

Integrated Multiple Access Protocols for Personal/Mobile Communication Networks Supporting Multimedia Applications

by

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Abstract

The success of personal communication networks will depend on their ability to accommodate the diverse traffic which will be generated from the numerous and diverse future multimedia applications and services. Towards this end, multimedia Medium access control (MAC) algorithms capable of accommodating this integrated traffic will play a major role. In this thesis, novel MAC protocols are proposed and their ability to efficiently utilize the available bandwidth is evaluated.

The proposed multimedia MAC protocols are based on random access algorithms and reservation policies using TDM and CDMA technologies. The MAC protocols also support multiple-priority mechanisms: an essential feature in communication systems supporting integrated services with diverse Quality of Service (QoS) requirements. One of the major characteristics of the priority mechanism is its simplicity of implementation. This adds to the practical value of the proposed MAC protocols. The TDMA and the CDMA versions of the protocols are evaluated for voice and data traffic and compared to other MACs proposed in the literature. The CDMA version of the protocol is also investigated for data, voice and video with multi-slot reservation for video. Network configurations servicing CBR (variable quality) and VBR (constant quality) video users are evaluated. The results verified the superiority of the proposed protocol: high channel utilization and ability to service effectively high volumes of integrated traffic. The new MACs are found to be stable, robust and easy to implement multimedia services in personal communication networks. The simplicity and high performance level of the proposed MACs makes it valuable for supporting integrated traffic wireless applications.

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Acronyms

ABR	Available Bit Rate
ARQ	Automatic Repeat Request
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband ISDN
BER	Bit Error Rate
C	Collision
CAC	Connection Admission Control
CB	Class Blocking
CBR	Constant Bit Rate
CCITT	Consultative Committee on International Telegraphy & Telephony
CDMA	Code Division Multiple Access
CIF	Common Intermediate Format
Cl-A	Class A
Cl-B	Class B
CRA	Collision Resolution Algorithm
CRC	Cyclic redundancy check
CRI	Collision Resolution Interval
CRP	Collision Resolution Process
DCB	Double Class Blocking
D_L	Downlink Channel
FC	Feedback Channel
FDMA	Frequency Division Multiple Access
FEC	Forward error correction

FH/SS	Frequency Hoping Spread Spectrum
FPLMTS	Future Public Land Mobile Telecommunications Services
GFC	Generic Flow Control
GOP	Group of Pictures
GSM	Global System Mobile
HDTV	High Definition Television
HEC	Header Error Control
ISDN	Integrated Service Digital Network
ISO	International Standards Organization
ITU	International Telecommunications Union
LAN	Local Area Network
MPEG	Moving Picture Expert Group
MPCS	Mobile and Personal Communications Service
MPCN	Mobile and Personal Communications Networks
MSAT	Mobile SATellite
NC	No Collision
PCN	Personal Communication Network
PCS	Personal Communication Services
PCM	Pulse Code Modulation
PTI	Payload Type Identifier
PMCN	Personal Mobile Communication Network
PSTN	Public Switch Telephone Network
PN	Public Network
PRMA	Packet Reservation Multiple Access
QCIF	Quarter CIF
QOS	Quality of Service
RMA	Reserved Multiple Access

SNR	Signal to Noise Ratio
SSA	Spread Slotted Aloha
SSRA	Spread Slotted Random Access
SA	Slotted Aloha
SF	System Frame
SFP	System Frame Period
SP	Slot Period
TC	Transmission Cell
TDMA	Time Division Multiple Access
TDM	Time Division Multiplexing
U_L	Uplink Channel
VAD	Voice Activity Detector
VBR	Variable Bit Rate
VF	Video Frame
VFP	Video Frame Period
VFS	Video Frame Speed
WC	Waiting Cell
WPBX	Wireless Public Branch Exchange
WLAN	Wireless Local Area Network

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Chapter 1

Introduction

The telecommunications industry is undergoing a fundamental revolution. Since the introduction of the cellular telephone over a decade ago, the growth of wireless communications has been rapid [Padg95]. A host of new communications systems, under the general heading of personal communications services (PCS), have recently been proposed. These systems are known by a number of different names, including Personal Communications Network (PCN), Future Public Land Mobile Telecommunications Services (FPLMTS), cellular phone, Wireless Public Branch Exchange (WPBX), Wireless Local Area Networks (WLANs), satellite personal communications and so on [Kobb93].

1.1 Background

Many emerging applications require the transmission of high bandwidth multimedia traffic, with various Quality of Service (QoS) requirements. Broadband ISDN (using the Asynchronous Transfer Mode switching technique, mostly known through its abbreviation -ATM-) is being developed to support these multimedia applications. Many of these new applications might require to support mobile users. Consequently, wireless networks

should be able to support multimedia communications and interface effectively with the ATM/BISDN network. The emergence of PCS/PCN and the proliferation of multimedia applications, on one hand, and the standardization of ATM as the high-speed multimedia transmission protocol, on the other hand, has created many debates as to whether a mobile, multi-user (multiple access) channel can support the same types of services as ATM does in a point to point link environment. In addition, PCNs should be compatible in design to these fixed communication networks.

First-generation wireless networks (supporting primarily voice communications) have sprung up worldwide. Examples are the North-American AMPS, IS-54 standards and the European GSM standard [Padg95], [Falc95]. Second-generation digital voice/data networks are under development, with some networks being deployed and undergoing trials in several areas. Third-generation networks designed ultimately to carry multimedia traffic- whether voice, video, images, files, data, or combinations of these - are under intensive study by the standardization organizations, such as International Telecommunications Union (ITU-R). Most of the research in the area of wireless communications reported in the open literature has tended to focus on physical layer issues such as TDMA versus CDMA and multiple access protocols for the wireless environment and on higher layer control issues such as call hand-off, hierarchical cell design, and dynamic allocation of channels among cells for the voice-call environment. [Rand94].

There are many issues and challenges to be addressed in order to support multimedia services in the mobile multi-user channel. The challenges that must be faced span a wide range of considerations and technical expertise. These include the architecture of the communications and information service infrastructure (base stations, network protocols, servers) necessary to support mobile communications, the demonstration of new and innovative infrastructure elements and services; the development of new design methods for power-sensitive, mobile, real-time, network-attached computer systems; the integration of wireless communications into the emerging national information

infrastructure. The wireless link issues include i) the efficient encoding of the signal to minimize the bit-error rate in the presence of severe channel impairments, such as interference, fading, shadowing and ii) the efficient use of the frequency spectrum. In order to extend emerging multimedia services over a mobile, multi-user channel, there is a need to include error detection/correction and multiple access mechanisms. The mobile channel tends to be noisy with erratic behavior due to fading, causing errors to occur. The bit error rate of a multi-user channel can be improved by appropriate use of forward error correction (FEC), automatic repeat request (ARQ), multi-user detection and other techniques. The channel bandwidth in the wireless is shared by many mobile users.

1.2 Motivation

In this thesis we focus on multiple access protocols designed to use in a multimedia environment. Among many of the challenges faced by the PCNs is the channel access problem. The broadcast nature of the radio channel requires the introduction of a medium access control (MAC) layer in order to co-ordinate the transmissions of the nodes which compose the network. Absence of a MAC to facilitate the sharing of channel resources among a set of geographically dispersed users with different quality of services (QoS) requirements is a serious difficulty in a mobile channel. The MAC protocol is a key design issue for reliable delivery of user data over a shared wireless channel. The MAC layer should support the flexibility of varying the assigned bandwidth to each connection according to the instantaneous requirements and without violating other connections allocated capacity. The MAC protocol should provide maximum utilization of the available radio spectrum and at the same time guarantee satisfactory transmission performance while taking into account the harsh radio environments. The seriousness of the MAC protocol in wireless applications has been addressed by the IEEE study group 802.11, which was

formed to establish a recommended international standard for WLANs that will support high bandwidth wireless data networking [Hars95].

In the past, numerous MAC protocols (e.g., ALOHA, TDMA, CDMA) have been analyzed, [Norm94]. However, these protocols were developed to handle one traffic type (e.g., data or voice). Recently proposed protocols for integrated traffic combine many features of the ALOHA and TDMA protocols; however, they do not address the issue of providing bandwidth on demand [Mitr93], [Weis95]. To support all ATM services (e.g., Constant Bit Rate (CBR), Variable Bit Rate (VBR), and Available Bit Rate (ABR)) to end users, a MAC protocol must be able to provide bandwidth on demand. VBR services bandwidth requirements vary over the duration of the connection; therefore a MAC protocol has to be able to assign bandwidth in a dynamic manner. For example, in a CDMA paradigm it is suggested that more than one code may be assigned to a single user. This, of course, increases the complexity of the system.

1.3 Objectives

This work deals with the design and evaluation of Medium Access Control (MAC) protocols, for PCNs carrying multimedia traffic. We propose two different types of MAC protocols. The first, (described in chapter 3) is based on TDM technology. The second (described in chapters 4 and 5), is based on CDMA, the spread spectrum technology. The protocols make efficient use of the bandwidth by using reservation schemes for delay sensitive, real time, constant and variable bit rate traffic, such as voice and video. We use some stable random multiple access algorithms for getting reservation and for transmission of data packets.

Both protocols are designed on the basis of a slotted structure. The channel consists of slotted time division frames, as illustrated in Figure 1.1. Acknowledging the importance of compatibility with ATM and the need for interconnectivity with the BISDN, we

packetize the information (i.e., voice, video or data) into ATM like packets (cells) with necessary overhead. Each packet fits into one slot.

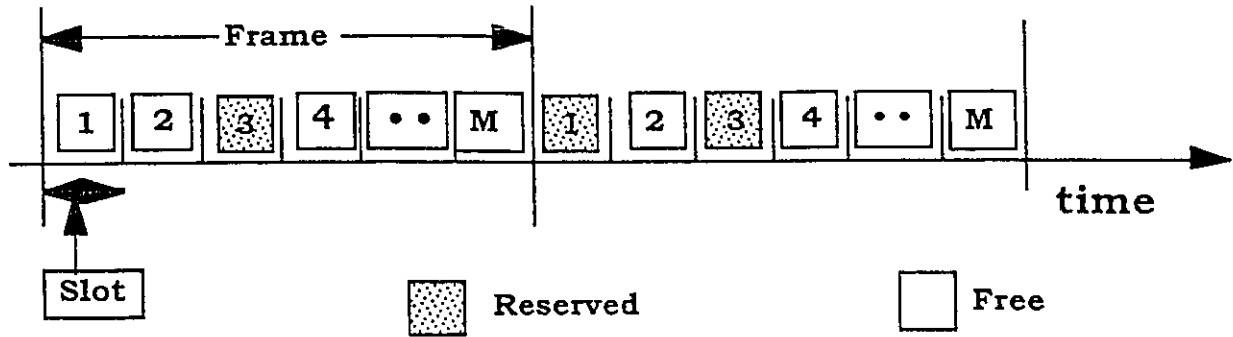


Figure 1.1 Slotted Channel Structure

In order to evaluate the performance of the proposed multiple access protocols, we developed and used simulation models based on the OPNET software-package [OPNT93]. It is worth of mentioning that when it comes to video, our simulations use real video sequences traces according to the H.261 video standard [H261]. Recently, several studies concluded that video traffic shows strong self-similar (fractal) behavior, which can not be captured by the various Poisson and Markovian models that were proposed in the past. By using actual video stream sequences in our simulations, we avoid the problem of evaluating our protocols using inaccurate traffic models.

1.4 Thesis Organization

This thesis is composed of six chapters. Following this introduction, chapter 2 gives the brief description of the cellular network architecture, the basics of TDMA/CDMA, 2-C collision resolution protocol, and the traffic source models used in our simulations.

Chapter 3 describes in detail the TDMA version of the proposed protocol, including the description of the used reservation scheme. General description of the random multiple access algorithms for multimedia traffic is followed by the simulation results obtained for integrated voice and data traffic. The performance of this protocol is also compared with that of the Packet Reservation Multiple Access (PRMA) protocol [Good89].

Chapter 4 provides a more detailed look at the CDMA protocol versions. Discussion of priority mechanisms for Spread Slotted Aloha (SSA) and Spread Slotted Random Access (SSRA) is followed by simulation analysis results for integrated voice and data traffic.

Chapter 5 describes the integration of video in the MAC protocol based on CDMA technology. Statistics from real time H.261 video sequences are used in the study. This chapter also explores the multi-slot reservation scheme and gives the general logic for its implementation. The overall conclusion and contributions of this work are summarized in Chapter 6.

Chapter 2

Wireless Network Technology and Traffic Models

2.1. Introduction

In the past several years, there has been a significant growth in the Mobile and Personal Communications Service (MPCS). Cellular radio is the fastest growing area of the communications industry today. The paging service has proven successful as well, while new services such as mobile-satellite and aeronautical communications have started to penetrate the market. In the future, the growth rate of the personal communications market is expected to increase further. This expectation of growth can be justified by the increased importance of mobility in our every day professional and personal activities, the development of new, more powerful and suitable equipment and the introduction of new services in the MPCS. New technology, such as wireless facsimile, hand-held telephones/devices with touch-sensitive displays and user interfaces, active badges etc. will appear in the market. There is also the general belief that the coming generations of enhanced Mobile and Personal Communication Networks (MPCNs) should be compatible, interconnect with the broadband ISDN (BISDN) and they should be able to support multimedia services, at least up to a certain degree (voice, low to medium bit rate video, text and image etc.).

2.2 Network Architecture

Different multimedia services span a very wide range of frequency allocations, expected data bandwidths, and assumptions about user mobility and location. For example, a high-mobility vehicular terminal (80 m/s) presents very different engineering challenges than one designed for medium (20 to 40 m/s) or low mobility (less than 10 m/s). In addition, systems designed for outside use must deal with fundamentally different constraints than those for indoor use, where wider bandwidth allocations are made possible through the use of low power signals. It appears that any wireless communications system is likely to be a hybrid, with high-bandwidth "islands" within buildings (using micro-cell, pico-cell architecture) and lower-bandwidth in-between spaces [LinC93].

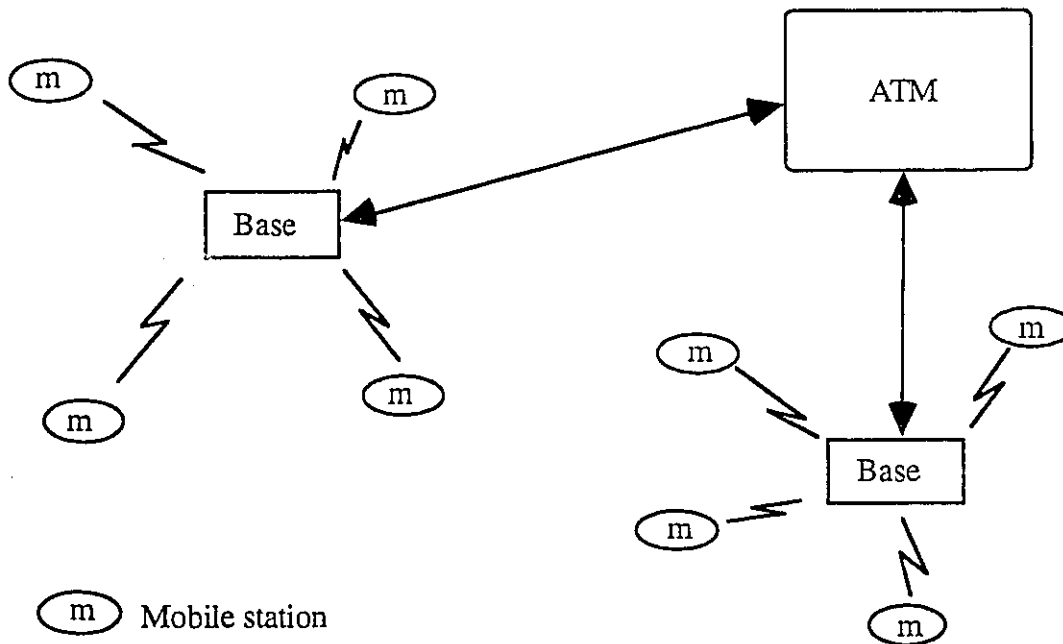


Fig. 2.1 Elements of the network

The basic elements of the network are shown in figure 2.1. The network consists of a number of cells, each one covering a specific area. Each cell has a base station (BS), which provides the radio interface with the wireless users as well as the network interface between the traffic generated within the cell and the public network (PN). Within the cell, there are two streams of traffic, one generated from the wireless users with the destination being the BS (up-link; from now on it will be abbreviated as U-L) and a second stream, starting from the BS and going towards the mobile and pedestrian users that are located within the cell communicating with the base station (down link; from now on it will be abbreviated as D-L).

The U-L consists of N_c different frequency channels or codes. The frequency assigned to these N_c channels is done with the objective to reduce the co-channel and adjacent channel interference between cells and between the channels themselves. The individual slots (equal sized) of each U-L channel are grouped in frames, each frame consisting of N_f consecutive slots. Slots can be marked by the system as "reserved" or "unreserved". D-L channel allocation is simple, as it is centrally controlled by the base station. In our analysis we would focus only on the channel access problem for the U-L.

Similarly, the slots of the D-L channels are grouped in frames. The size of the U-L and D-L frames depend on the bit rates of the channels, and characteristics of the users and the multimedia traffic carried through the cell.

2.3 Channel Allocation Schemes

In many communication networks it is difficult if not impossible to build permanent point to point channels [Stal85]. To make it feasible these transmission channels should be shared by many users. The U-L channel described in section 2.2 is shared channel. The main challenge in shared channels is that here the transmissions from different users interfere with each other, i.e., one transmission coinciding in time with another over the

same channel may cause none of them to be received. To make a successful transmission interference must be avoided or at least controlled. The channels thus become a shared resource whose allocation is critical for the proper operation of the network. The access schemes to allocate such channels are known as multiple access protocols.

Numerous multiple access protocols are suggested and there are various ways to classify them. As classified in [Raph90] we can broadly classify multiple access protocols as conflict free or contention based. In conflict free protocols a transmission would not be interfered by another transmission. Conflict free transmission can be achieved by allocating the channel to the users, either statically or dynamically. Here the channel resources can be divided by providing the entire frequency range (bandwidth) to a single user for a fraction of time as done in Time Division Multiple Access (TDMA), or providing the fraction of frequency range to every user all of the time as done in Frequency Division Multiple Access (FDMA), or providing every user a portion of the bandwidth for a fraction of the time as done in spread spectrum based systems such as Code Division Multiple Access (CDMA).

The contention based multiple access protocols must prescribe a way to resolve conflicts once they occur, so that all messages are eventually transmitted successfully. If probability of interference must be small, such as might be the case with bursty users, the wastage of bandwidth while resolving the conflicts is less than the wastage of bandwidth reserved for idle users in conflict free protocols. Hence in practice the contention based multiple access protocols are used for bursty traffic such as data traffic and conflict free protocols are used for stream traffic such as voice.

As this work deals with multimedia traffic types and transmission channel for mobile communication we adopt the following strategy for shared channel access schemes.

1. Use contention based protocols for data traffic types.
2. Use dynamic conflict free protocols for voice traffic on the talkspurt basis. Contention based protocols are used for first packet of the talkspurt.
3. Use dynamic bandwidth on demand, conflict free protocols for video traffic.

The basics of the contention based protocols used in this work are described in section 2.4. The basics of the TDMA and CDMA conflict free protocols are described in section 2.5.

2.4 Multiple Random Access Schemes

The Z-cell algorithm used to control the access to the channel in this thesis is a modification of the simple 2-cell algorithm proposed in [Geor82], [Geor85]. The 2-C algorithm is a limited sensing binary tree type protocol. The main feature of the 2-C protocol is that it does not allow all the newly generated packets to interfere with the packets already accepted by the system. The collision resolution process and the eligibility criteria to join the access process are described in the following sub-sections.

2.4.1 Collision Resolution Process

The 2-cell algorithm belongs to the family of limited sensing tree type protocols, where users attempting to access the channel can sense the status of their attempts via a feedback channel. The underlying channel is assumed to be slotted. All the users that are eligible for accessing the channel have to go through a collision resolution process. The means to determine the eligibility of a new user to join the resolution process will be explained in the next section. The collision resolution algorithm of the 2-C protocol can be described by using the two-cell concept, presented in Fig. 2.2. We assign to the system two cells. The first is called the Transmission Cell (TC) and the second the Waiting Cell (WC). Users participating in a competition process reside in one of these two cells. Users that during a slot belong to the TC are allowed to transmit in this slot. Users residing in the WC have to refrain from transmission.

All users eligible to join the competition process, starting at slot k , position themselves in the TC. If a collision occurs in the k^{th} slot, the FC carries a C status to the

users. Then, every user that participated in the collision, chooses with probability P_{TC} to remain in the TC and with probability P_{WC} to move in the WC. The users residing into the WC refrain from transmission during the $(k+1)^{th}$ slot. Those remaining in the TC attempt transmission. If a collision occurs, the process is repeated again, and statistically speaking, more users move in the WC for the $(k+2)^{th}$ slot. If instead, a NC is received through the FC, (i.e. during the $(k+1)^{th}$ slot at most one user was residing in the TC) all users residing in the WC move to the TC and attempt to transmit at $(k+2)^{th}$ slot. The reader can verify that when following this policy, the only way to receive two consecutive NC (i.e. NC-NC) is when both cells TC and WC, are empty. In which case all the competing users have been accommodated. The NC-NC pattern is the marker which permits the idle users to identify the termination of an on-going competition process.

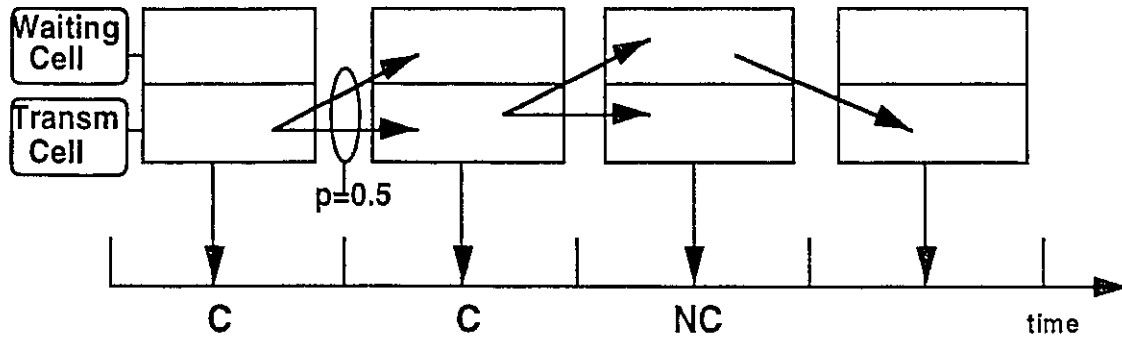


Fig. 2.2 Collision resolution Process

The algorithmic steps of the 2-C protocol can be implemented easily by the individual users as follows. Each user is equipped with a binary flag, its value being either 0 or 1. We represent the value of the flag of a specific user during the k^{th} slot as Z_k ($Z_k \in \{0, 1\}$). We represent the feedback status received by the user during the k^{th} slot as F_k , where $F_k \in \{NC, C\}$. A user attempts transmission during the k^{th} slot if $Z_k = 0$ and withholds from transmitting if $Z_k = 1$. A packet is successfully transmitted if $Z_k = 0$ and

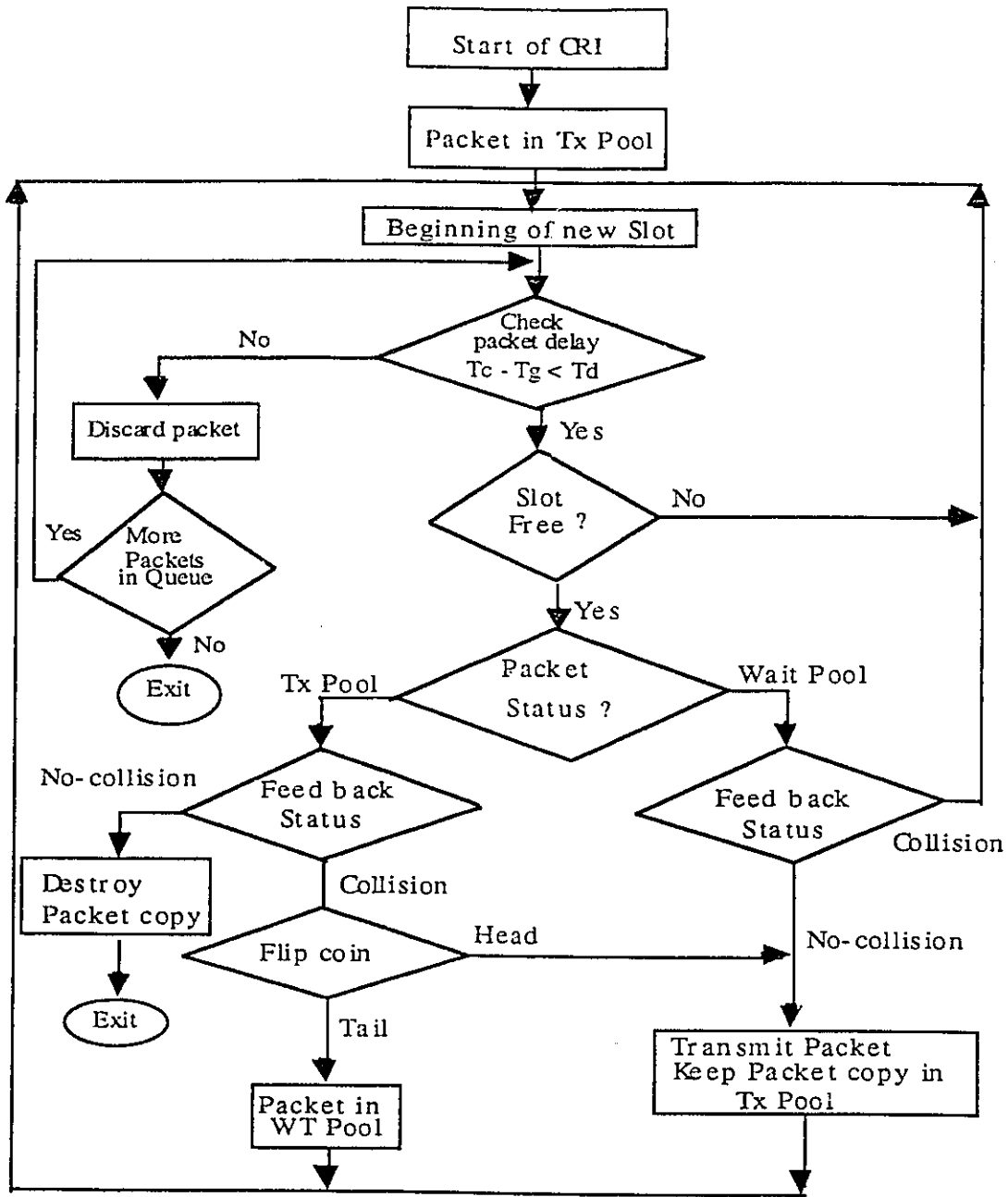
$F_k = \text{NC}$. When a user becomes active, it sets its flag to $Z_k = 1$ and starts monitoring the feedback channel. An idle user does not change the content of its flag until receiving two consecutive **NC** via the feedback channel. When this happens, a new user checks if it meets the qualifying criteria for joining the new competition process. If the decision is positive, the user switches the value of its flag to 0 and transmits at the first slot following the two consecutive slots marked **NC**. Let us assume that this happens at the n^{th} slot. From that point on, the user updates the flag Z_l as follows.

For $l \geq n$ and until the packet has been successfully transmitted:

1. if $F_l = \text{NC}$ and $Z_{l-1} = 1$, then $Z_l = 0$
2. if $F_l = \text{C}$ and $Z_{l-1} = 1$, then $Z_l = 1$.
3. if $F_l = \text{C}$ and $Z_{l-1} = 0$, then

$$Z_l = \begin{cases} 0 & \text{with probability } P_{ic} = 0.5 \\ 1 & \text{with probability } P_{wc} = 0.5 \end{cases} \quad (2.1)$$

Figure 2.3 depicts the logic diagram of algorithm of collision resolution process to be implemented by the users.



Legend:

Tx Pool = Transmission Pool

Wt Pool = Wait Pool

Tg = Packet Generation Time

Tc = Current Time

Td = Allowable Delay Before Packet Transmission

Figure 2.3 Flow chart representation of Collision Resolution Process

2.4.2 Window Protocol

As mentioned earlier, one of the main characteristics of the 2-C protocol is that it does not allow new users to enter an ongoing competition. They have to refrain from transmitting until all the packets engaged in the current competition are accommodated. When an ongoing competition process terminates, (let us assume that this occurs at slot $(k - 1)$) each new user verifies if it qualifies to join the next competition process. Two policies may be used to determine the eligibility of a user to join the next competition, namely 1) the window policy and 2) the probability of transmission policy.

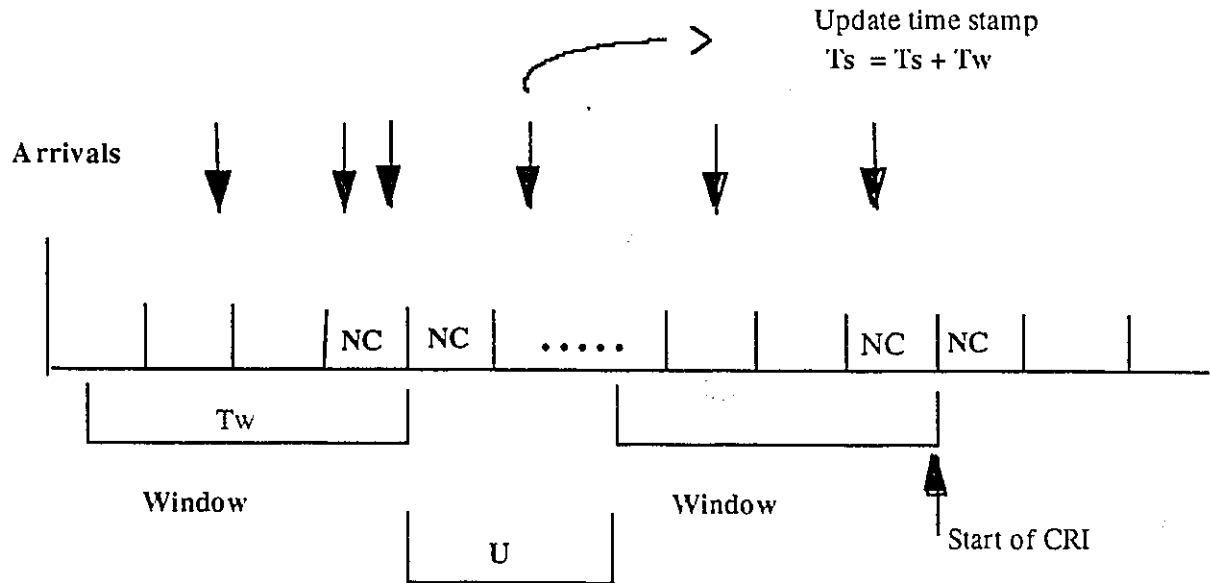
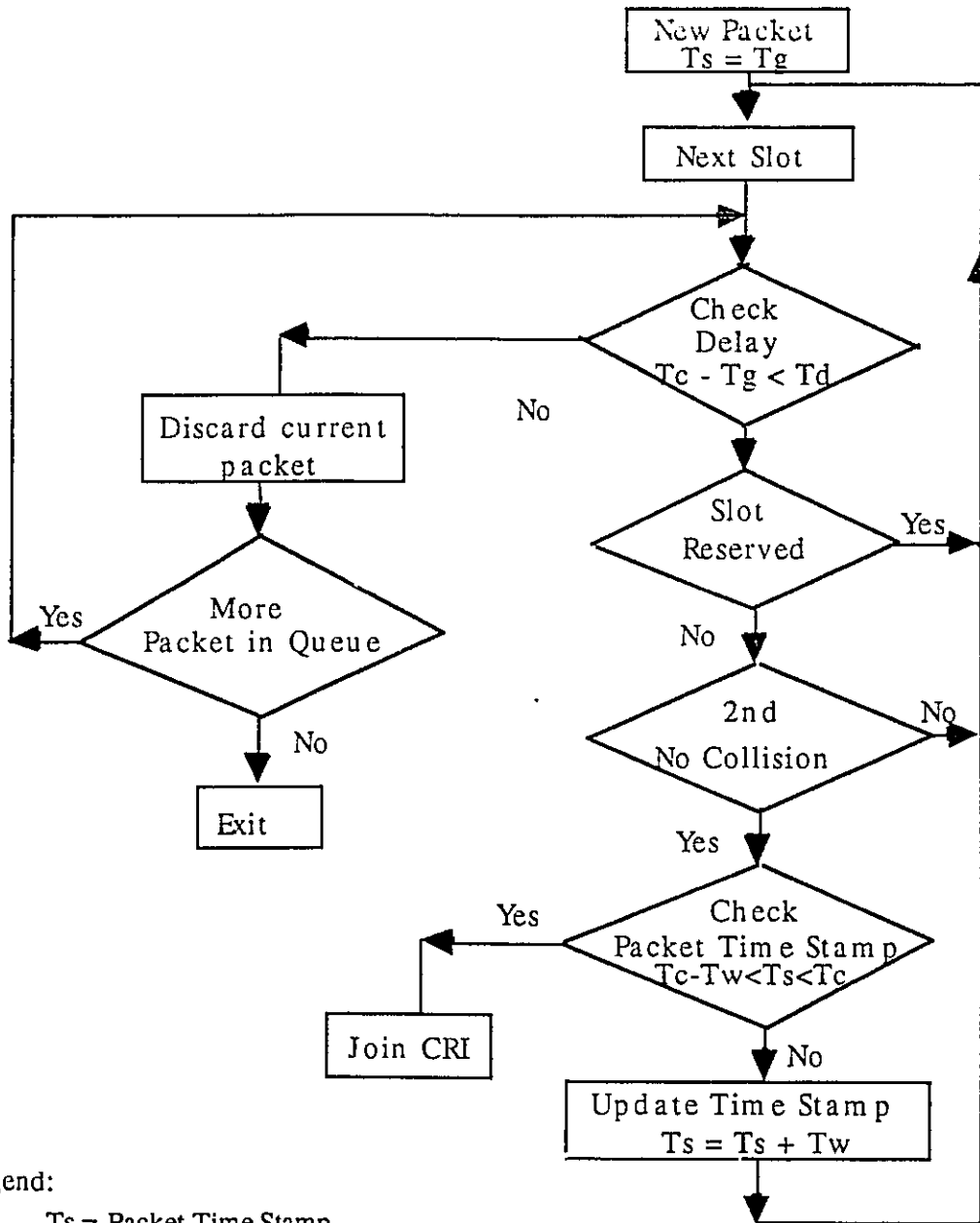


Figure. 2.4 Window algorithm



Legend:

T_s = Packet Time Stamp

T_w = Window Time Interval

T_c , T_g and T_d have the same significance as in figure 2.3.

Values of T_w and T_d are specific to traffic Types.

Figure 2.5 Logic Diagram of the Window Mechanism to Join CRI

In the window policy, the user examines the time elapsed between the moment the packet was generated and the end of the competition process. If the difference is less than a

certain value T_w the user is allowed to transmit. If not, the user refrains from transmission, however it updates the value marking the time of the packet generation by an amount equal to the window size, i.e., if the packet was marked as being generated at T_g , it updates the time of its generation to $T_g + T_w(m)$ as shown in figure 2.4.

If the probability of transmission process is used, every new user decides with probability P_d to join the new competition process. If the decision is positive, the user attempts transmission in the coming slot, i.e. k^{th} .

Figure 2.5 shows the procedure to follow by the users implementing the window mechanism.

2.5. Basics of TDMA and CDMA

The protocols analyzed in this work are based on two conflict free multiple access techniques TDMA and CDMA. These two basic techniques are described in the following subsections.

2.5.1 Time Division Multiple Access

Time Division Multiple Access (TDMA) scheme is well known and widely used in communication networks. Here we are giving only the brief overview of TDMA. Its detailed description can be found in many texts such as [KuoF81], [Stal85]. In TDMA the time axis is divided into time slots. These time slots are assigned to different users during connection set up time. Every user is allowed to transmit freely during the slot allocated to him. During the assigned slot the entire channel resources are devoted to that user. The slot assignment follows a predetermined pattern, that repeats itself periodically. Each such period is called a frame. The frame structure of TDMA is as shown in figure 2.6. In every frame each user have exactly fixed number of slots. For proper operation of TDMA

scheme, the users must be synchronized so that each one knows exactly when and for how long he can transmit.

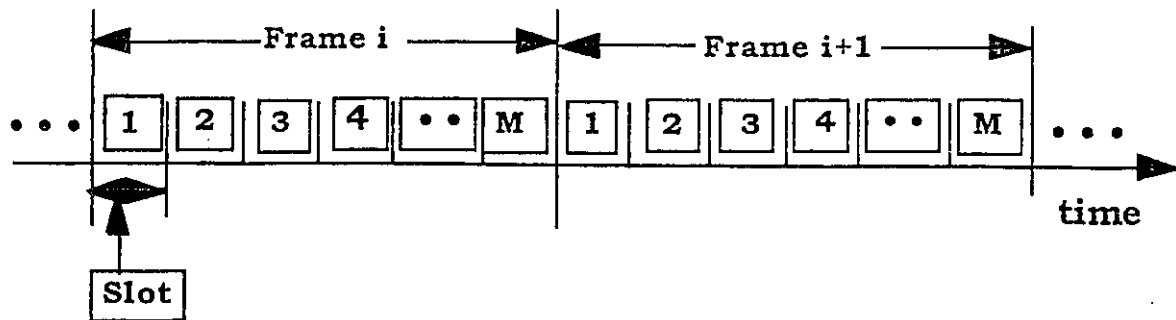


Figure 2.6 TDMA Frame Structure

2.5.2 Code Division Multiple Access

Spread spectrum transmission techniques have many advantages over traditional TDMA and FDMA techniques, especially in wireless environments. Some of its advantages are frequency diversity, soft hand-off capability, resistant to multipath effects and low probability of interference. Code Division Multiple Access (CDMA) is a strategy for accommodating multiple users per channel based on orthogonal spreading codes. Multiple users transmit simultaneously, yet they do not conflict with each other. Pilot tone techniques are used for synchronization and equalization purposes. The two most commonly used Spread Spectrum techniques are Direct Sequence Spread Spectrum (DS/SS) and Frequency Hopping Spread Spectrum (FH/SS).

CDMA signals overlap each other, in both the time and frequency domain. The signals are separated at the receiver by using a correlator which accepts only the signal energy from the selected binary sequence and despreads its spectrum. If the codewords are orthogonal and the synchronization is perfect, no interference by the other users appears at

the output of the receiver. If synchronization is not perfect and/or the spreading sequences are not orthogonal to each other, the accumulated effect of the signals of the other users

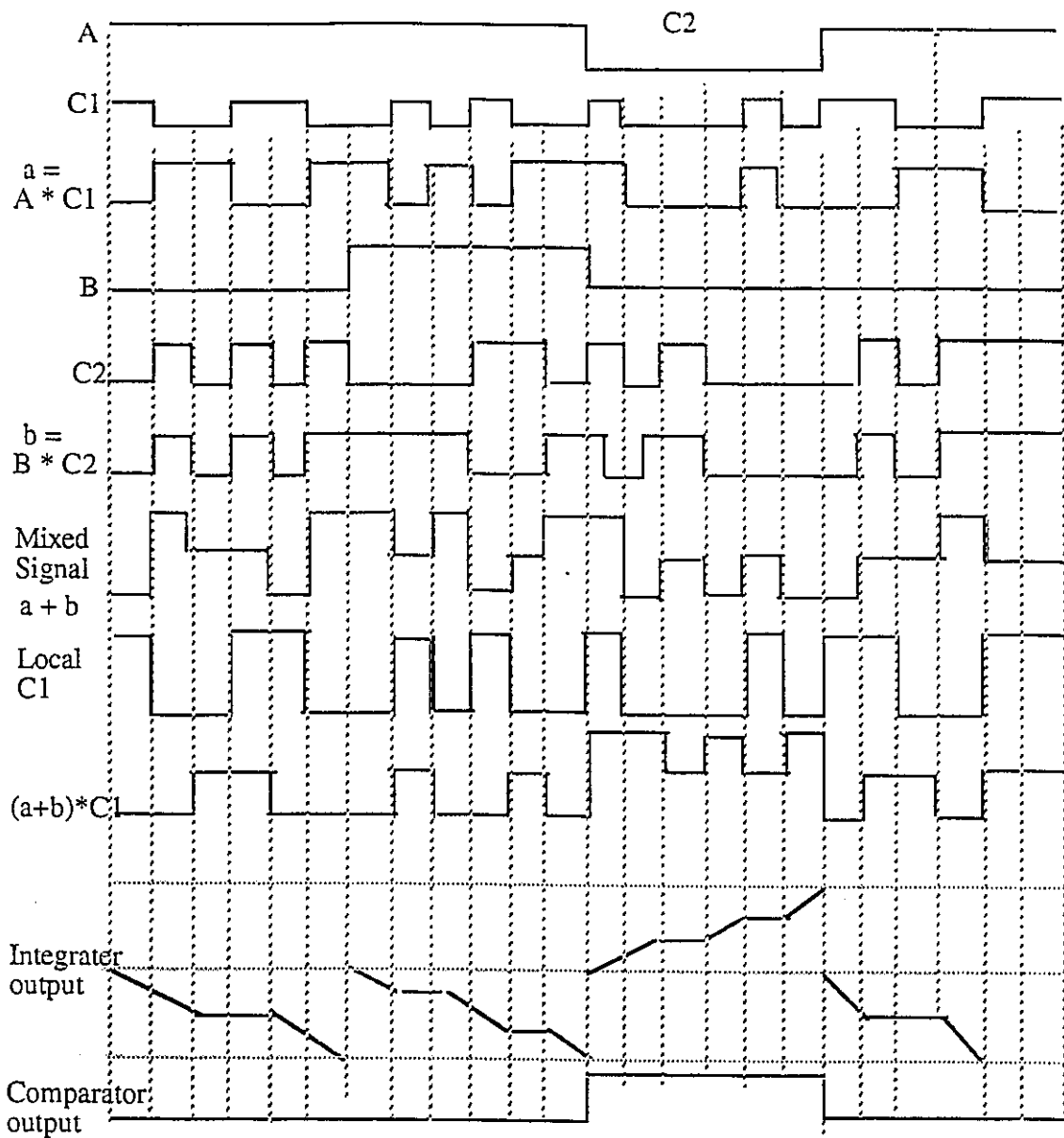


Figure 2.7 Signal recovery by the despreading process

appears as self-interference. Below we describe briefly the principle, of operation on which CDMA is based on.

With CDMA, the signal of each user is spread with a different pseudo random binary sequence. In this way, a large number of CDMA signals can share the same frequency spectrum. Figure 2.7 shows two signals A and B, spread with codes C1 and C2 respectively.

The received signal is given by

$$a + b = (A \times C_1) + (B \times C_2) \quad (2.2)$$

After decoding we get

$$(a + b) \times C_1 = (A \times C_1 + B \times C_2) \times C_1 = A \times C_1 \times C_1 + B \times C_2 \times C_1 = A \quad (2.3)$$

Similarly the other user using code C2 can decode the signal B from the mixed signal as

$$(a + b) \times C_2 = (A \times C_1 + B \times C_2) \times C_2 = A \times C_1 \times C_2 + B \times C_2 \times C_2 = B \quad (2.4)$$

As C1 and C2 are orthogonal to each other, hence in the above equations

$$C_1 \times C_2 = C_2 \times C_1 = 0 \quad (2.5)$$

Signal A is recovered by using Eqn. (2.3) as shown in figure 2.7. The signal-to-interference ratio is determined by the ratio of the desired signal power to the sum of the powers of all the other signals, and is enhanced by the system by the processing gain or the ratio of spread bandwidth to baseband data rate. The CDMA digital cellular system capacity is determined by the processing gain, the required E_b/N_0 , the voice duty cycle, the frequency reuse efficiency, and the number of sectors in the cell..

2.6 Traffic Source Models

2.6.1 Data Traffic

We assume that the aggregate data traffic follows the Poisson distribution. This assumption is based on our conviction that in the future, we will see a large number of data terminals supporting non-real time applications (i.e. paging, e-mail etc.), each one producing small amounts of traffic. Poisson traffic represents quite accurately the aggregated traffic produced by a large number of low activity users. In our simulation we refer to data load with different values of λ . Where λ is the average outcome of the Poisson distribution. Data load of $\lambda=0.1$ implies that the aggregate average number of data bits (or packets) coming into the network equals 10% of the channel capacity. Data packets should be delivered to the destination error free, however, data traffic is insensitive to delays. Hence flow control and automatic repeat request (ARQ) procedures can be employed for data traffic.

2.6.2 Voice Model

Voice communications represent the most popular telecommunications service. Voice communication signals are composed of periods with voice activity (talkspurts) and silence periods. In a conventional TDMA system, a voice user is allocated a specific slot in the frame that keeps it to himself/herself during the entire period of connection. However, the presence of silent periods during a conversation means that a user keeps a channel reserved even during time intervals where there is no signal to be carried through. Studies have shown that (in the average) the silent periods correspond to 57% -65% of the total voice communication time which means that during a connection for voice communications, more

than half of the network resources are wasted [Brad69], [Muis86]. Multiple access protocols have been developed to resolve this problem.

Voice communication signals are sampled with a sampling rate of 8,000 samples per second. This corresponds to a sampling period of 125 microseconds. The sample is digitized and a voice packet is produced. In our simulation analysis assuming speech coding rate of 32 kb/s and with 64 bits of header information, each active voice source generates one packet per frame, for frame duration of 16 ms.

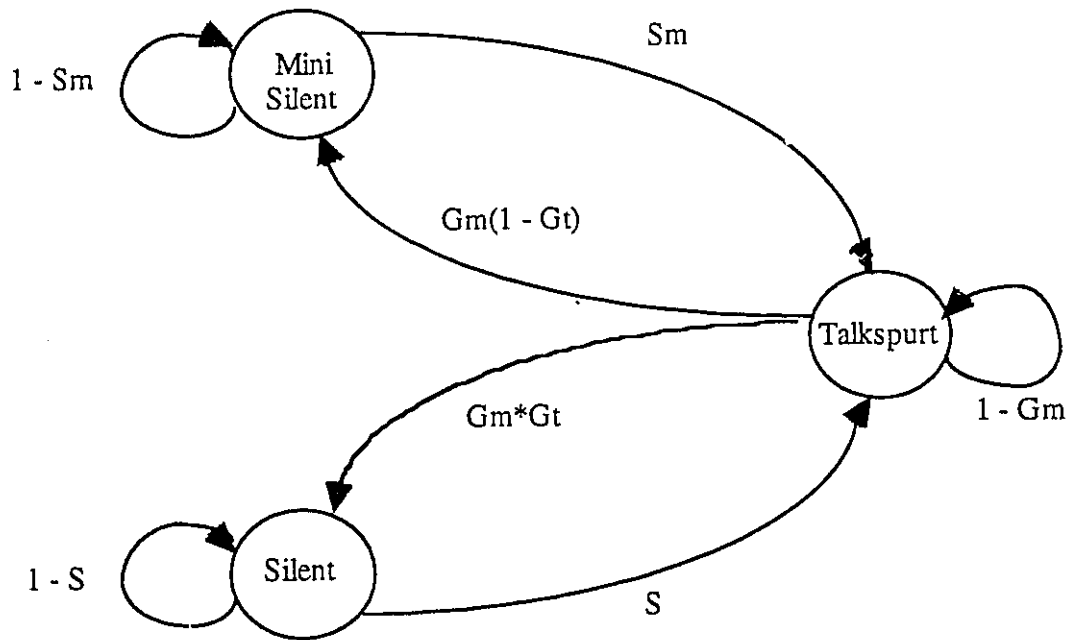


Figure 2.8 Markovian 3-State model for voice

Many systems employ voice activity detectors (VAD). VAD's can be classified as "slow" and "fast". Slow VAD can separate only talkspurts and silence periods. In addition to that, fast VAD are able to distinguish also between mini-talkspurts and mini-silence periods, residing within a talkspurt. The traffic generated by a voice user equipped with VAD can be described by a Markovian model: 2-state model for slow VAD, 3-state model for fast VAD. Figure 2.8 depicts the 3-state Markovian model for voice. The vocoder of the terminal samples and digitizes the voice signal during talkspurts and suppresses the silence

periods. The length of all the talkspurt and silence intervals is exponentially distributed. The average duration of these talkspurts and silence intervals is shown in table 2.1 [Good89].

State	Average Duration
Principal Talkspurt (T1)	1.00S
Principal Gap (T2)	1.35S
Mini Talkspurt (T3)	0.275S
Mini Gap (T4)	0.050S

Table: 2.1 Voice Source Parameters

2.6.3 Video Traffic

In the past many studies have used ON-OFF models for the modeling of video. The statistical properties of these models do not match with high accuracy the actual statistical behavior of the video signal. To overcome this problem, we use real compressed video sequence traces for our simulations. We use the statistics from ITU (formerly CCITT) standard video sequences, Grandma, Mother & Daughter and Salesman.

The personal communication networks will not have the bandwidth availability to support very high bit rate video. These will support low bit rate video applications such as video telephony. The variable bit rate video compressed according to CCITT standards H.261, H.263 or MPEG-4 [Gall91], [Ste94] are good candidates for use in PCN's. In our simulations we have used H.261 video in QCIF format at 15 frames/sec:

1. With feed back control from encoder transmission buffer, which gives constant bit rate (64 kb/s) but variable quality video.

2. Variable bit rate video without any feedback control (averaging) of the bit rate, which provides constant video quality.

H.261 is the CCITT recommendations for compressed video and is commonly known as px64. The purpose of using H.261 video is to show the capability of the proposed protocol to support variable bit rate traffic. Our protocol does not need to reserve the bandwidth at peak bit rate. In this way we can make efficient use of the bandwidth. The details of the video format can be found in the H.261 recommendation of the CCITT [H261].

CBR video is produced by dynamically changing the number of quantization levels the encoder uses. This control is done through a feedback mechanism whose input is the buffer occupancy of the output buffer. When the buffer occupancy exceeds a certain value, the control mechanism requests that the number of the quantization levels used by the encoder is reduced. This reduces the amount of produced data, however, the fidelity of the image is reduced as well. Consequently, CBR provides us with variable quality of video. We will test our system with both, CBR and VBR video sources using H.261 standard.

Chapter 3

Integrated Data/Voice Channel Access Protocol On a TDM System

3.1. Introduction

Due to the limited frequency allocation, the success of the today's cellular radio is directly linked to the bandwidth efficiency of the system. In the future, the demand for personal and mobile communication services is expected to increase and new wideband applications such as video and high quality audio will appear in the market. The introduction of these new services will require the efficient use of the available frequency resources. The communication industry is responding to this reality by constantly proposing and introducing systems and networks achieving higher spectral efficiency as compared to the previous generation wireless systems. For example, the analog North American cellular radio system (Advanced Mobile Phone System or AMPS) require a bandwidth of 30 KHz to accommodate the up or down link of a single user [Will89]. The first generation of Digital North American radio is based on Time Division Multiple Access (TDMA) and is able to allocate three customers in the same bandwidth that was before occupied only by one [Padg95]. Europe and Japan have developed their own digital cellular radio systems. In Europe we have the GSM(Global System Mobile), whereas Japan developed its own

cellular system [Padg95]. The same pattern of a continuous effort to develop more efficient schemes has appeared in other similar wireless services, such as the cordless telephone or the Mobile Satellite (MSAT) system .

The information rate produced by several types of information sources, such as voice, video is of variable bit rate (VBR) nature, and in many cases, the peak rate at which the information source might generate is considerably higher from its average rate. Most of today's systems are designed to allocate resources that can accommodate the maximum bit rate during the entire duration of the connection. However, this approach wastes useful resources. The coming generations of mobile and personal communication networks (MPCNs) are being designed to accommodate variable bit rate sources efficiently. Today two are the most popular directions followed in cellular and portable radio or satellite-based application. The first approach is based on the use of TDMA [Mitr90]. The second involves the design of systems based on Code Division Multiple Access (CDMA) [McTi94]. Because of its nature, CDMA allows the development of systems that can easily incorporate voice activation mechanisms in their operation.

In order to take advantage of the variable bit rate nature of voice communication sources, dynamic Reservation Multiple Algorithms (RMA) are being used in TDMA. The most popular of the proposed RMA seems to be the Packet Reservation Multiple Access (PRMA)[Good89], [Good91].

PRMA combines both, random access and reservation techniques and allows easy integration of voice and data. The random access scheme used by PRMA is the Slotted Aloha (SA) protocol. However, SA based protocols become progressively unstable as the number of users increases. This might create a problem to the performance of future MPCNs, where we expect to see quite sudden changes of the traffic volume. Consider for example a subway station at the time when a train arrives. Upon the train arrival, many passengers might try to use their personal communicators, increasing the volume of the traffic considerably in a very short period.

In order to resolve the weaknesses of PRMA, new stable reservation multiple access protocols are needed that are appropriate for the MPCNs of the future. The medium access protocols presented in this chapter have been developed by properly modifying a number of stable algorithms that have been proposed in the past [Pate89M], [Pate89S], so that they can accommodate effectively multimedia traffic and information sources with different types of priority, in a way, that they can guarantee the QoS requirements of the various multimedia applications and classes. Furthermore the protocols are evaluated when used in an integrated network environment: voice and data.

For the remainder, this chapter is organized as follows. Section 3.2 describes the Z-cell algorithm used to resolve collisions among competing packets. Section 3.3 describes the integration of various traffic types according to their priority. Section 3.4 provides general description of the reservation scheme. Section 3.5 deals with the performance evaluation of the proposed algorithms under data and voice traffic types. Finally, section 3.6 summarizes and concludes this chapter.

3.2 Z-cell Algorithm

In the present work, we proposed the use of the Z-C algorithm and variants of it for controlling the access to a shared channel. The Z-cell algorithm (Z-C) is a generalization of the 2-C algorithm proposed in [Pate89M] and discussed in detail in chapter 2. As for the case of the 2-C algorithm, we assume that users share a common TDM channel to communicate. A base station is used to inform to all users of the status of the TDM channel in a slot per slot basis. Slots are grouped into frames. The duration of the slot is equal to the time needed to transmit a packet. With voice encoding bit rate of 32 kb/s and packet size equal to ATM cell size, the frame period is fixed to 12 msec. In this way, a voice source may transmit a packet per frame. These values have been chosen to allow us to

compare the results for the proposed protocol with the results reported for PRMA [Good91].

Users compete to gain access to the channel in a slot per slot basis. Some of the slots in a frame may be reserved 'on demand' by voice users. The status of these slots is broadcasted by the base station to all nodes. All unreserved slots are open to be used by data or new voice users. The reservation scheme is presented in section 3.4.

We assume that the feedback regarding the collision no-collision status of packet transmission and the feedback information regarding reservations can be easily transferred through the reverse link channel.

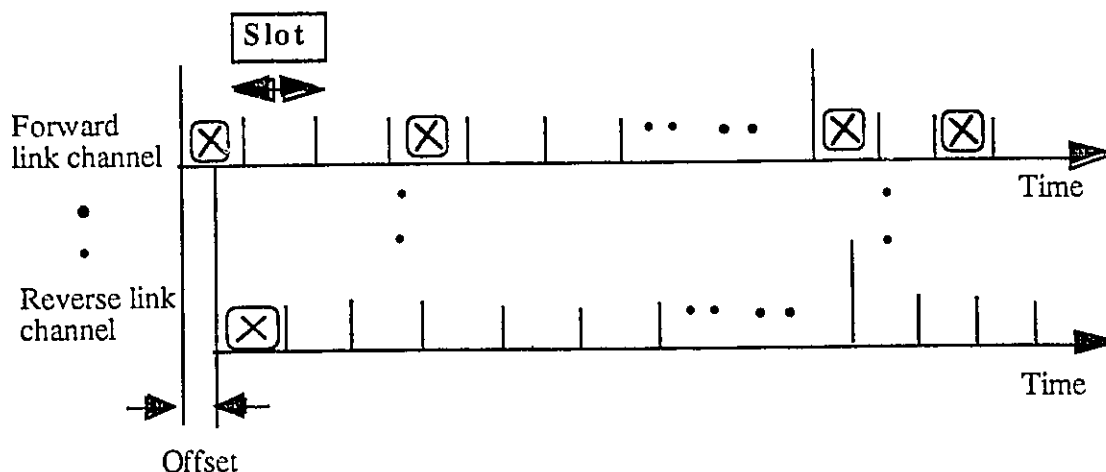


Figure 3.1 Offset in forward and reverse link channels to facilitate feedback.

In order to allow the users to process the feedback information two schemes are possible. In the first scheme, we assume that the slots of the FC and the TDM channel are time synchronized. In this case, a guard time interval is reserved to count for the time needed for the transmission and processing delay of the feedback information. In the second scheme, the forward and reverse link slots are offset as shown in figure 3.1.

Users implement the Z-C algorithm or variants of it as collision resolution protocols. Some modifications to the Z-C algorithm have deemed necessary to incorporate

additional priority mechanisms. The algorithm can be easily implemented. Every user is equipped with a counter, which takes values from zero to $Z-1$. At the beginning of a collision resolution interval (CRI), every user scheduled to compete at this CRI sets his counter, χ , to a value between 0 and Z . The user chooses to set $\chi = m$ with $0 < m < Z-1$ with probability $P_Z(m)$, where $\sum_{m=0}^{Z-1} P_Z(m) = 1$.

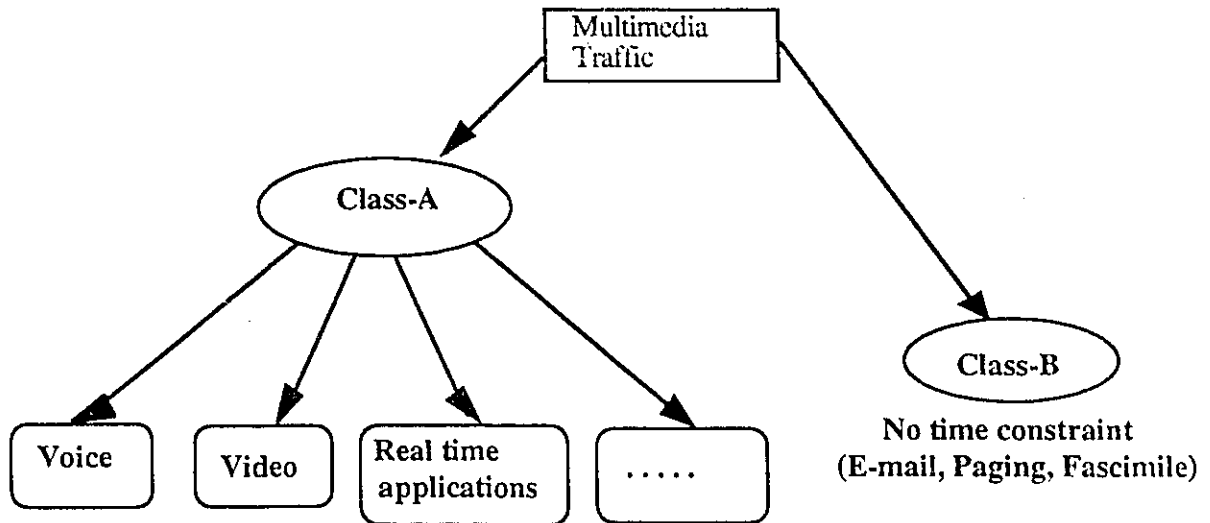
From that point, and until the CRI is finished only users with $\chi = 0$ are allowed to transmit in the first available slot. The remaining users refrain from transmitting. A user participating in the collision resolution process updates his counter as follows. If at the end of a contention slot the feedback channel indicates no collision (NC) then the user decrements the value of his counter, i.e., $\chi = \chi - 1$. If the feedback channel indicates collision (C), the update is done as follows. If $\chi = 0$ then the user sets his counters randomly to a value between 0 and $Z-1$. All values have equal probability, $1/Z$, to be chosen. If $\chi \neq 0$ the user leaves his counter unchanged. New arrivals (users) occurring during an ongoing CRI can follow one of the following two options. They can refrain from joining the competition or they set their counter to $\chi = 0$ and transmit in the first slot marked available. It can be shown that with this contention resolution algorithm, every CRI ends with Z consecutive (NC) slots. In other words, there is no way to have a string of Z consecutive NC and still packets in the competition process. This fact allows to identify the end of CRIs.

3.3 Traffic Integration

The Z-C algorithm can be used in an environment integrating different types of services: voice, video and data. In order to be able to satisfy the requirement of the different types of services, we can group the packets generated by the various types of services into two major classes. Class A (Cl-A) includes all packets with strict requirements in packet delay and packet dropping probabilities. Class B (Cl-B) includes all packets whose delivery is

not time-constrained, as long as the maximum delivery delay is finite. Packets falling in this category are produced by non-real time applications such as e-mail, facsimile, graphics, paging, etc. Some of these applications might only need one packet to transfer the information, others may require more than one packet to complete the transfer. In the first case, there is no need for performing a slot reservation. In the second case, the first packet to be sent competes and upon succeeding can reserve the slot for the transmission of the following packets.

In order to provide better access to Cl-A packets, it is more desirable to allocate slots to the Cl-B packets at the end of the frames. The allocation mechanism for Cl-B packets is activated only if all scheduled collision resolution processes for Cl-A packets have finished, and there is still space within the frame for Cl-B packets. Even more, if not all Cl-A packets have granted a reservation within a frame, any Cl-B reservation is canceled. In this case, Cl-B packets will have to go through the competition process at a latter time. The classification of packets generated by various types of services is shown in figure 3.2



Constraints:

- Time delay of packet delivery
- Acceptable levels of blocking probability

Figure 3.2 Multimedia Traffic Classes

The operation of the Z-cell algorithm for supporting multiple types of services can be described as follows. A different CRI is defined for each type of traffic. A user of type-Z chooses with probability $P_Z(i)$ ($1 < i < N_Z$) to participate in the CRI in progress during frame $(X + i)$ and after. P_Z and N_Z are parameters chosen according to the delay constraints of the traffic. By doing so, the traffic produced by type Z users is smoothed out. However, the protocol generates separate CRI's for each type, instead of allowing all the users to enter and compete during the same CRI.

At the beginning of the frame, the most time-sensitive packets, in our case voice, are allowed to start competing. Once the CRI for voice users is completed, the packets belonging to the next priority class are scheduled to compete from there on and up to the end of the current frame. In other words, the Cl-B packets are allowed to start to compete only after all the CRI's for Cl-A packets have been completed. This priority mechanism is

depicted in figure 3.3. As shown here the end of the CRI's for Cl-A packets is not fixed while the CRI for Cl-B packets are forced to terminate at the end of frame.

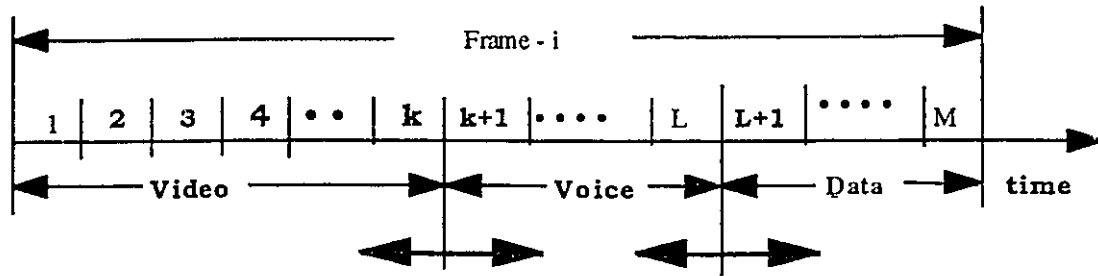


Figure 3.3 Traffic Integration Serving Individual Classes of Users separately

In every frame, the Cl-B packets are allowed to compete only after all the competing Cl-A packets have gone through. They do it according to the Z-cell algorithm described in section 3.2. In every CRI and similar to the 2-C algorithm (see chapter 2), only packets that have arrival times falling within a time window, defined with reference to the end of the previous CRI are allowed to compete. Those users that fall outside this window, update their arrival times properly (by shifting themselves forward in time for a period equal to the window size) and they wait their turn to compete. At the end of frame if the CRI for CL-B packets is still in process, it freezes and resumes in the following frame when all the Cl-A packets have gone through. Following this approach, we ensure the Cl-B packets never interfere with Cl-A packets.

3.4 Reservation Scheme

Data packet are not sensitive to delays. The only requirement is that they should be delivered to the destination correctly with a finite delay. On the contrary, voice communications are sensitive to time delays. If a voice packet does not go through the system within certain time, it ages and becomes useless for the destination. For this reason,

when voice packets exceed a certain amount of time without being successfully transmitted, they are dropped by the terminal to allow the following packet to proceed. Of course, the loss of packet degrades the quality of the voice communications, the degradation increasing with the increase in the packet dropping rate. Studies have shown that a packet dropping rate up to 1% - 2% can be tolerated and the quality of voice communications remains acceptable for the end user [Good89]. Now a days, a packet loss rate of 1% is targeted in as the acceptable level for the design of future wireless personal communication networks. If we allow all the voice packets to go through random access methods, it will be very difficult to deliver them within maximum allowable time period and achieve the targeted packet loss rate. Due to this sensitivity of voice packets to delays, we need to use some reservation methods for voice packets.

In packetized voice applications, a user produces packets at regular time intervals. This knowledge has been used in the past to implement some hybrid TDMA protocols which combine random access and reservation characteristics [Good89], [Mitr90]. Cellular systems are characterized by short propagation delays. This characteristic allows that the terminals to be able to receive feedback on the status of the channel from base station, enabling the system to be able to provide real time communications.

In order to avoid the wastage of resources we would use the reservation for voice connections on the talkspurt basis (see Chapter 2). When a voice user shifts from silence to talkspurt state, he starts by listening to the feedback channel in order to identify free slots, i.e., available for competition. When a free slot comes, the terminal decides to transmit the packet or refrain from doing so according to the rules of collision resolution algorithm. If this is the only packet that was transmitted in the slot, the packet is successfully received by the base station. Let us assume that this happens at the n th slot of the frame. The base station acknowledges that the packet transmitted in the n th slot was successfully received. Acknowledging that this was a voice packet, the base station reserves the n th slot in the coming frames for this particular voice user. The base station marks the slot unavailable for

open competition and reserves the use of the n th slot of each of the succeeding frames to as long as the user remains in the talkspurt state. The user, being made aware that the packet has been successfully transmitted, will take for granted that the n th slot in the coming frames is reserved for subsequent voice packets in the talkspurt.

When the talkspurt ends, the terminal releases the slot. This can be done in either one of two ways. In the first case, the user includes a field in each packet. This flag is set only if the packet is the last one in the talkspurt. The base station monitors this field. When it finds the flag set (last packet), the base station marks the n th slot as available for competition in the following frame. A second way to release the slot will be simply to leave the n th slot empty. The base station marks n th slot available for competition in the frame following an empty slot. The second approach ends up wasting a slot, however, it reduces the implementation complexity of the system. The second approach will be considered from now on.

3.5 Performance Evaluation

In the following results for protocols integrating voice and data under errorless channel conditions are presented. Voice traffic is generated for each voice user, according to the three-state Markovian model, corresponding to a terminal using fast VAD (see Chapter 2). For the data packets, we assume a Poisson process. In this work, we consider a TDM channel with frame duration 12 msec and 20 slots per frame. The maximum allowable delay for voice packets is 32 msec. Values used for these parameters are the same as those used in the analysis of PRMA [GOOD89]. If they exceed this delay, they are considered to be lost. The maximum acceptable blocking probability for voice packets equals 0.01 to 0.02.

We examined three different versions of Reservation Random Access Protocols. All three of them are based on the Z-Cell algorithm.

The first configuration, referred as Conf-1 from now on, allows the packets from active voice users to start competing only at the beginning of the frame. At the beginning of the first slot in the frame, all contending voice users set their counter to zero and transmit. When the CRI for voice users terminates, the voice users withdraw for the rest of the frame allowing data packets to compete.

The second configuration, Conf-2, is similar to the first one, with the only difference that at the beginning of the frame, every voice user decides with equal probability to set the counter χ to 0 or 1.

The third configuration (Conf-3) is also a variation of Conf-1. At the beginning of a CRI, all voice users reset their counters $\chi = 0$ and transmit in the first available slot. However, in this case, after the CRI for voice users has terminated, voice users who may have become active during the just finished CRI are allowed to start a new CRI. They do by detecting two consecutive NC indicating the end of the previous CRI and start a new CRI. Data packets do not enter into competition unless if they see three consecutive NC. The first two NC's mark the end of the CRI by the voice users. The third NC marks that no other voice user wants to initiate a new CRI for voice packets. With Conf-3, voice users have always priority for the entire duration of the frame. In all three configurations, the data packets use the Z-Cell algorithm with $Z=2$ and window size 10 slots.

Figures 3.4, 3.5, 3.6 display the blocking probability of voice packets vs. number of voice users. Each figure shows three curves. They correspond to data traffic loads of $\lambda = 0, 0.1$ and 0.2 packets per slot. In all these figures, the voice packets blocking probability is practically unaffected by the data load. This is to be expected, because the protocols are able to protect the voice packets from the presence of the data traffic, by not allowing data to contend in the same CRI's with voice packets. We see that the Conf-1 is

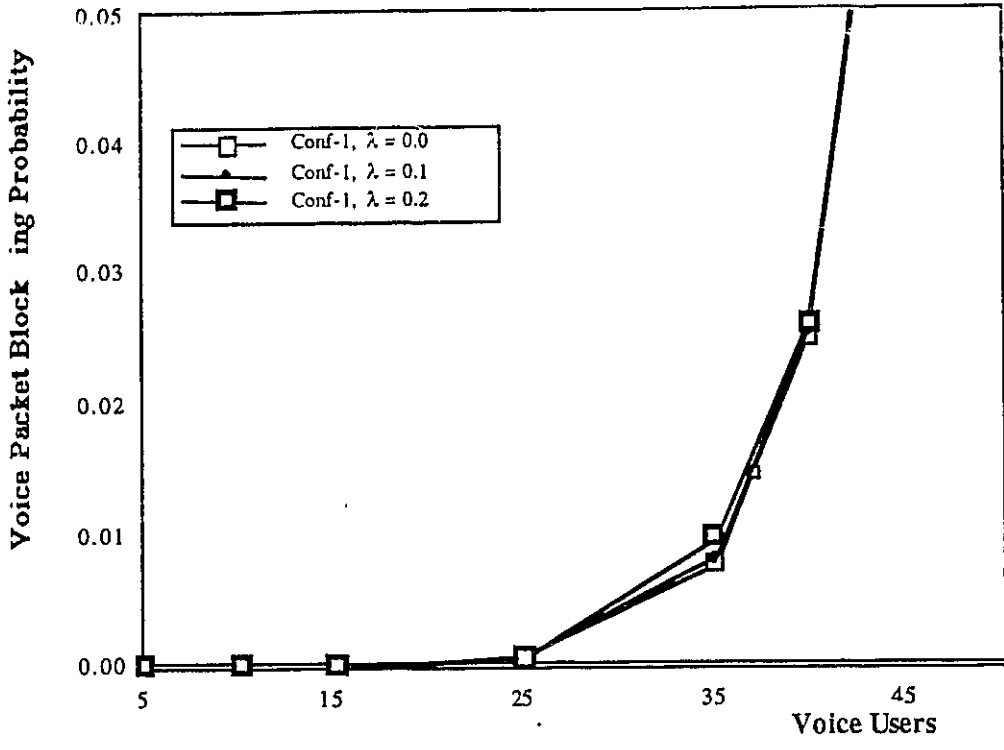


Figure 3.4 Voice Packet Blocking Probability (Configuration 1)

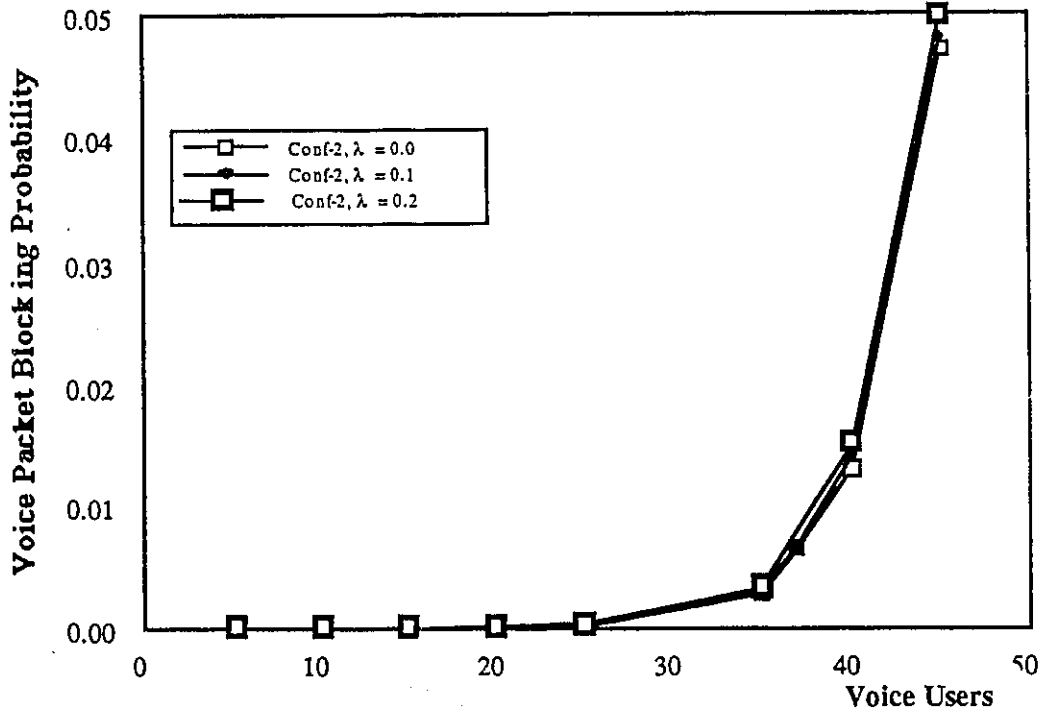


Figure 3.5 Voice Packet Blocking Probability (Configuration 2)

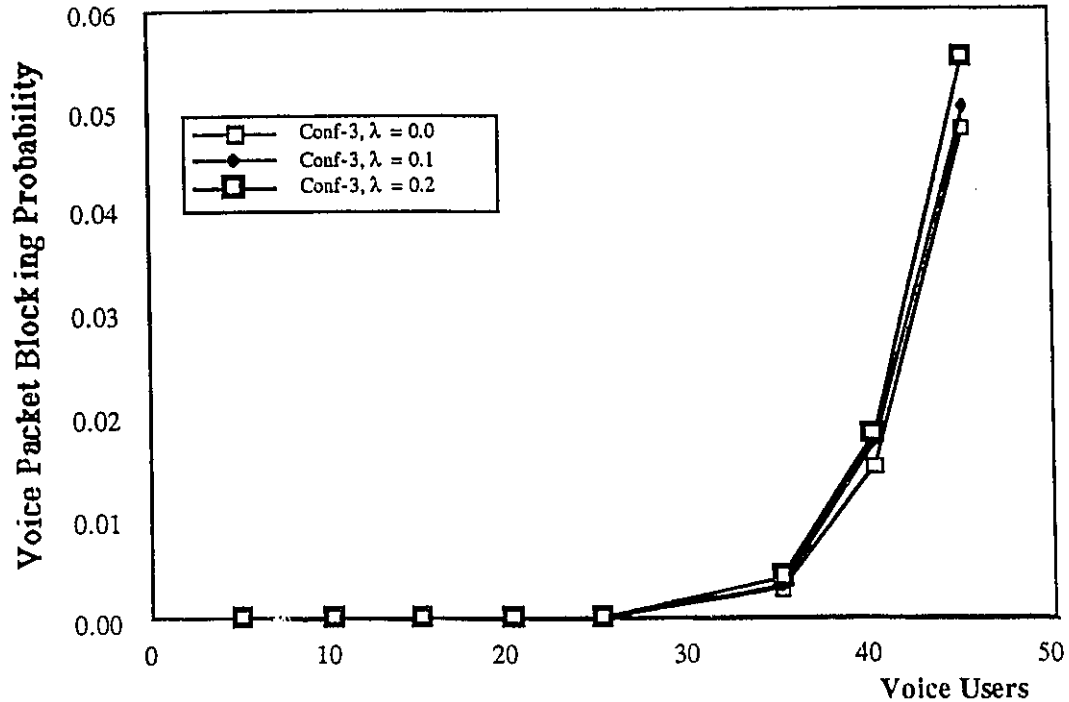


Figure 3.6 Voice Packet Blocking Probability (Configuration 3)

able to support at least 36 voice users, with a 0.01 packet loss rate. Conf-2 and Conf-3 go up to 38 voice users.

Fig. 3.7 represents the mean delay of data packets vs. number of voice users for each one of the three configurations and for $\lambda = 0.1, 0.2$. Also, Fig. 3.8 shows the percentage of blocked data packets vs. the number of voice users. For $\lambda = 0.1$. Conf-1 and Conf-2 show stability up to 40 voice users. Conf-3 remains stable with up to 25 to 30 voice users in the channel. For $\lambda = 0.2$ Conf-1 appears stable for up to a 25 voice users. The other two configurations trail back, showing stability for up to 15 voice users. It is interesting to examine the blocking probability of data packets vs. number of voice users curves. The results demonstrate that the performance of the algorithm degrades quite gradually as the load increases. This indicates its strong robustness in operation under overloading conditions. As already mentioned, this behavior is very useful for PCNs, where a very dynamic and bursty data traffic is expected.

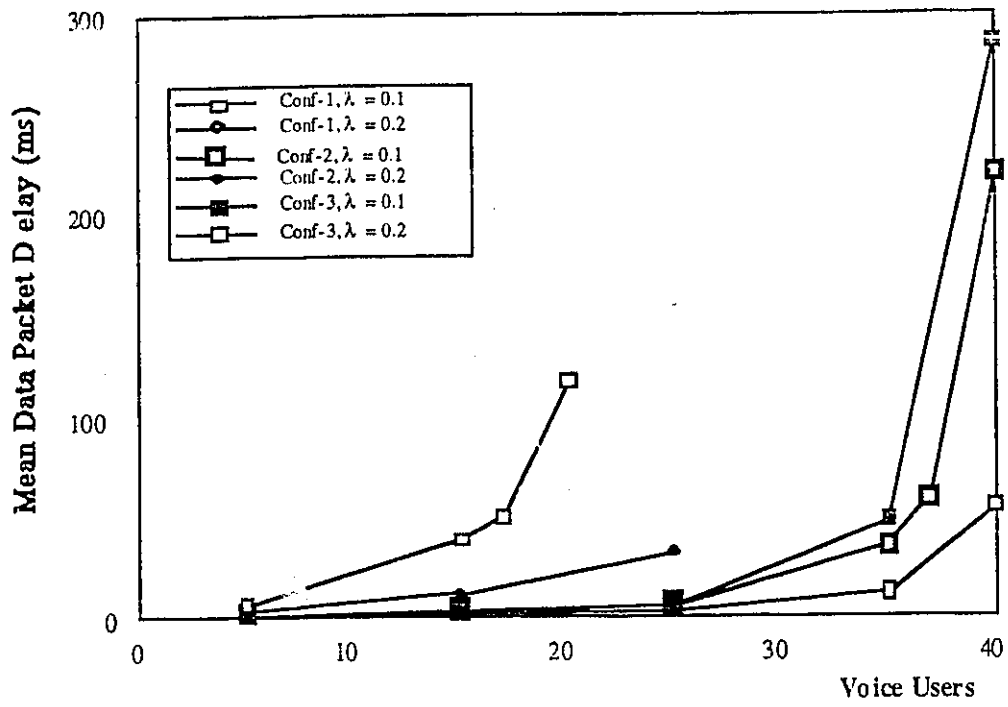


Figure 3.7 Mean Packet Delay (All Configurations)

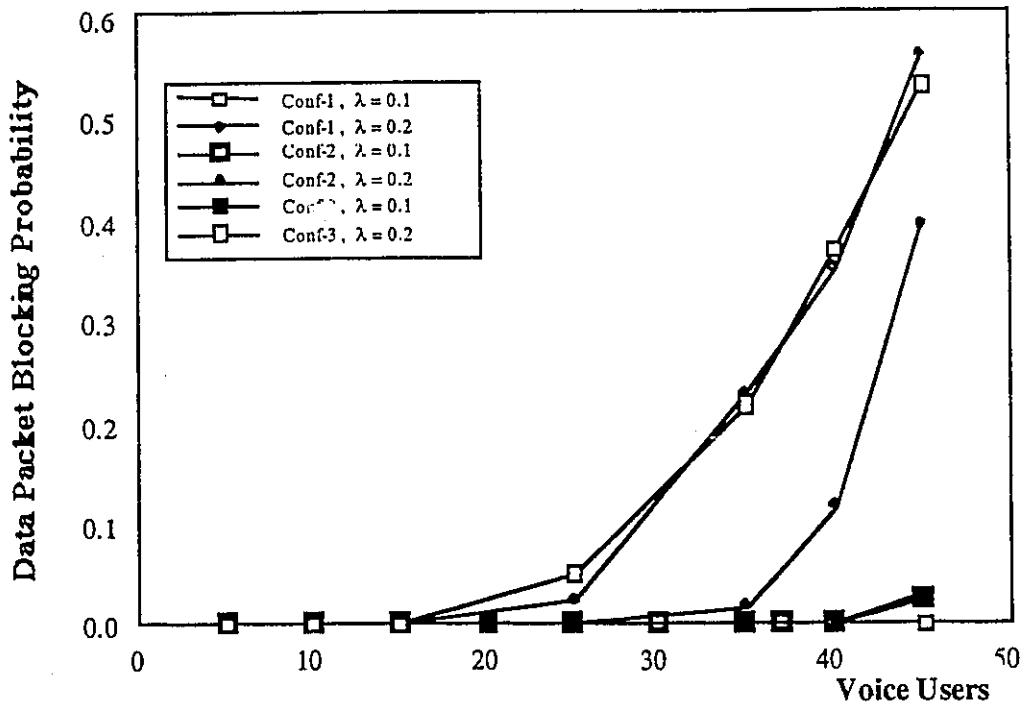


Figure 3.8 Data Packet Blocking Probability (All Configurations)

Figures 3.9, 3.10, 3.11 display the results for PRMA, for integrated voice and data traffic. In the analysis of PRMA in [Good89] simulation results were obtained only for voice traffic. In our analysis we drop data packets after 10 unsuccessful transmission attempts, to improve the unstable behavior of Slotted Aloha protocol. These results are obtained by using data transmission probability $P_D=0.1$ and voice transmission probability $P_V=0.3$ (same as described in [Good89]), in a free slot. Figure 3.9 display the blocking probability of voice packets vs. number of voice users, for data traffic loads of $\lambda = 0, 0.1$ and 0.2 packets per slot. Figures 3.10 show the data packet blocking probability for data loads $\lambda = 0.1$ and $\lambda = 0.2$. Figure 3.11 represents the mean delay experienced by successfully transmitted data packets.

PRMA has comparable performance when there is no data load ($\lambda = 0$). However, when data comes into the picture, its performance degrades rapidly, due to the instability of the Aloha protocol. For data loads $\lambda = 0.1$ and $\lambda = 0.2$, its performance is quite low. These data packets compete together with the voice users for the same slots. Even if we set the transmission probability of the data packets to considerably lower levels as compared to voice packets, the loading generated by data packets is considerable and capable to affect quite negatively the performance of PRMA. Table 3.1 and Table 3.2 show the performance of proposed protocols as compared to PRMA.

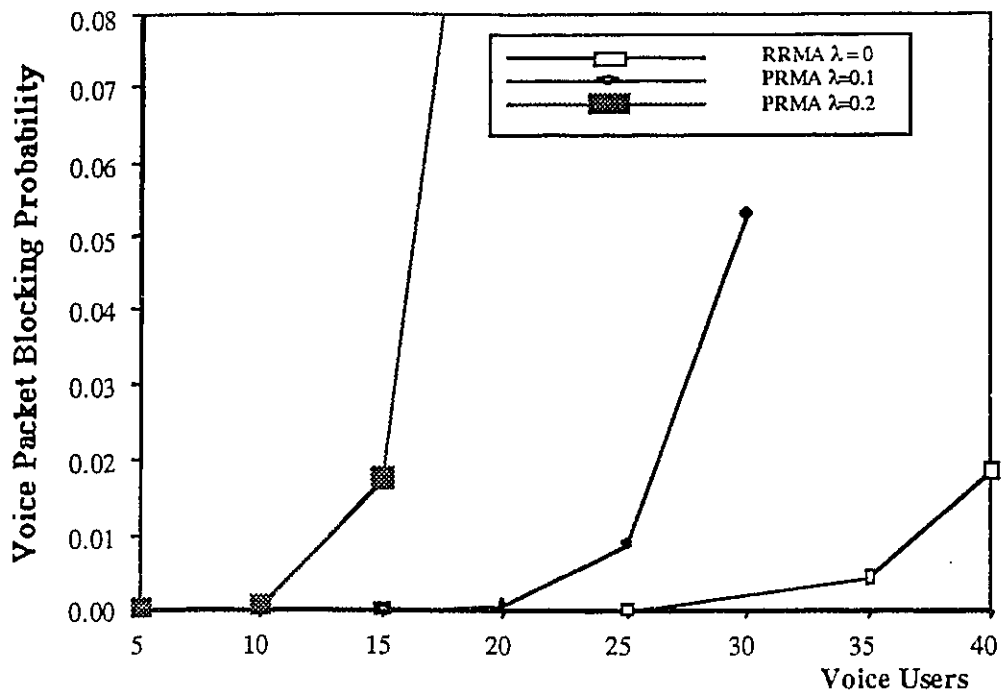


Figure 3.9 Voice Packet Blocking Probability (PRMA)

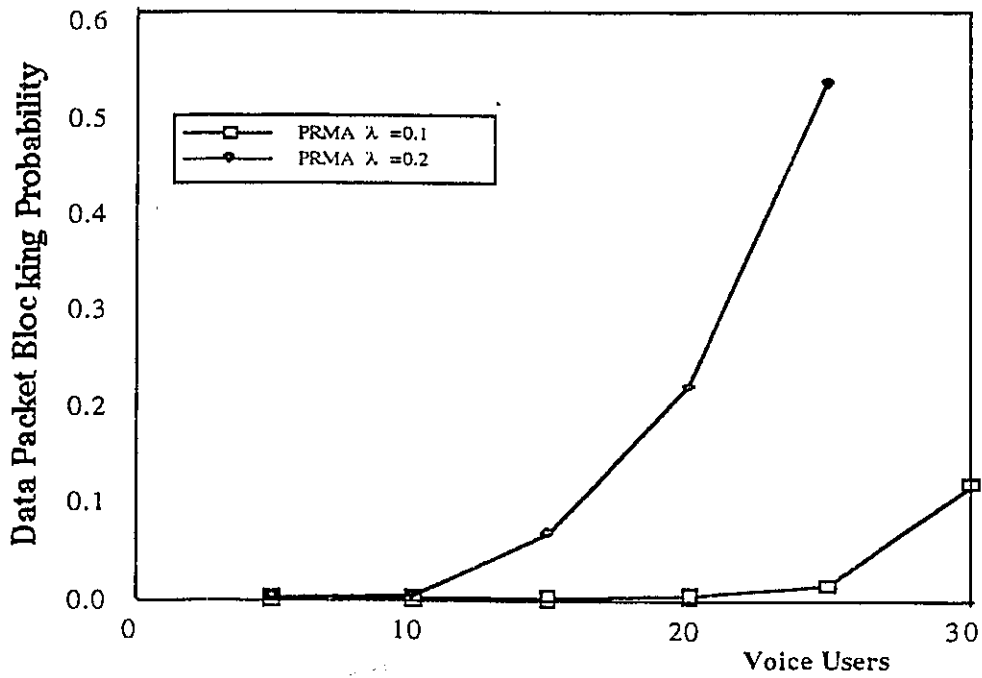


Figure 3.10 Data Packet Blocking Probability (PRMA)

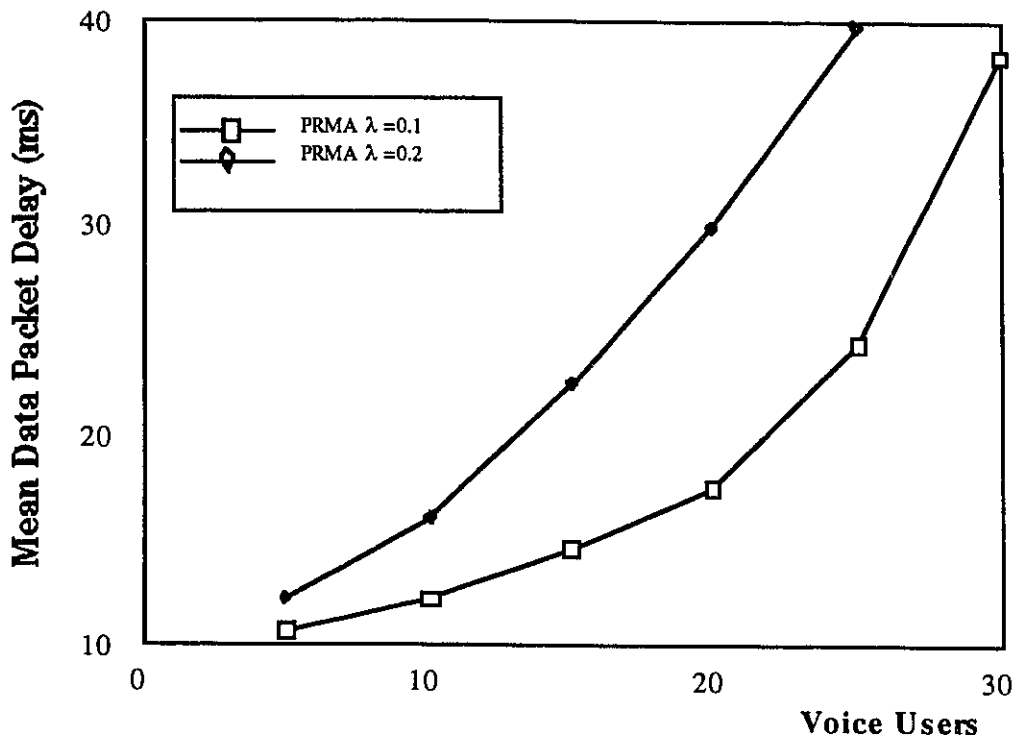


Figure 3.11 Mean Packet Delay (PRMA)

Table 3.1 Comparison of the PRMA and the Proposed Protocol (Config 1)

Data Load (packet/slot)	Proposed Protocol (Config.1)		PRMA	
	No. of Voice Users	Percentage of Data Packets Blocked	No. of Voice Users	Percentage of Data Packets Blocked
0.0	36	N/A	37	N/A
0.1	36	0.0	25	1.49
0.2	35	3.0	14	6.23

Before closing the chapter it is worth to mention that in [Jang94], a performance analysis of several Reservation Random Access protocols when applied in a network with

voice users has been performed. However, that work deals only with voice traffic and examines the performance of existing protocols by assuming that the transitions of voice users from silence to talkstart occur only at the beginning of a frame. In the present work we propose some new approaches for protocol design and deal not only with voice, but with integrated environment: voice/data. Furthermore, in our model for voice source, we have not made any assumption constraining the time when the various voice sources become active or inactive reflecting a more realistic situation.

Table 3.2 Comparison of the PRMA and the Proposed Protocol (Config. 2 & 3)

Data Load (packet/slot)	Proposed Protocols (Config. 2 and Config. 3)		PRMA	
	No. of Voice Users	Percentage of Data Packets Blocked	No. of Voice Users	Percentage of Data Packets Blocked
0.0	38	N/A	37	N/A
0.1	38	0.0	25	1.49
0.2	37	28.8	14	6.23

3.6 Conclusions

Simulation analysis of reservation random access protocols suitable for personal communication networks carrying integrated traffic is presented in this chapter. Z-cell algorithms provide an efficient statistical multiplexing for voice packets. Z-cell algorithm is robust and stable for different data loads integrated with voice. Furthermore, it is simple to implement and it can be enhanced easily to multiplex other traffic categories. The performance analysis of these protocols under various traffic intensities and patterns demonstrates the ability of these protocols to provide the quality of service required by different services.

Chapter 4

Collision Resolution Channel Access Scheme for CDMA Networks

4.1 Introduction

The future success of the personal communications service will depend on its ability to accommodate efficiently integrated traffic and service a variety of applications and communication sources with different quality of service requirements. As described in previous chapters efficient algorithms are needed to support integrated services in the personal and mobile communications environment.

Code Division Multiple Access (CDMA) is quite popular in the personal communications field [Gilh91], [Schi91], due to its robustness to several serious impairments of the wireless personal communications channel and system (multipath fading, interference, frequency selectivity, synchronization errors), its ability to take advantage of the voice activity factor with relatively simple hardware, and the higher peak power efficiency it demonstrates as compared to Time Division Multiple Access (TDMA). In addition to that, CDMA networks can share the same bandwidth with other narrowband signals, which allows for a more efficient use of the available radio frequencies spectrum. Presently, several hybrid networks considering the simultaneous operation of narrowband and CDMA systems over the same frequency bandwidth have been proposed [LinC93].

CDMA is based on spread spectrum, a well-established technology that has been primarily used over the years in military communications. It is only recently that CDMA and spread spectrum in general started seeing use in commercial applications. Presently, CDMA is used in portable telephone and has a very good chance to be used in digital cellular radio communications and other advanced wireless technologies. Its ability to co-exist with other narrowband applications will help to resolve the near-term capacity concerns of the wireless market and will assist the industry's long-term need for economic, efficient, and truly portable communications.

In [Makr92], [Makr93], a simple version of a CDMA based channel access scheme called Spread Slotted Aloha (SSA) has been examined. The analysis showed that the SSA network is able to perform well while its overall structure remains simple. However, similarly to other proposed schemes using CDMA technology, SSA has not taken into consideration the possibility of servicing more than one type of traffic. Furthermore, SSA uses the Slotted Aloha (SA) protocol to handle the competition between packets. SA, though simple, it becomes unstable when dealing with large user populations and highly dynamic traffic loads. A scenario that is expected to be seen in future PCNs.

In this chapter, we propose a number of modifications to SSA. First, we introduce a multi-priority mechanism into the structure of the protocol, in order to be able to accommodate non-homogeneous traffic. Second, we replace SA with a stable collision resolution algorithm (CRA) to handle the collisions occurring between competing packets. Several studies have appeared in the literature, dealing with the application of stable collision resolution algorithms to CDMA [Chih90]. However, these studies do not consider integrated traffic, neither they have included in the protocol the ability to service users and applications requiring different priorities. It is worth mentioning that in the proposed scheme priority is controlled exclusively by the users themselves, thus, there is no need for central control and intervention of the network. This reduces considerably the complexity of the procedure to be implemented in the Base Station (BS). In addition to that, the CRA

proposed in this work is different from algorithms examined in the past and has several advantages in terms of practical value and implementation efficiency.

Third, we have included the reservation option in SSA and we do the same for the proposed SSRA protocol as well. This is similar to work done in several access schemes based on TDM technology [Good89], [Mitr90] making use of a reservation policy to accommodate users which generate packets periodically when in the active state.

In this chapter, we assume errorless communications. For the remainder, this chapter is organized as follows. Section 4.2 gives the brief overview of the system architecture. Section 4.3 describes the priority mechanism that is incorporated into the protocol. Sections 4.4 and 4.5 describe the SSA and the proposed SSRA protocol respectively. Section 4.6 deals with performance issues. Finally, section 4.7 summarizes and concludes this chapter.

4.2. Description of the System

As explained in chapter 2 mobile and portable terminals located within a cell communicate their information to a BS. After receiving the information, the BS is responsible for directing it (through the wired and wireless network) to its destination. We define as user an application supported by a terminal. The same terminal can support more than one applications e.g. voice, data, video etc. The network perceives each application as different user. Consequently, the same terminal might have to support more than one user. We define active user as a user with a packet ready for transmission.

We consider a time-slotted channel with the size of the slot being equal to the size of the packet. We further assume that the network is allocated a total of N_c spreading codewords. In the present work we assume that a packet transmission attempt fails (thus the base station returns a C response through the Feedback Channel (FC)) if either or both of the following two conditions occurs: i) there is at least one more user that transmits during the slot using the same codeword; ii) a total of more than N_{II} users attempt

transmission during the slot. The second condition applies due to the interference limitation to the network. In a CDMA system users generate interference to each other [Lehn87]. The interference level increases as the number of users becomes large. We assume that if N_{IL} users transmit simultaneously, the interference increases to a level where the Bit Error Rate (BER) becomes unacceptable high. N_{IL} depends on the spreading gain, the operating signal to noise ratio (SNR), the requirements of the network, the type of spreading codewords, the synchronization properties of the transceivers and the quality of service of the different applications in terms of bit error rate (BER).

The underlying reservation scheme described here is similar to that used in chapter 3. When a user becomes active, he starts competing to gain access to the channel. He will persist until he succeeds in accessing the channel or he withdraws for reasons associated with the nature of the application or the operation of the system, i.e., due to the aging of the packet or because of exceeding the maximum allowable number of attempts. Reception of the first successful packet initiates a process that marks certain slots as reserved (R) and makes them available to the user. The reservation policy eliminates the need of competition for a large percentage of packets, thus it reduces the number of slots that end up being wasted due to collisions. When a user becomes inactive, the slot is freed and available for competition. Under the non-reservation scheme the users enter the competition process for every single packet to transmit.

In this chapter we evaluate different versions of medium access schemes based on SSA and SSRA protocols. Before explaining the different schemes under study, we explain how priority is allocated to the users of different classes and how the priority mechanism is incorporated into the protocols.

4.3. Description of the Priority Mechanism

Let us assume that the network accommodates M_g different classes of users. The segregation of the total user population to classes is based on the type of application supported by the user and/or the nature of the user. We define a number of N_c groups, G_1, G_2, \dots, G_{N_c} . Each group corresponds to a different level of priority. The groups are ordered in an upwards increasing priority level with G_1 having the lowest priority and G_{N_c} the highest one. The system does not have to implement all the groups i.e. M_g could be less than N_c . An active user joins one of the groups, depending on the application the user supports. If the user supports more than one application at the same time, i.e. voice and data or voice, data and video, the user joins one group for each individual application it services. Let us assume that a user belonging to the G_m group ($1 \leq m \leq N_c$), becomes active during slot $k-1$. The user accesses the network by choosing randomly m out of the N_c available codewords as shown in figure 4.1. The user follows the evolution of the competition processes taking place in the m codewords.

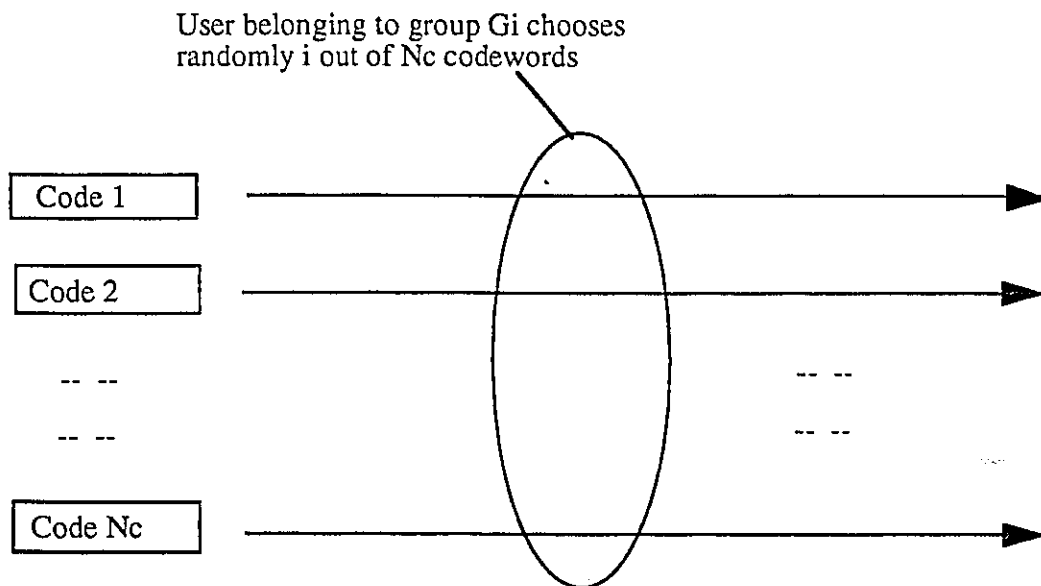


Figure 4.1 Priority of using codewords

The user follows the competition process by monitoring the information coming through the feedback channel. This allows the user to identify the earliest time to join one

of the m competition processes. The rules that qualify a user to join a competition process depend on the scheme being used to arbitrate conflicts in accessing the medium., i.e. SSA or SSRA. These schemes are explained in the following sections. In several cases, the user might be qualifying to join the competition process of more than one codeword. In this case, the user chooses randomly and equally probably one of these codewords. It should be clear at this point that users with higher priority are authorized to access more codewords and join the one that allows them to start competing at the earliest possible time. Notice that this priority mechanism is totally controlled by the users themselves, without any intervention from the network.

4.4. Spread Slotted Aloha with Priority

The operation of the SSA with priority is as follows. Let us assume that a user which belongs to the m^{th} priority class is active during the $(k-1)$ slot (k being an integer). Let us also assume that L out of the N_c codewords have already been reserved at the beginning of the k slot. The user decides to transmit the packet at the beginning of the k slot with probability $P(m, l, \Delta\tau_g)$, where m indicates the user class, l the number of times this packet was involved in failed transmission attempts in the past and $\Delta\tau_g$ the number of slots elapsing between the moment the packet was generated and the current slot. As mentioned earlier, in several applications a packet becomes useless if it arrives at its destination later than a certain time after its generation. In this case, if $\Delta\tau_g$ grows over a certain bound (its value depending on the application) the packet is withdrawn, i.e., $P(m, l, \Delta\tau_g) = 0$. Similarly, the protocol strategy could be such that a packet is withdrawn if it exceeds a number of failed attempts. A large number of consecutive collisions indicates that the network is overloaded. If the user decides in favor of transmitting during the k slot, he chooses randomly (and equally probably) $n = \min \{m, N_c - L\}$ out of the $(N_c - L)$ available codewords and afterwards he selects and uses one of them for spreading its

signal. The user receives through the FC information about the outcome of its attempt to transmit. If a Collision (C) is received, the user repeats at slot $k + 1$ the process described earlier.

In this work, we study three different medium access schemes based on SSA when supporting voice and data services. In the first scheme, we will be referring to this scheme as SSA/Conf-1, only the first voice packet in a talkspurt goes through competition process. The codeword and slot are reserved for the whole talkspurt. For this configuration, SSA/Conf-1, the codeword selection policy applies as follows. Voice users, when in active state and before committing themselves to a specific codeword, examine which codewords have unreserved slots and they randomly choose one of these unreserved slots. They repeat this procedure every time a competing voice packet attempts to transmit. On the contrary, data users commit themselves to a particular codeword and keep using this codeword until either, the packet has being successfully transmitted or it is withdrawn.

In the case of SSA without reservation, we consider two different schemes. In both of these schemes all the voice packets go through competition process. In the first, referred as SSA/Conf-2, the codeword selection policy is same as in Conf-1. A voice user chooses at random a codeword every time he has a packet available for transmission. A data user commits to a codeword at the first attempt and keeps it until either the packet goes through or is rejected. In the second scheme, both voice and data users choose a codeword at the beginning of their active period, and keep using it until the end, i.e., the talkspurt finishes or the data packet goes through or is rejected. This scheme will be called the SSA/Conf-3.

4.5. Spread Slotted Random Access Protocol with Multi-Priority

The behavior and performance of the SSRA protocol is determined by the policies followed by the users on: i) how they select the codeword(s) they plan to use for spreading their

signal; ii) how a user decides to join a competition process; iii) the algorithmic process to be followed by a competing user. The procedure followed by the user for the choice of spreading codeword(s) was discussed in section 3. In the present section we shall address the second and third items.

In the present work, the collision resolution protocols chosen to replace SA is the 2-C algorithm and modifications of it [Geor82], [Geor85]. Modifications have been applied to incorporate additional priority mechanisms in the structure of the 2-C protocol. The 2-C algorithm is stable and provides relatively high throughput. Under limit Poisson traffic, it can accommodate traffic load of 0.429 packets/slot guaranteeing finite average packet delay. It can be deployed with a limited sensing policy. 2-C has been chosen over a number of other stable protocols (see for example [Cape79], [Gall78], [Tsyb80J], [Raph90], [Tsyb80D], [Tsyb78], [Tsyb80V], [Geor82], [Geor85]) for the following reasons: i) its simplicity of implementation; ii) it is robust to errors in the feedback channel [Pate89S]; iii) it allows the users to be able to identify the ends of competition processes by simply observing the feedback channel. While several other protocols allow users to enter the competition process right away (after they become active), 2-C prevents them from doing so. This forces a more orderly evolution of the competition process. More about 2-C algorithm is described in chapter 2. Below we describe the 2-C protocol with embedded priority.

Let us assume that a user that belongs to the m^{th} priority class has a packet to transmit but he is not currently involved in the competition process (we will refer to this kind of user as new user). Let us also assume that the user sees a NC-NC pattern, marking the $k-1$ and k slots. Instead of attempting transmission, the user waits for an additional $(M_g - m)$ slots (i.e. the user waits to see a pattern consisting of $(M_g - m + 2)$ NC before it attempts a transmission). If a collision occurs before the user identifies $(M_g - m + 2)$ consecutive NC, it starts counting from the beginning (right after the slot marked as C). This waiting ensures that no users of higher priority want to use the network

resources. We call this policy Class Blocking (CB). A second level of priority can be included as follows. If the user joins a competition process and right after the first attempt it ends up with a **C**, it refrains from re-transmitting for $(M_g - m + 1)$ slots. If the **FC** marks these slots as **NC**, the user joins again and executes the CRA which will send eventually the packet through the network. If a **C** appears in one of these slots, the user refrains from further attempts, returns back to the idle mode and starts the process from the beginning. We call this strategy Double Class Blocking (DCB) policy. CB and DCB affect the priority by giving to users with higher priority the opportunity to join and form competition process earlier mainly among themselves

SSRA has been examined with and without reservation policies as well. In both cases, we examine configurations with and without CB policy. In this chapter we will consider four different configurations of SSRA. SSRA with reservation and CB policy (SSRA/Conf-1). SSRA with reservation but without CB policy (SSRA/Conf-2). SSRA without reservation but with CB policy (SSRA/Conf-3). SSRA without reservation and without CB policy (SSRA/Conf-4).

4.6 Performance Evaluation Results

We evaluate the SSA and SSRA protocols when transporting voice and data traffic. We assume that voice terminals are equipped with *fast* Voice Activity Detectors (VADs). These VADs are able to distinguish between mini-talkspurts and mini-silence periods that reside within a talkspurt. The traffic generated by a voice user equipped with a fast VAD can be described by a 3-state Markovian model [Good91]. The average duration of the silence and talkspurt periods have been set equal to 1.35 sec. and 1 sec. respectively. The average duration of mini-silences and mini-talkspurts, occurring within a talkspurt are set equal to 50 msec. and 0.275 sec. respectively.

Since digital voice is relatively insensitive to channel errors and packet losses, but highly sensitive to delays, packets should satisfy a certain delay bound in inter packet arrival distance. If a packet violates this limit, it becomes obsolete and the transmitter should discard it in favor of a more recent produced packet. In the present work, we assume that the maximum tolerable delay for voice packets is 32 ms, and the maximum acceptable packet dropping rate is 1%.

Data packets should be delivered to the destination error free, however, data traffic is insensitive to delays. We assume that the aggregate data traffic follows a Poisson distribution. This assumption is based on our conviction that in the future, we will see a large number of data terminals supporting non-real time applications (i.e. paging, e-mail etc.), each one producing a small traffic activity. Poisson traffic represents quite accurately the aggregate traffic produced by a large number of low activity users.

The simulated SSA and SSRA configurations use $N_c = 4$ codewords and $N_{IL} = 10$. Higher values of N_{IL} do not provide any substantial improvement. Voice users are allocated to G_4 and data users to G_1 . The duration of every slot is being set equal to 1.6 msec, thus a voice packet becomes obsolete 20 slots after the slot it was generated (i.e. for voice, $\Delta\tau_g = 20$). We investigated the performance of the various SSA and SSRA configurations by varying the number of voice users and data loads $\lambda = 0, 0.1$ and 0.2 packets/slot.

In section 4.4 we described three different versions of SSA (Conf-1 to Conf-3). For SSA, the following values of $P(1, m, \Delta\tau_g)$ (for data) and $P(4, m, \Delta\tau_g)$ (for voice) are used:

$$P(1, m, \Delta\tau_g) \begin{cases} 1 & \text{for } m = 0 \\ 0.02 & \text{for } 1 \leq m \leq 10 \\ 0 & \text{for } m \geq 10 \end{cases} \quad (4.1)$$

The transmission strategy described by Eqn. 1 is used (by data users) with all three examined data loads, i.e. $\lambda = 0, 0.1, 0.2$. For voice,

$$P(4, m, \Delta\tau_g) = \begin{cases} 1 & \text{for } m = 0 \text{ and } \Delta\tau_g \leq 20 \\ P_{rep} & \text{for } m \neq 0 \text{ and } \Delta\tau_g \leq 20 \\ 0 & \text{for } \Delta\tau_g > 20 \end{cases} \quad (4.2)$$

Where P_{ret} is the re-transmission probability of collided voice packets. For $\lambda = 0$ and $\lambda = 0.1$, all the SSA configurations (SSA/Conf-1 to -3) use $P_{ret} = 0.25$. For $\lambda = 0.2$ and when reservation policy is used (SSA/Conf-1) $P_{ret} = 0.15$. For $\lambda = 0.2$ and in the absence of reservation policy (SSA/Conf-2 and SSA/Conf -3) $P_{ret} = 0.5$. These values have been chosen after thorough investigation of the influence of $P(1, m, \Delta\tau_g)$, $P(4, m, \Delta\tau_g)$ on the performance of the various SSA configurations.

In this section we will provide results for the four SSRA configurations, described in section 4.5. In the reservation schemes, the data users use a window size $d_w(1)$ equal to 5 slots. In the non-reservation schemes, $d_w(1) = 2.5$ slots. In all the examined cases of the SSRA protocol, no windowing is applied to the voice users, however, voice packets are dropped if they have not been transmitted successfully within a time frame of twenty slots from the time they were generated. When CB policy is not applied, both, voice and data users are allowed to join the new competition process right after they have identified the termination of the previous one. Users detect the termination of a competition process by identifying two consecutive NC through the FC (see section 4.5). When CB policy is applied, voice users are allowed to transmit after two NC, while data users have to wait for three NC before attempting transmission.

The results concerning the performance of the various SSA and SSRA configurations are shown in Figs. 4.2 to 4.15. The results have been obtained through computer simulations. The simulations were developed using OPNET [OPNT93]. Figs. 4.2, 4.3 and 4.4 display the dropping rate of voice packets versus the number of voice

users in the channel, for data loads $\lambda = 0, 0.1$ and 0.2 respectively. Figs. 4.5 and 4.6 display the average delay experienced by data packets versus the number

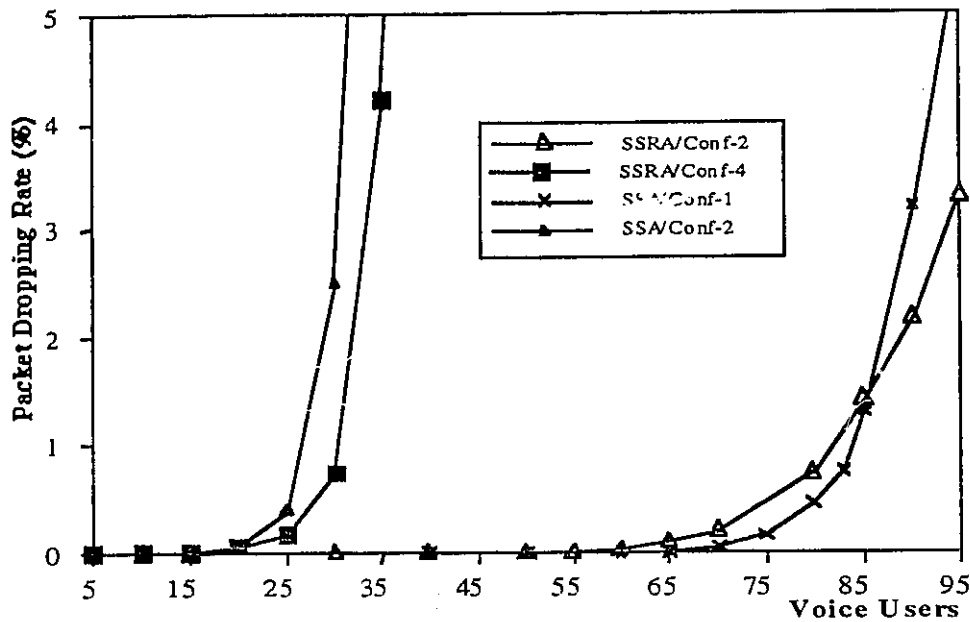


Fig. 4.2: Dropping rate of voice packets vs number of voice users. The data load is equal to 0.

of voice users, for data loads $\lambda = 0.1$ and 0.2 . Figs. 4.7 and 4.8 provide the blocking rate of the data packets versus the number of voice users, for $\lambda = 0.1$ and 0.2 respectively. Figs. 4.9, 4.10 show the channel utilization achieved by the various SSRA and SSA configurations for data loads $\lambda = 0.1, 0.2$. From the nine curves, the following points are evident.

SSRA is evidently superior to SSA in all accounts. For a 1% dropping rate of voice packets, SSA with reservation (SSA/Conf-1) accommodates 61 voice users when the data load is $\lambda = 0.1$ and 34 users for $\lambda = 0.2$. For $\lambda = 0.1$, SSRA/Conf-1 and SSRA/Conf-2 accommodate 70 voice users (a 15% improvement). For $\lambda = 0.2$, SSRA/Conf-1 accommodates 61 voice users (a 79.4% improvement over SSA/Conf-1). SSRA/Conf-2 accommodates 54 voice users (a 58.8% improvement over SSA/Conf-1). Notice that while for $\lambda = 0.1$, use of the CB policy gives practically no advantage (both SSRA/Conf-1 and

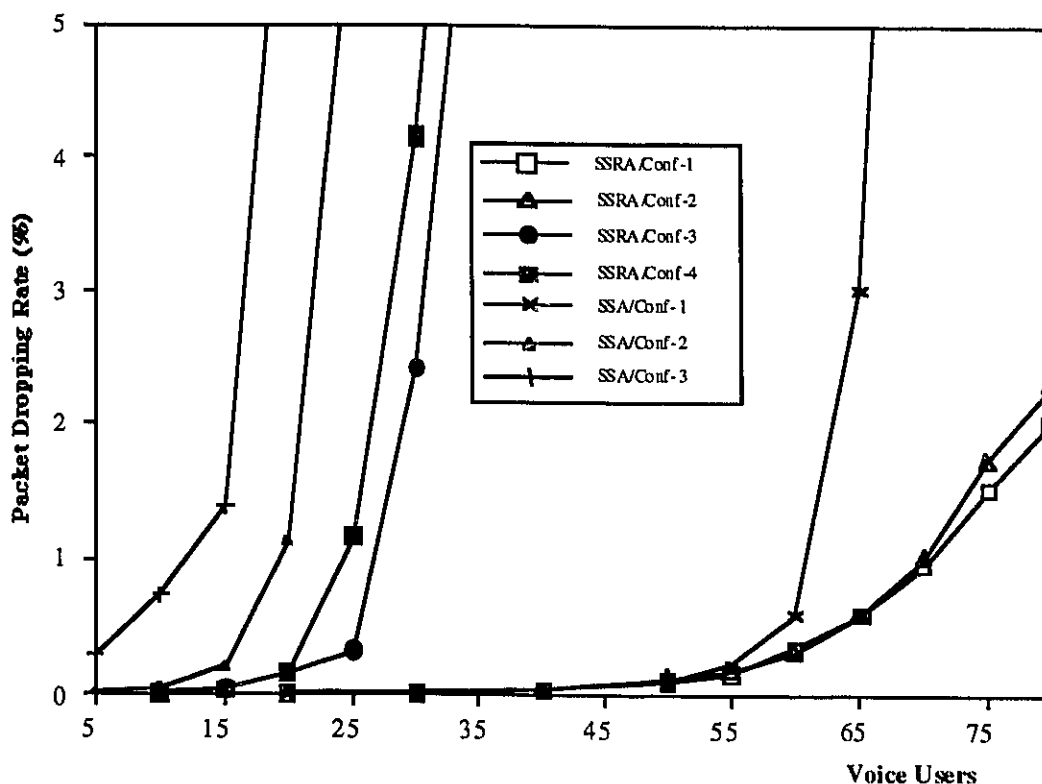


Fig. 4.3 : Dropping rate of voice packets vs number of voice users.
The data load is equal to 0.1 packets/slot.

SSRA/Conf-2 accommodate 70 voice users), for $\lambda = 0.2$, CB policy offers a considerable advantage. SSRA/Conf-1 accommodates 7 more voices users as compared to SSRA/Conf-2, i.e. 61 as compared to 54 (a 12% improvement). The benefits offered by CB increase as the data load λ becomes higher. For the non-reservation schemes the numbers stand as follows. When $\lambda = 0.1$, SSA/Conf-2 can accommodate up to 18 voice users without having the packet dropping rate exceed the 1% bound. SSA/Conf-3 accommodates only 11 users. When the data load is raised to $\lambda = 0.2$, these values become 11 users for SSA/Conf-2 and 2 users for SSA/Conf-3. For $\lambda = 0.1$, the non-reservation SSRA schemes achieve the following levels. SSRA/Conf-3 can accommodate up to 28 voice users (an improvement of 55% over SSA/Conf-2). SSRA/Conf-4 accommodates up to 24 users (33% improvement over SSA/Conf-2). For $\lambda = 0.2$, these numbers become: SSRA/Conf-3 accommodates 24 users and SSRA/Conf-4 18 users. Compared to SSA, SSRA achieves

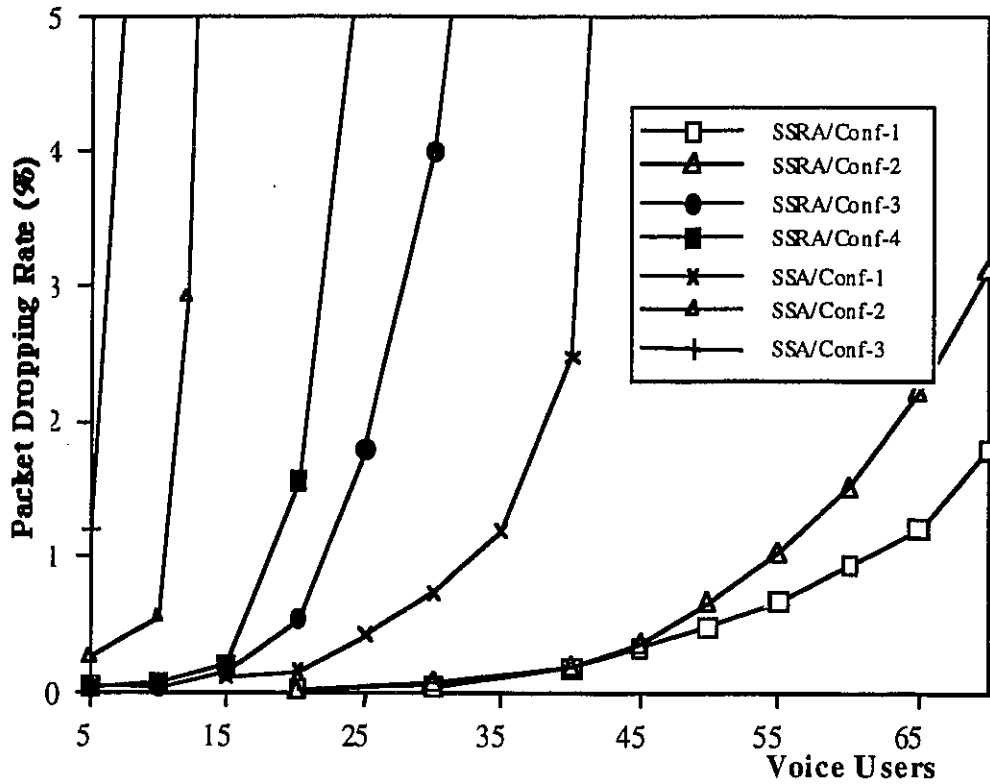


Fig. 4.4: Dropping rate of voice packets vs number of voice users. The data load is equal to 0.2 packets/slot.

higher level of channel utilization as well. This becomes evident from the curves of Figs. 4.10, 4.11.

It is evident from Figs. 4.5 to 4.8 that SSRA accommodates the data packets with less delay and achieves lower blocking probability as compared to SSA. The advantage of SSRA over SSA increases as the data load becomes higher. Comparison of the curves corresponding to SSRA configurations with CB policy (SSRA/Conf-1 and SSRA/Conf-3) with those of SSRA configurations that do not use CB policy (SSRA/Conf-2 and SSRA/Conf-4) shows that CB policy penalizes the data traffic. This is expected, since CB policy forces data to withhold transmission in favor of voice packets.

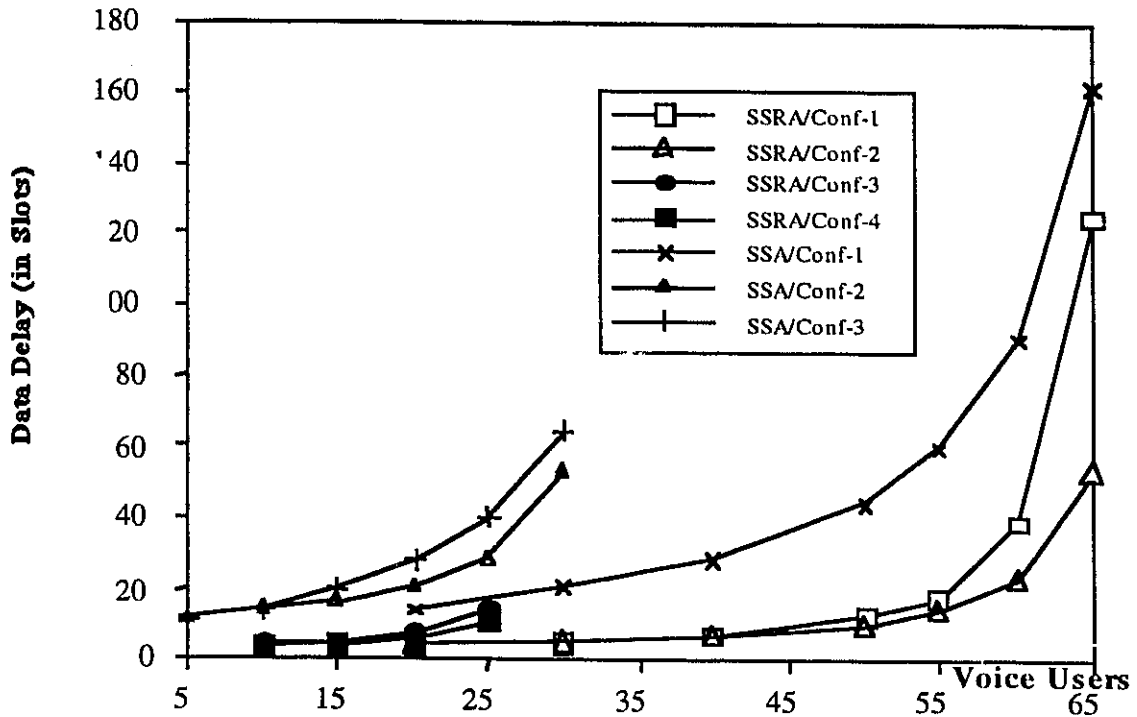


Fig. 4.5: Delay of data packets vs number of voice users. The data load is equal to 0.1 packets/slot.

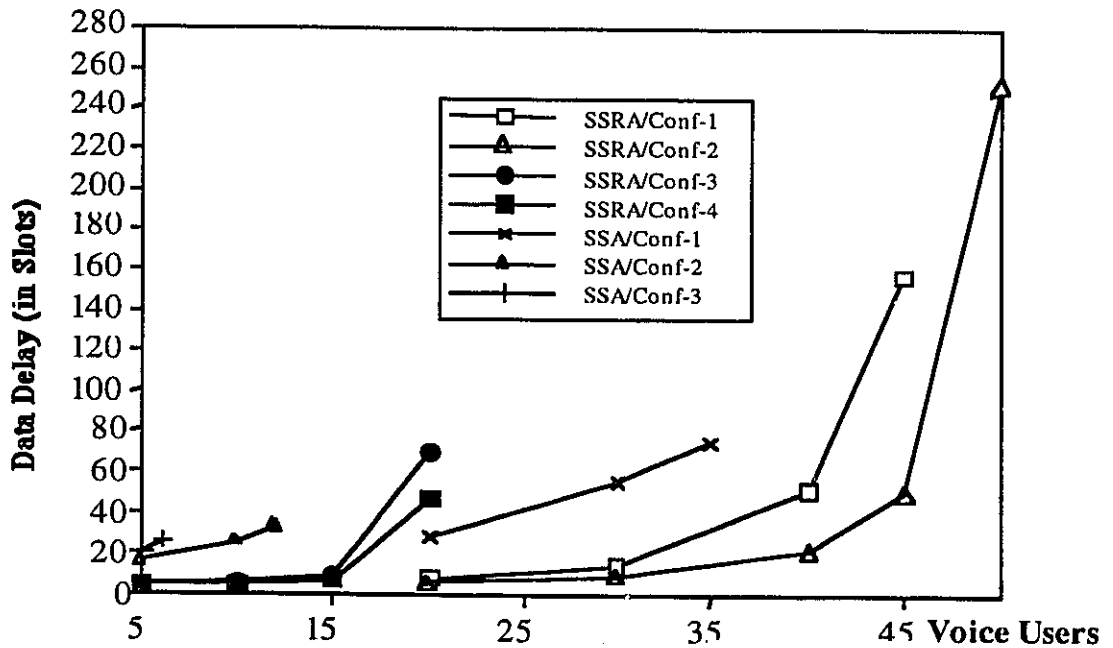


Fig. 4.6: Delay of data packets vs number of voice users. The data load is equal to 0.2 packets/slot.

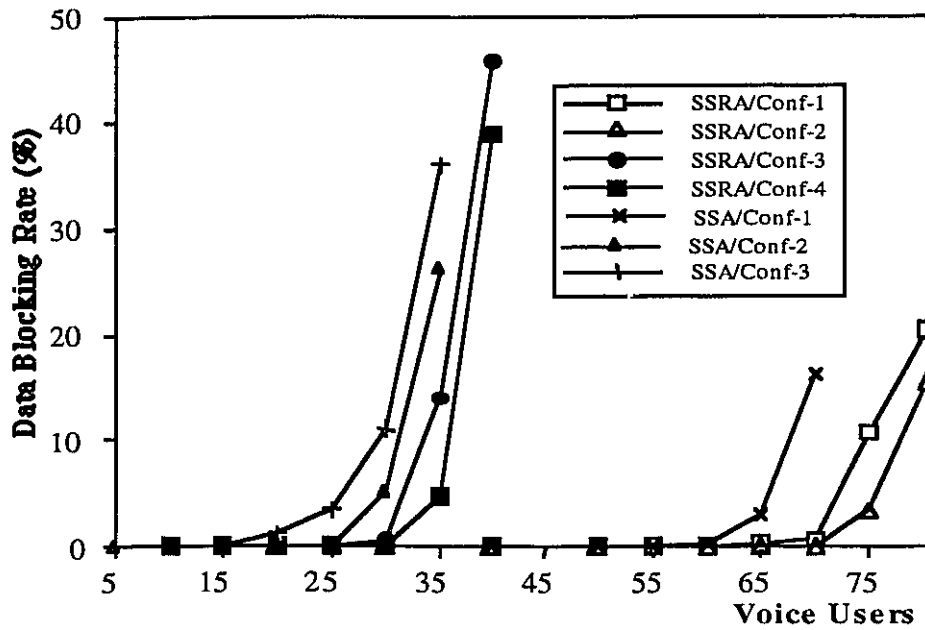


Fig. 4.7: Blocking rate of data packets vs number of voice users. The data load is equal to 0.1 packets/slot.

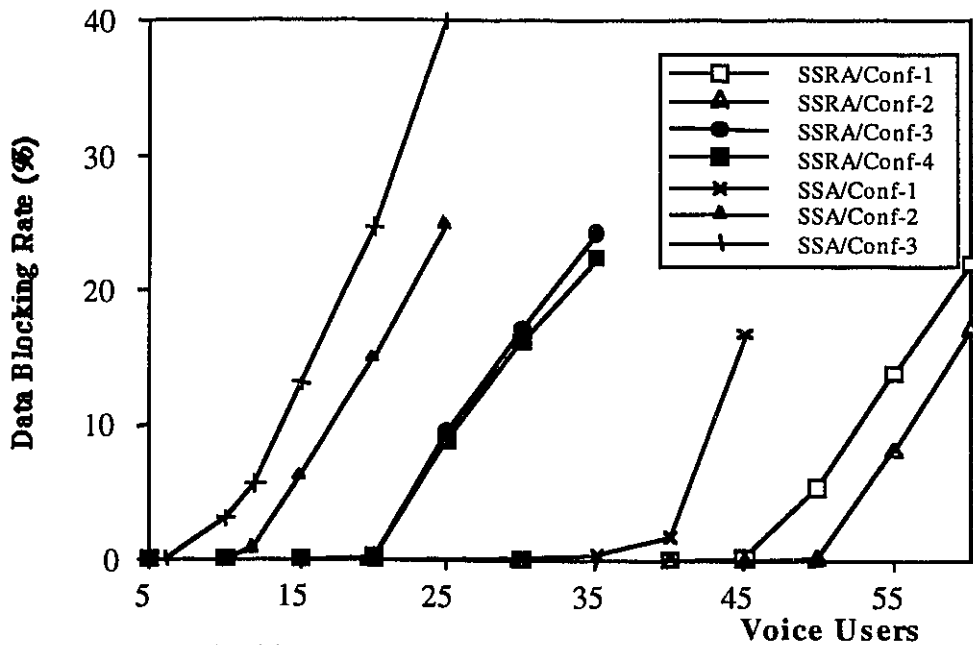


Fig. 4.8: Blocking rate of data packets vs number of voice users. The data load is equal to 0.2 packets/slot.

Reservation improves considerably the performance of both, the SSA and SSRA schemes. For data load $\lambda = 0.1$ and voice packet dropping rate of 1%, SSRA/Conf-1 accommodates 70 voice users versus 28 voice users accommodated by SSRA/Conf-3 (an improvement of 150%). For $\lambda = 0.2$, these numbers become 61 versus 24 voice users respectively (an improvement of 154%). These improvements are due to the reservation policy followed by SSRA/Conf-1. Reservation policy is absent from SSRA/Conf-3. Similar advantages exist for SSA as well. For $\lambda = 0.1$, SSA/Conf-1 accommodates 61 users versus 18 users accommodated by SSA/Conf-3 (an increase of 221%). For $\lambda = 0.2$, the numbers become 34 versus 11 (an increase of 238%). The same conclusions can be reached by observing the channel utilization levels achieved by the various configurations (see Figs. 4.9 and 4.10). For $\lambda = 0.1$, SSRA shows a channel utilization close to 80% and SSA close to 67%. In the absence of reservation, the utilization of both schemes drops. Without reservation, SSRA reaches a maximum utilization close to 39% and SSA close to 28%. For $\lambda = 0.2$, the maximum utilization levels of SSRA and SSA with reservation become 71% and 51% respectively. Without reservation, these values drop to 38% (SSRA) and 28% (SSA).

Figs. 4.11 to 4.15 provide us with some idea as to how the various design parameters of the system (N_{IL} , data window size $d_w(1)$, voice transmission probability $P(4, m, \Delta\tau_g)$) affect the performance. Fig. 4.11 gives an idea of how the value of N_{IL} affects the dropping rate of voice packets and blocking rate of data packets. The curves displayed in Fig. 4.11 correspond to SSRA with reservation but without CB (SSRA/Conf-2). The number of voice users in the channel has been set equal to 70, whereas the data load is $\lambda = 0.1$. We see that all the improvement in terms of lowering the loss rate of voice and data packets is achieved for $N_{IL} = 8$. Further increase of N_{IL} offers

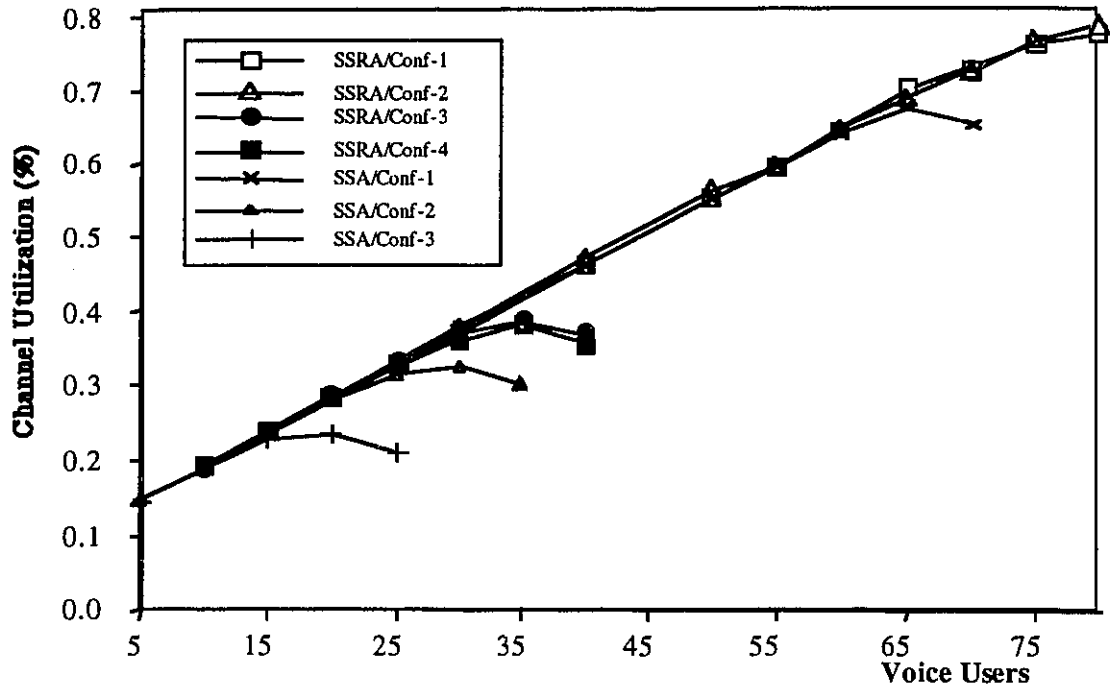


Fig. 4.9: Channel utilization vs number of voice users.
The data load is equal to 0.1 packets/slot.

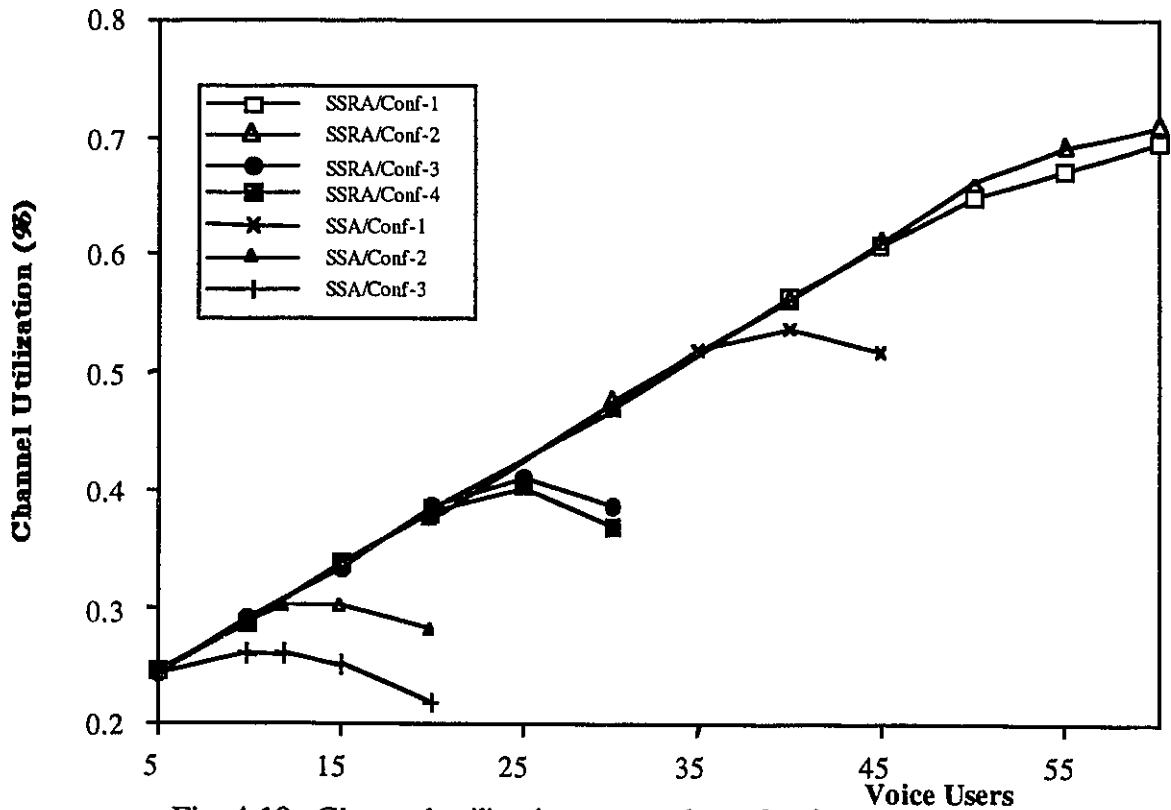


Fig. 4.10: Channel utilization vs number of voice users.
The data load is equal to 0.2 packets/slot.

only minimal additional improvements, that do not justify the additional complexity and/or waist of bandwidth that would result from an increase in the value of N_{IL} .

Figs. 4.12 and 4.13 display the dropping rate of voice packets and blocking rate of data packets versus the window size $d_w(1)$ used by the 2-C algorithm. The curves correspond to the four different configurations of SSRA described earlier (SSRA/Conf-1 to -4). All the results shown in these curves are for data load $\lambda = 0.2$. For the configurations with reservation (SSRA/Conf-1 & SSRA/Conf-2) we have assumed 50 voice users, whereas for the non-reservation configurations (SSRA/Conf-3 & SSRA/Conf-4) 23 voice users. From the displayed curves, it can be seen that for the reservation configurations

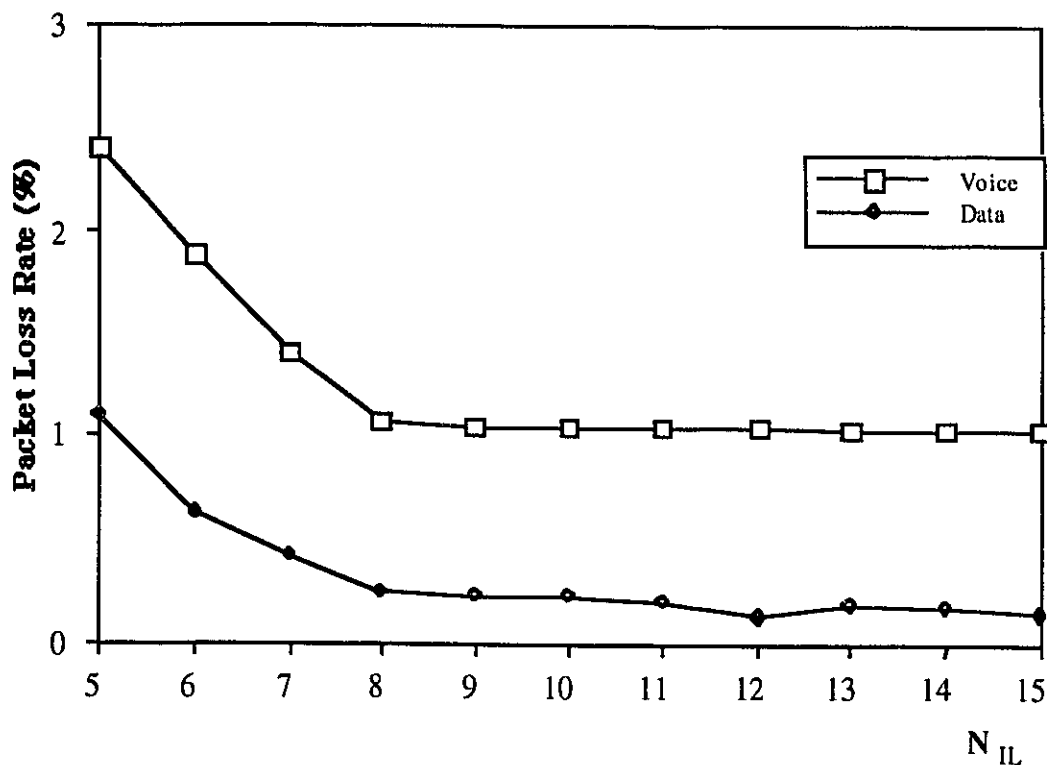


Fig. 4.11: Dropping rate of voice packets and blocking rate of data packets vs the maximum acceptable number of simultaneous transmissions. Number of voice users: 70. Data load: 0.1 packets/slot.

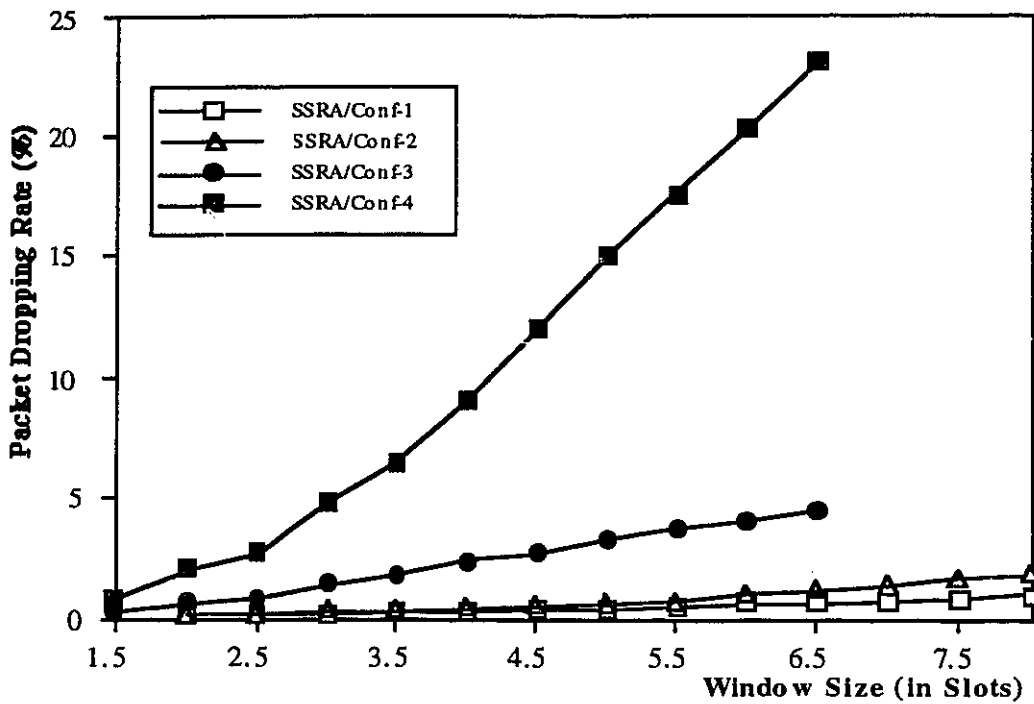


Fig. 4.12: Dropping rate of voice packets vs the window size.
The data load is equal to 0.2 packets/slot.

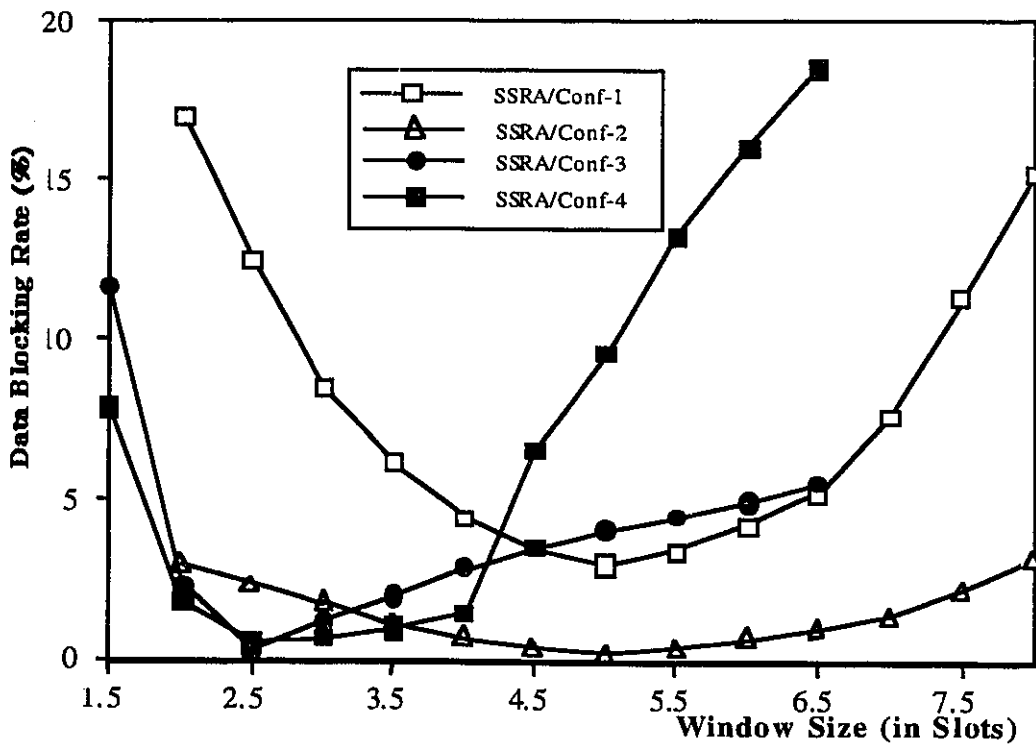


Fig. 4.13: Blocking rate of data packets vs the window size.
The data load is equal to 0.2 packets/slot.

(SSRA/Conf-1 & SSRA/Conf-2), the data packets experience the lowest blocking rate for $d_w(1)$ in the proximity of 5 slots. At the same time, for $d_w(1) = 5$, the dropping rate of voice packets remains below the 1% value that is required in order to sustain acceptable quality of voice communications. For the non-reservation configurations (SSRA/Conf-3 & SSRA/Conf-4), the data packets experience their lowest blocking rate for a window size $d_w(1)$ in the proximity of 2.5 slots.

Figs. 4.14, 4.15 display the dropping rate of voice packets (Fig. 4.14) and blocking rate of data packets (Fig. 4.15) versus the re-transmission probability of voice packets P_{ret} for SSA/Conf-1 and SSA/Conf-2. The curves appearing in these two figures,

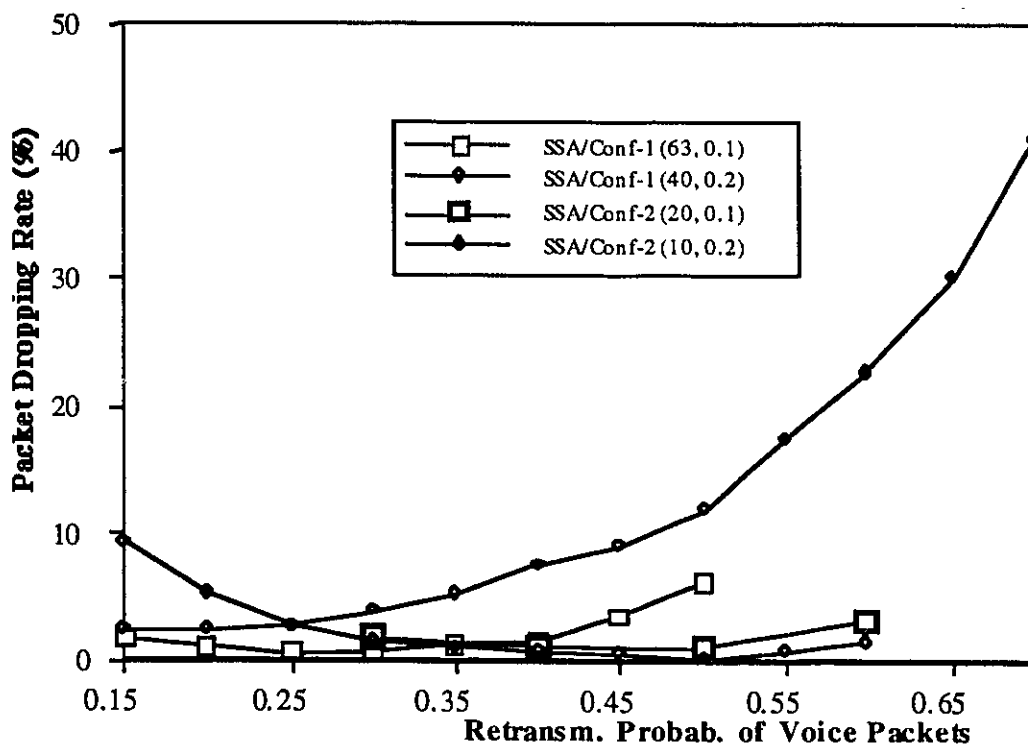


Fig. 4.14: Dropping rate of voice packets vs retransmission probability of voice packets.

are marked as SSA/Conf-1(μ, ξ), SSA/Conf-2(μ, ξ). μ refers to the number of voice users and ξ to the value of the data load which we used in the simulations that provided these curves. In all cases, the re-transmission probability of collided data packets has been set to

0.02. A data packet is blocked if it fails to go through after 9 re-transmissions. As can be seen, the SSA configuration with reservation (SSA/Conf-1) achieves the lowest dropping rate of voice packets for a re-transmission probability in the range of 0.15. The SSA configuration without reservation (SSA/Conf-2) achieves the lowest dropping rate of voice packets for re-transmission probability in the vicinity of 0.5.

The performance comparison of different versions of the SSRA and SSA protocols for data load $\lambda = 0.1$ and data load $\lambda = 0.2$ is shown in Table 4.1 and Table 4.2 respectively.

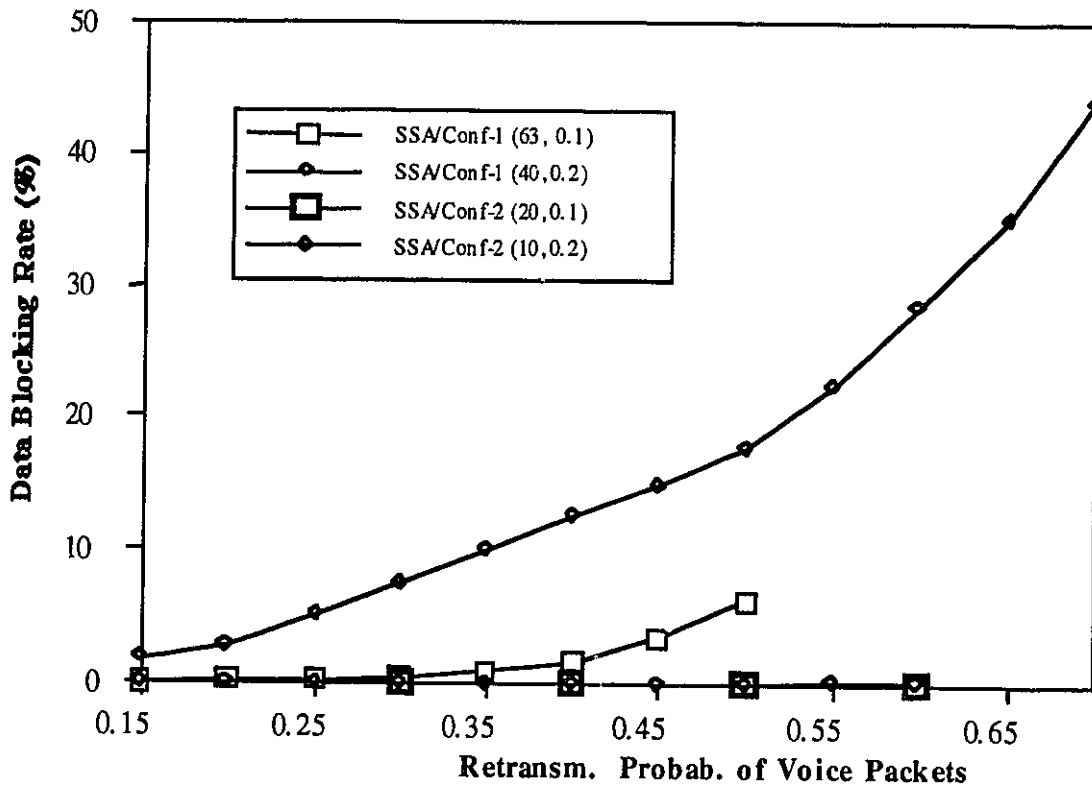


Fig. 4.15: Blocking rate of data packets vs retransmission probability of voice packets.

4.7 Conclusions

In this chapter, we propose a Spread Slotted Random Access protocol with a multi-priority service mechanism. The priority mechanism is implemented and controlled explicitly by the users and without any involvement from the network. This makes the allocation of priority

Table 4.1 Comparison of different versions of SSRA and SSA (Data load $\lambda = 0.1$)

Performance Metrics	SSRA			SSA		
	With reservation	Without reservation		With reservation	Without reservation	
		CB policy	No CB policy		Random code selection	Code assigned per talkspurt
No. of Voice User	70	28	25	61	18	11
Percentage of Data Packets Blocked	0.07	0.4	0.0	0.2	0.0	0.01
Channel Utilization	0.72	0.33	0.35	0.64	0.25	0.19

Table 4.2 Comparison of different versions of SSRA and SSA (Data load $\lambda = 0.2$)

Performance Metrics	SSRA			SSA		
	With reservation	Without reservation		With reservation	Without reservation	
		CB policy	No CB policy		Random code selection	Code assigned per talkspurt
No. of Voice User	60	22	17	32	5	0
Percentage of Data Packets Blocked	20	5	1.0	1.2	0.1	0.01
Channel Utilization	0.70	0.40	0.38	0.54	0.25	0.2

transparent to the network infrastructure and adds to the simplicity and practical value of the proposed scheme. We examine the proposed protocol under reservation and non-reservation policies and evaluate it for integrated data/voice traffic. Our results show that the protocol is able to achieve high levels of utilization and is capable of accommodating relatively large volumes of integrated traffic. Summarizing, we can claim that our results demonstrated the superiority of the SSRA protocol as compared to SSA. In addition to that, considerable gains in capacity and channel utilization are possible by adopting a reservation policy.

Chapter 5

An Integrated data/voice/video MAC for Wireless Communications

In the previous chapter we have introduced and analyzed a novel medium access control protocol integrating voice and data applications using CDMA technology. The proposed access mechanism uses the Spread Slotted Random Access (SSRA) protocol to resolve contention arising when two or more users attempt to simultaneously gain access to the channel. The access mechanism allows the definition of multiple priorities as well as the reservation of time slots. The main objective of this chapter is to extend the MAC protocol to support variable bit rate traffic.

5.1 Introduction

Digital video technology, and packetized video technology in particular, has received a lot of attention from industry and academia in the past few years. The main aim has been to define suitable compression algorithms to reduce the amount of information to store and distribute among computer-based systems interconnected via digital links. Various video compression schemes have been proposed and standardized in the last years. Since we are interested in defining MAC protocols for supporting integrated services (data/voice/video) over wireless channels, we have focused our study on using low bit rate video technology. In particular, we have considered the H.261 video compression standard [H261].

Depending on the type of the video encoder being used, a video source might be encoded into a Constant Bit Rate (CBR) or Variable Bit Rate (VBR) digital video streams.

In video terminology, the video stream resulting from the encoding process is organized into video frames (VF). A video stream can be perceived as a sequence of images, an image corresponding to a VF in the encoded digital video stream. The rate at which the images are encoded is referred as the video frame speed (VFS). The VFS might differ from one video source to the other, depending on the quality requirements and the nature of the application. In turn, the video frame period (VFP) is defined as the time elapsed between the beginning of the encoding of an image and the beginning of the encoding of the next image. Obviously, $VFP = 1/VFS$.

In a packetized video communications system supporting video conferencing applications, the digital video stream is encoded into VFs. The VFs are sent through the network in small data transport units, called packets. Upon receiving the packets, the receiver will feed them into the decoder which will reconstruct the images from the receiving VFs and played the VFs back at the rate at which they were generated.

5.2 Description of the System

We consider a communications system where the channel capacity is time-slotted. Slots are organized into frame. In order to avoid any confusion, we will refer to a time frame as system frame (SF). We further define the system frame period (SFP) as the duration of a frame; all frames having the same duration. In order to avoid wasting network resources in unnecessary competitions, we consider the use of reservation mechanism when dealing with video communications.

A personal communications network is expected to serve a variety of applications and information sources (voice, data, video, still image, text, graphics etc.). Each application can be associated with a specific set of parameters which define the Quality of Service (QoS) requirements of the connection. The set of parameters commonly considered are the following:

- the maximum acceptable delay in terms of establishing a connection, ΔT_c ,
- the acceptable percentage level of connection attempts experiencing a delay higher than ΔT_c when establish a connection, P_c ,
- the maximum acceptable packet interarrival delay, ΔT_p . Packets violating this delay become obsolete (useless to the application), they are dropped and considered as lost packets.
- the highest percentage value of lost packets that a connection can tolerate, without the service deteriorating below acceptable levels, P_p .

Since in the present chapter we will deal with already established connections, we will consider only the packet interarrival bound, ΔT_p , and the acceptable percentage level of packet losses, P_p .

For any time-constrained application, the delay constraint ΔT_p introduces a limitation in terms of SFP. Formally, this can be expressed as:

$$SFP \leq \min\{\Delta T_p\} + SP \quad (5.1)$$

where SP is the duration of a slot and $\min\{\Delta T_p\}$ represents the smallest of the inter packet delay limitations of all applications that are serviced by the network. This constraint ensures that the frame duration is adequately small to avoid losing systematically packets transmitted in consecutive SF. Eqn. 5.1 defines the upper bound on the system frame period that guarantee the quality of service required by the most stringent application to be supported by the system.

In order to determine the lower bound on SFP, we need to consider two important system parameters 1) the size of the slot to be used in our system and 2) the use of reservation mechanism. For the former, we need to bear in mind that one of the key requirements for the success of PCNs is their ability to interconnect efficiently with the

ATM Broadband ISDN public networks. An important step to this direction is to adopt the same basic packet format used in ATM. An ATM packet, cells in the ATM terminology, consists of 53 bytes, of which 48 carry user information. The remaining are used for control and management network purposes, such as cell priority, marking of Virtual Paths etc. In our work, we assume an ATM cell structure, i.e. every packet contains 48 bytes, equivalently $48 \times 8 = 384$ bits, of user information. The use of a reservation mechanism, allowing a source to make use of the same slot number in subsequent system frames can be justified by the relatively high bit rate required and time constraints of video and voice applications. To avoid loss of network resources in unnecessary competitions, we will apply reservation policies to the video and voice connections. The ability of reservation to improve network utilization was demonstrated in Chapter 4, when it was used with voice sources. In addition to the utilization benefits, reservation will be able to keep the losses of packets to a minimal.

In order to take advantage of the reservation mechanism and make efficient use of network resources, it is important that the sources accumulated during the frame period enough data to fully use the allocated slots. Let us assume that the lowest speed (in bits per second) at which a source making use of the reservation mechanism can generate information is Y_{\min} . Since the information load for a single slot is 384 bits, it is required that the **SFP** to be long enough to allow the source to produce this amount of information. This means that **SFP** has to satisfy the following relation:

$$SFP \geq SFP_{\min} = 384/Y_{\min} \quad (5.2)$$

where **SFP_{min}** is the smallest value of **SFP** that satisfies the above condition. Summarizing Eqns. (5.1) and (5.2) we end up with the following constraint regarding the duration of the SFP:

$$384/Y_{\min} \leq SFP \leq \min\{\Delta T_p\} + SP \quad (5.3)$$

Eqn.(5.3) defines the upper and lower bounds of the **SFP**.

5.3. Description of the Traffic Sources

As we mentioned earlier, we consider three different types of information sources: digital voice, digital compressed video and data.

Voice communications represent the most popular telecommunications service. Even though the importance of alternative ways of communication (video, data) increases, it is expected that voice based communications services will maintain their importance in the future. At the present time, compression technology allows us to transmit acceptable quality of digital voice at rates as low as 8 kbps. However, nowadays, most digital voice systems operate at speeds of 16 Kbps, 32 Kbps or 64 Kbps. Throughout this chapter, we will assume that every digital voice source produces data using one of these three rates. Similar to the parameters used in the previous chapter, the maximum acceptable packet delay ΔT_p for a voice application has been set equal to 32 msec and the maximum acceptable packet dropping rate P_p is set equal to 1% [Good89].

The increasing importance for bandwidth efficiency and the introduction of video-based services, products and applications has fuel a serious need for efficient compressed video algorithms. The development of compression standards such as MPEG and H.261 [H261] provides a solution to this problem. However, due to the reduction of redundancy from the video signal, the video service becomes more sensitive to, delay, delay variation and packet losses.

A good candidate for low bit rate video over mobile networks is the H.261 standard. H.261 is also known as px64, where $p = 1, 2, 3, \dots$ refers to the multiple of the (basic) 64 Kbps rate used by the encoder. For video conferencing applications, even 64

kbps of transmission rate ($p = 1$) is adequate to produce acceptable quality. In our work, we will be using signals compressed according to the H.261 video standard in QCIF format as well as H.261 video in CIF format, both at a medium speed of frame rate (15 frames/sec.). As mentioned earlier, we will test our system with both, CBR and VBR video sources. CCITT is in the process of defining the requirements of a new low bit rate object-based video coding (MPEG-4) for mobile applications, however, this work is in its initial stage at this point.

In a video encoder, CBR video is produced by dynamically changing the number of quantization levels the encoder uses. This control is done through a feedback mechanism whose input is the buffer occupancy of the output buffer. When the buffer occupancy exceeds a certain value, the control mechanism requests that the number of the quantization levels used by the encoder is reduced. This reduces the amount of produced data, however, the fidelity of the image is reduced as well. Consequently, CBR provides us with variable quality of video.

Since a video packet becomes obsolete if it is not delivered in time to be used for the reconstruction of the video image. We assume that a cell carrying video information is lost, if it is not delivered to the destination within one **VFP**. We assume that the highest acceptable rate of video packet losses is equal to 0.01%.

Video sources based on the H.261 standard can produce data streams at (average) rates multiples of 64 Kbps. Depending on the application, the frame rate is set between 10 fps and 30 fps. This gives a cell delay between 33 msec (for 30 fps) and 100 msec (for 10 fps). Video conferencing applications are usually in the low end of data speed and frame speed, i.e. in 64 Kbps to 128 Kbps and frame rates in the range of 10 fps to 15 fps.

The lowest data speed produced by sources of continuous nature is 16 Kbps (voice users). In order to fill a cell, the sources need 24 msec. Using these values in Eqn.(3) we find that the **VFP** has an upper bound of $(33 \text{ msec} + \mathbf{SP})$ and a lower bound of 24 msec. In our system we adopt a **VFP** = 24 msec.

5.4 Reservation Policies

5.4.1 Reservation Policy for Voice Users

For voice users, the reservation policy is the same one used in Chapter 4. Under this policy, a voice user joins the competition process for the right to transmit whenever he becomes active. Upon succeeding to use a slot, the user keeps the right to use the same slot during the following frames as long as the user remains active. When the user becomes idle, the base station marks the slot as, open for competition.

In Chapter 4 we dealt with homogeneous voice users, that is to say all voice users had the same characteristics in terms of data rate. The major assumption was that when in active state, all voice users required a slot per frame while in the active state. In this chapter, we relax this assumption and assume a population of non-homogeneous voice users. We assume that the basic data rate for voice users to be one slot per SFP. Other voice users will require higher rates, multiples of the basic rate, i.e., more than one slot per SFP. One of the major issues is therefore to define a simple and efficient reservation policy in a multiple rate user environment.

In our study, we have adopted the following reservation policy. Assume that a voice user requires M slots per SFP when active. Assume that the user is idle and it turns from its idle state to the active state. The first packet (cell) produced by the user starts competing following the rules of the Collision Resolution Algorithm (CRA) as defined in Chapter 4. Upon succeeding, the user requests from the base station to reserve for the user an additional $(M-1)$ slots. The user keeps the right to use these M slots as long as it remains active.

One major issue is to decide what to do when a user requiring M slots per SF does not get them. A number of different strategies can be followed, however, we have adopted one approach that maintains simplicity in the implementation of the protocol. Under this

strategy, users are classified in function of their needs in terms of slots per SF. For simplicity, we assume three different classes: users requiring one, two and four slots per SF.

During the operation of the system, the base station will keep track of the number of free slots. When the number of free slots during a SF becomes less than four the base station conveys this information to the users through the FC. Voice users requiring four slots should withhold their activities. The same happens when the number falls below two slots. This mechanism penalizes users requiring a larger number of slots. However, this unfairness can be compensated by allowing these high demanding users to access a larger number of codewords during the competition process. We will not address this last issue in this dissertation.

5.4.2 Reservation Policy for Video Users

For video traffic, we design and evaluate reservation policies for CBR and VBR video.

A. Reservation Policy for Constant Bit Rate Users

In CBR video, a video source produces packets at a constant rate. In this case, the appropriate number of slots per frame are allocated to each user: two slots for a 64 Kbps video connection and four slots for a 128 Kbps connection. In this case, the number of required slots are allocated to the connection for the whole duration of the communications. It is obvious that under this policy, CBR packets would not suffer any loss due to bandwidth. However, some kind of feedback mechanism can be implemented to throttle the source during periods of high demand. However, this issue is out of the scope of the work undertaken in this dissertation.

B. Reservation Policy for Variable Bit Rate Users

VBR sources produce a variable number of packets per video frame. In order to avoid waste of resources by following a bandwidth allocation policy based on peak rate traffic, the network should be able to allocate slots dynamically to the VBR video users according to their needs.

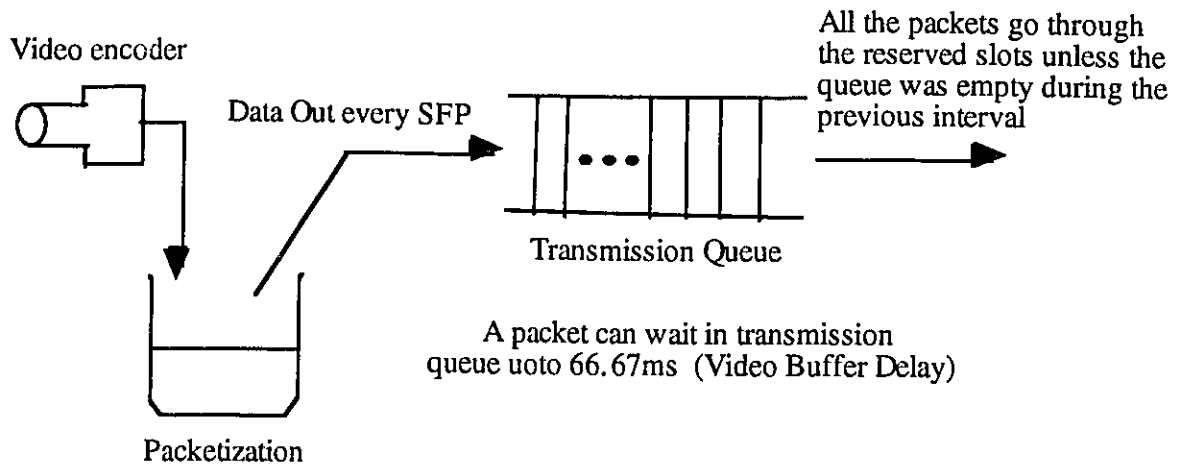


Figure 5.1 Transmission of video packets

Variable bit rate video generates a variable number of bits/s and hence variable number of packets (cells) per SF. Each video source has an output buffer. The data produced by the video encoder form cells and go into the buffer every macroblock time of the VF. The produced cells are serviced on a First In First Out (FIFO) approach. A cell remains into the buffer for a maximum duration of one VFP from the time it was generated. If it is not serviced during this period, it becomes obsolete and it is rejected from the buffer.

An active VBR video user is allocated several slots on the SF. These slots might not all be with the same codeword, however, the user can not make use of more than one codeword in the duration of the slot. This prevents the user from transmitting more than one packet simultaneously.

Let us assume that an active VBR video user has succeeded to send through the network (either through reservation and/or competition) a total of N_{res} packets. At the end of the present **SF**, the output buffer of the user has a total of N_{rem} packets. In the duration of the present video frame, the buffer received an additional N_{nar} packets. The total number of packets in the buffer at the beginning of the new frame is $N_{tot} = N_{rem} + N_{nar}$. The relation between N_{res} and N_{tot} determines the reservation policy of the user during the new **SF**. There are three possibilities:

1. If $N_{tot} = N_{res}$, then all the available N_{tot} packets are transmitted in the slots that are already marked as reserved for this video connection.
2. If $N_{tot} < N_{res}$, then out of the remaining N_{tot} packets, the first transmitted video packet informs the base station to cancel $N_{diff} = N_{res} - N_{tot}$ reservations.
3. If $N_{tot} > N_{res}$ then the user has to inform the base station that $N_{diff} = N_{tot} - N_{res}$ more reservations are required for this connection. There are two cases to take into consideration:
 - (a). If $N_{res} = 0$, then one of the available N_{tot} packets goes through the Collision Resolution Process (CRP). After receiving this packet, the base station knows the existence of $N_{tot}-1$ packets that require service and searches for free slots in the rest of that **SF** to service them through reservation. An alternative approach is to enforce the output buffer to transmit a "dummy" packet whenever no real information is available. This keeps always one slot reserved for the video user and it can send through this slot information for reservation of additional slots.
 - (b). The first packet that goes through an already reserved slot in the new **SF**, informs the base station about the requirement of an additional $N_{diff} = N_{tot} - N_{res}$ reservations.

The example below provides a numerical example of the reservation policy followed by the protocol.

Interval #	New packets (N_{nar})	Available packets (N_{tot})	Already reserved slots (N_{res})	Requested new slots N_{diff}	Request granted = $N_{tot} - N_{res}$	Packet goes through CRP
N+1	5	5	4	1	1	0
N+2	5	5	5	0	0	0
N+3	11	11	5	6	4	0
N+4	11	13	9	4	1	0
N+5	6	9	10	-1	-1	0
N+6	0	0	9	-9	-9	0
N+7	4	4	0	4	3	1

Table 5.1 Example of the reservation policy followed by VBR video sources

Notice that the request for additional slots comes with the first packet that goes through a reserved slot. However, it is possible that prior to this slot, other slots, available for competition might exist. An alternative approach would have been to allow the first packet in the output buffer to participate in the CRP and if successfully transmitted, give the request for the additional reservations to the base station. If the first slot reserved for the user comes before the packet has succeeded to go through the collision resolution process, it withdraws from competition and uses the reserved slot. In our work, we have evaluated the first approach. We remind the reader that slots that have being reserved for a VBR user in a SF, remain reserved for this user in the subsequent SF(s), unless if the user requests from the system to have (all or -most possibly- some of them) withdrawn.

5.4.3 Reservation Feedback

For voice and data users, the requirements concerning feedback information are quite simple. The Feedback Channel (**FC**) has to only inform the user if a Collision (**C**) or No-Collision (**NC**) occurred during the slot, as well as mark the slots as Reserved (**R**) or Non-Reserved (**NR**). The voice users themselves are aware if the reserved slot belongs to them or not (if a slot is reserved for a particular user, either, in the previous frame the same slot was reserved and used by the same user or the user succeeding to send a packet through competition in that slot). However, for VBR video users, the situation is different. Due to the reservation policy we follow in this case, VBR users have to know if a request for additional reservation has been granted and which of the slots have been reserved. Thus, in this case, it is necessary that the base station informs the user that the particular slot has been reserved and it is to its disposal. This introduces some additional complexity in the implementation of the network. The marking of the slots as reserved for a VBR video user can be performed either by sending information about new reservation at the beginning of the frame, or (which is the approach we adopted in our evaluated model), the assignment of a slot to a user becomes known at the end of the previous slot and before the new slot starts.

5.5 Performance Evaluation

As with the protocols evaluated in earlier chapters, the performance of the integrated voice, video and data systems is been assessed through computer simulations using the OPNET simulation package. We assume that each user can transmit at a maximum burst rate of 540 Kbps. The simulated configuration uses $N_c = 4$ codewords. The total network capacity is 1.5 times the T1 rate of 1.544 Mbps. Here we assume an overall 25% overhead information to be added at physical layer to improve the serious impairments of the wireless

channel. Considering this the total of equivalent to two T1 channel bandwidth is required by the network under study. The **SFP** is set equal to 24 msec. This corresponds to a total of 30 slots per **SF**, assuming that every slot carries a load of 432 bits (384 information bits and 48 bits of control information).

In our simulations, we have assumed that voice users are equipped with fast Voice Activity Detectors (VAD), as described in chapter 2. For modeling voice, we use the 3-state Markovian Model and values of parameters used in chapters 3 and 4. In our simulations, voice users, when active, produce information at the rate of 32 Kbps. This rate corresponds to packetized voice encoded using either Adaptive Differential Pulse Code Modulation or Adaptive Delta Modulation. The maximum tolerable delay for voice users has been set to 32 msec (which corresponds to 1.5 of the **SFP**) and the maximum acceptable voice packet dropping rate is being set to 1%. As in chapters 3 and 4, we generate data traffic using Poisson process generators.

Video traffic is produced using statistics from actual video sequences, compressed by a QCIF H.261 [H261] video encoder. In our work we evaluate both, CBR and VBR video traffic. In both cases, the video frame rate has been set to 15 fps. CBR sequences produce data at the rate of 64 kbps. VBR sequences are produced with a quantization step size of $Q = 11$. This gives us an average data rate of 66.7 kbps. We assume that a video packet becomes obsolete if it is not transmitted successfully within one **VFP**. In the case of 15 fps this corresponds to a delay of 66.7 msec.

To best of our knowledge, today there is not enough information to establish reliably an acceptable limit regarding the dropping rate of video packets without seriously impairing the video quality. This acceptable limit would depend on the video compression standard, transmission rate, the nature of encoder (CBR, VBR), as well as nature of application. As such we do not attempt to draw a final conclusion, such as how many video stations a particular network is able to support. We rather try to i) show the benefits of using a stable 2-C protocol for supporting multimedia communications. ii) see the relative

performance of CBR and VBR video iii) and also to see how the CBR and VBR video connections affect other applications (such as voice, data) and vice versa.

During the collision resolution process, video and voice users are allowed to access 4 codewords, whereas data is restrained to access only one codeword. We test the network using two different collision resolution algorithms, one SSA using the Aloha protocol, the second SSRA using the 2-C protocol.

The Aloha protocol is simple and allows easy implementation, however, it is unstable by nature when servicing limit Poisson traffic. The 2-C algorithm is a stable and simple protocol, capable of servicing considerable loading of user traffic with finite average delay. Below, versions of the network, using either, the Aloha or the 2-C protocol for collision resolution purposes will be designed and compared.

5.5.1 Spread Slotted Aloha

This section will report performance results for the Spread Slotted Aloha (SSA) protocol. To avoid allowing the protocol to collapse whenever high volume of the bursty data traffic appears, thus penalizing the voice and video users, we have adopted the following approach regarding the retransmission policy when a packet suffers from successive collisions. An unsuccessful data packet is allowed to attempt to transmit at the next available slot with probability $P_{ret} = 0.02$. Furthermore, if a data packet does not succeed after 10 transmission attempts, it is withdrawn from the collision resolution process and it is allowed to try again after a considerable amount of time. Network configurations servicing CBR or VBR video users have been evaluated. The evaluations have been performed for a number of video stations (N_{vj}) of 5, 10 and 15 (CBR or VBR) and for data load of $\lambda = 0.1$ and 0.2. The maximum number of packets that can be transmitted simultaneously, without exceeding the acceptable level of interference is set to $N_{il} = 10$.

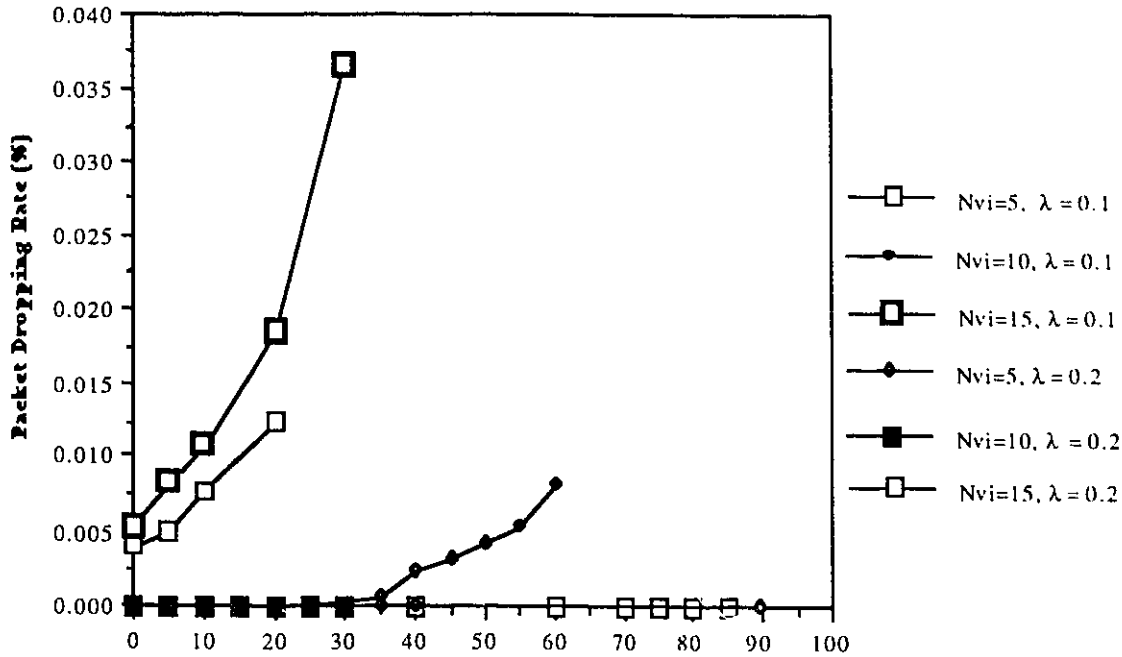


Fig. 5.2 Dropping rate of video packets vs number of voice users.
(SSA network, VBR video connections)

From our simulation results, CBR video users do not encounter any video losses when the number of video users were set to 5, 10 and 15. This is expected, since the reservation policy applied to CBR video users generates a virtual pipe that carries through the information. In this case, CBR video packets can be lost only if the interference limitation level is exceeded. However, even if this happens, video packets have adequate time to retry and be received before the expiration of the maximum allowable time delay for video packets (66.7 msec in our case), thus chances of losing a CBR packet in this case are quite small.

Fig. 5.2 (VBR) shows the dropping rate of video packets versus the number of voice users. The results depicted in Figure 5.2 for the case where VBR video connections are run through the network show an interesting trend. The packet dropping rate of video packets decreases as we move from an aggregate data traffic load of $\lambda = 0.1$ to $\lambda = 0.2$. In this case, voice users switching from idle to active have to request the reservation of slots.

To do so, they go through the competition process, and if successful, they reserve the required number of slots within the SF. They hold these slots as long as they remain active. When the data load is small, voice users have the opportunity and are able to place their reservation more easily. As the data load increases, the voice users have higher chances of failing in their competition, thus, they tend to remain for a longer period of time attempting to gain access to the channel. In this case, more slots are made available to the VBR video users who can achieve reservation by simply notifying the base station of their needs for additional slots through the reservation video packet. The base station searches for slots that have not been assigned (not reserved) and if available, assigns the required number to the video users. In a way, the increase of the data loading, acts as a negative catalyst that prevents voice users from achieving reservation, thus leaving more bandwidth to the disposal of the video users.

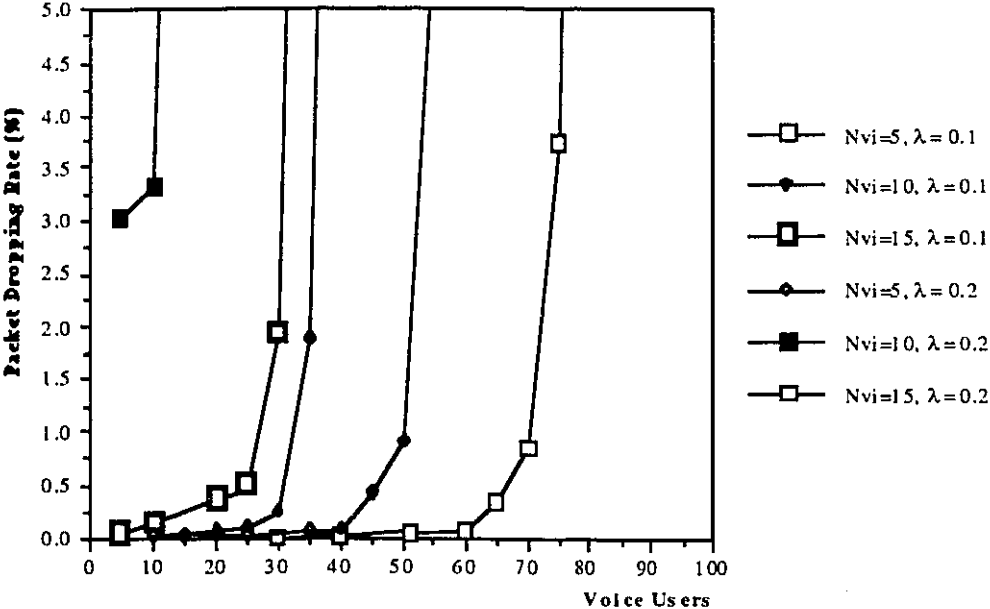


Fig. 5.3 Dropping rate of voice packets vs number of voice users.
(SSA network, CBR video connections)

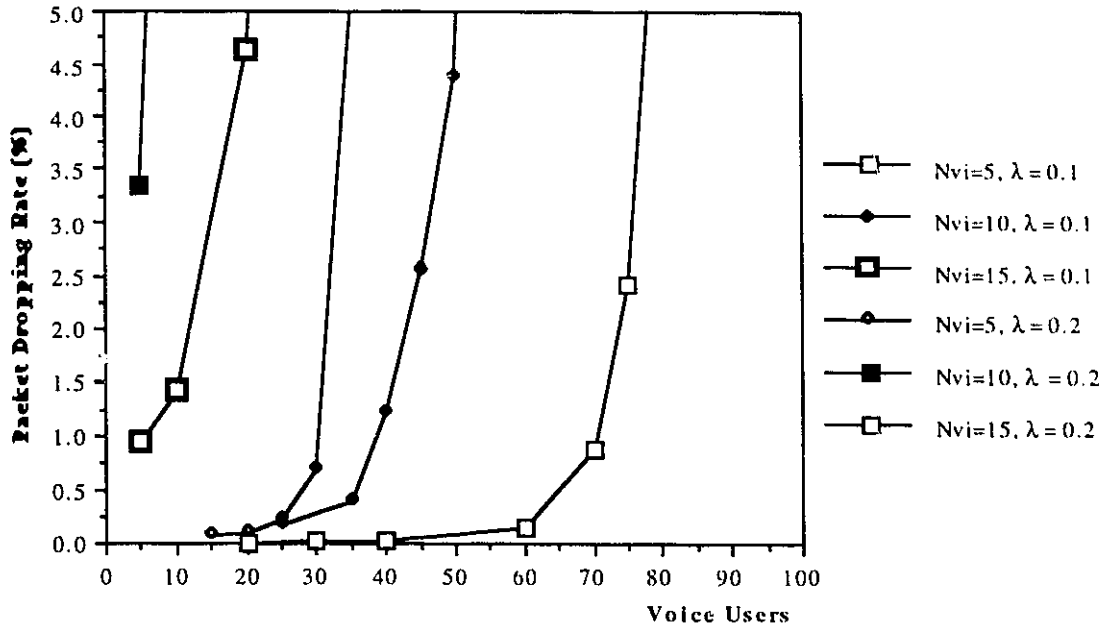


Fig. 5.4 Dropping rate of voice packets vs number of voice users.
(SSA network, VBR video connections)

Fig. 5.3 (CBR) and 5.4 (VBR) display the dropping rate of voice packets versus the number of voice users. The first conclusion that we can draw from the curves is that the data load affects quite considerably the number of users that the network can support without violating the 1% dropping rate of voice packets. While for $N_{vi} = 5$ and $\lambda = 0.1$ the network is able to support close to 70 users both under the presence of CBR or VBR video traffic, the figure drops to 30 when the traffic increases to $\lambda = 0.2$. Another point worth of mentioning is the effect VBR video traffic has on the performance of the network and its ability to satisfactory accommodate voice users. While for $N_{vi} = 5$, the performance of the network under VBR and CBR traffic is comparable, when the number of video stations increase to 10, presence of VBR traffic deteriorates the performance. For $N_{vi} = 10$ and $\lambda = 0.1$ packets/sec/slot, the number of voice connections that can be supported under CBR video traffic is 50 whereas under VBR traffic it is only 39. For $N_{vi} = 15$ and $\lambda = 0.1$,

these numbers become 25 and 5 respectively. For $N_{vi} = 10$ and data traffic $\lambda = 0.2$, the network is not able to support even 5 voice connections. From the results presented in the two last figures, it is reasonable to conclude that under the Aloha protocol, the network should not be dimensioned to accept more than 5 video users.

Figs. 5.5 (CBR) and 5.6 (VBR) show the blocking probability of data packet versus the number of voice users. The results demonstrate that for the same number of video stations and data load, the network can accommodate data without blocking for almost the same load of voice connections. However, after blocking starts appearing, the network supporting CBR video connections, shows a more gradual degradation of its performance in terms of blocking data packets, as compared to the network that supports VBR video user traffic. This behavior is due to the bursty nature of the VBR video traffic. At the time of increased VBR video traffic activity, network resources (i.e. slots) are withdrawn through the reservation policy followed for video packets. The effect of the

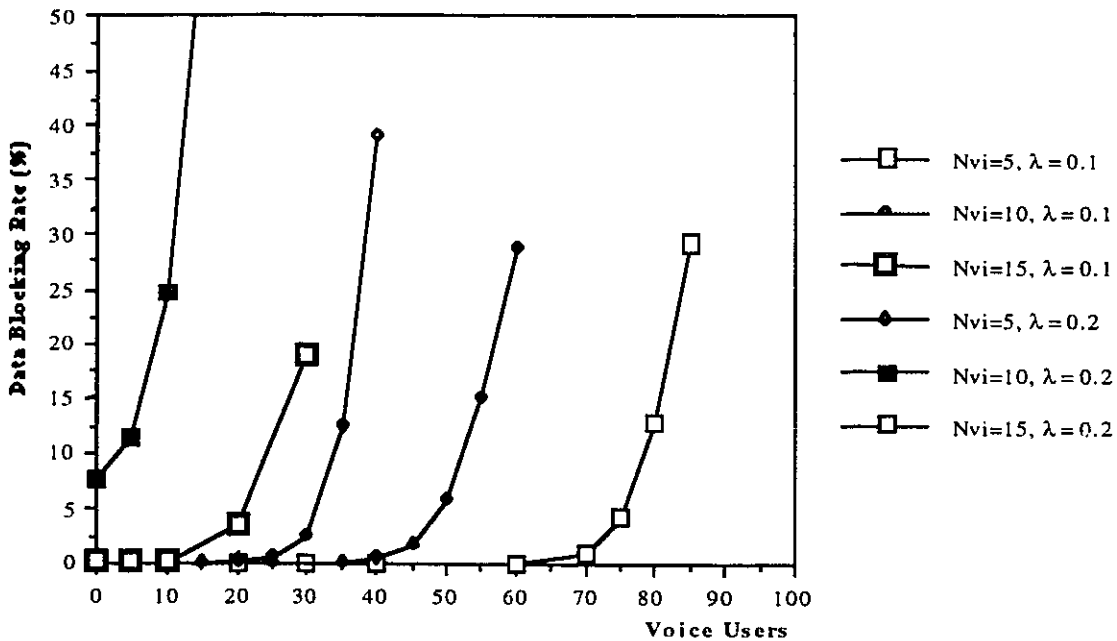


Fig. 5.5 Blocking rate of data packets vs number of voice users.
(SSA network, CBR video connections)

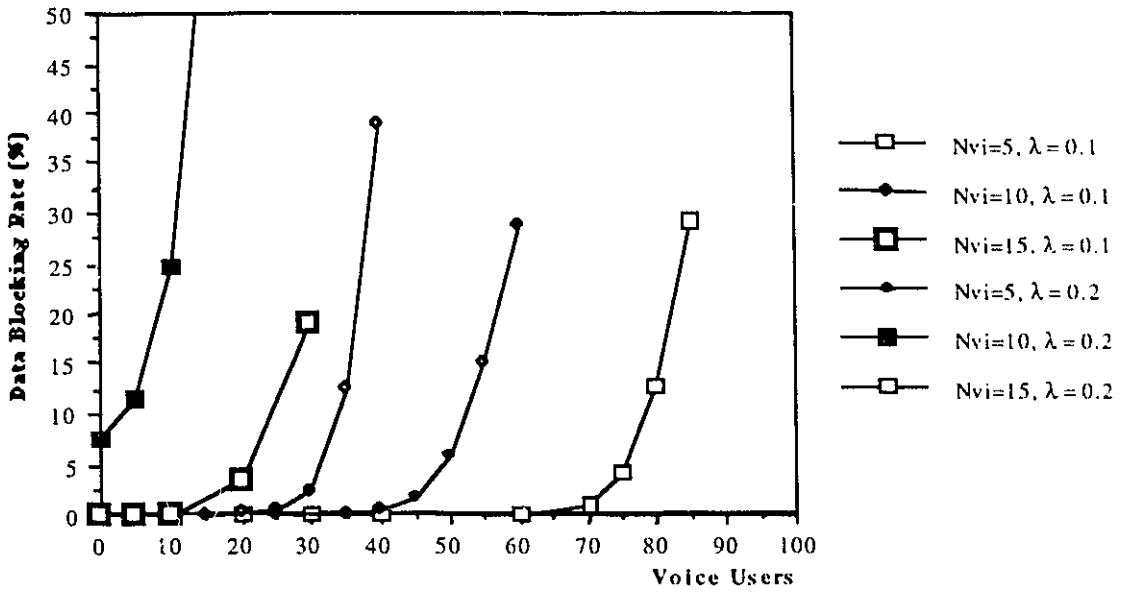


Fig. 5.6 Blocking rate of data packets vs number of voice users.
(SSA network, VBR video connections)

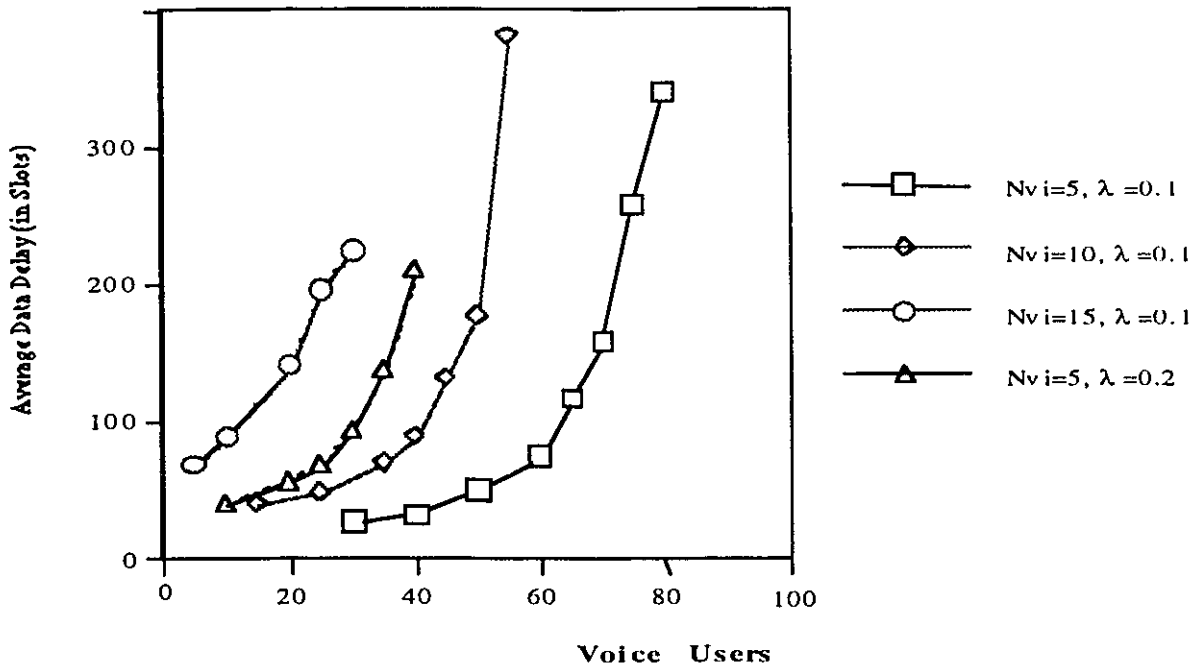


Fig. 5.7 Average delay of data packets vs number of voice users.
(SSA network, CBR video connections)

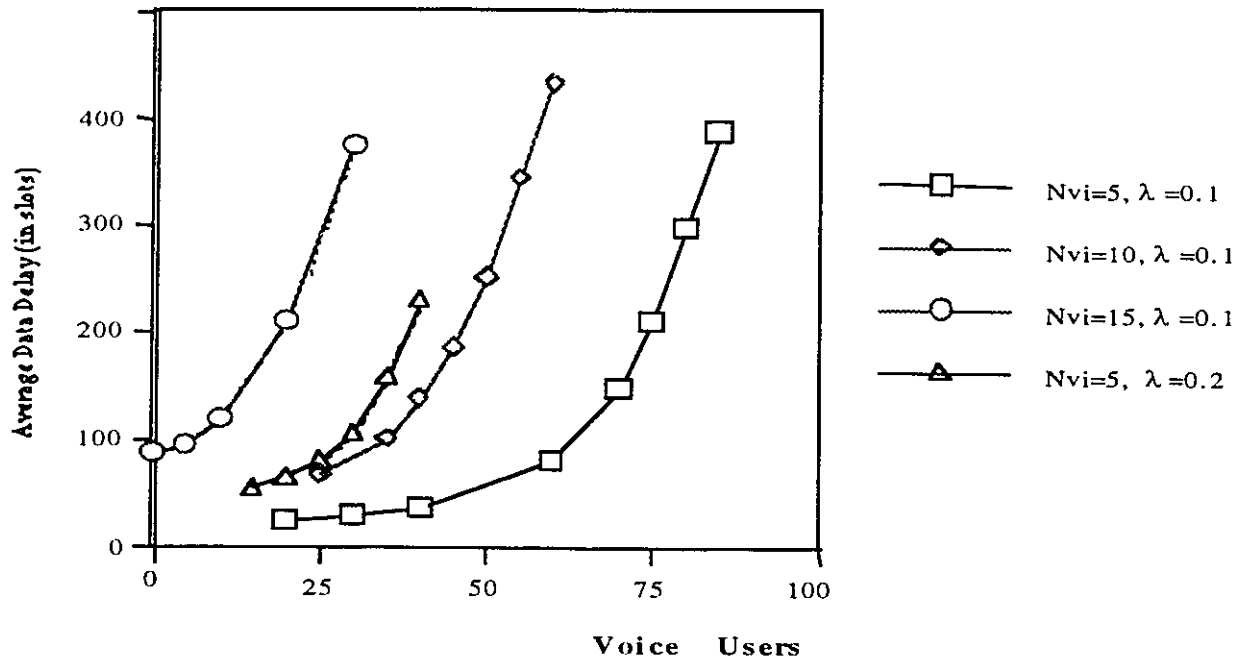


Fig. 5.8 Average delay of data packets vs number of voice users.
(SSA network, VBR video connections)

loss of resources tends to become more apparent when the network is under heavy traffic load conditions which is the case when we start seeing blocking of data packets. On other hand, CBR video traffic remains practically unchanged with the time, thus, it does not contribute to any time varying additional overloading of the network, which would drive it deeper to the saturation.

Figs. 5.7 (CBR) and 5.8 (VBR) display the average delay of data packets versus the number of voice users. For small number of video connections, the results are comparable for CBR and VBR types of video traffic. When the number of video connections increases, data packets face higher delay (in average) under the presence of VBR video traffic.

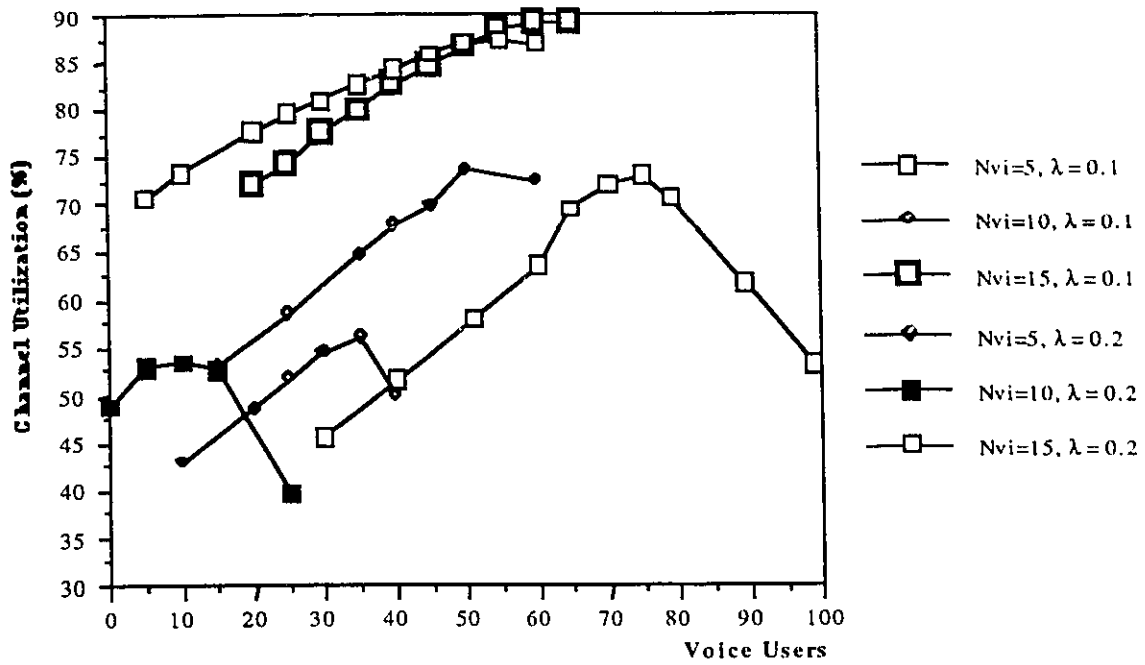


Fig. 5.9 Channel utilization vs number of voice users.
(SSA network, CBR video connections)

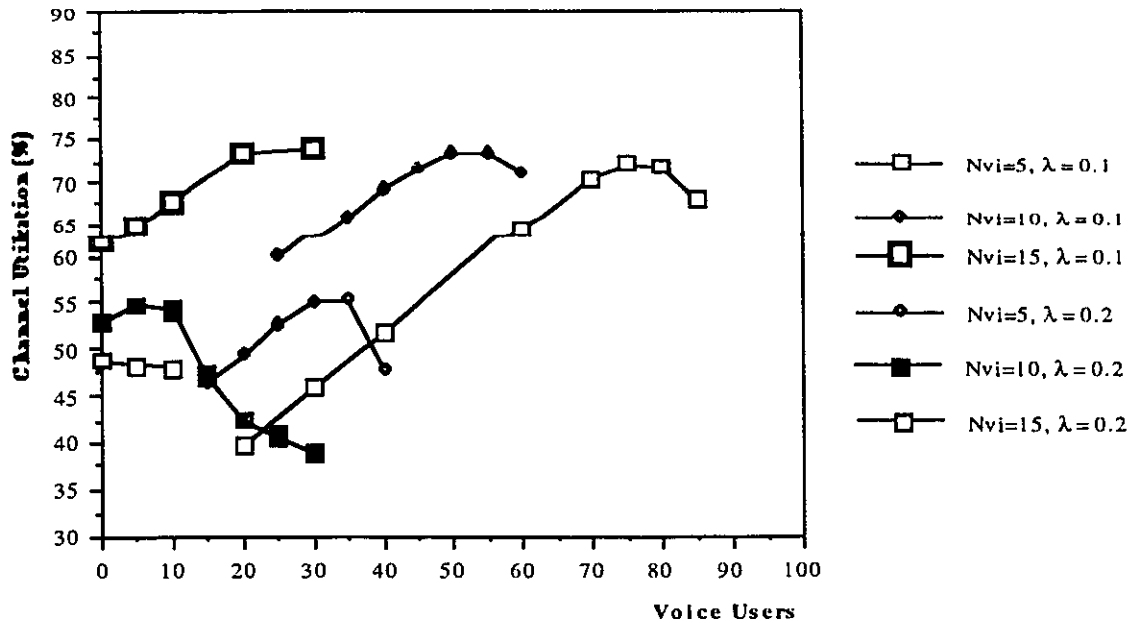


Fig. 5.10 Channel utilization vs number of voice users.
(SSA network, VBR video connections)

The network utilization versus the number of voice users is shown in Figs. 5.9 (CBR) and 5.10 (VBR). From these curves, we can draw the following conclusions. The network achieves higher levels of utilization under CBR video traffic. With CBR video, the network reaches utilization levels close to 90%. On the contrary, under VBR video traffic, none of the evaluated configurations was able to exceed the 75% utilization level in SSA. The higher utilization of the network under CBR video traffic is expected and explained by the way, CBR video users are serviced by the network. CBR video users are serviced by a virtual circuit established through the network. This, and the constant rate of the produced video traffic, i) eliminates the involvement of video packets in competitions which always could result in wasting the slot due to collision and ii) guarantees that all the slots allocated to CBR video users are used (they carry packets) and most of the times, they successfully deliver their load.

The curves displayed in Fig. 5.9 for CBR video, show that by increasing the number of video users, we see an increase in the utilization of the network. The explanation is the following. Every additional video user which is added to the network, guarantees that an additional part of the available bandwidth (i.e. the part that the network allocates to this user) will be utilized 100%. Obviously, by withdrawing these resources from users that are serviced through random access (data) or semi-reservation policies (voice) and by allocating them to users that utilize them 100%, the overall utilization of the network improves¹.

In VBR video, the utilization of the network seems to be determined by the load of the data packets. As VBR video and voice users have to compete to achieve reservation, the amount of data loading becomes an issue. High values of data loading prevents voice

¹Of course, this trade would continue up to the point that the network has enough resources to guarantee full accommodation of the CBR video users. We assume that the network will not accept a new video user if it does not have enough resources to accommodate this user.

users from achieving reservation and forces the system to waste considerable amount of resources due to unsuccessful competitions. The curves displayed in Fig. 5.10 show clearly that the utilization of the network configuration with $\lambda = 0.1$ is higher as compared to the utilization of the network when $\lambda = 0.2$.

5.5.2 Spread Slotted Random Access

This section reports results and analyzes the performance of network configurations that use the 2-C algorithm as core of the collision resolution algorithm. As in the previous case, the evaluation have been performed for a number of video stations $N_{vj} = 5, 10$ and 15 , for data loads $\lambda = 0.1$ and 0.2 and for a limit on maximum number of simultaneous transmission due to interference $N_{ij} = 10$.

In the case of CBR video, video users do not experience any video packet loss. This is as expected, since the network generates a virtual circuit for every CBR user, and allows the video user already adopted into the system, to be send information practically uninterrupted. On the other side we see that VBR video users experience dropping of packets. Fig. 5.11 (VBR) reports results concerning the dropping rate of video packets versus the number of voice users. The packet dropping rate increases as the number of voice users increases, however, it seems that the network is insensitive to the value of the data load. Doubling the data load from $\lambda = 0.1$ to $\lambda = 0.2$, did not have any noticeable effect on the packet dropping rate of the VBR video connections. To understand this fact we have to recall how the reservation of additional resources to VBR video users is done. The reservation of additional resources takes place through other video packets. The base station provides the additional bandwidth that was requested by a VBR video user, by withdrawing slots that were before marked as available for competitions. Thus, the

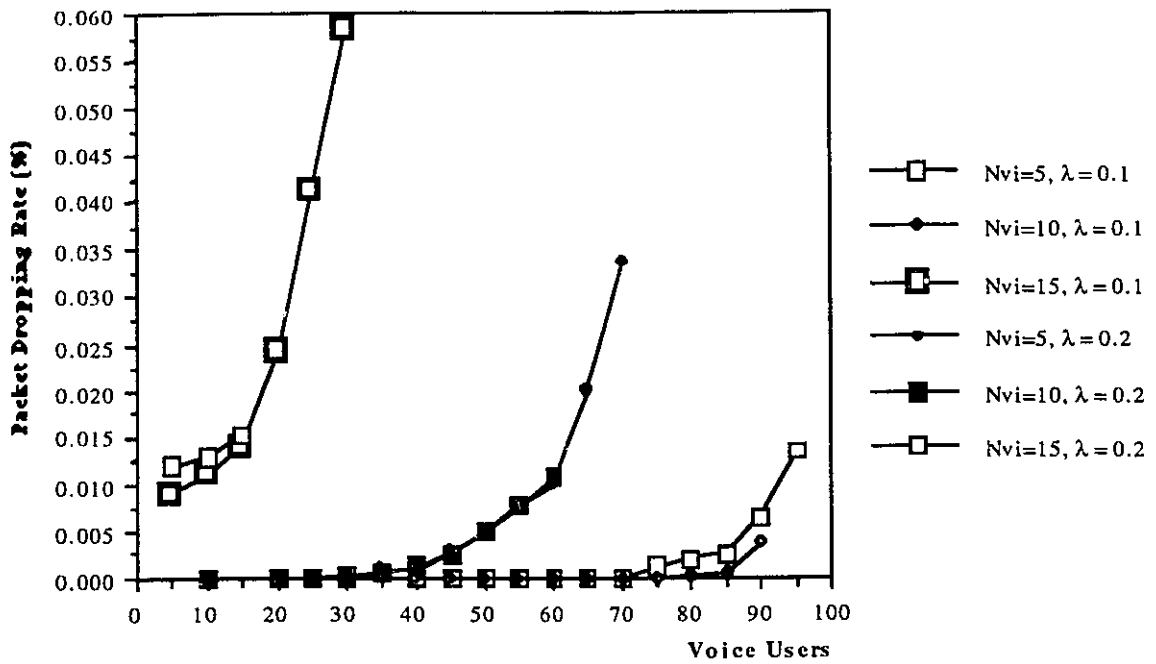


Fig. 5.11 Dropping rate of video packets vs number of voice users.
(SSRA network, VBR video connections)

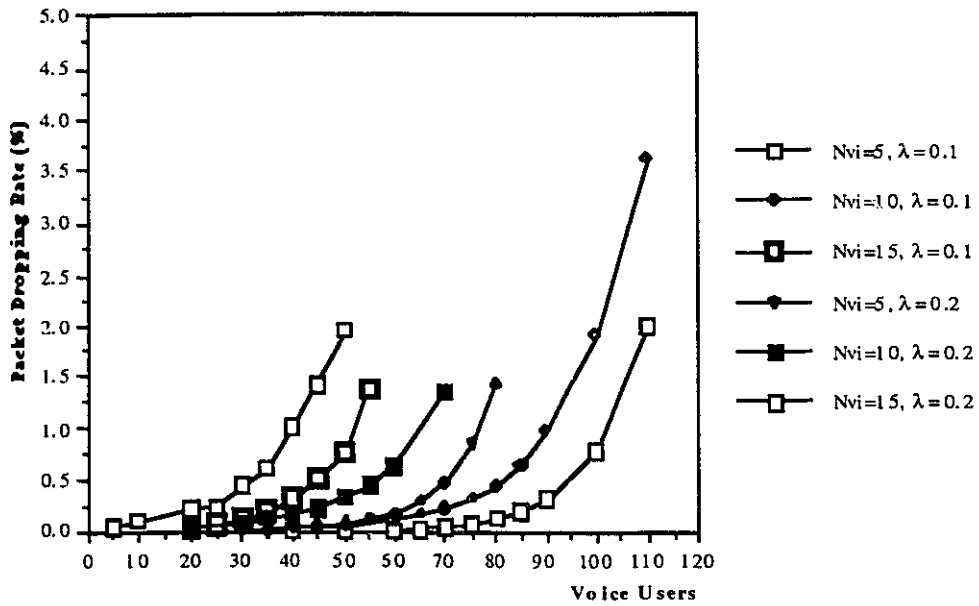


Fig. 5.12 Dropping rate of voice packets vs number of voice users.
(SSRA network, CBR video connections)

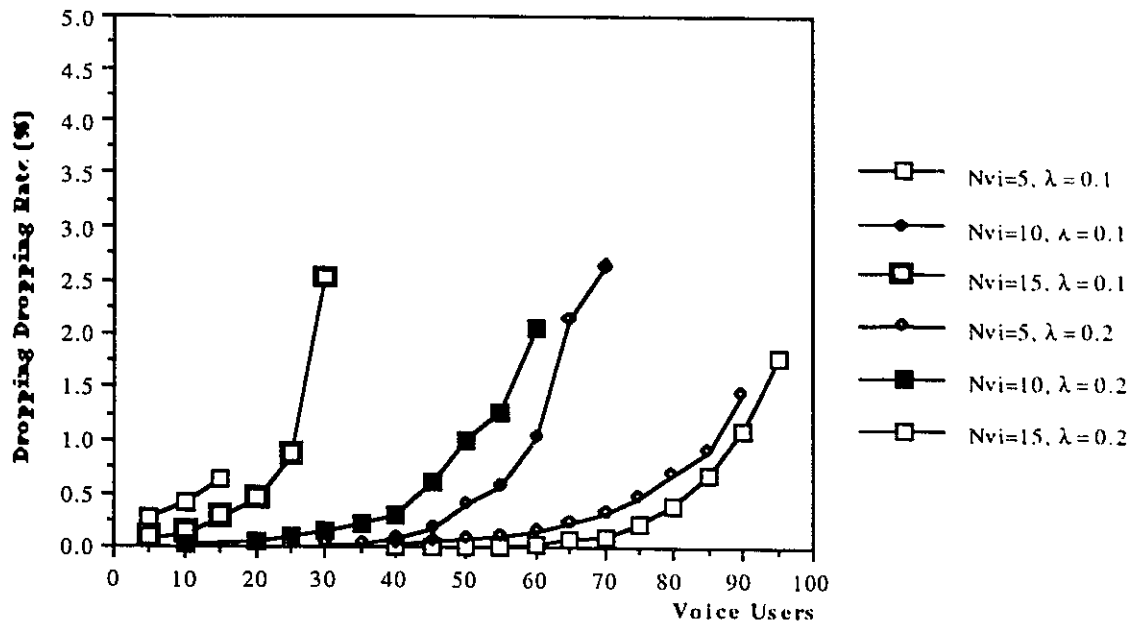


Fig. 5.13 Dropping rate of voice packets vs number of voice users.
(SSRA network, VBR video connections)

performance of the VBR video connections is affected primarily by the availability of unreserved bandwidth (slots). While the voice users request reservation and thus they consume bandwidth that could be needed by some VBR video users, data do not request any reservation, thus, they do not consume any such bandwidth, which remains available to be claimed by VBR video users whenever they need it.

In Figs. 5.12 (CBR) and 5.13 (VBR) we display the dropping rate of voice packets versus the number of voice users. From the curves, we can conclude the following. The nature of the traffic produced by the video users (VBR vs CBR) affects the performance of the voice users. The network seems to be able to accommodate higher number of voice users under CBR video traffic. The difference between the number of users that can be accommodated under CBR traffic to the number of voice users that can be accommodated under VBR video traffic increases as the number of video connections serviced by the

network increases. Of course, we should keep in mind that VBR video users produce an average bit rate of 66.7 Kbps as compared to the CBR video users which produce a 64 Kbps information rate. However, this difference is not enough to justify the considerably higher number of voice connections that can be accommodated when the video traffic is of CBR nature as compared to the number of voice users that can be supported under VBR video traffic.

For example, let us take the case of a network supporting 5 video stations ($N_{vi} = 5$) and with data load $\lambda = 0.1$. Under CBR video traffic, the network can support 101 voice users, without violating the 1% packet dropping rate limitation. Under VBR video traffic, this number becomes 90 voice users. Assuming an activity factor of 40% for voice users, every additional voice user which produces information at the rate of 32 Kbps (when active), loads the network with an additional (0.4×32 Kbps =) 12.8 Kbps average traffic. The additional 11 voice users that can be accommodated under CBR video traffic, give an additional total average traffic of (11×12.8 Kbps =) 140.8 Kbps. On the other side, the

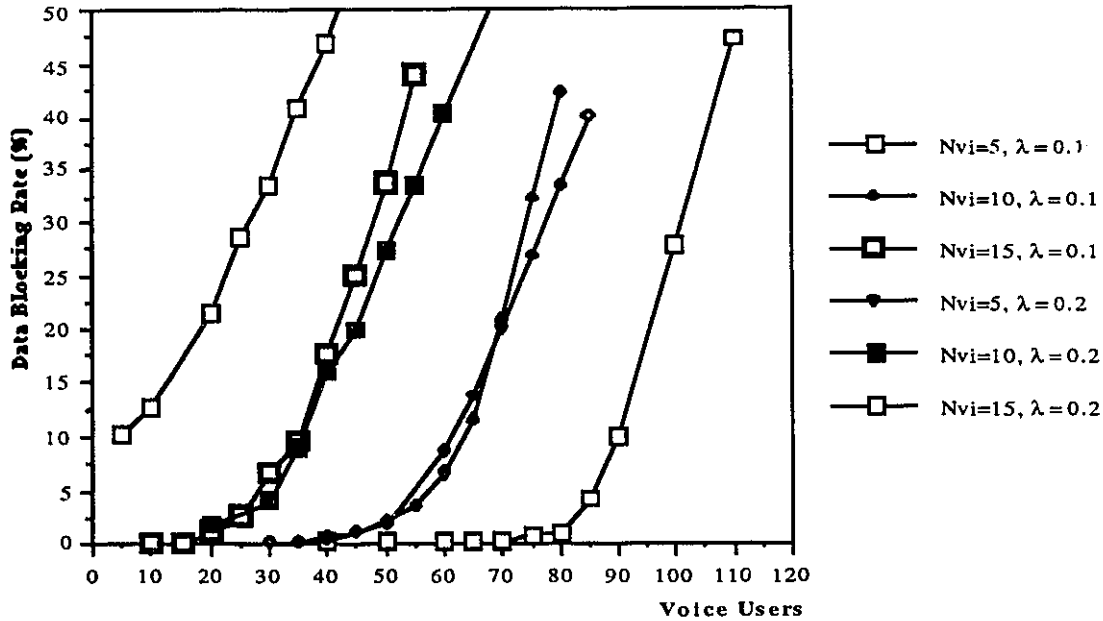


Fig. 5.14 Blocking rate of data packets vs number of voice users.
(SSRA network, CBR video connections)

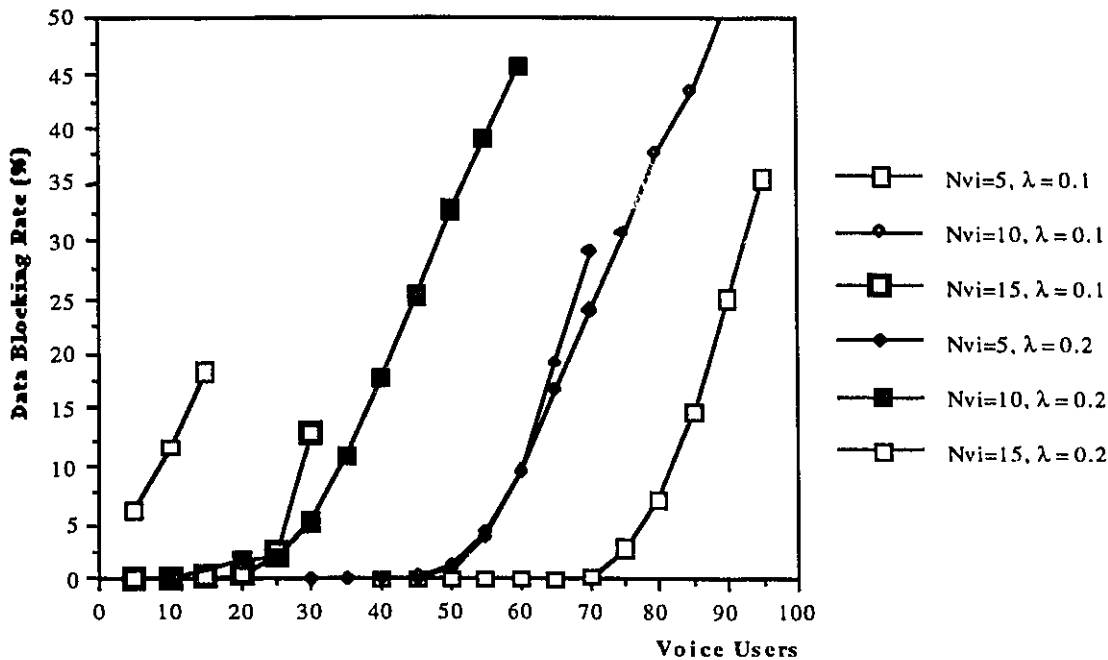


Fig. 5.15 Blocking rate of data packets vs number of voice users.

(SSRA network, VBR video connections)

additional traffic produced by the 5 VBR video users as compared to 5 CBR video users, equals $(66.7 \text{ Kbps} - 64 \text{ Kbps}) \times 5 = 13.5 \text{ Kbps}$. These number show that there is a definite reduction in the traffic that the network can carry when dealing with VBR video users. This will become again clear later, when we display curves on the network utilization.

The curves describing the blocking rate of data packets versus the number of voice users are displayed in Figs. 5.14 (CBR) and 5.15 (VBR). Comparison of the curves leads to the following conclusions. As expected, an increase in the number of voice and/or video users increases the blocking rate of data packets. However, an interesting observation is that the data blocking rate performance seems not to be influenced significantly by the nature of the video traffic (CBR vs VBR). Consideration that each VBR user produces an average traffic that is 2.7 Kbps rate higher compared to the traffic produced by a CBR video user, makes us realize that the results under VBR or CBR traffic are very comparable. This behavior can be explained as follows. While voice packets are sensitive

to time delays, data packets are not. As result, voice traffic is sensitive to the “dynamic” behavior of the network, with dynamic behavior referring to the statistical characteristics the network shows in terms of duration of high traffic time intervals, frequency they appear etc. On the contrary, data packets are affected more by the “average” characteristics of the network rather than the dynamic ones. Thus, while the bursty behavior of VBR video traffic is able to force a large number of voice packets to be dropped due to time delay violation, it does not affect data packets that much, since they can wait until the intensity of the traffic is reduced and bandwidth becomes again available for competition.

Figs. 5.16 (CBR) and 5.17 (VBR) display the curves describing the average delay (in slots) experienced by the data packets versus the number of voice users. Again, the comparison between the curves of these two figures reveals a comparative performance between the configurations transporting VBR or CBR video traffic. The data

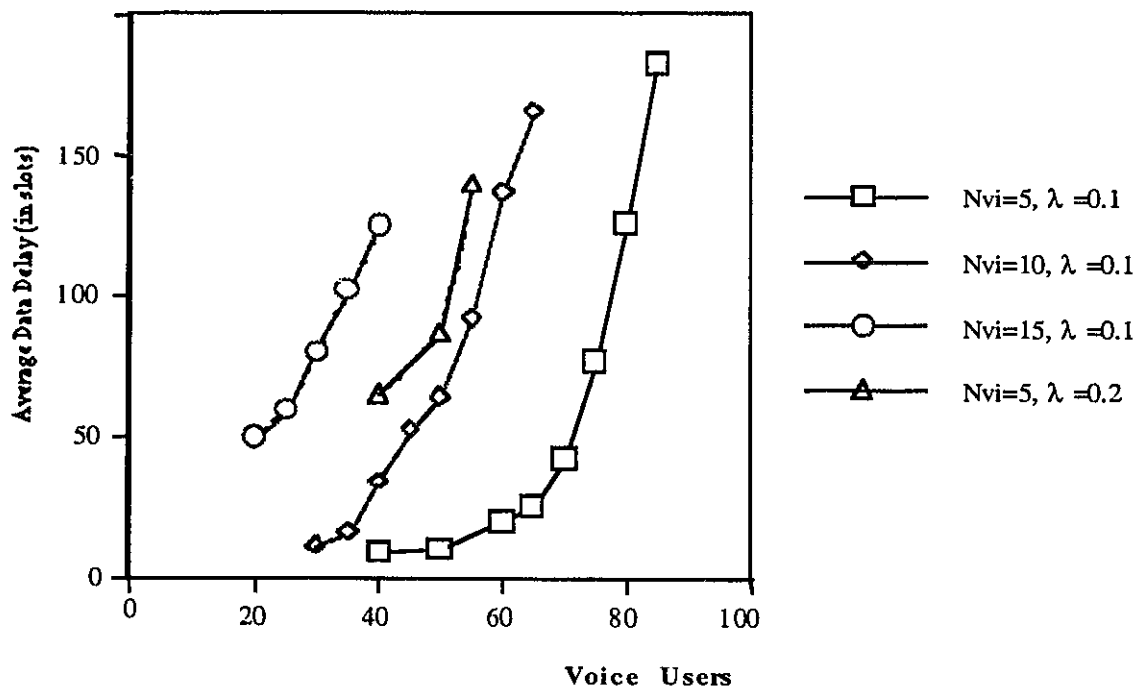


Fig. 5.16 Average delay of data packets vs number of voice users.
(SSRA network, CBR video connections)

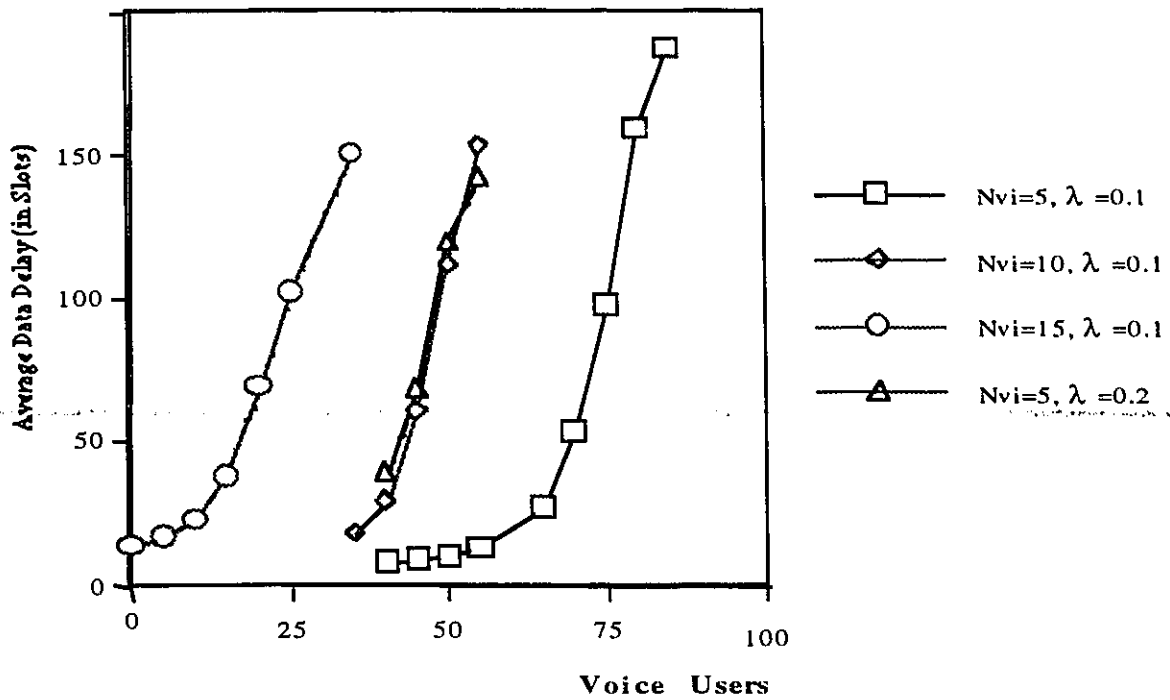


Fig. 5.17 Average delay of data packets vs number of voice users.
(SSRA network, VBR video connections)

packets serviced by networks carrying VBR video traffic experience a relatively higher delay, which can be explained by the fact that VBR video traffic, when at high levels, withdraws considerable amount of the resources from the network. Data packets have to wait until the volume of VBR video traffic subsides and then they are serviced.

Figs. 6.18 (VBR) and 6.19 (CBR) display the utilization curves of the network, when the 2-C algorithm is used for collision resolution purposes. It is evident from the two figures that the network achieves higher level of utilization under CBR traffic. We remind the reader that the same trend existed in the Aloha case. For an explanation why the network behaves this way, we refer the reader to the comments we made earlier in the Aloha section. Another trend which we show earlier in Aloha as well is the tendency the network has when carrying CBR traffic, to be achieving higher level of utilization, when

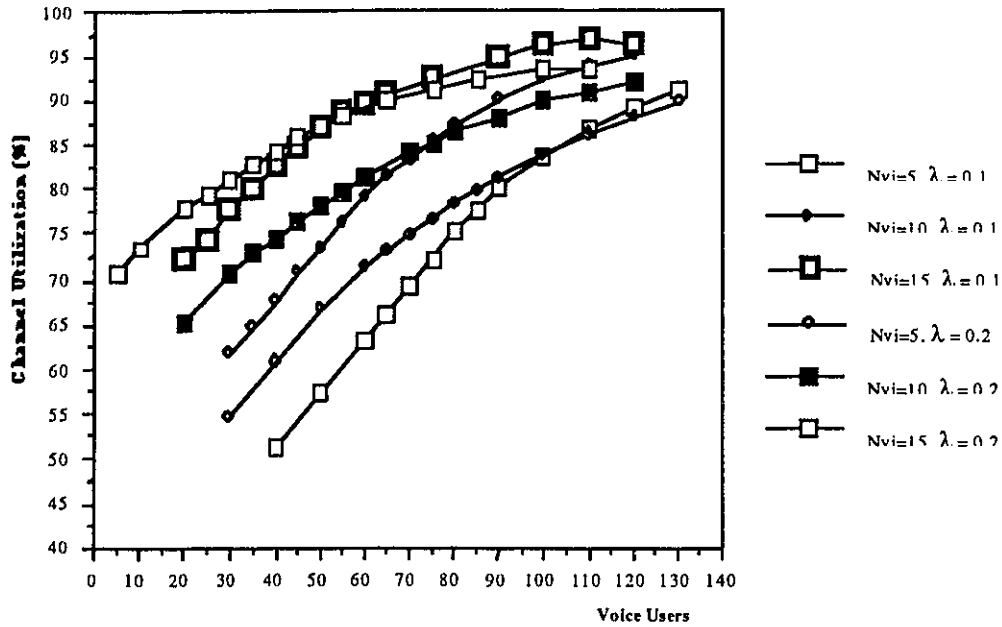


Fig. 5.18 Channel utilization vs number of voice users.
(SSRA network, CBR video connections)

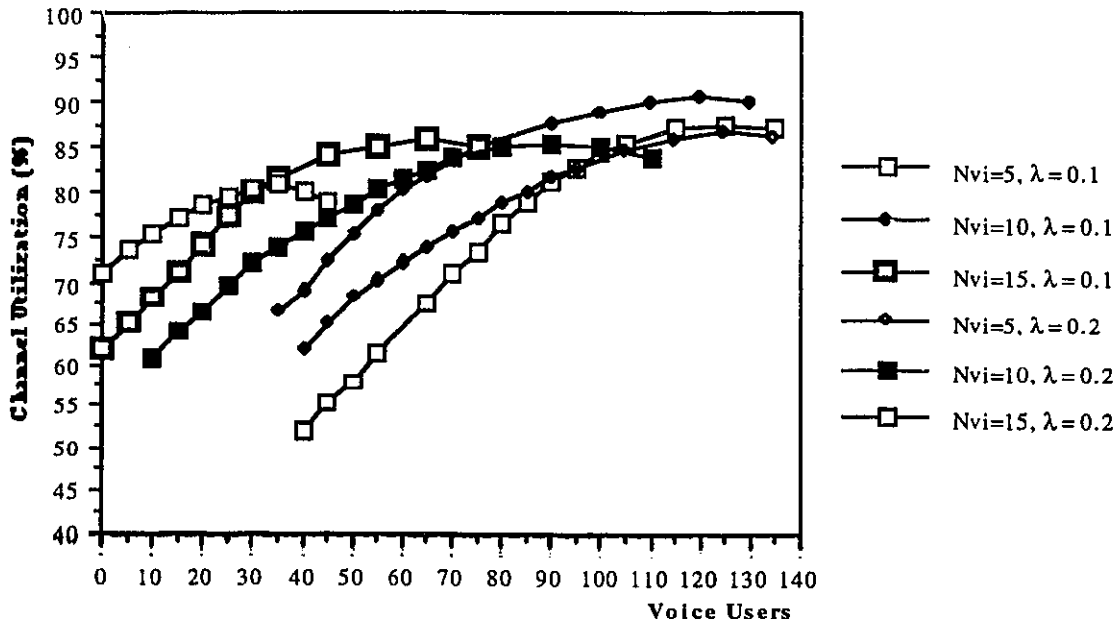


Fig. 5.19 Channel utilization vs number of voice users.
(SSRA network, VBR video connections)

the number of video users increases. The explanation for this behavior has been presented in the network utilization part of the Aloha section. We direct the interested reader to that section for a comprehensive explanation of this behavior. However, there is a change in behavior between the Aloha and the 2-C protocol and this concerns networks carrying VBR video traffic. We pointed out in the Aloha section, that the utilization of the networks carrying VBR video traffic, seems to be heavily influenced by the amount of data load. However, from the curves displayed in Fig. 6.20, we realize that the utilization of the network using the 2-C algorithm and carrying VBR video users, seems to be affected more by the number of video users rather than the data load. This explanation for the difference in behavior is the following. The 2-C protocol is superior to Aloha when dealing with a large population; the 2-C protocol is able to deliver higher loads of traffic without facing instability. This reduces the effect of data loading. We expect that if the data load keeps increasing, we will find a point where the effect of data loading on the 2-C based network will start becoming apparent as well.

Tables 5.2 and Table 5.3 show the maximum channel utilization achieved and the number of voice connections supported for easy comparison, for data load $\lambda = 0.1$ and $\lambda = 0.2$ respectively.

Table 5.2 Performance comparison of of SSRA and SSA (Data load $\lambda = 0.1$)

Number of video connections	Number of voice connections				Channel Utilization			
	SSRA		SSA		SSRA		SSA	
	CBR Video	VBR Video	CBR Video	VBR Video	CBR Video	VBR Video	CBR Video	VBR Video
5	102	87	70	70	0.90	0.87	0.86	0.73
10	76	60	50	38	0.94	0.89	0.72	0.73
15	52	25	27	5	0.97	0.86	0.89	0.73

Table 5.3 Performance comparison of of SSRA and SSA (Data load $\lambda = 0.2$)

Number of video connections	Number of voice				Channel Utilization			
	SSRA		SSA		SSRA		SSA	
	CBR Video	VBR Video	CBR Video	VBR Video	CBR Video	VBR Video	CBR Video	VBR Video
5	90	85	32	30	0.89	0.87	0.55	0.55
10	65	50	0	0	0.90	0.85	0.53	0.54
15	40	15	0	0	0.92	0.86	0.71	0.48

5.6 Conclusions

After having presented results concerning the performance of networks using either, Aloha or the 2-C protocol for collision resolution purposes, we are at the position of making the following comparisons and comments.

Both CBR (variable quality) and VBR (constant quality) video services can be efficiently supported by the proposed protocols. Network configurations using the 2-C protocol seem to outperform equivalent network configurations based on Aloha, in terms of capacity, utilization and level of service provided to the user (packet dropping rate, packet blocking rate, average delay). Networks based on the 2-C algorithm demonstrate a higher tolerance to increases of traffic (i.e. voice or data) and limit the effects of a class of users, such as data, exhibiting high load to other classes of users. This feature is a serious asset when considering to use these networks to service multimedia personal communications applications.

Overall, the use of CDMA and the 2-C algorithm, together with the imbedded (and transparent to the network) priority mechanism and reservation policies for video and voice users create a powerful system able to successfully carry multimedia services through the personal communication networks of the future.

Chapter 6

Conclusions and Future Research

Multimedia communication over wireless networks is the technology of the future. The requirement of multimedia transmission is dynamic bandwidth allocation. Voice is the major traffic carried by today's cellular networks. Need to support multimedia communications requires the redesign of many aspects especially the multiple channel access mechanisms. Performance studies show that by applying reservation policies and stable collision resolution algorithms we can easily integrate different traffic types in PCN's. Reservation schemes give the high channel utilization. Results and conclusions of this research work can be applied in several ways for PCN design.

6.1 Conclusions

Performance analysis of the various methods for uplink multiple channel access gives the following major conclusions:

1. Use of stable and efficient Z-cell algorithms (modifications of 2-cell algorithm) give better performance results than with using PRMA, for integrated voice and data traffic. PRMA uses Aloha protocol for collision resolution of the competing users. As described in chapter 3 we investigated both PRMA and the proposed algorithms for various data

loads over the slotted channel. Assuming that a base station is connected to Base Station Controller by a T1 line, and half of this one (i.e., 720 Kbits/s) for up link channel, we estimated the capacity of the system under condition of acceptable voice delay. Proposed protocols are found to support up to 38 voice users as compared to PRMA that supports up to 25 voice sources along with data load of $\lambda = 0.1$. For data load of $\lambda = 0.2$, it can support up to 37 voice stations while PRMA supports only 14 voice users. We can give high priority to the voice users by separating the CRI's for voice and data traffic.

2. In chapter 4 we have analyzed the above mentioned methods for spread spectrum technology. Actually this is a combination of CDMA and TDMA with each codeword having slotted channel. With the flexibility of accessing a codeword by different users in different time slots, the composite CDMA/TDMA gives better dynamic allocation of the bandwidth. Also we can give priority to high priority users by allowing them to transmit with more codewords. For voice and data integrated we have analyzed different versions of the protocol, using reservation and without reservation. We can claim that our results demonstrated the superiority of the SSRA (using modified 2-cell process as the CRP) protocol as compared to SSA (using Aloha as the CRP). In addition to that, considerable gains in capacity and channel utilization are possible by adopting a reservation policy.
3. The third part studied the integration of variable bit rate video in the CDMA system. Here we are using a multi-slot reservation scheme for VBR traffic. In this part we used the statistics gathered from real video sequences to drive our simulator. The simulation results show the effectiveness of these schemes. The results verified the superiority of the proposed protocol: high channel utilization and ability to service effectively high volumes of integrated traffic.

6.2 Thesis Contributions

Contributions of this thesis are as follows:

1. Design of multiple access protocol, that supports integrated (data, voice, video) traffic.
2. Use of reservation schemes for stream traffic showing the better PCN channel utilization and higher throughput.
3. Use of collision resolution processes that give stable performance for the multiple access protocol.
4. Use of different priority schemes to provide the required QoS for various traffic types.
5. Performance analysis of the protocol versions that makes use of both TDM and CDMA technologies.

Following is the list of papers published in conference proceedings and journals from the results of this research

1. D. Makrakis, R. S. Mander and L. Orozco-Barbosa, "An Integrated Traffic Protocol for Personal Communication Networks", Proc. of the IEEE Pacific Rim Conference (PACRIM'95), Victoria, B. C., pp. 20-25, May 1995.
2. D. Makrakis, R. S. Mander and L. Orozco Barbosa, "A Spread Slotted Random Access Protocol with Multi-Priority for Personal and Mobile Communication Networks Carrying Integrated Traffic", *Proc. of the Canadian Conference on Electrical and Computer Engineering*, Montreal, Canada, pp. 132-136, Sept. 1995.
3. D. Makrakis, R. S. Mander and L. Orozco Barbosa, "An Integrated Traffic Spread Slotted Random Access Protocol for Personal and mobile Communication Networks", *Wireless Networks*, in press.

6.3 Future Research:

1. Further study of various versions of the protocol with other possible reservation schemes for video.
2. Analysis in presence of impairments in the communications and feedback channels, i.e. effect of channel errors.
3. Effect on actual video streams of the video packet loss.
4. Study of control and signaling traffic generated within the cell by hand-offs.
5. Study of admission control and other connection setup procedures.
6. Capacity estimation of the system for channel bit rates according to the newly assigned PCN spectrum.
7. Analysis when packet capture applies.
8. Analysis of traffic integration using other versions of the Z-cell algorithm described in Chapter 3.

Appendix A: H.261 Overview

In 1984 the study group XV of the CCITT established a committee to work on the standard for the compression of moving pictures. In December 1990 the CCITT recommendation H.261 "Video Codec for Audiovisual Services at $p \times 64$ kbits/s" (CCITT 1990) was finalized. This recommendation is also known as $p \times 64$, because of the compressed data rate of $p \times 64$ kbits/s, at $p = 1, 2, \dots, 30$.

H.261 defines a very precise image format. Two resolution formats each with an aspect ratio of 4:3 are specified. The so called common intermediate format (CIF) has a resolution with the rate of 288 lines per frame and 352 pixels per line for the luminance (Y) signal. The resolution of the chrominance (Cb and Cr) components is, 144 lines per frame and 176 pixels per line. Quarter CIF (QCIF) format has exactly half the CIF resolution, i.e., 144x176 pixels for Y signal and 72x 88 pixels per line for Cb and Cr signals. In H.261 data units of size 8x8 pixels, known as blocks are used. A macro block consist of four blocks of the Y matrix and a corresponding block of each Cb and the Cr component.

H.261 uses both ways of coding: intraframe and interframe. The outline block diagram of the video encoder is shown in figure A.1. When operating with QCIF the number of bits created by coding any single picture must not exceed 64 kbits. The transmission encoder generates the bit stream that contains BCH Forward Error Correction Code. The video multiplexer is arranged in a hierarchical structure with four layers. From top to bottom the layers are: Picture Layer, Group of Blocks (GOB) Layer, Macroblock (MB) Layer, Block Layer. The video multiplex coder generates the header information for these layers, using information gathered from the source coder. The main elements of the video source algorithm are prediction, block transformation and quantization. The prediction error (INTER mode) or the input picture (INTRA mode) is subdivided into 8 pels by 8 line blocks that are segmented as transmitted or non-transmitted. Transmitted blocks are transformed (Discrete Cosine Transform) and resulting coefficients are quantized

and variable length coded. Data for 4 luminance blocks and two spatially corresponding color difference blocks is combined to form a macroblock and sent to the transmission buffer.

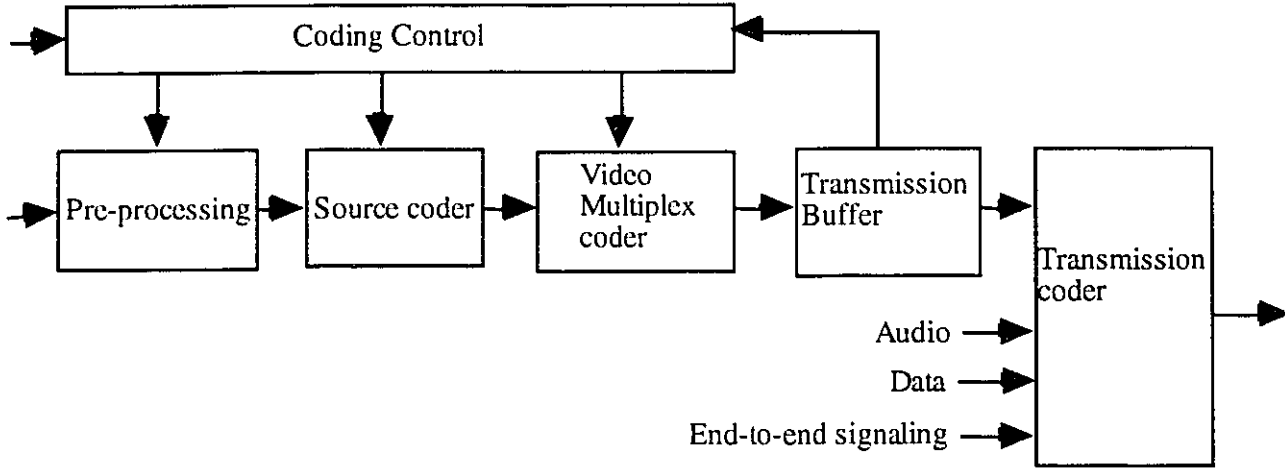


Figure A.1 : Outline block diagram of H.261 video encoder

Appendix B: A Brief Description of OPNET

The OPNET simulation package designed by MIL 3 Inc., is a modeling and analysis tool based on discrete event simulations. OPNET is designed to provide a comprehensive development environment supporting the modeling and performance evaluation of communication networks and distributed systems. The system behavior and performance are analyzed by performing discrete event simulations. The key features of the OPNET package are:

- **Object oriented**

OPNET models are specified in terms of objects, each with configurable sets of attributes. The package supports flexible definitions of new objects with programmable characteristics and behavior in order to address as wide a scope of systems as possible.

- **Specialized in communication networks and information systems.**

Model building blocks focus primarily on communications and information processing to accelerate development efforts for networks and distributed systems.

- **Hierarchical models**

The OPNET models are hierarchical to naturally parallel the structure of actual communication networks. There are four levels in the hierarchical models: the Network level, the Node level, the Process level and the Parameter level. This feature, plus the object oriented feature, makes the OPNET models easily to be designed and understood.

- **Graphical specification**

Wherever possible, models are entered through graphical editors that provide an intuitive mapping from the modeled system to the OPNET model specification.

- **Flexibility to develop detailed custom models**

The OPNET provides a flexible, high-level programming language with extensive support for communications and distributed systems. This environment allows the rapid modeling of communication protocols and algorithms.

- **Automatic generation of simulations**

Model specifications are automatically compiled into executable, efficient, discrete-event simulations implemented in the C programming language.

- **Integrated analysis tools**

Performance evaluation, and trade-off analysis require large volumes of simulation results to be interpreted. The OPNET includes a tool for graphical presentation and processing of simulation output.

- **Advanced debugging**

All OPNET simulations automatically incorporate support for program testing through a sophisticated interactive debugger.

All the results in this thesis were obtained by using the simulation models developed in OPNET. Apart from the programming and graphical model representation facilities OPNET has nice data analysis tools. The output scalar files can be selected and the scalar parameters from the same files can be plotted against each other. Confidence intervals can be computed and displayed for the scalar plots that are contained in analysis panels. In all the performance evaluation curves in this thesis, each point on the vertical axis represents the mean of a number of values obtained

from runs of the same simulation model (with same parameters but different random number seeds), with a close confidence interval.

For N statistically independent data values the confidence interval for each point is calculated using the relation

$$\bar{w} - t_{\alpha/2, f} \sigma \leq w \leq \bar{w} + t_{\alpha/2, f} \sigma \leq \bar{w} \quad (\text{B-1})$$

where \bar{w} is the average of the N values
 σ^2 is the variance of N values
 $t_{\alpha, f}$ is the 100(1- α) percentage of a t distribution with f degrees of freedom: that is, $t_{\alpha, f}$ is defined by $p(t \geq t_{\alpha, f}) = \alpha$.

In our analysis we have N = 4, $\alpha = 0.1$ and f = N-1 = 3. The 90% confidence interval obtained using these parameters is not displayed on the curves to retain the clarity of figures.

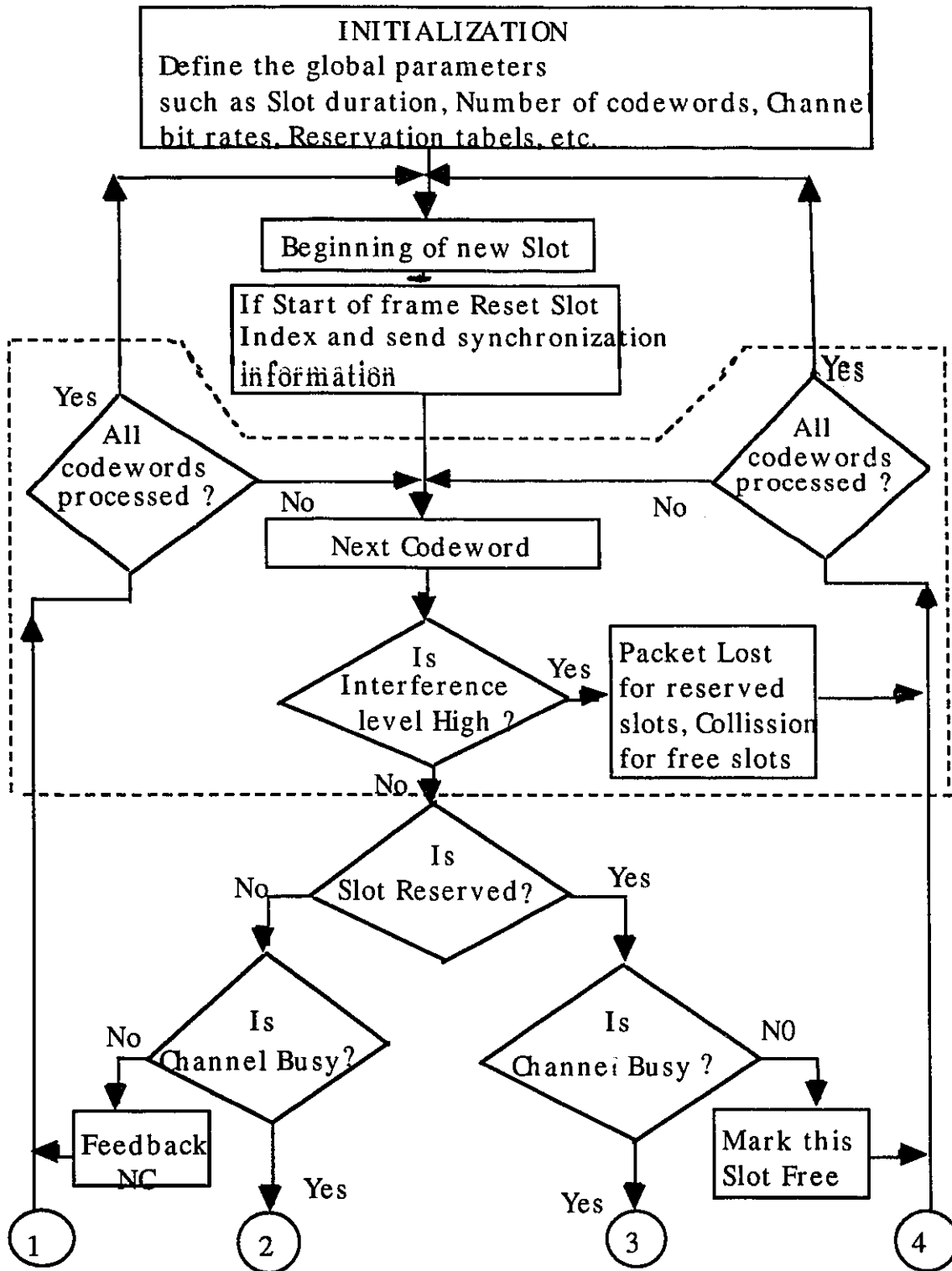
Appendix C. Simulation Models

C.1 Base Station Model

The base station needs a significant processing power, because of the many different requirements of the system and mobile users. The base station functionality may include, channel access control, hand off procedures, interface with other base stations and the wired communications network, error detection/correction, handling different traffic types, etc.

Figure C.1 displays the flow chart of the implementation of uplink channel access algorithm at the base station. This is the general flowchart algorithm for CDMA technologies. For the TDMA versions of the protocol the steps encircled by dotted lines is not required. For CDMA we assume the assignment of codewords on timeslot basis. Clearly the base station architecture is more complex for CDMA versions of the protocols. Also the model represented in this diagram handles multimedia traffic.

Here we assume two types of feedback. Firstly the base station broadcasts the status (Reserved, Free) of next slot. Secondly the base station gives feedback C and NC (about the outcome of the transmission), only for free slots.



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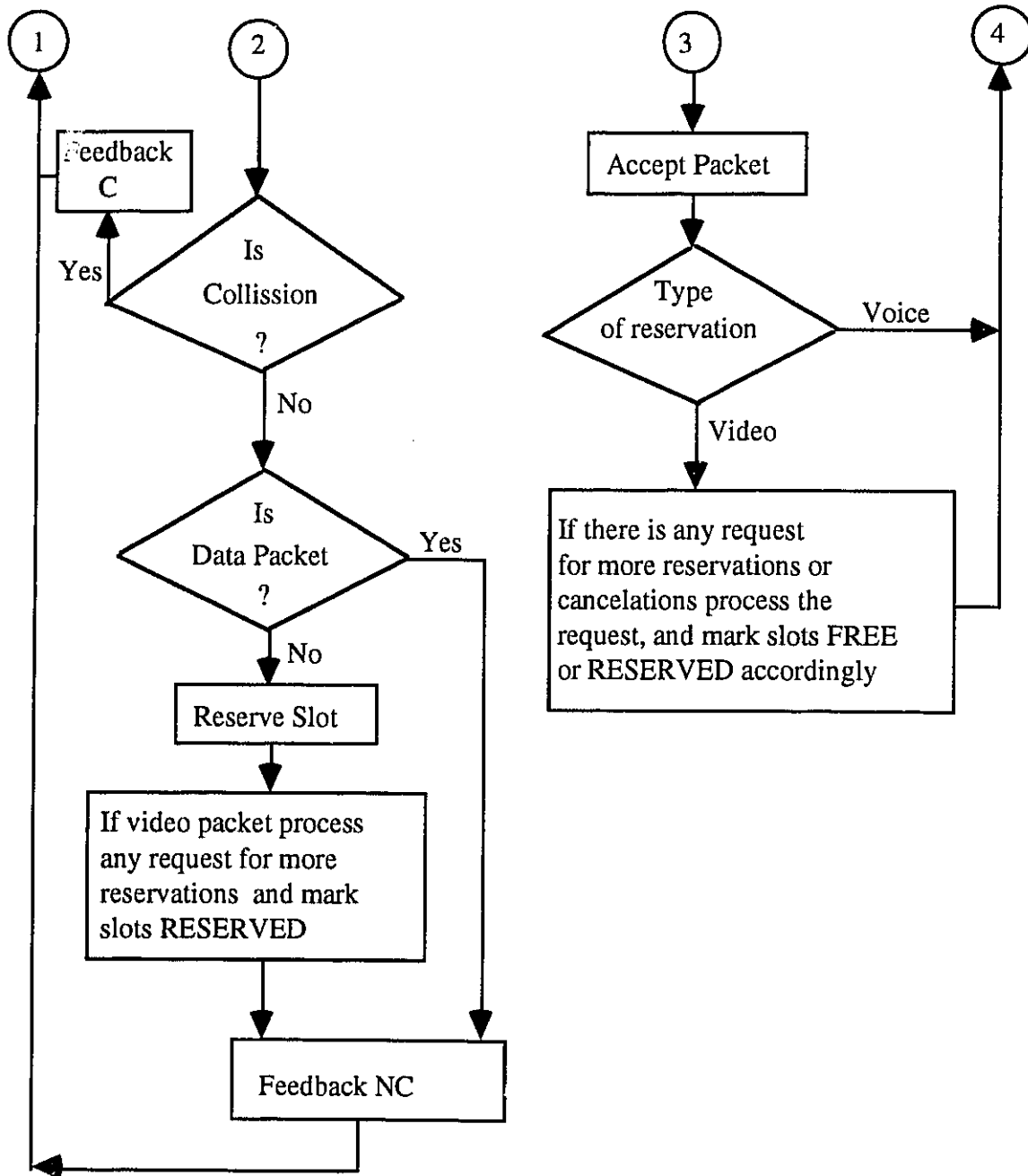


Fig. C.1 -- Flow chart representation of uplink channel reception part at Base Station

C.2 Mobile Data Terminal Model

Each data packet goes through the processing steps shown in figure C.2 implemented at mobile data terminal. Each newly generated data packet has to listen the feedback channel and wait for the completion of the ongoing CRP. At the start of new CRP this data terminal checks the eligibility of the packet to join the next CRP according to the steps in figure 2.5. Once the packet is able to join the CRP it follows the steps of figure 2.7 with appropriate Z-cell algorithm, to resolve the collisions among competing packets, until it goes through. Note that due to irregular arrival times of the data packets and their relative insensitivity to delay, the system does not reserve any time slots for them. Hence each data packet has to go through the collision resolution process.

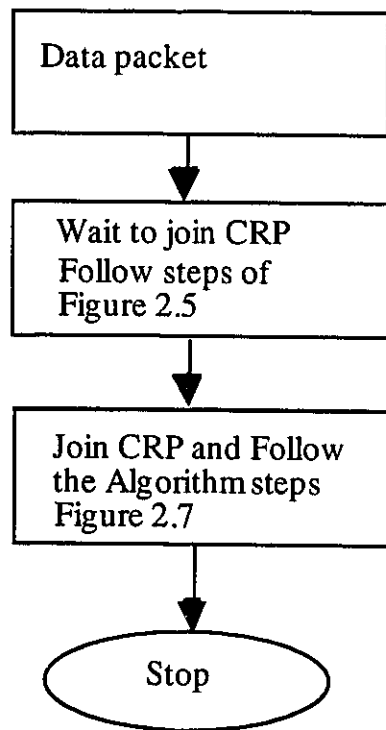


Figure C.2 Data Terminal Logic Implementation

C.3 Mobile Voice station Model

The processing steps at the mobile voice station, for the duration of a connection, are shown in figure C.3. In each frame the mobile voice terminal checks if there is any voice packet pending for transmission. If the voice encoder is in silence mode and there is not any packet then the voice station would wait for the next frame. If there was any reservation going on for this voice connection, then that slot would go free and this voice terminal would lose reservation.

If there is a packet in the current frame then two cases exist. In the first case a slot is already reserved for this voice connection. Then the voice station waits for this reserved slot and transmits packet in it. In the 2nd case there is no reservation for this connection. In this case the voice packet first waits for the start of new CRP according to the rules followed in figure 2.5. Then it follows the steps of figure 2.7 to resolve the collisions among competing packets, until it goes through. With the successful transmission of this packet, that slot would be reserved for this voice connection for the coming frames. Note that the packets going through the CRP ignore the reserved slots and the CRP rules apply only to the un-reserved slots.

C.4. Protocol Implementation In Mobile Video Terminal

The flow chart for the implementation of VBR video part of the CDMA version of proposed protocol at mobile station is shown in figure C.4.

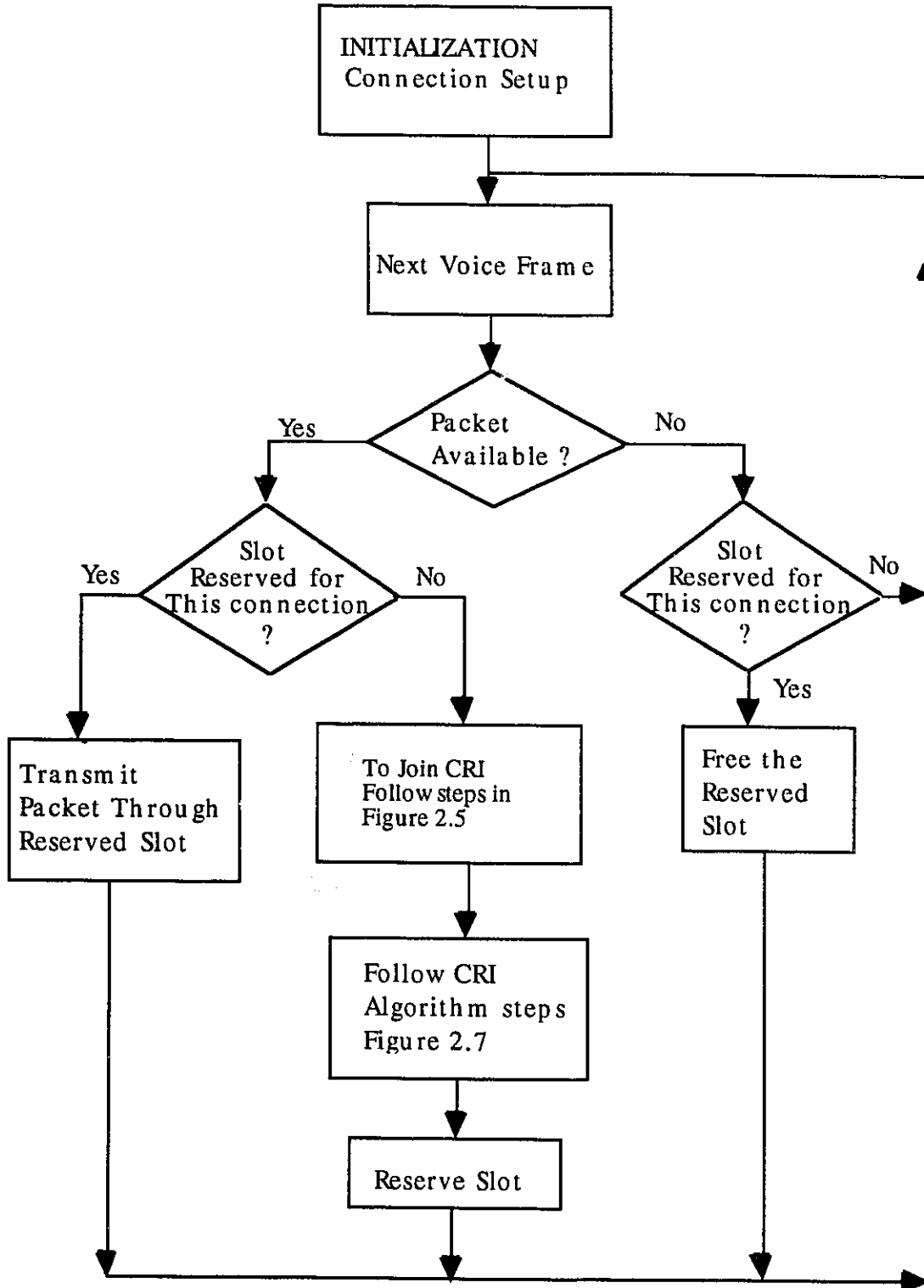


Figure C.3 Voice Station Logic Implementation

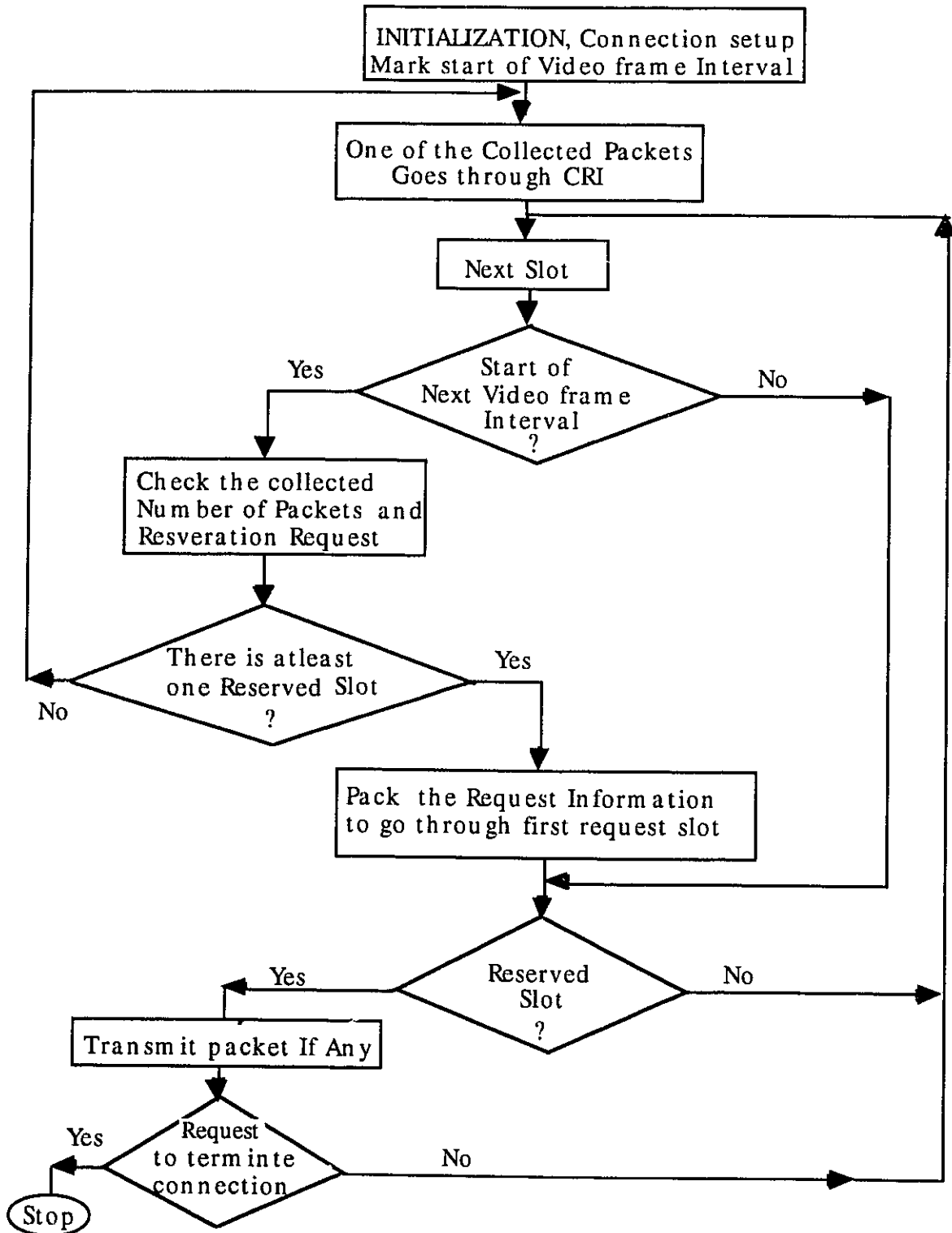


Figure C.4 Mobile VBR Video Station Logic Implementation

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