Multimedia Communications over 3G
Wireless Communication Systems

C. K. Kodikara Patabandi

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Unis

Centre for Communication Systems Research
School of Electronics and Physical Sciences
University of Surrey
Guildford, Surrey GU2 7XH, UK

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To my mother and my father
Summary

This thesis addresses the transmission of video over third generation radio access networks. The first part of the thesis investigates the capabilities of Universal Terrestrial Radio Access Networks (UTRAN) in terms of the provision of multimedia communications. The error performance and traffic requirements of real-time video transmission over circuit switched and packet switched connections are examined. The effect of network parameter settings upon video performances is evaluated, and optimum radio bearer configurations for the transmission of video are derived.

A method of estimating received video quality after transmission over error prone environments is developed. The quality estimation is based on a distortion model, which accurately models the overall distortion seen in decoder frame reconstruction. This includes quantisation distortion, concealment distortion, and error propagation. Based on the developed performance model, optimum MTU (Maximum Transfer Unit) size for efficient wireless video communications over a packet switched access network is derived.

The second part of the thesis investigates quality enhancement techniques for multimedia traffic transmitted over wireless channels. Quality enhancement is achieved at three levels of the transmission process: link level, application level and system level.

Link level quality enhancement techniques are designed to optimise the allocation of link level parameter values according to the media characteristics. A novel Unequal Error Protection scheme and a novel Unequal Power Allocation scheme are designed to exploit the inherent diversity in the subjective importance of different sections of compressed media. The algorithms are developed and analysed for transmission of video over 3G wireless systems. The effectiveness of these algorithms is demonstrated through the results of simulated transmission over a UMTS channel.

Application level quality enhancement techniques are designed to explore the time-varying nature of the wireless channel. A number of link adaptation schemes are proposed for real time video communication and real-time video streaming over 3G wireless systems. These algorithms are designed to enhance the perceptual video quality, and the system utilisation. This is achieved by adapting the allocated radio network parameters and the source parameters, according to a feedback channel condition. Simulation results show a significant performance improvement compared to non-adaptive schemes.

Finally, system level adaptation techniques are designed for efficient radio resource allocation in multi-user scenarios. Two adaptive resource allocation schemes are proposed and analysed for real-time video communications in a UMTS system. The proposed algorithms are shown to provide improved performances in terms of average received video quality and user satisfaction.

Key words: Wireless Communication, Multimedia communication, Video Transmission.
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Glossary of Terms

3G 3rd Generations
3GPP Third Generation Partnership Project
8-PSK 8-state Phase Shift Keying

ACDMA Adaptive CDMA
ACK ACKnowledgement
ACS Adaptive Coding Scheme
AIR Adaptive Intra Refresh
APN Access Point Name
ARQ Automatic Repeat reQuest
AVO Audio Visual Object
AWGN Additive White Gaussian Noise

BER Bit Error Rate
BLER BLock Error Rate
BMC Broadcast/Multicast Control
BSC Base Station Controller
BSS Base Station Subsystem
BTS Base Transceiver Station

CC Convolutional Code
CCSR Centre for Communication System Research
CDF Cumulative Distribution Function
CDMA Code Division Multiple Access
CIR Carrier to Interference Ratio
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<th>Term</th>
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<tr>
<td>CN-BS</td>
<td>Core Network Bearer Service</td>
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<tr>
<td>CRC</td>
<td>Cyclic Redundancy Check</td>
</tr>
<tr>
<td>CRTP</td>
<td>Compressing RTP/UDP/IP</td>
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<tr>
<td>CS</td>
<td>Circuit Switched</td>
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<tr>
<td>DBS</td>
<td>Direct Broadcasting Satellite</td>
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<td>DCCH</td>
<td>Dedicated Control Channel (DCCH)</td>
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<tr>
<td>DCH</td>
<td>Dedicated Channel (DCH)</td>
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<tr>
<td>DCS1800</td>
<td>Digital Cellular Network at 1800MHz</td>
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<tr>
<td>DCT</td>
<td>Discrete Cosine Transform</td>
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<tr>
<td>DMIF</td>
<td>Delivery Multimedia Integration Framework</td>
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<tr>
<td>DP</td>
<td>Data Partition</td>
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<td>DPCCH</td>
<td>Dedicated Physical Control Channel (DPCCH)</td>
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<td>DPCH</td>
<td>Dedicated Physical Channel (DPCH)</td>
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<td>DPDCH</td>
<td>Dedicated Physical Data Channel (DPDCH)</td>
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<td>DS-CDMA</td>
<td>Direct Sequence CDMA</td>
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<td>DSMA-CD</td>
<td>Digital Sense Multiple Access Collision Detection</td>
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<td>DSP</td>
<td>Digital Signal Processing</td>
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<tr>
<td>DTCH</td>
<td>Dedicated Transport (Traffic) Channel (DTCH)</td>
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<td>DVD</td>
<td>Digital Versatile Disk</td>
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<tr>
<td>EDGE</td>
<td>Enhanced Data rate for GSM Evolution</td>
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<tr>
<td>EGC</td>
<td>Equal Gain Combining</td>
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<td>EGPRS</td>
<td>Enhanced GPRS</td>
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<td>FDD</td>
<td>Frequency Division Duplex</td>
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<td>FDMA</td>
<td>Frequency Division Multiple Access</td>
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<td>FEC</td>
<td>Forward Error Correction</td>
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<td>Term</td>
<td>Definition</td>
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<tr>
<td>GERAN</td>
<td>GSM/EDGE Radio Access Network</td>
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<td>GGSN</td>
<td>Gateway GPRS Support Node</td>
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<tr>
<td>GMSK</td>
<td>Gaussian Minimum Shift Keying</td>
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<tr>
<td>GOB</td>
<td>Group Of Objects</td>
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<tr>
<td>GSM</td>
<td>Global System for Mobile communication</td>
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<tr>
<td>GTP-U</td>
<td>GPRS Tunnelling Protocol for User Plane</td>
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<tr>
<td>HDTV</td>
<td>High Definition TeleVision</td>
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<tr>
<td>HEC</td>
<td>Header Extension Code</td>
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<td>HSCSD</td>
<td>High Speed Circuit Switched Data</td>
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<td>IM</td>
<td>IP Multimedia</td>
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<td>IMT-2000</td>
<td>International Mobile Telecommunications 2000</td>
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<td>IP</td>
<td>Internet Protocol</td>
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<td>ISDN</td>
<td>Integrated Services Digital Network</td>
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<td>ISO</td>
<td>International Organisation for Standardisation</td>
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<td>ITU</td>
<td>International Telecommunication Union</td>
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<tr>
<td>ITU-R</td>
<td>ITU- Radio communication sector</td>
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<tr>
<td>JAS-UEP</td>
<td>Joint Adaptive Spreading gain control and UEP</td>
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<td>JSCC</td>
<td>Joint Source Channel Coding</td>
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<tr>
<td>LAA</td>
<td>Link Adaptation Algorithm</td>
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<tr>
<td>LLC</td>
<td>Logical Link Control</td>
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<td>MAC</td>
<td>Medium Access Control</td>
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<td>MB</td>
<td>Macro Block</td>
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<td>MBM</td>
<td>Motion Boundary Marker</td>
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<tr>
<td>Abbreviation</td>
<td>Description</td>
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<tr>
<td>MCS</td>
<td>Modulation Coding Scheme</td>
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<td>MDC</td>
<td>Multiple Descriptive Code</td>
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<td>MGW</td>
<td>Media GateWay</td>
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<td>MPEG</td>
<td>Moving Picture Expert Group</td>
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<td>MRC</td>
<td>Maximum Ratio Combining</td>
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<tr>
<td>MSC</td>
<td>Mobile Switching Centre</td>
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<td>MT</td>
<td>Mobile Terminal</td>
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<td>MTU</td>
<td>Maximum Transfer Unit</td>
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<tr>
<td>MV</td>
<td>Motion Vector</td>
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<td>NACK</td>
<td>Negative ACKnowledgement</td>
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<td>OCNS</td>
<td>Orthogonal Channel Noise Simulator</td>
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<td>OF</td>
<td>Orthogonality Factor</td>
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<td>OVSF</td>
<td>Orthogonal Variable Spreading Factor</td>
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<td>PACCH</td>
<td>Packet Associated Control Channel</td>
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<td>PAGCH</td>
<td>Packet Access Grant Channel</td>
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<td>PBCCH</td>
<td>Packet Broadcast Control Channel</td>
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<td>PC</td>
<td>Personal Computer</td>
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<tr>
<td>PCCCH</td>
<td>Packet Common Control Channel</td>
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<td>PCCPCH</td>
<td>Primary Common Control Physical Channel</td>
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<td>PCPCH</td>
<td>Physical Common Packet Channel</td>
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<td>PDCH</td>
<td>Packet Data Channel</td>
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<td>PDCP</td>
<td>Packet Data Convergence Protocol</td>
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<td>PDP</td>
<td>Packet Data Protocol</td>
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<td>PDSCH</td>
<td>Physical Downlink Shared Channel</td>
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<td>PDTCH</td>
<td>Packet Data Traffic Channel</td>
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<td>Term</td>
<td>Description</td>
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<tr>
<td>PDU</td>
<td>Protocol Data Unit</td>
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<td>PHY</td>
<td>PHYsical layer</td>
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<td>PRACH</td>
<td>Physical Random Access CHannel</td>
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<td>PS</td>
<td>Packet Switched</td>
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<td>PSNR</td>
<td>Peak Signal to Noise Ratio</td>
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<td>QCIF</td>
<td>Quarter Common Intermediate Format</td>
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<tr>
<td>QoS</td>
<td>Quality of Service</td>
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<td>QPSK</td>
<td>Quadrature Phase Shift Keying</td>
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<td>QSR-ARQ</td>
<td>Qos aware Selective Repeat ARQ</td>
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<td>RAB</td>
<td>Radio Access Bearer</td>
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<td>RLC</td>
<td>Radio Link Control</td>
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<td>RNC</td>
<td>Radio Network Controller</td>
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<td>RNS</td>
<td>Radio Network Subsystem</td>
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<tr>
<td>ROCCO</td>
<td>ROust Checksum based header COmpression</td>
</tr>
<tr>
<td>RRC</td>
<td>Radio Resource Control</td>
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<tr>
<td>RSS</td>
<td>Received Signal Strength</td>
</tr>
<tr>
<td>RSVP</td>
<td>ReSource reservation Protocol</td>
</tr>
<tr>
<td>RTP</td>
<td>Real Time Protocol</td>
</tr>
<tr>
<td>RVLC</td>
<td>Reversible Variable Length Code</td>
</tr>
<tr>
<td>SAD</td>
<td>Sum of Absolute Differences</td>
</tr>
<tr>
<td>SAP</td>
<td>Service Access Point</td>
</tr>
<tr>
<td>SCCPCH</td>
<td>Secondary Common Control Physical CHannel</td>
</tr>
<tr>
<td>SDP</td>
<td>Session Description Protocol</td>
</tr>
<tr>
<td>SDU</td>
<td>Service data Unit</td>
</tr>
<tr>
<td>SF</td>
<td>Spreading Factor</td>
</tr>
<tr>
<td>Acronym</td>
<td>Definition</td>
</tr>
<tr>
<td>---------</td>
<td>------------</td>
</tr>
<tr>
<td>SGSN</td>
<td>Serving GPRS Support Node</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiated Protocol</td>
</tr>
<tr>
<td>SNDCP</td>
<td>Sub-Network Dependent Convergence Protocol</td>
</tr>
<tr>
<td>SNR</td>
<td>Signal to Noise Ratio</td>
</tr>
<tr>
<td>SPW</td>
<td>Signal Processing Work system</td>
</tr>
<tr>
<td>SQBAS</td>
<td>Source Quality Based Adaptive Scheme</td>
</tr>
<tr>
<td>SR-ARQ</td>
<td>Selective Repeat ARQ</td>
</tr>
<tr>
<td>SRB</td>
<td>Signalling Radio bearer</td>
</tr>
<tr>
<td>TB</td>
<td>Transport Block</td>
</tr>
<tr>
<td>TBAS</td>
<td>Throughput Based Adaptive Scheme</td>
</tr>
<tr>
<td>TBF</td>
<td>Temporary Block Flow</td>
</tr>
<tr>
<td>TC</td>
<td>Turbo Code</td>
</tr>
<tr>
<td>TDD</td>
<td>Time Division Duplex</td>
</tr>
<tr>
<td>TDMA</td>
<td>Time Division Multiple Access</td>
</tr>
<tr>
<td>TE</td>
<td>Terminal Equipment</td>
</tr>
<tr>
<td>TFCI</td>
<td>Transport Format Combination Indicator</td>
</tr>
<tr>
<td>TFI</td>
<td>Transport Format Indicator or Temporary Flow Identity</td>
</tr>
<tr>
<td>TFT</td>
<td>Traffic Floe Template</td>
</tr>
<tr>
<td>TI</td>
<td>Transaction Identifier</td>
</tr>
<tr>
<td>TIA</td>
<td>Telecommunication Industry Association</td>
</tr>
<tr>
<td>TPC</td>
<td>Transmit Power Control</td>
</tr>
<tr>
<td>TS</td>
<td>Technical Specification or Time Slot</td>
</tr>
<tr>
<td>TTI</td>
<td>Transmission Time Interval</td>
</tr>
<tr>
<td>TU</td>
<td>Typical Urban</td>
</tr>
<tr>
<td>UDP</td>
<td>User Datagram protocol</td>
</tr>
<tr>
<td>UE</td>
<td>User Equipment</td>
</tr>
<tr>
<td>Term</td>
<td>Description</td>
</tr>
<tr>
<td>--------</td>
<td>--------------------------------------------</td>
</tr>
<tr>
<td>UEP</td>
<td>Unequal Error protection</td>
</tr>
<tr>
<td>UMTS</td>
<td>Universal Mobile Telecommunications System</td>
</tr>
<tr>
<td>UMTS-BS</td>
<td>UMTS Bearer Service</td>
</tr>
<tr>
<td>UPA</td>
<td>Unequal Power Allocation</td>
</tr>
<tr>
<td>UTRA</td>
<td>Universal Terrestrial Radio access</td>
</tr>
<tr>
<td>UTRAN</td>
<td>Universal terrestrial Radio Access Network</td>
</tr>
<tr>
<td>VBR</td>
<td>Variable Bit Rate</td>
</tr>
<tr>
<td>VLR</td>
<td>Visitor Location Registry</td>
</tr>
<tr>
<td>VLSI</td>
<td>Very Large Scale Integration</td>
</tr>
<tr>
<td>VOP</td>
<td>Video Object Plane</td>
</tr>
<tr>
<td>VP</td>
<td>Video Packet</td>
</tr>
<tr>
<td>WCDMA</td>
<td>Wideband CDMA</td>
</tr>
<tr>
<td>WMSA</td>
<td>Weighted Multi Slot Averaging</td>
</tr>
</tbody>
</table>
Chapter 1

1 Introduction

1.1 Preamble

Recent developments in wireless communication, multimedia technology, and microelectronics technology have created a new paradigm in mobile communications. Third generation wireless communication technologies promise significantly higher transmission rates and service flexibility, over a wide coverage area, than are possible with second generation wireless communication systems. High compression, error robust multimedia codecs have been designed to enable the support of multimedia application over error-prone bandwidth limited channels. The advances of VLSI and DSP technologies are preparing light weight, low cost, portable devices capable of transmitting and viewing multimedia streams. The above technological developments have shifted the service requirements of mobile communication from conventional voice telephony to business oriented multimedia services in third generation wireless communication systems.

Multimedia services by definition require the transmission of multiple media streams such as video, still picture, music, voice, and text data. A combination of these media types provides a number of value-added services, including video telephony, E-commerce services, multiparty video conferencing, virtual office and 3D video games. Different media types have different requirements on quality of service, and also enforce conflicting constraints on the communication
networks. Still picture and text data are categorised as background services and require high data rates but have no constraints on the transmission delay. Voice services are on the other hand characterised by low delay. However, they can be coded using fixed low rate algorithms operating in the 5-12 kbps range. In contrast to voice and data services, low-bit rate video coding involves rates at tens to hundreds of kilobits per second. Moreover, video applications are delay sensitive and impose tight constraints on system resources.

Mobile multimedia applications play an important role in the rapid penetration of 3G services, and the success of third generation communication systems. Even though the high transmission rates and service flexibility have made wireless multimedia communication possible over third generation wireless communication systems, many challenges remain to be addressed in order to support efficient communications in multi-user, multi-service environments. In addition to the high initial cost associated with the deployment of third generation systems, the move from telephony and low-bit rate data services to bandwidth consuming third generation services implies high system costs, as they consume a large portion of the available resources. Moreover, for rapid market evolvement, these wideband services should not be substantially more expensive than the services offered today. Therefore, efficient system resource (mainly the bandwidth limited radio resource) utilisation and Quality of Service (QoS) management are critical in 3G systems.

The provision of quality of service for multimedia applications and efficient resource management are in sharp conflict with each other. Of course, it is possible to provide high quality multimedia services with the use of a large amount of radio bandwidth and very strong channel protection. However, this is clearly inefficient in terms of system resource allocation. Moreover, the perceptual multimedia quality received by the end users depends on many factors such as source rate, channel protection, channel quality, error resilience techniques, transmission/processing power, system load and user interference. Therefore, it is difficult to obtain an optimal source and network parameter combination for a given source and channel characteristics. The time varying error characteristics of the radio access channel aggravate the problem.

This thesis investigates ways that multimedia communications may be achieved over third generation mobile networks in such a way as to offer service flexibility, and accepted quality of service, while maximising system resource utilisation.
Chapter 1. Introduction

1.2 Objectives and overall project description

The main objective of the research is the investigation of the capabilities of 3G radio access networks to support multimedia services. Even though multimedia streams consist of video, audio, speech and text data, the investigation will focus on to the provision of video communications, since video traffic requires higher network bandwidth and has stringent delay constraints. Among the variety of video applications, primarily, the performance of conversational video communication will be evaluated. Due to their requirements for low-latency and for the preservation of time relation between information entities of the stream, the conversational services place strict requirements on the underlying access network. Unidirectional streaming video applications, which have higher acceptable delay variation than that of conversational services, will also be described.

The overall project is divided into two main parts. The first part considers the investigation of optimal radio access bearer configurations for the transmission of video applications. The radio access bearers, which characterise the radio interface, are defined by a set of transport channel parameters such as transport block size, CRC (Cyclic Redundancy Check) code length, channel coding schemes, RLC (Radio Link Control) mode, MAC (Medium Access Control) type, radio frame size, rate matching, etc. The perceived quality of the application seen by the end user is greatly affected by the settings of these radio access parameters. The influences of network parameter settings on the performance of video communications are investigated, and the optimal combination of parameter settings for different application scenarios is derived.

A fundamental question seen in wireless multimedia communications is how to manage the radio resources in order to satisfy the end users' quality of service requirement, taking into consideration available system resources, system load, interference, propagation environment, and user mobility. The second part of the project will investigate the important issues of resource allocation and quality enhancement techniques for multimedia traffic transmitted over wireless channels. Quality enhancement is achieved at three levels of the transmission process, named as:

- Link level quality
- Application level quality
- System level quality.
Chapter 1. Introduction

The link level quality adaptation, which is mainly controlled by a Radio Resource Controller (RRC) within the network, is used to allocate the transport channel parameters according to the media characteristics and propagation channel characteristics. The ultimate goal is to provide each connection with the negotiated quality, while using as little radio resources as possible. Link level quality enhancement techniques, which are designed to exploit the inherent diversity in the subjective importance of different sections of compressed media, are developed and analysed for the transmission of video over 3G wireless systems. Compressed streams are separated into multiple streams, depending on their importance to visualisation and intelligibility of the decoded video. Separated streams are given different priority and are transmitted over different transport channels so as to increase the robustness of the transmitted information. Thus, the developed techniques result in unequal error protection and unequal power allocation techniques, and provide significantly improved video performances.

Application level quality adaptation is designed to explore the time-varying nature of wireless channels. The channel quality experienced at the receiver is measured and fed back to the transmitter. The source and the allocated link level parameters are adjusted according to estimated channel conditions to achieve optimal multimedia quality throughout the transmission. A number of link adaptation techniques, which is designed to enhance the perceptual video quality and the system throughput utilisation, are investigated for video telephony applications. Furthermore, the employment of link adaptation in video streaming applications is examined.

System level quality adaptation is used to allocate available system resources among different media users in such a way as to maximize the system throughput and user satisfaction. This is the most important entity for the success of a communication system. System level quality adaptation takes into account information such as current system load, other user interference, other cell interference, and user quality requirements when a new connection is granted, during negotiation/renegotiation of quality of service parameters, and adaptation of QoS parameters for a given application. System level quality adaptation and resource allocation techniques are discussed and analysed for video applications over 3G systems. The proposed algorithms provide improved system performances in terms of average received video quality and user satisfaction.
1.3 Original achievements

A number of publications have been produced as a result of the research described in this thesis. These publications are listed in Appendix A. The research achievements can be summarised as:

- Real-time PC based UMTS (Universal Mobile Telecommunication System) emulator software.
- Comprehensive study of video transmission over UMTS packet switched and circuit switched radio access bearers.
- Modelling rate-distortion characteristics of video applications over error prone packet/circuit switched networks.
- Analysis of optimal MTU (Maximum Transfer Unit) for efficient video communication over packet switched wireless networks.
- An optimal unequal error protection scheme for video communication based on an importance matrix.
- Energy efficient video telephony over UMTS. A combined unequal error protection scheme and unequal power allocation scheme is designed based on the predicted received video quality at the encoder.
- Link adaptation techniques, which are designed to maximise the received video quality and the system utilisation for transmission of real-time video over EDGE (Enhanced Data rates for GSM Evolution) networks.
- A link adaptation technique for real-time video streaming application over EDGE networks.
- Link adaptation techniques for video telephony applications over UMTS networks.
- System level perceptual quality enhancement techniques for video communication in multi-user scenarios.

1.4 Structure of thesis

The first chapter provides the rationale behind the research work and a brief description of the overall project while the final chapter, Chapter 9, summarises the research work carried out and research achievements. Furthermore, it discusses the potential areas for future research in efficient wireless multimedia communications. The work presented in the other chapters is summarised below.
1.4.1 Contents of chapter 2

Chapter 2 describes the background to the work carried out. The chapter starts with a discussion of the constraints and problems associated with the provision of multimedia communications. Then an introduction to multimedia compression technologies and a literature survey of advanced multimedia transmission technologies are given. The second half of the chapter discusses the enabling of third generation wireless communication systems, and their capabilities for multimedia transmission. In addition, third generation mobile services/applications and quality of service management architecture are presented. The chapter is concluded with an explanation of resource management strategies in wireless multimedia communications.

1.4.2 Contents of chapter 3

Universal Mobile Telecommunication System (UMTS), which is the leading third generation mobile communication system, is selected as the major transmission network for the multimedia provisioning and quality enhancement experiments carried out in this thesis. Chapter 3 describes the design and model implementation of a downlink physical link layer simulator. The developed simulator is validated against reference performance figures presented in relevant 3GPP (3rd Generation Partnership Project) documents. The actual radio interface experienced by users is emulated by integrating the physical link layer simulator with a protocol data flow model, which is designed in Visual C++. This facilitates real-time interactive testing of the effect of UMTS Radio Access Network (UTRAN) parameters upon the received multimedia quality.

1.4.3 Contents of chapter 4

Chapter 4 investigates the capabilities of UTRAN for video communications. The effects of radio network parameter settings on the performances of video telephony applications are experimentally determined by using the UTRAN emulator designed in Chapter 3. The video quality received by the end user depends on the sequence characteristics, network parameter settings and the propagation conditions. Video transmission over both circuit switched and packet switched connections are examined. Source throughput capabilities of UTRAN for video communications are derived for different radio bearer configurations.
1.4.4 Contents of chapter 5

The performances of many error resilience techniques can be improved if the quality of the reconstructed video at the decoder can be predicted prior to information transmission. A technique for estimating received video quality for given encoder parameters, decoder error concealment techniques and channel statistics is investigated in Chapter 5. The performance estimation is based on a distortion model, which takes into account quantisation distortion, concealment distortion, and distortion due error propagation over predictive frames. The accuracy of the model is evaluated by comparison with actual performances obtained for the transmission of video over the simulated UMTS channels. Using the developed performance estimation model, the optimal transport layer packet size, for efficient wireless video communications over packet switched networks, is derived.

1.4.5 Contents of chapter 6

Link level quality enhancement methods for transporting video over wireless networks are considered in Chapter 6. In particular, video data prioritisation techniques and Unequal Error Protection (UEP) schemes are explored. A novel optimal data prioritisation technique, which is based on the estimated received video quality and perceptual importance, is proposed. The performance improvement achieved with the proposed algorithm is demonstrated against a traditional MPEG-4 data partition based UEP scheme for video transmission over UTRAN. Furthermore, an Unequal Power Allocation (UPA) algorithm is proposed and analysed for energy efficient video communications in CDMA (Code Division Multiple Access) based systems.

1.4.6 Contents of chapter 7

In Chapters 3-6, video performances are investigated for the delivery of video services over multi-path channels. Fast fading channel characteristics is implemented in the transmission channel simulation, but user mobility is not contrived. Effect of user mobility on received video quality is investigated in Chapter 7. The influences of user mobility on channel characteristics are modelled considering shadow fading and the propagation loss of the transmission path. A number of link adaptation techniques, which exploit the time varying channel characteristics, are investigated. Two link adaptation algorithms are proposed for maximising the received video quality for fixed channel allocation. Another is designed to maximise the system throughput while guaranteeing a
required video quality. Algorithm performances are demonstrated for video transmission over EGPRS and UMTS networks.

1.4.7 Contents of chapter 8

Chapter 8 describes a system performance investigation for real-time video communications in multi-user scenarios. System performances are investigated for 3-sectored cell layout considering 24 hexagonal cells in the Vehicular environment. Three different resource allocation schemes are investigated taking into account the total base station power limitation and the number of available spreading codes (code limitation). Algorithms are designed to optimise the perceived video quality by the end users. System performance evaluation is carried out based on the received video quality and user satisfaction.
Chapter 2

2 Wireless Multimedia Communications

2.1 Introduction

This chapter provides a brief overview of recent developments in wireless video technologies and third generation wireless communication systems. The constraints and problems associated with wireless video transmission are addressed in Section 2.2. Advanced video processing technologies, which make video bit streams suitable for transmission over a time varying error prone wireless channels, are summarised in Section 2.3. This section also discusses QoS provisioning techniques for delivery of multimedia streams, while maintaining high system (power and spectral) efficiency. The second half of the chapter deals with third generation wireless communication technologies. Services and service capabilities of EGPRS and UMTS technologies for the transmission of multimedia are presented in Section 2.4. The chapter is concluded with a discussion of resource management issues in wireless multimedia communications in Section 2.5.

2.2 Constraints on wireless video communications

Reliable transmission of video over wireless links is becoming an increasingly important application requirement in mobile communications. However, supporting robust video
communications over wireless networks is a significant problem, primarily because of three factors:

- Scarcity of bandwidth
- Time varying error characteristics of the transmission channel
- Power limitations of wireless devices.

While the above constraints are present in many communications systems, the challenges they impose are particularly acute for video communications. Voice applications are delay sensitive by definition, however, the voice traffic requires low channel bandwidth in the 5-12 kbps range. On the other hand, data applications require high bit rates but tolerate high transmission latency, leaving the processor freedom to retransmit the packet if necessary, to achieve the adequate quality. In contrast to the above, low-bit rate video coding algorithms operate at rates ranging from tens to hundreds of kilobits per second. Moreover, video applications are delay sensitive and it may not be easy to use retransmission as much as in data applications.

Unlike wire line networks, where the increase in the number of users whose demands can be met by adding more fibre, cable or other similar wired media, the system capacity in wireless networks cannot be increased arbitrarily. Thus, due to the large quantity of data involved in video communications, compression is almost always used in the transmission and the management of digital video over wireless networks. Video compression technologies typically apply predictive coding and variable-length coding to achieve a higher degree of compactness. However, this has the undesirable side-effect of increasing the susceptibility of the media streams to errors. Predictive coding inevitably results frame-to-frame error propagation. The variable length decoding, on the other hand, tends to lose code word synchronisation in the presence of bit errors. This causes a large number of following symbols to be incorrectly decoded, regardless of their correct reception.

The problems arising from scarcity of bandwidth and the use of highly error susceptible video coding techniques, are aggravated by the fact that radio channels are highly unpredictable. This unreliability results from distance propagation phenomena such as multi-path fading, shadowing, path loss, noise and interference from other users, all of which have a multiplicative effect on the transmitted signal, causing it to deteriorate. Multi-path propagation caused by the superposition of radio waves reflected from surrounding objects, gives rise to frequency-selective fading, resulting in rapid fluctuations of the phase and amplitude of the signal. Shadowing caused by the presence of large physical objects, which preclude a direct line of sight between the radio transmitter and
receiver, is a medium-scale effect. Path loss causes the received power to vary gradually due to signal attenuation, determined by the geometry of the path profile in its entirety [RAPP-96]. The combined effect of these phenomena is that the receiver has to deal with a bit stream that is corrupted by both random bit errors and burst errors. Unless the video encoder and decoder take proper action to deal with a bit-stream, which is corrupted by both random bit and burst errors, the video communication system may totally break down.

Video coders are currently designed independently from channel coders. It does not take into account the joint source-channel coding adaptation and optimisation possible for resource allocation in a time varying channel environment. When independently encoded video is transmitted over a time varying channel such as typical wireless link, poor video performances are resulted. As shown in [BYST-00], joint source and channel coding can be implemented to optimise the performances of video communication over error prone environments.

At a higher layer, error protection schemes that use sophisticated convolutional or block codes may be employed to alleviate the problem associated with channel errors, but they aggravate bandwidth problems since more bits have to be added to the video bit stream. Similarly, automatic repeat request type retransmission procedures improve error recovery against burst errors, but aggravate latency problems. Equalization and interleaving techniques that reduce channel interference and the effect of burst errors can also be used to alleviate some of the problems [FREI-99]. However, performance improvements come at the expense of reduced system efficiency, increased system cost, and increased processing delay.

Evaluations of current cellular communications standards are biased towards integrated packet voice and data communications, even though digital video communications were considered as desirable. The characteristic of digital video was not accommodated in the development of these standards. For example, the selection procedure for the choice of radio technologies of UMTS specifies only speech and data traffic models to be used in the testing procedures, but no video traffic model is given [UMTS 30.03]. This provides inadequate systems for transmission of compressed video over wireless channels in terms of resource allocation algorithms, traffic scheduling mechanisms, and channel access protocols.

In second generation cellular standards, the radio frame length is set to be 20 ms, which is equivalent to the voice codec packet generation period. The connection is also guaranteed one
time slot per radio frame for the duration of a talk spurt in Time Division Multiplex Access (TDMA) based communication systems. Therefore, the resource allocation and quality management can be optimised for voice services. However, resource allocation based on 20 ms frame length or time slots has no meaning for video communication as video data arrival intervals depend on the video frame rate setting of the encoder. In addition, the data within a video frame is larger than a radio frame, and the video frame size varies from frame to frame due to the variable bit rate nature of video codecs. Thus, multiple slot resource allocation is necessary and it should be designed carefully to guarantee the quality requirement, while maximising the system utilisation.

Current cellular standards use separate protocol layer architectures to reduce implementation complexity. Each protocol layer is individually optimised for the defined functions and the protocol layer interaction is conducted through specified interfaces. The data within each protocol layer is transparent to each other. Therefore, layered protocol architecture may not provide optimal processing for the application data and may also add some redundant information at the lower layers. This reduces the amount of available radio resources, making cross layer optimisation important in multimedia communications.

Currently available compressed video, such as MPEG-4 coded video streams, can be separated into multiple streams with different quality of service requirements according to their importance to the perceptual quality of the decoded video. Most of the popular channel access protocols are unable to guarantee this multiple QoS requirements resulting in poor system performances and sometimes intolerable quality video. These multiple quality requirements of video should be incorporated into the channel access protocol design.

Video is characterised by high data rates and low delay tolerance, and it has different system requirements from voice and data services. Thus, video cannot be treated similarly to voice, it cannot be given a high priority in the network resource allocation. On the other hand it is not appropriate for video to be treated as data, as this would lead to unacceptable delays. Therefore, video has to be treated as a separate entity with its own set of requirements in network resource allocation schemes. While it is true that it is impossible to eliminate all the problems stated above, the work described in this thesis has taken some attempt in achieving improved quality video transmission over third generation wireless communication systems.
2.3 Video compression technologies

Work on video coding standards has proceeded primarily within the International Organisation for Standardisation (ISO) and the International Telecommunication Union (ITU). ISO has developed a series of video codec standards namely, MPEG-1, MPEG-2, and MPEG-4 [MPEG]. Each of these standards actually provide a set of specifications for different aspects of audio-visual compression, including audio coding, video coding, multiplexing and others. MPEG-1 operates at bit-rates of about 1.5 Mbit/s and targets storage and retrieval of multimedia information on a CD-ROM. MPEG-2 operates at bit rates around 3-15 Mbit/s and is designed for the compression of higher resolution video signals for broadcast applications. Applications of MPEG-2 include Direct Broadcast Satellite (DBS), Digital Versatile Disk (DVD) and High Definition TeleVision (HDTV). While MPEG-4 covers a wide range of multimedia applications, an important aspect of MPEG-4 is the focus on very low bit rate video coding, error resilience and concealment for transmission over error prone channels [MPEG4]. The ITU has also developed a series of video coding standards including H.261, H.263 and recently H.264. H.261 made videoconferencing feasible at 64 kbit/s using the capacity of one ISDN channel. H.263 and H.264 are designed for error robust low bit rate video communications. More details about video compression standards can be found in [ICT].

![Figure 2.1: Layout of a DCT based video encoder (taken from WANGb-00).](image)

Figure 2.1: Layout of a DCT based video encoder [taken from WANGb-00].
Most of the above mentioned standards share a common basic approach in achieving information compression. A hybrid motion compensated DCT structure in which $8 \times 8$ pixel blocks are encoded using the Discrete Cosine Transform is employed in the compression process. A block diagram of video encoding process is depicted in Figure 2.1. The DCT results in information energy being compacted into low frequency components. Thus, perceptually unimportant high frequency components can be discarded achieving a high compression ratio while minimising the visual quality degradation. More compression efficiency is achieved by eliminating redundancies in the temporal domain. The current video frame is compared to a reference video frame and motion, is estimated based on $8 \times 8$ or $16 \times 16$ pixel Macro-Blocks (MBs). Using the estimated motion information, the current video frame is predicted from the reference video frame. The difference between the current frame and the motion compensated frame is encoded. To maximise coding efficiency, both the motion estimation information and the transformed prediction error are represented using variable length codes.

Although most standards currently adopt a hybrid motion compensated DCT approach to video coding, alternative approaches such as waveform based, object and model based coding can offer a variety of benefits for transmission of compressed video especially at low bit rates. Waveform based codecs perform sub-band filtering in the compression process. Generally, sub-band filters are used for the spatial dimensions while a DCT derived filter bank is applied to the temporal dimension [EBRA-99, VASS-98, WILK-95]. This approach results in a scalable video output stream, however, the drawback is the high computational complexity.

Model based techniques rely mainly on the existence of an underlying model. The goal here is to find the best matched model together with its corresponding parameters from a database of different models [LAVA-00]. Therefore, the model-based technique is two fold: the first part is the analysis and second is the synthesis. Analysis is the most difficult task due to the complexity involved with most natural scenes. So far most effort has been concentrated on simple scenes such as head-and-shoulder sequences and simple models such as lips and eye models [STRE-96]. The synthesis part is much more straightforward.

In the object-based approach, objects are defined as regions with three associated parameters, these being shape, texture and motion. The parameters are obtained by image analysis based on either 2D or 3D objects. Typically, the shape information is represented by contour information [SPAA-97, WANG-95]. Object tracking techniques are incorporated in to object based codec in
order to avoid frequent transmission of object parameters. Because of this more natural representation of objects, compression can be more efficient.

2.3.1 Recommended video standards

Among many video codec standards, only MPEG-4 and H.263 are recommended by 3GPP [3GPP TS 26.235] to be used in video communications over third generation mobile communication systems. Both standards share a common basic approach, which uses hybrid motion compensation based DCT in compression. A number of different techniques, which are introduced in the above standards, are used to enable robust transmission of compressed video data over noisy communication channels. Most of the error resilience features introduced in H.263 are described in Annexes, but are not always implemented. Due to its enhanced error resilience capabilities included as core feature, MPEG-4 encoded sequences are used as the application in the experiments carried out in this thesis. Thus, no further discussion of H.263 is given. Interested readers can refer to [ITU-T-00, ITU-T-97].

2.3.2 MPEG-4 video codec

MPEG-4 was originally aimed at low-bit rate video communication. Later, attention was focused on producing a multi-media coding standard providing a variety of new features. MPEG-4 follows an object based compression methodology, where a scene is considered to consist of a number of Audio-Visual Objects (AVOs). These AVOs may represent a complete video frame, an object present in a video scene, a computer generated graphic, text, an image, an audio stream or speech. The AVOs can be encoded and transmitted as separate entities or in a combined form according to the requirement specified by the applications. For example, in an interactive application session, the user may require object modifications, thus separate encoding and transmission of AVOs are necessary.

A system layer, which is incorporated into the standard, describes the way these AVOs are synchronised, multiplexed and presented at the receiver providing appropriate Quality of Service for various applications. The system layer contains a synchronisation layer and a delivery layer as shown in Figure 2.2. The delivery layer is further divided into a Delivery Multimedia Integration Framework (DMIF) layer and a TransMux layer. The synchronisation layer specifies the synchronised delivery of streams from the source to the destination while exploiting the different
Chapter 2. Wireless Multimedia Communications

quality of services offered by the network. Media synchronisation is guaranteed by the insertion of a time stamp into elementary streams. The syntax of the synchronisation layer can be reconfigured according to the application requirement allowing its operation over a wide range of services and systems. The DMIF layer manages the efficient delivery of multimedia over the selected network. In particular stream grouping based on quality requirements, classification and multiplexing are performed at the DMIF layer using the MPEG-4 defined FlexMux tool. Low multiplexing overhead is achieved in these processes. Interfaces to a set of transport protocols are specified within TransMux layer [KOEN-00]. The choice of appropriate transport protocol is left open to the end users or the service providers. The RTP/UDP/IP interface is investigated for the transmission of video over the packet switched connection in the work described in this thesis.

![Figure 2.2: The MPEG-4 system layer model [KOEN-00].](image)

2.3.3 Error resilience tools

In addition to the features capable of supporting delivery of multimedia efficiently and flexibly over a variety of transport networks, many techniques have been incorporated into the MPEG-4 in order to make the coded video stream more resilient to channel degradation, when operating in error-prone environments [SCHA-98]. These techniques can be categorised into encoder error resilience tools, and decoder error concealment tools considering their implementation. There are four main error resilience tools introduced in to the MPEG-4 encoder and they are video packet resynchronisation, data partitioning, Reversible Variable Length Codes (RVLC) and header extension codes [TALL-98].
2.3.3.1 Video packet resynchronisation markers

Due to the characteristics of variable length codes, code synchronisation can be lost in the presence of bit errors. Resynchronisation markers, which are uniquely designed code words, can be used to regain synchronisation by placing them in the video bit stream. When the decoder detects an error it can then hunt for a resynchronisation marker, thereby preventing further error propagation. In previous video standards including H.261 and H.263, resynchronisation markers are inserted at the end of every Group Of Blocks (GOBs). A GOB contains a fixed (pre-set) number of consecutive macro blocks. A GOB may contain a different number of bits depending on the motion involved in the different parts of the image. In this approach, error localisation is limited to the size of the GOB and high motion parts of the image are thus highly sensitive to errors. MPEG-4 provides a different approach to resynchronisation: a packet based approach. The packet approach is based on providing periodic resynchronisation markers throughout the bit stream. In other words, the length of the video packets are not based on the number of macroblocks, but instead are based on the number of bits contained in that packet. If the number of bits contained in the current video packet exceeds a predetermined threshold, then a new video packet is created at the start of the next macro-block. Utilisation of this error-resilience tool can involve some small sacrifices in coding efficiency due to the extra overhead added by the synchronisation codes and the video packet headers.

2.3.3.2 Data partitioning

```
<table>
<thead>
<tr>
<th>VP header</th>
<th>M1</th>
<th>T1</th>
<th>M2</th>
<th>T2</th>
<th>M3</th>
<th>T3</th>
<th>M4</th>
<th>T4</th>
<th>M5</th>
<th>T5</th>
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<td></td>
</tr>
</tbody>
</table>

Non-data partition format

<table>
<thead>
<tr>
<th>VP header</th>
<th>M1</th>
<th>M2</th>
<th>M3</th>
<th>M4</th>
<th>M5</th>
<th>T1</th>
<th>T2</th>
<th>T3</th>
<th>T4</th>
<th>T5</th>
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</thead>
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<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Data partition format

M - Motion data & MB type
T - Texture data
- Motion Marker

Figure 2.3: MPEG-4 data partition format.
Resynchronisation markers help the decoder to deduce that the data error is within two resynchronisation markers. Further localisation of error can be achieved with data partitioning. Information data between two consecutive resynchronisation markers are grouped into finer logic units, such that each logic unit contains one group of information for all Macro-Blocks (MBs) in the whole packet. This is in contrast to the non-data-partitioned syntax where each MB contains its own motion and texture data together. Separation of data belonging to each logic unit is carried out by inserting a secondary motion marker between the logic units [TALL-98]. Data is more likely to be lost at the end of the packet, so less important texture data is placed at the end. This also facilitates multi-priority transmission, where more important motion data can be transmitted using highly protected channels. If texture data is lost in the transmission, the correctly received motion data can be used in concealing the errors within the texture part.

2.3.3.3 Reversible variable length codes

Reversible Variable Length Codes (RVLC) provide another error resilience technique for video applications. In this approach, the variable length code words are designed such that they can be read both in the forward as well as the reverse direction [WANGb-00]. If an error is detected during the decoding process of the received video stream, the decoding is immediately interrupted and the next re-synchronisation marker is searched. Once the next resynchronisation marker is located, instead of discarding the data between synchronisation markers, the decoding process is resumed in the reverse direction. This process is illustrated in Figure 2.4. RVLCs can provide more information on the position of the errors, thereby decreasing the amount of data unnecessarily discarded.
2.3.3.4 Header extension

Video frame headers (or Video Object Plane (VOP) headers) contain very important information that the decoder needs to be able to decode the video bit stream for that frame. These include information regarding the spatial dimension of the image, the time stamp associated with the decoding, and the mode in which the current video object is encoded. Without this information, decoding is impossible and the decoder has to discard the rest of the video stream regardless of its correct recovery. The Header Extension Code (HEC), introduced into the MPEG-4 standard, reduces the sensitivity of header information to channel errors by repeating important header information in selected video packets. The repetition is controlled by introducing a 1-bit field called the HEC bit to the video packet header field. If this bit is set to one, the important header information is repeated within that video packet. On one hand, this method allows the decoder to be certain about the header information by cross checking. On the other hand, if the video frame header is corrupted, the decoder can still decode the rest of the data in the video frame using the header information within the video packets [TALL-98]. However, frequent use of HEC reduces the compression efficiency due to the extra overhead.

2.3.3.5 Insertion of intra frame and adaptive intra refresh

Frame to frame error propagation is an inevitable in compression schemes based on predictive coding. Unless careful design strategies are taken to mitigate the effect of temporal error propagation, video communications can totally break down. Periodical insertion of intra-coded frames, which do not make use of any prediction information, is one way to limit temporal error propagation. This clears up any errors that may have been propagating in the video sequence. However, intra coded frames are on average much larger than predictive coded frames, reducing the compression efficiency. Also this approach produces a video bit stream with a highly variable bit rate, making resource allocation and network buffer management difficult.

One technique to mitigate error propagation, while avoiding the insertion of intra coded frames is adaptive intra refresh. This involves the insertion of a number of intra coded MBs in each frame. This provides not only an efficient technique for mitigating the effect of error propagation but also a much smoother output bit rate. A number of different methods can be used in selecting the Intra-MB locations and also the number of MBs to be intra coded in the current frame. Most of these methods use motion information in the MB selection. The experiment carried out in this thesis
uses an Adaptive Intra Refresh (AIR) algorithm developed at CCSR [WORRa-01]. The algorithm implementation is based on the Sum of Absolute Difference (SAD) calculation. SAD is used in the motion vector calculation to measure the differences between the pixels in current and reference frames. A detailed discussion of the algorithm implementation and performance evaluation is given in [WORRa-01].

2.3.4 Decoder error concealment

Error concealment capabilities should be included in decoders so that the severity of artefacts resulting from transmission errors can be minimized. Given the standard MPEG-4 data format, there are three types of information that need to be estimated in a damaged macro block. They are the texture information, for either non-predictive or predictive image blocks, the motion information for predictive blocks and the coding mode. Concealment techniques are categorised into two main categories as temporal error concealment and spatial error concealment.

One approach to spatial error concealment is the prediction of pixels in a damaged block from pixels in adjacent correctly received blocks. Most pixels in correctly received adjacent blocks are too far from the missing samples; therefore, interpolation is usually carried out in the prediction. Instead of interpolating pixel-by-pixel, estimation of the DC coefficient (of DCT) of a damage block can be performed by averaging the DC values of surrounding blocks [GALA-98]. If the motion vector information is received correctly, effective spatial concealment can be achieved by replacing the error block with the corresponding macro-block in the previously decoded frame. This scheme is highly effective for concealment in low motion video sequences, however, for high motion sequences, the pixel interpolation method shows better error concealment [TALL-98]. The methods described above ignore any correctly received DCT coefficients. Correctly received DCT coefficients can in fact be used to improve the accuracy of the estimation. Here the concealment algorithm can be formulated as an unconstrained optimisation problem. The estimation should be such that the recovered block is to be smoothly connected with its neighbouring pixels both in the spatial and temporal domains, subject to the constraint that the DCT on the recovered block would produce the same values for the received coefficients [LUO-95, SHIR-00].

Projection onto convex sets techniques [SUN-95, WANGb-00] have been widely proposed to reconstruct the corrupted blocks. However, as the solution can only be obtained through an iterative process, this method is not suitable for real-time applications.
The techniques described for spatial domain concealment can also be applied in recovering erroneous motion vector information. Simple operations include replacement of the lost motion vector with the corresponding block in the previous frame, use of the average or the median of the MV of adjacent blocks, and re-estimation of the MV using statistical information. Better, but more complex motion information concealment can be conducted using different motion information for different pixel regions in the macro block [LAM-93, CHEN-97].

One way to estimate the mode information is to use statistical information of the coding mode pattern of adjacent blocks and find a most likely mode given the modes of surrounding blocks. A simple widely used approach is to assume that the macro block is coded in the intra-mode, and to use only spatial interpolation in recovering the corresponding blocks.

2.3.5 Multimedia transmission issues in mobile networks

In addition to the encoder error resilience and decoder error concealment techniques, there are other techniques which are designed to improve the received quality of transmitted video over bandwidth limited error prone environments. This section examines some existing efficient transmission methodologies proposed by other researchers. The review is split into five areas:

- Joint source channel coding
- Advanced forward error correction
- Feedback based techniques
- Multiplexing
- Link adaptation

2.3.5.1 Joint source-channel coding

In order to reduce the effect of channel noise on the transmitted bit stream, some form of Forward Error Correction (FEC) is applied for information transmission over error prone channels. However, the appropriate level of FEC coding generally depends upon many parameters such as the type of media, the available bandwidth, and the channel bit error rate. Especially for video applications, the quality distortion seen in a decoded video sequence is determined by the distortion due to channel errors as well as the quantisation distortion at the source encoder. Therefore, a combined source-channel coding approach is necessary in optimal allocation of rates
between source and channel protection, subject to a fixed constraint on overall transmission bandwidth. Much research has been conducted in joint source-channel coding for image and video applications. Studies revealed that lower source rate and extra channel protection is favourable for image and video transmission over low quality channels [MODE-81]. The algorithms, which determine the optimal source-channel coding ratio for a particular source coding algorithm, channel coding scheme and set of channel characteristics have been proposed in [BYST-98, BYST-00, KURC-00]. The communication channel states can often be modelled as a hidden Markov process. Based on this statistical framework, the design of an optimal joint source-channel coding is discussed in [DYCK-99]. The method proposed in [DYCK-99] uses state estimation and minimum mean square error estimation procedure in the determination of optimal source-channel code ratio. The theoretical performance bounds in distortion-rate characteristics based optimal source-channel bit allocation has been derived in [BYST-00].

2.3.5.2 Advanced forward error correction

Advanced forward error correction methods aim to reduce the effect of channel error on video quality by minimising the amount of important information loss during the transmission. The output bit stream from the video encoder is modified or restructured in such a way to achieve maximum perceptual quality at the receiver. These techniques can be categorised into three main areas:

- Unequal error protection
- Unequal power allocation
- Multiple descriptive transmission

Unequal Error Protection (UEP) basically uses prioritised transmission. The encoder output bit stream is separated into several sub-streams according to their importance in perceiving video quality. Then, high priority streams are transmitted using highly protected channels, while low channel protection is used for low priority streams. A common method to split encoded data into separate streams uses the MPEG-4 data partition method [GHAN-93]. This method is simple to implement, yet provides a significant improvement in received video quality. Another obvious way of separating video information into multiple streams is scalable or layered video coding. A layered video encoder generally generates a number of bit streams. The most critical information is contained in the base layer. Other layers contain information that is needed for enhancing the quality of the base layer. Some other techniques used in bit stream splitting are information separation in the transform domain [GHAR-99].
Unequal Power Allocation (UPA) techniques for enhancing video quality have recently been proposed, especially for video communications over CDMA based cellular networks. The operation of unequal power allocation techniques is similar to that of the unequal error protection techniques described above. However, network compatibility is essential for practical implementation. In addition to the constraints imposed by the limited bandwidth or channel throughput, the transmit power also provide a limiting factor in the UPA algorithm design. Several UPA techniques for wireless video applications have been proposed in [EISE-02, KIM-03, ZHAO-02]. All of these techniques are optimised to achieve a target video quality while minimising the transmission power.

Multiple Descriptive Coding (MDC) is intended to achieve quality improvements by exploring the diversity of transmission links. Similar to layered coding, a multiple descriptive code generates multiple bit streams. However, unlike in layered coding, these multiple streams have equal priority. The streams are transmitted over separate transmission channels with equal channel characteristics, in terms of average channel quality. The transmitted streams can individually be decoded at the receiver. When they are combined, improved quality is achieved because of the transmission diversity gain [LEE-02]. For example, assume two streams are transmitted over radio channels with similar channel parameters as shown in Figure 2.5. Even though both channels have similar average channel characteristics, the instantaneous channel qualities experienced are very different due to the time, space and frequency parameters. Therefore, appropriate stream combinations can result in better received quality. MDC is especially favourable in ad-hoc networks.

![Figure 2.5: A typical MDC system with two descriptions.](image)
2.3.5.3 Feedback based techniques

In all of the above error resilience techniques, the encoder and the decoder operate independently of each other. If a feedback channel can be set up, the decoder can inform the encoder about the instantaneous behaviour of the transmission channel and the information regarding the corrupted packets. Thus the encoder can adjust accordingly to suppress or to minimise the effect of channel errors on the video stream. Feedback based techniques can be described under three main areas:

- Retransmission techniques
- Optimal encoder mode selection
- Error tracking

The automatic Repeat request (ARQ) method is a powerful technique, which is commonly used in conventional data transmission to retransmit the error packets when bit errors are not allowed. However, retransmission control as used in data transmission creates unacceptable delay and is not suitable for real time applications. Modifications of ARQ techniques, so that it is to be suitable for real time applications have been proposed in [LIU-97, MATO-98]. A common feature of these modifications is to use, Hybrid ARQ, which is the combination of ARQ and other error resilience techniques. The combination of FEC and ARQ for mobile video communications has been studied in [LIU-97]. These schemes request that the encoder retransmit any incorrectly received packets. Even with one bit error in a packet, the encoder is told to retransmit the entire packet. Therefore, algorithm efficiency is significantly low especially under burst errors. Matoba [MATO-98, MATO-97] has proposed Selective-Repeat ARQ/FEC (SR-ARQ), which is a modification of the FEC/ARQ method. Using the properties of layered coding and considering different QoS requirements of different data sections within a video stream, further modifications to SR-ARQ, called QoS aware Selective Repeat ARQ (QSR-ARQ), are proposed by Wang in [WANGa-00, WANGc-00, ZHEN-99]. Results show that QSR-ARQ improves the data link protocol performance, network throughput and bandwidth efficiency. Other approaches based on combined adaptive QoS control, optimal mode selection and delay-constrained hybrid ARQ, can be found in [DAPE-00, BATR-98].

Another way of achieving performance improvements with the use of feedback channel information is the reference picture selection method. If a reference frame is corrupted due to channel errors, the error propagates over the following video frames until the reference frame is refreshed by intra-coded frame. The error propagation can be mitigated by referencing to a correctly received video frame for encoding of future video frames. This concept is employed in
optimal reference picture selection methods proposed in [COTE-00]. The algorithm takes into account the channel condition and the error concealment method used by the decoder, to optimise video coding mode selection in the compressed bit stream. However, the algorithm should be carefully designed to avoid performance degradation caused by parameter mismatch between the parameters used by the encoder, the parameters associated with the actual channel condition and the decoder error concealment methods.

Error tracking methods also utilize a feedback channel to adapt the system to varying channel conditions. In this method, an error is localised to an image region. Decoder error concealment is employed to make transmission errors less visible, and unacceptable residual errors are compensated by coding the distorted regions in intra mode. Negative ACKnowledgments (NACK) are sent back to the transmitter for image part that could not be decoded successfully. The encoder evaluates the NACKs and makes the intra mode decision on a macro block basis. The reconstruction of spatial-temporal error propagation at the encoder permits for the compensation of errors that have propagated [LIU-97, CHOI-99, STEI-97].

2.3.5.4 Data stream multiplexing

Since video traffic is delay-sensitive, a resource reservation scheme seems to be the right choice in guaranteeing quality requirements for real-time video communications. However, due to the unpredictably bursty nature of Variable Bit Rate (VBR) video, the resource reservation task is complicated. If resources are reserved according to peak rates, the network is under-utilised most of the time. VBR not only causes low transmission efficiency, but also causes difficulties for QoS provisioning in multi-user communication systems.

Many research activities concentrate on minimising the effect of VBR on resource allocation. These are based on different multiplexing schemes and can be categorized in to three main classes [WANG-98]. The bursty traffic can be smoothed to become near constant bit rate traffic by using a buffer for each data stream. This method is called temporal statistical multiplexing and is adopted in some real-time MPEG encoders. Although, this is very easy to implement, this can cause delay variation leading to inconsistent video quality. A second method is called pre-fetching. It allows users to get data before the playback start time. This can be used for non real time streaming applications, utilizing the high link capacity at non-peak times. Spatial statistical multiplexing, the third method, allows sharing of bandwidth between several streams to achieve an aggregated constant bit rate like throughput. In this way, the streams running at peak rate
borrow the bandwidth from streams running at low rates. Delay variation is not observed with this method, but the arrangement of traffic is critical and difficult.

A scene-based multiplexing method proposed in [WANG-98] belongs to the spatial statistical multiplexing methods described above. Several video bit streams are multiplexed in to a single bit stream resulting in better bandwidth utilization without introducing delay variation. Scene information is used to decide the bandwidth requirement for the aggregated video streams. The basic assumption used in this system is that a significant video traffic bandwidth change could result from a visual scene change. Video trace synchronization is used to avoid several I frames being transmitted simultaneously. Simulation results show that scene based multiplexing achieves improve bandwidth utilization when transmitting MPEG1 and MPEG2 video streams.

Another form of spatial statistical multiplexing method is described in [BAHL-98], namely inter-frame statistical multiplexing. This method uses region segmentation or the MPEG-4 video object concept to segment the video frame in to regions or video objects of differing importance. The peak bit rate for the most important region is reserved at connection establishment time. It is likely that most of the time the compressor will produce bits far below this peak value. The bandwidth left over after the important region has been transmitted, is used to transmit the remaining regions and retransmission if required. Priority was given in the order of the lowest frequency sub-band of the main regions, followed by the lowest frequency sub band of the remaining regions, followed by the subsequent higher frequency sub bands. Any bits left over after the reserved bandwidth was used up, were transmitted using any available unreserved bandwidth. Experimental results illustrate the efficient bandwidth allocation. As the most important regions always reach the receiver, transmitted image frames can be displayed in any case.

Object prioritisation based on statistical multiplexing method is described in [CELL-99]. Using the object-oriented features of MPEG-4, the video frame is divided into separate objects. These objects are transmitted independently assigning higher priorities to the most important objects and are multiplexed at the receiver to get the combined frame. The use of object-based prioritisation schemes has been shown to give a noticeable perceptual quality improvement in received video signals over bandwidth limited and noisy channels.
2.3.5.5 Link adaptation

Link adaptation is employed for mitigating the effect of time varying radio channel conditions on the received media quality. This involves the modification of source and network parameters in response to the instantaneous channel and interference conditions. The concept of source and network parameter adaptation for system throughput maximisation has been considered for data communication in [NAND-00, BALA-99, MEHT-00, GUTI-00, ZHAO-01]. Ideally, the design of link adaptation algorithms for multimedia services should optimise the delivery of acceptable video quality, facilitating good quality video services with the minimum possible demand on network resources. Application of link adaptation techniques for video communication has been introduced by Hanzo and his associates in their series of paper publications [HANZa-00, HANZb-00, CHER-00]. In these schemes, adaptive modulation is considered. The appropriate modulation mode for the transmission is selected from set of different modulation schemes according to the instantaneous quality of the transmission channel. Adaptive schemes have been shown to provide better received video quality compared to the non-adaptive schemes for video transmission over narrowband and wideband channels.

2.3.6 Video performance assessment

Figure 2.6: Sample video frames from selected sequences.

‘Carphone’  ‘Suzie’

‘Foreman’

Figure 2.6: Sample video frames from selected sequences.
Three different ITU video sequences: namely "Suzie", "Carphone" and "Foreman" are selected as the test sequences for the evaluation of quality enhancement algorithms developed in this thesis. Sample video frames of each of these sequences are shown in Figure 2.6. These video sequences characterise typical real-time video telephony scenarios. The "Suzie" sequence, which features a woman in a telephone conversation, is a relatively low motion head-and-shoulder sequence with a stationary background. On the other hand, "Carphone" and "Foreman" sequences feature high activity scenes. The "Foreman" sequence is particularly demanding and is highly sensitive to errors, as the camera is non-stationary and pans around the scene. To produce longer sequences (30 seconds), these sequences are repeated, with even numbered sequences played in reverse order to guarantee a smooth transition. The selected sequences span from low motion activities to very-high motion activities. Therefore, the average over the selected sequences will give an assessment figure, which can be applicable to the majority of video scenes encountered in a typical video conversation.

For all the work described in this thesis, the performances are evaluated based on objective quality measurement. Subjective quality measurement may be more suitable for assessing perceptual video quality. However, subjective tests require large number of users to evaluate large number of decoded video sequences in order to achieve meaningful average performance figure. As this is clearly not a convenient option, the objective quality measurements are used.

Peak Signal to Noise Ratio (PSNR), which is a standard measurement indicating the amount of original signal remaining compared to noise introduced, is used as the objective quality measure. This is taken on a frame-by-frame basis and is expressed as a function of the Root Mean Squared Error (RMSE) as shown in Equation 2.1.

\[
RMSE = \sqrt{\frac{1}{M \times N} \sum_{i=1}^{M} \sum_{j=1}^{N} ([f(i,j)] - [\hat{f}(i,j)])^2}
\]

\[
PSNR = 20 \log_{10} \left( \frac{2^n - 1}{RMSE} \right)
\]

where, \(f(i,j)\) and \(\hat{f}(i,j)\) are the luminance value of pixels in source and reconstructed images respectively. Each image contains \(N\) by \(M\) pixels. \(n\) is number of bits used in the formatting. Thus, \(2^n - 1\) indicates the highest possible luminance value of a pixel. \(PSNR\) in Equation 2.2 is expressed as the ratio between the highest possible luminance value of a pixel and the average of the differences between the luminance values of corresponding pixels position in the two images.
For high quality sequences, where the relative difference seen compared to the original is low, this quality measure provides a high PSNR value. A PSNR value of 20 dB or above is considered as acceptable quality. This threshold can vary from sequence to sequence depending on the amount of motion and activity involved and also it is subjected to user preferences. As shown in Figure 2.7, 20 dB PSNR suggests a reasonable perceptual quality video for selected test sequences.

Figure 2.7: Sample decoded video frames with 20 dB PSNR. Coded at 64 kbps with 10fps.

Unless otherwise stated, each experiment describes in following chapters, was repeated 20 times to capture the effect of the bursty nature of the channel on the received video quality.

2.4 Third generation communication systems

The goal of wireless communication is to allow the user access to the required services at any time without regard to location or mobility. The user service requirement has been rapidly transformed from conventional telephony to multimedia services during the last few years, demanding higher capacity, higher data rates, and enhanced service flexibility from wireless
communication networks. Even though second generation radio access technologies, such as GSM and IS-95 have been successful for the delivery of telephony and low bit rate data services at rates up to 9.6 kbps, they are clearly not capable of providing high bit rate multimedia services including real-time video and audio transmission. In addition, network access was supported over circuit switched connection in second-generation wireless systems. This however, is inefficient for supporting service flexibility especially for bursty traffic, which would result in future multimedia services. To successfully meet the challenges set by multimedia service requirements and user demands, the International Telecommunication Union Radio communications (ITU-R) sector has established a set of service requirements for third generation radio access system [ITU]. They are

- Full area coverage and high data rates. Bit rates up to 144kbps are expected in a satellite radio and rural environment, up to 384kbps in urban radio environments and 2048kbps in indoor and low-range pico cell environments [3GPP TS 22.105].
- Symmetrical and asymmetrical data transmission. Symmetric transmission expects similar traffic loads in the uplink (from mobile terminal to the base station) and the downlink (from the base station to the mobile terminal). The network traffic generated from future multimedia applications is expected to be asymmetric in the uplink and downlink. These applications required asymmetric data transmission.
- Inclusion of circuit switched and packet switched services. This would facilitate efficient resource management for real-time delay critical as well as non-delay critical services.
- Good voice quality, comparable to wire-line quality.
- Greater capacity and improved spectrum efficiency
- Multiple simultaneous services to end users providing rich multimedia sessions.
- Global (international) roaming to provide discontinuous services to end users regardless of their location and mobility.
- The seamless incorporation of second-generation cellular systems in order to provide smoother transient.

ITU-R has elaborated on a framework for global third generation standard by recognising a limited number of radio access technologies. They are Universal Mobile Telecommunication System (UMTS), Enhanced Data rates for GSM Evolution (EDGE), and CDMA2000. UMTS is based on Wideband CDMA technology and is to be employed in Europe and Asia, using the frequency band around 2 GHz. EDGE is based on TDMA technology and uses the same air interface as the successful second generation mobile system GSM. General Packet Radio Service (GPRS) and High Speed Circuit Switched Data (HSCSD) are introduced by the Phase 2+ of the GSM standardisation process and support enhanced services with data rates up to 144 kbps in the packet switched and circuit switched domains respectively. GPRS has also been accepted by the
Telecommunication Industry Association (TIA) as the packet data standard for TDMA/136 systems. EDGE, which is the evolution of GPRS and HSCSD, provides third generation services up to 500kbps within GSM carrier spacing of 200 kHz [NILSa-99]. CDMA2000 is based on multi-carrier CDMA technology and it provides the upgraded solution for existing IS-95 operators, mainly in North America. The migration paths from existing second-generation standards to third generation standards are shown in Figure 2.8. The operation of the CDMA2000 system is not investigated in this thesis, thus no further detail of CDMA2000 system is presented.

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**Figure 2.8: Evolution toward global third-generation standards.**

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**Figure 2.9: Third generation wireless communication system [taken from NILSb-99].**
EDGE and UMTS are the most widely accepted third generation radio access technologies. It is being standardised by the 3rd Generation Partnership Project (3GPP). Even though EDGE and UMTS are based on two different multiple access technologies, both systems share the same core network. As shown in Figure 2.9, the evolved GSM core network will serve for a common GSM/UMTS core network that supports GSM/GPRS/EDGE and UMTS access.

The system architecture of the third generation wireless communication system is shown in Figure 2.10. The GSM/UMTS core network consists of four main components. They are the Serving GPRS Support Node (SGSN), the Gateway GPRS Support Node (GGSN), the Mobile Switching Centre/Visitor Location Register (MSC/VLR) and the Home Location Register (HLR). SGSN and GGSN handle the packet switched services. SGSN is connected to the Radio Access Network (RAN) and it is responsible for the delivery and management of packet switched services to mobile users. SGSN functions include the monitoring and tracking of user location, access control and provision of security. The SGSN is connected to the external IP network through GGSN. Between SGSN and GGSN IP is used as the transport protocol. Circuit switched services are handled by MSC/VLR and it is functions are similar to that of SGSN. SGSN and GGSN interwork with the MSC/VLR and the HLR through their specified interfaces.

![Diagram of the core network architecture of third generation wireless communication system.](image)

*Figure 2.10: The core network architecture of third generation wireless communication system.*
2.4.1 Third generation mobile services and applications

Future communications systems are intended to support a wide range of applications that possess different quality of services. Compared to the current single service network, future multimedia applications require multiple service operation in integrated network environments (see Figure 2.11). Application scenarios predicted in future systems include interactive multiparty conferencing, three dimensional video gaming, virtual office/home environment, augmented reality scenes and specialised multimedia business applications such as telemedicine, remote security surveillance and intelligent transportation systems. The characteristics of many of these applications span a wide spectrum and are currently undefined. Therefore, it is unrealistic to optimise the third generation systems only for a set of existing applications. 3GPP has established a new service architecture to enable the support of a wide range of application scenarios. The service architecture and requirement are specified in a series of Technical Specification documents [3GPP TS 22.011, 3GPP TS 22.101, 3GPP TS 23.110, 3GPP TS 22.105, and 3GPP TS 23.107]. The technical specifications given in 3GPP Release 4 are used in the work carried out in this thesis.

Figure 2.11: Current and future cellular services and application scenarios [taken from NILSb-99]

Applications and services have been categorised into different classes, depending on their operations. Four traffic classes have been identified. They are conversational, streaming, interactive and background class [3GPP TS 22.105]. The main distinguishing factors between these classes are the delay sensitivity and error tolerance. The conversational class is meant for delay critical traffic, while the background class is the most delay-insensitive. The required error
tolerance depends on the offered service characteristics. The fundamental characteristics of these traffic classes are summarised in Table 2.1. An example of different services to be offered within these traffic classes are shown in Figure 2.12.

<table>
<thead>
<tr>
<th>Traffic class</th>
<th>Conversational class</th>
<th>Streaming class</th>
<th>Interactive class</th>
<th>Background class</th>
</tr>
</thead>
<tbody>
<tr>
<td>Traffic characteristics</td>
<td>-preserve time relation between information entities of the stream</td>
<td>-preserve time relation between information entities of the stream</td>
<td>-request response pattern</td>
<td>-destination is not expecting the data within a certain time</td>
</tr>
<tr>
<td></td>
<td>-conversational pattern (stringent and low delay)</td>
<td></td>
<td>-preserve data integrity</td>
<td>-preserve data integrity</td>
</tr>
</tbody>
</table>

Table 2.1: The fundamental characteristics of different traffic classes [HOLM-01]

![Table 2.1: The fundamental characteristics of different traffic classes](image)

Figure 2.12: Summary of applications in terms of QoS requirements [taken from 3GPP TS 22.105]

The above applications and services can be divided into two main categories as real-time and non-real time services. Real time services include two-way conversational services such as interactive video conferencing, conventional telephony services and interactive video games. Streaming/retrieval services where users are allowed to access multimedia information servers and messaging services such as fax and e-mail, are considered under the non-real-time services. QoS requirements in terms of bit error rate and transfer delay for applications under these two categories in different system deployment environments are specified in [3GPP TS 22.105] and listed in Table 2.2.
Table 2.2: 3GPP QoS Requirement [taken from 3GPP TS 22.105]

<table>
<thead>
<tr>
<th>Operating environment</th>
<th>Real Time (Constant Delay)</th>
<th>Non Real Time (Variable Delay)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>BER/Max Transfer Delay</td>
<td>BER/Max Transfer Delay</td>
</tr>
<tr>
<td>Satellite</td>
<td>Max Transfer Delay less than 400 ms BER $10^{-3} - 10^{-7}$ (Note 1)</td>
<td>Max Transfer Delay 1200 ms or more BER $10^{-6}$ to $10^{-8}$ (Note 2)</td>
</tr>
<tr>
<td>(Terminal relative speed to ground up to 1000 km/h for plane)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Rural outdoor</td>
<td>Max Transfer Delay 20 - 300 ms BER $10^{-3} - 10^{-7}$ (Note 1)</td>
<td>Max Transfer Delay 150 ms or more BER $10^{-6}$ to $10^{-8}$ (Note 2)</td>
</tr>
<tr>
<td>(Terminal relative speed to ground up to 500 km/h) (Note 3)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Urban/Suburban outdoor</td>
<td>Max Transfer Delay 20 - 300 ms BER $10^{-3} - 10^{-7}$ (Note 1)</td>
<td>Max Transfer Delay 150 ms or more BER $10^{-6}$ to $10^{-8}$ (Note 2)</td>
</tr>
<tr>
<td>(Terminal relative speed to ground up to 120 km/h)</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Indoor/Low range outdoor</td>
<td>Max Transfer Delay 20 - 300 ms BER $10^{-3} - 10^{-7}$ (Note 1)</td>
<td>Max Transfer Delay 150 ms or more BER $10^{-6}$ to $10^{-8}$ (Note 2)</td>
</tr>
<tr>
<td>(Terminal relative speed to ground up to 10 km/h)</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

NOTE 1: There is likely to be a compromise between BER and delay.
NOTE 2: The Max Transfer Delay should be here regarded as the target value for 95% of the data.
NOTE 3: The value of 500 km/h as the maximum speed to be supported in the rural outdoor environment was selected in order to provide service on high speed vehicles (e.g. trains). This is not meant to be the typical value for this environment (250 km/h is more typical).

2.4.2 Quality of Service (QoS) management

2.4.2.1 QoS parameters

The applications, which belong to different traffic classes, have different demands on network performance in terms of bandwidth, delay and reliability, making resource allocation and management a difficult task. A new QoS management architecture is introduced by 3GPP in order to enable efficient resource allocation in UMTS systems. Compared to the Internet and the second-generation mobile networks, UMTS provides an improved and important feature, namely the ability of an application to choose between different levels of quality. This is carried out through the negotiation of a well defined set of QoS parameters. The main QoS parameters include:

- Traffic class
- Maximum bit rate
- Guaranteed bit rate
- Transfer delay
- Service Data Unit (SDU) error ratio
- Residual bit error ratio
- Maximum SDU size
- Delivery of erroneous SDU,
- Delivery order
- Traffic handling priority
- Allocation/retention priority

Traffic class is included as an attribute, allowing the network to make assumptions about the traffic source and to optimise the transport for that particular traffic type. Maximum bit rate is used to make code reservations in the radio interface and to limit the delivered bit-rate to applications or external networks, while guaranteed bit rate is defined to facilitate admission control and resource allocation. Maximum and guaranteed bit rates are into 10 distinctive rates, ranging from 8 to 2,048kbit/s, varying by a factor of 2 at each step [3GPP TS 24.008].

The transfer delay is defined by the time between a request and its delivery at the other Service Access Point (SAP), and is intended to specify the delay tolerated by the application. It allows a network to set the transport format and ARQ parameters in case of re-transmission. This is divided into three classes considering the delay range: 10 – 150 ms in 10ms increments, 200 – 950 ms in 50 ms increments and 1000- 4100ms in 100ms increments [3GPP TS 24.008].

The Service Data Unit (SDU) error ratio defined as the fraction of SDUs lost or detected as erroneous, while residual bit error ratio is defined as the undetected bit error ratio in the delivered SDUs. Together these parameters offer a mechanism to configure the protocols, algorithms and error detection schemes. The range for these attributes varies from $1 \times 10^{-1}$ to $6 \times 10^{-8}$. Other attributes, namely maximum SDU size, delivery order, delivery of erroneous SDU, traffic handling priority, and allocation/retention priority are defined in order to facilitate efficient admission control, QoS management, policing and charging for various services.

For each type of carrier there is a vast number of different attributes to be set. It is at the discretion of the operator to decide which of these the user will be able to manipulate. However, the operator will most likely provide a discrete set of traffic classes, with most attributes pre-set and others up to the user to decide.
2.4.2.2 QoS management architecture

The QoS architecture introduced in UMTS [3GPP TS 23.107] can be viewed as a layered architecture as illustrated in Figure 2.13. End-to-end QoS for an application is viewed as a series of chained services operating at different levels of the transmission networks. UMTS allows an application to negotiate bearer characteristics so that the best possible quality can be achieved. Renegotiation of bearer characteristics via a parameter renegotiation procedure is also allowed for active connections. A renegotiation command may be initiated by the application or the network operator. This enables efficient adaptive resource management and optimal service quality in a multi-user multi-service environment.

![Figure 2.13: UMTS bearer service architecture [3GPP TS 23.107].](image)

Application generated traffic should pass different networks on its way from the source to the destination. The end-to-end service on the application level is therefore realised over several different bearer services, namely a Terminal Equipment (TE) to Mobile Terminal (MT) local bearer service, a UMTS bearer service, and an external bearer service, such as the Internet.

The UMTS bearer service provides the quality of service in UMTS. It consists of two parts, the Radio Access Bearer (RAB) and the Core Network Bearer Service (CN-BS). The RAB service is
based on the characteristics of the radio interface and is maintained for moving MTs. The role of the CN-BS is to control and utilise quality within the core network.

In order to realize a certain quality of service, bearer services with clearly defined characteristics and functionality should be set up from the source to the destination of a service. These aspects include the control signalling, user plane transport and QoS management functions. Within the UMTS framework [3GPP TS 23.107], several QoS management functions are specified in order to establish, modify and maintain a UMTS bearer service with a specific QoS. The allocation of these functions to UMTS entities indicates the requirement for the specific entity to enforce the QoS commitments negotiated for the UMTS bearer service. The specific realisation of these functions is implementation dependent and has only to maintain the specific QoS characteristics. QoS management functions are divided into two main categories: control plane and user plane QoS management.

Control plane QoS management consists of a translator, a service manager and an admission/capability controller. Translator converts the QoS attribute between a UMTS bearer service and the external network and maintains QoS parameter mapping between two networks. The service manager is responsible for establishing, modifying and maintaining the service by providing all user plane QoS management functions with the relevant attributes. This may also perform an attribute translation to request lower layer services and may interrogate other control functions to receive permission for service provision. The admission/capability controller maintains a database of all available resources of a network entity and resources allocated to UMTS bearer services. The function checks whether the required resources can be provided for each UMTS bearer service request or modification. It also checks for the capability of the network entity, i.e. whether the specific service is implemented and not blocked for administrative reasons [3GPP TS 23.107].

The user plane QoS management functions maintain the signalling and user data traffic within certain limits, defined by specific QoS attributes. They include a parameter converter, a resource management unit, a classification unit and a traffic-coordinating unit. Within the UMTS network the parameter converter is responsible for converting service attributes between different bearers in order to maintain the intended QoS at the transfer. The resource manager distributes the available resources between services according to the required QoS. Examples of resource management functions at the radio interface include scheduling, bandwidth management and power control. The classification function is only used within the concept of multiple UMTS
bearer services establishment. This assigns incoming data units to the established services of a 
MT according to the related QoS attributes, saving system resources and speeding up the service 
performance. The appropriate UMTS bearer service is derived from the data unit header or from 
traffic characteristics of the data. Traffic conditioning is performed by policing or by traffic 
shaping. The policing function compares the data unit traffic with the related QoS attributes. The 
traffic shaper forms the data unit traffic according to the QoS of the service [3GPP TS 23.107].

2.4.2.3 End-to-end IP QoS management functions

For the backbone IP network, the end-to-end QoS is provided by a local mechanism in the UE, the 
UMTS Bearer Service (UMTS-BS), and the QoS mechanism of the IP network. To control the 
external IP Bearer Service (IP-BS), the IP BS manager is used. This manages the IP-BS according 
to standard IP mechanisms such as Diffserv, RSVP, and Intserv. The translation/mapping function 
provides inter-working between the IP-BS and UMTS-BS. The parameters used within the 
UMTS-BS are mapped to those used within the IP-BS. It is possible for the IP BS manager to be 
located both in the UE and the gateway node, or the gateway node only [3GPP TS 23.207].

Application level IP-based QoS is mapped to the UMTS QoS by a network element, such as the 
3G-GGSN. Within the UMTS network, an element called the Packet Data Protocol (PDP) context 
is used to negotiate and establish the QoS profile. The scope of the PDP context is from the UE to 
the GGSN, as shown in Figure 2.14.

![Figure 2.14: Network Architecture for QoS Conceptual Models [3GPP TS 23.207].](image)

The PDP context specified for UMTS Release 2000 is defined using the information sets held in 
the UE, SGSN, and GGSN that are used to specify the connection between one subscriber and the 
network. And it identifies an application, a PDP type and one QoS profile. More PDP contexts (up
to 16) with different QoS parameters can share the same PDP address. A Traffic Flow Template (TFT) is used to distinguish between them [3GPP TS 23.060]. The PDP context activation procedure is used to activate the first PDP context for a given PDP address and Access Point Name (APN), where all additional contexts associated with the same PDP address and APN are activated with the secondary PDP context activation procedure (see Figure 2.15).

The PDP context can be activated and released by either the UE or the Network. If a QoS requirement is beyond the capabilities of a communication network, the network negotiates the QoS profile as close as possible to the requested QoS profile. The Mobile Terminal can either accept the negotiated QoS profile, or deactivate the PDP context. After a successful PDP context deactivation, the associated Network layer Service Access Point Identifier (NSAPI) and Transaction Identifier (TI) values are released and can be reassigned to another PDP context [3GPP-24.007, 3GPP-24.008].

![Figure 2.15: Illustration of first and second PDP contexts [KARA-00].](image)

The PDP context modification procedure, which is activated by the network or by the UE, is used to change the QoS parameters, the Radio priority level or the TFT. These are negotiated during the PDP context activation procedure, the secondary PDP context activation procedure or at previously performed PDP context modification procedures.

Figure 16 shows the format of the Quality of Service information element, which specifies the QoS parameters for a PDP context. The explanation of these QoS attributes is given in Section 2.4.2.1 above. It indicates QoS parameter values in both the external network and in the radio access network and facilitates QoS parameter mapping from one network to another.
### Quality of Service IEI

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quality of service IEI</td>
<td></td>
</tr>
<tr>
<td>Length of quality of service IE</td>
<td></td>
</tr>
<tr>
<td>Delay class</td>
<td>0</td>
</tr>
<tr>
<td>Reliability class</td>
<td></td>
</tr>
<tr>
<td>Precedence class</td>
<td></td>
</tr>
<tr>
<td>Traffic Class</td>
<td></td>
</tr>
<tr>
<td>Mean throughput</td>
<td>0</td>
</tr>
<tr>
<td>Delivery order</td>
<td></td>
</tr>
<tr>
<td>Delivery of erroneous SDU</td>
<td></td>
</tr>
<tr>
<td>Maximum SDU size</td>
<td></td>
</tr>
<tr>
<td>Maximum bit rate for uplink</td>
<td></td>
</tr>
<tr>
<td>Maximum bit rate for downlink</td>
<td></td>
</tr>
<tr>
<td>Residual BER SDU error ratio</td>
<td></td>
</tr>
<tr>
<td>Transfer delay</td>
<td></td>
</tr>
<tr>
<td>Guaranteed bit rate for uplink</td>
<td></td>
</tr>
<tr>
<td>Guaranteed bit rate for downlink</td>
<td></td>
</tr>
</tbody>
</table>

#### 2.4.3 UMTS Terrestrial Radio Access Network (UTRAN)

![UTRAN Diagram](image)

Figure 2.16: Quality of Service information element [3GPP-TS 24.008].

Figure 2.17: Systems components in a UMTS [taken from NII.Sa-99]

The system components of UTRAN are shown in Figure 2.17. Functionally, the network elements are grouped into the Radio Network Subsystem (RNS), the Core Network (CN), and the User
Chapter 2. Wireless Multimedia Communications

Equipment (UE). UTRAN consists of a set of RNS’s connected to the core network through the Iu interface. The interface between the UE and the RNS is named Uu. An RNS contains a Radio Network Controller (RNC) and one or more Node Bs. The RNS handles all radio related functionality in its allocated region. A Node B is connected to a RNC through the Iub interface, and communication between RNS’s is conducted through the Iur interface. One or more cells is allocated to each Node B.

As explained in Section 2.4, the protocol within the CN is adopted from the evolution of GPRS protocol design. However, both the UE and UTRAN feature completely new protocol designs, which are based on the new WCDMA radio technology. WCDMA air interfaces have two versions defined for operation in Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD) modes. Only the FDD operation is investigated in this thesis. The modulation chip rate for WCDMA is 3.8 Mega chips per second. The specified pulse-shaping roll off factor is 0.22. This leads to a carrier bandwidth of approximately 5 MHz. The nominal channel spacing is 5 MHz, However, this can be adjusted approximately between 4.4 and 5 MHz, to optimise performance depending on interference between carriers in a particular operating environment. The FDD version is designed to operate in either of the following frequency bands [3GPP TS 25.101].

- 1920-1980 MHz for uplink and 2110-2170 MHz for downlink
- 1850-1950 MHz for uplink and 1930-1990 MHz for downlink.

All radio channels are code division multiplexed and are transmitted over the same (entire) frequency band.

<table>
<thead>
<tr>
<th>Table 2.3: WCDMA air interface parameters for FDD mode operation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Operating frequency band</td>
</tr>
<tr>
<td>Duplexing mode</td>
</tr>
<tr>
<td>Chip rate</td>
</tr>
<tr>
<td>Pulse-shaping roll-off factor</td>
</tr>
<tr>
<td>Carrier bandwidth</td>
</tr>
</tbody>
</table>

WCDMA supports highly variable user data rates with the use of variable spreading factors; facilitating the bandwidth on demand concept. Transmission data rates of up to 384 kbps are supported in wide area coverage, and 2 Mbps in local area coverage. The radio frame length is fixed at 10 ms. The number of information bits or symbols transmitted in a radio frame may vary,
corresponding to the spreading factor used for the transmission, while the number of chips in a radio frame is fixed at 38,400 [3GPP TS 25.101].

2.4.3.1 Radio interface protocol

The radio interface protocol architecture, which is visible in the UTRAN and the User Equipment (UE), is shown in Figure 2.18. Layer 1 comprises the WCDMA physical layer. Layer 2, which is the data link layer, is further split into Medium Access Control (MAC), Radio Link Control (RLC), Packet Data Convergence Protocol (PDCP), and Broadcast Multicast Control (BMC). The PDCP exists mainly to adapt packet-switched connections to the radio environment by compressing headers with negotiable algorithms. Adaptation of broadcast and multicast services to the radio interface is handled by BMC. For circuit-switched connections, user plane radio bearers are directly connected to the RLC. Every radio bearer should be connected to one unique instance of the RLC.

The Radio Resource Control (RRC) is the principal component of the network layer -Layer 3. This comprises functions such as broadcasting of system information, radio resource handling, handover management, admission control and provision of requested QoS for a given application. Unlike the traditional layered protocol architecture, where protocol layer interaction is only allowed between adjacent layers, RRC interfaces with all other protocols, providing fast local inter-layer controls. These interfaces allow the RRC to configure characteristics of the lower layer protocol entities including parameters for the physical, transport, and logical channels [HOLM-01]. Furthermore the same control interfaces are used by the RRC layer, to control the

![Figure 2.18: Radio-interface protocol architecture.](image-url)
measurements performed by the lower layers and by the lower layers, to report measurement results and errors to the RRC.

Figure 2.19: Interactions between RRC and lower layers [3GPP TS 25.301].

UTRAN supports both circuit switched and packet switched connections. In order to transmit an application’s data between UE and the end system, QoS enabled bearers have to be established between the UE and the Media GateWay (MGW). Figure 2.20 shows the User Plane protocol stack used for data transmission over packet switched connection in Release 4.

Figure 2.20: User Plane UMTS protocol stack for packet switched connection [3GPP 23.060].


2.4.3.2 Channel structure

Channels are used as a mean of interfacing the L2 and L1 sub-layers. Between the RLC/MAC layer and the network layer, logical channels are used. Between the RLC/MAC and the PHY layers, the transport channels are used, and below the PHY layer is the physical channel (see Figure 2.18).

Generally, logical channels can be divided into control and traffic channels. The paging control channel and the broadcast control channel are for the downlink only. The common control channel is a bi-directional channel shared by all UE's, while the common transport channel is a downlink only shared channel. Dedicated control channels and dedicated transport channels are unique for each UE.

Transport channels are used to transfer the data generated at a higher layer to the physical layer, where it gets transmitted over the air interface. The transport channels are described by a set of transport channel parameters, which are designed to characterise the data transfer over the radio interface. Each transport channel is accompanied by the Transport Format Indicator (TFI), which describes the format of data to be expected from the higher layer at each time interval. The physical layer combines the TFI from multiple transport channels to form a Transport Format Combination Indicator (TFCI). This facilitates the combination of several Transport channels, called Composite Transport Channel, at the physical layer and their correct recovery at the receiver [HOLM –01].

![Figure 2.21: Transport channel mapping.](image)
In UTRA, two types of transport channels, namely dedicated channels and common channels, exist. As the name suggests, the main difference between them is that a common channel has its resources divided between all or a group of users in a cell, while a dedicated channel reserves resources for a single user.

Transmission Time Interval (TTI) defines the data arrival period from higher layers to the Physical layer. TTI size has been defined to be 10, 20, 40, and 80 ms. Selection of TTI size depends on the traffic characteristics. The amount of data arrived in each TTI can vary in size, and is indicated in the Transport Format Indicator (TFI). In the case of transport channel multiplexing, TTIs for different transport channels are time aligned as shown in Figure 2.22.

![Figure 2.22: Transmission Time Intervals (TTIs) in transport channel multiplexing [HOLM-01].](image)

The physical channels are defined by a specific set of radio interface parameters such as scrambling code, spreading code, carrier frequency, and transmission power step. The channels are used to convey the actual data through the wireless link. The most important control information in a cell is carried by the Primary Common Control Physical CHannel (PCCPCH) and Secondary Common Control Physical CHannel (SCCPCH). The difference between these two is that the PCCPCH is always broadcasted over the whole cell in a well-defined format, while the SCCPCH can be more flexible in terms of transmission diversity and format. In the uplink, the Physical Random Access Channel (PRACH) and Physical Common Packet CHannel (PCPCH) are data channels shared by many users. The slotted ALOHA approach is used to grant user access in PRACH [QIU-01]. A number of small preambles precede the actual data, serving as power control and collision detection. The Physical Downlink Shared CHannel (PDSCH) is
shared by many users in downlink transmission. One PDSCH is allocated to a single UE within a radio frame, but if multiple PDSCHs exist they can be allocated to different UEs arbitrarily: one to many or many to one. The Dedicated Physical Data CHannel (DPDCH) and Dedicated Physical Control CHannel (DPCCH) together realize the Dedicated CHannel (DCH), which is dedicated to a single user [HOLMA-01].

2.4.3.3 Modes of connection

Figure 2.23 shows the possible modes of realizing the connections of the radio bearers at each layer. PDCP, RLC and MAC modes must be combined with physical layer parameters in a way that satisfies different QoS demands on the radio bearers. However, the exact parameter setting is a choice of the implementer of the UMTS system and the network operator.

The radio bearer can be viewed as either Packet Switched (PS) or Circuit-Switched (CS). A PS connection passes the PDCP, where header compression can be applied or not. The RLC offers three modes of data transfer. The transparent mode transmits higher layer Payload Data Units (PDUs) without adding any protocol information and it is recommended for real-time conversational applications. The un-acknowledged mode will not guarantee the delivery to the
peer entity, but offers other services such as detection of erroneous data. The acknowledged mode guarantees the delivery through the use of Automatic Repeat reQuest (ARQ) [3GPP TS 25.322].

The MAC layer can be operated in dedicated, shared or broadcast mode. Dedicated mode is responsible for handling dedicated channels allocated to a UE in connected mode, while shared mode takes the responsibility of handling shared channels. The broadcast channels are transmitted using broadcast mode. The physical layer follows the MAC in choosing a dedicated or shared physical channel [3GPP TS 25.321].

In UTRA spreading is based on the Orthogonal Variable Spreading Factor (OVSF) technique. The properties of spreading codes and code generation are discussed in Section 3.3.3. Quadrature Phase Shift Keying (QPSK) modulation is used for downlink transmission. Both convolutional and turbo coding are supported for channel protection. The maximum possible transmission rate in downlink is 5760 kbps. It is provided by three parallel codes with a spreading factor of 4. With 1/2 rate channel coding, this could accommodate up to 2.3 Mbps user data. However, the practical maximum user data rate is subject to the amount of interference present in the system and the quality requirement of the application.

2.4.4 GSM/EDGE Radio Access Network (GERAN)

![GERAN system architecture](image-url)

Figure 2.24: GERAN system architecture.
The system component of GERAN is shown in Figure 2.24. GERAN is connected to the core network through the Iu and A interfaces. Services based on second-generation systems are supported by the 2G SGSN and the 2G MSC. The 3G SGSN and 3G MSC take responsibilities for support of third generation mobile services. Each SGSN is in charge of several Base Station Subsystems (BSSs). A BSS contains a Base Station Controller (BSC) and one or more Base Transceiver Stations (BTSs). The BSC monitors and controls the BTSs in its BSS. MTs are connected to BTS via the Um air interface [GRAN-99].

GERAN uses the same air interface specified in GSM. Multiplex access is based on the combination of TDMA and FDMA techniques [STUC-03]. Frequency bands have been reserved around the 900 MHz for GSM 900 and around 1800 MHz for GSM 1800 to be used in Europe. Allocated frequency bands are

- 890-915 MHz uplink and 935-960 MHz downlink
- 1710-1785 MHz uplink and 1805-1880 MHz downlink

These bands are divided by frequency into a number of carriers, each spaced 200 kHz apart. One or more carrier frequencies are assigned to each base-station. The selection of carrier frequencies for each base station and the carrier reuse distance is decided by the cell planning procedure. Optionally, frequency hopping capabilities can be used. And the advantage is that the quality on all communication links is averaged through the interferer’s diversity. Each of the carrier frequencies is divided into eight time slots. The communication channel is described by its time slot number and a carrier frequency. One time slot is 577 μs long. Within a time slot, data is transmitted in bursts.

GERAN introduces a set of control and traffic channels to be used in packet switched connections. The Packet Common Control CHannel (PCCCH) comprises the common control and signalling channels used for the transfer of packet data. PCCCH contains The Packet Random Access CHannel (PRACH), Packet Paging CHannel (PPCH), Packet Access Grant CHannel (PAGCH), and Packet Broadcast Control CHannel (PBCCH). PRACH is used by the Mobile Terminal (MT) to initiate an uplink data transmission. PPCH is used to page MT prior to downlink data transfer. PAGCH is used to send resource assignment information to the MT, while PBCCH is used to broadcast packet data specific system information. The Packet Data Traffic CHannel (PDTCH) and Packet Associated Control CHannel (PACCH) form the traffic channels in GERAN. PDTCH is used to convey the traffic data to a specific MT. PDTCH is temporarily dedicated to a user. Users are allowed to use multiple simultaneous PDTCH to transfer data in multi-slot operation. Resource assignment/reassignment, adjustment in allocated capacity
for PDTCH and other necessary signalling information for each user is conveyed via PACCH [SARI-00].

![Transport & Network layer](image)

**Transport & Network layer**

---

**SNDCP**

---

**LLC**

---

**RLC/MAC**

---

**Physical Link Layer**

---

**Figure 2.25: E-GPRS Data Flow Diagram**

The user plane protocol layers of GERAN include the Transport/Network layer, SubNetwork Dependent Convergence Protocol (SNDCP) layer, Logical Link Control (LLC) layer, Radio Link Control (RLC) layer and Medium Access Control (MAC) layer. Transmission/reception data flow over the GERAN protocol stack is shown in Figure 2.25. The transfer of LLC PDUs on the packet data physical channel is supported by the use of a Temporary Block Flow (TBF). TBF is responsible for allocating radio resources on PDTCHs. As the name suggests, resource allocation based on TBF is temporary and is maintained only for the duration of the LLC PDU transfer. LLC PDUs are segmented into RLC/MAC blocks, and each RLC/MAC block is augmented with an 8 bit MAC header and a 16 bit block check sequence at the RLC/MAC layer. The Radio frame (or RLC/MAC block) is 20 ms long and it is transmitted over the wireless link using four bursts.

![GERAN User Plane protocol stack](image)
EGPRS introduces new Modulation-Coding Schemes (MCS) to achieve enhanced throughput capacity. These modulation coding schemes are referred to as MCS-1 to MCS-9. MCS-1 to MCS-4 are based on the Gaussian Minimum Shift Keying (GMSK) modulation scheme, as used in GSM speech services and GPRS PDTCHs. Schemes MCS-5 through MCS-9 employ a higher rate 8-state Phase Shift Keying (8-PSK) modulation scheme. Convolutional codes with different code rates are used for the channel protection. A summary of the coding parameters for the EGPRS coding schemes and the resulting information data rates for single slot downlink transmission are given in Table 2.4. Allowing a maximum of 8 time-slot operation, this channel structure supports a maximum data rate of 473.6 kbps for EGPRS downlink.

Table 2.4: Modulation-coding parameters for E-GPRS [extracted from 3GPP 03.64]

<table>
<thead>
<tr>
<th>Scheme</th>
<th>Convolutional code rate for data</th>
<th>Modulation</th>
<th>RLC blocks per Radio Block (20 ms)</th>
<th>Raw data within one Radio Block</th>
<th>Data rate kb/s</th>
</tr>
</thead>
<tbody>
<tr>
<td>MCS-1</td>
<td>0.53 GMSK</td>
<td>1</td>
<td>176</td>
<td>8.8</td>
<td></td>
</tr>
<tr>
<td>MCS-2</td>
<td>0.66 GMSK</td>
<td>1</td>
<td>224</td>
<td>11.2</td>
<td></td>
</tr>
<tr>
<td>MCS-3</td>
<td>0.80 GMSK</td>
<td>1</td>
<td>296</td>
<td>14.8</td>
<td></td>
</tr>
<tr>
<td>MCS-4</td>
<td>1.0 GMSK</td>
<td>1</td>
<td>352</td>
<td>17.6</td>
<td></td>
</tr>
<tr>
<td>MCS-5</td>
<td>0.37 8-PSK</td>
<td>1</td>
<td>448</td>
<td>22.4</td>
<td></td>
</tr>
<tr>
<td>MCS-6</td>
<td>0.49 8-PSK</td>
<td>1</td>
<td>592</td>
<td>29.2</td>
<td></td>
</tr>
<tr>
<td>MCS-7</td>
<td>0.76 8-PSK</td>
<td>2</td>
<td>2×448</td>
<td>44.8</td>
<td></td>
</tr>
<tr>
<td>MCS-8</td>
<td>0.92 8-PSK</td>
<td>2</td>
<td>2×592</td>
<td>54.4</td>
<td></td>
</tr>
<tr>
<td>MCS-9</td>
<td>1.0 8-PSK</td>
<td>2</td>
<td>2×592</td>
<td>59.2</td>
<td></td>
</tr>
</tbody>
</table>

2.5 Resource management strategy in wireless multimedia communications

In multimedia communications, there are several source and system parameters to be taken into account in the provision and maintenance of adequate service quality to multiple users in the system. They are

- Multimedia quality
- Source bit rate
- Error resilience
- Channel quality
- Channel protection
- Transmission power
- Processing power
- Mobility
Chapter 2. Wireless Multimedia Communications

- System capacity
- System load
- Coverage
- Total system transmit power
- Cell interference – cell planning
- Complexity.

Figure 2.27: Resource tradeoffs in wireless multimedia communication

All these parameters are inter-related and provide contradictory constraints on system performances. In conventional (system level) resource management schemes, only system capacity, system load, coverage, transmission power and interference are considered. Average
received channel quality is used to indicate the quality of service (multimedia quality) received by the user. However, for multimedia communications, average received channel condition does not always provide an accurate performance figure for perceptual quality. Moreover, a combination of source bit rate, error resilience, channel protection, transmission and processing power should be taken into account in optimising the received multimedia quality in systems with limited resources.

Figure 2.28: Graphical interpretation of resource allocation in wireless multimedia communications. (a). rate-distortion characteristics (b). Rate-power characteristics (c). Characteristics of channel coding and modulation. (d). Effect of error resilience (e). Channel quality vs system load and interference.
uses an Adaptive Intra Refresh (AIR) algorithm developed at CCSR [WORRa-01]. The algorithm implementation is based on the Sum of Absolute Difference (SAD) calculation. SAD is used in the motion vector calculation to measure the differences between the pixels in current and reference frames. A detailed discussion of the algorithm implementation and performance evaluation is given in [WORRa-01].

2.3.4 Decoder error concealment

Error concealment capabilities should be included in decoders so that the severity of artefacts resulting from transmission errors can be minimized. Given the standard MPEG-4 data format, there are three types of information that need to be estimated in a damaged macro block. They are the texture information, for either non-predictive or predictive image blocks, the motion information for predictive blocks and the coding mode. Concealment techniques are categorised into two main categories as temporal error concealment and spatial error concealment.

One approach to spatial error concealment is the prediction of pixels in a damaged block from pixels in adjacent correctly received blocks. Most pixels in correctly received adjacent blocks are too far from the missing samples; therefore, interpolation is usually carried out in the prediction. Instead of interpolating pixel-by-pixel, estimation of the DC coefficient (of DCT) of a damage block can be performed by averaging the DC values of surrounding blocks [GALA-98]. If the motion vector information is received correctly, effective spatial concealment can be achieved by replacing the error block with the corresponding macro-block in the previously decoded frame. This scheme is highly effective for concealment in low motion video sequences, however, for high motion sequences, the pixel interpolation method shows better error concealment [TALL-98]. The methods described above ignore any correctly received DCT coefficients. Correctly received DCT coefficients can in fact be used to improve the accuracy of the estimation. Here the concealment algorithm can be formulated as an unconstrained optimisation problem. The estimation should be such that the recovered block is to be smoothly connected with its neighbouring pixels both in the spatial and temporal domains, subject to the constraint that the DCT on the recovered block would produce the same values for the received coefficients [LUO-95, SHIR-00].

Projection onto convex sets techniques [SUN-95, WANGb-00] have been widely proposed to reconstruct the corrupted blocks. However, as the solution can only be obtained through an iterative process, this method is not suitable for real-time applications.
The techniques described for spatial domain concealment can also be applied in recovering erroneous motion vector information. Simple operations include replacement of the lost motion vector with the corresponding block in the previous frame, use of the average or the median of the MV of adjacent blocks, and re-estimation of the MV using statistical information. Better, but more complex motion information concealment can be conducted using different motion information for different pixel regions in the macro block [LAM-93, CHEN-97].

One way to estimate the mode information is to use statistical information of the coding mode pattern of adjacent blocks and find a most likely mode given the modes of surrounding blocks. A simple widely used approach is to assume that the macro block is coded in the intra-mode, and to use only spatial interpolation in recovering the corresponding blocks.

### 2.3.5 Multimedia transmission issues in mobile networks

In addition to the encoder error resilience and decoder error concealment techniques, there are other techniques which are designed to improve the received quality of transmitted video over bandwidth limited error prone environments. This section examines some existing efficient transmission methodologies proposed by other researches. The review is split into five areas:

- Joint source channel coding
- Advanced forward error correction
- Feedback based techniques
- Multiplexing
- Link adaptation

#### 2.3.5.1 Joint source-channel coding

In order to reduce the effect of channel noise on the transmitted bit stream, some form of Forward Error Correction (FEC) is applied for information transmission over error prone channels. However, the appropriate level of FEC coding generally depends upon many parameters such as the type of media, the available bandwidth, and the channel bit error rate. Especially for video applications, the quality distortion seen in a decoded video sequence is determined by the distortion due to channel errors as well as the quantisation distortion at the source encoder. Therefore, a combined source-channel coding approach is necessary in optimal allocation of rates
between source and channel protection, subject to a fixed constraint on overall transmission bandwidth. Much research has been conducted in joint source-channel coding for image and video applications. Studies revealed that lower source rate and extra channel protection is favourable for image and video transmission over low quality channels [MODE-81]. The algorithms, which determine the optimal source-channel coding ratio for a particular source coding algorithm, channel coding scheme and set of channel characteristics have been proposed in [BYST-98, BYST-00, KURC-00]. The communication channel states can often be modelled as a hidden Markov process. Based on this statistical framework, the design of an optimal joint source-channel coding is discussed in [DYCK-99]. The method proposed in [DYCK-99] uses state estimation and minimum mean square error estimation procedure in the determination of optimal source-channel code ratio. The theoretical performance bounds in distortion-rate characteristics based optimal source-channel bit allocation has been derived in [BYST-00].

2.3.5.2 Advanced forward error correction

Advanced forward error correction methods aim to reduce the effect of channel error on video quality by minimising the amount of important information loss during the transmission. The output bit stream from the video encoder is modified or restructured in such a way to achieve maximum perceptual quality at the receiver. These techniques can be categorised into three main areas:

- Unequal error protection
- Unequal power allocation
- Multiple descriptive transmission

Unequal Error Protection (UEP) basically uses prioritised transmission. The encoder output bit stream is separated into several sub-streams according to their importance in perceiving video quality. Then, high priority streams are transmitted using highly protected channels, while low channel protection is used for low priority streams. A common method to split encoded data into separate streams uses the MPEG-4 data partition method [GHAN-93]. This method is simple to implement, yet provides a significant improvement in received video quality. Another obvious way of separating video information into multiple streams is scalable or layered video coding. A layered video encoder generally generates a number of bit streams. The most critical information is contained in the base layer. Other layers contain information that is needed for enhancing the quality of the base layer. Some other techniques used in bit stream splitting are information separation in the transform domain [GHAR-99].
Unequal Power Allocation (UPA) techniques for enhancing video quality have recently been proposed, especially for video communications over CDMA based cellular networks. The operation of unequal power allocation techniques is similar to that of the unequal error protection techniques described above. However, network compatibility is essential for practical implementation. In addition to the constraints imposed by the limited bandwidth or channel throughput, the transmit power also provide a limiting factor in the UPA algorithm design. Several UPA techniques for wireless video applications have been proposed in [EISE-02, KIM-03, ZHAO-02]. All of these techniques are optimised to achieve a target video quality while minimising the transmission power.

Multiple Descriptive Coding (MDC) is intended to achieve quality improvements by exploring the diversity of transmission links. Similar to layered coding, a multiple descriptive code generates multiple bit streams. However, unlike in layered coding, these multiple streams have equal priority. The streams are transmitted over separate transmission channels with equal channel characteristics, in terms of average channel quality. The transmitted streams can individually be decoded at the receiver. When they are combined, improved quality is achieved because of the transmission diversity gain [LEE-02]. For example, assume two streams are transmitted over radio channels with similar channel parameters as shown in Figure 2.5. Even though both channels have similar average channel characteristics, the instantaneous channel qualities experienced are very different due to the time, space and frequency parameters. Therefore, appropriate stream combinations can result in better received quality. MDC is especially favourable in ad-hoc networks.

---

**Figure 2.5: A typical MDC system with two descriptions.**
2.3.5.3 Feedback based techniques

In all of the above error resilience techniques, the encoder and the decoder operate independently of each other. If a feedback channel can be set up, the decoder can inform the encoder about the instantaneous behaviour of the transmission channel and the information regarding the corrupted packets. Thus the encoder can adjust accordingly to suppress or to minimise the effect of channel errors on the video stream. Feedback based techniques can be described under three main areas:

- Retransmission techniques
- Optimal encoder mode selection
- Error tracking

The automatic Repeat request (ARQ) method is a powerful technique, which is commonly used in conventional data transmission to retransmit the error packets when bit errors are not allowed. However, retransmission control as used in data transmission creates unacceptable delay and is not suitable for real time applications. Modifications of ARQ techniques, so that it is to be suitable for real time applications have been proposed in [LIU-97, MATO-98]. A common feature of these modifications is to use, Hybrid ARQ, which is the combination of ARQ and other error resilience techniques. The combination of FEC and ARQ for mobile video communications has been studied in [LIU-97]. These schemes request that the encoder retransmit any incorrectly received packets. Even with one bit error in a packet, the encoder is told to retransmit the entire packet. Therefore, algorithm efficiency is significantly low especially under burst errors. Matoba [MATO-98, MATO-97] has proposed Selective-Repeat ARQ/FEC (SR-ARQ), which is a modification of the FEC/ARQ method. Using the properties of layered coding and considering different QoS requirements of different data sections within a video stream, further modifications to SR-ARQ, called QoS aware Selective Repeat ARQ (QSR-ARQ), are proposed by Wang in [WANGa-00, WANGc-00, ZHEN-99]. Results show that QSR-ARQ improves the data link protocol performance, network throughput and bandwidth efficiency. Other approaches based on combined adaptive QoS control, optimal mode selection and delay-constrained hybrid ARQ, can be found in [DAPE-00, BATR-98].

Another way of achieving performance improvements with the use of feedback channel information is the reference picture selection method. If a reference frame is corrupted due to channel errors, the error propagates over the following video frames until the reference frame is refreshed by intra-coded frame. The error propagation can be mitigated by referencing to a correctly received video frame for encoding of future video frames. This concept is employed in
optimal reference picture selection methods proposed in [COTE-00]. The algorithm takes into account the channel condition and the error concealment method used by the decoder, to optimise video coding mode selection in the compressed bit stream. However, the algorithm should be carefully designed to avoid performance degradation caused by parameter mismatch between the parameters used by the encoder, the parameters associated with the actual channel condition and the decoder error concealment methods.

Error tracking methods also utilize a feedback channel to adapt the system to varying channel conditions. In this method, an error is localised to an image region. Decoder error concealment is employed to make transmission errors less visible, and unacceptable residual errors are compensated by coding the distorted regions in intra mode. Negative ACKnowledgments (NACK) are sent back to the transmitter for image part that could not be decoded successfully. The encoder evaluates the NACKs and makes the intra mode decision on a macro block basis. The reconstruction of spatial-temporal error propagation at the encoder permits for the compensation of errors that have propagated [LIU-97, CHOI-99, STEI-97].

2.3.5.4 Data stream multiplexing

Since video traffic is delay-sensitive, a resource reservation scheme seems to be the right choice in guaranteeing quality requirements for real-time video communications. However, due to the unpredictably bursty nature of Variable Bit Rate (VBR) video, the resource reservation task is complicated. If resources are reserved according to peak rates, the network is under-utilised most of the time. VBR not only causes low transmission efficiency, but also causes difficulties for QoS provisioning in multi-user communication systems.

Many research activities concentrate on minimising the effect of VBR on resource allocation. These are based on different multiplexing schemes and can be categorized in to three main classes [WANG-98]. The bursty traffic can be smoothed to become near constant bit rate traffic by using a buffer for each data stream. This method is called temporal statistical multiplexing and is adopted in some real-time MPEG encoders. Although, this is very easy to implement, this can cause delay variation leading to inconsistent video quality. A second method is called prefetching. It allows users to get data before the playback start time. This can be used for non real time streaming applications, utilizing the high link capacity at non-peak times. Spatial statistical multiplexing, the third method, allows sharing of bandwidth between several streams to achieve an aggregated constant bit rate like throughput. In this way, the streams running at peak rate
borrow the bandwidth from streams running at low rates. Delay variation is not observed with this method, but the arrangement of traffic is critical and difficult.

A scene-based multiplexing method proposed in [WANG-98] belongs to the spatial statistical multiplexing methods described above. Several video bit streams are multiplexed in to a single bit stream resulting in better bandwidth utilization without introducing delay variation. Scene information is used to decide the bandwidth requirement for the aggregated video streams. The basic assumption used in this system is that a significant video traffic bandwidth change could result from a visual scene change. Video trace synchronization is used to avoid several I frames being transmitted simultaneously. Simulation results show that scene based multiplexing achieves improve bandwidth utilization when transmitting MPEG1 and MPEG2 video streams.

Another form of spatial statistical multiplexing method is described in [BAHL-98], namely inter-frame statistical multiplexing. This method uses region segmentation or the MPEG-4 video object concept to segment the video frame in to regions or video objects of differing importance. The peak bit rate for the most important region is reserved at connection establishment time. It is likely that most of the time the compressor will produce bits far below this peak value. The bandwidth left over after the important region has been transmitted, is used to transmit the remaining regions and retransmission if required. Priority was given in the order of the lowest frequency sub-band of the main regions, followed by the lowest frequency sub band of the remaining regions, followed by the subsequent higher frequency sub bands. Any bits left over after the reserved bandwidth was used up, were transmitted using any available unreserved bandwidth. Experimental results illustrate the efficient bandwidth allocation. As the most important regions always reach the receiver, transmitted image frames can be displayed in any case.

Object prioritisation based on statistical multiplexing method is described in [CELL-99]. Using the object-oriented features of MPEG-4, the video frame is divided into separate objects. These objects are transmitted independently assigning higher priorities to the most important objects and are multiplexed at the receiver to get the combined frame. The use of object-based prioritisation schemes has been shown to give a noticeable perceptual quality improvement in received video signals over bandwidth limited and noisy channels.
2.3.5.5 Link adaptation

Link adaptation is employed for mitigating the effect of time varying radio channel conditions on the received media quality. This involves the modification of source and network parameters in response to the instantaneous channel and interference conditions. The concept of source and network parameter adaptation for system throughput maximisation has been considered for data communication in [NAND-00, BALA-99, MEHT-00, GUTI-00, ZHAO-01]. Ideally, the design of link adaptation algorithms for multimedia services should optimise the delivery of acceptable video quality, facilitating good quality video services with the minimum possible demand on network resources. Application of link adaptation techniques for video communication has been introduced by Hanzo and his associates in their series of paper publications [HANZa-00, HANZb-00, CHER-00]. In these schemes, adaptive modulation is considered. The appropriate modulation mode for the transmission is selected from set of different modulation schemes according to the instantaneous quality of the transmission channel. Adaptive schemes have been shown to provide better received video quality compared to the non-adaptive schemes for video transmission over narrowband and wideband channels.

2.3.6 Video performance assessment

Figure 2.6: Sample video frames from selected sequences.
Three different ITU video sequences: namely “Suzie”, “Carphone” and “Foreman” are selected as the test sequences for the evaluation of quality enhancement algorithms developed in this thesis. Sample video frames of each of these sequences are shown in Figure 2.6. These video sequences characterise typical real-time video telephony scenarios. The “Suzie” sequence, which features a woman in a telephone conversation, is a relatively low motion head-and-shoulder sequence with a stationary background. On the other hand, “Carphone” and “Foreman” sequences feature high activity scenes. The “Foreman” sequence is particularly demanding and is highly sensitive to errors, as the camera is non-stationary and pans around the scene. To produce longer sequences (30 seconds), these sequences are repeated, with even numbered sequences played in reverse order to guarantee a smooth transition. The selected sequences span from low motion activities to very-high motion activities. Therefore, the average over the selected sequences will give an assessment figure, which can be applicable to the majority of video scenes encountered in a typical video conversation.

For all the work described in this thesis, the performances are evaluated based on objective quality measurement. Subjective quality measurement may be more suitable for assessing perceptual video quality. However, subjective tests require large number of users to evaluate large number of decoded video sequences in order to achieve meaningful average performance figure. As this is clearly not a convenient option, the objective quality measurements are used.

Peak Signal to Noise Ratio (PSNR), which is a standard measurement indicating the amount of original signal remaining compared to noise introduced, is used as the objective quality measure. This is taken on a frame-by-frame basis and is expressed as a function of the Root Mean Squared Error (RMSE) as shown in Equation 2.1.

\[
RMSE = \sqrt{\frac{1}{MN} \sum_{i=1}^{M} \sum_{j=1}^{N} (f(i,j) - \hat{f}(i,j))^2}
\]

(2.1)

\[
PSNR = 20 \log_{10} \left( \frac{2^n - 1}{RMSE} \right)
\]

(2.2)

where, \( f(i,j) \) and \( \hat{f}(i,j) \) are the luminance value of pixels in source and reconstructed images respectively. Each image contains \( M \) by \( N \) pixels. \( n \) is number of bits used in the formatting. Thus, \( 2^n - 1 \) indicates the highest possible luminance value of a pixel. PSNR in Equation 2.2 is expressed as the ratio between the highest possible luminance value of a pixel and the average of the differences between the luminance values of corresponding pixels position in the two images.
For high quality sequences, where the relative difference seen compared to the original is low, this quality measure provides a high PSNR value. A PSNR value of 20 dB or above is considered as acceptable quality. This threshold can vary from sequence to sequence depending on the amount of motion and activity involved and also it is subjected to user preferences. As shown in Figure 2.7, 20 dB PSNR suggests a reasonable perceptual quality video for selected test sequences.

![Sample decoded video frames with 20 dB PSNR. Coded at 64 kbps with 10fps.](image.png)

Figure 2.7: Sample decoded video frames with 20 dB PSNR. Coded at 64 kbps with 10fps.

Unless otherwise stated, each experiment describes in following chapters, was repeated 20 times to capture the effect of the bursty nature of the channel on the received video quality.

2.4 Third generation communication systems

The goal of wireless communication is to allow the user access to the required services at any time without regard to location or mobility. The user service requirement has been rapidly transformed from conventional telephony to multimedia services during the last few years, demanding higher capacity, higher data rates, and enhanced service flexibility from wireless
communication networks. Even though second generation radio access technologies, such as GSM and IS-95 have been successful for the delivery of telephony and low bit rate data services at rates up to 9.6 kbps, they are clearly not capable of providing high bit rate multimedia services including real-time video and audio transmission. In addition, network access was supported over circuit switched connection in second-generation wireless systems. This however, is inefficient for supporting service flexibility especially for bursty traffic, which would result in future multimedia services. To successfully meet the challenges set by multimedia service requirements and user demands, the International Telecommunication Union Radio communications (ITU-R) sector has established a set of service requirements for third generation radio access system [ITU]. They are

- Full area coverage and high data rates. Bit rates up to 144kbps are expected in a satellite radio and rural environment, up to 384kbps in urban radio environments and 2048kbps in indoor and low-range pico cell environments [3GPP TS 22.105].
- Symmetrical and asymmetrical data transmission. Symmetric transmission expects similar traffic loads in the uplink (from mobile terminal to the base station) and the downlink (from the base station to the mobile terminal). The network traffic generated from future multimedia applications is expected to be asymmetric in the uplink and downlink. These applications required asymmetric data transmission.
- Inclusion of circuit switched and packet switched services. This would facilitate efficient resource management for real-time delay critical as well as non-delay critical services.
- Good voice quality, comparable to wire-line quality.
- Greater capacity and improved spectrum efficiency
- Multiple simultaneous services to end users providing rich multimedia sessions.
- Global (international) roaming to provide discontinuous services to end users regardless of their location and mobility.
- The seamless incorporation of second-generation cellular systems in order to provide smoother transient.

ITU-R has elaborated on a framework for global third generation standard by recognising a limited number of radio access technologies. They are Universal Mobile Telecommunication System (UMTS), Enhanced Data rates for GSM Evolution (EDGE), and CDMA2000. UMTS is based on Wideband CDMA technology and is to be employed in Europe and Asia, using the frequency band around 2 GHz. EDGE is based on TDMA technology and uses the same air interface as the successful second generation mobile system GSM. General Packet Radio Service (GPRS) and High Speed Circuit Switched Data (HSCSD) are introduced by the Phase 2+ of the GSM standardisation process and support enhanced services with data rates up to 144 kbps in the packet switched and circuit switched domains respectively. GPRS has also been accepted by the
Telecommunication Industry Association (TIA) as the packet data standard for TDMA/136 systems. EDGE, which is the evolution of GPRS and HSCSD, provides third generation services up to 500kbps within GSM carrier spacing of 200 kHz [NILSa-99]. CDMA2000 is based on multi-carrier CDMA technology and it provides the upgraded solution for existing IS-95 operators, mainly in North America. The migration paths from existing second-generation standards to third generation standards are shown in Figure 2.8. The operation of the CDMA2000 system is not investigated in this thesis, thus no further detail of CDMA2000 system is presented.

Figure 2.8: Evolution toward global third-generation standards.

Figure 2.9: Third generation wireless communication system [taken from NILSb-99].
EDGE and UMTS are the most widely accepted third generation radio access technologies. It is being standardised by the 3rd Generation Partnership Project (3GPP). Even though EDGE and UMTS are based on two different multiple access technologies, both systems share the same core network. As shown in Figure 2.9, the evolved GSM core network will serve for a common GSM/UMTS core network that supports GSM/GPRS/EDGE and UMTS access.

The system architecture of the third generation wireless communication system is shown in Figure 2.10. The GSM/UMTS core network consists of four main components. They are the Serving GPRS Support Node (SGSN), the Gateway GPRS Support Node (GGSN), the Mobile Switching Centre/Visitor Location Register (MSC/VLR) and the Home Location Register (HLR). SGSN and GGSN handle the packet switched services. SGSN is connected to the Radio Access Network (RAN) and it is responsible for the delivery and management of packet switched services to mobile users. SGSN functions include the monitoring and tracking of user location, access control and provision of security. The SGSN is connected to the external IP network through GGSN. Between SGSN and GGSN IP is used as the transport protocol. Circuit switched services are handled by MSC/VLR and it is functions are similar to that of SGSN. SGSN and GGSN interwork with the MSC/VLR and the HLR through their specified interfaces.

Figure 2.10: The core network architecture of third generation wireless communication system.
Chapter 2. Wireless Multimedia Communications

2.4.1 Third generation mobile services and applications

Future communications systems are intended to support a wide range of applications that posses different quality of services. Compared to the current single service network, future multimedia applications require multiple service operation in integrated network environments (see Figure 2.11). Application scenarios predicted in future systems include interactive multiparty conferencing, three dimensional video gaming, virtual office/home environment, augmented reality scenes and specialised multimedia business applications such as telemedicine, remote security surveillance and intelligent transportation systems. The characteristics of many of these applications span a wide spectrum and are currently undefined. Therefore, it is unrealistic to optimise the third generation systems only for a set of existing applications. 3GPP has established a new service architecture to enable the support of a wide range of application scenarios. The service architecture and requirement are specified in a series of Technical Specification documents [3GPP TS 22.011, 3GPP TS 22.101, 3GPP TS 23.110, 3GPP TS 22.105, and 3GPP TS 23.107]. The technical specifications given in 3GPP Release 4 are used in the work carried out in this thesis.

Figure 2.11: Current and future cellular services and application scenarios [taken from NILSb-99]

Applications and services have been categorised into different classes, depending on their operations. Four traffic classes have been identified. They are conversational, streaming, interactive and background class [3GPP TS 22.105]. The main distinguishing factors between these classes are the delay sensitivity and error tolerance. The conversational class is meant for delay critical traffic, while the background class is the most delay-insensitive. The required error
tolerance depends on the offered service characteristics. The fundamental characteristics of these traffic classes are summarised in Table 2.1. An example of different services to be offered within these traffic classes are shown in Figure 2.12.

**Table 2.1: The fundamental characteristics of different traffic classes [HOLM-01]**

<table>
<thead>
<tr>
<th>Traffic class</th>
<th>Conversational class</th>
<th>Streaming class</th>
<th>Interactive class</th>
<th>Background class</th>
</tr>
</thead>
<tbody>
<tr>
<td>Traffic characteristics</td>
<td>-preserve time relation between information entities of the stream</td>
<td>-preserve time relation between information entities of the stream</td>
<td>-request response pattern</td>
<td>-destination is not expecting the data within a certain time</td>
</tr>
<tr>
<td></td>
<td>-conversational pattern (stringent and low delay)</td>
<td></td>
<td>-preserve data integrity</td>
<td>-preserve data integrity</td>
</tr>
</tbody>
</table>

**Figure 2.12: Summary of applications in terms of QoS requirements [taken from 3GPP TS 22.105]**

The above applications and services can be divided into two main categories as real-time and non-real-time services. Real time services include two-way conversational services such as interactive video conferencing, conventional telephony services and interactive video games. Streaming/retrieval services where users are allowed to access multimedia information servers and messaging services such as fax and e-mail, are considered under the non-real-time services. QoS requirements in terms of bit error rate and transfer delay for applications under these two categories in different system deployment environments are specified in [3GPP TS 22.105] and listed in Table 2.2.
Table 2.2: 3GPP QoS Requirement [taken from 3GPP TS 22.105]

<table>
<thead>
<tr>
<th>Operating environment</th>
<th>Real Time (Constant Delay)</th>
<th>Non Real Time (Variable Delay)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>BER/Max Transfer Delay</td>
<td>BER/Max Transfer Delay</td>
</tr>
<tr>
<td>Satellite</td>
<td>Max Transfer Delay less than 400 ms</td>
<td>Max Transfer Delay 1200 ms or more (Note 2)</td>
</tr>
<tr>
<td>(Terminal relative speed to ground up to 1000 km/h for plane)</td>
<td>BER $10^{-3} - 10^{-7}$ (Note 1)</td>
<td>BER = $10^{-5}$ to $10^{-6}$</td>
</tr>
<tr>
<td>Rural outdoor</td>
<td>Max Transfer Delay 20 - 300 ms</td>
<td>Max Transfer Delay 150 ms or more (Note 2)</td>
</tr>
<tr>
<td>(Terminal relative speed to ground up to 500 km/h) (Note 3)</td>
<td>BER $10^{-3} - 10^{-7}$ (Note 1)</td>
<td>BER = $10^{-5}$ to $10^{-6}$</td>
</tr>
<tr>
<td>Urban/ Suburban outdoor</td>
<td>Max Transfer Delay 20 - 300 ms</td>
<td>Max Transfer Delay 150 ms or more (Note 2)</td>
</tr>
<tr>
<td>(Terminal relative speed to ground up to 120 km/h)</td>
<td>BER $10^{-3} - 10^{-7}$ (Note 1)</td>
<td>BER = $10^{-5}$ to $10^{-6}$</td>
</tr>
<tr>
<td>Indoor/ Low range outdoor</td>
<td>Max Transfer Delay 20 - 300 ms</td>
<td>Max Transfer Delay 150 ms or more (Note 2)</td>
</tr>
<tr>
<td>(Terminal relative speed to ground up to 10 km/h)</td>
<td>BER $10^{-3} - 10^{-7}$ (Note 1)</td>
<td>BER = $10^{-5}$ to $10^{-6}$</td>
</tr>
</tbody>
</table>

NOTE 1: There is likely to be a compromise between BER and delay.
NOTE 2: The Max Transfer Delay should be here regarded as the target value for 95% of the data.
NOTE 3: The value of 500 km/h as the maximum speed to be supported in the rural outdoor environment was selected in order to provide service on high speed vehicles (e.g. trains). This is not meant to be the typical value for this environment (250 km/h is more typical).

2.4.2 Quality of Service (QoS) management

2.4.2.1 QoS parameters

The applications, which belong to different traffic classes, have different demands on network performance in terms of bandwidth, delay and reliability, making resource allocation and management a difficult task. A new QoS management architecture is introduced by 3GPP in order to enable efficient resource allocation in UMTS systems. Compared to the Internet and the second-generation mobile networks, UMTS provides an improved and important feature, namely the ability of an application to choose between different levels of quality. This is carried out through the negotiation of a well defined set of QoS parameters. The main QoS parameters include:

- Traffic class
- Maximum bit rate
- Guaranteed bit rate
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- Transfer delay
- Service Data Unit (SDU) error ratio
- Residual bit error ratio
- Maximum SDU size
- Delivery of erroneous SDU,
- Delivery order
- Traffic handling priority
- Allocation/retention priority

Traffic class is included as an attribute, allowing the network to make assumptions about the traffic source and to optimise the transport for that particular traffic type. Maximum bit rate is used to make code reservations in the radio interface and to limit the delivered bit-rate to applications or external networks, while guaranteed bit rate is defined to facilitate admission control and resource allocation. Maximum and guaranteed bit rates are into 10 distinctive rates, ranging from 8 to 2,048kbit/s, varying by a factor of 2 at each step [3GPP TS 24.008].

The transfer delay is defined by the time between a request and its delivery at the other Service Access Point (SAP), and is intended to specify the delay tolerated by the application. It allows a network to set the transport format and ARQ parameters in case of re-transmission. This is divided into three classes considering the delay range: 10 – 150 ms in 10ms increments, 200 – 950 ms in 50 ms increments and 1000- 4100ms in 100ms increments [3GPP TS 24.008].

The Service Data Unit (SDU) error ratio defined as the fraction of SDUs lost or detected as erroneous, while residual bit error ratio is defined as the undetected bit error ratio in the delivered SDUs. Together these parameters offer a mechanism to configure the protocols, algorithms and error detection schemes. The range for these attributes varies from $1 \times 10^{-1}$ to $6 \times 10^{-8}$. Other attributes, namely maximum SDU size, delivery order, delivery of erroneous SDU, traffic handling priority, and allocation/retention priority are defined in order to facilitate efficient admission control, QoS management, policing and charging for various services.

For each type of carrier there is a vast number of different attributes to be set. It is at the discretion of the operator to decide which of these the user will be able to manipulate. However, the operator will most likely provide a discrete set of traffic classes, with most attributes pre-set and others up to the user to decide.
2.4.2.2 QoS management architecture

The QoS architecture introduced in UMTS [3GPP TS 23.107] can be viewed as a layered architecture as illustrated in Figure 2.13. End-to-end QoS for an application is viewed as a series of chained services operating at different levels of the transmission networks. UMTS allows an application to negotiate bearer characteristics so that the best possible quality can be achieved. Renegotiation of bearer characteristics via a parameter renegotiation procedure is also allowed for active connections. A renegotiation command may be initiated by the application or the network operator. This enables efficient adaptive resource management and optimal service quality in a multi-user multi-service environment.

![Figure 2.13: UMTS bearer service architecture [3GPP TS 23.107].](image)

Application generated traffic should pass different networks on its way from the source to the destination. The end-to-end service on the application level is therefore realised over several different bearer services, namely a Terminal Equipment (TE) to Mobile Terminal (MT) local bearer service, a UMTS bearer service, and an external bearer service, such as the Internet.

The UMTS bearer service provides the quality of service in UMTS. It consists of two parts, the Radio Access Bearer (RAB) and the Core Network Bearer Service (CN-BS). The RAB service is
based on the characteristics of the radio interface and is maintained for moving MTs. The role of the CN-BS is to control and utilise quality within the core network.

In order to realize a certain quality of service, bearer services with clearly defined characteristics and functionality should be set up from the source to the destination of a service. These aspects include the control signalling, user plane transport and QoS management functions. Within the UMTS framework [3GPP TS 23.107], several QoS management functions are specified in order to establish, modify and maintain a UMTS bearer service with a specific QoS. The allocation of these functions to UMTS entities indicates the requirement for the specific entity to enforce the QoS commitments negotiated for the UMTS bearer service. The specific realisation of these functions is implementation dependent and has only to maintain the specific QoS characteristics. QoS management functions are divided into two main categories: control plane and user plane QoS management.

Control plane QoS management consists of a translator, a service manager and an admission/capability controller. Translator converts the QoS attribute between a UMTS bearer service and the external network and maintains QoS parameter mapping between two networks. The service manager is responsible for establishing, modifying and maintaining the service by providing all user plane QoS management functions with the relevant attributes. This may also perform an attribute translation to request lower layer services and may interrogate other control functions to receive permission for service provision. The admission/capability controller maintains a database of all available resources of a network entity and resources allocated to UMTS bearer services. The function checks whether the required resources can be provided for each UMTS bearer service request or modification. It also checks for the capability of the network entity, i.e. whether the specific service is implemented and not blocked for administrative reasons [3GPP TS 23.107].

The user plane QoS management functions maintain the signalling and user data traffic within certain limits, defined by specific QoS attributes. They include a parameter converter, a resource management unit, a classification unit and a traffic-coordinating unit. Within the UMTS network the parameter converter is responsible for converting service attributes between different bearers in order to maintain the intended QoS at the transfer. The resource manager distributes the available resources between services according to the required QoS. Examples of resource management functions at the radio interface include scheduling, bandwidth management and power control. The classification function is only used within the concept of multiple UMTS
bearer services establishment. This assigns incoming data units to the established services of a MT according to the related QoS attributes, saving system resources and speeding up the service performance. The appropriate UMTS bearer service is derived from the data unit header or from traffic characteristics of the data. Traffic conditioning is performed by policing or by traffic shaping. The policing function compares the data unit traffic with the related QoS attributes. The traffic shaper forms the data unit traffic according to the QoS of the service [3GPP TS 23.107].

2.4.2.3 End-to-end IP QoS management functions

For the backbone IP network, the end-to-end QoS is provided by a local mechanism in the UE, the UMTS Bearer Service (UMTS-BS), and the QoS mechanism of the IP network. To control the external IP Bearer Service (IP-BS), the IP BS manager is used. This manages the IP-BS according to standard IP mechanisms such as Diffserv, RSVP, and Intserv. The translation/mapping function provides inter-working between the IP-BS and UMTS-BS. The parameters used within the UMTS-BS are mapped to those used within the IP-BS. It is possible for the IP BS manager to be located both in the UE and the gateway node, or the gateway node only [3GPP TS 23.207].

Application level IP-based QoS is mapped to the UMTS QoS by a network element, such as the 3G-GGSN. Within the UMTS network, an element called the Packet Data Protocol (PDP) context is used to negotiate and establish the QoS profile. The scope of the PDP context is from the UE to the GGSN, as shown in Figure 2.14.

![Figure 2.14: Network Architecture for QoS Conceptual Models [3GPP TS 23.207].](image)

The PDP context specified for UMTS Release 2000 is defined using the information sets held in the UE, SGSN, and GGSN that are used to specify the connection between one subscriber and the network. And it identifies an application, a PDP type and one QoS profile. More PDP contexts (up
to 16) with different QoS parameters can share the same PDP address. A Traffic Flow Template (TFT) is used to distinguish between them [3GPP TS 23.060]. The PDP context activation procedure is used to activate the first PDP context for a given PDP address and Access Point Name (APN), where all additional contexts associated with the same PDP address and APN are activated with the secondary PDP context activation procedure (see Figure 2.15).

The PDP context can be activated and released by either the UE or the Network. If a QoS requirement is beyond the capabilities of a communication network, the network negotiates the QoS profile as close as possible to the requested QoS profile. The Mobile Terminal can either accept the negotiated QoS profile, or deactivate the PDP context. After a successful PDP context deactivation, the associated Network layer Service Access Point Identifier (NSAPI) and Transaction Identifier (TI) values are released and can be reassigned to another PDP context [3GPP-24.007, 3GPP-24.008].

Figure 2.15: Illustration of first and second PDP contexts [KARA-00].

The PDP context modification procedure, which is activated by the network or by the UE, is used to change the QoS parameters, the Radio priority level or the TFT. These are negotiated during the PDP context activation procedure, the secondary PDP context activation procedure or at previously performed PDP context modification procedures.

Figure 16 shows the format of the Quality of Service information element, which specifies the QoS parameters for a PDP context. The explanation of these QoS attributes is given in Section 2.4.2.1 above. It indicates QoS parameter values in both the external network and in the radio access network and facilitates QoS parameter mapping from one network to another.
Quality of service IEI
Length of quality of service IE
Delay class
Reliability class
Peak throughput
Precedence class
Peak throughput
Mean throughput
Traffic Class
Delivery order
Delivery of erroneous SDU
Maximum SDU size
Maximum bit rate for uplink
Maximum bit rate for downlink
Residual BER
SDU error ratio
Traffic Handling priority
Guaranteed bit rate for uplink
Guaranteed bit rate for downlink

Figure 2.16: Quality of Service information element [3GPP-TS 24.008].

2.4.3 UMTS Terrestrial Radio Access Network (UTRAN)

Figure 2.17: Systems components in a UMTS [taken from NILSa-99]

The system components of UTRAN are shown in Figure 2.17. Functionally, the network elements are grouped into the Radio Network Subsystem (RNS), the Core Network (CN), and the User
Equipment (UE). UTRAN consists of a set of RNS’s connected to the core network through the Iu interface. The interface between the UE and the RNS is named Uu. An RNS contains a Radio Network Controller (RNC) and one or more Node Bs. The RNS handles all radio related functionality in its allocated region. A Node B is connected to a RNC through the Iub interface, and communication between RNS’s is conducted through the Iur interface. One or more cells is allocated to each Node B.

As explained in Section 2.4, the protocol within the CN is adopted from the evolution of GPRS protocol design. However, both the UE and UTRAN feature completely new protocol designs, which are based on the new WCDMA radio technology. WCDMA air interfaces have two versions defined for operation in Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD) modes. Only the FDD operation is investigated in this thesis. The modulation chip rate for WCDMA is 3.8 Mega chips per second. The specified pulse-shaping roll off factor is 0.22. This leads to a carrier bandwidth of approximately 5 MHz. The nominal channel spacing is 5 MHz, However, this can be adjusted approximately between 4.4 and 5 MHz, to optimise performance depending on interference between carriers in a particular operating environment. The FDD version is designed to operate in either of the following frequency bands [3GPP TS 25.101].

- 1920-1980 MHz for uplink and 2110-2170 MHz for downlink
- 1850-1950 MHz for uplink and 1930-1990 MHz for downlink.

All radio channels are code division multiplexed and are transmitted over the same (entire) frequency band.

**Table 2.3: WCDMA air interface parameters for FDD mode operation**

<table>
<thead>
<tr>
<th>Operating frequency band</th>
<th>2110-2170 MHz</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>1930-1990 MHz downlink</td>
</tr>
<tr>
<td></td>
<td>1920-1980 MHz</td>
</tr>
<tr>
<td></td>
<td>1850-1910 MHz uplink</td>
</tr>
<tr>
<td>Duplexing mode</td>
<td>Frequency Division Duplex (FDD)</td>
</tr>
<tr>
<td>Chip rate</td>
<td>3.84 Mega chip per second</td>
</tr>
<tr>
<td>Pulse-shaping roll-off factor</td>
<td>0.22</td>
</tr>
<tr>
<td>Carrier bandwidth</td>
<td>5 MHz</td>
</tr>
</tbody>
</table>

WCDMA supports highly variable user data rates with the use of variable spreading factors; facilitating the bandwidth on demand concept. Transmission data rates of up to 384 kbps are supported in wide area coverage, and 2 Mbps in local area coverage. The radio frame length is fixed at 10 ms. The number of information bits or symbols transmitted in a radio frame may vary,
corresponding to the spreading factor used for the transmission, while the number of chips in a radio frame is fixed at 38,400 [3GPP TS 25.101].

2.4.3.1 Radio interface protocol

The radio interface protocol architecture, which is visible in the UTRAN and the User Equipment (UE), is shown in Figure 2.18. Layer 1 comprises the WCDMA physical layer. Layer 2, which is the data link layer, is further split into Medium Access Control (MAC), Radio Link Control (RLC), Packet Data Convergence Protocol (PDCP), and Broadcast Multicast Control (BMC). The PDCP exists mainly to adapt packet-switched connections to the radio environment by compressing headers with negotiable algorithms. Adaptation of broadcast and multicast services to the radio interface is handled by BMC. For circuit-switched connections, user-plane radio bearers are directly connected to the RLC. Every radio bearer should be connected to one unique instance of the RLC.

The Radio Resource Control (RRC) is the principal component of the network layer—Layer 3. This comprises functions such as broadcasting of system information, radio resource handling, handover management, admission control and provision of requested QoS for a given application. Unlike the traditional layered protocol architecture, where protocol layer interaction is only allowed between adjacent layers, RRC interfaces with all other protocols, providing fast local inter-layer controls. These interfaces allow the RRC to configure characteristics of the lower layer protocol entities including parameters for the physical, transport, and logical channels [HOLM-01]. Furthermore the same control interfaces are used by the RRC layer, to control the

SDU – Service Data Unit

Figure 2.18: Radio-interface protocol architecture.
measurements performed by the lower layers and by the lower layers, to report measurement results and errors to the RRC.

![Figure 2.19: Interactions between RRC and lower layers [3GPP TS 25.301].](image)

UTRAN supports both circuit switched and packet switched connections. In order to transmit an application's data between UE and the end system, QoS enabled bearers have to be established between the UE and the Media GateWay (MGW). Figure 2.20 shows the User Plane protocol stack used for data transmission over packet switched connection in Release 4.

![Figure 2.20: User Plane UMTS protocol stack for packet switched connection [3GPP 23.060].](image)
2.4.3.2 Channel structure

Channels are used as a mean of interfacing the L2 and L1 sub-layers. Between the RLC/MAC layer and the network layer, logical channels are used. Between the RLC/MAC and the PHY layers, the transport channels are used, and below the PHY layer is the physical channel (see Figure 2.18).

Generally, logical channels can be divided into control and traffic channels. The paging control channel and the broadcast control channel are for the downlink only. The common control channel is a bi-directional channel shared by all UE’s, while the common transport channel is a downlink only shared channel. Dedicated control channels and dedicated transport channels are unique for each UE.

Transport channels are used to transfer the data generated at a higher layer to the physical layer, where it gets transmitted over the air interface. The transport channels are described by a set of transport channel parameters, which are designed to characterise the data transfer over the radio interface. Each transport channel is accompanied by the Transport Format Indicator (TFI), which describes the format of data to be expected from the higher layer at each time interval. The physical layer combines the TFI from multiple transport channels to form a Transport Format Combination Indicator (TFCI). This facilitates the combination of several Transport channels, called Composite Transport Channel, at the physical layer and their correct recovery at the receiver [HOLM–01].

![Figure 2.21: Transport channel mapping.](image-url)
In UTRA, two types of transport channels, namely dedicated channels and common channels, exist. As the name suggests, the main difference between them is that a common channel has its resources divided between all or a group of users in a cell, while a dedicated channel reserves resources for a single user.

Transmission Time Interval (TTI) defines the data arrival period from higher layers to the Physical layer. TTI size has been defined to be 10, 20, 40, and 80 ms. Selection of TTI size depends on the traffic characteristics. The amount of data arrived in each TTI can vary in size, and is indicated in the Transport Format Indicator (TFI). In the case of transport channel multiplexing, TTIs for different transport channels are time aligned as shown in Figure 2.22.

![Figure 2.22: Transmission Time Intervals (TTIs) in transport channel multiplexing [HOLM-01].](image)

The physical channels are defined by a specific set of radio interface parameters such as scrambling code, spreading code, carrier frequency, and transmission power step. The channels are used to convey the actual data through the wireless link. The most important control information in a cell is carried by the Primary Common Control Physical Channel (PCCPCH) and Secondary Common Control Physical Channel (SCCPCH). The difference between these two is that the PCCPCH is always broadcasted over the whole cell in a well-defined format, while the SCCPCH can be more flexible in terms of transmission diversity and format. In the uplink, the Physical Random Access Channel (PRACH) and Physical Common Packet Channel (PCPCH) are data channels shared by many users. The slotted ALOHA approach is used to grant user access in PRACH [QIU-01]. A number of small preambles precede the actual data, serving as power control and collision detection. The Physical Downlink Shared Channel (PDSCH) is
shared by many users in downlink transmission. One PDSCH is allocated to a single UE within a radio frame, but if multiple PDSCHs exist they can be allocated to different UEs arbitrarily: one to many or many to one. The Dedicated Physical Data CHannel (DPDCH) and Dedicated Physical Control CHannel (DPCCH) together realize the Dedicated CHannel (DCH), which is dedicated to a single user [HOLMA-01].

### 2.4.3.3 Modes of connection

Figure 2.23 shows the possible modes of realizing the connections of the radio bearers at each layer. PDCP, RLC and MAC modes must be combined with physical layer parameters in a way that satisfies different QoS demands on the radio bearers. However, the exact parameter setting is a choice of the implementer of the UMTS system and the network operator.

![Figure 2.23: Inter-layer Modes of Operation](image-url)

The radio bearer can be viewed as either Packet Switched (PS) or Circuit-Switched (CS). A PS connection passes the PDCP, where header compression can be applied or not. The RLC offers three modes of data transfer. The transparent mode transmits higher layer Payload Data Units (PDUs) without adding any protocol information and it is recommended for real-time conversational applications. The un-acknowledged mode will not guarantee the delivery to the
peer entity, but offers other services such as detection of erroneous data. The acknowledged mode guarantees the delivery through the use of Automatic Repeat reQuest (ARQ) [3GPP TS 25.322].

The MAC layer can be operated in dedicated, shared or broadcast mode. Dedicated mode is responsible for handling dedicated channels allocated to a UE in connected mode, while shared mode takes the responsibility of handling shared channels. The broadcast channels are transmitted using broadcast mode. The physical layer follows the MAC in choosing a dedicated or shared physical channel [3GPP TS 25.321].

In UTRA spreading is based on the Orthogonal Variable Spreading Factor (OVSF) technique. The properties of spreading codes and code generation are discussed in Section 3.3.3. Quadrature Phase Shift Keying (QPSK) modulation is used for downlink transmission. Both convolutional and turbo coding are supported for channel protection. The maximum possible transmission rate in downlink is 5760 kbps. It is provided by three parallel codes with a spreading factor of 4. With 1/2 rate channel coding, this could accommodate up to 2.3 Mbps user data. However, the practical maximum user data rate is subject to the amount of interference present in the system and the quality requirement of the application.

2.4.4 GSM/EDGE Radio Access Network (GERAN)

![GERAN system architecture.](image)
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The system component of GERAN is shown in Figure 2.24. GERAN is connected to the core network through the Iu and A interfaces. Services based on second-generation systems are supported by the 2G SGSN and the 2G MSC. The 3G SGSN and 3G MSC take responsibilities for support of third generation mobile services. Each SGSN is in charge of several Base Station Subsystems (BSSs). A BSS contains a Base Station Controller (BSC) and one or more Base Transceiver Stations (BTSs). The BCS monitors and controls the BTSs in its BSS. MTs are connected to BTS via the Um air interface [GRAN-99].

GERAN uses the same air interface specified in GSM. Multiplex access is based on the combination of TDMA and FDMA techniques [STUC-03]. Frequency bands have been reserved around the 900 MHz for GSM 900 and around 1800 MHz for GSM 1800 to be used in Europe. Allocated frequency bands are:

- 890 -915 MHz uplink and 935-960 MHz downlink
- 1710-1785 MHz uplink and 1805-1880 MHz downlink

These bands are divided by frequency into a number of carriers, each spaced 200 kHz apart. One or more carrier frequencies are assigned to each base-station. The selection of carrier frequencies for each base station and the carrier reuse distance is decided by the cell planning procedure. Optionally, frequency hopping capabilities can be used. And the advantage is that the quality on all communication links is averaged through the interferer’s diversity. Each of the carrier frequencies is divided into eight time slots. The communication channel is described by its time slot number and a carrier frequency. One time slot is 577 μs long. Within a time slot, data is transmitted in bursts.

GERAN introduces a set of control and traffic channels to be used in packet switched connections. The Packet Common Control CHannel (PCCCH) comprises the common control and signalling channels used for the transfer of packet data. PCCCH contains The Packet Random Access CHannel (PRACH), Packet Paging CHannel (PPCH), Packet Access Grant CHannel (PAGCH), and Packet Broadcast Control CHannel (PBCCH). PRACH is used by the Mobile Terminal (MT) to initiate an uplink data transmission. PPCH is used to page MT prior to downlink data transfer. PAGCH is used to send resource assignment information to the MT, while PBCCH is used to broadcast packet data specific system information. The Packet Data Traffic CHannel (PDTCH) and Packet Associated Control CHannel (PACCH) form the traffic channels in GERAN. PDTCH is used to convey the traffic data to a specific MT. PDTCH is temporarily dedicated to a user. Users are allowed to use multiple simultaneous PDTCH to transfer data in multi-slot operation. Resource assignment/reassignment, adjustment in allocated capacity
for PDTCH and other necessary signalling information for each user is conveyed via PACCH [SARI-00].

The user plane protocol layers of GERAN include the Transport/Network layer, Subnetwork Dependent Convergence Protocol (SNDCP) layer, Logical Link Control (LLC) layer, Radio Link Control (RLC) layer and Medium Access Control (MAC) layer. Transmission/reception data flow over the GERAN protocol stack is shown in Figure 2.25. The transfer of LLC PDUs on the packet data physical channel is supported by the use of a Temporary Block Flow (TBF). TBF is responsible for allocating radio resources on PDTCHs. As the name suggests, resource allocation based on TBF is temporary and is maintained only for the duration of the LLC PDU transfer. LLC PDUs are segmented into RLC/MAC blocks, and each RLC/MAC block is augmented with an 8 bit MAC header and a 16 bit block check sequence at the RLC/MAC layer. The Radio frame (or RLC/MAC block) is 20 ms long and it is transmitted over the wireless link using four bursts.

Figure 2.26: GERAN User Plane protocol stack [from TS 23.060]
EGPRS introduces new Modulation-Coding Schemes (MCS) to achieve enhanced throughput capacity. These modulation coding schemes are referred to as MCS-1 to MCS-9. MCS-1 to MCS-4 are based on the Gaussian Minimum Shift Keying (GMSK) modulation scheme, as used in GSM speech services and GPRS PDTCHs. Schemes MCS-5 through MCS-9 employ a higher rate 8-state Phase Shift Keying (8-PSK) modulation scheme. Convolutional codes with different code rates are used for the channel protection. A summary of the coding parameters for the EGPRS coding schemes and the resulting information data rates for single slot downlink transmission are given in Table 2.4. Allowing a maximum of 8 time-slot operation, this channel structure supports a maximum data rate of 473.6 kbps for EGPRS downlink.

<table>
<thead>
<tr>
<th>Scheme</th>
<th>Convolutional code rate for data</th>
<th>Modulation</th>
<th>RLC blocks per Radio Block (20 ms)</th>
<th>Raw data within one Radio Block</th>
<th>Data rate kb/s</th>
</tr>
</thead>
<tbody>
<tr>
<td>MCS-1</td>
<td>0.53</td>
<td>GMSK</td>
<td>1</td>
<td>176</td>
<td>8.8</td>
</tr>
<tr>
<td>MCS-2</td>
<td>0.66</td>
<td>GMSK</td>
<td>1</td>
<td>224</td>
<td>11.2</td>
</tr>
<tr>
<td>MCS-3</td>
<td>0.80</td>
<td>GMSK</td>
<td>1</td>
<td>296</td>
<td>14.8</td>
</tr>
<tr>
<td>MCS-4</td>
<td>1.0</td>
<td>GMSK</td>
<td>1</td>
<td>352</td>
<td>17.6</td>
</tr>
<tr>
<td>MCS-5</td>
<td>0.37</td>
<td>8-PSK</td>
<td>1</td>
<td>448</td>
<td>22.4</td>
</tr>
<tr>
<td>MCS-6</td>
<td>0.49</td>
<td>8-PSK</td>
<td>1</td>
<td>592</td>
<td>29.2</td>
</tr>
<tr>
<td>MCS-7</td>
<td>0.76</td>
<td>8-PSK</td>
<td>2</td>
<td>2×448</td>
<td>44.8</td>
</tr>
<tr>
<td>MCS-8</td>
<td>0.92</td>
<td>8-PSK</td>
<td>2</td>
<td>2×544</td>
<td>54.4</td>
</tr>
<tr>
<td>MCS-9</td>
<td>1.0</td>
<td>8-PSK</td>
<td>2</td>
<td>2×592</td>
<td>59.2</td>
</tr>
</tbody>
</table>

2.5 Resource management strategy in wireless multimedia communications

In multimedia communications, there are several source and system parameters to be taken into account in the provision and maintenance of adequate service quality to multiple users in the system. They are

- Multimedia quality
- Source bit rate
- Error resilience
- Channel quality
- Channel protection
- Transmission power
- Processing power
- Mobility
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- System capacity
- System load
- Coverage
- Total system transmit power
- Cell interference - cell planning
- Complexity.

![Resource tradeoffs diagram](image)

(a) Conventional resource tradeoffs diagram

![Multimedia quality](image)

(b) Multimedia quality

Figure 2.27: Resource tradeoffs in wireless multimedia communication

All these parameters are inter-related and provide contradictory constraints on system performances. In conventional (system level) resource management schemes, only system capacity, system load, coverage, transmission power and interference are considered. Average
received channel quality is used to indicate the quality of service (multimedia quality) received by the user. However, for multimedia communications, average received channel condition does not always provide an accurate performance figure for perceptual quality. Moreover, a combination of source bit rate, error resilience, channel protection, transmission and processing power should be taken into account in optimising the received multimedia quality in systems with limited resources.

Figure 2.28: Graphical interpretation of resource allocation in wireless multimedia communications. (a). rate-distortion characteristics (b). Rate-power characteristics (c). Characteristics of channel coding and modulation. (d). Effect of error resilience (e). Channel quality vs system load and interference.
Figure 2.28 presents a graphical interpretation of the tradeoffs problem between source and system parameters in allocating resources for multimedia communications.

Figure 2.28(a) illustrates the relationship between distortions, source bit rate and channel quality for video transmission over a fixed bandwidth channel. Video distortion can be considered as a combination of quantisation distortion and channel induced distortion. Quantisation distortion decreases with increases in source rate. The distortion due to the channel errors increases with increases of bit rate. This is because the amount of channel protection depends on the channel coding rate, which is decided by the available channel bandwidth and the source rate. A higher source rate requires a reduction in channel protection when operating over a fixed bandwidth channel. The resulting total distortion is as shown in Figure 2.29. The optimal source rate which gives minimum distortion, varies according to the channel quality. The setting of optimal source and channel coding rates for a fixed bandwidth channel is provided by joint source-channel coding, which is described in Section 2.3.5.1.

Mobile terminal power consumption can also be divided into two parts: processing power and transmission power. The transmission power is directly proportional to the source rate, if the transmission bit energy is kept constant during transmission. Processing power is often ignored in resource allocation schemes. However, processing power is significant in current and future multimedia source coding as much signal processing is involved in compression and resilience algorithms. Mobile computation and communication platforms must rely on the battery power. And future mobile terminals should offer longer battery life in an attractive smaller terminal. Therefore, the processing power should be considered as a parameter in resource allocation and
the combination of processing and transmission power should be minimised for a given transmission [LAN-03]. Figure 2.30 shows the relationship between processing power, transmission power and source rate. And a three dimensional illustration is given in Figure 2.28(b).

![Figure 2.30: Rate-power characteristics.](image)

Channel coding and modulation provide different degrees of channel protection and source throughput in a given channel environment. Variation of bit error rate characteristics with channel coding and modulation is shown in Figure 2.28(c). Figure 2.28(d) depicts the effects of error resilience on multimedia quality, while influences of system load and interference on channel quality are shown in Figure 2.28(e). Resource allocation in multimedia communication should consider all these factors in achieving optimal received quality. The complexity of such a system is considerably high, thus a sub-optimal approach is often followed. Moreover, due to the dynamic nature of communication systems, the system parameters fluctuate with time. Therefore, adaptive resource allocation must be implemented in order to maintain the required service quality at an acceptable level. In the following chapters, these resource management issues are directly and indirectly exploited for the transmission of video over third generation wireless communication systems.

### 2.6 Conclusion

This chapter has given an overview of the technologies and concepts involved in provisioning multimedia communications over wireless access networks. The first half of the chapter deals
Chapter 2. Wireless Multimedia Communications

with the recent advances in multimedia technologies. A description of the constraints involved in wireless multimedia communications is given, such as the scarcity of bandwidth, the error characteristics of wireless channels, and lack of network support. The role of media compression technologies, in particular error robust low bit rate video codec design, is discussed. This includes the importance of encoder based error resilience schemes, decoder based error isolation, and concealment schemes. A variety of techniques have been proposed for increasing the quality of the transmitted media stream over error prone wireless links. These techniques are categorised into joint-source-channel coding, advanced forward error correction, feedback based techniques, stream multiplexing, and link adaptation. A discussion of each of these techniques and their use in multimedia communication is presented in Section 2.3.5.

The evolution of wireless communication systems is described in the second half of the chapter. Starting with the technological transition from second generation wireless communication systems to third generation wireless communication systems, the service capabilities and application of third generation communication systems are presented. GERAN and UTRAN are chosen as the radio access networks for use in the experiments described in this thesis. GERAN provides a suitable migration path from second generation to third generation wireless systems, and is based on the combined TDMA and FDMA multiple access technologies. UTRAN on the other hand uses WCDMA based radio access network. They both share the same core network, which is the evolution of GSM/GPRS core networks. The above radio technologies offer end-to-end packet transfer capabilities, service flexibility, high data rates, symmetrical/asymmetrical data transmission, and multiple simultaneous services. Those features are necessary for supporting rich multimedia services, such as multi-party video conferencing, virtual office environments, 3D video games, etc.

The last section of the chapter discusses the resource management issues in wireless multimedia communications. The tradeoffs between source rate, channel protection, error resilience, transmission power, processing power, complexity and perceptual quality are discussed for the allocation of network resources for transmission of multimedia over given channel conditions. The effects of system parameters such as system capacity, coverage, interference, mobility and system load on multimedia performances in a multi-user scenario should also be taken into account in system resource allocation. The remainder of this thesis will examine video transmission and performance enhancement techniques that may be used to reconcile the conflicting requirements between the quality of service, radio access network characteristics and system characteristics of third generation wireless networks.

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Chapter 3

3 Design and Development of a UMTS Radio Access Simulator

3.1 Introduction

This chapter presents the design and implementation procedure for a multimedia evaluation testbed of the UMTS forward link. A WCDMA physical link layer simulator has been implemented using the Signal Processing WorkSystem (SPW) software simulation tools developed by cadence Design System Inc [CADE-02]. The model has been developed in a generic manner that includes all the forward link radio configurations, channel structures, channel coding/decoding, spreading/de-spreading, modulation parameters, transmission modelling and their corresponding data rates according to the UMTS specifications. The performance of the simulator model is validated by comparison with figures presented in the relevant 3GPP documents [TSGR4-578, TSGR4-581] for the specified measurement channels, propagation environments and interference conditions in [3GPP TS 25.101]. Using the developed simulator, a set of UMTS error pattern files is generated for different radio bearer configurations in different operating environments, and the results are presented. Furthermore, UMTS link level performance is enhanced by implementing a closed loop power control algorithm. A UMTS radio interface protocol model, which represents the data flow across the UMTS protocol layers, is implemented in Visual C++. It is integrated with the physical link layer model to emulate the actual radio interface experienced by users. This
allows for interactive testing of the effects of different parameter settings of the UMTS Terrestrial Radio Access Network (UTRAN) upon the received multimedia quality.

3.2 UMTS model description

The physical link layer parameters and functionality of the downlink for the FDD mode of the UMTS radio access scheme is described in this section. The main issues addressed are transport/physical channel structures, channel coding, spreading, modulation, transmission modelling and channel decoding. Only the dedicated channels are considered, as the end application is real-time multimedia transmission for dedicated users. The implementation closely follows the relevant 3GPP specifications. A closed loop fast power control method is also implemented.

Figure 3.1: UMTS Physical link Layer Model.
Chapter 3. Design and Development of UMTS Radio Access Simulator

The developed model simulates the UMTS air interface. Figure 3.1 is a block diagram of the simulated physical link layer. It can be seen that the transmitted signal is subjected to a multi-path fast fading environment, where the power-delay profiles are specified in [UMTS-30.03]. In addition, an AWGN source is presented after the multi-path propagation model. Co-channel interferers are not explicitly presented in the model. This is because, as explained in Section 3.4.2, the loss of orthogonality of co-channels due to multi-path propagation can be quantified using a parameter called the “orthogonality factor”, [HOLM-01], which indicates the fraction of intra-cell interfering power that is perceived by the receiver as Gaussian noise. The multipath-induced inter symbol interference is implicit in the developed chip level simulator. By changing the variance of the AWGN source, the bit error and block error characteristics can be determined for a range of carrier-to-interference (C/I) ratios and Signal-to-Noise ratios (S/N) for different physical layer configurations. The simulator only considers a static C/I and S/N profile, and no slow fading effects are implemented. However, as described in Section 8.3, slow fading can be easily implemented by concatenating the data sets describing the channel bit error characteristics of different, static C/I levels.

Each Radio Access Bearer (RAB) is normally accompanied by a Signalling Radio Bearer (SRB) [3GPP TS 34.108]. Therefore, in the simulator, two dedicated transport channels are multiplexed and mapped on to a physical channel.

3.2.1 Channel coding

UTRA employs four channel coding schemes offering flexibility in the degree of protection, coding complexity and traffic capacity available to the user. The available channel coding methods and code rates for dedicated channels are 1/2 rate convolutional code, 1/3 rate convolutional code, 1/3 rate turbo code, and no coding.

1/2 rate and 1/3 rate convolution coding is intended to be used with low data rates, equivalent to the data rates provided by second-generation cellular networks [HOLM -01]. For high data rates 1/3 rate turbo coding is recommended and typically brings performance benefits when large enough input block sizes are achieved. The channel coding schemes are defined in [3GPP TS 25.212] and are outlined here.
3.2.1.1 Convolutional coding

Convolutional codes with constraint length 9 and coding rates 1/3 and 1/2 are defined. Channel code block size is varied according to the data bit rates. The specified maximum code block size for convolutional coding is 504. If the number of bits in a Transmission Time Interval (TTI) exceeds the maximum code block size then code block segmentation is performed. In order to achieve similar size code blocks after segmentation, filler bits are added to the beginning of the first block.

![Diagram of block segmentation at the channel encoder]

Figure 3.2: Example of block segmentation at the channel encoder.

8 tail bits with binary value 0 are added to the end of the code block before encoding and initial values of the shift register are set to 0's when starting the encoding. The generator polynomials used in the encoding as given in [3GPP TS 25.212] are:

Rate 1/2 convolutional coder;

\[
\begin{align*}
G_0 &= 1 + D^2 + D^3 + D^4 + D^8 \\
G_1 &= 1 + D + D^2 + D^3 + D^5 + D^7 + D^8
\end{align*}
\]

Equation 3.1

Rate 1/3 convolutional coder,

\[
\begin{align*}
G_0 &= 1 + D^2 + D^3 + D^5 + D^7 + D^8 \\
G_1 &= 1 + D + D^3 + D^4 + D^7 + D^8 \\
G_2 &= 1 + D + D^2 + D^5 + D^8
\end{align*}
\]

Equation 3.2

Note that the UTRA uses two different sets of generator polynomials to achieve two different convolutional code rates. If \( K_i \) denotes the number of bits in the \( i \)th code block before encoding, then the number of bits after encoding, \( Y_i \), is

\[
Y_i = 2K_i + 16 \text{, with } \frac{1}{2} \text{ rate coding}
\]

\[
Y_i = 3K_i + 24 \text{, with } \frac{1}{3} \text{ rate coding}
\]
3.2.1.2 Turbo coding

Turbo codes employ two or more error control codes, which are arranged in such a way to enhance the coding gain. They have been demonstrated to have close approach to the Shannon capacity limit on both AWGN and Rayleigh fading channels. Traditionally, two parallel or serial concatenated recursive convolutional codes are used in the encoder implementation. Bit interleaver is used in between the encoders. Generated parity bit streams from two encoders are finally multiplexed to produce output turbo coded bit stream. Turbo decoding is carried out iteratively. The whole process results in code that has powerful error correction properties.

The defined turbo coder for use in UMTS, is a parallel-concatenated convolutional code with two 8-state constituent encoders and one turbo code internal interleaver. The coding rate of the turbo coder is 1/3. Figure 3.3 shows the configuration of the turbo coder.

The transfer function of the 8-state constituent code is defined as,

\[
G(D) = \begin{bmatrix}
1, & g_1(D) \\
g_0(D)
\end{bmatrix}
\]

where \( g_0(D) = 1 + D^2 + D^3 \),

\( g_1(D) = 1 + D + D^3 \)

Equation 3.3

Equation 3.4

Figure 3.3: Structure of rate 1/3 turbo coder [3GPP TS 25.212].
Chapter 3. Design and Development of UMTS Radio Access Simulator

The initial values of the shift registers are set to 0’s at the starting of the encoding. Output from the turbo code is read as $X_i$, $Z_i$, $Z'_i$, so on. Termination of the turbo coder defined in UMTS is performed in a different way to conventional turbo code termination, which uses 0 incoming bits to generate the trellis termination bits. Here, the shift register feedbacks after all information bits are encoded are used to generate the termination bits. To terminate the first constituent encoder, the switch A in Figure 3.3 is set to lower position while the second constituent encoder is disabled. Likewise, the second constituent encoder is terminated by setting the switch B in Figure 3.3 to a lower position while the first constituent encoder is disabled.

The turbo code internal interleaver arranges incoming bits into a matrix. If the number of incoming bits is less than the number of bits that the matrix could contain, padding bits are used. Then intra-row and inter-row permutations are performed according to the algorithm given in [3GPP TS 25.212]. Pruning is performed at the output, so the output block size is guaranteed to be equal to the input block size. If $K_i$ denotes the number of input bits in a code block, then the number of turbo code output bits $Y_i$ is $Y_i = 3K_i + 12$ for 1/3 code rate.

The minimum block size and the maximum block size for turbo coding are defined to be 40 bits and 5114 bits respectively. Data rates below 40 bits can be coded with turbo codes however, in such a case, dummy bits are used to fill the 40 bits minimum size interleaver. If the incoming block size exceeds the maximum size then segmentation is performed before the channel coding.

3.2.2 Rate matching

Rate matching is used to match the incoming data bits to available bits on the radio frame. Rate matching is achieved either by bit puncturing or repetition. If the number of incoming data is larger than the bits which can be accommodated in a single frame then bit puncturing is performed. Otherwise, bit repetition is performed. In the case of transport channel multiplexing, rate matching should take into account the number of bits arriving in other transport channels.

Rate matching algorithm depends on the channel coding applied. The corresponding rate matching algorithms for convolutional and turbo coding are defined in [3GPP TS 25.212]. In the simulated under discussion, rate matching is only performed for signalling bearer. As signalling data is protected using convolutional codes, the rate matching algorithm is implemented only for convolutional coding.


3.2.3 Interleaving

In UTRA, data interleaving is performed in two steps, the first and second interleaver. They are also known as inter-frame interleaving and intra-frame interleaving respectively. The first interleaving is a block interleaver with inter-column permutations (inter-frame permutation) and is used when the delay budget allows more than 10 ms of interleaving. In other words, the specified Transmission Time Interval (TTI), which indicates how often data arrives from higher layers to the physical layer, is larger than 10 ms. The TTI is directly related to the interleaving period and it could take values of 10, 20, 40 or 80 ms. Table 3.1 shows the inter-column permutation patterns for 1st interleaving. Each column contains data bits for 10 ms duration.

Table 3.1: Inter-columns permutation patterns for 1st interleaving [3GPP TS 25.212]

<table>
<thead>
<tr>
<th>TTI</th>
<th>Number of columns</th>
<th>Inter-column permutation patterns</th>
</tr>
</thead>
<tbody>
<tr>
<td>10 ms</td>
<td>1</td>
<td>&lt;0&gt;</td>
</tr>
<tr>
<td>20 ms</td>
<td>2</td>
<td>&lt;0,1&gt;</td>
</tr>
<tr>
<td>40 ms</td>
<td>4</td>
<td>&lt;0,2,1,3&gt;</td>
</tr>
<tr>
<td>80 ms</td>
<td>8</td>
<td>&lt;0,4,2,6,1,5,3,7&gt;</td>
</tr>
</tbody>
</table>

The second or intra-frame interleaving performs data interleaving within a 10 ms radio frame. This is also a block interleaver with inter-column permutations applied. Incoming data bits are input into a matrix with \( n \) rows and 30 columns, row by row with a starting position of column 0 and row 0. The number of rows is the minimum integer \( n \), which satisfies the condition:

\[
\text{Total number of bits in radio block} \leq n \times 30
\]

If the total number of bits in the radio block is less than that is necessary to fill the whole matrix, then bit padding is performed. The inter-column permutation for the matrix is performed based on the pattern shown in Table 3.2. Output is read out from the matrix column by column and finally, pruning is performed to remove padding bits that were added to the input of the matrix before the inter-column permutation.

Table 3.2: Inter-columns permutation patterns for 2nd interleaving [3GPP TS 25.212]

<table>
<thead>
<tr>
<th>Number of columns</th>
<th>Inter-column permutation patterns</th>
</tr>
</thead>
<tbody>
<tr>
<td>30</td>
<td>&lt;0,20,10,5,15,25,3,13,23,8,18,28,1,11,21,6,16,26,4,14,24,19,9,29,12,2,7,22,27,17&gt;</td>
</tr>
</tbody>
</table>
3.2.4 Spreading and scrambling

The spreading in the downlink is based on the channelisation codes and it is used to preserve the orthogonality among different downlink physical channels within one cell (or sector of a cell) and to spread the data to the chip rate, which is 3.84 Mcps. In UTRA spreading is based on the Orthogonal Variable Spreading Factor (OVSF) technique. The OVSF code tree is illustrated in Figure 3.4.

Typically only one OVSF code tree is used per cell sector in the base station (or Node B). The common channels and dedicated channels share the same code tree resources. The codes are normally picked from the code tree, however, there are certain restrictions as to which of the codes can be used for a downlink transmission. A physical channel can only use a certain code from the tree, if no other physical channel is using a code that is on an underlying branch. Neither can a smaller spreading factor code on the path to the root of the tree be used. This is because even though all codes from the same level are orthogonal to each other, two codes from different levels are orthogonal to each other only if one of them is not the mother code of the other. The radio network controller in the network, manages the downlink orthogonal codes within each base station.

Figure 3.4: Example of OVSF code tree used for Downlink.
Chapter 3. Design and Development of UMTS Radio Access Simulator

The spreading factor on the downlink may vary from 4 to 512 (an integer power of 2), depending on the data rate of the channel. Table 3.3 summarizes the channel bit rates, data rates and spreading factors for downlink dedicated physical channels.

### Table 3.3: Downlink Dedicated channel bit rates [extracted from HOLM-01]

<table>
<thead>
<tr>
<th>Spreading Factor</th>
<th>Channel bit rate (kbps)</th>
<th>User data rate with 1/2 rate coding (approx.)</th>
</tr>
</thead>
<tbody>
<tr>
<td>512</td>
<td>15</td>
<td>1-3 kbps</td>
</tr>
<tr>
<td>256</td>
<td>30</td>
<td>6-12 kbps</td>
</tr>
<tr>
<td>128</td>
<td>60</td>
<td>20-24 kbps</td>
</tr>
<tr>
<td>64</td>
<td>120</td>
<td>45 kbps</td>
</tr>
<tr>
<td>32</td>
<td>240</td>
<td>105 kbps</td>
</tr>
<tr>
<td>16</td>
<td>480</td>
<td>215 kbps</td>
</tr>
<tr>
<td>8</td>
<td>960</td>
<td>456 kbps</td>
</tr>
<tr>
<td>4, with 3 parallel codes</td>
<td>1920</td>
<td>936 kbps</td>
</tr>
<tr>
<td>4</td>
<td>5760</td>
<td>2.3 mbps</td>
</tr>
</tbody>
</table>

In addition to the spreading, a scrambling operation is performed in the transmitter. This is used to separate base stations (cell sectors) from each other. As the chip rate is already achieved with spreading, the symbol rate is not affected by scrambling. The downlink scrambling uses the Gold codes [3GPP TS 25.213]. The number of primary scrambling codes is limited to 512, simplifying the cell search procedure. The secondary scrambling codes are used in the case of beam steering and adaptive antenna techniques [SAUN-99].

### 3.2.5 Modulation

Quadrature Phase Shift Keying (QPSK) modulation is applied on time-multiplexed control and data streams on the downlink. Each pair of two consecutive symbols is serial-to-parallel converted and mapped on to I and Q branches. The symbols on I and Q branches are then spread to the chip rate by the same real-valued channelaisation code. The spread signal is then scrambled by a cell specific complex-valued scrambling code.

Figure 3.5 shows the spreading and modulation procedure for a downlink physical channel. A square-root raised cosine filter with a roll-off factor of 0.22 is employed for pulse shaping and the pulsed shaped signal is subsequently up-converted and transmitted.
Figure 3.5: Downlink modulation [3GPP TS 25.213].

3.2.6 Physical channel mapping

The frame structure for a downlink dedicated physical channel is shown in Figure 3.6. Each radio frame has 15 equal-length slots. The slot length is 2560 chips. As shown in Figure 3.6, the DPCCH and DPDCH are time multiplexed within the same slot [3GPP TS 25.211].

Each slot consists of pilot symbols, Transmit Power Control (TPC) bits, Transport Format Combination Indicator (TFCI) bits, and bearer data. The number of information bits transmitted in a single slot depends on the source data rates, the channel coding used, the spreading factor and the channel symbol rate. The exact number of bits in the downlink DPCH fields is given in [3GPP TS 25.211] and is summarised in Table 3.4.
Table 3.4: DPDCH and DPCCH fields (extracted from 3GPP TS 25.211)

<table>
<thead>
<tr>
<th>Spreading factor</th>
<th>DPDCH (Bits/Slot)</th>
<th>DPCCH (Bits/Slot)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>(N_{\text{data1}})</td>
<td>(N_{\text{data2}})</td>
</tr>
<tr>
<td>512</td>
<td>0</td>
<td>2</td>
</tr>
<tr>
<td>256</td>
<td>2</td>
<td>6</td>
</tr>
<tr>
<td>128</td>
<td>6</td>
<td>22</td>
</tr>
<tr>
<td>64</td>
<td>12</td>
<td>48</td>
</tr>
<tr>
<td>32</td>
<td>28</td>
<td>112</td>
</tr>
<tr>
<td>16</td>
<td>56</td>
<td>232</td>
</tr>
<tr>
<td>8</td>
<td>120</td>
<td>488</td>
</tr>
<tr>
<td>4</td>
<td>248</td>
<td>1000</td>
</tr>
</tbody>
</table>

### 3.2.7 Propagation model

![Figure 3.7: Four-path frequency selective fading channel.](image)

The channel model used in the simulator is the multi-path propagation model specified by IMT2000 in [UMTS-30.03]. This model takes into account that the mobile radio environment is dispersive, with several reflectors and scatterers. For this reason, the transmitted signal may reach the receiver via a number of distinct paths, each having different delays and amplitudes. The multipath fast fading is modelled by the superposition of multiple single faded paths with different arrival times and different average powers for specified power-delay profiles in [UMTS-30.03]. Each path is characterised by Rayleigh distribution (first order statistic) and classic Doppler spectrum (second order statistic).

Figure 3.7 shows a block diagram of a four-path frequency selective fading channel. UTRAN defines three different multipath power-delay profiles for use in different propagation environments. There are Indoor Office environments, Outdoor to Indoor and pedestrian...
environments and vehicular environments. All of these models are implemented in the simulator and the tapped-delay-line parameters for the vehicular environment are shown in Table 3.5. Mobile channel impulse response is updated 100 times for every coherence time interval.

Table 3.5: Vehicular A Test Environment [UMTS 30.03]

<table>
<thead>
<tr>
<th>Taps</th>
<th>Delay (nsec)</th>
<th>Power (dB)</th>
<th>Doppler Spectrum</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0</td>
<td>0</td>
<td>Classic</td>
</tr>
<tr>
<td>2</td>
<td>310</td>
<td>-1.0</td>
<td>Classic</td>
</tr>
<tr>
<td>3</td>
<td>710</td>
<td>-9.0</td>
<td>Classic</td>
</tr>
<tr>
<td>4</td>
<td>1090</td>
<td>-10.0</td>
<td>Classic</td>
</tr>
<tr>
<td>5</td>
<td>1730</td>
<td>-15.0</td>
<td>Classic</td>
</tr>
<tr>
<td>6</td>
<td>2510</td>
<td>-20.0</td>
<td>Classic</td>
</tr>
</tbody>
</table>

After the multi-path channel shown in Figure 3.1, white Gaussian noise is added to simulate the effect of overall interference in the system, including thermal noise and inter-cell interference.

### 3.2.8 Rake receiver

The rake receiver is a coherent receiver that attempts to collect the signal energy from all received signal paths that carry the same information. The rake receiver therefore can significantly reduce the fading caused by these multiple paths.

![Block diagram of a Rake receiver.](image)

The operation of the rake receiver follows three main operating principles (see Figure 3.8). First being the identification of the time delay positions at which significant energy arrives and time alignment of rake fingers for combining. The second step is the tracking of fast-changing phase and amplitude values originating from the fast fading process within each correlation receiver and removal of them from the incoming data. Finally, it combines the demodulated and phase-
adjusted symbols across all active fingers and passes them to the decoder for further processing. The combination can be processed using three different methods.

- **Equal Gain Combining (EGC)**, where the output from each finger is simply summed with equal gain.
- **Maximal Ratio Combining (MRC)**, where finger outputs are scaled by a gain proportional to the square-root of the signal to noise ratio of each finger before combining [VUCE-00]. In this case, finger output with higher power dominates in the combination.
- **Selective combining (SC)**, where not all finger outputs are considered in the combination, but only some fingers are selected according to the received power on each finger for combining.

Two types of rake receivers have been developed for the downlink.

1. Ideal rake receiver
2. Quasi-ideal rake receiver

The following provides details of the rake receiver design.

### 3.2.8.1 Ideal rake receiver

![Figure 3.9: Ideal Rake Receiver.](image)

The block diagram of an ideal rake receiver is shown in Figure 3.9. Here, perfect channel estimation and perfect finger time alignment is assumed. This is implemented by storing the entire fast fading channel coefficients as a complex vector, where the vector length equals the number of frequency selective fading paths. This vector is then fed from the channel directly to the ideal receiver. At the receiver, the coefficients for each path are first separated and then applied to each
rake finger after being time aligned in accordance with the delay (from channel delay-profile) in each reflected path. Out of three rake finger combination methods, EGC is selected for ideal receiver as it gives the optimal performance in the presence of ideal channel estimation and perfect time alignment.

3.2.8.2 Quasi-ideal rake receiver

![Diagram of Quasi-ideal Rake Receiver](image)

Figure 3.10: Quasi-ideal Rake Receiver.

The quasi-ideal rake receiver resembles the practical rake receiver in terms of implementation. However, as depicted in Figure 3.10, ideal finger search for the Rake receiver is assumed. That is, each finger in the receiver is assumed to have perfect synchronization with the corresponding path in the channel. First the received data is time aligned according to the channel delay-profile. Then data on each finger undergoes a complex-correlator process to remove the scrambling code and spreading code. As in an actual receiver implementation, pilot bits are used to estimate the momentary channel state for a particular finger. Channel estimation is achieved through the use of a matched filter, which is employed only during the period in which the pilot bits are being received. The pilot bits are send in every transmit time slot, therefore, the maximum effective channel updating interval is equivalent to half a slot. Output from the matched filter is further refined by using a complex FIR filter. Here, Weighted Multi-Slot Averaging (WMSA) technique as proposed in [HIGU-00] is employed to reduce the noise variance and also to track fast channel variation between consecutive channel estimates. Two different sets of weighting for the WSMA filter are used for low vehicular speeds and high vehicular speeds respectively. This is because, the limiting factor in channel estimation errors at low vehicular speed is the channel noise rather than the channel variations. Therefore, the noise averaging effect is more desirable at low vehicle speed. Whereas, at high vehicle speed, channel variation becomes the limiting factor hence, weighting based on interpolation should be considered. The WMSA technique requires a delay of
an integer number of time slots for the channel processing. The time-varying channel effect is
removed from the de-scrambled and de-spreaded signal before it is sent to the signal combiner.
Maximum Ratio Combining is used for the rake finger combination as it gives better performance.
Inter-symbol-interference due to the multi-path is implicit in the resulted output.

3.2.9 Channel decoding

In the implementation, a soft-decision Viterbi algorithm is used for decoding of the convolutional
codes. Turbo decoding is based on the standard LogMap algorithm (which is provided in SPW),
which tries to minimise bit error rate rather than the block error rate [VUCE-00]. 8 iterations are
performed.

3.3 Model verification for forward link

The theoretical formula for the BER probability with an order L MRC diversity combiner is given
in [OLMO-00] and is stated as,

\[ P_b = \frac{1}{2} \sum_{k=1}^{L} \pi_k \left[ 1 - \sqrt{1 + \gamma_k} \right] \]  

\[ \text{Equation 3.5} \]

where \( P_b \) is the bit error probability, \( L \) denotes the number of diversity path and \( \gamma_k \) is the mean
\( E_b / n \) for \( k^{th} \) diversity path. \( \pi_k \) is given by

\[ \pi_k = \prod_{i=1, i \neq k}^{L} \frac{\gamma_k}{\gamma_k - \gamma_i} \]  

\[ \text{Equation 3.6} \]

If the Rake receiver is assumed to behave as an order L MRC diversity combiner, then Equation
3.5 gives the lower bound of the BER performance. Other test conditions assumed in the Equation
3.5 are,

- perfect channel estimation,
- no inter-symbol-interference presence and
- each propagation path has a Rayleigh envelope.

Using Equation 3.5, the theoretical lower bound of the performance for the power delay profile
that is specified in the Case 3 outdoor performance measurement test environment in Annex B
[3GPP TS 25.101] is calculated and depicted in Figure 3.11. Here, a mean SNR value for each
individual path is calculated from the global $E_b/N_0$ by simply multiplying it with the fraction of power carried by each path (given in power delay profile). The number of rake fingers equal the number of propagation paths, which is 4 in this case.

Figure 3.11 shows the performance in terms of raw-ber (uncoded) for varying spreading factors in the above described test environment. A single active connection is considered. The dashed lines show the performance obtained by Olmos and Ruiz in [OLMO-00] for similar test conditions. Figure 3.11 clearly illustrates the close match of results obtained from the described model to those given in [OLMO-00]. As the spreading factor reduces, the performance deviates from that of the theoretical bound due to the presence of inter-symbol-interference.

For non-ideal channel estimation, raw-ber performance (see Figure 3.12) deviates considerably from the ideal channel estimation performance. Performance degradation is about 3-4 dB for operation at lower $E_b/N_0$ and increases gradually as $E_b/N_0$ increases. It should be emphasised here, that the channel-coding algorithm can correct almost all of the channel error occurrences if the raw-ber values are less than $10^{-2}$. Therefore, the region that is interesting for multimedia applications is limited to top left hand corner in Figure 3.12.
3.3.1 Model performance validation

Reference performance figures for the downlink dedicated physical channels are given in [3GPP TS 25.101]. These allow for the setting of reference transmitter and receiver performance figures for nominal error rates, sensitivity levels, interference levels and different propagation conditions. Reference measurement channel configurations are specified in Annex A [3GPP TS 25.101], while the reference propagation conditions are specified in Annex B [3GPP TS 25.101]. A mechanism to simulate the interference from other users and control channels in the downlink (named Orthogonal Channel Noise Simulator (OCNS)) on the dedicated channel, is shown in Annex C [3GPP TS 25.101]. The performance requirements are given in terms of Block Error Rate (BLER) for different multi-path propagation conditions and data rates (hence spreading factors) and channel coding schemes. For example, Table 3.6 is listed the required upper bound of BLER for the reference parameter setting shown in Table 3.7. The power-delay profile of the multi-path fading propagation condition used in the reference test is given in Table 3.8. Physical channel parameters, transport channel parameters, channel coding, and channel mapping for the 64 kbps reference test channel are depicted in Figure 3.13. As in a typical operating scenario, two transport channels, the data channel and the signalling channel, are multiplexed and mapped on to the same physical channel.
Table 3.6: BLER performance requirement [3GPP TS 25.101]

<table>
<thead>
<tr>
<th>Test Number</th>
<th>DPCH $E_{c}/I_{o_r}$</th>
<th>BLER</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>-11.8 db</td>
<td>$10^{-2}$</td>
</tr>
<tr>
<td>2</td>
<td>-8.1 db</td>
<td>$10^{-1}$</td>
</tr>
<tr>
<td></td>
<td>-7.4 db</td>
<td>$10^{-2}$</td>
</tr>
<tr>
<td></td>
<td>-6.8 db</td>
<td>$10^{-3}$</td>
</tr>
<tr>
<td>3</td>
<td>-9.0 db</td>
<td>$10^{-1}$</td>
</tr>
<tr>
<td></td>
<td>-8.5 db</td>
<td>$10^{-2}$</td>
</tr>
<tr>
<td></td>
<td>-8.0 db</td>
<td>$10^{-3}$</td>
</tr>
<tr>
<td>4</td>
<td>-5.9 db</td>
<td>$10^{-1}$</td>
</tr>
<tr>
<td></td>
<td>-5.1 db</td>
<td>$10^{-2}$</td>
</tr>
<tr>
<td></td>
<td>-4.4 db</td>
<td>$10^{-3}$</td>
</tr>
</tbody>
</table>

Table 3.7: Reference parameter setting [3GPP TS 25.101]

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Unit</th>
<th>Test 1</th>
<th>Test 2</th>
<th>Test 3</th>
<th>Test 4</th>
</tr>
</thead>
<tbody>
<tr>
<td>$I_{o_r}/I_{o_c}$</td>
<td>dB</td>
<td>-3</td>
<td>-3</td>
<td>3</td>
<td>6</td>
</tr>
<tr>
<td>$I_{o_c}$</td>
<td>dBm/3.84 MHz</td>
<td>-60</td>
<td>-60</td>
<td>-60</td>
<td>-60</td>
</tr>
<tr>
<td>Information data rate</td>
<td>kbps</td>
<td>12.2</td>
<td>64</td>
<td>144</td>
<td>384</td>
</tr>
</tbody>
</table>

DTCH

DCCH

Figure 3.13: Channel coding of DL reference measurement channel (64 kbps) [3GPP TS 25.101]
Table 3.8: power-delay profile for Case 3 test environment [3GPP TS 25.101].

<table>
<thead>
<tr>
<th>Relative delay[ns]</th>
<th>Average power [dB]</th>
<th>Fading</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>Classical-Doppler</td>
</tr>
<tr>
<td>260</td>
<td>-3</td>
<td>Classical-Doppler</td>
</tr>
<tr>
<td>521</td>
<td>-6</td>
<td>Classical-Doppler</td>
</tr>
<tr>
<td>781</td>
<td>-9</td>
<td>Classical-Doppler</td>
</tr>
</tbody>
</table>

3.3.1.1 Calculation of $E_b/N_0$ and $DPCH\_EC/I_{or}$

Reference test settings are given in terms of the ratio of energy per chip to the total transmit power spectral density of the Node B antenna connector. The relationship between $E_b/N_0$ and the setting of the variance ($\sigma$) of the AWGN source, and the conversion of $DPCH\_EC/I_{or}$ to $E_b/N_0$ is given in Equations 3.7 and 3.8 respectively. The derivations of these equations are given in Appendix B.

$$\frac{E_b}{N_0} = \frac{R_C \cdot ch\_os}{2 \cdot R_b \cdot \sigma^2}$$  \hspace{1cm} \text{Equation 3.7}

$$\frac{E_b}{N_0} = \frac{DPCH\_EC}{I_{or}} \cdot \frac{R_C \cdot ch\_os}{R_b \cdot \frac{I_{oc}}{I_{or}}}$$  \hspace{1cm} \text{Equation 3.8}

where, $R_C$ and $R_b$ are the chip rate and the channel bit rate respectively. $ch\_os$ denotes the channel over sampling factor. Equation 3.8 is equivalent to the equation proposed by Ericsson in [TSGR4-578].

3.3.1.2 UMTS DL model verification for convolutional code use

Reference interference performance figures [3GPP TS 25.101] for convolutional code with 12.2 kbps data were compared to the results obtained using the designed simulation model, which is referred to as the CCSR model. A comparison is given in Table 3.9. The reference results are given in terms of the upper bound of the average downlink power, which is needed to achieve the specified block error rate value.

The results listed in Table 3.9, show that the CCSR model performance is within the performance requirement limits in all propagation conditions. The performance requirements specified in 3GPP are limited to a single value, which is insufficient to test the model performance over a range of
propagation conditions. Therefore, performance tests were carried out for a range of $DPCH_{E_c/I_\text{or}}$ for different reference propagation environments, and the results are compared to those obtained by Ericsson [TSGR4-578] and NTT DoCoMo [TSGR4-581]. These results are shown in Figure 3.14.

Table 3.9: Performance validation for convolutional code use

<table>
<thead>
<tr>
<th>Propagation environment</th>
<th>$DPCH_{E_c/I_\text{or}}$ at BLER = 1%</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>3GPP -upper bound</td>
</tr>
<tr>
<td>AWGN</td>
<td>-16.6 dB</td>
</tr>
<tr>
<td>Case1</td>
<td>-15.0 dB</td>
</tr>
<tr>
<td>Case2</td>
<td>-7.7 dB</td>
</tr>
<tr>
<td>Case3</td>
<td>-11.8 dB</td>
</tr>
</tbody>
</table>

Figure 3.14 clearly illustrates that the close performance of the CCSR model to the results given in the above references. BLER curves for Case 1 and Case 3, are virtually identical to those given in the above references. In the Case 2 propagation environment, the CCSR model out-performs the other two reference figures. The reason for this may be the use of an EGC rake receiver in CCSR model. Case 2 represents an imaginary radio environment, which consists of three paths with large relative delays and equal average power. Use of EGC in this environment combines energy from all three paths with equal gain resulting in maximum power output and shows optimal performance. The path combiner structures used in the reference models are un-known. Performance over AWGN environment shows slight variation at low quality channels. However the performance gets closer to that of reference figure when channel quality gets better.
b). 12.2 kbps measurement channel over Case 1 environment

c). 12.2 kbps measurement channel over Case 2 environment
3.3.1.3 UMTS DL model verification for turbo code use

[3GPP TS 25.101] also presents the upper bounds for performance of turbo codes over different propagation conditions. The results generated with the CCSR model at a BLER of 10% and 1% were compared to the above reference figures and listed in Table 3.10 for the 64kbps test channel and in Table 3.11 for the 144kbps test channel.

Table 3.10: Performance validation for 64kbps channel

<table>
<thead>
<tr>
<th>Propagation environment</th>
<th>DPCH/Ior at BLER = 10%</th>
<th>DPCH/Ior at BLER = 1%</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>3GPP-upper bound</td>
<td>CCSR model</td>
</tr>
<tr>
<td><strong>AWGN</strong></td>
<td>-13.1 dB</td>
<td>-15.2</td>
</tr>
<tr>
<td>Case1</td>
<td>-13.9 dB</td>
<td>-15.0</td>
</tr>
<tr>
<td>Case2</td>
<td>-6.4 dB</td>
<td>-10.5</td>
</tr>
<tr>
<td>Case3</td>
<td>-8.1 dB</td>
<td>-10.9</td>
</tr>
<tr>
<td>DPCH/Ior at BLER = 1%</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>AWGN</strong></td>
<td>-12.8 dB</td>
<td>-14.95</td>
</tr>
<tr>
<td>Case1</td>
<td>-10.0 dB</td>
<td>-10.7</td>
</tr>
<tr>
<td>Case2</td>
<td>-2.7 dB</td>
<td>-7.5</td>
</tr>
<tr>
<td>Case3</td>
<td>-7.4 dB</td>
<td>-10.1</td>
</tr>
</tbody>
</table>
Table 3.11: Performance validation for 144kbps channel

<table>
<thead>
<tr>
<th>Propagation environment</th>
<th>DPCH/Ior at BLER = 10%</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>3GPP-upper bound</td>
</tr>
<tr>
<td>AWGN</td>
<td>-9.9 dB</td>
</tr>
<tr>
<td>Case1</td>
<td>-10.6 dB</td>
</tr>
<tr>
<td>Case2</td>
<td>-8.1 dB</td>
</tr>
<tr>
<td>Case3</td>
<td>-9.0 dB</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th></th>
<th>DPCH/Ior at BLER = 1%</th>
</tr>
</thead>
<tbody>
<tr>
<td>AWGN</td>
<td>-9.8 dB</td>
</tr>
<tr>
<td>Case1</td>
<td>-6.8 dB</td>
</tr>
<tr>
<td>Case2</td>
<td>-5.1 dB</td>
</tr>
<tr>
<td>Case3</td>
<td>-8.5 dB</td>
</tr>
</tbody>
</table>

The results listed in the tables clearly show that under the propagation conditions described in [3GPP TS 25.101], the performance of the CCSR model at a resulting BLER of 10% and 1% is within the required maximum limits. As for convolutional codes, performance for turbo codes are evaluated and compared to performance figures obtained by Ericsson [TSGR4-578] and NTT DoCoMo [TSGR4-581]. The performance traces under different conditions are shown in Figure 3.15 for 64 kbps and in Figure 3.16 for 144kbps. Dashed-dotted lines denote the results by Ericsson while dashed lines with star marks show the results by NTT DoCoMo.
b). 64 kbps measurement channel over Case 1 environment

c). 64 kbps measurement channel over Case 2 environment
Result for Case 1 is very close to the results given in the above references. Result over AWGN channel shows closer performance to Ericsson figures. As with the convolutional code, the CCSR model out-performs other two models in the operation over Case 2 propagation environment. However, for the 144kbps channel over Case 3, the CCSR model results do not closely follow the reference traces. Moreover, even the reference traces do not show close performance in this environment. There are several reasons for this behaviour. Case 3 resembles a typical outdoor vehicular environment. Mobile speed is set to 120 km/h in this condition. Therefore the effect of time varying multi-path channel conditions and inter-cell interference are more evident in this condition resulting in a variation in performances. Second, the implementation of the inter-cell interference could be different in each model. In the CCSR model, inter-cell interference is evaluated mathematically and is mapped on to the variance of the Gaussian noise source. Thirdly, the decoding algorithm used for turbo iterative decoding in the CCSR model is the LogMap algorithm, while the reference models use the MaxLogMap algorithm. Even though these two algorithms show similar performance for AWGN channels, when applied over multi-path propagation conditions, their performance depends on other conditions, such as the length of the turbo-internal interleaver, input block length and input signal amplitude [VUCE-00].
a). 144 kbps measurement channel over AWGN environment

b). 144 kbps measurement channel over Case 1 environment
Chapter 3. Design and Development of UMTS Radio Access Simulator

c). 144 kbps measurement channel over Case 2 environment

d). 144 kbps measurement channel over Case 3 environment

Figure 3.16: Comparison of reference performances for 144kbps channel.
3.4 UMTS physical link layer simulator

From a user point-of-view, services are considered end-to-end, this means from one terminal equipment to another terminal equipment. An end-to-end service may have a certain quality of service, which is provided for the user by the different networks. In UMTS, it is the UMTS bearer service that provides the requested QoS through the use of different QoS classes as defined in [3GPP TS 23.107]. At the physical layer, these QoS attributes are assigned a Radio Access Bearer (RAB) with specific physical layer parameters in order to guarantee quality of service over the air interface. RABs are normally accompanied by signalling radio bearers SRB. Typical parameter sets for reference RABs, signalling RBs and important combinations of them (down link, FDD) are presented in [3GPP TS 34.108]. In the simulation, 3.4 kbps SRB, which is specified in [3GPP TS 34.108], is used for the Dedicated Control Channel (DCCH). Transport channel parameters for the 3.4 kbps SRB are summarised in Table 3.12.

<table>
<thead>
<tr>
<th>RLC</th>
<th>Logical channel type</th>
<th>DCCH</th>
<th>DCCH</th>
<th>DCCH</th>
<th>DCCH</th>
</tr>
</thead>
<tbody>
<tr>
<td>RLC mode</td>
<td></td>
<td>UM</td>
<td>AM</td>
<td>AM</td>
<td>AM</td>
</tr>
<tr>
<td>Payload size, bit</td>
<td></td>
<td>136</td>
<td>128</td>
<td>128</td>
<td>128</td>
</tr>
<tr>
<td>Max data rate, bps</td>
<td></td>
<td>3400</td>
<td>3200</td>
<td>3200</td>
<td>3200</td>
</tr>
<tr>
<td>AMD/UMD PDU header, bit</td>
<td></td>
<td>8</td>
<td>16</td>
<td>16</td>
<td>16</td>
</tr>
<tr>
<td>MAC</td>
<td>MAC header, bit</td>
<td>4</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>MAC multiplexing</td>
<td></td>
<td>4 logical channel multiplexing</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Layer 1</td>
<td>TrCH type</td>
<td>DCH</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>TB sizes, bit</td>
<td>148</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>TTI, ms</td>
<td>40</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Coding type</td>
<td>CC 1/3</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>CRC, bit</td>
<td>16</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Max number of bits/TTI before rate matching</td>
<td>516</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Careful examination of parameter sets for RABs and SRBs, which are specified in [3GPP TS 34.108] shows that the minimum possible rate matching ratios for RABs and SRBs vary with the physical layer spreading factor being used. This is because, when a higher spreading factor is used, it adds transmission channel protection to the transmitted data in addition to the channel protection provided by the channel coding. Therefore, the channel bit error rate reduces with increase in spreading factor and it is possible to allow higher puncturing in these scenarios without loss of performance. Table 3.13 shows the variation of calculated minimum rate-
matching ratios (maximum puncturing ratio) with a spreading factor for FDD downlink control channels.

<table>
<thead>
<tr>
<th>SF</th>
<th>Minimum rate matching ratio</th>
</tr>
</thead>
<tbody>
<tr>
<td>128</td>
<td>0.690</td>
</tr>
<tr>
<td>64</td>
<td>0.73</td>
</tr>
<tr>
<td>32</td>
<td>0.99</td>
</tr>
<tr>
<td>16-4</td>
<td>1.0</td>
</tr>
</tbody>
</table>

Table 3.13: Minimum rate matching ratios for SRB

In the simulation, rate matching attributes for a SRB are set according to the minimum rate matching ratios shown in Table 3.13 for different physical channels. The actual information data rate is a function of spreading factor, rate-matching ratio, type of channel coding, channel coding rate, number of CRC bits and Transport Block size. Therefore the information data rates are calculated according to the simulation parameter settings and are shown in section.

Table 3.14 is a list of all the parameters that are user-definable, either by modifying the parameters of hierarchical models, by changing the building blocks that constitute the model, or by using different schematics.

Table 3.14: UTRAN simulator parameters

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>CRC attachment</td>
<td>24, 16, 12, 8 or 0</td>
</tr>
<tr>
<td>Channel coding scheme supported</td>
<td>No coding, ½ rate convolutional coding, 1/3 rate convolutional coding, 1/3 rate turbo coding</td>
</tr>
<tr>
<td>Interleaving</td>
<td>1st interleaving: Block interleaver with inter-frame permutation</td>
</tr>
<tr>
<td></td>
<td>2nd interleaving: Block interleaver with inter-columns permutation</td>
</tr>
<tr>
<td></td>
<td>[Permutation patterns are specified in 3GPP TS 25.212]</td>
</tr>
<tr>
<td>Rate matching</td>
<td>The algorithm (for convolutional rate matching) as specified in 3GPP TS 25.212. Rate matching ratio (repeat or puncturing ratio) is user definable.</td>
</tr>
<tr>
<td>TrCH multiplexing</td>
<td>Experiments were conducted for two Transport Channels.</td>
</tr>
<tr>
<td>Transport format detection</td>
<td>TFCI based detection</td>
</tr>
<tr>
<td>Spreading factor</td>
<td>512, 256, 128, 64, 32, 16, 8, 4</td>
</tr>
<tr>
<td>Transmission Time interval</td>
<td>10 ms, 20 ms, 40 ms, 80 ms</td>
</tr>
<tr>
<td>Pilot bit patterns</td>
<td>As specified in 3GPP TS 25.211</td>
</tr>
<tr>
<td>Interference/Noise Characteristics</td>
<td>User defined values are converted to the variance of AWGN source at receiver.</td>
</tr>
<tr>
<td>Fading Characteristics</td>
<td>Rayleigh fading mobile channel impulse response is updated 100 times for every coherence time interval.</td>
</tr>
<tr>
<td>Multipath Characteristics</td>
<td>Vehicular, Pedestrian</td>
</tr>
<tr>
<td>Mobile terminal velocity</td>
<td>User definable. Constant for the simulation run.</td>
</tr>
</tbody>
</table>
### Chapter 3. Design and Development of UMTS Radio Access Simulator

<table>
<thead>
<tr>
<th>Chip rate</th>
<th>3.84 Mcps</th>
</tr>
</thead>
<tbody>
<tr>
<td>Carrier Frequency</td>
<td>2000 MHz</td>
</tr>
<tr>
<td>Antenna Characteristics</td>
<td>0 dB gain for both transmitter and receiver antenna.</td>
</tr>
<tr>
<td>Receiver characteristics</td>
<td>Rake receiver with Maximum ratio combining, Equal gain combining or Selective combining. Number of rake fingers is user definable.</td>
</tr>
<tr>
<td>Transmission Diversity</td>
<td>Closed loop fast power control [3GPP TS 25.214]</td>
</tr>
<tr>
<td>Channel decoding</td>
<td>Soft-decision Viterbi convolutional decoder, Standard LogMap turbo decoder, Number of turbo iteration is user definable.</td>
</tr>
<tr>
<td>Performance measures</td>
<td>Bit Error Patterns and Block Error Patterns</td>
</tr>
<tr>
<td>Simulation length</td>
<td>3000 – 6000 Radio Frames equivalent to 30-60 sec duration</td>
</tr>
</tbody>
</table>

#### 3.4.1 BLER/BER performance of simulator

Simulations were carried out for different radio bearer configuration settings. For higher spreading factor realisations, the simulation period was set to 60 seconds duration. However for spreading factor 8, the simulation period was limited to 30 seconds. This is to compensate the higher processing time requirement seen at high data rates. 6000 – 3000 radio frames (10ms) accommodate about 20000 – 15000 RLC blocks in a generated bit error sequence and that is sufficiently long enough to obtain a meaningful BLER average. Furthermore, experimental results show that the selected simulation duration is sufficient to capture the bursty nature of the wireless channel and its effect on the perceptual quality of received video.

#### 3.4.1.1 Effect of spreading factor

For the purpose of a performance comparison of the effect of spreading factor variation, experiments were conducted for various spreading factor allocations. The other physical channel parameters are set to their nominal values, which are shown in Table 3.15. The calculated possible information data rates are based on the specified SRBs for a given composite transport channel (consisting of one signalling channel and one dedicated data channel) for FDD downlink channels and are presented in Table 3.16. Table 3.16 also lists the RLC payload setting used in different bearer configurations.
Table 3.15: Nominal parameter settings

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Spreading factor</td>
<td>32</td>
</tr>
<tr>
<td>Transmission Time Interval</td>
<td>20 ms</td>
</tr>
<tr>
<td>CRC attachment</td>
<td>16 bits</td>
</tr>
<tr>
<td>Channel coding</td>
<td>½ CC, 1/3 CC, 1/3 TC</td>
</tr>
<tr>
<td>Mobile speed</td>
<td>3 km/h, 50 km/h</td>
</tr>
<tr>
<td>Rate matching ratio</td>
<td>1.0</td>
</tr>
<tr>
<td>Operating environment</td>
<td>Vehicular A, Pedestrian B</td>
</tr>
</tbody>
</table>

Table 3.16: Channel throughput characteristics.

<table>
<thead>
<tr>
<th>Spreading factor</th>
<th>Convolutional coding (1/2 rate)</th>
<th>Convolutional coding (1/3 rate)</th>
<th>Turbo coding (1/3 rate)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>RLC payload</td>
<td>Rate(kbps)</td>
<td>RLC payload</td>
</tr>
<tr>
<td>128</td>
<td>45</td>
<td>15.75</td>
<td>49</td>
</tr>
<tr>
<td>32</td>
<td>320</td>
<td>96.0</td>
<td>320</td>
</tr>
<tr>
<td>16</td>
<td>320</td>
<td>192.0</td>
<td>320</td>
</tr>
<tr>
<td>8</td>
<td>640</td>
<td>416.0</td>
<td>640</td>
</tr>
<tr>
<td>4</td>
<td>640</td>
<td>896.0</td>
<td>640</td>
</tr>
</tbody>
</table>

Figure 3.17 shows the BER performance for the transmission of un-coded data (raw ber/ channel ber) over Vehicular A propagation environment. It clearly illustrates the error-flow characteristics experienced due to the inter-symbol-interference in multi-path channels. The effect of inter-symbol-interference increases with a reduction of spreading factor. However, the error-flow characteristic is not very pronounced in terms of coded BER performance, apart from with very low spreading factor allocations (see Figure 3.18). This is due to the effect of the channel coding algorithm, which tends to correct most of the errors if the channel bit error rate is lower than $10^{-2}$.

Figure 3.17: Performance of un-coded channel over Vehicular A environment.
Figure 3.18: Spreading factor effect for Vehicular A environment; a). 1/3 rate convolutional code b). 1/3 rate turbo code.

Figure 3.18 (a) shows the performance of convolutional code, while the performance of turbo code is shown in Figure 3.18 (b). The effect of spreading factor variation on the performance of turbo codes shows a similar behaviour to that of convolutional code. However, the performance for spreading factor 8 shows closer performance to that of other spreading factors compared to the convolutional coding case. This is mainly due to the behaviour of turbo codes. It is known that the
higher the input block size of the turbo code, the better the performance. A high bit rate (with low spreading factor) service can accumulate more bits in a TTI than a low bit rate service. The better performance of the turbo code seen with large input block sizes compensates for the reduced robustness against interference provided by low spreading factor realisations. In Figure 3.18 (a) & (b), the performance for 128 spreading factor is worse than that for 16 and 32 spreading factors. A possible reason is the poor performance of the interleavers (the first interleaver in convolutional coding and the first interleaver and the turbo internal interleaver in turbo coding) in the presence of smaller input block sizes. Furthermore, experimental results (shown in Appendix C) for the half-rate convolutional code also show similar performance.

3.4.1.2 Effect of channel coding

Figure 3.19: Effect of channel coding scheme a) BER b). BLER performance.
Figure 3.19 illustrates the effect of channel coding schemes on the block error rate and bit error rate performances. Vehicular A channel environment is considered as the test environment, while the spreading factor is set to 32. As expected, turbo coding shows better performance than that of the other channel coding schemes, while the 1/3 rate convolutional code out-performs the 1/2 rate convolution code. It must be emphasised that the plots shown are the BLER/BER performances vs $E_b/N_0$. If the BLER/BER performances are viewed vs transmitted power, significant improvements will be visible for the 1/3 rate-coding scheme compared to the 1/2 rate-coding scheme. This is because, the transmit power is directly proportional to the source bit rate. As 1/2 rate coding supports higher source rates, the corresponding curve will be shifted more to the right compared to others. Furthermore, the convolutional code and the turbo code show closer BLER performance (shown in Figure 3.19 (b)) compared to the BER performance shown in Figure 3.19(a). This is due to the properties of the implemented LogMap algorithm at the turbo decoder, which is optimised to minimise the number of bit errors rather than block error rate [VUCE-00].

3.4.1.3 Effect of channel environment

![Graph showing BLER vs E_b/N_0 for different spreading factors.]

Figure 3.20: 1/3 rate convolutional coding performance for the Pedestrian B environment.

Experiments were conducted to investigate the BLER performance for the Pedestrian B channel environment. The mobile speed is set to 3 km/h. Results for 1/3 rate convolutional code with different spreading factors are shown in Figure 3.20. As is evident from the figure, the resulting
performance over the Pedestrian B channel is much lower than that over the Vehicular A channel environment when operating without fast power control. This is due to the slow channel variation associated with low mobile speeds. A larger number of consecutive information blocks can experience a long weaker channel condition during the transmission. This reduces the performance of block based de-interleaving and channel decoding algorithms resulting in a high block error rate. On the other hand, a faster Doppler effect results in alternating weak and strong channel conditions of short durations at high vehicular speeds. This effect behaves as a time domain transmit diversity technique and enhances the performance of block based interleaving and channel coding algorithms. The employment of a fast power control algorithm and the enhancement of BLER performance at low vehicular speeds are discussed in Section 3.6.1.

3.4.2 $E_b/N_0$ to $E_b/I_0$ and C/I conversion

The BER performance of UMTS-FDD systems depends on many factors, such as mean bit energy of the useful signal, thermal noise, and interference. Interference can be divided into three main parts as inter-symbol-interference, intra-cell interference, and inter-cell interference. Due to the multiple receptions, the signal is received with significant delay spread in a multi-path propagation environment. This causes the inter-symbol-interference. Orthogonal codes are used to separate users in the downlink. Without any multi-path propagation, these codes can be considered as perfectly orthogonal to each other. However, in a multi-path propagation environment, orthogonality among spreading codes deviates from perfection, due to the presence of delay spread of the received signal. Therefore, the mobile terminal sees part of the signal, which is transmitted to other users as interference power and it is named as intra-cell interference. The interference power seen among users in neighbouring cells is quantified as the inter-cell interference.

BER performance is commonly written as a function of the global $E_b/\eta$, where the definition is given as:

$$\frac{E_b}{\eta} = \frac{E_b}{N_0 + \chi + (1 - \rho) \cdot I_o + \eta_{ISI}}$$  \hspace{1cm} \text{Equation 3.9}$$

$$E_b = \left( \frac{E_b}{N_0} \right)^{-1} + \left( \frac{E_b}{\chi} \right)^{-1} + (1 - \rho) \cdot \left( \frac{E_b}{I_o} \right)^{-1} + \left( \frac{E_b}{\eta_{ISI}} \right)^{-1}$$  \hspace{1cm} \text{Equation 3.10}$$

where

$E_b$ - Received energy per bit of the useful signal
Chapter 3. Design and Development of UMTS Radio Access Simulator

\( N_0 \) - The power density representing the system generated thermal noise.
\( \eta \) - The global noise power spectrum density
\( \chi \) - The inter-cell interference power spectral density
\( \rho \) - The orthogonality factor
\( I_o \) - The intra-cell interference power spectral density
\( \eta_{ISI} \) - The power spectral density of the inert-symbol-interference of the received signal

These factors depend on

- the operating environment,
- the number of active users per cell,
- the used spreading factors in the code tree,
- the cell site configurations,
- presence of the diversity techniques,
- the mobile user locations
- the type of radio bearer and
- the voice activity factor.

The loss of orthogonality between simultaneously transmitted signals on a WCDMA downlink is quantified by the Orthogonality Factor (OF). The lower the value of the OF, the smaller the interference; an OF of 1 corresponds to the perfect orthogonal case while an OF near 0 indicates considerable downlink interference. The introduction of the orthogonality factor in modelling intra-cell interference allows the employment of the Gaussian hypothesis. It is employed, where the equivalent Gaussian noise with power spectral density, is equal to \((1-\rho)\) times the received intra-cell interference power, and is simply added at the receiver input. The statistic of the orthogonality factor is normally derived from measurement data gathered through extensive field trial campaigns. In the designed UTRA down link simulator, the orthogonality factors, which are derived based on the gathered channel data and are presented in [HUNU-02], are used to simulate the intra-cell interference power.

Table 3.17: Orthogonality factor variation for different cellular environments [HUNU-02]

<table>
<thead>
<tr>
<th>Environment</th>
<th>Code orthogonality factor</th>
</tr>
</thead>
<tbody>
<tr>
<td>Urban Small</td>
<td>Mean 0.571</td>
</tr>
<tr>
<td>Urban Large</td>
<td>Std 0.169</td>
</tr>
<tr>
<td>Rural Large</td>
<td>Mean 0.514</td>
</tr>
<tr>
<td></td>
<td>Std 0.212</td>
</tr>
<tr>
<td></td>
<td>Mean 0.626</td>
</tr>
<tr>
<td></td>
<td>Std 0.190</td>
</tr>
</tbody>
</table>
The inter-cell interference can also be modelled with the Gaussian hypothesis. The inter-cell interference power spectral density and the intra-cell interference power spectral density can be explicitly obtained through system level simulations or analytical calculations based on simplifying assumptions and cell configuration [HOLM-01]. However, inter-symbol-interference can only be obtained by chip-level simulation and it does not depend on other factors apart from used spreading factor, propagation condition and mobile speed. Therefore, it is sufficient to obtain the $E_b/\eta$ performance for one single connection ($\chi = 0, I_o = 0$) of each of all the possible bit rates or spreading factors by chip level simulation. Then the $E_b/I_o$ performances can easily be derived from $E_b/\eta$ using Equation 3.11. $N_o << (1-\rho)\cdot I_o$ is assumed and the inter-symbol-interference is implicit in the simulation.

$$\frac{E_b}{I_o} = (1-\rho) \cdot \frac{E_b}{\eta}$$

Equation 3.11

Equation 3.12 shows relationship between average energy per bit and average received signal power, $S$.

$$S = E_b \cdot R$$

Equation 3.12

where $R$ denotes data bit rate.

Therefore,

$$SIR_{\chi} = R \cdot \frac{E_b}{\chi}$$

Equation 3.13

$$SIR_I = R \cdot \frac{E_b}{I_o} = (1-\rho) \cdot R \cdot \frac{E_b}{N_o}$$

Equation 3.14

$$SIR_{Total} = \left[ SIR_{\chi}^{-1} + SIR_I^{-1} \right]^{-1}$$

Equation 3.15

where $SIR_{\chi}, SIR_I$ and $SIR_{Total}$ denote signal to inter-cell interference ratio, signal to intra-cell ratio and signal to interference ratio respectively.

Assume $N_o = 0.0002$, $\chi = 0.005$ and Vehicular A propagation environment. From Table 3.17, the orthogonality factor is 0.514. The $E_b/I_o$ for $E_b/\eta$ values shown in Figure 3.18.(a) are calculated from Equation 3.9 and are shown in Figure 3.21.
3.5 Performance enhancement techniques

3.5.1 Fast power control for down link

As can be seen in Figure 3.20, data transmission over the Pedestrian B slow-speed propagation environment shows worse performance compared to transmission over the high-speed propagation environment. This is mainly because the error-correcting methods are based on the interleaving and block based methods, which do not work effectively in the presence of long duration weak channel conditions caused by Rayleigh fading at low mobile speeds. This condition (weak long radio link) can be improved by the application of a fast power control algorithm. A closed-loop fast power control algorithm is designed and is incorporated in the simulator. The implementation and the resulting performance improvement are described below.

3.5.1.1 Algorithm implementation

A block diagram representation of the implemented power control algorithm is depicted in Figure 3.22. According to the measured received pilot power at the receiver, the UE generates
appropriate Transport Power Control (TPC) commands (whether to adjust transmit power up or
down) to control the network transmit power and sends them in the TPC field of the uplink
Dedicated Physical Control CHannel (DPCCH). The TPC command decision is made by
comparing the average received pilot power (averaged over an integer number of slots to mitigate
the effects of varying interference and noise) to the pilot power threshold, which is pre-defined by
the UTRAN, based on the outer-loop power control [3GPP TS 25.104]. Upon receiving the TPC
commands, UTRAN adjusts its downlink DPCCH/DPDCH power according to Equation 3.16.

\[ P(k) = P(k-1) \pm \Delta_{TPC} \]  

Equation 3.16

Where, \( P(k) \) denotes the downlink transmit power in \( k \)th slot, \( \Delta_{TPC} \) is the power control step size.
\( \pm \) is decided from the uplink TPC command. This algorithm is executed at a rate of 1500 times
per second for each mobile connection. Settings used in the implementation are listed in Table
3.18.

<table>
<thead>
<tr>
<th>Power control step size</th>
<th>0.5dB ± 0.25dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>Aggregated power control step change</td>
<td>4-6 dB</td>
</tr>
<tr>
<td>Power averaging window size, n</td>
<td>4</td>
</tr>
<tr>
<td>Feedback delay</td>
<td>3 slots</td>
</tr>
<tr>
<td>Algorithm execution frequency</td>
<td>1.5 kHz</td>
</tr>
</tbody>
</table>

Table 3.18: power control parameter settings [Extracted from 3GPP TS 25.104]

Note: the aggregated power control step is defined as the required total changes in a code channel
in response to ten multiple consecutive power control commands.

The control algorithm adjusts the power of the DPCCH and DPDCH, however the relative power
difference between the two is not changed.
3.5.1.2 Power control algorithm performance

Figure 3.23: Characteristic of fast power control algorithm

Figure 3.23 shows how a downlink closed-loop power control algorithm works on a fading channel at low vehicular speed. Node B transmit power varies inversely proportional to the received pilot power. This closely resembles the time varying channel at low mobile speeds. Transmit power cut off values are defined by the maximum and minimum power limits set by the Node B. Receive power at the receiver shows very little residual fading.

Figure 3.24 illustrates the performance of the power control algorithm for data transmission over the Vehicular A propagation environment with a spreading factor 16 and 1/3 rate convolutional coding. The experiment was carried out at three different mobile speeds settings namely 3, 50 and 120 km/h. The performance improvement by power control is evident at low speed, while at high mobile speed, the improvement is most insignificant. This is because a transmission diversity gain is provided by the highly time varying channel at high mobile speed.

Simulation results for different combinations of channel coding schemes, spreading factors and channel environments (Vehicular A, Pedestrian B) with fast power control for a UTRA-FDD down link are shown in Appendix C.
3.6 Radio interface data flow model

The designed physical link layer simulator alone provides a necessary experimental platform to examine the effects of the radio link upon the data transmitted through the physical channel. However, in order to investigate the effect of channel bit errors upon the end-application, the application performance must be validated in an environment as close as possible to that of the real world. Therefore, not only the effect of the physical link layer but also the effect of UMTS protocol layer operation on multimedia performance should be investigated. A UMTS data flow model was designed in Microsoft Visual C++ to emulate the protocol layer behaviour. The design criterion follows a modular design strategy. Each of the protocol layers was implemented separately and protocol interaction is performed through the specified interfaces. This allows individual protocol layer optimisation or improvement and testing of novel performance enhancement algorithms in the presence of a complete system.

In this section, only the protocol layer effect on multimedia performance is considered. The protocol layers implemented include the application layer, transport layer, PDCP layer, RLC/MAC layer and Layer 1. The block diagram of the data flow model is shown in Figure 3.25.
The presence of protocol headers and their effects on application performance were emulated by allocating dummy headers.

The application consists of a full error-resilience enabled MPEG-4 video source. In addition to the employed error-resilience techniques, the TM5 rate control algorithm is used in order to achieve a smoother output bit rate. An Adaptive Intra Refresh algorithm, as described in Section 2.3.3.5, is also implemented to stop temporal error propagation and to achieve a smoother output bit rate. The output source bit rate is set according to the guaranteed bit rate, which is a user defined QoS parameter. For packet switched connections, encoded video frames are forwarded to the transport layer at regular intervals, as defined by the video frame rate. Each video frame is encapsulated into an independent RTP/UDP/IP packet [RFC-1889] for forwarding down to the PDCP layer. The PDCP exists mainly to adapt transport layer packet to the radio environment by compressing headers with negotiable algorithms [3GPP TS 25.323]. The current version of the data flow model, implements the resulting compressed header sizes, but not the actual header compression.

Figure 3.25: UTRAN data flow model.
algorithms. More extensive and detailed examination of different header compression algorithms and their effect on multimedia performance can be found in [CELL-03]. For circuit switched connections, the output from the video encoder is directly forwarded to the RLC/MAC layer.

At the RLC layer, forwarded information data is further segmented into RLC blocks. The size of the RLC block is defined by the Transport Block (TB) size, which is an implementation dependent parameter. As explained in Section 4.4.1, optimal setting of TB size should include many factors such as, application type, source traffic statistic, allowable frame delay-jitter, and RLC buffer size. RLC block header size depends on the selected RLC mode for the transmission. Transparent mode adds no header as it transmits higher layer payload units transparently. Unacknowledge mode and acknowledgement mode add 8 bits and 16 bits headers to each RLC block respectively [3GPP TS 25.322]. Apart from the segmentation and addition of a header field, other RLC layer functions such as error detection and retransmission of erroneous data are not implemented in the current version of the model as the main use of the model is to investigate the performance of conversational type multimedia applications. The MAC layer can be either dedicated or shared. A dedicated mode is responsible for handling dedicated channels allocated to a UE in connected mode, while shared mode takes responsibility for handling shared channels. If channel multiplexing is performed at the MAC layer, then a 4 bit MAC header is added to each RLC block [3GPP TS 25.321].

Layer 1 attaches a CRC to forwarded RLC/MAC blocks. According to the specified TTI length, higher layer blocks are combined to form the TTI blocks and store them in a buffer for further processing before transmitting over the air interface. The number of higher layer PDUs to be encapsulated within the TTI block depends on the selected channel coding scheme, the spreading factor and the rate-matching ratio. In a practical system, the selection of channel coding scheme, TTI length and CRC size are normally performed by the radio resource management algorithm according to the end user quality requirement, application type, operating environment, system load, and so on. For experimental sake, these parameters are to be user definable in the designed data flow model. An error prone radio channel environment is emulated by applying generated bit errors from the physical link layer simulator to the information data at Layer 1.

The receiver side is emulated by reversing the described processing. Layer 1 segments the TTI blocks received over the simulated air interface in to RLC/MAC blocks. After detaching CRC bits, RLC/MAC blocks are passed on to the RLC/MAC layer. At the RLC/MAC layer the received data is reassembled in to PDCP data units for packet switched connections. If
Chapter 3. Design and Development of UMTS Radio Access Simulator

IP/UDP/RTP headers are detected to be corrupted, data encapsulated within that packet is dropped at the network layer. Finally the received source data is displayed using an MPEG-4 decoder.

This layered implementation of the UMTS protocol architecture allows the investigation of the effect of physical layer generated bit errors upon different fields of the payload data units at each protocol layer. In other words, the data flow model can be used to map channel errors on to different PDU fields and to optimise protocol performance for the given application.

3.7 Real-time emulator

![Diagram of UMTS emulator architecture.](image)

The above-described UMTS data flow model is integrated with the physical link layer model to form the UTRAN emulator. The emulator software suite provides a graphical user interface for connection set-up, radio bearer configuration and performance monitoring. The emulator model considers the emulated system to be a black box, whose input-output behaviour intends to
reproduce the real system without requiring knowledge of the internal structure and processes. It has also being designed for accurate operation in real time with moderate implementation complexity. The emulator was implemented in Visual C++, as it provides a comprehensive graphical user interface design environment.

Figure 3.26 depicts the block diagram of the designed emulator architecture. It consists of three main parts namely, content server, UMTS emulator and mobile client. An "MPEG-4 file transmitter" is used as the content server. It selects the corresponding video sequence, which is encoded according to the requested source bit rate, frame rate and other error resilience parameters and transmits the video to the UMTS radio link emulator. At the emulator the received source data passes through the UMTS data flow model and the simulated physical link layer, and is finally transmitted to the mobile client for display. Here, a PC based MPEG-4 decoder is used to emulate the display capabilities of the mobile terminal.

![Figure 3.27: QoS parameter option page.](image)

The UMTS configuration options dialog box is designed for interactive radio bearer configuration for a particular connection. The QoS parameter page shows the user requested quality of service parameters, such as type of service, traffic class, data rates, residual bit error rate and transfer
delay. In addition, operator control parameters, connection type, PDCP connection type and the number of multiplexed transport channels are shown.

The transport channel parameter page for the data channel shows the transport channel related network parameter settings (see Figure 3.28). Logical channel type, RLC mode, MAC channel type, MAC multiplexing, and Layer 1 parameters, TTI, Channel coding scheme, CRC are user definable emulator parameters, while TB size and Rate matching ratio are calculated and displayed from the other input parameter values. If the number of multiplexed transport channels is set to 1, then the transport channel parameter page for the control channel is disabled. Otherwise it shows the transport channel parameters that are related to the control channel.

The appropriate spreading factor for transmission is calculated based on the requested QoS parameters and it is displayed on the physical/Radio channel parameter page (see Figure 3.29). Radio channel related settings (carrier frequency, channel environment, mobile speed) and receiver characteristics (number of rake fingers, rake combining, diversity techniques, power control) are to be selected on the Physical/Radio parameter page.
Figure 3.29: Physical/Radio Channel Parameter option page.

Figure 3.30 illustrates the user interfaces of the designed emulator. In addition to the radio bearer configuration parameter pages described so far, the emulator also displays the instantaneous performance in terms of Eb/No, Eb/Io, C/I and BER. Furthermore, it allows interactive manipulation of the number of users in the cell, (hence co-channel interference) and monitoring of the video performance in a more realistic operating environment.

3.8 Conclusion

This chapter has described the design and evaluation of a UMTS-FDD simulator for the forward link. The CCSR model gives performances that satisfy the requirements shown in 3GPP performance figures. Furthermore, the performance of the CCSR model closely follows the performance traces published by different terminal manufactures on most test configurations. However, some performance variation was visible for operation over the Case 2 propagation environment and 144kbps reference channel over the Case 3 test environment. As mentioned earlier, there are several possible reasons, which could contribute to this. The most likely reason is the different implementation strategies followed in the receiver design and in channel decoding. Other contributors are the different simulation techniques used for propagation modelling and interference modelling. The differences seen in the performances of turbo codes is greater
compared to the performances of convolutional codes, where the coding /decoding technology is fairly stable and consolidated. In addition, the performance of the LogMap algorithm implementation (provided in the SPW package) is highly sensitive to the amplitude of the decoder input signal. The input amplitude setting was based on the conducted experimental results, and may cause slight performance degradation.

Even though, the CCSR model matches the reference performance traces or comes very close to them under the particular bearer configurations tested, the bit error sequences generated by the CCSR model should be considered, to a certain extent, as worst case performance figures. The designed quasi-ideal Rake receiver employs non-ideal channel estimation based on Weighted Multi-Slot Averaging (WMSA) techniques. The settings of the WSMA filter parameters might not be considered as optimal settings in varying simulation environments. Also the employment of
advanced power control techniques can result in improved performances compared to the less complex fast power control algorithm implemented. In fact, the CCSR model shows about 3 dB performance loss compared to the results published by Olmos [OLMO-02] using a non-ideal Rake receiver and fast power control.

The intention of the UMTS physical layer simulation model was to facilitate investigation of multimedia performance over the UMTS air interface. Although, the performance test results were presented as block error rate values, the output of the physical link simulator produces bit error sequences to characterise the actual physical link layer. After integrating with the UMTS protocol data flow model, the physical link layer generated bit error patterns can be used in multimedia transmission experiments as described in the following chapters. In addition to the high degree of correlation shown between the performance of the CCSR model and the quoted reference figures, generated bit error patterns are nevertheless suitable for employment in a radio bearer optimisation for multimedia communication, as they exhibit relative differences between various network parameter and interference settings, despite the type of receiver architecture implemented.
Chapter 4

4 Real-time Video Communications over UTRAN

4.1 Introduction

UMTS access networks were designed from the outset to provide a wide range of bearer services with different levels of quality of service suitable for multimedia applications with bit rates of up to 2Mbit/s. The bearer services are characterised by a set of transport channel parameters, which include:

- transport block size
- CRC code length
- channel coding schemes
- RLC mode
- MAC type
- transport time interval
- rate matching
- spreading factor.
The perceived quality of the application seen by the end user is greatly affected by the settings of these transport channel parameters. The optimal parameter settings depend highly on the characteristics of the application, the propagation conditions, and the end user QoS requirements. This Chapter will examine the effect of these transport channel (network) parameter settings upon the performance of MPEG-4 coded video telephony applications and will investigate the optimal radio bearer design for real-time video transmission over UTRAN. The influence of the network parameter settings and different channel and interference conditions upon the received video quality and network performance will be assessed experimentally using the real-time UMTS emulator described in Chapter 3. Furthermore, differences between packet-switched and circuit-switched radio bearer configurations for conversational video applications will be investigated.

4.2 3G real-time video requirements

The most challenging form of communication class in terms of application requirement, is the conversational service class. The real time conversational scheme is characterised by two main requirements: very low end-to-end delay and the preservation of time relations between information entities in the stream. The maximum end-to-end delay is decided by human perception. Therefore, the limit for acceptable delay is very strict, as failure to provide sufficiently low delay will result in unacceptable quality. Conversational traffic is symmetric or nearly symmetric in nature. The characteristics of the conversational applications as specified in [3GPP TS 22.105] are shown in Table 4.1.

<table>
<thead>
<tr>
<th>Application example</th>
<th>Videophone</th>
</tr>
</thead>
<tbody>
<tr>
<td>Degree of symmetry</td>
<td>Two-way</td>
</tr>
<tr>
<td>Data rates</td>
<td>32-384 kbps</td>
</tr>
<tr>
<td>One-way end-to-end delay</td>
<td>&lt; 150 msec preferred</td>
</tr>
<tr>
<td></td>
<td>&lt; 400 msec limit</td>
</tr>
<tr>
<td>Frame jitter</td>
<td>&lt; 100 msec for Lip-synch</td>
</tr>
<tr>
<td>Information loss</td>
<td>&lt; 1% FER</td>
</tr>
</tbody>
</table>

In order to allocate the scarce radio resources fairly and flexibly between different types of services with their respective quality demands, an end-to-end Quality of Service (QoS) architecture, which is described in Section 2.4.2.2, is used in UMTS. Here, QoS is viewed as a series of chained services operating at different levels of the mobile environment and the required QoS is realised through several different bearer services. A Radio Access Bearer (RAB), which is based on the characteristics of the radio interface, provides the service quality over the radio interface. Clearly defined QoS attributes (parameters) are used to characterise the services and
functionality of the application. The specified QoS attributes for the conversational class are shown in Table 4.2.

Table 4.2: Value ranges of Radio Access Bearer Service Attributes for the conversational class [3GPP TS 23.107]

<table>
<thead>
<tr>
<th>QoS attributes</th>
<th>Value range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Traffic class</td>
<td>Conversational class</td>
</tr>
<tr>
<td>Maximum bitrate (kbps)</td>
<td>&lt; 2048</td>
</tr>
<tr>
<td>Correct delivery order</td>
<td>Yes/No</td>
</tr>
<tr>
<td>Maximum SDU size (octets)</td>
<td>≤ 1500 or 1502</td>
</tr>
<tr>
<td>Delivery of erroneous SDUs</td>
<td>Yes/No/-</td>
</tr>
<tr>
<td>Residual BER</td>
<td>(5 \times 10^{-2}, 1 \times 10^{-2}, 5 \times 10^{-3}, 1 \times 10^{-4}, 1 \times 10^{-6})</td>
</tr>
<tr>
<td>SDU error ratio</td>
<td>(1 \times 10^{-2}, 7 \times 10^{-3}, 1 \times 10^{-4}, 1 \times 10^{-5})</td>
</tr>
<tr>
<td>Transfer delay (ms)</td>
<td>80 – maximum value</td>
</tr>
<tr>
<td>Guaranteed bit rate (kbps)</td>
<td>&lt; 2048</td>
</tr>
<tr>
<td>Allocation/Retention priority</td>
<td>1, 2, 3</td>
</tr>
</tbody>
</table>

Table 4.3: Transport channel parameters for 64kbps Conversational radio bearer (DL/CS RAB) [3GPP TS 34.108]

<table>
<thead>
<tr>
<th>RLC</th>
<th>Logical channel type</th>
<th>DTCH</th>
</tr>
</thead>
<tbody>
<tr>
<td>RLC mode</td>
<td></td>
<td>TM</td>
</tr>
<tr>
<td>Payload sizes, bit</td>
<td></td>
<td>640</td>
</tr>
<tr>
<td>Max data rate, bps</td>
<td></td>
<td>64000</td>
</tr>
<tr>
<td>TrD PDU header, bit</td>
<td></td>
<td>0</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>MAC</th>
<th>MAC header, bit</th>
<th>0</th>
</tr>
</thead>
<tbody>
<tr>
<td>MAC multiplexing</td>
<td></td>
<td>N/A</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Layer 1</th>
<th>TrCH type</th>
<th>DCH</th>
</tr>
</thead>
<tbody>
<tr>
<td>TB sizes, bit</td>
<td></td>
<td>640</td>
</tr>
<tr>
<td>TTI, ms</td>
<td>20(or. 40)</td>
<td></td>
</tr>
<tr>
<td>Coding type</td>
<td>TC</td>
<td></td>
</tr>
<tr>
<td>CRC, bit</td>
<td>16</td>
<td></td>
</tr>
<tr>
<td>Max number of bits/TTI after channel coding</td>
<td>3948(alt. 7884)</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>DPCH</th>
<th>Spreading factor</th>
<th>32</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Downlink</th>
<th>DPCCH</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of TFCI bits/slot</td>
<td>8</td>
<td></td>
</tr>
<tr>
<td>Number of TPC bits/slot</td>
<td>4</td>
<td></td>
</tr>
<tr>
<td>Number of Pilot bits/slot</td>
<td>8</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>DPDCH</th>
<th>Number of data bits/slot</th>
<th>140</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of data bits/frame</td>
<td>2100</td>
<td></td>
</tr>
</tbody>
</table>
For physical realisation of the intended QoS for the transfer, all these QoS parameters must be mapped on to the transport/physical channel parameters such as spreading factor, transport format, ARQ parameters, channel protection and error detection schemes, RLC/MAC type and rate matching. The effectiveness of radio resource allocation algorithms and the system performances are greatly affected by the settings of these radio access network parameters. The optimal settings depend highly on the characteristics of the application, the propagation conditions, and the end user quality requirements. For illustration purposes, an example of the settings of a radio access bearer for 64 kbps conversational video applications are shown in Table 4.3.

4.3 Video traffic characteristics

The video encoding process can be considered under two main categories, namely open loop encoding and closed loop encoding. In the open loop encoding process, the input video sequence is encoded with fixed (pre-set) quantisation settings for each frame. To prevent the propagation of error in an encoded video sequence, intra-coded frames are inserted at regular intervals, which do not make use of any information from previously encoded frames. This process leads to highly variable encoded video frame sizes and is often referred to as Variable Bit Rate (VBR) encoding.

![Video Encoding Process Diagram](image)

**Figure 4.1: Video Encoding Process**

(a) Open loop (or Variable Bit Rate) encoding.

(b) Closed loop encoding

Closed loop encoding sends the encoded video frames into an output buffer, which is located at the encoder (see Figure 4.1 (b)). The buffer threshold setting controls the output video bit rate. If the input bit rate to the buffer exceeds the output bit rate set by the controller, then the encoder is told to adjust the quantisation step size of the current frame or macro-blocks in a way to realise the required video bit rate. Therefore, closed loop (rate controlled) encoding generates a more or less constant bit rate output.
The intra coded frames are much larger in size compared to the inter (predictive) coded frames. An increase of quantisation step size in order to achieve the required bit rate will result in poor quality intra coded frames. Moreover as they are referenced by following frames, a dramatic drop in the quality of the entire video sequence is visible. In order to mitigate the error propagation and to minimise the quantisation distortion while achieving the target output bit rate, Adaptive Intra Refresh (AIR) and Cyclic Intra Refresh (CIR) algorithms are often used in conjunction with the rate control algorithms. In both algorithms, only a selected number of macro blocks within each video frame are intra coded. The AIR technique refreshes the macro-blocks belonging to high activity regions of the video frame. The CIR refreshes macro-block in cyclic manner starting with the first macro-block of the video frame. This avoids the propagation of errors over low motion field.

Figure 4.2 shows the traffic characteristics of the “Suzie” sequence, which is encoded with and without rate control enabled. The quantisation step size is selected for the open loop encoding so as to achieve same average bit rate as the close loop encoding output. In addition to the TM5 rate control algorithm, AIR/CIR algorithms as described in Annex E1.5 of MPEG-4 standard [ISO-14496-2] are used in the close loop encoding. The number of macro blocks selected for AIR and CIR algorithms are 8 and 2 respectively. This provides a frame refresh rate that is equivalent (on average) to the insertion of one intra coded frame every 10 frames. Figure 4.2 clearly illustrates the variable frame sizes caused by open loop encoding. In fact the intra coded frames are roughly 6 times the size of inter coded frames.
Figure 4.3: Frame quality variation of MPEG-4 coded video. The circled areas represent high activity region.

(a) pdf of frame size for open loop encoding

(b) pdf of frame size for closed loop encoding

(c) pdf of frame PSNR for open loop encoding

(d) pdf of frame PSNR for closed loop encoding

Figure 4.4: Frame size and PSNR statistics of MPEG-4 coded video
Figure 4.3 shows the corresponding frame Peak Signal to Noise Ratio (PSNR) values for the open and the close loop encoding. The probability distribution of frame sizes and frame PSNR values are also shown in Figure 4.4. As can be seen in Figure 4.4(b), the variance of the video frame size distribution is greatly reduced by the closed loop encoding process. However, this improvement is obtained at the expense of video quality. As demonstrated in Figure 4.3 and 5(d), the resulting frame PSNR from the closed loop encoding is highly variable compared to that of the open loop encoding. Referring to Figure 4.3, open loop encoding gives higher frame PSNR while quality drop is visible in the close loop encoding in high motion section (circled in the figure). This is due to the need for a course quantiser to compensate for the relatively large number of information bits generated in high motion regions.

Basic comparisons of tradeoffs and potentials between the open loop and the closed loop encoded video transmission can be found in [LAKS-98]. A smoother output bit rate is favoured in fixed channel allocation schemes, where the radio channel is allocated according to the target rate setting at the rate controller. Due to the relatively high video quality and smoother frame quality variation seen at the open loop encoding, perceived quality may be improved by an adaptive channel allocation scheme. The closed loop video encoding will be used in all the experiments described in this chapter.

4.4 Real-time video communications over UTRAN

UMTS is designed to support both circuit switched and packet switched multimedia applications. 3GPP [3G TS 26.111] describes a modified version of the H.324 interface in order to provide circuit switched multimedia applications over UMTS networks. Packet switched multimedia communication is realised using standard IETF defined protocols IP, UDP and RTP. The Session Initiation Protocol (SIP) and Session Description Protocol (SDP) are used to negotiate and open an IP connection between the terminals [3GPP TS 26.235].

Two radio bearers are considered for conversational multimedia applications. Media data is transmitted over the specified Radio Access Bearer (RAB) while the necessary control information is conveyed over the accompanying Signalling Radio Bearer (SRB) [3GPP TS 34.108]. The corresponding transport channels for radio bearers are separately channel protected and formatted. The transport channels are finally multiplexed on to the same physical channel at the Physical layer in order to transmit the data over the air interface.
RLC transparent mode operation is considered for both circuit switched and packet switched transmission. In other words no re-transmission mechanism is considered. Unlike in other media applications, where the received corrupted data packets are more likely to be dropped at the RLC layer, for video applications it is necessary to pass all received data (including the corrupted data) to the application layer. This is because, the error resilience/concealment mechanisms implemented in the MPEG-4 decoder can be used to recover/conceal the corrupted data bits up to some extent. In this scenario, the use of CRC bits at the RLC packet level is not only redundant but also reduces the available bandwidth for the source data. Hence, no CRC attachment is considered.

Full error-resilience enabled MPEG-4 coded video transmission is considered in the experiments discussed in this Chapter. QCIF (176×144) formatted sequences are transmitted over the specified channels for 30 sec duration. For video sequences captured at 30fps, this allows transmission of 900 frames. To capture the effect of the bursty nature of the channel on the received video quality each experiment was repeated 10 times. The results for all three selected sequences were averaged to obtain a meaningful figure. That means each point represents an average frame PSNR value of 27000 frames of the original (captured at 30 fps) video sequences.

### 4.4.1 Circuit-switched bearers

Typical parameter sets for reference RABs, SRBs and important combinations of them for conversational multimedia applications are presented in [3GPP TS 34.108]. In the simulation a 3.4 kbps SRB, which is specified in [3GPP TS 34.108] and described in Section 3.4, is used for a Dedicated Control CHannel (DCCH). For the reasons already stated in Section 3.4, rate-matching ratios for DCCH are set according to Table 3.13. The throughput available to the application depends mainly on the spreading factor, the rate-matching ratio and the channel-coding scheme used for the bearer configuration. The CRC attachment, the protocol layer operation modes and the Transport Block size also influence the available application throughput. However, as mentioned earlier, transparent mode is selected and zero CRC attachment is considered for video telephony applications. With no CRC bits, the transport block size can be assumed to have an insignificant effect on the application throughput. No network protocol layer overhead is added for circuit switched connections. The calculated information data rates, according to the simulation parameter settings for circuit-switched connections are shown in Table 4.4.
Table 4.4: UMTS Traffic capacity (kbps) for Circuit switched radio bearers.

<table>
<thead>
<tr>
<th>Spreading Factor</th>
<th>CC 1/2</th>
<th>CC 1/3</th>
<th>TC 1/3</th>
<th>No Coding</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>RM 1.0</td>
<td>RM 0.9</td>
<td>RM 1.0</td>
<td>RM 0.9</td>
</tr>
<tr>
<td>128</td>
<td>16.2</td>
<td>18</td>
<td>10.6</td>
<td>11.8</td>
</tr>
<tr>
<td>64</td>
<td>39.5</td>
<td>44</td>
<td>26.10</td>
<td>29.1</td>
</tr>
<tr>
<td>32</td>
<td>97</td>
<td>107.75</td>
<td>64.5</td>
<td>71.85</td>
</tr>
<tr>
<td>16</td>
<td>206.1</td>
<td>229</td>
<td>137.4</td>
<td>152.6</td>
</tr>
<tr>
<td>8</td>
<td>442.8</td>
<td>492</td>
<td>295.1</td>
<td>327.6</td>
</tr>
<tr>
<td>4</td>
<td>915.75</td>
<td>1016.8</td>
<td>610</td>
<td>679.05</td>
</tr>
</tbody>
</table>

CC - Convolutional Code  
TC - Turbo Code  
RM - Rate matching ratio

4.4.1.1 Channel utilisation and delay-jitter variation

Low frame delay variation (frame jitter) is an important requirement for video telephony applications. For accurate lip synchronisation, the maximum allowed frame delay variation is limited to 100 ms [3GPP TS 22.105]. As can be seen in Figure 4.4(b), the output video frame sizes vary around the mean value by a considerable amount even for the closed loop (rate controlled) encoding process. This could lead to frame delay variation when transmitted over fixed rate channels. This section investigates frame delay (jitter) variation resulting from the source traffic characteristics, and its effect on channel utilisation.

The video encoding frequency is set to 10fps. That means that the RLC/MAC-layer receives an encoded video frame every 100 ms. The received video frame is segmented into RLC blocks according to the specified RLC block size, which is equivalent to the transport block size for transparent mode operation. If the information bits in a video frame do not fit into an integer number of RLC blocks then zero padding bits are added at the last RLC block. The segmented blocks are channel coded and stored in a transmitter buffer to be transmitted over the physical channel in every Transmit Time Interval (TTI), which is an integer multiple of 10 ms. As in a practical system, if the number of data blocks in the transmitter buffer is less than the required number of data blocks in a TTI frame, then dummy bits are used to complete the current TTI frame. Another media stream can be used instead of dummy bits and two media streams can be multiplexed into same TTI frame in order to maximise the system utilisation as recommended by 3GPP [3GPP TS 25.321]. However, the intention of this work is to examine the effect of video traffic characteristics alone on the system performance. Therefore, no media multiplexing is performed.
Channel utilisation is calculated in terms of the ratio between the total number of useful bits transmitted and the total number of available bits for a given RLC block size. The resulting channel utilisation is shown in Figure 4.5 for spreading factor 32 realisation and 1/3 rate channel coding. As can be seen from the figure, the channel utilisation decreases with an increase in the RLC block size for both long duration (30 sec) and short duration (5 sec) connections. Experiments show similar performances for other spreading factor realisations.

![Figure 4.5: Protocol efficiency in circuit switched video telephony.](image)

Figure 4.6 illustrates the effect of RLC block size upon the frame delay variations. The definition of the frame jitter variation follows the statistical delay-jitter bound with 95% probability as given in [RFC-1193].

\[
\text{prob}(J_i \leq J_{\text{max}}) \geq U_{\text{min}} \text{ for all } I
\]

Where, \(J_i\) and \(J_{\text{max}}\) represents the frame jitter of the \(i^{th}\) frame and the maximum frame jitter value respectively. \(U_{\text{min}}\) is the lower bound of the probability that \(J_i\) be within its limit. Frame jitter is defined as

\[
J_i = |D_i - D| \text{ for all } I
\]

\(D\) denotes the ideal or target frame delay and \(D_i\) is the actual frame delay resulting from the transmission.

Short duration transmission is more robust to jitter variation compared to long duration transmission as it benefits from the allowable start-up delay, which is set to 300 ms in the experiment. It is evident from Figure 4.6, that the resulting frame jitter for most RLC block size settings is not acceptable for conversational video applications. The motion involved in the video
sequence also has significant effects on the jitter variation. As can be seen in Figure 4.7, the high motion "Foreman" sequence shows higher frame delay compared to the low motion "Carphone" and "Suzie" sequences. This is because the sudden changes of pictures experienced in high motion sequences are too fast for a rate controller to smooth out, resulting high variation in output bit rates.

![Figure 4.6: Frame delay jitter characteristics of circuit switched video telephony.](image)

![Figure 4.7: Variation of frame delay jitter characteristics with video sequence type.](image)

The frame jitter variation caused by the variable nature of the application throughput can be controlled by reducing the source bit rate or managing the RLC buffer at the transmitter. Both of these methods have weakness associated with them. Reducing the source throughput results in an inefficient system. The buffer management techniques control the size of the transmitter buffer in order to maintain time correlation between the transmitting frames. Information data that cannot
be transmitted to the receiver within the required time window, is dropped at the transmitter. As even the most important data, such as VOP header information, can be dropped at the transmitter, buffer overflow could lead to tremendous degradation of receive video quality, unless an interaction between the application layer and the RLC layer is conducted.

The maximum affordable transport block sizes that satisfy the delay requirements (< 100 ms) are calculated for different spreading factor realisations and are listed in Table 4.5. The table also shows the resulting channel utilisation for the above channel configurations.

Table 4.5: Transport block size & channel utilisation; Frame delay jitter 95m percentile -100ms

<table>
<thead>
<tr>
<th>Spreading Factor</th>
<th>Maximum RLC block size [bits]</th>
<th>Channel Utilisation [%]</th>
</tr>
</thead>
<tbody>
<tr>
<td>32</td>
<td>216</td>
<td>98.4</td>
</tr>
<tr>
<td>16</td>
<td>560</td>
<td>98.25</td>
</tr>
<tr>
<td>8</td>
<td>1000</td>
<td>98.2</td>
</tr>
</tbody>
</table>

However, for optimal performance of interleaving and convolutional coding algorithms, the input block size to the channel coder should be less than 504 bits. Otherwise, block segmentation is performed (see Section in Chapter 3) at Layer 1. Therefore for optimal performance and system utilisation, the RLC block size should be set to less than 504 bits for spreading factor 16 and 8 (with convolutional code) realisations.

4.4.1.2 Video performance in error free environments

The optimal video quality that can be delivered to the user at different source rates was experimentally derived for the selected test sequences. The sequences are coded with source rates varying from 26kbps to 300 kbps while the video frame rate is set to 10 fps. The frame rate of 10fps is considered to be sufficient to provide acceptable quality for relatively low motion video conferencing applications [3G TS 26.110]. Video performances over the error free environment shows (see Figure 4.8) a linear relationship between mean frame PSNR and video source rate at higher source rate operations. At lower source rates, the mean frame PSNR decreases dramatically falling of video rates. It can be seen that the quality of the “Suzie” sequence is constantly 3-4 dB better than that of the “Carphone” sequence. Similarly, the “Carphone” sequence shows better performance than the “Foreman” sequence throughout. This is mainly due to the differences in motion activities involved in the sequences.
Figure 4.8: MPEG-4 performance over error free channel.

Figure 4.9: MPEG-4 performance with various frame rate over error free channel.

Figure 4.9 illustrates the effect of frame rates on the video performance over an error free environment. Transmission of the “Suzie” sequence is considered in the experiment. Low frame rates result in low quantisation distortion, hence high frame PSNR. However, frame PSNR calculation only demonstrates the relative quality of individual decoded frames compared to the original video frame. For video communication, the quality seen in the time domain is also important in assessing the performance. When the frame rate is too low, motion becomes jerky when viewing the decoded video stream. This behaviour is more disturbing and tends to deteriorate the perceptual video quality.
The above findings can be used in conjunction with the UMTS traffic capacity states shown in Table 4.4, to determine the optimum settings of source and network parameters for conversational video communications over UTRAN. The throughput capacity for a spreading factor 128 realisation, which is approximately 12 kbps, is too low to support video applications. In fact, the minimum source rate that a high motion sequence (such as “Foreman” and “Carphone”) can be encoded at, is limited to 28-29 kbps by the maximum quantisation step size for settings of 10 fps. A spreading factor of 64 can only support source rates around 26 kbps (with 1/3 rate channel coding). This means that even a spreading factor of 64 may not be adequate to realise video communications over UTRAN. Therefore, only spreading factor settings of 32, 16 and 8 are considered for the investigation of video performances over error prone environments. Table 4.6 summarises the maximum average frame PSNR value that can be achieved with various radio bearer configurations for the video telephony application over UTRAN. Note that video encoded at 5 fps is used for transmission over the spreading factor 64 channels.

Table 4.6: Maximum frame PSNR value variation with radio bearer configurations.

<table>
<thead>
<tr>
<th>Spreading Factor</th>
<th>Video frame rate (fps)</th>
<th>CC 1/2</th>
<th>CC 1/3</th>
<th>TC 1/3</th>
</tr>
</thead>
<tbody>
<tr>
<td>64</td>
<td>5</td>
<td>29.94</td>
<td>28.31</td>
<td>28.37</td>
</tr>
<tr>
<td>32</td>
<td>10</td>
<td>34.43</td>
<td>32.48</td>
<td>32.51</td>
</tr>
<tr>
<td>16</td>
<td>10</td>
<td>38.02</td>
<td>36.06</td>
<td>36.13</td>
</tr>
<tr>
<td>8</td>
<td>10</td>
<td>41.65</td>
<td>39.71</td>
<td>39.78</td>
</tr>
</tbody>
</table>

4.4.1.3 Video performance in error prone environments

This section describes the performance of video telephony over simulated error-prone channel conditions using the developed UTRAN emulator. The channel environments investigated are Vehicular A (with 50 km/h mobile speed) and Pedestrian B (with 3 km/h mobile speed) multi-path propagation conditions as defined in [UMTS-30.03]. The source bit rates are set according to the values given in Section 4.4.1. Full error resilience enabled, rate controlled (using TM5 rate control algorithm) MPEG-4 coded video is considered in the experiments. Influences of MPEG-4 codec parameter settings on the received video quality over W-CDMA based radio networks are discussed in [SOAR-00]. The results show that the video pack size of around 500-700 bits provides the optimal settings. Furthermore, a fixed setting of 2 and 10 intra coded macro blocks in each frame for Cyclic Intra Refresh (CIR) and Adaptive Intra Refresh (AIR) respectively are shown to provide the optimal video quality over the range of channel error rates considered likely.
for W-CDMA mobile networks. Therefore, in the experiments discussed in this section, the number of intra coded MBs per frame is set to 2 and 10 in CIR and AIR algorithms. And the video packet size of 600 bits is used.

Experiments were conducted to investigate the influences of network parameter settings on video performance. Mainly the effect of spreading factor allocation, channel coding schemes and influence of fast power control were examined. The maximum frame PSNR value that can be achieved for a given source rate is shown in Table 4.6. These values can be considered as reference performance figures for the experiment results described in this section.

4.4.1.3.1 Effect of source frame rate

Despite the temporal domain quality degradation discussed earlier, low frame rate encoded video shows higher spatial quality (Frame PSNR) at error free (good) channel conditions. However, in the presence of channel errors, high frame rate video shows better performance than low rate. This is mainly because of the prevention of temporal error propagation. For example, settings of 12 intra MB’s per frame results refreshment of one complete frame in every 550 ms at 15fps frame rate while refreshment of one complete frame will be resulted in every 1650ms at 5fps frame rate. However, as can be seen in Figure 4.10, the performance of 15fps shows only a slight improvement compare to that of 10fps. Therefore, video sequences are encoded at 10fps frame rate for the experiment described in below sections.
4.4.1.3.2 Effect of spreading factor

The received video quality for different spreading factor realisation is measured in terms of average frame PSNR and the results are depicted in Figure 4.11. Video sequences are coded at the appropriate rates listed in Table 4.4. 1/3 rate convolutional code is used to protect the video data and the protected data is transmitted over the simulated Vehicular A channel environment. As expected, allocation of SF 32 provides slightly better performance than others at poor channel conditions due to the better channel protection capability of higher spreading factors. At better channel conditions, allocation of SF 16 provides superior video quality compared to others. Both SF 16 and SF 32 reach the maximum achievable video quality at channel Eb/No value equals to 10 dB. SF 8 considerably under-performs all other schemes even with good conditions. This is due to the inter-symbol-interference experienced in multi-path channels. The channel coding algorithm tends to mitigate the inter-symbol-interference effect. However, significant performance degradation is visible with low spreading factors (such as 8).

![Figure 4.11: MPEG-4 performance over Vehicular A environment. 1/3 CC & no power control.](image)

Similar performances are visible for spreading factors 32 and 16 with turbo coding when operating over the Vehicular A propagation environment. However, the performance of spreading factor 8 shows much better performance for turbo coding compared to that of convolutional coding. As stated earlier in Section 3.4.1.1, the main reason is the better performance of the turbo encoder/decoder in the presence of large input block sizes. Even though both SF 16 and SF 32 approach the maximum achievable frame PSNR values (36.13 dB and 32.51 dB respectively) at
an Eb/No value of 10 dB, the SF 8 results do not reach the maximum PSNR value (39.78 dB) within the Eb/No range considered in the experiment.

Figure 4.12: MPEG-4 performance over Vehicular A environment with 1/3 TC and no power control.

Figure 4.13: MPEG-4 performance over Pedestrian B environment with 1/3 CC and no power control.

With no fast power control, the bit error rate characteristics of the channel with the Pedestrian B environment is significantly low. This is because of the poor performance of block based interleaver and channel decoder in the presence of long, weak bursts of channel condition at low mobile speeds. Therefore, video telephony over the Pedestrian B environment shows poor performance relative to that resulting from the Vehicular A environment. For 1/3 rate convolutional code, a spreading factor of 32 gives the best performance, while a spreading factor
of 8 shows the worst performance. The performance for the 1/3 rate turbo code shows similar behaviour, however, performance differences between each spreading factor are reduced compared to that of the convolutional coding scheme. None of the coding schemes or spreading factor realisations approach the maximum possible frame PSNR values within the Eb/No value range considered for video transmission over the Pedestrian B environment.

Figure 4.14: MPEG-4 performance over Pedestrian B environment with 1/3 TC and no power control.

4.4.1.3.3 Effect of channel coding

Figure 4.15: Effect of channel coding scheme on MPEG-4 performance over the Vehicular A channel (SF-32).
Figure 4.15 shows the effectiveness of different channel coding schemes for video applications over the Vehicular A environment. The allocated spreading factor is 32 and video source rates are set according to the values shown in Table 4.4. Figure 4.15 clearly illustrates the performance improvement achieved by turbo coding for operation over low quality channels. Frame PSNR resulted for turbo coding is about 4-5 dB better compared to the 1/3 rate convolutional coding and 9-10 dB better compared to 1/2 rate convolutional coding. For good channel conditions, 1/2 rate convolutional coding out-performs others by a PSNR value of 2 dB mainly because of the low quantisation distortion seen at higher source rates. All three coding schemes obtain the maximum possible frame PSNR values at a channel Eb/No of around 10 dB.

![Figure 4.15: Effect of channel coding scheme on MPEG-4 performance over the Pedestrian B channel (SF-32).](image)

The expected performance improvement with turbo coding is also visible for video transmission over the Pedestrian B propagation environment (see Figure 4.16). The performance improvement achieved with turbo coding is about 2-3 dB relative to the 1/3 convolutional code and 4-5 dB relative to the 1/2 convolutional code for transmission of video over the Pedestrian B propagation environment.

### 4.4.1.3.4 Effect of fast power control

As outlined in Section 3.5.1.2, the employment of a fast power control algorithm can improve the bit error characteristics of the propagation environment. Experiments were carried out to investigate the influences of fast power control on the performance of video applications over various propagation environments. The average frame PSNR values for sequences protected with...
various channel-coding schemes and a spreading factor of 32 over Vehicular A and Pedestrian B channels conditions, are shown in Figure 4.17 and Figure 4.18 respectively. Dashed lines in Figure 4.17 and Figure 4.18 indicate the performances achieved without the application of power control. Slight performance improvements can be seen at Vehicular A propagation environment. For example, turbo code protected transmission achieves its maximum video quality at a $E_b / N_0$ setting of around 9 dB without fast power control. However, with fast power control the maximum value is reached at an $E_b / N_0$ setting of 8 dB.

Figure 4.17: MPEG-4 performance over Vehicular A channel with fast power control (SF-32).

Figure 4.18: MPEG-4 performance over Pedestrian B channel with fast power control (SF-32).
More noticeable improvement is visible over the Pedestrian B propagation environment. The relative quality improvement seen in terms of frame PSNR for turbo code realisation, with and without power control, varies between 3-4 dB with a good channel and 8-9 dB with moderate channel conditions. About 5-8 dB frame PSNR improvement is visible for the 1/3 rate convolutional code. Even better improvement can be seen for the 1/2 rate convolutional code.

Figure 4.19: MPEG-4 performance over the Pedestrian B channel with fast power control and 1/3 convolutional code.

Figure 4.20: MPEG-4 performance over the Pedestrian B channel with fast power control and 1/3 turbo code.
Tests were carried out to evaluate the video quality that can be achieved with fast power control for other spreading factor realisations over the Pedestrian B channels. Figure 4.19 and Figure 4.20, illustrate the obtained results for 1/3-rate convolutional coding and 1/3 rate turbo coding respectively. Solid lines illustrate performances with the power control algorithm, while dashed lines denote performances without the power control algorithm. The figures clearly illustrate the performance improvement achieved by employing fast power control over the Pedestrian B channels. For both figures, spreading factor 8 shows better performance than the others with good channel conditions. As in the Vehicular A environment, spreading factor 32 marginally outperforms the others at channel Eb/No lower than 9 dB. As shown in Figure 4.20, the maximum possible frame PSNR values, 32.51 dB and 36.13 dB, are achieved at Eb/No 8 dB and 10 dB for SF 32 and SF 16 respectively over the power controlled Pedestrian B channels.

So far, the performances of video telephony over UTRAN under various radio bearer configurations and different propagation environments have been presented in terms of average frame PSNR values vs channel Eb/No conditions. As outlined in Chapter 2, even though PSNR is the standard measure of judging video quality, it has some limitations. Video sequences can generally be characterised by their spatial information and temporal information. Spatial information is described by frame size, resolution, chrominance and luminance accuracy. The temporal domain is described by the frame rate, the motion involved and scene changed. The PSNR calculation is based only on the spatial information of the video frames, hence it provides no indication about the temporal quality of the video. Although it may be used as an assessment figure in performance comparisons, absolute (perceptual) quality of video should be judged based on subjective quality tests. Unfortunately, there is no standard subjective quality measurement method specified for video applications. However, experimental results show that if the average frame PSNR value is higher than 20 dB, the video output shows an acceptable visual quality. In order to put the above PSNR performance figures in to perspective, the minimum Eb/No requirement needed to deliver perceptually acceptable video to end users, is calculated for all radio bearer configurations investigated. The average frame PSNR value of 20 dB is considered as the reference limit and the obtained results are listed in Table 4.7.

The minimum required $E_b/N_o$ values for acceptable quality video with turbo coding, is lower compared to those required for the convolutional codes. 1/3 rate convolutional codes requires lower $E_b/N_o$ than that required for 1/2 rate convolution code. Spreading factors 16 and 32 show similar $E_b/N_o$ requirements for acceptable video quality, while spreading factor 8 shows relatively higher minimum $E_b/N_o$ requirement. $E_b/N_o$ values as low as 4.5- 5.0 dB are
sufficient to provide acceptable quality video with 1/3 rate turbo coding over UMTS multi-path propagation environments. Perceptually acceptable video quality can be obtained around an $E_b/N_o$ of 7 dB with 1/2 rate convolutional code over fast power controlled channels. 1/3 rate convolutional code requires a minimum $E_b/N_o$ of around 5.5 – 6.5 dB for reasonable quality video.

Table 4.7: Minimum Eb/No requirement for acceptable video quality over UTRAN

<table>
<thead>
<tr>
<th>Spreading factor</th>
<th>Vehicular A</th>
<th>Pedestrian B</th>
</tr>
</thead>
<tbody>
<tr>
<td>1/2 CC</td>
<td>1/3 CC</td>
<td>1/3 TC</td>
</tr>
<tr>
<td>32</td>
<td>7.2</td>
<td>6.3</td>
</tr>
<tr>
<td>16</td>
<td>-</td>
<td>6.5</td>
</tr>
<tr>
<td>8</td>
<td>-</td>
<td>8.5</td>
</tr>
</tbody>
</table>

With fast power control

<table>
<thead>
<tr>
<th>Spreading factor</th>
<th>Vehicular A</th>
<th>Pedestrian B</th>
</tr>
</thead>
<tbody>
<tr>
<td>½ CC</td>
<td>1/3 CC</td>
<td>1/3 TC</td>
</tr>
<tr>
<td>32</td>
<td>6.6</td>
<td>5.5</td>
</tr>
<tr>
<td>16</td>
<td>-</td>
<td>5.8</td>
</tr>
<tr>
<td>8</td>
<td>-</td>
<td>7.7</td>
</tr>
</tbody>
</table>

It is necessary to emphasise that the values shown in Table 4.7 are the required minimum information bit energies per received noise spectral density ($E_b/N_o$), and not the transmitted power per received noise spectral density ($P_t/N_o$). The relationship between $P_t/N_o$ and $E_b/N_o$ can be shown as:

$$P_t/N_o = E_b/N_o \cdot R$$  \hspace{1cm} \text{Equation 4.3}

where $R$ is the video source rate. Therefore, minimum $P_t/N_o$ requirements can be obtained by multiplying the $E_b/N_o$ values shown in Table 4.7 by the corresponding source rates shown in Table 4.4. As 1/2 rate code can support higher source rates compared to the 1/3 codes, the minimum $P_t/N_o$ requirement for acceptable video quality will be much higher for the 1/2 rate convolutional code compared to that required for 1/3 rate codes. Also the source rate increases with decreasing spreading factor. That means that the required $P_t/N_o$ for acceptable quality increases with a decrease in spreading factor for each channel-coding scheme. This concludes that a spreading factor of 32 with 1/3 rate turbo coding provides acceptable video quality with the least transmit power over UMTS networks.
4.4.2 Packet-switched bearers

Packet switched conversational multimedia applications are considered within the 3GPP specified IP Multimedia Subsystem (IM Subsystem) framework. The service architecture, call control and media capability control procedures for the packet switched multimedia application have been defined in [3GPP TS 24.229]. These functions are defined based on a modified version of the IETF Session Initiated Protocol (SIP) and the Session Description Protocol (SDP). Packet switched multimedia terminals are supposed to have the above functions inbuilt to support packet switched based multimedia telephony.

In packet switched conversational multimedia applications, the individual media types are independently encoded and packetised separately into Real-time Transport Protocol (RTP) packets. The encapsulated packets are then transmitted end-to-end over IP connections inside UDP datagrams. Inter-media synchronisation among received media streams are performed based on the RTP time stamps at the receiver.

To allow communication over heterogeneous networks, the same multimedia codecs (as in circuit switched telephony) are selected to be implemented in the packet switched domain. A thorough discussion of codec selection and codec format for packet switched conversational multimedia applications can be found in [3GPP TS 26.235]. For packet switched based video telephony applications, MPEG-4 coded video frames are separately encapsulated within RTP packets following the RTP fragmentation rules specified in [RFC 3016]. The RTP packet size is defined by the Maximum Transmission Unit (MTU) setting. One RTP packet can only contain an integer number of video packets and every video frame should start in a new RTP packet. In other words, a video packet can not be split over multiple of RTP packets. And also data belongs to different video frames should not be packetised into the same RTP packet. This allows use of RTP time stamp to indicate the VOP time framing. The resulting RTP packets may contain information bits numbering slightly less than the limit defined by the path-MTU setting. Finally, each RTP packet is encapsulated within a UDP/IP packet for transmission. The RTP header and the UDP header contain 96 bits and 32 bits respectively. Header size of the IP (version 4) is 192. Therefore, a total of 320 bits overhead is added to each RTP payload from the transport and network layer. No network layer header compression is assumed at the PDCP layer.
4.4.2.1 Traffic capacity for packet switched bearer

The presence of network layer overhead reduces the throughput available to the application. The available application throughput in this scenario is experimentally determined and the results are presented in Figure 4.21 in terms of overhead percentage vs. channel bit rate for two specific MTU sizes (576 and 288 Bytes). As expected, the network layer overhead percentage decreases with an increase in MTU size. Also the overhead percentage shows approximately exponential decay behaviour with respect to the channel bit rate. At low channel bit rates, the resulting overhead percentage can go up to 11% and 17% for radio bearer realisation with 576 byte and 288 byte MTU sizes respectively.

![Figure 4.21: Throughput loss due to network layer overhead in packet switched connections.](image)

The experimentally determined source traffic capacities for packet switched video telephony over UMTS are listed in Table 4.8. An MTU size of 576 bytes is assumed in the calculation. A significant reduction in source throughputs compared to equivalent circuit switched connections is visible at high spreading factor realisations.

**Table 4.8: UMTS Traffic capacities [kbps] for packet switched connections (MTU – 576 bytes).**

<table>
<thead>
<tr>
<th>Spreading Factor</th>
<th>CC 1/2</th>
<th>CC 1/3</th>
<th>TC 1/3</th>
<th>No Coding</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>RM 1.0</td>
<td>RM 0.9</td>
<td>RM 1.0</td>
<td>RM 0.9</td>
</tr>
<tr>
<td>64</td>
<td>35.85</td>
<td>39.82</td>
<td>23.24</td>
<td>26.12</td>
</tr>
<tr>
<td>32</td>
<td>89.16</td>
<td>99.56</td>
<td>58.75</td>
<td>65.71</td>
</tr>
<tr>
<td>16</td>
<td>191.30</td>
<td>212.56</td>
<td>127.19</td>
<td>141.22</td>
</tr>
<tr>
<td>8</td>
<td>412.17</td>
<td>457.97</td>
<td>274.63</td>
<td>306.31</td>
</tr>
<tr>
<td>4</td>
<td>856.23</td>
<td>950.71</td>
<td>570.35</td>
<td>634.91</td>
</tr>
</tbody>
</table>

CC – Convolutional Code  
TC – Turbo Code  
RM – Rate Matching ratio
4.4.2.2 Video performance over packet switched bearers

A packet switched connection provides operational flexibility for media services, particularly with respect to data delivery over the Core Network. However, careful design strategies should be followed to avoid throughput loss, which occurs due to the extra overhead added at the network layer, and also due to extra information loss, resulting from corrupted network layer header information in the radio access networks.

For variable length coded information sequences, such as MPEG-4 video, the effective probability of information loss is significantly different from the actual percentage of information loss occurring in the presence of channel errors as shown in [WORRb-01]. The effective probability of loss depends on many factors, such as error resilience/concealment techniques, data format, temporal error propagation, probability of packet header corruption, and the probability of VOP header loss. The channel distortion arising from corrupted packet headers also depends on the above-mentioned factors. In this section, experiments are carried out to investigate real-time video performance over packet switched connection in UMTS radio access networks. Chapter 5 will further explain these issues based on a mathematically derived distortion model for video transmission over an error-prone channel.

Figure 4.22: MPEG-4 performance over packet switched connection.

Figure 4.22 reveals the obtained results for video transmission over packet switched connections with selected MTU sizes. A spreading factor of 32 and 1/3 rate convolutional code realisation is used in the radio bearer configuration, and the Vehicular A propagation environment is assumed.
The figure shows the total received video quality over a packet switched connection is always less compared to the performance over circuit switched connections. For good channel conditions, a slight performance loss (about 1-0.5 dB) is visible. This is mainly because of the source throughput reduction resulting from the network layer overhead. However, for low channel quality, the performance loss due to corrupted packet headers becomes the dominating factor and about 3-4 dB PSNR loss is noticeable compared to circuit switched performance. Although the experimental results are shown only for a specific radio bearer setting, similar performance loss figures can also be expected with other radio bearer configurations.

4.5 Conclusion

The results presented in this chapter, highlight a number of key factors, which should be considered in the provision of real time video applications over UMTS networks. Compared to delay insensitive media applications, the strict delay requirements seen in real-time video applications restrict the use of some error resilience and congestion control mechanisms. Powerful Automatic Repeat reQuest (ARQ) techniques, which are widely used in streaming video applications in enhancing received video quality, may not be suitable for real-time conversational video applications. Also, the encoding/decoding and error resilience should not be too complex to avoid extra processing delay. Results shown in Section 4.4.1.1 illustrate that the delay requirements stated in [3GPP TS 22.105] for real-time video communication can be satisfied with careful selection of network parameter settings for the rate control of MPEG-4 encoded video transmission. The networks should satisfy not only the delay requirement but also the error rate requirement, in order to achieve acceptable quality video transmission over error prone channels. Bit error rates of around $10^{-3}$ are the maximum tolerable bit error rate by a DCT based source codec such as MPEG-4. As demonstrated in Section 4.4.1.3, an $E_b/N_0$ value of 5-6 dB is sufficient to provide acceptable video quality over the UMTS multi-path channels.

Video quality can be degraded due to two main reasons when transmitted over error prone environments. First is the un-recoverable quantisation distortion resulting from the operation of compression algorithms at the encoder, and the second is the channel distortion due to the information loss from transmission over error prone channels. A compromise between source throughput capacity and the error-correcting capabilities is necessary to obtain the optimal video quality for a given channel condition. In addition, the transmission power requirement plays an important role in W-CDMA based UMTS networks. This is because UMTS is an interference-limited system. In other words, increases in transmit power for one user, increases the interference.
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power for other users, hence it tends to reduce the overall system capacity. Therefore, careful selection of channel coding schemes, spreading factor, source throughput, and transmit power are essential in achieving optimal video quality and maximum system capacity for video applications over UMTS.

Even though high operational flexibility can be achieved with packet switched connections, especially in the Core Network, careful design criteria should be followed to avoid quality degradation seen over the radio interface. Experimental results illustrate that there is a 1-3 dB performance loss (in average frame PSNR value) for packet switched operation, compared to circuit switched operation. The performance loss seen with good channels is mainly due to the source throughput loss resulting from the network layer overhead. The main reason for the additional performance loss seen in the presence of channel errors is the information loss resulting from the corrupted packet headers. As will be explained in Chapter 5, a compromise between throughput loss and information loss should be considered in selecting the optimal packet size for video transmission over error prone channels.

Radio link quality can be improved with the use of performance enhancement techniques such as fast power control for down link transmission. Experimental results demonstrate a tremendous improvement in video quality with fast power control for slow mobile speed channels. Other transmission diversity techniques such as space-time transmit diversity, close loop transmit diversity may further enhance video performance over UMTS networks.
Chapter 5

5 Distortion Modelling and Optimal MTU Calculation

5.1 Introduction

Low bit rate video codecs, such as MPEG-4, employ combined block-based motion compensation together with variable length coding to maximise the compression efficiency. However, variable length coding is more sensitive to channel errors and motion prediction promotes error propagation through the coded sequence, increasing the susceptibility to channel errors. Carefully designed error resilience techniques can be incorporated into the compression algorithm in order to make the video codec more resilient to channel degradations. Information prioritisation can also be used to enhance the performance by mitigating the effect of channel errors. In other words, the video information is prioritised according to its importance and during encoding, is processed in such a way as to reduce the channel distortion in high priority data. Widely recognised methods are codec/network parameter adaptation [FABR-00], content adaptation [SHIH-01], unequal error protection [KUNG-01], and unequal power allocation [EISE-02]. In order to adapt encoder/network parameters and to prioritise the image region in a way that maximises the quality of the received video at the decoder, channel distortion or relative regional distortion seen at the decoder, must be determined at the encoder prior to transmission of the information. This chapter describes a method of estimating received video quality for given encoder parameters, decoder concealment techniques and channel statistics. The performance
estimation is based on a distortion model, which accurately models the overall distortion seen in decoder frame reconstruction including quantisation distortion, concealment distortion and error propagation. The accuracy of the developed distortion model is evaluated for video transmission over simulated UMTS channels. In addition, using the developed model, the effect of packet size on the performance of video communications over packet switched wireless networks is investigated.

5.2 Video performance modelling

Extensive research has been carried out in the area of performance modelling and its application in enhancing the quality of transmitted video over error-prone environments. The estimation of channel distortion based on a simple rate-distortion framework is proposed in [ORTE-98, HONG-02]. The incorporation of error concealment in the computation of decoder distortion at the encoder has been described in [SUH-01, WENG-99]. All these distortion models use some simplified assumptions in the calculation, especially in the calculation of error propagation. The method proposed in [HONG-02] use only the quantisation distortion in the calculation, while the methods proposed in [WENG-99] ignore error propagation beyond one frame. Furthermore, the total block distortion is approximated as a simple summation of the quantisation distortion of the block and weighted concealment distortion of corresponding blocks in the previous frame. Zhang et al, proposed a recursive algorithm for an optimal distortion estimate of the decoder output with pixel level precision, and its application in optimal Intra/Inter mode selection in [ZHAN-00]. This method is shown to have more accurate distortion estimation. However, the application of the recursive algorithm is limited by its complexity. Under the assumption that the video reconstruction errors are uniformly distributed, a relatively simple distortion model is proposed in [KIM-03] for optimum power management for transmission of video in a CDMA system. The distortion induced by the reference frame misalignment is expressed in terms of the quantisation distortion, distortion caused by the corruption of motion information, and the distortion induced by error concealment. The distortion method adopted in this chapter is similar to the method proposed in [KIM-03]. However, the distortion induced due to the error propagation is calculated differently, even though the same assumption, namely the uniform distribution of video reconstruction error, is made. Also, the model proposed in this chapter uses Adaptive Intra Refresh (AIR) techniques instead of just intra frame refreshment.
5.2.1 MPEG-4 video format

The bit stream syntax of MPEG-4 uses a hierarchical structure. Each video frame is partitioned into smaller rectangular regions called “Macro-Blocks (MBs)”, which are 16×16 pixels in size. Each MB is coded either in inter-mode or intra-mode. Intra-mode MBs are transform-coded directly without applying motion compensation while inter-mode MB’s uses motion compensation.

Compressed video data is ordered according to a specified format before the transmission. The MPEG-4 data format is shown in Figure 5.1. A Video Packet (VP) is formed by grouping an integer number of coded consecutive MB’s. Synchronisation markers are used to isolate them from each other. The video packet sizes are more or less fixed by user defined encoder settings. The number of MB’s within a video packet depends on the video sequence characteristics. For example, high motion sequences produces more information within a MB, hence fewer number of MB’s are contained in a VP. Following the concept of data partitioning, data within a video packet is further divided in to two main parts, namely the first partition and second partition. The motion related information for all the MB’s contained in a given VP are placed in the first partition and the relevant DCT data is placed in the second partition. Motion Boundary Marker (MBM) is used to indicate end of first partition [TALL-98].

Combination of VPs in a video frame forms the Video Object Plane (VOP). The most important information the decoder needs to know prior to the decoding of compressed video data, is placed in the VOP header part. This includes the spatial dimensions of the video frame, the time stamps associated with the current frame, presentation information and the mode in which the current frame is coded. If some of this information is corrupted due to channel errors, the decoder has to discard all the information belonging to the current video frame whether or not it has been received correctly. The Header Extension Code (HEC), which is included in the MPEG-4 standard, reduces the sensitivity of header information to channel errors by repeating important header information in selected video packets. The repetition is controlled by introducing a 1-bit field called the HEC. If the HEC is set to one, header information is repeated. This method allows the decoder to be certain about the header information by cross checking. If the VOP header is corrupted, the decoder can still decode the rest of the data in the video frame using the header information within the video packets.
Chapter 5. Distortion Modelling and Optimal MTU Calculation

5.2.2 Performance modelling

Video performance is often shown as a combination of quantisation distortion and channel distortion [KIM-03]. Quantisation distortion is mainly a function of the video source rate, resulting in low distortion at high source rates and is un-recoverable at the decoder. Channel distortion can further be divided into three parts; namely spatial concealment distortion, temporal concealment distortion and distortion caused from error propagation over predictive frames.

If the video frame configuration information and video packet headers are lost during transmission, the decoding process is impossible and the data belonging to that video packet has to be discarded at the decoder. However, the error concealment tools implemented at the decoder replaces the discarded data with error-concealed data from neighbouring packets. The distortion resulting from this process is called spatial- concealment distortion, as only spatial error concealment is involved in the process. On the other hand, if the configuration information, video packet header information and motion information are received correctly, but the DCT information is corrupted, then the decoder only discards the corrupted DCT data and replaces it with the corresponding concealed data from the previous frame. The distortion resulting from motion compensated error concealment is called temporal concealment distortion. Research has shown that temporal error concealment is more powerful than the spatial error concealment,
especially in low motion video sequences, resulting in better perceptual video quality [WANGb-00].

Errors can propagate in two ways, either in the temporal domain or in the spatial domain. Frame to frame error propagation through motion prediction and temporal concealment is called temporal domain error propagation. This can be prevented by the regular insertion of intra-coded frames or the use of adaptive intra refresh techniques. Propagation of errors from neighbouring video packets via spatial concealment is considered as spatial domain error propagation. Compared to temporal error propagation, spatial domain error propagation is less severe and only a small region of the video frame is affected.

Taking the video packet as the base unit, the expected frame quality and the total frame distortion can be written as

\[ E(Q^j_f) = 10 \cdot \log \left( \frac{g}{E(D^j_f)} \right) \]
\[ E(D^j_f) = \sum_{i=0}^{I} E(D_{pv}^{i,j}) \]

where \( E(Q^j_f) \) and \( E(D^j_f) \) are the expected quality and the total expected distortion, \( E(D_{pv}^{i,j}) \) is the expected distortion and \( I \) is the total number of video packets in the video frame. \( i \) and \( j \) represent the \( i^{th} \) video packet of the \( j^{th} \) video frame, \( g \) is a constant defined by the dimension of the frame. For QCIF format video \( g = 255^2 \times 176 \times 144 \).

Now, \( E(D_{pv}^{i,j}) \) can be written as

\[ E(D_{pv}^{i,j}) = E(D_{Q, pv}^{i,j}) + P_{u, pv}^{i,j} E(D_{s, con, pv}^{i,j}) + P_{d, pv}^{i,j} E(D_{t, con, pv}^{i,j}) + f_{tp}^{i,j} + f_{sp}^{i,j} \]

\( E(D_{Q, pv}^{i,j}), E(D_{s, con, pv}^{i,j}) \) and \( E(D_{t, con, pv}^{i,j}) \) represent the expected quantisation distortion, spatial concealment distortion and temporal concealment distortion respectively. \( P_{u, pv}^{i,j} \) denotes the probability of receiving an un-decodeable video packet. This includes the corruption of VOP header, VP header or motion data. The probability of receiving a decodeable video packet but with errors, where the DCT data is corrupted but other information is received correctly is denoted by \( P_{d, pv}^{i,j} \). \( f_{tp}^{i,j} \) and \( f_{sp}^{i,j} \) represent the distortion cause from the propagation of errors in the temporal domain and spatial domain respectively.
5.2.2.1 Probability calculation

Assuming that the probability of receiving a VOP header with errors is $p_{l,J_{VOP}}$, the probability of receiving the video packet header and motion information with errors is $p_{l,J_{M}}$ and the probability of finding an error in the DCT part is $x_{l,J}$. Then

$$p_{l,J_{VOP}} = 1 - (1 - p_{l,J_{VOP}})(1 - p_{l,J_{M}})$$ \hspace{0.5cm} \text{Equation 5.3}

$$p_{l,J_{d,pv}} = (1 - p_{l,J_{VOP}})(1 - p_{l,J_{M}})x_{l,J}$$ \hspace{0.5cm} \text{Equation 5.4}

If the probability of channel bit errors is denoted by, $p_b$, then it can be shown,

$$p_{l,J_{VOP}} = \sum_{v=1}^{V} (1 - p_b)^{v-1} p_b = (1 - (1 - p_b)^V)$$ \hspace{0.5cm} \text{Equation 5.5}

where $V$ represents the VOP header size. Similarly,

$$p_{l,J_{M}} = 1 - (1 - p_b)^{Z_M}$$ \hspace{0.5cm} \text{Equation 5.6}

$$x_{l,J} = \sum_{z=1}^{Z_{DCT}} (1 - p_b)^{Z_{DCT} - z} p_b^z = 1 - (1 - p_b)^{Z_{DCT}}$$ \hspace{0.5cm} \text{Equation 5.7}

where $Z_{DCT}$ and $Z_M$ denote the length of the DCT and the length of the video packet header and motion information (the first partition) of the $i^{th}$ video packet of the $j^{th}$ frame respectively.

5.2.2.2 Distortion calculation

5.2.2.2.1 Quantisation distortion

Let $\phi_{k,i,j}(u,v)$ and $\tilde{\phi}_{k,i,j}(u,v)$ be the original and reconstructed luminance values of the $(u,v)$ pixel of the $k^{th}$ Macro Block (MB) of the $i^{th}$ video packet in the $j^{th}$ video frame respectively. Video MB reconstruction follows the inverse of the encoding process as shown in Figure 5.2. The quantisation distortion of a MB can now be defined as

$$E(D^{k,i,j}_{Q,MB}) = \sum_{u=1}^{16} \sum_{v=1}^{16} (\phi_{k,i,j}(u,v) - \tilde{\phi}_{k,i,j}(u,v))^2$$ \hspace{0.5cm} \text{Equation 5.8}

and the total quantisation distortion of video packet is given by

$$E(D^{i,j}_{Q,pv}) = \sum_{k=1}^{K} E(D^{k,i,j}_{Q,MB})$$ \hspace{0.5cm} \text{Equation 5.9}

where $K$ is the total number of MB's in the $i^{th}$ video packet.
5.2.2.2 Concealment distortion

Concealment distortion is computed in a similar manner to quantisation distortion. As shown in Figure 5.3, the transmitted video data belonging to each MB is corrupted using the noise generator located at the encoder. Corrupted data is replaced by the concealed data and data belonging to the original and concealed MB's is compared. Thus concealment distortion of a MB can be written as

\[ E(D_{\text{con},MB}^{k,l}) = \sum_{\nu=1}^{16} \sum_{\nu=1}^{16} (\phi_{k,l,j}(u,\nu) - \tilde{\phi}_{k,l,j}(u,\nu))^2 \]

Equation 5.10

where \( \phi_{k,l,j}(u,\nu) \) indicates the luminance values of the \((u,\nu)\) pixel of the \(k^{th}\) MB after the concealment. And the total concealment distortion of the video packet is

\[ E(D_{\text{con},pv}^{l,j}) = \sum_{k=1}^{K} E(D_{\text{con},MB}^{k,l}) \]

Equation 5.11
In the case of spatial concealment distortion calculation, only the spatial concealment algorithms are used to generate the concealed data. On the other hand, only the temporal concealment algorithm is used to conceal the corrupted data in the temporal concealment distortion calculation. The correct reception of neighbouring video packets and reference video frames is assumed in the calculation of $E(D_{s\_con,pv}^{i,j})$ and $E(D_{t\_con,pv}^{i,j})$.

### 5.2.2.3 Propagation loss modelling

The expected distortion calculation described in previous section assumes error free neighbouring packets and reference video frames. Correlation, between the corrupted video packets in the same frame and the distortion due to the MB mismatch in adjacent video frames, is quantified by the spatial and temporal error propagation terms in Equation 5.2.

The temporal propagation loss, $f_{tp,mb}^{k,j}$, represents the propagation of corrupted information through predictive coding. An Adaptive Intra Refresh (AIR) algorithm, which uses a selected number of intra coded MBs in a video frame, is used to prevent the error propagation. The temporal error propagation is calculated at MB level. Let $f_{tp,mb}^{k,j}$ be the temporal error propagation of $k^{th}$ MB in $j^{th}$ frame. $f_{tp,mb}^{k,j}$ depends on the coding mode used in the encoding of the MB. $f_{tp,mb}^{k,j}$ is calculated as

$$
egin{align*}
 f_{tp,mb}^{k,j} &= P_{sp,mb}^{k,j} \cdot E(D_{s\_con,mb}^{k,j}) + P_{d,mb}^{k,j} \cdot E(D_{t\_con,mb}^{k,j}) \\
 &+ f_{sp,mb}^{k,j} + (1 - P_{sp,mb}^{k,j}) \cdot f_{tp,mb}^{k,j-1} & \text{for inter coded MBs} \\
 f_{tp,mb}^{k,j} &= P_{sp,mb}^{k,j} \cdot E(D_{s\_con,mb}^{k,j}) + f_{sp,mb}^{k,j} & \text{for intra coded MBs}
\end{align*}
$$

Equation 5.12

where $E(D_{s\_con,mb}^{k,j})$ and $E(D_{t\_con,mb}^{k,j})$ represent the spatial concealment distortion and temporal concealment distortion of the MB respectively. As before, $i$ and $j$ represent the $i^{th}$ video packet in $j^{th}$ video frame. $f_{sp,mb}^{k,j}$ is the spatial error propagation of the $k^{th}$ MB in $j^{th}$ video frame. $P_{sp,mb}^{k,j}$ quantifies the fraction of distortion of the Macro Block, which should be considered in the propagation loss calculation. The $P_{sp,mb}^{f}$ is a function of channel bit error rate, $\rho_b$, and experimentally it is found to be approximated to
5.2.3 Evaluation of the accuracy of the performance estimation

The frame PSNR value of every frame can be estimated at the encoder by using the developed distortion model shown in Section 5.2.2. However, the accuracy of the model is limited to certain range of channel bit error probabilities due to the assumption made in the derivation. We assumed that the quantisation distortion, temporal concealment distortion, and spatial concealment distortion are uncorrelated in Equation 5.2. At the presence of high channel bit error rates this assumption may not be valid thus, the expected frame PSNR may deviate from the actual performance. Also, the spatial and temporal domain error propagations are calculated under the assumption that video reconstruction errors are uniformly distributed in the frame. However, this is somewhat unreasonable for video performance estimation in a burst error environment such as a UMTS multi-path operating environment. Therefore, the accuracy of the model is evaluated by comparing the estimated performance and the actual video performance over a simulated UMTS environment.
In the simulation, two ITU test sequences, namely low motion “Suzie” sequence and high motion “Foremen” sequence are used. Sequences are encoded at 64 kbps and 137 kbps, which are corresponding to spreading factor 32 and 16 operations. Frame rate is set to be 10 fps. Other video encoding parameter settings are shown in Table 5.1.

MPEG-4 encoded sequences are transmitted over the simulated Vehicular A environment. 1/3 rate convolution coding with spreading factor 32 and 16 operations are considered. Experiments were carried out for a range of channel conditions, where the channel BLock Error Rates (BLER) experienced by the information data vary from 0.0003 to 0.0130 and from 0.0010 to 0.0240 for the spreading factor 32 and 16 respectively. The frame PSNR values estimated by the developed model are compared to the measured frame PSNR values. Each experiment was repeated 20 times in order to capture the effect of bursty channel errors on the performance, and average values are plotted in Figure 5.4 and Figure 5.5 for the “Suzie” and “Foreman” sequences respectively.
Figure 5.4: performance comparison between theoretical and actual frame PSNR for transmission of “Suzie” over Vehicular A channel.
Figure 5.5: performance comparison between theoretical and actual frame PSNR for transmission of "Foreman" over Vehicular A channel.
Chapter 5. Distortion Modelling and Optimal MTU Calculation

The estimated PSNR values closely match the actual PSNR values in good channel conditions. However, as expected, the estimation error becomes larger for operation in low quality channels. Compared to the high source rate, low source rates show closer performances to the actual values. Referring to Figure 5.4(f) and Figure 5.5(f), the model shows better performance estimation for low motion sequences especially at high error rate channel conditions. It should also note that 20 simulation runs might not be sufficient to capture the effect of bursty channel errors on the performance fully, especially at poor quality channels.

5.3 Video performance modelling over packet switched networks

As shown in Section 4.4.2.2, the performance of video communication over wireless channels is greatly influenced by the introduction of IP packetisation. The resulting video quality over packet switched connections is degraded by a considerable amount when compared to that of circuit switched connections. In this section, the derived distortion model is modified in such a way that the presence of packet headers is taken into account for video communications over packet switched networks.

5.3.1 Video packetisation issues

The output bit stream of a video encoder should be segmented into transport layer packets for operation over packet switched networks. The video packetisation should be conducted so as to enhance the error resiliency provided by the error resilience techniques incorporated in the MPEG-4 encoder. In other words, each transport layer packet should consist of number of independently decodable video units. This requirement mitigates the error propagation from one corrupted transport layer packet to another.

Video packetisation uses the RTP packetisation rules specified for MPEG-4 visual content in [RFC 3016]. The segmentation rules recommend not to map more than one VOP in an RTP packet allowing unique indication of VOP frame timing by RTP time stamp. However, in the case of very small VOPs, the segmentation rules, permits the concatenation of multiple VOPs in a RTP packet to increase the protocol efficiency. RTP packet size is defined by the setting of Maximum Transmission Unit (MTU). If the VOP size exceeds the allowed MTU size for the transmission, packet segmentation is performed. Each RTP packet contains an integer number of video packets.
Figure 5.6 shows the RTP packet format for MPEG-4 visual content. The VOP header follows the transport packet header information, which is placed at the beginning of the packet and proceeds with video packet information. The number of video packets encapsulated into one RTP packet is calculated in such a way that the resulting transport packet is not larger than the selected MTU size ($L_m$). The video packet starts with the header extension code (if Header Extension Code is enabled) or the video packet header; followed by the motion information, texture information, and ends with the resynchronisation marker. One video frame is encapsulated into $M$ RTP packets. The VOP header is repeated $N$ times. Assume $N \leq M$ and no more than one VOP header extension code is present in an IP packet.

![Figure 5.6: Video packet format and video packetisation.](image)

<table>
<thead>
<tr>
<th>$1^{st}$ RTP packet</th>
<th>$2^{nd}$ RTP packet</th>
<th>$3^{rd}$ RTP packet</th>
<th>…</th>
<th>$M^{th}$ RTP packet</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video frame</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

$X$ - RTP header size  
$Y$ - VOP header size  
$Y + \Delta Y$ - Video Packet

$Z_m$ - length of the first partition
$Z_{DCT}$ - length of the second partition

Number of VP's in a RTP packet $= \left\lfloor \frac{L_m}{Y} \right\rfloor$
5.3.2 Distortion model modification to packet switched connection

In previous sections video performances are determined frame by frame for transmission over circuit switched connection. Equation 5.1 is rewritten for average frame PSNR as

\[
\bar{E}(Q_f) = 10 \cdot \log \left( \frac{g}{\bar{E}(D_f)} \right) \quad \text{Equation 5.18}
\]

\[
\bar{E}(D_f) = I \cdot \bar{E}(D_{pv})
\]

where \(\bar{E}(Q_f)\) and \(\bar{E}(D_f)\) are the expected average frame quality and the expected average frame distortion, \(\bar{E}(D_{pv})\) denotes the expected average distortion of a video packet. \(I\) is the average number of video packets in a frame.

In the presence of network layer header corruption, the information data encapsulated into that packet should be discarded. This causes the loss of large amounts of data, hence accurate concealment is difficult, resulting in severe spatial concealment distortion. A new spatial concealment distortion term is introduced in the total distortion calculation. The expected average video packet distortion \(\bar{E}(D_{pv})\) for packet switched connections can be written as

\[
\bar{E}(D_{pv}) = \bar{E}(D_{Q_{pv}}) + \bar{P}_{h_{pv}} \bar{E}(D_{s_{con_{pv}}}) \\
+ \bar{P}_{d_{pv}} \bar{E}(D_{l_{con_{pv}}}) + \bar{P}_{um_{pv}} \bar{E}(D_{sm_{con_{pv}}}) \\
+ \bar{f}_{lp} + \bar{f}_{sp}
\]

Equation 5.19

where \(\bar{E}(D_{sm_{con_{pv}}}\) represents the expected spatial concealment distortion due to the IP packet corruption. \(\bar{P}_{um_{pv}}\) is the probability of network layer packet header corruption. Others are average values of the terms defined in Equation 5.2.

The probability calculations of Equation 5.18 are directly influenced by the introduction of IP packetisation, as the corruption of IP packet headers results in discarded video packets. Because of the IP packet overhead, the resulting \(\bar{E}(D_{Q_{pv}})\) is greater than that of non-packetised scenarios. \(\bar{E}(D_{s_{con_{pv}}}\), \(\bar{E}(D_{sm_{con_{pv}}}\) and \(\bar{E}(D_{l_{con_{pv}}}\) are mainly decided by the decoder concealment techniques. The presence of IP packetisation can be assumed to have an insignificant effect on the \(\bar{E}(D_{s_{con_{pv}}}\) and \(\bar{E}(D_{l_{con_{pv}}}\). However, as the spatial concealment algorithm under-performs in circumstances involving large amounts of information loss, \(\bar{E}(D_{sm_{con_{pv}}}\) will be greatly

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affected by the IP packetisation. Mathematical derivations of terms used in Equation 5.18 in the presence of IP packetisation are discussed below.

5.3.2.1 Probability calculation

The influence of the network layer packet header on the distortion calculation can be quantified by the introduction of term $\tilde{\rho}_{IP}$, which represents the probability of receiving an IP header with errors.

Then, $\tilde{\rho}_{u,pv}, \tilde{\rho}_{d,pv}$ and $\tilde{\rho}_{um,pv}$ for packet switched connections can be written as:

$$\tilde{\rho}_{u,pv} = 1 - (1 - \tilde{\rho}_{IP})(1 - \tilde{\rho}_{VOP})(1 - \tilde{\rho}_M)$$  
Equation 5.20

$$\tilde{\rho}_{d,pv} = (1 - \tilde{\rho}_{IP})(1 - \tilde{\rho}_{VOP})(1 - \tilde{\rho}_M)\tilde{\chi}$$  
Equation 5.21

$$\tilde{\rho}_{um,pv} = \tilde{\rho}_{IP}$$  
Equation 5.22

For a given probability of channel bit error, $\rho_b$, and RTP header size, $X$, $\tilde{\rho}_{IP}$ can be computed as,

$$\tilde{\rho}_{IP} = \sum_{i=1}^{X}(1 - \rho_b)^{i-1}\rho_b = 1 - (1 - \rho_b)^X$$  
Equation 5.23

$\tilde{\rho}_{VOP}, \tilde{\rho}_M$ and $\tilde{\chi}$ are computed as,

$$\tilde{\rho}_{VOP} = \sum_{i=1}^{\nu}(1 - \rho_b)^{i-1}\rho_b = 1 - (1 - \rho_b)^\nu$$

$$\tilde{\rho}_M = 1 - (1 - \rho_b)\tilde{Z}_M$$  
Equation 5.24

$$\tilde{\chi} = \sum_{z=1}^{Z_{DCT}} \left( Z_{DCT} \right) (1 - \rho_b)^{\tilde{Z}_{DCT} - z} \rho_b^z = 1 - (1 - \rho_b)^{\tilde{Z}_{DCT}}$$

$\tilde{Z}_M$ and $\tilde{Z}_{DCT}$ are the average lengths of the video packet header and motion information (the first partition) and DCT respectively.

For fixed IP header sizes (assume no IP header compression) $\tilde{\rho}_{IP}$ is clearly independent of the MTU size. Similarly $\tilde{\rho}_{VOP}$ only depends on the size of the VOP header part and the occurrence of a header extension within the video packet. $\tilde{\rho}_M$ and $\tilde{\chi}$ are slightly affected by the MTU size as it
might change the encoder quantisation step size, hence $Z_{dct}$ and $Z_m$. However, for fixed video packet size (as used in the experiments discussed in this chapter), the probability terms $\bar{\rho}_M$ and $\bar{\chi}$ can be considered to be independent of MTU size.

### 5.3.2.2 Distortion modelling

Both the quantisation distortion and concealment distortion are calculated as described in Section 5.2.2.2. The temporal concealment distortion and spatial concealment distortion of the corrupted video packet assume the correct reception of neighbouring video packets, while the calculation of $E(D_{sm\_con,pv})$ relies on the correct reception of neighbouring transport layer packets.

Average distortion calculations can be written as,

$$
\bar{E}(D_{Q,pv}) = \frac{1}{J} \sum_{j=1}^{J} \frac{1}{I'_{j}} \sum_{i=1}^{I'_{j}} E(D_{I'_{j},Q,MB}) \\
\text{Equation 5.25}
$$

$$
\bar{E}(D_{con,pv}) = \frac{1}{J} \sum_{j=1}^{J} \frac{1}{I'_{j}} \sum_{i=1}^{I'_{j}} E(D_{I'_{j},con,MB}) \\
\text{Equation 5.26}
$$

where, $J$ and $I'$ are the total number of video frames and the total number of video packets in $j^{th}$ frame respectively.

### 5.3.2.3 Propagation loss modelling

The average temporal propagation loss, $f_{tp}$, is computed as

$$
f_{tp} = P_{tp} \cdot [\bar{\rho}_{u,pv} \bar{E}(D_{s\_con,pv}) + \bar{\rho}_{d,pv} \bar{E}(D_{i\_con,pv}) + \bar{\rho}_{um,pv} \bar{E}(D_{sm\_con,pv}) + \bar{f}_{sp}] \\
\text{Equation 5.27}
$$

$$
P_{tp} = f_{refresh} \cdot (1 - (1 - \rho_b)^F) \\
\text{Equation 5.28}
$$

where $f_{refresh}$ is the effective intra frame refreshing frequency and $F$ is the average frame size.

Similarly, the average spatial error propagation, $f_{sp}$ is modelled as,

$$
f_{sp} = P_{sp} \cdot [\bar{\rho}_{u,pv} \bar{E}(D_{s\_con,pv}) + \bar{\rho}_{um,pv} \bar{E}(D_{sm\_con,pv})] \\
\text{Equation 5.29}
$$

$$
P_{sp} = N_{sp} \cdot (1 - (1 - \rho_b)^{IP}) \\
\text{Equation 5.30}
$$
where $N_{vp}$ and $VP$ represent the average number of video packets in a frame and the average size of a video packet respectively.

5.3.3 Experimental results

Transmission of the "Suzie" sequence over a simulated UMTS channel is considered in the experiments. Video performances over circuit switched and packet switched connections are shown in terms of average frame Peak Signal to Noise Ratio (PSNR) vs varying channel conditions in Figure 5.7. The theoretical calculations are shown in thick lines while the actual performance curves are shown in dashed lines. The performance calculation for circuit switched connections use the same distortion model with $\tilde{\rho}_{IP} = 0$. For packet switched connections, an MTU size of 576 bytes is used. Figure 5.7 clearly illustrates the close match between the experimental and the theoretical results. Furthermore, the figure shows the video quality loss resulting from the use of packet switched connections instead of circuit switched connections ranges from 1 – 3 dB.

![Figure 5.7: comparison of PS and CS performances over Vehicular A channel. SF- 32 and 1/3 rate convolution coding. Source rate 64 kbps.](image-url)
5.4 Optimal MTU size

Extensive research activity has been carried out in the area of optimisation of IP and its underlying network protocols (e.g. RTP/RTCP, UDP) for efficient network utilisation [VARS-00, STOC-03, WORRb-01]. Although IP allows network nodes to fragment packets that are too large to be forwarded, fragmentation not only results in poor network performance but can also lead to total communication failure in some circumstances [KENT-97]. On the other hand, datagrams that are much smaller than the allowed MTU size waste network resources, and result in sub-optimal system throughput. IP path MTU discovery [MOGU-90] is used to obtain the optimal maximum packet size for a given connection while avoiding datagram fragmentation along the transmission path. Design criteria mainly follow the response time requirement, where the interactive response time is limited to 100 to 200 ms [JACO-90]. However, little attention is given to issues regarding the higher loss probability of the wireless link on the design and optimisation of the IP path MTU discovery algorithms. The effect of packet headers on video performance has been analysed in [HONG-02] without considering error resilience or error concealment techniques. This section analyses the effect of packet size on the performance of wireless video applications. By contrast to the method proposed in [HONG-02], full error resilience MPEG-4 coded video is considered in the investigation. Using the developed distortion model, the optimal packet size for a given channel condition is derived. Theoretical analysis was validated by comparing to the actual performances.

5.4.1 Problem formulation

To achieve better video quality in bandwidth constrained error prone packet networks, it is necessary to maximise the source bit-rate, and at the same time minimise the effects of packet loss. A larger transport packet reduces the packet overhead resulting in higher source throughput. However, accommodating a large amount of information in one packet results in a large amount of information loss at one time if the packet is lost due to packet header corruption. Therefore, the optimal packet size is a function of packet overhead and packet loss ratios and it greatly depends on the operating environment.

The goal set here is to find the optimal IP packet size (MTU size), which maximises the received video quality for video transmission over a bandwidth limited error prone channel. The channel is
characterised by the probability of channel bit errors and the channel bandwidth. The optimisation
problem can formally be written as

\[
\text{Minimise } \quad \mathcal{E}(D_f) = I \cdot \mathcal{E}(D_{pv})
\]

subject to \( OH_{MTU} + R_{source} = R_{channel} \)

where, \( OH_{MTU} \), \( R_{source} \) and \( R_{channel} \) denote overhead rate due to IP packetisation, source rate and
channel bit rate respectively. Other terms are as specified in Section 5.3.2.

The distortion \( \mathcal{E}(D_{Q,pv}) \) with varying packet sizes is experimentally evaluated and results are
shown in Figure 5.8. ITU sequence "Suzie", which is encoded at 64 kbps and 137 kbps are used
in the investigation. As expected \( \mathcal{E}(D_{Q,pv}) \) decreases with the increasing MTU size. The relative
increase of \( \mathcal{E}(D_{Q,pv}) \) with small MTU size is more evident for low bit rate channels.

![Figure 5.8: Variation of \( \mathcal{E}(D_{Q,pv}) \)](image)

Figure 5.9, Figure 5.10 and Figure 5.11 illustrate the effect of MTU size upon the expected
distortions; \( \mathcal{E}(D_{l\_con,pv}) \), \( \mathcal{E}(D_{s\_con,pv}) \) and \( \mathcal{E}(D_{sm\_con,pv}) \) respectively. \( \mathcal{E}(D_{s\_con,pv}) \) and
\( \mathcal{E}(D_{l\_con,pv}) \) show more or less constant values with varying MTU sizes. This is because, both
terms assume the correct reception of IP header in the calculation. Hence the introduction of IP
packetisation does not directly affect the concealment performance. However, \( \mathcal{E}(D_{sm\_con,pv}) \),
where the corruption of IP headers is taken into account, is greatly affected by the IP packetisation. Figure 5.11 shows a dramatic increase in $\bar{E}(D_{sm\_con,pv})$ with increasing MTU size for relatively low MTU sizes. However, it reaches an asymptotical maximum value when MTU size gets closer to the VOP size. As VOPs are segmented into separate IP packets, a significant number of IP packets are resulted much smaller than the specified MTU size. This reduces the effectiveness of MTU size upon $\bar{E}(D_{sm\_con,pv})$ at large MTU sizes.

Figure 5.9: Variation of $\bar{E}(D_{t\_con,pv})$

Figure 5.10: Variation of $\bar{E}(D_{x\_con,pv})$
5.4.2 Results and analysis

Using the developed distortion model, the optimal packet sizes for the transmission of video over the UMTS multi-path propagation environment with varying channel conditions are calculated and the results are shown in Figure 5.12. The total available channel bit rates of 64 kbps and 137 kbps (corresponding to spreading factors of 32 and 16) are used in the simulation. The Vehicular A propagation environment with 1/3 rate convolution code is used. The corresponding channel block error rates for a spreading factor of 32 are listed in Column 2 of Table 5.3.
Chapter 5. Distortion Modelling and Optimal MTU Calculation

The performance can be divided into three main regions. Small packet sizes show better performance for poor channel conditions, while long packet sizes are preferable with good channel conditions. Packet size variation from small to long is visible with moderate channel conditions.

This behaviour can be further explained based on the expected distortion values, which are described in Section 5.3.2. Grouping the distortion terms, Equation 5.18 can be re-written as

\[ \tilde{E}(D_{pv}) = \lambda_1 + \lambda_2 + \lambda_3 \]  

where

\[ \lambda_1 = \tilde{E}(D_{Q,pv}) \]
\[ \lambda_2 = \tilde{E}(D_{s,con,pv}) \cdot \beta_{u,pv}(1 + P_{ip} + P_{sp} + P_{dp}) \]
\[ \lambda_3 = \tilde{E}(D_{s,m,con,pv}) \cdot \beta_{s,m,pv}(1 + P_{ip} + P_{sp} + P_{dp}) \]

The calculated \( \lambda_1, \lambda_2, \) and \( \lambda_3 \) values for 576 bytes MTU size with varying channel conditions are listed in Table 5.3. The last column in the table shows the Dominating distortion Factor (DF) for a given channel Block Error Rate.

<table>
<thead>
<tr>
<th>Eb/No</th>
<th>BLER</th>
<th>( \lambda_1 )</th>
<th>( \lambda_2 )</th>
<th>( \lambda_3 )</th>
<th>DF</th>
</tr>
</thead>
<tbody>
<tr>
<td>12</td>
<td>0.0000</td>
<td>61044</td>
<td>0</td>
<td>0</td>
<td>( \lambda_1 )</td>
</tr>
<tr>
<td>10</td>
<td>0.0010</td>
<td>61044</td>
<td>992.6</td>
<td>1091.6</td>
<td>( \lambda_1 )</td>
</tr>
<tr>
<td>9</td>
<td>0.0030</td>
<td>61044</td>
<td>5197.2</td>
<td>5385.2</td>
<td>( \lambda_1 )</td>
</tr>
<tr>
<td>8</td>
<td>0.0130</td>
<td>61044</td>
<td>7760.1</td>
<td>56880.5</td>
<td>( \lambda_2 )</td>
</tr>
<tr>
<td>7</td>
<td>0.0460</td>
<td>61044</td>
<td>34667.4</td>
<td>341978.1</td>
<td>( \lambda_2 )</td>
</tr>
<tr>
<td>6</td>
<td>0.1371</td>
<td>61044</td>
<td>1157580</td>
<td>1155251</td>
<td>( \lambda_3 )</td>
</tr>
<tr>
<td>5</td>
<td>0.3173</td>
<td>61044</td>
<td>2702683</td>
<td>3025136</td>
<td>( \lambda_3 )</td>
</tr>
<tr>
<td>4</td>
<td>0.5335</td>
<td>61044</td>
<td>4752671</td>
<td>6085720</td>
<td>( \lambda_3 )</td>
</tr>
<tr>
<td>3</td>
<td>0.7828</td>
<td>61044</td>
<td>7332380</td>
<td>12002987</td>
<td>( \lambda_3 )</td>
</tr>
</tbody>
</table>

For channels where \( E_b/N_0 \) is less than 6 dB, \( \lambda_3 \) dominates the others. \( \lambda_3 \) is proportional to the \( \tilde{E}(D_{s,m,con,pv}) \). As can be seen in Figure 5.11, \( \tilde{E}(D_{s,m,con,pv}) \) increases with increasing MTU size, hence a small MTU size is expected to have a better performance for poor channel conditions (channel \( E_b/N_0 \) less than 6 dB). For good channel conditions, \( \lambda_3 \), which is the quantisation distortion, significantly dominates \( \lambda_2 \) and \( \lambda_3 \). The quantisation distortion decreases with increasing packet size. Therefore, long packets are preferable for good channel conditions. However, for moderate channel conditions where channel \( E_b/N_0 \) varies from 6 to 8 dB, \( \lambda_2 \) becomes significant compare to the others. According to Equation 5.27, \( \lambda_2 \) is proportional to
\( E(D_{s\_con,pv}) \) and \( E(D_{t\_con,pv}) \), which are independent of MTU size. Therefore, video performance is expected to be independent of the packet size for moderate channel conditions. These theoretical analyses are verified by the experimental results shown in Figure 5.13.

A performance improvement of 2-3 dB of frame PSNR can be achieved by using optimal packetisation for good channels, while an improvement of 1-2 dB is possible for poor channel conditions. The performance is independent of the packet size with moderate channel conditions. As stated in Chapter 4, the resulting video output is unacceptable in terms of visual quality if the average frame PSNR value is lower than 20 dB. Taking 20 dB as the marginal value, we can conclude that the use of the largest possible MTU size provides optimal results in terms of acceptable quality video over wireless networks.

![Figure 5.13: Video performance over packet networks](image)

### 5.5 Conclusion

This chapter described a frame-by-frame distortion estimation method for video communications over wireless networks. By contrast to most of the performance estimation methods proposed in the literature, the model was designed for full error resilience enabled MPEG-4 video applications. A comparison between the actual performances, which are obtained for video
transmission over simulated UMTS propagation environments, and the theoretical performance, verified the model accuracy for applications over mobile networks.

As shown in Chapter 4, the performance of video applications over packet switched networks degrades in terms of perceptual quality, because of the information loss caused by corrupted packet headers, and the throughput loss due to the packet header overhead. Larger packet size results in less throughput loss. However, a large amount of information data will be lost in the presence of packet header corruption. On the other hand, a smaller packet size increases the overhead rates. Therefore, the optimal packet size should be carefully designed to minimise the distortion seen with error prone channels. Using the developed distortion model, the optimal packet size for video transmission over wireless network is obtained. The experiment results show that a significant performance gain can be achieved with the use of optimal packet sizes in packet switched networks.

Frame by frame distortion estimation at the encoder facilitates a wide variety of techniques, which are capable of enhancing the received video quality. Novel unequal error protection and unequal power allocation techniques, which are based on the developed distortion model for video communications over UMTS networks will be proposed and analysed in Chapter 6.
Chapter 6

6 Link Level Quality Enhancement Techniques

6.1 Introduction

As mentioned in Section 2.3, error resilience techniques incorporated into the video compression algorithm can improve the robustness of video data to channel degradation. The error resilience tools provide a significant quality improvement under error prone channel conditions, particularly with channel bit error rates in the order of $10^{-3}$ to $10^{-4}$. This chapter will introduce new link level quality improvement techniques, which can further enhance the error resiliency of information data when transmitted over mobile channels. These techniques exploit different levels of importance of information data for perceptual quality. Two different methods are used to prioritise information data for each transmission. Method 1 uses the MPEG-4 data-partitioning concept to separate data into two streams. More accurate data prioritisation is achieved with Method 2, which uses a frame-by-frame distortion calculation (as described in Chapter 5) to estimate the relative importance of multimedia data. Based on the priority level, Unequal Error Protection (UEP) and Unequal Power Allocation (UPA) techniques are applied to enhance the quality of received data at the decoder. Algorithm performances are assessed for MPEG-4 coded video transmission over the simulated UMTS network. Results illustrate a significant performance improvement in terms of received video quality as well as of transmission efficiency.
6.2 Data Partition based Unequal Error Protection (DP based UEP) scheme

UEP techniques have been widely used to enhance video performance over error-prone channels [GHAR-01, RABI-98]. They can be designed with the use of rate compatible codes at the encoder [RABI-98]. Here, the source block length is fixed, and differential protection is achieved via varying channel block length. Using an iterative descent algorithm, optimal error control is obtained. Although this gives a performance enhancement, the algorithm shows some limitations for practical implementation. As the channel protection mechanisms are coupled with the video compression formatting at the application layer, this requires modification of all underlying network protocol layers, limiting the video transmission in heterogeneous networks.

This problem can be solved by prioritisation of different parts of the video bit stream by sending them using multiple bearers. The data partitioning used in MPEG-4 coders may be exploited as a method of information prioritisation [WORRe-01]. The data belonging to the first partition is sent over a highly protected channel while the second partition data is transmitted using a less reliable channel.

6.2.1 Information prioritisation

When transmitting a single video sequence using a number of streams, it is necessary to ensure synchronisation between the streams. At the receiver, the streams should be combined to form a single decodable stream. In other words, the stream should be formatted in such a way that the combining algorithm can determine whether the data in multiple streams corresponds to the same video frames and video packets.

Video frame time stamps are presented in the Video Object Plane (VOP) header. For frame synchronisation, it is necessary to include VOP headers in both streams. Video packet level synchronisation information, including the macroblock number of the first macroblock contained in the packet, is coded within the Video Packet (VP) header. As the VP header is located within the first partition, it is unnecessary to place any extra information on the first partition stream. However, VP header information and a resynchronisation marker are added to the second partition data as shown in Figure 6.1.
Chapter 6. Link Level Quality Enhancement Techniques

<table>
<thead>
<tr>
<th>VOP start code</th>
<th>H E C</th>
<th>Header extension information</th>
</tr>
</thead>
<tbody>
<tr>
<td>a). Second partition stream frame header</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Resync marker</th>
<th>MB num</th>
</tr>
</thead>
<tbody>
<tr>
<td>b). Second partition stream packet header</td>
<td></td>
</tr>
</tbody>
</table>

Figure 6.1: Extra data added to second partition stream [WORRe-01].

The extra overhead incurred by this technique mainly depends on the video frame rate, $f$, video packet size, $V_p$, and source rate, $R_s$. If VOP and VP header sizes are denoted by $X$ and $Y$ respectively, then the approximate overhead incurred, $OH$, is given by

$$OH = \left(\frac{R_s}{V_p \cdot f} \cdot \frac{Y \cdot f}{R_s} + \frac{X \cdot f}{R_s}\right) \times 100\%$$

Equation 6.1

For the sequences tested, the extra overhead generated amounted to approximately 3-4% of the overall rate. The overhead was calculated using the parameter values shown in Table 6.1.

Table 6.1: source parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frame rate, $f$</td>
<td>10 fps</td>
</tr>
<tr>
<td>Video packet size, $V_p$</td>
<td>700 bits</td>
</tr>
<tr>
<td>Source rate, $R$</td>
<td>60 – 90 kbps</td>
</tr>
<tr>
<td>VOP header size, $X$</td>
<td>56 bits</td>
</tr>
<tr>
<td>VP header size, $Y$</td>
<td>24 bits</td>
</tr>
</tbody>
</table>

The ratio between amounts of data in the first partition and in the second partition, $\phi$, mainly depends on the characteristics of the sequence. As shown in Figure 6.2, $\phi$ is higher in the high motion "Foreman" sequence compared to that for "Suzie" for a given source rate. The source rate supported by data partition based UEP is limited by the channel bandwidth, $R_{ch}$, the channel coding rate of the high priority channel, $1/X_{ch1}$, the channel coding rate of the low priority channel, $1/X_{ch2}$, the extra overhead incurred, $OH$, and the sequence characteristics ($\phi$). The source rate, $R_s$, for a given bearer configuration can be calculated as

$$R_s = \frac{(1 + \phi) \cdot R_{ch}}{X_{ch2} + \phi \cdot X_{ch1}} \cdot \frac{1}{(1 + OH)}$$

Equation 6.2

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Figure 6.2: Ratio between first partition and second partition ($\varphi$) variation with source rate.

6.2.2 Application in UMTS networks

![Diagram showing data partition based UEP over UMTS]

Realisation of the data partition based UEP algorithm for a UMTS network is shown in Figure 6.3. At the encoder, each encoded video frame is separated into two streams based on MPEG-4
data partitioning. The separated streams are mapped on to two transport channels. The high priority data is sent over the highly protected channel, which is protected with a 1/3 rate convolutional code. A 1/2 rate convolutional code is used to protect the lower priority channel. At the physical layer, the information on the transport channels is multiplexed on to the same physical channel for transmission over the air interface.

A spreading factor of 32 is used in the physical channel configuration. This permits 64 kbps and 97 kbps information rates with 1/3 rate and 1/2 rate convolutional coding respectively. For the data partition based UEP scheme, video coded at 88 kbps and 82 kbps (calculated from Equation 6.2) provides the appropriate source-channel coding ratios for the “Suzie” and “Foreman” sequences respectively.

### 6.2.3 Video performances

Video sequences are encoded according to the MPEG-4 simple profile [3GPP TS 26.110] format. This includes error resilience tools such as video packetisation, data partitioning, and reversible variable length coding. The first video frame is intra coded while all others use Inter coding. The TM5 rate control algorithm is used to achieve a smoother output bit rate, while an adaptive intra refresh algorithm is used to stop temporal error propagation. Other source parameters are as shown in Table 6.2.

<table>
<thead>
<tr>
<th>Source parameter</th>
<th>QCIF</th>
<th>MPEG-4/ IPPPP</th>
<th>Simple profile</th>
<th>Foreman</th>
<th>Suzie</th>
</tr>
</thead>
<tbody>
<tr>
<td>Test sequence</td>
<td>“Foreman”</td>
<td>“Suzie”</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Source rate</td>
<td>82 kbps</td>
<td>88 kbps</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Channel parameter</th>
<th>Vehicular A</th>
<th>50 km/h</th>
<th>½ CC, 1/3 CC</th>
<th>32</th>
<th>OFF</th>
</tr>
</thead>
<tbody>
<tr>
<td>Channel BLER</td>
<td>$E_b/N_0$</td>
<td>1/2 CC</td>
<td>1/3 CC</td>
<td>$E_b/N_0$</td>
<td>1/2 CC</td>
</tr>
<tr>
<td>3 dB</td>
<td>0.92</td>
<td>0.78</td>
<td>8 dB</td>
<td>0.047</td>
<td>0.013</td>
</tr>
<tr>
<td>4 dB</td>
<td>0.78</td>
<td>0.53</td>
<td>10 dB</td>
<td>0.0020</td>
<td>0.0010</td>
</tr>
<tr>
<td>6 dB</td>
<td>0.31</td>
<td>0.13</td>
<td>12 dB</td>
<td>0.0010</td>
<td>0.000</td>
</tr>
</tbody>
</table>

Table 6.2: Parameter settings used in the experiment
The performance of the DP based UEP scheme is shown in terms of average frame peak signal to noise ratio (PSNR) vs channel condition in Figure 6.4. The performances of 1/2 rate and 1/3 rate Convolutional Codes (CC) without application of UEP are also shown in the figure. Each point on the graphs corresponds to the average PSNR taken from 20 simulations. The figure clearly illustrates the performance improvement of the data partition based UEP method over non-UEP methods. In good channel conditions, the performance of the UEP scheme is closer to that of the 1/2 rate channel coding scheme. A considerable quality improvement with the UEP scheme, compared to others, is visible for moderate channel conditions. However, the 1/3 rate
convolutional code outperforms the UEP scheme for poor channels. This is because, the channel corruption seen in the low priority channel is severe, and hence decoder error concealment is difficult resulting in high distortion for poor channel conditions. For the Suzie sequence, only 20% of the total amount of data is protected with the 1/3 rate code, while 30% of the data is protected with the 1/3 rate code in the high motion Foreman sequence. Therefore, the “Foreman” sequence is expected to perform better than the “Suzie” sequence for poor channel conditions. Thus, the cross over point between the performance trace for the 1/3 rate non-UEP scheme and the data partition based UEP scheme is expected to occur at a lower Eb/No value than that for “Suzie”. As can be seen in Figure 6.4, the experimentally obtained cross over point for “Suzie” is 6 dB, while it is 5.3 dB for “Foreman” in terms of channel Eb/No.

6.3 Perceived importance based unequal error protection scheme

The data partition based UEP scheme is simple to implement, as it requires little modification to the encoder data format and none to the network protocols. However, it is a sub optimal scheme considering the fact that MPEG-4 data partitioning does not always reflect the relative importance of data in different video packets. In other words, it is possible to have more important second partition data in some packets compared to the first partition data in others, especially in high motion scenes. As shown in Section 6.2.1, the source rate with the data partition based UEP scheme is fixed by the sequence characteristics for a given bearer configuration. High activity sequences require lower source rates while low motion sequences can be encoded at high source rates for a given channel throughput. Therefore, a Joint-Source Channel Coding (JSCC) scheme [ZHIH-02], which varies the source rate and effective channel protection according to the channel bit error rate characteristics can not be used with data partitioning based UEP techniques for given channel coding schemes.

To overcome the limitations of the data partition based UEP scheme, a novel model driven Unequal Error Protection scheme is proposed and analysed in this section. Similar to the data partition based UEP scheme, the proposed UEP scheme deploys differential protection by sending prioritised data over multiple radio bearers. Hence, it is compatible with the codec standard and is also transparent to the underlying network. Data prioritisation is conducted based on the perceived importance of bits, which is estimated using the developed distortion model (as described in Chapter 5) at the encoder. In contrast to the data partition based method, the proposed scheme combines JSCC and UEP schemes and obtains optimal video quality over a wide range of channel conditions.
Chapter 6. Link Level Quality Enhancement Techniques

6.3.1 Stream prioritisation algorithm

The proposed UEP algorithm is summarised in Figure 6.5. The optimum possible source rate, $R_s$, is a function of channel bit rate, $R_m$, channel coding rate, $1/x$, number of radio bearers, $N$, and channel bit error rate, $\eta_r$, of each radio bearer.

![Figure 6.5: Proposed UEP algorithm.](image)

A JSCC technique is applied to determine the optimum source rate for given channel conditions at the start of the transmission. Even though it requires extra processing delay, it is acceptable as video communications can afford a relatively long start-up delay. As explained in Chapter 5, video performance can be modelled as a combination of quantisation distortion, $E(D_{Q_{PV}})$, spatial concealment distortion $E(D_{S_{con,PV}})$, temporal concealment distortion $E(D_{T_{con,PV}})$, and distortion due to error propagation over predictive frames and video packets ($f_{wp}$ and $f_{ap}$).
The expected distortion of the $i^{th}$ video packet in the $j^{th}$ video frame, $E(D_{pv}^{ij})$, can be written as

$$E(D_{pv}^{ij}) = E(D_{Q,pv}^{ij}) + \rho_{u,pv}^{ij}E(D_{con,pv}^{ij})$$

$$+ \rho_{d,pv}^{ij}E(D_{con,pv}^{ij}) + f_{lp}^{ij} + f_{sp}^{ij}$$

Equation 6.3

$\rho_{u,pv}^{ij}$ denotes the probability of receiving an un-decodeable video packet. This includes the corruption of VOP headers, VP headers or motion data. The probability of receiving a decodeable video packet but with errors, where the DCT data is corrupted but other information is received correctly is denoted by $\rho_{d,pv}^{ij}$. The expected distortion calculation and the evaluation of probability terms are as described in Chapter 5.

The JSCC scheme can be described by grouping the distortion terms in Equation 6.3 and re-writing it as

$$\sum_{i=0}^{l} E(D_{pv}^{ij}) = \lambda_1 + \lambda_2 + \lambda_3$$

Equation 6.4

where

$$\lambda_1 = \sum_{i=0}^{l} E(D_{Q,pv}^{ij})$$

$$\lambda_2 = \sum_{i=0}^{l} (E(D_{con,pv}^{ij}) \cdot \rho_{u,pv}^{ij}(1 + P_{lp} + P_{sp} + P_{pp} \cdot P_{lp}))$$

$$+ E(D_{con,pv}^{ij}) \cdot \rho_{d,pv}^{ij}(1 + P_{pp}))$$

$$\lambda_3 = \sum_{i=0}^{l} (E(D_{con,pv}^{ij}) \cdot \rho_{u,pv}^{ij}(1 + P_{lp} + P_{sp} + P_{pp} \cdot P_{lp}))$$

Equation 6.5

The derivation of $P_{lp}$ and $P_{sp}$ are given in Section 5.2.2.3.

It is assumed that the channel characteristics are known at the start of transmission. This is a reasonable assumption, as the channel characteristics can be roughly estimated using the information received over the common channels at connection set up.

First, two video frames are encoded with a pre-defined source rate. The expected distortions of the first and second frames are computed using the developed distortion model under the assumption that the first frame is transmitted over the highest priority channel. Effective channel bit error rates are used in the distortion calculation of the second frame. Based on the estimated distortion of the second frame, the optimal source rate for the given channel condition is recursively computed following the algorithm shown in Figure 6.6.
According to the selected source rate, the effective channel bit error rate, $\mu$, is computed.

$$\mu = \frac{1}{R} \sum_{n=1}^{N} \eta_n \cdot R_n$$

Equation 6.6

$$R = \sum_{n=1}^{N} R_n$$

$$R_{ch} = \sum_{n=1}^{N} x_n \cdot R_n$$

where $n$ represents the $n^{th}$ radio bearer. The total channel bit rate is denoted by $R_{ch}$. The number of data bytes in each stream is determined in such a way as to satisfy the channel bit rate requirement. If the size of the $j^{th}$ video frame is $S'$ then, the amount of data allocated in the $n^{th}$ channel, $B_n$, is

$$B_n = \left| R_n \cdot S' / R \right|$$

Equation 6.7

Using the developed distortion estimation model, the distortion caused by the corruption of each data partitioned video packet within the current video frame is calculated. For each video packet, the distortion caused by the corruption of texture data, $\beta$, is grouped separately from that caused by the corruption of motion data, $\alpha$. This information provides a relative importance measurement figure for video data belonging to the current frame. Relative importance is divided into $N$ levels, where $N$ is the number of radio bearers. Next, each partition is marked with relative importance levels, according to the estimated distortion. In this calculation the second partition of some video packets may have a higher importance level than the first partition of others as illustrated in Section 6.3.3. The effective bit error rate is used for the distortion calculation in the first iteration.
Actual channel bit error rate values are used in following iterations. Finally, the information is prioritised into separate streams according to their calculated importance and separated streams are transmitted using different radio bearers.

### 6.3.2 Stream format

To facilitate stream data combination at the video frame level, time stamp and video frame configuration information should be present in each stream. Therefore, the VOP header is repeated in every stream at the start of the frame. The proposed UEP algorithm may transmit

1. a complete video packet in a stream
2. the first partition of the video packet alone in a stream
3. the second partition of the video packet alone in a stream.

If condition one or two occurs, then all the necessary information needed to decode the video packet is received within itself, and therefore no format modification is needed. When condition three occurs, VP header information is added to the second partition to ensure video packet synchronisation. The resulting video packet formats are shown in Figure 6.7.

![Video packet format for condition 1](header motion dct)

![Video packet format for condition 2](header motion)

![Video packet format for condition 3](header dct)

Header size - 24 bits

**Figure 6.7: Video packet format in multi-stream.**

The extra overhead arising from this technique depends on the source rate, sequence characteristic, video packet size and channel condition. The amount of overhead incurred is high in moderate channel conditions compared to others (see Table 6.3). For the video sequences tested in this paper, the generated extra overhead is approximately 1-2% of the overall rate.
6.3.3 Algorithm performances

For the sake of performance comparison with the DP based UEP method, only two radio bearers are used for transmission. High priority data is sent over a highly protected channel, which uses a 1/3 rate convolutional code. A 1/2 rate convolutional code is used to protect the lower priority channel. The source rate permitted by the proposed UEP algorithm varies from 64 kbps to 97 kbps according to the channel characteristics. Settings of other source and channel parameters used in the experiment are given in Table 6.2.

Experiments were conducted with a range of channel conditions, for video transmission using the proposed UEP schemes. Table 6.3 lists the occurrence of first and second partition video data in the high priority channel for transmission of the “Suzie” sequence with the proposed UEP scheme. As can be seen in the table, a considerable amount of second partition data is sent over the high priority channel. The last column of Table 6.3 shows the resulting overhead percentage.

<table>
<thead>
<tr>
<th>$E_b/N_0$</th>
<th>1st partition in high priority channel (%)</th>
<th>2nd partition in high priority channel (%)</th>
<th>Overhead %</th>
</tr>
</thead>
<tbody>
<tr>
<td>11 dB</td>
<td>12.98</td>
<td>7.37</td>
<td>0.27</td>
</tr>
<tr>
<td>10 dB</td>
<td>28.64</td>
<td>12.82</td>
<td>0.76</td>
</tr>
<tr>
<td>9 dB</td>
<td>48.01</td>
<td>30.41</td>
<td>1.04</td>
</tr>
<tr>
<td>8 dB</td>
<td>64.80</td>
<td>41.14</td>
<td>1.13</td>
</tr>
<tr>
<td>7 dB</td>
<td>77.95</td>
<td>58.74</td>
<td>0.91</td>
</tr>
<tr>
<td>6 dB</td>
<td>88.25</td>
<td>80.90</td>
<td>0.35</td>
</tr>
</tbody>
</table>

The video performances are shown in terms of average frame peak signal to noise ratio (PSNR) vs channel condition in Figure 6.8. The performance of data partition based UEP, 1/2 rate and 1/3 rate CC without application of UEP is also shown in the figure. The figure clearly illustrates the performance improvement of the proposed method over DP based UEP methods as well as non-UEP methods. With poor channel conditions, significant quality improvement is visible. Although the average frame PSNR values of the proposed scheme match with those of the DP based method at Eb/No 9 dB, the frame PSNR values show a smaller variance than the proposed UEP scheme. No significant performance differences are noticeable between low and high motion sequences.
The proposed algorithm is designed to provide better protection for the perceptually important regions in the video frame. Quality variation in a video frame is demonstrated in Figure 6.9 and Figure 6.10 for “Suzie” and “Foreman” respectively. Unequally error protected sequences are subjected to the same channel errors. As it is visible in the figures, distortion in the frame which is protected using the proposed UEP scheme is distributed in the background without disturbing the perceptual quality.
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6.4 Unequal power allocation scheme

Efficient transmission power utilisation is an important design criterion in interference limited cellular systems, such as UMTS networks. System capacity is limited by the total interference
experienced within the cell coverage. Thus, the optimisation of power consumption for an individual user can provide an increase in system capacity, as well as in the quality of service experienced by the user.

Recently, several energy minimisation techniques for wireless video applications have been proposed [EISE-02, KIM-03, ZHAO-02]. All of these techniques are optimised to achieve a target video quality while minimising the transmission power. In [EISE-02], joint error resilience and transmission power management at the video frame level is proposed. However, as the video frame quality is variable in nature (even in an error free environment), controlling the transmission power to achieve a target frame quality at the video frame level is inaccurate and would result in poor system performance. This problem can be solved by minimising the total consumed power within a certain period, while achieving the optimal average video quality. The method proposed in [KIM-03], employs this concept for intra-refreshed video sequences, thus video performance is optimised for a fixed intra-refresh period. However, this method can not be applied in conjunction with rate controlled Adaptive Intra Refresh (AIR) techniques [ISO-14496-2], which are commonly used to produce smoother output bit rates for transmission over fixed bandwidth channels. Another issue that should be considered in the design of transmit power optimisation schemes is the support of network compatibility and interoperability between different networks and platforms. Transmit power is normally allocated at the physical layer for a given Transmit Time Interval (TTI) [3GPP TS 25.214]. If the power allocation algorithms are closely coupled with the video compression formatting as in [EISE-02], it is impossible to implement such an algorithm at the physical layer without modifying the entire protocol stack of the existing network.

The algorithm proposed in this section takes these issues into account in the design and implementation. In contrast to the method in [KIM-03], the proposed scheme can equally be applied to rate controlled AIR video sequences as well as intra-refreshed sequences. The proposed method combines a UEP technique and a UPA technique to obtain energy efficient video transmission. A data partition based UEP scheme, as described in Section 6.2, is used to achieve unequal error protection. At the start of every video frame, the transmission energy for different bearers is selected in such a way as to achieve the maximum expected video frame quality for an increment in the transmission power step. Frame quality is estimated at the encoder using the developed distortion model in Chapter 5.
6.4.1 Energy optimised UEP scheme

Let the user requested video quality in terms of average frame PSNR be $Q_{\text{target}}$. Total channel interference and the noise experienced is denoted by the noise power spectral density, $N_o$. The minimum required transmission energy for an information bit to satisfy the user quality requirement under a given channel condition is $E_{b_{\text{min}}}$. The expected video frame quality, $E(Q_{f_{\text{min}}})$, is computed using Equation 5.1. It is assumed that the data on both higher priority and lower priority channels are transmitted with equal bit energies, $E_{b_{\text{min}}}$. 

Table 6.4: Possible combinations of transmission energy allocation.

<table>
<thead>
<tr>
<th>point</th>
<th>Energy level on high priority channel</th>
<th>Energy level on low priority channel</th>
<th></th>
<th>Energy level on high priority channel</th>
<th>Energy level on low priority channel</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>$E_{b_{\text{min}}}$</td>
<td>$E_{b_{\text{min}}}$</td>
<td>F</td>
<td>$E_{b_{\text{min}}}+1$</td>
<td>$E_{b_{\text{min}}}+2$</td>
</tr>
<tr>
<td>B</td>
<td>$E_{b_{\text{min}}}$</td>
<td>$E_{b_{\text{min}}}+1$</td>
<td>G</td>
<td>$E_{b_{\text{min}}}+2$</td>
<td>$E_{b_{\text{min}}}$</td>
</tr>
<tr>
<td>C</td>
<td>$E_{b_{\text{min}}}$</td>
<td>$E_{b_{\text{min}}}+2$</td>
<td>H</td>
<td>$E_{b_{\text{min}}}+2$</td>
<td>$E_{b_{\text{min}}}+1$</td>
</tr>
<tr>
<td>D</td>
<td>$E_{b_{\text{min}}}+1$</td>
<td>$E_{b_{\text{min}}}$</td>
<td>K</td>
<td>$E_{b_{\text{min}}}+2$</td>
<td>$E_{b_{\text{min}}}+2$</td>
</tr>
<tr>
<td>E</td>
<td>$E_{b_{\text{min}}}+1$</td>
<td>$E_{b_{\text{min}}}+1$</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Figure 6.11: Calculation of optimal transmission energy levels.

Point A in Figure 6.11 represents the estimated quality, $E(Q_{f_{\text{min}}})$. Point E shows the expected quality, if the transmission energy on both channels is incremented by 1 dB. The goal is to find the combination of transmission energies in the two channels, which maximise the current video quality.
frame quality for an increment in the transmission energy. Therefore, all the possible combinations of transmission energy allocation in different priority channels are considered (see Table 6.4). Even though, it is valid to consider other transmission energy levels, the settings listed in Table 6.4 are more likely to provide optimum energy allocation. Moreover, this simplifies the searching procedure. The expected video frame qualities are computed for the energy settings listed in Table 6.4 and the corresponding points are shown in Figure 6.11. The transmission energy levels corresponding to the point that shows the highest gradient from point A provide the optimal energy levels for transmission of the current video frame.

This algorithm operates at the video frame level to find the optimal operating point. It guarantees the end user quality requirement and the optimal energy setting throughout the transmission.

6.4.2 Simulation setup

Realisation of the proposed power optimised UEP algorithm over a UMTS network is shown in Figure 6.12. As explained in Section 6.2, each video frame is separated into two streams at the encoder. The separated streams are mapped on to two transport channels with different error protection capabilities. At the physical layer, the information on the transport channels is allocated the selected transmission bit energy and is multiplexed on to the same physical channel for transmission over the air interface.
6.4.3 Results and analysis

Experiments were conducted for a range of channel conditions for the proposed power optimised UEP scheme. Settings of source and channel parameters are as shown in Table 6.2. The results are shown in terms of average $E_b / N_0$ vs average frame PSNR in Figure 6.13 (a) and (b) for the transmission of “Suzie” and “Foreman” respectively. The performances of 1/2 rate and 1/3 rate convolutional code without application of UEP and DP based UEP scheme are also shown. The figures clearly demonstrate that efficient energy utilisation is achieved with the proposed method.
compared to the data partition based UEP scheme as well as the non-UEP schemes. Video performances are limited by the quantisation distortion with good channel conditions. In such situations, increasing the transmit power will not further increase the performance. This effect is well captured by the proposed algorithm and the highest allocated transmit $E_b/N_0$ is limited to $11.2$ dB. For the allocation of higher transmission energies, the proposed algorithm shows close performance to that of the DP based UEP scheme. However, the proposed algorithm considerably out-performs the UEP scheme at lower transmit energies. For example, for transmission of the "Suzie" sequence, the achieved average frame PSNR with the UEP scheme is $18$ dB for transmit bit energy to noise ratio of $6$ dB. The proposed power optimised UEP scheme results in average frame PSNR of $24$ dB for the same transmit bit energy allocation. No significant performance differences are seen between the two different sequences.

![Figure 6.14](image)

**Figure 6.14:** a). Performance and b). Transmit power allocation of proposed algorithm.

Figure 6.14(a) displays the frame-by-frame PSNR of the Suzie sequence using the power optimised UEP scheme, compared to the non-power adaptive UEP scheme. The PSNR values are the average of 20 simulations over the Vehicular A channel. Transmit bit energy to noise ratio is
set to 7 dB for non-power adaptive UEP. Frame-by-frame transmit energy allocation for energy optimised UEP is shown in Figure 6.14(b). The resulting average bit energy to noise ratio on the power optimised UEP equals 7.15 dB. Figure 6.14(a) clearly demonstrates the improved performance achieved with power optimised UEP throughout the transmission.

6.5 Conclusion

An optimum Unequal Error Protection scheme is proposed for video communications over wireless networks. The video data is prioritised according to the relative perceived importance, which is estimated using the developed distortion model at the encoder. Prioritised streams are transmitted over the air interface by using multiple radio bearers with different error protection capabilities. The experiments that are carried out over the simulated UMTS systems show significant performance improvements with the proposed UEP scheme compared to the traditional UEP, as well as non-UEP schemes. Furthermore, the data prioritisation of the proposed system is transparent to the underlying network protocols, and hence can be used in heterogeneous networks. The implementation of the proposed system also suggests that easy collaboration is possible with a content adaptation system for enhancing subjective video quality.

Secondly, an energy efficient network compatible performance enhancement method is proposed for video communications over direct-sequence CDMA cellular networks. The proposed scheme combines data partition based UEP and Unequal Power Allocation in order to minimise the required transmit power for acceptable video quality. Prioritised video information is transmitted over two different transport channels with different error protection capabilities. Transmit energy for each bearer is selected to maximise the expected frame quality for an increment in transmit power. The experiment show significant performance improvement with the proposed algorithm compared to the traditional UEP schemes and non-UEP schemes.
Chapter 7

7 Application Level Quality Enhancement Techniques

7.1 Introduction

Previous chapters investigated the performance of video in error-prone environments and quality improvement techniques, which can be applied at the link level where network resources are allocated for the transmission. However, the time-varying channel conditions resulting from the mobility of the terminal and the surroundings were not considered in the work described in previous chapters. The effect of time-varying channel conditions (slow fading) on video performance and the techniques that can be used to mitigate the effects of time-varying channel conditions are investigated in this chapter. A performance enhancement method for real-time video communications by employing link adaptation is discussed. A novel link adaptation algorithm, which is based on feedback channel information including BLock Error Rate (BLER), Received Signal Strength (RSS) and the first order statistic of the RSS is proposed and analysed. Two approaches for link adaptation are investigated. First, the effects of a link adaptation scheme at the video frame level, which aims to optimise video quality by varying the channel-coding scheme and video source rate for a fixed channel allocation, is studied. Second, a link adaptation algorithm with a goal of maximising the overall access network throughput is developed at the radio block level. Another link adaptation algorithm is proposed for real-time streaming video applications. Link adaptation is not generally considered suitable for multi-user streaming video,
because it usually requires interaction between the link-layer protocol and the encoder to perform source rate adaptation. However, the technique presented here facilitates stream switching, hence requires no such interaction with the encoder, thereby significantly simplifying the link adaptation system. The benefits of the link adaptation algorithms are demonstrated for MPEG-4 coded video transmissions over the simulated Enhanced General Packet Radio Service (EGPRS) access network. In addition, the effects of feedback delay, a noisy feedback channel and bursty channel errors on the algorithms’ performances are investigated. Finally, a link adaptation algorithm, which makes use of variable spreading factor assignment in UMTS network, is examined for real-time video communications. Algorithm performance is further enhanced by combined application of Unequal Error Protection (UEP) and link adaptation techniques. Results illustrate a significant performance improvement in perceptual video quality.

7.2 Time varying channel model design

The time varying channel model used in the simulator consists of three main components, which are fast fading, shadow fading and propagation loss. The fast fading model follows the description of multi-path propagation models described in [3GPP-05.05] and [UMTS 30.03]. In these models, it is assumed that the mobile radio environment is dispersive with several reflectors, scatters and different distances from the line-of-sight path between the mobile terminal and the base station. Shadow fading and propagation path loss are modelled as described below.

7.2.1 Shadow fading model

Shadowing is modelled as a log-normally distributed random variable with correlated consecutive samples. The form of autocorrelation for the shadowing process depends on the user velocity, \( v \), and the correlation distance, \( d_c \), of the particular channel. The auto-covariance, \( C_\xi(r) \), of the shadowing process can be modelled as [SAUN-99]

\[
C_\xi(r) = \sigma^2 e^{-\nu r/d_c}
\]

where, \( \sigma \) is variance of log normal distribution. The shadowing components for each instant of the simulation are generated in two different ways. In the first method, a white Gaussian process is generated and is filtered through a first-degree filter with a pole at \( e^{\nu d_c} \). In the second method, following the exponential form of the auto-covariance function, the shadowing process is
represented by a first order autoregressive process where the $n^{th}$ shadowing sample is given by the expression [SAUN-99]:

$$ S(n) = \alpha + S(n-1) + \beta W(n) $$

Equation 7.2

where $W(n)$ is a sample of a white Gaussian process, $\alpha = e^{-\nu v/d_c}$ and $\beta = \sigma \sqrt{(1-\sigma^2)}$. Figure 7.1 shows the auto-covariance plots for a shadowing process at 3 km/h vehicular velocity, $\sigma = 7$dB and $d_c = 5$m. The star marked line indicates the auto-covariance plot corresponding to the samples generated from the filter approach. The dashed line corresponds to the autoregressive model, while the solid line is generated by the theoretical formula. For the simulation model under discussion, the autoregressive model was used for generating the lognormal shadowing components considering its' fast computation.

![Figure 7.1: Autocorrelation of shadowing process](image)

7.2.2 Path loss model

The COST 231-Walfish-Ikegami model [GSM 03.30] was used to approximate the path loss experienced in urban environment when the cell radius is less than 5 km. The following parameters have been used:

- Width of the road, $= 20$m
- Height of building roof tops, $= 15$m
- Height of base station antenna $= 17$m
- Height of mobile station antenna $= 1.5$ m
Road orientation to direct radio path \( = 90^\circ \)

Building separation \( = 40 \text{ m} \)

For GSM 900 the corresponding propagation loss is:

\[ L = 132.8 + 38 \log(r_f) \]

For DCS 1800 the corresponding propagation loss is:

\[ L = 142.9 + 38 \log(r_f) \text{ for medium sized cities} \]
\[ L = 145.3 + 38 \log(r_f) \text{ for metropolitan centres} \]

The path loss model used for UMTS Vehicular test environment is [UMTS 30.03]:

\[ L = 128.1 + 37.6 \log(d) \]

where \( L \) is in dB while \( d \) is in km.

The following parameter values are assumed in the UMTS test environment:

- The difference between the mean building height and the mobile antenna height \( = 10.5 \text{ m} \)
- The height difference between the base station antenna and the mean building rooftops height \( = 15 \text{ m} \)
- The horizontal distance between the mobile and the diffracting edges \( = 15 \text{ m} \)
- The average separation between rows of buildings \( = 80 \text{ m} \)
- Carrier frequency \( = 2000 \text{ MHz} \)

### 7.2.3 Mobility model

A pseudo random mobility model with semi-directed trajectories is used to model the user mobility. The Mobile Terminal’s (MT) position is updated according to the de-correlation length and the direction is changed at each position update according to the given probability [UMTS 30.03].

The mobility model is defined by the following parameters:

- Speed (assume to be constant at): \( 3 \text{ km/h, 50km/h} \)
- Probability to change direction at position update: \( 0.2 \)
- Maximal angle for direction update: \( 45^\circ \)
Chapter 7. Application Level Quality Enhancement Techniques

De-correlation length: 5 m (corresponding to 3 km/h mobile speed), 20m (corresponding to 50 km/h mobile speed)

Mobile terminals are uniformly distributed and their direction is randomly chosen at initialisation.

7.2.4 CIR calculation

The capacity of a cellular radio network is interference limited. The network operator is assigned a band of frequencies by the regulatory authorities, and must strive to reuse the frequency band in an efficient manner to maximise the number of subscribers that can use the service. Cells are tessellated to form clusters, and each cluster may use the entire allocated spectrum. Cells in neighbouring clusters use the same frequencies, and hence mobile terminals in these cells may interfere with each other, causing co-channel interference. To reduce the co-channel interference, cells are further divided into sectors within each cluster. In GSM [3GPP TS 05.05] three main cluster/sector configurations, namely 4/12, 3/9 and 1/3 are defined.

Only the C/I ratio for downlink is investigated in this study. Each base station is assumed to be transmitting equal power ($P_T$). As the BS transmitted power is independent of mobile position frequency hopping does not affect the calculation of C/I ratio. C/I ratio is calculated for each of the different cluster sizes and sectorization configurations defined in GSM. Because the C/I ratio in the downlink is location dependent, the C/I for different locations within the cell are also calculated.

Consider a mobile that receives in the $k^{th}$ slot on carrier $f_i$ in the $0^{th}$ cell. The $j^{th}$ co-channel interfering BS creates an interference with this mobile in the $0^{th}$ cell when it communicates to the mobile using $k^{th}$ slot of carrier $f_i$ in its own cell. The geometrical arrangement is shown in Figure 7.2.

![Figure 7.2: Down-link interference from the $j^{th}$ BS to MS in $0^{th}$ cell](image-url)
State that the path loss component is represented by \( L(f, d, x) \) where \( f \) is transmitting frequency, \( d \) is distance between transmitter and receiver and other factors such as geometric orientation, heights of antennas are represented by \( x \). Shadow fading, \( S \), is a function of velocity, \( v \), decorrelation length, \( d_c \), and log-normal variance, \( \sigma \). Assume that the fast fading is effectively combated by the use of channel equalisation, frequency hopping and signal processing. Thus, the received power by the mobile is

\[
P_R = P_T \cdot L(f, d, x) \cdot S(v, d_c, \sigma)
\]

Equation 7.3

Where \( P_R \) and \( P_T \) are received power and transmitted power respectively.

Referring to Figure 7.2,

The total carrier power received by the mobile in 0\(^{th}\) cell is

\[
P_C = P_T \cdot L_0(f, d_0, x_0) \cdot S_0(v, d_c, \sigma)
\]

Equation 7.4

The total interfering power received by the mobile is

\[
P_I = \sum_j P_T \cdot L_j(f, d_j, x_j) \cdot S_j(v, d_c, \sigma)
\]

Equation 7.5

where subscript \( j \) represents the \( j^{th} \) interfering cell.

The application of voice activity detection results in discontinuous transmission, thereby reducing the interfering power received by the mobile. In order to consider the effect of voice activity detection on C/I ratio, a voice activity variable \( v_j \) is introduced. Variable \( v_j \) is defined as [LEE-95],

\[
v_j = \begin{cases} 1, & \text{with a probability of } \mu \\ 0, & \text{with a probability of } 1 - \mu \end{cases}
\]

Equation 7.6

where the mean value of the voice activity random variable is given by \( E[v_j] = \mu \). Following the above argument, the equation (7.5) can be modified as

\[
P_I = \sum_j \mu \cdot P_T \cdot L_j(f, d_j, x_j) \cdot S_j(v, d_c, \sigma)
\]

Equation 7.7

Therefore,

\[
\frac{C/I}{P_T} = \frac{P_C}{P_I} = \frac{L_0(f, d_0, x_0) \cdot S_0(v, d_c, \sigma)}{\mu \cdot \sum_j L_j(f, d_j, x_j) \cdot S_j(v, d_c, \sigma)}
\]

Equation 7.8
7.2.4.1 Calculation of C/I variation due to path loss

For a hexagonal cell pattern, there are always six close interferers, irrespective of the number of cells per cluster, as can be seen from Figure 7.3. However, the distance between co-channel cells depends on the number of cells \((N)\) in a cluster and is given by [WILL-86]

\[
D = R \cdot \sqrt{3N}
\]

Equation 7.9

where \(D\) is the distance between co-channel cell centres and \(R\) is the cell radius.

Figure 7.3 also illustrates the three-sector configuration. Assuming perfect sectorisation, (i.e. No electromagnetic wave penetration to other sectors) the number of close interferes is reduced to two.

Using the COST 231-Walfish-Ikegami path loss model and hexagonal cell structure, the variation of C/I due to path loss is calculated for each of cluster/sector configurations specified in GSM [3GPP TS 05.05]. For 3/9 cluster/sector configuration, the graph of C/I for moving alone the path \(x - y\) is shown in Figure 7.4.
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Figure 7.4: Down-link: 3/9 cluster/sector configuration CIR for MS at normalised distance, r, from the BS.

The worst C/I condition occurs when the mobiles are located near the cell boundary. The difference between the two extreme cases, where the mobile is located near the BS or near the cell boundary, is about 35 dB.

Figure 7.5 illustrates the effect of different cluster/sector configurations on C/I ratio. In the case of 1/3 configuration, the effect of a second set of interferers is also considered. 4/12 cluster/sector configuration gives the best performance while 1/3 cluster/sector configuration gives the worst performance. The difference between these two cases is about 10 dB.

Figure 7.5: Down-link: CIR for MS at normalised distance, r, from the BS. Line with triangular mark- 3/9 configuration; circle mark- 1/3 configuration; square mark- 4/12 configuration.
7.2.4.2 Calculation of C/I variation due to shadow fading

Consider a single interferer case. Equation 7.8 can be re-written as

\[ C/I = L_0 + S_0 - L_f - S_f \]  

Equation 7.10

All the terms are in dB.

As shadowing is modelled as a log normal process with zero means and \( \sigma \) variance, terms \( S_0 \) and \( S_f \) in equation (7.12) are random variables with normal distribution. Following the properties of a normal distribution it can be shown that the distribution of \( S_0 - S_f \) (\( S_{0-f} \)) is also a normal distribution with zero mean and variance \( \sigma_{0-f} \) given by [SAUN-99],

\[ \sigma_{0-f}^2 = \sigma_0^2 + \sigma_f^2 - 2 \cdot \rho \cdot \sigma_0 \cdot \sigma_f \]  

Equation 7.11

where \( \rho \) defines the shadowing correlation coefficient. \( \rho = 0 \) indicates no shadowing correlation resulting in the worst-case scenario. The variance, \( \sigma_{0-f} \), decreases as the correlation increases, reaching a minimum value when \( \rho = 1 \). When \( \rho = 1/2 \) and \( \sigma_0 = \sigma_f \), \( \sigma_{0-f} \) becomes \( \sigma \). Following the above argument, it can be stated that \( \sigma \) always indicates one instance of the resultant variance \( \sigma_{0-f} \) of the shadowing process even in the scenario where more than one interferes are taken in to account.

![Figure 7.6: Simulated channel CIR for 30 sec duration.](image)

Applying the mobility model, each position of the mobile is estimated for the duration of the conversation. The C/I ratio due to the path loss is calculated at each position of the mobile and the shadowing process is simulated separately as explained above for \( \sigma = 7 \) dB, \( v = 3 \) km/h and \( d_c = 5 \) m. The total C/I ratio was estimated by superimposing the simulated shadowing process over the calculated C/I ratio due to the path loss. Figure 7.6 shows the estimated C/I ratio for the 30 sec
duration when the mobile travels at 3km/h taking into account both propagation loss as well as shadowing. The parameters used, are as listed in Table 7.1.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Log-normal variance</td>
<td>7 dB</td>
</tr>
<tr>
<td>De-correlation distance</td>
<td>5 m</td>
</tr>
<tr>
<td>Radius of hexagonal cell</td>
<td>200 m</td>
</tr>
<tr>
<td>Propagation frequency</td>
<td>900 MHz</td>
</tr>
<tr>
<td>Vehicular speed</td>
<td>3 km/h</td>
</tr>
<tr>
<td>Channel environment</td>
<td>TU3</td>
</tr>
<tr>
<td>CIR margin</td>
<td>9 dB</td>
</tr>
<tr>
<td>Fading Characteristics</td>
<td>Raleigh fading</td>
</tr>
<tr>
<td>Cell configuration</td>
<td>4/12</td>
</tr>
<tr>
<td>Frequency hopping</td>
<td>Ideal frequency hopping</td>
</tr>
<tr>
<td>Path loss model</td>
<td>COST 231-Walfish-Ikegami model</td>
</tr>
</tbody>
</table>

### 7.3 Link-adaptation for real-time video communications

One fundamental characteristic of cellular systems is the time-varying channel conditions experienced by mobile users due to differences in distances to the base station, slow/fast-fading characteristics of the channel and other user/cell interference. However, real-time multimedia services such as audio and video require maintenance of a certain CIR (Carrier to Interference Ratio) to give good perceptual quality. To keep the performance at a desirable level, traditionally, a communication system is designed for the average or worst-case situation. This, however, results in a severely under-utilized system when the channel is in a good state.

Power control and diversity techniques can be used to mitigate the effects of the time varying nature of the channel. However, power control may add extra interference to other users, while diversity techniques may require complex processing. Another way to increase the robustness of the radio link to varying channel quality is to employ link adaptation techniques [FURU-99]. The main idea of link adaptation is to adapt the modulation and coding levels according to the feedback channel information. Link adaptation techniques have attracted a lot of attention recently. Many data rate adaptation techniques based on the estimated signal to interference noise
ratio have been proposed to adapt coded modulation schemes, thus improving data throughput of the mobile channels [CHER-00, MEHT-00]. The theoretical basis for optimal switching in practical mobile systems involving several different adaptation parameters is investigated in [TANG-01]. However, these previous investigations were based on data and voice transmission with simplifying assumptions such as coherent detection, perfect mode synchronization on the transmission modes, zero feedback delay and noiseless feedback channels. In contrast to that, the performance of the proposed adaptive system for real-time video communication is evaluated based on practical considerations of channel estimation noise, feedback noise and feedback delay.

7.3.1 **Link adaptation for real-time video communication in EGPRS networks**

![Flow diagram of the proposed link adaptation scheme](image)

![Radio block structure in EGPRS channels](image)

Figure 7.7: (a) Flow diagram of the proposed link adaptation scheme (b) Radio block structure in EGPRS channels, one RLC block per 20 ms [extracted from 3GPP TS 03.64].
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The flow diagram of the proposed scheme is shown in Figure 7.7(a). The EGPRS Physical layer, including interleaver, modulator, equalizer, de-modulator and de-interleaver, was implemented as explained in [FABR-01] to simulate the reception performance of the EGPRS receiver. The effect of error upon the EGPRS protocols at the radio interface was simulated by integrating the physical layer model with a radio access data flow model.

In addition to the EGPRS protocol layer units, the system consists of a channel state predictor and a link adaptation decision unit at the transmitter. The channel state predictor estimates the channel condition based on the feedback information supplied. According to this estimate, the link adaptation decision unit commands the RLC/MAC layer protocol and source encoder to vary the modulation-coding scheme, allocated number of time-slots and source bit rate for the current transmission.

At the receiver, a channel quality measurement unit is located at the RLC/MAC layer. This measures the channel quality in terms of Received Signal Strength (RSS) and radio block error occurrence and feeds them back to the transmitter via a noisy feedback link with certain delay. Mode synchronisation is attained with a closed-loop method as described in [3GPP TS 03.64]. A control word describing the transmission modes for the radio block is embedded into the Radio-Block header as illustrated in Figure 7.7(b). The header format is indicated by the Stealing Bits (SB) of the block. To ensure strong header protection, the header part of the radio block is independently convolutionally coded from the data part of the radio block according to the code rate specified in Table 7.2. Finally, the header is interleaved over four bursts and transmitted. At the receiver, the header is de-interleaved and decoded first, followed by the rest of the radio block, which is, decoded with the indicated transmission modes.

<table>
<thead>
<tr>
<th>Scheme</th>
<th>Header rate</th>
<th>Code rate</th>
<th>Data/ Radio Block</th>
</tr>
</thead>
<tbody>
<tr>
<td>MCS-1</td>
<td>0.53</td>
<td>0.53</td>
<td>176</td>
</tr>
<tr>
<td>MCS-2</td>
<td>0.53</td>
<td>0.66</td>
<td>224</td>
</tr>
<tr>
<td>MCS-3</td>
<td>0.53</td>
<td>0.80</td>
<td>296</td>
</tr>
<tr>
<td>MCS-4</td>
<td>0.53</td>
<td>1.0</td>
<td>352</td>
</tr>
<tr>
<td>MCS-5</td>
<td>0.33</td>
<td>0.37</td>
<td>448</td>
</tr>
<tr>
<td>MCS-6</td>
<td>0.33</td>
<td>0.49</td>
<td>592</td>
</tr>
<tr>
<td>MCS-7</td>
<td>0.36</td>
<td>0.76</td>
<td>2×448</td>
</tr>
<tr>
<td>MCS-8</td>
<td>0.36</td>
<td>0.92</td>
<td>2×544</td>
</tr>
<tr>
<td>MCS-9</td>
<td>0.36</td>
<td>1.0</td>
<td>2×592</td>
</tr>
</tbody>
</table>
7.3.1.1 Link adaptation algorithms

The link adaptation algorithms adapt each radio link to one of nine modulation-coding schemes. Adaptation intervals and switching thresholds are determined by the particular algorithm used. Let $CIR_{est(k)}$ be the estimated channel condition at the $k^{th}$ radio block. Modulation-coding scheme mode $m$ is chosen if $CIR_{est(k)} \in [\xi_{th(m)}^{(m)}, \xi_{th(m+1)}]$. Where $\xi_{th(m)}$ and $\xi_{th(m+1)}$ indicate channel CIR threshold values corresponding to mode $m$. Note that $\xi_0 = 0$ and $\xi_\infty = \infty$. Two approaches for link adaptation, one based on source quality (referred to as the Source (video) Quality-Based Adaptation Scheme, SQBAS) and the other based on system throughput (referred to as the Throughput-Based Adaptation Scheme, TBAS), are investigated. Table 7.3 depicts the differences between SQBAS and TBAS.

<table>
<thead>
<tr>
<th>Operation at</th>
<th>SQBAS</th>
<th>TBAS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Optimisation based on</td>
<td>Received Video quality</td>
<td>Total system throughput</td>
</tr>
<tr>
<td>Video source rate</td>
<td>Variable</td>
<td>Fixed</td>
</tr>
<tr>
<td>Modulation-coding scheme</td>
<td>Variable</td>
<td>Variable</td>
</tr>
<tr>
<td>No. of time-slots allocation</td>
<td>Fixed</td>
<td>Variable</td>
</tr>
</tbody>
</table>

7.3.1.1.1 Source (video) Quality Based Adaptation Scheme (SQBAS)

The proposed source (video) quality-based adaptation scheme is designed to maximise the video quality by varying the source rate and channel-coding scheme accordingly, while maintaining fixed channel allocation throughout the transmission.

Figure 7.8 shows the effect of channel errors upon the quality of MPEG-4 encoded video in a TU3 (Typical Urban multi-path, mobile terminal velocity 3 km/h) propagation environment. Video sequences have been encoded for operation with three Time-Slots (TS) usage. The video source rate for each channel-coding scheme has been set according to Table 7.4. Average performances over a number of different video sequences are shown.
Figure 7.8: Video Quality at TU3 900MHz with ideal frequency hopping at 3 time slots operation [FABR-01].

Table 7.4: EGPRS Multi-slotting capacity for video (kbit/s) [FABR-01]

<table>
<thead>
<tr>
<th>Scheme</th>
<th>1 TS</th>
<th>2 TS</th>
<th>3 TS</th>
<th>4 TS</th>
<th>5 TS</th>
<th>6 TS</th>
<th>7 TS</th>
<th>8 TS</th>
</tr>
</thead>
<tbody>
<tr>
<td>MCS-1</td>
<td>7.5</td>
<td>15</td>
<td>22.5</td>
<td>30</td>
<td>37.5</td>
<td>45</td>
<td>52.5</td>
<td>60</td>
</tr>
<tr>
<td>MCS-2</td>
<td>9.6</td>
<td>19.2</td>
<td>28.8</td>
<td>38.4</td>
<td>48</td>
<td>57.6</td>
<td>67.2</td>
<td>76.8</td>
</tr>
<tr>
<td>MCS-3</td>
<td>12.6</td>
<td>25.2</td>
<td>37.8</td>
<td>50.4</td>
<td>63</td>
<td>75.6</td>
<td>88.2</td>
<td>100.8</td>
</tr>
<tr>
<td>MCS-4</td>
<td>15</td>
<td>30</td>
<td>45</td>
<td>60</td>
<td>75</td>
<td>90</td>
<td>105</td>
<td>120</td>
</tr>
<tr>
<td>MCS-5</td>
<td>19</td>
<td>38</td>
<td>57</td>
<td>76</td>
<td>95</td>
<td>114</td>
<td>133</td>
<td>152</td>
</tr>
<tr>
<td>MCS-6</td>
<td>25.2</td>
<td>50.4</td>
<td>75.6</td>
<td>100.8</td>
<td>126</td>
<td>151.2</td>
<td>176.4</td>
<td>201.6</td>
</tr>
<tr>
<td>MCS-7</td>
<td>38</td>
<td>76</td>
<td>114</td>
<td>152</td>
<td>190</td>
<td>228</td>
<td>266</td>
<td>304</td>
</tr>
<tr>
<td>MCS-8</td>
<td>446.2</td>
<td>92.4</td>
<td>138.6</td>
<td>184.8</td>
<td>231</td>
<td>277.2</td>
<td>323.4</td>
<td>369.6</td>
</tr>
<tr>
<td>MCS-9</td>
<td>50.3</td>
<td>100.6</td>
<td>150.9</td>
<td>201.2</td>
<td>251.5</td>
<td>301.8</td>
<td>352.1</td>
<td>402.4</td>
</tr>
</tbody>
</table>

As can be seen in Figure 7.8, MCS-1 gives better performance than MCS-2 at all CIR values up to 22.5 dB. MCS-5 provides superior video quality than that of other schemes at CIR values better than 22.5 dB. Optimal video quality in these propagation conditions is therefore achieved by selecting MCS-1 when the channel CIR is lower than 22.5 dB and MCS-5 otherwise. The proposed source quality-based link adaptation algorithm is summarised in Figure 7.9.
7.3.1.1.2 Throughput Based Adaptation Scheme (TBAS)

For the throughput-based adaptation scheme, the switching threshold is selected so as to guarantee a target received video quality over a range of CIR. The scheme is designed to maximise the system throughput by selecting a channel-coding scheme with low protection, and high throughput, at better channel conditions for fixed source rate operation.
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Figure 7.10 shows the results that are obtained for video transmission over the TU3 propagation environment with a fixed source rate of 38 kbps. The number of time slots is allocated according to Table 7.4. For example, if target video quality in PSNR is 30 dB, an appropriate switching policy would be to select MCS-5 when the channel CIR is higher than 18 dB and MCS-1 otherwise, thereby achieving target video quality. This results in a saving of 3 timeslots when switching from MCS-1 to MCS-5.

Even though four channel coding schemes (except MCS-3) are equally valid for use in the switching algorithm, for comparison purposes with SQBAS only MCS-1 and MCS-5 are used in the experiment.

![Figure 7.10: Video Quality at TU3 900MHz with ideal frequency hopping at 38 kbps fixed source rate operation.]

7.3.1.2 Channel state estimation

The channel estimation algorithm is constructed by partitioning the range of the BLock Error Rate (BLER) measurements into a finite number of intervals (I) and mapping them on to the actual channel carrier to interference ratio. As Figure 7.11 illustrates, the measured BLER is a non-linear function of channel CIR and also depends on the channel-coding scheme used. This is because the BLER flag is set if any bit of the radio block is in error after the channel decoding. Measured BLER is averaged over n radio blocks in order to reduce the effect of burst errors on the channel.
estimation. Thus the calculated BLER which to be used in the channel estimation algorithm is

\[ B_{cal,j} = \frac{1}{n} \sum_{k=1}^{n} B_{meas,j} \]

where \( B_{cal,j} \) and \( B_{meas,j} \) are calculated and measured BLERs respectively. Subscript \( j \) represents the channel-coding scheme used.

In addition, the mean (\( R_{mean} \)), the variance (\( R_{var} \)) and the gradient (\( R_{grad} \)) of Received Signal Strength (RSS) are used in the channel prediction calculation. The measurement window size is set to be equal to the estimated processing delay, \( \Delta_{est} \), which is assumed to be constant for a given application. Using linear prediction, predicted RSS is

\[ R_{pre} = R_{grad} \cdot \Delta_{est} + R_{mean} \]

Predicted RSS is also partitioned in to \( I \) intervals. Let \( CIR_{est}^{(k)} \), \( B_{cal,j}^{(k)} \) and \( R_{pre}^{(k)} \) be the estimated channel condition, calculated BLER and predicted RSS at \( k^{th} \) radio block and \( CIR_i \) be the corresponding CIR value for \( i^{th} \) (\( i \in \{1,2,\ldots,I\} \)) interval.

\[
CIR_{est}^{(k)} = \begin{cases} 
CIR, & B_{cal,j}^{(k)} \in [\nu_{th}(i),\eta_{th}(i+1)], R_{pre}^{(k)} \in [\nu_{th}(i),\eta_{th}(i+1)] \\
CIR_{est}^{(k-1)}, & R_{var}^{(k)} < \gamma \theta h \\
CIR, & B_{cal,j}^{(k)} \in [\nu_{th}(i),\eta_{th}(i+1)], R_{pre}^{(k)} \in [\nu_{th}(m),\eta_{th}(m+1)]
\end{cases} \quad \text{where } l < m
\]

where \( \mu_{th}(i) \), \( \nu_{th}(i) \) and \( \gamma \theta h \) indicate corresponding RSS, BLER and variance of RSS threshold values.

Figure 7.11: Performance of EGPRS networks at TU3 900MHz with ideal frequency hopping.
7.3.1.3 Feedback techniques

The feedback channel is used to carry measured channel information, RSS and BLER. The measured RSS is quantised with 8-bits uniform quantisation, and the measured BLER is represented by either zero or one. The 9 bits of each quantised sample are further protected with a selected modulation-coding scheme and are fed back to the transmitter via a noisy channel. The total information rate on the feedback channel is around 450 bits/s. This is very low compared with the forward data rate of 22 – 60 kbits/s. Therefore in-band signalling is possible for two-way videotelephony.

7.3.2 Results and discussion

The QCIF test sequence Suzie was selected as the source signal. The video codec parameters used in the experiment are listed in Table 7.5. Each video frame is considered as one transport and network layer data payload unit in the EGPRS protocol implementation. This is because the size of each video frame is below the specified maximum LLC-PDU size, which is 1520 octets [3GPP TS 04.64].

Table 7.5: Video codec parameters used in the experiment.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video codec</td>
<td>MPEG-4</td>
</tr>
<tr>
<td>Frame rate</td>
<td>10 fps</td>
</tr>
<tr>
<td>Video packet size</td>
<td>600 bits</td>
</tr>
<tr>
<td>Reversible variable length code (RVLC)</td>
<td>enable</td>
</tr>
<tr>
<td>Data partitioning</td>
<td>enable</td>
</tr>
<tr>
<td>Number of intra-Macro-Block used in AIR</td>
<td>10 (fixed)</td>
</tr>
<tr>
<td>Video quality measure</td>
<td>Peak Signal to Noise Ratio (PSNR)</td>
</tr>
<tr>
<td>Rate adjustment</td>
<td>MP4</td>
</tr>
</tbody>
</table>
7.3.2.1 Performance of link adaptive algorithms

7.3.2.1.1 Zero feedback delay

(a) Average frame PSNR vs. average channel CIR

(b) CDF of frame PSNR at CIR 21.5 dB

Figure 7.12: Performance of the quality based adaptation scheme.
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The performance of the Source (video) Quality-Based Adaptation Scheme (SQBAS) is compared to the performance of three other schemes. Two of them involve the use of fixed coding schemes; MCS-1 with the source rate fixed at 22.5 kbits/s and MCS-5 with the source rate fixed at 57 kbits/s. The third (CBAS) uses both the MCS-1 and MCS-2 coding schemes and switching between MCS-1 and MCS-5 is made based on the actual CIR of the transmission channel. Figure 7.12(a) clearly illustrates the improvement in video quality under all channel conditions when the Adaptive Coding Schemes (ACS) are used. For example, the achieved quality improvements with SQBAS relative to the MCS-1 and MCS-5 for an average channel CIR of 20.4 dB are 4.13 dB and 3.14 dB respectively. The result also shows that the proposed algorithm, SQBAS, slightly outperforms CBAS. For clarity, the video quality variation throughout a transmission is illustrated in Figure 7.12 as the Cumulative Distribution Functions (CDF) of averaged frame PSNR corresponding to the points a1, a2, a3 and a4 on Figure 7.12(a) (average CIR of 21.5 dB). The quality improvement is also visible when viewing the decoded video sequences (Figure 7.13).

Figure 7.13: Perceptual quality comparison.
Performances of the Throughput-Based Adaptation Scheme (TBAS) were measured in terms of system throughput savings ($T_{\text{adapt,mcS}1}$) and service quality improvements ($Q_{\text{adapt,mcS}5}$) which are defined as:

$$Q_{\text{adapt,mcS}5} = \frac{Q_{\text{adapt}} - Q_{\text{mcS}5}}{Q_{\text{mcS}5}} \times 100\% \quad \text{Equation 7.13}$$

$$T_{\text{adapt,mcS}1} = \frac{T_{\text{mcS}1} - T_{\text{adapt}}}{T_{\text{mcS}1}} \times 100\% \quad \text{Equation 7.14}$$

where $Q$ and $T$ indicate the median frame PSNR of the decoded video sequence and the number of time slots used in the transmission respectively. The resulting normalised quality improvement ($Q_{\text{adapt,mcS}5}$) and normalised system throughput savings ($T_{\text{adapt,mcS}1}$) show a linear relationship for operation at a range of fixed source rates (Figure 7.14). The normalised quality improvement achieved tends to decrease with the increase in source rate. This can be explained taking an example. Assume video sequences encoded at 38 kbps and 60 kbps are transmitted over the same time-varying channel and that the same mode switching threshold is used. The percentage of time slot saving in switching from MCS-2 to MCS-5 for lower rate transmission is 60%. The resulting timeslots saving is 50% for high source rate transmission. In order to achieve the same timeslot saving for high source rate transmission, the switching threshold should be lowered. This requires the transmission of more video data using low quality MCS-5 mode. Thus, it reduces the normalised video quality seen in high source rate transmission.

![Figure 7.14: Performance of throughput based adaptation](image-url)
7.3.2.1.2 Effect of feedback delay

The difference in quality, between zero-delay feedback (in terms of average frame PSNR), and finite delay feedback is shown in Figure 7.15. The SQBAS is robust to feedback delay up to a delay of 180ms. Above that, performance decreases with delay, resulting in 0.5 dB quality reduction at 240ms. TBAS under-performs SQBAS in terms of robustness to feedback delay, resulting in 0.5 dB quality reduction at 100ms. However, as the TBAS operates at radio block level, the expected feedback delay is limited to the duration of a few radio-blocks, which is less than 100ms. Therefore, expected drop in quality is small for both schemes.

![Figure 7.15: Effects of feedback delay on algorithm performance. Channel characteristics: ave. CIR - 14.4 dB, std. CIR - 4.2 dB. Channel allocation: 3 time-slots fixed for SQBAS, 3.8 time-slots on average for TBAS.](image)

7.3.2.1.3 Effect of noisy feedback

The effect of noisy feedback on the performance of SQBAS is illustrated in Figure 7.16. The scheme is relatively robust to noisy feedback with average PSNR degradation being around 0.5 dB. The effect of noisy feedback is similar to the effects of delayed feedback, in the sense that an inaccurate channel estimate for mode decision is made. In SQBAS, the mode decision interval, which equals to the video frame rate, is usually wide enough to accommodate reasonable prediction errors of channel states.
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73.2.1.4 Effect of burst errors

The effect of burst error on the performance of the quality based adaptation scheme is investigated in terms of standard deviation of frame PSNR over a number of runs for a set of simulated time-varying channels. The calculated standard deviation of frame PSNR is averaged over 1500 frames in each case and results are depicted in Table 7.6. Fixed rate MCS-1 is most robust to the burst errors while MCS-5 shows least robustness. Adaptive algorithms show moderate robustness, however, due to the averaging property in channel estimation algorithm, SQBAS is more robust than CBAS.

Table 7.6: Effect of burst channel errors

<table>
<thead>
<tr>
<th>Method</th>
<th>Std [dB]</th>
</tr>
</thead>
<tbody>
<tr>
<td>MCS-1</td>
<td>0.7516</td>
</tr>
<tr>
<td>MCS-5</td>
<td>1.5908</td>
</tr>
<tr>
<td>CBAS-variable rate</td>
<td>1.4999</td>
</tr>
<tr>
<td>SQBAS</td>
<td>1.0294</td>
</tr>
</tbody>
</table>

73.2.1.5 Performance comparison

The video source rates and the channel-switching thresholds for TBAS were selected in such a way as to achieve on average 3-time slots allocation, thereby allowing easy comparison with the SQBAS, which guarantees fixed 3 time slot operation.
Figure 7.17 shows that the performance of SQBAS is much better than that of TBAS. The figure also shows the effect of source rates and channel coding rates on video performances over error prone fixed bandwidth channels. The video quality decreases with increase of source rate as provided channel protection decreases.

Figure 7.17: Performance of SQBAS and TBAS for transmission of video over a time-varying channel with an average CIR of 14.4 dB and a standard deviation of 4.2 dB.

7.4 Link-adaptation for streaming video communications

Ideally, the design of link adaptation algorithms for multimedia services should optimise the delivery of acceptable video quality, facilitating good quality video services with the minimum possible demand on network resources. The previous section showed that appropriate design of link adaptation algorithms results in improved perceptual quality for one-to-one conversational services compared to non-adaptive schemes. In this section, a method of applying a link adaptation technique to streaming (one-to-many) services is proposed. To achieve link adaptation for streaming applications, the algorithm should ideally be able to switch between a number of source coding rates without performing custom encoding for each user. One way to do this would be to use scalable coding [3GPP TS 04.64]. However, scalable coding tends to be much less compression efficient than a single stream [SUN-01], and is therefore not very efficient in low bit rate channels. Therefore, to improve the efficiency of the link adaptation, the scheme proposed in this paper switches between pre-encoded streams stored at the server. No extra user interaction is required and the link adaptation algorithm is transparent to the end user.
Figure 7.18 shows how the EGPRS packet-switched system architecture is modified for this proposal. The most important service entities for streaming are the content server and the streaming client, which together with the EGPRS core network entities ensure correct media delivery to the user. Link quality and QoS profile storage is the element that must be added to the system to provide link adaptation. It stores the most current measurements provided by the BSC (Base Station Controller), regarding the time-varying radio link. In addition, it stores hand over and requested QoS parameters for each individual user. This recorded information can be used by a link adaptation algorithm.

**Figure 7.18:** The system architecture: EGPRS packet switched streaming services.

### 7.4.1 Stream-switched link adaptation

As in Section 7.3.1.1.1, the proposed Link Adaptation Algorithm (LAA) is designed to improve video quality by varying the source rate and the channel-coding scheme according to measured channel condition, while using a fixed number of EGPRS Time Slots (TS’s). This means that although the source rate is allowed to vary, the allocated resources across the radio link do not change. This section describes how streams can be switched at the server, and how the link adaptation algorithm operates.

#### 7.4.1.1 Stream switching

The most obvious way of providing different source rates for each user would be to use scalability. However, this is inefficient in terms of compression. An alternative is to encode “on-the-fly” for each user, which requires significant computation, and does not scale easily.
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The approach proposed here involves the use of pre-stored streams on the server (see Figure 7.19). The server buffers encoded video frames from each stream, and transmit the frame corresponding to the rate specified by the link adaptation algorithm.

![Figure 7.19: Example of Stream-Switched link adaptation, where $VF_n$ is the nth video frame in the sequence, and $B$ is the buffer (see section ).](image)

Use of the AIR technique (described in Section 2.3.3.5) is critical to the success of this switching method. When switching is performed between streams, there is a mismatch between the encoder and decoder in terms of the reference frames used to predict future frames. This can potentially lead to drift [DOGA-01]. However, the AIR technique limits the effects, and prevents it becoming a detectable problem.

### 7.4.1.2 Link Adaptation Algorithm (LAA)

$M \times N$ number of buffers are allocated for each video sequence in the content server, where $N$ is the number of Time-Slots (TS) in a radio frame (8 for EGPRS) and $M$ is the number of Modulation-Coding Schemes (MCS’s) supported (9 for EGPRS). Buffers are labelled:
In practice, it is not necessary to use all possible combinations of TS's and MCS's, as demonstrated in Section 7.3.1.1.1. Minimum possible source rate, where high motion “Foreman” sequence can be encoded at, is 27 kbps. Therefore, MCS-1, which supports 22.5 kbps with 3-time slots, can not be used in the adaptive scheme for 3-time slot operation. Thus, MCS-2 and MCS-5 are used in the algorithm implementation.

According to the video bit rate, \( R_n \), and assuming MCS-5 is used, \( n \) number of time-slots, \( TS_{n,h} \), are selected for user \( i \). As can be seen from Table 7.4, for selected \( TS_{n,h} \) there are \( M \) different video source rates (corresponding to different MCS schemes), which can be applied. The link adaptation algorithm is designed to adapt each radio link to one of \( M \) MCS schemes. Let \( CIR_{k,t} \) be the measured channel condition at the \( k^\text{th} \) radio block for user \( i \). Modulation-coding scheme mode \( m \) and video source rate \( R_{n,m} \) are chosen if:

\[
CIR_{k,t} \in [\xi_{th}(m), \xi_{th}(m+1)]
\]

where \( \xi_{th}(m) \) and \( \xi_{th}(m+1) \) indicate corresponding channel threshold values. The main steps of the proposed link adaptation algorithm are listed below.

1. Select \( n \) number of TS's needed to satisfy the user's source bit rate requirement.
2. Check for the start of the \( j^\text{th} \) video frame. If start go to step 3.
3. Estimate the channel condition for next radio frame from channel condition measurements. Check for channel-code switching conditions. Select the video source rate \( R_{n,m} \) and the MCS for the \( j^\text{th} \) video frame according to the estimated channel condition.
4. Switch to buffer \( B_{n,m} \) and use frame header information to find the start of the \( j^\text{th} \) video frame.
5. Transmit data for the \( j^\text{th} \) video frame from buffer \( B_{n,m} \) with modulation-coding scheme \( m \) using \( n \) TS's of the current radio frame.
6. If there are more video frames to be transmitted, return to step 2.

The sequences are MPEG-4 coded using all of the error resilience options and with AIR enabled. Sequences are compressed with different non-varying bit rate outputs using the TM5 [ISO/IEC 13818-2] rate control algorithm. TM5 varies the quantisation to ensure a particular target rate is achieved and does not drop frames. To achieve the lower bit rates, it is necessary to reduce the number of non-predictive AIR macroblocks per frame. This means that while 10 AIR MB's are used for MCS-5, only 5 are used for MCS-2.
7.4.2 Results and discussion

7.4.2.1 CIR based LAA

Tests were performed as described in section 4 using the actual CIR value of the communications link. Accurate determination of the CIR is difficult in a practical system, so the performance of this LAA scheme should be seen as ideal. No delay or feedback corruption was used.

The test results are shown in Figure 7.20. It should be noted that the CIR value on the x-axis of the graphs is the average CIR of a time varying channel. The results clearly show that ideal LAA outperforms both of the fixed schemes over almost the entire CIR range. Figure 7.20 gives an indication of the difference in quality that can be expected from using LAA. CIR LAA improves PSNR by more than 2dB for certain channel conditions.
Another noticeable element of the results in Figure 7.20 is the similar performance of MCS-2 to MCS-5 at low average CIR’s. This is due to the lower error recovery properties of the MCS-2 bitstream: each frame contains fewer AIR blocks than in the MCS-5 sequence.

7.4.2.2 BLER based LAA

Further tests were carried out using BLER based measurement of channel quality, which is a more practical measurement method for implementation. The results in Figure 7.21 show that the performance of BLER LAA is only slightly lower than that of the ideal CIR LAA. For high average CIR, BLER LAA outperforms CIR LAA with Suzie. The gain in performance of BLER LAA over the fixed schemes is significant, and provides only slightly lower performance gain to the CIR LAA described in Section 7.4.2.1.
7.4.2.3 Sensitivity of switching threshold

The switching threshold used in the experiments here was determined experimentally. However, the optimum threshold was slightly different for the Suzie and Foreman sequences, indicating that
threshold switching can only be performed optimally by considering the content. In a practical implementation this is not feasible. Because precise optimisation of switching threshold according to the encoded content is not possible, it is important to examine the LAA scheme's sensitivity to different threshold values.

Figure 7.22: Simulation results using different threshold values for CIR LAA switching.
Figure 7.22 shows the results of tests conducted using different switching thresholds for CIR based LAA. For the Suzie sequence, the results show that 22.5 dB and 21.5 dB are both good choices for switching threshold. For Foreman, the situation is more complex, as the best performing threshold appears to change depending on the average CIR. However, the thresholds of 22.5 dB and 21.5 dB appear to be good choices. The results show that there is some tolerance for setting a sub-optimal threshold value, and that it is preferable to set a threshold value slightly lower than the optimal rather than slightly higher.

7.4.2.4 Sensitivity to feedback delay and errors

![Graph showing sensitivity to feedback delay and errors](attachment:graph.png)

(a) Feedback delay (no feedback corruption)

![Graph showing sensitivity to feedback corruption](attachment:graph2.png)

(b) Feedback corruption (140 ms delay)

Figure 7.23: Simulation results, at an average CIR of around 20 dB, showing the sensitivity of BLER LAA to delay and error.
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The BLER LAA described here uses channel prediction estimates to minimise the effects of delay on the link adaptation performance. Tests were conducted using a variety of delay values to evaluate the scheme's sensitivity to delay. Additional tests were performed to evaluate the effects of feedback corruption on system performance by simulating transmission of the feedback bitstream using the same channel model described in section 3. A delay of 140 ms was used in these additional tests.

Figure 7.23(a) shows the results of feedback delay on the Suzie and Foreman sequences. The results differ slightly between the two sequences, but it is clear that the algorithm remains beneficial in the face of typical EGPRS network delays. However, even with this scheme it is preferable that delay is kept as low as possible. Figure 7.23(b) shows only minor variation with increasing Bit Error Rate (BER).

7.4.2.5 Comparisons with power control techniques

Power control techniques for TDMA radio access systems typically use the same measurements as those used for the link adaptation scheme presented here. This means that it is possible to use both schemes on the same transmission, where a decision would have to be made on whether to switch streams, or change the power settings. In this section, the performance of the link adaptation algorithm is compared to that of power control techniques and comments are made concerning the possible improvements that can be achieved by combination of power control and link adaptation.

The simple power control algorithm implemented for this task, closely follows the feedback error rate power control algorithm proposed in [CHER-96]. However, instead of using the feedback error rate as an indication of channel quality, here the actual channel carrier to interference ratio is used in making the power stepping decision. Therefore, the achieved performance should be considered as ideal performance. Another simplification of the algorithm compared to the paper by Cherriman et al [CHER-96] is the use of fixed power stepping decision threshold and power step size. Similar to [CHER-96], the dynamic range of the algorithm is limited by two settings: the maximum power (30 dBm) and the power transmission dynamic range (64 dB).
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(a) Comparison of CIR at MS with and without power control.

(b) Base station transmission power.

Figure 7.24: Example power control scenario.

An example of the performance of the power control algorithm is shown in Figure 7.24. The figure shows the CIR of the channel that would be experienced by the MS in the absence of power control techniques. For the same channel, it also shows the channel conditions experienced by the MS and the transmission power variation in the power control scenario.

Power control simulations were carried out using fixed MCS-2 and MCS-5. Figure 7.25 shows the power control results compared to the CIR LAA. The results show that CIR LAA performance is similar to the best-case ideal power control scenario for average channel CIR value up to 24 dB. However, at high CIR values, the power control technique out-performs the link adaptation performance. This is due to the fact that the power control algorithm tends to prevent significant periods of low CIR, while the link adaptation algorithm switches to MCS-2 at low CIR values, reducing the effective source bit rate. However, for channel conditions in the range of 12-24 dB, which is the average channel condition experienced by most mobile users, both techniques perform similarly.
If only one scheme is implemented, then the results appear to make LAA a more attractive proposition, given the problems with power control mechanisms (e.g. reduced system capacity due to increased interference with other users). The results also suggest that combining both schemes would result in an improved Link Adaptation scheme.

7.5 Link adaptation for UMTS

As shown in Section 7.3 & 7.4, adaptive modulation coding schemes provide a powerful means of exploiting the time varying channel quality fluctuations of wireless channels. Spreading gain provides the key variable in determining user data rates and associated channel quality in CDMA based communication systems. Therefore, in addition to channel coding schemes, adaptive spreading gain can also be used to exploit time varying channels in CDMA systems. In this section, performance gain, which can be achieved by means of adaptive spreading gain control for real-time video communications over UMTS, will be discussed. Further perceptual quality enhancement is achieved via the joint application of unequal error protection techniques, which are described in Chapter 6, and adaptive spreading gain control.
A variety of adaptive rate schemes have been proposed in the literature for CDMA based communications systems. An Adaptive Code Division Multiple Access (ACDMA) scheme is proposed in [ABET-96]. The transmission rate was modified by varying the channel code rate and the processing gain of the CDMA user according to the instantaneous channel conditions. The performance of a transmitter power adaptation and information rate adaptation scheme was compared in [KIM-99]. It was concluded that rate adaptation provided the higher average information rate for a given average transmit power and given BER condition. Spread Adaptive Quadrature Amplitude Modulation was proposed in [KUAN-00]. It exploited the time variant channel quality of mobile channels by switching either the modulation mode or the spreading factor on a burst-by-burst basis. The multi-user joint detector and the successive interference cancellation receiver gain were analysed and compared in the context of these adaptive schemes. Bit rate adaptation with the aim of solving local coverage problems for uplink transmission was discussed in [FIOR-00]. Evaluations were based on the estimated BLER experienced by the user and the mobile output power. System level simulation showed that an increment in cell coverage can be achieved with bit rate adaptation while maintaining a requested BLER. Also if the coverage is fixed then, link quality improvement can be gained for uplink transmission. A dynamic spreading code assignment algorithm, which efficiently shares a WCDMA downlink between data traffic sources and different Quality of Service requirements have been presented in [FOSS-02, KAM-01]. Both analytical and simulation results showed that the dynamic code allocation algorithm provides higher bandwidth utilisation compared to a non-adaptive code allocation scheme. An adaptive rate and power allocation algorithm for uplink throughput maximisation has been proposed in [JAFA-03]. Results concluded that the optimum rate and power allocation performs significantly better than a scheme that uses power adaptation alone.

The above mention spreading gain control related research has been carried out mainly at the system level. The main goal of the research was the improvement in system level performances, which can be categorised into

- system capacity maximisation,
- system throughput maximisation,
- quality improvement in terms of average channel BER or BLER,
- service flexibility and service multiplexing,
- higher system utilisation
- cell coverage.

However, little attention was given to the application level performances, such as received video quality in multimedia communications. As shown earlier in this thesis, perceptual video quality is a function of quantisation distortion, concealment distortion and distortion due to error propagation, thus received video quality greatly depends on the encoder format, error resilience
techniques, and error concealment techniques applied. Therefore, it is necessary to investigate the effect of spreading gain control on multimedia performance at the application level, and to produce an optimum scheme for video application.

7.5.1 Adaptive spreading gain control for real-time video communications

Adaptive spreading gain control techniques attempt to improve the received video quality by switching between different spreading codes levels depending on the state of the transmission channel. Source bit rate is varied according to the selected spreading factor within the Transmission Time Interval (TTI), while keeping the chip rate constant. In good channel conditions, quantisation distortion becomes the dominating factor in received video quality. Therefore, in order to reduce the quantisation distortion, a code with a low spreading factor, which supports a higher source rate, is selected in favourable channel conditions. Conversely, in hostile channel conditions, a high spreading factor is used to minimise the channel distortion.

The rate switching threshold is selected according to the link level simulation results, which are explained in Chapter 4. The transmission power is kept at a constant level for a certain channel SNR. That ensures that the interference power experienced by other users is not affected by the adaptive spreading gain control techniques. \( \frac{E_b}{N_0} \) and SNR are inter-related, and the derivation of SNR from \( \frac{E_b}{N_0} \) is as shown in Equation 7.16.

$$ SNR = \frac{R \cdot E_b}{W \cdot N_o} $$

\[ \text{Equation 7.17} \]

where \( W \) and \( R \) denote the channel bandwidth and source rate respectively.

For clarity, Figure 4.12 is re-drawn in Figure 26 as average PSNR vs SNR instead of \( \frac{E_b}{N_0} \). As the figure illustrates, the condition used to switch between spreading factors (hence source rates) is set according to received video quality:

$$ \text{Switching mode} = \begin{cases} \text{SF16} & \text{for } SNR \geq -6dB \\ \text{SF32} & \text{for } SNR < -6dB \end{cases} $$

\[ \text{Equation 7.18} \]

Due to the varying value of \( R \), the spreading gain control technique results in a variation of transmit bit energy (Eb) over the duration of transmission. Selected transmission modes and corresponding source rates are listed in Table 7.7.
Table 7.7: characteristics of transmission modes.

<table>
<thead>
<tr>
<th>Mode</th>
<th>Mode 1</th>
<th>Mode 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Spreading factor</td>
<td>SF 32</td>
<td>SF16</td>
</tr>
<tr>
<td>Source rate</td>
<td>64.5 kbps</td>
<td>137.4 kbps</td>
</tr>
</tbody>
</table>

MPEG4 performance over Vehicle A channel with 1/3 rate convolutional channel coding.

Figure 7.26: Video performance over Vehicular A channel with 1/3 rate convolutional channel coding.

7.5.1.1 Experiment and results

Table 7.8: Parameter values used in UMTS channel simulation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Propagation environment</td>
<td>VehA</td>
</tr>
<tr>
<td>Mobile speed</td>
<td>50 km/h</td>
</tr>
<tr>
<td>Log-normal variance</td>
<td>10 dB</td>
</tr>
<tr>
<td>De-correlation length</td>
<td>20 m</td>
</tr>
<tr>
<td>Hexagonal cell radius</td>
<td>2 km</td>
</tr>
<tr>
<td>Fading Characteristics</td>
<td>Raleigh fading</td>
</tr>
<tr>
<td>Orthogonality factor</td>
<td>0.6</td>
</tr>
<tr>
<td>Channel coding</td>
<td>CC1/3</td>
</tr>
</tbody>
</table>
Similarly to the EGPRS time varying channel model explained in Section 7.2.4, a time varying channel model is developed for the UMTS down link. Table 7.8 lists the simulation parameters used. A detailed description of the time varying channel model for UMTS considering multi user scenarios is given in Section 8.3.

Figure 7.27: Performance of adaptive spreading gain control scheme. (a). Suzie (b). Foreman
Perfect Signal to Noise Ratio (SNR) estimation is assumed. Therefore, the obtained results show an upper-bound of performance estimates. Full error resilience MPEG-4 coded video sequences were transmitted over simulated time varying channels. The performance results in terms of average frame PSNR vs average channel SNR (of time varying channel) are presented in Figure 7.27. Each point is produced by averaging frame PSNR values for 50 different runs over at least 15 different channel profiles with the same mean SNR. For comparison, the performances of the non-adaptive schemes with spreading factors 32 and 16 are also shown in the figure. The adaptive spreading gain control scheme shows significant quality improvement for the transmission of both low and high motion video sequences. However, higher gain is achieved for the low motion sequence. As the theory suggests, the performance of the adaptive scheme gets closer to that of spreading factor 16 operation in good channel conditions. With poor channel conditions, the performance gets closer to that of spreading factor 32 operation.

7.5.2 Joint Adaptive Spreading gain control and Unequal Error Protection scheme (JAS-UEP)

Unequal error protection schemes exploit the different level of importance of transmitted data for perceptual quality, and applies different levels of channel protection accordingly in order to achieve maximum received quality for given channel conditions. The link adaptation techniques, on the other hand, exploit time varying channel quality by adaptive control of the transmission mode. Logically, combination of these two techniques should result in improved performances. In this section, the combination of adaptive spreading gain control and unequal error protection schemes is examined for real-time video communications over UMTS systems.

An MPEG-4 data partition based UEP scheme, (see Section 6.2) is combined with the developed adaptive spreading gain control scheme. UEP supports higher source bit rates than the 1/3 convolutional coding scheme. As given in Equation 7.16, channel SNR increases with increase of source rate. In order to maintain the same SNR, information bits should be transmitted with lower energy in JAS-UEP scheme compare to adaptive spreading gain control scheme. This results in lower video quality for a given channel SNR. This problem is over come with the use of unequal bit energy allocation for different bearers in the JAS-UEP scheme. The overall joint adaptive spreading gain control and unequal error protection scheme is depicted in Figure 7.28.
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Each encoded video frame is separated into two streams based on MPEG-4 data partitioning. The high priority data stream (the first partition) is channel protected with 1/3 convolutional coding while 1/2 rate convolutional code is used to protect the low priority data stream (the second partition). Channel coded streams are allocated with different transmit bit energy levels based on the JAS-UEP decision command. Finally, transport channels are multiplexed into the same physical channel for transmission over the wireless link. Feedback channel information is used to predict the channel conditions. According to the predicted channel conditions, transmission mode selection decision is made.

Two transmission modes are considered. As explained in Section 6.2, the supportable source rate by the UEP scheme depends on the sequence characteristics, channel bit rate, and ratio between the size of the first partition and the second partition. Re-writing Equation 6.2:

\[
R_s = \frac{(1 + \varphi) \cdot R_{ch}}{X_{ch2} + \varphi \cdot X_{ch1}} \cdot \frac{1}{(1 + OH)}
\]

Equation 7.19

where \( R_s, R_{ch}, 1/X_{ch1}, \) and \( 1/X_{ch2} \) denote source rate, channel bit rate, channel coding rate of first stream and channel coding rate of second stream respectively. \( \varphi \) is the ratio between the amount of data in the first partition and the amount of data in the second partition. \( OH \) is the extra
overhead incurred. Average bit energy for the transmission can be calculated from Equation 7.16 as,

$$E_b = \frac{SNR \cdot W \cdot N_0}{R_s}$$  \hspace{1cm} \text{Equation 7.20}

Let $E_{b1}$ and $E_{b2}$ denote the allocated bit energy for Stream 1 and Stream 2 respectively. In order to satisfy the average bit energy requirement:

$$\frac{\varphi}{1 + \varphi} \cdot E_{b1} + \frac{1}{1 + \varphi} \cdot E_{b2} = E_b$$  \hspace{1cm} \text{Equation 7.21}

Say $E_{b1} = E_b + x$ and $E_{b2} = E_b - y$, then

$$\frac{x}{y} = \frac{1}{\varphi}$$  \hspace{1cm} \text{Equation 7.22}

Based on the experimental results, $y$ is selected to be $0.25E_b$. The calculated source rate and transmit bit energy for the two operation modes for transmission of the “Suzie” sequence is listed in Table 7.9.

**Table 7.9: Operation modes of JAS-UEP scheme.**

<table>
<thead>
<tr>
<th>Video sequence</th>
<th>Suzie</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mode</td>
<td>Mode 1</td>
</tr>
<tr>
<td>Stream 1</td>
<td>32</td>
</tr>
<tr>
<td>Spreading factor</td>
<td>$88$ kbps</td>
</tr>
<tr>
<td>Source rate</td>
<td>$E_{b1} + 2.33$</td>
</tr>
<tr>
<td>Channel code</td>
<td>CC 1/3</td>
</tr>
<tr>
<td>Bit energy (dB)</td>
<td>$E_{b1} + 2.33$</td>
</tr>
</tbody>
</table>

Where, $E_{b1}$ and $E_{b2}$ indicate the average bit energy for mode 1 and mode 2 respectively.

### 7.5.2.1 Experimental results

The experiments carried out in Section 7.5.1.1, are repeated for the transmission of the “Suzie” sequence over the Vehicular A UMTS channel with the application of the JAS-UEP scheme. Results are plotted in Figure 7.29. The JAS-UEP scheme outperforms the adaptive spreading gain control scheme in good channel conditions. However, slightly lower performance is visible compared to that of the adaptive spreading gain control scheme with poor channel conditions.
This is due to the poor performance of MPEG-4 data partition based UEP scheme in extreme channel conditions, which is described in Section 6.2. Also the transmit energy levels on two streams are kept constant for selected transmission mode independent of the instantaneous channel quality. These energy levels can optimally be controlled according to the characteristics of the channel, thereby optimal performance gain can be obtained over a wide range of channel conditions.

![Figure 7.29: Performance of the JAS-UEP scheme.](image)

### 7.6 Conclusion

This chapter explores application level adaptive techniques, which can be applied to enhance video quality while the media is being delivered. A novel link adaptation scheme, based on measurable quantities in a practical cellular system, was introduced to exploit the time-varying nature of the mobile radio channel. Two modes of operation: the Source (video) Quality-Based Adaptation Scheme (SQBAS) and the Throughput-Based Adaptation Scheme (TBAS) are introduced for variable application source rate and fixed application source rate operation respectively. The algorithms' performances are evaluated for video communications over EGPRS systems. Our results reveal that when offered with similar traffic loads over similar channel environments, the quality-based scheme can provide noticeable improvements in video quality, compared to the throughput-based scheme and any fixed coding scheme. The throughput-based
adaptation scheme shows an approximately linear relationship between the system throughput-savings and the video quality improvements. The proposed algorithms based on BLER, RSS and first order statistics of RSS, perform as well as the CIR based adaptive scheme. Furthermore, the investigation shows that the proposed algorithms are robust against feedback delay, noisy feedback and burst of channel errors. These results reveal that proper control of link adaptation mechanisms can be used to maximise the system throughput while maintaining adequate service quality for real-time video communications over wireless networks.

A technique that provides delay robust link adaptation for streaming video over EGPRS mobile networks is presented. The proposed scheme uses the Adaptive Intra Refresh (AIR) technique in MPEG-4 to mitigate the drift effect resulting from stream switching. The switching is performed between two pre-encoded bit streams, intended for use with two different modulation-coding schemes. This removes the need for the encoder-decoder interaction normally associated with link adaptation. By using a prediction method for future channel conditions, the technique is also robust to delay in the feedback channel. Tests were performed using EGPRS channel models, comparing the results of fixed modulation-coding scheme scenarios to the link adaptation method. Results show significant quality improvements with the adaptive scheme. Further tests reveal that the feedback data is acceptably robust to channel errors. The CIR based link adaptation algorithm is shown to perform similarly to the best-case ideal power control scenario, and the results indicate that combining power control with the proposed LAA would result in a better scheme than ideal power control.

Finally, link adaptation techniques in UMTS networks are examined. An adaptive spreading gain control algorithm is proposed and analysed for real time video communications. Conducted experiments show significant improvements in received video quality. Furthermore, adaptive spreading gain control scheme is applied in conjunction with unequal error protection scheme. It was shown that the joint adaptive spreading gain control and unequal error protection scheme can be used to achieve improved video performance in wireless networks.
Chapter 8

8 System Level Quality Enhancement Techniques

8.1 Introduction

The video transmission experiments and quality enhancement techniques, introduced in Chapters 4-7, demonstrated the quality of received video by individual users for video transmission for given channel conditions. The quality of communication channels in the cellular system is greatly influenced by the presence of other users in the system due to the generated interferences. Especially in CDMA based cellular systems, system capacity and coverage depend on the generation of interference from other users in the same cell and also in neighbouring cells. Therefore, it is necessary to investigate the performance of video communication at the system level, taking into account system parameters such as system load, total base station transmit power (power budget) and the number of spreading codes assigned for each cell (code budget). Video performances in multi-user scenarios for down link transmission are investigated in this chapter. Two system level quality enhancement techniques are discussed. The first method is intended to maximise the average received video quality for a power limited system. A power control technique, which maximises the total Signal to Noise ratio received by users in the cell is designed and implemented. The second method combines the adaptive spreading gain control technique, proposed in the previous chapter, with the power control technique to further enhance perceptual video quality in time-varying error-prone propagation environments. System
performances are evaluated in terms of the number of satisfied users in the system and the average received quality for video transmission in a simulated multi-user UMTS network.

8.2 Channel quality calculations in multi-user environments

In second generation cellular systems, which are based on FDMA/TDMA technologies, users are separated in the time and frequency domains by allocating separate transmission channels. Therefore, ideally the signal that is transmitted to one user does not appear as interference to other users in the cell. In other words, system capacity is fixed and it is defined by the number of channels allocated to each cell by the cell planning procedure. The system coverage depends on the propagation characteristics and the base station/terminal power specifications. In these systems, adaptive power control can be used to mitigate the effect of time-varying channel conditions for individual users, without affecting the quality of the communication channels experienced by other users.

On the other hand, in CDMA systems, users are separated by their allocated spreading code for the transmission. Even though these spreading codes are orthogonal to each other in narrow band channels, they can not be considered to be perfectly orthogonal in multi-path propagation environments [HOLM-01]. This means that users in the same cell and neighbouring cells interfere with each other. The system capacity and the cell coverage are therefore defined by the total system traffic load, carrier transmit power, maximum base station transmit power and the number of available codes. The channel conditions, experienced by the users, varies according to the system traffic load at a given time. In this section, the channel quality calculation, under multi-user scenarios in a UMTS down link, is presented.

8.2.1 Modelling of down link SNR

A variable rate downlink data transmission system for UMTS is considered. Assume that the number of Mobile Terminals (MTs) in cell $j$ is $N_j$. The MTs are assumed to be randomly distributed. The radio propagation is modelled by path loss, shadow fading and multi-path propagation. Suppose that the transmission power from the base station $j$ to the $i^{th}$ MT in the $j^{th}$ cell is $P_{ij}^f$. State that the path loss component is represented by $L_{ij}^f(f,d,x)$, where $f$ represents transmitting frequency, $d$ is the distance between transmitter and receiver, and other
factors such as geometric orientation, and heights of antennas are represented by $x$. Shadow fading, $S_{ij}^f$, is represented by a log-normal distribution, and is a function of velocity, $v$, decorrelation length, $d_c$, and log-normal variance, $\sigma$. Assume that the fast fading effect is denoted by $F_{ij}^f(f,v)$. Thus, the received power, $P_{R(i,j)}^f$, by the $i^{th}$ MT in the $j^{th}$ cell is

$$P_{R(i,j)}^f = P_{T(i,j)}^f \cdot L_{ij}^f(f,d,x) \cdot S_{ij}^f(v,d_c,\sigma) \cdot F_{ij}^f(f,v)$$  

Equation 8.1

Suppose that the number of paths in the multi-path channel is $L$ and path gain is denoted by $a_{l(i,j)}^f$. Then,

$$F_{ij}^f(f,v) = \sum_{l=1}^{L} a_{l(i,j)}^f \cdot F_{l(i,j)}^f(f,v)$$  

Equation 8.2

where $F_{l(i,j)}^f(f,v)$ denotes the fast fading effect of the $l^{th}$ path.

As explained in Section 3.5.2, interference from users in the same cell can be represented by a concept called "orthogonality factor" in the downlink. Let $\mu$ be the downlink orthogonality factor for the propagation environment under consideration. $\mu$ takes values between 0 and 1. One corresponds to a perfectly orthogonal system, while zero corresponds to an un-orthogonal system. In practice, $\mu$ depends on the location of the mobile and the characteristics of the propagation environment [HUNU-02]. Here, for simplicity, an average value of $\mu$ is used in the SNR calculation.

Let power on common control channels (broadcast & paging channels) and thermal noise power be denoted by $P_c$ and $N_0$ respectively. Then received signal to noise ratio, $SNR_{i,j}$ at the $i^{th}$ MT in the $j^{th}$ cell can be computed as

$$SNR_{i,j} = \frac{P_{T(i,j)}^f \cdot L_{ij}^f \cdot S_{ij}^f \cdot F_{ij}^f}{N_0 + (1-\mu) \cdot P_c \cdot \bar{L}_{ij}^f \cdot \bar{S}_{ij}^f \cdot \bar{F}_{ij}^f + \sum_{k=1,k \neq i}^{N} \left( (1-\mu) \cdot P_{T(k,j)}^f \cdot \bar{L}_{ij}^f \cdot \bar{S}_{ij}^f \cdot \bar{F}_{ij}^f \right) + \sum_{j' \neq j}^{I} \sum_{j=1}^{I} \left( P_{T(i,j')}^f \cdot \bar{L}_{ij}^f \cdot \bar{S}_{ij}^f \cdot \bar{F}_{ij}^f \right)}$$  

Equation 8.3

$P_T^f$ denotes the total transmitted power of the $j^{th}$ base station. $I$ is the number of interfering cells in the system. $\bar{L}, \bar{S}$, and $\bar{F}$ denote the effect due to path loss, shadow fading and fast fading on an interfering signal in the same cell while $L, S$, and $F$ represent those for common control signals.
CDMA systems are generally interference-dominant and, especially at the maximum capacity utilisation, thermal noise term, $N_0$, can be ignored. $S_{i,j}$ and $\bar{S}_{i,j}$ (also $\bar{S}_{i,j}$) are assumed to be correlated considering the synchronous transmission. However, no correlation is assumed between $S_{i,j}$ and $S_{i,j}$.

8.2.2 System capacity limits in downlink UMTS

The capacity of WCDMA based cellular communications systems are limited by two main factors. They are,

- Interference and power limitation
- Code limitation.

As explained in Section 8.2.1, the received channel quality at the user terminal (in downlink transmission) is affected by the number of users in the system and the carrier transmit power. Therefore, the maximum base station transmit power is a major parameter in downlink system specification. Let maximum base station transmit power of cell $j$ be $P^j_M$. Assume the minimum transmit power needed to satisfy the user quality requirement for the $i^{th}$ user is $P^{l(i,j)}_M$. If the maximum number of permitted users is $N_j$, then guaranteeing their quality requirements demands that

$$P^j_M \geq \sum_{i=0}^{N_j} P^{l(i,j)}_M + P_c \quad \text{Equation 8.4}$$

where $P_c$ is power allocated on common control channels.

Apart from the interference and power limits observed in CDMA based cellular systems, the total number of codes available also limits system resources. In UMTS, sets of codes from different code families are defined for use in downlink transmissions. The base stations in the system (or sector antenna in the sectored system) are assigned individual scrambling codes to prevent excessive interference from neighbouring cells. Orthogonal spreading codes are used to separate users in the same cell (or cell sector). Data belonging to different users in the same cell is transmitted with different spreading codes and scrambled with the same scrambling code. Each base station (or sector antenna) is allocated an Orthogonal Variable Spreading Factor (OVSF) code tree. The OVSF code tree and code selection procedure are described in Section 3.3.3. Since the base stations (or sector antennas) are uniquely defined by the allocated scrambling code, the same spreading code tree is reused by different base stations in the system. The number of users
that can be accommodated in a cell may exceed the number of available downlink spreading codes, especially in the case where the interference power from neighbouring cells is significantly low. In this case, the system is considered to be a code limited system. Several different techniques to increase the number of available downlink codes have been presented in the literature to avoid the code limitation. Use of larger quasi orthogonal code families has been presented in [AMAD-02]. Another way to tackle the problem is the reuse of spreading codes allocated to inactive users. This concept is used in the operation of UMTS shared channels.

In the system level simulator described below, each sector antenna is assigned an OVSF code tree. Given the source bit rate, $R_s$, channel coding rate, $1/X_c$, and WCDMA chip rate, $R_c$, then the maximum number of available spreading codes, $N_{code}$ for the specific service can be calculated as

$$N_{code} = 2^\mu \text{ such that } x = \frac{R_c}{R_s \cdot X_c} \leq 2^\mu$$

Equation 8.5

$\mu$ is the lowest integer value which satisfies the above condition and $2^\mu$ is equivalent to the allocated spreading factor for the transmission. For example, let the source rate be 64 kbps. Assume 1/3 rate convolutional coding is used and the WCDMA chip rate is 3.84 Mcps. The maximum number of spreading codes available for the specified service is calculated to be 32.

8.3 Joint link and system level simulation model

The simulation model discussed here combines the link level simulation model developed in Chapter 3 with a system level simulation model to emulate the effects of a complete system on multimedia transmission performance. A single complete simulation model is not feasible due to the complexity of such a simulator. Therefore, the system and link level simulators are developed separately and are combined to form the complete system. As discussed in Chapter 3, the link level simulator is developed at chip level, with 3.84 Mcps time resolution. Normally, system level simulators operate at fast power control frequency, which is set to 1.5 kHz in a WCDMA system. In our simulator, the fast power control technique is incorporated into the link level simulator. Therefore, the system level simulator is designed to operate at the Transmission Time Interval (TTI) level. This reduces the required system simulation time.
8.3.1 System level simulator

The Vehicular A environment is considered in the simulation. The system lay out implemented is the recommended 3-sectored cell layout for testing procedures in [UMTS 30.03]. The simulation area is divided in to 24 hexagonal cells as shown in Figure 8.1 each with a 2km cell radius. The system includes seven base stations and each illuminates three sectors (cells). Perfect sectorisation and uniform antenna radiation pattern for each sector antenna is assumed. The distances between base stations are equal to 6 km.

The positions of the mobiles and their direction of travelling are randomly chosen at initialisation. Cell selection is conducted based on the geographical location of the mobile. User location based hard handover is assumed in base-station-to-base-station and sector-to-sector hand over in the power and interference calculation.

![Figure 8.1: Simulation cell layout.](image)

The received signal and interference power at the mobile terminal is calculated considering the effect of path loss, slow fading and the mobility of the user. The methods used in modelling of these effects closely follows the 3GPP recommended models and are described below.

8.3.1.1 Path loss model

The path loss model used for the UMTS Vehicular test environment is [UMTS 30.03]:

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\[ L = 128.1 + 37.6 \log(d) \]

where \( L \) is in dB while \( d \) is in km. The parameter values used in the derivation are given in Section 7.2.2.

### 8.3.1.2 Shadow fading

The shadowing model described in Section 7.2.1, is used to generate the slow fading component of the channel. Slow fading is modelled as a log-normally distributed random variable with correlated consecutive samples and 10 dB variance. The de-correlation length is set to 20m in the Vehicular environment.

### 8.3.1.3 Mobility model

Users are assigned a mobile speed, which is kept constant for the duration of the simulation. The direction of the mobile is randomly selected when the call is set up and it is updated at a regular intervals defined by the de-correlation length. The direction is changed with a probability of 0.2 to a randomly selected direction. The maximum angle for direction update is set to 45°.

### 8.3.1.4 System specification

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>Channel type</td>
<td>Vehicular A</td>
</tr>
<tr>
<td>Cell configuration</td>
<td>3-sectored</td>
</tr>
<tr>
<td>Cell radius</td>
<td>2km</td>
</tr>
<tr>
<td>Distance between base station</td>
<td>6km</td>
</tr>
<tr>
<td>Mobile speed</td>
<td>50 km/h</td>
</tr>
<tr>
<td>Antenna diversity</td>
<td>No</td>
</tr>
<tr>
<td>Downlink orthogonality factor</td>
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</tr>
<tr>
<td>Path loss model</td>
<td>Yes</td>
</tr>
<tr>
<td>Shadow fading</td>
<td>Log-normal fading</td>
</tr>
<tr>
<td>Shadow fading variance</td>
<td>10 dB</td>
</tr>
<tr>
<td>De correlation length</td>
<td>20 m</td>
</tr>
<tr>
<td>Mobility model</td>
<td>Directional mobility model</td>
</tr>
<tr>
<td>Power settings</td>
<td></td>
</tr>
<tr>
<td>Max BS transmit power</td>
<td>20 W</td>
</tr>
<tr>
<td>Max carrier transmit power</td>
<td>1 W</td>
</tr>
<tr>
<td>HO algorithm</td>
<td>Hard hand over</td>
</tr>
</tbody>
</table>

Table 8.1: Parameters for system simulations.
Sectors are separated by allocating different scrambling codes. Users in the same sector are separated by allocated OVSF codes, which are taken from the OVSF code tree described in Section 3.3.3. Maximum transmit power per sector antenna is set to 20 W, while maximum carrier transmit power is set to 1 W. The orthogonality factor for the Vehicular environment is assumed to be 0.6. These system parameter values are taken from [UMTS 30.06]. All of the parameter values used in the system simulator are summarised in Table 8.1.

### 8.3.2 Combination of link and system level simulators

As the simulator is constructed in two separate parts, a method of interconnection is necessary to form the complete system. Conventionally, link level performances are incorporated into system level simulators through an average value interface. In this case, the link level performances are described in terms of BER/BLER performances for a given average Eb/No requirement. This interconnection method is suitable for derivation in statistical performance figures, which are used in cell/coverage planning procedures, service/mobility modelling and system planning. The intention of the simulator described here is the investigation of multimedia performances in UMTS systems. Therefore, the effect of instantaneous channel quality variation at chip level must be incorporated into the simulator.

<table>
<thead>
<tr>
<th></th>
<th>Link level simulator</th>
<th>System level simulator</th>
</tr>
</thead>
<tbody>
<tr>
<td>Multi-path model</td>
<td>yes</td>
<td>indirectly</td>
</tr>
<tr>
<td>Receiver characteristics</td>
<td>yes</td>
<td>indirectly</td>
</tr>
<tr>
<td>Channel coding/interleaving</td>
<td>yes</td>
<td>indirectly</td>
</tr>
<tr>
<td>Slot formatting</td>
<td>yes</td>
<td>indirectly</td>
</tr>
<tr>
<td>Slow fading model</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>Path loss model</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>Mobility model</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>Interference</td>
<td>Modelled as AWGN</td>
<td>Model practical situation</td>
</tr>
<tr>
<td>System load</td>
<td>Single user</td>
<td>Multi-user, multi-service</td>
</tr>
<tr>
<td>Cell configuration</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>User location</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>Actual traffic</td>
<td>no</td>
<td>yes – use actual video traffic</td>
</tr>
<tr>
<td>Power control</td>
<td>Fast power control</td>
<td>yes</td>
</tr>
<tr>
<td>Radio resource management</td>
<td>no</td>
<td>yes</td>
</tr>
<tr>
<td>Time resolution</td>
<td>Chip level with 3.84 Mcps</td>
<td>TTI level – 10-40 ms</td>
</tr>
</tbody>
</table>
Because the system level simulator is developed at the TTI level, one sample point value (average SNR value) is generated for each TTI in the system level. Rapid variation of radio link quality within each TTI is generated using a link level simulator for a corresponding SNR or Eb/No value provided by the system level simulator. This method captures both system and link level quality variations, and provides accurate modelling of receiver performances with significantly reduced computational complexity. The simulation tasks for link and system level simulators is summarised in Table 8.2.

8.4 System level quality enhancement techniques

System level quality enhancement techniques are designed to improve system performances as a whole. System performances are often measured in terms of:

- Total system throughput
- Average delay experienced by users
- Number of satisfied users in the system
- Increase in system coverage.

Limited radio resources should be efficiently allocated between users in order to achieve maximum company profit. In second generation communication systems, the total system capacity (in terms of number of channels) is fixed by the system planning procedure and does not vary with the current system load. In contrast, third generation communication systems are interference-limited and the system capacity is defined by the current system load. Therefore, radio resource management in third generation systems is significantly more complex than in second generation systems.

A variety of radio resource management schemes have been proposed in the literature for CDMA based communication systems. Two important concepts followed in efficient radio resource management schemes are:

- Adaptive power control
- Adaptive rate allocation.

The effect of dynamic spreading control on multiple access interference, spectral efficiency and the quality of service experienced by users in multiple-service DS-CDMA system is investigated in [OH-99]. The relationships between the optimal retransmission probability, system throughput, multiple access interference, quality of service and radio resource allocation were derived. A combined adaptive power control and forward error correction coding approach for scheduling in
Chapter 8. System Level Quality Enhancement Techniques

CDMA downlinks is presented in [LU-99]. The overall optimisation is divided into three hierarchical levels: system, cell and user levels. The goal is to optimise the system utilisation, while satisfying the user’s quality requirement. Dynamic transmitted power and processing gain allocation for efficient resource allocation for multiple class packet data in CDMA systems is proposed in [KIM-00]. Here, the relationship between average packet delay and processing gain is derived for both a single class of data users and for two classes of data and voice users assuming the use of ARQ with forward error corrections.

Another joint rate and power allocation scheme is proposed in [KIMb-03] for resource allocation in a WCDMA system. The optimality criterion used is the maximisation of sum-rate capacity in terms of the average number of radio link level frames transmitted per adaptation interval under the constraint of SIR and power limits in the base station transmitter. Furthermore, performances have been investigated in the presence of user mobility and correlated slow fading for both the uniform and non-uniform traffic load scenarios. Improved MAC protocols which provide flexible support for real-time/non-real-time variable/constant bit rate services, while maximising the system utilisation in WCDMA based communication systems are proposed and analysed in [FANT-00, HANG-03]. Efficient call admission control strategies and scheduling algorithms for multiple service CDMA systems can be found in [ROME-02, JEON-02]. A mathematical derivation of system capacity and required transmission power of WCDMA systems based on downlink pole equation is analysed in [SIPI-00]. A detailed mathematical formulation regarding adaptive rate and power for system throughput maximisation, optimum admission procedure and service quality for multi-class CDMA systems can be found in [KIM-99, JAFA-03, CHO-01]. More references for adaptive spreading gain control for system level performance enhancement are given in Section 7.5.

As mentioned earlier, conventionally, system level performances are evaluated for cell/system planning and system budget evaluation. Only the statistical or average values of link level performances have been considered in these calculations. The goal of the research work described in this thesis is the evaluation and enhancement of multimedia quality received by end users of the system. Therefore, in contrast to the system level research referenced above, the overall system performances are investigated for end user received quality. In other words, the evaluation criterion is based on the actual video quality received by the end users, measured in terms of average Peak Signal to Noise Ratio (PSNR). Two adaptive resource allocation schemes for video communications are investigated. The first scheme uses adaptive transmit power control, while the second scheme is based on combined adaptive spreading gain control and power control.
8.4.1 Reference resource allocation scheme

A resource allocation scheme, without adaptive power control or adaptive spreading code allocation, is implemented as a reference scheme for performance comparison purposes. Speech channels use the maximum allowable power, ensuring the highest priority for speech users. In addition, the common control channels are transmitted with maximum power to cover the whole cell area. Video users are classified into separate classes according to their required transmission rate. Users in the same class are allocated equal transmit power levels. Let $P_M, P_s$ and $P_c$ denote maximum base station transmit power, total power on speech channels, and total power on common control channels respectively. Assuming that video users require the same quality of service, the allocated transmit power on a video channel, $P_v$, is calculated as

$$P_v = \begin{cases} \frac{P_M - P_s - P_c}{N} & \text{if } \frac{P_M - P_s - P_c}{P_{\text{max}}} < P_{\text{max}} \\ \frac{N}{P_{\text{max}}} & \text{otherwise} \end{cases} \quad \text{Equation 8.6}$$

where $N$ equals the number of video users in the system and $P_{\text{max}}$ is the maximum carrier transmit power. The equation satisfies the requirement on the total base station transmit power and maximum allowable transmit power per channel.

8.4.2 Adaptive power control scheme

This section describes the development of an adaptive power allocation scheme. The design goal is to maximise the number of satisfied users in the system, for given user quality requirements. Thus, the power allocation algorithm is designed to maximise the total SNR received by users in the system, while satisfying the constraints on total base station power and maximum carrier transmit power. The algorithm is divided into three main steps. In the first step, the initial carrier transmit power is decided. Then the required transmit power setting is computed for each user based on their individual quality requirements and channel conditions experienced. Finally, the allocated carrier power is modified considering the constraints on system load, base station total transmit power and total received SNR by users in the system. The proposed adaptive power control scheme is illustrated in Figure 8.2 and Figure 8.3.
The initial transmit power allocation algorithm starts with the setting of equal transmit power on video channels and maximum transmit power on speech and common control channels. Using Equation 8.3, the received SNR for each user is calculated. Then, available transmit power is fairly distributed among video users based on individual received SNR. For example, assume that four video users are in the system. Their received SNR with equal transmit power allocation are represented by $SNR'_{1,v}, SNR'_{2,v}, SNR'_{3,v}$ and $SNR'_{4,v}$. Say $SNR'_{1,v} < SNR'_{3,v} < SNR'_{4,v} < SNR'_{2,v}$.

Then transmit power is calculated as

\[ P_{1,v} = P_{T,v} \frac{SNR'_{2,v}}{\sum_{i=1}^{4} SNR'_{i,v}} \]  
\[ P_{2,v} = P_{T,v} \frac{SNR'_{1,v}}{\sum_{i=1}^{4} SNR'_{i,v}} \]  
\[ P_{3,v} = P_{T,v} \frac{SNR'_{4,v}}{\sum_{i=1}^{4} SNR'_{i,v}} \]  
\[ P_{4,v} = P_{T,v} \frac{SNR'_{3,v}}{\sum_{i=1}^{4} SNR'_{i,v}} \]

where $P_{T,v}$ is available total transmit power for video users. Then allocated transmit power, $P_{i,v}$, on each is given by

Figure 8.2: Initial transmit power setting for video users.
where $P_{\text{min}}$ and $P_{\text{max}}$ denote minimum and maximum carrier transmit power respectively.

In this power setting, users with poor quality channels are assigned higher transmit powers, while compensating powers on users with good channels. Note that speech and common control channels are allocated maximum initial transmit power.

The proposed adaptive power allocation scheme is illustrated in Figure 8.3. It operates at TTI level. Initially, carrier power, $P_{i,v}$, is assumed to be equal to the transmit power on the previous TTI. Based on these power settings, the channel quality, $\text{SNR}_{i,v}$, received by each user is estimated. Also the summation of $\text{SNR}_{i,v}$'s, $S'$ is computed. If the estimated channel quality, $\text{SNR}_{i,v}$, is less than the required channel quality, $\text{SNR}_{\text{req}}$, for the service, then transmit power is incremented by one power control step, $P_{\text{step}}$. Otherwise, transmit power is reduced by one power control step. Before the increment and reduction of transmit power, the requirement for aggregate power, $P_{i,v,\text{agre}}$, and conditions for the maximum transmit power, $P_{\text{max}}$ and minimum transmit power, $P_{\text{min}}$ are checked. The aggregate power determines the maximum number of power increments or reduction in 10 consecutive steps and is described in detail in Section 3.5.1.1. After estimating the required power on each channel, the requirement on total base station transmit power is checked. If the computed total base station power, $P$, is larger than the maximum transmit power setting, $P_{T}$, then the transmit powers on high quality video channels are reduced. When the constraints on the total base station power are satisfied, the channel quality, $\text{SNR}_{i,v}^{\text{w}}$, received by the user for the current power setting is estimated. The summation of $\text{SNR}_{i,v}^{\text{w}}$ values, $S''(t)$, is also computed. This value is compared with the total estimated channel qualities, $S'(t)$, which could have been achieved if no power adaptation is applied for the current TTI. In the case where the $S'(t)$ is larger than the $S''(t)$, then the powers on high power video channels are reduced until the local optimal $S''(t)$ is attained. Finally, the video data is transmitted with the computed transmit power, $P_{i,v}$. Note that transmit power on control channels is kept constant, while transmit powers on speech channels are adjusted based on individual quality requirements and channel quality. No further power adjustment is carried out on speech channels based on total received system SNR. This ensures high priority for speech users in the system.
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Set
\[ P_{i,v}(t) = P_{i,v}(t-1) \]

Calculate \[ SNR_{i,v}^i(t) \ & S'(t) = \sum SNR_{i,v}^i(t) \]

If \[ SNR_{i,v}^i(t) < SNR_{req} \]

\[ P_{i,v}(t) < P_{max} - P_{step} \]
\[ P_{i,v}(t) = P_{max} \]

\[ P_{i,v}(t) > P_{max} + P_{step} \]
\[ P_{i,v}(t) = P_{min} \]
\[ P_{i,v}(t) = P_{i,v}(t) - P_{step} \]
\[ P_{i,v}(t) = P_{i,v}(t) + P_{step} \]

Calculate \[ P = \sum P_{i,v}(t) \]

If \[ P < P_T \]

Calculate \[ SNR_{i,v}^* = \sum SNR_{i,v}^* \]

If \[ S'(t) < S'(t) \]

Reduce \[ P_{i,v}(t) \] on high power video channel

Allocate \[ P_{i,v}(t) \] for the transmission

Calculate \[ P = \sum P_{i,v}(t) \]

Reduce power on high quality video channel

Note: \( P_{agre2} \) and \( P_{agre2} \) represent the maximum and minimum aggregate power respectively. Other terms used in the diagram are defined in the previous paragraph.

Figure 8.3: Adaptive power control scheme.
Figure 8.4: Performance of adaptive power control scheme.

Figure 8.4 shows the total received SNR in a 14 user system. The power control parameter values used in the simulation are listed in Table 8.3. The parameter values chosen are the recommended values to be used in the system level simulation by 3GPP [UMTS 30.06]. The quality requirement for the video application is set according to the experimental results obtained in the link level simulation (see Chapter 4). Video source rate is set to 64 kbps, while a spreading factor 32 and 1/3 rate convolutional channel code is used. As can be seen in the figure, the total base station transmit power fluctuates slightly during the transmission interval. However, most of the time, the base station operates at full transmission power. The total received SNR is computed for both employment of the reference scheme, where power is distributed equally between video users and the proposed adaptive power allocation scheme. The experimental results show that the received total SNR with the adaptive power allocation scheme is higher than that resulting from equal power allocation.

The developed adaptive power allocation scheme uses the actual SNR of the communication channel as the channel quality measurement. Actual channel SNR is not available in practical systems. Therefore, the channel quality should be estimated based on the measurable quantities. A number of channel quality estimation techniques have been proposed in the literature. Some of these techniques use BLER and measurements on received signal strength in estimating channel conditions. Others use channel prediction in determining the actual channel quality. The Kalman
filter method [LEUN-01, ANDE-03] is often used in interference prediction and has been shown to provide significant performance improvement in power control algorithms.

<table>
<thead>
<tr>
<th>parameter</th>
<th>value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum base station transmit power</td>
<td>43 dBm (20W)</td>
</tr>
<tr>
<td>Maximum carrier power</td>
<td>30 dBm (1W)</td>
</tr>
<tr>
<td>Minimum carrier power</td>
<td>20 dBm</td>
</tr>
<tr>
<td>Power on broadcast channels</td>
<td>35 dBm</td>
</tr>
<tr>
<td>Power step size</td>
<td>1 dB</td>
</tr>
<tr>
<td>Aggregate power limit</td>
<td>12 dB</td>
</tr>
<tr>
<td>Number of speech users</td>
<td>6</td>
</tr>
<tr>
<td>Quality requirements in Eb/No</td>
<td></td>
</tr>
<tr>
<td>Video at 64 kbps</td>
<td>5.5 dB</td>
</tr>
<tr>
<td>Video at 138 kbps</td>
<td>5.7 dB</td>
</tr>
<tr>
<td>Speech at 8 kbps</td>
<td>5.0 dB</td>
</tr>
</tbody>
</table>

### 8.4.3 Adaptive spreading code allocation scheme

The adaptive spreading code allocation scheme, described in Section 7.5.1 for a single user environment, is modified here for multi-user scenarios. The number of available orthogonal codes is limited in the downlink of a UMTS system. Therefore, the constraint on the number of available codes should be taken into account in the design of adaptive code allocation techniques. The proposed system level code allocation, which operates at TTI level, is depicted in Figure 8.5. First, each user is assigned a code with a certain spreading factor according to their received channel quality. If the number of assigned codes falls within the system code budget, then each user is allocated that spreading code for the current transmission. Otherwise, the transmission code for the user with the minimum channel SNR within the spreading factor 16 group, is changed to a higher level (spreading factor 32 in this case). Both transmission modes used in the adaptive spreading gain control algorithm (see Section 7.5.1) show similar video quality near the channel switching threshold. Therefore, a reduction in transmission rate for a user experiencing a channel condition close to the switching threshold would minimise the (system) performance degradation in a code limited multi-user scenario.
Chapter 8. System Level Quality Enhancement Techniques

Figure 8.5: Adaptive spreading gain control scheme.

The adaptive spreading code allocation algorithm is applied in conjunction with the adaptive power control algorithm. Note that the operations of both algorithms are based on the received channel SNR at the terminal. The adaptive spreading gain control scheme maintains constant transmit power by varying the bit energy allocated in each transmission mode. Therefore, it does not affect the operation of adaptive power control.

The implementation of the proposed adaptive spreading code allocation scheme assumes perfect code assignment and re-assignment in a system. However, in an active system with a large number of users, it is difficult to keep track of the allocated code for each user. This causes a so-called "code blocking condition" in UMTS systems. There has been a lot of research conducted in efficient code assignment/re-assignment, which enhances statistical multiplexing gain and spectral efficiency while avoiding code blocking in variable rate multi-services UMTS systems. More information regarding efficient code assignment and re-assignment can be found in [MINN-00, AMIC-02, ASSA-02, ROUS-02, PARK-02, FOSS-02, TSEN-01].
8.5 System performances

8.5.1 Experimental setup

The number of voice users in the system is assumed to be 6 users per sector and is kept constant for the simulation duration. Six spreading codes with a spreading factor of 128 are assigned for voice users. The actual voice traffic is not modelled. However, the presence of voice users and their effect on the system load (and the influences on the performance of video users), is simulated via the transport channels, which are assigned to be used for voice traffic. Similarly, 5 channels with a spreading factor of 256 are assigned for control channels. The total transmit power on these control channels is kept constant at 3.2 dB following the recommendations given in [UMTS 30.06]. The number of video users permitted in each sector is varied from 1 to 40.

The goal of this work is the investigation of the received video quality by end users and the video quality enhancement achieved by adaptive resource allocation in a multi-user environment. Therefore, it is necessary to use actual video traffic in the simulation. Fully error resilience enabled MPEG-4 coded ITU test sequences “Suzie” and “Foreman” are used in the experiments. The above mentioned code allocations on voice and broadcasting channels, permits a maximum number of 14 channels for video transmission with 138 kbps source rate. A maximum of 29 channels are available for video application with 64 kbps transmission rate.

The system performances are evaluated based on the video performances of users in the middle cell marked as “A” in Figure 8.1. In the intra-cell interference calculation, the first set of interferer and some of the second set of interferes are considered. Neighbouring cells are assumed to be operating at full system load. Therefore, the results shown below illustrate the performances seen in a worst case scenario.

Initially, users are randomly placed in the cell. Then the system interference and the received channel quality, in terms of SNR for each user, are calculated. Selected video sequences are transmitted over the simulated channels. In order to consider the effects of fast fading, the simulation is repeated 3-5 times for each SNR profile. The user location and shadow fading also influence the end user’s received quality. The effect of the geographical location and propagation condition is fully captured by repeating the above described experiments for 4 to 50 times.
depending on the simulation computation required. This means that each of the points shown in
the plots, described below, is computed from at least 150 different simulation runs.

![Figure 8.6: Average video performances in power limited system.](image)

Video quality is measured in terms of average frame PSNR for each user. The received quality is
averaged over a number of simulation runs and the results are presented in Figure 8.6 for the
“Suzie” sequence. No code limitation is assumed. As expected, average received quality decreases
with increases in the number of users in the system. The reference scheme with a spreading factor
of 16 shows the worst performances. Spreading factor 32 realisation performs better than
spreading factor 16 realisation. Adaptive power control enhances the performances by 2-3 dB.
Better performance improvement with adaptive power control is visible in systems with large
numbers of users compared to those with fewer users. This is due to efficient channel
multiplexing among large numbers of transmission channels. Adaptive spreading gain control
further enhances the performance by 1-2 dB, and provides the best overall performance.

The average performances of algorithms in a code limited system for the transmission of “Suzie” and
“Foreman” sequences are illustrated in

Figure 8.7. For the reasons detailed above, the maximum number of supportable users with a
spreading factor of 16 is limited to 14, while is limited to 29 for spreading factor 32 operation.
Compared to the non-code limited scenario, the gain achieved by adaptive spreading gain control
in the code limited scenario is significantly reduced if large numbers of users are present in the
system. Similar performances are seen for transmission of the “Foreman” sequence. However, the
average performance is about 2-3 dB lower compared to that of “Suzie”. This is because of the high motion scene characteristics of “Foreman” sequence.

Figure 8.7: Average video performances in power/code limited multi user system; a) Suzie b) Foreman.
Figure 8.8: User satisfaction in power/code limited multi user system; a) Suzie b) Foreman.

Figure 8.8 shows the system performances in terms of user satisfaction for transmission of "Suzie" and "Foreman" sequences over the simulated time varying propagation channels. Both system capacity limitations (code limit and power limit), are considered. The users are considered to be satisfied if the average received frame PSNR value is greater than 20 dB. The reason for the selection of 20 dB frame quality threshold in assessing video performances is discussed in Section 2.3.6. The percentage of satisfied users in the system closely follows the average performance figures shown above. Without adaptive power control and spreading gain allocation, video communication can be supported only for one video user in the system with 90% satisfaction for
the transmission of the "Suzie" sequence. For the high motion "Foreman" sequence the maximum user satisfaction without adaptive power control and spreading gain control is 80%. The application of adaptive power control and spreading gain allocation increases the number of supported users (with 90% of satisfaction) to 7 and 5 for transmission of Suzie sequence and Foreman sequence respectively. User satisfaction decreases with an increase in the number of users in the system. In a system with 15 video users, only 50 -60% of them are guaranteed to be satisfied with the received video quality. As illustrated in Figure 8.8, lower user satisfaction is expected for high motion video sequences compared to that of low motion video sequences.

The distribution of quality received by video users in the system is demonstrated in Figure 8.9 for the transmission of “Suzie” and “Foreman” sequences respectively. The figure presents the Cumulative Distribution Function (CDF) of received video quality in terms of average frame PSNR for the deployment of 14 concurrent video users in the system. Adaptive schemes tend to increase the received video quality of users who experience poor channel conditions. The number of high quality channels or number of users who receive high quality video (with PSNR value above 30 dB) remains fairly constant across the resource allocation schemes.
8.6 Conclusion

This chapter analysed the video performances in a multi-user downlink UMTS system. The performance investigation is carried out for the transmission of fully error resilience enabled MPEG-4 coded video over a simulated multi-user UMTS system. The developed multi-user UMTS simulator combines a system level simulator and the link level simulator developed in Chapter 3. In contrast to conventional system level simulator design, which incorporates average link level performance figures, the developed simulator uses instantaneous link quality. Therefore, it provides more accurate modelling of receiver performances with significantly reduced computational complexity.

System level video performances are investigated under three different resource allocation schemes. Scheme 1 allocates equal transmit power for video users, who require the same quality of service. The allocated power is kept constant throughout the transmission duration. Scheme 2
adapts the allocated transmit power of each user according to the individual received channel qualities. Further power adjustment is conducted to maximise the overall system performances based on the summation of the received channel SNR by individual users in the system. In Scheme 3, an adaptive spreading gain control scheme, which is designed to maximise the received video quality by individual users in the system is applied in conjunction with Scheme 2.

Video performances are measured by averaging the frame PSNR received by individual users. Based on individual user quality, the system performances are evaluated in terms of mean received video quality and the number of satisfied users in the system. The performances are investigated under constraints on the base station total transmit power (power limitation) and the code availability per carrier (code limitation) in the system simulation. Results show that better video quality can be achieved using adaptive resource allocation schemes compared to fixed resource allocation. Scheme 2 results in about 2-3 dB quality improvement in average frame PSNR over Scheme 1. Scheme 3 further enhances the average performances by 1-2 dB. However, the performance gain achieved by Scheme 3 is significantly reduced if large numbers of users are present in the system. Scheme 1 only supports one video user with 90% satisfaction in the deployment scenario considered. This number is increased to 5-7 video users with 90% satisfaction with the application of adaptive power and spreading gain control.
Chapter 9

9 Conclusion

9.1 Overview

Mobile and multimedia communication technologies have experienced rapid growth and commercial success during the last decade. Driven by the powerful vision of being able to communicate from anywhere at any time with any type of data, the integration of multimedia and mobile technologies is currently under-way. Third generation communication systems will support a wide range of communication services for mobile users from any geographical location, in a variety of formats such as voice, data, images and video. Among those, video communication is particularly demanding due to the stringent requirements on quality of service and the enormous amounts of data involved. This thesis has examined and investigated resource allocation and quality enhancement methods for the transmission of video services over third generation radio access networks.

Radio resource management algorithms in UMTS are designed to utilise air interface resources in such a way as to guarantee required service quality, while maintaining high capacity over the specified coverage area. They can adjust link level parameters such as spreading factor, transmit power, and channel coding for each radio bearer according to the varying traffic, system interference and channel conditions. Conventionally, these parameters are adapted based on the estimated received channel quality at the receiver. Research carried out in this thesis showed that optimal resource allocation for video applications requires adaptive resource allocation based on
the actual received video quality, rather than the average received channel quality. Furthermore, combined adaptive source, network, and system parameter optimisation is essential in guaranteeing service quality, service fairness and service flexibility, for supporting next generation multimedia applications, while maximising the capacity of the system.

The capabilities of UMTS and suitability of different radio bearer configurations for supporting real-time video communications are shown in Chapter 4. A bearer configuration with a spreading factor of 8 is clearly unsuitable for video applications due to high channel induced errors and high transmission power requirements. The lower bound of $E_b/N_0$, which could support acceptable video quality, is found to be 8.5 to 10 dB for a spreading factor of 8 with a 1/3 rate convolutional code channels in Vehicular and Pedestrian environments. This is not achievable in a multi-user system, unless very sophisticated diversity techniques and interference suppression techniques are employed. The minimum $E_b/N_0$ requirement was reduced down to 6.3-8.0 dB for spreading factor 32 and 16 channels. Fast power control can lower the minimum $E_b/N_0$ requirements further by 0.8-3 dB in tested operating environments. Perceptual quality measurements show that spreading factor 32 outperforms others for video transmission over similar operating conditions. However, this limits the operating source rate to 64 and 97 kbps with 1/3 and 1/2 rate channel coding respectively. System level performances demonstrate the necessity of advanced antenna techniques, adaptive resource allocation and combined source and channel quality enhancement techniques in achieving 95% user satisfaction for real-time video services even for source rates as low as 64 kbps.

In addition to the investigation of capabilities of UTRAN for the transmission of video, perceptual quality enhancement methods, which exploit the adaptive source and network parameters for varying transmission conditions, are explored in the work carried out in this thesis. The most efficient way to use limited radio resources is the deployment of class or priority based channel allocation. In this way the most important information is transmitted over a high priority channel, while a low priority channel is used to transport low priority data. The syntax of the video format and the video representation can be used in information prioritisation. Data partition formatting and object based video coding provide simple methods of data separation in multi-priority transmission. However, a perceptual quality estimation based data separation algorithm provides an adaptive and flexible information prioritisation, mechanism which results in optimal video performance. The priority of the transmission channel can in fact be based on the channel protection, in terms of channel coding rate, modulation, spreading factor and transmission power allocation. A combination of these channel parameters provides optimal prioritised transmission.
In addition, link adaptation, where source and network parameters are adjusted according to the time varying channel condition at the receiver, is necessary in maintaining the perceptual video quality received by the end user. Link adaptation can be designed either to enhance the received video quality for a given network configuration, or to enhance the system performance in terms of system capacity or coverage, while guaranteeing the end users' quality of service requirements. Either way provides improved performances compared to non-adaptive transmission schemes.

9.2 Research achievements

The background work in this thesis is described in Chapter 2 while the original work is set out in Chapters 3-8. The main achievements of the research are summarised below.

1. Real-time PC based UMTS emulator software and a comprehensive study of video transmission over UMTS radio access bearers

A UMTS-FDD forward link physical layer simulation model was designed using the Signal Processing WorkSystem (SPW) software simulation tools. The model includes all the radio configurations, channel coding/decoding, modulation parameters, transmission modelling, and their corresponding data rates for a dedicated physical channel according to the UMTS specifications. The physical link simulator generated bit error patterns, which correspond to various radio bearer configurations. These error patterns were integrated with the UMTS protocol data flow model, designed in Visual C++. This integration was implemented in a real-time emulator so as to allow for interactive monitoring of the effects of network parameter settings upon the received multimedia quality.

Using the developed emulator, a comprehensive study of the network parameters required for the transmission of video over UTRAN, under different channel and interference conditions, was carried out for different source parameter settings. Results showed that the delay requirement for real time communication can be satisfied with careful selection of network parameters for rate controlled MPEG-4 encoded video applications. Received video quality depends upon the source rate as well as the transmission channel condition (error rate). Therefore, an appropriate compromise between video source rate and error correcting capabilities of the channel should be considered in the radio bearer optimisation, for transmission of video over a given channel.
condition. In addition, the performance differences for video communications over packet switched and circuit switched connections are investigated. Experimental results show a 1-3 dB performance loss in terms of average frame PSNR for packet switched operation compared to the operation over circuit switched connection. The performance loss is because of the source throughput loss due to the transport layer overhead, and the information loss due to the corrupted transport layer packet headers.

2. Modelling of rate-distortion characteristics and analysis of optimal MTU settings for efficient wireless video communications

Received video quality is estimated frame-by-frame at the encoder based on a distortion model that accurately models the overall distortion seen in decoder frame reconstruction for given encoder parameters, decoder concealment techniques, and channel statistics. The model is designed for full error resilience enabled MPEG-4 video applications. Overall distortion is modelled as a combination of quantisation distortion and channel distortion. Channel distortion is divided into spatial concealment distortion, temporal concealment distortion and distortion caused by error propagation over predictive frames. The model is validated for a range of channel characteristics and different encoder parameter settings. The results verify the model accuracy for video services over mobile networks.

Video performances over packet switched networks are greatly influenced by the introduction of transport layer packetisation. Source throughput is reduced compared to circuit switched operation due to the transport layer overhead. Moreover, if the packet header is corrupted, the information data encapsulated within the packet should be discarded. This causes the loss of a large amount of data and results in severe spatial concealment distortion. The developed distortion model is modified in such a way that the presence of a packet header is taken into account for video communications over packet switched networks. The model is used in deriving the optimal Maximum Transmission Unit (MTU) for wireless video communications.

3. Unequal Error Protection (UEP) and Unequal Power Allocation (UPA) schemes for video communication
An Unequal Error protection scheme, which is based on the MPEG-4 data partition format, was designed for video transmission over UTRAN. The video information within a frame is split into two streams. The data belonging to the first partition forms stream 1 and is highly protected using a 1/3 rate convolutional code. The second partition data is placed in stream 2, and uses less reliable channel protection, a 1/2 rate convolutional code. Channel coded data in the two streams is multiplexed onto the same physical channel for transmission over the wireless link. Extra information is added to the second stream to guarantee accurate stream synchronisation. This method is simple to implement, as it requires little modification to the encoder data format and no modification to the network protocols. It shows a performance improvement of 1-2 dB in terms of average frame PSNR in moderate channel conditions.

Even though, MPEG-4 data partitioning indicates the relative importance of information within a video packet, it does not reveal the relative importance of data in different video packets. Also, the source rate is fixed by the sequence characteristics in the data partition based UEP scheme. Therefore, a joint-source channel coding (JSCC) scheme, which varies the source rate and channel coding rate according to the channel bit error rate characteristics, can not be combined with data partitioning based UEP techniques. This suggests that the data partition based UEP scheme provides a sub-optimal method. An optimal UEP scheme was designed to achieve more accurate data prioritisation. The proposed method uses the perceived importance of bits, estimated using the developed distortion model. Differential protection is deployed by sending prioritised data streams over multiple radio bearers, with different channel protection capabilities. The proposed scheme is compatible with the codec standard and is also transparent to the underlying network. The proposed scheme combines JSCC and UEP schemes and obtains optimal video quality over a wide range of channel conditions.

In addition to the channel coding schemes, transmission power allocation can be used in achieving differential information protection. This concept is used in the design of an energy efficient network compatible performance enhancement algorithm for video communication over direct-sequence CDMA cellular networks. Here, a UEP scheme is combined with an Unequal Power Allocation (UPA) scheme to achieve optimal performances. Prioritised video information is transmitted over two different transport channels with different error protection capabilities. Transmit energy for each bearer is selected to maximise the expected frame quality, which is estimated using the developed rate-distortion model, for an increment in transmit power. The experiment carried out over the simulated UMTS system, shows that minimum transmission
energy is needed to achieve a target video quality with the proposed joint UEP-UAP scheme, compared to the traditional UEP schemes, and non-UEP schemes.

4. Link adaptation techniques for transmission of video over EDGE and UMTS networks

Link adaptation techniques, which adjust source rate, channel modulation, and channel coding schemes in response to the instantaneous channel and interference conditions, are designed to exploit the time varying nature of the mobile radio channel. Two link adaptation algorithms are implemented. One is designed to maximise the received video quality, and is named quality-based adaptation scheme. The other is designed to maximise the system throughput, while guaranteeing a required video quality. This is called throughput based adaptation scheme. Source and network parameter adaptation is conducted according to the estimated instantaneous channel quality, based on measured channel Block Error Rate, Received Signal Strength (RSS), and the first order statistics of RSS. Algorithm performances are demonstrated for video telephony over an EGPRS network. The adaptive schemes showed better performances than those of non-adaptive schemes. When offered similar traffic load and channel environments, the quality based adaptation scheme outperforms the throughput based scheme. Furthermore, the investigation shows that the proposed algorithms are robust against feedback delay, noisy feedback, and burst channel errors.

The above described link adaptation algorithms are not suitable for video streaming applications as they require separate encoding of video sequences for each user. To overcome this problem, the video sequence is encoded at different output bit rates, which are stored in a buffer at the server. Link adaptation is performed by switching between pre-encoded streams, according to the instantaneous channel condition. When switching is performed between streams, there is a mismatch between the encoder and decoder in terms of the reference frames used to predict future frames. This can potentially lead to drift. However, the proposed scheme uses appropriate selection of intra coded blocks (AIR technique) in video frames to limit the drift effects, and prevents it becoming a detectable problem. Experiments, which were performed using EGPRS channel models, show significant quality improvements for the adaptive scheme, compared with the results for fixed modulation-coding schemes.

Spreading gain provides the key variable in determining user data rates and associated channel quality in CDMA based communication systems. Therefore, in addition to the channel coding schemes, adaptive spreading gain can be used to exploit time varying channels in CDMA systems.
Link adaptation techniques, based on adaptive spreading gain control, are proposed and analysed for real-time video communications in UMTS networks. Source rate and spreading code levels were varied depending on the state of the transmission channel. Adaptation is based on the actual channel signal to interference ratio, which is calculated at every Transmission Time Interval. The transmission power is kept at a constant level and transmission bit energy is adjusted according to the selected spreading code level. That ensures that the interference power experienced by other users does not affect the adaptive spreading gain control techniques. The conducted experiments show 2-3 dB frame PSNR improvement compared to non-adaptive schemes. Further performance improvements are achieved by the combined application of adaptive spreading gain control, and the unequal error protection scheme.

5. System level perceptual quality enhancement techniques for video communication in multi-user scenarios

A joint system-link UMTS simulator, which combines a system level simulator, and the developed link level simulator, is designed for the analysis of video performances in a multi-user downlink UMTS system. Video performances are investigated for the transmission of fully error resilience enabled MPEG-4 coded video under constraints on the base station total transmit power (power budget), and the number of codes available per carrier (code budget). Video performance is measured in terms of the average frame PSNR of received video by individual users. System performances are shown in terms of the mean video quality, which is the average video quality received by users, and the number of satisfied users in the system.

Three different resource allocation schemes are implemented. Scheme 1 is a non-adaptive scheme, and it allocates equal transmit power for video users, who require the same quality of service. Scheme 2 adapts the allocated transmit power for each user according to their individual received channel quality. Scheme 3 combines an adaptive spreading gain control scheme with scheme 2 to maximise the video quality received by individual users in the system. Experiments conducted over the simulated system show about 2-3 dB quality improvement in terms of average PSNR with adaptive power allocation (Scheme 2), compared to fixed resource allocation. Further performance enhancement, in terms of average quality and user satisfaction, is achieved with Scheme 3.
9.3 Areas for future research

This section describes some of the issues, which remain to be tackled in the provision of wireless multimedia, and outlines the author's view of the future of mobile multimedia communications.

The more immediate research activities can be carried out with the link level quality enhancement techniques described in Chapter 6. Here, a novel optimal unequal error protection algorithm with data prioritisation based on the estimated perceptual quality is proposed. The algorithm performances are evaluated for video transmission using two radio bearers. This can be extended to any number of radio bearers, depending on the priority levels required for the service and the network compatibility. Furthermore, the algorithm is implemented only for simple frame level video representation. However, it can be implemented in conjunction with video segmentation, (object based or shape coding representation) and thus achieve improved transmission efficiency.

In Chapter 3, the data flow model of the UMTS protocol stack was implemented following a modular design strategy. This allows fast and easy implementation of the optimised algorithm within each protocol layer. Only the protocol layer effect on multimedia performance is considered in the implementation. Even though the presence of protocol layer headers, and their effects on application performances were emulated with the allocated dummy headers, no actual layer functions are implemented. Further model improvement can be achieved with the implementation of actual layer functions. In particular, the implementation of header compression algorithms, such as Compressing RTP/UDP/IP (CRTP) and ROBust Checksum-based header Compression (ROCCO) in the PDCP layer, would provide a test platform for efficient multimedia communication over all-IP based infrastructure.

The work carried out in this thesis used a layered protocol architecture for multimedia delivery. No protocol layer interaction and cross protocol optimisation is considered. Data within each layer was transparent to the other layers. In other words, the lower layers do not have any knowledge about the applications and the delivered bit streams are treated in the same way for all services. This will reduce the efficiency of the communication system. Transmission buffer management at the RLC layer for variable bit rate video application is a good example. Current buffer management works in a first-come-first-served basis, and if the buffer is full, the incoming data is discarded, regardless of its importance to the visualisation of video at the decoder. If the RLC layer is informed about the importance of the incoming data (in other words layer interaction
between the application layer and the RLC layer), some of the low priority data, already within the buffer, can be discarded, freeing buffer space for more important data. Also, the CRC attachment can be controlled in an efficient manner if the service characteristics are known at the RLC layer.

Another area of research is combined channel source decoding. More efficient source decoding is possible if soft information about channel characteristics is available at the decoder. In particular, the decoding of variable length codes shows significant improvement with prior knowledge of the probabilistic characteristics of the channel. Currently, channel characteristics are used only in channel decoding, and no soft information is passed on to the application layer. As in joint-source-channel encoding, the combined channel source decoding approach will lead to better source decoding than is available today.

Chapter 7 examines the issues regarding the design of link adaptation algorithms for video telephony and streaming video applications. New application level quality enhancement methods are proposed and analysed. The designed algorithm for streaming video exploits the time varying nature of the transmission channel by adjusting the network parameters. The pre-encoded stream switching is used as an alternative to encoder parameter adjustment. A backward-error correction scheme, which can be used in enhancing received quality for delay tolerant streaming applications is not implemented. Therefore, further research needs to be carried out, particularly in investigating the performances of combined link adaptation and the backward-error correcting techniques, such as Automatic Repeat reQuest (ARQ), for video streaming applications.

The importance of efficient resource allocation in wireless multimedia communication is demonstrated in Chapter 8. The resource allocation schemes implemented are designed to optimise the perceptual video quality. The allocated resources for each user were adapted based on the actual channel conditions experienced by the user. In a practical system, the channel conditions should be estimated from the channel measurements taken at the physical layer, such as received signal power, bit error rate, and interference power. Thus, more accurate system performance evaluation should be conducted for the implementation of channel estimation in conjunction with the proposed resource allocation scheme.

Most of the research carried out to date on system resource allocation employs resource adaptation based on the direct or indirect indication of the channel conditions experienced by the
user. Even though the resource allocation schemes proposed in Chapter 8 are intended to maximise the perceived quality, it uses a channel condition based resource adaptation. The received video quality depends on many factors such as sequence characteristics, error resilience, source rate, channel protection and transmission bit energy as explained in detail in Section 2. Therefore, the received channel condition does not always reflect the quality of received video accurately. More appropriate, resource allocation scheme should in fact be designed based on the estimated video quality for video communication. The distortion model developed in Chapter 5 can be used to estimate the quality of the reconstructed video at the decoder. This model can be used at the system level (Radio Resource Controller) to achieve efficient multimedia resource allocation. Furthermore, the incorporation of link level quality enhancement techniques, developed in Chapter 6, and system level resource allocation will result in improved system performances.

The resource allocation and quality enhancement techniques investigated in this thesis were based on video telephony services. Only video representation and error resilience algorithms specified under the MPEG-4 simple profile and QCIF video format, have been considered in this work. This is because initially only simple video services will be offered to a small handset. However, future multimedia services are expected to be more complex than viewing transmitted video on a smaller screen. These services may require viewing of multiple streams, content manipulation, visualisation of 3-D images, and reconstruction of virtual images. Radio resource management and quality enhancement techniques should in fact be designed according to the application requirement for efficient multimedia communications. Even though characteristics of future multimedia services are not yet fully defined, research activities in the area of multimedia coding technologies is advancing towards the fulfilment of rich multimedia services. For example, standardisation activities carried out in the ITU and ISO/IEC moving picture experts groups have recently standardised a new video codec called H.264/AVC for two dimensional video representation. H.264/AVC introduces a number of new coding techniques for achieving high compression performances, and the provision of network friendly video representation for conversational and non-conversational (streaming, broadcasting, and storage) applications. There are many research programs being conducted in 3-D video coding, and also in the representation and synthesis of virtual scenes. Therefore, it is necessary to conduct parallel investigation for efficient resource management and quality enhancement techniques, such as media multiplexing, content adaptation, and application-aware radio bearer optimisation, for emerging multimedia services.
Future multimedia systems will need to offer flexible operation and multiple services using different source codes over a number of different communication platforms. Example scenarios are inter-operability between conventional broadcast, wireless LAN and cellular radio systems. It is necessary to ensure information transfer at the optimum quality according to environment and to enable the decoding of a high data rate source by equipment with limited capability. Bit rate variations, network parameter variation and propagation characteristics arising from communication across differing networks using differing protocols can generate complex interactions, which significantly influence the quality of the received video. Thus, appropriate control of system resources and efficient network parameter mapping is essential in guaranteeing quality of service for multimedia communications over heterogeneous networks. A lot of research effort has already been dedicated to the provisioning of QoS in IP-based multimedia transportation, which is the common transport mechanism used in many of the future communication systems. However, QoS parameter mapping between IP networks and other wireless networks would require much attention.
Appendix A

List of Publication


7. C. Kodikara, S. Worrall, A. Kondoz and A.H. Sadka “combined adaptive spreading gain control and unequal error protection for real-time video communications over WCDMA system” The Fifth International Workshop on Image Analysis for Multimedia Interactive Services (WIAMIS’2004), Lisbon, Portugal, 21-23 April 2004 (Accepted for publication).

Appendix B

**Eb/No and DPCH Ec/Io Calculation**

\[
E_b = \frac{\text{total signal power}}{R_b}
\]

\[
N_o = \frac{2\sigma^2}{R_C \cdot \text{ch} \_os}
\]

Where, \( R_C \) and \( R_b \) are chip rate and channel bit rate respectively. \( \text{ch} \_os \) denotes the channel over sampling factor. Setting \( \text{total signal power} \) to be 1, \( E_b/N_o \) becomes

\[
E_b = \frac{R_C \cdot \text{ch} \_os}{2 \cdot R_b \cdot \sigma^2}
\]

\( I_{or} \) - the received power spectral density of the down link as measured at the UE antenna connector

\( I_{oc} \) - the power spectral density of a band limited white noise source as measured at the UE antenna connector.

\( I_{or} \) - the total transmit power spectral density of the down link at the Node B antenna connector.

\( DPCH \_E_c \) - average energy per chip for DPCH

\[
DPCH \_E_c = \frac{\text{transmit signal power}}{\text{chip rate}}
\]

assume no path loss
\[ E_b = \frac{\text{transmit signal power}}{\text{bit rate}} = \frac{\text{DPCH factor} \cdot I_{\text{or}} \cdot \text{chip rate}}{\text{bit rate}} \]

total noise power = power on OCNS + AGWN
\[ = (\text{OCNS factor} \cdot I_{\text{or}} + I_{\text{oc}}) \cdot \text{chip rate} \]

code orthogonality is assumed.

\[ \text{total noise power} = N_0 \cdot \text{chip rate} \cdot \text{ch os} \]
\[ E_b = \frac{\text{DPCH factor} \cdot I_{\text{or}} \cdot \text{chip rate}^2 \cdot \text{ch os}}{\text{bit rate} \cdot (\text{OCNS factor} \cdot I_{\text{or}} + I_{\text{oc}}) \cdot \text{chip rate}} \]
\[ N_o = \frac{\text{DPCH factor} \cdot I_{\text{or}} \cdot \text{chip rate} \cdot \text{ch os}}{\text{bit rate} \cdot (\text{OCNS factor} \cdot I_{\text{or}} + I_{\text{oc}})} \]
\[ E_b \quad \frac{\text{DPCH } E_c \cdot \text{chip rate} \cdot \text{ch os}}{I_{\text{or}} \cdot \text{bit rate}} \quad \frac{N_o}{\text{bit rate} \cdot (\text{OCNS factor} + I_{\text{oc}} / I_{\text{or}})} \]

as \( I_{\text{oc}} \big/ I_{\text{or}} \gg \text{OCNS factor} \)
\[ E_b = \frac{\text{DPCH } E_c \cdot \text{chip rate} \cdot \text{ch os}}{I_{\text{or}} \cdot \text{bit rate}} \quad \frac{N_o}{\frac{I_{\text{oc}}}{I_{\text{or}}}} \]
Appendix C

UMTS Radio Access Simulator: Results

TC  – Turbo Code
CC  – Convolutional Code
VehA – Vehicular A environment
PedB – Pedestrian B environment
BER – Bit Error Rate
BLER – Block Error Rate
PC  – Power Controlled

TC 1/3 50kmh VehA
Channel coding

Veha sf32 50kmh

Power Control

CC1_2 50kmh VehA PC

CC1/3 50kmh VehA PC
TC1/3 50kmh VehA PC

CC1/2 3kmh PedB PC

CC1/3 3kmh PedB PC
TC1/3 3kmh PedB PC
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