PERFORMANCE EVALUATION OF VOICE SERVICE OFFERED BY HIGH-SPEED WIRELESS PACKET DATA

By

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ABSTRACT

Good quality voice service is characterized by low delay and low frame error rate. Currently cellular networks provide voice services by using circuit-switched technology. There is an increasing demand for multi-media wireless services. New generations of wireless networks are required to support various data services, such as web browsing, FTP and Video on Demand. These services require high-speed data transmission. In response to such demands, new wireless multiple access techniques are being proposed. New techniques such as AMCS, hybrid ARQ, shared channel and fast-scheduling algorithms based on channel variation had been proposed and evaluated by the 3G wireless networks.

In this thesis, an attempt is made to explore the possibility of transmitting the voice application using the proposed high-speed packet transmission algorithms and techniques [1]. This possibility could eliminate the need of having different technologies for voice and high-speed data application.

This thesis shows that voice service offered by proposed high-speed scheduling techniques, when employed in the forward link, could meet the performance offered by circuit-switched voice application for a certain number of users in the cell. However, these techniques become inefficient, in terms of throughput, when employed to support voice applications. This thesis also proposes the required changes in the transmission times scheduling rules specified in [1] in order to improve the throughput and delay performance of the proposed high-speed packet data transmission techniques.

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Acronyms

8-PSK	8-Phase Shift Keying
16-QAM	16-Quadrature Amplitude Modulation
ACK	Acknowledgement
AMC	Adaptive Modulation and Coding
ARQ	Automatic Repeat Request
BS	Base Station
BSC	Base Station Controllers
bps	Bits Per Second
C/I	Carrier to Interference Ratio
CDMA	Code Division Multiple Access
cdma 2000	3 rd Generation Standard for CDMA Networks
c.l.	Cell loading
CN	Core Network
CSMA/CD	Carrier Sense Multiple Access/ Collision Detection
DRC	Data Rate Request Channel
DTE	Data Terminal Equipment
FDM	Frequency Division Multiplexing
FEC	Forward Error Correction
FER	Frame Error Rate
FIFO	First-In-First-Out
FPDCH	Forward Packet Data Channel
FTP	File Transfer Protocol
HDR	cdma2000 High Rate Packet Data Air Interface Specifications
IR	Incremental Redundancy
IS-95	2 nd Generation CDMA Standard
ITU-T	International telecommunication Union- Telegraph and Telephone
kbps	Kilobits Per Seconds
LAN	Local Area Network
MAC	Medium Access control

Mbps	Mega Bits Per Seconds
MSC	Mobile Switching Center
msec	Milli Second
MT	Mobile Terminal
NACK	Negative Acknowledgement
PCSSR	Proposed Changes in Specified Scheduling Rules
PDN	Packet Data Network
PDR	Packet Discard Rate
PFS	Proportional Fair Scheduler
PN	Pseudonoise
PPP	Point-to-Point Protocol
PSTN	Public Switched Telephone Network
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RAN	Radio Access Network
RC	Radio Configuration
RF	Radio Frequency
RLP	Radio Link Protocol
RS1	Data Rate Set 1 for IS-95
RS2	Data Rate Set 1 for IS-95
RSVP	Resource Allocation Protocol
RTP	Real Time Protocol
RTCP	Real Time Control Protocol
SMTP	Simple Mail Transfer Protocol
SSR	Specified Scheduling Rules
TCP/IP	Transmission Control Protocol/Internet Protocol (Protocol Stack)
TDM	Time Division Multiplexing
UDP	User Datagram Protocol
MIU	Mobile Interface Module
WAN	Wide Area Network
WWW	World Wide Web

CHAPTER 1

INTRODUCTION

1.1 Motivation

Traditionally, mobile and cellular networks use circuit switched technologies to provide good quality voice and low rate data services to their users. There is an increasing demand for multi-media wireless services. New generations of wireless networks are required to support various packet-switched data services, such as web browsing, FTP and Video on Demand. These services require high-speed data transmission.

The characteristics of packet data applications are different from those of low speed circuit data. A session of a packet data application is sustained for longer period of time than a voice call. But activity of the session is relatively lower than that of the voice call. Therefore, circuit-switched technology becomes inefficient when used to support high-speed data applications. Currently there are several ongoing efforts to define cellular network architectures that would enable fully packet-switched IP based service delivery; i.e. not only data, but also speech service would be provisioned over IP bearer. Such networks architectures are usually referred as “All IP networks”.

The 3rd generation Partnership Project 2(3GPP2) is a consortium of national standards bodies, tasked with developing architectures and standards for third generation cellular networks. It has developed a new packet data architecture built on the CDMA 2G and 3G air interface data services [2]. This architecture is shown below:

Figure 1.1 3GPP2 All IP network [2]

Mobility support will be provided using IETF's Mobile IP Protocol. Access to 2G networks and PSTN will be provided via gateways. The authentication, authorization, and accounting (AAA) infrastructure will enable the visited data network to query the home data network for authentication credentials and will ensure payments for services rendered. This is in addition to the existing home/visiting location register (HLR/ VLR) based authentication.

Packet data serving node (PDSN) implements the features of a network access server (NAS) and a Mobile IP FA. PDSN is similar to existing NASs that provide dial up Internet access in the sense that both terminate a link-layer protocol and are directly connected to the Internet. It will be the first hop IP router and perform Diffserv shaping, ingress filtering, handoff between IP based radio access network node, called Packet Control Function (PCF). An IP network will connect PCF and PDSN.

Both the base station controller (BSC)/SDU and BTS are contained in PCF. It means that BSC will be a router based IP node containing some critical radio control functions (e.g. power control, soft handoff, frame selection). PCF is responsible for relaying data to and from PDSN and for insulating the PDSN from certain wireless –specific aspects of the architecture such as the current state of the air interface. The remaining control functions, such as call/session control, mobility management, and gateway functions, are moved to the managed IP network.

Air interface must be capable of supporting the higher data rates necessary for packet data applications. The proposed architecture will make use of cdma2000 air interface, the high-speed Code Division Multiple Access (CDMA) standard, that is an evolution of IS-95.

Cdma2000 High-rate packet data air interface (HDR) has been specified to support data applications on a separate carrier. 1xEV-DV air interface specification of cdma2000 will be used to support high-speed data mode that does not require circuit-switched connection. Voice will still be transmitted over dedicated circuit-switched channel while high-speed data applications users will share a packet-switched common channel. Both voice and data applications will share the same bandwidth.

To support high-speed data transmission reliably and efficiently, scheduling algorithms such as Proportional Fair and maximum C/I have been proposed. These algorithms make use of proposed high-speed data transmission techniques such as Adaptive Modulation and Coding (AMC), Hybrid ARQ and fast scheduling based on channel condition.

In this thesis, assuming the cdma2000 High-rate Packet data specifications (HDR) [1], we explore the possibility of supporting the voice service by the same set of techniques. We focus our work on the common downlink (from base station to mobile user) channel and we use most widely proposed high-speed data transmission scheduling algorithms and techniques. This possibility might eliminate the need of having separate technologies for voice and data applications.

Voice traffic is characterized by low and constant delay. For better voice quality, packet transfer delay between end users should be kept to a certain limit. Studies (the ITU-T G.114 Recommendation [3]) show that one-way packet transfer delay for a telephone conversation should be below 150 ms to maintain a coherent conversation between the end users. A round trip delay of 300ms is not discernable to even the most sensitive listener. Round trip delays of about 500ms result in a loss of conversational efficiency of about 25% [p.113, 3]. A wireless network offers a good quality voice service using circuit-switched technology with 40msec [4] constant delay if a voice call is between a telephone and a mobile user. It also offers a good quality voice service using circuit-switched technology with 80msec constant delay if a voice call is between two mobile users via telephone network.

Another measure of quality is packet loss rate. In CDMA 2G systems, frame error rate of 1% is considered acceptable for voice service.

In this thesis, we measure the performance of the voice service by packet transfer delay and packet discard rate. We also measure the performance of the proposed high-speed packet data transmission techniques, in terms of supportable data rates and effective data rate. We conduct this study by simulation.

The simulations show that the known scheduling algorithms, with the help of the proposed high-speed packet data transmission techniques, could meet the performance offered by circuit-switched voice service application for a cell loading of 0.52 (15 users).

However, we observed that the proposed high-speed packet data techniques are inefficient, in terms of achievable throughput, when employed to support voice applications.

Maximum C/I performs worse than PFS and FIFO as far as delay performance is concerned. Delay performance of FIFO is better than that of PFS, but it performs worse than the other two scheduling algorithms as far as supportable data rates are concerned. The PFS not only offers the delay performance, comparable to that of FIFO, but also offers supportable data rate performance, comparable to that of the Maximum C/I. The PFS could be a suitable choice to meet the contrasting lower delay requirement of the voice applications and the higher throughput requirement of the data applications.

1.2 Contributions

In this thesis, we compare the delay and packet loss performances offered by specified scheduling rules (SSR) [1] with our proposed changes in specified scheduling rules (PCSSR) using the widely proposed scheduling algorithms. We observed that with

PCSSR, delay and packet discard rate performances of the voice services offered by the proposed algorithms improve.

We also compare the voice service performances offered by the proposed scheduling algorithms using PCSSR with the voice service performance offered by circuit-switched technology.

We compared the voice service performances offered by the proposed scheduling algorithms. We observed that the PFS could be suitable choice between the contrasting lower delay requirement of the voice applications and the higher throughput requirement of the data applications.

This thesis provides the values of residual capacity and available delay budgets that could be used to explore the possibility of fully IP-based voice service between two 3G-wireless network users via Internet and to explore the possibility of transporting the mix of voice and data traffic.

1.3 Organization

This document is organized as follows:

In chapter 2, we present the background, which consists of overviews of circuit-switched and packet-switched technologies. We also describe CDMA technology, techniques to support high-speed data transmission, options to support data applications in traditional (IS-95) and proposed cdma2000 specifications. Finally, we describe relevant features of High-rate packet data specifications [1] that will be used as packet-switched air-interface for cdma2000.

In chapter 3, we present our simulation model and the assumptions we made about the high-speed data transmission techniques. We describe the scheduling algorithms and list the causes of delay in schedulers.

In chapter 4, we describe the simulation and the simulation results.

In chapter 5, this document is completed with concluding remarks and suggestions for future work.

CHAPTER 2

BACKGROUND AND RELATED WORKS

2.1 Introduction

An end-to-end communication link generally consists of a series of switching nodes. The first and the last switching nodes are connected with data terminals. These terminals are used to transmit and receive the information. The switching nodes within the network are connected with other switching nodes via links, which are normally multiplexed using either frequency division multiplexing (FDM) or time division multiplexing (TDM). A data terminal is given a frequency or a time slot, which is called a channel. Two different technologies are available to transfer the data among users connected with the network: Circuit-switching and packet switching. In section 2.2, an overview of circuit-switched technology is presented. In section 2.3, an overview of packet switched technology is presented, taking the Internet as an example. In section 2.4, we describe circuit-switched cellular network and Code Division Multiple Access (CDMA) access techniques. In section 2.5, we describe the mechanisms and techniques, proposed for wireless networks, to support high-speed packet data applications. In section 2.6, we describe how CDMA systems support high-speed packet data applications.

2.2 Circuit-switched Technology

Communication in circuit-switched networks is done in three phases. First an end-to-end (between two users) dedicated connection via one or more switching nodes is established.

In the second phase, the information is exchanged between the users. In the third and final phase, when the information has been exchanged, the sender or receiver sends a message to the network to terminate the connection. In response to the message, the network releases the connection. An example of a circuit-switched connection is shown in figure 2.1.

Figure 2.1: An example of a circuit-switched connection.

Circuit-switched technology has the following three characteristics:

1. The capacity is reserved for the duration of a call, whether it is used or not.
2. Long connection setup delay and virtually no switching delay.
3. Works at constant rate. End terminals engaged in data transfer must transmit and receive at the same data rates.

New data services such as web browsing and email are highly bursty in nature. They consist of a short information transfer period following the long idle period. During the idle period, link capacity reserved for the session will be wasted when circuit-switched technology is used.

2.3 Packet-switched technology

In packet data networks, information is transferred as data blocks, which are called packets. No call set up procedure is required. Each packet carries all the information required for routing and switching between sender and receiver. A brief overview of packet switched technology is given in the following paragraphs taking the example of the Internet, which is the most widely used packet switched network.

In the Internet, data is transferred through several tasks. These tasks are organized in a layered structure. Each layer is responsible for a specific task. The Internet uses the TCP/IP protocol stack, shown in figure 2.2.

Figure 2.2: TCP/IP Reference Model [p. 36, 5].

The physical layer is responsible for activating, maintaining, and deactivating of physical circuits between data terminals (DTE). The data link layer provides framing, error control and flow control across one communication link. Each data terminal (DTE) is assigned a hardware address. This layer is responsible to identify the hardware address associated with the incoming packet.

The Job of the network layer is to route the traffic between networks. IP is the key protocol at this layer. It provides routing and addressing facilities. Switching nodes in the network, route the packet based on the IP address present in a packet.

The transport layer resides in the end-terminals and provides end-to-end reliability and congestion control. TCP and UDP are the two main protocols in this layer. TCP provides connection-oriented services. Using retransmission schemes, TCP tries to correct packet errors and packet loss. It also provides flow control. UDP provides connectionless service. It can detect error but does not use retransmission for error recovery.

The application layer provides direct interface to users for utilizing the services of lower layers. Some of the most widely used applications are File Transfer Protocol (FTP), Simple Mail Transfer Protocol (SMTP) and World Wide Web (WWW). Data terminals run the full protocol stack. Switching nodes, in the network, use the lower three layers.

The Internet is composed of two types of networks. These are Local Area Networks (LAN) and Wide Area Networks (WAN). In a LAN, computers are attached to a shared link. A switching node, which is called a router, is also connected to the same link. To implement a LAN, many technologies are available. One of the most common technologies is **Ethernet**. It works this way: Each computer controls its own transmission using a protocol called **Carrier sense multiple access/ collision detection (CSMA/CD)**. It senses the link and transmits if no other computer on the link is transmitting. All the computers on the link receive the packet. They discard the packet if it is not addressed to them. Only the computer, whose hardware address is present in the packet, accepts the packet. If a packet is destined to a computer attached to a remote LAN, the router on the

same link receives the packet and, based on the destination IP address present in the packet, switches the packet to one of the output buffers, it has for each link.

Packet switched technologies running between LANs are called WANs. A WAN consists of routers, which typically support several WAN technologies like IP, SNA, AppleTalk and X.25. Routers that connect WAN and LAN have physical layers of both LAN and WAN interfaces.

The Internet was designed for data applications. It provides highly efficient and robust services for data applications. But the Internet has major problems to support real-time applications. These problems include:

1. No session admission control is implemented in the Internet. This results in unpredictable queuing delays in the routers along the path between two users. And, packet transfer delay is not constant.
2. If packets arrive too quickly for a router to process them, then the buffer becomes full, and the following packets are discarded. Therefore packet loss occurs.
3. No dedicated path is set between the sender and the receiver. The routing decision at each router is made based on some defined criteria, such as link utilization and delay. It is not uncommon that different routes to be chosen for the different packets in a data transfer between one user to another. Therefore, the packets may follow the different routes to reach the same destination. Due to the different queuing delays at routers in different routes, packet transfer delays for the packets for the same user are different. Therefore, the packet order is disturbed.
4. Routers normally work on “first in first out” basis. They do not classify the traffic. No specific throughput, delay or packet loss is guaranteed.

Various mechanisms, such as TCP, are available to combat against congestion and packet disorder. But all of these mechanisms and schemes for reliable transmission introduce additional delay in addition to queuing, transmission and propagation delays.

New Internet models (integrated services [6], Differentiated services [7], label switching [3], guaranteed QoS [8], Controlled load [9] and new protocols (RTP [3], RTCP[3], RSVP[10] etc.) have been proposed to provide guaranteed quality of service (such as bounded and constant delay) to the real-time application users.

2.4 Cellular Network

Wireless cellular networks use circuit-switched technology and provide access to a wire-line communication network, such as PSTN. The major components of a cellular network are shown in the figure 2.3.

Figure 2.3: Major components of a cellular access network

MIU stands for Mobile Interface Module. It is a standard interface, which connects the information source to a Mobile Terminal (MT). MT performs information coding (e.g.

speech coding), channel coding, and modulation etc. The Radio Access Network (RAN) consists of Base stations (BS) and a Base station Controller (BSC). The Core Network (CN) consists of Mobile switching centers (MSC) and other relevant identities, such as various databases. In second-generation wireless networks (2G), the MSC is the main hub that interfaces with other external networks (e.g. PSTN, Internet etc.).

The coverage area is divided into blocks, which are called cells. Each cell supports a number of mobile users. The BS provides speech coding, channel coding and modulation etc. and transmits information to the users. The BS also receives the information from the mobile users and retransmits the information to the CN. It means that transmission of information in a cell is centrally controlled through BS, as opposed to the Internet. The wireless link from BS to the mobile users in a cell is called **“forward link”** or **“downlink”**. The wireless link from a mobile user to the base station is called **“reverse link”** or **“uplink”**.

To eliminate the interference and to uniquely identify the users in the system, three types of access techniques are available: Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA) and Code Division Multiple Access (CDMA). In FDMA, the users in a cell are assigned to separate frequency bands. In TDMA, a specified number of users in a cell are assigned to the same frequency band and separate time slots.

As cdma2000 (the proposed air interface for 3GPP2 All IP Networks) makes use of the CDMA technology, we describe it in the following paragraphs:

In CDMA, the users in a cell can transmit on the same frequency band and are allowed to transmit at the same time. Unique bit sequences, called binary chip sequences, are used as

a multiple access technique in a CDMA system. This technique in frequency domain is shown in the figure 2.4. In a BS, Information bits of each user in the system are multiplied with the assigned sequence. This multiplication process is called spreading. After the spreading, information bits of each user become a noise-like signal. These noise-like signals are added together, and then, transmitted toward the users. When a user multiplies the received noise-like signals with the sequence assigned to it, the original information bits are recovered. The multiplication process at the receiver is called de-spreading. Received signals destined for the other users are not despread, and remain noise like signals.

Figure2.4: CDMA technique

2.5 Mechanisms and techniques to support high-speed packet data applications

To achieve high performance for high-speed data rate applications, key techniques that have been proposed include **higher order modulation** to achieve high data rate, **fast scheduling** (supported with features like data rate selection based on C/I feed back and hybrid ARQ.) centered at base station and the use of **shared channel**.

2.5.1 Adaptive modulation and coding Scheme (AMCS)

In cellular communication systems, the quality of a signal received by a receiver depends upon a number of factors—the distance between the desired and interfering base stations, path loss exponent, log-normal shadowing, short term Rayleigh fading and noise. In order to improve system capacity, peak data rate and coverage reliability, the signal transmitted to and by a particular user is modified to account for the signal quality variation through a process, commonly referred to as link adaptation.

Traditionally, CDMA systems have used fast power control as the preferred method for link adaptation. Adaptive Modulation and Coding (AMC) offers an alternative link adaptation method to raise the overall system capacity. AMC provides the flexibility to match the modulation-coding scheme to the average channel conditions for each user. With AMC, the power of the transmitted signal is held constant over a frame interval, and the modulation and coding format is changed to match the current received signal quality or channel conditions. AMC combined with time domain scheduling offers the opportunity to take advantage of short-term variations in a receiver's fading envelope so that a receiver is always being served on a constructive fade.

2.5.2 Scheduling

Wireless links are subject to time and location-dependant signal attenuation, fading, interference and noise that result in bursty errors and time-varying channel capacities. Scheduling algorithms are important components in the provision of efficient channel utilization. One way to achieve effective channel utilization, which has been proposed in the literature for the schedulers used in the third generation wireless networks (3G), is as follows:

On the basis of the received transmission from an access point (e.g. BS), each access terminal (e.g. MT) can measure carrier to interference ratio (C/I). The data rate that can be supported to each user is proportional to its received C/I . This may change continuously, especially for mobile users. The value of the C/I is then mapped to a value representing the maximum data rate for a given level of error performance.

Scheduling algorithms are also responsible to provide guaranteed quality of service. It should provide throughput and delay bound guarantees for individual sessions. It should provide fairness among sessions with good channel conditions and sessions with bad channel conditions.

2.5.2.1 Selection of users for transmission

The scheduler selects a user based on a selection criterion and schedules the time of transmission and allocates the required resources. Many user selection algorithms have been proposed in the literature. We will describe the scheduling algorithms assumed in this thesis in chapter 3.

2.5.3 Hybrid ARQ scheme

The implementation of AMC offers several challenges. First, AMC is sensitive to measurement error and delay. In order to select the appropriate modulation, the scheduler must be aware of the channel quality. Errors in the channel estimate could cause the scheduler to select the wrong data rate. Delay in reporting channel measurements also reduces the reliability of the channel quality estimate due to the constantly varying mobile channel. Furthermore changes in the interference add to the measurement errors. Automatic Repeat Request (ARQ) enables the implementation of MCS by reducing the sensitivity to measurement error and traffic fluctuations.

In a conventional ARQ scheme, if a receiver detects errors in the received information block, the receiver discards the information block, and sends a request to the transmitter to retransmit the information block. In the ARQ scheme based on **'stop-and-wait'**, after sending an information block, a transmitter waits for feedback from the receiver before sending the next information block. If the transmitter receives a negative acknowledgement (NACK) from the receiver, the transmitter then retransmits the same information block. This process continues until the correct reception is done and the transmitter receives the ACK from the receiver. In the ARQ scheme based on **'go back n'**, a transmitter sends more than one outstanding information block without waiting for feedback from the receiver. In case of error, the receiver discards all the subsequent information blocks and sends no acknowledgements. When the transmitter learns this (using a timer), it retransmits all the subsequent blocks. In the ARQ scheme based on **'selective repeat'**, a transmitter sends more than one outstanding information block

without waiting for feedback from the receiver. The transmitter retransmits only those information blocks, which are reported to be erroneous by the receiver.

The conventional ARQ schemes can be effective for a burst-error channel. But if a channel remains bad, for a longer period of time, then the conventional approach performs worse in terms of throughput and delay. The ARQ scheme that has been proposed to support high-speed data in a wireless environment is hybrid ARQ (ARQ combined with FEC). In a HARQ scheme, a receiver makes use of the coding information in the retransmissions and tries to optimize the code rate for an unknown channel capacity. The two fundamental forms of HARQ are Incremental Redundancy (IR) and code combining: In the code combining, the same encoded packet is retransmitted for a specified number of retransmission. The receiver tries to decode the packets combining the received packet in each retransmission. Code combining is a special case of IR. In an IR scheme, each retransmission provides new code bits from the mother code to build a lower rate code. In the following sub-section, we describe a proposed hybrid ARQ scheme.

2.5.3.1 Incremental Redundancy

In an IR scheme, information is encoded in many sub-blocks called encoder blocks shown in figure 2.5. The IR operation steps are as follows [11]:

- First encoder sub-block is transmitted.
- If the intended receiver decodes the received sub-block successfully, it sends an ACK to the sender. On receiving the ACK, the sender discards the remaining encoder blocks (that are waiting for transmission).

- If the intended receiver does not decode the received sub-block successfully, it sends an NACK to the sender.
- The sender sends the next encoder block.
- The receiver tries to decode by combining the coding information in these two encoder blocks. If the decoding is successful, it transmits an ACK to the sender. Otherwise it sends a NACK again.
- This process continues until the receiver decodes successfully or the last encoder sub-block is transmitted.

In adaptive IR, coding rates for encoder sub-blocks can be chosen from the given set depending upon the channel conditions. Referring to figure 2.5, encoder sub-block#1 can be coded with code rate of 0.2 or encoder sub-block#4 can be coded with code rate of 0.8. Redundancy can be obtained by repetition instead of coding.

Figure2.5: An example of coding for Incremental redundancy [11]

2.6 Support for data applications in CDMA systems

The following sections describe the evaluation of CDMA systems to support high-speed packet data applications.

2.6.1 Options to support packet data applications in IS-95

IS-95 is a CDMA cellular standard that was designed for voice and low-speed data applications. It specifies 64-chip orthogonal sequences called Walsh function for channelization. In addition to that, one pilot sequence is specified for base station identification and for synchronization purposes. For user identification, a unique chip sequence called long sequence is used. Each user is assigned the same long sequence with a different phase.

Figure 2.6 shows the support for packet data applications available in IS-95. Voice or user data is first put into packets. These packets include protocols headers of TCP/IP/PPP or UDP/IP/PPP. Then the packet is delivered to the Radio Link Protocol (RLP). The RLP is used for framing and ARQ. It delivers the information packets to the physical layer, which performs channel coding, spreading and modulation operations. Then packets are transmitted through the air on an assigned dedicated traffic channel.

In IS-95, the maximum supported data rate per traffic channel is 9.6 kbps for Rate set 1 (RS1) and 14.4 kbps for rate set 2 (RS2). To support higher data rates, more dedicated traffic channels can be assigned to a user. These additional channels are called **supplemental code channels**. Up to 7 supplemental code channels can be assigned to a user.

Figure 2.6: Support for packet data applications in IS-95 [p.12, 12]

Typically, the tasks of efficient resource utilization and provision of Quality of Service (QoS) is given to the MAC layer in a wireless system. In IS-95, the MAC layer has two states. These states are **active** and **dormant**. In the active state, the traffic channels are assigned; RLP, PPP and higher protocols are setup. When the user becomes inactive, the MAC layer remains in the active state, and waits for a specified period for the user to become active again. During this waiting period, the user keeps the resources. These idle and assigned resources cannot be assigned to other users.

When the waiting period ends, the MAC layer enters into the dormant state. In this state, traffic channels are released; RLP, PPP state information is discarded; and no resources

are allocated for the user in the BS or the MSC. If the user becomes active again, a traffic channel has to be assigned again, RLP and PPP protocols have to be setup, and the necessary resources have to be allocated and setup in the BS and the MSC. This reallocation of resources introduces substantial delays noticeable to the user.

For high-speed data applications, that have shorter activity periods and longer inactivity periods than voice application, the two-state MAC model becomes more inefficient. IS-95 does not include the techniques and features such as AMC technique and fast scheduling to provide reasonable performance to high-speed data rate applications.

2.6.2 Options to support packet data applications in cdma2000

cdma2000 [12] is a set of specifications for 3rd generation mobile systems, which employs circuit-switched technology and which is the evolution of the IS-95 CDMA systems. Figure 2.7 shows the services provided by IS-95 and the new capabilities provided by cdma2000. cdma2000 is designed to provide various user services like voice (packet and circuit modes), packet data, circuit data and simultaneous voice and packet data. It provides protocols and services that correspond to the bottom two layers of the TCP/IP reference model shown in figure 2.2.

cdma2000 provides a variety of physical layers, called Radio Configuration (RC), to support various quality-of-service requirements. Its two physical layers called RC1 and RC2 are the same as in IS-95. Its physical layers RC3 to RC5 uses 1.25MHz RF bandwidth and support higher data rates and throughput than RS1 and RS2 using variable length Walsh codes, and a new powerful channel coder called turbo coder. The target data rate on 1.25 MHz carrier is 230.4kbps. A further increase in data rate is possible

using higher RF bandwidths (3.75, 7.5, 11.25 and 15MHz.), Multi-carrier and new supplemental channels. The maximum target data rate is 1Mbps.

Figure 2.7: IS-95 and cdma2000 layer structure [p.12, 12].

To improve the resource utilization and to provide better support for packet data applications, two more MAC states have been proposed in the specifications. These new additional MAC states are “**control hold**” and “**suspended**” states. Active state is very similar to active state in IS-95. When a user enters inactive state, the MT goes to dormant state in steps.

In the first step, it enters into the control hold state. In this state, traffic channels are released. But control channels are maintained. RLP and PPP states are also maintained. If

a user wants to get back to active state, only traffic channels need to be allocated. Fast traffic channel assignment message is possible over control channels.

In the next step, if the user continues to be inactive, MT enters from the control hold to the suspended state. In this state, control channels are released. But RLP and PPP states are maintained. If the user wants to go back to active state, the MT enters the control held state first, and then enters into the active state. The return from the suspended state to active state is not as fast as the return from the control hold state to the active state. But as the RLP and the PPP states are maintained in the suspended state, the assignment of traffic channel is faster as compared to IS-95.

If the user remains inactive, then finally the MT enters into the dormant state. This state is similar to the dormant state in IS-95. But as the PPP state is maintained, the traffic channel assignment is relatively faster as compared to IS-95. In this state, the MT is allowed to transfer short data bursts over common channels without requiring the setup of a dedicated traffic channel.

In this way with the help of the additional MAC states, reduction in the delay in re-allocating the resources and improvement in the link utilization are expected.

2.6.3 cdma2000 High-Rate Packet Data (HDR)

To provide a high-speed data rate transmission using packet-switched technology, the air interface standard, which is referred to as HDR [1] has been proposed as an extension of the existing cdma2000. It is aimed to provide 2.4Mbps peak data rates for packet-switched data applications while sharing the transmission time between the users. The specified bandwidth is 1.25MHz. A new RC and a new common traffic channel FPDCH (Forward Packet data channel) have been specified. The standard specifies the adaptive

modulation and coding scheme and hybrid ARQ scheme to support high-speed data rates and to improve the link utilization. The standard also specifies the scheduling rules to use the resources.

2.6.3.1 Forward Packet Data Channel

FPDCH is shared by packet data users and consists of time slots of length 2048 chips (1.6667ms). Each slot is divided into two half slots. The slot structures in active and idle states are shown in figure 2.8 and figure 2.9. Within each slot, the Pilot Channel, the Forward Medium Access Control (MAC) channel and the Forward Traffic Channel or the Control Channel are time-multiplexed. The Traffic Channel carries user physical layer packets. The Control Channel carries control messages, and it may also carry user traffic. Each channel is further decomposed into code-division-multiplexed quadrature Walsh channels.

The pilots bursts are transmitted in each half slot, and are used to provide the access terminal with a means of initial acquisition, phase recovery, timing recovery and prediction of the receive C/I for the purpose of access-terminal directed forward data rate control of the Data channel transmission.

Figure 2.8-Active slot structure [p.9-66, 1]

Figure 2.9-Idle slot structure [p.9-66, 1].

2.6.3.2 The Forward Traffic Channel

The forward Traffic Channel is a packet-based variable rate channel. The user physical layer packets for an access terminal are transmitted at a rate that varies from 38.4 kbps to 2.4 Mbps, depending upon the estimate of the received forward C/I, for the duration of the next possible forward link packet transmission, reported by the access terminal. In the following paragraphs, we provide a summery of the relevant features of the Forward Traffic Channel, described in [1].

Modulation and Coding Scheme (MCS)

Channel state information is fed back to the base station on the reverse link data rate request channel (DRC) and updated every 1.67msec. Depending upon the selected data rate, the appropriate modulation and coding scheme is selected to achieve the given link level performance. The modulation and coding parameters are shown in the table 2.1.

Sequence repetition

If the number of modulation symbols is less than the required number of modulation symbols to achieve a constant modulation rate of 1.2288MHz, the complete modulated symbol sequences (I and Q) are repeated as many full-sequence times as possible followed by a partial sequence, if necessary.

Table 2.1. Specified modulation and coding parameters

Data rate(kbps)	Code rate	Modulation type
38.4	1/5	QPSK
76.8	1/5	QPSK
153.6	1/5	QPSK
307.2	1/5	QPSK
614.4	1/3	QPSK
307.2	1/3	QPSK
614.4	1/3	QPSK
1228.8	1/3	QPSK
921.6	1/3	8-PSK
1843.2	1/3	8-PSK
1228.8	1/3	16-QAM
2457.6	1/3	16-QAM

If the required number of modulation symbols is less than the modulation symbol provided, then only the first portion of the input modulation symbol sequence is used.

The sequence repetition factor, as specified, is shown in the table 2.2.

Table 2.2: sequence repetition

Selected data rate (kbps)	Packet size (bits)	No. of modulation symbols provided	No. of modulation symbols needed	Repetitions factor
38.4	1024	2560	24576	9.6
76.8	1024	2560	12288	4.8
153.6	1024	2560	6144	2.4
307.2	1024	2560	3072	1.2
614.4	1024	1536	1536	1
307.2	2048	3072	6272	2.04
614.4	2048	3072	3136	1.02
1228.8	2048	3072	1536	1
921.6	3072	3072	3136	1.02
1843.2	3072	3072	1536	1
1228.8	4096	3072	3136	1.02
2457.6	4096	3072	1536	1

Walsh code covering

The resulting sequences are de-multiplexed into 16 parallel streams. Each is covered with a 16-ary orthogonal function. All the streams are added together to form a single in-phase and a single quadrature stream at chip rate of 1.2288 Mcps. Then the resulting chips are time-division multiplexed with the preamble, Pilot Channel and MAC channel chips.

Packet duration

The transmission of the physical layer packets is scheduled in 1 to 16 time slots, depending upon the selected data rate and encoder packet size, as shown in the table 2.3.

Table 2.3 Slot assignments for the supportable data rates

Data rate (kbps)	Encoder packet size (bits)	Number of slots
38.4	1024	16
76.8	1024	8
153.6	1024	4
307.2	1024	2
614.4	1024	1
307.2	2048	4
614.4	2048	2
1228.8	2048	1
921.6	3072	2
1843.2	3072	1
1228.8	4096	2
2457.6	4096	1

When more than one slot is scheduled, then the scheduled slots are 3-slot apart in time (4-slot interlacing). Therefore, the scheduled packet transmission duration is equal to the product of total number of slots required to interlace assigned slots and slot duration. The slot assignments are made as follows (assuming slot index =1):

- Scheduled time slots =16

$$\text{Scheduled packet duration} = (16 \times 4) \times \text{slot duration} = 64 \times 1.667 \times 10^{-3} = 106.667 \text{ msec}$$

Interlaced slot sequence: 1, 5, 9, 13, 17, 21, 25, 29, 33, 37, 41, 45, 49, 53, 57, and 61.

- Scheduled time slots = 8

Scheduled packet duration = $(8 \times 4 - 3) \times \text{slot duration} = 29 \times 1.667 \times 10^{-3} = 48.333 \text{ msec}$

Interlaced slot sequence: 1, 5, 9, 13, 17, 21, 25, and 29.

- **Scheduled time slots = 4**

Scheduled packet duration = $(4 \times 4 - 3) \times \text{slot duration} = 13 \times 1.667 \times 10^{-3} = 21.667 \text{ msec}$

Interlaced slot sequence: 1, 5, 9, and 13.

- **Scheduled time slots = 2**

Scheduled packet duration = $(2 \times 4 - 3) \times \text{slot duration} = 5 \times 1.667 \times 10^{-3} = 8.333 \text{ msec}$

Interlaced slot sequence: 1, 5.

- **Scheduled time slots = 1**

Packet duration = $(1) \times \text{slot duration} = 1.667 \text{ msec}$

Interlacing is not required.

Early termination of scheduled slots for a packet

As specified in the standard, with the help of an ARQ scheme, early termination of scheduled transmission for a packet is possible. For the case of the data rate of 153.6 kbps, transmission of a packet with an early termination of scheduled slots is shown in the figure 2.10

Figure 2.10: Multi-slot transmission of a packet with Early Termination. [9-68, 1]

In this example, the MT transmits an ACK response on the ACK channel after the third slot is received indicating that it has correctly received the physical layer packet. When the scheduler receives such an ACK response, it cancels the transmission of the remaining scheduled slots of the physical layer packet. These released slots can be used for further scheduling and transmission of subsequent physical layer packets. In this way, with the help of early termination, the transmission duration of a packet is reduced, which increases the effective transmission data rates.

CHAPTER 3

SYSTEM MODEL DESCRIPTION

3.1 Introduction

Our simulation study is based on the wireless scheduler model shown in figure 3.1. In the following sections we describe the modeling of the components of this scheduler model. Section 3.2 describes the modeling of voice sessions. Sections 3.3 to 3.6 describe the assumptions made about the scheduler and the high-speed data transmission techniques. Section 3.7 describes the assumptions made about user distances. Section 3.8 describes the causes of packet transfer delay and packet discard in the schedulers and section 3.9 describes the simulation program.

Figure 3.1: A typical wireless scheduler [13]

3.2 A simulation model for a session of voice source

The following sub sections describe the modeling of a voice session.

3.2.1 Single voice source

A voice source is characterized by two states [14] as shown in figure 3.2. These are called “speech burst” and “silence period”. In the speech burst state, the voice source generates packets at a fixed rate. The number of packets in a speech burst is characterized by a geometrically distributed random variable with mean ‘ n ’. The duration of a speech burst is equal to the product of the number of packets in a speech burst and the fixed inter-arrival time. In the silence state, no packets are generated. The length of a silence period is characterized by an exponential random variable with mean ‘ $1/\lambda$ ’, where λ is the arrival rate of the speech bursts.

- Geometrically distributed number of packets in a speech burst

Figure 3.2: A voice source model

The values of the parameters of random variables used to characterize a voice source are taken from [15], [16] and are listed below:

Mean number of packets in a speech burst = 22

Mean silence period = 0.65sec

Packet arrival time :In a typical CDMA system, voice packets are transmitted in a fixed 20msec frame. Therefore we assumed that the packets arrival every 20msec during a speech burst from a voice source.

A session for a voice user was modeled as described above. Sessions to generate packets for other users in the cell are modeled as separate independent and identical voice sources instead of one aggregate source. Simulating separate voice sessions for other users is beneficial because it takes advantage of the voice traffic pattern and provides statistical multiplexing. During the silence periods of inactive voice sessions, other active sessions of voice users can be served. In our simulation model, other user sessions were generated to simulate congestion.

To provide the desired quality of service (described in section 3.8) to the voice service users using packet-switched technology, it is necessary to set higher priority for the voice service users. Therefore, we considered only voice traffic. We assume that residual capacity, not used by the voice applications, could be given to other traffic such as data applications.

3.2.2 Packet size

The protocols, used in IP-based technologies, to transport information dictate the size and format of the packets that can be used. The following sections describe the required packet and frame format at various layers.

3.2.2.1 Packet format

Voice is sampled at 64kbps. In CDMA system, voice samples are collected during 20msec. Then speech coding is employed to reduce the number of bits required to transmit the voice information and therefore to save the bandwidth. Cdma2000 uses several types of vocoders, which operate at two different rates: 9.6kbps and 14.4kbps. At 9.6kbps, the data rates are 4.8, 2.4 or 1.2 kbps depending on the voice activities. These data rates are called Rate Set 1(RS1). It generates 160 bits in 20msec frames while operating at full rate (9.6kbps). The 14.4kbps vocoders use data rates of 14.4, 7.2, 3.6 or 1.8kbps. These data rates are called Rate Set 2(RS2). The 14.4kbps vocoder generates 267 bits in 20msec frames while operating at full rate (14.4kbps). We assumed the speech vocoder that works at 9.6kbps. Therefore, 20bytes of voice information is delivered to the lower layers.

Real time services are delay sensitive. These services, normally, do not tolerate the delay associated with retransmissions. Therefore the UDP protocol could be more suitable than TCP. In figure 2.7, Real Time Protocol (RTP), Real Time Control Protocol (RTCP) and Ipv6 are not shown, however such protocols are required to support IP-based voice service. Therefore, it is assumed that these protocols could be used. The packet formats of the assumed protocols are shown in figure 3.3.

Figure 3.3. Packet encapsulation

RLP frame: Non transparent mode

RLP frame: Transparent mode

Figure 3.4: RLP frame structures

Header compression

Figure 3.3 shows the RTP/UDP/Ipv6/PPP protocols headers, which constitute 70 bytes.

Figure 3.4 shows that RLP payload can take only 160 bits when it works in transparent mode. So, deep header compression is required as mentioned in the 3G Network Design Guideline 3.1 [p.88, 17].

According to the header compression technique mentioned in [18], RTP/UDP/IP headers can be reduced to 5 bytes. PPP header cannot be further reduced below 5 bytes: framing byte (2), protocol byte (1) and checksum (1) should be present. Therefore a voice packet delivered to the RLP layer will consist of 10 bytes of headers and 20 bytes of voice information. Figure 3.5 shows this header compression process.

Figure 3.5: Header Compression

3.2.2.2 RLP frame format

After header compression, the size of the packet delivered to the RLP layer is 240 bits.

From figure 3.4, it is clear that the RLP frame header size is 11 bits. It means that the required RLP frame size should be 251bits. Therefore, we assume that the RLP frame is 251 bits. This hypothetical frame format is shown in figure 3.6.

Figure 3.6: Hypothetical frame format of RLP protocol in transparent mode

3.2.2.3 Encoder packet size

As specified in [1], depending upon the supportable data rates, the encoder packet can carry 1024, 2048, 3072 or 4096 bits. As described in the previous section, size of the RLP frame is assumed to be 251 bits. Therefore, we assume that, depending upon the selected size of the encoder packet and the number of the voice frames waiting in the queue for the selected user, one to sixteen voice frames could be transported in one encoder packet. For example, for the data rate of 38.4 kbps, the encoder packet size is 1024 bits. The last 6 bits in the TAIL field of the encoder packet are discarded. It means that, up to four voices frames (251x4 bits) can be filled in the encoder packet. We assume that the remaining space in the encoder packet will be padded with zeros. The table 3.1 shows this arrangement.

3.3 Estimation of channel conditions

As described earlier, to achieve the efficient utilization of the specified 1.25MHz channel, on the basis of the received transmission from an access point (e.g. BS), each access terminal (e.g. MT) can measure carrier to interference ratio (C/I). The value of the C/I is then mapped to a value representing the maximum data rate that each user can support for a given level of error performance.

Table 3.1. Voice packets in the specified encoder packets

Data rate (kbps)	Encoder packet size (bits)	Number of voice packets
38.4	1024	1 to 4
76.8	1024	1 to 4
153.6	1024	1 to 4
307.2	1024	1 to 4
614.4	1024	1 to 4
307.2	2048	1 to 8
614.4	2048	1 to 8
1228.8	2048	1 to 8
921.6	3072	1 to 12
1843.2	3072	1 to 12
1228.8	4096	1 to 16
2457.6	4096	1 to 16

To simulate this behavior, we assume C/I distributions [19] as models for channel conditions for users located at various distances (described in section 3.7) in the cell. These C/I distributions are the results of a simulation of 19 hexagonal 3-sector cells with path loss, shadowing and interference from other transmitters. The effects of Rayleigh and Doppler fading were not considered.

We further assume table 3.2 for the mapping of the C/I values to the corresponding supportable data rates, for 1% Packet Error Rate (PER), when the MCS (table 3.3) are employed.

Table3.2: Mapping between data rate and C/I values for 1% PER [20]

Data rate (kbps)	C/I(dB)
38.4	-12.5
76.8	-9.5
153.6	-6.5
307.2	-4.0
614.4	-1.0
921.6	1.3
1228.8	3.0
1843.2	7.2
2457.6	9.5

3.4 Modulation and Coding Scheme

As described in section 3.3, each user in the cell sends a request of appropriate data rate corresponding to its channel conditions. Based on the requested supportable data rate, the modulation and coding parameters are selected, which achieve the supportable data rate for a target PER. The modulation and coding parameter values corresponding to the specified MCS could not be available. Therefore we assume the MCS (which achieves the specified data rates for 1%PER) available in the related literature. It is shown in table 3.3.

Table3.3: Modulation and coding scheme [20]

Data rate(kbps)	Code rate	Modulation type
38.4	1/4	QPSK
76.8	1/4	QPSK
153.6	1/4	QPSK
307.2	1/4	QPSK
614.4	1/4	QPSK
307.2	1/4	QPSK
614.4	1/4	QPSK
1228.8	1/3	QPSK
921.6	3/8	QPSK
1843.2	1/2	8-PSK
1228.8	1/2	QPSK
2457.6	1/2	16-QAM

As described in chapter 2(section 2.6.3.1), an encoder packet is coded, modulated and repeated/punctured as needed, and de-multiplexed in 16 streams. Then these streams are covered with 16-ary Walsh function, scaled and added together. Then the packet (we call it physical layer packet) is time multiplexed with the other specified channels and transmitted in one or more scheduled time slots.

3.5 The scheduler

As described in chapter 2 (section 2.6.3), the HDR forward link is defined in terms of frames of length 26.67msec. Each frame consists of 16 time slots of 1.67msec duration. Frames and time slots are used as units of time. The users in the cell share the link and the transmissions of users' packets are scheduled in time slots. Depending upon the data rate, 1, 2, 4, 8 or 16 time slots could be scheduled for the transmission of one encoder packet. These scheduled slots are 3 slots apart in time (refer to section 2.6.3.5). The following sub-sections describe the rules to schedule the time slots and the scheduling algorithms assumed:

3.5.1 The Specified Scheduling Rules (SSR)

- The scheduler will work on a per slot basis.
- If the current time slot has already been scheduled in the previous scheduling event(s), no scheduling decision will be made in the current time slot.
- More than one time slot could be scheduled for an encoder packet (refer to section 2.6.3.5).
- The scheduled time slots will be 3 slots apart in time, if multi-slot scheduling is made.
- If the transmission of an encoder packet, in the scheduled time slots, is not successful, the packet will enter in the competition again.

3.5.2 The Proposed Changes in the SSR (PCSSR)

With SSR, a scheduler commits to provide service in the scheduled time slots. During the transmission of the scheduled slots, the channel conditions may change. This scheme causes two problems: packets suffer higher delay in the scheduler, and for the case of the Proportional Fair Queue (PFS) scheduling algorithm, the delay is reduced as the number of users increases in the cell. To solve these problems, we propose the following changes, called “**Proposed Changes in Specified Scheduling Rules (PCSSR)**”:

- Scheduling will be done on a per slot basis.
- No multi-slot scheduling will be made.
- If the transmission is not successful in the current slot, the packet will enter in the competition again.

3.5.3 The scheduling algorithms

Scheduling algorithms are important components in the provision of efficient channel utilization. The most widely used proposed algorithms are:

- Proportional fair scheduler
- Maximum C/I scheduler
- The First-in-first-out scheduler (FIFO)

In this thesis, we assumed these algorithms to evaluate the performance of voice service that could be offered using the proposed high-speed data transmission techniques. We describe these algorithms as follows:

3.5.3.1 Proportional Fair Scheduler (PFS)

In the following paragraph, we provide a summary of the information presented in [21]. This scheduling algorithm takes advantage of variations of the channel characteristics and schedules the transmission when the access terminals experience better channel conditions. The selection criteria is based on a parameter defined as:

Selection parameter = DRC/R

Where

DRC = Requested data rate in a given time slot from an access terminal

R = Moving average of data rate received by an access terminal over a window of appropriate size.

A user with highest DRC/R is selected at each decision time. If two terminals have the same DRC/R value, then a user is picked randomly. Users with no data to send are not considered in the selection phase.

Each user's moving average is updated at each decision time. Moving averages of those users who have no data to send or those who are not selected at the current decision time are also updated. Moving average is updated as follows:

$$R_i(t+1) = (1 - 1/tc) \times R_i(t) + 1/tc \times \text{Current_transmission_rate_of_user}_i$$

Value of *Current_transmission_rate_of_user_i* is '0' for those users who are not receiving service in the current transmission time and those who have no data to send. In this way those users who are not receiving service accumulate credits with the scheduler and their priority to be selected at the next decision instant is increased.

' tc ' is a system parameter used to specify the size of the moving average window. It is related to the maximum amount of time a user can wait. Higher value of ' tc ' allows the scheduler to wait longer for a user's channel condition to improve. This improves the overall system throughput. On the other hand, a large value of ' tc ' means a specific user's packets may be delayed. Figure 3.7 shows the effect of moving average window size on the scheduling delay for a user at a distance of 1110m (section 3.7) away from the base station, when the cell loading is 0.92.

Figure 3.7: Effect of moving average window size on the scheduling delay for a user at a distance of 1110m.

We observed that with the window size of 24, the mean scheduling delay is at its lowest with the value of 17.09msec. With the window size of 36, the scheduling delay is

17.25msec. From table 3.4, we observe that the average transmitted data rate, for the case of window size of 24, is 1.64Mbps. For the case of window size of 36, the average transmitted data rate is 1.80Mbps. Therefore, as a compromise between mean transmitted data rate and the mean scheduling delay, we chose the moving window size of 36 for our simulation model.

Table 3.4: Effect of moving window size on maximum delay and data rate

Window size (slots)	Mean delay (msec)	Maximum delay (msec)	Data rate (Mbps)
12	25.38	220.17	1.57
24	17.09	77.06	1.64
36	17.25	73.5	1.8
48	17.30	85.9	1.8
60	17.28	95.44	1.8
72	17.54	114.17	1.79
75	17.49	115.17	1.79
84	17.54	117.66	1.79
100	17.79	123.89	1.78
150	17.85	135.24	1.75
200	18.50	157.49	1.73

3.5.3.2 Maximum C/I Scheduler

This scheduler has been described in [22]. It takes advantage of temporal variations of the channel. The selection criterion is based on C/I. A user with the highest C/I value at the

decision instant is selected. Users who are experiencing bad channel condition or who have no data to transmit do not accumulate credits with the scheduler and their priority does not increase. This scheduler uses the advantage of the channel condition over the short term and increases link utilization and throughput.

3.5.3.3 First-in-first-out scheduler [23]

The FIFO base scheduler has been used in the literature, e.g. in [23], for the performance comparison of the proposed algorithms described above. This scheduler tries to serve the users on a ‘first in, first out’ basis. It selects a user whose head-of-the-line packet carries the time stamp (arrival time) of minimum value. It chooses the transmission data rate based on the channel condition (table 3.2). But, it does not take advantage of the short-term variation of the channel.

3.6 Early termination of scheduled transmission

As specified in 2.6.3, an early termination of the scheduled transmission of an encoder packet is possible with the help of a hybrid ARQ (2.5.3). For an encoder packet that requires multi-slot transmission, retransmission of the copies of the encoder packet continues in the scheduled slots until the scheduler receives an ACK from the user. The user’s terminal combines the coding information present in the retransmissions and tries to decode the received packet. The user sends a NACK, if the user is unsuccessful to decode after receiving a (re) transmission. If the user succeeds in decoding the packet, it sends an ACK. When the scheduler receives an ACK from the user, the transmission of the encoder packet in the remaining scheduled slots, if any, is cancelled. This process is shown in the table 3.5 for the data rate of 153.6 kbps.

The table 3.5 shows the actual number of scheduled slots used to transmit a packet successfully. It also shows the actual transmission time (packet duration) and the saving in transmission time that could be achieved using the early termination of transmission of the scheduled slots for a packet. The following relations are used for the table3.4:

- Initial scheduled transmission time: number of scheduled slots * slot duration
- Actual transmission time= number of slots transmitted * slot duration.
- Reduction in transmission time = Initial scheduled transmission time - actual transmission time.
- n = current slot

Table 3.5: Early termination of the scheduled transmission for 153.6 kbps

<ul style="list-style-type: none"> - Initial scheduled transmission time = 21.71msec - Number of scheduled time slots = 4 					
Slot index	Attempt #	ACK/NACK	Actual # of slots used	Actual transmission time (msec)	Reduction in transmission time
n ,	1	If ACK	1	1.67	20.04
$n+4$	2	If ACK	2	8.35	13.36
$n+8$	3	If ACK	3	15.03	6.68
$n+12$	4	If ACK	4	21.71	0

To simulate the behavior of the early termination of transmissions in the scheduled time slots for an encoder packet, we need to determine the probability of failure of transmission in the scheduled time slots for the encoder packet. We assume that the

decoding of an encoder packet, received in successive transmissions in the scheduled time slots for the encoder packet, is independent of each other. And, we further assume that the probability of decoding failure in a scheduled time slot is the same. We define the probability of decoding failure in each scheduled transmission for a specified maximum number of scheduled transmissions as follows:

$$p = (PER)^{1/N}$$

Where

p = probability of decoding failure in a transmission (in a scheduled slot)

N = maximum number of allowed transmissions for a specified data rate (number of scheduled slots) to achieve a target PER.

PER= target packet error rate (1% in this thesis)

With each failure, the probability of failure in the next retransmission decreases. If the transmission in the last scheduled slot fails, then the packet re-enters in the queue for the rescheduling. This gives us a simple ARQ scheme and provides an upper bond on the performance of the proposed hybrid ARQ schemes in the literature.

3.7 Users distances in the cell and coverage area

Channel condition and hence C/I values are related to distance. Therefore user distances from the BS are an important factor in any realistic simulation. The values of the C/I distributions [19] were for the distances of 10, 110, 210, 310, 410, 510, 610, 710, 810, 910, 1010, 1110, 1210, 1310, 1410, 1510 and 1610m. As mentioned in [24], the specified site-to-site distance for HDR is 2000m. Therefore, we assumed the user distances of 10, 110, 210, 310, 410, 510, 610, 710, 810, 910, 1010, and 1110m. The

distance of the user, for which we measure the performance of the scheduling algorithms and scheduling techniques, is assumed to be 1110m as the worst case. We expect that the users at distances lower than 1110m could be offered similar or better performance than the performance that could be offered to the user at the distance of 1110m away from the base station.

3.8 Measures of Performance

For a good quality voice service, packet transfer delay and packet loss rate are considered as measures of performance. From the system point of view resource utilization is important. In the following sub-sections we describe the causes of delay and packet loss in a scheduler.

3.8.1 Causes of delay in a packet scheduler

Arriving packets exhibit delay in the packet scheduler due to the following:

- As mentioned in section 2.6.3, a packet can be transmitted in one or more slots. The minimum service time a packet can take is 1.6667msec, and the maximum service time that can be given is 101.667msec. An arriving packet waits for the completion of service being given to a packet.
- If a packet is being served (scheduled in more than one inter-laced slots), following packets for the same destination will not be scheduled until the transmission of the already scheduled packet is done.
- If the channel condition between the base station and the user for which the arriving packets are destined, is such that it can not support the specified data rates (table 3.2), the scheduler will not transmit the packets and will wait for the

channel condition to be supportive of specified data rates, even when no one else is waiting.

- When more than one user are active, a packet scheduler selects packet(s) from users' queues based on one of the selection criteria described in section 3.5. When a user loses the competition in the current slot, packets belonging to that user will wait for the next slot and contend again. This contention will continue until the user is selected.

All the delays described above, including service time will be called '**scheduling delay**'.

3.8.2 Causes of packet loss

- Transmission errors occur due to channel impairment. To achieve a target Packet Error Rate (PER) of 1 %, one of the modulation and coding schemes (shown in table 3.1) is selected based on the channel condition reported by a user (in terms of received C/I). ARQ scheme is also employed to help MCS. After the transmission of a packet for the specified number of times, if the receiver is still unable to decode the packet successfully, the packet is discarded. Therefore, we assume that packet error rate due to the transmission error is 1%.
- Because of the scheduling delay described in section 3.8.1, packets could still be discarded if they exceed a specified delay constraint.

3.8.3 Performance constraints

As mentioned earlier, voice quality is characterized by low and constant packet transfer delay and frame error rate (FER). In the following section, we establish the values of these constraints.

3.8.3.1 Delay constraints

To evaluate the performance of voice service that could be offered by a downlink wireless access network using scheduling techniques for high-speed data applications, when voice application could be treated as data application, we describe the following possibilities:

Delay constraint#1

In a circuit-switched wireless cellular network, there is a fixed service time (transmission time) of 20msec and no queuing delay. Therefore, to match the delay performance of a cellular network, a high-speed packet-switched wireless cellular network should meet the delay constraint of 20msec.

Delay constraint #2

ITU-T G114 [3] recommends that, for real-time voice service, end-to-end packet transfer delay should not be greater than 150msec. For a fully IP-based packet switched voice call, between two 3G packet-switched CDMA cellular networks users via the Internet, delay constraint#2 could be considered as a measure of delay performance. In this thesis, we do not explore the reverse wireless link, and packet transfer delay in the Internet. However, this simulation study estimates the available delay budget for the Internet and the 3G packet-switched networks at the other side of the Internet.

3.8.3.2 Packet loss constraint

As described in section 3.4, the MCS are specified to achieve a target PER of 1%. Because of the delay constraints we described in section 3.8.3, packets could be discarded

if a delay constraint is not met. Combining these two causes together, we assumed 1% packet discard rate (PDR), as the packet loss constraint.

3.8.4 User Throughput and average attempted data rate

Referring to [24] the user throughput for a given user ‘ j ’ is defined as follows:

$$S_{user} = \frac{\text{good bits in } [0, T] \text{ for user } j}{T}$$

Therefore, we define the following:

Real-time voice applications are considered less tolerant to delay. Therefore, for voice applications, good bits mean the successfully received information bits after packet discard due to excessive delay. For that case, to evaluate the performance that could be offered to a user, we will be using the term ‘**user throughput**’ for a desired user.

As data applications are more tolerant to delay than the real-time voice applications, usually received packets are not discarded due to delay. Therefore, for data applications, good bits mean the successfully received information bits. In that case, we will be using the term ‘**Effective data rate**’. Effective data rate for all the users in a given sector could be defined as follows [24]:

$$S_{service} = \frac{\text{good bits in } [0, T] \text{ for sector } j}{T}$$

Similarly in [24], the average data rate for all the users in a given sector, termed as OTA throughput, is defined as follows:

$$S_{ota} = \frac{\sum \text{data rate attempted}}{\text{total transmission attempts}}$$

Average data rate includes all the transmissions (successful or not). It reflects the ability of the system to adapt the channel conditions and to vary the transmission data rate accordingly.

3.9 Simulation program

We describe our simulation program as follows:

- **Inputs to the simulation program:**
 - Number of users: {1,5,10,15,20,25, 30},
 - The distance of a specified user for which performance is being measured. We call the user as the **intended user**: {1110m}
 - Slot duration: 1.67msec
 - Delay constraint: {20msec, 150 msec}
 - Packet loss constraint: 1% PER
 - Slot scheduling rules {SSR, PCSSR}
 - The Scheduling algorithm {PFS, Maximum C/I, FIFO}.
 - Moving window size, when the PFS scheduler is used: 36 slots.
 - Length of a simulation run: 50,000 slots (83sec).
 - Number of independent simulation runs: 5
 - Simulation seeds for each simulation run.

In each simulation run, the following procedures are executed:

- **Initialization:**
 - The users, other than the intended user, are placed randomly in the cell. The intended user is placed at the given distance.

- All the random variables used are initialized, using the simulation seed for the simulation run.
- Voice sessions are generated for the duration of a simulation run.
- The states of the voice sessions are determined. It is determined that which of the voice sessions are in the talk-spurt state and which of the voice sessions are in the silence state at the start of the simulation.
- C/I ratio distributions are loaded from a file.
- Event list is populated i.e. next packet generation time for the voice sessions in talk-spurt state, arrival times of the talk-spurts for the voice sessions in silence states and the scheduling instant are entered in the event list.
- **Execution of events:**

Two types of events are executed: voice packet generation and scheduling:

- Voice packet generation:
 - A packet is generated and inserted in the queue of the user, to which the packet will be transmitted.
 - If this is the last packet of the current talk-spurt, arrival time for the next talk-spurt is entered in the event list.
 - If this is not the last packet of the current talk-spurt, arrival time of the next packet is entered in the event list.
 - The packet generation time is recorded and the state of the corresponding queue is updated.
- Scheduling event
 - Packet discard procedure:

- a) The scheduler looks for voice packets which have exceeded the given delay constraint.
 - b) The scheduler uses the relation to discard a packet: for a voice packet, if (the queuing delay for the packet + expected service time (one time slot) > the given delay constraint), discard the packet.
 - c) The scheduler discards those packets and updates the users' queues
 - d) The scheduler records the number of packets discarded.
- Estimation of channel conditions and supportable data rates
 - a) In each slot interval, C/I values are picked from a lookup table of the C/I distributions. Based on the C/I values, supportable data rates are selected (refer to section 3.3).
 - b) Supportable data rates for the users, whose C/I values are below the minimum value (table 3.2), are set to zero.
 - c) Supportable data rates for the users, whose queues are empty, are set to zero.
- Selection of a user
 - a) If the current time slot has already been scheduled in the previous decision instant(s), the selection process terminates.
 - b) A user whose queue is empty is not considered in the selection.
 - c) A user whose supportable data rate is set to zero is not considered in the selection.

- d) A user who is still being served in more than one scheduled slot, assigned in previous decision times, is not taken in the competition.
- e) Among the users, who qualify for the selection, a user is selected based on the selection criterion (PFS, Maximum C/I or FIFO)(refer to section 3.5.3).
- f) If more than one user is selected, a user (among the selected users in 'e') is selected randomly.

Selection of an encoder packet size

- g) An encoder packet size is selected based upon the supportable data rate for the selected user (table 3.1).
- h) In the encoder packet, depending upon the queue status of the selected user, one or more voice packets are combined (table 3.1).

Scheduling of transmission times

- a) For the case of the SSR, based on the supportable data rate for the selected user, one or more time slots are scheduled to the user (refer to section 3.5.1). In the case of multi-slot scheduling, the scheduled slots are 3-slots apart in time.
- b) For the case of the PCSSR, transmission in the current time slot is scheduled to the user (refer to section 3.5.2).

ARQ and early termination

For the case of the SSR

- a) It is determined how many scheduled slots are required to be transmitted in order for the successful reception of the encoder packet for the selected user (refer to section 3.6).
- b) If the actual required slots are less than the scheduled slots, the remaining scheduled slots are cancelled.
- c) If the actual required slots are more than the scheduled slots, the voice packets in the encoder packet are re-entered in the competition.

For the case of the PCSSR

- a) It is determined whether the transmission of the encoder packet, for the selected user, in the current time slot is successful or not.
- b) If not, then the voice packets in the encoder packet are re-entered in the competition.
- o Next decision time is entered in the event list (current time +slot duration).

Execution of the events continues till the end of a simulation run.

In each simulation run, we measure queue delay, service time and total delay in the scheduler for the intended user. The measured values of the total delay in the scheduler are used to examine the quality of service, in terms of packet delay and packet discard rate, which could be provided. In each slot interval, the transmitted data rate and the selected user are recorded. The state of the current slot (whether the slot is active or idle) is also recorded. (These statistics are used to estimate the average transmitted data rate,

fairness and scheduler utilization). In each simulation run, we measure the effective data rate as well.

For a better estimation of measures of performance, multiple independent simulations run were conducted for a given number of randomly placed users in the cell and a given distance of the intended user from a base station. In each simulation run, more than 1000 packets (1000 trials) are transmitted to the intended user.

To compare the performance that could be offered using SSR and the performance that could be offered using PCSSR, simulations were completed for the PFS only. Three sets of simulations were conducted, one for each scheduling algorithm, to compare the performances offered by the scheduling algorithms with the performance offered by the voice service using circuit-switched technology.

CHAPTER 4

RESULTS AND OBSERVATIONS

4.1 Introduction

This chapter describes the simulation results about the performances of voice services that could be offered by the scheduling techniques described in the previous chapter. In section 4.2 we present a comparison between performances that could be offered using the SSR and the PCSSR. In section 4.3 to 4.5 we present the performances, in terms of scheduling delay and packet discard rate, of the scheduling algorithms, when the PCSSR is used. In section 4.6 we present a comparison among the performances of the schedulers.

4.2 Comparison between PCSSR and SSR

In this section, we present a comparison between the performances that could be offered with the SSR (section 3.5.1) and with the PCSSR (section 3.5.2) for the PFS, in terms of the packet discard rate, the maximum scheduling delay and the maximum number of transmissions for an encoder packet.

Figure 4.1 shows the comparison, in terms of PDR, between SSR and PCSSR for a user at a distance of 1110m away from the base station. As shown in the figure, PCSSR performs much better than SSR. The lowest value of PDR for the case of SSR is 5.46%, whereas the lowest value for the case of PCSSR is 0.23%.

Figure 4.1: Comparison of packet discard rate for the cases of SSR and PCSSR

Figure 4.2: Comparison of maximum number of transmission attempts for the cases of SSR and PCSSR

With SSR, the scheduler commits multiple transmission attempts for an encoder packet for the selected user in the scheduled time slots, not realizing the potential gains of the a-posteriori link adaptation. Figure 4.2 shows the maximum number of transmission attempts made to transmit an encoder packet successfully. It shows that, for the case of SSR, at least 11 attempts were made. As described earlier, each attempt is 3-slots apart. Therefore each retransmission consumes 6.68msec(four time slots). The higher number of retransmissions causes excessive delay in the scheduler. Figure 4.3 shows the comparison of SSR and PCSSR, in terms of maximum delay exhibited by encoder packets in the scheduler.

Figure 4.3: Comparison of maximum delays for the cases of SSR and PCSSR

With PCSSR, if the current transmission fails, the scheduler assesses the channel conditions again prior to the next retransmission. It means that, for each retransmission of the same encoder packet, the modulation and coding parameters, appropriate to the latest

estimation of the channel conditions, are chosen. Figure 4.2 shows that the number of retransmissions, for the case of PCSSR, is 4 or lower. These lower number of retransmissions results in lower values of maximum delays, as shown in the figure 4.3. The reduced delays, in turn, result in reduced PDR, as shown in the figure 4.1.

Figure 4.4 shows the comparison between SSR and PCSSR in terms of user throughput.

Figure 4.4: Comparison of user throughput for the cases of SSR and PCSSR

We observed the similar results (not shown) for the cases of the Maximum C/I and the FIFO.

Therefore, we conclude that with PCSSR, the delay and user throughput performances of the high-speed packet data transmission techniques are expected to improve. In the following sections we will show the performances using PCSSR.

4.3 Performance of voice service offered with PFS

The following sub-sections present the performance of voice service that could be offered with PFS, when PCSSR is used.

Figure 4.5 shows the delay performance of the PFS, in terms of cumulative distribution function (CDF) for an intended user's distance of 1110m. We observe that for the cell loading of 0.03, 0.18 and 0.34, the probabilities of scheduling delays remain similar. As the cell loading further increases, the intended user, at the edge of the cell, experiences more delays and the probabilities of the same delays, that could be experienced by the intended user when the cell loading is below 0.34, are lower. We will be using this graph to evaluate the performance of the scheduler for the described delay constraints.

Figure 4.6 shows the Packet Discard Rate (PDR) performance of the PFS, for a user at a distance of 1110m, as a function of the scheduling delay in the scheduler. We observe that the PDR increases with the increase in cell loading. We will be using this graph to evaluate the performance of the scheduler for the described PDR constraint.

Figure 4.5: Comparison of CDFs of delay in the PFS for a user at 1110m

Figure 4.6: Distribution of PDR in the PFS for a user at 1110m

Figure 4.7 shows the performance of the voice service offered by the PFS as compared with the performance of the voice service offered by the circuit-switched wireless access network (delay constraint#1: 20msec).

Figure 4.7: Comparison of PDR in the PFS for a user at 1110m

We observe that up to the cell loading of 0.5 (15 users), the PDR remain below or equal to 0.34%. Then it starts to rise up. For the cell loading of 0.66, the observed PDR is 1.45%. Therefore, we conclude that the packet-switched voice service, comparable to the circuit-switched voice service, could be offered up to cell loading of 0.5 (15 users), when the PFS is used.

We observe that the maximum number of transmissions required for an encoder packet is 4, up to the cell loading of 0.66. Above the c.l. of 0.66, the maximum number of transmission attempts tends to decrease. We observe that, for the cell loadings of 0.8 and 0.92, the maximum number of attempts is 3. This is due to the fact that the PFS tends to

serve the users at peak data rates. As number of users in the cell increases, the probability of selecting the intended user with the peak data rates also increases. At peak data rates, the probability of decoding failure is less than that at lower data rates. Therefore, the maximum number of transmission attempts tends to decrease. We observe, from figure 4.5, that the maximum scheduling delay is $\leq 28\text{msec}$ for the cell loading ≤ 0.5 whereas, the observed maximum scheduling delay is $\geq 48\text{msec}$ for the cell loading >0.5 .

Therefore for the cell loading ≤ 0.5 , the major cause of the scheduling delay seems to be the transmission attempts. And for the cell loading >0.5 , the major cause of the scheduling delay seems to be the congestion caused by the users.

To evaluate the performance of the voice service offered by the PFS, as a part of an end-to-end fully IP based wireless voice service via the Internet (delay constraint of 150msec), we observe the scheduling delays, from figure 4.5, that are experienced by the voice packets for the target PDR of 1%. Table 4.1 shows the values of scheduling delays for the target PDR of 1%. It also shows the delay margins that are available for the up-link wireless access network and the Internet, involved in the fully IP based voice conversation.

Table 4.1: Delay budget for the delay constraint#2 with PFS

PDR(%)	Scheduling delay for 1%PDR	Cell loading	Available delay budget (msec)
1.0	16.7	0.03-0.52	133.3
1.0	22.8	0.67	127.2
1.0	36.4	0.82	113.6
1.0	45.8	0.92	104.2

4.4 Performance of voice service offered with Max. C/I

The following sub-sections present the performance of voice service that could be offered with the Maximum C/I scheduler, when PCSSR is used.

Figure 4.8 shows the delay performance of the Maximum C/I scheduler, in terms of cumulative distribution function (CDF) for an intended user's distance of 1110m. We observe that for the cell loading of 0.03 and 0.18, the probabilities of scheduling delays are close to each other. As the cell loading further increases, the intended user, at the edge of the cell, experiences more delays and the probabilities of the same delays, that could be experienced by the intended user when the cell loading is below 0.34, are lower. We will be using this graph to evaluate the performance of the scheduler for the described delay constraints.

Figure 4.9 shows the Packet Discard Rate (PDR) performance of the PFS, for a user at a distance of 1110m, as a function of the scheduling delay in the scheduler. We observe, as expected, that the PDR increases with the increase in cell loading. We will be using this graph to evaluate the performance of the scheduler for the described PDR constraint.

Figure 4.10 shows the performance of the voice service offered by the PFS as compared with the performance of the voice service offered by the circuit-switched wireless access network (delay constraint#1: 20msec).

Figure 4.8: Comparison of CDFs of delay in the Max. C/I for a user at 1110m

Figure 4.9: Distribution of PDR in the Max. C/I for a user at 1110m

Figure 4.10: Comparison of PDR in the Max. C/I for a user at 1110m

We observe that, from figure 4.10, up to the cell loading of 0.5 (15 users), the PDR remain below or equal to 0.83%. Then it starts to rise up. For the cell loading of 0.66, the observed PDR is 3.10%. Therefore, we conclude that the packet-switched voice service, comparable to the circuit-switched voice service, could be offered up to cell loading of 0.5 (15 users), when the Maximum C/I scheduler is used.

We observe that the maximum number of transmissions required for an encoder packet is 4, up to the cell loading of 0.66. Above the c.l. of 0.66, the maximum number of transmission attempts tends to decrease. We observe that, for the cell loadings of 0.82 and 0.92, the maximum number of attempts is 3. This is due to the fact that the Maximum C/I scheduler tends to serve the users at peak data rates. As the number of users in the cell increases, the probability of selecting the intended user with the peak data rates also increases. At peak data rates, the probability of decoding failure is less than that at lower

data rates. Therefore the maximum number of transmission attempts tends to decrease. On the average, the number of transmission attempts remains at 1 for all the values of the cell loading.

We observe that the maximum scheduling delay is $\leq 40.8\text{msec}$ for the cell loading ≤ 0.5 (15 users) whereas, the observed maximum scheduling delay is $\geq 98.5\text{msec}$ for the cell loading >0.67 (20 users).

Therefore, for the cell loading ≤ 0.5 , the major cause of the scheduling delay seems to be the transmission attempts (and hence channel conditions). And for the cell loading >0.5 , the major cause of the scheduling delay seems to be the congestion caused by the users.

To evaluate the performance of the voice service offered by the Maximum C/I, as a part of an end-to-end fully IP based wireless voice service via the Internet (delay constraint of 150msec), we observe, from the figure 4.8, the scheduling delays that are experienced by the voice packets for the target PDR of 1%. Table 4.2 shows the values of scheduling delays for the target PDR of 1%. It also shows the delay margins that are available for the up link wireless access network and the Internet, involved in the fully IP based voice conversation.

Table 4.2: Delay budget for the delay constraint#2 with Max C/I

PDR(%)	Scheduling delay for 1%PDR (msec)	Cell loading	Available delay budget (msec)
1.0	19.0	0.03-0.52	131.0
1.0	33.0	0.67	117.0
1.0	87.5	0.82	62.5
1.0	>150	0.92	0

4.5 Performance of voice service offered with FIFO

The following sub-sections present the performance of voice service that could be offered with the FIFO, when PCSSR is used.

Figure 4.11 shows the delay performance of the FIFO, in terms of cumulative distribution function (CDF) for an intended user's distance of 1110m. We observe that for the cell loading of 0.03, 0.18, 0.34 and 0.52, the probabilities of scheduling delays are closer to each other. As the cell loading further increases, the intended user, at the edge of the cell, experiences more delays and the probabilities of the same delays, that could be experienced by the intended user when the cell loading is below 0.52, are lower. We will be using this graph to evaluate the performance of the scheduler for the described delay constraints.

Figure 4.12 shows the Packet Discard Rate (PDR) performance of the FIFO, for a user at a distance of 1110m, as a function of the scheduling delay in the scheduler. We observed that the PDR increases with the increase in cell loading. We will be using this graph to evaluate the performance of the scheduler for the described PDR constraint.

Figure 4.11: Comparison of CDFs of delay in the FIFO for a user at 1110m

Figure 4.12: Distribution of PDR in the FIFO for a user at 1110m

Figure 4.13 shows the performance of the voice service offered by the FIFO as compared with the performance of the voice service offered by the circuit-switched wireless access network (delay constraint#1: 20msec).

Figure 4.13: Comparison of PDR in the FIFO for a user at 1110m

We observe that up to the cell loading of 0.67 (20 users), the PDR remain below or equal to 0.95%. Then it starts to rise up. For the cell loading of 0.82, the observed PDR is 6.68%. Therefore, we conclude that the packet-switched voice service, comparable to the circuit-switched voice service, could be offered up to cell loading of 0.67 (20 users), when the FIFO is used.

We observe that the maximum number of transmissions required for an encoder packet remains the same with the value of 4, for all the values of the cell loading, as opposed to the other schedulers. This is due to the fact that the FIFO does not tend to serve the users at peak data rates. On the average, the number of transmission attempts remains at 1 for

all the values of the cell loading. We observe that the maximum scheduling delay is $\leq 31.99\text{msec}$ for the cell loading ≤ 0.67 whereas, the observed maximum scheduling delay is $\geq 36.76\text{msec}$ for the cell loading >0.67 .

Therefore, for all the values of cell loading, delay contribution from the retransmission remain the same. And as cell loading is increased, additional delay contribution comes from the congestion caused by the users.

To evaluate the performance of the voice service offered by the FIFO, as a part of an end-to-end fully IP based wireless voice service via the Internet (delay constraint of 150msec), we observe the scheduling delays, from the figure 4.11, that are experienced by the voice packets for the target PDR of 1%. Table 4.3 shows the values of scheduling delays for the target PDR of 1%. It also shows the delay margins that are available for the up link wireless access network and the Internet, involved in the fully IP based voice conversation.

Table 4.3: Delay budget for the delay constraint#2 with FIFO

PDR(%)	Scheduling delay for 1%PDR (msec)	Cell loading	Available delay budget (msec)
1.0	≤ 20	0.03-0.67	130.0
1.0	27.5	0.82	122.5
1.0	31.0	0.92	119.0

Tables 4.1-4.3 also show the residual capacities that could be used to explore to support data applications.

4.6 Performance comparison among the schedulers

In the following paragraphs, we present a comparison between the performances offered by the PFS, Maximum C/I and FIFO, in terms of PDR, average transmitted data rates and the effective data rates.

4.6.1 Comparison of PDR performance

Figure 4.14 shows the comparison among the PDR performances of the schedulers. We observe that up to the cell loading of 0.52 (15 users), all of the three schedulers could achieve the target PDR of 1%. The FIFO could achieve the target PDR for the cell loading of 0.67 (20 users) as well.

As described earlier, the FIFO selects a user based on the first in, first out basis. If the current transmission for a voice packet fails, it enters the competition with the highest

Figure 4.14: Comparison of PDR performances for a user at 1110m

priority (as being the oldest packet). With the PFS and the Maximum C/I schedulers, if the current transmission for a voice packet fails, the voice packet returns in the competition and contends again. Therefore, we observe that delay performance of the FIFO is better than the other two schedulers.

The PFS comes 2nd, and performs better than the Maximum C/I scheduler, because it not only tends to select a user who can support peak data rate but also serve a user who has not received the service for a specified duration. The Maximum C/I scheduler performs worse than the other two schedulers, because it does not consider the delay experienced by the users in the queue. It only considers the channel conditions for the selection of a user. Figure 4.15 summarize the performances in terms of PDR and user throughput.

Figure 4.15: User throughput vs. PDR for a user at 1110m

In figure 4.15, performance of circuit-switched service is also shown for reference. We observe that up to 15 users (c.l of 0.52), all the three scheduler match the performance of the circuit-switched voice. When the number of users increases above 15, then the performance offered by the high-speed data techniques start to degrade. PDR increases with the increase in number of users and hence user throughput starts of decrease.

We might conclude that, all the three schedulers could meet the delay performance, offered by the circuit-switched voice service, for the cell loading of 0.52 (15 users).

4.6.2 Comparison of attempted data rates performances

Figure 4.16 shows the comparison of the average transmitted data rates (link adaptation capability) offered by the schedulers. We can observe that, for the cell loading of 0.92 (30 users), the attempted transmitted data rate is around 2.2 Mbps for the cases of the Maximum C/I and the PFS schedulers whereas for the case of the FIFO the attempted data rate is around 1.9 Mbps.

As expected, the Maximum C/I performs better than the other two schedulers, though the PFS matches the performance of the Maximum C/I scheduler. Both the PFS and the Maximum C/I schedulers take advantage of the short- term channel variations and try to serve the users who can support peak data rates.

The FIFO though chooses the modulation and coding parameters based on the channel conditions around the selected user, does select a user based on the channel conditions. It chooses a user with the earliest packet in the queues. Therefore its attempted data rate performance is worse than that of the other two schedulers.

Figure 4.16: Comparison of attempted data rates

4.6.3 Comparison of Effective data rate performances

Figure 4.17 shows the comparison of effective data rate (system's capability of successful transmission and reception, not including PDR) performances of the schedulers. The figure shows that achievable effective data rate is much lower than the attempted transmitted data rates shown in the figure 4.16.

This inefficiency is not the result of the retransmission attempts or the excessive delay in the queue. We observe that, on the average, one attempt is required to transmit an

Figure 4.17: Comparison of effective data rate performances

encoder packet (containing voice packets) successfully. It means that, on the average, the transmission duration for an encoder packet is around 1.67 msec (one time slot).

The inefficiency in the effective data rate is the result of the nature of the voice traffic pattern and voice packet size. As described in chapter 3, a voice packet of 251 bits is delivered to the physical layer every 20msec during speech bursts. Whereas, the schedulers are capable of transmitting at least 1024bits in one time slot (1.67msec).

For the cell loading of 0.92 (fully loaded cell with 30 users), we observe that the effective data rate is 150kbps. This value of effective data rate suggests that, on the average, one voice packet in each encoder packet was transmitted. It means that, the voice application provides less information bits to the schedulers than the schedulers are capable of transmitting successfully.

As shown in the table 3.1, an encoder packet can carry 1024, 2048, 3072 or 4096 information bits, depending upon the supportable data rates and queue status. If in each encoder packet, on the average, 1024 bits could be delivered to the physical layer, then for the similar conditions, the achievable effective data rate would be around 614.4kbps. Similarly for the case of 2048 information bits, the achievable effective data rate could be around 1228.8 kbps, and so on.

For data applications, packets are not discarded due to the delay. Therefore, under the condition of no packet discard, the values of effective data rates could be considered as sector throughput as well. And, we may infer that the achievable sector throughput would be 614kbps or more.

Therefore, we might conclude that supporting voice applications by using the proposed high-speed packet data transmission techniques results in inefficiency of effective data rates and throughput.

To summarize, all of the three schedulers and the associated high-speed packet data techniques could meet the delay performance, offered by the circuit-switched voice service, for the cell loadings of up to 0.52. The FIFO could meet the same performance for the cell loading of up to 0.67. The delay performance of the PFS is better than that of Maximum C/I scheduler.

As far as attempted served data rate is concerned, the Maximum C/I scheduler performs better than the other two schedulers. We observe, from figure 4.16, that the performance of the PFS, in terms of the attempted data rates, is matched closely to that of the Maximum C/I. The performance of the FIFO, in terms of the attempted data rates, is worse than that of the other two schedulers.

It is interesting that the delay performance of the PFS is comparable to that of the FIFO. And the attempted data rate performance of the PFS is almost the same as that of the Maximum C/I. According to [21] the throughput performance of FIFO based schedulers, offered to the data applications, is worse than that of the PFS and the Maximum C/I. Therefore, we may conclude that the PFS may be a suitable choice to meet the contrasting lower delay requirement of the voice applications and the higher throughput requirement of the data applications.

CHAPTER 5

CONCLUSIONS AND FUTURE WORK

New and popular data applications put demands on wireless networks to support high-speed data applications. New wireless service models have been proposed to address this challenge, which are based on high-speed packet data transmission techniques. In this document, we explored the possibility of using the same high-speed data transmission technology for voice applications, in search of a common technology for both voice and data services.

In this thesis, we proposed the required changes in the specified scheduling rules to improve the performance of proposed high-speed data transmission techniques. We observed that, with these proposed changes, the delay performances of the schedulers improve to such an extent that they could meet the performance of circuit-switched voice services, when there are up to 15 users anywhere in the cell. On the other hand, without these proposed changes, the quality of voice service could not be provided, that is offered using circuit-switched technology.

All of the three schedulers could meet the delay performance, offered by the circuit-switched voice service, for the cell loadings of up to 0.52. The FIFO could meet the same performance for the cell loading of up to 0.67. The delay performance of the PFS is better than that of the Maximum C/I scheduler. As far as attempted served data rate is concerned, the Maximum C/I scheduler performs better than the other two schedulers. The performance of the PFS, in terms of the attempted data rates, is matched closely to that of the Maximum C/I. The performance of the FIFO, in terms of the attempted data

rates is worse than that of the other two schedulers. We may conclude that the PFS may be a suitable choice to meet the contrasting lower delay requirement of the voice applications and the higher throughput requirement of the data applications.

Voice service between the users of 3G IP-based CDMA wireless networks via the Internet is dependant upon the performances of the Internet and the proposed IP-based downlink and uplink wireless networks. We provided the tables of available system capacities and available delay budgets to explore the possibilities of fully IP-based voice services and of a mix of voice and data traffic.

A downlink wireless access network, which employs circuit-switched technology offers a packet transfer delay of 40msec with a 1% FER. We found that the performance of the packet-switched voice service, with the help of high-speed data transmission techniques, is comparable to the performance of the circuit-switched voice services for the cell loading of up to 0.52 (15 users) and for the coverage area of up to 1110m. We observed that the proposed high-speed packet data techniques are inefficient, in terms of achievable throughput, when employed to support voice applications

We observed the performance improvements with the changes in the specified scheduling rules. Studies may be conducted to optimize the specified scheduling rules. We also observed that the maximum number of transmission attempts for an encoder packet was 4, which resulted by using the specified modulation and coding schemes for the target packet error rate of 1%. The possibility of better modulation and coding schemes may be explored in order to minimize the number of transmission attempts. This could help in reducing the delay exhibited by the packets in the schedulers.

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