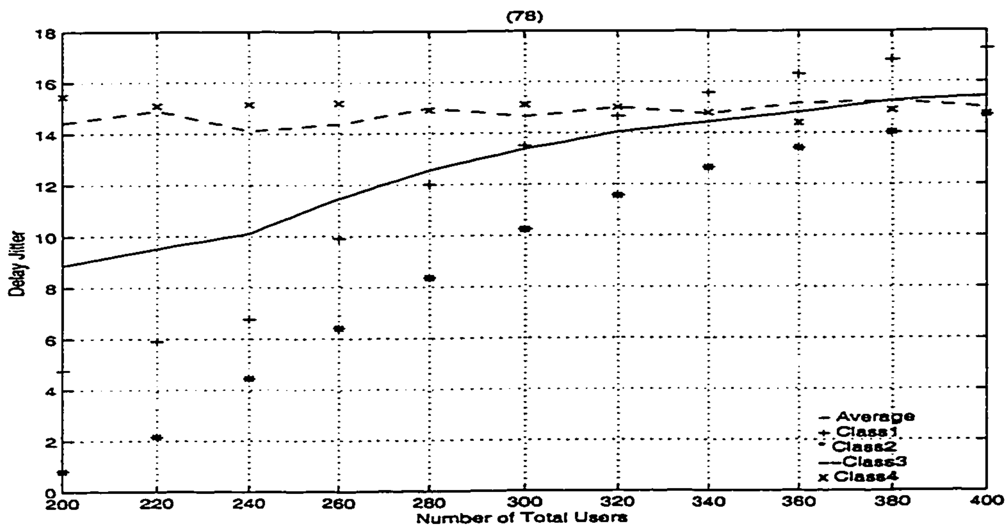


Fig. 3.47: Delay jitter vs. system users where total users are equally distributed in each class.

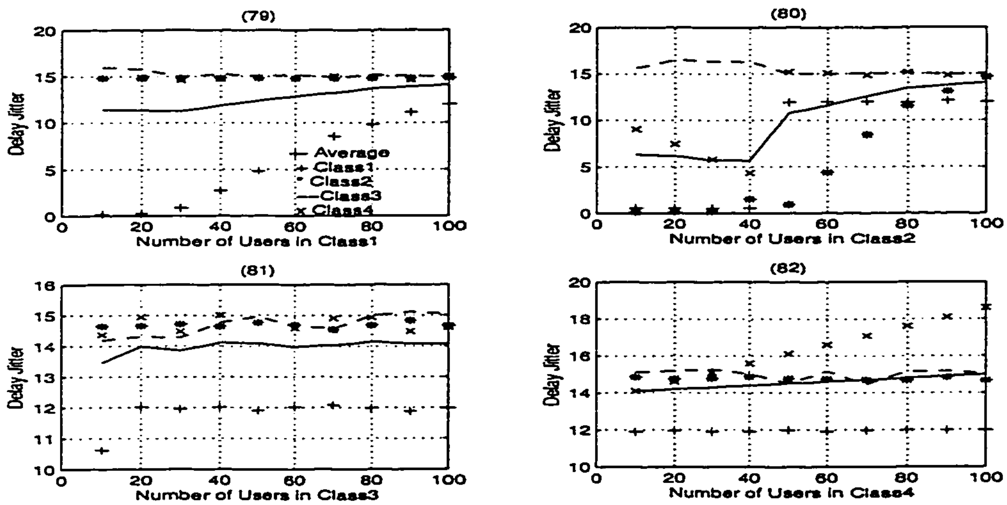


Fig. 3.48: Delay jitter Vs each class of users where: (79) for Class 1, (80) for Class 2, (81) for Class 3 and (82) for Class 4.

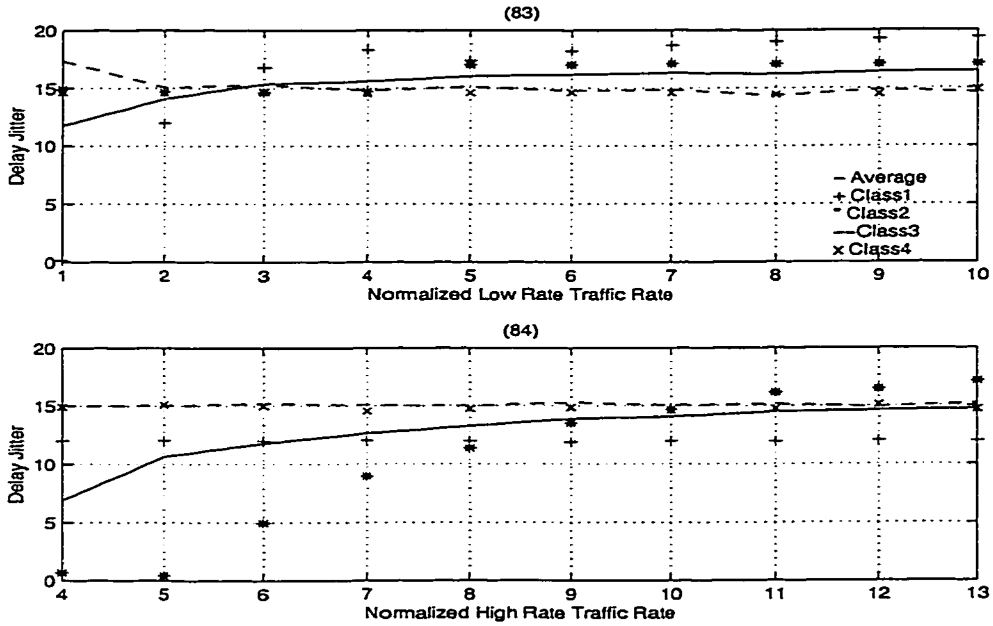


Fig. 3.49: Delay jitter vs. traffic rate where: (83) for low rate traffic and (84) for high rate traffic.

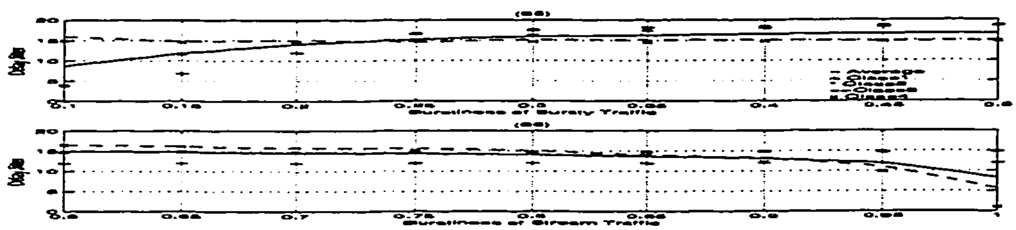


Fig. 3.50: Delay jitter vs. traffic burstiness where: (85) for bursty traffic and (86) for stream traffic.

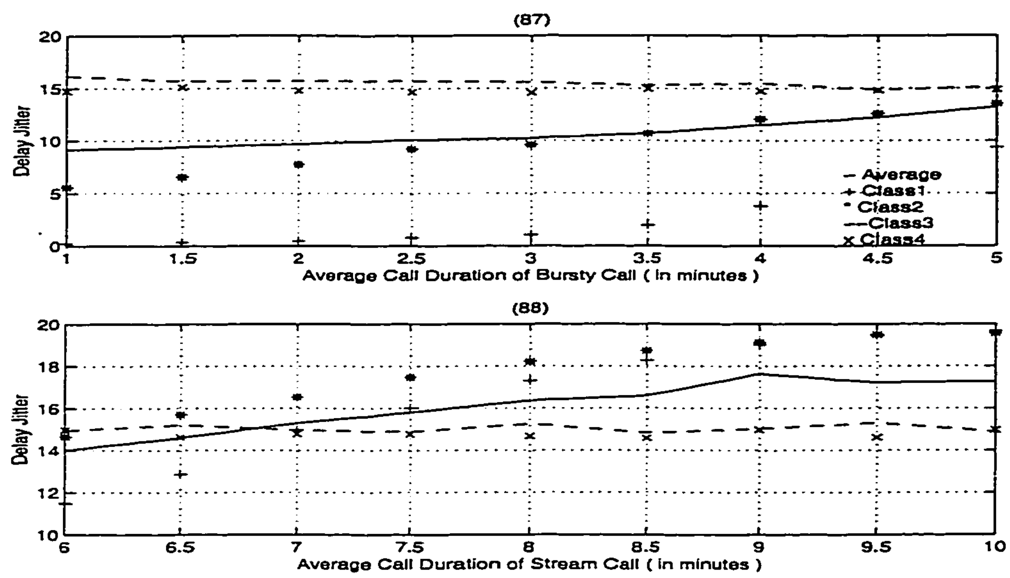


Fig. 3.51: Delay jitter vs. call duration where: (87) for short calls and (88) for long calls.

The simulation results of average throughput in an MFHMAC system with FEC coding are shown in Fig. 3.52-Fig. 3.56

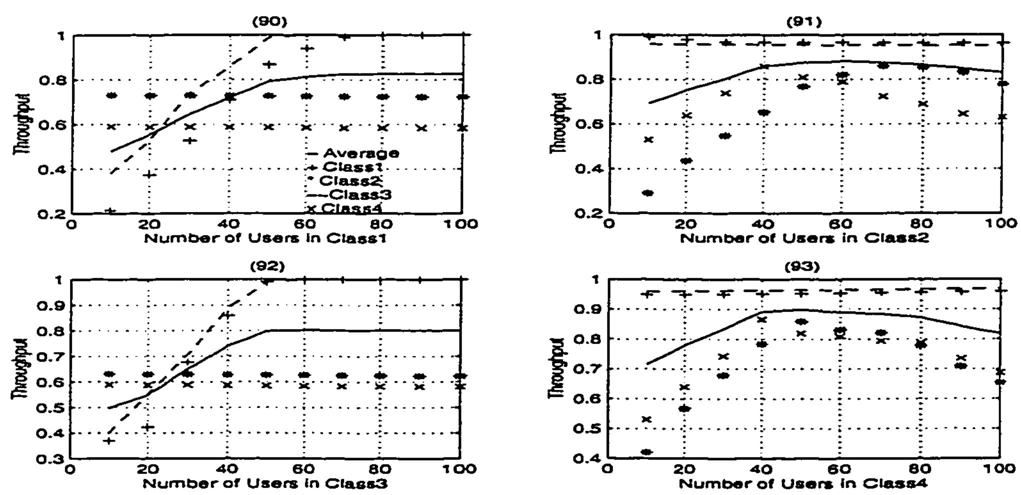


Fig. 3.52: Throughput vs. system users where total users are equally distributed in each class.

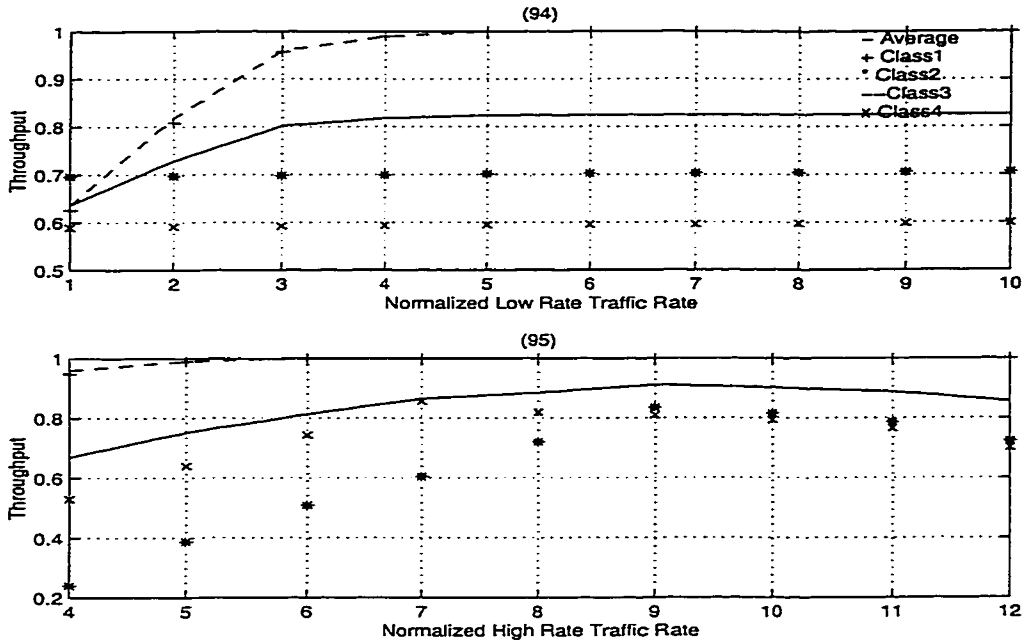


Fig. 3.53: Throughput Vs. each class of users where: (90) for Class 1, (91) for Class 2, (92) for Class 3 and (93) for Class 4.

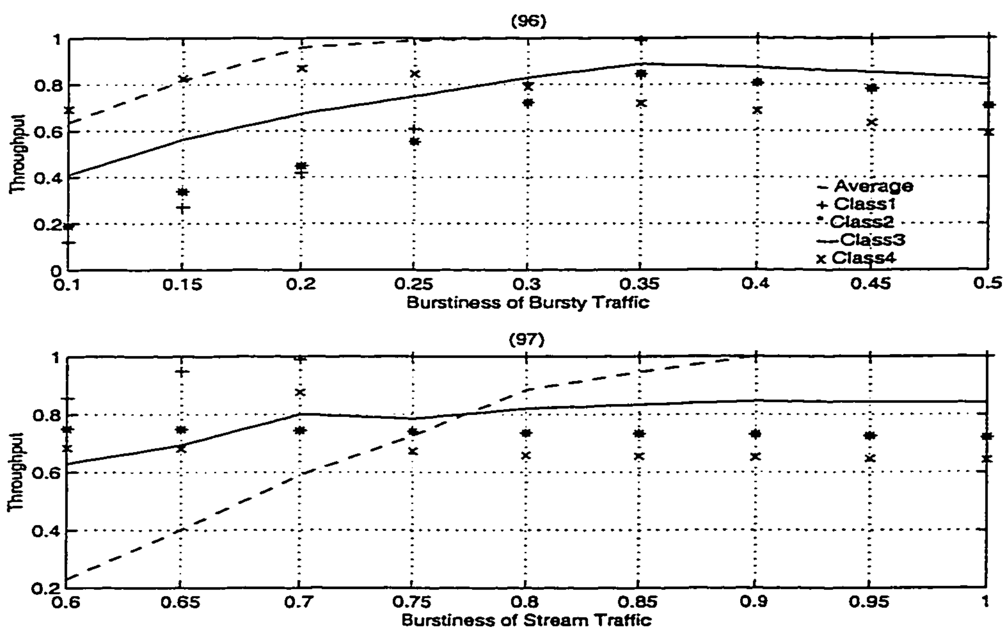


Fig. 3.54: Throughput vs. traffic rate where: (94) for low rate traffic and (95) for high rate traffic.

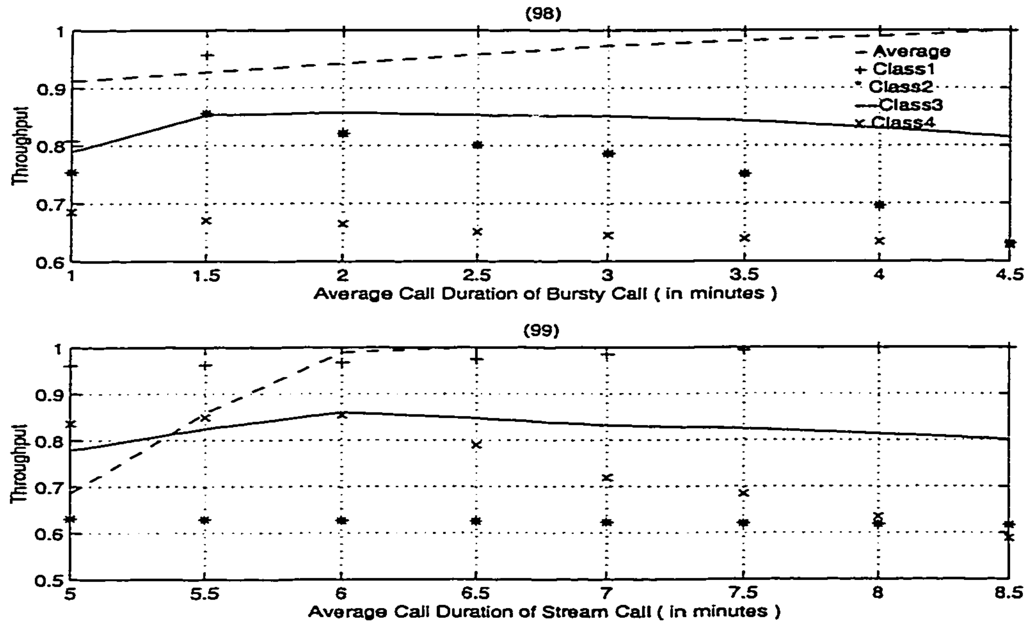


Fig. 3.55: Throughput vs. traffic burstiness where: (96) for bursty traffic and (97) for stream traffic.

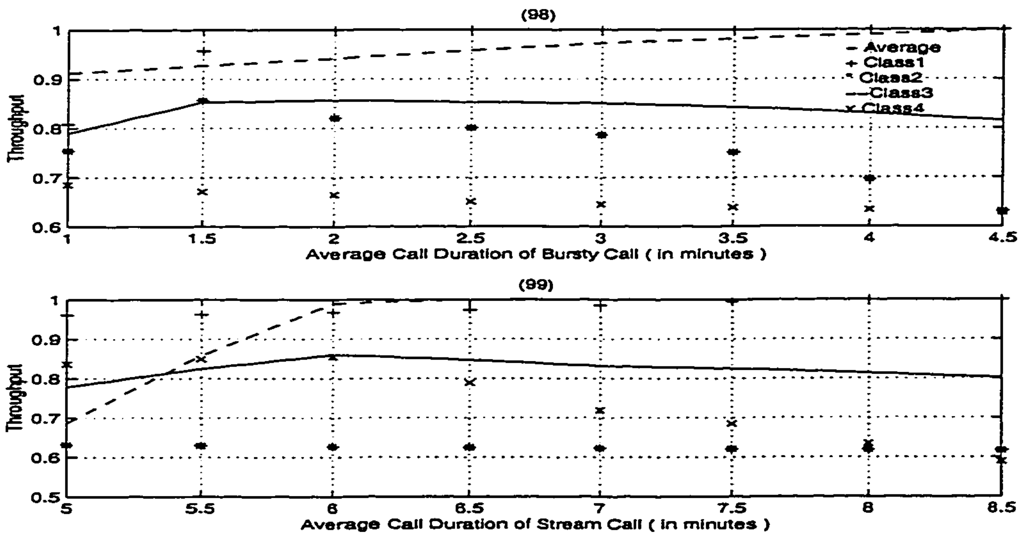


Fig. 3.56: Throughput vs. call duration where: (98) for short calls and (99) for long calls.

The simulation results of buffer overflow in an MFHMAC system with FEC coding are shown in Fig. 3.57-Fig. 3.61.

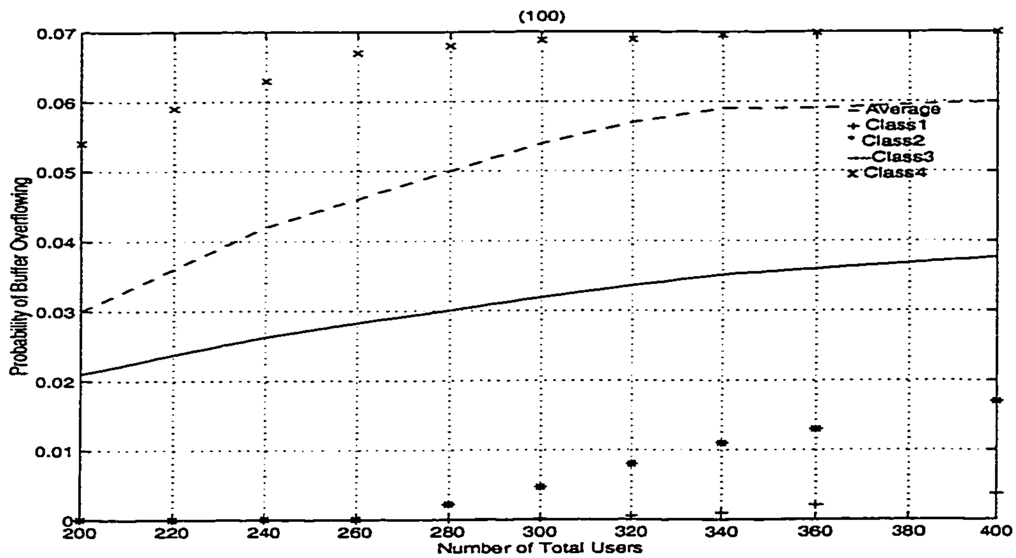


Fig. 3.57: Buffer overflow vs. user system users, where total users are equally distributed in each class.

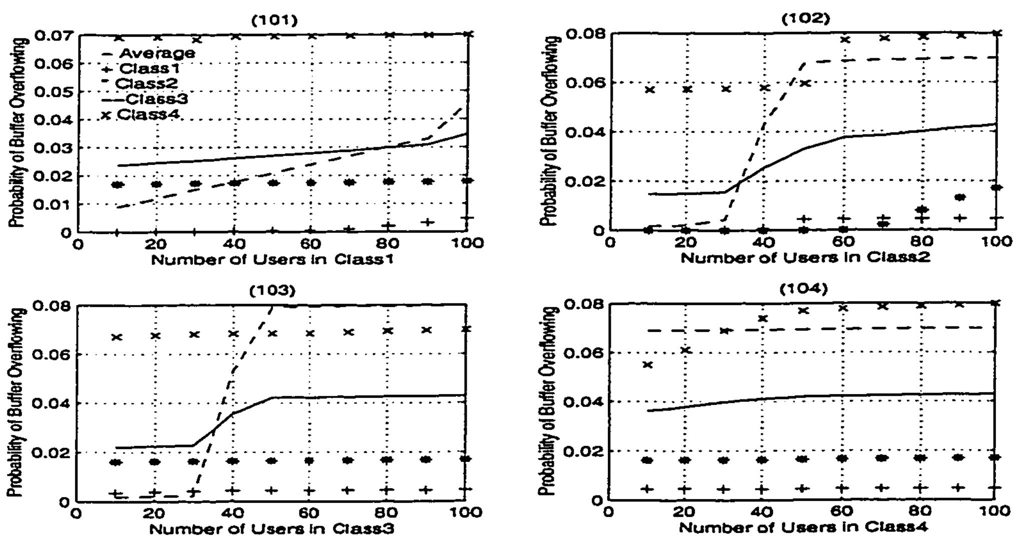


Fig. 3.58: Buffer overflow vs. each class of users where: (101) for Class 1, (102) for Class 2, (103) for Class 3 and (104) for Class 4.

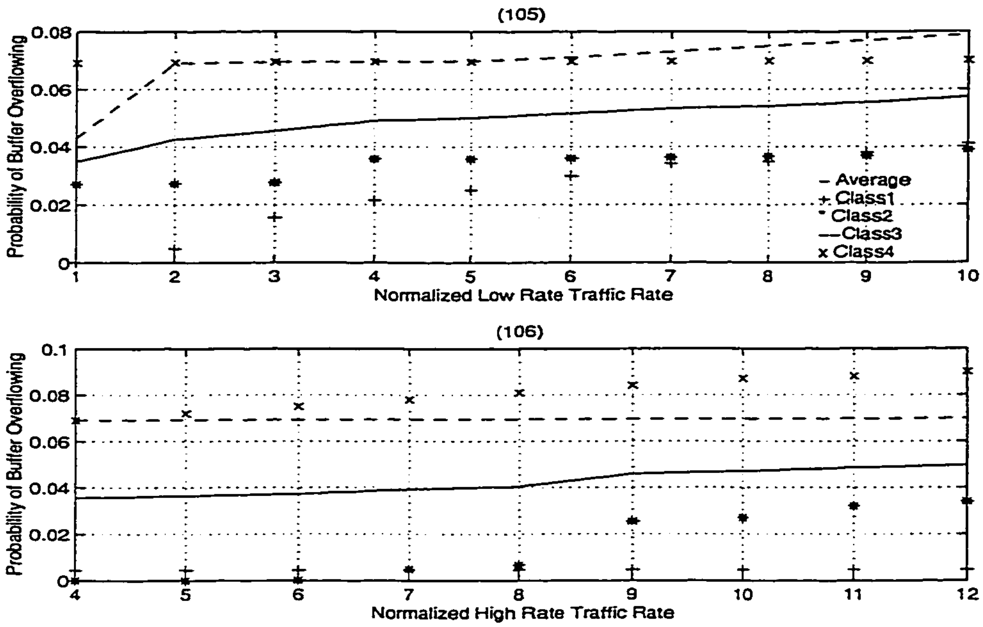


Fig. 3.59: Buffer overflow vs. traffic rate where: (105) for low ate traffic and (106) for high rate traffic.

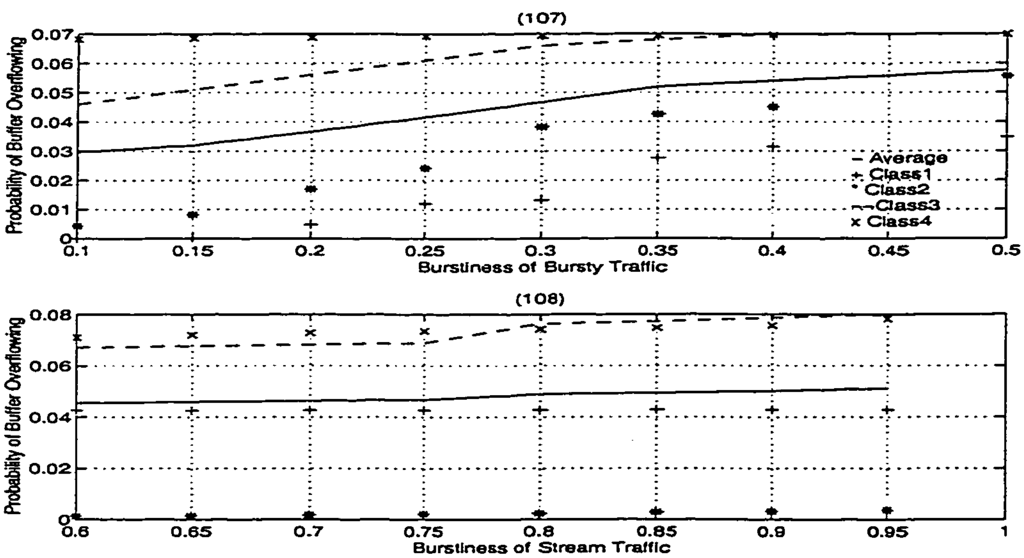


Fig. 3.60: Buffer overflow vs. traffic burstiness where: (107) for bursty traffic and (108) for stream traffic.

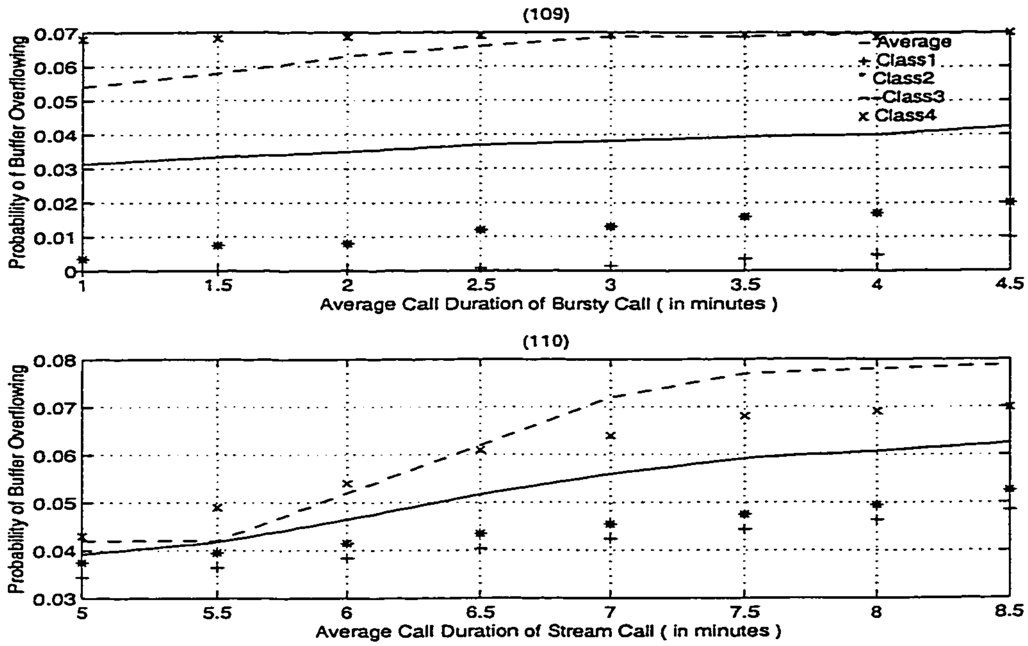


Fig. 3.61: Buffer overflow vs. call duration where: (109) for short calls and (110) for long calls.

From above observation, we noted that with utilization of FEC, the system will have better performance in call blocking, average delay, delay jitter, throughput and buffer overflow.

For an interesting observation, a comparison of FFHMAC with MFHMAC and MFHMAC with coding are shown in Fig. 3.62-Fig. 3.94. Fig. 3.62-Fig.72 compare the call blocking probabilities in three schemes with different situations. In Fig. 3.62, we can see the MFHMAC system average call blocking probability performs better than the FFHMAC system (0.013), when the system total users equally increase in each class and the MFHMAC with FEC coding even better than MFHMAC without coding (0.012).

More explicitly, we can see, from Fig. 3.63, as Class 1 and Class 2 users increase (bursty traffic load), MFHMAC with FEC coding and without FEC coding both perform better than FFHMAC (0.02-0.03). Also from Fig. 3.64 to Fig. 3.66, it is easy to see that when the traffic rate increases the MFHMAC with FEC with coding and without FEC coding will have better performance than FFHMAC, especially the QoS in Class 1 and Class 2 will improve a lot. From Fig. 3.67-F.3.69, we can see clearly that as the traffic becomes more streamed (load increase) the average call blocking probability in MFHMAC with FEC coding is much better than the FFHMAC system, especially for Class 1 and Class 2 users. From Fig. 3.70-Fig. 3.72 we can see when the caller's call duration increase, the call blocking will increase in both FFHMAC system and MFHMAC system, however, the MFHMAC technique will perform better than FFHAMC.

Fig. 3.73-Fig. 3.83 show the comparison of average delay in MFHMAC and in FFHMAC system. Clearly, MFHMAC with FEC coding and without FEC coding has less delay than FFHAMC technique. For more specifically, when the number of users (total load) increase, MFHMAC (with FEC coding and without FEC coding) will have less 0.45 (0.55) than the FFHMAC technique. This is shown in Fig. 3.73 and Fig. 3.74. Other scenarios shown in Fig. 3.75-Fig. 3.77 is that when the traffic rate increases the MFHMAC (with FEC coding and without FEC coding) also has less delay than the FFHMAC technique (0.45). and from Fig. 3.78 to Fig. 3.80 we can see that the MFHMAC scheme (with and without FEC coding) has less delay than the FFHMAC (0.23) when the traffic became more stream and from Fig. 3.81 to Fig. 3.83 we find that when the bursty call's duration increase, MFHMAC scheme (with and without FEC coding) will have less delay (0.65).

From Fig. 3.84-Fig. 3.94 we can see the comparison of the delay jitter in MFHMAC (with and without DFEC coding) and FFHMAC system. In Fig. 3.84 and Fig.

3.85 very clearly that the MFHMAC (with coding) has less delay jitter than the FFHMAC scheme (25 percent) when the system users (load) increase, especially for Class 1 and Class 2 traffic (bursty traffic)..

From Fig. 3.86 to Fig. 3.88 the observation can be found that when the traffic rate increases the MFHMA (with FEC coding) has less than 26% delay jitter than the FFHMAC. And from Fig. 3.89-Fig. 3.91 we can see that when traffic becomes more stream the MFHMAC (with FEC coding) has less than 30% delay jitter, and from Fig. 3.92-Fig. 3.94 we also can find that the MFHMAC (with FEC coding) will have less than 35% delay jitter than FFHMAC when the user's call duration increases.

Since the throughput and the buffer overflow of the FFHMAC system we evaluated in this thesis are just in theoretical model, we didn't put these comparisons in the figure.

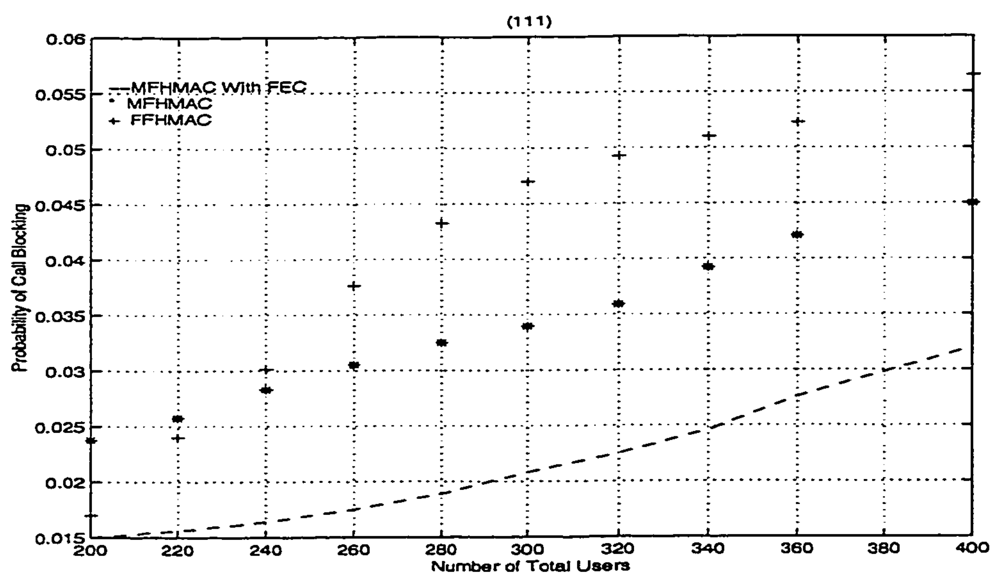


Fig. 3.62: Comparison of call blocking in MFHMAC and FFHMAC vs. system users.

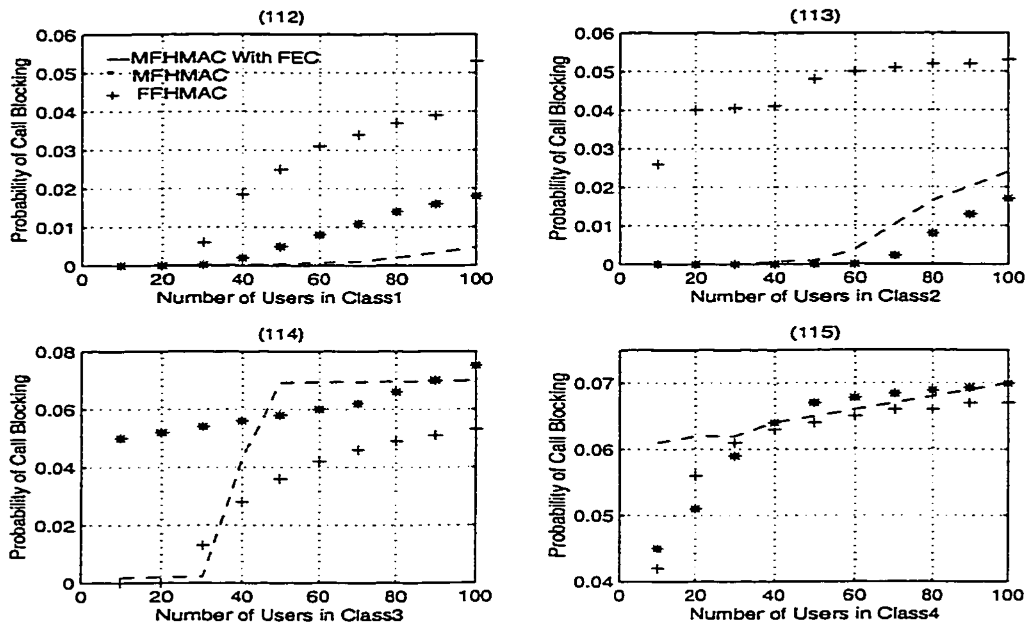


Fig. 3.63: Comparison of call blocking in MFHMAC and FFHMAC vs. each class of users where: (112) for Class 1, (113) for Class 2, (114) for Class 3 and (115) for Class 4.

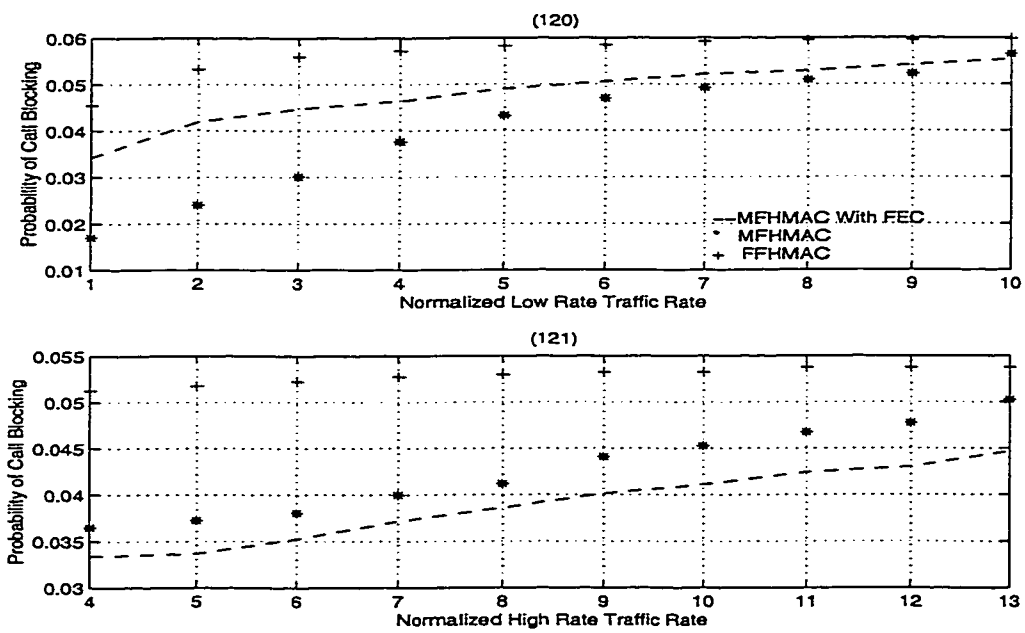


Fig. 3.64: Comparison of call blocking in MFHMAC and FFHMAC vs. traffic rate, where: (120) for low rate traffic and (121) for high rate traffic.

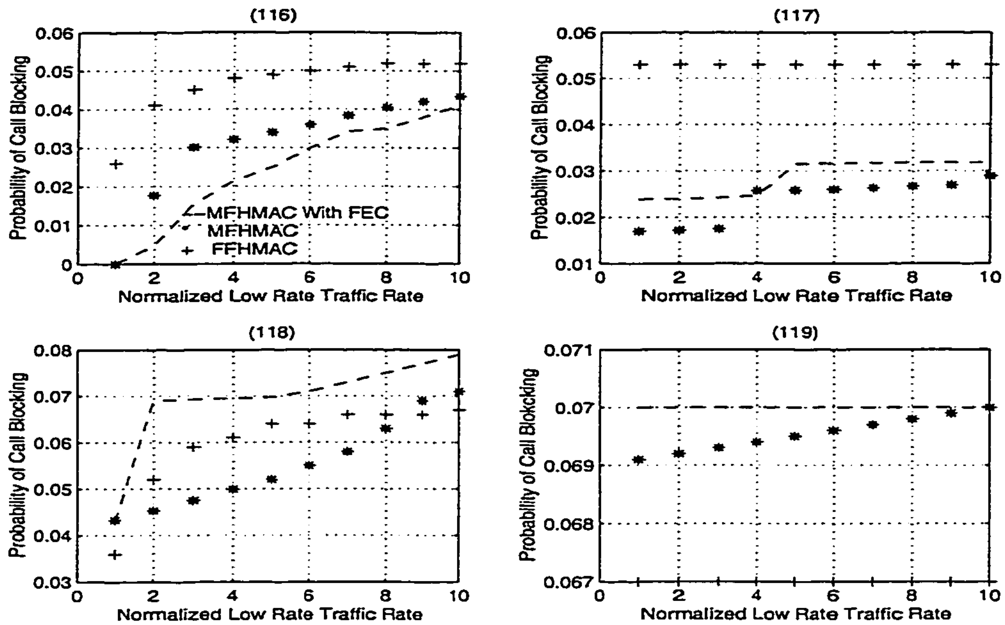


Fig. 3.65: Comparison of each class call blocking in MFHMAC and FFHMAC vs. low rate traffic where: (116) for Class 1, (117) for Class 2, (118) for Class 3 and (119) for Class 4.

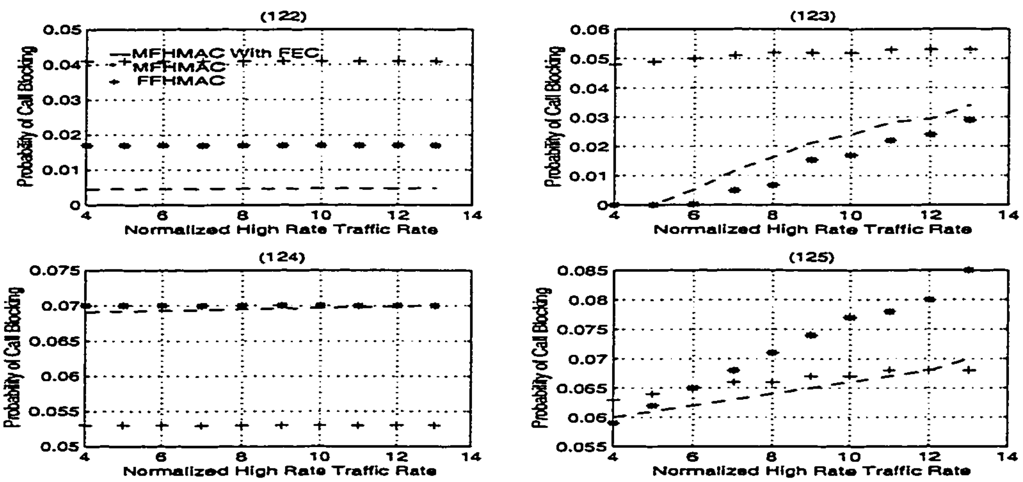


Fig. 3.66: Comparison of each class call blocking in MFHMAC and FFHMAC Vs. high rate traffic where: (122) for Class 1, (123) for Class 2, (124) for Class 3 and (125) for Class 4.

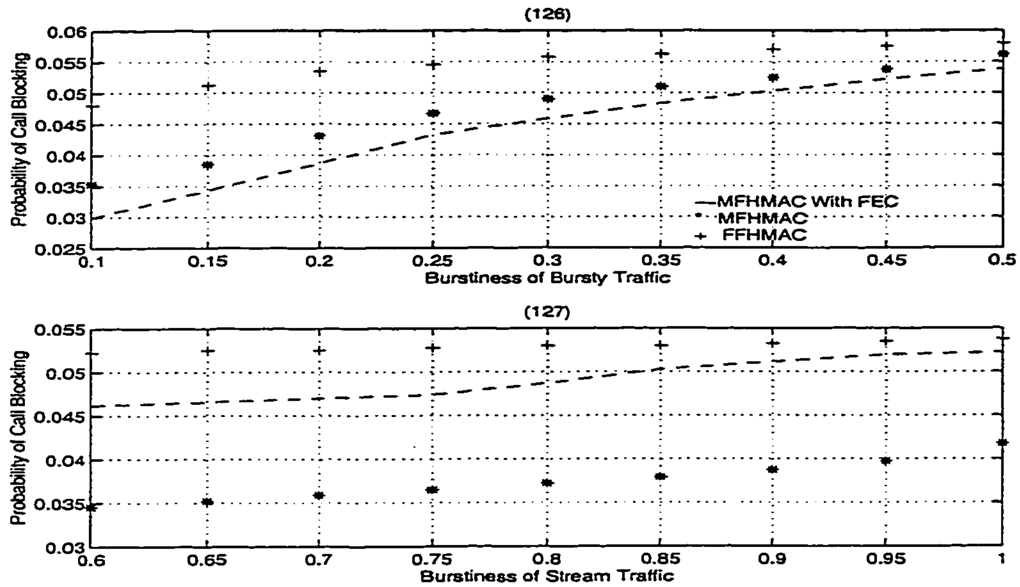


Fig. 3.67: Comparison of system call blocking in MFHMAC and FFHMAC vs. traffic burstiness where: (126) for bursty traffic and (127) for stream traffic.

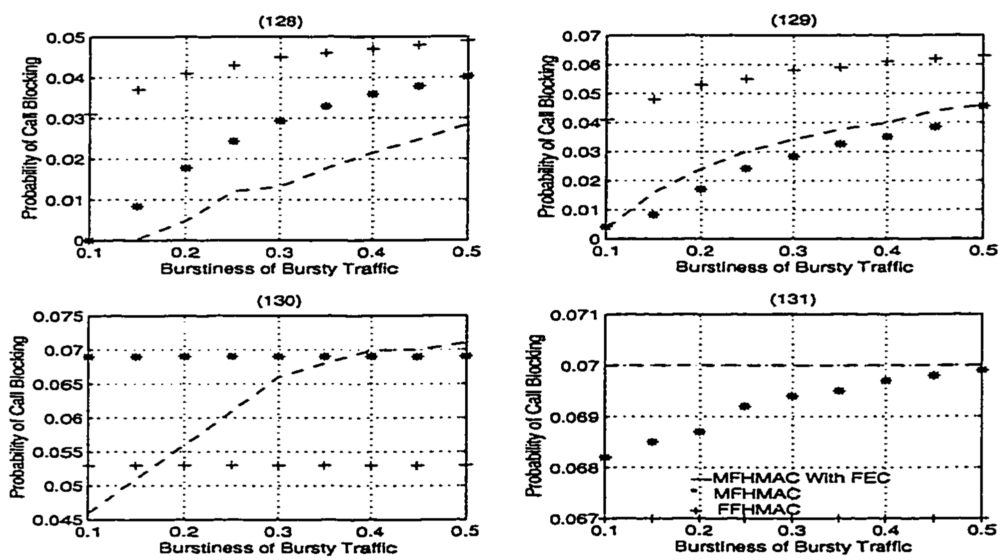


Fig. 3.68: Comparison of each class of call blocking in MFHMAC and FFHMAC Vs. bursty traffic burstiness where: (128) for Class 1, (129) for Class 2, (130) for Class 3 and (131) for Class 4.

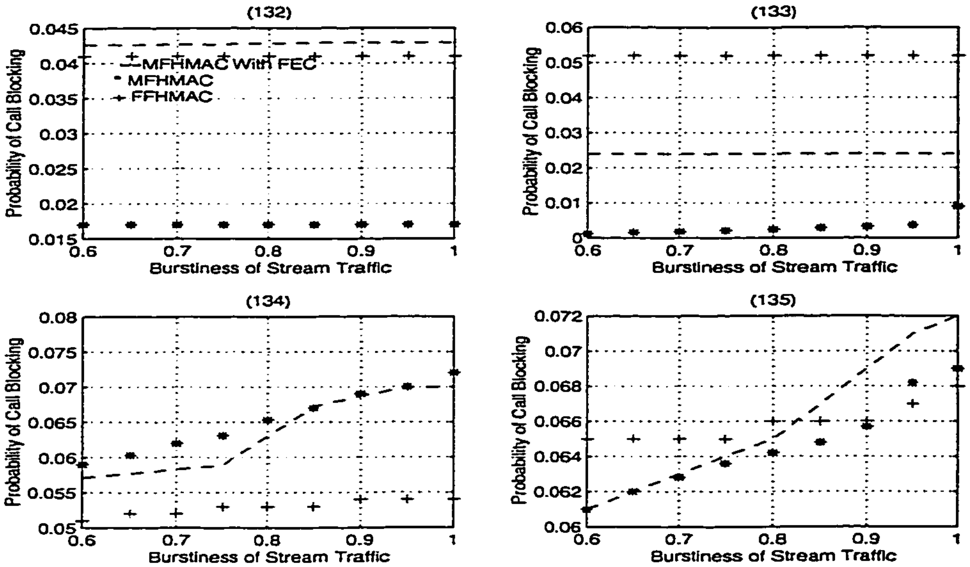


Fig. 3.69: Comparison of each class call blocking in MFHMAC and FFHMAC Vs. stream traffic burstiness where: (132) for Class 1, (133) for Class 2, (134) for Class 4 and (135) for Class 4.

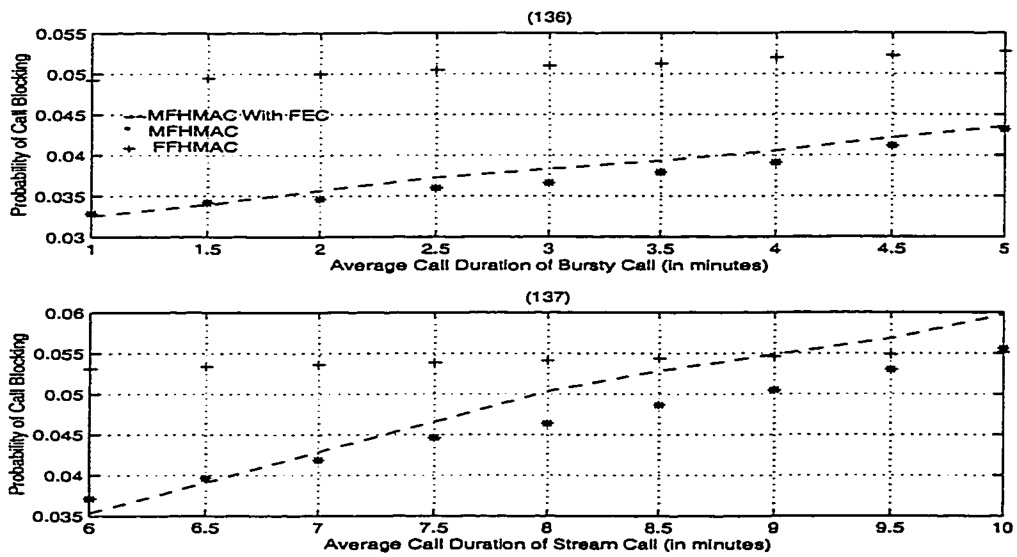


Fig. 3.70: Comparison of system call blocking in MFHMAC and FFHMAC Vs. call duration: (136) for short calls and (137) for long calls.

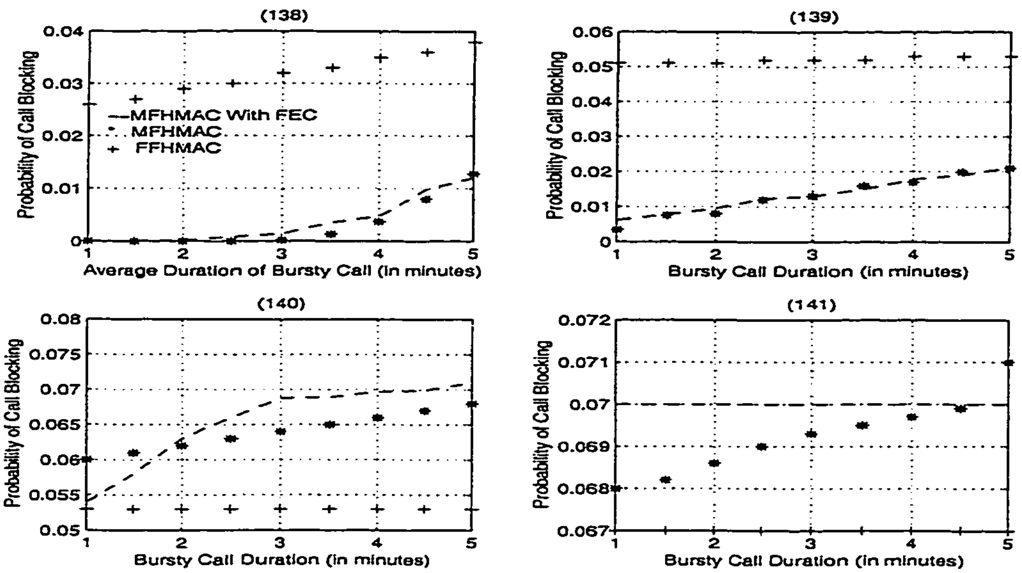


Fig. 3.71: Comparison of each class call blocking in MFHMAC and FFHMAC vs. short call duration: (138) for Class 1, (139) for Class 2, (140) for Class 3, and (141) for Class 4.

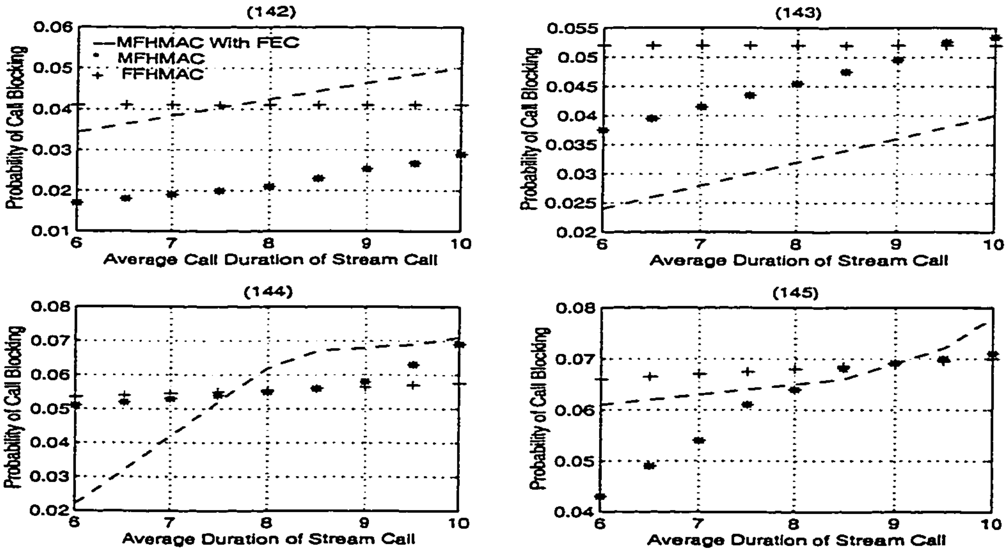


Fig. 3.72: Comparison of each class call blocking in MFHMAC and FFHMAC vs. long call duration: (142) for Class 1, (143) for Class 2, (144) for Class 3 and (145) for Class 4.

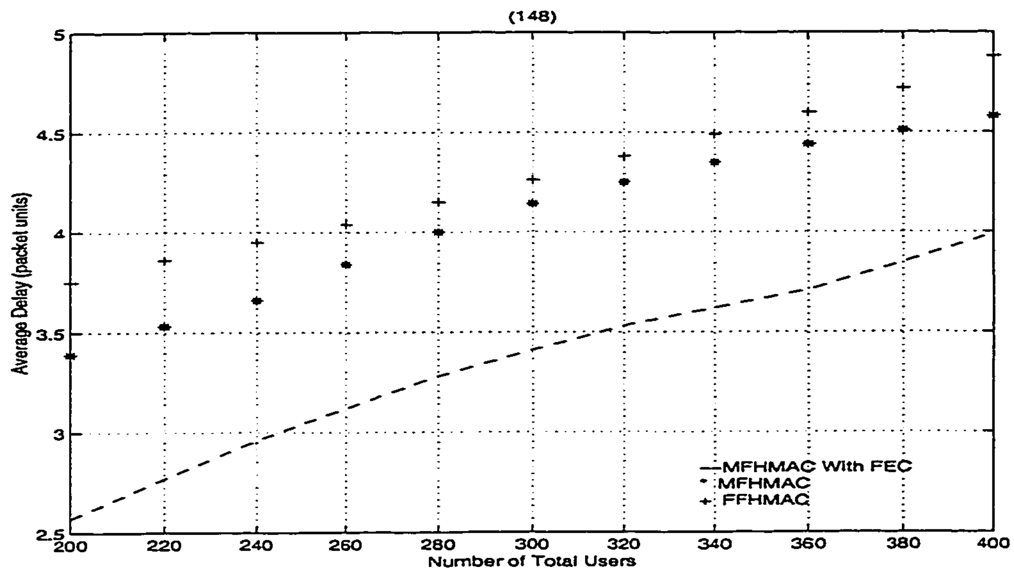


Fig. 3.73: Comparison of system average delay in MFHMAC and FFHMAC vs. system users.

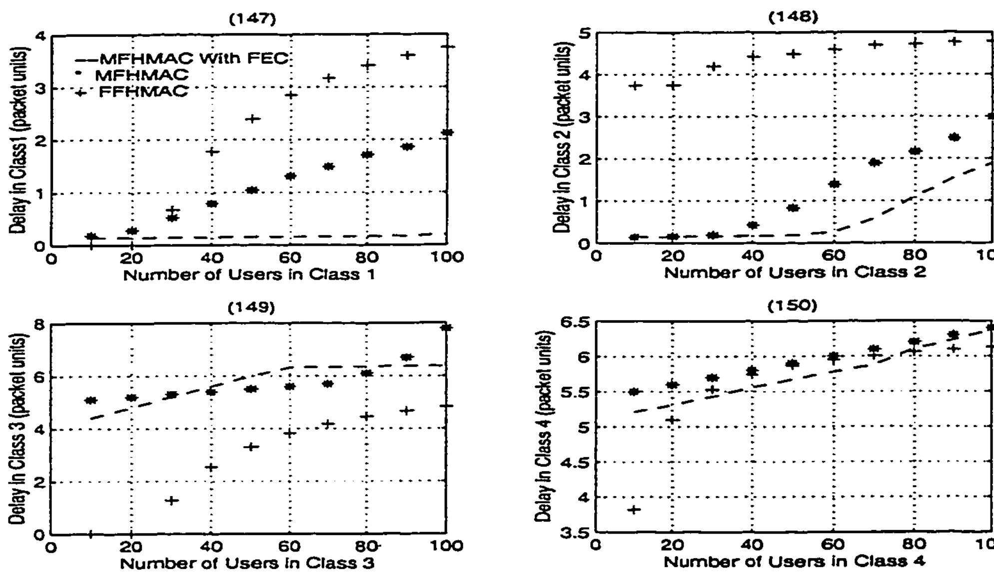


Fig. 3. 74: Comparison of average delay in each class in MFHMAC and FFHMAC Vs. each class users: (147) Class 1, (148) for Class 2, (149) for Class 3 and (150) for Class 4.

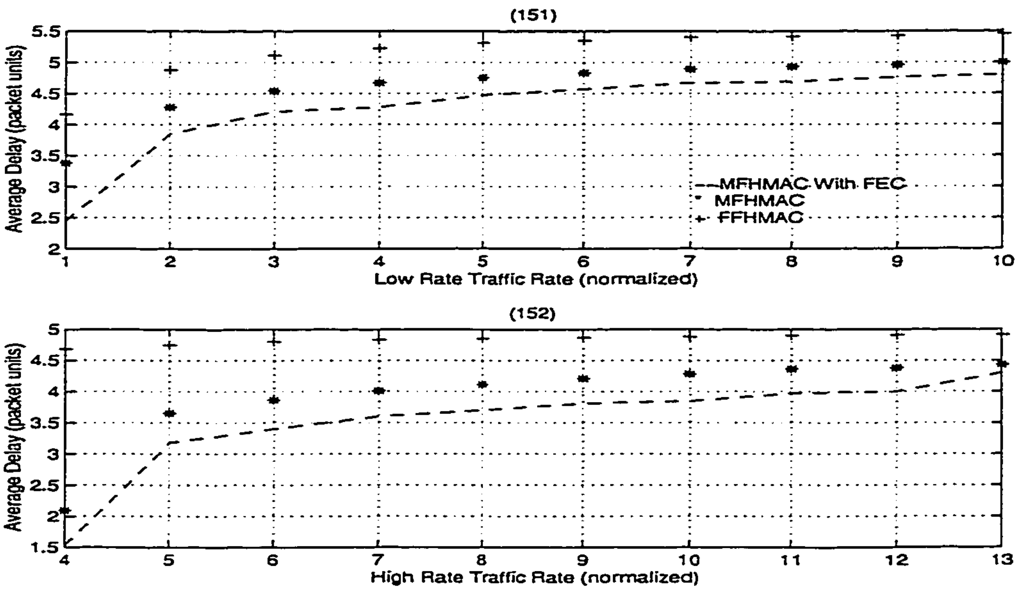


Fig. 3.75: Comparison of systems average delay in MFHGMAC and FFHMAC vs. traffic rate: (151) for low rate traffic and (152) for high rate traffic.

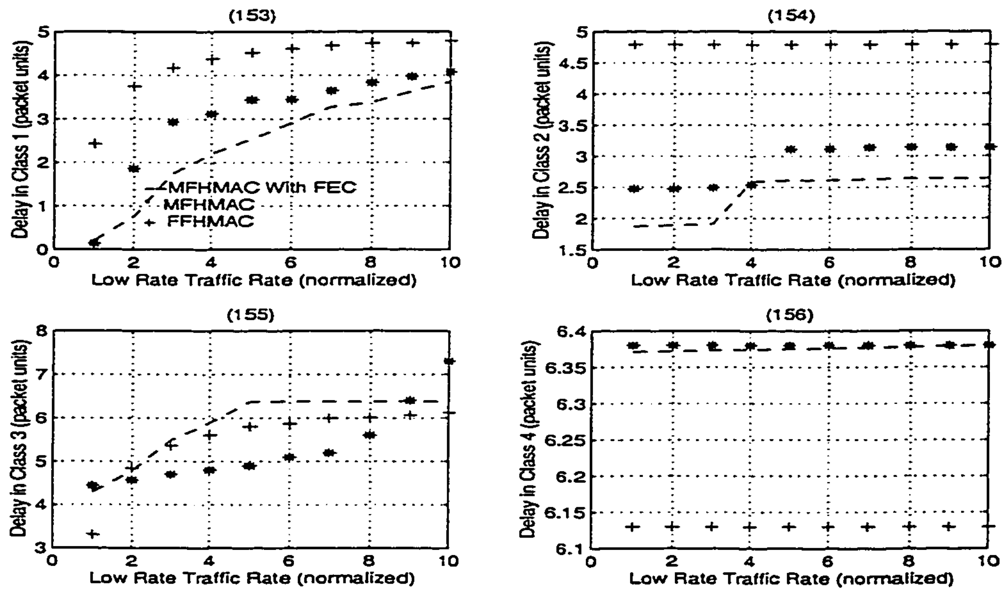


Fig. 3.76: Comparison of each class average delay in MFHMAC and FFHMAC vs. low rate traffic: (153) for Class 1, (154) for Class 2, (155) for class 3, and (156) for Class 4.

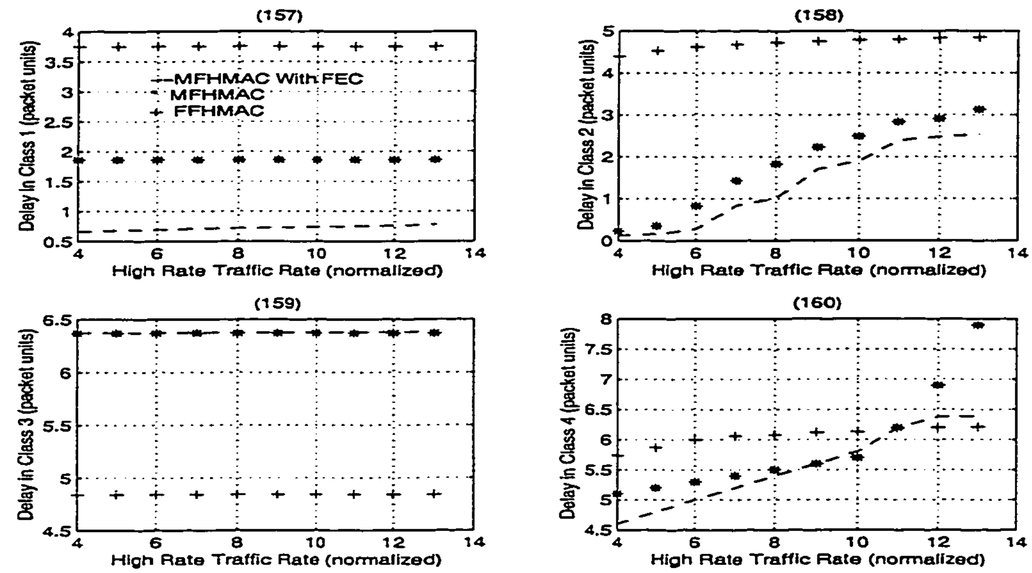


Fig. 3.77: Comparison of each class average delay in MFHMAC and FFHMAC vs. high rate traffic: (157) for Class 1, (158) for Class 2 (159) for Class 3 and (160) for Class 4.

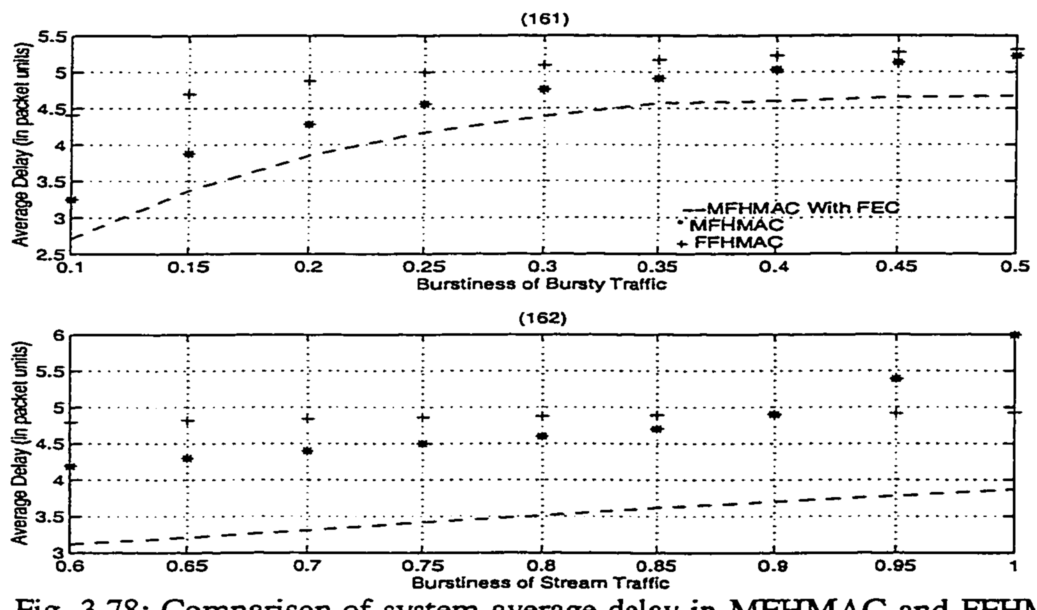


Fig. 3.78: Comparison of system average delay in MFHMAC and FFHMAC vs. traffic burstiness: (161) for bursty traffic and (162) for stream traffic.

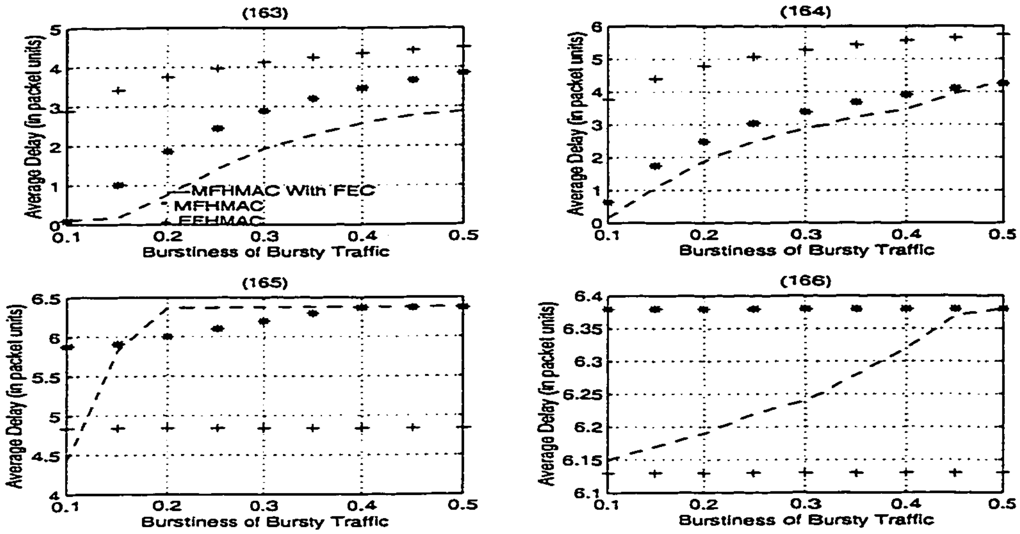


Fig. 3.79: Comparison of each class average delay in MFHMAC and FFHMAC vs. bursty traffic burstiness: (163) for Class 1, (164) for Class 2, (165) for Class 3 and (166) for Class 4.

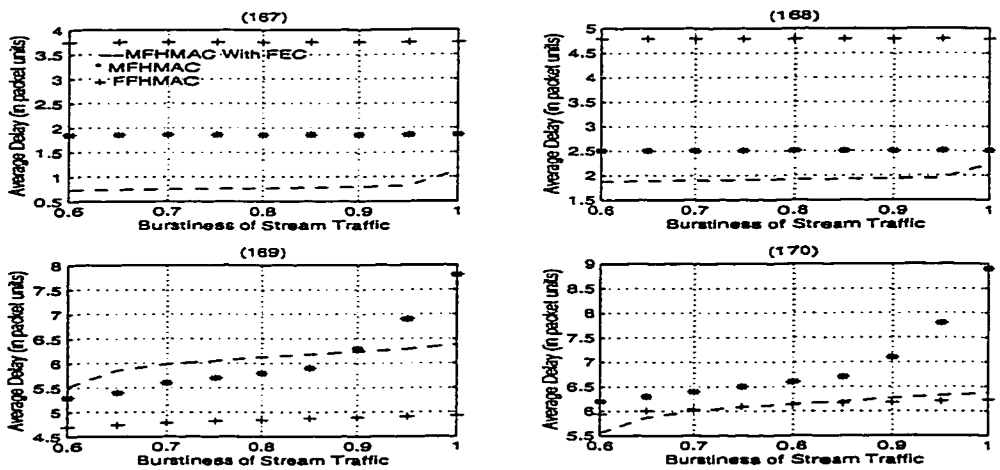


Fig.3.80. Comparison of each class average delay in MFHMAC and FFHMAC vs. stream traffic burstiness: (167) for class 1, (168) for class 2, (169) for class 3 and (170) for class 4.

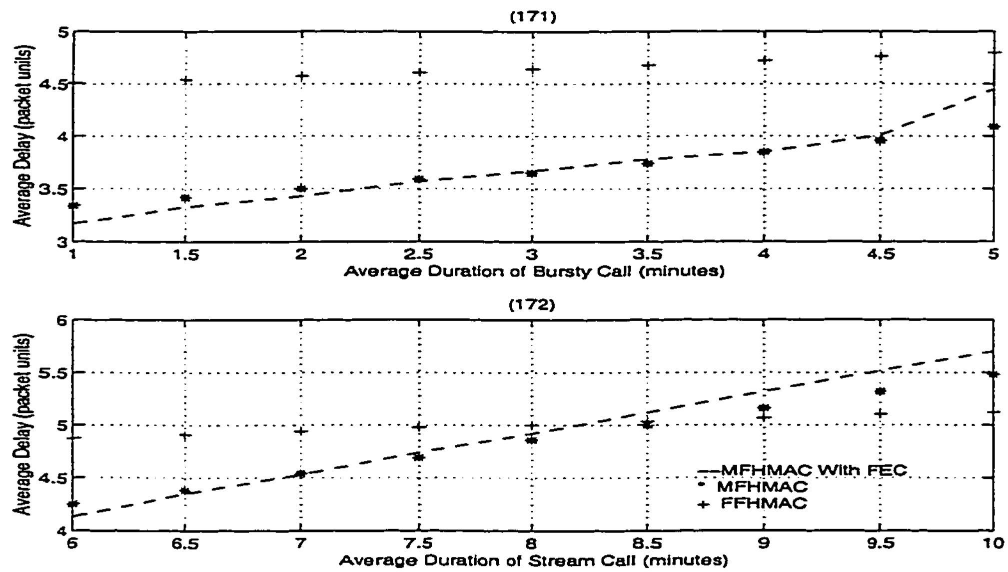


Fig. 3.81: Comparison of system average delay in MFHMAC and FFHMAC vs. call duration: (171) for short calls and (172) for long calls.

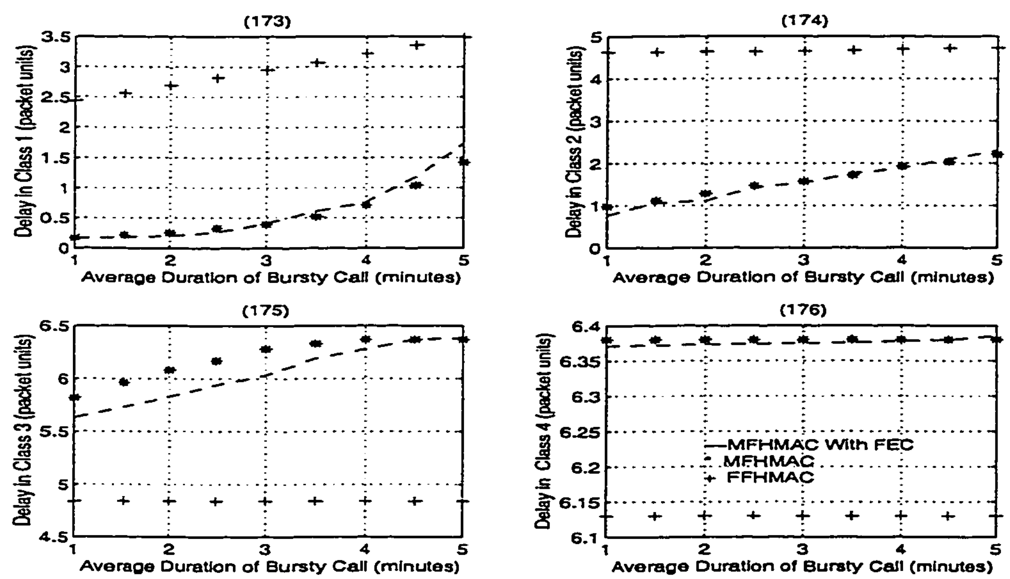


Fig.3.82. Comparison of each class average delay in MFHMAC and FFHMAC vs. short call : (173) for Class 1, (114) for Class 2, (175) for Class 3 and (176) for Class 4.

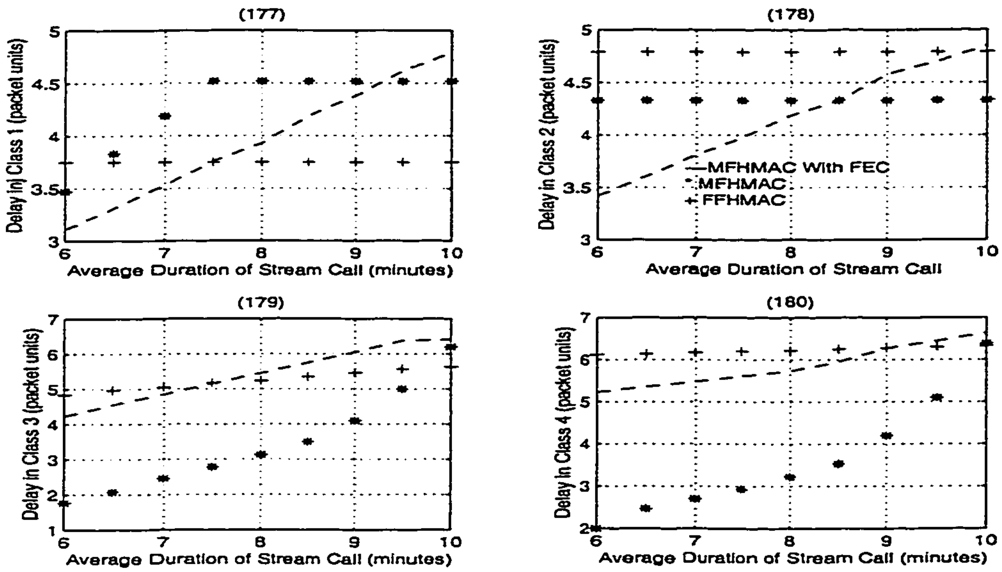


Fig. 3.83: Comparison of each class average delay in MFHMAC and FFHMAC vs. long call duration: (177) for Class 1, (178) for Class 2, (179) for Class 3 and (180) for Class 4.

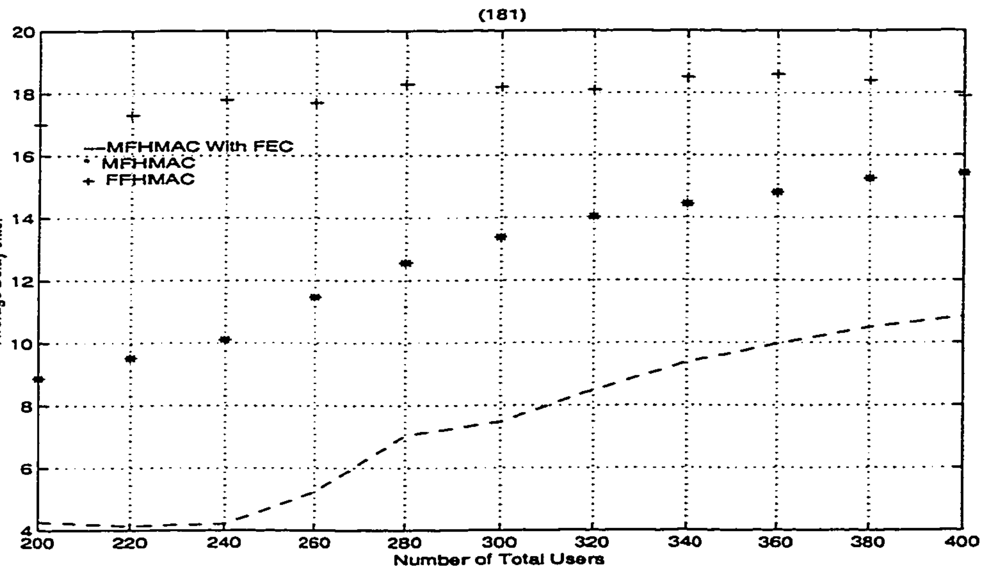


Fig.3.84. Comparison of system delay jitter in MFHMAC and FFHMAC vs. system users.

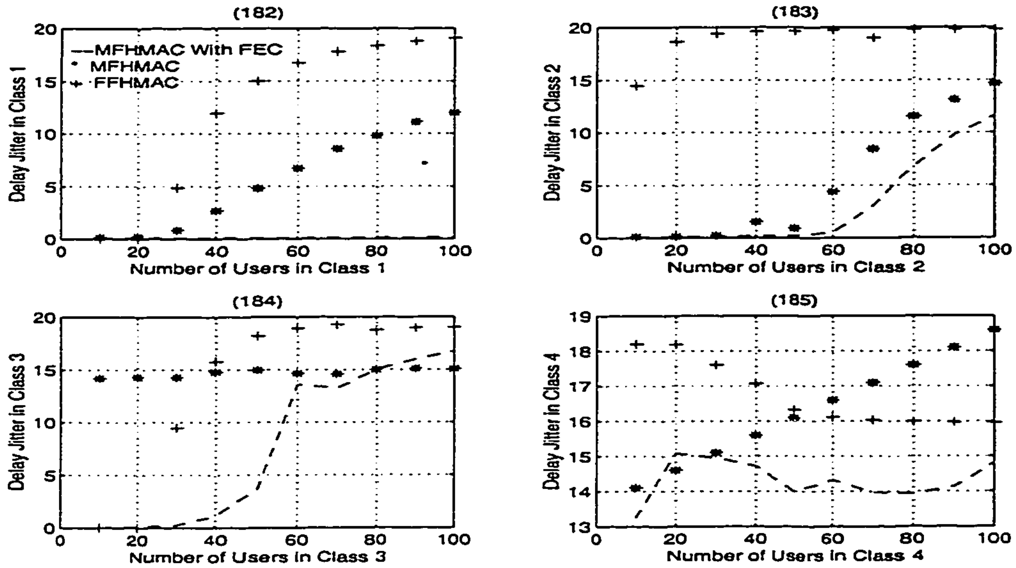


Fig. 3.85: Comparison of each class delay jitter in MFHMACV and FFHMAC vs. each class users: (182) for Class 1, (183) for Class 2, (184) for Class 3 and (185) for Class 4.

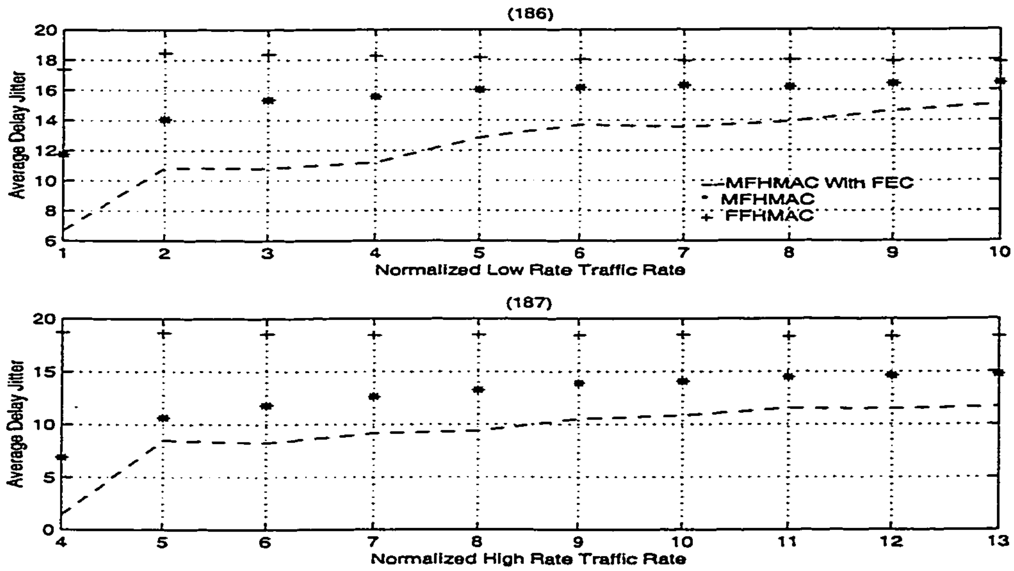


Fig.3.86. Comparison of system delay jitter in MFHMAC and FFHMAC vs. traffic rate: (186) for low rate traffic and (187) for high rate traffic.

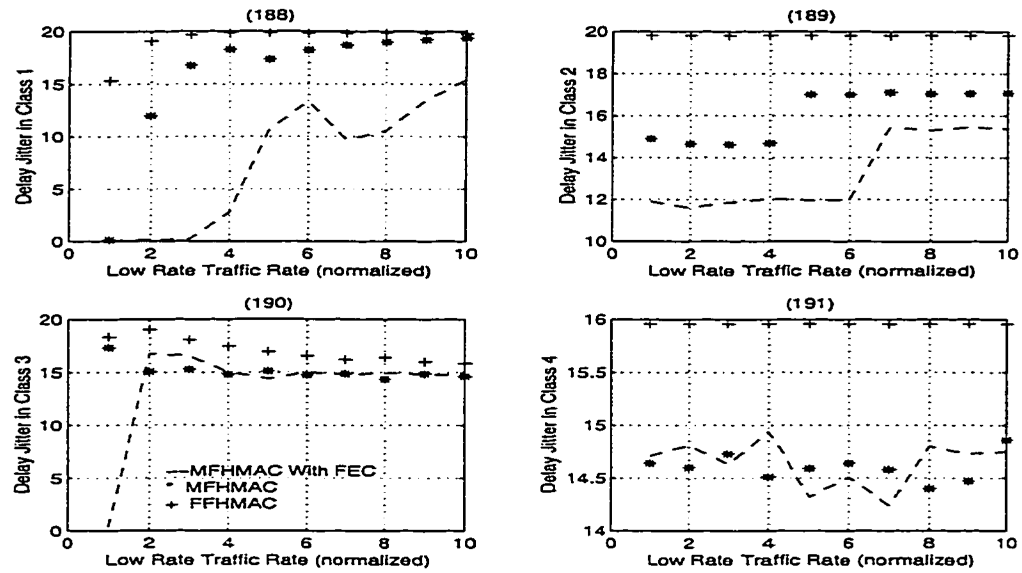


Fig. 3.87: Comparison of each class delay jitter in MFHMAC and FFHAMC vs. low rate traffic: (1988) for Class 1, (189) for Class 2, (190) for Class 3 and (191) for Class 4.

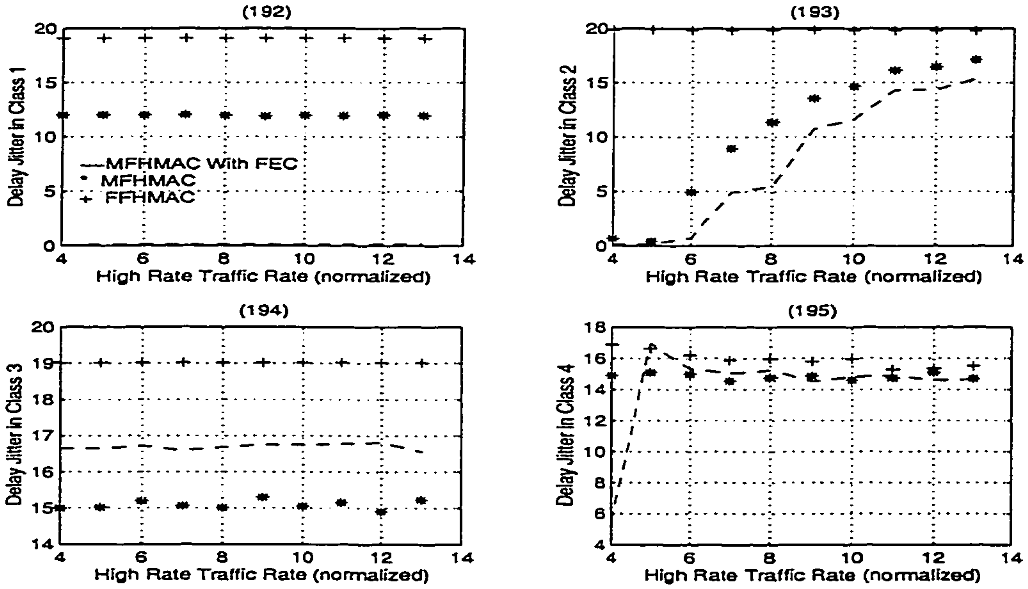


Fig.3.88. Comparison of each class delay jitter in MFHMAC and FFHMAC vs. high rate traffic: (192) for class 1, (193) for class 2, (194) for class 3 and (195) for class 4.

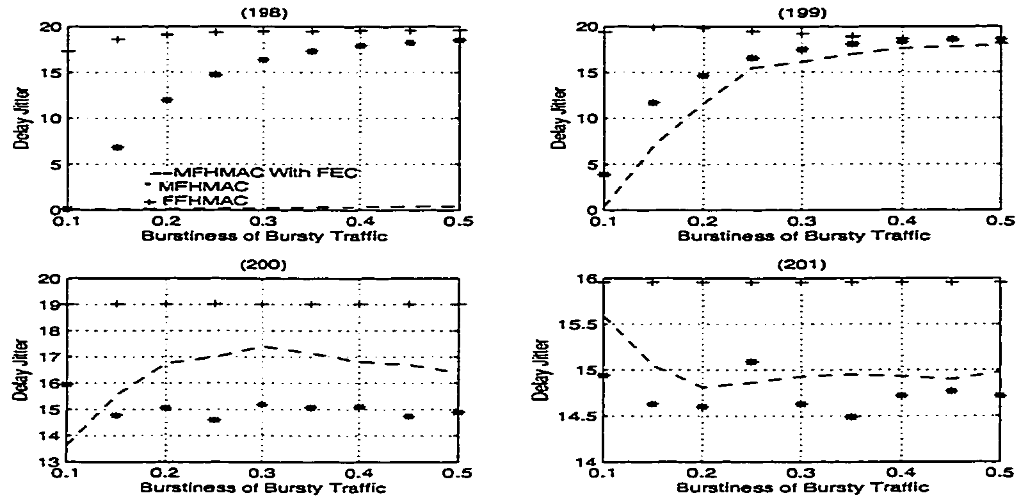


Fig. 3.89: Comparison of system delay jitter in MFHMAC and FFHMAC vs. traffic burstiness; (196) for bursty traffic (197) for stream traffic.

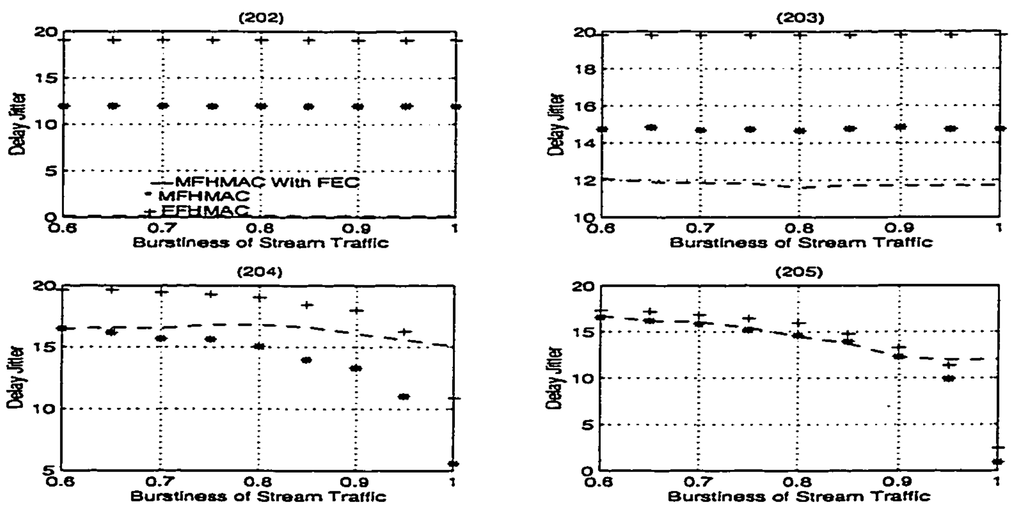


Fig.3.90. Comparison of each class delay jitter in MFHMAC and FFHMAC vs. bursty traffic burstiness: (198) for Class 1, (199) for Class 2, (200) for Class 3 and (201) for class 4.

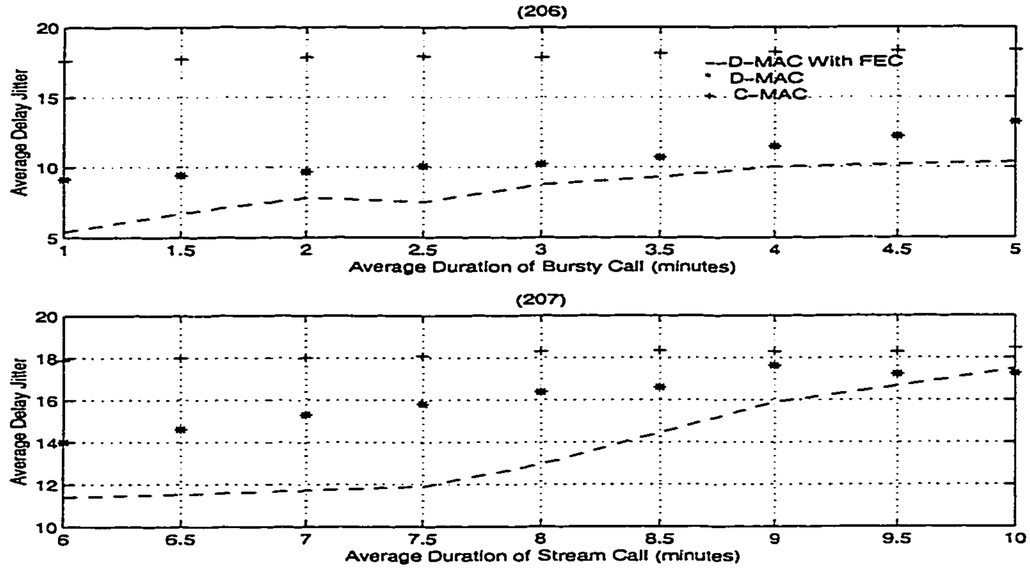


Fig. 3.91: Comparison of each class delay jitter in MFHMAC and FFHMAC vs. stream traffic burstiness: (202) for Class 1, (203) for Class 2 , (204) for Class 3 and (205) for Class 4.

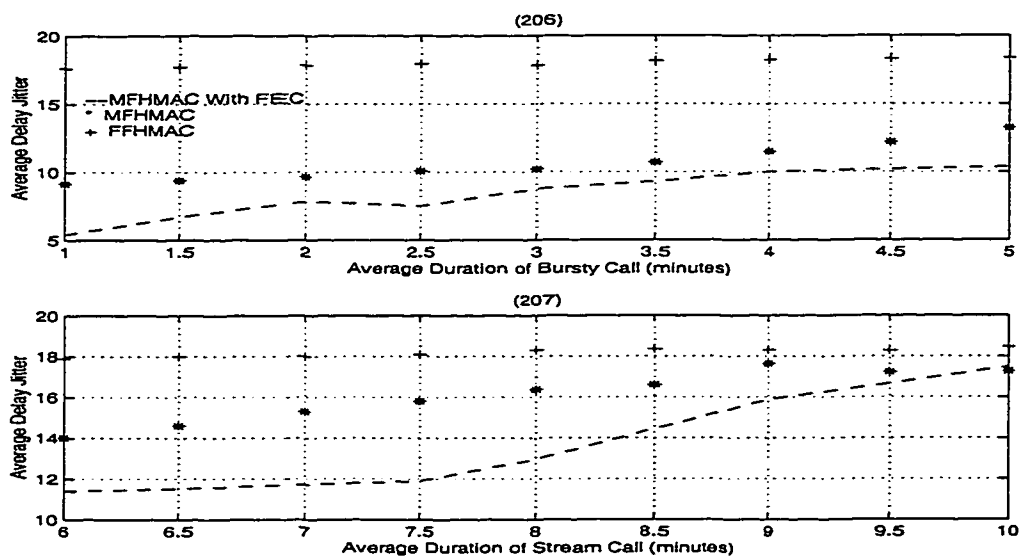


Fig.3.92. Comparison of system delay jitter in MFHMAC and FFHMAC vs. call duration. (206) for short call, (207) for long call.

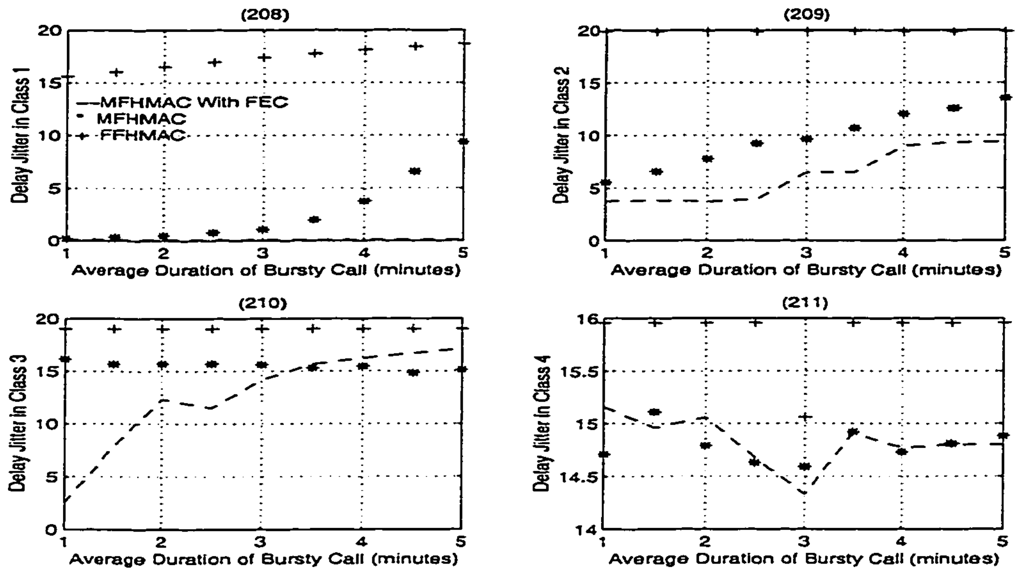


Fig.3.93: Comparison of each class delay jitter in MFHMAC and FFHMAC vs. short call duration. (208) for class1, (209) for class2, (210) for class3, and (211) for class4.

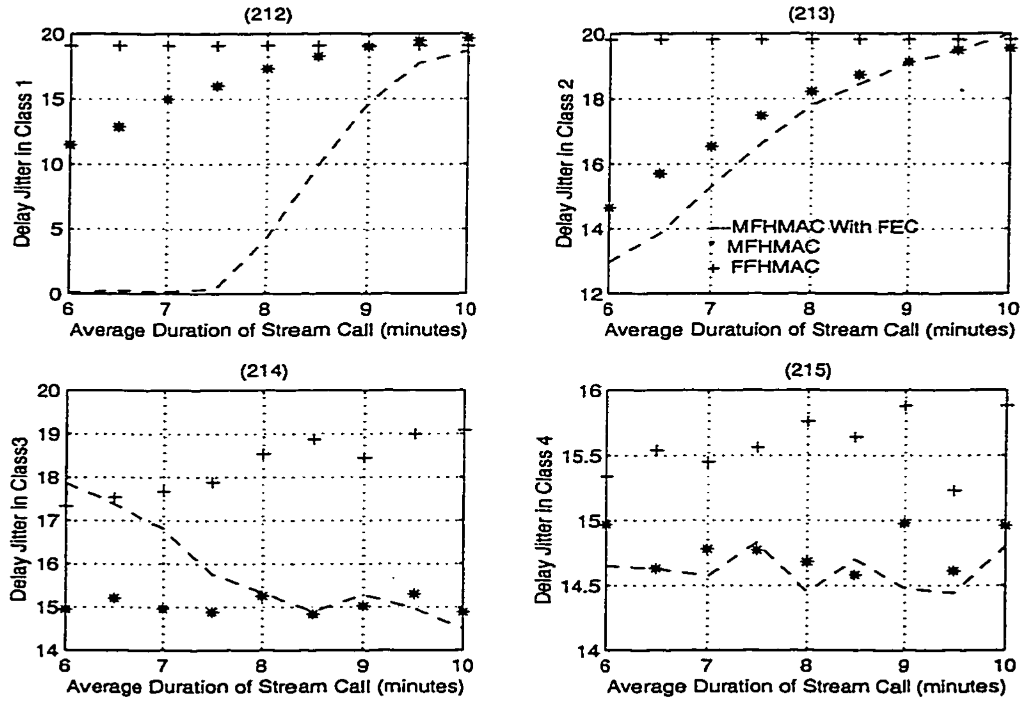


Fig.3.94: Comparison of each class delay jitter in MFHMAC and FFHMAC vs. long call duration. (212) for class1, (213) for class2, (214) for class3, (215) for class4.

CHAPTER 4

SUMMARY AND CONCLUSIONS

In Chapter 1, the basic elements of the integrated service multi-rate multimedia wireless network were introduced. The main approach and basic requirements for the wireless network design were discussed.

In Chapter 2, a short review of conventional multiple access protocol was presented. A comparison of some MAC techniques used in different type of networks were also listed.

In the first part of Chapter 3, the concept of radio software is presented. In turn a multi-MAC protocol for wireless software radios which named Moveable Frame Hybrid Multiple Access Control (MFHMAC) was introduced. The basic idea of a "moveable frame" in this technique is that each user in a certain cell can be assigned a flexible channel to transmit his information packet. This is based on the each user's unique traffic characteristics and the access processing will follow the MFHMAC policy. The goal of this technique is the efficient use of the limited radio source and improving the link quality of service.

From simulation results of probability of call blocking, we noted that class 4 users have high-test call blocking probability due to the have traffic load (stream and high rate). Class 2 users have lowest call blocking probability due to the light traffic (bursty) and the first priority of service. Results showed that even only one class users increase, all other classes' call blocking are also increased. These are coursed by the flexible adjustment bandwidth allocations. One important factor is that when traffic load increase, probability of call blocking didn't change too much. This means our techniques are robust for any type of traffic. However, we also noted that when the low rate users' (class 1 and class 3) arriving packet rate increase, the probability of call blocking in high rate class (4) almost constant due to the high loading existence in basic and reservation FDMA band. The effects of burstiness θ and average call duration Td are showed very similarly.

From the delay observation, we noted that delay in each class have same characteristic as call blocking scenarios. One class users increase, while all other classes' users fixed, average packet delay in each class is also increased as well. However, low rate users' (class 1 and class 3) packet arriving rate (δ_1 and δ_3) increase, high rate users' (class 2 and class 4) average delay are just changed a little, vice versa. The effects of traffic burstiness and call duration have the same characteristics as traffic rates.

Interesting observation is that with the traffic load increasing the system throughput increase as well, but after reach a certain value, the throughput will decrease , this was coursed by the collision retransmissions from CSMA band. We noted that class 1 and class 3 users have higher throughput due to the high utilization of CDMA and TDMA techniques. Eventually, class 1 and class 3 users' throughput can reach 1. Class 2 and class 4 users may access radio channels by CSMA band, so, after reach maximum throughput, with the high collisions the throughput are decreasing. This also course the average throughput in the whole system perform the same property. Other conditions which represent the traffic characteristics have the same effects as traffic load.

We noted from that the jump of delay jitter for a small changing is number of users. Class 4 users don't have much variations in delay due to every packet undergo longer delay. Class 2 and Class 3 users have some jump points due to the channel adjustment: with light traffic loading, class 2 and class 3 users with the high priority to access basic or reservation band without longer queuing. After the traffic build up, the class 2 and class 3 users are also undergo longer queuing delay even with high priority. Class 1 users always carrying light traffic, so class 1 users have very lower delay jitter. Other effects such as δ, θ and Td are same as the effects in the evaluation of call blocking, delay and throughput.

By observation of probability of buffer overflow, we noted that the probability of buffer overflow in each class almost perform same as the probability of call blocking in each class. This due to the use of finite buffer in each class.

The simulation results showed that in the MFHMAC system bursty traffic (Class 1 and Class 2 users) will have better performance than the stream traffic (Class 3 and Class 4 users), even Class 2 traffic has a higher rate than Class 3 traffic. It is well known that when the number of users becomes large, the quality of service in the system will decrease due to the Multiple Access Interference (MAI). From simulation it can be seen that when the number of users in one class becomes large, the performance of other classes will also be affected. However, when the number of users with a low rate character (Class 1 and Class 3 traffic) increase, it didn't affect the system performance too much.

The other components, such as traffic rate, traffic bursty level and each caller's call duration (all of them can characterize the traffic in different types) also effect the system performance as well.

In the second part of Chapter 3, the effects of the FEC coding in MFHMAC systems was considered through simulations. Like DS-CDMA, FH-CMA and other CDMA network systems [4, 12, 15], FEC also played good in MFHMAC system due to the channel error correlation capability.

At the end of Chapter 3, some comparisons and comments were made. By these comparisons we can conclude that the MFHMAC is a more suitable access technique for packet switched connection-oriented wireless network, especially when the majority of traffic in the system is bursty data traffic. This is quite one feature of the third generation communicants.

Suggestions for Future Works:

- In our work we didn't derive the Markov Model of the MFHMAC scheme. So as a possible continuation of this work the performance of MFHMAC can be examined by a mathematical model and the results can be compared with a simulation model.
- The hybrid access techniques can be deployed by choosing some other more suitable combination of other schemes such as PRMA or Mascra, etc.

REFERENCES

[1] R. Pichna and Q. Wang, "A Medium-Access Control Protocol for a Cellular Packet CDMA Carrying Multirate Traffic," *IEEE Journal on Selected Areas in Comm.*, Vol. 14, No. 9, pp. 1728-1736, December 1996.

[2] C. Romans and J. Tourrilhes, "A Medium Access Protocol for Wireless LANS Which Supports Isochronous and Asynchronous Traffic," Hewlett-Packard Company, February 1998.

[3] D. Agyagari and A. Ephremides, "Admission Control with Priorities: Approaches for Multi-rate Wireless Systems," *Mobile Networks and Applications*, 4 (1999) 209-218.

[4] A.K. Elhakeem, R.D. Girolamo, I.B. Bolira and M. Talla, "Delay and Throughput Characteristics of TH, CDMA, TDMA and Hybrid Networks for Multipath Feded Data Transmission Channels," *IEEE Journal on Selected Areas in Comm.*, Vol. 12, No. 4, pp. 622-637, May 1994.

[5] T.K. Apostolopoulos and E.N. Protonotarios, "Queuing Analysis of Buffered CSMA/CO Protocols," *IEEE Transactions on Communications*, Vol. COM-34, No. 9, pp. 898-905, September 1986.

[6] K.S. Gilhousen, I.M. Jacobs, R. Padovani, A.J. Viterbi, L.A. Weaver Jr., and C.E#. Wheatley, "On the Capacity of a Cellular CDMA System," *IEEE Transactions on Vehicular Technology*, Vol. 40, No. 2, pp. 303-312, May 1991.

[7] M. Schwartz, *Broadband Integrated Networks*. Prentice-Hall PTR, 1996.

[8] T.S. Rappaport, *Wireless Communications Principles and Practice*. Prentice-Hall, 1996.

[9] T.N. Saadawi, M. Ammor and A.K. Elhakeem, *Fundamentals of Telecommunication Networks*. New York: Wiley, 1994.

[10] H.P. Stern. "Design Issues Relevant to Developing an Integrated Voice/Data Mobile Radio System," *IEEE Transactions on Vehicular Technology*, Vol. 39, No. 4, pp. 281-292, November 1990.

[11] K. Zhang and K. Pahlavan, "An Integrated Voice/Data System for Mobile Indoor Radio Networks," *IEEE Transactions on Vehicular Technology*, Vol. 39, No. 1, pp. 75-82, February 1990.

[12] A.E. Brand and A.H. Aghvami, "Performance of a Joint CDMA/PRMA Protocol for Mixed Voice/data Transmission for Third Generation Mobile Communication," *IEEE Journal of Selected Areas in Communications*, Vol. 14, No. 9, pp. 1698-1707, December 1996.

[13] L. Tan and Q.T. Zhang, "A Reservation Random-Access Protocol for Voice/Data Integrated Spread-Spectrum Multiple-Access Systems," *IEEE Journal on Selected Areas in Communications*, Vol. 14, No. 9, pp. 1717-1727, December 1996.

[14] A. Demers, S. Shenker and L. Zhong. "MACAW: A Media Access Protocol for Wireless LANS," SIGCOMM 94-8/94 London, England, U.K.

[15] C.J. Chang, B.W. Chen, T.Y. Liu and F.C. Ren, "Fuzzy/Neural Congestion Control for Integrated Voice and Data DS-CDMA/FRMA Cellular Networks,"

IEEE Journal on Selected Areas in Communications, Vol. 18, No. 2, pp. 283-293, February 2000.

[16] G. Bianchi and A.T. Campbell, "A Programmable MAC Framework for Utility-Based Adaptive Quality of Service Support," *IEEE Journal on Selected Areas in Communications*, Vol. 18, No. 2, pp. 244-255, February 2000.

[17] I.E. Kassotakis, M.E. Markaki and A.V. Vasilakos, "A Hybrid Genetic Approach for Channel Reuse in Multiple Access Telecommunication Networks," *IEEE Journal of Selected Areas in Communications*, Vol. 18, No. 2, pp. 234-242, February 2000.

[18] H. Tong and T.X. Brown, "Adaptive Call Admission Control Under Quality of Service Constraints: A Reinforcement Learning solution," *IEEE Journal on Selected Areas in Communication*, Vol. 18, No. 2, pp. 209-221, February 2000.

[19] F. Davoli and P. Margni, "A Two-Level Stochastic Approximation for Admission Control and Bandwidth Allocation," *IEEE Journal on Selected Areas in Communication*, Vol. 18, No. 2, pp. 222-232 February 2000.

[20] *IEEE Journal on Selected Areas in Communications*, Special Issue, February 2000. Guest Editorial.

[21] Z. Dziong, M. Jia and P. Mermelstein, "Adaptive Traffic Admission for Integrated Services in CDMA Wireless-Access Networks," *IEEE Journal on Selected Areas in Communications*, Vol. 14, No. 9, pp. 1737-1747, December 1996.

[22] J.W. and R. Kohno, "Performance Evaluation of Wireless Multimedia CDMA Networks Using Adaptive Transmission Control," *IEEE Journal on Selected Areas in Communications,* Vol. 14, No. 9, pp. 1688-1697, December 1996.

[23] P. Mermelstein, A. Jalali, and H. Leib, "Integrated Services on Wireless Multiple Access Networks," in Proc., IEEE Int. Conf. Commun. ...'93 Geneva Switzerland, May 1993, pp. 863-867.

[24] K. Pahlavan, "Wireless Intro-office networks," ACM Trans. OIS, Jan. 1998.

[25] K. Pahlavan, "Wireless Office Information Network," *IEEE Commun. Soc. Mag.*, June 1985.

[26] D.C. Cox, H.W. Arnold and P.T. Porter, "Universal Digital Portable Communications: A System Perspective," *IEEE Journal on Selected Areas in Communications*, June 1987.

[27] T.X. Brown, "Adaptive Access Control Applied to Ethernet Data," in Advance in N2PS, M. Mozer, M. Jordan and T. Petsche, Eds. Cambridge, MA: MIT Press, 1997, pp. 932-938.

[28] A. Hiramatsu, "ATM Communications Network Control by Neural Networks," *IEEE Trans. Neural Networks*, Vol. 1, No. 1, pp. 122-130, 1990.

[29] S. Verma, R.K. Pankaj and A. Leon-Garcia, "Call Admission and Resource Reservation for Guaranteed Quality of Services in the Internet," *Computer Commun.*, Vol. 21, pp. 360-374, Apr. 1998.

[30] T.X. Brown, "Adaptive Statistical Multiplexing for Broadband Communications," in Proc. Fifth IFIOP Workshop Performance Modeling and Evaluation of ATM Networks, Ilkley, U.K. July 1997.

[31] S. Lin, D.J. Costello and M.L. Miller, "Automatic-Repeat Request Error-control Schemes," *IEEE Communications Magazine,* Vol. 22, No. 12, pp. 5-17, 1984.

[32] H.S. Chin, J.W. Goodge, R. Griffiths and D.J. Parish, "Statistics of Video Signals for Viewpoint type Applications," *IEEE Journal on Selected Areas in Communications*, pp. 826-832, 1989.

[33] " IEEE Journal on Selected Areas in Communications, April 99" special issue.

[34] J. Mitola, "Software Radio Architecture: A mathematical perspective," *IEEE Journal on Selected Areas in Communications*, April 1999.

[35] J. E. Gunn, K. S. Barron, W. Ruczczyk, "A Low-Power DSP Core-Based Software Radio Architecture," *IEEE Journal on Selected Areas in Communications*, April 1999.