PERFORMANCE ANALYSIS OF A MOBILE ROBOTIC TELE-ULTRASONOGRAPHY SYSTEM OVER 2.5G / 3G COMMUNICATION NETWORKS

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THE THESIS IS BEING SUBMITTED IN PARTIAL FULFILMENT OF THE REQUIREMENTS OF THE UNIVERSITY FOR THE APPROPRIATE DEGREE OF DOCTOR OF PHILOSOHPY

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Submitted: January 2006

Abstract

The concept of merging the state of the art mobile and wireless communication technologies and health care is the research subject of this thesis. The emerging concept represents the evolution of M-health systems from traditional desktop 'telemedicine' platforms to wireless and mobile configurations. Current developments in wireless communications integrated with developments in pervasive and ultrasound monitoring technologies will have a radical impact on future healthcare delivery systems. The work in this thesis formed part of developing an end-to-end mobile robotic tele-ultrasonography system called (OTELO), that brings together the evolution of merging wireless communications and network technologies with the concept of 'connected healthcare' anytime and anywhere. OTELO system allows an Expert to examine a distant patient by remotely and virtually controlling robotic ultrasound probe, that produces ultrasound images transmitted to the Expert side in a real-time environment. The research objectives represent the performance analysis and validation of the system over both 2.5G and 3G networks. Real-time robotic tele-ultrasonography over the mobile networks is a challenging task in terms of reliable, delay sensitive and medically acceptable quality of service.

The approaches made to fulfil the requirements for the functional modalities of the system, were based on the performance matrices of the system on both the simulated and real-network environments. These testing matrices were covering the performance of the wireless path, wired path and the end-to-end connectivity of the system, and can be summarised as the; compression ratio of the transmitted medical ultrasound images, data throughput, Latency, delay Jitter, Round Trip Time and Packet loss. The major part of the study concentrated on the asymmetry nature of the end-to-end data interaction, therefore the Uplink channel characteristics of the Patient station, were under comprehensive investigations on its feasibility for the system medical QoS over the both communication networks (2.5G and 3G). The research tasks were implemented over both simulated environment and on real operating network, and most of the data dealt with were real data acquired from the field. The achieved results were analyzed and furthermore comparative performances between simulated and real network were discussed and justified. The first approach made, was addressing the capability of the GPRS (2.5G) network and its limitations to perform real-time ultrasonography operation. That was an essential subtask investigation towards specific and deeper analysis on the performance of the system over the promising UMTS (3G) network, where the controlled ultrasound data transmission in real-time were investigated and the results thoroughly analyzed. The achieved results analysis of the subtasks mentioned, formed the bases to study the ultrasound transmission objectively, that fulfilling the medical QoS requirements when performing real-time tele-ultrasound medical session, these are precisely the image size, image quality and frame rate. To improve the medical QoS over relatively unreliable environments (wireless), a new adaptation technique for enhanced wireless ultrasound streaming was developed for OTELO environment and the performance results presented.

The results of this research show the successful transmission of robotically acquired medical images and diagnostically acceptable quality medical video streams in 3G wireless network environment. It provides an important and essential knowledge on M-health systems, when close loop robot control, Delay sensitive and Real-time Telemedicine is required. Future work in this area is also presented for enhanced performance of this mobile robotic telemedical system especially for future use in 3.5G and 4G mobile environments.

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Abbreviations

2.5G	Second Generation + (GPRS)
3G	Third Generation
3GPP	3rd Generation Partnership Project
4G	Fourth Generation
A/D	Analogue to Digital Convertor
AAL	ATM Adaptation Layer
ADSL	Asymmetric Digital Subscriber Line
ALF	Application Layer Framing
API	Application Program Interface
ARQ	Automatic Repeat request
ATM	Asynchronous Transfer Mode
BCCH	Broadcast control channel
BDP	Bandwidth Delay Product
BER	Bit Error Rate
BLER	BLock Error Rate
BMC	broadcast/multicast control
BPSK	Binary Phase Shift Keying
BRI	Basic Rate Interface
BS	Base Station
BSC	Base Station Controller
BTS	Base Transceiver Station
BW	Bandwidth
CCCH	Common control channel
CCH	Control channel
CCPCH	Primary and secondary common control physical channels
CDMA	Code Division Multiple Access
CGSN	Combine GPRS Support Node
CIF	Common Intermediate Format
CN	Core Network
cnwd	congestion window
CPICH	Primary and secondary common pilot channels
CR	Compression Ratio
CRE	Centre for Radiologist Education
CS	Coding Scheme
CS	Circuit Switched
CT	Computerised Tomography
CTCH	Common traffic channel
DCCH	Dedicated control channel
DCH	Dedicated Channel
DCT	Discrete Cosine Transform

DICOM	Digital Imaging and Communications in Medicine
DLC	Data Link Control
DoF	Degree of Freedom
DPCCH	dedicated physical control channel
DPCH	Downlink dedicated physical channel
DPDCH	dedicated physical data channel
DSCH	Physical downlink shared channel
DSL	Digital Subscriber Line
DTCH	Dedicated traffic channel
DWT	Discrete Wavelet Transform
EBCOT	Embedded Block Coding with Optimized Truncation
ECG	Echocardiogram
EDR	Echographic Diagnosis Robot
EEG	ElectroEncephaloGram
EIR	Equipment Identify Register
E-UDP	Enhanced-User Datagram Protocol
FBI	feedback information
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FER	Frame Error Ratio
FGS	Fine Granular Scalability
FGS	Fine Granularity Scalability
FoB	Flok of Bird
FPS	frames Per Second
FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GP	General Practitioners
GPRS	General Packet Radio Service
GSM	Global System for Mobile communications
GSN	GPRS Support Node
GUI	Graphical User Interface
H.263	Video CoDec
HLR	Home Location Register
HSDPA	High Speed Downlink Packet Access
HS-DSCH	high-speed downlink shared channel
ICMP	Internet Control Message Protocol
IEEE	Institute of Electrical and Elictronic Engineers
IMT	International Mobile Technology
IP	Internet Protocol
ISDN	Integrated Service Digital Network
ISO	International Organisation for Standardisation
ISP	Internet Service Provider

ITU	International Telecommunication Union
JBIT	Java Based Interface for Telerobotics
JPEG	Joint Photographs Expert Group
JVT	Joint Video Team
LAN	Local Area Network
LEO	Low Earth Orbit
LLC	Logical Link Control Protocol
MAC	medium access control
MAN	metropolitan area network
MANET	mobile ad hoc networks
MB	Mega Bytes
Mbit	Mega Bits
MCM	Multi-Channel Modem
ME	Mobile Equipment
MF-TDMA	Multi-Frequency Time Division Multiple Access
MIDSTEP	Multimedia Interactive DemonStrator Tele-Presence
M-JPEG	Motion JPEG
MOMEDA	Mobile Medical Data
MPEG	Motion Picture Expert Group
MRI	Magnetic Resonance Image
MSC	Mobile-services Switching Centres
MSE	Mean Square Error
MSS	Mobile satellite services
MTU	Maximum Transit Unit
NAK	Negative Acknowledgement
OCCCH	ODMA common control channel
ODCCH	ODMA dedicated control channel
ODTCH	ODMA dedicated traffic channel
OTELO	mObile Tele-Echography using Ultra Light rObot
OVSF	Orthogonal Variable Spreading Factor
PACCH	Packet Associated Control Channel
PACS	Picture Archiving and Communication System
PAGCH	Packet Access Grant Channel
PAN	personal area networking
PCCH	Paging control channel
PCMCIA	Personal Computer Memory Card International Association
PCPCH	physical common packet channel
PCU	Packet Control Unit
PDCH	Packet Data Channel
PDCP	packet data convergence protocol
PDN	Packet Data Network
PDN	Packet Data Network
PDP	Packet Data Protocol
PDP	Packet Data Ptotocol

PDTCH	Packet Data Traffic Channel
PDU	Protocol Data Unit
PGM	Portable Graymap
PLMN	Public Land Mobile Network
PMS	Personalized Medical System
PPCH	Packet Paging Channel
PPP	Point to Point Protocol
PRACH	Packet Random Access Channel
PRACH	physical random access channel
PS	Packet Switched
PSNR	Peak Signal to Noise Ratio
PSTN	Public Service Telephone Network
PTM-SC	Point-to-Multipoint Service Centre
QAM	Quadrature Amplitude Modulation
QCIF	Quadrature Common Intermediate Format
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RC	relative compression
RLC	radio link control
RLC/MAC	Radio Link Control/Medium Access Control
RNC	Radio Network Controller
ROI	Region of Interest
RR	receiver reports
RRC	radio resource control
RTCP	Real-Time Control Protocol
RTO	Retransmission Time Out
RTP	Real Time Protocol
RTSP	Real-Time Streaming Protocol
RTT	Round Trip Time
SCH	Synchronization channel
SDP	Session Description Protocol
SDU	Service Data Unit
SF	Spreading Factor
SGSN	Serving GPRS Support Node
SHCCH	Shared channel control channel
SNDCP	Sub Network Dependent Convergence Protocol
SNR	signal to noise power ratio
SS7	Signalling System 7
STD	Standard Deviation
TCH	Traffic channel
TCP/IP	Transmission Control Protocol / Internet Protocol
TDD	Time Division Duplex
TDMA	Time-Division Multiple-Access
TFCI	transport-format combination indicator

TPC	transmit power-control
TS	Time Slot
TTI	Transmission Time Interval
UDP	User Datagram Protocol
UE	User Equipment
UMTS	Universal Mobile Telecommunication Service
US	Ultrasound
USB	Universal Serial Bus
USIM	UMTS Subscriber Identify Module
UTRA	Universal Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
VBR	Variable Bit Rate
VCEG	Video Coding Experts Group
VLR	Visitors Locator Registers
VoIP	Voice-over-IP
VPBR	Virtual Patient Body Representation
VPN	Virtual Privaté Network
WAN	Wide Area Network
WAP	Wireless Application Protocol
WCDMA	Wide-band Code Division Multiple Access
WiMAX	Worldwide Microwave Interoperability Forum
WLAN	Wireless LAN
WPAN	Wireless PAN

Acknowledgments

I would like to acknowledge all my colleagues and individuals whom contributed to this work.

I owe a special gratitude to my supervisor Prof. Robert S. H. Istepanian. He patiently guided me throughout the entire period of this research and was always helpful and ready to give his advice and support when requested.

I want to thank the following members and researchers of MINT Lab / Kingston University for their kind cooperation and help; Mrs. Nada Philip, Mr. Ala Sungoor, Mr. Nilesh Prag, Mrs. Ying Zuo, Dr. David Wertheim, Mr. Tom Geake and Dr. Henry Wang. Thanks also to the faculty of Computing, Information Systems and Mathematics (CISM) / Kingston University, I should also remember here Ms. Bee Tang and Prof. Tim Ellis.

I am also grateful to our colleagues within the OTELO project consortium from different partners; Dr. Fabien Courreges for his collaboration during his research visit to MINT Lab, also Prof. Pierre Vieyres, Dr. Natali Smith-Guérin, Ms. Lema Al Bassit, Mr. Gerard Poisson, Ms. Cecil Delgorge, Dr. Philip Arbeille, and Mr. Cerile Novales of Laboratoire Vision & Robotique (LVR), Université d'Orléans, France. Furthermore I would like to thank the other individuals of OTELO consortium partners who provided support; Sinters SA, France., Kell s.r.l, Italy., Universite de Tours, France., Brunel University, UK., Elsacom S.p.a, Italy., Corporacio Sanitaria Clinic, Spain., EBIT Sanita S.p.a, Italy., and Informatics and Telematics Institute, Greece.

My gratitude also goes out to Mr. Haris Zisimopoulos of Vodafone Research Group, Newbury, UK, for his great help and providing all the possible networking facilities to perform the experimental works. My thanks to his colleague Dr. Phil Gosset.

I am grateful to Prof. Dariush Nassiri of St. George's Hospital, London, UK, for his great medical advice where we accomplished successfully many robotic clinical demonstrations on OTELO.

My thanks also to Dr. Mosa Ali Abu-Rgheff of Faculty of Technology, University of Plymouth. Thanks to my friends, Mrs. Hanar Atroshi and Dr. Haytham Al Nahi for their help and support throughout this work.

I am also grateful to the European Union for the support and funding of OTELO project (EU IST-2001-32516).

Finally, special thanks to my parents and my wife Zainab as well as my Daughters for being very supportive and patient with me throughout these research years and the nights of studying.

CHAPTER 1

Literature Review on M-health

1.1 Introduction

The recent advances in Mobile and Network technologies will present a paradigm shift in future health care delivery systems. The fusion of the two terms 'mobility' and 'healthcare' within the next decade, will introduce new areas of research and development that will bring new medical applications and services closer to large scale realization that were not possible in the 20^{th} century. Telecommunication technologies have presented themselves as a powerful tool to break the barriers of time and space. With the introduction of high-bandwidth wireless communication technologies, it is possible to deliver simultaneously audio, video, and waveform data to wherever and whenever needed [1, 8].

Since mid-90's telemedicine was forecast to grow substantially and was dubbed as the solution to many healthcare service problems. However, a number of factors have retarded this growth, including technology readiness, legal and regulatory issues and medical acceptance. It is expected that the global population aged over 65 will increase by 88% within the next 25 years. Coupling this with the increasing incidence of medical conditions soon becomes obvious that current health resources and methods of care are very shortly going to be pushed to the limits as stated by [2]. Does the healthcare industry now require telemedicine? This question may lead to a conclusion that e-health and telemedicine are not exactly representing the same meaning, so it is convenient to distinguish between the terms telemedicine and e-health.

Until recently, telemedicine was defined as interactive video systems that linked rural patients with urban specialists. It had required bulky television equipment with cameras and monitors at both ends of a real-time system [3]. Telemedicine itself has been defined as the use of telecommunication to provide diagnostic and therapeutic medical information between patient and doctor without either of them having to travel [4]. However e-Health is broader than telemedicine and can be described as an emerging field in the intersection of medical informatics, public health and business, that enables health services and information to be delivered or enhanced through the Internet and related technologies [5].

Recently these definitions involved in the identification of "e-health" as an umbrella term, with definitions such as "a new term needed to describe the combined use of electronic communication and information technology in the health sector, the use in the health sector of digital data, transmitted, stored and retrieved electronically for clinical, educational and administrative purposes, both at the local site and at distance" [6].

In a further recent editorials [2, 6], a new difference is discovered; telemedicine remains linked to medical professionals, while e-health is driven by non-professionals, namely patients (or in the e-health jargon, consumers) that with their interests drive new services even in the healthcare field mostly for their empowerment through access to information and knowledge.

Therefore, the term e-health will be used in this thesis as an umbrella term to telemedicine. The term 'M-health' is also introduced and will be defined in the following section. The structure of this chapter presents the following introductory topics to the work in this thesis:

- The concepts of the M-health and next generation wireless telemedicine systems are defining and categorising the e-health applications and their significance to the health care, as in section 1.1.
- Sections 1.2 and 1.3 present a general literature review on tele-chography systems and specifically a review on the tele-robotic in medicine and in particular wireless tele-echography systems.
- Sections 1.4 and 1.5 present the scope of the thesis in terms of the follow-up chapter structures. Additionally the contribution of the research to the science and knowledge, especially regarding advanced M-health systems and applications.

1.1.1 M-health and Evolution of Telemedicine

M-health can be defined as "mobile computing, medical sensor, and communications technologies for healthcare." This emerging concept represents the evolution of e-health systems from traditional desktop "telemedicine" platforms to wireless and mobile configurations [8]. The term M-health was first introduced implicitly in 2000 as "Unwired e-med" in the first special issue of the 'IEEE Transaction on Information Technology in Biomedicine' on wireless telemedicine systems by [9].

The increased availability, miniaturization, performance, enhanced data rates, and the expected convergence of future wireless communication and network technologies around mobile health systems will accelerate the deployment of M-Health systems and services within the next decade [8]. These will have a powerful impact on some of the existing healthcare services and will reshape some of the mechanisms of existing healthcare delivery routes [8].

It is evident that organizations and the delivery of health care are being underpinned by the advances in M-health technologies. These advances are giving rise to a range of reforms in the way in which some healthcare services are currently delivered. In the near future, the increasing medical data traffic and demand from different clinical applications and mobile medical scenarios will be compatible with the data rates of the current third generation (3G) systems. Specifically, in a society penetrated by 3G systems, home medical care and remote diagnosis will become common. Check-up by specialists and prescription of drugs will be enabled at home and in under-populated areas based on high resolution image transmission technologies and remote surgery, and virtual hospitals with no resident doctors will be realized. Preventive medical care will also be emphasized: for individual health management, data will constantly be transmitted to the hospital through built-in sensor and monitoring systems, e.g., in the patient's watch, accessories, or other items worn daily, and results will be fed back to the patient [8].

To realise the concept of using these technologies to improve the next generation healthcare system, we approach few of the different definitions been given earlier to the term telemedicine:

> "Telemedicine is a revolutionary means of providing healthcare to remote and underserved populations. telemedicine works symbiotically within the growing globalization. It simultaneously thrives within and enables the globalization", define by [10].

"Information technology has taken giant strides in developing effective and efficient tools for delivering health services to widely dispersed populations, much remains to be done on the parts of both the technology industry and the health system", defined by [11].

"Telemedicine means, organizing and integrating information technology in such a way that resources outside the local organization can be used systematically in the activities of health service", defined by [12].

"Telemedicine generally refers to the use of telecommunications and medical technologies to provide any or all of the following forms of information exchange, i.e., data, audio and/or visual communication between physician and patient or between physician and health care professional in geographically separate locations. Moreover, to facilitate the exchange of information for medical, health care, research and/or educational and training purposes, such exchanges can be on-line or off-line", states by [13].

Technically a telemedical application always has sender(s) and receiver(s) of information, located in distant places. Both sender and receiver have an adequate infrastructure with computers, storage units, e.g. Picture Archiving and Communication System (PACS) or electronic patient record, input/output units and medical devices, e.g. X-ray, CT or Magnetic Resonance Image (MRI), endoscopy, ultrasound and monitoring. Sender and receiver are connected by some electronic transmission path, e.g. Integrated Service Digital Network (ISDN) or the Internet, and these also require some security architectures in case of transfer sensitive personal medical data. The roles of sender and receiver may change during telemedical transaction. For example in case of a teleconsultation of two medical sites with the same rights, information flow is bidirectional. In case of information retrieval from databases by a physician or patient main information, flow is unidirectional [13].

1.1.2 E-health Terminology and Applications

It is convenient to characterise the e-health applications according to its maturity and based on their performance especially in specific medical settings [14]. In this section we categories the e-health applications into the following categories:

- *Emerging applications*; these include Telesurgery, Telepediatrics, and Emergency medicine. This set is labelled as emerging because of the combination of recent applications in the field, limited research, and limited professional acceptance at this stage, stated by [14].

- *Maturing applications;* these include tele-psychiatry, Teledermatology, Telecardiology, and Teleophthalmology. There has been substantial research and development work in these specialities states [15]. Substantial information is available about technical specifications, clinical effectiveness, and cost analysis of teledermatology [16]. Today successful transmission of echocardiogram images is feasible in both real-time and asynchronous modes, where ECG transmission via the internet has been reported in orcoronary-care-unit patient-monitoring data and to check patients being monitored at homes [17]. Furthermore, teleophthalmology research results are available from several studies in the United States, especially the Joslin Vision Network [18].

- *Established applications;* these include Teleradiology, and Telepathology, which represent the most mature and well established clinical specialities within the e-health categories. Teleradiology has been the leading clinical application in developing the Digital Imaging and Communications in Medicine (DICOM) standards [14].

A study in Finland investigated whether teleradiology consultations would reduce unnecessary patient transportation and thereby save on opportunity and treatment costs, 81% of the patients examined via teleradiology, avoided unnecessary transportation, and 75% of those transported to a central hospital were operated on, immediately on arrival without further radiological study. Moreover, real-time interaction and audio capabilities related to image acquisition, transmission, and display are fundamental to teleradiology and telepathology. For example in telepathology there existing robotic systems and real-time transmissions is to circumvent errors in field selection and depth of focus and to improve diagnostic accuracy. Ultrasound in teleradiology also requires real-time capabilities, synchronous transmission of video and audio signals are especially crucial in Doppler ultrasound applications and echocardiography [19].

1.1.3 The Significance of E-health

The current pace of change from industrial to information and telecommunication society is fast evolving medicine. Both patients and health care professionals should have a benefit from this development. In the near future, any health care provider going to be connected to the data highways. A great part of medical information will be transmitting in a fast and efficient manner; this will further improve patient care as stated by [13].

Globalisation and rapid progress of communication technology is merging the use of e-health in almost every aspect of health care from education, clinical consultation and diagnostic purposes to remote surgery. Apart from health education and training prospective, e-healthcare system can remarkably reduce cost, time and improve efficiency. Many studies and researches have innovated and fulfilled the promises of such issues [14].

Healthcare systems in developed countries is moving away from episodic care to concentrating on continuity of care, especially for patients with chronic disease, who will cause the greatest disease burden in the future. There is a gradual move in many countries away from a focus on the service provider to that of the informed patient, and from an individual approach to treatment to a team approach. Increasingly there is a focus less on the treatment of illness and more on the need of wellness promotion and illness prevention, which of course parallels a shift away from the traditional care to community healthcare, states clearly [4].

For example in recent years and especially in the last decade, a major stimulus to e-health was provided by European Commission (EU) funded research programmes such as ESPRIT and RACE. Most of the European countries are involved in healthcare telematics projects, many of them encouraged by national or regional healthcare authorities. In addition a very important contribution came from the EU Structural and Regional Funds. In spite of the barriers, telemedicine initiatives are now proliferating across Europe [7].

Across EU and United States, large number of e-health projects has been implementing in practically all areas of clinical medicine as well as medical education. Applications in various level of maturity from teleradiology, telepathology, telepsychiatry, telecardiology to emerging applications like telesurgery and emergency medicine has been substantially investigated and developed, where there are evidences of acceptance by professionals and on going standards and protocols development taking place [14].

1.2 Tele-robotic and Tele-operation Systems

In this section we present one of the important and growing fields of telemedicine that are relevant to this thesis, namely tele-robotic in medicine.

It is well known that a robotic structure with a master-slave tele-operation system has been used in various applications where autonomous robots cannot perform given tasks successfully because of the complexity of the tasks or uncertainty of the environments, states [21].

In general, detailed discussion on the use of robotic telemedicine based on Master-Slave teleoperation is explained [21, 43]. This system describes control schemes of master and slave arms using one simple Degree of Freedom (DoF) for each operation mode based on the new bilateral control scheme which has been previously proposed by the authors. The outcome of this experiment reflects how crucial the time delay is when the data size transfer between the master and slave machines can not be negligible. On another level, tele-operation systems for example have made it possible for man to reprocess plutonium, service satellites in space, and perform deep-sea salvage operations from safe control room by using remote manipulators and communication links [22]. These systems are typically operated by teams of specialists acting on data exchange from walls full of computer displays; hence tele-operation can make a major contribution for example to tele-surgery in medicine applications.

Tele-surgery is the provision of surgical care over a distance, with direct real-time visualisation of the operative field, and it may be categorised as described in [23] as follows:

- *Telepresence surgery;* uses a computerised interface to transmit the surgeons actions at a surgical workstation to the operative site at the remote surgical unit, with haptic (force feed back) input to transmit to the surgeon the tactile environment of the operative field.
- *Telerobotics;* remote control with a robotic arm, usually in conjunction with a laparoscope, without haptic feedback.
- *Telementoring;* an experienced surgeon acts as a preceptor for a remote inexperienced surgeon by observing the surgeon via interactive video.

1.2.1 Tele-echography Systems

As shown in earlier section, the research of this thesis is within the advances on robotic tele-echography (tele-ultrasonography) domain. A brief overview of these systems and their applications is presented here. Across the globe, large populations live far away from the main expert hospital facilities [24], hence medical doctors in these cases have a limited amount of data (echography and radiography) to evaluate the gravity of the clinical symptoms and to make the decision to whether to send the patient, by ambulance, helicopter or to the closest hospital or to keep they at home. In approximately 50% of the cases, patients are transferred to the hospital centre at night, to receive an ultrasound examination, and are sent back home a few hours later. Medical scenarios requesting an echography examination as a first evaluation arise several times a day in main hospital centres [24].

From above, the definition of tele-echography systems can be defined as "the use of modern information technology to transmit ultrasound images via a communication link, both in real time and stationary modes between primary care clinicians and specialists at some distance from each other".

In recent studies in this area [25, 26], a continuous ultrasound session carried out between General Practitioners (GP) and radiologists are based on two modes either; on store-and-forward image transmission or real-time interaction although the last mode lacked the necessary bandwidth used to transmit the scans. In total of 64 patients were scanned over an ISDN link, the results presented with an array of clinical conditions; 22 (34%) were abdominal cases, 18 (28%) were gynaecology and 18 (28%) obstetrics [25,26]. Altogether 229 store-and-forward images were transmitted, evaluated and compared with paper prints for the quality by the consultant radiologist, interacted with multiple videoconferencing sessions carried out between the Health centres and the Centre for Radiologist Education (CRE) at University of Portsmouth via ISDN link. Patients found the interaction pleasant, innovative and useful, and GPs were able to scan with confidence and were able to demonstrate the potential of the system to other colleagues, although in real-time capability there was a lag between the GP scanning and the radiographer at CRE receiving the images [25, 26]. However it is not clear from this study the approximate image transmission delay and the data rates of the ISDN line is not specified.

The Fraunhofer institute for Computer Graphics (IGD) has developed TeleinVIVO tele-echography project for visualisation, processing and analysing volume of 3D ultrasound imaging data, mainly for medical applications developed by [27]. With the built in flexible telecommunication channel (phone line ,Internet, ISDN, and Global System for Mobile communications (GSM) or satellite), the acquired 3D ultrasound data set are transferred to remote experts, who can be located virtually everywhere in the world to perform "virtual echography" in each site and viewing identical images on their screens in relatively real-time. A wavelet-based image compression algorithm has been implemented for the size reduction of the image data sets to be transferred; the compression is lossy but very efficient in cases where bandwidth is a crucial issue. The resulting system has low price, low weight, is transportable, non-radiating and supports a very large range of applications varying from gynaecology over pathology to abdominal scans, and when the diagnosis proposed by the reading site was compared with clinical pathological, follow-up or the previous diagnosis of the patients, there were 27 true positive, 46 true negative, 2 false positive, and 4 false negative diagnosis, corresponding to 96% specificity, 92% sensitivity, and an accuracy of 93%.

However this system still lacks the ability of two ways interaction during the consultation process, and did not solve the problem of the presence of the specialist in the remote area, where the action of one user introduced at a time, needs high ranking specialists on both sites, and most important thing that the range of the communication links proposed are significantly varying in capabilities and limitations, where the performance of the system can not be reliable.

The Multimedia Interactive DemonStrator Tele-Presence (MIDSTEP) was a tele-presence research project within EU developed by [28], the project had the principal objective of realising two surgical tele-manipulation demonstration systems; one over Local Area Network (LAN) and one for remote tele-manipulation over Wide Area Network (WAN). Both utilised remote tele-presence manipulation of an ultrasound probe producing images subject to compression factors of 15:1 and 30:1 based on Joint Photographs Expert Group (JPEG) codec, and transferring data through Asynchronous Transfer Mode (ATM) as a communication technology. The general objective of MIDSTEP was to make use of an ultrasound radiologist to remotely perform simple biopsies and examinations. The project demonstrates the ability to transfer high quantities of video and control data to control a Robot in performing delicate tasks, and satisfactory receiving images compressed up to 15:1. The application in this project, based on using high bandwidth links (LAN or WAN, which provide high data rates; 10, 100Mbps). These systems require fixed lines and locations for both Expert and Remote stations.

In general most of the tele-ultrasonography telemedicine studies in the UK, describe the use of ISDN connections ranging from ISDN2 (128kbps) up to ISDN30 (2Mbps) [29]. The cost of ISDN30 (thirty lines) telemedicine is often costly. Therefore, the acceptability of transmitting dynamic ultrasound images at transmission speeds of 384Kbps and 128Kbps, were investigated and examined. The results was that an average of 90% of dynamic ultrasound images transmitted at 384kbit/s were rated as diagnostically acceptable by the four observers, compared with 32% of those transmitted at 128Kbps. There was complete diagnostic agreement between the original images and the 384Kbps transmissions in 85% of the cases. [30, 31] states that there was complete diagnostic agreement between the original and received images, at 128Kbps transmissions in 78% of the cases. The later rate over 128Kbps, considered very convenient for pre-diagnostic purposes for OTELO system medical requirements.

1.2.2 The OTELO System: An overview

Currently, ultrasound examinations are only performed by well-trained specialists in the main clinical centres (public hospital, private centre... etc). Studies have demonstrated the potential use of echography to perform a quick and efficient diagnosis for many types of pathologies that sometimes require immediate care: appendicitis, internal abdominal bleeding, kidney stones...etc [32,33]. Unfortunately, ultrasound experts are scarce and not always on hand when needed. mObile Tele-Echography using Ultra Light rObot (**OTELO**) is a fully integrated end-to-end mobile tele-echography system dedicated to population groups that are not served locally, either temporarily or permanently, by medical experts. OTELO was funded by the EU programme (Contract Number: IST-2001-2004, 32516). There are partners participating in the project works from different EU countries (France, Spain, Italy and UK), each of those partners performed a specific part of the project. Kingston University is representing the UK partner, performed the research works on the wireless and mobile communications issues of the OTELO system, and that is my research covered by this thesis.

OTELO offers an alternative to specialist medical centres that lack ultrasound specialists; it is a portable ultrasound probe holder robotic system, associated with new mobile communications technologies, that reproduces the expert's hand movements to perform at a distance an ultrasound examination [24]. Although being held by non-specialised paramedic on the remote site, the Patient system brings in real time good ultrasound image quality back to the Expert station where force feedback control is combined with virtual reality for the rendering of the distant environment.

OTELO is a remotely controlled system designed to guarantee a reliable echographic diagnosis in an isolated site far away from an expert (Medical Doctor) located at the expert clinical site (University Hospital, Ultrasound expert centre...etc), a non ultrasound specialist is present (e.g. nurse) next to the patient, at the isolated site or in rescue vehicle, and a reliable communication link to connect the two sites. The research work of this thesis forms part of the OTELO project, the general objectives of the project were [34]:

1. To design and develop a new ultrasound 6 DoF probe-handler robot that enables, via state of the art mobile and fixed communication links, remote ultrasound examinations with no need to medical expert on the spot of examination.
2. To develop a new methodology for real-time tele-echography, tele-robotic clinical examinations.
3. Widening the range of the communication channels choices operating for the tele-echography Systems to mobile and IP based connectivity in addition to the satellite link and land lines.
4. To develop techniques for reliable and efficient performance, of the system functional modalities.
5. Examining new compression techniques, clinically acceptable for medical diagnosis purposes and still subject to the standardisation adopted by wireless telecommunication institutions.
6. Innovative approach to enhance the robotic tele-echography applications, with powerful tools to expand the range of functions and performances for multiple purpose clinical remote echography.

A- OTELO System Architecture

The Expert / Patient terms will replace the Master / Slave terminology used to define the robotic system ends as mentioned in the previous section. That will better explain their usability for the medical examinations purposes, and especially for OTELO tele-echography system. The general architecture of OTELO system is shown in figure 1.1 below [34, 47]:

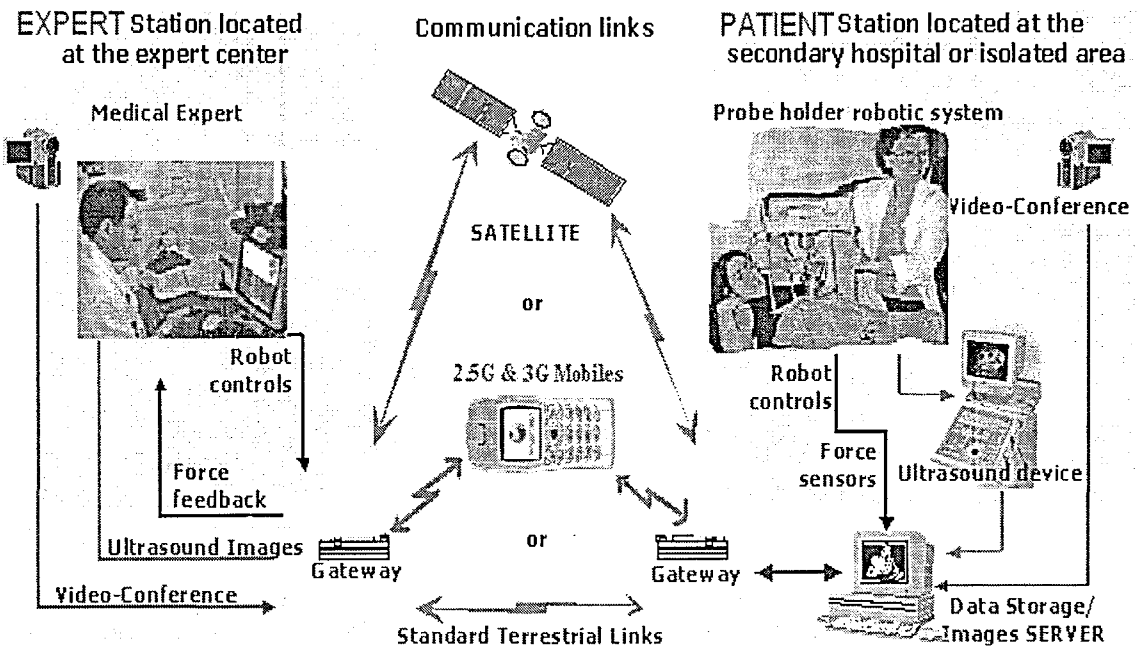
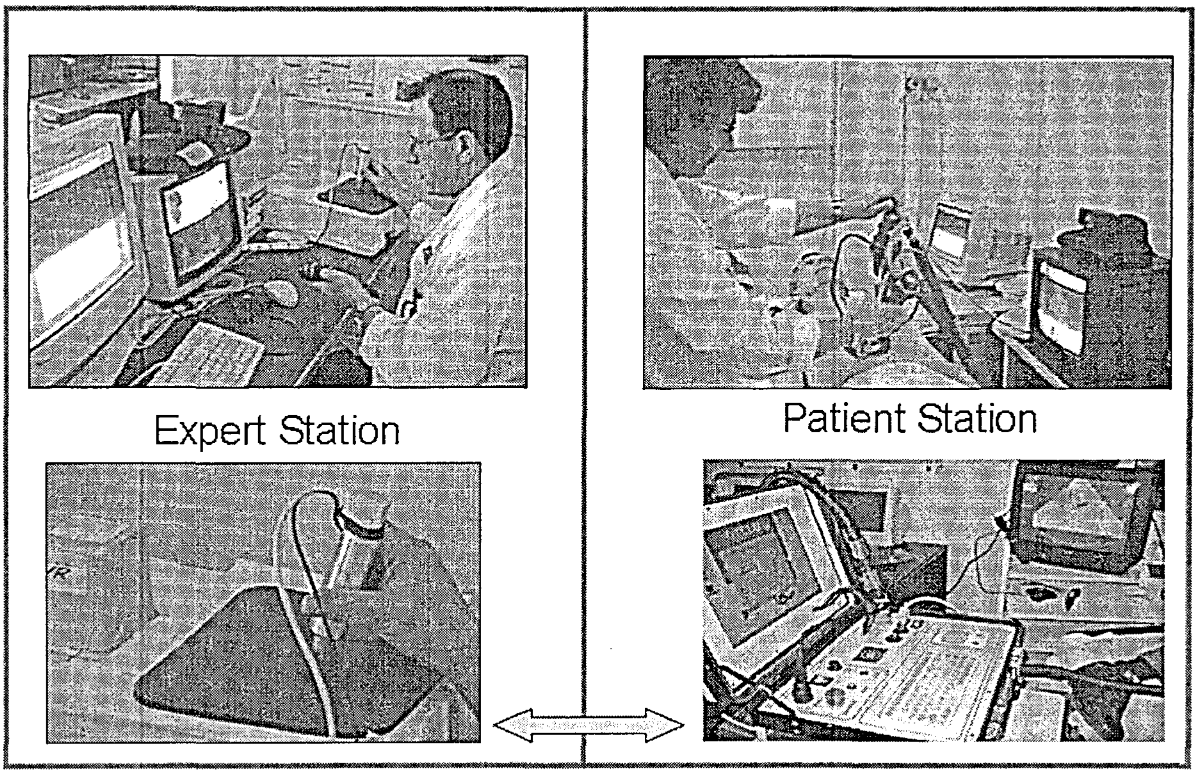


Figure 1.1 OTELO Tele-echography System and the Communication Links.

OTELO system includes two major parts:

The Expert Station:
At the expert station site, the clinical expert's role is to control and tele-operate the distant robot by holding a fictive probe. The expert should be able to visualize in real-time the patient ultrasound images on a control screen and continuously receives force feedback information from the patient station. The haptic information is a necessity for the specialist, as ultrasound examination requires good hand-to-eye co-ordination, and the ability to integrate the acquired information over time and space.

The Patient Station:
At the patient site, a six degree-of-freedom probe holder robot reproduces the movements performed by the expert on the fictive probe. The system is able to handle various types of manufactured ultrasound probes. Its mechanical structure is lightweight, easy to handle and has to maintain a continuous contact with the patient's skin, thanks to a force control. The actuators driving the robotic system are chosen to give the best response time despite delays introduced naturally by communication links which could be constant, time-varying or even random. Figure 1.2 below shows the Expert and the Patient stations:



(a) (b)

Figure 1.2 Both ends of OTELO system, (a) The Expert station, where the clinical expert is operating the fictive probe. (b) The Patient station, where the paramedic is holding the ultrasound robot.

The communication link:

In OTELO system, and to perform a Real-Time clinical session, the following data needs to be exchanged between the two sites, based on the clinical test scenarios; Stream of ultrasound images, ultrasound still images, ambient video, voice, and robotic control data. To transmit the above data, a reliable communication link that can fulfil the demand of a large number of patients, is required. This communication link allows discussion on medical information or robot positioning between the expert and the paramedic, and also enables the expert to comfort the patient throughout the examination. The communication link options for OTELO connectivity could be; ISDN line, Asymmetric Digital Subscriber Line (ADSL) line, Satellite link, wireless and mobile link. In this thesis, the research work will focus on the performance analysis of the General Packet Radio Service (GPRS) and Universal Mobile Telecommunication Service (UMTS) wireless connectivity within OTELO telemedical environment.

B- OTELO Services

The main objective of The OTELO project is dedicated to the development of an advanced tele-echography system, which will bring to population groups preventive care support using the latest mobile robotic based ultrasound techniques [24, 35].

Services to Physicians and Specialists:

OTELO allows doctors and ultrasound specialists to create the personalised patient data records and tele-echography imaging profile including information regarding diagnosis, examination plan and any other information about the remote patients and from their offices. After completing the review of the developed diagnosis, the physician will forward the plan to the patient via OTELO service, which will then allow the remote and mobile nurse or the non-specialist to take the proper actions and log the patient's case for further analysis or transfer to specialist unit if necessary.

Services to patients:
Will be provided by means of patient-to-OTELO, portable and compact unit. With such service, the patient at home or remote area can be in complete interaction with the specialist during the remote in-house examination, and can request disease-specific information. The service is also important for a second opinion diagnosis that can be provided from another distant expert centre.

OTELO targets the world market of personal health services. The area is very wide, and the application of advanced technologies poses an important business opportunities. OTELO is a service that will be primarily provided to hospitals, clinics, or other health institutions and physicians dealing with emergent or non-hospitalised patients. National health systems and Hospitals (private and public) are considered the most important "client groups" to whom OTELO will target first. Though for commercial exploitation purposes NHS and Hospitals are analysed as primary client of OTELO. The project target groups can be defined as follows:

- Public and or/ communal health authorities
- Private health service providers (private hospitals and clinics)
- Individual Physicians
- E-learning ultrasound departments

1.3 M-health

The evolution of current 3Gwireless communication and mobile network technologies will be the major driving force for future developments in M-health systems. 3G wireless technology represents the convergence of various second-generation wireless systems. One of the most important aspects of 3G technology is its ability to unify existing cellular standards, such as Code-Division Multiple-Access (CDMA), GSM and Time-Division Multiple-Access (TDMA) under one umbrella states [34].

Today with wireless technology, patient records could be accessed by healthcare professionals from any given location by connection to the institution's information system. Physicians' access to patient history, laboratory results, pharmaceutical data, insurance information, and medical resources would be enhanced by mobile technology, thereby improving the quality patient care. Handheld devices can also be used in home health care, for example to fight diabetes through effective monitoring. A comprehensive overview of some of these existing wireless telemedicine applications and research can be found in recent publications in this area [9, 36, 37, 38]. However, there are some limitations to existing wireless technologies that mostly depend on GPRS technologies and on their deployment strategies health care. Some of these issues can be summarized as follows [39]:

1) The lack of an existing flexible and integrated "M-health-on-demand" linkage of the different mobile telecommunication options and standards for e-Health services. This lack of linkage and compatibility for telemedical services exists due to the difficulty of achieving operational compatibility between the telecommunication services, terminals and devices standards, and "M-health protocols."

2) The high cost of communication links, especially between satellites and global mobile devices and the limitation of existing wireless data rates especially for the globally available 2.5G and third-generation (3G) services for some e-Health services. This is also combined with the availability of secure mobile Internet connectivity and information access especially for e-health systems.

3) Healthcare is a very complex industry that is difficult to change. Organizational changes are very often required for healthcare institutions to benefit from e-Health and M-health services.

4) The short-term and long-term economic consequences and working conditions for physicians and healthcare experts using these technologies are not yet fully understood or properly investigated.

5) The methods of payment and reimbursement issues for e-Health and M-health services are not yet fully developed and standardized.

6) There is a lack of integration between existing e-Health services and other information systems, e.g., referral and ordering systems, medical records, etc.

7) The demonstration projects so far have failed to show that M-health services result in real savings and have cost effective potential.

These represent some of the factors that have hindered the wider applications of M-health technologies thus far across healthcare systems. However, it is hoped that the current deployment of UMTS networks globally will alleviate some of these issues and will provide a better and more effective platform for mobile healthcare services.

The most important and critical issue in any successful M-health or e-health system is the communication medium required to provide effective and accurate diagnosis and treatment. It is well known that the evolution of mobile telecommunication systems from 2G to 2.5G (GPRS) then to 3G (UMTS) will facilitate the provision of faster data-transfer rates, thus enabling the development of telemedicine systems that require high-data rates. In wireless telemedicine systems, a comprehensive overview of recent wireless telemedicine systems has been reported in [32]. In the following sections we will review some of the research works on wireless telemedicine systems.

1.3.1 An overview of M-health Systems

The availability of prompt and expert medical care can improve the healthcare services in rural and isolated areas. The provision of effective emergency telemedicine and home monitoring solutions is the major field of interest of the Ambulance HC1001 and Emergency 112 HC4027 projects, which were partially funded by the E.U / DGXIII Telematics Application Programme [32].

The aim of the AMBULANCE project was the development of a portable emergency telemedicine device, which supports real-time transmission of critical bio-signals, as well as still images of the patient, using a GSM link. The system comprises of two different modules: The mobile unit, which is located in an ambulance vehicle near the patient: and the consultation unit, which is located at the hospital site, and can be used by the expert in order to give instructions. Communication was performed using Transmission Control Protocol / Internet Protocol (TCP/IP). Transmission rate were limited to 9.6 Kbps, which is the maximum transmission rate for GSM. ECG data were sampled at 200 samples/s, thus resulting in a generation of 1.6 Kbps per lead, thus for real-time biosignal transmission, the available GSM bandwidth was adequate under normal network congestion conditions. Images were captured at 320 x 240 pixel resolution and compressed using the JPEG compression algorithm. The resulting data set was approximately 2.5-3 KB in size, thus resulting 3-5 Sec for transmission.

EMERGENCY – 112, which was the extension of the ambulance project, aimed at the development of an integrated portable medical device for emergency telemedicine. The system enables the transmission of critical bio-signals (ECG, BP, HR, SpO2 and Temperature), and still images of the patient, from the emergency site to an emergency call centre. The system was designed to operate over several communication links, such as satellite, GSM and IDSN. The system was used to provide emergency health care from ambulances, rural hospital centres or any other remotely located health centre and for patient home monitoring. The above two systems are not proposing technically real-time operation, and due to the fact that the GSM is one of the proposed communication link, where the maximum rates for GSM is around 9.6 Kbps. Transmitting sound signal over GSM, will take priority and will occupy most of the available bandwidth, where there will be no chance for the ECG signals to be transferred in a real-time. Furthermore it is known that the TCP/IP protocol is not the preferred communication protocol for real-time applications.

Traditionally, the "wireless concept" is associated closely with "biomonitoring." These have been used extensively in the last two decades to perform different data acquisition tasks mostly, without timely integration of data into the medical record; thus, no immediate action occurs if abnormalities are detected. Typical examples are Holter monitors that are routinely used for ECG and electroencephalogram (EEG) monitoring [34]. A study to find the diagnostic accuracy of a telecardiology service was evaluated, where each practitioner was equipped with a 12-lead ECG protable electrocardiograph (Card-Guard 7100), which was connected to a mobile GSM or fixed connection. The study concluded that the telecardiology service when compared to emergency-department admission, showed a sensitivity of 95%, a specificity of 97.5% and a diagnostic accuracy of 92.5% as stated by [40]. Although no means of signal compression mentioned in this experiment, the achieved results shows good performance rates since the ECG signal bandwidth requirements is very close to normal sound data rates (around 9 Kbps) over GSM connection,

Adequate, continuous information about patients during advanced medical procedures is a major factor in the patients overall satisfaction. Moreover, patients, while hospitalized, need to continue their daily routine, or they at least need to be able to communicate through modern means with the outside world. On the other hand, a specialized physicians who are moving inside or outside the hospital need to have complete and continuous information about a patient's record, in order to be able to provide the best medical practice.

These are the main issues addressed by the Mobile Medical Data (MOMEDA) HC4015 telemedicine project, which was partially funded from the European Commission. The system consisted of two modules, the patient's information modules and the doctor's information module. The main objective of the patient's information module was the development of a demonstration unit called the Personalized Medical System (PMIS). This system allows access to customized disease-specific information from patients about their medical problem, the planned medical procedures...etc.

The main objective of the doctor's information module was the development of a demonstration unit that allows the consulting physician to access electronic patient records, using a hand-held companion device, connected to the GSM network. A module based on the Nokia Communicator 9110 was developed and tested through the project. The user of this module was able to connect to the hospital's main server, and to receive electronic records and medical images, such as MRIs. The project was successfully tested in three European countries (Filand, Italy, and Greece) [32]. This system is non real-time data transmission system, due to the use of a low data rate communication channel (GSM).

1.3.2 M-health and tele-echography Systems

In this section, we review some of the recent works on wireless tele-echography systems and discuss their relevance to this research work. Earlier work in this area is reported by [41], a group of researchers in department of medical informatics, Ehime University, Japan. The team experimented a tele-diagnosis system to control Echographic Diagnosis Robot (EDR), which is developed in the university laboratory. The places of the examiner and the patient connected by wireless network, the robot developed to performe 6DoF motion of three dimensional rotation and translation. Two wireless LAN bridges SB-1100 (ICOM Co. Ltd) to connect Ehime University and temporal examination room that is located 1.4Km apart from the hospital at 10Mbps. Line speed was sufficient to communicate at data rate up to 6Mbps.

In this experiment ITU T.120 standard data protocols were used that enabled high quality conference with real time moving echogram. The time delay was less than 1sec when the image size was Common Intermediate Format (CIF) 352x288 pixel for echogram by 512Kbps. The results of this work are based on wireless LAN, short distance field and not over a wireless telecommunication channel (e.g. UMTS), where the added noise in terms of interference and mobility can significantly effect the performance.

A telerobotic system, in which internet users can access and command a 2 DoF robot in a real time closed loop over the Internet, receiving both visual and force feedback, is described in [42]. Java Based Interface for Telerobotics (JBIT) project, aimed at demonstrating the feasibility of Internet-based telerobotics equipment. The real-time video is realised by using H.263 compression, which brings a 4 frame/s performance over a 28.8 Kbps modem connection, and the obtained results confirmed that real-time robotic control loop operation over the Internet is possible. Concerning this application, although practically the data rates of 2 DoF and 6 DoF robot movements are almost close to each other, nevertheless when transmitting ultrasound images, the above bandwidth criteria will be insufficient for end-to-end real-time robotic tele-echography application.

A virtual-reality-based tele-rehabilitation system with force feedback was developed by [44] for use at home, the system has a Pentium II PC with graphics accelerator, a polhemus tracker, and a multipurpose haptic control interface. This interface is used to sample a patient's hand positions. The real-time software running on the embedded Pentium reads and filters data from the sensors through an analogue de-multiplexer and A/D board and transforms it into the patient's joint angles. Data sent to the host contain joint angles, measured forces, or device state, while received data from the host include commands or forces to be displayed to the patient. At a rate of 57.5 Kbps, the RS232 line can transmit up to 187 sensor position data set per second or 166 data sets that contain both position and finger force readings every second. Videoconferencing tools installed at server and client sites use CuSeeMe software, and during several teleconferencing system trials they obtained uneven performances, in some cases with only 2-3 frames Per Second (fps).

The above system depends entirely on the Internet as a communication medium, and apart from that, RS232 with 57.5 Kbps is very convenient for the robot data transmission in a real-time, but loading the line with ambient video will degrade the system performance, add to that the effect of the time delay variability of the Internet network.

In OTELO project, a major collaboration of various technological aspects functioned to validate the system functional modalities. A robotic control arm of 6DoF with force feedback, gives the sense of virtual reality in terms of felling the pressure on the patient body. Implementing GPRS and UMTS wireless communication technology as the preferable communication links, this is fully investigated for its suitability to OTELO system. Widening the range of the operating bandwidth that OTELO system requirement has been validated on to prove the ability of the system for effectiveness and flexibility to perform at different networks and test environments.

1.4 Contributions of the Research

The main objectives and contributions of the research work in this thesis can be summarized as follows:

1. Performance analysis and validation of a mobile robotic tele-ultrasonography system (OTELO), over 2.5G and 3G Mobile communication networks. Comparative studies on the performance of the system in both simulated and real-network conditions are also presented.

2. Concentrated study on M-health systems that requires close loop robot control, delay sensitive and real-time medical data transmission to work together reliably in a wireless and mobile communication environment.

3. Design and development of new quality adaptation algorithm for enhanced video streaming in mobile robotic tele-ultrasonography environments.

1.5 Scope of the Thesis

The work in this thesis is divided into seven chapters, the following presents the outline and content of these chapters:

- Chapter-1: This chapter presents an overview on telemedicine and e-health. It is also provides a comprehensive literature review on wireless telemedicine and tele-echography systems. The chapter also outlines the OTELO systems and presents a brief overview of this tele-robotic echography system.

- Chapter-2: In this chapter we present the architecture of the OTELO system and the connectivity of its stations over the communication link. Also this chapter is setting up the test matrices to be implemented based on the system functional modalities.

- Chapter-3: This chapter presents a brief theoretical overview 2.5G wireless communication systems, and analysing the OTELO system performance in both simulated environment and over operating GPRS network. The chapter also presents the physical layer limitations of GPRS system specifically the Up-link bandwidth limitations.

- Chapter-4: The first section presents a theoretical background on the 3G wireless communication technologies, its capability, limitations and QoS issues. Moreover it presents comprehensive and comparative system performance analysis on both simulated and real network environments regarding the Real-time medical data transmission over (W-CDMA) air interface. Analysing the performance of the OTELO system in terms of the end-to-end delay, packet loss, jitter, main and maximum throughput and many other functional modalities are presented.

- Chapter-5: This chapter introduce the objective performance analysis of transmitting OTELO ultrasound images, over both networks environments (2.5G and 3G networks). It presents the relevant test methods for evaluating the ultrasound transmission objectively. It also briefly presents the quality evaluation of three types of ultrasound image format that OTELO system is dealing with (4CIF, CIF and QCIF) in lossy and losseless compression modes.

- Chapter-6: This chapter presents a new quality adaptation technique, specifically designed to fulfill OTELO system medical tele-ultrasonography requirements to work on a wider range of operating channel data rates. The adaptation technique operation should accommodate the minimum QoS requirements of the system medical ultrasound stream, in terms of the image quality and frame rate.

- Chapter-7: This chapter summarises the conclusions of the thesis. The future research directions are also introduced for future work in this area of advanced mobile telemedical systems.

CHAPTER 2

mObile Tele-Echography using ultra Light rObot (OTELO)

2.1 Introduction

This chapter presents an introduction of the structure of the OTELO system. Section two presents a general overview and the major building blocks of the system.

Section three presents the general communication links requirements to perform the remote clinical examinations and diagnostic scenarios, this section presents as well the asymmetrical data traffic requirements for each data type implemented by the system.

Section four presents the model design of the OTELO system using GPRS and UMTS networks and the connectivity options over the above environments and the testing mechanism, hardware and the software implementation issues. This section also briefly introduces the performance of the OTELO system by using the ISDN and satellite links and the connectivity issues between the Expert and Patient stations of the system. These represent alternative wired and terrestrial communication links for the OTELO system, other than the wireless options.

2.2 OTELO System

The OTELO system's aims are to bring to patient population groups in rural areas or in more isolated geographical regions, preventive care support, using the latest ultrasound investigation techniques by a remote ultrasound specialist. Co-operative tasks with others for image analysis can also be achieved by developing an advanced tele-echography system using fixed links, satellite facilities and mobile links [51]. Figure 2.1 shows general OTELO system connectivity over the relevant communication links:

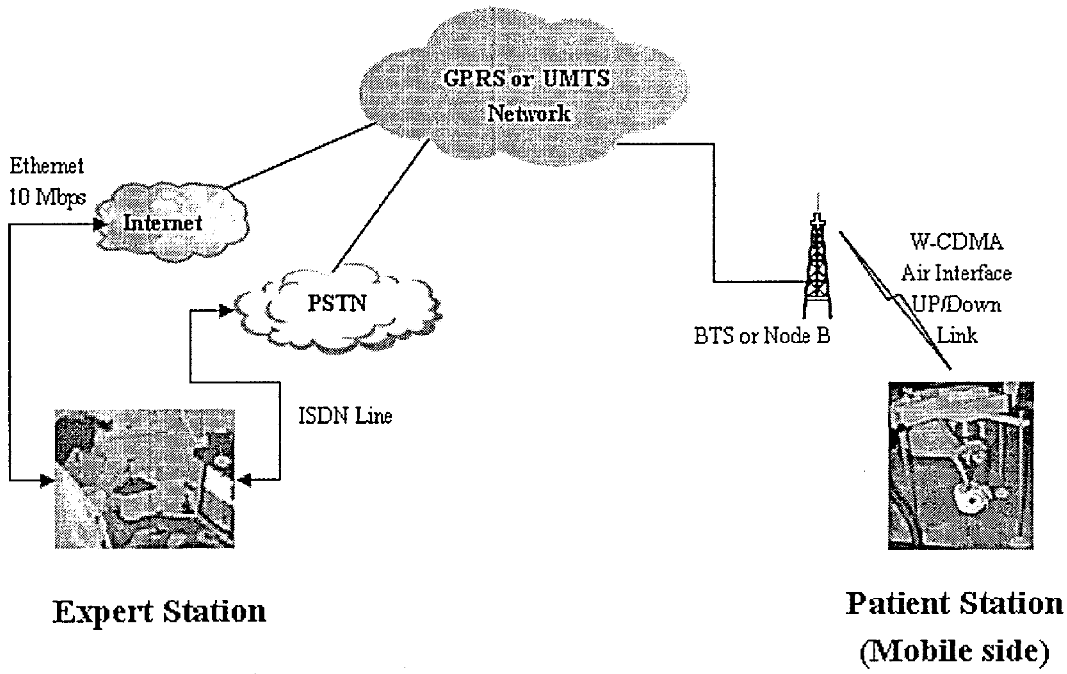


Figure 2.1 General OTELO system end-to-end connectivity.

The main structure of the OTELO system consists of two sub-systems:

1. The communication network
2. OTELO Robotic and Remote Ultrasound imaging interaction sub-systems.

OTELO System Architecture:
The system has three different robotic system prototypes and generations (namely OTELO 1, OTELO 2 and OTELO 3). They were developed and manufactured in France by SINTER and University of Orealans [46]. Most of the work of this thesis is applied using OTELO 1, as it was available for testing for the duration of this research work. The OTELO robotic system is comprised of the following sub-systems:

1. The Expert Station (ES): to be delivered to medical centres and operated by the Expert. It comprises of the following:

   a) A robotic arm (fictive probe) used by the medical expert to control remotely the tele-operated 6DoF robotic arm located at the patient station.
   b) Hardware, including a PC or Laptop for interfacing, and software for robot control, ultrasound images, audio and video data transfer over the selected communication links.
   c) High-performance group videoconferencing system up to 15 fps at 128 Kbps, with Ethernet / Internet / Intranet connectivity.
   d) An Electromagnetic device called Flok of Bird (FoB) to initialise the Fictive Probe position prior to the test.

   Fig. 2.2 shows the relevant parts of the Expert Station, that will be connected to the Patient Station through the mobile communication link [47], and it is not suggesting the communication link to the other station, since there will be more than one communication link option as we will see in more details in the following chapters:

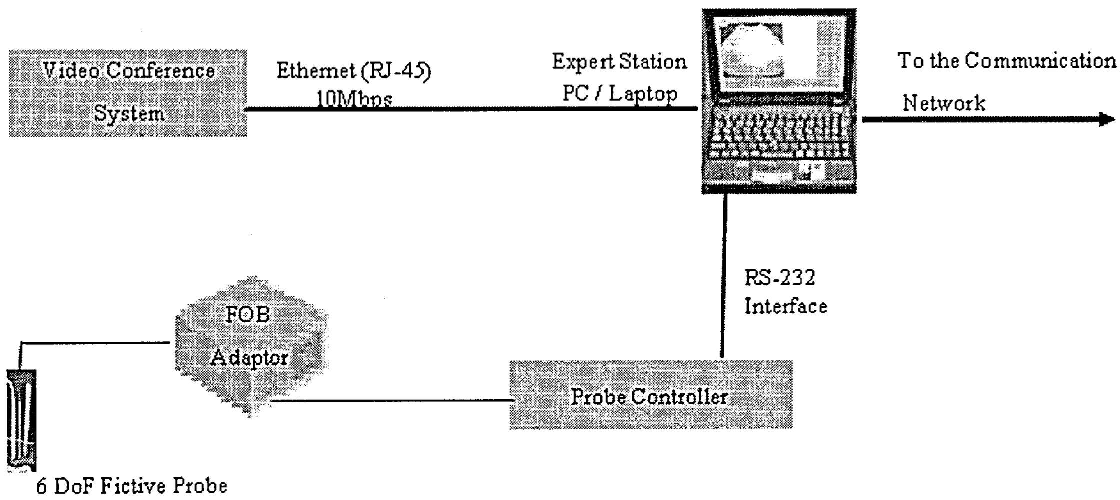


Figure 2.2 Expert Station components.

2. <u>The Patient Station (PS):</u> to be delivered to remote medical centres (secondary hospital, dispensary, ship... etc) where the ultrasound scanning is taking place. The Patient station comprises of the following items:

    a) A light ultrasound probe-holder robot with 6DoF and tele-operated by the clinical Expert at the Expert Station side via the communication link.
    b) An ultrasound acquisition system (the system will investigate different module examples, TRINGA 50S, 240 PARUS).
    c) A couple of general purposes Ultrasound probes, with a diameter of 35cm.
    d) Hardware, including a PC or Laptop for interfacing and software for robot control, ultrasound images, audio and video data transfer over the selected communication links.
    e) Same videoconferencing specifications as mentioned above.

The figure below shows the basic diagram for the patient station. It is reflecting version-1 (OTELO-1) [47]:

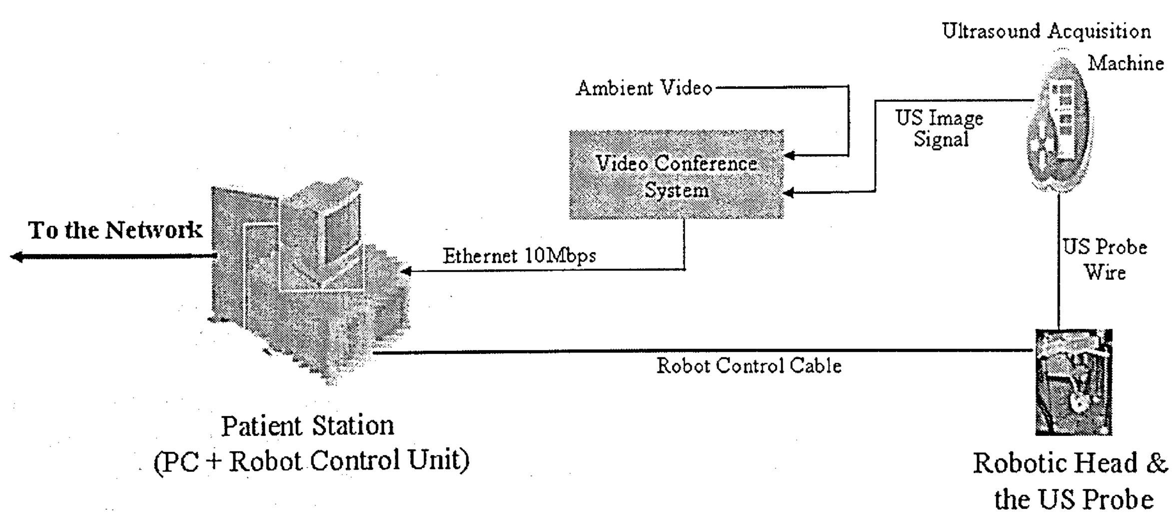


Figure 2.3 Patient Station Components

The following communication links and connectivity options are used between the Expert and the Patient stations:

1) A terrestrial link (ISDN, ADSL lines), with a wide range of operating data rates depending on the number of lines installed.

2) A terrestrial link as above from one end and a Satellite link at the other end, where the Expert / Patient station can establish a kind of Virtual Private Network (VPN). The Patient station will be the end connected to the satellite link.

3) GPRS & UMTS wireless options; GPRS is the supporting packet switching within the mobile network. Based on the radio resources, Quality of Services (QoS), Coding Schemes (CS) applied and the operating bandwidth provided by the service provider, the GPRS could be the third communication link option for the OTELO system. More details will be in Chapter 3.

4) UMTS is the wireless and packet switching most promising mobile communication link candidate for OTELO system, which will be fully investigated in Chapter 4, for its bearer services, different QoS classes and dynamic bandwidth allocation capabilities.

2.3 Communication Options and Traffic Classes of OTELO

During the tele-echography examination, ultrasound images have to be transmitted from the Patient site to the Expert site. The received ultrasound images will bring to the expert relevant information to move the fictive probe and to control the distant probe-holder robotic system, in order to analyse different anatomical regions or to search for a region of interest [48]. These images will be transmitted via various communication links that have different bandwidths, i.e. data rates (from as low as 56 Kbps up to 384 Kbps; Patient station uplink).

A- OTELO Stations Connection:

- Fixed link (Expert Site) ⇔ mobile link (Patient Site)

Or

- Mobile link (Expert Site) ⇔ mobile link (Patient Site)

B- Transmitting Four Types of Traffic:

1) Robot control data associated with fictive feedback sensors data (in real-time).
2) Still Ultrasound Images transfer only in one direction, from Patient station to Expert station (not in real-time).
3) Ultrasound streams of image transfer only in one direction, from Patient station to Expert station (in real-time).
4) In addition to full duplex ambient video and sound (videoconferencing, to take place after the medical session).

C- Robot Real Time Controls:

Time delay of less than 350ms is acceptable. User Datagram Protocol (UDP), Real Time Protocol (RTP) or E-UDP (Enhanced UDP) is suggested to be used for robotic control, since it is not advisable to use connection oriented protocol like TCP depending on retransmission mechanism, for real-time robot control.

2.3.1 OTELO Clinical Scenarios

The clinical session remotely controlled by the Expert side is designed to have the following sequence of operations, which will then be structured over different scenarios depending mainly on the communication links and the available bandwidth [49, 50]:

1- During the hardware and session preparation, only voice and text messages exchange are required between both ends.

2- The medical expert would remotely control the robot (which would already be properly orientated above the patient body), with the help of receiving ultrasound images or ambient video information alternatively. Throughout the process, the Expert will be adjusting the force needed to place pressure on the patient body.

3- Through the ultrasound scanning and organs location and diagnosis, a satisfactory Ultrasound image stream transfer on real-time is needed, no ambient video is necessary.

4- After locating the area of examination, high quality of still Ultrasound images transfer need to be received. This is because the data rate used for the communication link, makes a difference in conjunction with the compression standards implemented.

5- The expert will then analyse and validate the incoming images and information clinically, in co-operation with the patient side with the help of high quality videoconferencing. Then real-time continuous communication can still be available until one end terminates the connection (more likely the patient, as the customer who needs the service).

The basic procedure carried out by the Expert operator during a tele-consultation session may have corresponding operative options related to the availability of bandwidth, which mainly needs to be increased as we approach the scenario of full real-time data transmission as shown in Table 2.1:

Table 2.1 OTELO scenarios and data flow.

	Description	Operating Bandwidth Options 1), 2), 3), 4) (Low-to-Higher)
1	*Communicate to Patient station of standard setup:* The patient position (standardised) for the examination of a specific anatomic area. This position of the patient and the US probe is shown in a virtual and simplified patient representation on the Expert PC screen, by the Virtual Patient Body Representation (VPBR).	
2	*Resize the Operation Field Area (OFA):* To be framed by the remote video-camera to improve other parameters. (E.g. frame rate, colours). The camera co-ordinates; the camera focus and the camera zoom may be loaded in any moment.	Frame rate, max size, colours have to be automatically configured on the basis of the bandwidth used and according to the kind of exam to be done: Abdominal and Cardiac examinations
3	*Do remote US test:* Scan to define the Region of Interest (ROI), inside the 512*512 Pixel US image examination window. The Expert freezes during this scan a test cine-loop. The R.O.I. will be defined within this loop.	*1) The RO.N.I. remain static during the R.O.I. definition. The OFA is stopped during this phase.* 2) The RO.N.I. will be shown in real time at resolution lower than the R.O.I. (bigger pixels) during the examination. The OFA display is stopped 3) Like 2) with active OFA. 4) Like 3) with R.O.I. = 512*512 Pixel.
4	*Physician starts the effective examination:* The probe movements are showed on the VPBR. The real time US examination is always shown in the R.O.I. When an interesting US sequence is found static images or loops can be frozen and received by the Expert station. The same images or loops but Lossless and full screen are always forwarded to the OTELO central archive in non Real-Time mode after the teleconsultation, when the stations are not used.	1) The R.O.I. of frozen images is forwarded lossless to the Expert station during the on-line teleconsultation. OFA display is not active. 2) Like 1) for static images. Loops of prefixed size (ex. 35-40 frames) are forwarded loss to Expert station during the on-line teleconsultation. OFA display is not active 3) Like 2) but with OFA display active simultaneously.
5	On the basis of the quality of exam in real time (frame rate, R.O.I. size, etc) the Expert may decide to go to step 1- or 2- or 3- to improve it.	

Legend:

OFF-LINE: images transmission mode after the real-time teleconsultation end.

ON-LINE: images transmission mode during the real-time teleconsultation.

O.F.A: Operational Field Area, is the area of the remote operational field framed by the video- camera and transmitted in real time or near real time to the expert station.

R.O.I: Exam Region Of Interest, is the area of the US image transmitted in real time or near real-time from the Patient station to the Expert station.
RO.N.I: Region of No Interest; is the US images area around the R.O.I.
V.P.B.R: Virtual Patient Body Representation.

2.3.2 OTELO Asymmetric Traffic Requirements:

From the clinical requirements and OTELO proposed scenarios, we can conclude that the data type and the medical data rates required to maintaining a continuous remote examinations, are not transmitted symmetrically i.e. in other word the Patient station has to transmit at higher data rates with respect to the Expert station. This is true for all scenarios especially scenario (3) in table 2.1 for full real time remote clinical examinations [56, 50]. Figure 2.4 below shows the direction, the data type and approximate data sizes transmitted from each station:

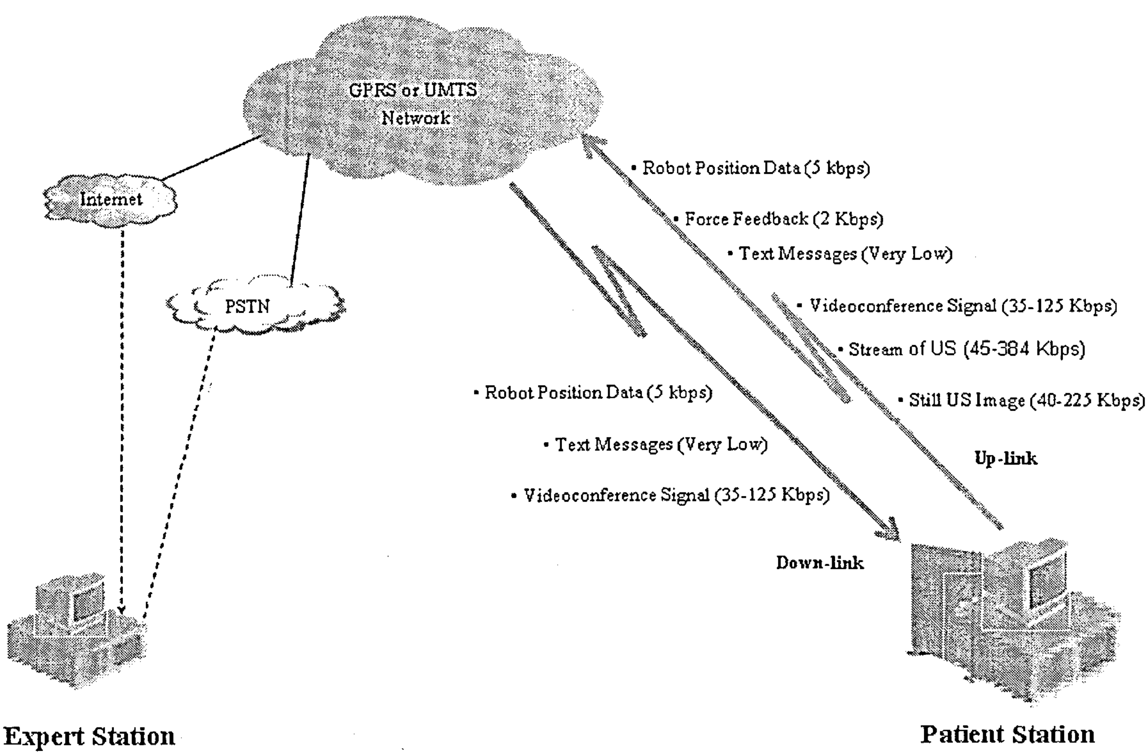


Figure 2.4 OTELO Medical data distribution on Up-link & Down-link

This fact raises an important issue, that the Patient station requires much higher data rates at the Up-link connection compared to much lower bandwidth (data rate) at the Down-link connection. This issue is more problematic in the case of using low bandwidth channel, especially for the performance of GPRS link (171.2 Kbps maximum theoretical data rate).

Table 2.2 summarizes the different functional data rates and operating bounds of the OTELO system:

Table 2.2 Medical data types and data rates requirements of the OTELO System.

Medical Data Flow				Scenarios		
Medical Data Type	**Data Description**	**Data Rates Required (Kbps) (fps)**	**Data Flow**	**1 Voice contact. Receive & Store Images. Question / Answer**	**2 US Images Stream (Low Quality) Videoconf. (Low Quality) Receive & Store Images. Question/Answer**	**3 US Images Stream. (High Quality) Videoconf. (High Quality) Receive & Store Images. Question/Ans--wer**
Still US Images	Gray Scale, 512x512 Pxl.	14-97 Kbyte	Up-link P-to-E	T. time/ Image < 30 Sec.	T. time/ Image < 3 Sec.	
Still US Images	Gray Scale, 768x576 Pxl.	43-232 Kbyte	Up-link P-to-E		T. time/ Image < 7 Sec.	T. time/ Image < 1.2 Sec.
Stream US Images	Gray Scale, 176x144 Pxl.	7.5 fps	Up-link P-to-E		Approx. 45 Kbps	
Stream US Images	Gray Scale, 200x200, ROI	10 fps	Up-link P-to-E	Approx. 45 Kbps		Approx. 130 Kbps OR ↓
Stream US Images	Gray Scale, 400x400 Pxl.	25 fps	Up-link P-to-E			Approx. 384 kbps
Robotic Arm Control	100-200 Hz	9-10 kbps	Down-link E-to-P		4.5 – 5.5 Kbps	4.5 – 5.5 Kbps
	100-200 Hz	10-11 kbps	Up-link P-----E		4.5 – 5.5 Kbps	4.5 – 5.5 Kbps
Force Feedback	Frequency 100-200 Hz		Up-link P-to-E			< 2Kbps
Low Qlty. Ambient Video.	QCIF 176x144	5 fps	Up & Down P-----E E-----P	< 30 Kbps	< 30 Kbps	
Medium Qlty. Ambient Video	QCIF 176x144	7.5 fps	Up & Down P-----E E-----P			Approx. 35 Kbps OR
High Qlty. Ambient Video.	CIF 352x288	5 fps	Up & Down P-----E E-----P			Approx. 125 Kbps
Question / Answers	Text message	Text length	Up & Down P-----E E-----P	Text Length dependent	Text Length dependent	
Approximate Required Bandwidth		**Min.**		48 Kbps	94 Kbps	225 Kbps
		Max.		128 Kbps	184 Kbps	750 Kbps

None Real-time transmission
Real-time transmission

T. Time: Transmission Time, estimated.
P – to – E: Patient station to Expert station
P-------E: Patient station to Expert station and vice versa (E------P).

During the tele-echography examination, still and stream of Ultrasound images must be transmitted from the patient site to the expert site. Ambient video and sound (videoconferencing) have to be interacted between both sides, yet we know the whole operation is controlled by the robotic mechanism. Apart from videoconferencing and robot control data, the received ultrasound images will bring the expert relevant information to move the fictive probe and to control the distant probe-holder robotic system, in order to analyse different anatomical regions or to search for a region of interest.

These images will be transmitted via various communication links that have different available bandwidths (from 56 Kbps up to 384 Kbps). Very important studies therefore require adequately compressing the acquired ultrasound images, so that they can be sent through the chosen communication link, with the minimum quality loss in order for the medical expert to be able to propose a pre-diagnosis or to move the fictive probe to another desired location on the patient's body [48].

The focus of this section will be on choosing the ultrasound data acquisition techniques and compression standards adopted for ultrasound images and ambient video transmission. These will be implemented later in this study for data transmission on the OTELO system.

A- Ultrasound Images transmission:
For the OTELO system, six lossless compression methods are selected and three lossy types including the JPEG2000 standard. These techniques have been compared to determine which one would provide the best compromise between coding performance and image quality as studied by [51].

Results: The survey performed, was on 10 ultrasound images (size 768*576 pixels) to cover reasonable range of different scan locations for diagnosis purpose. These images have been acquired by an AU3 ultrasound scanner (ESAOTE) at a rate of 15 images per second and then digitised, the computing was achieved by a Pentium III with 450 MHz, under Windows NT. The results present an average quality measure of Peak Signal to Noise Ratio (PSNR) that compare lossy or losseless rebuilt images to the original ones of the database. A criterion of 30dB considered to be the lower value for a good PSNR. For a compression rate greater than 5%, JPEG-LS gave the best image quality, with regard to the PSNR and for a rate lower than 5%, JPEG2000 becomes the optimal method according to [51]. Overall JPEG presented the least successful results.

The study concluded that JPEG-LS gives the best compression rate in the lossless case (33% vs 80%). For a high compression rate, JPEG2000 also shows its efficiency. Experimental results performed on ten ultrasound images establish that the JPEG-LS technique seems to be the best lossless method for our tele-medicine application.

The suggested compression types, compression ratio, resolution and the distortion levels, which fulfil OTELO medical requirements for still ultrasound image transmission, are shown in the shaded areas of table 2.3 below:

Table 2.3 Still Ulltrasound images quality based on using JPEG2000.

OTELO	Compression	Sample Image requirement			
		Compression Ratio	Frame Size	Distortion	Image size (bytes)
Ultrasound image: with Medical information text	Uncompressed	0 %	768x576	-----	442.36
	Lossless	37 %	768x576	> 85 dB	276.26
	Near lossless	47 %	768x576	85 dB	232.53
	Lossy	90 %	768x576	35 dB	43.0
Ultrasound Image: without medical information text	Uncompressed	0 %	512x512	-----	262.14
	Lossless	62.6 %	512x512	> 85 dB	97.97
	Near Lossless	65.8 %	512x512	85 dB	89.65
	Lossy	94.4 %	512x512	35dB	14.5

B- Dynamic (Stream) Ultrasound image requirements:

The new standard JPEG2000 is a fully scalable (in spatial resolution and quality) still picture compression algorithm and is suitable for medical image coding. A novel medical image sequence compression scheme based on 2D Wavelet transform (lossy mode), is proposed by the author of [52]. The medical image sequence coder is endowed with Fine Granular Scalability (FGS) properties. The proposed video Coding scheme was experimentally evaluated for the transmission of a typical echogram image sequence (resolution 400 x 400 pixels, 25 fps).

The above video coding technique has compared with a coder based on the JPEG2000 image Coding standard. Specifically, every frame in the sequence is compressed independently using the JPEG2000 coder (no motion compensation is deployed) [52].

The bit-stream comprises of two layers: the base and the enhancement layer. It has been concluded that FGS guarantees a minimum acceptable video quality over heterogeneous networks and at the same time achieves visually lossless quality whenever there is a high-rate channel (1-2 Mbps).

C- Ambient Video coding standards:

H.263: The H.263 standard is based on the framework of H.261. Due to progress in compression technology and the availability of the high performance desktop computers at reasonable cost, International Telecommunication Union (ITU) decided to include more computationally intensive and efficient algorithms in the H.263 standard. The development of the H.263 has three phases. The technical work for the initial standard finished in November 1995. An extension of the H.263, nicknamed H.263+, was incorporated into the standard in September 1997, The results of the third phase, nicknamed H.263++, were folded into standard in 1999 and formally approved in November 2000 [51].

H.26L: The ITU-T Video Coding Experts Group (VCEG) has initiated the work on the H.26L standard in 1997. The end of 2001 witnessed the superiority of video quality offered by H.26L-based software over that achieved by the existing most optimized MPEG-4 based software. ISO/IEC MPEG joined ITU-T VCEG by forming a Joint Video Team (JVT) that took over the H.26L project of the ITU-T. The JVT objective is to create a single video coding standard that would simultaneously result in a new part of the MPEG-4 family of standards and a new ITU-T (likely H.264) recommendation. The first version of H.26L finalized in 2003. Table 2.4 shows a comparison between H263 and H26L with respect to Wavelet-based method:

Table 2.4 Subjective quality assessment of different video coding schemes.

Coder	Time efficiency	Quality
H263	Fastest	Good
H26L	Very fast	Best
Wavelet-based method	Fast	Very good

It has been concluded that H263 coding scheme shall be used for videoconferencing as we require the fastest coding time efficiency and highest quality video. Table 2.5 shows the possible frame formats and frame rates compressed by H.263 codec, which will be implemented by OTELO system for ambient video between the Expert and Patient stations, where in average 7.5 frames per second would be satisfactory:

Table 2.5 encoded video formats for communication over mobile channel.

Frame format	Frame rate (fps)	Image Quality	Applications	Throughput
Sub-QCIF	5	Low	Low-quality videoconferencing	20 kbit/s
QCIF (176 x 144)	5	Low/ Medium	Videoconferencing	30 kbit/s
QCIF (176 x 144)	7.5	Medium	General purpose TV/ film trailers	50 kbit/s
QCIF (176 x 144)	10	Medium/ High	Sport footage	100 kbit/s
CIF (352x288)	7.5	Medium	High quality videoconferencing	120 kbit/s

2.4 Experimental Set-up of the OTELO System

The following sub-sections present the detailed testing methodologies for implementing and evaluating the functional modalities of the OTELO clinical session scenarios over the 2.5G (GPRS) and the 3G (UMTS) wireless communication networks. The test will be based on two environments:

1. The OTELO system on GPRS network is performed in the London area, using the Vodafone GPRS operating network and 2.5G terminals supplied to us by Vodafone / UK.
2. OTELO system on UMTS network testing, by using off-the shelf 3G mobile phones with the collaboration of VF / UK.

OTELO functional modalities require four types of medical and robotic data to be transmitted synchronously or simultaneously between the Expert station and the Patient station:

a) <u>*Robotic control data*</u>: The round trip time for the transmitted signals between the Expert and Patient sides should be less than 350ms in order to achieve real-time interaction. The UDP, RTP or E-UDP (Enhanced UDP) suggested being use for End-to-End connection.

b) *Ultrasound still images:* no real-time transmission, and transferred in only one direction (Patient-to-Expert station).

c) *Stream of Ultrasound images:* to be synchronized with the movement of the Robot control data based on either; *move and wait* (GPRS case) or Real-time interaction (UMTS case).

d) *Videoconference:* Variable quality of videoconference interaction has to be performed, based on the available bandwidth and the users' requirements as shown in table 2.2.

The radio link specifications that OTELO system will work on:

GPRS network radio interface resources, which can be shared dynamically between speech and data services as a function of traffic load and operator preferences [53]. Channel Coding scheme 2 (CS2) is specified to allow bit rates of up to 53.6Kbps Downlink; to 12.4Kbps and 24.8Kbps Uplink (Patient-to-Expert station).

Or

The 3G Wideband Code Division Multiple Access (W-CDMA air interface) capable of bearing broadband services like Internet, Video, and other multimedia services, which should support a minimum data rate of 144 Kbps in all radio environments and up to 2 Mbps in low-mobility and indoor environments [54].

2.4.1 GPRS Experimental Set-up

GPRS overview: GPRS supports multi-user Network sharing of individual radio channels and time slots, thus radio resources (channels) are allocated based on demand. GSM contains 8 time slots per frame, therefore 1 to 8 time slots can be allocated (up link and Down-link independently) based on demand. When all eight times slots of GSM radio channel are dedicated to GPRS, an individual user is able to achieve as much as 171.2 kbps [55]. Chapter three presents an overall theory background on GPRS network.

To support packets switching, two new nodes are required to modify the existing GSM network to support GPRS radio packets:

Serving GPRS Support Node (SGSN): Keeps track of the individual MS's location, performs security functions and access control.

Gateway GPRS Support Node (GGSN): Provides inter-working with external packet-switched networks; Connected with SGSN's via IP-based GPRS backbone network.

On GPRS Network connection option, both system ends are connected to the network as follows:

- Expert side (fixed line: within LAN, ADSL or ISDN line) ⇔Patient side (**one** T68i Sony Ericsson GPRS mobile terminal, using Vodafone GPRS Network).

OR

- Expert side (fixed line: within LAN, ADSL or ISDN line) ⇔Patient side (**two** T68i Sony Ericsson GPRS mobile terminal, using Vodafone GPRS Network).

We have to bear in mind here, that all the proposed mobile terminals in the test procedure can be replaced with the equivalent Personal Computer Memory Card International Association (PCMCIA) wireless cards, with the same specifications of the terminals. Figure 2.5 shows the GPRS connectivity of the OTELO system:

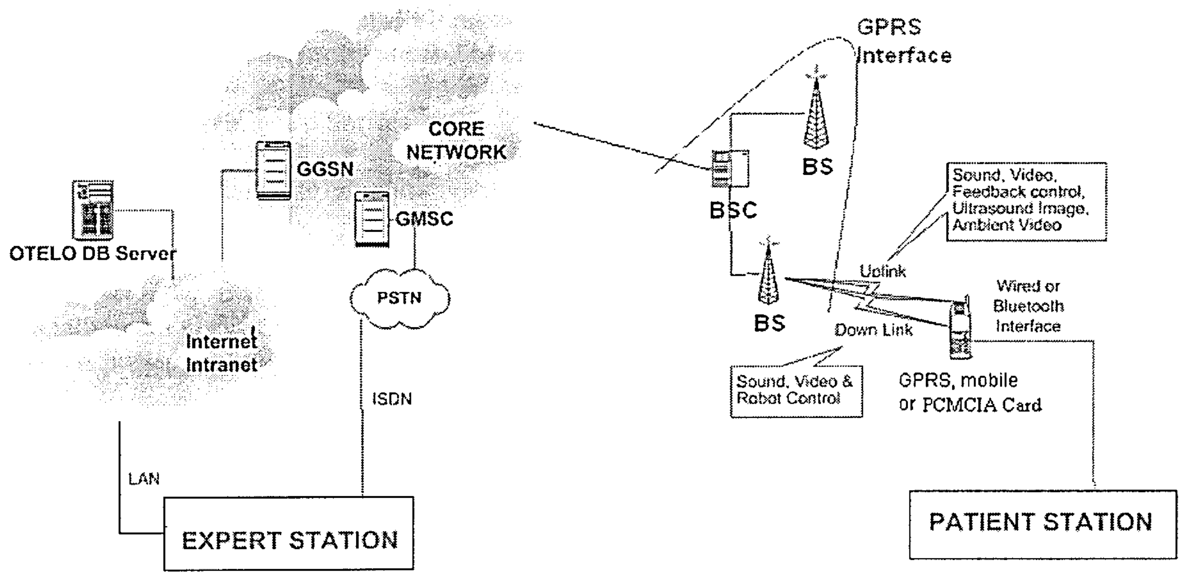


Figure 2.5 OTELO system design using GPRS Network.

Expert station; connected to the PSTN through ISDN line, Digital Subscriber Line (DSL). Or located within a Hospital or clinic and connected through a LAN to the Internet.

Patient Station; after integrating the Robotic arm, videoconferencing equipment and computer interface, the station connected to the GPRS network through either 2.5G terminal (GPRS terminal) hardwired to the computer interface or 2.5G PCMCIA card.

The GPRS test designed to study the performance of the system based on the following performance matrix:

- Data rates for Uplink, Downlink, and the asymmetrical performance.
- Packet delay and path losses
- Objective evaluation of the received data (Robotic control data, dynamic ultrasound and ambient video).
- Effectiveness and efficiency of the used compression techniques.

The service provider operating bandwidths of the GPRS terminal are as follow:

3+1 Channels, i.e. 3 Channels Download + one Channel Upload,
OR
4+2 Channels, i.e. 4 Channels Download + two Channel Upload

2.5G Hardware & Software requirements;

- The **Vodafone Sony Ericsson T68i** mobile terminals, provided by Vodafone / UK for the project research purpose, they were adjusted to work with the GPRS Coding Scheme two (CS-2), as discussed in more details in chapter three.
- USB Plug, connection to PC, ADSL High-speed line, and RJ45 end-to-end cables.

- Window 2000 Operating system.
- Communication management Module (Client - Expert Programme / OTELO1 prototype interface).
- J-Hospital Interface program (Kell registered software).
- Vodafone, ConnectMe software and Microsoft Netmeeting.

2.4.2 UMTS Experimental Set-up

The 3G link represents the wireless communication link that provides the real-time functional requirements of the system.

UMTS overview; Brief overviews on UMTS concepts are presented here for completeness, especially from OTELO's perspective. The detailed performance analysis of UMTS for OTELO system is presented later in chapter 4 of this thesis:

- High-speed access capable of bearing broadband services like Internet, video, and other multimedia services.
- Support of symmetrical and asymmetrical traffic.
- Packet-access. Most of the traffic in 3G networks originated from data communications. Therefore packet switched communications must be provided in addition to a circuit switched mode to ensure efficient resource usage.
- High flexibility to support new types of services. 3G systems will provide basic communications services, with applications and services being independent from the underlying transport layers.

UMTS network is fundamentally divided into three parts [54]:

1- CN, (Core Network)
2- UTRAN, (UMTS Terrestrial Radio Access Network)
3- UE, (user Equipment)

The supported theoretical bit rates by UMTS are [54]:

- 144Kbps / subscriber in rural outdoor radio environments.
- 384Kbps / subscriber in urban / suburban outdoor radio environments.
- 2048Kbps / subscriber in indoor / low range outdoor radio environments.

The test plan approached for the OTELO system is based on full end-to-end system performance over W-CDMA air interface. The proposal that we developed for the OTELO system is based on two connectivity choices:

Stationary Connection; *Fixed Expert station ⇔ Mobile Patient station,* and the hardware connection to the network as follow and seen in figure 2.6:

- Expert side (fixed line: within LAN, ADSL or ISDN line) ⇔ Patient side (Motorola A835, 3G mobile terminal).

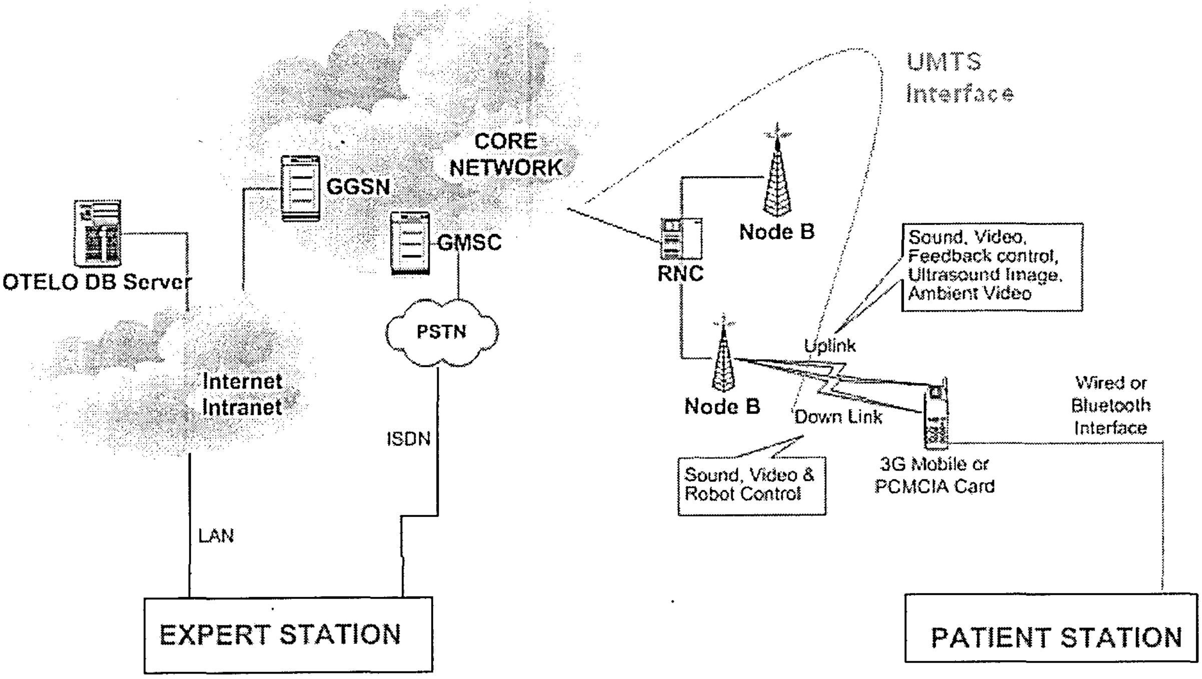


Figure 2.6 OTELO system design based on UMTS Network (stationary connection).

Mobile Connection; *'Mobile Expert station ⇔ Mobile Patient station,* and the hardware connection to the network as follow and seen in figure 2.7:

- Expert side (Motorola A835, 3G mobile terminal) ⇔ Patient side (Motorola A835, 3G mobile terminal).

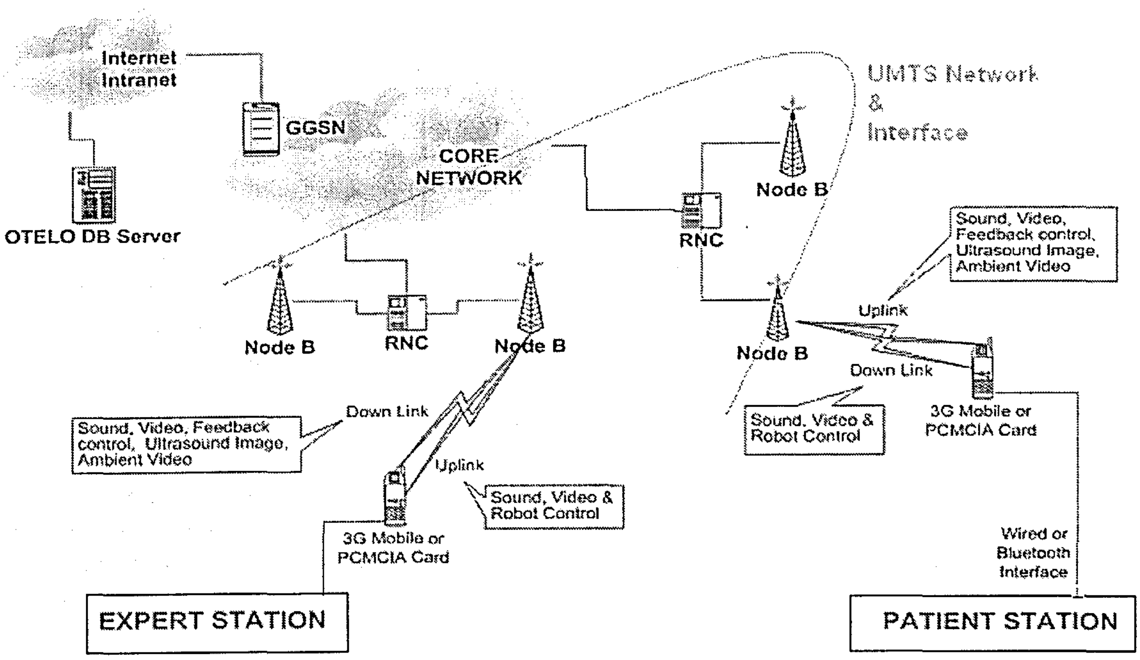


Figure 2.7 Full mobile OTELO system design based on UMTS network (Mobile connection).

3G Interface Requirements;

Expert Station: The expert station is connected to the public network as shown in figure 2.6 through:

A Broadband Modem connected to ADSL Line of the following bit rates:

- Downstream bandwidth = 1024 Kbps
- Upstream Bandwidth varying from 64 Kbps to 256 Kbps
- One Static IP address

These specifications are very convenient for the Expert side functional modalities.

Patient Station: USB Plug serial interface used as a standard for the 3G-phone connection to the Patient side PC (data transfer at up to 1.5 Mbps).

The 3G (UMTS) test is performed in the Lab site. That means the Robot function and full end-to-end stream ultrasound images transmission tested in the laboratory environment to validate the performance of the system based on the following matrices:

- Data rates for Uplink, Downlink, and the asymmetrical performance.
- Packet delay and path losses
- Average IP packet transfer rate
- Real-time interaction
- Peak and average throughput
- The issue of dynamic bandwidth allocation
- Patient station low mobility issues
- Objective evaluation of the received data (still images, dynamic ultrasound and ambient video)
- The effectiveness and efficiency of the used compression schemes.

Testing Hardware & Software;

- Motorola A835 (3G Phone) as seen in figure 2.7 (a), dual Mode W-CDMA 2100, Tri-Band (900/1800/1900 MHZ) GPRS, communicate with PC via:
- RS 232, IrDA, USB, Bluetooth and the support of the Wireless Application Protocol (WAP) for mobile Internet access.
- Window 2000 Operating system.
- Communication management Module, (Patient/Expert Programme, OTELO1 prototype interface, **Appendix D**).
- J-Hospital Interface program [113].
- Microsoft Netmeeting.

2.4.3 Land line and Satellite Communication Links

In this section, we briefly describe the OTELO system performance over ISDN and Satellite communication links studied by [35, 50]. The brief description is given in this section for comparative purpose with my research on wireless communication networks (GPRS and UMTS) that presented in the following chapters.

A- ISDN Link Based Design Performance:

ISDN is a telecommunication technology that allows the transport of voice and data on-demand. ISDN is a dial-up (not dedicated, but used on a call-by-call basis) digital connection to the telecommunication carrier. An ISDN line can carry information at nearly five times the fastest rate achievable using analogue modems over Public Service Telephone Network (PSTN). Basic Rate Interface (BRI) defines an ISDN digital communications line consisting of three independent channels: two bearer (or B) channels, each at 64 Kbps, and one data (or D) channel at 16 Kbps. The B channels are used for carrying the digital information, whether computer data, digitised voice, or motion video.

With appropriate equipment these B channels can be linked together to provide an aggregate 128 kbps data channel [50]. A clinical test has been performed between Expert and Patient stations located in two towns connected through the public network by ISDN lines. The figure below shows the networked stations:

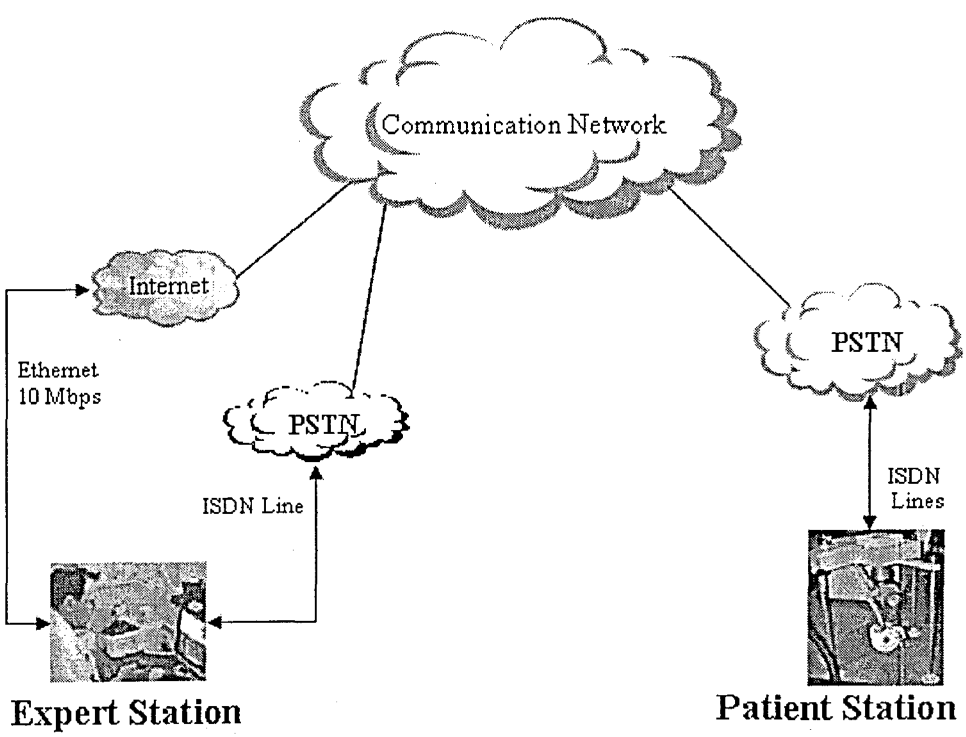


Figure 2.8 OTELO testing system using ISDN lines.

A technical validation of ISDN communication link was performed and showed that the tele-echography system is transparent to that link. The bandwidths were:

- 256 Kbps (4 ISDN lines)for Ultrasound and Ambient data and
- 64 Kbps (one ISDN line) for Robot control data.

US images format of CIF (352x288) for still images and QCIF (176x144) for the ambient video has been chosen to optimise the real-time transmission with the video Codec; H323 and H263 [52].

B- Satellite Link Based Design Performance:
Satellite communication is a technique, which has been used in telemedicine, often for educational purposes. Mobile satellite services (MSS) provide two-way voice and data communication from hand-held terminals, where the final link to the subscriber is by satellite [57]. The satellite used in this test called (Globalstar) provides mobile and fixed satellite based voice and data services using a network of Low Earth Orbit (LEO) satellites and state-of-the-art spread spectrum technology. Globalstar provides reliable voice and data services, to customers, routing calls through existing public and private telephone companies according to [58].

<u>Globalstar for the OTELO System:</u>
The figure below highlights the system's architecture using the Globalstar system. The secondary hospital (Patient Site) connected with the Clinical centre (Expert Site) through the Globalstar (Multi-Channel Modem) MCM-8 of (16) terminal. This terminal will simulate a Virtual Private Network between the two sites:

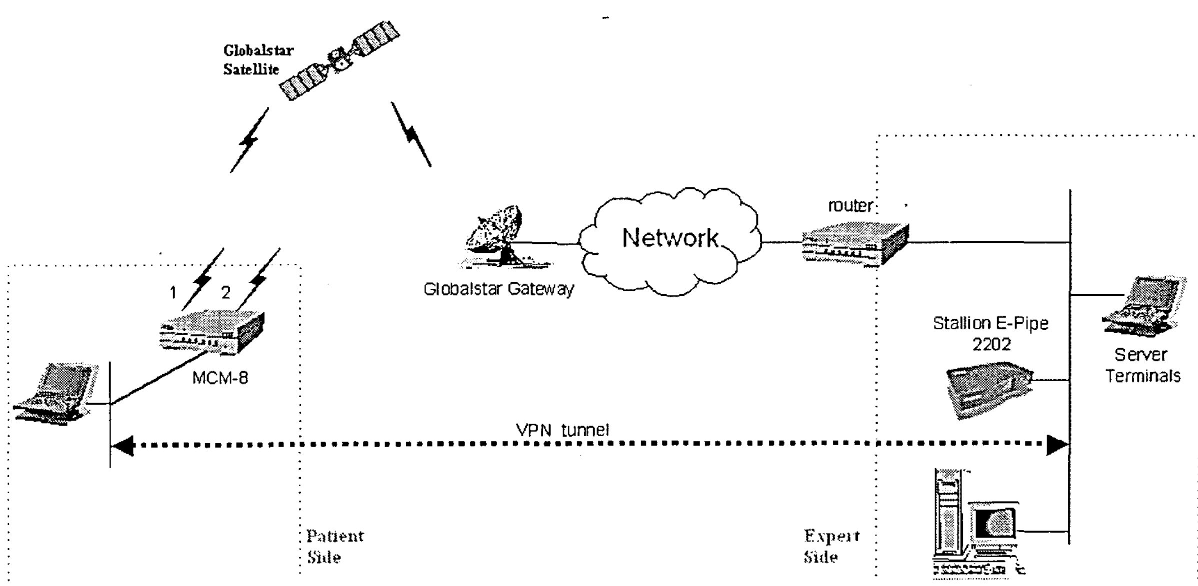


Figure 2.9 OTELO telecommunication architecture using the Globalstar System [57].

The VPN mode operates in conjunction with a Stallion e-Pipe 2202, which is located within the clinical centres private data network. The gross bandwidth of the VPN tunnel will be the aggregated bandwidth of all the satellite data modem connections that are concurrently established [58]. For example if all eight/sixteen Satellite Data Modem connections are established, the VPN tunnel provides approximately 64 (128) Kbps bandwidth (less any bandwidth that is being used at any instant, by any of the other modes used by the MCM).

When used in the VPN mode, each MCM is capable of establishing up to fourteen different VPN tunnels i.e. it may concurrently connect with fourteen different Stallion e-Pipe 2202.s within the clinical centre private data network. The establishment of a VPN tunnel for each MCM will allow the remote client terminals attached to the MCM to be considered as belonging to a remote sub-net within the clinical centre private data network. The terrestrial data network will probably be an ISDN (2 x 64 Kbps).

DSAT 2000 used for OTELO System:

EUTELSAT is currently deploying a new wide band Demand Assigned Multiple Access service named D-SAT 2000 ATM based on a technology developed by Lockheed Martin Global Telecommunications in the United State. The key concept of this new offering is the creation of an "open network", where each member can connect with everybody else, in an "on demand" basis as much as in a "leased line" framework. The D-SAT 2000 ATM can be positioned as far as the other services of EUTELSAT are concerned in the high end of the market. A possible network architecture using DSAT 2000 for OTELO is shown in figure 2.10:

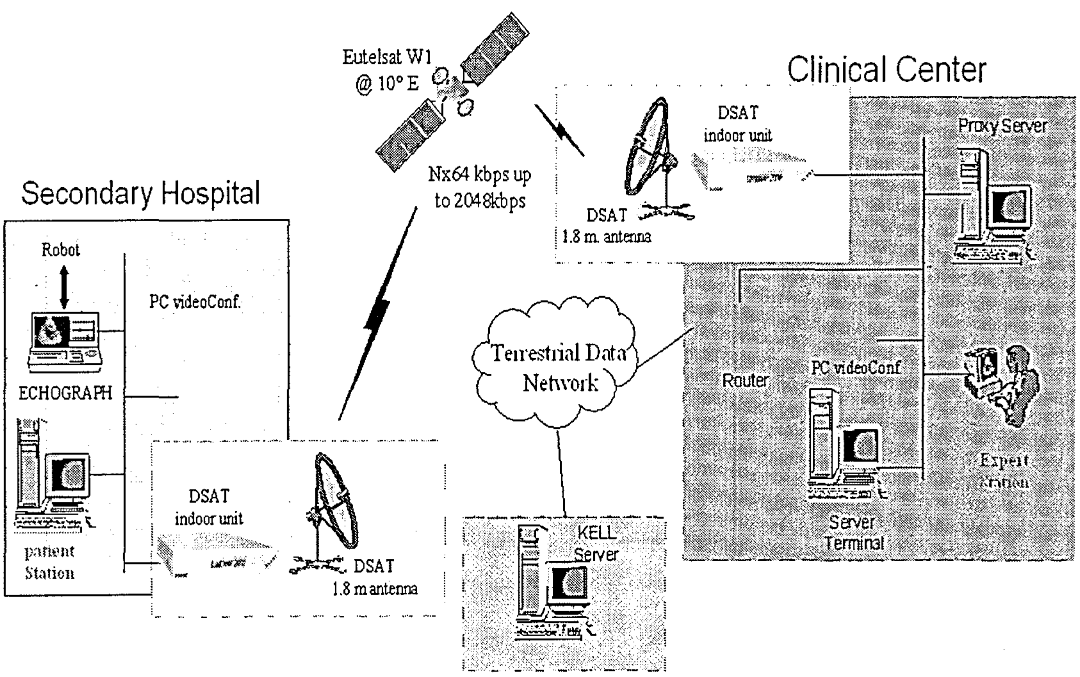


Figure 2.10 OTELO telecommunication architecture using DSAT 2000 system [35].

D-SAT 2000 ATM is a multi-service on-demand satellite system based on terminals providing ATM, frame relay, ISDN, and Signalling System 7 (SS7) interfaces for state of the art mesh network connectivity. The user terminal enable single-hop, mesh satellite networks with switched bandwidth-on-demand services for voice, video, data, and multimedia applications, unlike conventional VSAT systems, the system capacity is allocated for both circuit and packet-switched traffic in a dynamic fashion.

Multi-Frequency Time Division Multiple Access (MF-TDMA) is used to achieve very high efficiency and flexibility in satellite bandwidth management. Multiple-beam operation provides excellent opportunities for international network expansion. MF-TDMA is using to achieve very high efficiency and flexibility in satellite bandwidth management. Multiple-beam operation provides excellent opportunities for international network expansion.

C- TCP Issues over Satellite Link:

TCP applications typically do not run well over ANY link that has significant delay (i.e., greater than 100ms). This is often called the TCP-Window or TCP Bandwidth Delay Product (BDP) problem. It is inherent to all acknowledgement-based guaranteed protocols. The end-to-end Round Trip Time (RTT), as a window of data is sent every RTT when an acknowledgement is received, can determine the TCP throughput. With a window size of 8 KB, this places a limit on TCP throughput at 8 KB/RTT. For most terrestrial networks, this is not a limitation, as RTT<100 ms, so the throughput limit is ~ 640 Kbps, far greater than the line speed of most WAN connections.

For a geostationary satellite connection, however, with RTT = ~ 500 ms, the theoretical TCP throughput limit with 8 KB TCP windows is = 128 Kbps, which is often less than the satellite link data rate. It is important to note that this limit is only for a single TCP session connection, and any real network will have hundreds of TCP sessions occurring simultaneously [59].

D- Bandwidth and Delay limitations:
The transmission performance of the OTELO over Globalstar and DSAT 2000 shows high delay, although DSAT 2000 shows better performance with the 2Mbps bandwidth. Still DSAT as a geostationary system, has a very high delay, around 500 ms to transfer a bit of information. In figure 2.11 below a comparative calculation of transferring still ultrasound images using DSAT 2000 (2 Mbps), ISDN (384 Kbps) and Globalstar (MCM-16 at 128 Kbps):

		duration	GEO sat	duration	G*	duration	ISDN
Probe Movement	[ms]		0		0		0
Uncompressed Image	[ms]	9	9		9		9
Image Compression	[ms]	9	18		18		18
Image transmission Delay	[ms]	313	331	5000	5018	1667	1685
Propagation delay	[ms]	250	581	40	5058	50	1735
Image recover	[ms]	10	591		5068		1745

compressed image size	[KB]	80
GEO sat bandwidth	[Kbit/s]	2048
G* bandwidth	[Kbit/s]	128
ISDN bandwidth	[Kbit/s]	384

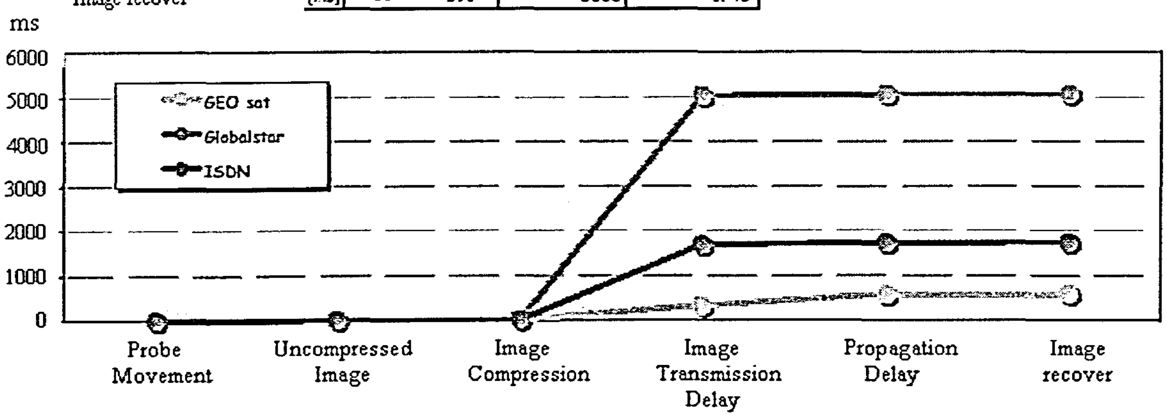


Figure 2.11 Illustration of different channel bandwidth and the achieved time delay [59].

It has been requested by OTELO operational modalities that the end-to-end delay between the Patient robotic probe movement and the reception of the video Image at the Expert station, should be as close as possible to 350 ms [50, 42].

2.5 Summary

This chapter presented the necessary tools in terms of the system connections layout over the 2.5G and 3G networks, and the functional modalities for the following chapters that will study in specific details the performance analysis of OTELO system. Additionally the chapter briefs the performance of the system over satellite link and land line.

CHAPTER 3

OTELO GPRS Performance Analysis

3.1 Introduction

In this chapter we present the performance analysis of OTELO system over GPRS network and its limitations for real-time operation. Section two summarise the basic architecture and relevant issues of the GPRS system, a brief overview of the relevant blocks and elements that are relevant to this thesis work are addressed. Section three describes the experimental testing bed and the network analysis tools used, followed by the performance analysis results of the tele-ultrasonography system conducted over both simulated and real GPRS network. Additionally the end-to-end delay analysis is addressed. Section four presents a comparative analysis of the simulated GPRS results in comparison to real network tests. Section five summarises the overall results and conclusions on the work of this chapter.

3.2 General Packet Radio Service (GPRS); Architecture and Overview

GPRS is the packet-switched data service introduced in GSM Phase 2+ (2.5G) [53]. Packet-switched services are much more suited to the bursty nature of data traffic than circuit-switched services. It is a new non-voice value added service that allows information to be sent and received across a mobile telephone network, it is an end-to-end mobile packet communication system which makes use of the same radio architecture as global system for mobile (GSM) communication [60].

GPRS uses the same resources as the basic GSM system, i.e. the Frequency and Time Division Multiple Access FDMA – TDMA radio interface respectively. A resources unit is a timeslot-frequency pair; however GPRS allows multi-slot operation within a single TDMA frame served from a mobile station. Also resource units can be allocated for Uplink and Downlink separately. GPRS channels allocate resources only for the time that data is ready to transfer and releases the resources immediately when they are not needed.

These results in efficient resource usage, multiple users can use the same resources in time multiplex [53]. Resources are used only when there is data to transmit, otherwise they can be used for other services or users. GPRS allows packet switching in the complete mobile network including the radio interface, and the hosts can be "always on".

3.2.1 Network Architecture

The classic GSM network does not provide sufficient capabilities for routing packet data. For this reason, the conventional GSM structure has been extended by introducing a new class of logical network entity called GPRS Support Node (GSN). The GSN nodes, shown in figure-3.1 below, manage interconnection with the other networks and perform a variety of functions, including subscriber management, billing and security, mobility managements, roaming and geographic re-routing, virtual connection control and packet transmission [61]. The figure below shows the OTELO sub-system equivalent parts and the possible location connections to the circuit or packet switched domain of the GPRS network:

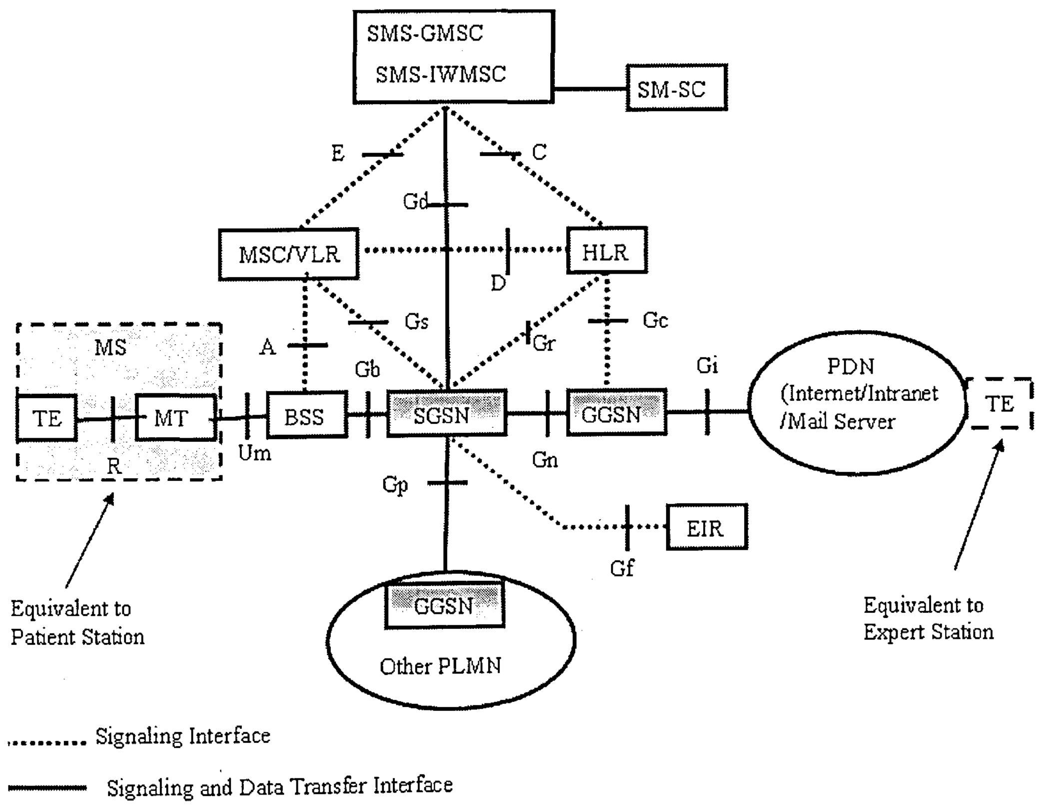


Figure 3.1 Block diagram of the GPRS Network Logical Architecture [61].

The GSN logical nodes implemented on the GSM structure through the addition of two network nodes [62]:

Serving GPRS Support Node (SGSN); is connected to the access network and is at the same hierarchical level as the switching centres Mobile-services Switching Centres (MSCs), and Visitors Locator Registers (VLRs). SGSN is the node that serves the GPRS mobile terminal, retaining location information and carrying out functions related to communication security and access control.

Gateway GPRS Support Node (GGSN); supports session management, charging and provides interconnect points to Internet Service Provider (ISPs). Packets arriving from the external networks are delivered to the GGSN in the GPRS network to which the mobile terminal belongs [61]. In the network architecture the following network elements can be identified:

BTS - Base Transceiver Station
BSC - Base Station Controller
MSC - Mobile Switching Centre
GMSC – Gateway MSC
HLR - Home Location Register
VLR- Visitor Location Register
EIR- Equipment Identify Register
SGSN - Serving GPRS Support Node
GGSN - Gateway GPRS Support Node
PTM-SC - Point-to-Multipoint Service Centre
PDN – Packet Data Network

***A- Radio Interface*:** GPRS is logically implemented on the GSM structure through the addition of GSN nodes, therefore several new interfaces have been defined as shown in figure 3.1 and they are:

a) Gb – Connects BSC with SGSN
b) Gn – SGSN – SGSN/GGSN (in the same network)
c) Gp – SGSN – SGSN/GGSN (in different networks)
d) Gf – For equipment querying at registering time
e) Gi – Connects PLMN with external Packet Data Networks (PDNs)
f) Gr – To exchange User profile between HLR & SGSN
g) Gs – To exchange Database between SGSN & MSC
h) Gd – Interface between SMS & GPRS

B- GPRS Channels:

a) PRACH – Packet Random Access Channel, uplink, used to initiate uplink transfer.

b) PPCH – Packet Paging Channel, downlink, BSC uses this to page the MS before downlink transmission.

c) PAGCH – Packet Access Grant Channel, downlink, resource assignments are sent on this channel.

d) PDTCH – Packet Data Traffic Channel, up & downlink, used to send data packets.

e) PACCH – Packet Associated Control Channel, Up & Downlink, used to convey signalling along with PDTCH.

3.2.2 The GPRS Protocol Stack

The GPRS Protocol architecture makes a distinction between the transmission and signalling planes. The transmission plane consists of a layered protocol structure providing user information transfer, along with associated information transfer control procedures (e.g. flow control, error detection, error correction and error recovery) [62, 53].

Figure 3.2 shows the protocol stack for the GPRS, and it consist the transmission/user and signalling/control planes:

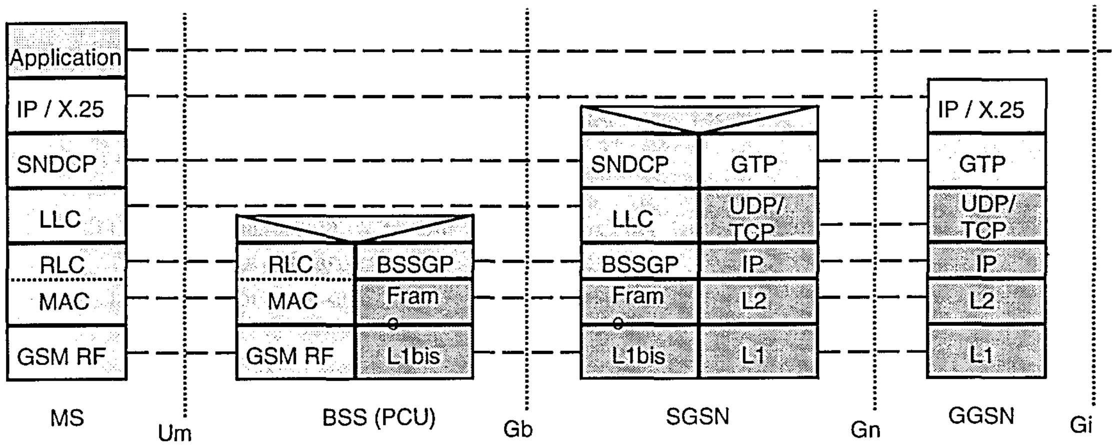


Figure 3.2 Protocol Stack for the GPRS Interface [63].

A- Sub Network Dependent Convergence Protocol **(SNDCP)**

- Convergence from different protocols to single link supported by LLC
- Multiplexing different sources onto one link
- Header Compression
- Data Compression
- Fragmentation of large packets

B- Logical Link Control Protocol **(LLC)**

- Establishes a link between Mobile station & SGSN
- It may work either in acknowledged or unacknowledged modes

C- Radio Link Control/Medium Access Control **(RLC/MAC)**

RLC

- Works in Acknowledge mode
- Using sliding window mechanism for flow control
- Uses Packet data Traffic Channel (PDTCH), 8 PDTCHs form a Packet Data Channel (PDCH), each PDCH correspond to one timeslot (TS) in GSM TDMA frame.

MAC

- MAC controls the access of a device to a given transmission medium.
- In GPRS, MAC is applicable to the air interface access, sharing and release of the physical medium [71].

3.2.3 The QoS Profile in GPRS Network

In this section we summarise the GPRS (QoS) issues as they are relevant to this work. QoS in packet-switched data transfer is defined in a different way to QoS in circuit-switched environment. In GPRS five QoS parameters are available [53], which are *service precedence, delay, reliability, peak throughput and mean throughput.* The users can select or negotiate the best suited QoS parameters for their applications [53]:

1- Service Precedence / Priority:

- Under normal network conditions, all users shall be served equally, however in case of network congestion; those users with a higher priority level will enjoy a privileged handling of their transactions opposed to lower priorities.

- Thus, a user with a low priority level may encounter higher delay times or even data losses in case of network overload.
- Three precedence levels are defined, High priority, Normal priority and Low priority

2 - Delay class: It is a requirement that the service delay for GPRS is competitive with existing data networks, both proprietary and standardized [64]:

- The delay class relates to the maximum delay times that a data packet may encounter while transported through the GPRS network.
- This delay does not consider delay times that are caused by effects from outside the Public Land Mobile Network (PLMN).
- Four delay classes need to be distinguished with '1' offering the lowest delay times and '3' bearing the highest risk for delays.
- Delay class '4' relates to 'best effort' which means that all transactions are handled according to the "first-in-first-out" principle.

Table 3.1 below shows the maximum delay values for each delay class:

Table 3.1 maximum allowed delay on GPRS network [53].

Delay Class	**Delay (maximum values)**			
	Service Data Unit (SDU) 128 Bytes		Service Data Unit (SDU) 1024 Bytes	
	Mean Transfer Delay (sec)	95 percentile Delay (sec)	Mean Transfer Delay (sec)	95 percentile Delay (sec)
1. Predictive	< 0.5	< 1.5	< 2	< 7
2. Predictive	< 5	< 25	< 15	< 75
3. Predictive	< 50	< 250	< 75	< 375
4. Best Effort	Unspecified			

3 - Reliability Class:

- Reliability relates to the probability of data loss, data corruption or out-of-sequence delivery of data packets.
- Five different reliability classes have been defined, whereas the differences among them are due to the data protection measures, being applied by the underlying GPRS protocols like LLC or RLC/MAC.

4 - Peak throughput:
The throughput specifies the expected bandwidth the user requires for a transfer. The peak throughput is measured in bytes per second. There is no guarantee that the peak throughput negotiated at the beginning is ever reached; only the mean throughput must be maintained during the lifetime of a connection. Nine different peak throughput rates are defined, offering transfer rates from 8 Kbps up to an impressive 2 Mbps [53, 78]. Note that the peak throughput classes are defined up to 2 Mbps, while the GPRS radio interface only supports a maximum of 171.2 Kbps [53]. In this study the maximum negotiated peak throughput class depends on the GPRS terminal used. In the real test, we will use a dual band, Class 10 GPRS Card.

5 - Mean throughput:
Is measured in units of bytes per hour and therefore presents an average value, and may be limited by the network even if more resources were available.

No less than 19 different mean throughput rates have been defined, ranking from 0.22 bps up to 111 Kbps, which equivalent to 100 bytes per hour and 50 Mbytes per hour respectively.

It is well known that GPRS like other wide-area wireless networks exhibits many of the following characteristics: low bandwidth, high and variable latency, Ack compression, link blackouts and rapid bandwidth fluctuations over time [65].

3.2.4 GPRS Coding Schemes

The GPRS Coding Scheme (CS) is outlined briefly here for completeness. Various radio channel CS are specified to allow bit rates from 9 Kbps to more than 150 Kbps per user [63]. GPRS introduce four channel coding schemes, namely CS-1 to CS-4, which allow for different user data rates per timeslot [53]. CS-1 is a coding scheme already used in GSM, CS-2 – CS-4 offer higher throughput rates but less protection against transmission errors.

Note that coding schemes CS-2 – CS-4 are applicable only for PDTCHs while information on all other PDCHs will be encoded using CS-1. The mobile station needs to be able to process all coding schemes while the network is only required to support Coding Scheme-1 [53, 66]. Depending on reception quality and error rate, the coding schemes can be dynamically adjusted during a transaction. Starting with CS-1, the used coding scheme may be changed to CS-2, 3 or 4 during a transaction [71].

Transmission Rates:
Coding in GPRS is always done for a single RLC/MAC radio block which always has a coded length of 456 bits. This block is interleaved over 4 normal bursts, each covering 114 data bits, hence the interleaving is much more reduced compared to circuit-switched data (interleaving over 19 bursts). The radio block before coding depends on the coding scheme and varies from 181 to 428 bits [53]. The coding schemes and achievable net user data rates for a few multi-slot combinations are summarized in table 3.2 below:

Table 3.2 GPRS coding schemes and net throughput for different TS.

Coding Scheme	Code rate	Uncoded radio block, payload (bit)	Coded Bits (bit)	Punctured Bits (bit)	Data Rates (Kbps)			
					1 TS	2 TS	4 TS	8 TS
CS-1	½	181	456	0	9.05	18.1	36.2	72.4
CS-2	≈ 2/3	268	588	132	13.4	26.8	53.6	107.2
CS-3	≈ 3/4	312	676	220	15.6	31.2	62.4	124.8
CS-4	1	428	456	0	21.4	42.8	85.6	171.2

Hence the GPRS radio interface can theoretically support up to 171.2 Kbps [53]. Throughout all the simulated works on GPRS network, time slots 1TS, 2TS and 4TS were considered based on CS-2, as shown by the shaded areas of the table above.

3.3 GPRS Simulation and Real Network Performance Studies

The works in this section describe the performance analysis of the OTELO system over GPRS network. The section describes the GPRS capacity to perform an end-to-end connectivity and transmission of the system's operational modalities and data.

3.3.1 Simulation Set-Up

The simulated studies were carried out by setting and configuring Wide Area Network Emulator (CLOUD emulator, **Appendix A**). This software environment is used to configure the connectivity of OTELO system option (mobile Patient station-to-fixed Expert station), for the **Uplink** studies. The Mobile terminals are classified according to the number of time slots they are capable of operating on, simultaneously [68, 65].

For example, most current mobile terminals are classified as (3+1). This means that they can simultaneously listen to 3 downlink channels (from base station to mobile), but only transmit on 1 uplink channel to the base station. Assuming CS-2 coding is in use; this corresponds to a maximum downlink bandwidth of 40.2 Kbps and uplink bandwidth of 13.4 Kbps. Hence the Emulator is set to the above parameters whenever possible to simulate the real operating GPRS conditions performed by Vodafone network.

Furthermore the setting of the Emulator upgraded to; '3+2' and '8+3' whenever necessary, as shown in table 3.2, to examine the limitations of the GPRS network in case of expanding the user data rates. A considerable number of parameters have to be considered for the WAN Emulator to match as close as possible the real GPRS network. Figure 3.3 shows the block diagram of the GPRS simulated environment:

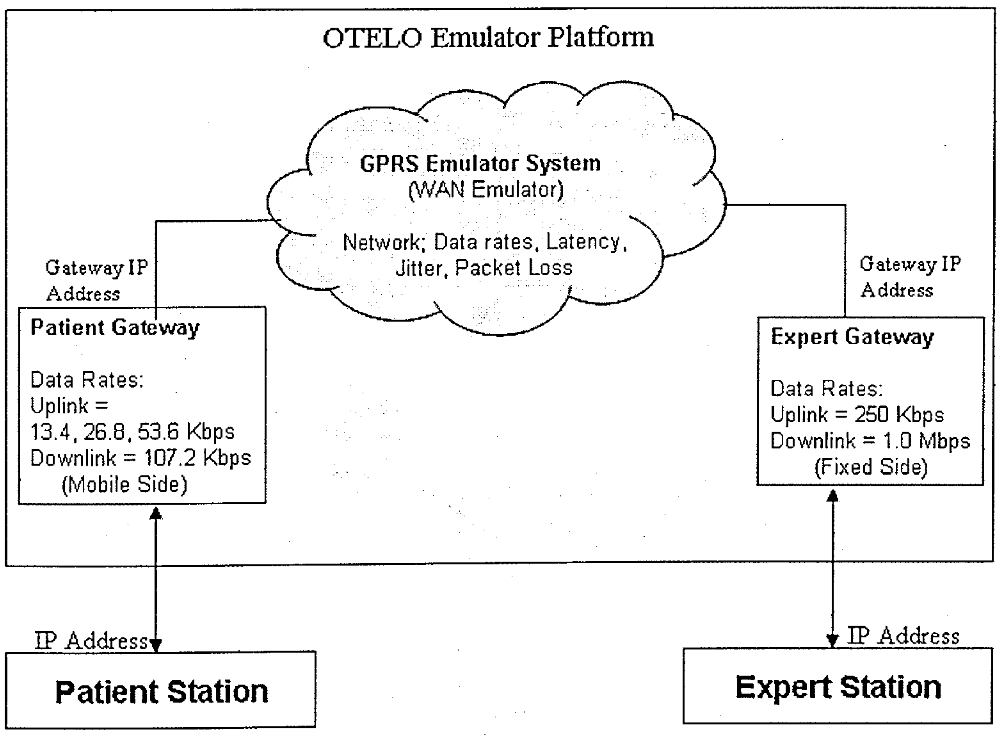


Figure 3.3 GPRS Emulation Set-up of OTELO system.

The OTELO system emulation based on the Lab environment, has an advantage of controlling the network operational conditions in terms of the load, number of users, data rate, bandwidth allocation, whereas with a real operating GPRS network the performance of the experiment depends on the status of the network with unknown number of users and on the service class used [53].

3.3.2 GPRS Test Bed and Network Settings

In this thesis the experimental works on OTELO system performed using the following GPRS mobile terminals (wireless Modems) supplied by VF, UK; i) Ericsson T68I, ii) Vodafone 2.5G PCMCIA card (**Appendix B**). The experiments conducted, were based on the present operating channel conditions of Vodafone network (3+1) channels as described in previous section. To provide the necessary IP Internet connectivity, the connection protocol to the service provider backbone network was Point to Point Protocol (PPP). The Virtual Private Network connection to the Expert station on the ADSL line was through the Internet. This will add further delay to the overall end-to-end delay on the mobile operation of the system.

In the OTELO system end-to-end connection over the standard GPRS network, the base station is linked to the SGSN, which in turn connected to a GGSN. In the current Vodafone configuration, both SGSN and GGSN nodes are collocated in a Combine GPRS Support Node (CGSN) [68]. During the testing cycle, no dedicated circuit switched voice call conducted to or from the GPRS terminal, as this should give it priority on the IP packets transmission [64].

Figure 3.4 shows the experimental end-to-end OTELO stations over the GPRS link. In this set-up the application data (only) path of mobile Patent station to the Expert station can be recognized as follow; Mobile station (Um Air Interface) to, BTS (A-bis Interface) to, Packet Control Unit (PCU) (Gb Interface) to, SGSN (Gn Interface) to, GGSN (Gi Interface) to Packet Data Network (PDN) then routing to an ADSL line to the Expert station:

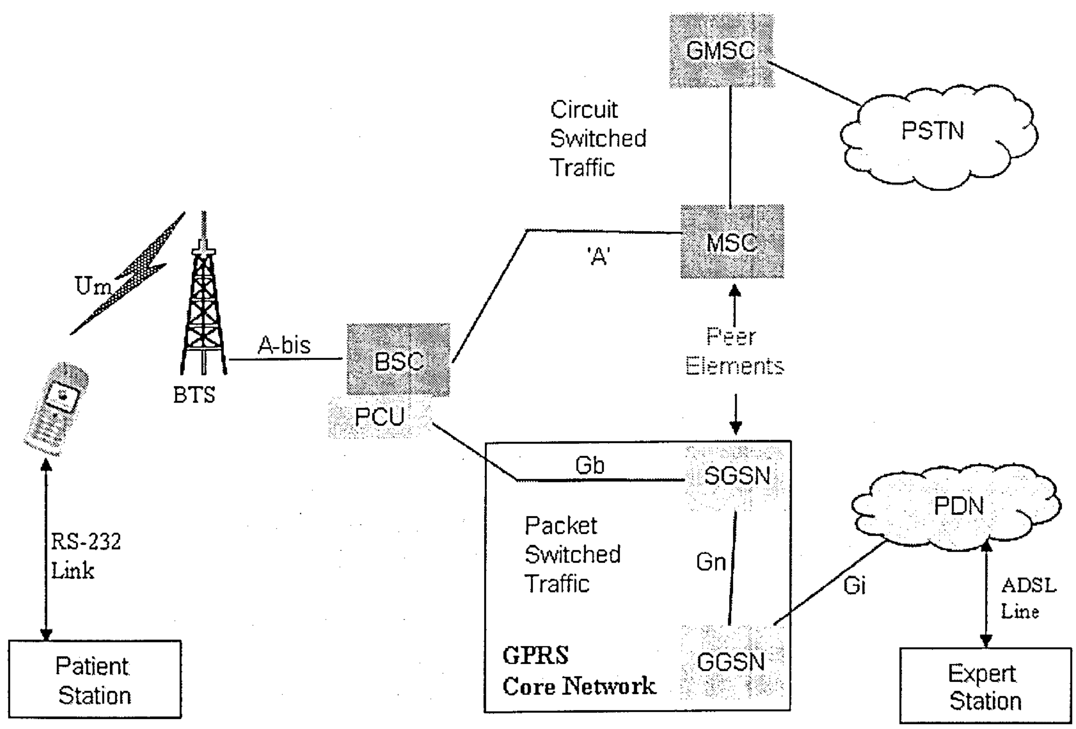


Figure 3.4 OTELO GPRS Network connectivity and experimental Set-up Environment.

The relevant performance analysis studies over the GPRS network were conducted in Laboratory environment. Software tools such as Ethereal and NetMonitor (NetMon) are used to capture the packets movement between the end points, depicting the protocols used (application layer, transport layer and link layer). Throughput, RTT, Transmission delay and other network performance parameters can then be derived (**Appendix C**).

It is well known that TCP protocol is dominating both wired and wireless networks; therefore we will use it throughout the test for Ultrasound still image transmission, bearing in mind that its operation over wireless link is not as reliable as in wired link [68], since TCP mechanism of retransmission depends highly on network congestion, while the case in wireless network, other parameters has to be considered like channel fading and interference [53].

The Microsoft Netmeeting model is also used to establish the system end-to-end connection, and the ultrasound images of table 3.3 above, were uploaded to the Expert station by running Netmeeting File Transfer Protocol (FTP) of the application layer. Generally the performance of the application protocols used in these measurements will be separated, whenever possible from the derived network performance parameters.

3.3.3 OTELO Medical Data

(i) Ultrasound Still Images: Samples of uncompressed ultrasound images acquired from ultrasound medical test, with image size ranging from 256 KB to 470 KB of *.pgm (Portable Graymap) format. These are compressed over wide range (losseless and lossy modes), using JPEG2000 codec (JP2 – format).

These images are transmitted from the Patient station to the Expert station (Uplink direction), as a group of ten images each time. Table 3.3 shows the characteristics details of the US medical images used in the studies:

Table 3.3 10 Ultrasound image compressed at different compression ratios.

Group No.	Sizes of files	Bit/Pixels	Compression Ratio
1	1196 KB	3.0	03:1
2	994 KB	2.0	04:1
3	546 KB	1.0	08:1
4	381 KB	0.7	11:1
5	272 KB	0.5	16:1
6	163 KB	0.3	27:1

(ii) Ultrasound Stream: From tables 2.2 and 2.5 in chapter two, the sample ultrasound streams of different frame formats, quality and frame rates have been implemented and mostly compressed and decoded by H263 codec at both system ends. A new scalable coding mechanism, called Fine Granularity Scalability (FGS) was proposed for compressing the ultrasound acquired data for OTELO Patient station, nevertheless during the time of the test it was not integrated in the system.

(iii) Ambient Video: From table 2.5, these represent low quality 'videoconferencing' required for OTELO function. Videoconferencing data were transmitted simultaneously with the ultrasound stream, bearing in mind the expected performance limitations of the GPRS network.

3.3.4 Performance Analysis of the Medical Ultrasound Still Images

During the data transmission (Ultrasound still images) in OTELO system, voice data transmission from the Patient to Expert station assumed to be very low or not used due to the limitation of GPRS channel. Therefore the voice channel is deactivated to reduce loading the voice IP packets over the communication channel. A brief default and setting parameters of the Emulator set-up are shown in table 3.4 to turn the platform as close as possible to real GPRS network:

Table 3.4 Set-up parameters for GPRS simulation.

Parameters	Default and assigned value
TCP header length:	20 bytes
IP maximum packet size:	65 536 bytes
IP header length:	20 bytes
LLC maximum packet size:	1500 bytes
LLC mode:	unacknowledged
LLC header, and check length:	3 bytes
Number of users:	single
Latency (uniform distribution latency)	300 ms – 450 ms
Congestion:	none to Low
Packet Loss	Zero – 2 %
Coding Scheme:	CS-2
Uplink channels TS:	1, 2 and 4
Downlink channel TS:	3, 8
ADSL side data rates	Downlink = 1024 Kbps, Uplink = 256 Kbps
BER:	variable, (2.00E-05 to 1.00E-07)
Data source size:	variable, 1196 Kbytes to 163 Kbytes

However, it is well known that TCP/IP performance tend to degrade over wireless links, where losses are mostly non-congestive and predominantly due to external environmental factors such as fading, interference etc. Wireless networks also suffer from a number of other discrepancies [65, 69]. Hence the TCP performance has the following disadvantages:

- A sluggish slow-start that takes many seconds (due to high RTTs) for the window to ramp-up and allow full link utilization.
- Excess queuing over the downlink can result in gross unfairness to other TCP flows, and a high probability of timeouts during initial connection request,
- Spurious TCP timeouts due to occasional link `stalls' and,
- Slow recovery (many seconds) after timeouts.

Fundamental IP service-performance metrics have already been defined for verifying the quality of packet networks [70]. These are throughput, packet loss, latency and jitter, and can be applied to the performance analysis of transmitting ultrasound still images over GPRS networks. As mentioned in chapter two, 10 differently compressed ultrasound images is used in this test. This proposed number of images is to fulfil many requirements from the analysis point of view; First; during the practical medical session the Patient station will send around this number of images to the Expert station in nearly each transmission session, Second; investigating and analysing this number of images transmitted and received sequentially is much more convenient to probe the network behaviour in terms of time delay and general network reliability than sending single images.

There are two basic end-to-end performance measurements that characterise the expected data service performance; throughput and RTT [71]. Throughput is the number of bytes per second that can be sent through a GPRS connection. Hence this can be considered an important parameter for IP users, the higher the throughput, the faster the IP packets can be downloaded [70]. In a good network condition the throughput per time slot for CS-2 coding is less than 12 Kbps, this represent the overhead introduced by upper layers and the GPRS procedures. The application throughput per TS is including the overhead introduced by the LLC layer, when no data compression used [71].

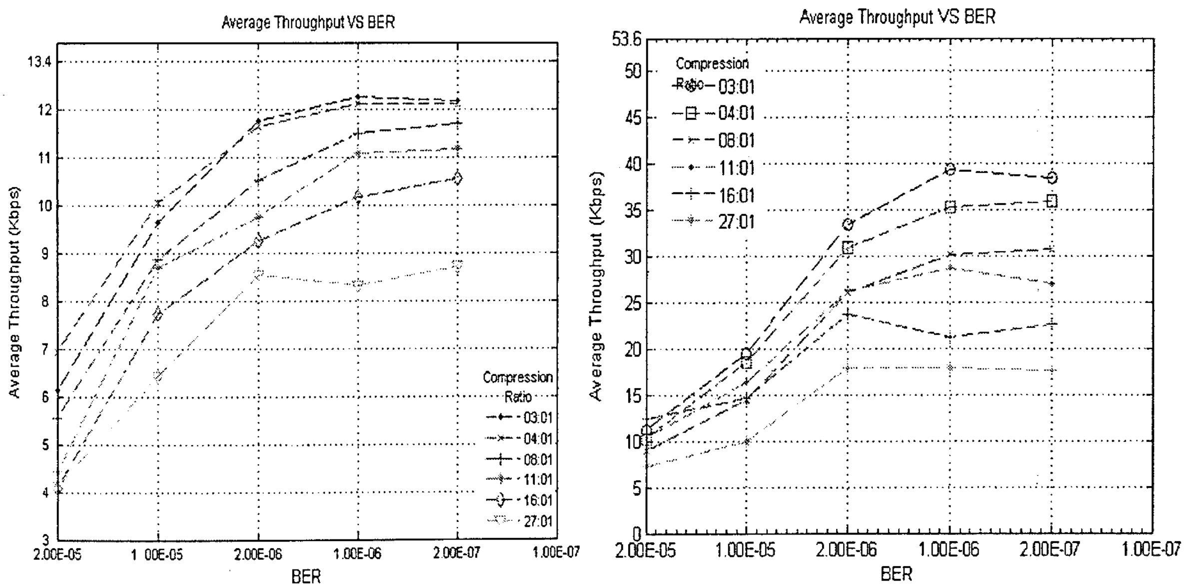
(i) GPRS Throughput Performance:

In order to determine best-effort data with a QoS criterion, it is necessary to define the maximum utilisation of free TS so that the amount of TS multiplexing measured in terms of reduction factor

[71], is such that a minimum quality criterion is met. Considering average user throughput as QoS criterion, the following equation can be used to check whether a certain amount of free TSs fulfils the QoS criterion (Reduction Factor = 99%), when no significant timeslot multiplexing take place [71, 53]:

$$\text{Average User Throughput} = \text{Number of TSs} \times \text{TS Capacity} \times \text{Reduction Factor} \qquad (3.1)$$

To approach a realistic average throughput rate for the Patient station Uplink direction, we consider the net TCP packets throughput captured for the transmitted data by the Patient side, and not the total traffic seen by network analyser called (Ethereal), which is capturing the uploaded and downloaded traffic movement. Efficient channel usage as can be seen from Figure 3.5 depends on the level of Compression Ratio (CR) and Bit Error Rate (BER):



(a) (b)

Figure 3.5 Simulated Average Throughput performances in terms of BER at Uplink channel data rate of, (a) 13.4 kbps (b) 53.6 Kbps.

The throughput performance found to be below the maximum link bandwidth, where the TCP uses loss as an indication of congestion, and hence a signal that it should halve its congestion window [68]. The throughput model of a TCP connection, specifically the throughput can be characterized by the following formula [72]:

$$T_{TCP} = \frac{1.22 \times MTU}{RTT \times \sqrt{P}} \qquad (3.2)$$

Where:

T_{TCP} Throughput of a TCP connection;

MTU (maximum transit unit) is the packet size used by the connection;

RTT Round-Trip Time for the connection;

P Packet loss ratio experienced by the connection.

It is sensible to assume that even for best-effort services, there is certain minimum QoS that must be fulfilled (this is required already by some higher-level protocols like TCP), QoS criteria are specified in terms of minimum throughput and maximum delay achieved by 90% of the throughput and delay samples [71]. The *(MTU)* transmitted over the simulated connection was 1514 Bytes, these include TCP header of 20 Bytes. The *RTT* fluctuations depend on the number of the TS implemented and the transmitted still image size as can be seen later. Moreover multiplying the time slots is not necessarily multiplying the performance of the channel on transmitting compressed Ultrasound Images, although in general TCP delay decreases as the number of time slots used increased. From figure 3.5, we can derive the utilization efficiency of the bandwidths assigned, at lowest BER (2×10^{-7}):

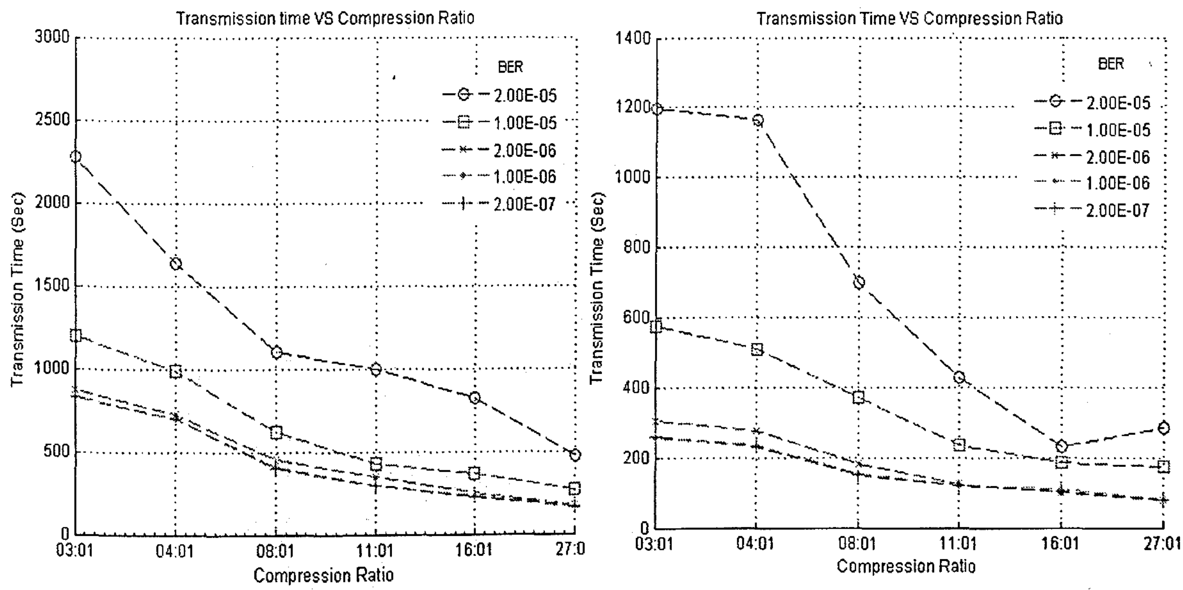
67% to 85% at single time slot (13.4 Kbps) and
35% to 72% at multiple time slot as in (b) 53.6 Kbps

A further indication on influence of the Compression Ratio on the throughput in both cases of number of TS assigned; the higher the CR, the lower is the average throughput. The advantage of multiplying the timeslots on the average throughput and channel utilisation, can be summarised in a comparative table for both data rates setting conditions, at BER = 1×10^{-6}:

Table 3.5 Throughput and channel capacity performance at 13.4 Kbps and 53.6 Kbps.

Compression Ratio (CR)	Average Throughput (Kbps)		Channel Utilisation (%)	
	At 13.4 Kbps	At 53.6 kbps	At 13.4 Kbps	At 53.6 Kbps
0.3	8.32	17.97	62	36
3.0	12.26	39.4	91	73

This degradation in performance against the expected throughput after increasing to 4 timeslots is probably due to the maximum window size of the TCP protocol is being 64132 Bytes (ranging in Ethereal from 17520 – 64132 Bytes). As compared for the same transmitted images over one or four timeslots and this window size should be enough to operate with 4 timeslots [53]. It has been shown in few applications that doubling the timeslots also doubling the average throughput, although adding time slots up to (8 TS) does not necessarily increase the throughput with the same proportion [53]. In conclusion, it is clear that the GPRS transmission time needed for the medical still image is strongly dependent on CR, BER and channel data rates, as shown in figure 3.6. It is obvious that BER variation has the highest influence on the time needed for each file transmission process:



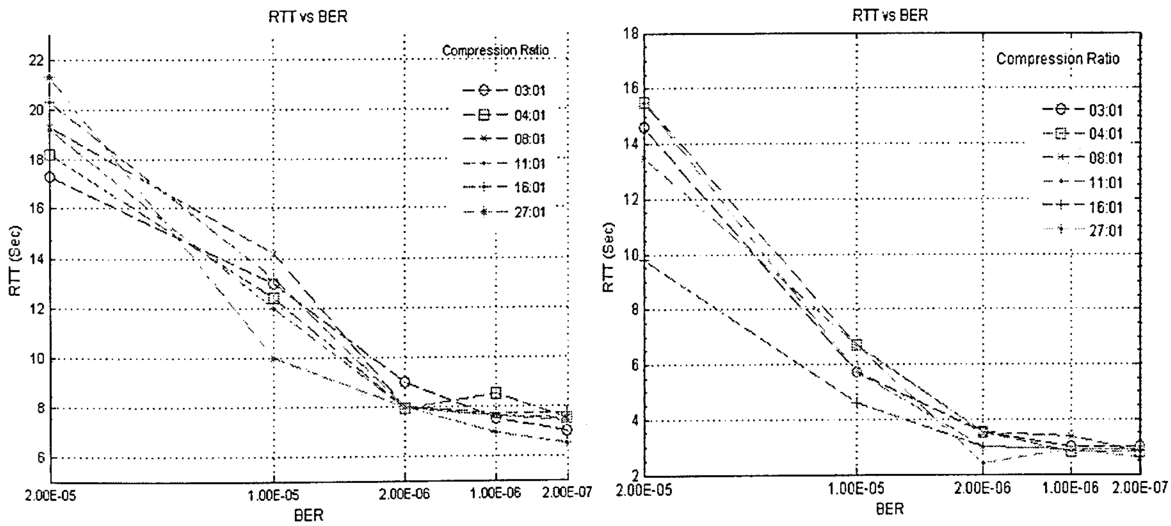
(a) (b)

Figure 3.6 Simulated performance of the transmission time at different BER in terms of different compression ratios at Uplink channel data rate of, (a) 13.4 Kbps, (b) 53.6 Kbps.

The effective range of BER is between 2 x 10^{-5} to 2 x 10^{-7}. The transmission time of images compressed at CR = 08:1 is about 625 sec; Patient-to-Expert station at BER = 1 x 10^{-5} as in figure 3.6 (a), i.e. in average 62.5 sec for each image, whilst this was reduced to about 360 sec at the same BER, i.e 36 sec per image as in part (b) when the uplink TSs increased to 4 TS. At low CR, the influence of low BER is clearly minimised the transmission time needed for the same CR, due to the fact that with small image size, TCP requires less number of IP packets transmission, which will then suffer less retransmission probability as a function of packet loss on congestion happening on wired network.

(ii) GPRS Latency and RTT Performance:

Round Trip Time is defined as the time since a short packet is sent from the Mobile Station, received by the application server and finally received back at the Mobile Station. It characterises the end-to-end latency, which is important for time-critical application [71]. As in figure 3.7 nearly different compression ratios sharing the same RTT although the BER decreases to more practical values, that the real GPRS network is operating on:



(a) (b)

Figure 3.7 Effect of both BER and the CR on the RTT of the transmitted packets with the channel Uplink channel settings of, (a) 13.4 Kbps and (b) 53.6 Kbps.

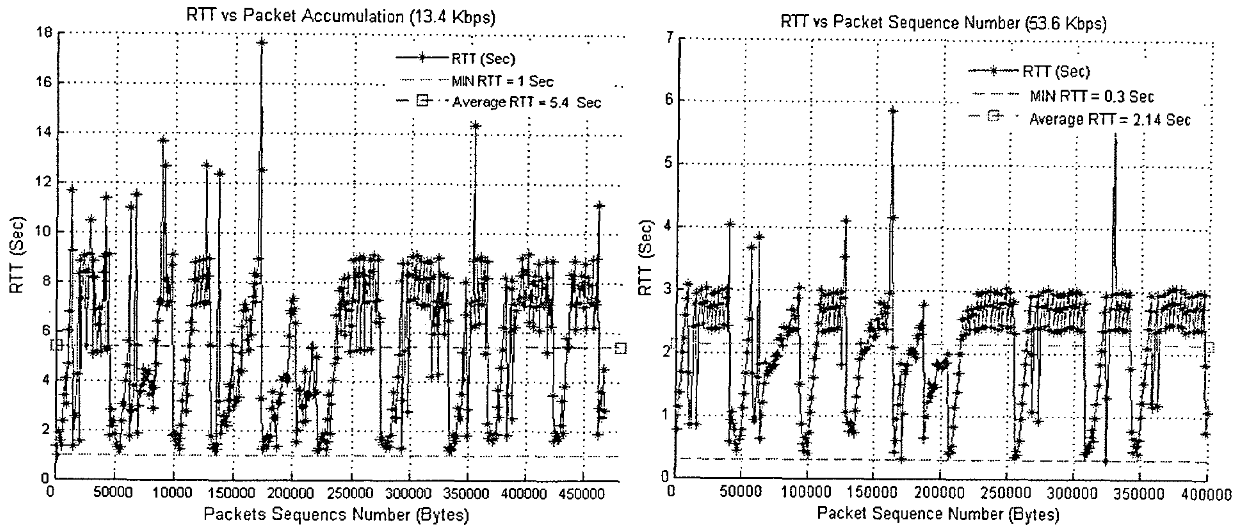
Some other literatures define the RTT experienced by a connection, as the sum of the uplink and downlink distributions [68]. Figure 3.7 shows that at BER = 2 x 10^{-7} the RTT variation range is about 6.5 to 7.7 seconds as in part (a), whilst in part (b) the range become narrower and the RTT is about 2.6 to 3 sec as the channel data rate increased.

To calculate the maximum RTT encountered by the communication channel, without taking in to consideration the wireless part of the link, a standard TCP maximum windows size of 65536 Bytes gives maximum RTT as follow:

$$RTT_{\max}(Sec) = \frac{Windowsize(Bytes)}{Bandwidth(Bytes/Sec)} \qquad (3.3)$$

Higher values of congestion window (cnwd) leads to higher values of RTTs [68], these values incrementing follows the slow start mechanism of the TCP to probe the network condition. Additionally as in figure 3.7, higher compression ratios does not have significant reduction in RTT values especially at improved channel conditions.

The OTELO GPRS simulated studies conclude that the measurements of RTTs values is > 1.2 sec and exceeding the average RTT shown in figure 3.8 (a) to about 12 sec in bad connection conditions. Due to proportionally lower congestion window and higher bandwidth, a reliable complete transmission of US image from the slow start to maximum RTT of around (3 Sec) as in part (b) of figure 3.8 is achieved. This can be shown if for example the packet sequence 200000 – 250000 Bytes is taken from part (b):



(a) (b)

Figure 3.8 RTT measurements at the Patient station, Uplink channel; (a) 13.4 Kbps (b) 53.6 Kbps.

In the GPRS system, the start of the connection allows the sender to adopt the slow-start algorithm to probe the available bandwidth. The slow-start stops right after a congestive packet loss happens. After slow-start, the sender adjusts its sending rate based on the congestive packet loss ratio, RTT and Retransmission Time Out (RTO) [73].

When the TCP throughput drops to a very low values, the packets RTT fluctuate sharply as seen, and this is reflecting the large delay variation introduces by the wireless channels [73]. Therefore choosing a single packet can not accurately reflect the variation in RTT. The results of the analysis above are comparable to a high and variable latency observed over GPRS network, as high as 600 ms to 3000 ms for the downlink and 400 ms-1300 ms on the uplink. Round-trip latencies are 1000 ms or more [65].

We can derive a table concluded from the above performance plots for the communication channel at BER = 1.00E-06, and for both channel bandwidth to show the average transmission time and average throughput at different compression ratio. The table shows the values for each image in each case:

Table 3.6 Compression performance of Transmission time and Average throughput of the Still images.

Image size Bytes / Uncompressed 416 KB	Compression Ratio CR	1 timeslot = 13.4 Kbps		4 timeslots = 53.6 Kbps	
		Transmission Time Sec.	Average throughput Kbps	Transmission Time Sec.	Average throughput Kbps
119 KB	2.7:1	83	12.2	25.5	39.4
99.4 KB	4.0:1	69.7	12.1	23.7	35.3
54.6 KB	8.0:1	40.6	115	15.5	30.2
38.1 KB	11.4:1	29.7	11.1	12.1	28.8
27.2 KB	16.0:1	23.1	10.1	11.1	21.2
16.3 KB	26.7:1	17.3	8.30	8.1	17.6

Other interesting area of analysis is to see the TCP packet accumulation at the Patient station and its corresponding accumulation behaviour when received by the Expert station. The time-sequence graphs (tcptrace) in figure 3.9 shows the TCP packets sequence and distribution over the transmitted time, where it can be found that the event of retransmission because of packet loss or congestion is precisely with lower data rates.

Figure 3.9 shows the TCP sequence of packets transmitted over channel data rate of 13.4 Kbps, (a) shows the behaviour of packet transmission at the Patient side, while (b) shows the received packets at the Expert station:

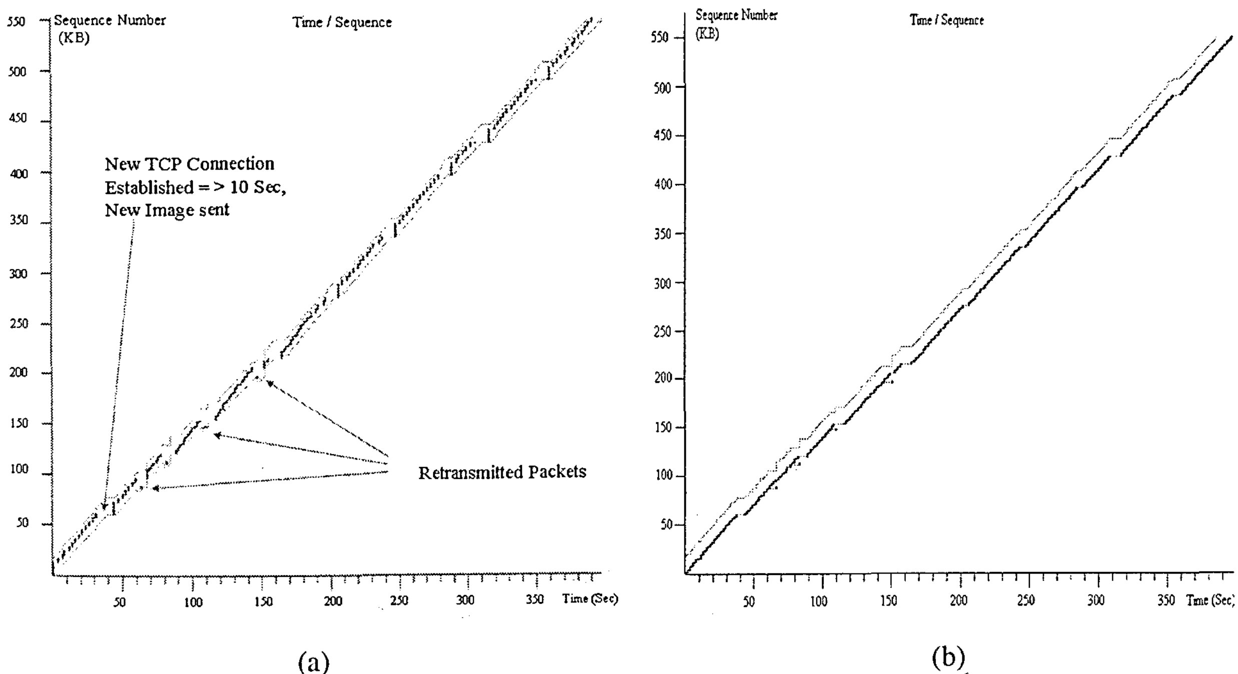
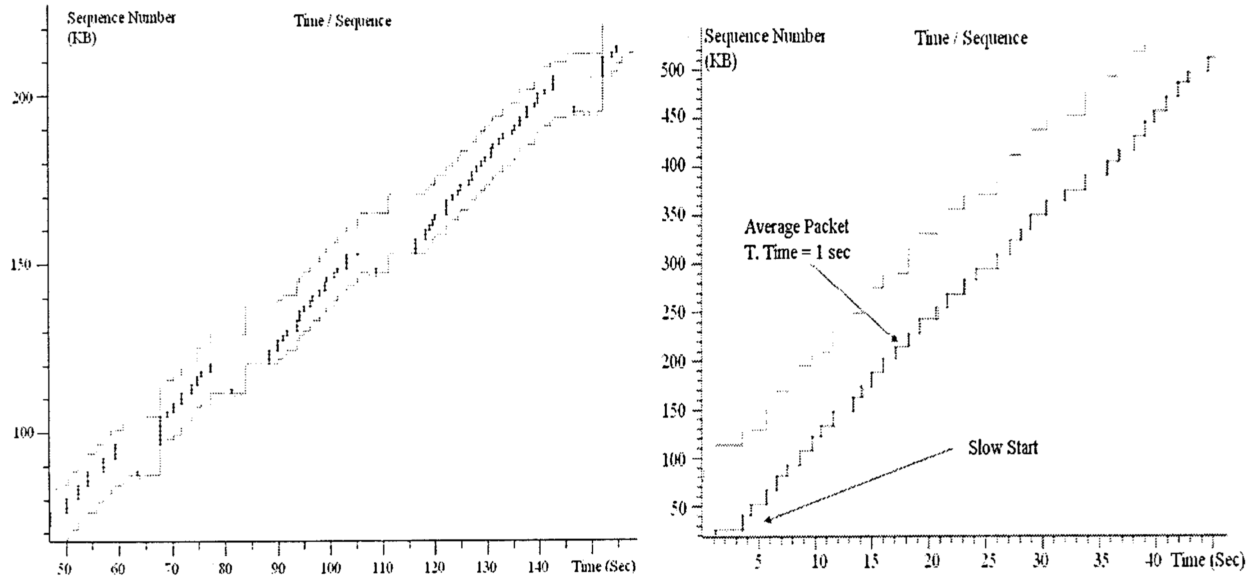


Figure 3.9 Accumulation of the transmitted packets at f CR = 8:1 and BER = 1×10^{-6} versus the time of the entire connection at 13.4 Kbps Uplink channel. (a) Patient side, (b) Expert side.

It can also be noted the zero throughput time length taken to establish new connection for the next image file (10 images transmitted in each group) furthermore we can estimate the time taken for each image fully received by the Expert station. The majority of the retransmitted packets suffer from single retransmission and about 3% suffer from duplicate retransmission and in worse link condition, retransmission may occur for third time. Also RTT can be predicted from figure 3.9 (a), where it has been proved in [53] that the radio interface (in our case GPRS like air interface) seems to have only minor influence on the delay of the packets.

Results from our link characterization measurements indicate that losses are relatively rare. However, some losses do occur, and can cause TCP problems [68]. Figure 3.9 (a) shows such cases in which a retransmission observed. The main point to note is the large amount of time (> 10 sec) it takes TCP to recover from the loss:

Zooming in the figure above shows a close look of the first few seconds of a connection under slightly worse radio conditions as in figure 3.10. In part (a) the time taken for retransmitting packets is around (8 sec) in average, furthermore it shows the recovery time taken by the TCP mechanism to resume sending packets burst:



(a) (b)

Figure 3.10 Simulated retransmission events, the sample is at CR = 8:1and BER = 1 x 10^{-6}, (a) Patient side, (b) Expert side.

For the TCP connection to fully utilize the link bandwidth, its congestion window must be equal or exceed the Bandwidth Delay Product of the link [65]. We can observe from (b) of figure 3.10 that the slow start in the case of our setting condition took about 5 seconds to ramp the congestion window up to a value of link BDP from the time the initial connect request (TCP's SYN) was made. A further point to note in figure 3.10 (a) is that the sender releases packets in bursts in response to groups of four ACKs arriving in quick succession. Receiver-side traces show that the ACKs are generated in a smooth fashion.

3.3.5 GPRS Performance Analysis of Ultrasound Streaming

In this section we study the performance of the Ultrasound Image stream transmission in terms of the following performance matrix:

a) Average throughput (Ultrasound stream and Robot control data)
b) End-to-end packet delay
c) Frame delay
d) Jitter

Ultrasound stream of 13 fps down to 5 fps were used with the following resolutions; CIF and QCIF. Robotic data flow; is bursting from the Patient station at 16 Bytes (payload) packet size of 70 ms interval time.

Implementation of Streaming:
It is well known that the protocols for streaming media are designed and standardized for communication between clients and streaming servers. They provide such services as network addressing, transport, and session control [72]. According to their functionalities, the protocols can be classified into three categories:

1. Network-layer protocol such as Internet protocol (IP)

2. Transport protocol such as user datagram protocol (UDP)
3. Session control protocol such as real-time streaming protocol (RTSP).

In this section we focus on the Transport protocol family as these govern the overall application performance analysis. The transport protocol family for media streaming includes UDP, TCP, RTP, and RTCP protocols. UDP and TCP provide basic transport functions while RTP and RTCP run on top of UDP/TCP. UDP and TCP protocols support such functions as multiplexing, error control, congestion control, or flow control. These functions can be briefly described as follows [74, 75]:

a) UDP and TCP can multiplex data streams from different applications running on the same machine with the same IP address.

b) For the purpose of error control, TCP and most UDP implementations employ the checksum to detect bit errors. If a single or multiple bit errors are detected in the incoming packet, the TCP/UDP layer discards the packet so that the upper layer (e.g., RTP) will not receive the corrupted packet. On the other hand, different from UDP, TCP uses retransmission to recover lost packets. Therefore, TCP provides reliable transmission while UDP does not.

c) TCP employs congestion control to avoid sending too much traffic, which may cause network congestion. This is another feature that distinguishes TCP from UDP.

d) TCP employs flow control to prevent the receiver buffer from overflowing while UDP does not have any flow control mechanism [72]. Since TCP retransmission introduces delays that are not acceptable for real-time streaming applications with stringent delay requirements, UDP is typically employed as the transport protocol for video streams.

Therefore for OTELO real-time and near real-time performance, the use of the TCP protocol is not acceptable. In addition, since UDP does not guarantee packet delivery, the receiver needs to rely on upper layer (i.e., RTP) to detect packet loss.

RTP is an Internet standard protocol designed to provide end-to-end transport functions for supporting real time applications. RTCP is a companion protocol with RTP and is designed to provide QoS feedback to the participants of an RTP session. In other words, RTP is a data transfer protocol while RTCP is a control protocol.

Therefore RTP/UDP/IP protocol is implemented for the ultrasound stream to investigate the GPRS possibility of real-time interaction between the Expert station probe movement and the reception of the ultrasound stream from the Patient station. And UDP/IP protocol used for Robot data transmitted on both directions.

Figure 3.11 shows the Ultrasound stream throughput in reaching the Expert station. The simulation considered real latency readings parameter, captured by sending (250) Bytes to the Expert station at (500) ms intervals from the Patient station to approach analysis on real network conditions:

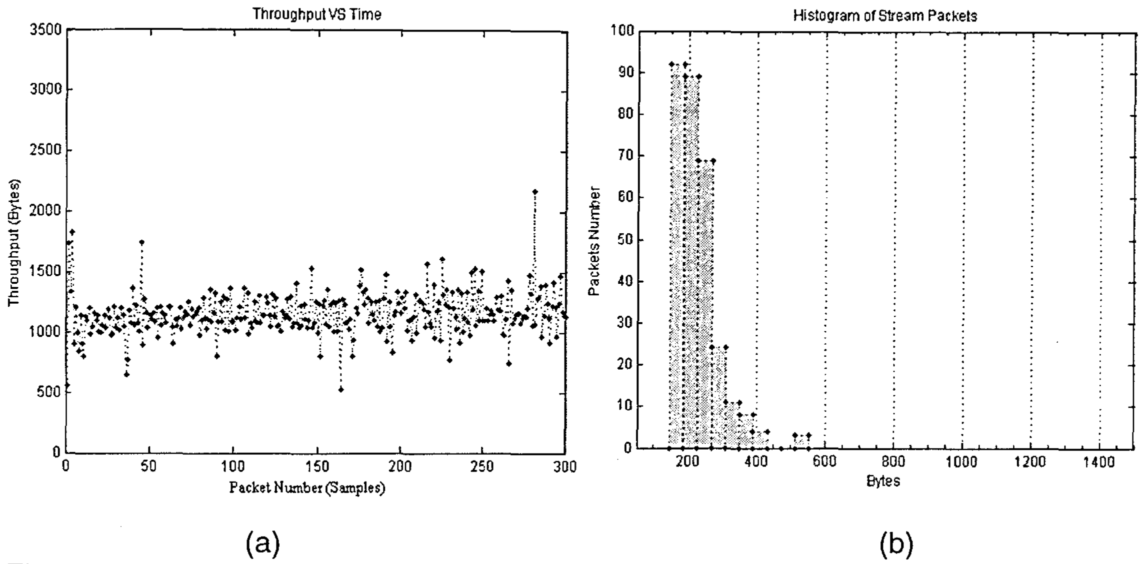


Figure 3.11 Instantaneous throughput at the Expert station and throughput distribution, (a) Peak throughput (Bytes/sec), (b) stream packets distribution.

For the purpose of analysis, (300) packet sample were considered. The average throughput is about 1200 bytes/sec, fluctuating marginally whenever the ultrasound probe moves producing new streams. In conclusion, the probe movement lead to an increase in the throughput to about 15%. Part (b) shows the histograms of the received packet number. The figure shows the dominant packet sizes of around 180, 200, 250 Bytes, and few packets of sizes above 400 Bytes produced by the Patient station. First stream of ultrasound packets has found to take around 9 – 13 sec to reach the Expert station.

Quality of the ultrasound reception is a robotic data performance dependent, where the real-time robotic system process requesting at least < 500 ms time lag between the Expert probe movement and the first Ultrasound packet received at the same station:

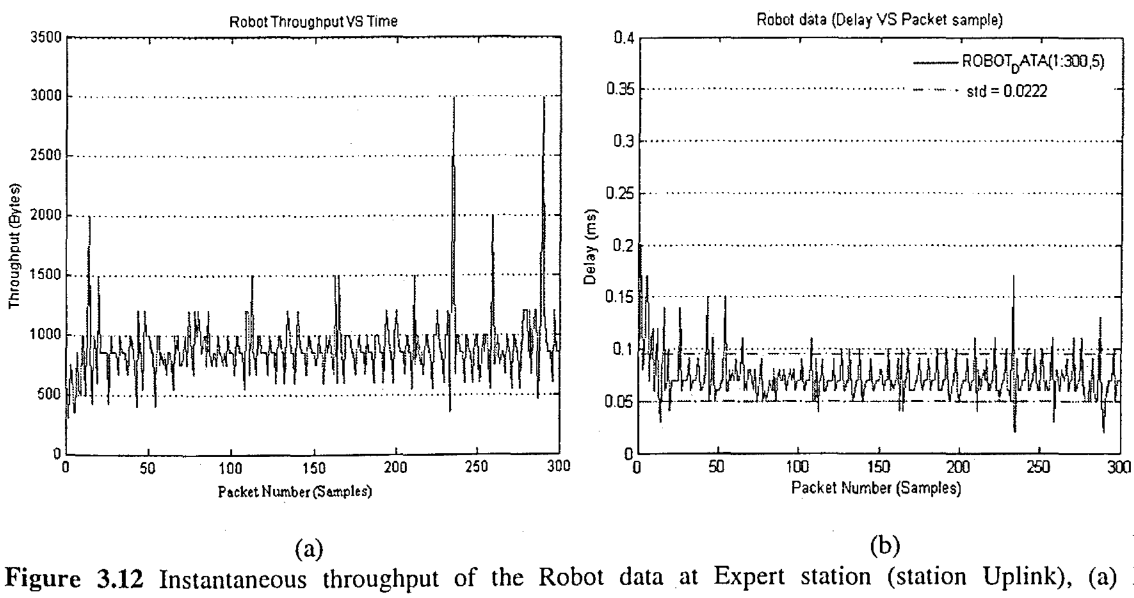


Figure 3.12 Instantaneous throughput of the Robot data at Expert station (station Uplink), (a) Peak throughput, (b) Packets Delay.

The throughput performance in figure 3.12 part (a), shows throughput variation from 600 to 1000 Bytes/sec. this phenomenon has no major influence on real-time performance of the robot control data interaction only, with a maximum round trip robot cycle (end-to-end) was < 700 ms. Robot packets show standard deviation on the delay at (b) around 0.022 sec, as the robot packet sizes are equal to 58 Bytes including the header length.

Since OTELO system ultrasound transmission in a real-time has very strict requirements: a transfer delay below 350 ms and acceptable packet loss ratio, are sufficient to receive US stream with acceptable quality at the Expert station. These requirements are close to the voice transmission in a real time service, which has very strict requirements of quality of service: a low transfer delay (about 200 ms for an acceptable quality, reaching 400 ms in some circumstances) and a loss ratio less than 5 % of the packets [76].

A statistical analysis of the distribution of packet latencies shows significant delay in the network. The 300 samples taken in figure 3.13, shows the pattern of the packet delay in terms of the time difference between any received packet and the predecessor packet (Delta Time). It has been noticed that the first few packets (around 50), gives higher latencies due to the end-to-end connection establishment, consequently lower throughput produced if we refer to the same packet range shown in figure 3.11 part (a).

Most packets experienced latencies of 140-220 ms, and very little shows latency as high as 400 ms. Standard deviation of around 60 ms on the latencies shown in figure 3.13 (a), explains the delay jitter on the network:

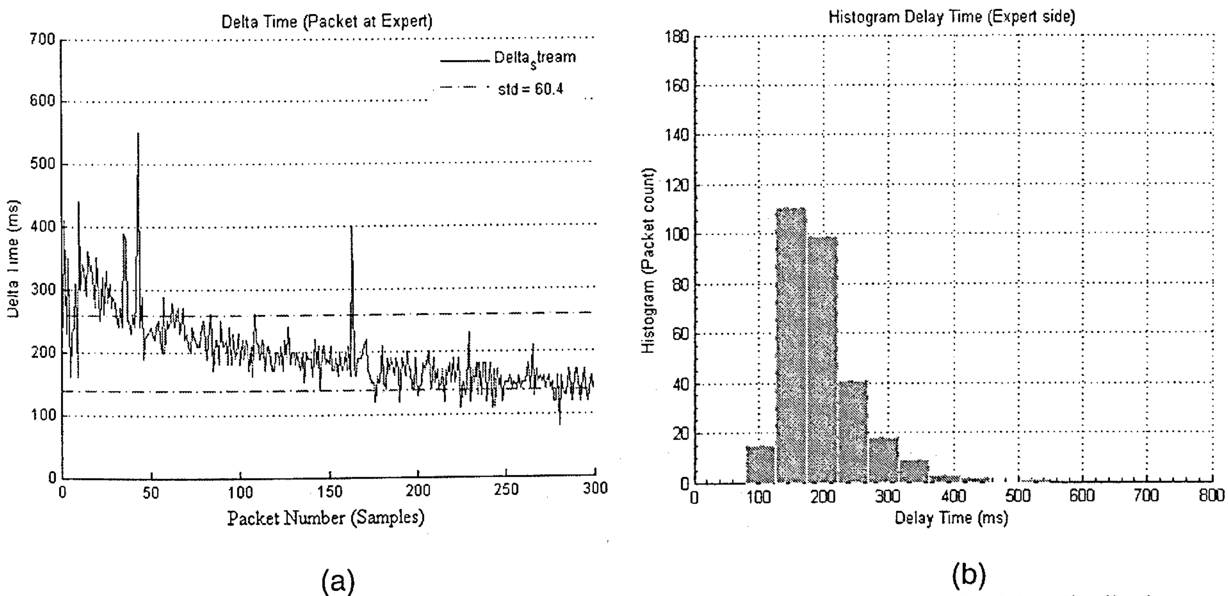


Figure 3.13 Delay time on the packets received at Expert station, (a) Packets delay (b) delay distribution.

The delay spikes shown in (a) explain the momentary network congestion lasting just for few seconds. While loss defines the percentage of these packets that are lost by the network, Latency is the time it takes for an IP packet to propagate across the network and jitter is the statistical variation in the measurement of latency (so jitter is said to be high if the measured latency varies a great deal) [70].

Jitter is often quantified with a statistical standard deviation. Adding the delay jitter values to the stream packets delay, would result obvious unreliability on the ultrasound reception. Consider the first 50 packets of figure 3.14, delay jitter shows around 40 ms comparing to an average jitter around 25 ms at the steady state performance, although this is not the optimum performance that fulfilling the system ultrasound scanning requirements:

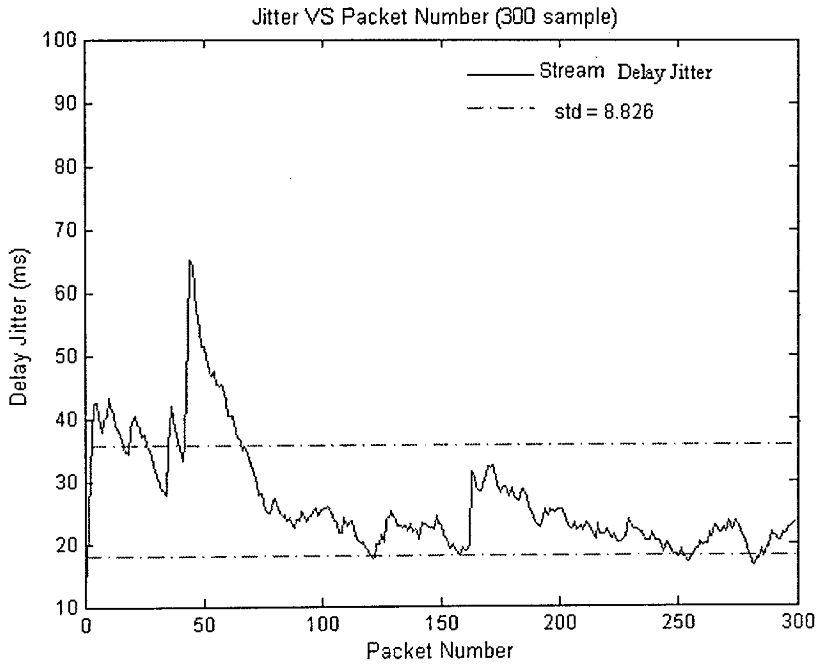


Figure 3.14 Delay jitter on the packets delay, received by the Expert station.

Generally jitter behavior represents the large variation in end-to-end delay for video streaming. Usually, streaming protocols adjust sending rate based on the estimated packet loss ratio and round-trip time. To reduce traffic retransmission, many streaming protocols such as (RTP) send only a single acknowledgment back to the sender to measure the RTT during a predefined period of time [77].

3.3.6 GPRS Real Network Performance Analysis

In this section we analyze the OTELO system performance under real GPRS network conditions. The implemented experiments were on Vodafone / UK GPRS network. The performance of GPRS may differ in different networks due to different topology and propagation environment [71].

A- Ultrasound Still Image:

The wireless mobile connectivity has been carried out with the following settings:

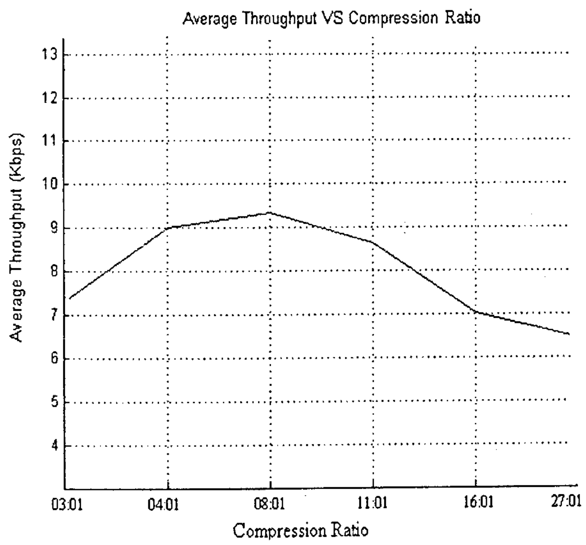
- Erickson T 68I, stationary in the Lab and connected to the Patient station Master PC.
- Channel data rates = '3 + 1' which means 3 TSs for Downlink + 1 TS for Uplink.

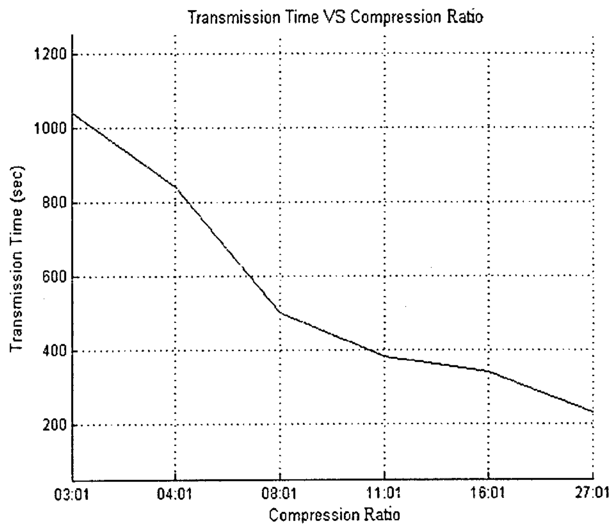
The TS capacity measures shows figures close to a maximum of 12 Kbps. For example when the average block error rate (BLER) in downlink equal 5.68%, the TS capacity can be computed as 12 x (1-BLER) = 11.2 Kbps per TS [71]. Where the BLER, is a block in error if at least one bit within this block is in error. This can be defined as [53]:

$$BLER = 1 - (1 - BER)^n \qquad (3.4)$$

Where n is the block length. The performance of the GPRS network on transmitting ultrasound still images, was investigated in two occasions; during of peak hours and during the day time. The drawback of testing on real network is that we do not have any control on the network parameters, except choosing the testing time and probably the data rates of one system end; furthermore the service class used in measurements will strongly affect performance [53].

Figure 3.15 (a) presents the performance of the link at different Compression Ratios. CRs > 04:1 shows quite similar results to the simulated readings if we refer to figure 3.5 (a), except that at lower compression ratio (high volume transmitted file) the throughput dropped down to around 7.4 Kbps. This can be referred to the delay introduced by the mobile network as well as the external packet data network [78], where we mentioned that the Internet is an intermediary IP network on the route between the Patient station and the Expert station:





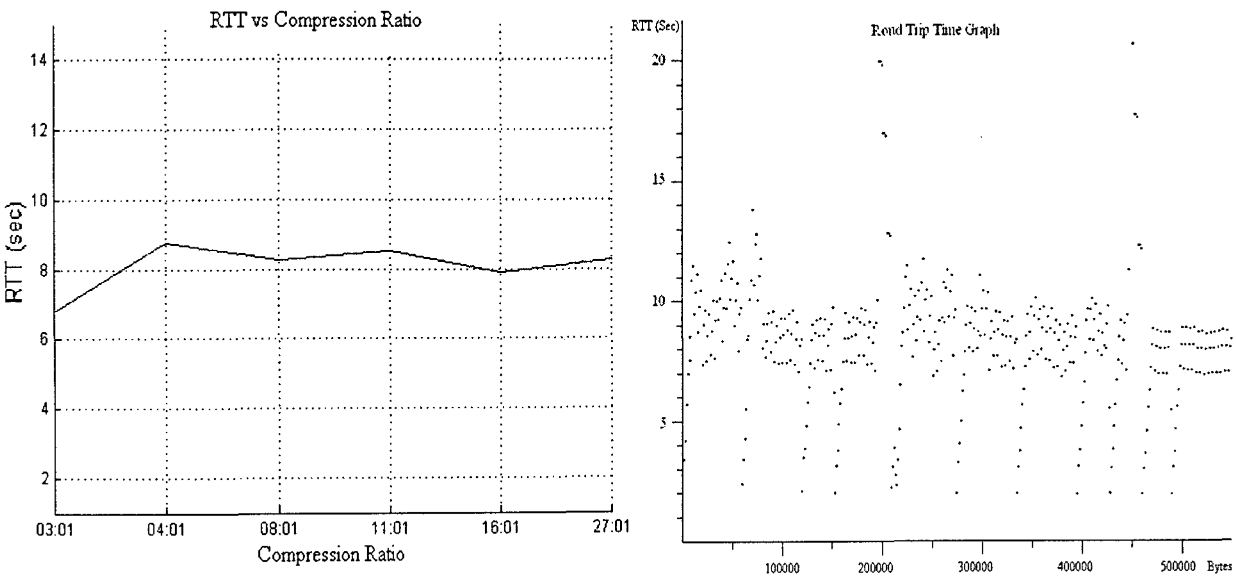
(a) (b)

Figure 3.15 Average throughput and Transmission time of the transmitted still images, (a) Average throughput (b) Transmission time.

The transmission time taken for group of a 10 ultrasound still images at different CRs, shows time delay varying around the simulated measurements. Although it is true that high-quality GPRS networks rely on signal quality, there are many new factors that can affect the performance of GPRS services. For GPRS to function as a viable Internet-access technology, it must be tested as a wireless IP network [70]. In wireless Internet, the overall RTT generally consists of two parts: wireless part and wired part, which can be denoted as [70]:

$$RTT = RTTwired + RTTwireless \qquad (3.5)$$

These RTTs in OTELO connectivity can be defined that the wireless RTT; is the time taken by a TCP packet sent by the Patient station to the GPRS core network plus the acknowledgement packet time back to the sender, and the wired RTT; is the time taken by a TCP packet sent by the Expert station to the Internet or PSTN backbone plus the acknowledgement packet time back to the sender. In a relatively small time scale, the RTT in wired network part will not vary sharply [73]. Figure 3.16 (b) presents clearly the network instantaneous RTT behaviour for the end-to-end connection as a function of the Uplink performance of the Patient station, including of course the wireless part, where each dot seen in part (b) of the figure is a TCP segment (packet) sent by the Patient station:



(a) (b)

Figure 3.16 Average RTT and instantaneous RTT behavior, (a) Average (b) instantaneous.

In two occasions the RTT jumps to as high as 20 sec. A perfect network link can be seen on transmitting the tenth image (at sequence numbers above 5 Mbytes). Part (a) shows an interesting outcome, that all the compression ratios sharing nearly the same RTT readings, and this is identical to the results in figure 3.8 (a), where RTT is narrowing as the BER approaching very low values range 2 x 10^{-6} to 1 x 10^{-7}.

The signal quality leads to significant variations in bandwidth perceivable by the receiver, sudden signal quality fluctuations (good or bad) impacts GPRS link performance [65]. Figure 3.17 shows the TCP packets burst over the GPRS link:

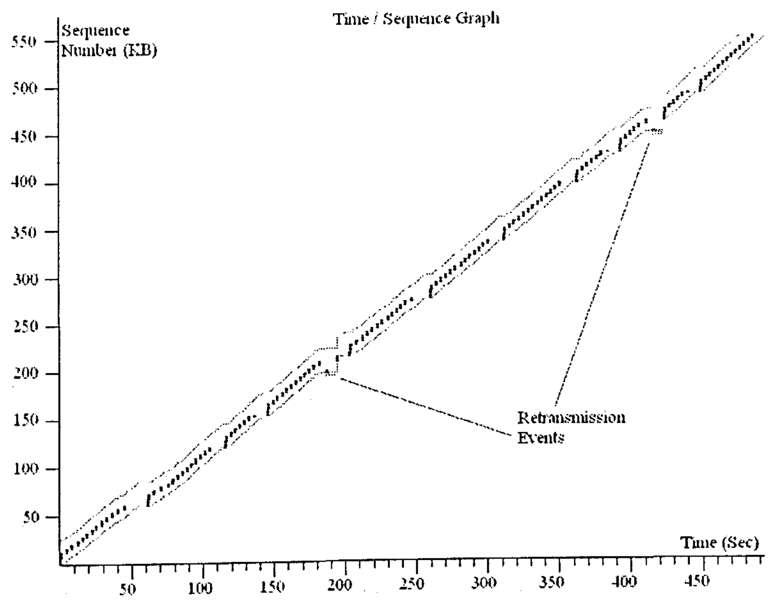


Figure 3.17 Sequence number of the still images packet transmission at Patient station (Uplink).

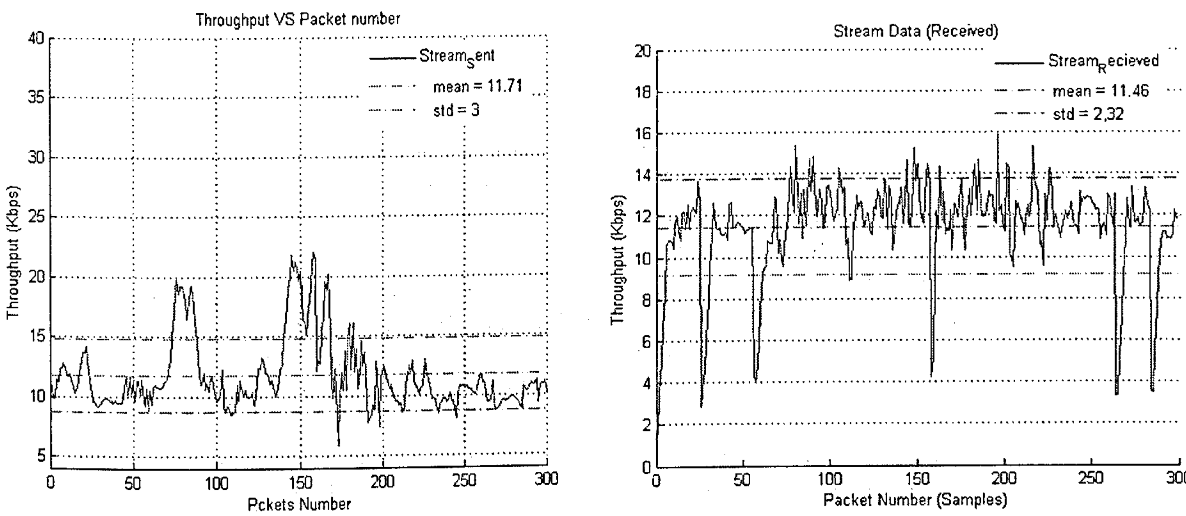
Figure 3.17 plots the sequence of packet accumulation of the transmitted ultrasound images compressed at 8:1, where we can recognize the excessive time taken before establishing new connection for the next Image, it is around 13 sec. The occasional link outages cause particular problems for TCP, outages can freeze data transfer over the link during the outage interval, which can lead to RTO triggered retransmissions that later turn out to be spurious when the data is finally released over the link [68]. Loss is an important performance metric because although applications that use TCP seamlessly re-transmit lost data, loss has a tremendous impact on the performance of the application layer [70]. The figure also shows retransmission events on two places, almost all the retransmission occasions show only one retransmission each time. Observations of TCP over GPRS conclude that the high underlying RTT on GPRS links results in suboptimal performance, particularly for short-lived sessions. Furthermore, the TCP causes excess queuing, leading to variety of performance problems [68].

B- Ultrasound Stream Performance:

The experimental Set-up of the OTELO system has the following cellular conditions to perform the ultrasound streaming:

1. Both the Patient station and Expert station are in the same location (MINT Lab / Kingston University, London).
2. Patient station was moved to a location nearly 20 miles away from the Expert station (St. George's Medical School), to provide different cellular geographical coverage.

The ultrasound streaming packets were separated through the analysis from the background traffic and other applications IP packets movement, which we have no control on it, conducted by the end-to-end wireless plus wired network. The RTP streaming protocol has been implemented for the OTELO streaming service. RTP does not guarantee QoS or reliable delivery, but rather provides the following functions in support of media streaming [72]; Time-stamping; Sequence numbering; Payload type identification and Source identification. The instantaneous throughput of the ultrasound stream leaving the Patient station and the same RTP packets receipt at the Expert station, are shown in figure 3.18:



(a) (b)

Figure 3.18 Peak throughput of the transmitted ultrasound stream, (a) at Patient side (Uplink) (b) at Expert side (Downlink).

Part of the stream fluctuation shown at both ends in figure 3.18, normally reveals the ultrasound head movement and consequently the H263 codec compression of the new frames produced. The available bandwidth often varies with time as radio conditions change [68], providing we know that the time slots are fixed by the operator for Up/Down links.

The Number of stream packets taken for the analysis is equal 300 stream packets. Figure 3.18 shows the peak throughput at the Patient station and Expert respectively. The packets received by the Expert station are greatly varying in the arrival time (network latency), while feeding them at the Patient side buffer was depending solely on the video encoder and the condition of the GPRS link (the wireless part). The negative throughput spikes approaching nearly 3 Kbps at the Expert side were due to congestion on the wired network.

We also noticed that the mean throughput is close to each other at both ends, approximately 11.5 Kbps, while the Standard Deviation was 2.32 Kbps at Expert station and 3.0 Kbps at the Patient station. Distribution of the packets shows interesting outcome, that nearly > 40 % of peak throughput is approaching 12.5 Kbps, as seen in figure 3.19 below:

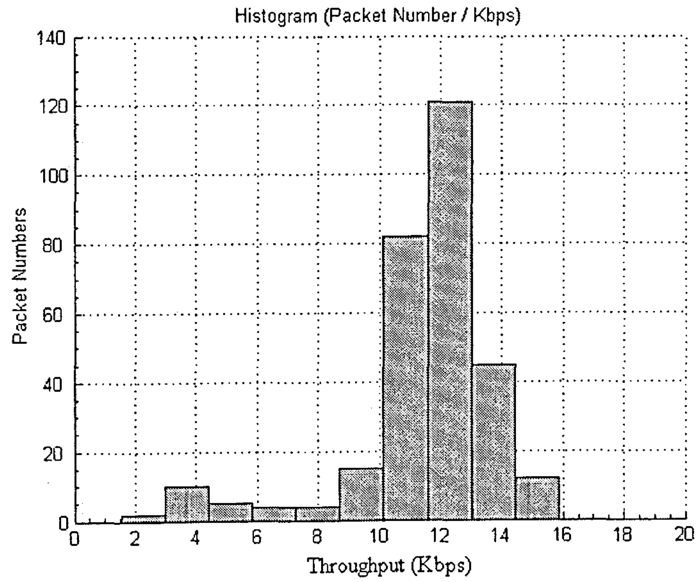


Figure 3.19 Peak throughput distribution of the Expert side.

Packet loss does occur over GPRS links in both the downlink and uplink directions, but the incidence is relatively rare, and hard to quantify [73]. Nevertheless out of 906 RTP packets captured by the Expert side; total lost was 3 packets i.e. 0.33%. Retransmission is usually dismissed as a method to recover lost packets in real-time interaction since a retransmitted packet may miss its play-out time [72]. The delay introduced by the GPRS air interface is mainly caused by three factors: the delay access on each ON period *(variable)*, the delay of the multiplex process *(variable)* and the transmission delay [76]. These factors represent the delay on the wireless part of OTELO Patient/Expert connection.

Figure 3.20 shows the Delta time (which is the time difference between the any received packet and the predecessor one) of the ultrasound packets received at the Expert side, which are influenced by the GPRS air interface (wireless link delay behavior):

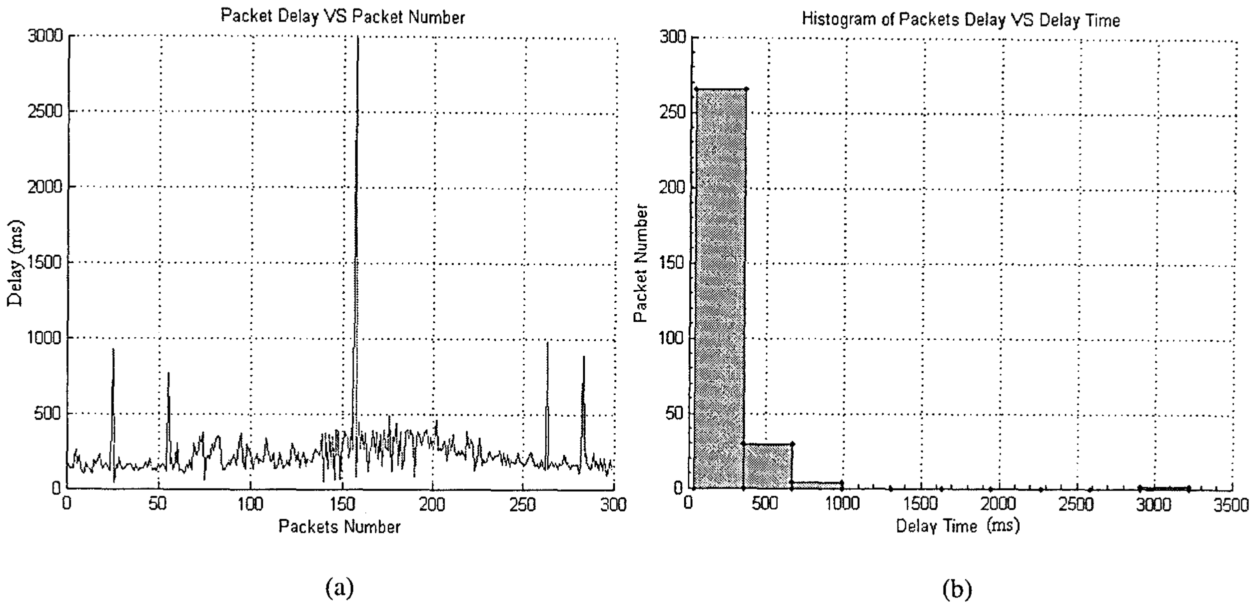


Figure 3.20 Delay and delay distribution of the packets captured at the Expert station, (a) packet delay, (b) Delay distribution.

Streaming video requires bounded end-to-end delay so that packets can arrive at the receiver in time to be decoded and displayed. If a video packet does not arrive in time, the play out process will pause, which is annoying to human eyes. A video packet that arrives beyond its delay bound (e.g. its play out time) is useless and can be regarded as lost [72]. Since the Internet introduces time-varying delay, to provide continuous play out, a buffer at the receiver is usually introduced before decoding. Close look to the delay distribution of the delay, figure 3.20 (b), 88.3 % of the packets are facing in average 250 ms delay with respect to the previously received packet.

Although the latency is one of the most important key metric for real-time applications such as voice-over-IP (VoIP) and media streaming [79], the system performance requires as low as possible statistical variation in the delay i.e. jitter. Therefore the delay jitter across our system link is also a key factor for reliable medical ultrasound stream reception at the Expert station. Figure 3.21 present the delay jitter of the packets at the Expert station:

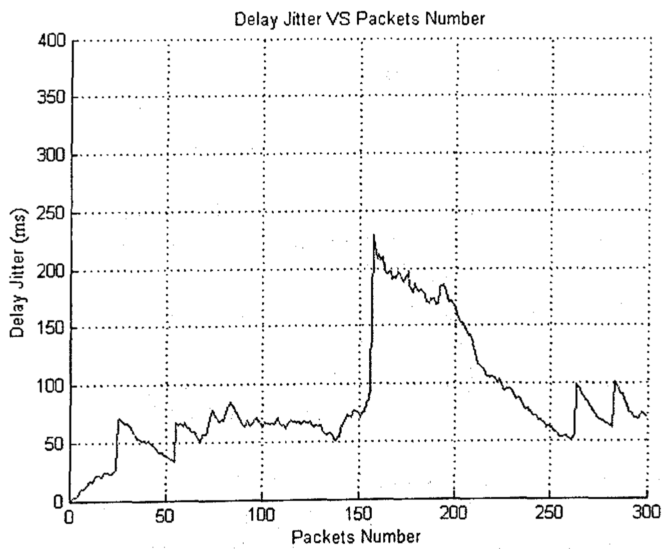


Figure 3.21 The delay jitter of the packets received by the Expert station.

The delay jitter countered on the packets received at the Expert side explains the unreliability of the ultrasound images seen at the Expert station, in terms of variable frames per second. Packets numbers 150 – 250 shows high jitter above the average of 60 ms. This phenomenon happens as a result of dramatic fluctuation of network bandwidth and delay.

By referring to the streaming packet delay characteristics shown in figure 3.20 specifically part (b), and obtaining the packet size distribution of 900 transmitted packets, we found that the highest percentage of packet sizes were varying between 200 – 300 bytes in average as shown in figure 3.22:

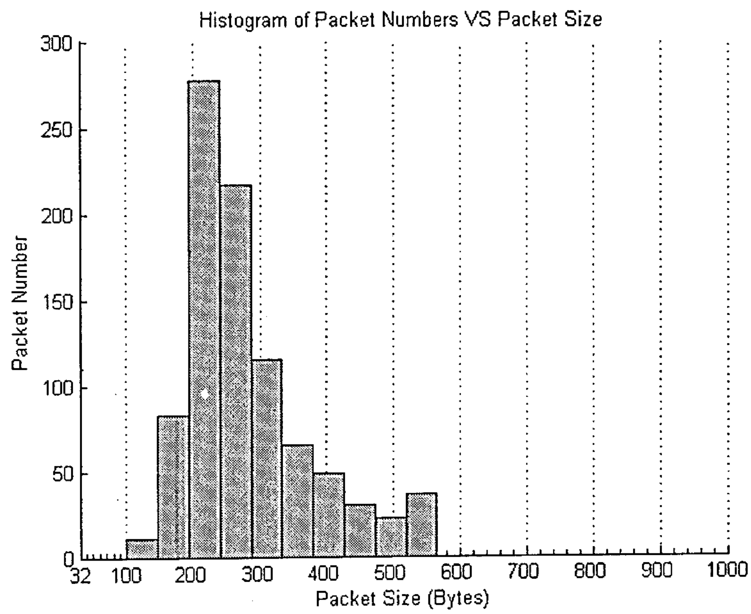


Figure 3.22 Packet distribution of the Ultrasound stream Throughput shown in figure 3.19.

3.3.7 End-to-End Delay Performance

In this section we focus on the Uplink (UL) transmission analysis of the Patient station as it is the important direction for the OTELO system wireless connection, although the Downlink transmission effect on the Patient station, is also included in the analysis of latency. A common way to benchmark the network latency is to evaluate its RTT [71], RTT can also be defined as; the time it takes to transmit one packet from e.g. a server to a terminal plus the time it takes for the corresponding packet to be sent back from the terminal to the server [71].

The definition of RTT regarding OTELO system functionality is the function of the packet size transmitted to the Expert station plus the corresponding packet sent by the Expert to the Patient station. Different packet sizes may be used to characterize various performance degradation effects, packet sizes 32, 100, 330 and 500 Bytes are used.

The characterisation of the minimum network latency can be estimated by pinging 32 bytes from the Patient station to Expert station, where it is also estimate the performance degradation associated to the transport and application layers establishment protocols [71]. In the mean time pinging packet sizes around 500 Bytes is required to determine the latency associated with the transmission of the relatively large ultrasound data packet over the GPRS link to the Expert side.

ICMP:
The Internet Control Message Protocol (ICMP), provides some error detection mechanisms. It can be used to send error messages or other messages for network diagnosis [53]. ICMP packets are sent as IP datagrams, and it uses the basic support of IP to send datagrams as if it were a higher-level protocol, however the ICMP is actually an integral part of IP. ICMP has been used to ping the Expert station with the packet sizes above at 500 ms interval time from the Patient station.

The test performed with channel setting of '3+1' used in the real GPRS network. Figure 3.23 shows the end-to-end latency break down between the wired and wireless link:

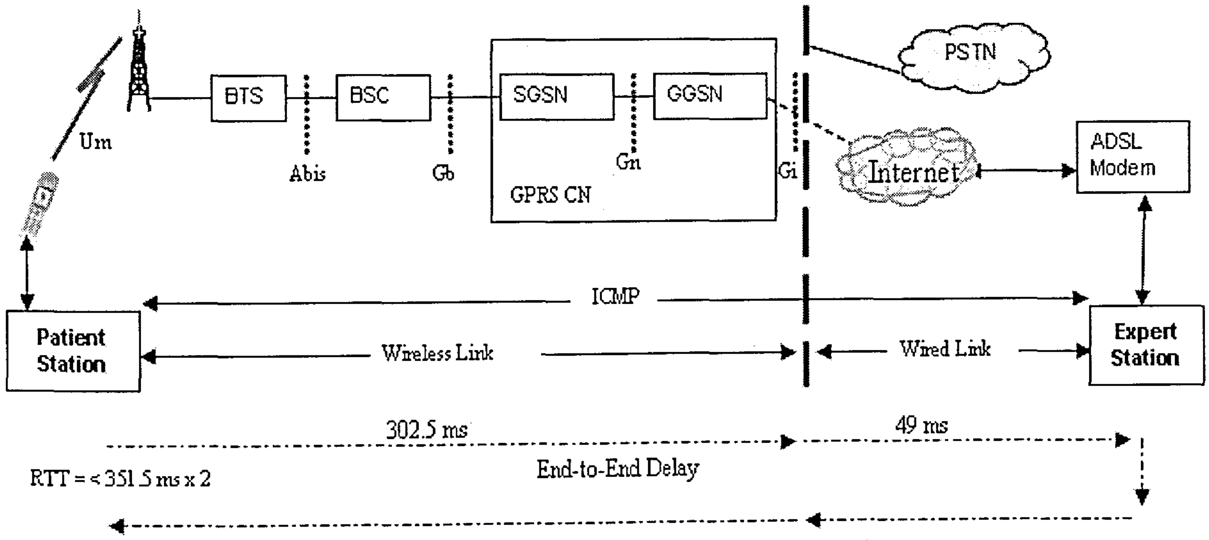


Figure 3.23 End-to-End GPRS Latency configuration of OTELO system, at a specific transmitted packet sizes of 100 to 330 Bytes.

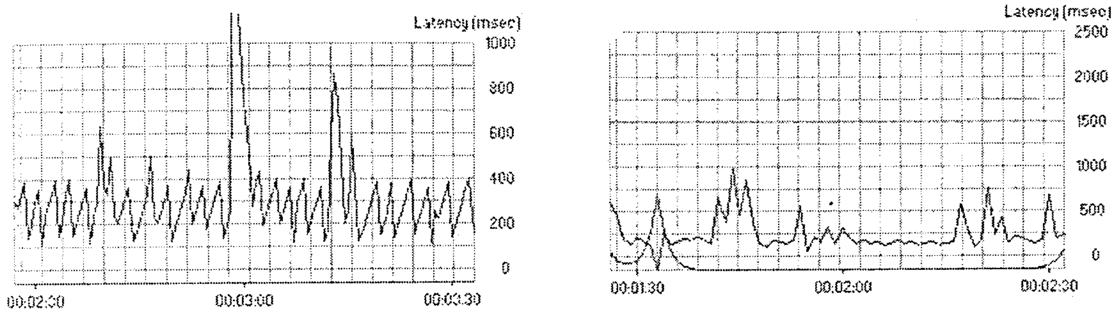
To differentiate the latencies caused by only the wireless part of the complete link, and the latencies over the wired part; the test performed as follow:

1. Sending the packets to the GPRS service provider backbone network, by pinging the server, where our Patient station is connected.

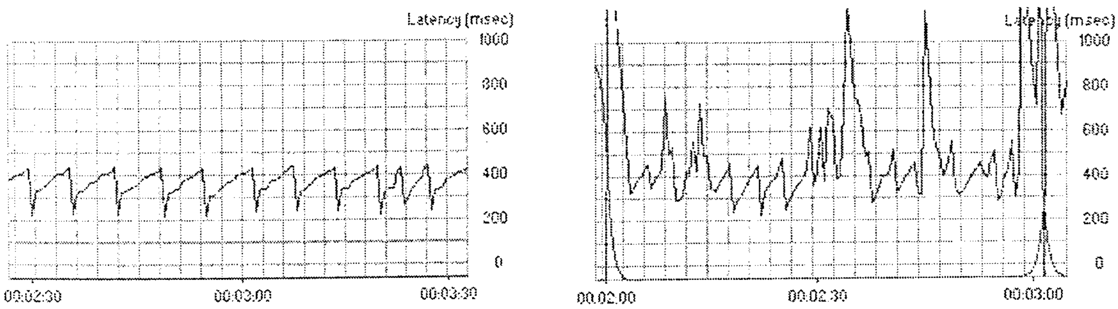
2. Sending the packets directly to the Expert station, and in this case pinging the full Patient / Expert stations path including the Internet as intermediary network as mentioned before.

The packet latencies measurements have been repeated several times in order to approach very close results to the delay performance of the network during reliable connection, i.e. avoiding the cases of high packet loss and congestion.

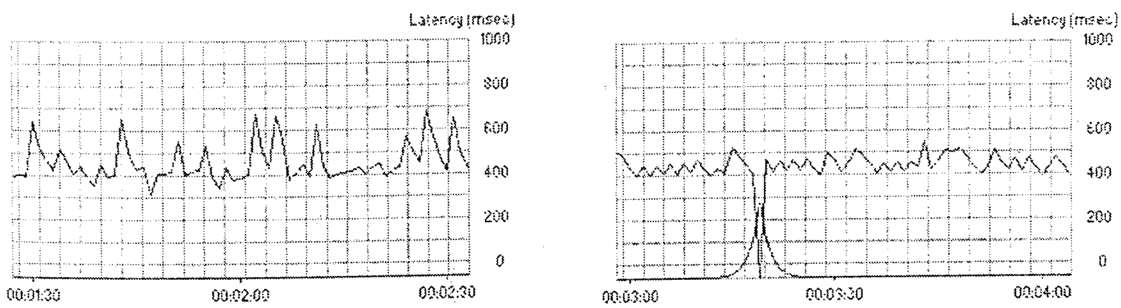
Figure 3.24 shows latency performance traces in the form of Minimum / Average / Maximum within two groups:



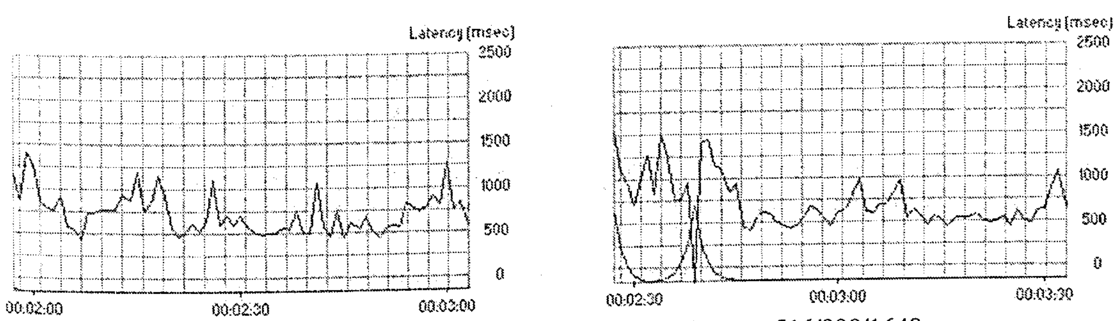
163/372/1495 (Packet size sent = 32 Bytes) 195/411/1641



265/402/500 (Packet size sent = 100 Bytes) 273/632/1562



340/482/761 (Packet size sent = 330 Bytes) 430/484/1577



508/868/1554 (Packet size sent = 500 Bytes) 516/890/1648

(a) GPRS Server side (b) Expert Station side

Figure 3.24 Packets delay performance at different packet sizes, when pinging different packet sizes over the GPRS Link; Left column to the GPRS Server, Right column to the Expert Station.

The left column shows the latencies of the transmitted packets from the Patient station to the GPRS server, while the right column presents the full end-to-end (Patient-to-Expert) latencies including the delay over the wireless part. Figure 3.24, left column, (packet size = 32), shows the propagation variation of 32 Bytes sent to the GPRS server, the average of 372 ms gives an idea of the minimum network latency [64], although during the ultrasound transmission, these lower sizes are rarely coded by the stream decoder (example H263 Codec).

The first few packets sent in the Uplink direction requires the radio network to establish a new radio connection; this entails protocol interaction with the phone [70]. We noticed that the first packets are resulting in latencies that generally introduce the Maximum latency readings, after the first few seconds the latencies approaching the average. The spikes seen across the latency traces shown above can be attributed to the following:

a) The small wider ones mostly due to burst packets sending that incurred delay because of queuing [70].
b) The sharp spikes, which are mostly seen on the traces represents the latencies over the end-to-end link. It is also noticed that when repeating the test several times these spikes were more often seen when the wired link is involve. These are due to momentary congestion and load on the wired link. This is including the delay across the Internet.

From the results, the delay in the wireless link is the largest; however the corresponding packet loss mostly happened within the wired part of the complete link. The results above are comparable to literatures, where it shows that the RTT of an GPRS network can change between 0.5 sec and more than 1 sec depending on the user throughput, while typical RTT of fixed networks can be 1ms for LAN, 20 ms for ISDN, 20 to 200 ms for analogue modem and 125 ms for average Internet [71]. The corresponding data of the plots above is represented in the table below:

Table 3.7 Latencies values of different Packet sizes; to GPRS Server; to Expert Station.

Packet Sizes (Bytes)	Latencies (ms)						Average delay on The Wireless Link w.r.t The end-to-end Link (%)
	Over Wireless Link			Over Wireless + Wired Link			
	Min	Avg	Max	Min	Avg	Max	90.5 %
32	163	372	1495	195	411	1641	90.5 %
100	265	402	500	273	630	1562	63.0 %
330	340	482	761	430	484	1577	99.0 %
500	508	868	1554	516	890	1648	97.5 %

Combining the average latencies over the wireless part shown in table 3.7 with packet sizes histogram shown above, we can map the following approximation:

(i) Packet sizes 100 – 330 Bytes, has an average Min. Delay = 351.5 ms (Patient-to-Expert).
And = 302.5 ms (Wireless part).

(ii) Delay on the fixed part of the network equal nearly 49 ms.

Therefore the average delay of the common sizes of the ultrasound packets transmitted to the Expert station, is around 351 ms, on '3+1' GPRS channel setting. Apart from the Uplink TS assignment, GPRS therefore represent the bottleneck of the Uplink direction bandwidth.

3.4 Comparative Analysis of the Results Achieved

This section present selected comparative results between the simulated and real GPRS network experimentations. Average throughput, transmission time, RTT and packet delay results, selected for both ultrasound still images and ultrasound stream. A measurement environment for GPRS network performance can only be set on a simulated platform, where the set-up parameters can be adjusted, compared to real GPRS network [53]. Isolated measurement for a single user will be difficult; furthermore the service class used in the real network will strongly affect the performance. Our GPRS simulations included only a single user.

3.4.1 Comparative Analysis Results

In this section we compare some of the results obtained in both simulated and real GPRS network environments. The average throughput performance over real GPRS network shows closer results to the simulated conditions, as in figure 3.25:

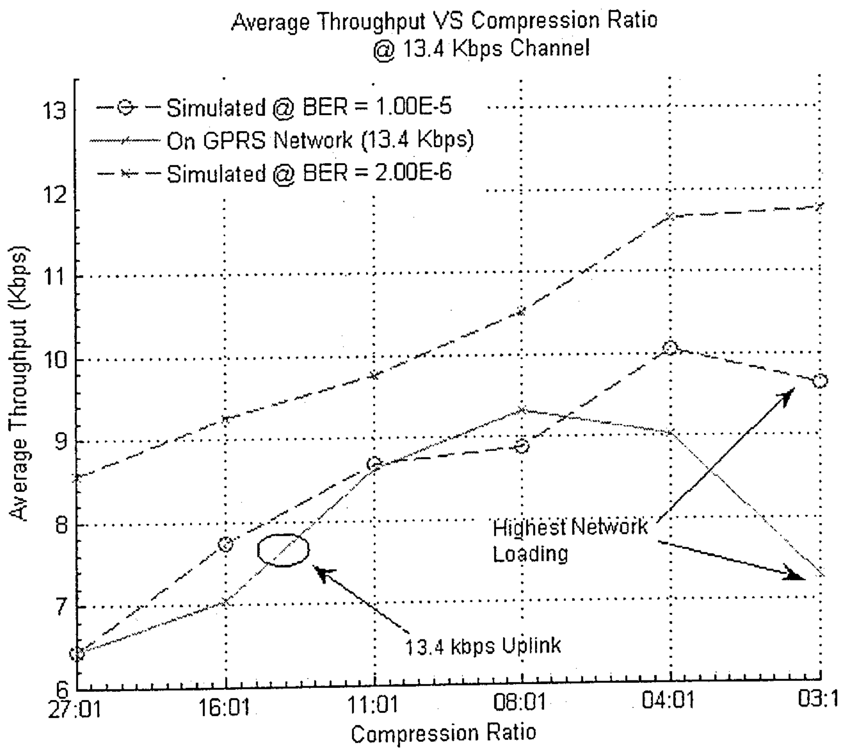


Figure 3.25 Comparative average throughput performances in real and simulated GPRS networks.

It is well known that (BERs < 1 x 10^{-5}) are practically far from the average operating conditions of the real GPRS networks [71, 53]. We noticed throughput reduction at higher TCP segments flow (at very low image compression), as the cell load increases, there are two main reasons for throughput reduction; increasing interference levels and Time Slots sharing between users [71], where evaluation of mean throughput analysis shows throughput reduction for all CSs and for load increasing. We did not use TCP/IP header compression in our GPRS simulation, where this is normally applicable in a real GPRS network, therefore the maximum throughput can increase compared to the simulation condition.

Under low data traffic (no TS multiplexing), a GPRS user receives typically 12 Kbps over each allocated TS in downlink. Other effects like IP overhead, TCP slow start and cell reselections may reduce in practice, the end-to-end throughput [71].

When moving to the instantaneous throughput of the ultrasound stream images, and although the simulated network parameters has been imported from a real network based on transmitting (330) Bytes packet length as an average encoded US packet length, a difference of around 2.2 Kbps between the mean throughputs was noticed, as in figure 3.26:

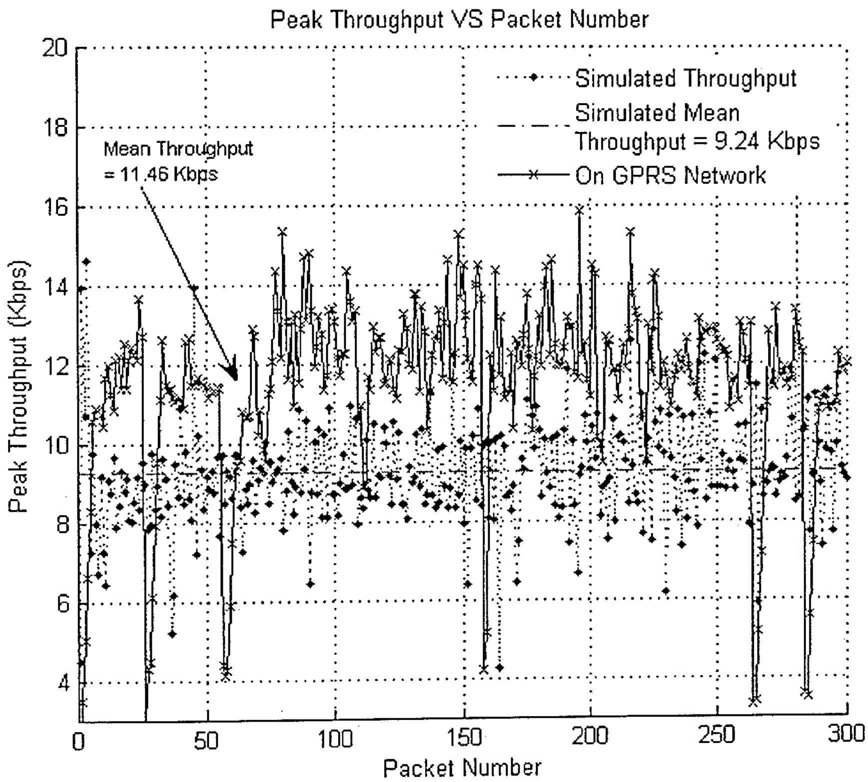


Figure 3.26 Comparative peak throughput of ultrasound stream.

This difference in the comparative mean throughput results can be attributed to the following:

a) The delay parameters specified for 330 bytes packet length were functioning in compatibility for all other packet lengths transmitted to the Expert station on the simulated environment, whilst with the real network, each packet length will suffer proportional latency, therefore affecting the resulting overall throughput.

b) The streaming encoder (H.263) performances in each environment were not exactly identical. The relatively high delay parameters assigned to the simulated setting, caused the H.263 encoder to reduce the ultrasound frame size (higher compression), to function properly with the buffering process at the Patient station, therefore the packet length produced by the encoder effects the received peak throughput at the Expert station.

The transmission time needed to transmit the ultrasound still images at different compression ratios, shows marginal differences between the simulated results compared to the real network results, although the latencies simulated parameters have been chosen to reflect relatively worst case operation scenarios:

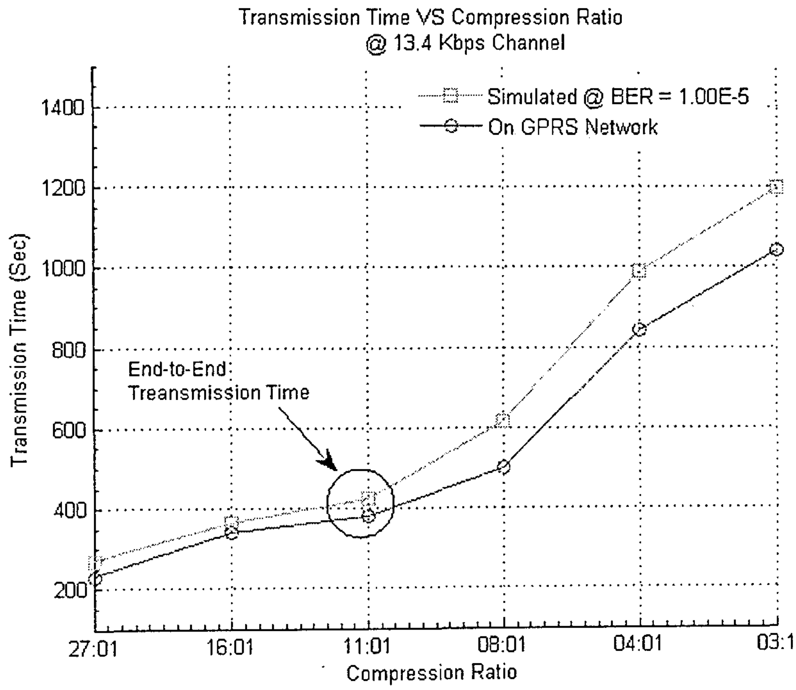


Figure 3.27 Transmission time for the ultrasound still images received by the Expert side, on simulated and real GPRS network.

We have observed typically high latencies over the GPRS link, and the propagation time of the packets across the network varies highly. This explain the higher standard deviation of about 208 ms at real network compared to 61.6 ms on the simulated network, although the latencies over both environments are very close in terms of the mean delta time as in the figure 3.28:

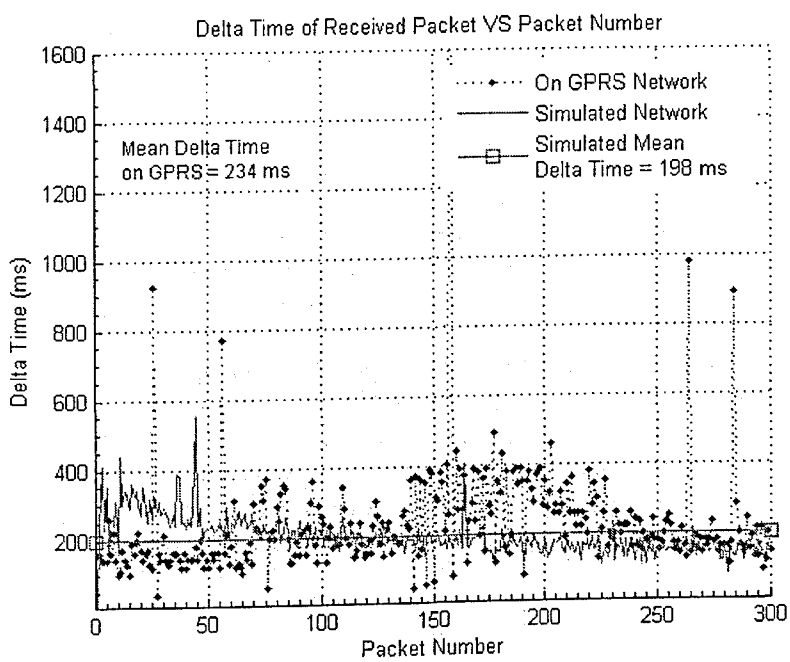


Figure 3.28 Comparative ultrasound packets delay at the Expert station, when simulated and on real network.

The comparative results of the RTT measures, which are the average values between the maximum and the minimum readings on both cases, on transmitting ultrasound still images.

An obvious differences between the simulated and the real test RTT values are shown below:

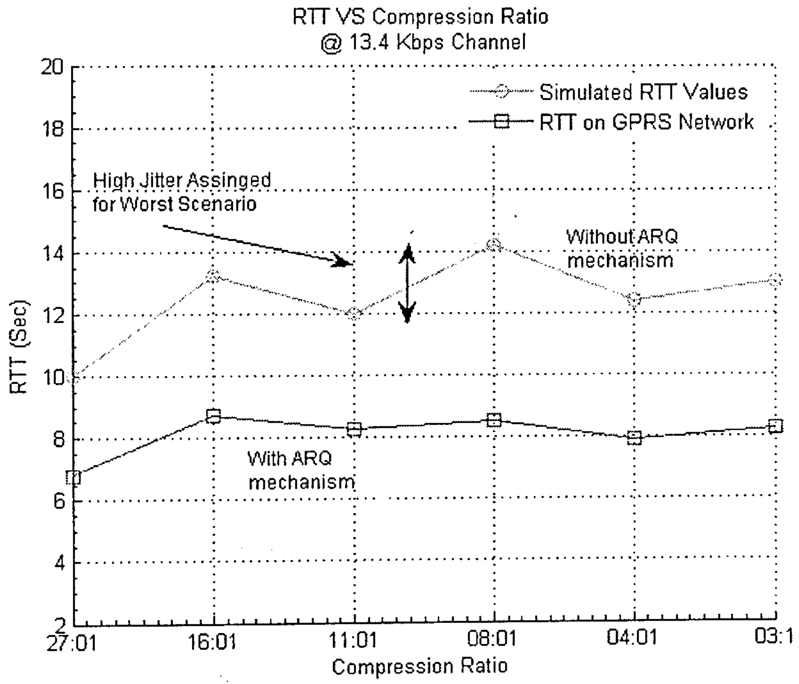


Figure 3.29 comparative RTT measures on simulated and real GPRS network.

Figure 3.29 show simulated RTT measurements of around 2 sec above the RTTs measurements on the real GPRS network. Although both measurements are within comparable values to previous results [69, 65, 71]. Nevertheless the difference above can be justified by the following important issues:

a) The performance set-up parameters on the simulated network, considered worst scenarios condition. Therefore relatively higher latencies (delay) and specifically the delay jitter parameters were assigned to the simulation case. The average delay of 482 ms, equivalent to pinging 330 Bytes packet length was assigned, while great deal of the TCP ACK packets of 40 Bytes length normally took place.

b) TCP offers mechanism for congestion avoidance ensuring network stability for the Internet. It provides sliding window based Automatic Repeat request (ARQ) including an adaptive time-out mechanism for ensuring reliable data transmission. The ARQ is additionally supported by a function called Fast Retransmit, which allows the retransmission of single lost packets without waiting for time-out. Nevertheless, it can be concluded that TCP works very stable over GPRS [69]. This ARQ function is not operating in the simulation platform.

3.5 Summary

In this chapter we have presented a detail results and performance analysis of OTELO medical data end-to-end transmission over simulated and real GPRS network conditions. The results of the ultrasound images transmission using TCP/IP protocol were discussed in terms of the performance analysis over both simulated and real network. Furthermore the achieved results were referenced to applicable literatures for comparison.

The performance analysis of the ultrasound stream transmission has also been evaluated in details in simulated and real GPRS network conditions. The performance matrices of peak throughput, average throughput, end-to-end latency and delay jitter were evaluated. The recommended streaming transmission protocols were presented. The Patient-to-Expert station RTT of the full robotic control cycle has been analysed and measured in order to investigate the limitations of the GPRS network for real time OTELO performance.

CHAPTER 4

OTELO 3G Performance Analysis

4.1 Introduction

In this chapter we present a brief introduction to the 3G mobile network and the relevant details of the air interface protocol architecture, in particular the physical layer of the UMTS network required for the 3G performance analysis of OTELO system. The three sections describe the system performance over simulated and real-time 3G mobile environment. These include studies on different 3G QoS classes and performance test matrices including; throughput, transmission time, delay, delay jitter and end-to-end round trip time.

4.2 An overview of 3G Network and Standardisation

In this section we present brief overview on 3G network and standardisation. UMTS is a wideband circuit and packet based transmission system of text, digitised voice, video and multimedia with data rates up to (and possibly greater than) 2 Mbps [80, 66].

It aims to offer a consistent set of services to mobile computer and mobile phone users no matter where they are located in the world [80]. Based on GSM communication standards and endorsed by major standards bodies and manufacturers worldwide, UMTS has become the dominating 3G standard for mobile users even before its introduction in 2001 and 2002.

The frequency spectrum for UMTS has been identified as terrestrial systems, where the frequency ranges of 1,885- 2,025 MHz and 2,110-2,170 MHz has been assigned for International Mobile Technology (IMT-2000) systems, and frequency ranges of 1,980-2010 MHz and 2,170-2,200 MHz assigned for the satellite part of the UMTS systems [66].

WCDMA technology is the most widely adopted third generation air interface, its specifications has been created in 3rd Generation Partnership Project (3GPP), which is the joint standardisation project of the standardisation bodies from Europe, Japan, Korea, USA and China [66]. Within 3GPP, WCDMA is called Universal Terrestrial Radio Access Network (UTRAN).

4.2.1 WCDMA Air Interface

WCDMA offers superior performance in terms of higher data rates and capacity [80]. Table 4.1 lists the parameters of WCDMA air interface [62]:

Table 4.1 WCDMA characteristics.

Channel bandwidth	5 MHz
Duplex mode	FDD and TDD
Downlink RF channel structure	Direct spread
Chip rate	3.84 Mbps
Frame length	10 ms
Spreading modulation	Balanced QPSK (downlink) Dual-channel QPSK(uplink) Complex spreading circuit
Data modulation	QPSK (downlink) BPSK (uplink)
Channel coding	Convolutional and turbo codes
Coherent detection	User dedicated time multiplexed pilot (downlink and uplink), common pilot in the downlink
Channel multiplexing in downlink	Data and control channels time multiplexed
Channel multiplexing in uplink	Control and pilot channel time multiplexed I&Q multiplexing for data and control channel
Multirate	Variable spreading and multi-code
Spreading factors	4–256 (uplink), 4–512 (Downlink)
Power control	Open and fast closed loop (1.6 kHz)
Spreading (downlink)	OVSF sequences for channel separation Gold sequences 218-1 for cell and user separation (truncated cycle 10 ms)
Spreading (uplink)	OVSF sequences, Gold sequence 241 for user separation (different time shifts in I and Q channel, truncated cycle 10 ms)
Handover	- Soft handover - Inter-frequency handover

WCDMA has two modes characterized by the duplex method; Frequency Division Duplex (FDD) and Time Division Duplex (TDD). These are the names that WCDMA use for operating with paired and unpaired bands, respectively. The chip rate of the system is 3.84 Mcps. The frame length is 10 ms and each frame is divided into 15 slots (2560 chip/slot at the chip rate 3.84 Mcps).

Spreading factors range from 256 to 4 in the uplink and from 512 to 4 in the downlink. Thus, the respective modulation symbol rates vary from 960 Ksymbols/s to 15 Ksymbols/s. For separating channels from the same source, Orthogonal Variable Spreading Factor (OVSF) channelization codes are used [62].

For the channel coding three options are supported; convolutional coding, turbo coding, or no channel coding. Channel coding selection is indicated by upper layers. Bit interleaving is used to randomize transmission errors. The modulation scheme is Quadrature Phase Shift Keying (QPSK). The carrier spacing has a raster of 200 kHz (which means that the carrier frequency must be a multiple of 200 KHz.) and can vary from 4.2 to 5.4 MHz [81]. The different carrier spacing can be used to obtain suitable adjacent channel protections depending on the interference scenario.

Larger carrier spacing can be applied between operators than within one operator's band in order to avoid inter-operator interference.

4.2.2 3G Protocol Architecture

The overall radio interface protocol architecture is shown in figure 4.1 below. This figure contains only the protocols that are visible in UTRAN:

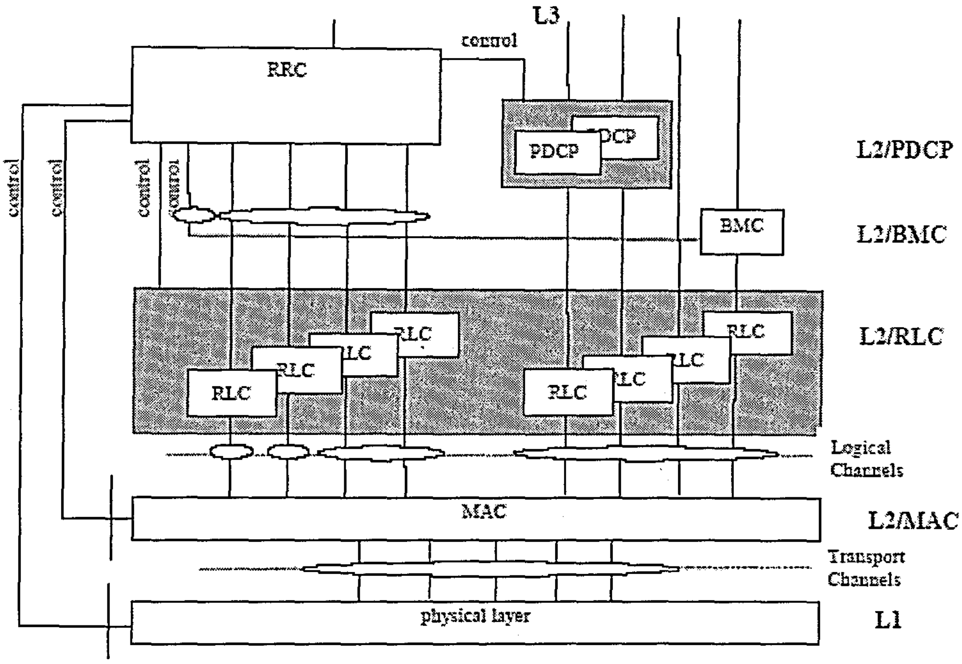


Figure 4.1 UTRAN FDD Radio Interface Protocol Architecture [66].

The protocol architecture is similar to the current ITU-R protocol architecture, ITU-R M.1035. The air interface is layered into three protocol layers:

1. The physical layer (layer 1, L1);
2. The data link layer (layer 2, L2);
3. Network layer (layer 3, L3).

Layer 1; The physical layer interfaces the Medium Access Control (MAC) sub-layer of layer 2 and the Radio Resource Control (RRC) layer of layer 3. The physical layer offers different transport channels to MAC. A transport channel is characterized by how the information is transferred over the radio interface. Transport channels are channel coded and then mapped to the physical channels specified in the physical layer. MAC offers different logical channels to the Radio Link Control (RLC) sublayer of layer 2. A logical channel is characterized by the type of information transferred [62].

Layer 2; is split into following sublayers: MAC, RLC, Packet Data Convergence Protocol (PDCP) and Broadcast/Multicast Control (BMC).

Layer 3 and RLC; are divided into control and user planes. PDCP and BMC exist in the user plane only. In the control plane, layer 3 is partitioned into sub-layers where the lowest sub-layer denoted as RRC, interfaces with layer 2. The RLC sub-layer provides ARQ functionality closely coupled with the radio transmission technique used.

4.2.3 Logical Channel and Transport Channel

The MAC layer provides data transfer services on logical channels. A set of logical channel types is defined for different kinds of data transfer services as offered by MAC. Each logical channel type is defined by the type of information that is transferred. Logical channels are classified into two groups [62]:

1. Control channel (CCH) include: Broadcast control channel (BCCH), Paging control channel (PCCH), Dedicated control channel (DCCH), Common control channel (CCCH), Shared channel control channel (SHCCH), ODMA dedicated control channel (ODCCH), and ODMA common control channel (OCCCH).

2. Traffic channel (TCH) include: Dedicated traffic channel (DTCH), ODMA dedicated traffic channel (ODTCH), and Common traffic channel (CTCH).

A transport channel is defined by how and with what characteristics data is transferred over the air interface. There exist two types of transport channels; Dedicated channels and Common channels. There is one dedicated transport channel, the Dedicated Channel (DCH), which is a downlink or uplink transport channel. The DCH is transmitted over the entire cell or over only a part of the cell using beam-forming antennas. The DCH is characterized by the possibility of fast rate change (every 10 ms), fast power control, and inherent addressing of mobile stations. The mapping between logical channels and transport channels is shown in figure 4.2:

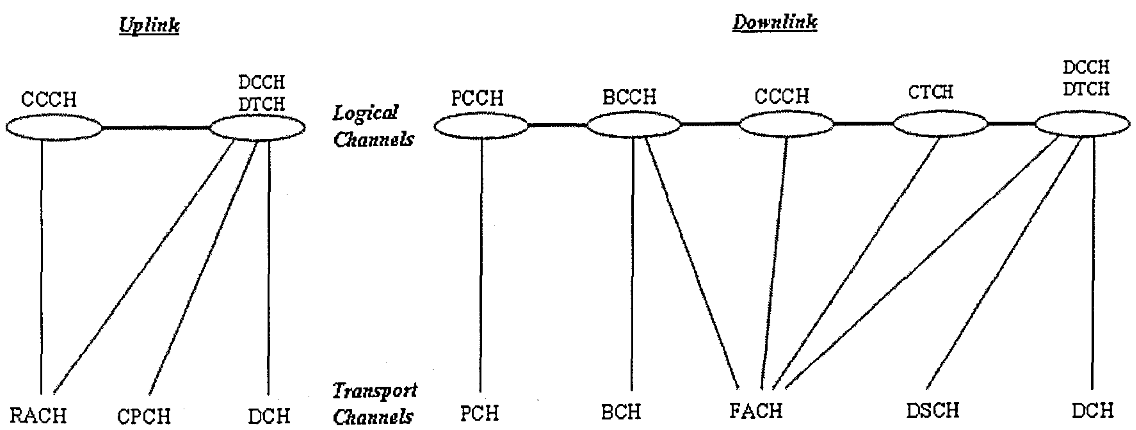


Figure 4.2 Mapping between logical channels and transport channels, uplink and downlink direction [66].

4.2.4 Physical Layer

The transport channels are channel coded and matched to the data rate offered by physical channels. Thereafter, the transport channels are mapped on the physical channels. Physical channels consist of radio frames and time slots. The length of a radio frame is 10 ms and one frame consists of 15 time slots. A time slot is a unit, which consists of fields containing bits. The number of bits per time slot depends on the physical channel. Depending on the symbol rate of the physical channel, the configuration of radio frames or time slots varies [62].

A- Uplink Physical Channels:

There are two uplink dedicated physical and two common physical channels:

a) The uplink dedicated physical data channel (uplink DPDCH) and the uplink dedicated physical control channel (uplink DPCCH);
b) The Physical random access channel (PRACH) and Physical common packet channel (PCPCH).

The uplink DPDCH is used to carry dedicated data generated at layer 2 and above (i.e., the dedicated transport channel (DCH)). There may be zero, one, or several uplink DPDCHs on each layer 1 connection. The uplink DPCCH is used to carry control information generated at layer 1. Control information consists of known pilot bits to support channel estimation for coherent detection, transmit power-control (TPC) commands, feedback information (FBI), and an optional transport-format combination indicator (TFCI).

The transport-format combination indicator informs the receiver about the instantaneous parameters of the different transport channels multiplexed on the uplink DPDCH, and corresponds to the data transmitted in the same frame. For each layer 1 connection, there is only one uplink DPCCH. Figure 4.3 shows the principle frame structure of the uplink dedicated physical channels:

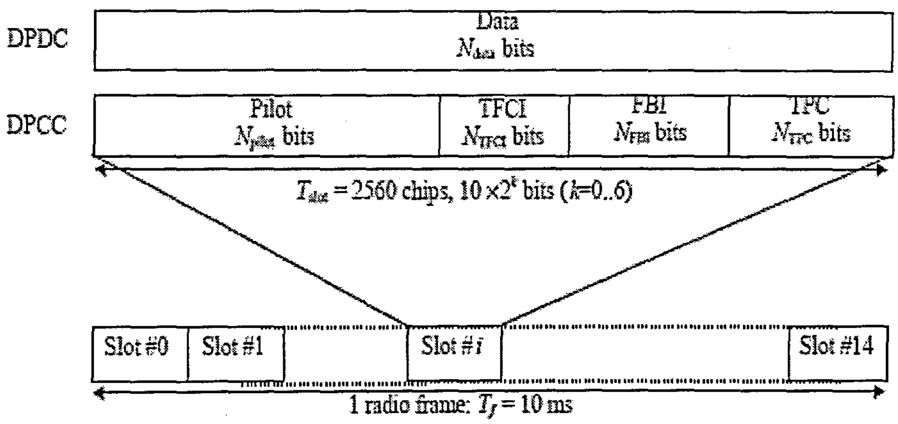


Figure 4.3 Frame structure of Uplink DPDCH/DPCCH channels [66].

Each frame of length 10 ms is split into 15 slots, each of length T_{slot} = 2560 chips, corresponding to one power-control period. The parameter k in Figure 4.3 determines the number of bits per uplink DPDCH/DPCCH slot. It is related to the Spreading Factor (SF) of the physical channel as; SF = $256/2^k$. The DPDCH spreading factor may thus range from 256 down to 4. An uplink DPDCH and uplink DPCCH on the same layer 1 connection generally are of different rates and thus have different spreading factors.

B- Downlink Physical channels:

There is one downlink dedicated physical channel, one shared and five common control channels; figure 4.4 shows only the frame structure of the DPCH channel:

a) Downlink dedicated physical channel (DPCH);
b) Physical downlink shared channel (DSCH);
c) Primary and secondary common pilot channels (CPICH);

d) Primary and secondary common control physical channels (CCPCH);
e) Synchronization channel (SCH).

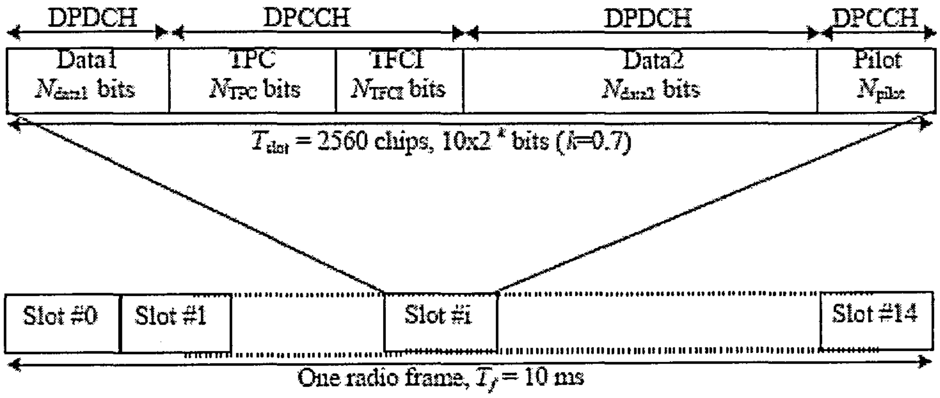


Figure 4.4 Frame structure for Downlink DPCH channel [62].

On the DPCH, the dedicated transport channel is transmitted time multiplexed with control information generated at layer 1 (known pilot bits, power-control commands, and an optional transport-format combination indicator). DPCH can contain several simultaneous services when TFCI is transmitted or a fixed rate service when TFCI is not transmitted. The network determines if a TFCI should be transmitted [62]. When the total bit rate to be transmitted exceeds the maximum bit rate for a downlink physical channel, multi-code transmission is employed (i.e., several parallel downlink DPCHs are transmitted using the same spreading factor). In this case, the layer 1 control information is put on only the first downlink DPCH. In this section we described WCDMA (UTRAN FDD) physical layer.

4.2.5 Spreading and Modulation

WCDMA applies a two-layered code structure consisting of an orthogonal spreading codes and pseudo-random scrambling codes. Spreading is performed using channelization codes, which transforms every data symbol into a number of chips, thus increasing the bandwidth of the signal. Orthogonality between the different spreading factors can be achieved by the tree structured orthogonal codes. Scrambling is used for cell separation in the downlink and user separation in the uplink [62].

In the uplink, either short or long spreading (scrambling) codes are used. The short codes are used to ease the implementation of advanced multiuser receiver techniques; otherwise, long spreading codes can be used. Short codes are S (2) codes of length 256 and long codes are Gold sequences of length 2^{41}, but the latter are truncated to form a cycle of a 10-ms frame. IQ / code multiplexing used in the uplink leads to parallel transmission of two channels, and therefore, attention must be paid to modulated signal constellation and related peak-to average power ratio (crest factor) [66].

By using the complex spreading circuit shown in Figure 4.5, the transmitter power amplifier efficiency remains the same as for QPSK transmission in general:

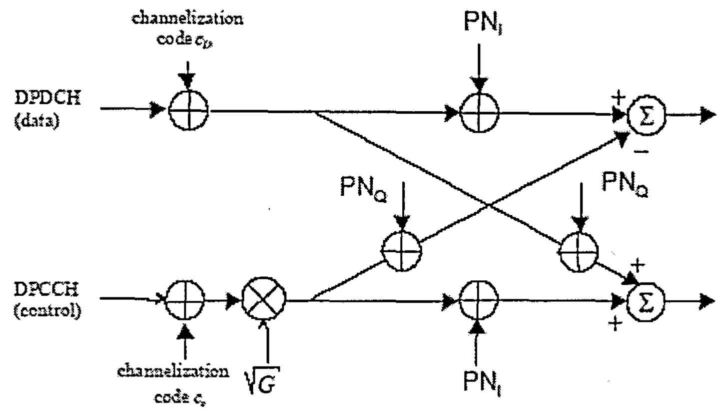


Figure 4.5 IQ/ code multiplexing with complex spreading circuit [62].

In the downlink, the same orthogonal channelization codes are used as in the uplink. For scrambling, Gold codes of length 2^{18} are used, but they are truncated to form a cycle of a 10-ms frame (i.e., 384,000 chips). To form a complex-valued code, the same truncated code is used with different time shifts in I and Q channels. It is possible to generate 2^{18}-1 scrambling codes, but only 8191 of them are used. Each cell is allocated one primary scrambling code. In order to reduce the cell search time, the primary scrambling codes are divided into 512 sets. Thus, the mobile station needs to search at maximum 512 10-ms-long codes [62]. The complex-valued chip sequence generated by the spreading process is QPSK modulated.

Figure 4.6 illustrates the modulation principle used in the uplink and downlink. The pulse shaping is root-raised cosine with roll-off factor 0.22 and is the same for the mobile and base stations:

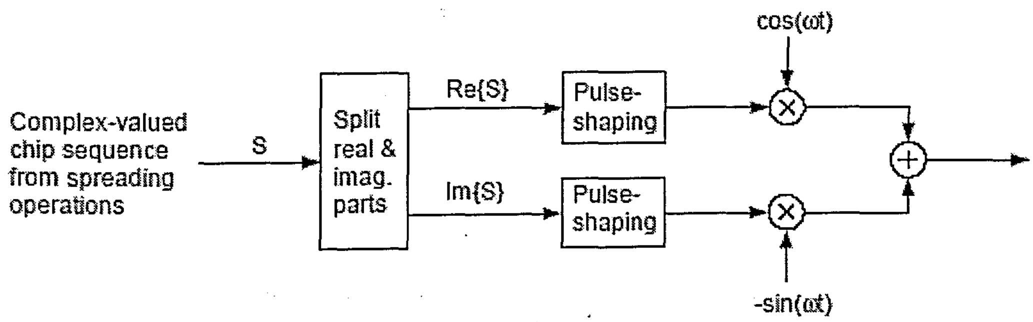


Figure 4.6 Modulation principle, (reproduced with permission from ETSI.) [62].

4.2.6 UMTS Network Architecture

The overall architecture of the UMTS system is shown in figure 4.7. The entire system can be divided into two main segments; the access network, called the UMTS Terrestrial Radio Access Network or UTRAN, and the switching and routing infrastructure, or Core Network (CN) [61]. The UMTS architecture consists of two network domains; the Circuit Switched domain which centres on the MSCs, and the Packet Switched domain, which centres on the GPRS Support Nodes (GSNs). The two domains thus rely on two separated and parallel backbones. The first backbone based on ISDN derived technologies, carries voice traffic, while the second is based on technologies derived from the IP world and transports data traffic:

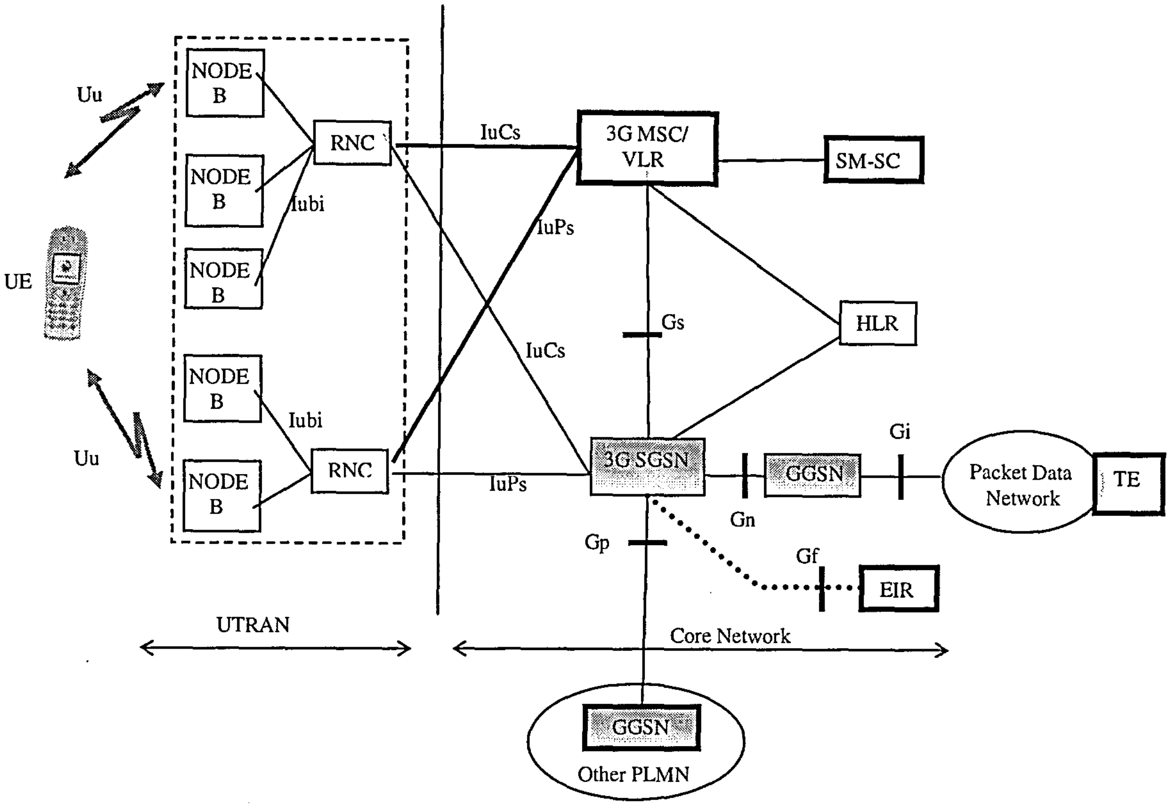


Figure 4.7 UMTS system architecture [61].

It should be emphasised that the circuit switched UMTS backbone is derived directly from the classic GSM network infrastructure, whereas the packet switched UMTS backbone derives from the infrastructure used to introduce GPRS in the GSM network. In fact while the UMTS access network is entirely new and separate from that used for GSM, the core network infrastructure is a direct evolution of the GSM infrastructure [61]. The core network (CN) could be connected to another network like PLMN through the GGSN.

1- UTRAN Elements and Interfaces:

As can be seen from the figure above, that the radio access network is bounded by two interfaces; on one side, the Uu radio interface stands between UTRAN and the mobile terminals, while on the other side, the Iu interface connects UTRAN to the core network. Actually the latter interface connects UTRAN to the circuit based CN (CS service) and the interface connecting UTRAN to the packet based Core Network (PS service). UTRAN consists of a set of Radio Network Subsystems (RNSs), which in turn consists of a controller, the Radio Network Controller (RNC) and one or more entities called Nodes B, which are connected to the RNC through the Iub interface. The RNSs connected to CN via the Iu interface [61].

A Node B superintends a set of cells which may be FDD, TDD or mixed. It should be kept in mind that the RNC is the boundary between the radio domain and the rest of the network. The protocols opened in the terminal to manage the air link (i.e. the radio protocols which cross the Iub and Iur interfaces) are terminated in the RNC. The UMTS standards are structured so that internal functionality of the network elements is not specified in details. Instead, the interfaces between the logical network elements have been defined. The following main open interfaces are specified [66]:

a) Cu Interface. This is the electrical interface between the UMTS Subscriber Identify Module (USIM) smartcard and the Mobile Equipment (ME). The interface follows a standard format for smartcards.

b) Uu Interface. This is the WCDMA radio interface, which is the subject of the main part of this book. The Uu is the interface through which the UE accesses the fixed part of the system, and is therefore probably the most important open interface in UMTS.

c) Iu Interface. This connects UTRAN to the CN similarly to the corresponding interfaces in GSM. The open Iu interface gives UMTS operators the possibility of acquiring UTRAN and CN from different manufacturers.

d) Iur Interface. The open Iur interface allows soft handover between RNCs from different manufacturers, and therefore complements the open Iu interface.

e) Iub Interface. UMTS is the first commercial mobile telephony system where the Controller-Base Station interface is standardised as a fully open interface. Like the other open interfaces, open Iub is expected to further motivate competition between manufacturers in this area.

2- Protocol Architecture:

The entire protocol architecture is divided into a Control Plane and User Plane. Actual data transfer is done within the user plane. We will restrict the overview on the protocol architecture to packet-switched data transmission modes. Circuit-switched data bearer uses the conventional GSM system structure in the CN and thus also uses those protocols [53]. The overall UMTS protocol architecture for user plane is shown in figure 4.8, and it is similar to the GSM GPRS architecture, especially in the core network:

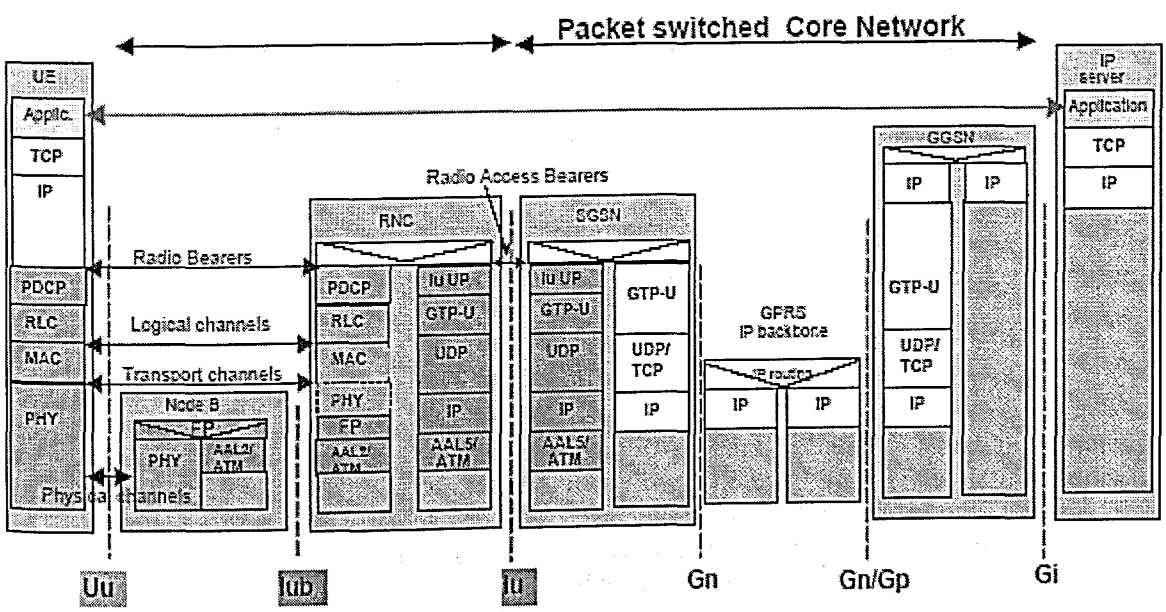


Figure 4.8 UMTS protocol architecture (user plane) [82].

4.2.7 UMTS QoS and Delay Distribution

End-to-End network services (i.e. between two terminals) are characterised by a certain QoS, which is provided to the user, who will thus have a personal perception of this quality. To provide a given network QoS, it is necessary to establish a bearer service, with specified characteristics and capabilities, from the service's source to its destination [61]. The bearer classes, bearer parameters and parameter values are directly related to an application as well as to the networks that lie between the sender and the receiver. The set of parameters should be selected so that negotiation and renegotiation procedures are simple and unambiguous [66].

1- QoS Classes:

In general, applications and services can be divided into different groups, depending on how they are considered. Like new packet-switched protocols, UMTS attempts to fulfil QoS requests from the application or the user. In UMTS four traffic classes have been identified [66]:

a) Conversational, **b)** Streaming, **c)** Interactive, and **d)** Background classes.

The main distinguishing factor between these classes is how delay-sensitive the traffic is: The conversational class is meant for very delay-sensitive traffic, while the background class is the most delay-insensitive. The UMTS QoS classes are summarised in table 4.2 [66]:

Table 4.2 UMTS QoS classes.

Traffic class	Conversational	Streaming	Interactive	Background
Fundamental characteristics	Preserve time relation (variation) between information entities of the stream Conversational pattern (stringent and low delay)	Preserve time relation (variation) between information entities of the stream	Request response pattern Preserve data integrity	Destination is not expecting the data within a certain time Preserve data integrity
Example of the application	Voice, videotelephony, video games	Streaming multimedia	Web browsing, network games	Background, download of emails

One of the key requirements of the UMTS air interface is the support for multiplexing different services with different QoS requirements on a single connection [82].

2- UTRAN Delay; Components and Estimation:

The retransmission of data streams will not take place over real time bearers. When retransmission is used in non-real time services, guaranteed delivery over the radio interface is performed by the RLC. The amount of retransmissions needed for a single transport block is a multiplication factor for delay, i.e. if it takes two re-transmissions to transfer a transport block successfully, then twice the physical layer delay would be added plus the delay needed to send the Negative Acknowledgement (NAK) information back to the MAC [83]:

$$\textit{Re-transmission delay} = N_{retransmissions} * \textit{Round trip delay}_{UE\text{-}SRNC} \quad (4.1)$$

The maximum number of allowed re-transmissions defines the weight introduced by this component according to the following formula [84]:

$$Max\ Re\text{-}transmission\ Delay = N_{retransmissions}Max * (ITC + PPO + MAC + MDC + AAL + MD + SD)*2. \qquad (4.2)$$

Where:

ITC = Interleaving and Turbo Coding delay
PPO = Packetisation, De-packetisation and End-System Play-Out delay
MAC = MAC Scheduling delay
MDC = Macro-diversity Combining delay
AAL = ATM Adaptation Layer Packetisation, Multiplexing and De-packetisation delay
MD = Media delay
SD = Switch delay

The following figure shows the position of the delay components described previously, considering also the components not included in the Access Stratum:

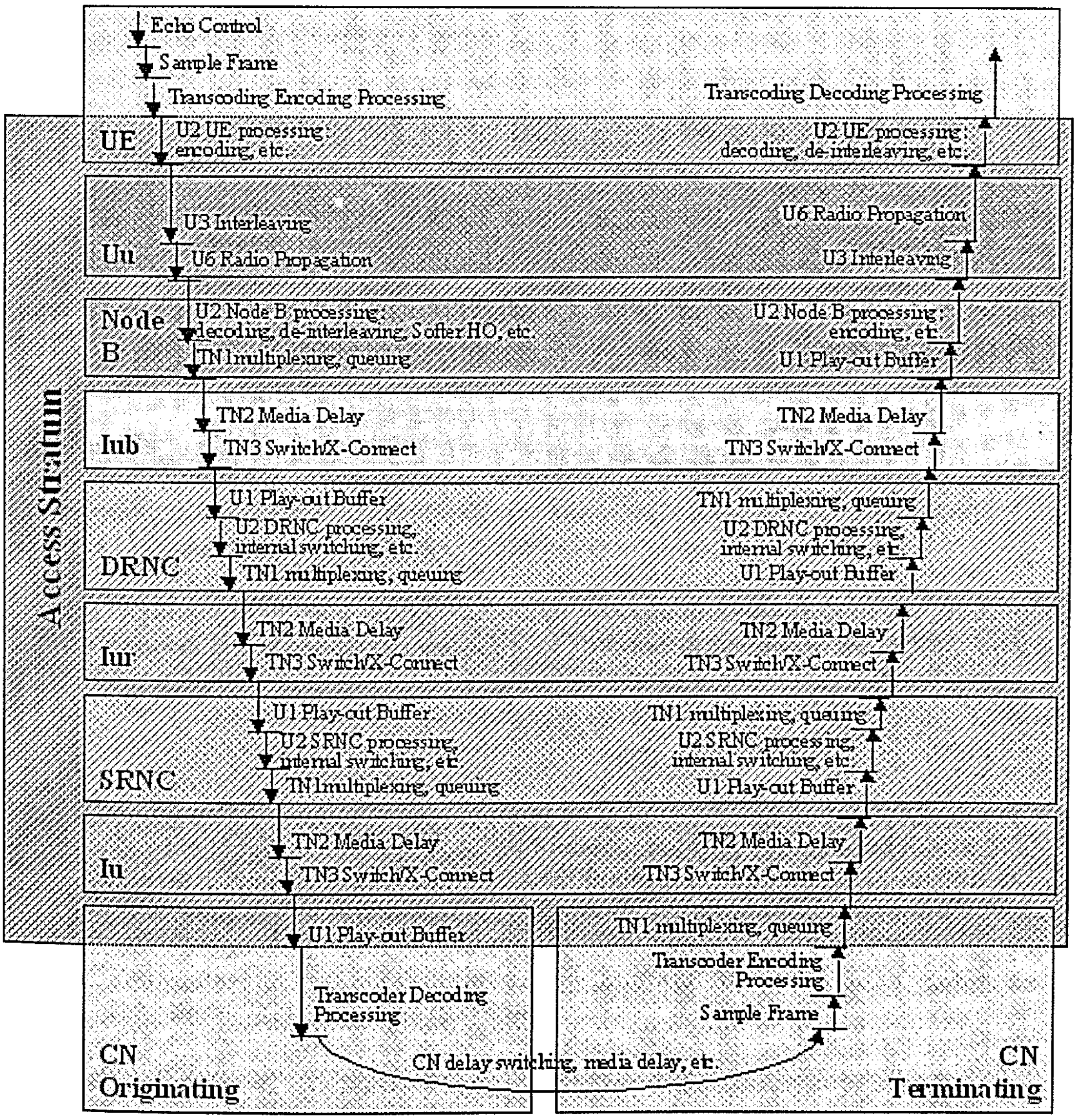


Figure 4.9 End-to-End Delay Chain [83].

This delay component can be reduced if prioritization mechanisms are used for data re-transmission. Node internal processing delay is due mainly to SW processing and to information transfer inside nodes. This component has to be considered as the engineered capacity of the network nodes, it is heavily implementation dependent and, therefore, only a rough global evaluation can be given.

To derive the following figures, [84] has been used as a guideline; from this starting point technological considerations and the above mentioned aspects have been considered to provide the following estimations:

For RNC (5 ms),
For DL Node B (2 ms),
For UL Node B, no turbo decoding (15 ms),
15 ms + 0.15ms x Throughput for UL Node B, with turbo decoding (throughput in kbit/s, evaluated by applying Maximum A-posteriori Probability (MAP) algorithm on a state-of-the-art DSP),
For UL UE (15 ms),
For DL UE, no turbo decoding (15 ms), and
For DL UE, with turbo decoding (20 ms).

4.3 UMTS OTELO System Simulation Studies

This section presents analytical study on the performance of the OTELO system over UMTS simulated environment. Since the Patient station Up-link is the BW demanding air interface, the Up-link end-to-end connectivity and transmission issues of the Ultrasound still images, Ultrasound stream of images, robotic control data and multiplexed data transmission of full system operational modalities has been studied. The simulated environment is configured using WAN Emulator / CLOUD emulator (**Appendix** A).

4.3.1 Simulation Environment

The simulation environment is applicable to the overall performance analysis test of the system, taking in to consideration the following issues:

a) Network parameters set-up was set to be as close as possible to the actual operating 3G network characteristics provided in our case and at the time of the test, by Vodafone/UK.

b) The Emulator represent homogeneous end-to-end network simulator, while the system connectivity on the operating network is heterogeneous, i.e. Wired end + Wireless end system.

c) Specific simulation environment parameters, example; Latency and Packet Loss have been imported from a real UMTS network behaviour under OTELO system characteristics, rather than applying theoretical parameters.

d) Transmitted packets overhead set to PPP overhead (6 Bytes to each packet) for the Packet Data Protocol (PDP) of the user plane.

The UMTS simulation environment for the end-to-end OTELO system connectivity is shown in figure 4.10:

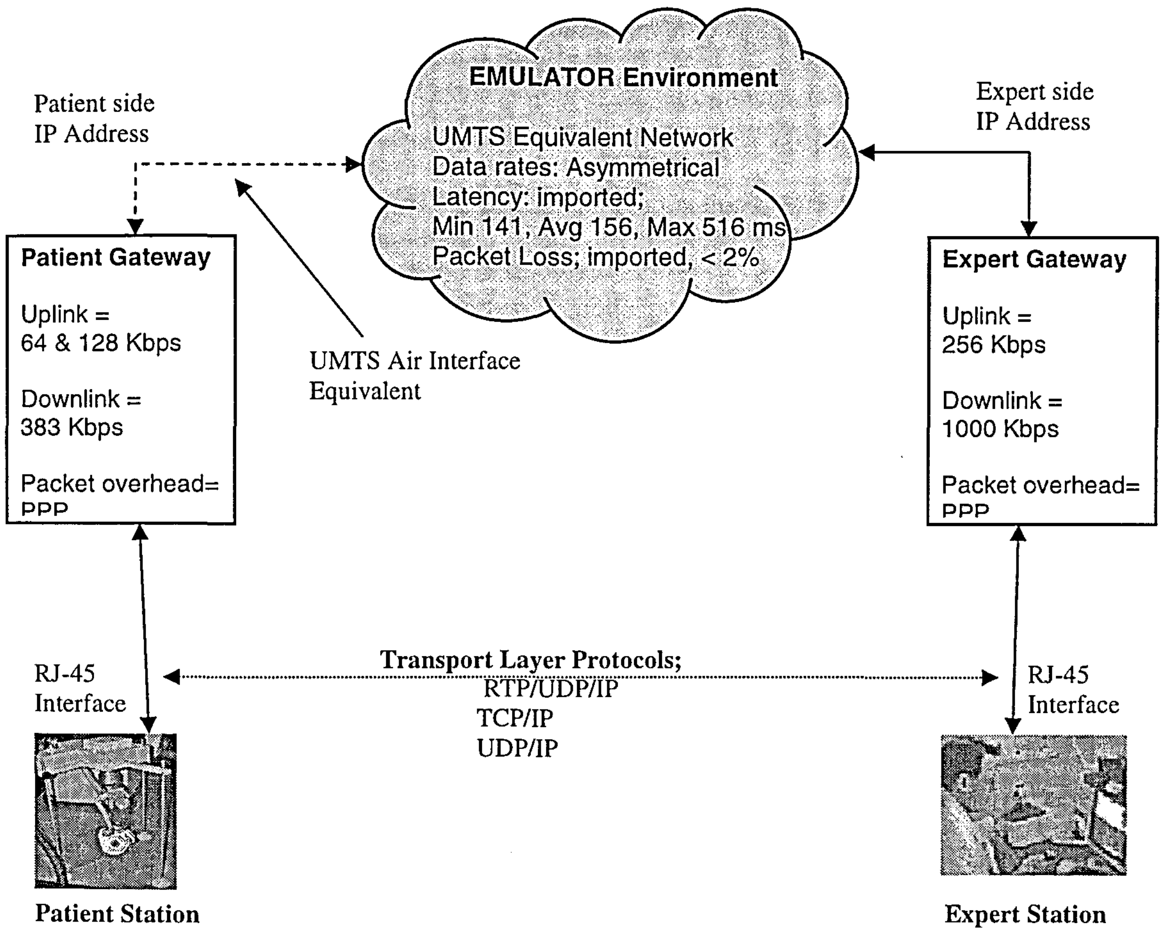


Figure 4.10 OTELO system simulator for UMTS equivalent network.

The classification of the OTELO traffic is mapped to the following traffic classes defined by the 3GPP UMTS QoS Classes [66]:

1. Streaming RT Class: the best-suited UMTS QoS class for video streaming, which preserves the time relation (variation) between information entities of the stream.

2. Nevertheless for medical image sequence with real-time (RT) requirements, the 'Conversational RT Class' would be necessary since our medical session implementation requires ultrasound data reception in conjunction with the Expert remote scanning. In addition to preserving the time relation between entities of the stream, it has conversational pattern (stringent and low delay) which is preferable for real-time operation.

The following functional modalities are considered for OTELO UMTS studies:

a) The same group of US medical imaging files with the compression specifications specified in table 3.3 of Chapter-3, has been implemented in the test for UMTS over both simulated environment and test-bed of real network. These arranged in five groups based on the compression ratios, transmitted from the Patient station to Expert station (Uplink direction).

b) The functional modalities needed for real-time ultrasound stream of images specifications specified in table 2.2 of Chapter-2 has been implemented over the same environments mentioned above.

4.3.2 Simulated End-to-End Performance of US Still Images Transmission

With the advent of data applications in mobile communications, TCP is making a de facto move to emerging 3GPP UMTS User Equipment. Quite often, those TCP flows originating from mobile terminals traverse some wired portions of the network. End-to-end TCP modeling should thus take into account the wired as well as wireless, mobile portions of the end-to-end path. The wired portion of the network is typically characterized by loss and TCP is well tuned to react, via retransmission, to this sole indicator of congestion [85].

We investigated the end-to-end performance at different Patient station Uplink data rates settings; 64 and 128 Kbps.

A- Throughput:

The traffic is generated by sending a bulk of Ultrasound images to the TCP layer, which performs flow control. The TCP stack was configured so that the maximum TCP segment is 1500 Bytes. Maximum buffer sizes has been used at both ends to avoid any packet discard on buffer over loading, since the focus is on the performance of the communication link rather the end-to-end UEs. Total throughput measurements in our experiment can be defined as the total throughput on the TCP layer accounting for TCP overhead and all retransmission.

The TCP throughput equation is given in Eq. 4.1 as a function of Packet loss rate and RTT for bulk transfer TCP flow [86]:

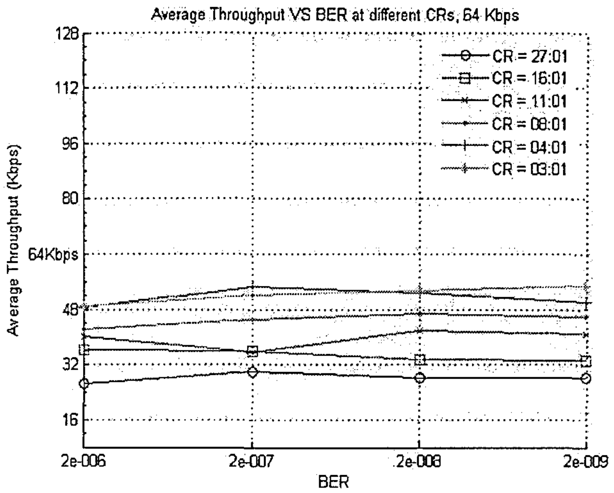
$$TCPthroughput = \frac{MTU}{RTT\sqrt{\frac{2P}{3}} + T_O\sqrt{\frac{27P}{8}}P\left(1+32P^2\right)} \qquad (4.3)$$

Where; (T_O) is the retransmission time out and *(P)* is the packet loss rate. This module captures not only the behaviour of TCP's fast retransmit mechanism but also the effect of TCP's timeout mechanism on throughput.

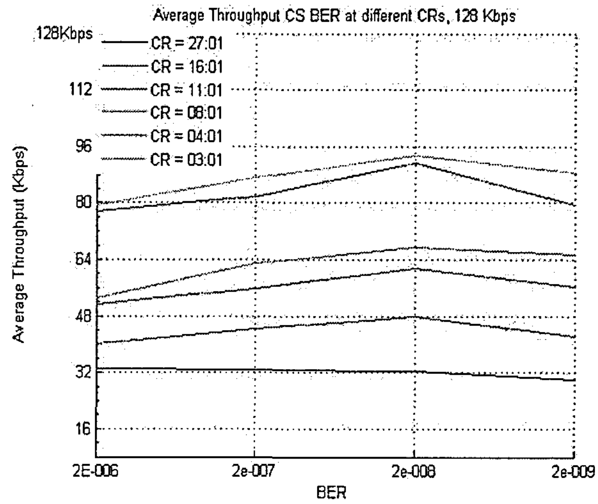
Increasing the TCP (MTU) size reduces the TCP/IP header overhead, thus improving throughput, but also increasing the interactive response time [78]. TCP header over head (η_0) can be defined by the following equation [53]:

$$\eta_o = \frac{Throughput(bit/s)}{PacketLength(bit/Packet)} * OverheadPerPacket(bits/Packet) \qquad (4.4)$$

Figure 4.11 below present comparative average throughput of the ultrasound image transmission as a function of the BER on the full communication path (which will then influence the RTT and the packet loss), and as a function of different compression ratio:



(a)



(b)

Figure 4.11 Average Throughput of the Ultrasound transmission. (a) over 64 Kbps. (b) over 128 Kbps.

Four different levels of BER have been chosen, these mainly cover the BERs rang of reliable operating 3G network. Clearly that BER spanning from 2 x 10^{-6} to 2 x 10^{-9} has no major influence on the average throughput as much as the change of the compression ratios for this system. The lower the compression ratio (i.e. higher image file size) the higher the TCP packets (segments) produced.

We noticed that the channel throughput efficiency on transmitting highly compressed images is not proportional to the channel bandwidth increment, for example the results at BER = 2 x 10^{-8} of the figure above with the corresponding bandwidth usage efficiency can be represented in the table 4.3:

Table 4.3 Summarise the results of figure 4.11 at BER = 2x 10^{-8}

Bandwidth Setting (Kbps)	Compression Ratio (CR)	Average Throughput (Kbps)	Bandwidth Usage Efficiency (%)
64	Lossy, CR = 27:1	28	43.75
	Lossless, CR = 03:1	53	82.8
128	Lossy, CR = 27:1	32	25
	Lossless, CR = 03:1	92	71.87

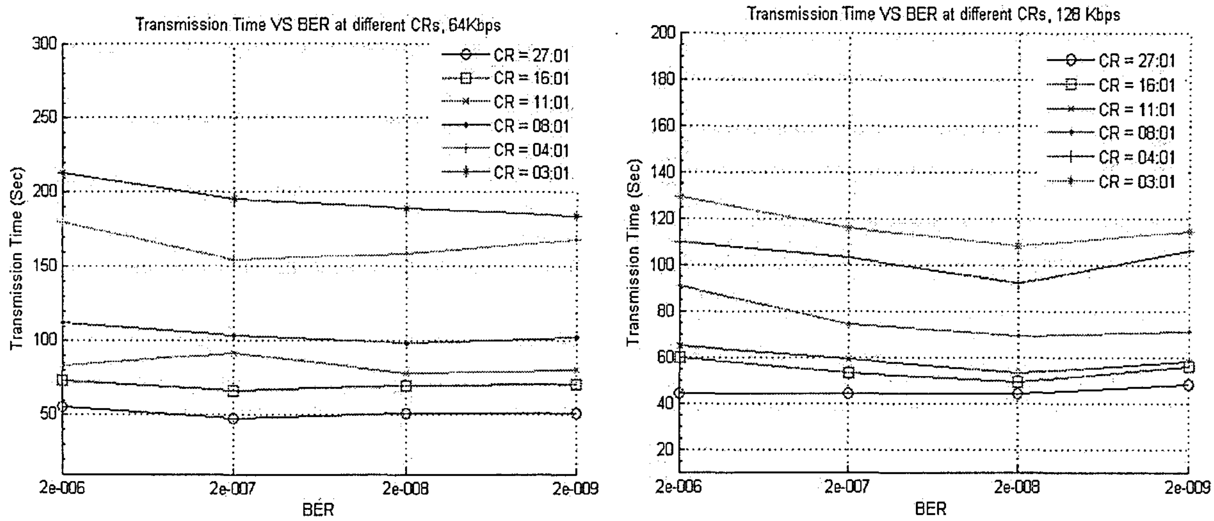
This shows that increasing the channel bandwidth, maximise remarkably the throughput efficiency on transmitting bigger image size rather than the smaller sizes, and this is due to; less significant number of ACK packets (ACK packet size is 52 Bytes, comprising 40 + timestamp) sent from the Expert side acknowledging Patient side TCP segments.

B- Transmission Time:

The transmission time at the bottleneck link in our experiment (the Patient station Up-link) is equivalent to:

$$TransmissionTime(Uplink) = \frac{MTU * 8}{Bandwidth}(ms) \tag{4.5}$$

If the wireless link delay were constant, the TCP ACKs arriving at the source would be evenly spaced with a constant duration, because of the delayed ACK feature of TCP (every 3 packets are acknowledged rather than every packet, as noticed in our test). The transmission time of the transmitted ultrasound images on both channel BW settings are shown in figure 4.12 below:



(a) (b)

Fig 4.12 Transmission time of each group of US images files with respect to BER, starting with lowest CR to as high as 27:1, (a) at 64 Kbps, (b) at 128 Kbps.

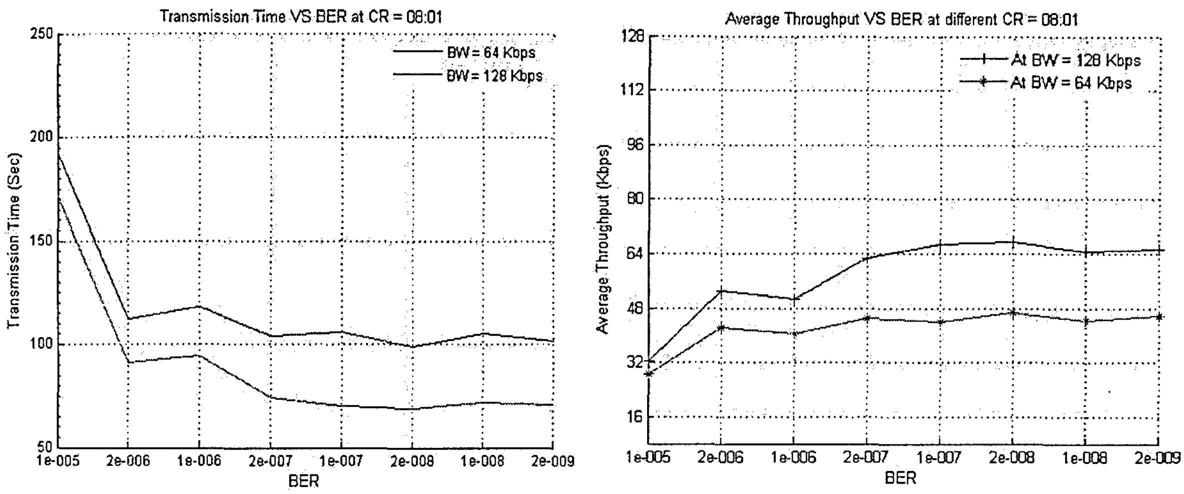
The transmission time calculation based on measuring the time needed for group of ten ultrasound images to reach the Expert station, and this took in consideration the time needed for TCP to establish new connection session for new image to be transmitted, which revealed additional delay not related to the channel transmission.

If we take the case when CR = 08:1 at BER = 2x 10^{-8}, the transmission time of a single MTU is; *(MTU * 8) / BW = (ms)*. This case can be summarised in the table below:

Table 4.4 Summarise the results of figure 4.12 at CR = 08:1 and BER = 2x 10^{-8}

Bandwidth Setting	MTU Transmission Time	Total TCP Segments Transmission Time
64 Kbps	185 ms	100 sec
128 Kbps	91.5 ms	70 sec

Practically the transmission time of a single image transmission (the first image chosen) is 4 sec at 64 Kbps and 7.5 sec at 128 Kbps channel bandwidths. Closer view on the effect of increasing the BW on the average throughput and the transmission time over wider range of BER, can be seen in figure 4.13, which present a comparative results over wider range of BER = 1 x 10^{-5} to 2 x 10^{-9}:



(a) (b)

Figure 4.13 Patient station uplink channel performance. (a) Transmission time. (b) Average throughput.

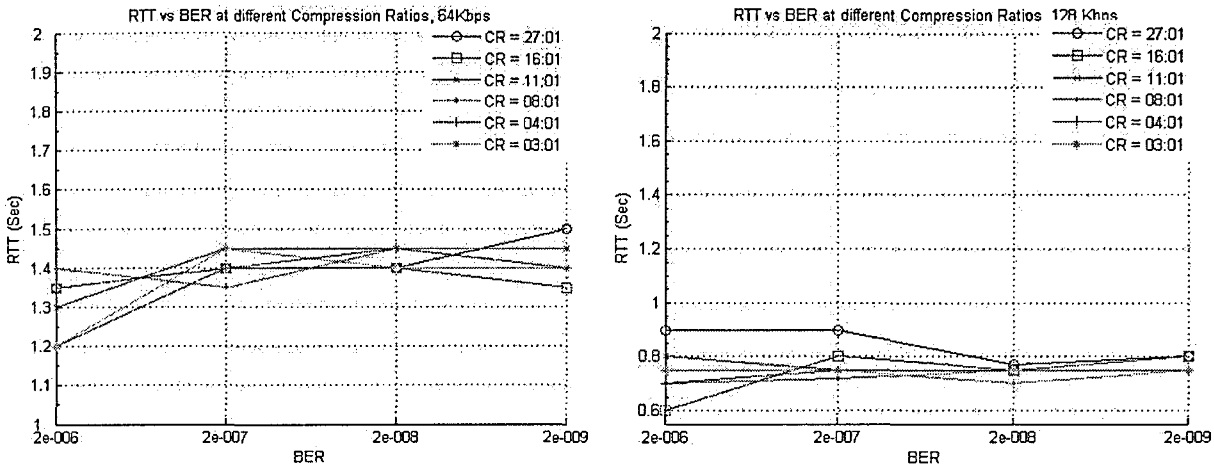
A dramatic improvement on the transmission time performance, once the BER dropped to 1 x 10^{-6} and onward, where very little efficiency noticed after then as the BER reduced to a values as low as 2 x 10^{-9}, and this explain that the loss probabilities are very rear at beyond BER = 2 x 10^{-7} as long as the MTU is around 1514 bytes with user data size of 1460 Bytes. Average throughput shown in part (b) of the figure above has been improved to about (68/128) when the BW increased to 128 Kbps.

3- Round Trip Time (RTT):

In the mobile and wireless context, RTT_{TCP} accounting for both air channel and wired network. Since the noisy wireless channel introduces large delay variations, the packet RTT will fluctuate sharply. Therefore, choosing a single packet as an observation sample cannot accurately reflect this variation [73].

In our analysis the method used to measure the "average" RTT during a period of time, is based on taking the average RTT of all TCP segment transmitted. RTT is used to maintain the TCP congestion window; which is an estimate of number of packets that can be in transit without causing congestion [78].

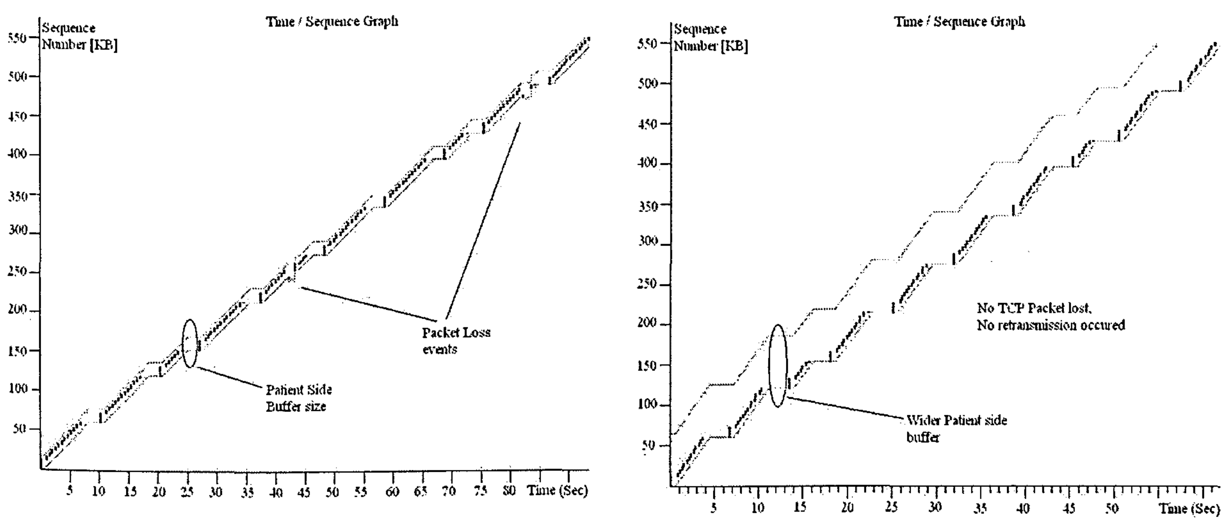
As noticed from figure 4.14 below the RTT measurements for different compression ratios of the ultrasound images, are neither BER nor compression ratio related:



(a) (b)

Figure 4.14 shows the RTT of TCP packets transmission; at different BERs and CRs. (a) 64 Kbps. (b) 128 Kbps.

Here the RTT measurements are much closer to the theoretical values of the channel performance on increasing the BW to 128 Kbps, as the RTT is directly depending on the transmitted TCP segment size and the available channel data rates. Compression Ratio has no effect what so ever on the RTT values. The issue above can be explored in close view by comparing the performance of the TCP segment transmission over 64 Kbps and 128 Kbps radio bearers, as in figure 4.15 below:



(a) (b)

Figure 4.15 TCP segments sequence number of the transmitted images with CR = 08:1 at Patient station. (a) 64 Kbps. (b) 128 Kbps.

Figure 4.15 shows the following characteristics:

a) The transmission time and the bytes accumulation of each individual image transmitted.
b) Events of re-transmission due to packet loss can be seen in (a) with 64 Kbps data rate, while this probability is much less with 128 Kbps data rate as in (b).
c) The Buffer size available for transmission over 128 Kbps still valid for higher throughputs as the Bandwidth Delay Product (BDP) of the link is approximately 5.13 KB.

Since the average ping latency of 1514 Bytes from the Patient to Expert side is around 0.319 sec (one way pinging 1514 Bytes) and an average ACK transmission time of 1.63 ms (represent 52 bytes / 256 Kbps for the Downlink BW) for 52 bytes packet length, the resulting BDP is [88]:

(319 + 1.63) ms * 128 Kbps = 5.13 KB, or about 3 packet size.

Concluded table of images transmission performance over the two radio bearers of interest, can shows the average throughput and the approximated transmission time. The measurements are based on BER = 2 x 10^{-8}:

Table 4.5 Transmission time and Average throughput of the Still images, over 64 & 128 Kbps.

Image size Bytes	Compression Ratio (CR)	64 Kbps		128 Kbps	
Uncompressed 416 KB		Transmission Time / Image Sec.	Average throughput Kbps	Transmission Time / Image Sec.	Average throughput Kbps
119 KB	03:1	18.9	53.4	10.8	93.4
99.4 KB	04:1	15.9	52.8	9.2	91.2
54.6 KB	08:1	9.9	46.9	6.9	67.3
38.1 KB	11:1	7.8	41.8	5.3	61.5
27.2 KB	16:1	7.0	33.5	4.9	47.8
16.3 KB	27:1	5.1	28.0	4.4	32.5

4.3.3 Simulated End-to-End Ultrasound Streaming

The same testing platform shown in figure 4.10, is used for the investigation of the performance of OTELO system on full real-time end-to-end ultrasound streaming described in chapter two. That means on the Patient station Uplink channel;

- Robot Control, Ultrasound stream and or Ambient video and Sound transmitted, and
- Robot Control and Sound received from the Expert station.

The simulation platform does not support QoS management; hence the best effort of QoS has been considered within the application software at both system ends. We have assessed the Patient station Uplink channel performance over both radio bearers 64 & 128 Kbps, in terms of the following performance matrices:

a) Peak throughput of the Ultrasound stream seen by the Expert station.
b) Distribution of the Data Rates over chosen number of transmitted Ultrasound packets.
c) Delta Time delay of the received packet at the Expert station. This implies the speed of the packet reception at the Expert station.
d) Robot control peak throughput and the packets data rates distribution.
e) Delay Jitter of the received ultrasound packets at the Expert station.

The 3GPP PS multimedia streaming service is being standardized based on control and transport IETF protocols as Real-Time Streaming Protocol (RTSP), RTP and Session Description Protocol (SDP), as figure 4.16 shows. RTSP is an application level client-server protocol, which is used to control the delivery of real-time streaming data. RTP transports media data flows over UDP, in the same way as its related control protocol RTCP. RTP carries data with real time requirements while RTCP conveys information of the participants and monitors the quality of the RTP session [89]:

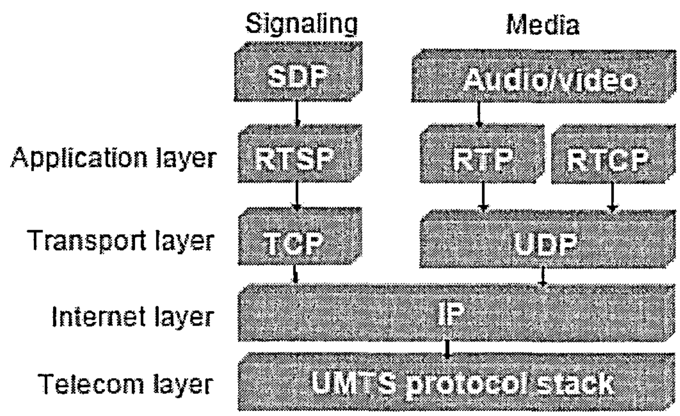


Figure 4.16 Protocol stack for signalling and media flows of 3G streaming services [89].

A- Uplink Throughput Efficiency:

The ultrasound stream seen by the Expert station represents the ultrasound packet with source coded at the source by H263 decoder, and transmitted over the heterogeneous and time variant nature of an IP network [90], this stream typically has bandwidth delay and loss requirements [72]. Figure 4.17 shows comparative results of the peak throughput of the ultrasound packets received by the Expert station:

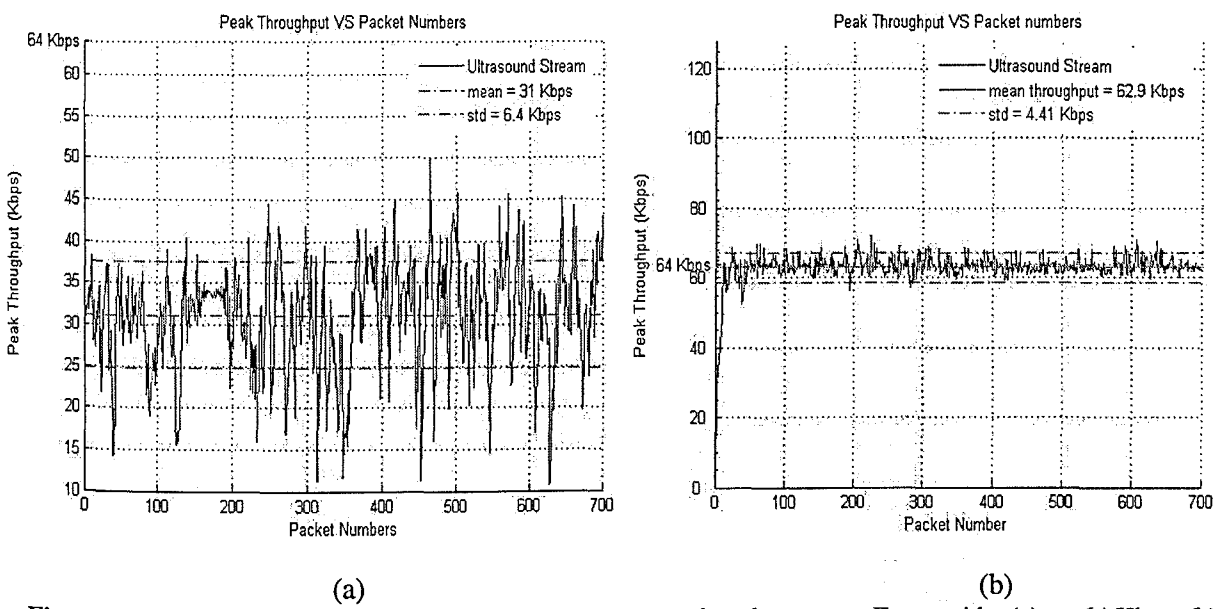
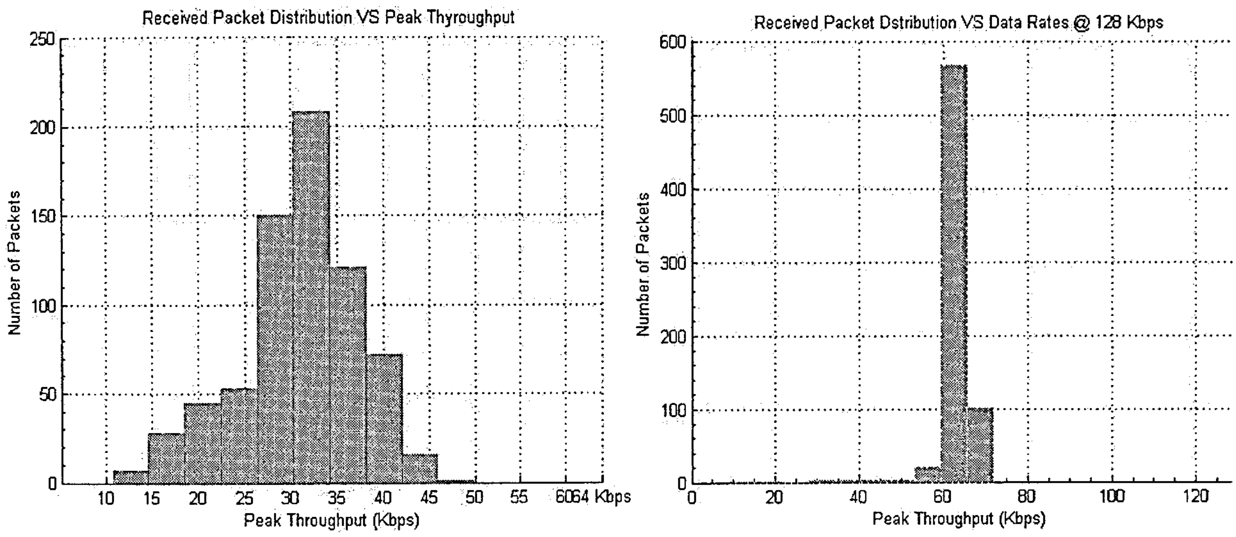


Figure 4.17 Instantaneous (Peak) throughput of the ultrasound packet seen at Expert side, (a) on 64 Kbps, (b) on 128 Kbps.

In both cases the throughput fluctuation is governed by many parameters; first that the Patient side multiplexing and transmitting simultaneously three streaming data types, Ultrasound, Robot Control and sound over IP; second the coding algorithm for ultrasound image stream dictates most of the QoS requirements (our trans-coder is H263); third is the ultrasound motion intensity, which is varying depending on the ultrasound scanning process. 700 packets received over 140 second have been taken as a sample for the analysis.

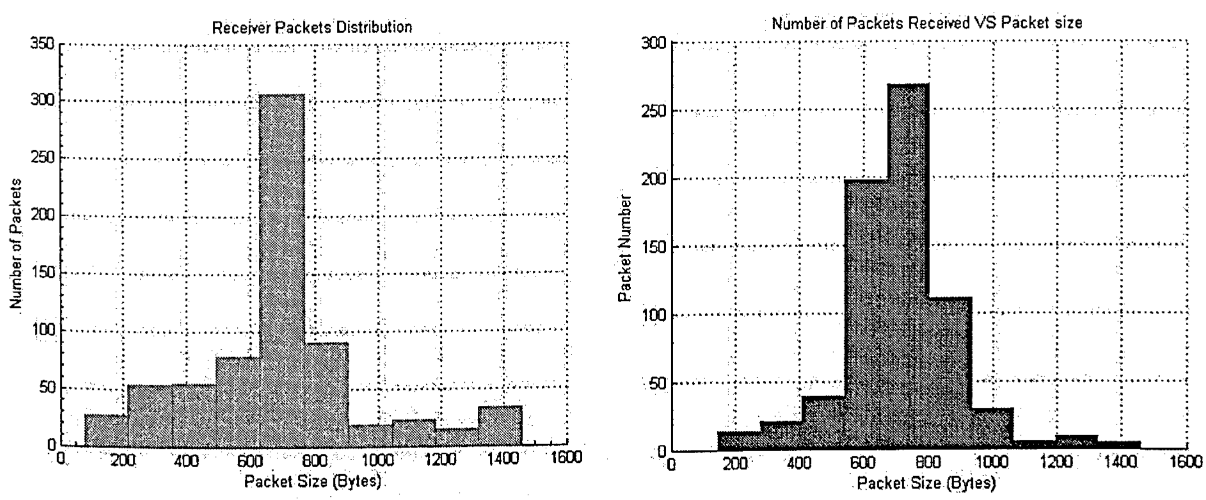
Although the channel usage efficiency is around 50% in both cases, nevertheless the radio bearer 128 Kbps shows improved efficiency in terms of lower standard deviation (equal to 4.41 Kbps). Figure 4.18 is further explaining the channel usage efficiency by showing the distribution of peak throughput at each case:



(a) (b)

Figure 4.18 Peak throughput distributions of 700 packet samples, (a) on 64 Kbps (b) on 128 kbps.

81% of the packets on the 128 Kbps radio bearer are received at main speed of around 65 Kbps, while the case with 64 Kbps radio bearer, only 30% of the packets received, were at a peak throughput of 31 Kbps. Although the video trans-coder produces higher packet sizes as shown in figure 4.19 (b), where a comparative ultrasound packet size shows the distribution of 700 packets over the packet sizes over both radio bearers cases:



(a) (b)

Figure 4.19 Distribution of the ultrasound packet sizes on both channel data rates, (a) 64 Kbps, (b) 128 Kbps.

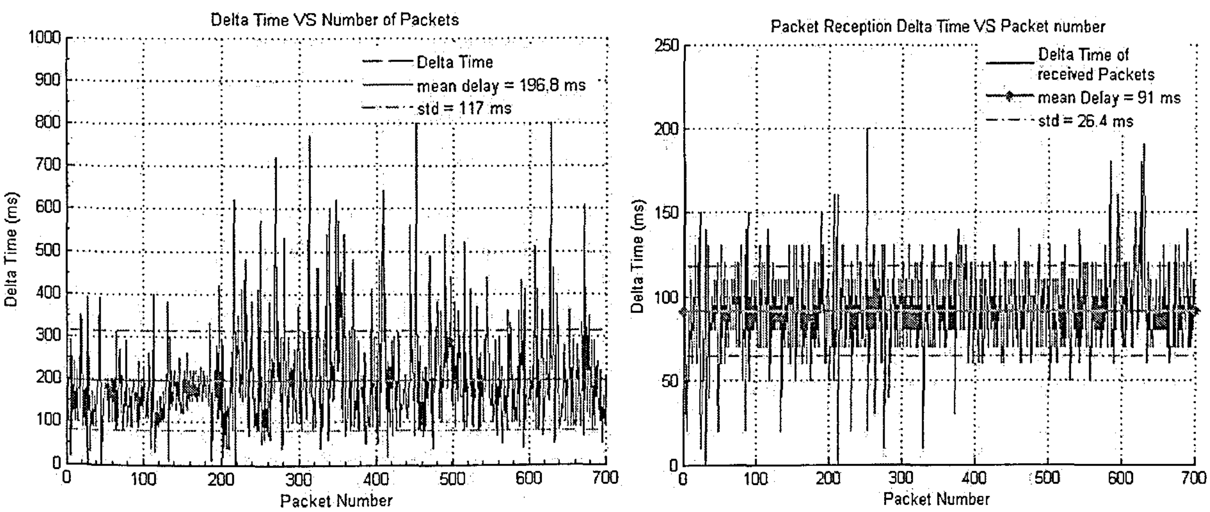
It is known that every transcoding process has some disadvantages; transcoding always adds some additional delay to the stream, and the RTP (packet number, payload type, flags, etc) as well as the RTCP information (packet and byte counter, jitter, etc) have to be recalculated [92].

B- RTP Packet Reception Delay:

As mentioned previously, OTELO system real-time performance has very restrict latency requirements as its performance classified within the conversational class. Our experiment assumes that the connection is established continuously for medical data transmission; hence the delay encountered is excluding the delay due to connection establishment. A comparative analysis of packet latencies shows the delta time of the ultrasound packets arrival at the Expert station, which was subject to the following effecting parameters:

a) The buffering process, at the Patient station and through the communication link.
b) The sound and robot control data random transmission, which occupies the channel bandwidth and degrade the ultrasound packets arrival.
c) Channel bandwidth reliability, especially at low data rates.

Figure 4.20 presents the ultrasound packets delta time on both radio bearers. Since we were using RTP/UDP/IP protocol, then no out of order packets found by the Expert station, the packets received at the Expert side followed the same sequence on leaving the Patient side, therefore the delta time shown in the figure below is proportional to the delta time at the uplink channel of the patient side, except that the network path added latencies to each packet correspond to its length and depending on the channel condition in terms of congestion and routing algorithm:



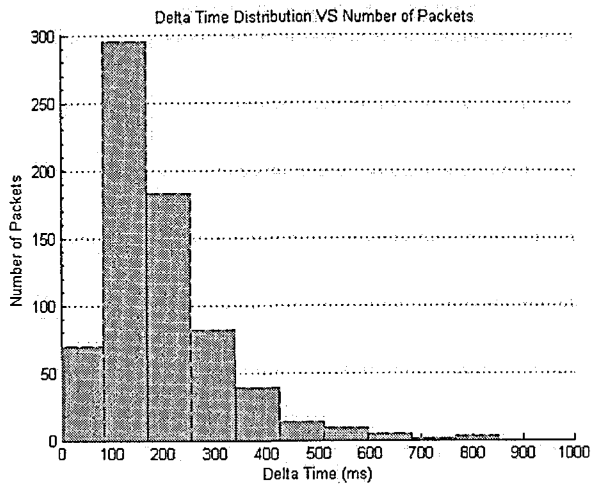
(a) (b)

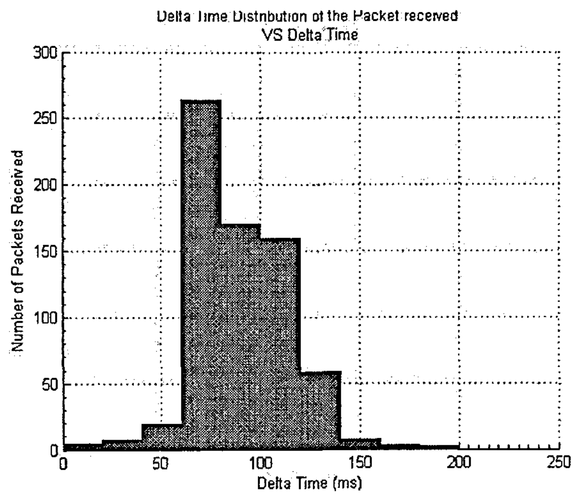
Figure 4.20 Delta time of the Ultrasound packets received by the Expert station, (a) 64 Kbps, (b) 128 Kbps.

The delta time above does not represent the end-to-end delay time across the network, it represent the time difference between a specific packet received and the previous one. This is very important factor in allowing the Expert side to play the received ultrasound stream packets as close as possible to real-time. In figure 4.20 (a) the variation in time is very obvious, as it shows high standard deviation = 117 ms, furthermore the high delta time spikes are due to channel reliability and congestion other than packet sizes. In (a) above during the packet sequence number 110 – 190, the best reliability found, which correspond to the main peak throughput and highest packet distribution as showed in figure 4.17 (a) and figure 4.18 (a) respectively.

Moving to figure 4.20 (b), the gap between the higher and lower delta time is narrowing to give a standard deviation of about 26 ms, which reveals in general 65 to 117 ms minimum to maximum delta time. Therefore increasing the bandwidth is not just improving the mean delta time;

additionally improves the deviation of that delay. A comparative delta time distribution is shown in figure 4.21:



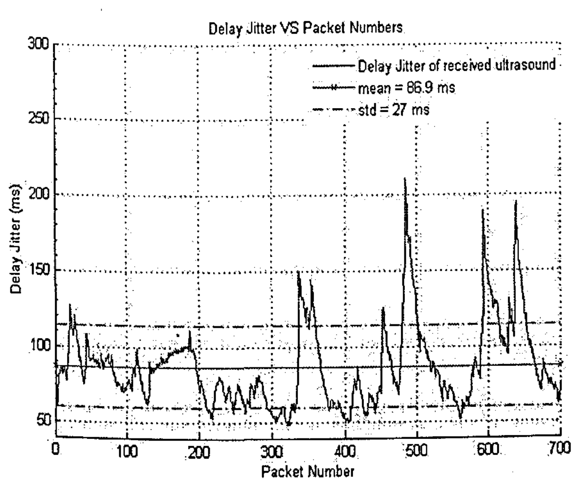


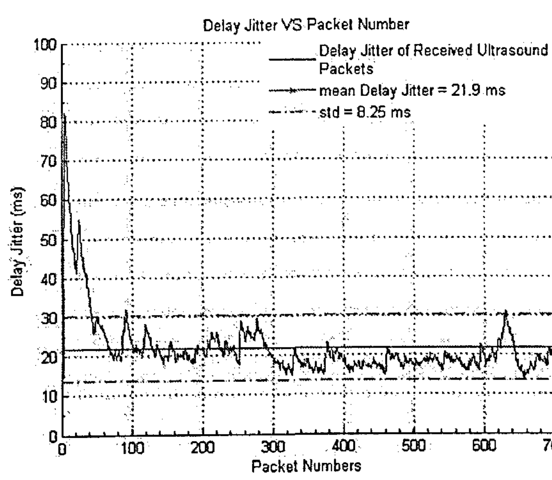
(a) (b)

Figure 4.21 Ultrasound packet delta time distributions, (a) on 64 Kbps and (b) on 128 Kbps channel.

In (b) the simulation results indicates that nearly 92% of the packets, has a delay distributed over a narrow delta time (approx. 60 to 140 ms), if compared to figure 4.17 (a), 88% of the packets has a delay distribution over 50 to 300 ms. This can be attributed to less packet queuing delay through the buffering across the link, when assigning higher bandwidth, therefore we noticed from figure 4.17 (b) higher main throughput = 62.9 Kbps, nearly twice as large as the resulting main throughput at 64 Kbps assigned bandwidth.

The parameters that guarantee a real time ultrasound streaming performance are end-to-end delay, congestion and packet loss, and most effectively in this case is the delay jitter. Jitter is often quantified with a statistical standard deviation. The transmission over 64 Kbps data rate channel shows much higher delay jitter compared to the delay jitter encountered by the test over 128 Kbps channel, as shown in figure 4.22:





(a) (b)

Figure 4.22 Delay jitter measurements of the ultrasound packets received by the Expert station, (a) on 64 Kbps channel and (b) on 128 Kbps channel.

Generally jitter behavior represents the large variation in end-to-end delay for video streaming. We noticed that the spikes of the delay jitter over 64 Kbps could last over wide packet range (packets sequence from 590 to 670) as in figure 4.18 (a), that is received over 60 sec, whilst the wider spike of delay jitter over 128 Kbps was for around 30 packets (see the samples 250 to 280 of figure 4.22 (b)), these are received over about 7 sec.

Transmitting the ultrasound packets over 128 Kbps, improve the delta time by around 50%, in addition to that it is remarkably improved the delay jitter by up to 75% by reducing the mean delay jitter to 21.9 ms and the corresponding standard deviation to 8.25 ms, as seen in figure 4.22 (b).

4.3.4 Robotic Control Data Analysis

Robotic control data exchanged by both system ends in conjunction with the ultrasound stream transmitted from the Patient station, is characterise the QoS class of OTELO system under the conversational class. Robot control data is highly mitigated to channel instability, due to its constant packet size, smallest packet size (payload equal to 16 Bytes) and constant source generation (every 70 ms). Robotic control application software was implementing UDP/IP protocol; since re-transmission is impractical with real-time robot control process, where out of order packet may cause robot control malfunction.

Figure 4.23 present the peak throughput behaviour of 200 control packets transmitted, and received by the Expert station (this data is multiplexed at the application layer with other medical data at the Patient side):

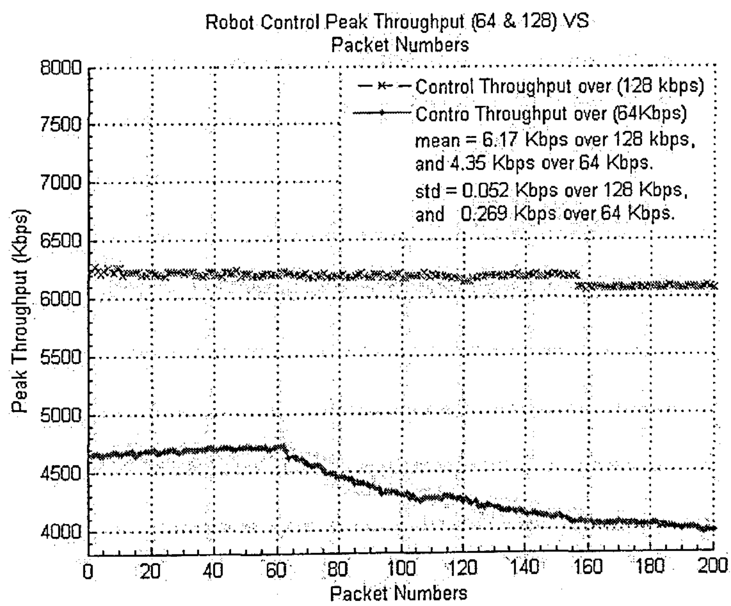
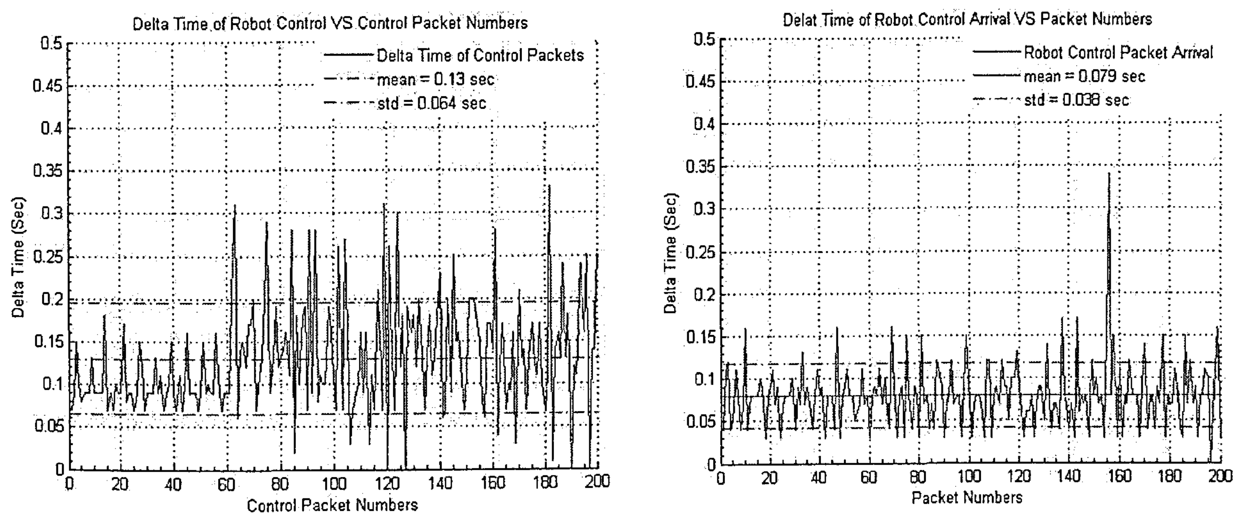


Figure 4.23 Robot control data peak throughput over 64 & 128 Kbps radio bearers (at Expert station).

Average throughput of 4.35 Kbps is achieved for the robot control over 64 Kbps channel, was sufficient to perform real-time robotic control, since the average end-to-end of the robot control data is not exceeding 130 ms as found by ping the Expert station from the Patient station. Accordingly and due to low peak throughput standard deviation (around 269 bps), real-time robot control has been accomplished in even slightly unreliable channel data rates (the case with 64 Kbps). The trace related to the performance on 64 Kbps radio bearer, has been chosen to show the minimum and maximum throughput fluctuation, 4 to 4.7 Kbps respectively.

Performing over 128 Kbps radio bearer shows more efficient peak throughput and much higher reliability in terms of low standard deviation = 52 bps, and peak throughput ranging from 6.1 Kbps to 6.25 Kbps. In this case a trade off between the quality of the robotic control and the received ultrasound stream at the Expert station can be managed by increasing the delta time (time interval) of the control data generating source to around 100 ms instead of 70 ms, which would then spare a very marginal portion of the bandwidth to the ultrasound streaming.

The peak throughput of the robot control data seen in figure 4.23, is the Patient station robotic head axes positioning information continuously updating the Expert station with the new position of the robot head (which is holding the ultrasound probe). Therefore the delta time variation shown in figure 4.24 is not purely related to the latency and delay jitter countered by only the link condition, furthermore by the ultrasound packets formation and transmission rate:



(a) (b)

Figure 4.24 Delta time of the robot control data on (a) over 64 Kbps channel, (b) over 128 kbps channel.

Control data sent by the Patient station, is generated every 70 ms at a payload of 16 Bytes. Apart from the end-to-end delay, the link is adding further delay to each packet with respect to the previous one:

- At 64 Kbps; mean Delta time – 70 ms = 130 - 70 = 40 ms and standard deviation is 64 ms.
- At 128 Kbps; mean delta time equal to 9 ms and standard deviation is 38 ms.

The robot control data streaming is considered as the background traffic to the uplink channel of the Patient station, since the average performance characteristics are much lower with respect to the ultrasound stream leaving the Patient station as presented in the table below:

Table 4.6 Comparative results on the performance of the simulated Patient station Uplink Channel.

Data	Radio Bearer	Mean Throughput (Kbps)	Mean Delta Time (ms)	Standard deviation	
				Delta Time (ms)	Mean Throughput (Kbps)
Ultrasound Stream	64 Kbps	31	196.8	117	6.4
	128 Kbps	62.9	91	26.4	4.41
Robot Control Data	64 Kbps	4.35	130	64	0.269
	128 Kbps	6.71	79	38	0.052

4.4 Real-Time 3G Performance Analysis

This section presents the experimental results of OTELO system operating on real 3G (UMTS) network, provided by Vodafone / UK. The wireless connectivity of OTELO system is depicted in Figure 4.25. The Expert Station can also be connected to an IP Multimedia Network or to ISDN, ADSL network to provide the wired connectivity part between the remote (Patient side) and the Expert side if such connectivity options are required and available [91]:

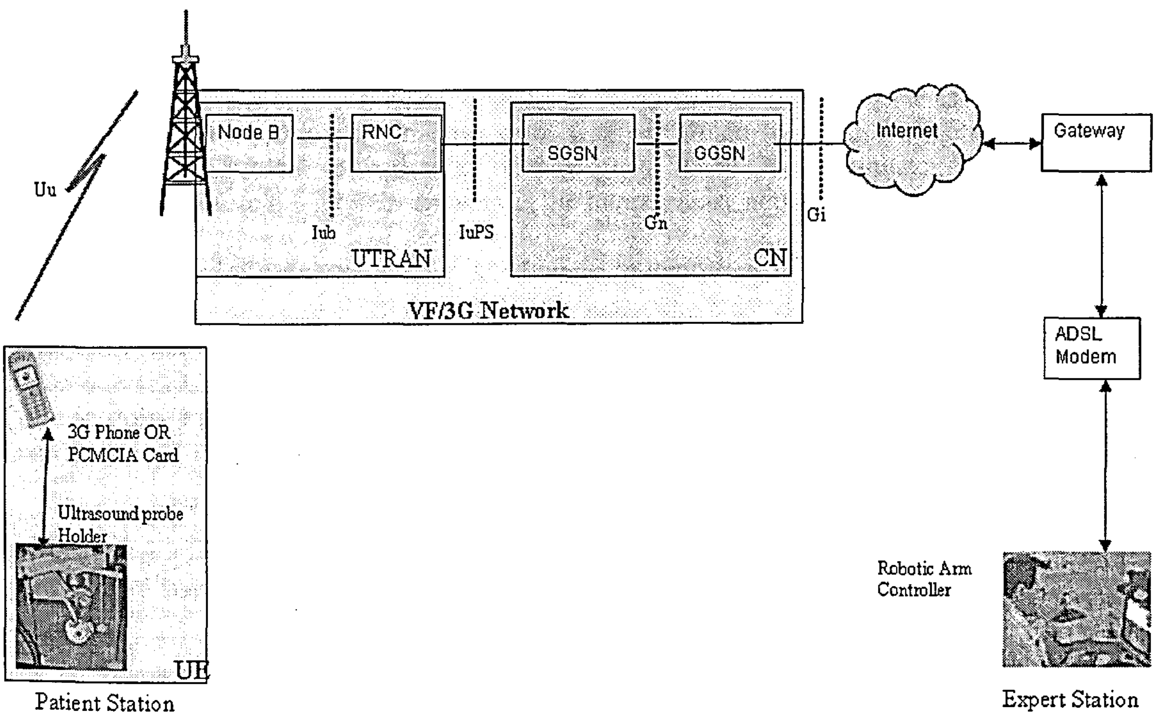


Figure 4.25 3G (UMTS) Wireless OTELO Connectivity scheme.

The radio interface of the UMTS network is identified by the UTRAN part of the connectivity above. The UMTS User Equipment (UE) is connected to the (RNC) of the UTRAN via a Dedicated Channel (DCH) in both Up-link and Downlink directions. The RNC is connected to the Internet via the 3G-SGSN and the 3G-GGSN of the cellular system's CN [93]. Finally the UE establishes a data connection with the Expert station connected to the Internet via ADSL line.

The 3G experimental setup is designed to measure the End-to-End system performance when transmitting individually either ultrasound still Images or ultrasound image stream, by analyzing the data characteristics at the other end (Expert station), which would then clarifies the performance of the patient station uplink and the wireless communication link effects on the performance of the real-time robotic system functionalities. In particular the following performance matrices are measured:

a) Average and Peak throughput of ultrasound still images, ultrasound stream and Robot control data transmission.
b) End-to-end transmission time and Delta Time of the received packets.
c) End-to-end delay and delay Jitter.

Both Patient and Expert stations are located on the same site (Lab / Kingston University, London). The traffic analysis of the end-to-end wireless / wired link was carried out where the desired packets were separated from the background traffic including the undesirable IP packets generated by other applications using the link. The expert station is connected to the public network through a broadband ADSL Modem of the following ADSL line specifications:

a) Downstream bandwidth = 1024 Kbps
b) Upstream Bandwidth varying from 64 Kbps to 256 Kbps
c) One Static IP address

To establish the connection to the 3G network, the Patient station is hardwired serially (USB connector) to a 3G Phone or to a PCMCIA card when a Laptop is used as the patient station functional computer. Our experimental work was performed by using Vodafone 3G PCMCIA (**Appendix B**) operational network card, operating as a 3G Modem to connect the Patient station to the Internet, therefore the internet network is the intermediary network between both system ends.

The wireless portion introduces new deficiencies, notably error, and new performance measures, such as Frame Error Ratio (FER) or BER. It is under the control of the Data Link Control (DLC) layer error detection/retransmission mechanisms, particularly ARQ, as well as Forward Error Correction (FEC) [94]. The connection protocol to the service provider backbone network was the Point to Point Protocol (PPP), which is specified in RFC 1661 [95]. The user plane for Packet Data Protocol (PDP) type PPP over UMTS network is shown in the figure 4.26:

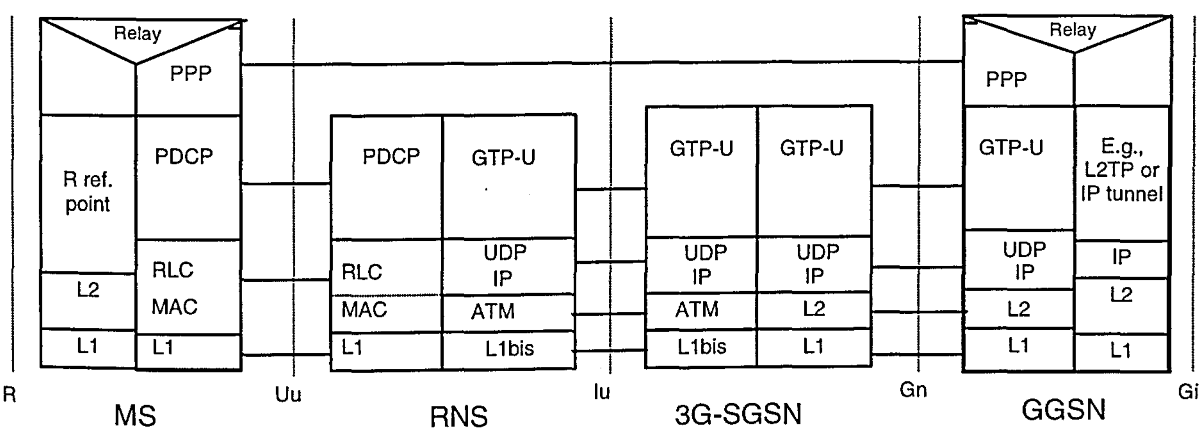


Figure 4.26 UMTS User Plane for PDP Type PPP protocols [95].

The user plane for the PDP type PPP consists of a PPP protocol stack above SNDCP for GSM or above PDCP for UMTS in the MS, and above GTP in the GGSN. The GGSN may either terminate the PPP protocol and access the packet data network at the IP level or further tunnels PPP PDUs via e.g., L2 Tunnel Protocol.

To establish a continuous and maintained connection between both system ends, a remote control manager program establishes a real-time connection and both ends exchange at the beginning of the connection, TCP/IP messages that keep the state of both workstations in synch. Once the connection established the manager switch to use UDP/IP protocol to minimise the latency between the system ends, and furthermore UDP is the convenient protocol to be used for the robot control with reasonable packet loss, instead of TCP due to practical fact that control packet re-transmission is irritating and confusing the robot control process [96].

Separate conferencing program is also running to tele-conference the system ends either through the ultrasound stream transmission and or the ambient video connectivity.

4.4.1 Ultrasound Still Image Transmission (Uplink)

Group of ultrasound images were uploaded to the Expert station via WCDMA air interface of the 3G operating network. The transmission protocol used is TCP/IP, which is the most popular transport layer protocol on the Internet, which offers a reliable byte stream service. TCP provides transparent segmentation and reassembly of user data and handles flow and congestion control [87]. As 3G networks provide pervasive internet access, good performance of TCP over these wireless links will be critical for end user satisfaction [88]. In order to mitigate the effects of losses, 3G wireless link designers have augmented their system with extensive local retransmission mechanisms, for example link layer retransmission protocol such as RLC is used in UMTS, this mechanism ensure packet loss probability of less than 1% on wireless link, thereby mitigating the adverse impact of loss on TCP [88].

It is known that TCP functionality becomes problematic over wireless link, the TCP assumption that all losses are due to congestion is quite problematic over wireless links. The prime concern for TCP is congestion, which occurs when routers are overloaded with traffic that causes their queues to build up and eventually overflow leading to high delays and packet losses [78]. The same group of ultrasound images that compressed between 03:1 to lower compression of 27:1 shown in table 3.3 of chapter 3, has been used. The test performed at different network peak time and the results shown to be satisfactory and close to each other.

A- Throughput:

The total numbers of MTUs transmitted were significantly image sizes dependant, the MTU selected was 1486 Bytes with user data of 1423 Bytes. The buffer at the RNC assume to be larger than the TCP window size [88], thus the transfer resulted in minimal (as found with our test) TCP packets and a maximal throughput [88]. To approach a realistic average throughput rate for the Patient station Uplink direction, we consider the net TCP packets throughput captured for the transmitted data by the Patient side, which presents the average throughput of 10 ultrasound images, transmitted sequentially to the Expert station as seen in figure 4.27:

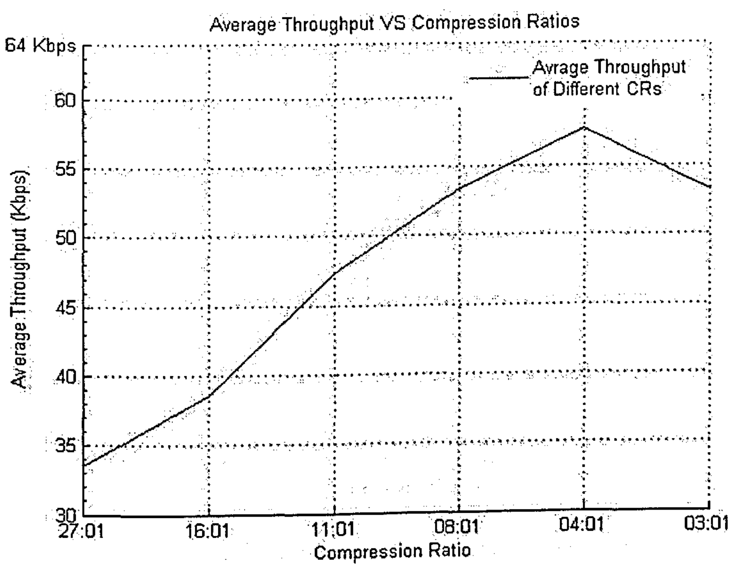


Figure 4.27 Average throughput of the ultrasound images group sent by Patient station at different CRs.

The throughput at the transport layer was evaluated using multiple FTP sessions (ten ultrasound images) from the UE (Patient station) to the Expert station. That means, the only data going in the Patient station Downlink is TCP ACKs. The FTP session run for as long as it takes to complete ten ultrasound image transmissions. TCP throughput is defined as the amount of successfully received TCP segments at the TCP layer in bit per second [97].

These results indicates that the images compressed at 04:1 gives higher throughput, while compression ratio of 27:1 gives around 33 Kbps throughput, although the net throughput for each images with different compression looks very close. Selected image sizes performance can be summarised in the table below:

Table 4.7 Net throughput versus different CRs.

Image size (KB)	Compression Ratio (CR)	Net throughput (Kbps)
18	27:1	62.5
153	03:1	66

Hence for TCP to maintain congestion window, it starts at one packet, with new ACK causing it to be incremented by one, thus doubling after each RTT, this is the slow start phase (exponential increase) [78]. Therefore lower size images does not help the congestion mechanism to maintain the slow start gradually and reaching the maximum throughput, while higher images sizes can do so. Lower sizes ultrasound images can be transmitted even within the process of TCP slow start mechanism.

B- Transmission Time:

One of the critical issues of ultrasound images transmission over UMTS network is analysing the transmission time for each image separately considering the compression ratio applied. Figure 4.28 presents the total time taken to transmit ten ultrasound images compressed at different CRs:

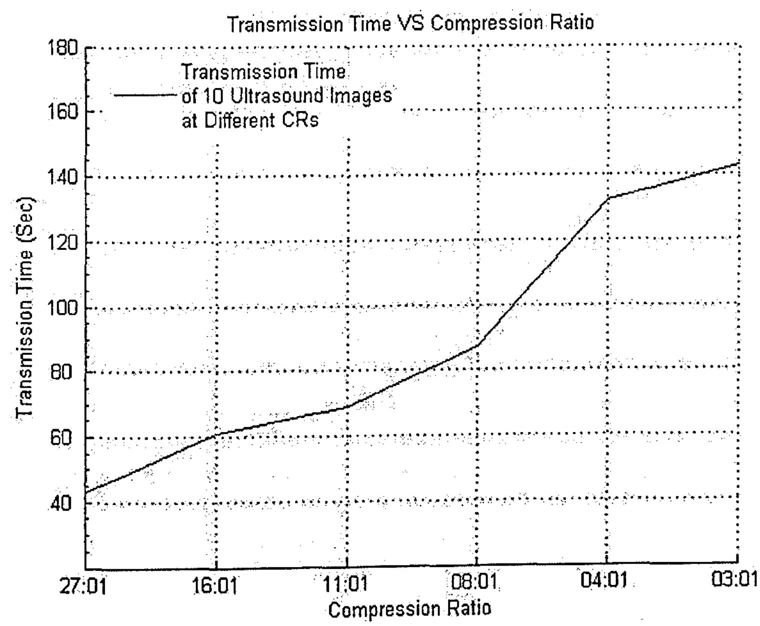


Figure 4.28 3G transmission time verses Compression Ratios US images.

The transmission time spanning between 42 sec for compression rate at 27:1 to 142 sec for compression rate at 03:1. This gave an average image transmission time of 4.2 sec and 14.2 sec in each case respectively. Images with low compression of 08:1 and below (towards 03:1), shows sharp increase in transmission time. Image size of 59 Kbytes (compressed at 08:1) took 7.4 sec to be transmitted to expert station and image size of 153 Kbytes (compressed at 03:1) took 22.17 sec to be transmitted to the Expert station. The total transmission time is also including around 2.3 sec for new TCP session establishment before each image transmission.

Many important parameters have to be considered when experimenting over real network; First its difficult to predict on the availability of a stable data rates and delay variability, which translates to burst of ACKs arrivals (also called ACK compression) at the TCP source (Patient station). Since TCP uses ACK clocking to probe for more bandwidth, bursty ACK arrival leads to release of a burst of packets form the source [88]. When this burst arrives at a link with variable rate or delay, it could result in multiple packet losses, these losses significantly degrade TCPs throughput; second, repeating the test does not necessarily end with exactly same results mainly due to the facts mentioned above.

3- RTT:

Another important metric of performance analysis is the RTT measurements. The TCP sender estimates the RTT of a connection by measuring the time between the sending of a packet and the receipt of the ACK for this packet. For each congestion window, the TCP sender can get a sample of the RTT. The new RTT and its deviation are calculated below [139]:

$$RTT = \alpha * RTT + (1-\alpha) * M \qquad (4.6)$$

$$D = \alpha * D + (1-\alpha) * |RTT - M| \qquad (4.7)$$

Where D is the mean deviation of RTT; *M is the sample of the RTT;* α is the control factor. The optimal value of α (as suggested by RFC of IETF) equals to 0.875 in order to have small oscillation between two RTTs.

In TCP protocol, the RTO value of the sender is calculated as a function of the average and mean deviation of the RTT, where:

$$RTO = RTT + 4 * D \qquad (4.8)$$

Most TCP implementations use a 500 ms timer granularity for the retransmission timeout. In heterogeneous networks, the local retransmission at the wireless link can cause the end-to-end RTT to increase significantly in a short time due to the long processing time and the low data transmission rate on the wireless link. As a result, big time gaps in the receiving ACK sequence are observed at the sender when local retransmission occurs [139].

The group of images compressed at 16:1 are selected to show the behaviour of RTT samples that present the time taken by the Patient station to receive acknowledgment from the Expert station for the TCP packets transmitted, as shown in figure 4.29:

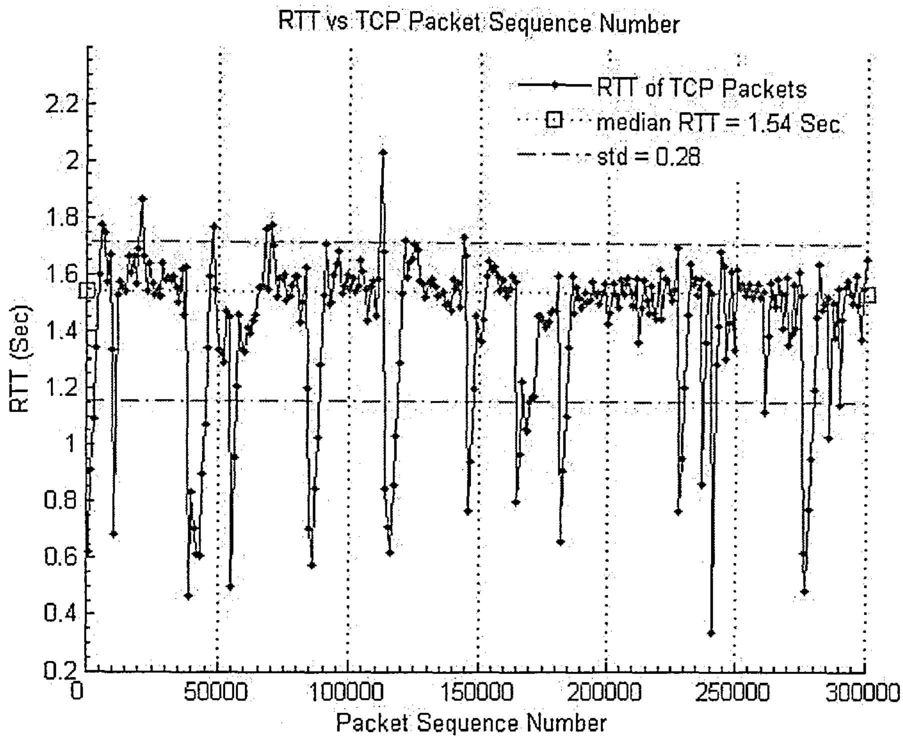


Figure 4.29 The RTT values versus sequence number for transmission of Images compressed at 16:1, each value represent the time for the Patient station TCP packet to be ACK by the Expert station.

New RTT samples increment, presents new image transmission until it reaches the average RTT values, which span between 1.4 and 1.65 sec as shown in the figure above. For the performance of the ARQ mechanism, the RTT on the MAC-layer is an important system characteristic [93].

The RTT measurements between both system ends was including the RTTs of the wireless side of the path for both Uplink and Downlink, which are related to the service time of the IP packets within the wireless part of the link. Because of the different Transmission Time Intervals (TTI) Up- and Downlink, two different RTTs can be obtained as following [93]:

One for the path (Patient station-RNC-Patient station), where;

$$RTT_{Patient\text{-}RNC\text{-}Patient} = \text{UL delay} + \text{wait next TTI} + \text{DL delay} \quad (4.9)$$

And one for the path (RNC-Patient station-RNC), where;

$$RTT_{RNC\text{-}Patient\ station\text{-}RNC} = \text{DL delay} + \text{wait next TTI} + \text{UL delay} \quad (4.10)$$

Therefore for specific part of the RTT path (Patient station air interface only) based on the literature above, RTT measurements can be illustrated in the table below, since the Uplink TTI is twice as large as in downlink direction and based on the following measured parameters:

Table 4.8 Patient station-to-RNC corresponding RTT measurements.

RTT Path	RTT measurement (ms)	UL Delay (ms)	TTI (ms)	DL (ms)
$RTT_{Patient\text{-}RNC\text{-}Patient}$	155	68	2	45
$RTT_{RNC\text{-}Patient\ station\text{-}RNC}$	118 (128)	68	5 (15)	45

The average RTT delay of the transmitted packet can be driven as the average delay measured of the maximum group of the IP packets shown by the sequence number of packets for differently compressed group of images. Figure 4.30 shows the average RTTs at different CRs:

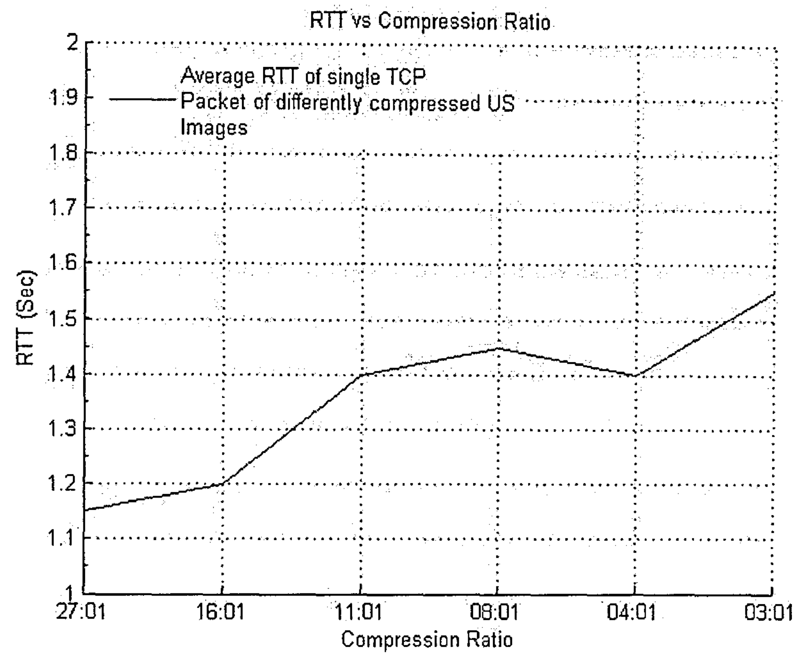


Figure 4.30 Average RTT measurements verses Compression Ratio for group of Ultrasound images transmitted to the Expert station.

The measurements in the curve above represents the RTT values of the maximum number of the ultrasound images packets at a specific compression ratio. These measurements are including the IP packet service time for the Patient station Up-link and Downlink channels.

4.4.2 3G Ultrasound Streaming Results

During the tele-echography examination, ultrasound streams transmitted from the Patient site to the Expert site conveying to the expert the relevant information to move the fictive probe and to control the distant probe-holder robotic system. Furthermore the received information enables the expert to analyze different anatomical regions or to search for a region of interest. The ultrasound scanner produces data stream at a rate of 13 fps and resolutions of (320x240) pixels with videoconferencing format of QCIF pixels have been used. The Robotic data flow bursts from the Expert station at 16 Bytes (payload) every 70 ms, and the received robot control data stream from the Patient station is continuously updating the Expert side with the new Robotic head (Ultrasound Probe) position.

By referring to figure 4.26, the service activation from the Patient station viewpoint can be described as follows; first the Patient side initiates the conversational application, which connects the patient side to the UMTS network by using a socket Application Program Interface (API) [89]. Furthermore PPP protocol is used to maintain the connection between the Patient station and the Expert station during the life of the connection. Once the video stream activated by the application program, the RTP flow will start running through the streaming PDP context. The Patient station sends the ultrasound stream in form of RTP flow likewise, RTCP traffic is exchanged for the QoS control of the corresponding RTP data flow [89].

Since TCP retransmission introduces delays that are not acceptable for real time streaming applications with stringent delay requirements especially for transmission over fading wireless links, the UDP is typically employed as the transport protocol for video streams over such fading channels [91]. Therefore TCP protocol is not acceptable for OTELO real-time performance; moreover since UDP does not guarantee packet delivery, the receiver needs to rely on upper layer (i.e., RTP) to detect packet loss [98] as shown in figure 4.16. RTP is an Internet standard protocol designed to provide end-to-end transport functions for supporting real time applications. RTCP is a companion control protocol with RTP and is designed to provide QoS feedback to the participants of an RTP session [89].

Therefore for OTELO streaming, the RTP/UDP/IP protocol is implemented for the ultrasound stream to investigate the feasibility of the 3G network for real-time interaction between the Expert station probe movement and the reception of the ultrasound stream from the Patient station. UDP/IP protocol is used for the Robot data that is transmitted in both directions. We have to keep in mind that additional bandwidth consumed by the headers of the protocol used, RTP, UDP and IP [99].

A- Ultrasound Stream Codec:

The video compression standard used in our application is H263 Codec, which has a wide range of applications including medical consultation and diagnosis at a distance [119]. H.263 is aimed particularly at video coding for low bit rates (typically 20-30 Kbps and above). A major drawback of this scheme is the additional delay introduced by transcoding which, in fact requires decoding and encoding [92].

B- Uplink Throughput Variation:

When testing on real network, an important issue has to be considered, that is the number of user occupying the same cell where the patient station is located. In order to guarantee the desired quality of service, it is necessary to guarantee a given signal-to-noise ratio [92]. The ratio which is generally used for this purpose is *Eb/Io,* defined as the ratio of energy per information bit to interference spectral density (assuming that thermal noise is negligible). Let N be the number of users, and C the effective power value of the reference communication, then this can be expressed:

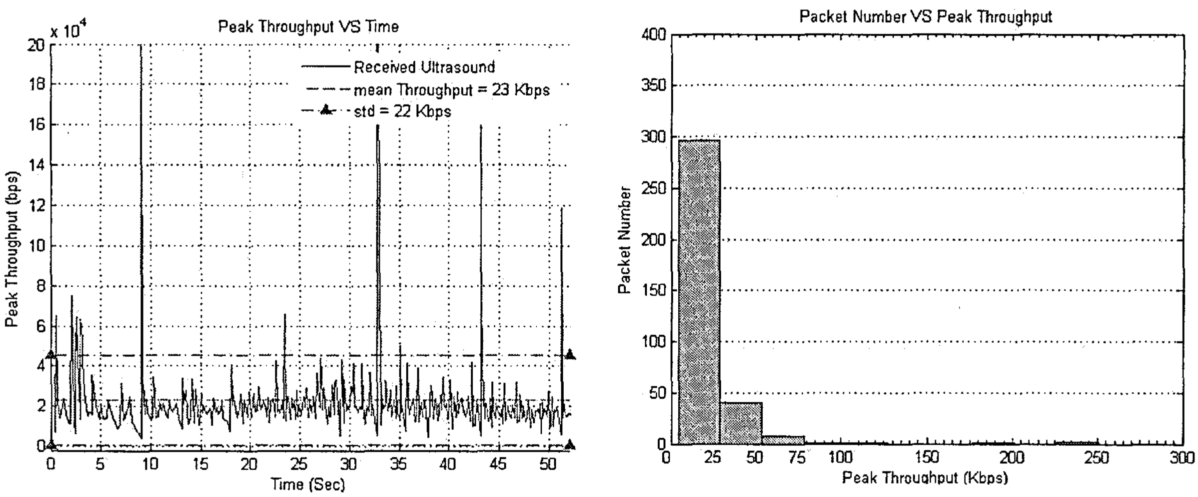
$$\frac{Eb}{Io}=\frac{C/R}{I/W}=\frac{W}{R}\frac{C}{I}=\frac{W}{R}\frac{1}{N-1} \qquad (4.11)$$

Where (W) is the signal chip rate and (R) is the information source bit rate. The (W/R) is the processing gain, the number of users who can be served in the cell is then:

$$N \cong \frac{W}{R}\frac{1}{\frac{Eb}{Io}} \qquad (4.12)$$

As can be seen, the number of users (N) is influenced by two factors: the processing gain and the *Eb/Io* ratio. In particular any technique that makes it possible to lower the *Eb/Io* ratio automatically entails a gain in terms of capacity.

It is assumed in some literatures, that the mean number of video users in the same cell ranges from 15 to 30, the considered QoS parameters for voice and video transmission are a maximum delay of 400 ms and a packet loss of 1% [82]. Therefore multiple test has been performed at different timing, where noticed that there is no dramatic change in the peak throughput then the average throughput from time to time, providing that the signal strength is satisfactory and the data rates provided by the operator, is reliable. Figure 4.31 shows the instantaneous throughput of the ultrasound stream captured by the Expert station and the relevant packet distribution:



(a) (b)

Figure 4.31 Peak throughput of the transmitted ultrasound stream captured by the Expert station, (a) Peak Throughput (b) Received packet throughput distribution.

The throughput performance above is for the transmitted ultrasound stream by the Patient station, and we have to bear in mind that the number of users within the Patient station Node-B is unknown. The mean throughput is = 23 Kbps and (std) = 22 Kbps, in fact this is the Patient station uplink average throughput performance influenced by the complete path including the 3G streaming bearer, this bearer is solely dedicated to RTP traffic and provides a guaranteed QoS for streaming applications in terms of bit rate and delay jitter [86]. Apart from the source signal changes, the stream fluctuation shown in figure 4.31 is normally reveals the ultrasound head movement and consequently the H263 codec frames produced.

The number of stream packets taken for the analysis is 350 stream packets; approximately 82% of the packets has been received at a rate of 18.5 Kbps at the Expert station, and the rest of the packets shows data rates up to a maximum of 60 kbps, hence the average throughput performance of our system for this specific test can be approximated to 23 Kbps for the calculations of end-to-end system delay and delay jitter performance.

C- RTP Packet Loss:

Packet loss does occur over the 3G link (wireless link) in both the downlink and uplink directions more often due to the harsh environment of the wireless environment compared with the wired part of the link, but for moderate to high signal to noise power ratio (SNR), the incidence of packet loss is relatively rare, and hard to quantify [73]. In 3G wireless communication systems, we can deduce a packet loss caused by wireless errors based on the information provided in the radio link control layer (RLC).

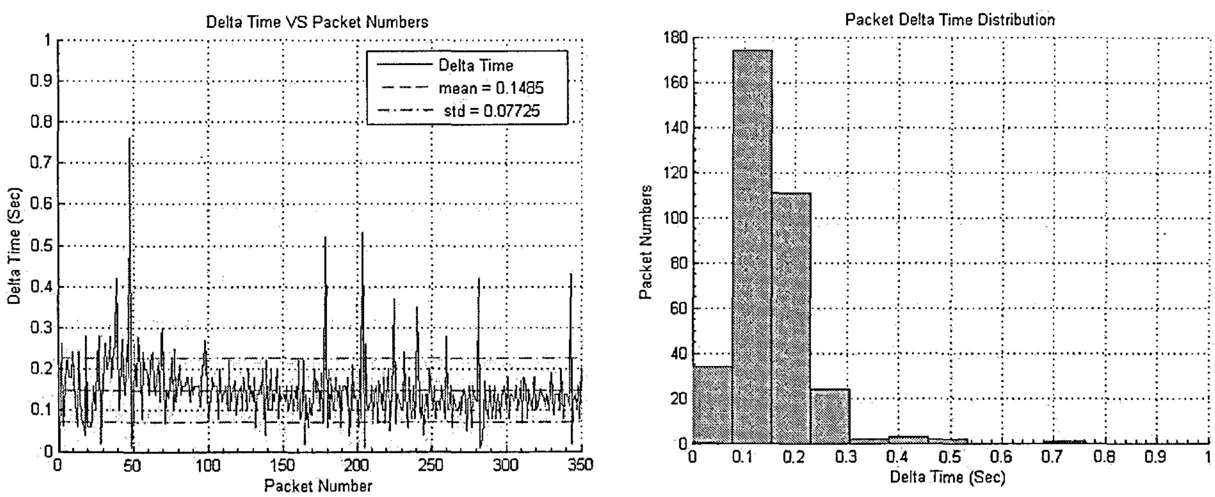
The RLC layer adopts a selective ARQ scheme. It segments an IP packet into several RLC frames before transmission and reassembles them into an IP packet on the receiver side. An IP packet loss occurs when any RLC frame belonging to the IP packet fails to be delivered, when this happens the receiver knows the RLC frames reassembly fails and the IP packet is lost due to wireless error. Meanwhile, the sender knows that the retransmission time of the frame reaches the maximum so it discards all the RLC frames belonging to the IP packet [73].

Nevertheless out of 360 RTP packets captured by the Expert side; total lost was 0 packets, whereas in a different test, out of 1192 RTP packets captured, only 2 packets (0.17%) were lost. Retransmission schemes are usually inappropriate to recover lost packets in real-time interaction since a retransmitted packet may miss its play-out time [98].

4.4.3 End-to-End Delay Performance

The delay introduced by the air interface is one of the elements which define the QoS requirements to be supported by 3G bearer services [83]. The factors causing the delay across the wireless link are; the delay access on each ON period *(variable)*, the delay of the multiplex process *(variable)* and the transmission delay. The connection between both system ends is always on (during the entire clinical session).

Fig. 4.32 shows the Delta time of the ultrasound packets received at the Expert side, which are influenced by the factors above (wireless link delay behavior):



(a) (b)

Figure 4.32 Delta time of the packet delay at the Expert station, (a) Delay Time. (b) Delay distribution of the captured packets.

Streaming video requires bounded end-to-end delay so that packets arrive at the receiver in a timely fashion to be decoded and displayed. If a video packet does not arrive on time, the play out process will pause, which is annoying to human eyes [72]. An average packet delta time delay and relatively low standard deviation enhance the feasibility of real-time system performance. Relevant performance analysis parameters shown in figure 4.32 part (a) and (b) can be summarized and illustrated in the table below:

Table 4.9 Delay measurements of the packet received by Expert station.

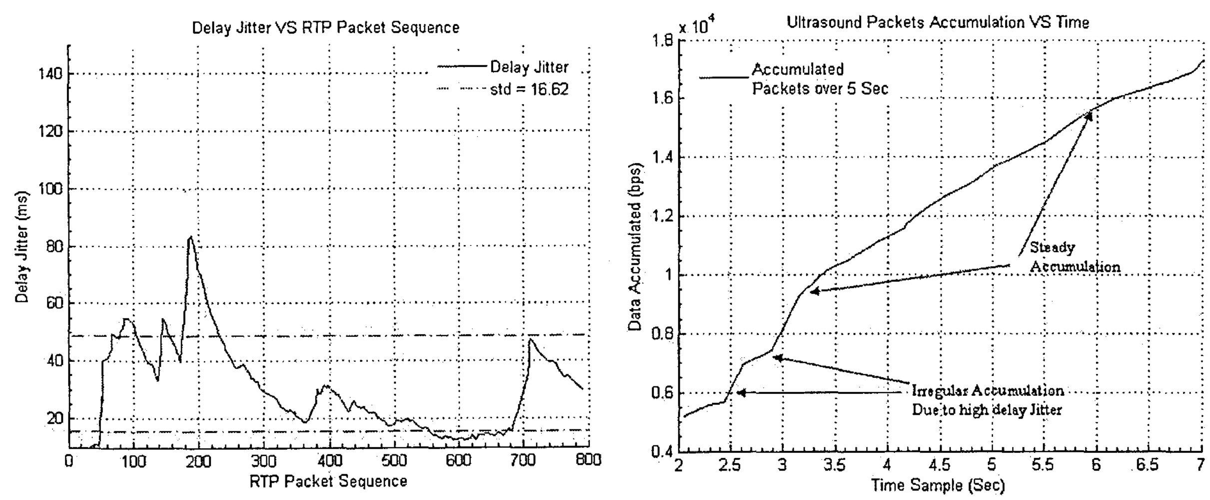
Mean Delta Time (Sec)	Standard Deviation (Sec)	Packet delta time distribution			
0.148	0.077	Maximum (Sec)	0.3	Average (Sec)	0.15
		Packet percentage (%)	8.3	Packet percentage (%)	58

A video packet that arrives beyond its delay bound (e.g. its play out time) is useless and can be regarded as lost [98]. Since the Internet introduces time-varying delay, to provide continuous play out, a buffer at the receiver is usually introduced before decoding.

A- Delay Jitter:

Although the latency is the key metric for real-time applications such as voice-over-IP (VoIP) and media streaming [100], latency is also required to have as low as possible statistical variation in the delay i.e. delay jitter. Therefore the jitter across our system link is the key factor to have a reliable real-time ultrasound stream reception at the Expert station.

Figure 4.33 (a) presents the delay jitter of the packets at the Expert station supported by the packet accumulation in (b), where the influence of the delay jitter can be seen.



(a) (b)

Figure 4.33 Delay jitter of the packets received by the Expert station, part (a). Zooming on ultrasound packets accumulation over 5 second period, part (b).

In the figure above, part (b) shows 5 second reception period, the worse delay jitter influence on the packet accumulation trace is within the 2.0-3.0 sec interval. The delay jitter encountered on the packets received at the Expert side degrades the reliability of the ultrasound images seen at the Expert station in terms of irregular packets accumulation, therefore producing variable frames per second. Generally dramatic fluctuation on 3G network bandwidth has not been noticed, except occasional coverage weaknesses.

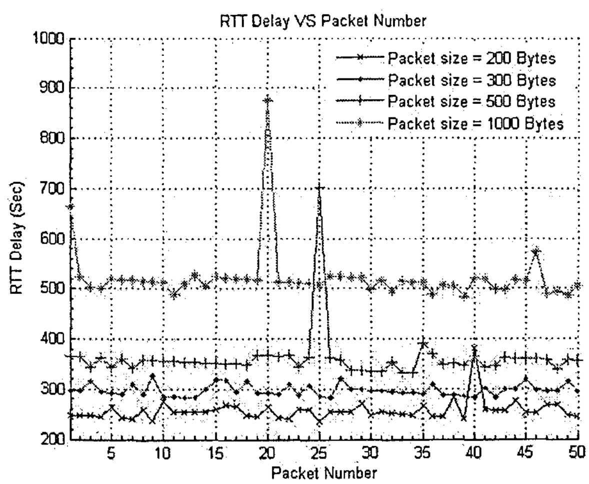
B- RTT Delay:

The analysis in this section focused on the Uplink transmission of the Patient station since it is the dominant direction for OTELO system, although the Downlink transmission to the Patient station is included in the analysis of latency from the robot control viewpoint considering the control data that is sent by the Expert station. A common way to benchmark the network latency is to evaluate its RTT [101]. Generally the network RTT can be defined as the time it takes to transmit one packet from say a server to a terminal plus the time it takes for the corresponding packet to be sent back from the terminal to the server [101]. The definition of RTT delay regarding OTELO system is as follow:

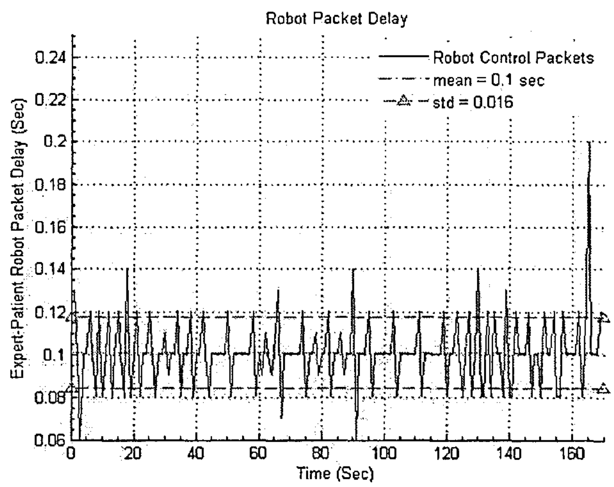
Expert side generates and sent robot control data (Expert-Patient link Delay) +
Patient station response by transmitting Ultrasound stream of images (Patient-Expert link Delay).

Different packet sizes 200, 300, 500 and 1000 Bytes are used to characterize various performance degradation effects and these sizes are chosen to cover the possible size range of the packets generated by the H263 decoder. The characterization of the minimum network latency can be estimated by pinging 32 Bytes from the Patient to Expert station, where it is also used to estimate the performance degradation associated with the transport and application layers establishment protocols [101].

ICMP protocol has been used to ping the Expert station with the packet sizes given above, at 500 ms interval time from the Patient station. ICMP provides some error detection mechanisms and it can be used to send error messages or other messages for network diagnosis [53]. The RTT delay test is performed over Vodafone's 3G network, with the Patient station side data rates of 256 Kbps for Downlink and up to 64 Kbps for Uplink channel. Figure 4.34 part (a), shows the round trip time delay fluctuating between around 250 ms to 525 ms for transmitted packets sizes ranging 200 to 1000 Bytes as seen respectively:



(a)



(b)

Figure 4.34 End-to-end delay system performance, (a) RTT delay of pinging 200, 300, 500 & 1000 Bytes, Patient-to-Expert, (b) end-to-end delay of robot control data, Expert-to-Patient.

Part (b) of the figure above, presents the robotic control packets (robot payload packet size = 16 Bytes) transmitted by the Expert station to the Patient side, its performance is consistence and shows a mean delay of 0.1 sec and very low standard deviation, which can be neglected.

We can conclude that the latencies caused by only the wireless part of the complete link as the highest percentage (around 80% to 90%) of the total delay.

Depending on the network conditions, the tests have been repeated several times in order to approach as close as possible to the RTT delay performance of the network during reliable connection, i.e. avoiding the cases of high packet loss and congestion. When the packet loss measurements are analysed, it is clear that significant numbers of packets were lost when the throughput dropped to zero [102], which is very rear occasion with the real current 3G networks due to the implementation of ARQ mechanism.

4.4.4 Evaluation of RTT (Robotic-Ultrasound Interaction)

We primarily intended to derive a round trip response to the Expert hand movement as the person controlling the Robotic head, in the mean time, is receiving continuous ultrasound streams in real-time or very near real-time to investigate the annoying time lag recognition by the Expert hand movement. Figure 4.35 shows the complete round trip path delay over Expert-Patient-Expert links:

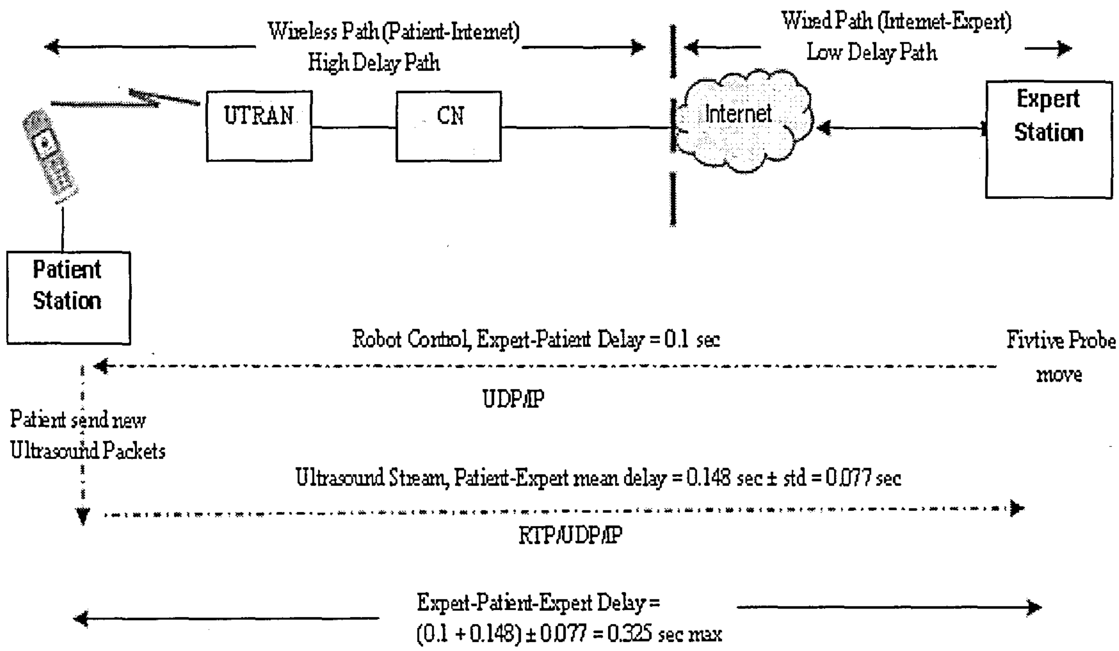


Figure 4.35 RTT Latency measurements Test Bed for the Ultrasonography system over 3G Link.

The end-to-end analysis shown in the figure above, is based on the average transmission delta time found using figure 4.32 (a), which is closely correspond to RTT/2 of pinging 300 Bytes from Patient to Expert, since the average ultrasound transmitted packet size is around 300 Bytes.

Although our analysis of several experiments shows that the wireless link encountered the highest delay (nearly 80% of the delay time is across the wireless path), nevertheless it has been noticed that most of the packet loss events happened on the wired part of the complete link.

We have found that network delay jitter variation was still within the boundaries of maintaining real-time interaction. The maximum delay of Probe movement-to-Received image (Expert-Patient-Expert) path is around 325 ms, in these specific ultrasound encoding conditions.

4.5 Comparative Results and Analysis

This section presents selected comparative performance results for the real network and simulated 3G environments. The case of 64 Kbps channel bandwidth was selected, since the operating data rate of the Patient station Up-link on the real UMTS network, was up to 64 Kbps. The performance metrics of Throughput, Transmission time, end-to-end delay and RTT have been chosen for this comparative analysis.

4.5.1 Ultrasound Still Images

Comparative average throughput of ultrasound images transmission (group of 10 images each time) is illustrated in table 4.10. This shows the average throughput behaviour on simulated network (figure 4.11) at different BER, with respect to the results on the UMTS network (figure 4.27):

Table 4.10 US still images average throughput on simulated and real UMTS network.

Compression Ratio (CR)	Average Throughput at different BERs (Simulated) @ 64 Kbps Uplink BW (Kbps)				Average Throughput at Patient Station Uplink (Real 3G Network) (Kbps)
	BER				
	2e-006	2e-007	2e-008	2e-009	
27:1	26.30	29.84	28.09	28.12	33.45
16:1	36.08	35.76	33.52	33.09	38.58
11:1	40.04	35.49	41.84	40.79	47.31
08:1	42.24	44.92	46.92	45.58	53.43
04:1	48.46	54.36	52.82	50.00	57.64
03:1	48.74	52.05	53.41	54.85	53.17

In general the real time performance shows relatively higher throughput at each compression level, this sustain our simulating parameters scheduling for worst-case performance assumption. The best channel efficiency was at CR = 03:1, where in average simulated throughput gave about 81.6%, whilst the real time one shows 83%. The corresponding comparative analysis of the images transmission time is shown in table 4.11, that summarises the simulated results of figure 4.12 and the real time results of figure 4.28, at different BERs:

Table 4.11 US still images transmission time on simulated and real UMTS network.

Compression Ratio (CR)	Transmission Time at different BERs (Simulated) @ 64 Kbps Uplink BW (Sec)				Transmission Time, Patient-to-Expert Station (Real Network) (Sec)
	BER				
	2e-006	2e-007	2e-008	2e-009	
27:1	55	48	51	51	43
16:1	73	66	70	71	61
11:1	83	92	78	80	69
08:1	112	104	99	102	87
04:1	180	155	159	168	132
03:1	213	195	189	184	143

At lower CRs, the real network performance shows remarkable difference in the transmission time needed. This was due to relatively high latencies assignment for smaller TCP packets < 1460 Bytes on the simulated environment. In the real time this could be contributed to the WCDMA packet access, which allows non-real time bearers to use common, dedicated or shared channels dynamically [66].

The round trip time for the TCP packets are summarised in the table below, that represents the RTTs measurements on simulated environment shown in figure 4.14 and the RTTs on real time performance shown in figure 4.30:

Table 4.12 RTTs of different US still images CRs on simulated and real UMTS network.

Compression Ratio (CR)	Round Trip Time of TCP Packets at different BERs (Simulated) @ 64 Kbps Uplink BW (Sec)				Round Trip Time of TCP Packets at Patient Station (Real Network) (Sec)
	BER				
	2e-006	2e-007	2e-008	2e-009	
27:1	1.35	1.4	1.4	1.5	1.15
16:1	1.35	1.4	1.4	1.35	1.2
11:1	1.3	1.45	1.45	1.45	1.4
08:1	1.4	1.35	1.45	1.45	1.45
04:1	1.2	1.4	1.45	1.4	1.4
03:1	1.2	1.45	1.4	1.4	1.55

Since the overall RTT measurements either in simulated environment or on real time test were corresponding mostly to the behaviour of the biggest TCP packet size segmented (around 1460 Bytes), therefore we found that the performance over both environments were very close.

4.5.2 Ultrasound Streaming

This section presents a comparative peak throughput for the ultrasound stream seen at the Expert station. 350 ultrasound stream packets have been taken to represent the results on figures 4.17 and figure 4.31 respectively as shown in the figure below:

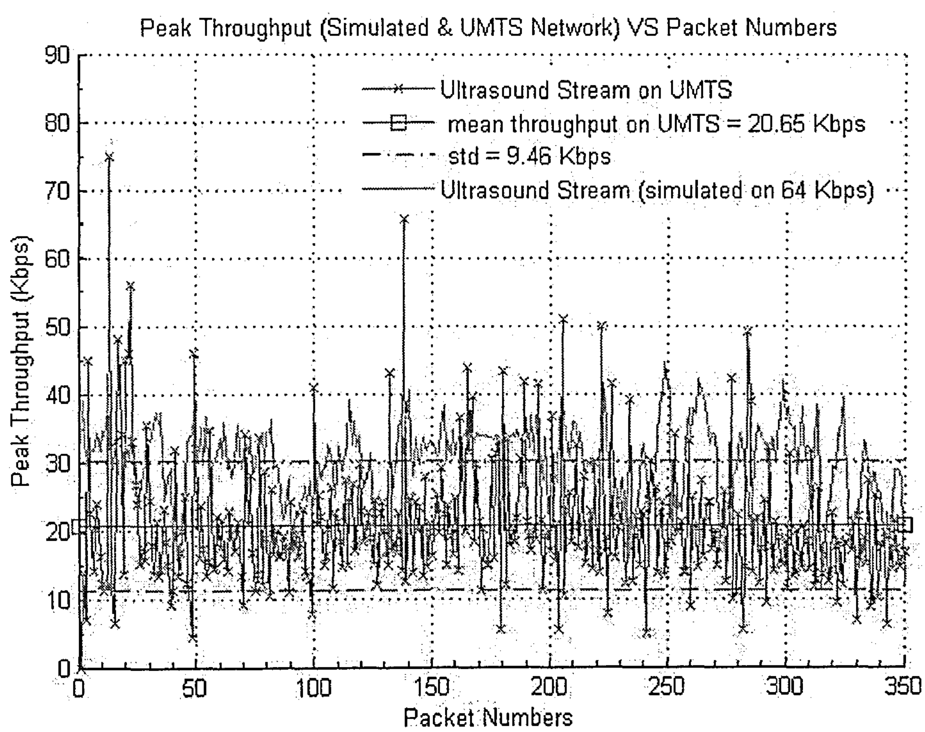


Figure 4.36 Comparative peak throughputs on 64 Kbps, for real and simulated cases.

Both plots shows high standard deviation with respect to the mean throughput of 20.65 Kbps and 29 Kbps for UMTS network and simulated respectively. The physical layer of the UMTS air interface introduces jitter [103], this is increasing the variation in throughput (std = 9.5 Kbps), therefore reducing the mean throughput on the real UMTS test comparing to the simulated results, where the std = 6.4 Kbps. In the mean time the active number of users using the cell is affecting the throughput, where ultimately the resources available and the QoS required would reduce the bit rates. The main number of video users in the same cell ranges from 15 to 30. The considered QoS parameters for voice and video transmission are a maximum delay of 400 ms and a packet loss of 3% and 1% respectively [104].

The ultrasound stream packet indicates higher variation at the Expert station affected by the end-to-end path and the encoding mechanism at the source (Patient station). The figure below shows the comparative packet delta time performance on the simulated and the real-time UMTS networks:

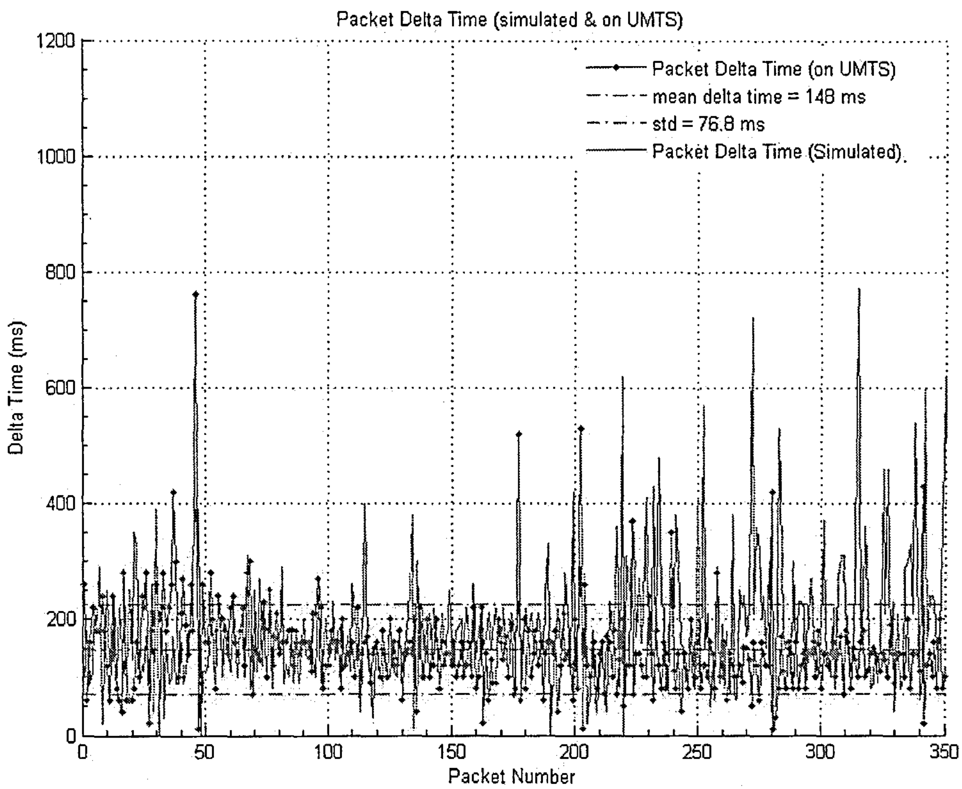


Figure 4.37 Comparative received packet delta time on 64 Kbps for real and simulated cases.

Although in general the delta time shows parallel results for both simulated and real time test. Nevertheless the samples of the real time were more reliable in terms of mean delta time over longer packet number. This could be contributed to packet sizes produced by the encoder at the time of the test as well the packet delta time distribution with respect to its distribution when testing on the simulated network, as summarised in the table below:

Table 4.13 Distribution of the packet delta time received by Expert station, on simulated and on real UMTS Network.

Distribution of the Packet Delta Time received by Expert station		Delta Time on Simulated Network (ms)	Delta Time on UMTS Network (ms)
Percentage of Packet Number	Average	67 % shows 190 ms	82 % shows 145 ms
	Max	11.4 % shows 300 ms	6.3 % shows 270 ms

The Patient station ultrasound decoder is producing wide range of US packet sizes, these sizes have different round trip time (Patient-Expert-Patient) that also affect the packet delta time illustrated in table 4.13.

Summarised comparative results on the RTTs for different packet sizes in terms of mean RTT delay and standard deviation is shown in the table below as well as the robotic control data, which has constant payload size of 16 Bytes:

Table 4.14 Comparative RTT results of different US stream packet size and robotic packet on real UMTS Network.

OTELO System Packets RTTs	Ultrasound image Packet Size (Bytes)						Robot Control Data, (payload = 16 B)
	100 B	200B	300B	500B	1000B	1400B	
Mean RTT Delay (ms)	206	257	297	360	521	638	100
Std (ms)	41	20.77	11.5	50	57	35.6	16

The values above comprising the RTT of the path 'Patient station Up-link-to-Expert station + the Expert station Uplink-to-Patient station'. It is well known that in real-time services, the packet data tolerate less delay than non real-time services in WCDMA radio access [66].

4.6 Summary

The work in this chapter presented detailed performance analysis results of OTELO medical data end-to-end transmission over simulated and real UMTS network. The results of the ultrasound still images transmission were analysed and discussed in a comparative manner over both simulated and real network.

Feasibility of real-time ultrasound streaming performance has been evaluated and concluded that the UMTS network can form a good platform for a robotically controlled tele-ultrasonography performance. For that purpose, a wide range of performance modalities has been dealt with, that formed the testing matrix. Furthermore reliable real-time robotic control has been achieved including ultrasound stream interaction.

CHAPTER 5

Quality Image Analysis of OTELO System

5.1 Introduction

This chapter introduces the objective performance analysis of OTELO system's medical imaging and US video data. The following sections present the relevant compression techniques with the lossy and losseless modes used in these studies. Furthermore the analysis of the Region of Interest (ROI) is also introduced in this chapter.

5.2 OTELO System Compression Issues

When compressing data it is important to take into account the type of data and how it is interpreted. For example pixels in an image may be distorted but it would still contain the required information. Whereas, a single erroneous bit in a computer data file could cause severe problems. Video and sound images are normally compressed with a lossy compression whereas computer- type data has a lossless compression. The basic definitions are [105]:

- Lossless compression – where the information, once uncompressed, will be identical to the original uncompressed data. This will obviously be the case with computer-type data, such as data files, computer programs, and so on. Any loss of data will cause the file to be corrupted.
- Lossy compression – where the information, once uncompressed, cannot be fully recovered. Lossy compression normally involves analyzing the data and determining which information has little effect on the resulting compressed data.

For example, there is little difference, to the eye, between an image with 16.7 million colors (24-bit color information) and an image stored with 1024 colors (10-bit color information), but the storage will be reduced to 41.67%. Compression of an image might also be used to reduce the resolution of the image [105].

5.2.1 Introduction of JPEG & JPEG2000

JPEG is a superior compression technique that produces lossy compression (although in one mode it is lossless). A typical standard for image compression has been devised by the Joint Photographic Expert Group (JPEG), a subcommittee of the ISO/IEC, and the standards produced can be summarized as follows "It is a compression technique for gray-scale or color images and uses a combination of discrete cosine transform, quantization, run-length and Huffman coding". JPEG has three main modes of compression [140, 75]:

1. Progressive mode – this mode allows the viewing of a rough outline of an image while decoding the rest of the file. It is useful when an image is being received over a relatively slow transfer channel (such as over a modem or from the Internet). There are two main methods used: spectral-selection mode and successive-approximation mode. The successive- approximation mode first sends the high-order bits of each of the encoded values and then the lower-order lower bits. Spectral-selection mode first sends the low-frequency components of each of the 8´8 blocks then sends the high-frequency terms.

2. Hierarchical mode – in this mode the image is stored in increasing resolution. For example a 1280´960 image might be stored as 160´120, 320´240, 640´480 and 1280´960. The viewing program can then show the image in increasing resolution as it reads (or receives) the file. Most systems do not implement this facility.

3. Lossless mode – in this mode the data is allowed to be stored and recovered in exactly in its original state. It does not use Discrete Cosine Transform (DCT) conversion or sub-sampling.

The JPEG standards have made some impact on the coding of moving images [75]. JPEG2000 was developed to provide a more efficient successor to the original JPEG. It uses the Discrete Wavelet Transform (DWT) as its basic coding method and hence has similarities to the still Texture coding tools of MPEG-4 visual. The (DWT) performs a sub-band decompression [106], these are:

a) Improve compression performance at low bit rates.

b) Increase suitability for progressive transmission.

c) Meets the requirements for transmission in noisy environments (e.g. robustness to bit error).

d) Offers protective image security.

e) Codes region of interest.

Previous works have indicated that at low data rates JPEG2000 provides about 20% better compression efficiency for the same image quality than JPEG [107]. JPEG2000 also offers a new set of functionalities. These include error resilience, arbitrarily shaped region of interest, random access, losseless and lossy coding as well as a fully scalable bit stream.

Despite the fact that JPEG is considered an earlier technology and is now outperformed by JPEG2000 and many proprietary compression formats, it is still very widely used for storage of images in digital cameras, PCs and on web pages [75].

1- Quality Index:

Peak Signal to Noise Ratio (PSNR) is a well know quality measure in image processing studies. It is widely used to compare the 'quality' of compressed and decompressed video images. The PSNR suffers from number of limitations. PSNR requires an unimpaired original image for compression but this may not be available in every case and it may not be easy to verify that the original image has perfect fidelity [75].

PSNR is measured on a logarithmic scale and depend on the Mean Squared Error (MSE) between an original and an impaired images or video frame, relative to $(2^n - 1)^2$ (the square of the highest possible signal value in the image, where n is the number of bits per image sample) [74]:

$$PSNR_{dB} = 10\log_{10}\frac{(2^n - 1)^2}{MSE} \qquad (5.1)$$

MSE is commonly converted to PSNR in the video coding studies. PSNR can also be defined as $10\log_{10}(255^2 / MSE)$, where 255 corresponds to the peak-to-peak range of the encoded and decoded video signal (each quantised to 256 levels). The logarithmic scale provides a better correlation with subjective quality.

As a rule of thumb for low bit rate video coding (with clearly visible distortions), a difference of 1 dB generally corresponds to a noticeable difference, while acceptable picture quality requires values greater than 30dB. PSNR does not correlate well with video quality measures. For a given image or image sequence, high PSNR usually indicates high quality and low PSNR usually indicate low quality [75]. Currently, the PSNR and MSE are still employed "universally" regardless of their performance quality [108].

Various measures have been used by literatures to evaluate compression efficiency in video streaming and imaging researches [109]. In this work the following measures have been used: compression ratio (CR) and relative compression (RC) [110]. These are defined as:

$$CR = \frac{t}{L} \qquad (5.2)$$

$$RC = \frac{(t - L)}{t} x100\% \qquad (5.3)$$

Where t is the original file size and L is the compressed file size. Image compression methods usually involve a tradeoff between compression efficiency and the fidelity of uncompressed images. For any lossy compression algorithm, a measure is required to evaluate the image quality and fidelity after the decompression step.

5.2.2 Ultrasound Still Images Compression

There are several approaches to ultrasound image compression including both losseless and lossy compression methods. It is well known that Losseles compression is limited to 03:1 and 07:1 for Gray-Scale and color ultrasound images, respectively. JPEG2000 provide superior compression performance to JPEG and does not exhibit the characteristic blocking artifacts of a DCT-based compression method [107]. JPEG2000 is developed to provide better coding efficiency for both lossless and lossy compression [106].

The OTELO system was implementing 4CIF (768x576x8 bit) ultrasound still image format, for transmission to the Expert station. These images have to be compressed prior to transmission over the low bit rate (mobile network).

JasPer is the software-based implementation of the image Codec specified in the emerging JPEG2000 standard [106, 111].

This software has been used to compress the above original image size (4CIF of jp2 extension) to a multiple resolutions, and then measuring the quality of the new compressed image with respect to the original, by using the PSNR quality index. The table below shows different compression rates and the resulting quality measures:

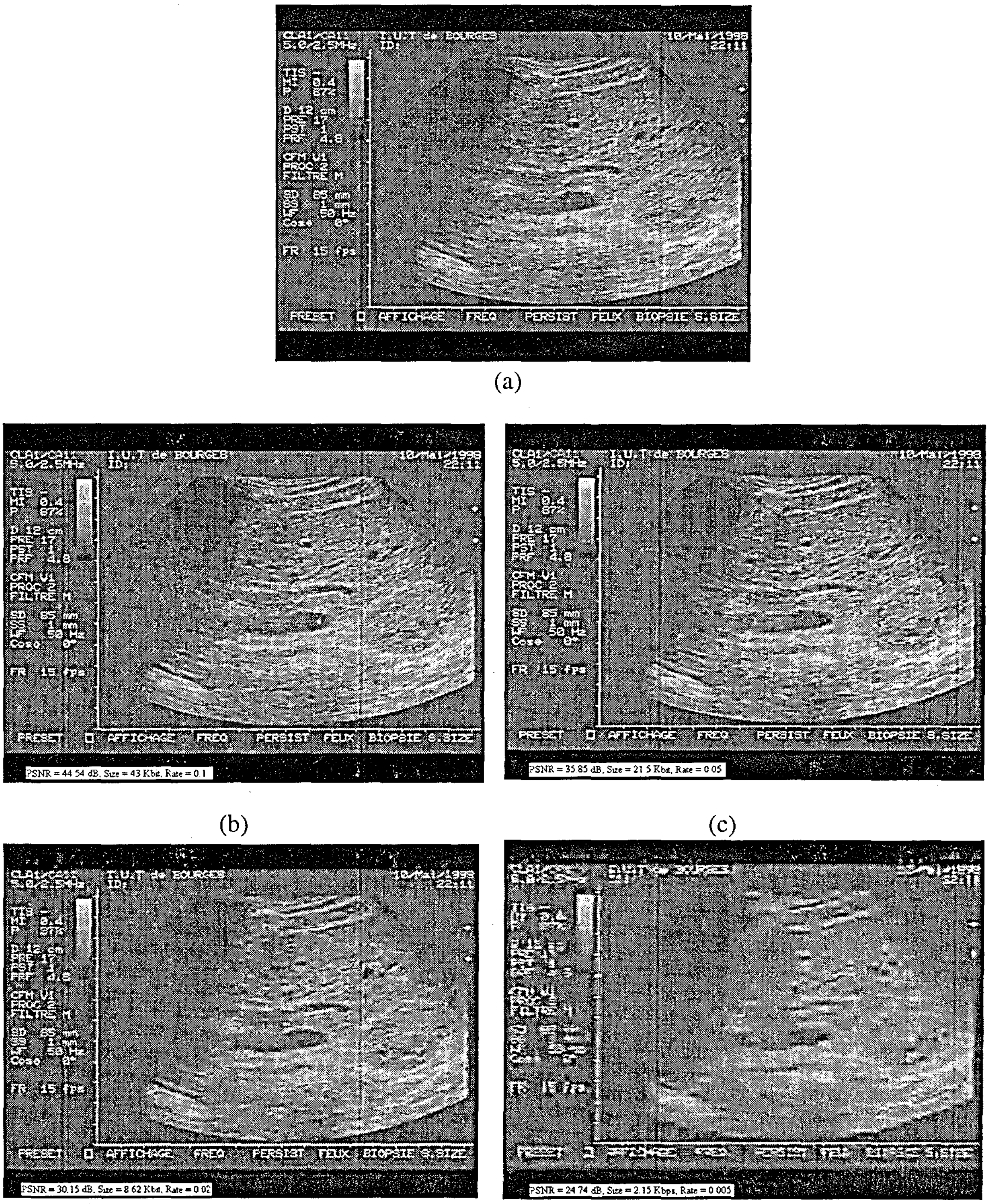
Table 5.1 Image Quality as a function of Compression Rates.

Compression Rate = 1 – 0.005	Ultrasound Image Original JPEG2000 image size = 135 Kbits. Resolution (768x576) pixels.	Quality (Distortion) PSNR
bpp	Kbits	dB
1	135	93.12
0.75	135	93.12
0.5	135	93.12
0.4	135	93.12
0.3	129	60.11
0.2	86.2	48.08
0.1	43	44.54
0.08	34.5	39.56
0.05	21.5	35.85
0.03	12.9	32.45
0.02	8.62	30.15
0.01	4.28	26.64
0.005	2.15	24.74

Lossy compression using JPEG at 20:1 was found to be diagnostically acceptable by comparing the quality of digitally compressed images to images stored and retrieved from video tape [112]. Quality evaluation of the ultrasound images transmitted by OTELO system, suggested that the minimum threshold for objectively accepted medical US image quality is PSNR = 35 dB and above [113, 114].

Hence in the current test and experimental set-up, the threshold of the compression ratio used was not exceeding 15:1, this is supported by literatures recommendation of up to that ratio for acceptable resulting quality [115].

However the qualitative evaluation of the compressed images is not necessarily proportional to the objective evaluation measures. Figure 5.1 shows four descending quality ultrasound images compared with the original US image (size 135 Kbits, PSNR = 93.12 dB), as described in table 5.1, these images were acquired by OTELO system:



(a)

(b) (c)

(d) (e)

Figure 5.1 Samples for ultrasound Image for abdomen, acquired by OTELO system. (a) Original image, size = 135 Kbits, resolution (768x576x8), uncompressed, (b) compression ratio 3.1:1, image size = 43 Kbits, PSNR = 44.54, (c) compression ratio 6.3:1, image size = 21.5 Kbits, PSNR = 35.85, (d) compression ratio 15.6:1, image size = 8.62 Kbits, PSNR = 30.15, (e) compression ratio 62.8:1, image size = 2.15 Kbits, PSNR = 24.74.

Although the quality measurements between (b) and (c) shows a 8.7 dB difference, nevertheless no visual artifacts and deficiency was concluded from the visual medical inspection. It is clear that the images in (d) and (e) are not acceptable from the objective view point and more likely the same from the medical Doctors. Further experiments shows that at low compression ratio, the differences between the JPEG2000 compressed images and the original images are negligible [106].

5.2.3 Ultrasound Stream; Lossy Compression Techniques

Ultrasound stream of images in fact is a video stream with exceptionally high resolution and quality requirements for medical diagnosis purposes. Ultrasound and generally video coding tries to achieve compression by eliminating redundancy in the stream data. There are two types of redundancy present in the stream data; the first type is spatial, while the second type is temporal. Spatial redundancy refers to the correlation present between different parts of a frame. Removal of spatial redundancy thus involves looking within a frame and is hence referred to as *intra coding*.

Temporal redundancy, on the other hand, is the redundancy present between the frames. At a sufficiently high frame rate it is quite likely that successive frames in the video sequence are very similar. Hence, removal of temporal redundancy involves looking between frames and is called *inter coding*. Spatial redundancy is removed through the use of transform coding techniques. Temporal redundancy is removed through the use of motion estimation and compensation techniques [75]. Compression involves removal of spatial and temporal redundancy.

One of the techniques that eliminating redundancy in the stream data is; Motion JPEG (M-JPEG), a non-standard method of compressing a sequence of video frames using JPEG, and it is popular for applications such as video capture, PC-based video editing and security surveillance [75]. This technique is depending on using JPEG to compress each frame as independent still image, under heavy packet loss, error propagation may render the use of predictive coding inefficient, and a solution based on *intra-only* coding may be preferable [75].

Because JPEG frames are typically larger than the underlying network's path Maximum Transmission Unit, frames must often be fragmented into several packets. One approach is to allow the IP layer to perform the fragmentation [116]. Fragmentation increases the effects of packet losses while imposing extra work on the network elements. If the size of a packet can be changed based on the MTU while staying consistent with Application Layer Framing (ALF), fragmentation can be avoided. To reduce the packet header and packet processing overheads, it is desirable to have packet sizes as close to the MTU a possible. However, particularly for low bit rate applications, the acceptable delay may limit the largest packet size to a value much smaller than the MTU. In such cases, header compression may be used to reduce the overhead. In applications that use more than one medium, multiplexing these may provide another solution [75].

ITU study group 16 has adopted feedback-based error control in their effort toward mobile extensions of the successful recommendation H.263 for low bit rate video coding. The first version of H.263 already included *error tracking,* a technique that allows the encoder to accurately estimate inter-frame error propagation and adapt its encoding strategy to mitigate the effects of past transmission errors [117, 118].

A- H.263 Codec:

The H263 standard uses the (DCT) to remove spatial redundancy and motion estimation and compensation to remove temporal redundancy. Motion compensation involves removing the temporal redundancy present in video sequences. There is a great amount of similarity between neighboring frames.

It is more efficient to code the difference between frames, rather than the frames themselves [75, 107]. H.263 is a new improved standard for low bit-rate video (wide range of low bit rates, 32 Kbps up to 128 Kbps), adopted in March 1996. H.263 includes various optimal coding modes that can be used for error resilience in many combinations. On the other hand the operation encoder is not standardised, such that the present syntax can be used in a very flexible way. The clever combination of existing options that may not even be intended for error resilience can significantly increase the performance of a wireless video system [119, 75].

This technique is very convenient to be used for OTELO ultrasound stream image transmission for the specifications mentioned above. H.263 uses the transform coding for intra-frames and predictive coding for inter-frames, and it has the following advance options:

1. Half-pixel precision in motion compensation.
2. Unrestricted motion vectors
3. Syntax-based arithmetic coding
4. Advanced prediction and PB-frames

In addition to CIF and QCIF, H. 263 could also support SQCIF, 4CIF, and 16CIF. The table below summaries the video formats supported by H.263:

Table 5.2 Images formats supported by H.263 Codec.

Video format	Luminance Image Resolution (pixels)	Chrominance Image Resolution (pixels)	H.263 support	Bit-rate (Mbit/s) (if uncompressed, 30 fps)	
				B / W	Color
SQCIF	128 x 96	64 x 48	Required	3.0	4.4
QCIF	176 x 144	88 x 72	Required	6.1	9.1
CIF	352 x 288	176 x 144	Optional	24.3	36.5
4CIF	704 x 576	352 x 288	Optional	97.3	146.0
16CIF	1408 x 1152	704 x 576	Optional	389.3	583.9

B- Input Format and Rate Control:

To achieve high compression, as required for transmission over mobile radio channels at low bit rates, both the spatial resolution and the frame rate must be reduced compared to standard television pictures [75]. In our implementation the highlighted formats in the table above were used as follow:

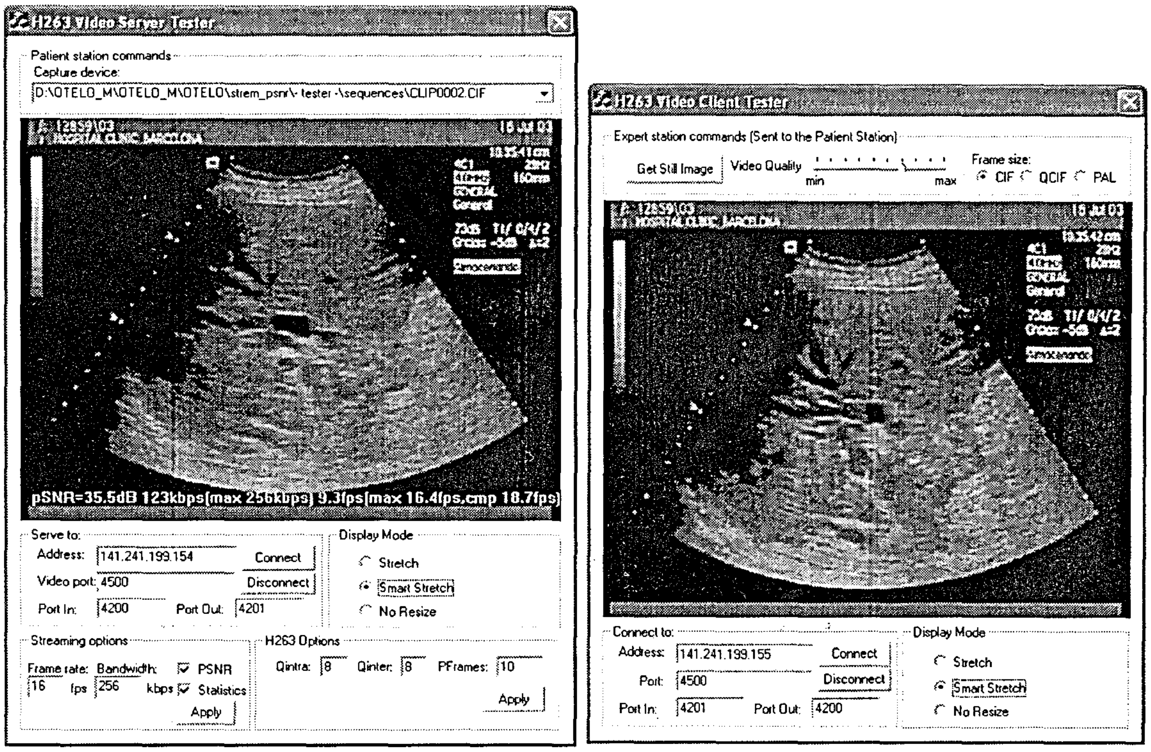
1. Gray scale CIF, which is the most common input format over the range of bit rates considered, as well as its acceptability by OTELO medical criteria. For minimum acceptable frame rate of 5 fps for medical quality evaluation. The CIF format encoded at 5 or 10 fps, at rates of 53 Kbps, 64 kbps, 128 Kbps and 256 Kbps as required by the GPRS and UMTS network data rates.

2. Gray Scale QCIF, which is used as another option under the criteria of the ROI, encoded at 10 fps, at a rates of 53 Kbps, 64 Kbps, 128 Kbps and 256 Kbps.

In this test scheme a fixed encoding rate has been used, to study the effect of the OTELO medical ultrasound stream compression on the performance of the system. The criteria of the minimum objective measures and frame rate (fps) are considered.

5.3 OTELO Ultrasound Images Evaluation over 2.5G and 3G Networks

The aim of this section is to evaluate objectively the performance of OTELO system; first over real-time GPRS network and also to validate the same performance over simulated 3G network. Expert / Patient evaluation programme has been developed by OTELO research group as seen in figure 5.2, is used for evaluation purposes.



(a) (b)

Figure 5.2 Image capture Expert/Patient software used H.263 ultrasound Codec, (a) Patient side, (b) Expert side [113].

The Patient side application transmits the ultrasound stream and computes the relevant statistics of PSNR, data rates in Kbps and the frame rate (fps) for the transmitted data. For evaluation purposes, the transmitted ultrasound sequence can be either pre-stored or acquired by hardwiring the ultrasound machine signal to the Patient PC data input; these are described earlier in details in chapters three and four.

5.3.1 Quality Matrices and Limitations over 2.5G Mobile Network

The quality of received ultrasound over mobile channels may be improved by matching the channel characteristics to those of the compressed ultrasound syntax [120]. The bit rate reduction was essential to enable the ultrasound streams to transport over the GPRS network and channel conditions. The amount of user data for the transport over GPRS is strictly limited depending on the selected channel-protection scheme [60]. However the feature of time slots multiplication in GPRS, may improve for some extent the ultrasound streaming over GPRS.

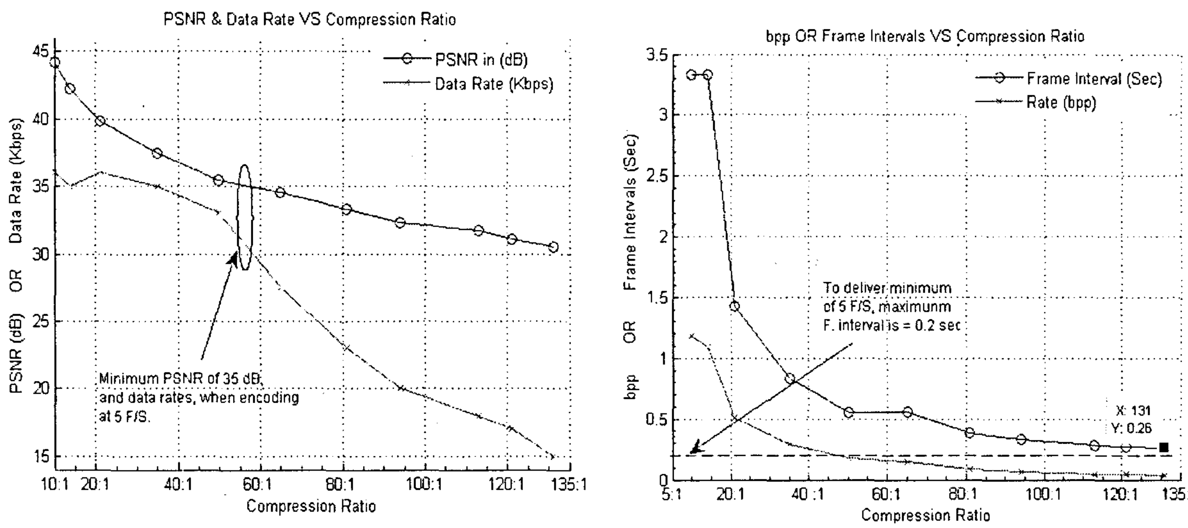
The end-to-end system was using three timeslots (3 TS) assigned to the Patient station uplink, which was based on CS-2 as the operating GPRS coding scheme of the service provider (VF). Hence the available data rate for uplink is equal to 13.4 Kbps x 3 = 40.2 Kbps. The experimental set-up and GPRS connectivity were explained in chapter three. Table 5.3 summarise the GPRS network specifications for this test:

Table 5.3 2.5G Network experimental set-up and test parameters.

Ultrasound Stream Encoding		H.263 Compression Options		US Image Size	GPRS Uplink channel BW (kbps)	Stream Length (Sec)
Frame Rate (fps)	Bit rates (Kbps)	Intra	Inter	CIF (352x288x8) Gray Scale	3 TS = 40.2 Kbps	4 seconds, 16 fps, (Repeatable)
5	53	Yes	Yes			

US Medical Streaming Results:

As discussed earlier, the minimum requirements for real-time ultrasound operation in terms of the minimum frame rate and objective quality should be > 5 fps and > 35 dB respectively. The testing of the OTELO system ultrasound stream transmission has been performed several times to reach realistic results. Figure 5.3 shows the quality measure (PSNR), the average data rate, the resulting bpp and the arrival Frame Intervals as functions of the compression ratios:

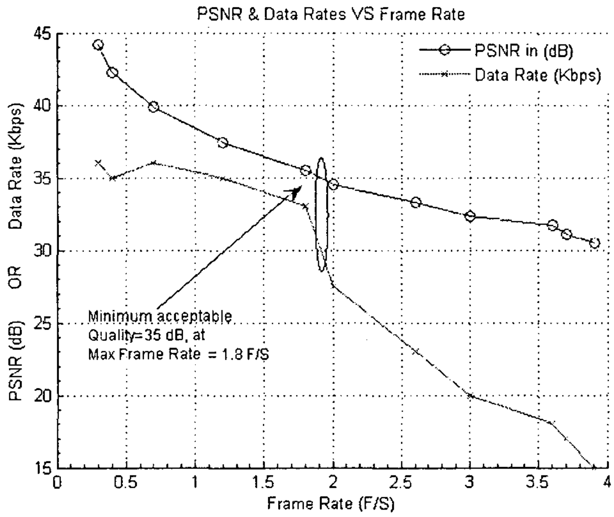


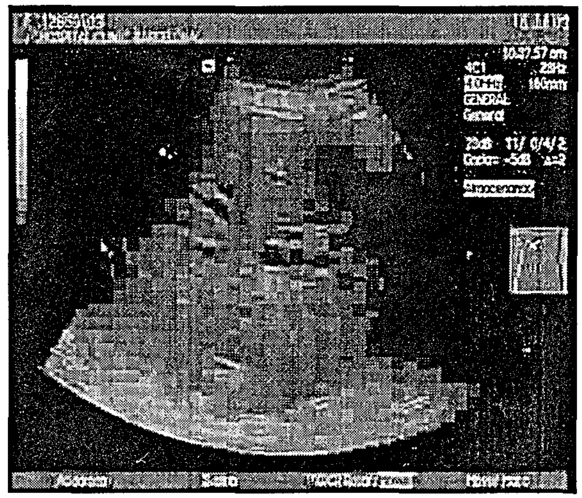
(a) (b)

Figure 5.3 Evaluation measures of the transmitted Ultrasound stream as a function of the CRs. (a) PSNR and Data Rate curves, (b) Bit per pixels and the Frame intervals curves. At Expert station side.

Figure 5.3 shows that the PSNR value varies above and below the threshold (35dB) within the range of (31 – 44 dB), whilst the channel usage efficiency is above 77% (31 Kbps / 40.2 Kbps) at the threshold quality value, achieved by encoding at CR = 55:1 as in (a). This represent good measure providing that minimum frame interval at the Expert station maintained at maximum of 200 ms (Frame Interval = 1/5 fps), in order to reliably decode 5 fps at the Expert Screen. As expected an increase in timeslot allocation allows for an improvement in the average quality of the received ultrasound [120]. Nevertheless the major drawback of the OTELO wireless test is the frame interval increases to 500 msec at 55:1 as shown in part (b) of the figure above, although the image resolution was at good rate of around 0.18 bpp at the specified CR. With this specific test parameters, compression ratios to as high as CR = 130:1, did not improve the frame interval to be below the maximum acceptable.

From the above analysis, we conclude that improving the frame rate on such a limited wireless link capacity (GPRS) will result in degradation in the clinical quality of the images transmitted. This has been verified by a corporate clinician, and this is already shown in figure 5.4 (b):





(a) (b)

Figure 5.4 In (a) Quality and Data Rate, as a function of the frame rate, (b) CIF (352X288) format Ultrasound sequence, at average data rate (throughput) = 16.5 kbps, PSNR = 29.2 dB, Frame Rate = 5 fps.

The PSNR performance of 35 dB and the corresponding data rate of 30 Kbps, produces an average frame rate of 1.8 fps as shown in figure 5.4 (a). Such a low frame rates inevitably can not be satisfactory for even pre-diagnosis medical requirements, where the appropriate target rate should be above 5 fps. Now if trading of between the quality measure and the frame rate could increase the frame rate received by the Expert to 5fps as shown in figure 5.4 (b), however in the mean time this will degrade the objective quality to PSNR = 29.2 dB, as seen in figure 5.4 (b).

Despite the multi-slotting feature, GPRS rates are still far to low for ultrasound real-time communications if frame droppings are not employed [60], therefore another important fact can be approached in this analysis which is the frame delivery ratio (DR), which is defined as the percentage of the frames received at the receiver to the frame generated at the sender [77]:

$$DeliveryRatio = \frac{F_{\text{Received}}}{F_{Generated}} * 100\% \quad (5.4)$$

The objective quality measures, frame rates and the delivery ratio as a function of the compression ratio as discussed above, summarised in the table below:

Table 5.4 Ultrasound stream quality measures with respect to the CR.

CR	PSNR (dB)	Frame Rate (F/S)	DR % ($F_{generated}$ = 5 fps)
10:1	44.2	0.3	6
14:1	42.2	0.3	6
21:1	39.9	0.7	14
35:1	37.4	1.2	24
50:1	35.5	1.8	36
65:1	34.5	1.8	36
81:1	33.3	2.6	52
94:1	32.3	3	60
113:1	31.7	3.6	72
121:1	31.1	3.7	74
131:1	30.5	3.9	78

It can be seen from table 5.4 that the frame delivery ratio drops at the acceptable quality measures. In the experiment above, we conclude that although with multiple time slot allocation, the frame rate of OTELO system is limited by the GPRS network frame rate and not the Codec used. It is known in literature that if the network is the limiting factor, therefore the codec that provides the best possible quality at the network-limited frame rate is recommended [112].

5.3.2 Qualitative Evaluation of 3G US Image Transmission

It is well known that OTELO system ultrasound transmission is based on real-time ultrasound data capturing and transmission to the Expert station, therefore the following sections will present background on the live US images capturing, and the quality evaluation results.

A- Ultrasound Image Captures:

Assume that the encoding and decoding delays are constant and thus the two delay components that can result in violation of decoder buffer underflow constraint, are the sender (Patient station) buffer delay and the transmission channel delay, defined as follow [75]:

Patient station buffer delay:
This represent the delay due to the time needed to drain live video data corresponding to a particular frame, after it has been placed in the transmitter buffer by the encoder, this delay exist only if the channel bandwidth is limited and does not match the data produced for all frames, and let this delay be δ_{th}. It is known that the source buffer should not introduce delay so large that it eats into delay budget of the network; this would make the network less attractive for real-time services. It is assumed that there is a sufficiently large play-out buffer at the Expert side to overcome delay jitter. Hence the primary concern for the work is the aggregate delay introduced in the source buffer and the network. In literatures an overall (one way) delay budget around 200-300 ms is acceptable [75].

Transmission channel delay:
This is defining the delay suffered by packets of ultrasound data being transmitted through the network. This delay may be variable in numerous scenarios, such as transmission over a shared network or transmission over a lossy link (here delay is assumed to include the time needed for retransmission of lost data if applicable), and let this delay be δ_{ch}. Therefore assume that the first frame experiences delays of δ^1_{th} and δ^1_{ch} after the time the frame has been captured and compressed. Normally δ^1_{th} will tend to be very small or zero, since there are no other frames waiting to be transmitted. A delay may exist, however because time is needed to set up the transmission or because the encoder decides to store a few frames in the encoder buffer before transmission actually starts. Thus the first frame will experience, from capture to display, an overall end-to-end delay ΔT, which can be written as follow [75]:

$$\Delta T = \delta^1_{th} + \delta^1_{ch} + \Delta T_d \tag{5.5}$$

Where ΔT_d is the time the decoder at the Expert station waits before starting the decoding process, measured from the time the first bits of ultrasound data were received. That is the time between the capture and encoding of frame i and its decoding and display will be ΔT for any i. It is important to note that even though all the delay components will vary over time, so that frame i will experience delays δ^i_{th} and δ^i_{ch}, i.e. each frame could experience different delay [75].

Assuming that ultrasound sequence with total of M frames is transmitted at a fixed number of frames per second (5 or 10 fps). Let Ri be the number of bits assigned to the ith frame (this formulation could be easily adapted to having a variable number of frames per second). Let $td = 0$ be the time at which the first bit of the ultrasound sequence is received by the decoder. The overall delay ΔT must remain constant. This is illustrated in figure 5.5 below:

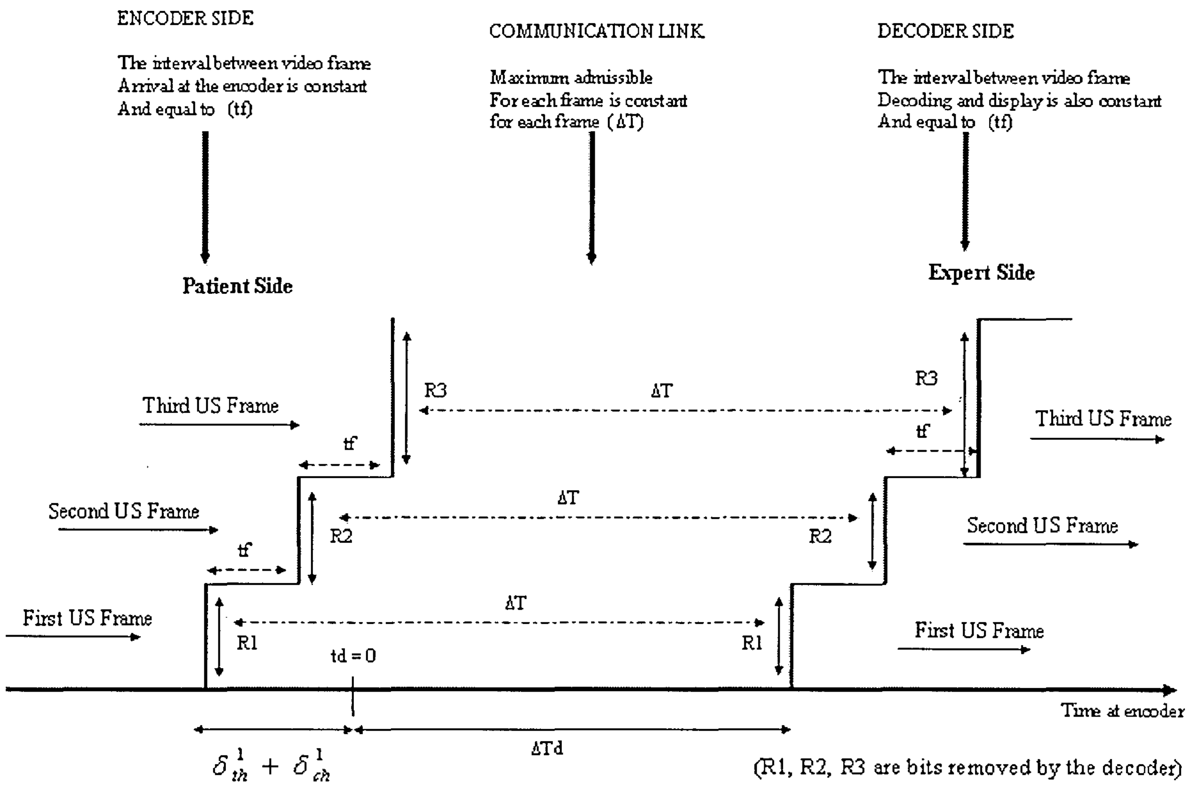


Figure 5.5 Delay constrain in the live ultrasound capturing and transmission.

Time is measured in units of number of frames at the decoder. Thus td = I corresponds to i frame intervals having passed, where one frame interval lasts δ_f seconds. For example considering our minimal frame rate of 5 fps are being displayed at the Expert side, then $\delta_f = 1/5$. Therefore in this real-time encoding case, frame arrival at the Expert side, begins decoding and displaying frames after ΔN_d frame intervals or:

$$\Delta T_d = \Delta N_d * \delta_f \qquad (5.6)$$

Clearly, given that ΔT determines the maximum delay a frame experiences, having a large ΔT chosen will tend to reduce the chance of violating the delay constraint. In this case each frame spends exactly ΔT seconds in the system, and there are always exactly $\Delta N = \Delta T / \delta_f$ frames in the system [121].

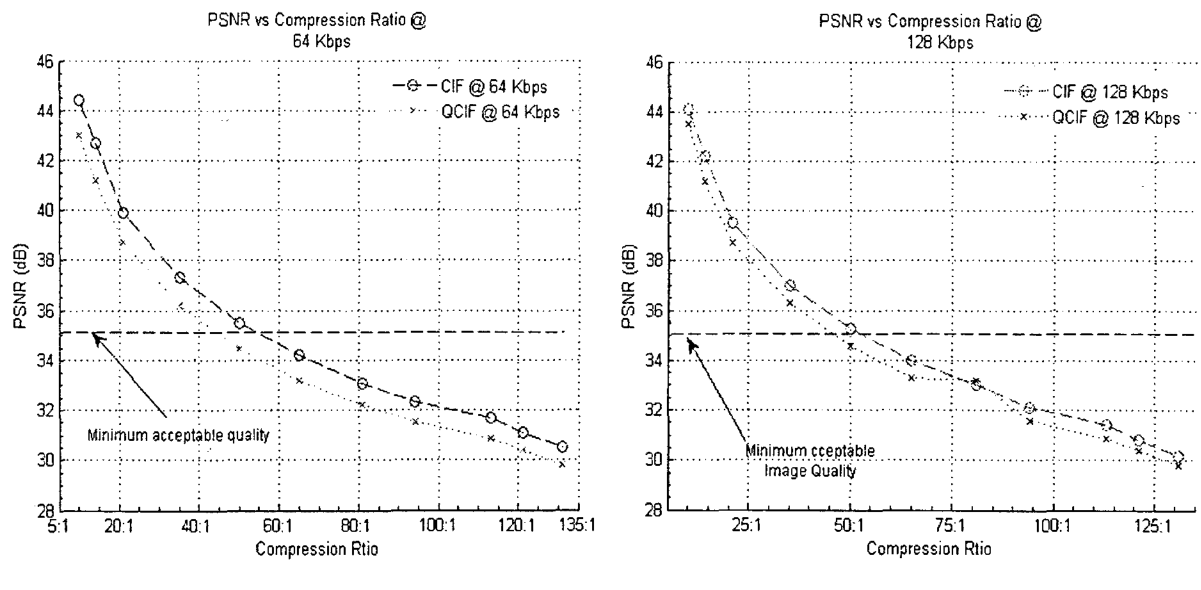
B- 3G Experimental Set-up and Results:

The simulation was carried out using the Cloud Emulator, illustrated in chapters three and four. Table 5.5 summarises the 3G experimental set-up parameters of the test:

Table 5.5 Experimental set-up parameters.

Ultrasound Stream Encoding (Patient Station)		H.263 Compression Coding Options (Patient station)		Format (Gray scale) Stream Length = 4 seconds, 16 fps, Repeatable		Emulator Channel Settings			
F. Rate (fps)	Data Rate (Kbps)	Intra	Inter			Patient side BW (Kbps)	Delay (ms)	Delay Jitter (ms)	P. Loss %
10	64	Yes	Yes	CIF	QCIF	64 UP + 384 Down	180	10	2 %
10	128	Yes	Yes	CIF	QCIF	128 UP + 384 Down	100	10	1 %

The ultrasound images used in the experiment show close performance for both formats over both channel data rates as in figure 5.6. However the CIF format outperformed the QCIF format by an average of 1 dB irrelevant to the channel data rate assigned:



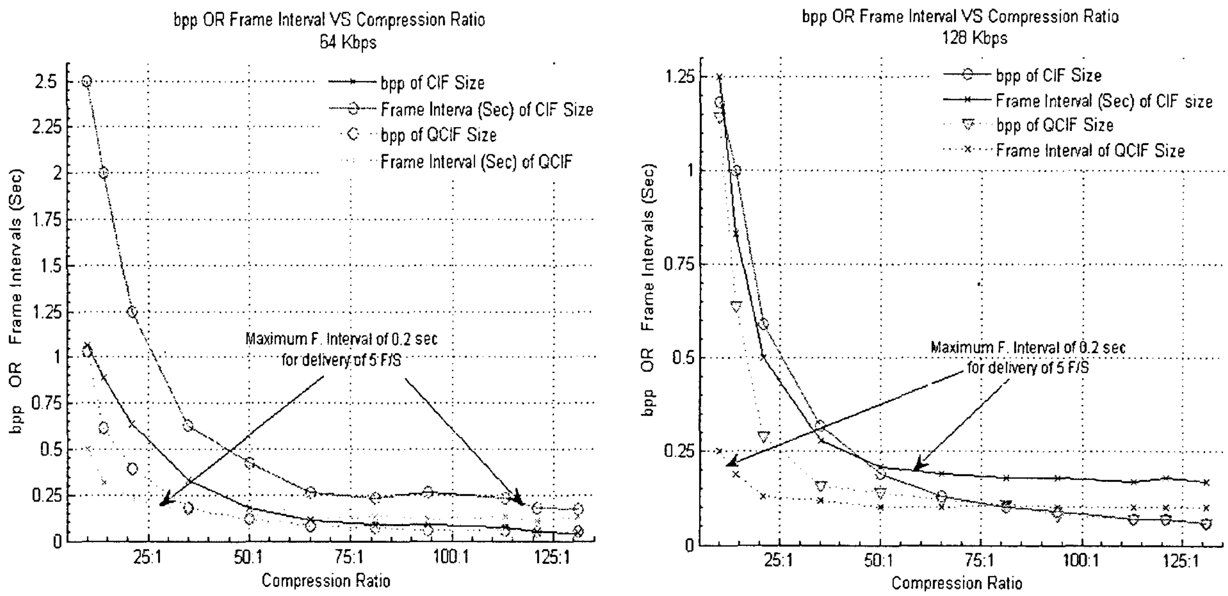
(a) (b)

Figure 5.6 Evaluation measures of the transmitted Ultrasound stream as a function of the CRs, for CIF & QCIF formats, (a) Quality curves over 64 Kbps, (b) Quality curves over 128 Kbps.

It is known that the problem of transmitting ultrasound stream over noisy wireless channels, involves both source and channel coding. The classic goal of source coding is to achieve the lowest possible distortion for a given target bit rate [75]. Figure 5.6 shows that the lowest possible distortion or the minimum acceptable quality achieved was at CR = 50:1. This is based on implementing similar encoded frame rates (10 fps) over two different channel modules with 64

Kbps and 128 Kbps bandwidth respectively. This in turn degrades the target data rates in terms of degrading the delivery ratio and bit per pixel, especially the case with 64 Kbps channel bandwidth.

For OTELO system, an appropriate target bit rate had to be determined prior to any clinical validation. Experimentally, this target was set-up to 0.18 bpp. This rate resulted in good coded ultrasound quality, above the minimum (35 dB). The figure below plots an important relationship of the resulting bpp and frame interval as a function of the CRs, where the higher channel data rate obviously improves the performance by approaching the functional boundaries earlier than with 64 Kbps as the compression went above 50:1, especially with QCIF format:



(a) (b)

Figure 5.7 Bits per pixels and the Frame Intervals of the received QCIF & CIF Image formats, (a) over 64 Kbps, (b) over 128 Kbps.

To address the possibility of performing over 64 Kbps channel rate, for CIF format transmission, figure 5.7 (a) shows clearly high frame interval time (0.4 sec) although the relative target bpp of 0.18 is achieved in this case, at the corresponding CR. On other hand QCIF format shows satisfactory result just before approaching CR of 50:1. table 5.6 below summarised a comparative result of figure 5.7 above:

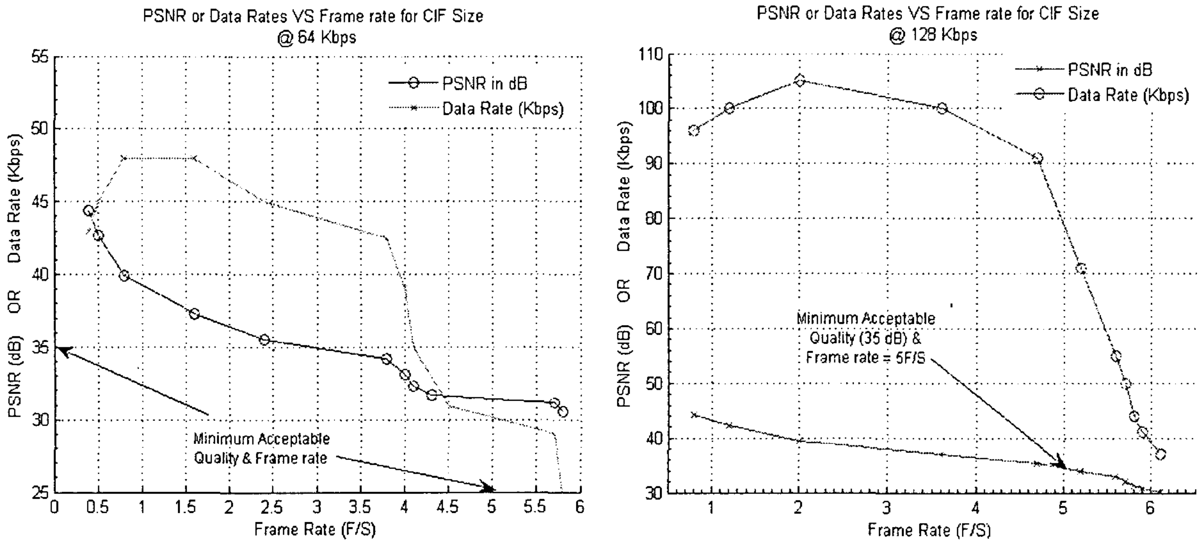
Table 5.6 derived performance results of figure 5.7.

Format	Channel Data Rate (Kbps)	CR	bpp	F. Interval (Sec)
CIF	64	50:1	0.18	0.45
QCIF	64	50:1	0.12	0.18
CIF	128	50:1	0.18	0.2
QCIF	128	50:1	0.15	0.1

It can be concluded that at 64 Kbps channel, it is not possible for the decoder to provide the minimum acceptable medical US image quality as well as frame rate of 5 fps for CIF format, where the main boundary was the limited channel data rate, whilst the QCIF format met the quality requirements at a very narrow range of CRs. CIF format has a specified compression ratio of 50:1, in order to perform good enough to comply with the minimum objective. The shaded values table 5.6 shows how critical is changing the CR above or below 50:1 threshold. However, the 128 Kbps channel can guarantee better performance on transmitting QCIF format.

Handling reliable frame intervals in reality affected directly by the delay jitter of the operating network. Therefore intelligent use of play out buffers at the receiver side is fundamental to a successful ultrasound stream delivery [75].

The outcome of the compressed ultrasound image transmission is translated in terms of quality and data rate as a function of the frame rate; these are ultimately representing the overall performance of the specified application. Figure 5.8 chose CIF format to present a comparative PSNR and data rate performance over both channel bandwidth settings:



(a) (b)

Figure 5.8 Ultrasound image quality and data rate received by the Expert station as a function of the frame rate, (a) at 64 Kbps, (b) at 128 Kbps.

In figure 5.8 (a) the minimum acceptable quality and frame rate was not fulfilled at 64 Kbps channel data rate, where at PSNR = 35 dB, the frame rate is 3 fps, this rate can only be acceptable for very preliminary medical diagnosis from OTELO viewpoint. To increase the frame rate to 5 fps, the quality of the encoded ultrasound stream should be reduced to about 31.5 dB, which in turn unacceptable clinically. Therefore there is no chance to trade of between the frame number and the quality when transmitting over 64Kbps, in other word the minimum acceptable criteria can not be reached.

In part (b) the performance improved, where the channel data rate increased up to 128 Kbps, and at 5 fps the PSNR = 35 dB. These results are critical as no trading between frame rate and the quality was obtained. These minimum acceptable limits were achieved using 63% of the channel bandwidth. The relationship of maximum frame rate with regards to the effective channel data rate of 128 Kbps, is by dividing the network bandwidth by the size of an image at each bit rate [112]. Therefore at bit rate produces 5 fps and PSNR of 35 dB:

> The image size is nearly 80 kbps/5fps = 16 Kbits per frame, hence the maximum frame rate is 128 Kbps/16 kbits/F = 8 fps.

Providing that the ultrasound CIF format encoded at 10 fps, therefore at maximum network efficiency, 2 fps will be lost. One solution to avoid frame loss is by reducing the encoded rate at the Patient station to 8 fps maximum.

5.4 ROI Evaluation Issues

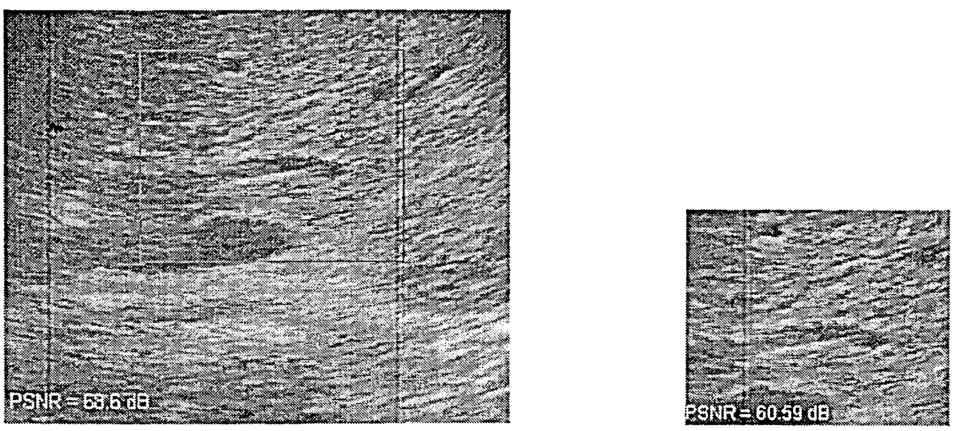
This section presents relevant results of the Region of Interest (ROI) for the OTELO system. It is well known, that in medical imaging any deficiency in diagnostically important regions (ROI), can effect the selection of the correct compression ratio. The general theme is to preserve the quality for diagnostically important regions, whereas the remainder of the image (background) is compressed with higher values. ROI coding usually support progressive transmission by quality, which may further reduce the transmission time and storage cost [122]. There are three considerations for ROI coding systems:

1. ROI rate distortion performance.
2. Background quality and
3. Algorithm computational complexity and implementation cost.

The main reason for preserving regions other than ROI is to let the viewer more easily locate the position of the critical regions in the original image, and to evaluate possible interactions with surrounding organs. Some applications used motion compensation coding and it produced a bit rate as low as 0.018 bpp for an acceptable error level for coding the non- ROI regions [123]. The same approach is thus used in the OTELO wireless testing environment as described later.

5.4.1 QCIF Format as ROI

For OTELO system, the first consideration for ROI coding system was approached. The ROI rate distortion has the first priority over low data rates channel and considering the background area as a second priority, when working on real-time mode. Moreover a complete high quality (lossless or near lossless compressed) still image transmission can take place over non real-time operation mode of OTELO system. Figure 5.9 shows QCIF format as a ROI area of the CIF format, where both equally compressed with respect to its original corresponding image acquired by OTELO system:



(a) (b)

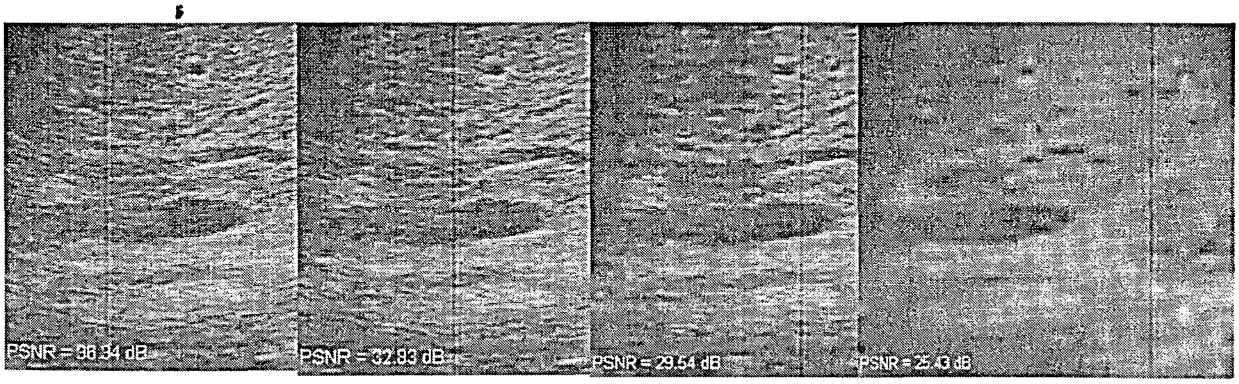
Figure 5.9 Ultrasound images for size comparison and evaluation, (a) CIF format compressed at compression rate = 1, PSNR = 68.6 dB, (b) ROI (QCIF) format with compression rate = 1, PSNR = 60.59 dB.

The figure shows resulting QCIF format of less quality compared to the CIF format of about 8 dB higher at compression rate = 1 bpp for both, although in terms of the file size a remarkable bandwidth saving obtained as the file size (13.6 Kbit) of the QCIF formats reduced to about ¼ of the CIF format size (54.2 Kbit) for the corresponding compression rate. Table 5.7 summarizes the results of ROI format compression compared to the CIF format:

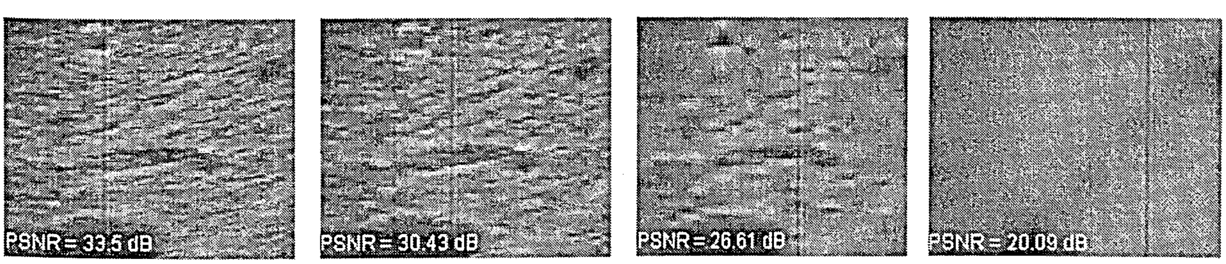
Table 5.7 Results of the ROI quality measure and the resulting ultrasound image size compared to CIF.

Rate = 0.005 - 1	Original Image Size = 101Kbit - Resolution CIF	PSNR	- Original Image Size = 26 Kbit - Resolution QCIF **(ROI)**	PSNR
bpp	Kbit	(dB)	Kbit	dB
1	54.2	68.6	13.6	60.59
0.75	54.2	68.6	13.6	60.59
0.5	49.3	51.71	12.1	50.44
0.4	39.3	47.88	9.87	46
0.3	29.5	44.38	7.22	42.98
0.2	19.6	41.18	4.77	39.66
0.1	9.68	36.34	2.11	33.5
0.08	7.62	34.91	1.94	33.04
0.05	4.9	32.83	1.16	30.43
0.03	2.92	30.98	0.726	28.1
0.02	1.97	29.54	0.5	26.61
0.01	0.988	27.57	0.249	20.09
0.005	0.494	25.43	0.233	14.65

The above still ultrasound images have been compressed using JPEG2000 at different compression ratios. The JPEG2000 standard, is superior to the current standards at low bit rate (below 0.25 bpp). This significantly improved low bit-rate performance should be achieved without sacrificing performance on the rest of the rate-distortion spectrum [124]. The last four highlighted images compressed at rates of 0.1, 0.05, 0.02 and 0.005 bpp represented in figure 5.10 below to show the deficiencies in PSNR quality values of both CIF and ROI (QCIF) for the same compression rates:



(a)



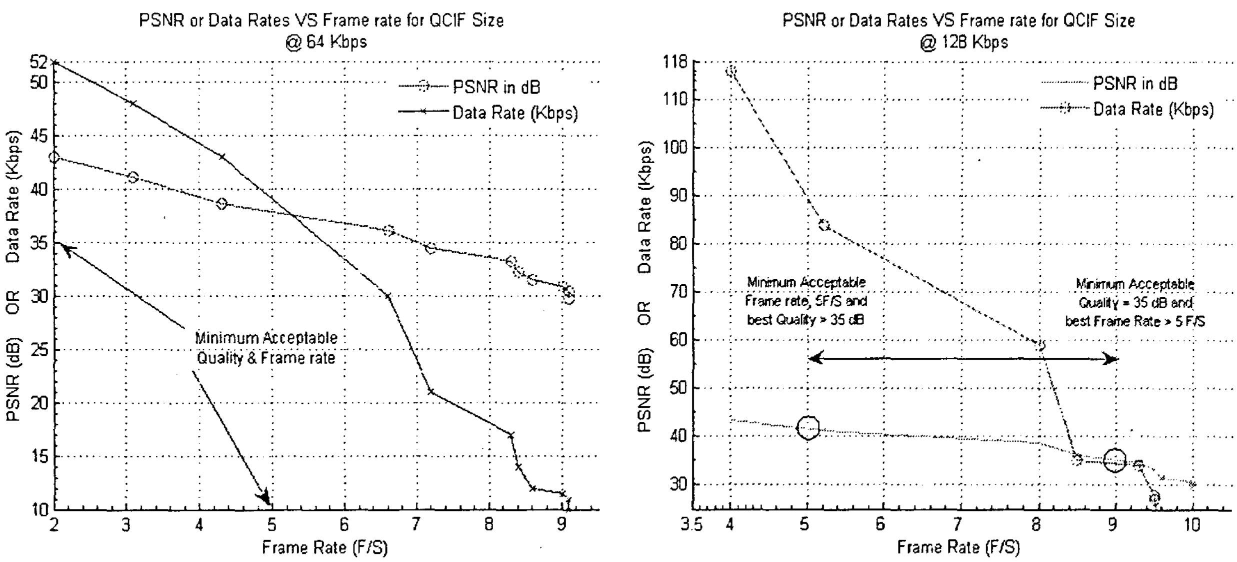
(b)

Figure 5.10 comparative quality evaluations of CIF and ROI compressed at 0.1, 0.05, 0.02 and 0.005 bpp sequentially from left to right, (a) CIF, (b) ROI.

For none diagnostically quality evaluation, it is clear the level of degradation of ROI quality compared to its CIF pair, which are compressed to the same rate. Therefore further improve in the quality of the ROI is required to maintain same PSNR measures for all image formats, although this is increasing the ROI image size proportionally further. For example CIF format of PSNR = 36.34 dB, shows better quality of about 2.8 dB to its corresponding ROI format although they both compressed to the same compression rate (0.1 bpp) as in table 5.7.

5.4.2 Evaluation of ROI Image Quality

The ROI images transmitted by OTELO system, under the same experimental set-up parameters discussed earlier and applied to the CIF format, were investigated. Figure 5.11 presents the performance of the ROI images received by the Expert station in terms of the PSNR, data rate as a function of the frame rates achieved:



(a) (b)

Figure 5.11 Quality and the data rate of the received ROI uultrasound image by the Expert station as a function of the frame rate, (a) at 64 Kbps, (b) at 128 Kbps.

Simulation results for 64 Kbps channel bandwidth, shows that receiving QCIF format of ultrasound stream is possible, where 5 fps to 7 fps obtained with an acceptable PSNR range of 38 dB to 35 dB respectively as shown in figure 5.11 (a). Therefore objective evaluation of transmitting QCIF format can be met providing that the Expert station buffering mechanism is to reduce the packet loss due to delay, using the minimum possible buffer size (or overall delay) [75]. Frame loss arises from the loss of one or more packets, depending on the encoding.

Figure 5.11 part (b) shows good performance of QCIF format over the 128 Kbps channel bandwidth ranging from minimum acceptable to higher frame rates without degrading the PSNR values. At 5 fps the encoder can produce the best quality of around 41 dB, and as the frame rates increased the quality decreased to its minimum limit (35 dB).

Furthermore the channel efficiency can be deduced from part (b), where it shows 70% at 5 fps, and as the frame rates increases the channel efficiency decreases since that involve objective quality reduction, as seen when the frame rate approaches 9 fps.

Channel data rate of 128 Kbps was therefore the threshold bandwidth required for OTLEO system to meet the real-time performance requirements. Table 5.8 summarizes the results discussed earlier and shows the acceptable ROI values to be use in OTELO 3G testing environments:

Table 5.8 objective evaluation of CIF &QCIF format over 128 Kbps channel bandwidth.

CR	RC %	Encoded Rate (F/S)	Ultrasound Stream Image Format							
			CIF over 128 Kbps				QCIF over 128 Kbps			
			PSNR (dB)	F. Rate (fps)	Data Rate (Kbps)	DR %	PSNR (dB)	F. Rate (fps)	Data Rate (Kbps)	DR %
4:1	75	-	-	-	-	-	-	-	-	-
10:1	90.0	10	44.1	0.8	96	8	43.5	4	116	40
14:1	92.8	10	42.2	1.2	105	12	41.2	5.2	84	52
21:1	95.2	10	39.5	2	103	20	38.7	8	59	80
35:1	97.1	10	37	3.6	100	36	36.3	8.5	35	85
50:1	98.0	10	35.3	4.7	91	47	34.6	9.7	34	97
65:1	98.4	10	34	5.2	71	52	33.3	9.6	28	96
81:1	98.7	10	33	5.6	55	56	33.2	9.3	27	93
94:1	98.9	10	32.1	5.7	50	57	31.6	9.6	20	96
113:1	99.1	10	31.4	5.8	44	58	30.9	9.7	18	97
121:1	99.2	10	30.8	5.6	41	56	30.4	9.9	17	99
131:1	99.23	10	30.2	5.8	37	58	29.8	9.8	15	98

The table clearly shows that ROI format offer more operating options compared to the CIF format, where there is flexibility in trade-of between the quality (PSNR) and the frame rate as shown in the shaded area of the table. Hence, this can be provide the best comparative options for the best performance of the OTELO imaging in the specified 3G wireless link.

5.5 Summary

The chapter presents the experimental and simulated studies dealt with the objective quality evaluation of the transmitted ultrasound still image and ultrasound streaming. The quality evaluation matrix adopted include the frame rate, PSNR measure, Bit per Pixels and the data rate performance at different compression rates and channel conditions. It can be concluded that the GPRS network with its present capability is not suitable for OTELO tele-ultrasonography functionalities to provide minimum quality requirements. However the experimental studies over simulated UMTS network shows good capability of 3G environment to fulfill the minimum quality requirements, and above the minimum with improved ROI performance.

CHAPTER 6

Quality Adaptation Technique for Enhanced OTELO video Streaming

6.1 Introduction

This chapter presents new quality adaptation technique designed to optimize the medical ultrasound and video streaming in a robotic ultra-sonography system. The major objective is the accommodation of the medical QoS requirements in terms of the image quality and frame rate. In the first section we describe the adaptation technique used on the application level and the relevant video streaming methods. In section two we illustrate the problem formulation and the design of the control algorithm. Section three presents the results and discussion.

6.1.1 Mobile Ultrasound and Video Streaming Adaptation Issues

In wireless telemedical environment with bandwidth-restricted, optimum video delivery is a demanding but a technically challenging issue. Exploiting the trade-off between the image-quality and the frame-rate in adapting to varying channel bandwidth is an attractive solution [125]. In packet-switched networks, packets may be discarded due to buffer overflow at intermediate nodes of the network, or may be considered lost due to long queuing delays. This problem become more severe and the packet loss rate in Internet communications for example may reach 20%. Robustness to packet loss (congestion control) becomes a crucial requirement [126].

The feedback information in a heterogeneous communication network, that identifies the source of the packet loss, can be summarised as follow:

a) *Receiver Feedback Information;* The feedback information provided in RTCP packets in receiver reports (RR) is used to derive loss rates at the sender [127]. The source then uses a control algorithm to adjust the maximum rate of the coder (λ). The adjustment algorithm uses the principles of linear increase and multiplicative decreases of λ to ensure that reasonable efficiency and fairness are maintained. Congestion in the network is deduced from the RR reports. When the loss rate reported by the RTCP exceeds a high-loss threshold (e.g. 5%), the source declares that the network is congested and that λ need to be reduced. When the loss rate drops below a low-loss threshold (e.g. 2%), the source declares that the network is uncongested:

If(congested)
$\lambda = max\ \{\ \lambda * \alpha,\ minimum\_rate\};$
if(uncogested)
$\lambda = min\ \{\ \lambda + \beta,\ maximum\_rate\};$

Where two variables are defined; reduction factor; $\alpha \leq 1$ and an increase factor; β. The algorithm ensures that the source coder's rate neither goes below a *minimum_rate* nor exceeds a *maximum_rate*.

b) *Streaming Agent:* the end-to-end feedback information alone is ineffective for congestion control purposes since it is not possible to identify where losses occur. Specially, if losses occur in the wireless link due to poor wireless condition, it is not helpful for the sources to reduce their transmission rate. On other hand, if losses occur in the wired network due to congestion, the source should reduce their transmission rate [128]. One of the limitations of using only end-to-end feedback is the long time for the feedback to arrive, in today's 3G wireless network typical one way delay of radio link is quite large, on order of 100ms without link layer retransmission. The problem above can be solved simultaneously using a special agent called a streaming Agent (SA), located at the junction of the wired and wireless link [128].

c) *Transmitter Buffer:* This is based on UDP, since UDP uses simple datagram with no congestion control, this task is performed at the application layer. When congestion occurs, the UDP sender buffer will overflow and the information transmitter will drop the frame that causes the error to ease the congestion [77]. Therefore the judgment on transmission attempts is based on the availability of the transmitter buffer.

Since the redundancy level varies from frame to frame in the ultrasound images, therefore it is clear that the number of bits per frame in such scenario must be variable. A fundamental result in video coding is that maintaining a constant (perceptual of objective) quality, throughout and sequence requires a variable rate allocation [75]. For optimal OTELO system performance over variable channel bandwidth and relatively high data rates varying between 64 Kbps to 512 Kbps as in 3G wireless environment, the M-JPEG Codec is preferable comparing to H.263 that was described in chapter five, due to the following specifications:

(i) It is widely recognized that intra-coding is an important tool for mitigating the effects of packet loss, However, the robustness provided by intra-coding may be costly, as it typically requires a higher bit rate than inter-coding (with prediction) [126]. M-JPEG uses no inter-frame compression, which results in low latency transmissions. The quality of the image will vary greatly based on the speed and quality of the codec as well as the availability of bandwidth for transmission [129]. That means; M-JPEG architecture uses normal JPEG tiles, where the errors or packet loss on the network only impact a tile, or row of tiles, in an image, and the error does not propagate for several frames.

(ii) M-JPEG does not impose stringent requirements on network bandwidth. The frame-size, compression quality, and frame-rate of M-JPEG images can be adjusted to suit a given network capacity. Moreover, M-JPEG is much easier to synchronize with audio using time-stamps [125].

(iii) M-JPEG has become a demanding choice of video delivery due to many inherent attractive features. It is a choice for applications that require high-quality individual images at a fair rate rather than low-quality smooth video. Irrespective of the underlying transport protocol, the end applications should consider certain issues for optimal M-JPEG delivery [125].

Many approaches addressed different methods for efficient transmission of stored M-JPEG video over bandwidth varying channels; moreover very few works have addressed the real-time and end-host performance issues. A multimedia system for homecare and developed a packet loss based static methodology for switching M-JPEG streaming between TCP and UDP, is provisionally addressed. Other methods describe the M-JPEG frame rate for transmission over UDP by dynamically measuring the available bandwidth [125]. The adaptation technique presented here is based on the practical experimentations indicates that the optimal frame rate and image quality can not be easily achieved especially in a medical robotic environment such as OTELO, due to the following:

- Over relatively low-data rates wireless channel, the network load and the variation on the packet loss is not predictable from the user end so as to adapt the quality setting accordingly. For example rates below 64 Kbps, may lead to annoying blocking artifacts or require operation at low frame rates, resulting in low temporal resolution and long end-to-end delay [130].

- Channel data rate availability and the congestion variation may lead to a fluctuating quality performance above and below the quality setting of the user. This is quite possible for end users like the Expert and the patient station operator in OTELO system environment.

Hence the adaptation algorithm applied to enhanced M-JPEG performance in OTELO environment is implemented within the application layer of both system ends:

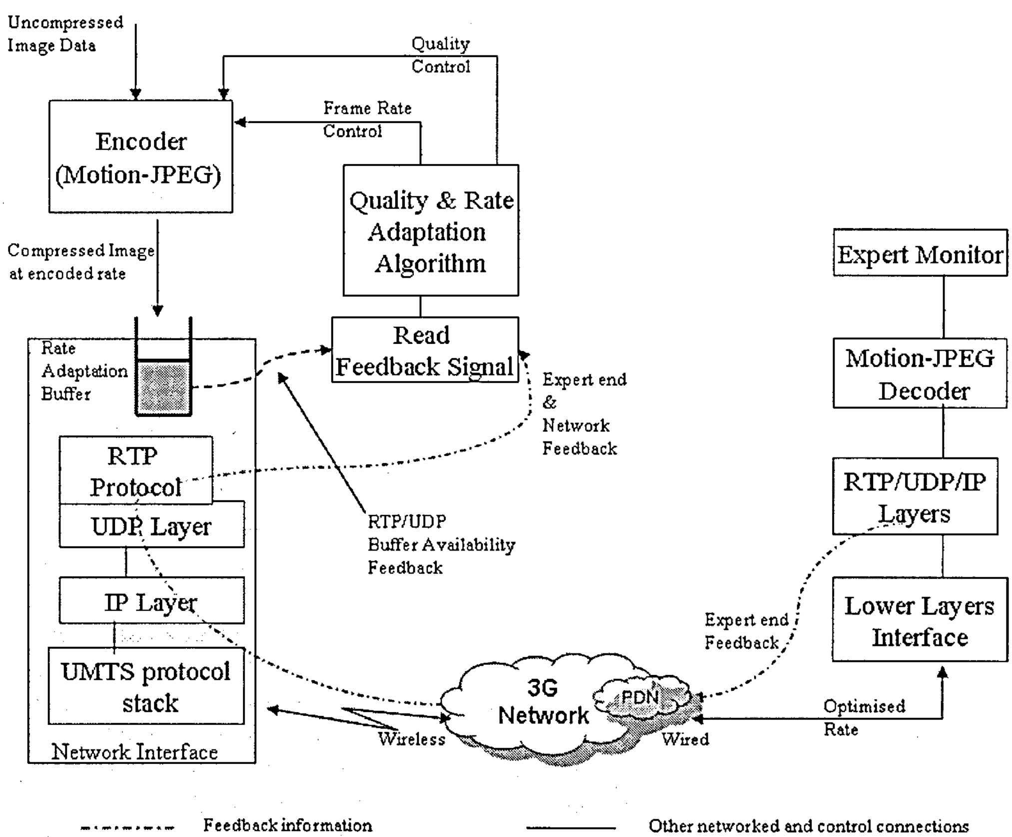


Figure 6.1 Adaptation for enhanced M-JPEG performance in 3G wireless environment.

The algorithm is allocated in the Patient station, where the encoding process performed by M-JPEG and controlled by feedback information acquired from different location within the network, as illustrated in figure 6.2.

6.1.2 Adaptation Technique and Rate Control

The minimum QoS requirements of medical ultrasound stream discussed earlier in chapter five are considered as the basic requirements for lower quality adaptation. It is known that the requirement for video on the network transport layer is to have very small packet loss probability, in the range of 0.1-5%, depending on the user. This is quite difficult to achieve, unless the network is considerably over provisioned [130]. Our approach will work over narrower packet loss range; (below 2%) for ultrasound stream and (below 5%) for good quality ambient video measure.

It is assumed that the packet loss rate is available at the M-JPEG encoder, this can be either specified as part of the initial negotiations, or adaptively calculated from information provided by the transmission protocol. For example, the RTCP provides the encoder with information for calculation of packet loss rate, packet delay, and delay jitter [126]. The output of the M-JPEG encoder is fed to a rate adaptation buffer at the source that is used to accommodate the variability in demand from the encoder [75]. It also accommodates the difference between the rate coming into the buffer and the output rate from the source into the network. The parameters of the proposed adaptation algorithm can be explained as follow:

1. Control Objectives;

Although we assume that both system ends are using sufficient memory, implementing good buffering system and reasonable consumption of CPU power, nevertheless the algorithm tend to avoid unnecessary operating system loading for reliable performance. M-JPEG encoding process has direct effect on the issues mentioned above, where the adaptation software has to deal with the following:

a) Adaptation to network bandwidth: This is directly related to the change in the frame size, image-quality, and frame-rate of the M-JPEG stream.

b) Adaptation receiver performance: since the UDP does not guarantee reliable delivery of the ultrasound packets, therefore the receiver end (Expert station) should implement capable CPU and good buffer management, as well as the negotiation already running by the GUI of the Patient/Expert application software.

c) Real-time delivery: that is our main algorithm objective, can be achieved by appropriate adaptation to the network congestion and buffer management at the sending socket.

d) Between the peer UDP layers, this guarantee is not extended up to the end application due to problems like CPU overloading. Further, the image frame decoding might fail due to lose of header synchronization etc. Therefore, it is essential to perform application level negotiation to avoid useless delivery, and thus enhance the transmission efficiency.

e) CPU overloading at the sender: it is essential to be prevented, since CPU power is heavily consumed when capturing images from the camera and for subsequent compression. Running the CPU at the fullest capacity leads to instabilities in the application itself as well as causes the PC irresponsive to secondary applications [125].

2. Video and Ultrasound Streaming Parameters:
These are the outcome of the adaptation process and are summarized as follow:

a) Image resolution: as we will see the control algorithm is adaptable to either OTELO system ambient video, or ultrasound stream (highest priority) control. It has been found that image resolution of QCIF is sufficient for the system ambient video quality requirement. For ultrasound stream imaging, CIF format is the minimum size required.

b) Image quality: for JPEG compression implementation, it has been found that reduction of quality below 20%, deteriorates the image-quality, although it reduces the file size remarkably (as found by the works in chapter five). In the mean time increasing the quality above 80%, would increase remarkably the file size. Hence and precisely for the ultrasound streaming we will work within quality region that guarantee the QoS over good network conditions.

c) Frame rate: The frame-rate for the US image is bounded by minimum acceptable quality of 5 fps and up to 10 fps as upper bound, where beyond that will unnecessarily overload the network. And for the ambient video 3 fps to 5 fps are the lower and upper bounds respectively. In both cases the Patient station operating system power is considered, especially if we know that this end represents the bottleneck of the entire system capability in terms of the available bandwidths.

We believe that for such a consistent medical diagnosis and based on expert opinions the option of ultrasound stream quality and frame rate adaptation to the communication channel changes is diagnostically convenient, rather than changing the image frame size. Changing the image frame size based on the network condition may disturb the medical diagnosis process.

6.2 Enhanced M-JPEG Streaming Adaptation Approach

In this section we formulate the quality levels targeted by the new adaptation algorithm. The algorithm for adaptive M-JPEG streaming involves basically two control levels. These levels will be implemented in correct sequence in case of non-congested network, and in a reverse sequence in case of congested network as follow:

1- Quality rate control (level one); this is based on adaptation to the available channel capacity provided by the feedback information in terms of uncongested network (Packet loss, level one), where initially the frame rate will be kept by the encoder at its minimum rate (5 fps) and gradual increase in the ultrasound image quality will take place after the first packet loss threshold (P_{th1}), to reach the first quality level of 30%.

2- Quality rate control (level two) plus Frame rate control; at the second level of uncongested network (Packet loss level two), the increase of the quality will continue up to a certain level (upper bound, that is up to 50%). Furthermore if the network stays in the same condition, the algorithm will increment the frame rate to its upper bound, which is 10 fps. Implementing this mode is at packet loss threshold (P_{th2}).

The functionality of the presented method is shown in figure 6.2, where OTELO system can accommodate multiple congestion monitoring tools for the network condition. In this figure the US stream and the ambient video are the system data subject to the quality control at specified relevant QoS:

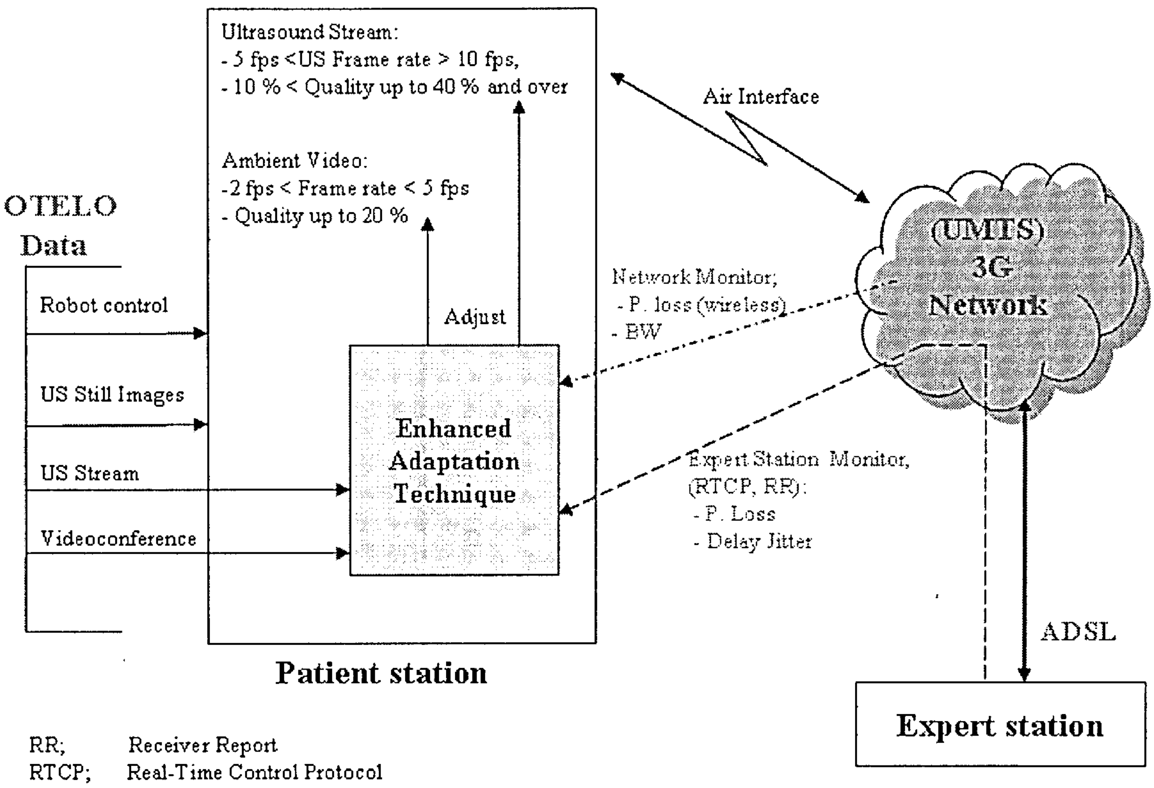


Figure 6.2 Quality adaptation technique for end-to-end OTELO system.

6.2.1 Adaptation for OTELO Real-time Performance

In this approach the sender (Patient station) encoder is made free from the user instruction specifying the quality and the frame rate of the required medical images, where some implementation follows this adaptation scenario as in [77]. Hence, this approach will allow the Patient station to adapt itself to real-time or near real-time operation. The algorithm initially mitigates the changes in the network condition (congestion), and then operates within the boundaries of the QoS required by OTELO system.

In real-time environment, for the decoder to have data to decode, requires the transmitted bit stream to comply with a series of rate constrains. These constrains depend on the type of transmission environment considered, including the variability, or lack of it, in the transmitted rate, and the end-to-end delay [75]. Thus, after a transmission has been chosen, the video encoder will be responsible for producing a bit stream that meets the relevant delay constrains.

The algorithm implemented is based on Variable Bit Rate (VBR) video encoding that is defined as: a video encoder is VBR if it produces a variable number of bits per frame. Where it has been found that VBR transmission provides benefits in lower delay, higher video quality, and increased network utilization [75].

The proposed adaptation algorithm should be able to detect the stream signal to be encoded, weather ambient video or ultrasound stream signal by reading the sending port or specified manually by the operator of the sender. Furthermore after detecting the applied signal, the algorithm will implement the relevant QoS adaptation, as seen in the illustrative flow chart in figure 6.3:

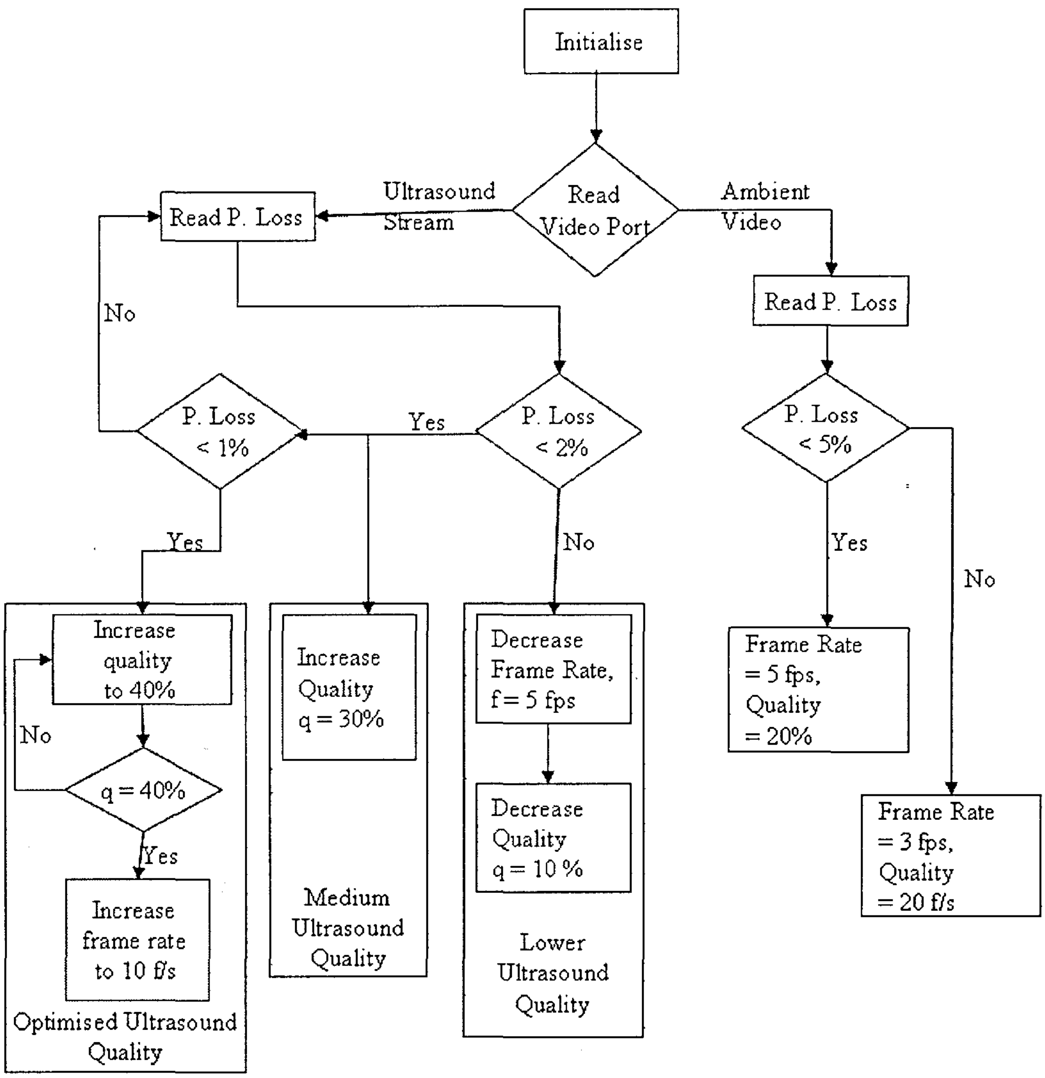


Figure 6.3 Flow chart representation of the adaptation Algorithm.

As shown from figure 6.3, the adaptation algorithm provides two levels for ambient video quality control and three levels for the ultrasound stream control. Furthermore it specifies the operating image quality limits.

6.2.2 Quality Rate and Frame Rate Selection

The adaptation algorithm as described earlier, involves basically the adaptation to the available channel data rate through monitoring the packet loss feedback information (e.g. RTCP receiver report), to be compared to two packet loss thresholds. The adaptation algorithm priorities the image quality first then if the communication channel is still uncongested, the algorithm increases the frame rate. From earlier works, the image-quality increase has much lower impact on the channel consumption if compared to the higher impact of the frame-rate increases [125, 128]. The algorithmic notations are summarized in the table 6.1:

Table 6.1 Notations of the adaptation algorithmic parameters.

Control notations	Description
P_l	Packet loss
P_{th1}	Packet loss threshold 1 (2 %)
P_{th2}	Packet loss threshold 2 (1 %)
P_{th3}	Packet loss threshold 3 (5 %, Ambient)
F_{opt}	Optimised frame rate
F_r	Current frame rate
Q_{opt}	Obtimised quality
Q_r	Current quality
F_{LL}	Frame Rate Lower Limit (5 f/s)
F_{UL}	Frame Rate Upper Limit (10 f/s)
Q_{LL}	Quality Lower Limit (10 %)
Q_{30}	Quality = 30 %
Q_{UL}	Quality Upper limit (40 %)

Quality Rate Control:
In this case, the adaptation algorithm will respond first to the quality rate increment as long as the packet loss reading is not exceeding the first threshold, therefore at this stage the system adopt to the channel variation by controlling the M-JPEG compression rate (lossy mode; generally accepted JPEG compression ratio for echocardiology is up to 20:1 for still images, and up to 30:1 for motion images) [75]. The quality-adaptation algorithm, specified for ultrasound stream is based on the following:

1- Initialization: set Packet Loss threshold; P_{th1} & P_{th2}, initialize Frame rate & Quality limits.

2- Decrease frame rate then quality rate;

```
Pl >= Pth1;
        if (Fr >= FLL) then;
           Fopt = max (Fr -1, FLL)

else

        if ((Fr = FLL ) && (Qr > QLL))) then;
           Qopt = max (Qr - 10, QLL)
```

3- Perform Quality rate control (level one):

```
Pl <= Pth1;
        if ((Qr < QUL) && (Fr = FLL) then;
           Qopt = min (Qr + 5, Q30);

else    decrease quality;

        if ((Fr = FLL) && (Qr > QLL) then;
           Qopt = max (Qr – 10, Q30)
```

Therefore at this moderate packet loss range (1% < P. Loss < 2%), the Expert station should expect frame rate equal to 5 fps and ultrasound steam quality up to 30% respectively. These enhanced limits are following the network condition rather the user setting.

Frame rate Control:
The adaptation algorithm is carrying on comparing the packet loss with the threshold values, this process is updating the algorithm with the new packet loss rate every 2 sec. to avoid rapid and unnecessary control and in the mean time not to consume the sending computer CPU vigorously. If the packet loss found to be < 1%, the following code of the algorithm will then be executed:

4- Quality rate control (level two), then frame rate control:

```
Pl <= Pth2;
        if(Qr < QUL) then
           Qopt = min (Qr +5, QUL)

else

        if((Qr = QUL) && (Fr <= FUL)) then;
           Fopt = min (Fr +1, FUL)
```

At this stage the algorithm bring the overall quality measures to its ultimate values, i.e. maximum quality of 40% and maximum frame rate of 10 fps.

6.3 Simulation Set-up and Results Discussion

The adaptation algorithm is practically implemented in the Windows 2000 operating system environment. We assume that the receiver side implements good buffer size to avoid buffer management and any delay might be added by the receiver. The operating processor should be capable to accommodate the reception of the images frame packets. It is also recommended that minimum second priority applications to be run at the Expert station side.

A simulation environment representing OTELO station has been set-up. The simulation platform of this approach has been performed using Cloud Emulator (as discussed earlier in chapters three and four) as a network simulator. Hospital/Patient software developed in [77] has been used as application software. We implemented the adaptation algorithm as an embedded control software in the Hospital/Patient software mentioned, this software is implemented on a commercially development platform—Labview for Windows operating system [131].

For the stream data acquisition, the Patient station unit equipped by a PC of Pentium-4 (1.7-GHz), and connected to a high resolution USB port Camera or IEEE 1394 compatible camera for ambient video, the ultrasound signal is fed to second USB port through Belkin video input/Serial output adaptor, where the ultrasound machine is connected.

The video and ultrasound signals are captured in real-time from the relevant interface. For this work UDP/IP protocol has been used by the platform, since UDP is not a connection-based protocol such as TCP, we do not need to establish a connection with a destination before sending or receiving data. In theory datagrams of any size can be sent by using UDP, however we will not send large datagrams (maximum 1000 Bytes) since UDP is not as reliable as TCP [131].

The Hospital (Expert station) unit; implemented by Pentium III 1.2 GHz Laptop connected to the simulation set-up through Ethernet card. The Labview (IMAQ tool kit) for multipurpose image processing and machine vision capabilities has been used to the stream data acquisition.

Simulation Results and Discussion:

The results obtained, summarize the performance analysis of the adaptation technique. Figure 6.4 shows a typical run of this simulation results as applied for the presented algorithmic approach:

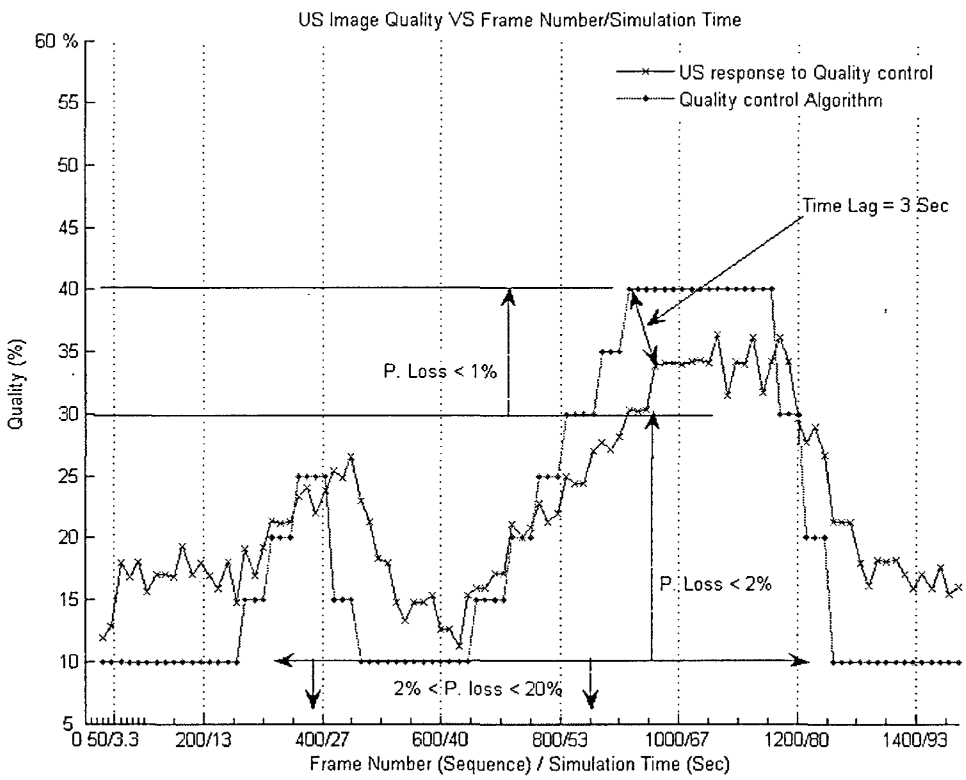


Figure 6.4 Simulated US stream quality performance with the adaptation algorithm.

Two major areas are chosen for the relevant analysis; during the first one the packet loss lasted for 13 sec and during the second one the packet loss lasted for 20 sec. A time lag of maximum 3 sec seen between the second control cycle compared to the corresponding quality response. The algorithm suggests waiting about 2 sec to avoid algorithm oscillation before changing to new state. When the packet loss exceeds 2% the algorithm should maintain the quality and the frame rate to its minimum values, 10% and 5 fps respectively.

The packetisation of the ultrasound frame packet length of 250 Bytes (data only) has been set at both ends. In a lossy packet environment; the use of appropriate packetisation techniques can improve error resilience substantially [77]. Packetising the JPEG frame at lower packet sizes has an advantage of avoiding fragmentation that the IP layer is usually accomplishing [77], which may lead to higher packet losses, and in the mean time enhance the real-time reception of the US stream, although on other hand this process involve high number of packet headers.

Figure 6.5 shows the ultrasound packet accumulation based on the mentioned packet length as a function of the quality rate:

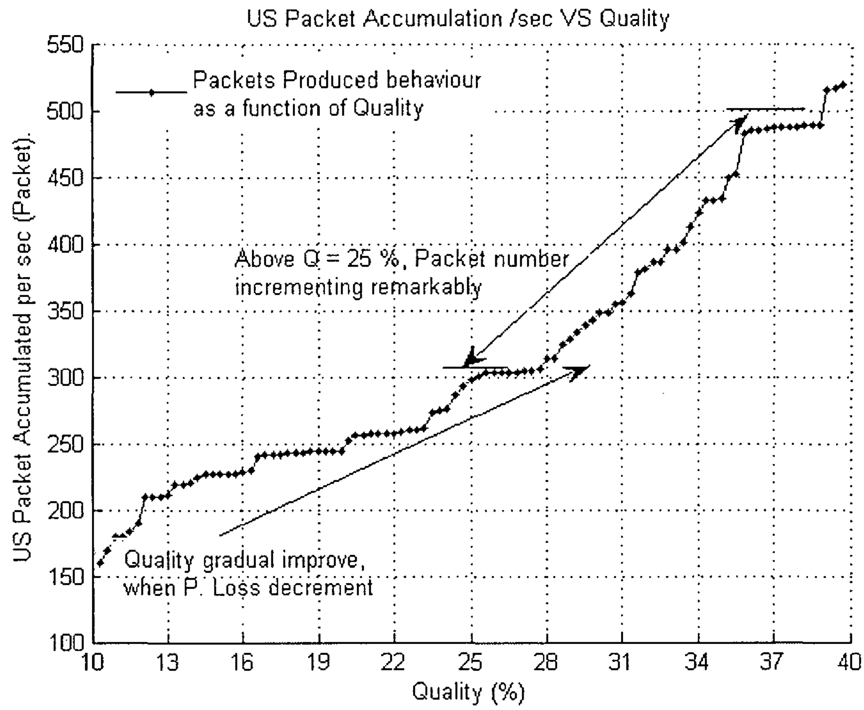


Figure 6.5 US packet accumulation plot at different Quality rates.

The accumulated packet trace, shows areas when the control algorithm responding to different packet loss levels in terms of number of packets/second, as summarized in table 6.2 for selected quality ranges:

Table 6.2 Comparative performance of US packets at selected Quality ranges.

Quality Rate Range (%)	13 - 16	22 - 25	28 - 31	37 - 40
US Packets/Sec. (packets)	215 - 230	260 - 290	310 - 360	480 - 520

From the table we can conclude that packet size increases due to improving the image quality causes less network loading compared to the frame rate multiplication. The higher level of adaptation is when the packet loss < 1%, that means and providing that the quality rates at its maximum values (40% or 50%, these are optional upper values set by the operator before the start), then the frame rate increment can take place. The timing relation between the quality rate and the frame rate is shown in figure 6.6:

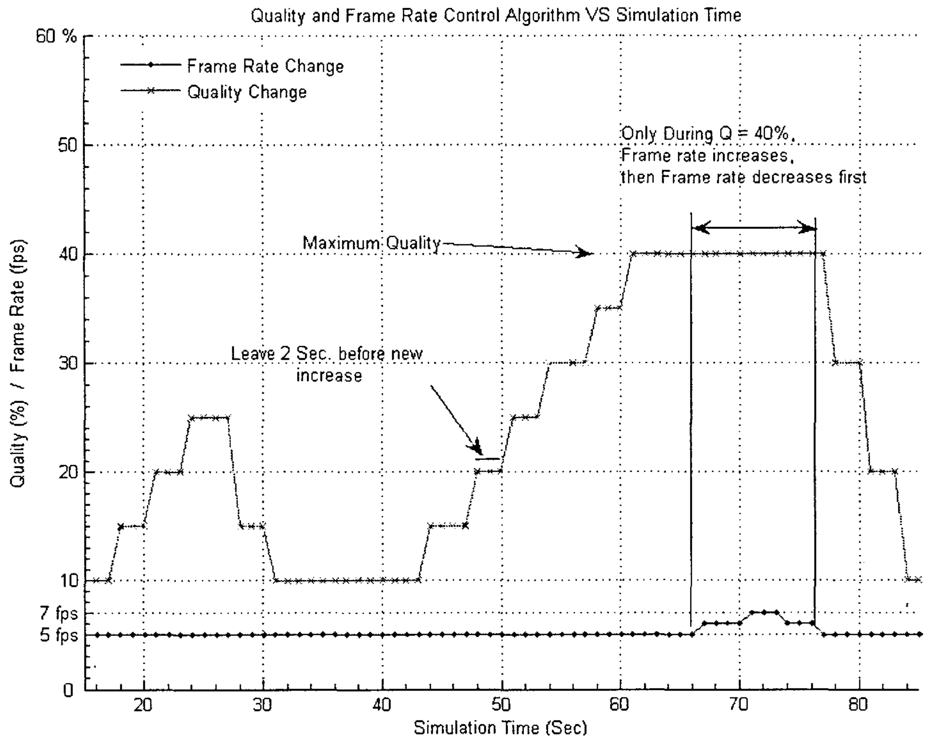


Figure 6.6 The simulated timing relation between the quality and frame rate adaptation process.

The figure represents the complete adaptation control cycle, starting with quality then frame control. We have implemented two cycles of packet loss detection:

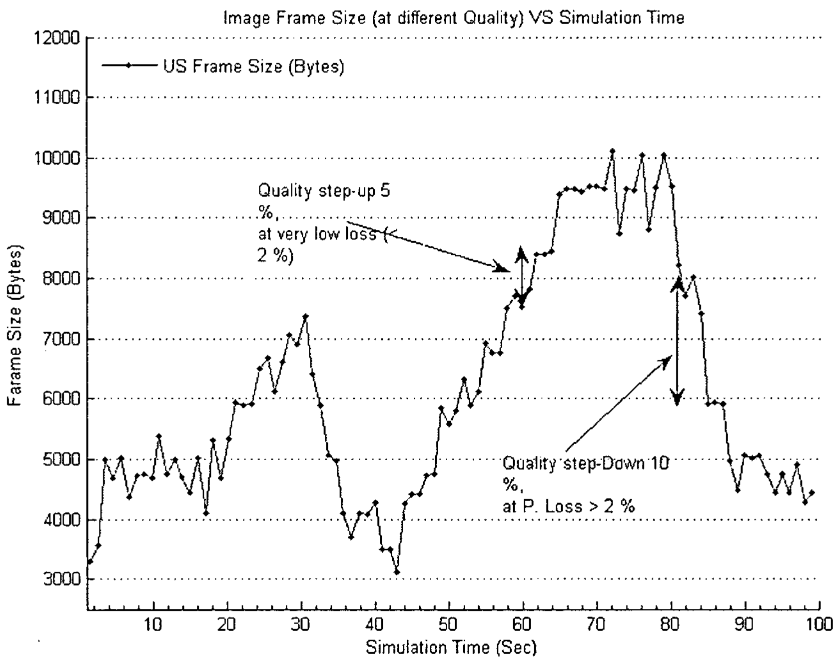
The first one lasted for 9 seconds of packet loss < 2%, where the maximum quality reached was about 25% and increases 5% at each step. Then packet loss detected to be > 2% for about 4 seconds, therefore the quality dropped this time 10% each step as the algorithm specify.

The second control cycle lasted for about 18 seconds, detecting that the packet loss was < 2%, and stays good enough to bring the quality to medium value (30%). The packet losses still improving to be < 1%, therefore the algorithm brought the quality to its maximum value (40%).

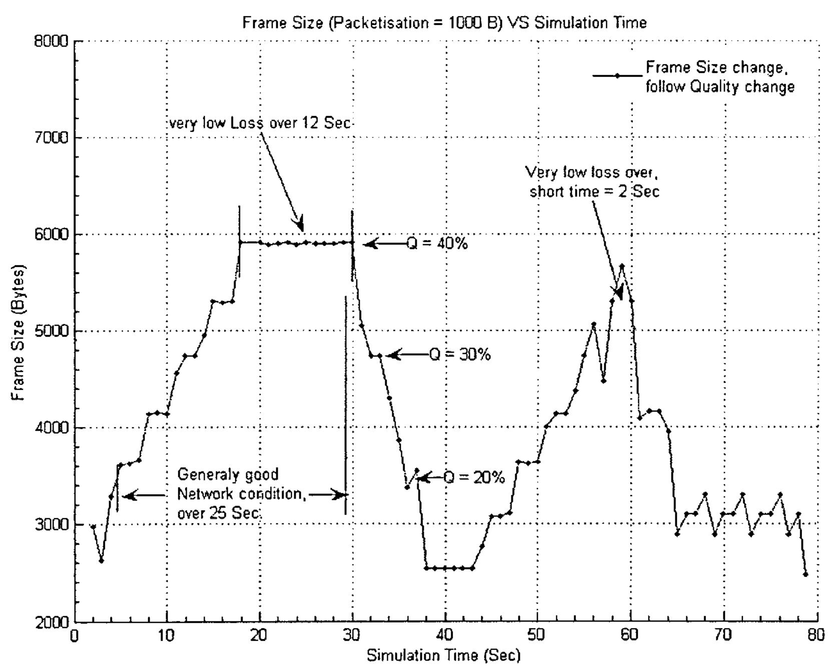
Now if the packet loss stays < 1%, for about 5 second (this time is important to probe the network before the frame increase), then as seen from figure 6.6 the ultrasound frames will be incremented by one frame each step, starting from 5 fps to maximum of 10 fps. In the example of figure 6.6 the increment reaches 7 fps then packet loss detected to be > 1%, therefore the number of frames reduced to its minimum, and immediately after that the quality started decrementing.

The frame increment has a crucial effect on the network congestion, therefore we choose to increase one frame each time, and avoiding multiple decrementing for the reliability of our application. We have to mention here that in real network operation the algorithm should follow the congestion status in terms of the packet loss that is happening randomly and completely unpredictable.

One of the important issues to be considered in this application is the packet length (packetisaion) of the stream data subject to M-JPEG compression and the effect of the adaptation algorithm on the final frame size produced [105, 74]. Figure 6.7 shows a comparative frame sizes resulting from packetisation at 250 Bytes and 1000 Bytes respectively as a function of controlled quality rate:



(a)



(b)

Figure 6.7 Frame size optimised as a function of different packet lengths, (a) 250 Bytes. (b) 1000 Bytes.

Clearly that 1000 Bytes packet length is consuming much less network bandwidth, compared to 250 Bytes packet length. For example at figure 6.7 (a) the time taken for the quality to span over the range 10% to 40%, was about 19 sec. whilst the same quality range took about 14 sec in part (b) for 1000 Bytes packet length.

Nevertheless small packet lengths is preferable to avoid IP fragmentation, where if any one of the small IP blocks is corrupted, the original big packet will be dropped completely, thus increase the effect of packet loss rate [77], consequently the quality of the received image.

A comparative results based on figure 6.7 summarises the frame size (payload + headers) at different packet length as a function of the quality rate control is shown in table 6.3 below:

Table 6.3 Comparative performance of different frame sizes.

		Quality Rate			
		At 10%	At 20%	At 30%	At 40%
Frame Size (Bytes)	At 250 Bytes packet length	4500 B	5900 B	7900 B	9500 B
	At 1000 Bytes Packet length	3100 B	3500 B	4700 B	5900 B

Depending on the encoding, the medical video has varying amounts of tolerance to frame loss. Typically, no more than a very small probability of frame loss is acceptable. Loss of one or several packets can lead to frame loss; the sensitivity is depending on the packetization.

6.4 Summary

This chapter presented a new quality adaptation algorithm for improved performance of OTELO system. The simulation results using simulated 3G network shows good medical streaming quality response to network congestion. The adaptation shows that improving the streaming quality should take the priority, then increasing the frame rate. The work here is focused on non-scalable medical video quality as scalable video quality is generally not desirable for medical ultrasound diagnostic purposes and provides problematic and unnecessary congestion on the operating 3G network.

CHAPTER 7

Conclusions and Future Work

7.1 Introduction

This chapter presents the main conclusions of the work carried out in this thesis and the performance of OTELO system in 2.5G and 3G networks. Suggestions for future work in this area are also addressed in this chapter for future improvement on the wireless performance of this mobile robotic system especially in 3.5G and 4G environments.

7.1.1 Conclusions on GPRS performance

- From the simulation studies and since that GPRS system is capable of transmitting good quality still medical US images up to data rate of 53.6Kbps (Patient station Uplink) with proper JPEG2000 compression ratios, therefore GPRS network can be the none-real time standby network for OTELO system functionality.

- The compression ratio of the transmitted Images directly effects the transmission time, data transmission speed and the ratio of the TCP number of packets needed to complete the transmission. A trade-of is required between the Image file size and the compression ratio to improve the overall performance of the GPRS communication link.

- The results indicates that the GPRS link can efficiently perform ultrasound image transmission (non real-time) at compression ratio (CR = 08:1), where the disadvantage of higher compression was degradation of the image quality, in the mean time the disadvantage of compression lower than (08:1), was less average throughput, hence lower channel usage (efficiency).

- Larger delay and the delay jitter over the heterogeneous networks maintaining the connectivity of OTELO system ends, was around 83% over the GPRS link.

- In this study, we conclude that the variation of the compression ratio of the transmitted ultrasound images has no major effect on the RTT taken for a single packet to be acknowledged. Increasing the data rate of the channel to 53.6 Kbps (4 TS) will further enhance the GPRS capability to fulfil OTELO end-to-end medical user requirements.

- In the GPRS link, reliable real-time ultrasound streaming will not be possible, regardless the number of time slots assigned to the Patient station Uplink. Although US stream is packetized at lower sizes, nevertheless even near real-time performance was not possible.

- For the OTELO robotic control performance over the GPRS link, real-time robotic data operation only was possible with one GPRS TS assigned to the Patient station Uplink.

- In GPRS system, mixed traffic (packet-switched and circuit-switched), has significant influence on the performance of TCP traffic of the ultrasound still images transmission.

- There are two main performance parameters that disqualify the GPRS link from being reliable communication link for OTELO system requirements; these are the end-to-end packet delay and the delay jitter.

7.1.2 Conclusions on UMTS Performance

- The medical US still images showed better performance compared to the GPRS link, since the TCP packets retransmission events was very rear over the UMTS link, that should encourage to transmit ultrasound images with lower compression. In case of retransmission, there was no more than one retransmission event happening for the packet that acknowledged lost.

- The 3G experimental test results for transmitting wireless medical ultrasound streams encoded with Quarter CIF (QCIF) size images using the H.263 Codec, indicate the successful transmission with the following rates; 5 fps and objective quality measure of PSNR = 35dB. These values represent the minimum bounds that are clinically acceptable by the medical experts using the OTELO system for pre-diagnostic requirements. These results are based on Patient station Uplink data rate of up to 64 Kbps.

- The results also indicate that the network delay jitter variations are still within the acceptable boundaries of maintaining high quality real-time interaction between both system ends. The maximum delay of Probe movement-to-Received image (Expert-to-Patient-to-Expert) path is shown to be approximately 325 ms for the specified ultrasound encoding conditions.

- The robotic control of the system based on UDP/IP protocol over UMTS network, indicate very reliable system performance with minimal packet loss of < 0.5%, at Patient station uplink of 64 Kbps.

- Test results indicate that during OTELO system ultrasound scan, the possibility of transmitting voice or other data over the link depends on the availability of the 3G link resources (data rates). However, we noticed that the voice packet takes priority and causes degradation on the ultrasound reception. In addition, during the 'ultrasound scan', voice is rarely needed by either the patient or the expert. If the available 3G bandwidth can accommodate both medical data types and other videoconference data, in such case all data can be transmitted simultaneously, in addition the medical streaming is the most data rate demanding elements in the system.

- The video codec H.263 is much more convenient for the videoconference performance rather than the ultrasound stream compared to M-JPEG, where with H.263 the loss in the image propagates over the network due to the criteria of *inter* and *intra* coding nature. Therefore M-JPEG found to be more efficient for ultrasound streaming although this involve higher data rates requirements.

- The implementation of ROI combined with a quality adaptation algorithm can boost further the performance of the system to very efficient levels.

7.1.3 General Conclusions

From the results and conclusions, we can summarize the overall performance issues of OTELO system as follow:

- The overall test results of GPRS link performance, indicate the difficulty of performing clinically acceptable diagnostic US image quality for such robotic ultra-sonography system using the current GPRS operational network conditions.

- Adaptation techniques to mitigate the mobile network condition changes, can improve remarkably the performance of ultrasonography system especially in unreliable wireless communication environments, by mitigating the system performance to the changes (congestion and delay) happening in the network.

- The initial experimental performance results of the system between an Expert hospital and remote area (within the London area), has shown that OTELO system with current 3G operational network and traffic conditions can perform successfully with medically acceptable Ultrasonography stream and imaging data transmission. In general we can conclude that such advanced mobile robotic telemedical systems can successfully perform clinically acceptable Tele-ultrasonography sessions in 3G environments.

7.2 Future Work

A major goal toward the 4G Wireless evolution is providing pervasive computing environments that can seamlessly and ubiquitously support users in accomplishing their tasks, in accessing information or communicating with other users at anytime, anywhere, and from any device [132]. Fitting OTELO robotic m-health system in this environment, is requiring intensive works and adaptation to full mobility remote ultra-sonography, which can be the heart of the wireless Hospital environments. The OTELO system can be used in future hospitals, employing High Speed Downlink Packet Access (HSDPA) which is optimized for shared data and WiMAX technologies are going to be the research area for seamless wireless hospital environments;

i) Application of WiMAX and HSDPA Technologies in OTELO Environment

UMTS does have one huge advantage in that it is already deployed. There are operators, manufacturers, regulators and most importantly there is a customer base, with a huge pool of 2G users who will migrate in the future. The air interface of the UMTS, which is WCDMA has extend its specifications with HSDPA according to 3GPP Release 5. HSDPA adds a new transport channel, the high-speed downlink shared channel (HS-DSCH), which is optimized for shared data. It also provides higher-order modulation (Quadrature Amplitude Modulation or QAM), short TTI, fast link adaptation, fast scheduling, and fast hybrid (ARQ) [133]. HSDPA is already a real and demonstrable technology, and to a large extent delivers what WCDMA was always supposed to have been: a technology optimized for high speed, reliable, cost efficient data services.

WiMAX is a wireless metropolitan area network (MAN) technology that can connect IEEE 802.11 (Wi-Fi) hotspots with each other and to other parts of the Internet and provide a wireless alternative to cable and DSL for (last km) broadband access [134]. IEEE 802.16 provides up to 50 km of linear service area range and allows connectivity between users without a direct line of sight. Note that this should not be taken to mean that users 50 km away without line of sight will have connectivity. Practical limits from real world tests seem to be around 5 to 8 km. The technology has been claimed to provide shared data rates up to 70 Mbps, which, according to WiMAX proponents, is enough bandwidth to simultaneously support more than 60 businesses with T1-type connectivity and well over a thousand homes at 1Mbps DSL-level connectivity. Real world tests, however, show practical maximum data rates between 500 Kbps and 2 Mbps, depending on conditions at a given site [133].

WiMAX covers both fixed wireless access (16d) and a mobile broadband version (16e). There are many examples of how the two technologies – HSDPA and WiMAX – are entwined. For instance, WiMAX is an ideal technology for backhaul applications because it eliminates expensive leased line or fibre alternatives [133]. An HSDPA Pico-cell with wireless back connection would be very cheap and very easy to deploy, and could offer voice as well as high-speed, high-quality data, inside a corporate office or at a super hot spot.

A second example is the provision of seamless service, always using whichever is the better connected technology. For example, a passenger on a train waiting in a station would connect using WiMAX, and when the train starts moving into the countryside, there would be a smooth transition between networks to HSDPA for full high-speed mobility with handoff using the cellular network.

Figure 7.1 shows the topology represent a combination of WiMAX and Wi-Fi mesh-network topology provides the best solution for this situation. WiMAX can be used to aggregate the community centers. WiMAX extends the reach of broadband, while the proprietary Wi-Fi mesh network available today can provide mobile client access throughout the community centers and park:

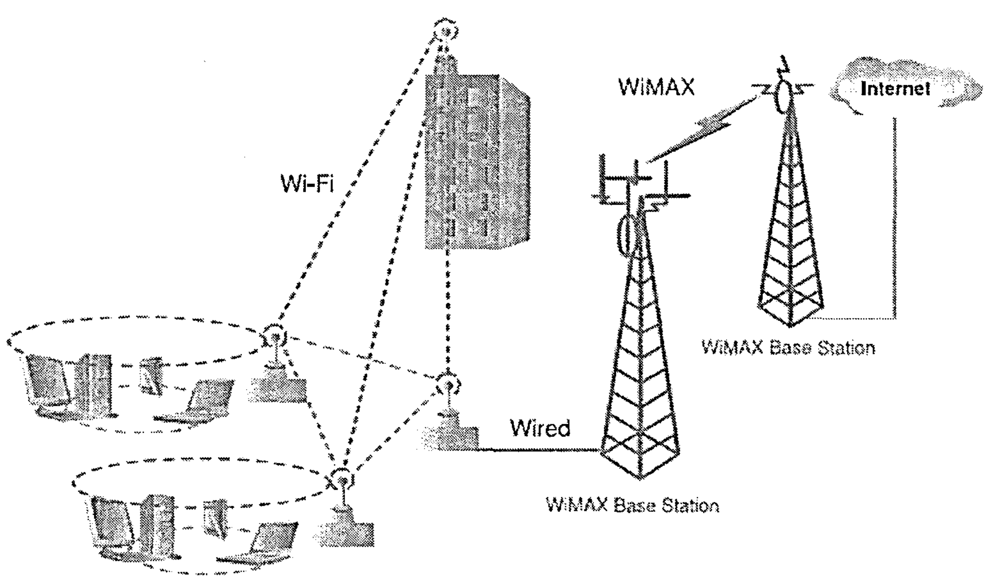


Figure 7.1 WiMAX connection between Internet and Wi-Fi mesh topology [137].

There is also interesting potential for interoperability of WiMAX with legacy cellular networks [135]. WiMAX antennas can "share" a cell tower without compromising the function of cellular arrays already in place. WiMAX antennae may be even connected to an Internet backbone via either a light fiber optics cable or a directional microwave link. Some cellular companies are evaluating WiMAX as a means of increasing bandwidth for a variety of data-intensive applications.

Although the cost-effectiveness of WiMAX in a remote application will be higher, it is definitely not limited to such applications, and may in fact be an answer to expensive urban deployments of T1 backhauls as well [137, 138]. Although, even with the upcoming mobile version of standard, WiMAX cannot be as wide area as 2G/3G, it delivers far higher rates and, with sufficiently widespread deployment, could significantly cut into the usage of cellular networks in many areas.

Quality of Service Issues:
WiMAX has a data oriented MAC compared to the essentially circuit-switched MACs of HSDPA and WCDMA. WiMAX can also take advantage of multiple duplexing modes, including TDD dynamic asymmetry; this allows the uplink/downlink bandwidth to be allocated according to current traffic conditions. HSDPA has added shared channel transmission. HSDPA is an improvement over WCDMA, since it allows TDD and FDD multiplexing, although it can only be used in the downlink transmission [134].

The new terminals that are required to take advantage of HSDPA, has to be with following specifications:

- PC-cards will be the first on the market (end of 2005)
- In the 1st phase terminals will offer: Download 3,6Mbps end user throughput, and Upload 384 kbps.
- Hand-held terminals will follow
- In a 2nd phase, peak data rates are increased to: Download 14 Mbps.

The PC-card will further robust the operation of OTELO system full mobility, since specifically the Patient station Uplink can approach (384 Kbps). The anticipation for OTELO system of full mobility deployment will take in to account the simplicity in deployment makes mobile ad hoc networks (which do not require any infrastructure) suitable for a variety of applications such as collaborative computing, disaster recovery, battle field communication. These characteristics are quite beneficial for full OTELO system mobility and adaptation to seamless network topologies.

With the proliferation of communications and computing devices such as mobile phones, laptops, or PDAs, personal area networking (PAN), which is an ad hoc networking-based technology, has recently gained much interest. The 4G networks are touted as hybrid broadband networks that integrate different network topologies and platforms. The overlapping of different network boundaries represents the integration of different types of networks in 4G. Figure 7.2 present a vision to the heterogeneous networks topology (4G based) that OTELO system can be implemented with Hospital environment:

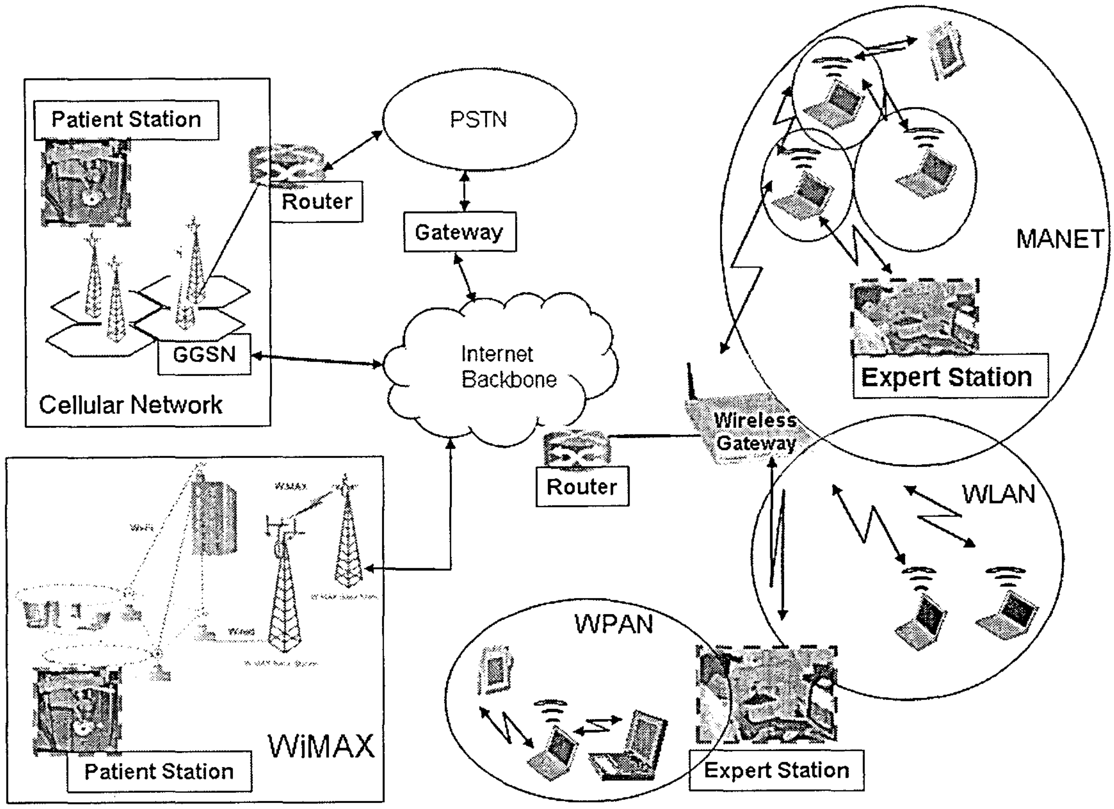


Figure 7.2 OTELO system full connectivity options over high speed data and mobile environments.

The figure propose two mobile locations for the Patient station as well as the Expert station, the dotted lines of OTELO stations within the figure represent the flexibility of the connectivity and not the physical locations. The Patient station is roaming between the cellular network and WiMAX environment, whereas the Expert station is routing within the wireless environment between the MANET, WLAN or WPAN depending on the location and the mobility of the Doctor or the expert.

There are two levels of integration;

First is the integration of heterogeneous wireless networks with varying transmission characteristics such as Wireless LAN (WLAN), WAN, PAN, as well as mobile ad hoc networks (MANET). MANET is a collection of heterogeneous communications and computing devices which can communicate with one another (within their transmission range) without any central coordination. It is essentially infrastructureless and there is no need for any fixed radio-base station or router. Such a network is self-organizing and adaptive. At the second level, the integration of wireless networks with the fixed network backbone infrastructure, the Internet, and PSTN. Much work remains to enable a seamless integration, for example that can extend IP to support mobile network devices [132].

In general, communication between a Base Station (BS) and a mobile node in a centralized architecture is performed by using a deterministic MAC protocol such as FDMA, TDMA, or CDMA. MAC protocols play an important role in the performance of the (MANETs), it is defines how each mobile unit can share the limited wireless bandwidth resource in an efficient manner [136].

ii) Optimized Ultrasound Video Streaming Issues

The adaptation technique designed and implemented in chapter six, represent a preliminary approach to a proposed optimization algorithm that can optimize the desired values of the ultrasound stream quality and the image frame rate. The objectives are to find the optimal dynamic streaming policy that satisfies the network and user constraints (QoS provisioning). We formulate the video streaming adaptation as a Markov Decision Process (MDP) to find the optimal streaming policy that can maximise the network revenue (in our case minimize the network packet loss) and guarantee the user medical QoS requirements. The proposed method is a real time direct adaptive method called Q-learning. The main advantage of this method is that it does not require a priori knowledge of the state transition probabilities (explicit state transition model) in mobile communication networks, which are difficult to estimate due to the large fluctuation in the link bandwidth of mobile networks.

iii) 3D Ultrasound Imaging Wireless Transmission

This involves using the most innovative 3D medical imaging solutions, and the integration of this technology within OTELO robotic environment. Future works involves further studies of the future 4G bandwidth and the QoS issues relevant to 3D imaging technology. This study will further enhance the issue of full Virtual Reality Ultrasonography performance that is under development.

iv) Further Clinical Validation

Future work on further clinical validation and hospital trials of OTELO services are also required using 3.5G/ 4G technologies for the potential and wider use of OTELO system in UK NHS services.

REFERENCES

[1] Tachakra, S., Wang, X. and Istepanian, R. (2003), "Mobile e-health: The Unwired Evolution of Telemedicine," *Tlemedicine Journal and e-health,* Vol. 9, No. 3, pp. 247-257.

[2] Morphing, A. A. (2000), "Telemedicine - Telecare - Telehealth - eHealth. Telemed Today", *Special issue: 2000 Buyer's Guide and Directory* (1): 43.

[3] Bauer, J.C. (2002), "Insights on telemedicine: How big is the market?" *Journal of healthcare information Management,* Vol. 16, No.2, Spring

[4] Yellowlees, P. (2005), "Global Broadband e-Health Services," *Business Briefing: Global Healthcare 2002* Issue 3, Available: <http://www.healthcare.com.au>

[5] Eysenbach, G. (2005), "What is e-Health? [editorial]. *Journal of Medical Internet Research,* Available: <http://www.jmir.org/2202/2/e20/> (Accessed: 2003, September 20).

[6] Della Mea, V. (2005), "What is e-health (2): The Death of Telemedicine?," [editorial]. *Journal of Medical Internet Research,* Available: <http://www.jmir.org/2001/2/e22/> (Accessed: 2004, Oct. 10).

[7] Kenrr, K., Dew, K. and Abernethy, D. (2002), "Telemedicine in Europe – the TELEPLANS Project", *Journal of telemedicine and Telecare,* Vol 8, No. 5, pp. 308-310

[8] Istepanian, R., Jovanov, E. and Zhang, Y. (2004), "Introduction to the Special Section on M-Health: Beyond Seamless Mobility and Global Wireless Health-Care Connectivity", *IEEE transaction on information Technology in Biomedicine,* Dec. 2004, Vol. 8, No. 4, pp. 405-414

[9] Istepanian, R., and Laxminaryan, S. (2000), "UNWIRED, the next generation of wireless and internetable telemedicine systems-editorial paper", *IEEE Transaction on Information Technology and Biomedicine*, Sept. 2000, Vol. 4, pp. 189–194.

[10] Bangert, D., Doktor, R., and Warren, J. (1999), "Introducing Telemedicine as a Strategic Intent," *in Proceedings of the 32nd Hawaii International Con. on System Sciences,* 1999. pp. 4034

[11] Ackerman, M., and Ferrante, F. (2002), "Telemedicine Technology," *Telemed journal & e-health,* Vol. 8, No.1, pp. 72-78.

[12] Ingenerf, J. (1999), "Telemedicine and Terminology: Different Needs of Context Information," *IEEE Transaction on Information Technology in Biomedicine,* June. 1999, Vol. 3, No. 2, pp. 92-100

[13] Horsch, A., and Balbach, T. (1999), "Telemedical information systems," *IEEE (TITB), Special Issue,* Sep. Vol. 3, No. 3, 166-75

[14] Krupinski, E., Nypaver, M., Sapci, H., Ellis, D., Safwat, R., and Sapci, H. (2002), "Clinical application in Telemedicine/Telehealth," *Telemedicine Journal &E-health*, Vol. 8, No. 1, pp. 13-34

[15] Thomas, D. F. (1973), "Telepsychiatry: Psychiatric consultation by interactive television," *Am Psychiatry,* Vol. 130, pp. 865-869.

[16] Krupinski, E. A., Lesueuer, B., Ellsworth, L., and Levine, N. (1999), "Diagnostic accuracy and image quality using a digital camera for teledermatology," *Telemedical Journal and e-health,* Vol. 5, pp. 257-263.

[17] Hermandez, A., Mora, F., Villegas, G., Passariello, G., and Carrault, G. (2001), "Real-time ECG Transmission Via Internet for Nonclinical Application," *IEEE transaction on information Technology in Biomedicine,* Sep. 2001, Vol. 5, No. 3, pp. 253-257.

[18] Liesenfeld, B., Hohner, E., Piehlmeier, W., and Kluthe, S. (2000), "A telemedical approach to the screening of diabetic retinopathy: Digital fundus photography," *Diabetes Care,* Vol. 23, pp. 345-348.

[19] Malone, F. D., Athanassiou, A., Nores, J., and Dalton, M. E. (1998), "Effect of ISDN bandwidth on image quality for telemedicine transmission of obstetric ultrasonography," *Telemedical Journal and e-health,* Vol. 4, pp. 161-165.

[20] Michael A. and Mandil, S. (2002), "Telemedicine Technology, Chapter 6," *Telemedicine Journal & E-health,* Vol. 8, No. 1.

[21] Yokokohji, Y., Ogawa, A., Hasunuma, H., and Yoshikawa, T. (1993), "Operation Modes for Cooperating with Autonomous Functions in Intelligent Teleoperation Systems," *Proc. IEEE International Conference on Robotics and Automation,* pp.510 - 515.

[22] Hill, J., and Jensen, J. (1998), "Telepresence Technology in Medicine: Principle & Applications, " *Proceedings of the IEEE,* March, Vol. 86, No. 3, pp. 569-580.

[23] Allen, D., Jones, G., and Bowerson, J. (1997), "Telesurgry, Telepresence, Telementoring, Telerobotics," *Telemedicine Today,* May, pp. 18-19.

[24] Courreges, F., Vieyres, P., Istepanian, R., Arbeille, P., and Bru, C. (2005), "Clinical trials and Evaluation of a mobile robotic tele-ultrasound System," *Journal of Telemedicine and Telecare,* Vol. 11, Supp. 1, pp. 46-49

[25] Hussain, B. (1998), "Tele-ultrasound a preliminary report on a BT-supported project," *RAD Magazine,* June 1998, pp. 41-42

[26] Li, X., Hu, G., and Gao, S. (1999), "Design and implementation of a novel compression method in a tele-ultrasound system," *IEEE Transactions on Information Technology in Biomedicine,* Sep. 1999, Vol. 3, No. 3, pp. 205-213

[27] Kontaxakis, G., Walter, S., and Sakas, G. (2005), "Mobile tele-echography systems: TeleInViVo - a case study," In: *M-Health: Emerging Mobile Health Systems,* Istepanian, R., Laxminarayan, S., and Pattichis, C., (editors), Kluwer Academic/Plenum Publishers, 2005.

[28] Boctor, E. M., Fischer, G., Choti, M. A., Fichtinger, G. and Taylor, R. H. (2004), "A Dual-Armed Robotic System for Intraoperative Ultrasound Guided Hepatic Ablative Therapy:A Prospective Study," *Proceedings of the 2004 IEEE International Conference on Robotics & Automation,* pp. 377-382

[29] Brebner, J., Ruddick, H., Brebner, E., Gilbert, F., Madeani, R., Ritchie, L., Smith, P., and Thompson, A. (1999), "Low-bandwidth tele-ultrasound," *Journal of Telemedicine and Telecare,* Vol. 5, Supp. 1, pp. 75-76.

[30] Chan, F., Taylor, A., Soong, B., Martin, B., Clark, J., Timothy, P., Lee-Tannock, A., Begg, L., Cincotta, R., and Wootton, R. (2002), "Randomized comparison of the quality of realtime fetal ultrasound images transmitted by ISDN and by IP videoconferencing," *Journal of Telemedicine and Telecare,* Vol. 8, pp. 91-96.

[31] Wooton, R., Dornan, J., Harper, A., Barry, C., Kyle, P., Smith, P., and Yates, R. (1997), "The effect of transmission bandwidth on diagnostic accuracy in remote fetal ultrasound scanning," *Journal of Telemedicine and Telecare,* Vol. 3, pp. 209-214.

[32] Istepanian, R., Pattichis, C. S., Kyriacou, E., and Voskarides, S. (2002), "Wireless Telemedicine Systems: an overview," *IEEE Antenna's Propagation Magazine,* April, Vol. 44, No. 2, pp. 143-153

[33] Butner, S., and Ghodoussi, M. (2001), "A Real-Time System for Tele-surgery," In: IEEE Computer Society, (eds) *Proceedings of the The 21st International Conference on Distributed Computing Systems*, Washington DC, USA, pp. 236

[34] Istepanian, R., Courreges, F., and Vieyres, P. (2004), "Advances in Robotic Tele-echography Services – The OTELO System," *The 26th Annual International Conference of the IEEE Engineering in Medicine and Biology Society,* San Francisco / USA, September

[35] Bove, A. (2004), "Mobile Tele-echography using an ultra-light Robot," *International Conference on e-health in Developing Countries [e-HDC 2004],* Rome, Italy, 31st May-1st June

[36] Pattichis, C. S. Kyriacou, E., Voskarides, S., Pattichis, M. S., Istepanian, R., and Schizas, C. N. (2002), "Wireless telemedicine systems: An overview," *IEEE Antennas Propagat. Mag.*, April, Vol. 44, No. 2, pp. 143–153.

[37] Istepanian, R., Laxminarayan, S., and Pattichis, C. [eds] (2005), "M-Health: Emerging Mobile Health Systems," *Kluwer / Plenum*, New York

[38] Istepanian, R., and Wang, H. (2003), "Telemedicine in UK," In: L. Beolchi, (eds) *European Telemedicine Glossary of Concepts, Standards Technologies and Users, 5th ed*, European Commission— Information Society Directorate-General, Brussels, Belgium, pp. 1159–1165

[39] Istepanian R., and Lacal, J. (2003), "M-Health systems: Future directions," *Proc. 25th Annu. Int. Conf. IEEE Engineering Medicine and Biology*, Cancun, Mexico, September, pp. 17–21

[40] Scalvini, S., Zanelli, E., Gritti, M., Pollina, R., Giordano, A., and Glisenti, F. (2000), "Appropriateness of Referral to the Emergency Department Through a Telecardiology Service, Boario Home-Care Researchers," *Ital Heart J.* Suppl. 7, July, pp. 905-9

[41] Masuda, K., Kimura, E., Ishihara, K., Tateishi, N., and Suzuki, Y. (2002), "Robotic Tele-diagnosis System of Echography & Wireless Experiment for Mobile Telemedicine," In: *Proceeding of the second joint EMBS/BMES Conf.* Houston, TX USA, October, pp. 23-26

[42] Oboe, R. (2001), "Web-Interfaced, Forced-Reflecting Teleoperation Systems," *IEEE Transactions on Industrial Electronics,* Vol. 48, No. 6, December, pp. 1257-1265

[43] Zivanovic, A., and Davies, B. (2000), "A Robotic Sytem for Blood Sampling," *IEEE Transactions on Information Technology in Biomedicine,* Vol. 4, No. 1, March, pp. 8-14

[44] Popescu, V., Burdea, G., Bouzit, M., and Hentz, V. (2000), "A Virtual-Reality – Based Telerehabilitation System with Force Feedback," *IEEE transaction on information Technology in Biomedicin,* Vol 4, No 1, March, pp. 45-51

[45] Yao, W., Istepanian, R.S.H., Salem, Z., Zisimopoulo, H., and Gosset, P. (2003), "Throughput Performance of a WCDMA System Suporting Tele-Echography Services," *Proceedings of the 4th. International IEEE_EMBS Special Topic Conference on Information Technology Applications in Biomedicine*, Birmingham, UK, April 24-26, pp. 310-313

[46] Al Bassit, L., Vieyres, p., and Toure, Y. (2003), "Modelling and optimisation of a 6DOF robot for tele-operation," *Journées Doctorales et Nationales d'Automatique,* Valenciennes / France, 30 June – 03 July

[47] Garawi, S., Istepanian, R., Wertheim, D., Vieyres, P., and Geake, T. (2004), "A Wireless Tele-echography Robotic System using a GPRS Link," *Medicon & Health Telematics,* Italy, 31 July – 5 August

[48] Gilabert, V. M., Courreges, F., Delgorge, C., Novales, C., Poisson, G, Vieyres, P. and Bru, C. (2004), "Clinical Validation of Tele-operated Mobile Ultrasound Scanner using a Light Weight Robot, OTELO Project," *European Congress of Radiology,* Vienna, Austria, March 5-9

[49] Collyda, C., Kyratzi, S., Triantafyllidis, G.A., Boulgouris, N.V., and Strintzis, M.G. (2002), "The Graphical User Interface for the Otelo Tele-Echography System," *SCI 2002*, Orlando, USA, July 14-18

[50] Smith-Guérin, N., Al Bassit, L., Courrèges, F., Poisson, G., Delgorge, C., Arbeille, P., and Vieyres, P. (2003), "Clinical validation of a mobile patient-expert tele-echography system using ISDN lines," *4th annual IEEE-EMBS Special topic Conference on information technology applications in biomedicine ITAB 2003*, Birmingham, UK, April 24-26

[51] Al-fahoum, A. S. and Reza A. M. (2001), "Combined Edge Crispiness and Statistical Differencing for Deblocking JPEG Compressed Images," *IEEE Transaction on Image Processing,* Vol. 10, No. 9, September, pp. 1288-1298

[52] Delgorge, C., Vieyres, P., Poisson, G., Rosenberger, C., and Arbeille P. (2001), "Comparative Survey of Ultrasound Images Compression methods Dedicated to a Tele-echography robotic System," *IEEE Engineering in Medicine and Biology Society conference EMBS 2001*, Istanbul, Turkey, October

[53] Taferner, M., and Bonek, E. (2002) *Wireless Internet Access over GSM and UMTS*, New York, Springer-Verlag

[54] Springer, A., and Weigel, R. (2002) *UMTS, the physical layer of the Universal Mobile Telecommunication Systems.* New York, Springer-Verlag

[55] Zelmer, D. (2001), "Improving the performance of GSM & TDMA Wireless by Packet Capabilities," *SUPERCOMM 2001GPRS, EDGE, & GERAN*, Atlanta, Georgia, June 6th

[56] Garawi, S., Courreges, F., Istepanian, R., Gosset, P., and Zizimopoulos, H. (2004), "Performance Analysis of a Compact Robotic Tele-Echography E-Health System over Terrestrial and Mobile Communication links," *IEE Fifth International Conference on 3G Mobile Communication Technologies,* London, UK, Oct. 18-20, pp. 118-122

[57] EBIT, ELSACOM, and UTOURS. (2002), "OTELO Project presentation," *Italian Society of Radiology (SMAU salutec)*, March 25

[58] ELSACOM. (2002), "Presentation: Telecommunication issues for OTELO," *3GSM World Congress*, Cannes, France, February

[59] Courreges, F., Vieyres, P. and Istepanian, R.S.H. (2004), "Advances in Robotic TeleEchography Services. The OTELO System," *Proceedings of the 26th Annual International Conference of the IEEE-EMBS*, San Francisco, CA, USA, September 1-5, pp. 5371- 5374

[60] Dogan, S., Cellatoglu, A., Uyguroglu, M., Sadika, A. H. and Kondoz, A. M. (2002), "Error-Resilient Video Transcoding for Robust Internetwork Comunications Using GPRS," *IEEE Transactions on Circuits and Systems for Video Technology*, Vol. 12, No. 6, June, pp. 453-464

[61] Muratore, F. (2001) *UMTS Mobile Communications for the Future.* Chichester, U.K. John Wiley & Sons Ltd

[62] Ojanpera, T. and Prasad, R. (2001) *WCDMA; Towards IP Mobility and Mobile Internet.* Boston, USA, Artech House

[63] Lin, Y. B., Rao, H. C. H. and Chlamtac, I. (2001), "General Packet Radio Service (GPRS): architecture, interface and deployment," *Wireless comunication & Mobile computing,* Vol. 1, pp. 77-92

[64] TR 101 186. (1998), "Digital cellular telecommunications system (phase 2+); GPRS Requirements specification of GPRS," *ETSI; European Telecommunications Standards Institute* V6.0.0, GSM 01.60

[65] Graja, H. and Perry, P. (2004), "An Analysis of Small IP Packet Delay in Cellular Networks," *IEE Fifth International Conference on 3G Mobile Communication Technologies,* London, UK, Oct. 18-20, pp. 347-351

[66] Holma, H., Toskala, A. (2000) *WCDMA for UMTS, Radio Access for Third Generation Mobile Communications.* Chichester, U.K. John Wiley & Sons Ltd

[67] Nicopolitidis, P., Obaidat, M.S., Papadimitrios, G. I. and Pomportsis, A. S. (2003) *Wireless Networks.* Chichester, U.K. John Wiley & Sons Ltd

[68] Chakravorty, R. Cartwright, J and Pratt, I. (1999), "Practical Experience with HTTP and TCP over GPRS," Proceedings of *IEEE Wireless Communications and Networking Conference,* WCNC, pp. 1248-1252

[69] Meyer, M. (1999), "TCP Performance over GPRS," *Proc. IEEE WCNC'99*, pp. 1248-1252

[70] Orvis J. (2002), "GPRS Calls for new QoS testing," Wireless Europe wireless.iop.org, 31-32, Nov.

[71] Halonen, T., Romero, J. Melero, J. (eds.) (2003) *GSM, GPRS and EDGE Performance.* Chichester, U.K. John Wiley & Sons Inc.

[72] Wu, D., Hou, Y. T., Zhu, W., Zhang, Y. Q. and Peha, J. M. (2001), "Streaming Video over the Internet: Approaches and Directions," *IEEE Transactions on circuits and systems for Video Technologies,* Vol. 11, No. 3, March, pp. 282-300

[73] Yang, F., Zhang, Q., Zhu, W., and Zhang, Y. (2004), "End-to-End TCP-Friendly Streaming Protocol and Bit Allocation for Scalable Video Over Wireless Internet," *IEEE Journal on Selected Areas in Communications*, Vol. 22, No. 4, May, pp. 777-790

[74] Richardson, I. E. G. (2003) *H264 and MPWG-4 Video Compression Video coding for next generation Multimedia*, Chichester, U.K. John Wiley & Sons Ltd

[75] Sun, M. T. and Reibman, A. R. (2001) *Compressed Video over Networks*, New York. Marcel Dekker Inc.

[76] Shen, Q. (2003), "Performance of VoIP over GPRS," Advanced Information Networking and Applications, AINA 2003. 17th International Conference, March 27-29, pp. 611 - 614

[77] Chu, Y. C. and Ganz, A. (2004), "A Mobile Teletrauma System using 3G Networks," *IEEE Transaction on Information Technology in Biomedicine,* Vol. 8, No. 4, Dec. pp. 456-462

[78] Korhenen, J., Olli, A., Gurtov, A. and Laamanen, H. (2001), "Measured performance of GSM HSCSD and GPRS," *In Proceedings of the IEEE International Conference on Communications*, Helsinki, Finland, June

[79] Dafonte, C., Gomez, A. and Arcay, B. (2002), "Intelligent agents technology applied to tasks control in ICU telesupervision," *Proceeding of the second joint EMBS/BMES Conference,* Huston, TX USA, October 23-26

[80] Huber, A. J. and Huber, J. F. (2002) *UMTS and mobile Computing*, London, U.K. Artech House

[81] ETSI TS 131 102. (1999), "Characteristics of the USIM Application," *Technical Specification, 3GPP TS 31.102,* Version 3.7.0 Release 1999

[82] Gállego, J. R., Hernández-Solana, A., Canales, M., Lafuente, J., Valdovinos, A and Fernández-Navajas, J. (2005), "Performance Analysis of Multiplexed Medical Data Transmission for Mobile Emergency Care over the UMTS Channel," *IEEE Transaction on Information Technology in Biomedicine,* Vol. 9, No. 1, March, pp. 13-22

[83] 3GPP TR 25.853. (2001),"Delay Budget within the Access Stratum," *Third Generation Partnership Group (TSG)* V4.0.0, Release 4 .0.0

[84] 3G TR 101 631, (1997), "Technical performance objectives," *GSM 03.05* Version 6.0.0 Release 1997

[85] Featherstone, W. and Willington, J. P. (2004), "TCP Performance Over UMTS Rate Controlled DCHs," *IEE Fifth International Conference on 3G Mobile Communication Technologies,* London, UK, Oct. 18-20, pp. 168-172

[86] Huang, L., Horn, U., Hartung, F. and Kampmann, M. (2002), "Proxy-based TCP-friendly streaming over mobile networks," *WoWMoM'02,* Atlanta, Georgia, USA, September, PP. 17-24

[87] Xylomenos, G., Polyzos, G., Saaranen, M. and Mahonene, P. (2001), "TCP Performance issues over Wireless Links," *IEEE Communications Magazine,* Vol. 39, No. 4, pp. 52-58

[88] Chan, M. C., and Ramjee, R. (2002), "TCP/IP Performance over 3G Wireless Links with Rate and Delay Variation," *MOBICOM'02,* Atlanta, Georgia, USA, September

[89] Etoh, M and Yoshimura, T. (2005), "Wireless Video Applications in 3G and Beyond," *IEEE Wireless Communications*, Volume 12, No. 4, August, pp. 66-72

[90] Conklin, G. J., Greenbaum, G. S., Lillevold, K. O., Lippman, A. F. and Reznik, Y. A. (2001),"Video Coding for Streaming Media Delivery on the Internet," *IEEE Transactions on Circuit and Systems for Video Technology,* Vol. 11, No. 3, March

[91] Garawi, S., Istepanian, R.S.H. and Abu-Rgheff, A. (2005), "3G Wireless Communications for Mobile Robotic Tele-Ultrasonography systems," *IEEE Communication Magazine, Special Issue*, November, [submitted 2005]

[92] Sallent, O., Romero, J.P., Agusti, R., and Casadevall, F. (2003), "Provisioning Multimedia Wireless Network for Better QoS: RRM Strategies for 3G W-CDMA," *IEEE Communication Magazine,* Vol. 41, Feb. 2003

[93] Mutter, A., Necker, M. C., Luck, S. (2004), "IP-Packet Service Time Distributions in UMTS Radio Access Networks," *EUNICE 2004*, Tampere, Finland

[94] Canton, A. F.and Chahed, T. (2001), "End-to-end Reliability in UMTS: TCP over ARQ," *Globecom,* San Antonio, November

[95] 3GPP TS 23.060, (2001), "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; General Packet Radio Service (GPRS); Service description," *Stage 2, V 4.1.0,* Release 4

[96] Oboe, R. (2001), "Web-Interfaced, Forced-Reflecting Teleoperation Systems," *IEEE Transaction on industrial Electronics*, Vol. 48, No. 6, December, pp. 1257-1265

[97] Chan, M. and Zakhor, A. (2005), "Rate Control for Streaming Video over Wireless," *IEEE Wireless Communications*, pp. 32-41

[98] Conklin, G. J., Greenbaum, G. S., Lillevold, K. O., Lippman, A. F. and Reznik, Y. A. (2001), "Video Coding for Streaming Media Delivery on the Internet," *IEEE Transactions on Circuit and Systems for Video Technology*, Vol. 11, No. 3, March, pp. 269-281

[99] Rejaie, R., Handley, M and Estrin, D. (1999), "An end-to-end rate-based congestion control mechanism for real time streams in the internet," *Proc. of the IEEE INFOCOM*, March. 1999, Vol. 3, pp. 1337-1345

[100] Dafonte, C. and Gomez, A. (2002), "Intelligent agents technology applied to tasks control in ICU telesupervision," *IEEE Proceeding of the second joint EMBS/BMES Conference, Huston,* TX USA, October 23-26

[101] Halonen, T., Romero, J. and Melero, J. (2003) *GSM, GPRS and EDGE Performance,* John Wiley & Sons, Ltd, USA

[102] Orvis, J. (2002), "GPRS Calls for new QoS testing," *Wireless Europe wireless.iop.org*, Nov.

[103] Sze, H. P., Liew, S. C., Lee, J. Y. B. and Yip, D. C. S. (2002), "A multiplexing scheme for H.323 voice-over-IP applications," IEEE Journal on Selected Areas in Communications, Vol. 20, No. 7, pp. 1360 - 1368

[104] 3GPP Technical Specification 23.107. (2003), "Quality of Service (Qos) concept and architecture," V5.9.0, (2003-06)

[105] Stalling, W. (2003) *Data Communications and Networks.* New York, Prentice Hall

[106] Wanigasekara, N.R., Zeng, Y., Ding, S. and Yan, Z. (2002), "Quality Evaluation For JPEG 2000based Medical Image Compression," *IEEE proceedings of the second joint EMBS/BMES Conference*, Houston, USA, Oct 23-26, pp 1019 -1020

[107] Zeng, W. and Wen J. (2002), "3G Wireless Multimedia: technologies and practical isues," Wireless Communications and Mobile, Wirel. Commun. Mob. Comput, Vol. 2, pp. 563-572

[108]Wang, Z., Bovik, A. C. (2002), "A Universal Image Quality Index," *IEEE Signal Processing letters,* Vol. xx, No .Y, March

[109] Howard, P. G., Vitter, J.S. (1992), "New methods for losseless image compression using arithmetic coding," *inform. Proc. Manage,* Vol. 28, pp. 765-779

[110] Feng, D., Cai, W. and Fulton, R. (2002), "Dynamic Image Data Compression in the Spatial and Temporal Domains: Clinical Issues and Assessment," *IEEE Transaction on Information Technology in Biomedicine,* Vol 6, No 4, PP 262-268

[111] Lawson, S. and Zhu, J. (2002), "Image Compression Using Wavelets & JPEG2000: A tutorial," *Electronic & Communication Engineering Journal,* June

[112] Cabral, J., Linker, D. and Kim, Y. (2000), "Ultrasound Telemedicine system supporting compression of pre-scan-converted data," *In Medical Imaging 2000: Image Display and Visualization, Seong K. Mun, Editor, Proceeding of SPIE* Vol, 3976.

[113] Cañero, C., Thomos, N., Triantafyllidis, G. A., Litos, G. C. and Strintzis, M. G. (2005), "Mobile Tele-Echography: User Interface Design," *IEEE Transaction on Information Technology in Biomedicine*, Vol, 9, No. 1, March, pp. 44-49

[114] Delgorge, C., Courrèges, F., Al Bassit, l., Novales, C., Rosenberger, C., Smith-Guerin, N., Brù, C., Gilabert, R., Vannoni, M., Poisson, P and Vieyres, P. (2005), "A Tele-Operated Mobile Ultrasound Scanner Using a Light-Weight Robot," *IEEE Transaction on information Technology in Biomedicine,* Vol, 9, No. 1, March, pp. 50-58

[115] Chiu, E., Vaisey, J., Stella Atkins, M. (2001), "Wavelet-Based Space-Frequency Compression of Ultrasound Images," *IEEE Transaction on Information Technology in Biomedicine*, Vol 5, No 4, Dec, PP 300-310

[116] Cote, G., Gallant, M. and Kossenitini, F. (1999), "Semi-fixed length motion vector coding for H.263-based low bit rate video compression," *IEEE Transactions Image Process,* Oct. pp.1451-1455.

[117] Steinbach, E., Farber, N. and Girod, B. (1997), "Standard compatible extension of H.263 for robust video transmission in mobile environments," *IEEE Trans Circuits Sys video Technology,* Dec. pp. 872-881

[118] Reyes, G. L., Reibman,A. R., Chang, S. F. and Chuang, J C. I. (2000), "Error-Resilient Transcoding for Video over Wireless Channels," IEEE Journal on Selected Areas in Communications, Vol. 18, No. 6, June, pp. 1063-1074

[119] Rijkse, K. (1996), "H.263: Video Coding for Low-bit Rate Communication," *IEEE Cominunications Magazine*, Dec. pp. 42-45

[120] Kodikara, C., Worrall, S., Sadika, A. and Kondoz, A. M. (2004), "Performance Enhancement of Real-Time Video Communications over UTRAN by Adaptation Resource Allocation: System Level Simulation," *IEE Fifth International Conference on 3G Mobile Communication Technologies,* London, UK, Oct. 18-20, 427-431

[121] Hsu, C. Y., Ortega, A. and Khansari, M (1999), "Rate Control for robust video transmission over wireless channels," *IEEE Journal of Selected Areas in Communications*, May, pp.756-773

[122] Tai, H. M., Long, M., He, W., Yang, H. (2002), "An efficient region of interest coding for Medical Image Compression," *IEEE proceedings of the second joint EMBS/BMES Conference*, Houston, USA, Oct. 23-26

[123] Penedo, M. Pearlman, W. A. Tahoces, P. G. Souto, M. and Vidal, J. J. (2003), "Region-based wavelet coding methods for digital mammography,", *IEEE Transactions on Medical Imaging,* Vol. 22, No. 10, pp. 128-1296

[124] Saxe D. M., Foulds, R. A. (2002), "Robust Region of Interest Coding for Improved Sign Language Telecommunication," *IEEE Transactions on information Technology in Biomedicine*, Vol. 6, No. 4, Dec. pp. 310-316

[125] Nishantha, D., Hayashida, Y. and Hayashi, T. (2004), "Application Level Rate Adaptive Motion-JPEG Transmission for Medical Collaboration Systems," *IEEE Proceedings of the 24th International Conference on Distributed Computing Systems Workshops (ICDCSW'04),* March 23-24, pp. 64-69

[126] Zhang, R., Regunathan, S. L., Rose, K. (2000), "Video Coding with Optimal Inter/Intra-Mode Switching for Packet Loss Resilience," *IEEE Journal on Selected Areas on Communications,* Vol. 18, No. 6, June

[127] Bolot, J. C., Turletti, T. (1998), "Experience with control mechanism for packet video in the Internet," *Computer Communications*, January, Rev. 28(1), pp. 4-15

[128] Cheung, G., Tan, W. and Yoshimura, T. (2005), "Real –Time Video Transport optimization using Streaming Agent over 3G Wireless Networks," *IEEE Transaction on Multimedia*, Vol. 7, No. 4, August

[129] Videoconferencing Cookbook, Version 4.1, (2005) *Motion JPEG (MJPEG),* [online]. Available at http://www.videnet.gatech.edu/cookbook.en [Accessed 11 Nov. 2005]

[130] Rijkse, K. (1996), "H.263: Video Coding for Low-bit Rate Communication," *IEEE Cominunications Magazine*, Dec., pp. 42-45

[131] N.I. LABVIEW <http://www.ni.com/aap/> [Accessed 09 Dec. 2005]

[132] Issariyakul, T., Hossain, H. and Kim, D.I. (2003), "Medium access control protocols for wireless mobile ad hoc networks: issues and approaches," *Wireless Communications and Mobile Computing*, Vol. 3, pp.935–958

[133] Baines, R. (2005), "The Roadmap to Mobile WiMAX," *IEE Communications Engineer*, Aug-Sep.

[134] Intel, Broadband Wireless Access. (2004) *Understanding WiMAX and 3G for Portable/Mobile Broadband Wireless.* A Technical Overview and Comparison of WiMAX and 3G Technologies

[135] Agis, E., Mitchel, H., Ovadia, S., Aissi, S., Bakshi, S., Iyer, P., Kibria, M. Rogers, C. and Tsai, J. (2004), "Global, Interoperable Broadband Wireless Networks: Extending WiMAX Technology to Mobility," *Intel Technology Journal,* Vol. 8, Issue 3, August

[136] Parkall, S., Englund, E., Helmersson, K. W., Samuelsson, M., Peisa, J., Torsner, J. Malm, P., Edvardson, M. and Edholm, C. (2004), "WCDMA Evolved – High-Speed Data Access," *IEE Fifth International Conference on 3G Mobile Communication Technologies,* London, UK, Oct. 18-20, pp. 212-216

[137] Intel, Wi-Fi and WiMAX Solutions (2003) Understanding Wi-Fi and WiMAX as Metro-Access Solutions. Intel Corp., *IEEE 802.16 and WiMAX:* Broadband Wireless Access for Everyone

[138] Ghosh, A., Wolter, D. R., Andrews, J. G. and Chen, R. (2005), "Broadband Wireless Access with WiMax/8O2.16: Current Performance Benchmarks and Future Potential Gabriel," *IEEE Communication Magazine,* February, pp. 129-136

[139] Hu, J. H., Feng, G. and Yeung, K. L. (2003), "Hierarchical Cache Design for Enhancing TCP Over Heterogeneous Networks With Wired and Wireless Links," *IEEE Transaction on Wireless Communications*, Vol. 2, No. 2, March, pp. 205-217

[140] Korhenon, J. (2001) *Introduction to 3G Mobile Communications.* Boston, USA, Artech House

APPENDIX A

Cloud Network Simulator

Introduction

Wide area networks such as the Internet and complex Intranets are characterized by the variety of network infrastructures they support. Making sure that communications products function well and have reasonable response times in such a varied environment is a difficult task. Parameters such as physical distance, time of the day, connection speed, and the quality of the Internet services can greatly affect the speed and efficiency of data transfer. By emulating a WAN, SHUNRA\Cloud reduces much of the pain and expense associated with the testing of different network configurations.

SHUNRA\Cloud emulates point-to-point WAN in laboratory conditions. The software is used to introduce network parameters such as bandwidth, latency, and packet loss that characterize WANs, to a LAN environment. A patent-pending tool, SHUNRA\CLOUDCatcher, can be used to record real WAN data, which can be incorporated into SHUNRA\Cloud for emulation purposes.

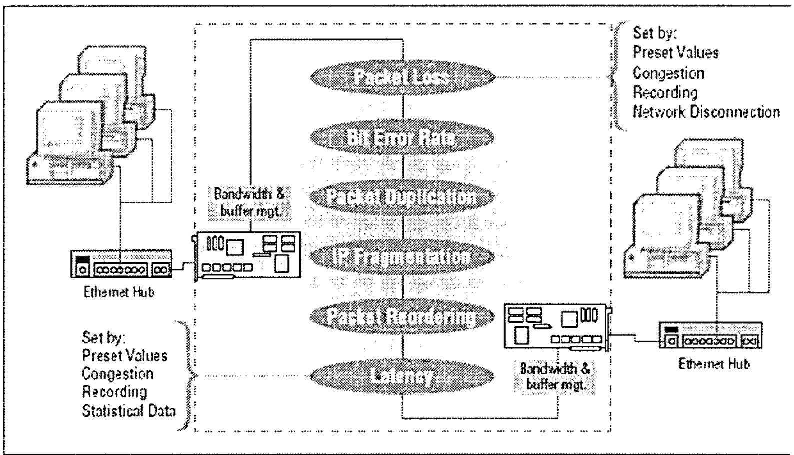
For the academic tester SHUNRA\Cloud also offers standard mathematical algorithms and the ability to import any model from a simple text file.

Once the emulated WAN is defined, you can activate your emulation and exchange data through various applications such as Voice over IP, as you would do on the real WAN. You can then observe and analyze the effects of the different WAN settings on the quality and stability of the applications or products that you are testing.

The CLOUD™ WAN SIMULATOR

MAIN FEATURES:

- Simple yet powerful setup gives the user full control over the emulated WAN pipe
- Real time, real connection performance playback through the simulator
- Stand alone, real time, real connection performance recording module included
- Uses Standard Ethernet Card/s for LAN interface/s
- Create custom latency profiles and packet loss setting as simple text files
- Scripting support through a simple command line interface
- Software runs on Microsoft® Windows® NT 4.0 and Windows® 2000
- Low host CPU usage
- Basic and advanced modes
- Includes Simple Network Analyzer for real time statistics



EMULATION FEATURES:

LATENCY

Internet Recording & Playback—Record and Playback mechanism used to accurately emulate Internet Connections.
Custom Profile—Create custom latency profiles and packet loss settings as simple text files.
Fixed Latency—Any value between 0 and 8,000 miliseconds.
Uniform Distributed Latency—Randomly changing latency between a set Minimum and Maximum. Also supports a limit on the maximum change of the latency.
Normal Distributed Latency—Randomly changing latency according to a given Average and Standard deviation.
Linear Dynamic Latency—Latency ramps between defined values and repeats.

PACKET LOSS

Periodic Loss—Cycle a lost packet every so many packets.
Random Loss—Each packet has the same chance of being lost according to a predefined probability.
Burst Loss—A stream of packets is lost according to a predefined probability. The burst size (in packets) is selected randomly from a range defined by the user. Granularity down to 0.01%.
Gilbert-Elliot Model—Alternating between good and bad scenarios.

PACKET EFFECTS

Out Of Order—Specify the chance of a packet to change its location within the packet stream. The offset of the packet from its original position is selected randomly from a user defined range.
Duplicate Packets—Specify the chance of a packet to be duplicated. The number of times a packet is duplicated is selected randomly from a user defined range.
IP Fragmentation—Specify the chance of a packet to be fragmented due to a short Maximum Transmission Unit (MTU). The Cloud can ignore the Do not Fragment (DF) bit on incoming packets, to allow for easier reproduction of fragmentation problems.

LINK FAULTS

Bit Error Rate—Allows for emulation of physical line errors. The user can set the frequency of the error and the range of the bit stream length to toggle. Granularity down to 1 in 10^{-12}.
Network Disconnection—Emulates relatively long times of network downtime. The user can set the average frequency o the disconnection events and the range o the disconnection time.

CONGESTION

Allows the user to define network congestion events. The user can set the average frequency of the event, its time span and the fixed latency and packet loss during the congestion event.

GATEWAY BANDWIDTH LIMITATION

- Symmetric or Asymmetric Bandwidth limitation.
- Any bandwidth from 2.4 kbps up to 10 Mbps.
- Insertion delay can be adjusted to support different physical medium (PPP, etc.).

continued on other side

GATEWAY BUCKET (BUFFER) EMULATION

- Provides emulation of router buffers for accurate emulation.
- Limited and unlimited bucket size.
- Byte mode and packet mode.
- Drop tail bucket management.
- Random Early Detection (RED) bucket management.

HOST SYSTEM REQUIREMENTS:

- Microsoft Windows NT 4.0 Service Pack 3 or higher and Windows 2000.
- Pentium 233MHz or higher.
- 64 Mbytes RAM or more.
- 5 Mbytes free disk space.
- One or more Ethernet adapters.

APPENDIX B

Vodafone 2.5G Terminals and 2.5G/3G PCMCIA Card Specifications

Sony Ericsson T68i

Info | User Reviews | Shop | Forum

This update to Ericsson's ground-breaking T68 model adds full MMS capability, including a photo album that can store up to 200 photos, and compatibility with a camera accessory. Otherwise similar. Notable features include small size, color display, world roaming, GPRS high-speed data, MMS, EMS, Bluetooth, voice dialing, full organizer features, and extensive customization options.

Offered By:

AT&T Wireless (Cingular) Discontinued
Cingular Discontinued
T-Mobile Discontinued

BUY this phone

BUY compatible stuff

Specifications

Compare vs...

Modes	GSM 900 / GSM 1800 / GSM 1900
Weight	2.96 oz (84 g)
Dimensions	3.94" x 1.89" x 0.79" (100 x 48 x 20 mm)
Form Factor	Bar Internal Antenna
Battery Life	Talk: 7.50 hours (450 minutes) Standby: 384 hours (16 days)
Battery Type	LiPolymer 700 mAh
Display	Type: LCD (Color STN) Colors: 256 (8-bit) Size: 101 x 80 pixels
Platform / OS	(N/A)

Memory	400 KB (built-in, flash shared memory)
Phone Book Capacity	510 plus SIM card memory
FCC ID	PY71130202 (Approved Mar 6, 2002)

Features

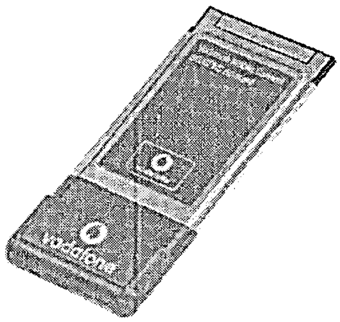
Show Missing Features

Alarm	works even when phone is off
Bluetooth	Yes
Calculator	Yes
Calendar	Yes
Custom Graphics	built-in picture editor / wallpaper, screen savers, MMS, picture ID, and photo album / supports GIF, JPEG, WBMP / exchange via IR, Bluetooth, MMS, or e-mail
Custom Ringtones	built-in melody composer / exchange via WAP, Bluetooth, IR, or PC cable
Data-Capable	Yes
Email Client	Protocols Supported: POP3, IMAP, SMTP
EMS / Picture Messaging	Yes
Games	Number of games: 7 Arimona, ContraryErix, Four Piles, Naval Fleet, North Territory, Q, Yukon Struggle
High-Speed Data	Technology: GPRS also HSCSD
Infrared (IR)	Yes
MMS	Yes
Multiple Languages	Yes
Multiple Numbers per Name	Yes

Vodafone - Mobile Connnect 3G Datacard review

- Hardware reviews
- > Networking hardware reviews
- > Vodafone Mobile Connnect 3G Datacard

fast Internet access for laptop users (11/05/2004)



After the years of controversy, nay-saying and all round pessimism that have dogged the development of 3G mobile phone services, it still seems a little surprising that the technology ever made it into actual, real live products. Nevertheless, the third generation mobile networks are finally up and running, albeit with limited coverage, and early adopters all over the UK are learning that the novelty of live video chat wears off a lot quicker than you'd think.

Mobile phone behemoth Vodafone has launched its first 3G product into the IT market. Rather than offering a 3G mobile handset the company has released a datacard that allows laptop users to access the Internet via the new high-speed network. The premise is simple: plug this PCMCIA card into your laptop and you'll get broadband Internet access whenever you're in range of Vodafone's 3G network. Before we get into the details, let's answer the most important question: yes, it works, and it works pretty damned well.

We tested the card in several locations around London and, even though the signal strength varied considerably between areas, the card provided a decent connection. In areas where the reception was at full strength we achieved download speeds of around 380kbps. This is not quite as fast as a typical cable or DSL connection (512kbps) but plenty fast enough for the kind of applications mobile workers are likely to use and certainly a massive improvement over the GPRS connection mobile Internet users have been stuck with thus far.

When you move out of 3G range the card switches over to a GPRS connection, which offers speeds similar to a standard dial-up modem. We didn't expect it to work as smoothly as it did, but we have to admit that the transition between the two connections is fairly seamless nearly all of the time. Vodafone's 3G network is currently restricted to major cities but is expanding fairly rapidly, and you can get a GPRS connection just about anywhere in the country. A coverage map is available on the company's Web site.

Installation is a doddle. We set the card up on a Compaq Armada E500 running Windows 98SE (the minimum Windows version supported) and had a live connection within ten minutes. The card is managed via Vodafone's Mobile Connect dashboard, a simple application that tells you everything you need to know about your connection and even allows you to send SMS messages from your PC.

Our main gripe about the card is that in order to boost network performance, the system performs extra compression on nearly all Web page images. This includes buttons and logos as well as photographs, and there's no option to turn it off. While it's not usually that noticeable, the over-compression can sometimes make images look bad and if the quality of images in web pages is very important to you then we recommend that you try the card out in a Vodafone store before parting with cash.

The only other negative that's worth mentioning is the large antenna that comes with the card. It's about five inches long and needs to be positioned vertically for optimal performance. That said, when attached to the top of your laptop's screen it's bound to score some bonus gadget-points from other commuters when you're checking your e-mail on the train into work.

Vodafone - Mobile Connnect 3G Datacard features - Verdict

This could be the start of a revolution in mobile Internet access, adding a fast new option to the existing wireless networking solutions available to laptop users. If and when the 3G coverage improves, datacards such as this are likely to become a staple component of the mobile worker's inventory.

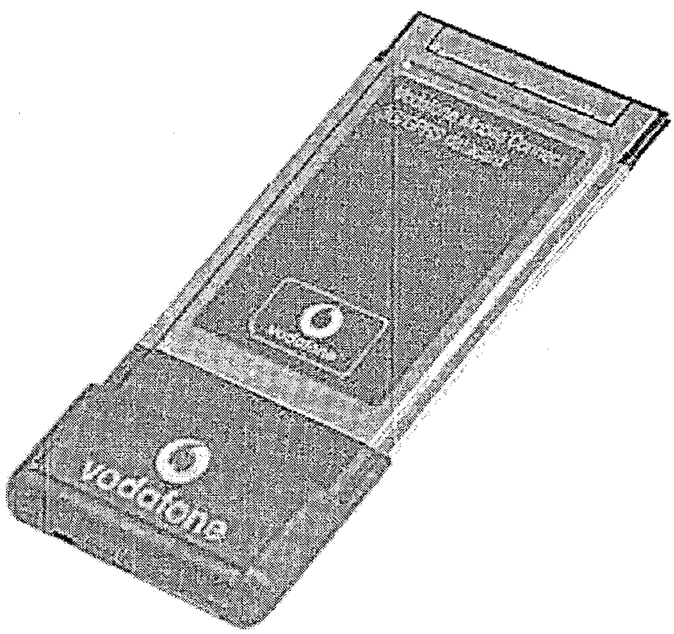
Vodafone - Mobile Connnect 3G Datacard price

Buy Vodafone Mobile Connnect 3G Datacard securely online at a bargain price

£100 - £200 inc. VAT depending on contract. Contracts range between £11.75 and £99.99 inc. VAT per month.

Vodafone: 08700 700 191

www.vodafone.co.uk



APPENDIX C

Network Monitoring Software (Ethereal / Netmon)

What is `Ethereal`?

Every network manager at some time or other needs a tool that can capture packets off the network and analyze them. In the past, such tools were either very expensive propietary, or both. However, with the advent of Ethereal, all that has changed.

`Ethereal` is perhaps one the best open source packet sniffers available today. The following are some of the features Ethereal provides:

- Available for UNIX and Windows.
- Capture and display packets from any interface on a UNIX system.
- Display packets captured under a number of other capture programs:
  - tcpdump
  - Network Associates Sniffer and Sniffer Pro
  - NetXray
  - LANalyzer
  - Shomiti
  - AIX's iptrace
  - RADCOM's WAN/LAN Analyzer
  - Lucent/Ascend access products
  - HP-UX's nettl
  - Toshiba's ISDN routers
  - ISDN4BSD *i4btrace* utility
  - Microsoft Network Monitor
  - Sun snoop
- Save captures to a number of formats:
  - libpcap (tcpdump)
  - Sun snoop
  - Microsoft Network Monitor
  - Network Associates Sniffer
- Filter packets on many criteria.

- Search for packets using filters.
- Colorize packet display based on filters

However, to really appreciate its power, you have to start using it.

Figure 1-1 shows `Ethereal` having captured some packets and waiting for you to examine the packets.

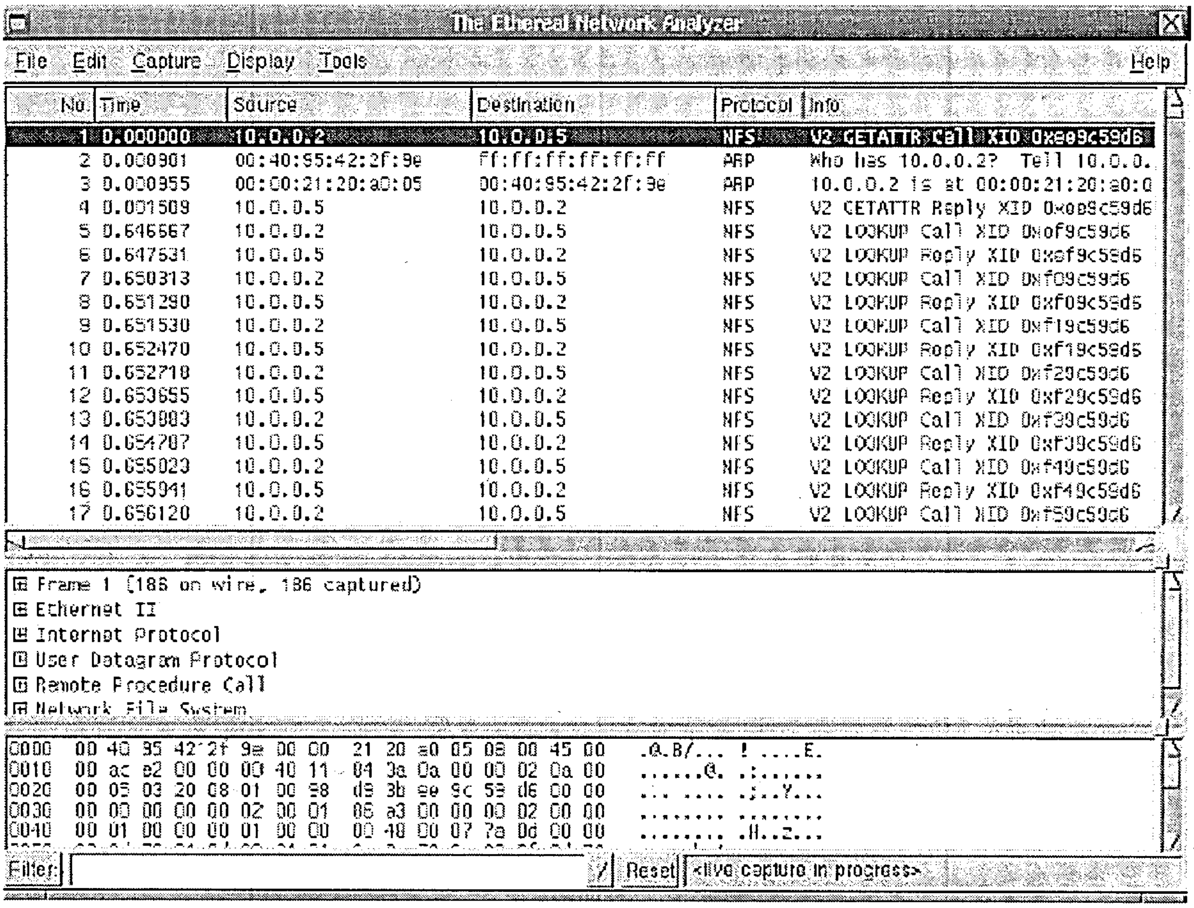


Figure 1-1. `Ethereal` **captures packets and allows you to examine their content.**

In addition, because all the source code for Ethereal is freely available, it is very easy for people to add new protocols to Ethereal, either as modules, or built into the source.

There are currently protocol decoders (or dissectors, as they are known in Ethereal), In addition, you can find a PDF version of the guide at: http://www.ns.aus.com/ethereal/user-guide/user-guide-a4.pdf [11] in A4 and http://www.ns.aus.com/ethereal/user-guide/user-guide-usletter.pdf [12] in US Letter.

Providing feedback

Should you have any feedback about this document, please send them to the author at rsharpe@ns.aus.com[13].

Notes

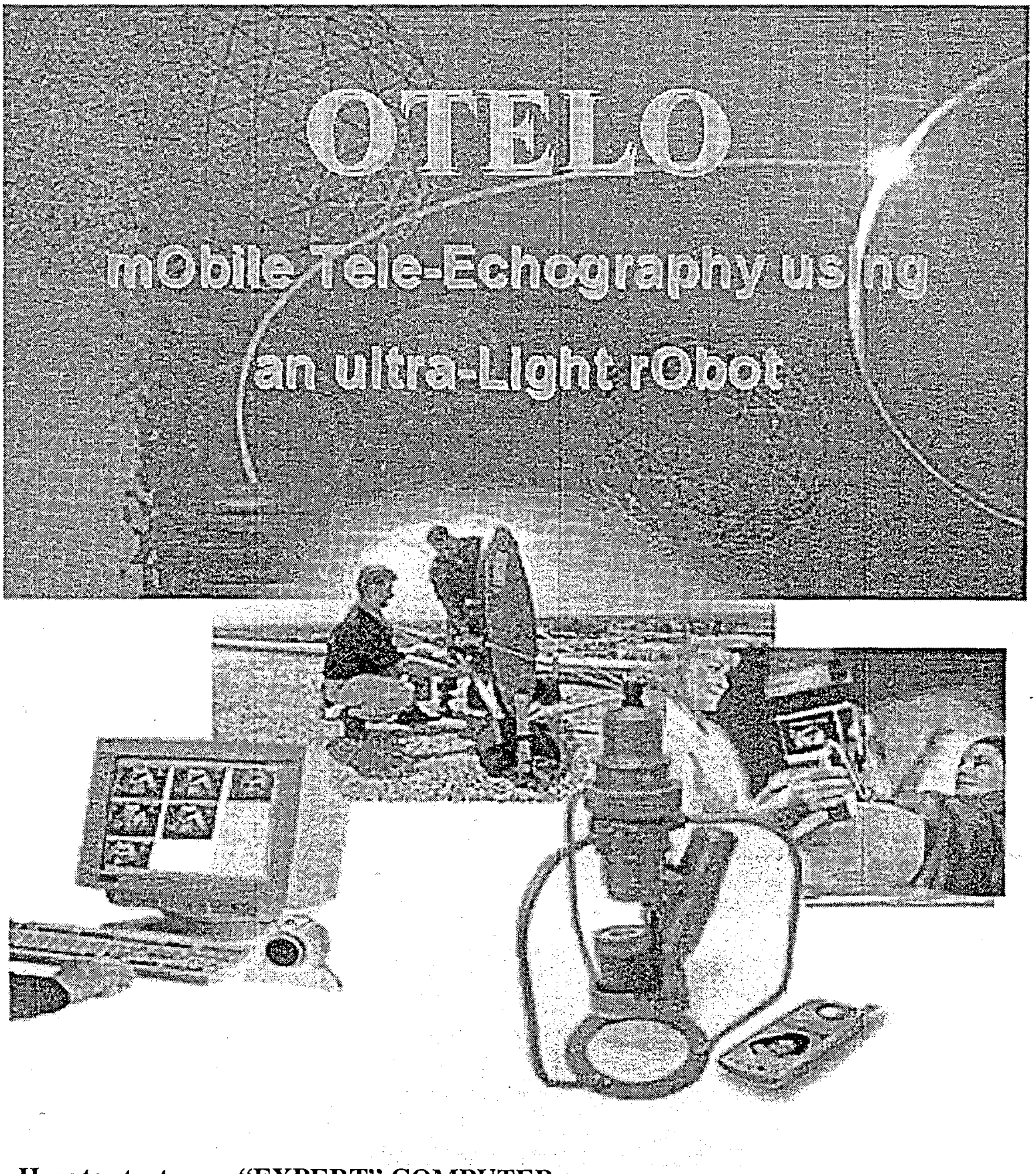
1. mailto:rsharpe@ns.aus.com
2. mailto:hagbard@physics.rutgers.edu
3. http://www.gnu.org/copyleft/gpl.html
4. http://www.ethereal.com
5. http://www.ethereal.com/introduction.html#authors
6. http://www.ethereal.com
7. http://www.ethereal.com
8. http://www.ethereal.com
9. http://www.ns.aus.com/ethereal/user-guide/book1.html
10. http://www.ethereal.com
11. http://www.ns.aus.com/ethereal/user-guide/user-guide-a4.pdf
12. http://www.ns.aus.com/ethereal/user-guide/user-guide-usletter.pdf
13. mailto:rsharpe@ns.aus.com

APPENDIX D

OTELO Brochure and Patient / Expert GUI Programme

OTELO1 « HOW-TO » BOOK

Version Otelo1_1.1

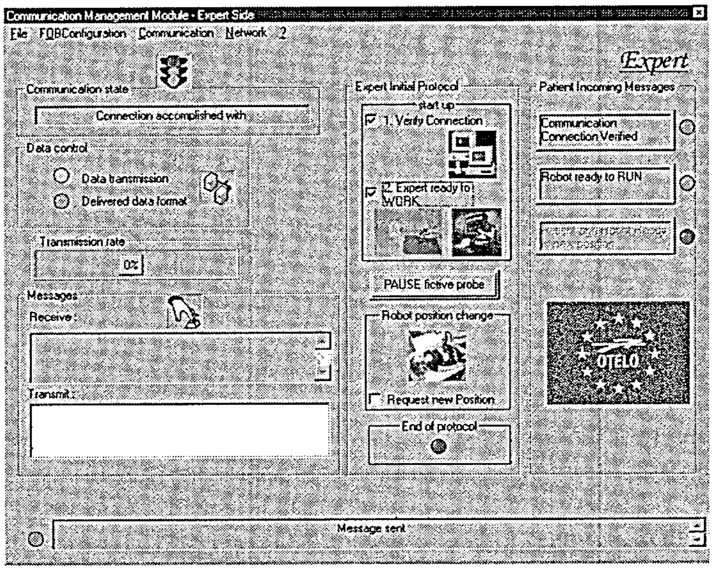


How to start your "EXPERT" COMPUTER

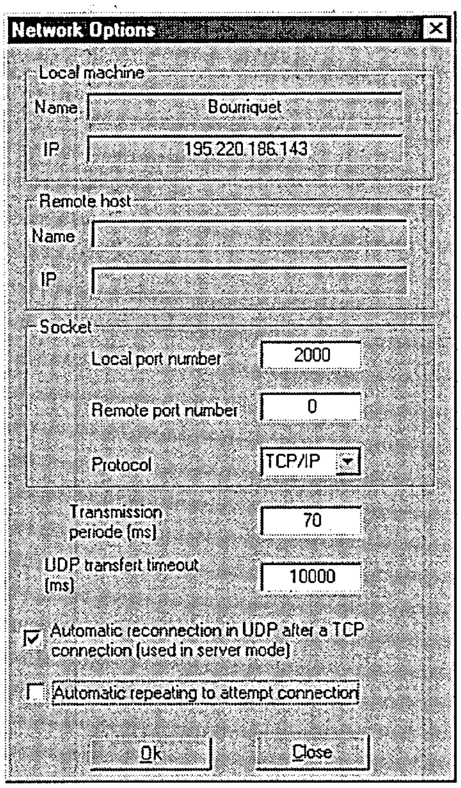
Turn the system ON
Your USERNAME ("*Utilisateur*") is : Administrateur
Type in your password ("*mot de passe*"): " OTELO" (respect CAPITAL lettering, avoid space),
Click on OK

How to start your TELE-ECHOGRAPHY

1. Double click on the EXPERT-OTELO ICON (the window below appears), wait for the assistant IP request



➢ Go To NETWORK menu to find your IP number when requested by your Assistant



➢ Tell your IP number to your ASSISTANT at the patient station using the video-conference

➢ Wait for the PATIENT STATION to get connected with you : all LED on the right are GREY

- Tick the first box to verify the Connection link and wait for the PATIENT answer : Green LED and message box "Communication connection verify" appears. The com link with the PATIENT is now OPEN
- Tick second check box " Expert ready to work " . This will start the fictive probe and FOB sensor initialisation :
    - Turn the FOB system on **FLY**
    - Initialise the FOB sensor by pressing INITIALIZE
    - Wait for the "FOB INITIALIZED" message
    - Hold the ficitve probe vertically and on the "X" mark located at the centre of the mouse pad.
    - The wire of the FOB device attached on the ficitive porbe has to go out on your left
    - Go to calibration for XY and Z sequentially by pressing on (**at least twice**) :
        - CALIBRATION XY (do not move the fictive probe)
        - CALIBRATION Z (do not move the fictive probe)
    - Then PRESS OK
- Wait for the PATIENT STATION answer : "Robot ready to work " green LED
- You can now start tele-echography by CLICKING on the " PAUSE FICTIVE PROBE" button

At any time during the examination, you can check if the fictive probe is correctly initialised (calibration XY and Z). For this action, go to MENU
FOB configuration
Configuration then proceed with the calibration actions describe above.
This action can be done when for example the robot is being relocate to another position on the patient's skin

How to terminate your TELE-ECHOGRAPHY

Inform your assistant with the video-conference you want to terminate
Go to the menu bar: select COMMUNICATION, and then Select DISCONNECT.
Go to the menu Bar: select FILE, end then Select QUIT
On the FOB device, switch from **FLY** to **STBY**

How to turn your COMPUTER OFF

PRESS simultaneously CTL+ALT+SUPPR
Select "Arrêter le système" (*Stop the system*)
Tick "Arrêter le système"(*Stop the system*) and then OK
Your computer system is now turned OFF / Turn the Screen OFF

Your tele-echography Act is now finished;

you can now turn the videoconference OFF.

IF YOU ARE THE ASSISTANT, please follow these instructions:

How to start your "PATIENT" COMPUTER

Part I

✍ If you are using the OTELO tele-echography system for **the very first time**, please follow step 1 to 7 on the **SET_UP_for OTELO1 BOCHURE, then go to Part II**

✍ If your robotic system **is already connected**, CONNECT the batteries CHARGER to the main 220V outlet
Then apply step 3, step 5, step 6 and step 7 of the **SET_UP_for OTELO1 BOCHURE, then go to Part II**

Part II

PRESS simultaneously CTL+ALT+SUPPR
The username (*Nom d'Utilisateur*) is Administrateur, DO NOT change it
Input your password (*Mot de passe*) : "OTELO"
Click OK

How to MAKE A CALL with the expert

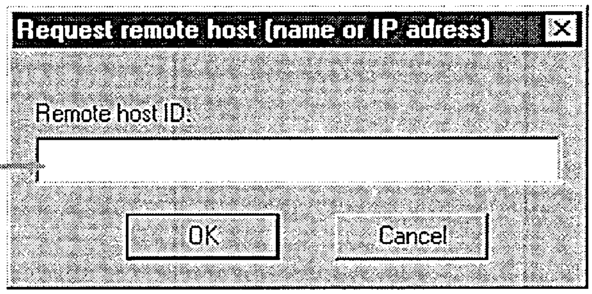
Go to "DEMARRER" (to start)
Select "OTELO I"
Click on "Call expert with ISDN"
Network remote access menu pops Up
Select your EXPERT
Click on COMPOSER ("DIAL")
A new window pops up:
The username (*Nom d'Utilisateur*) is Administrateur, DO NOT change it
Input your password (*Mot de passe*) : "OTELO"
Leave the *domaine* field blank.
Do not tick the "*Enregistrer le mot de passe*" (password auto-save)
Click OK

The Expert number is dialed
A question window appears click on "*Accepter*" (Accept)
Another question Windows appears, simply click on OK

All windows disappear, the ISDN link is established with EXPERT

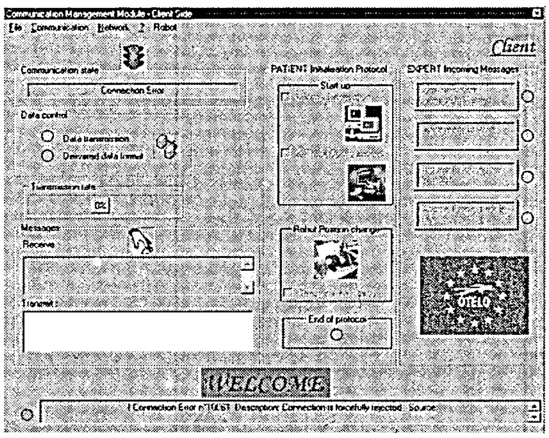
How to RUN OTELO1

Go to "DEMARRER"
Select "OTELO I"
Click on "OTELO I run" (the window below appears)



Wait until the " REQUEST REMOTE HOST" dialog box pops up, Ask the Expert for his IP number (to be found on **his expert window** under Network menu)

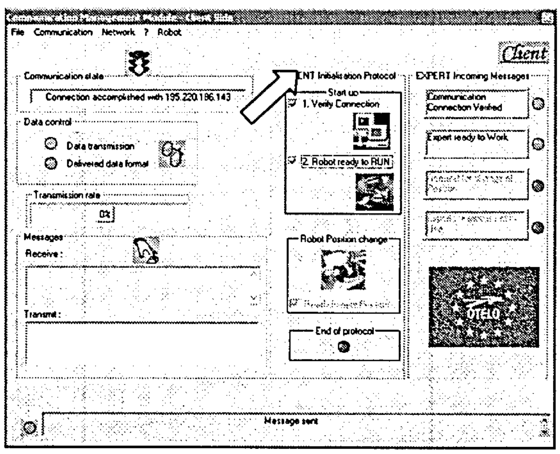
Enter the IP number and Click on "OK" (the window below appears)



- PATIENT STATION interface pops up : The communication link with the expert is now OPEN

- Tick "1. Verify Connection" (see white arrow), in the PATIENT initialisation protocol - all LED turn RED This is to verify the Connection link

- Wait for the EXPERT answer : the LED turns Green and a message box "Communication connection verify" pops up



How to get the ROBOT READY for the tele-echography

Set robot in vertical position, make sure there is no obstruction for its movements
Tick the Second check Box "Control of the Robot Now authorised"
The initialisation box of the Robot pops up
PRESS OK to Initialise the robot positions
Wait until the message "INITIALISATION FINISHED" appears on the bottom banner
POSITION the robot on the anatomical location requested by the EXPERT

You are now ready to start the tele-echography - Hold the robot firmly –
Keep alert for any Expert request – Be ready to remove the robot OFF the patient's body

How to change the ROBOT POSITION on the patient

The request has to be stated by a window poping up on the Expert command.
You can only move the robot when this request is received
When robot is on a new anatomical area, Tick the check box "Robot in new position"
Hold the Robot Firmly

How to UNLOCK the screen saver or the computer at the patient station

PRESS any Key : the OTELO screen should go off , a "lock" message appear
To unlock (*"Deverrouiller"*) , PRESS CTRL+ALY+SUPPR
Input you password ("mot de passe") : **OTELO** (respect CAPITAL lettering, avoid space),
PRESS OK

How to stop the ROBOT in case of EMERGENCY

PRESS on the **RED EMERGENCY STOP Button** on the OTELO POWER UNIT

How to turn your ROBOT equipment OFF

Go to Menu, choose File, click on QUIT
PRESS on the **RED EMERGENCY STOP Button to stop the ROBOT**

How to turn your COMPUTER OFF

PRESS simultaneously CTL+ALT+SUPPR
Select "Arrêter le système" (*Stop the system*)
Tick "Arrêter le système"(*Stop the system*) and then OK
Your computer system is now turned OFF
PRESS on the **GREEN ON/OFF** button to stop the whole system
PRESS on the **RED EMERGENCY STOP Button**

Author's Publications

1. **Garawi, S.**, Istepanian, R.S.H. and Abu-Rgheff, A. (2006), "3G Wireless Communications for Mobile Robotic Tele-Ultrasonography systems," *IEEE Communication Magazine, Special Issue*, April, Vol. 44, No. 4, pp. 91-96

2. Istepanian, R., **Garawi, S.** Philip, N. and Courreges, F. (2005), "3G Traffic Analysis of Wireless Tele-Echography Robotic System," MECHCI Conference, Dubai, April 9-10

3. **Garawi, S.**, Courreges, F., Istepanian, R., Gosset, P., and Zizimopoulos, H. (2004), "Performance Analysis of a Compact Robotic Tele-Echography E-Health System over Terrestrial and Mobile Communication links", *IEE Fifth International Conference on 3G Mobile Communication Technologies*, London, UK, Oct. 18-20, pp. 118-122

4. Courreges, F., Istepanian, R.S.H. and **Garawi, S.** (2004), "Wireless Challenges for Mobile Robotic Tele-Echography System," *Proceedings of the 2nd International Conference on Smart home and health Telematics-ICOST04*, Sept 2004, Singapore, In ' Toward a Human –Friendly Assistive Environment, Zhang, D. and Mokhtari M. (Eds.), IOS Press, pp.15-22

5. **Garawi, S.**, Istepanian, R., Wertheim, D., Vieyres, P., and Geake, T. (2004), "A Wireless Tele-echography Robotic System using a GPRS Link", *Medicon & Health Telematics 2004*, Italy, 31 July – 5 August 2004.

6. Yao, W., Istepanian, R.S.H., **Salem, Z.**, Zisimopoulo, H., and Gosset, P. (2003), "Throughput Performance of a WCDMA System Suporting Tele-Echography Services," *Proceedings of the 4th. International IEEE_EMBS Special Topic Conference on Information Technology Applications in Biomedicine*, Birmingham, UK, April 24-26, pp. 310-313

Other Publications:

1. E-health-Insider. (2004) *Tele-echograpgy robot allows remote patient scans*, [Available at www.e-health-insider.com/news/], News letter [Accessed Nov. 2005]