Brunel University London

INTEROPERABILITY OF WIRELESS COMMUNICATION TECHNOLOGIES IN HYBRID NETWORKS: EVALUATION OF END-TO-END INTEROPERABILITY ISSUES AND QUALITY OF SERVICE REQUIREMENTS

A THESIS SUBMITTED FOR THE DEGREE OF

DOCTOR OF PHILOSOPHY

BY

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Abstract

Hybrid Networks employing wireless communication technologies have nowadays brought closer the vision of communication *"anywhere, any time with anyone"*. Such communication technologies consist of various standards, protocols, architectures, characteristics, models, devices, modulation and coding techniques. All these different technologies naturally may share some common characteristics, but there are also many important differences. New advances in these technologies are emerging very rapidly, with the advent of new models, characteristics, protocols and architectures. This rapid evolution imposes many challenges and issues to be addressed, and of particular importance are the interoperability issues of the following wireless technologies: Wireless Fidelity (Wi-Fi) IEEE802.11, Worldwide Interoperability for Microwave Access (WiMAX) IEEE 802.16, Single Channel per Carrier (SCPC), Digital Video Broadcasting of Satellite (DVB-S/DVB-S2), and Digital Video Broadcasting Return Channel through Satellite (DVB-RCS). Due to the differences amongst wireless technologies, these technologies do not generally interoperate easily with each other because of various interoperability and Quality of Service (QoS) issues.

The aim of this study is to assess and investigate end-to-end interoperability issues and QoS requirements, such as bandwidth, delays, jitter, latency, packet loss, throughput, TCP performance, UDP performance, unicast and multicast services and availability, on hybrid wireless communication networks (employing both satellite broadband and terrestrial wireless technologies).

The thesis provides an introduction to wireless communication technologies followed by a review of previous research studies on Hybrid Networks (both satellite and terrestrial wireless technologies, particularly Wi-Fi, WiMAX, DVB-RCS, and SCPC). Previous studies have discussed Wi-Fi, WiMAX, DVB-RCS, SCPC and 3G technologies and their standards as well as their properties and characteristics, such as operating frequency, bandwidth, data rate, basic configuration, coverage, power, interference, social issues, security problems, physical and MAC layer design and development issues. Although some previous studies provide valuable contributions to this area of research, they are limited to link layer characteristics, TCP

performance, delay, bandwidth, capacity, data rate, and throughput. None of the studies cover all aspects of end-to-end interoperability issues and QoS requirements; such as bandwidth, delay, jitter, latency, packet loss, link performance, TCP and UDP performance, unicast and multicast performance, at end-to-end level, on Hybrid wireless networks.

Interoperability issues are discussed in detail and a comparison of the different technologies and protocols was done using appropriate testing tools, assessing various performance measures including: bandwidth, delay, jitter, latency, packet loss, throughput and availability testing. The standards, protocol suite/ models and architectures for Wi-Fi, WiMAX, DVB-RCS, SCPC, alongside with different platforms and applications, are discussed and compared. Using a robust approach, which includes a new testing methodology and a generic test plan, the testing was conducted using various realistic test scenarios on real networks, comprising variable numbers and types of nodes. The data, traces, packets, and files were captured from various live scenarios and sites. The test results were analysed in order to measure and compare the characteristics of wireless technologies, devices, protocols and applications.

The motivation of this research is to study all the end-to-end interoperability issues and Quality of Service requirements for rapidly growing Hybrid Networks in a comprehensive and systematic way.

The significance of this research is that it is based on a comprehensive and systematic investigation of issues and facts, instead of hypothetical ideas/scenarios or simulations, which informed the design of a test methodology for empirical data gathering by real network testing, suitable for the measurement of hybrid network single-link or end-to-end issues using proven test tools.

This systematic investigation of the issues encompasses an extensive series of tests measuring delay, jitter, packet loss, bandwidth, throughput, availability, performance of audio and video session, multicast and unicast performance, and stress testing. This testing covers most common test scenarios in hybrid networks and gives

recommendations in achieving good end-to-end interoperability and QoS in hybrid networks.

Contributions of study include the identification of gaps in the research, a description of interoperability issues, a comparison of most common test tools, the development of a generic test plan, a new testing process and methodology, analysis and network design recommendations for end-to-end interoperability issues and QoS requirements. This covers the complete cycle of this research.

It is found that UDP is more suitable for hybrid wireless network as compared to TCP, particularly for the demanding applications considered, since TCP presents significant problems for multimedia and live traffic which requires strict QoS requirements on delay, jitter, packet loss and bandwidth. The main bottleneck for satellite communication is the delay of approximately 600 to 680 ms due to the long distance factor (and the finite speed of light) when communicating over geostationary satellites.

The delay and packet loss can be controlled using various methods, such as traffic classification, traffic prioritization, congestion control, buffer management, using delay compensator, protocol compensator, developing automatic request technique, flow scheduling, and bandwidth allocation.

Dedication

This Thesis is dedicated to my late parents, they have given me encouragement especially my mother, she brought me up to this stage with unconditional love and unlimited support. She suffered from cancer and died during my research. I could not forget those sad moments in my life.

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This research would not have been possible without the support and patience of my family, who have endured more that I should have ever asked them. Thanks go to my incredible wife Ghazala Abbasi, my sons Sarmad and Ashir Abbasi for their understanding while I was doing research. Thanks also go to my elder brother Rifat Abbasi and my sister for moral support, my nephew Basit and Tufail Abbasi and my cousin Amna Waqas.

Despite many of my activities including job and family have overlapped significantly with research but eventually by the grace of Allah I became successful to achieve this.

List of Publications

This thesis gives an account of the research undertaken solely by the author.
Some of the material contained therein has been presented as shown below:

Abbasi, M.; Stergioulas, L.K.; El-Haddadeh, R.; Kretschmer, M.; Pitsilis, V.; Tsiolis, I.; Zagkos, D.; Hatziefremidis, A.; "Interoperability and testing in broadband satellite networks". Signal Processing for Space Communications, 2008. SPSC 2008. 10th International Workshop on, pp.1-6, 6-8 Oct. 2008.

Abbasi, M.; Stergioulas, L.K.; Kretschmer, M.; Pitsilis, V.; Khalid, Z.; Khan, N.; "Heterogeneous satellite-terrestrial technologies: Quality of service and availability testing." Emerging Technologies, 2008. ICET 2008. 4th International Conference on, pp.138-145, 18-19 Oct. 2008.

Stergioulas, L.K.; Abbasi, M.; Pitsilis ,V. ; Makropoulos,C. ; Kretschmer, M.; "Satellite Enabled Education for Geographically isolated communities of farmers and maritime workers". International Conference on Digital Divide, Athens May 2008.

Abbasi, M.; Stergioulas, L.K; "Hybrid Wireless Networks for e-learning and digital literacy- testing and evaluation". International Journal of Digital Literacy and Digital Competence, paper accepted (on 12th April 2011) and in press.

Abbasi, M.; Stergioulas, L.K.; "Interoperability issues, testing tools and methodology for hybrid wireless networks". International Journal of Computer Science and Information Technology, paper accepted on 28th March 2011.

Abbasi, M.; Stergioulas, L.K.; "Studying Interoperability testing for Satellite Terrestrial Networks." International Journal of Satellite Communications and Networking, submitted in July 2010, under review.

Abbasi, M.; Stergioulas, L.K.; "Hybrid wireless communication technology: testing and evaluation for Maritime Community". Bahria Journal of communications and networks, paper submitted in April 2011, under review.

Abbasi, M.; Stergioulas, L.K; “Integration and Testing of Wi-Fi/ WiMAX with DVB-RCS”. International Journal of information and communication engineering, submitted in January 2011, under review.

Acronyms

3G PP	– 3rd Generation Partnership Project
AAA	– Authentication, Authorization, Accounting
AAL	– ATM Adaptation Layer
ACK	– Acknowledgement
ADPCM	– Adaptive Differential Pulse Code Modulation
AES	– Advanced Encryption Standard
AMPS	– Advanced Mobile Phone System
AP	– Access Point
APSK	– Asymmetric Phase Shift Keying
ARP	– Address Resolution Protocol
BE	– Best Effort
BEC	– Backward Error Correction
BER	– Bit Error Rate
BPSK	– Bipolar(Binary)Shift Keying
BS	– Base Station
BSS	– Broadcast Satellite Service
BWA	– Broadband Wireless Access
CCK	– Complimentary Code Keying
CDMA	– Code Division Multiple Access
CLIX	– Corporate Learning and Access Platform
COFDM	– Coded Orthogonal Frequency Division Multiplexing
CRA	– Continuous Rate Assignment
CRC	– Cyclic Redundancy Check
CSMA	– Carrier Sensing Multiple Access
CSMA/CA	– CSMA/ Collision Avoidance
CSMA/CD	– CSMA/ Collision Detection
DARPA	– Defense Advanced Research Projects Agency
DBS	– Direct Broadcasting Satellites
DCF	– Distributed Coordination Function
DCS	– Dynamic Channel Selection
DECT	– Digital Enhanced Cordless Telecommunications
DFS	– Dynamic Frequency Selection

DHCP	– Dynamic Host Configuration Protocol
DPSK	– Differential Phase Shift Keying
DQPSK	– Differential Quaternary Phase-Shift Keying
DSL	– Digital Subscriber Line
DSSS	– Direct Sequence Spread Spectrum
DVB-C	– Digital Video Broadcasting – Cable TV
DVB-H	– Digital Broadcasting Handheld
DVB-S/S2	– Digital Video Broadcasting –Satellite
DVB-RCS	– Digital Video Broadcasting Return Channel through Satellite
DVB-T	– Digital Video Broadcasting – Terrestrial
EAP	– Extensible Authentication Protocol
EDGE	– Enhanced Data Rates for GSM Evolution
EIRP	– Effective Isotropic Radiated Power
ETSI	– European Telecommunications Standards Institute
ERC	– European Radio communications Committee
EVDO	– Evolution-Data Optimized
FCA	– Free Capacity Assignment
FCC	– Federal Communications Commission
FDD	– Frequency Division Duplex
FDM	– Frequency Division Multiplexing
FDMA	– Frequency Division Multiple Access
FEC	– Forward Error Correction
FER	– Forward Error Rate
FH SS	– Frequency Hoping Spread Spectrum
FSS	– Fixed Satellite Service
FTP	– File Transfer Protocol
GHz	– Giga Hertz
GMSK	– Gaussian Minimum Shift Keying
GPRS	– General Packet Radio Service
GSM	– Global System for Mobile Communications
HARQ	– Hybrid Automatic Repeat Request (HARQ),
HIPERLAN	– High Performance Radio Local Area Network
HSPA	– High Speed Packet Access (HSPA)

HSPDA –High Speed Downlink Packet Access

HSUPA – High Speed Uplink Packet Access

HTTP – Hypertext Transfer Protocol

ICMP – Internet Control Message Protocol

IEEE – Institute of Electrical and Electronics Engineers

IETF – Internet Engineering Task Force

IMT-2000 – International Mobile Telecommunications-2000

IMTAdv – IMT advanced

IP – Internet Protocol

IR – Infrared

ISP – Internet Service Provider

ITU – International Telecommunications Union

Kbps – Kilo Bits per second

LAN – Local Area Network

LCMS – Learning Content Management Services

LLC – Logical Link Control

LDPC – Low-density parity-check

LS OFMD – Loosely synchronous OFDM

LTE – Long Term Evolution

MAC – Medium Access Control

MAN – Metropolitan Area Networks

MANET – Mobile Ad-hoc Networks

Mbps – Megabits per second

MFTDMA – Multifrequency Time Division Multiple Access

MHz – Mega Hertz

MPDS – Mobile Packet Data Service

MPEG –Moving Pictures Expert Group

MPLS –Multiprotocol Label Switching

MiMo – Multiple input Multiple output

MoWLAN – Mobile over WLAN

MTBF – Mean Time Between Failure

MTTF – Mean Time To Failure

MTTR – Mean Time to Repair/Recovery

NAT – Network Address Translation

NIC	– Network Interface Cards (NICs)
OFDMA	– Orthogonal Frequency Division Multiple Access
OFDM	– Orthogonal frequency-division multiplexing
PAN	– Personal Area Networks
PDA	– Personal Digital Assistant
PDU	–Protocol Data Unit
PEP	– Protocol Extension Protocol
POP	– Post Office Protocol
PHY	– Physical Layer
PSK	– Phase Shift Keying
QAM	– Quadrature Amplitude Modulation
QOE	– Quality of Experience
QOP	– Quality of Perception
QoS	– Quality of Service
QPSK	– Quadrature Phase Shift Keying
RLLS	– Return Link Sub System
RBDC	– Rate-Based Dynamic Capacity
RTP	– Real Time Protocol
RTPS	–Real Time Pooling Service
RF	– Radio Frequency
RTT	– Round Trip Time
SCFDMA	– Single-Carrier FDMA
SCPC	– Single Channel Per Carrier
SDMA	– Space Division Multiple Access
SIP	– Session Initiated Protocol
SIT	– Satellite Interactive Terminal
SNMP	– Simple Network Management Protocol
TCP	– Transmission Control Protocol
TDMA	– Time Division Multiple Access
UDP	– User Datagram Protocol
UHF	– Ultra High Frequency
UMTS	– Universal Mobile Telecommunications System
URL	– Uniform Resource Locator
UWB	– Ultra Wide Band

VBDC	– Volume-Based Dynamic Capacity
VNC	– Virtual Network Connection
VHF	– Very High Frequency
VoIP	– Voice over IP
VRRP	– Virtual Router Redundancy Protocol
VSAT	– Very Small Aperture Terminal
WAN	– Wide Area Network
WAS	– Wireless Access Systems
WCDMA	– Wideband Code Division Multiple Access
WECA	– Wireless Ethernet Compatibility Alliance
WEP	– Wireless Equivalent Privacy
Wi-Fi	– Wireless Fidelity
WiMAX	– Worldwide Interoperability for Microwave Access
Wireless HUMAN	–Wireless High speed Unlicensed Metropolitan Area Network
WiSIP	– Wi-Fi Session Initiated Protocol
W-ISPs	– Wireless Internet Service Providers
WLAN	–Wireless Local Area Network
WMAN	– Wireless Metropolitan Area Network
WOFDMA	–Wideband Orthogonal Frequency Division Multiple Access
WRED	–Weighted Random Early Detection
WTDMA	– Wideband Time Division Multiple Access

Table of Contents

List of Figures

List of Tables

Chapter 1

Introduction to the Research

1.1 Basics of Wireless Communication Technologies

Wireless communication technologies hold the potential to establish or extend network connections in a very cost effective way and provide the capability to serve new users living in sparsely populated and/or remote areas that can't be reached practically using the existing technologies infrastructure. Emerging wireless technologies are now empowering various industries to discover new business models and generate new services.

Wireless communication technologies include various devices and systems with different standards, protocols, architectures, modulation and coding techniques. Some important ones are: Global System for Mobile Communications (GSM) (Bekkers *et al.* 2002), General Packet Radio Service (GPRS) (Ghribi and Logrippo 2000), Wireless Fidelity (Wi-Fi) IEEE 802.11 (IEEE802.11 2009), Worldwide Interoperability for Microwave Access (WiMAX) IEEE 802.16 (IEEE802.16 2006). A classification of the various wireless technologies is shown in figure 1.1.

Due to rapid evolution in hybrid networks, Transmission Control Protocol (TCP), Hypertext Transfer Protocol (HTTP), User Datagram Protocol (UDP), Internet Protocol (IP) and architecture in basic OSI model are modified (as shown in figure 1.2). The additional layers are also evolving in basic OSI models, and other emerging wireless technologies.

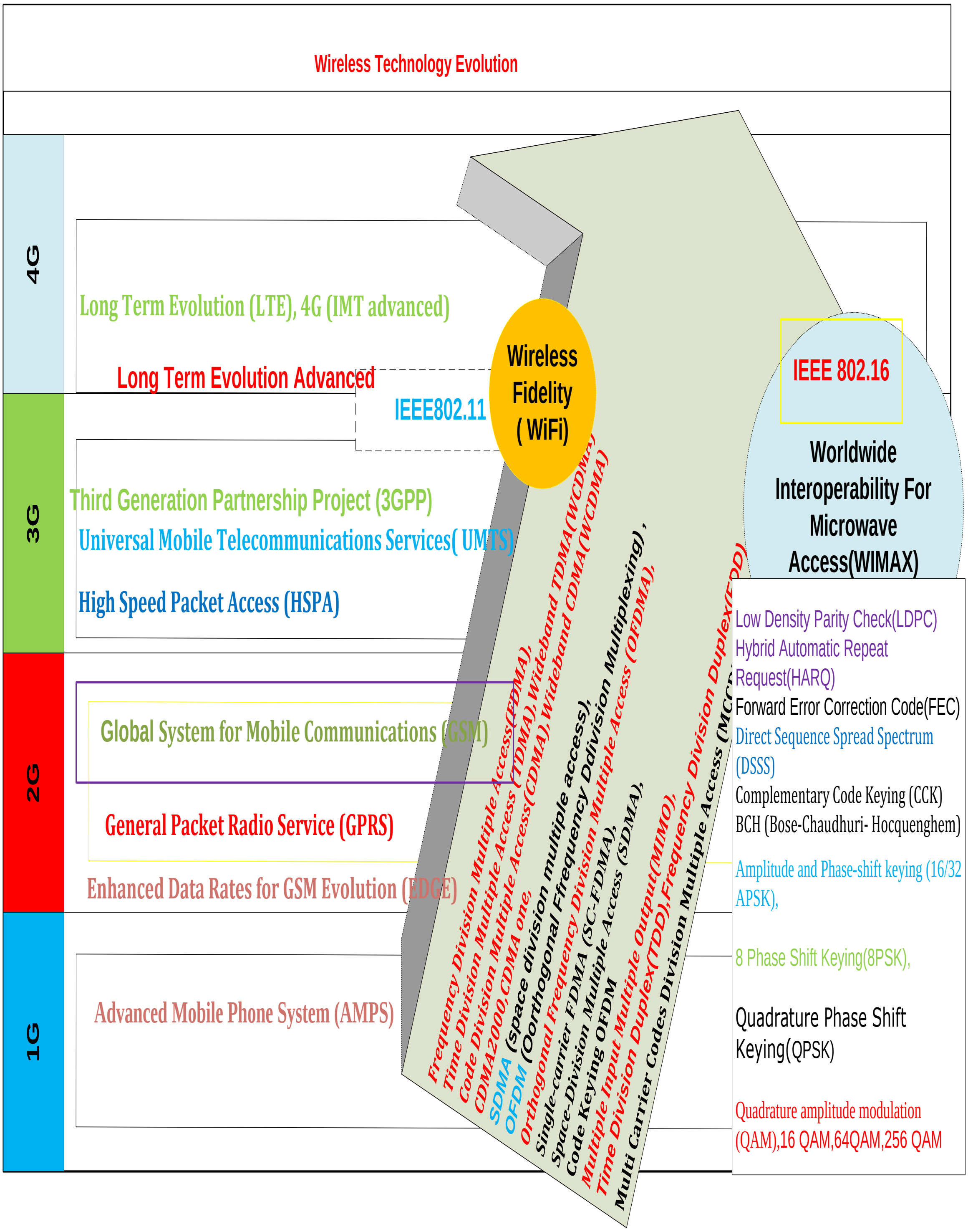


Figure 1.1: The landscape of Wireless Technology

Application	HTTP,FTP,SMTP,NCP,Telnet,VTP, FTAM
Presentation	GIF,JPEG, MPEG, ANS.1,SMB
Session	NetBios, TCP 3 way handshake
Transport	TCP, UDP,SPX
Network	X25, IP,IPX,DDP
Data Link	LLC(802.2), Ethernet(802.3), Token ring(802.5),Token bus(802.4), GARP, VLAN,CIF , ATM(802.2u),802.15,FDDI, Frame Relay,HDLC,PPP MAC, (CSMA/CD},802.1P,802.1Q
Physical	RS 232,DS1,DS3,UTP,Optical Fibre, Coaxial, SONET,X21,IEEE802.3 ,802.4,802.5

Figure 1.2: The OSI Seven Layer Architecture

As various wireless communication systems and technologies are being developed with different standards and protocols, the primary beneficiary will be the user who will get most benefit from the innovation potential that comes with interoperable solutions. So far there has been little progress in the field of end-to-end interoperability issues and Quality of Service (QoS) requirements in hybrid network. Understanding the interoperability issues and Quality of Service requirement for these technologies and systems can make the processes of selection, plan, design, development, implementation, integration and interconnection easier.

New international standards are also emerging very rapidly in wireless technology. Wireless communication systems such as blue tooth, cellular, wireless local area networks (WLAN), wireless metropolitan area networks (WMAN), are now in widespread use and have become essential in everyday life. Their popularity is extremely high due to the advantages of ubiquitous communication (i.e. anywhere, anytime with anyone).

Wireless Communication Networks are classified into four different network types according to their range: Wireless Personal Area Network (WPAN), Wireless Local Area Network (WLAN), Wireless Metropolitan Area Network (WMAN), and Wireless Wide Area Network (WWAN) (Fourty *et al.* 2005). The Wi-Fi and WiMAX are WLAN and WMAN Network technologies respectively, while Satellite is considered as WWAN. This study examines and determines end-to-end interoperability issues of wireless communication technologies in hybrid network and assess end-to–end QoS requirements in hybrid wireless networks. The classification of wireless network is shown in figure 1.3.

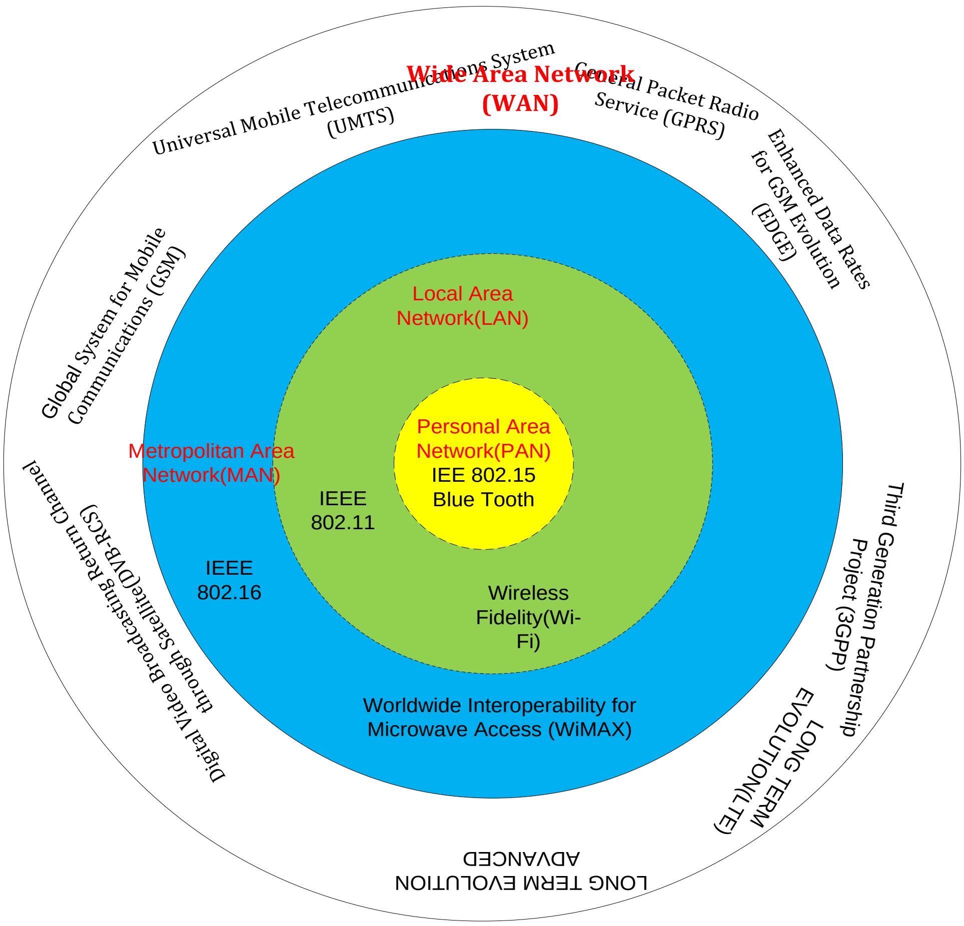


Figure 1.3: Classification of Wireless Networks

This chapter is organised as follows: Section 1 presents a brief introduction to mainstream wireless communication technologies such as Wi-Fi, WiMAX, 3GPP LTE/3GPP2, DVB-RCS, S/S2, and SCPC. Section 2 provides the rationale and motivation behind the presented research. Section 3 sets the research aim and objectives. Section 4 formulates the research questions; Section 5 outlines the research plan; Section 6 describes in more detail the thesis organisation and provides brief summaries of each chapter; finally Section 7 provides a summary of this chapter and a quick-glance overview of the thesis layout.

1.2 Research Rationale

Recent advances in wireless communication technologies have covered significant ground towards satisfying the main user requirements of high data rate, bandwidth and low latency. Based on these requirements, a key feature is that these systems and devices should communicate effectively without any issue. However, there are several standards, protocols, platform, modulation and coding techniques for wireless communications are in use, which are required to be interoperable with each other. In order to communicate effectively it is important to consider interoperability issues and Quality of Service requirements.

Interoperability issues arise for any layer of the end-to-end communication of a hybrid wireless network. These issues are needed to be investigated, examined, established and addressed. This research evaluates end-to-end interoperability issues and determines QoS requirements.

Our objective is to come up with a methodology for testing of hybrid wireless networks with combination of diverse test scenarios, nodes and network traffic. This approach can then be used to address any issue in heterogeneous – hybrid Satellite-Terrestrial - wireless networks. The motivation is to study and analyse hybrid networks which employ wireless technologies and enable access ***anywhere, anytime with anyone***.

The main contribution is to investigate and highlight the issues for needed capability and challenges for hybrid network and gives future direction and recommendations to communication industries, network providers and the research community.

1.3 Research Aim and Objectives

The aim and objectives of this research are as follows:

1.3.1 Aim

To investigate and evaluate end-to-end interoperability issues and Quality of Service requirements for wireless communication technologies in hybrid networks.

1.3.2 Objectives

1. *To study and compare technologies, protocols, standards, architectures and services used in hybrid wireless networks.*
2. *To investigate the interoperability issues and compare test tools for Hybrid Wireless Networks.*
3. *To prepare a comprehensive test plan and methodology which allows testing of all possible Hybrid Wireless Network scenarios.*
4. *To analyse results and investigate interoperability issues and QoS requirements.*
5. *To provide recommendations for best practice in the design of hybrid communication networks and directions for further research.*

1.4 Research Questions

The following research questions are addressed by this thesis:

- *What are the major issues for end-to-end interoperability and how different communication systems and technologies can be interoperable with each other?*
- *What are the categories (such as user-level or network-level) and types of QoS requirements for hybrid wireless networks?*
- *What are the most suitable testing tools to test interoperability issues in such networks?*

- *What is a suitable test plan, testing methodology for assessing end-to-end interoperability and QoS requirements of wireless communication technologies in hybrid networks (satellite and terrestrial)?*
- *Based on the results for hybrid wireless networks, what are the recommendations to address various interoperability issues?*

1.5 Research Plan

The present research was carried out in five phases:

Phase 1. In this phase, a literature review was carried out and various technologies and standards were compared with regard to interoperability issues and Quality of Service requirements for existing as well as emerging wireless communication technologies in hybrid network. The standards, protocols, data rate, bandwidth, modulation, coding techniques, models and architectures for Wi-Fi, WiMAX, and DVB-RCS were compared. Various internet-based services, applications and platforms such as Skype, MSN Messenger, NetMeeting, Clix, Isabel and collaboration tools (like Remote Desktop Publishing and Virtual Network Connection) were also compared.

Phase 2. A detailed outline of the interoperability issues and QoS requirements such as delay, jitter, packet loss, latency, throughput measurement, availability and bandwidth was prepared. Some of the important software testing tools, including Beacon/multibeacon, Ethereal/Wireshark, httperf, Iperf, Kismet, Mgen, MRTG, Netmeter, OWAMP, PRTG, Pathchar, Pathload, Pathrate, Pchar, Ping, SNMP, Tcpdump, Tcptrace, Tracemate, and Traceroute, were researched and studied. Most of software tools are open-source, flexible, easy to use and economical. A comparison of testing tools was conducted. A research plan, process and testing methodology was also prepared.

Phase 3. In this phase, a test bed was built and a testing plan was implemented and carried out. The test bed of this study was a network developed in the Broadband access satellite enabled education ($BASE^2$ EU) project, which designed and deployed a hybrid satellite broadband and terrestrial wireless-based network infrastructure, and learning services for geographically isolated communities. In particular, BASE² focused on the empowerment (enabling learning) of the agrarian and maritime geographically isolated communities. The BASE² network architecture was deployed and tested to supports the different modes of learning such as live virtual classroom, video conference, offline asynchronous learning, collaborative learning, individual learning, management and delivery over different network technologies to a large number of sites for agrarian communities in Greece and Cyprus as well as maritime communities on ships. The overall objective of this project was to implement an end-to-end system for tele-education applications. Twelve sites (10 in Greece and 2 in Cyprus) were involved with full network and service deployment. The end-to-end broadband infrastructure was an integration of broadband terrestrial networks with satellite broadband technologies using DVB-RCS, SCPC (VSAT), Wi-Fi, WiMAX, CLIX management and delivery frameworks and Isabel application. The test bed scenarios include Satellite only, SCPC (VSAT), Satellite (DVB-RCS) and WiMAX and finally Satellite (DVB-RCS), WiMAX and Wi-Fi. All testing was completed in this phase. A series of testing was carried out in order to determine the interoperability requirements for hybrid wireless network technology with the objective to examine live measurements from various scenarios by using measurement and analysis tools as elaborated and planned in Phase 2.

Phase 4. Results were analysed for various Interoperability issues and QoS requirements using a range of different tools which were studied in Phase 2. The objective of the testing was to investigate any interoperability issues and QoS requirements to ensure the hybrid network performance. This enables us to examine and determine the capabilities needed and the issues faced when striving for the seamless operation of the heterogeneous satellite-terrestrial wireless networks. Various types of traffic data, files, packets and traces were captured, and filtered.

Phase 5. Summary, discussions, findings, and recommendations were completed in this phase.

1.6 Thesis Organisation

This thesis is organised as follows:

Chapter 1 presents the research background and a basic introduction to wireless communication technologies, provides the research rationale, highlights the aim and objectives, and formulates the research questions. This chapter also provides an overview of this thesis.

Chapter 2 gives an overview of various wireless communication technologies. Compares standards, protocols, and modulation techniques for Wi-Fi, WiMAX, DVB-RCS, DVB-S/S2, 3G and LTE. This chapter also comprises literature review and research background, which was conducted in the area of wireless communication technologies in hybrid networks (satellite and terrestrial). Several studies on Wi-Fi, WiMAX, DVB-RCS, and VSAT have been reviewed, discussed and critically analysed.

Chapter 3 explains the Interoperability and QoS Requirements for Wi-Fi, WiMAX, DVB-S/S2 and DVB-RCS and describes different internet-based applications such as Skype, MSN Messenger, NetMeeting, Clix, Isabel and collaboration tools (such as Remote Desktop Publishing and Virtual Network Connection testing).

Chapter 4 focuses on interoperability issues such as bandwidth, delay, jitter, latency, packet loss, throughput and other performance issues.

This chapter also examines and describes suitable test tools required for interoperability testing of Hybrid Networks. Some of the important testing software tools are discussed and compared such as D-ITG, Ethereal/Wireshark, Iperf, Httperf, Mgen, MRTG, Netmeter, Ping, Tcpdump, Tcptrace, and Trace mate.

Chapter 5 presents the employed research methodology and the developed test plan and testing methodology for hybrid wireless networks. A testing flow diagram was created to illustrate the process of generic requirements and considerations for tests which includes purpose of test, test specifications, time estimate for test, time of any day test, number of nodes to be tested, list of suggested steps, list of issues and errors to pay attention, expected results and methodology to test various wireless technologies in hybrid wireless networks. A test plan was generated after considering various interoperability issues such as delay, jitter, bandwidth, packet loss, throughput, latency, TCP, UDP Performance, link characteristics, and availability issues. This test plan is complemented by a systematic test process and testing methodology for hybrid wireless networks to test various DVB-RCS, SCPC, WiMAX, and Wi-Fi scenarios and nodes.

Chapter 6 conducts, analyses and evaluates the results from live hybrid wireless communication network scenarios and nodes. The objective of the testing was to identify and examine any end-to-end interoperability issues and to ensure that the performance objectives are met. This enables to identify the needed capabilities and main challenges for the seamless operation of the heterogeneous satellite-terrestrial wireless networks and their Quality of Service (QoS) requirements. Testing was carried out in various stages. The testing scenarios included Wi-Fi, WiMAX, DVB-RCS and SCPC. Various traffic data, files, packets and traces were captured, and filtered. All the results are presented, analysed and discussed in this chapter. This chapter also presents the results on the Quality of Service requirements from various network sites used during the testing phase of the BASE^2 project.

Chapter 7 provides a summary of the present research and concludes this thesis by discussing and reviewing the findings. This chapter provides recommendations towards improving and achieving good end-to-end interoperability and end-to-end Quality of Service and suggests possible directions for future research in this field.

1.7 Summary

This chapter presented an introduction to wireless communication technologies and standards. The Wi-Fi, WiMAX, DVB-RCS, SCPC, 3G and 4G LTE standards and evaluation process were elaborated and discussed. A list of research questions is formulated. This chapter also furnishes the research rationale and motivation behind this study. A research plan outlines with various phases are shown. The summaries of each chapter given at the end. An overview of the thesis layout for the achievements in each chapter against objectives is shown in Figure 1.4.

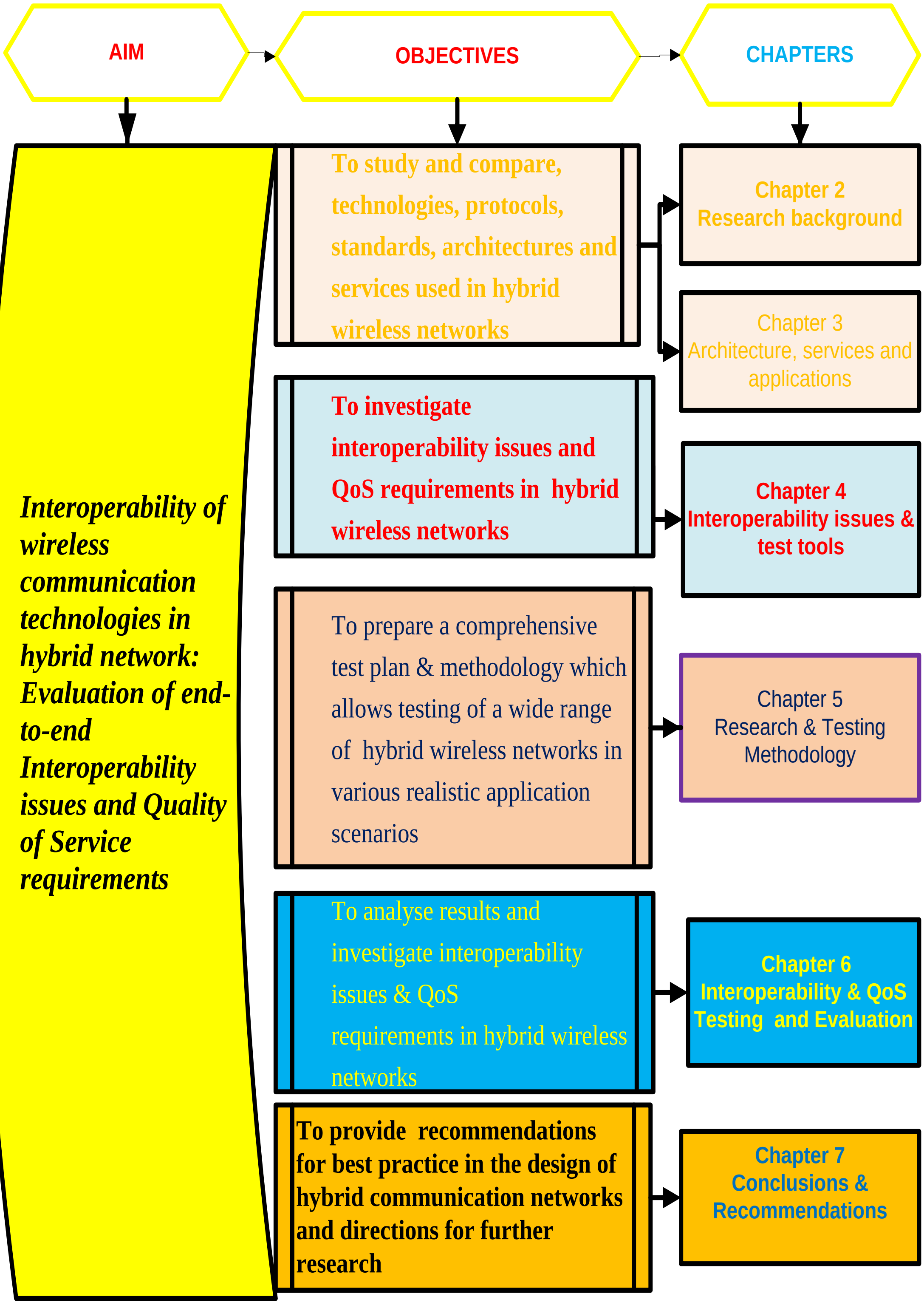


Figure 1.4: Overview of Thesis Layout

Chapter 2

Research Background on Interoperability and Quality of Service Requirements

2.1 Overview

Wireless communication technology requirements have been changing rapidly due to the immense demand for better QoS; thus standards are also continuously evolving due to the requirements of bandwidth, scalability, spectrum flexibility, efficiency, reliability, and performance.

The 1st generation (1G) communication technology (Lyytinen and Fomin 2002) completed its lifecycle supporting analogue Advanced Mobile Phone System (AMPS) standard using the Frequency Division Multiple Access (FDMA) technique. This technology has almost been superseded by the 2nd generation (2G) technology (Haug 2002). Although TDMA and FDMA techniques have been used in 1G technology, however, this technology couldn't handle high data rates and could support voice services only.

The Global Standard for Mobile communication (GSM) technology, which is also known as the 2nd generation (2G) standard for mobile communication, is available all over the world allowing maximum bit rate of 14.4 kbits/s (Ghribi and Logrippo 2000) using Time Division Multiple Access (TMDA) and Code Division Multiple Access (CDMA) techniques.

The GSM technology standard rapidly evolved into the GPRS and Enhanced Data rates for GSM Evolution (EDGE) standards which became 2.5 G technology. Meanwhile the hybrid combination of 2.5G and 3G also came into existence using CDMA 2000 (Rao *et al.* 2000; Kalavakunta and Kripalani 2005).

The technology has been further developed to the Universal Mobile Telecommunications System (UMTS) (Lindemann *et al.* 2002) using Wideband

Code Division Multiple Access (WCDMA), which supports 14 Mbps with High-Speed Downlink Packet Access (HSDPA) and 42Mbps (Shah 2008) with HSPA+. Both CDMA2000 and WCDMA air interface systems are accepted by the International Telecommunication Union (ITU) as a part of the IMT-2000 family of 3G standards. User can access e-mail and internet using High Speed Packet Access (HSPA).

The Long Term Evolution (LTE) technology is based on new Orthogonal Frequency Division Multiple Access (OFDM) air interface techniques instead of WCDMA and offers higher data rates. OFDMA and Single Carrier Frequency Division Multiple Access (SCFDMA) are now becoming popular due to better QoS. In addition to OFDMA which is used in WiMAX, SCFDMA, and Multi Carrier Codes Division Multiple Access (MCCDMA) are gaining more popularity. This technology is called 4th Generation (4G) and is based purely on packet switching instead of circuit switching (Akyildiz *et al.* 2010), or a mixture of both as in 3rd Generation (3G).

The aim and objectives of this review of research background and related literature is to study and review the literature and previous research which has already been carried out in the area of wireless communication technologies in hybrid networks. This review will assist to analyse existing research in the area of Interoperability of wireless communication technologies in Hybrid Networks, find the gaps and guide the process of establishing the requirements for seamless interoperability of all realistic network architectures, and of assessing end-to-end performance in Hybrid communication networks.

This literature review covers wireless communication technologies of both terrestrial and satellite type, and particularly Wireless Fidelity (Wi-Fi), Worldwide Interoperability for Microwave Access (WiMAX), Very Small Aperture Terminal / Single Channel Per Carrier (VSAT/SCPC), Digital Video Broadcasting for Satellite (DVB -S/S2) and Digital Video Broadcasting Return Channel Through Satellite (DVB-RCS) technologies. It further focuses on interoperability issues and Quality of Service requirements. A part of the literature review also covers Global System for Mobile Communications (GSM), General Packet Radio Service (GPRS), Third Generation (3G) and 4th Generation Long Term Evaluation Advanced (4G LTE advance) technologies.

This chapter is organised as following: Section 2.2 shows literature's review methodology adopted. Section 2.3 provides a brief introduction to various wireless technologies, standards and development stages for Wi-Fi, WiMAX, SCPC, DVB-S/S2, and DVB-RCS. This section also presents the literature review of Wi-Fi, WiMAX, 3G, LTE, SCPC, DVB-S/S2, DVB-RCS, interoperability and QoS requirements and a comparison of relevant standards. Section 2.4 presents critical reviews of previous studies on Wi-Fi, WiMAX, DVB-S/S2, DVB-RCS, SCPC, Interoperability and QoS requirements. Finally, section 2.5 summarises this chapter.

2.2 Literature Review Methodology

The first step in the adopted literature review methodology includes generation of a template, selection of data databases to be searched, selection of the key search words, literature search criteria, inclusion/exclusion criteria, extraction of relevant information from selected literature regarding Wi-Fi, WiMAX, DVB-RCS, DVB-S/S2, SCPC, interoperability and Quality of Service requirements on hybrid wireless networks, studying and critical reviewing the literature and summarising of the literature review.

A list of key words to cover the field of interoperability of wireless communication technologies and Quality of Service in Hybrid networks was prepared. These key words were used for searching published literature from the Electronics, Computer engineering and Information systems data bases available online through the Brunel University library. The selected literature was also searched for data such as Bandwidth, delays, jitters, packet loss, latency and throughput. Literature searches were carried out across 16 different electronic bibliographic databases, more than 20 International Journals. In addition to these, World Wide Web search engines such as google.com, yahoo.com, and ask.com were also searched. The author of this study considered a number of relevant research books for potential reference, but only few books were found slightly helpful. Therefore, literature searches primarily included searches of electronic databases and web search engines.

The literature search inclusion and exclusion criteria were set based on the latest research and relevant material on interoperability and Quality of Service requirements of wireless communication technologies particularly Wi-Fi, WiMAX, DVB-RCS, DVB-S/S2 and SCPC, wireless communication, broadband communication, and satellite communication. The abstract of each of the selected papers was read in the first instance. After reading the selected paper abstract, if the study was relevant to the research area, then it was selected for the literature review. The aforementioned databases were searched using key words and search criteria, but the results were not very encouraging and then an attempt was made to find different white papers by relevant research forums which were very helpful, particularly from the WiMAX, Wi-Fi, 3G, and Forum/Organization and IEEE standard organizations.

A template was developed for data inclusion/ exclusion which was helpful for the research process as shown in Appendix “A”. This appendix includes detail of database and journals mostly selected material after year 2000, exclusion particularly Global Positioning System (GPS) and blue tooth, list of key words such as interoperability, QoS, wireless communication, Satellite communication, etc. The close search was carried out for delay, jitter, bandwidth, packet loss, latency, and throughput.

Reference management of studies and research is a very important and essential in order to cite the study, author with published reference journal, and dates but difficult task to perform, particularly during the literature review process. In most cases the cited reference it has been recorded manually and later on added on Refworks due to the compatibility issue as some journal and data base do not offer export of all information to Refwork.

In summary, a total of 390 papers were selected through searches across above mentioned different electronic bibliographic databases and search engines. After reviewing each paper's abstract, introduction and conclusion, it was found that 315 papers were not relevant and were therefore excluded. The remaining 90 papers were reviewed fully.

2.3 Research Background and Literature Review

The following sections provide background and literature review of Wi-Fi, WiMAX, SCPC, DVB-S/S2, and DVB-RCS. The details for each technology literature review and previous research studies are as follow:
Wireless Fidelity (Wi-Fi) which is a wireless technology uses the frequency range between 2.4 to 5.8 GHz and is typically based on one of the IEEE 802.11 standards. The Wi-Fi technology covers Wireless Local Area Network (WLAN). This is taking place of Local Area Networks (LAN). The Wi-Fi most common standards are IEEE 802.11a, b and g for WLAN.

WiMAX is a metropolitan area network using Orthogonal Frequency Division Multiple Access (OFDMA) technique and facilitates the network access for remote or suburban area.

Digital Video Broadcast for Satellite (DVB-S) and its successor DVB-S2 are both mainly broadcast technologies for television and radio programmes, but they are also widely used within data communications.

Digital Video Broadcasting - Return Channel through Satellite (DVB-RCS), Digital Video Broadcasting-satellite/second Generation(S/S2), and Single Channel Per Carrier (SCPC) are technologies for satellite communication networks.

The DVB-RCS is a geostationary earth orbit satellite interactive network which provides interactive broadband access via geostationary satellites. It is an open standard for providing two-way broadband access over satellite. The forward link is in DVB-S format, whereas the return link is based on Multifrequency–Time Division Multiple Access (MF-TDMA).

Wi-Fi and WiMAX are more sophisticated computer systems supporting broadband internet services which can be integrated with Satellite communication equipment such as SCPC (VSAT), DVB-S/S2 and DVB-RCS for interconnecting large numbers of users in remote areas. WiMAX is one of the 4G technologies. The 4G technology (Rinne and Tirkkonen 2010) supports higher data rate, but it is at the development stage.

2.3.1 Wireless Fidelity (Wi-Fi)

The Wi-Fi task group was created in 1997 and released the first set of specifications for Wi-Fi operating at 2.4 GHz. The IEEE 802.11 group created several task forces such as a, b, g, f, e, h, I, n. Wi-Fi which is an effective wireless local area network (Bahr 2006), officially within an effective range of up to 100 metres. Wi-Fi is called the IEEE 802.11 standard developed by the IEEE standard committee working group 11. The IEEE 802.11 task group issued the first set of specifications in 1997 for Wi-Fi working at a frequency of 2.4 GHz. The IEEE 802.11 task group comprised several task force named a, b, g, e, h, I ,n to address the user needs (Skordoulis *et al.* 2008), regarding security, speed, Quality of Service (QoS) and throughput, etc. (Zhang *et al.* 2010).

Three standards most commonly used by Wi-Fi: 802.11 a, b, and g. These standards define the Physical (PHY) Layer and the Media Access Control (MAC) Layer. The MAC layer is the same for these three Wi-Fi IEEE 802.11 standards but the PHY layer differs among them.

IEEE 802.11b use to be the most common and popular WLAN standard; it uses frequency range of 2.4 GHz-2.4835 GHz (Celebi *et al.* 2007; Mahasukhon *et al.* 2009). Data rates supported by this standard are 1, 2, 5.5 and 11Mbps (Celebi *et al.* 2007; Mahasukhon *et al.* 2009) maximum data rate using Direct Sequence Spread Spectrum (DSSS).

IEEE 801.11a is a high speed WLAN capable of providing speeds up to 54 Mbps (Al-Khusaibi *et al.* 2006) in the 5 GHz band (Liu *et al.* 2009). It uses modulation technique known as OFDM which reduces the multipath interferences.

The IEEE 802.11g standard is an extension of the 802.11b standard operating in the 2.4 GHz band (Dao and Malaney 2007), but has advantages over 802.11a and 802.11b. It supports upto 54 Mbps because of the combination of OFDM and Complementary Code Keying (CCK). The advantages of OFDM technique include reduced multipath effects and increased spectral efficiency. The range of IEEE 802.11g is 100 meters indoor.

The IEEE 802.11e standard has come to fill the QoS gap. Many real time applications such as multimedia and teleconference session need higher priority over standard network traffic. The IEEE 802.11e standard creates different classes of transmission and provides priority for audio, video and data transmissions.

IEEE 802f is used for Handover extension. The IEEE 802.11h standard is to improve other WLANs IEEE 802.11 standards operating in the 5 GHz band.
IEEE 802.11i solves the security issues that has become significant after various attacks made on the IEEE 802.11b standard with improper configuration (Potter 2003). The details of each standard are shown in Table 2.1.

Table 2.1: Wi-Fi standards

Wi-Fi Standards	***802.11a***	***802.11b***	***802.11g***	***802.11n***
Frequency (GHz)	5 .15-5.25	2.4 -2.4835	2.4	2.4-5.5
Modulation	OFDM(64 channel)	CCK DSSS/FHSS	Code Keying OFDM	MiMo
Channel Bandwidth (MHz)	20	25	20	40
Data rate (Mbps)	(6-54) max 54	11	(5.5 -54) 54	320
Maximum range (Meter)	50	100	100	200

IEEE 802.11n (Gross *et al.* 2009; 2010) is enhanced to 40 MHz channel bandwidth width double the throughput to increase the bandwidth of IEEE 802.11 to a minimum of 100 Mbps (Ashtaiwi and Hassanein 2010) using MIMO (Multiple input, Multiple output) technology. The 802.11n standard is compatible with the technology standards 802.11b and g.

The modulation and coding techniques used for the 802.11b standard are Differential Phase Shift Keying (DPSK), Differential Quaternary Phase-Shift Keying (DQPSK), and Complementary Code Keying (CCK), for 802.11g are Orthogonal Frequency Division Multiplexing (OFDM), with BPSK, QPSK, 16 Quadrature amplitude modulation (QAM), 16QAM, 64QAM, and for 802.11a are OFDM with BPSK, QPSK, 16QAM, 64QAM. Wi-Fi operating channels in US are 2.412 to 2.462 GHz (11 channels) and 5.15 to 5.25 GHz (4 channels). In Europe these channels are from 2.412 to 2.472 GHz (13 channels) and 5.15 to 5.25 GHz (4 channels). The data rates for 802.11b are: 1, 2, 5.5, and 11 Mbps and for 802.11a/g: 6, 9, 12, 18, 24, 36, 48, and 54 Mbps.

Wi-Fi has experienced a remarkably rapid market acceptance, growing and replacing Ethernet extremely fast. The availability of low-cost Wi-Fi devices and the increasing trend towards built-in Wi-Fi computers/devices is accelerating this growth.

Suitor (2011) gives a brief introduction of the IEEE 802.11 Wi-Fi standards and its maximum range. This study discusses three factors: range, QoS and security of Wi-Fi. The author asserts that the range is significantly reduced if there are obstacles. To overcome this issue, the author recommends that when building an urban canopy, service providers need to build a considerable number of wireless Points of Presence with a transport network using either Ethernet or wireless networks delivering the bandwidth to each access point. However, in terms of QoS and security, most of standards and recommendations became out-dated with the emergence of new standards and protocols such as IEEE 802.11e, IEEE 802.11i and IEEE 802.11n (WiFi.org 2011).

A study (Bahr 2006) discusses the advantages of wireless mesh networks such as greater flexibility, increased reliability and improved performance over conventional Wireless LANs. This study presents a detailed overview on the proposed routing of the first draft version of the IEEE 802.11s standard for WLAN mesh networks. This study does not cover backward compatibility with the previous version of IEEE802.11 standards.

Lorchat and Noel (2006) introduced a replacement Medium Access Control layer (MAC) for wireless communications using the IEEE 802.11 standard. The authors explained the limitations of the 802.11 standard relating to power efficiency and suggested to solve it. The authors proposed Time Division Multiple Access (TDMA)-based MAC protocol and carried out modelling for power consumption of the wireless network interface by using four different states: idle, transmitting, receiving and sleeping by delivering 10,000 packets from 13 transmitting stations to the base station. Their analysis shows that stations can save up to 90% of their battery power according to the network traffic. However, this TDMA based MAC layer has been modified and superseded and new techniques such as Orthogonal Frequency Division Multiplexing (OFDM) and Multiple- input, Multiple-output (MiMo) are added in recent updated IEEE802.11 standards which are also emerged.

A study (Wei 2006) discusses the factors predicting the adoption of Wi-Fi in a work place organizational setting. This study is based on the existing literature and shows various impacts on individual characteristics of the Wi-Fi attributes such as, benefits, social pressure, response to new innovation, individual influence, technology influence, methods, and measures to be taken. However, this study does not offer new idea or novelty.

Henry and Hui (2002) discuss challenges for Wi-Fi engineers and proposed solution for these challenges. This study particularly discuss the security for those using hotspots and focus on Wi-Fi service to support travelling professionals and identified four perspectives such as: ease of use, security, mobility, and network management. However, due to advancements in the technology the security issues have been addressed largely.

Liu and Zarki (2006) propose an adaptive delay and synchronization control scheme for Wi-Fi based Audio Video (AV) conferencing applications. This scheme employs a distributed timing function, which monitors the application-level Quality of Service, and regulates the delay by virtual clock to keep balance between synchronization and delay requirements. The authors claim that this scheme can reduce up to 100 ms end-to-end delay, however, this study is limited to WLAN IEE802.11b and has not been empirically proven and also does not considers delay for satellite communications in hybrid networks.

Li et al. (2006) define a novel performance parameter named as Product of successful transmission Probability and saturation Throughput (PPT) for IEEE 802.11 Distributed Coordination Function (DCF). The performance of DCF-PPT is simulated with different stations in terms of saturation throughput, successful transmission probability and PPT. The simulation result indicates that DCF-PPT can largely increase the PPT and successful transmission probability on the condition that the saturation throughput is not decreased, comparing to IEEE 802.11 DCF. This study doesn't show the performance at different load condition. Therefore, this research contribution is limited.

Almuhtadi (2005) provides a summary of the Basic configuration of Wi-Fi broadband system point-to-multipoint, its performance, and recommendations for use in the field. The Wi-Fi Basic Configuration has been designed, examined and tested. Direct RF interference between nearby circuit boards for Access Points and Bridges was identified as a problem. This problem was solved by using enclosures with appropriate RF shielding.

A study (LaRoche and Zincir-Heywood 2006) presents a genetic programming based detection system for data link layer attacks on a IEEE 802.11 Wi-Fi network and explore two functions to achieve a high detection rate, as well as a low false positive rate. This study focuses on a specific subset of Denial of Services (DoS) attacks, however, new data encryption algorithm and advanced encryption standard covers all of these aspects of security (Masadeh *et al.* 2010; Olteanu and Yang 2010).

Another study (Sun *et al.* 2005) conducts and implements an in-depth evaluation on the effectiveness of Ado Probe which is an end-to-end path capacity estimation tool for multi-hop adhoc wireless networks. According to this study, the sending time is stamped on every packet, One Way Delay (OWD) and path capacity estimation is performed at the receiver end. The receiver measures the OWD of each packet by calculating the difference between the receiving time and the sending time. However, the proposed methodology is highly questionable and is based on set of empirical rules defined by the authors without any strong support to justify their use.

A study (Rajasekhar *et al.* 2006) proposes algorithms to compute QoS paths with maximal path-capacity-to-hop count ratio from a super-peer to all other super-peers in a Wi-Fi peer-to-peer network. Although existing routing solutions provide an effective way of dealing with path selection using shortest possible path selection but this study claims 15% better reliability with proposed algorithms, however, results are showing 9 % successful rate.

Calvagna et al. (2003) tested the system in a real test bed using Personal Digital Assistance iPaq devices using Linux operating system. This study focuses on the problem that arises when two or more IEEE 802.11b access zones are isolated. The approach taken in the experiments is investigating the impact of handoffs on User Datagram Protocol performance. The structure planned and implemented to allow wireless users experience service continuity during moving in the gap between WLAN IEEE 802.11b non adjacent areas in an effective way by a GPRS access. However, reader could not learn new methodologies or results and the IEEE 802.11b standard became out-dated.

Hoene et al. (2003) investigate the packet loss and delay at MAC level for Wi-Fi 802.11b. However, there have been advancements in this standard such as IEEE 802.11g and n (Dao and Malaney 2007; IEEE802.11n 2010).

Kyasanur and Vaidya (2006) propose a protocol to manage multiple channels and assign them to network interfaces. It is thought that M interfaces are available at each node and these are divided in two subsets. At each node some K of M interfaces are assigned for long intervals of time to some K channels. Different nodes assign their K fixed interfaces to a different set of K channels, using different values of K and M. There is a possibility to vary K with time. If an unicast packet is received for transmission, the fixed channel of destination is looked up in the neighbour table and the packet is added to the corresponding channel queue. If the packet received is broadcast, it is added in every channel queue. The fixed interface transmits packets queued up for transmission on fixed channel. Packets are transmitted on all other channels using the switchable interface. The interface is switched to a new channel with the oldest queued data. Each node maintains a channel usage list and neighbouring table containing the number of nodes in its two hops neighbourhood using each channel as their fixed channel. Each node broadcasts “hello” packet which contains the fixed channel. When a node receives a hello packet from a neighbour, it updates the neighbour table. However, it would be more significant, if the information presented were described to add value to the results.

Corvaja (2006) proposes a novel scheme of vertical handover based on a profile of objective parameters which has been considered in an overlay Bluetooth/Wi-Fi network. The scenario consists of one Bluetooth and one Wi-Fi network both covering the same service area. An approximation for the handover probabilities has been derived and the influence of main parameters on the network performance has been studied. To validate the approximate analysis and to obtain the performance of the network throughput, and packet delay, a simulation has been developed, where the position of the users is randomly generated. This study suggests that integrated system should be viable and cost effective.

The most of the above studies describe and discuss Wi-Fi's standards, operating frequency, bandwidth, data rate and coverage. The other studies provide a summary of the basic configuration of Wi-Fi; discuss range of Wi-Fi, power, interference, social issues, security problems, physical and MAC layers.

Some of the studies (Sun *et al.* 2005; Rajasekhar *et al.* 2006) conducts and implements an in-depth evaluation on the effectiveness of Adhoc Probe which is an end-to-end path capacity and address the concept of capacity to hop count ratio, which is used in computing QoS paths. Few other studies (Li *et al.* 2006; Liu and Zarki 2006) discuss issues such as delay, throughput and gives recommendation to improve. However, none of the above studies cover interoperability issues and QoS requirements such as bandwidth, jitter, packet loss, latency, link performance, TCP and UDP performance for Wi-Fi technology for end-to-end level for Hybrid wireless networks.

2.3.2 Worldwide Interoperability for Microwave Access (WiMAX)

The IEEE 802.16 family of standards often referred to as WiMAX. The IEEE 802.16 working group has the responsibility for WiMAX associated standards and its amendments. The 802.16 working group initial interests was in the 10-66 GHz range but has changed to 2-11 GHz in IEEE 802.16a, d and e standards.

WiMAX is defined as "Worldwide Interoperability for Microwave Access" (WiMAXforum 2008), to promote standardisation and interoperability of the IEEE 802.16 standard, which is also known as Wireless MAN.

WiMAX is an effective metropolitan area access technique with many encouraging features such as cost efficiency, flexibility, and fast networking (Sayenko et al. 2006), which provides wireless access as well as serves as a wireless expanding for wired network access. This facilitates the network access for remote or suburban areas. The coverage area of WiMAX spans 30 to 50 kilometres. It provides high speed of data rates more than 100 Mbps in 20 MHz bandwidth (Ghosh et al. 2005).

The WiMAX standard which defines the Wireless MAN air interface was first approved as the IEEE 802.16-2001 standard (ieee.org 2009) and published in 2002. The IEEE 802.16a standard was published in April 2003; its frequency range is 2-11 GHz (IEEE802.16 2003).

The 802.16d standard employs three kinds of physical layer technologies, which are: Single carrier (SC) applied in the frequency range of 10-66 GHz, OFDM 256 points in frequency range of 2-11 GHz fixed wireless access, and OFDMA 2048 points with frequency up to 11 GHz for long distance between operator point of presence and Wireless Local Area Network (Eklund *et al.* 2002).

Mobile WiMAX is based on the IEEE 802.16e standard and operates in the spectrum bands of 2.3 GHz, 2.5 GHz, 3.3 GHz, and 3.4-3.8 GHz. This standard is focusing mainly an enhancement of MAN-OFDM. WiMAX uses Orthogonal Frequency Division Multiple Access (OFDMA) which has a performance edge in delivering IP services as compared to 3G wireless technologies.

The main advantages of WiMAX when compared to other access network technologies are the more sophisticated support of Quality of Service (Theodoros and Kostantinos 2007).

The comparison of WiMAX standard is shown in Table 2.2.

Table 2.2: WiMAX standards

WiMAX Standards	***802.16***	***802.16a***	***802.16-2004 d***	***802.16.e-2005***
Release Time	2001.12	2003.1	2004.7	2005
Classifications	OFDM	OFDM	Air interface standard OFDMA	SOFDMA
Frequency (GHz)	10-66	2-11	2-66, 2-11	2-6
Modulation (QAM)	Single carrier, QPSK,16 ,64	SCa OFDM, using QPSK,16QAM, 64QAM,,256QAM	OFDM, using BPSK,QPSK,16,64	OFDMA, QPSK,16,64
Channel Bandwidth (MHz)	14, 28-	1.25-20	3.5-20	5,7,8.75,10 1.5- 5
Spectrum allocation (GHz)	5.725-5.825	2.5-3.5, 5-5.8	2.5-2.69 & 3.4-3.6 Unlicensed, 5.725-5.85 Licence	2.3,2.5,3.3,3.5
Spectrum usage (bps/Hz)	4.8	3.75	3.75	3
Data rate (Mbps)	155	70	70	100
Maximum range (km)	8	50	50	10
Security	Triple-DES (128-bit) and RSA	Triple-DES (128-bit) and RSA	Triple-DES (128-bit) and RSA	Triple-DES (128-bit) and RSA
	(1024-bit)	(1024-bit)	(1024-bit)	(1024-bit)
Mobility	No	No	No	Yes
LOS	No	No	No	Yes

WiMAX is a metropolitan areas network “last mile access” technology. The operating frequencies of 2.5 and 3.5 GHz require a license; however, 5.86 GHz frequency is an unlicensed band. WiMAX addresses the requirements of those users who want to use a broadband connection regardless of location which are not covered with DSL and cable technologies.

IEEE 802.16e is designed to support mobility and asymmetrical Link, Voice and Video, as well as centrally enforced QoS.

The IEEE 802.16 standard group is currently working on IEEE 802.16m (Ahmadi 2011) to define a high mobility interface which offers data transfer speeds up to 1 Gbps, frequency channel bandwidth of **5**,7,8.75,10,20 MHz and reduced latency, and is compatible to all existing WiMAX standards.

WiMAX provides high speed upto 100 Mbps. The IEEE 802.16d products work on both licensed 2.5-2.69 and 3.4-3.6 GHz and unlicensed 5.725-5.85 GHz. The IEEE 802.16 standard defined two types of Orthogonal Frequency Division Multiplexing (OFDM) systems for WiMAX i.e. OFDM and OFDMA. OFDM is a multicarrier modulation technique whereas OFDMA is multiple access scheme. The WiMAX 802.16d standard has three different kinds of physical layer technologies, which are: Single carrier (SC) applied in the frequency band of 10-66 GHz fixed wireless access system, OFDM 256 points used in band of 2-11 GHz Fixed wireless access, and OFDMA 2048 points with frequency range of up to 11 GHz for long distance links between operator point of presence. Mobile WiMAX is based on the WiMAX IEEE 802.16e standard and operates in the frequency spectrum bands of 2.3 GHz, 2.5 GHz, 3.3 GHz, and 3.4 to 3.8 GHz range.

WiMAX can inter work with satellite and terrestrial wireless existing as well as emerging technologies. WiMAX also serves as a backbone for Wi-Fi hotspots for connecting to the broadband Internet. The IEEE 802.16n standard which offers higher network reliability and the IEEE802.16p standard, which supports machine to machine applications, are in developing stage.

WiMAX offers true broadband connections which support multiple scenarios, including fixed, portable and mobile wireless access and covers a range up to 40 km for Line of Sight (LOS) operation and up to 10 km range for Non Line of sight (NLOS) operation.

The WiMAX network architecture is more flexible, encourages inter working and roaming and is more cost effective. Its applications include broadband internet access, tele-presence, information access, inter machine communication, intelligent shopping, and location-based services.

Jing and Raychaudhuri (2006) investigate the feasibility of spectrum coexistence between the Wi-Fi IEEE 802.11b standard and the WiMAX IEEE 802.16a standard using both reactive interference avoidance methods and the Common Spectrum Coordination Channel (CSCC) protocol. This study considers the important emerging scenario in which both metropolitan area WiMAX IEEE 802.16 and local area Wi-Fi IEEE 802.11 technologies could co-exist in the same unlicensed band sharing 2.4 GHz spectrum with a small amount of coordination. Both simple scenarios with one IEEE 802.16a cell and one IEEE 802.11b hotspot and more realistic scenarios with multiple hotspots were simulated using an ns-2 simulator. Results demonstrate that CSCC power adaptation can help maintaining IEEE 802.16 quality at the expense of a modest decrease in IEEE 802.11 throughput in the hidden-receiver scenario (Xiangpeng and Raychaudhuri 2006). However, 802.11b has been superseded by 802.11g; and 802.16a has been superseded by 802.16g and 802.16n.

A study (Ghosh *et al.* 2005) mentions the IEEE 802.16 standard for fixed WiMAX 802.16d. This standard is also known as 802.16-2004, which delivers high data rate over a large area by reusing frequency. However, this study measures only the throughput and does not consider delay, jitter, latency and packet loss. Another study (Fong *et al.* 2004) presents an overview of various features of Broadband Wireless Access to support a fast growing network. This study combines the scalability with WiMAX standards such as IEEE 802.16 and a hierarchical Local Mobile Distribution Service structure. This study recommends high level of scalability by optimizing various network resources, such as utilizing the available bandwidth efficiently and making a minor enhancement to an existing system. However, the authors did not consider other factors such as interference and weather condition. The study is also limited to 10-66 GHz fixed wireless access system and excludes OFDM range 2-11 GHz fixed wireless access.

Eklund et al. (2002) describe the history of WiMAX standards and its advantages such as MAC of IEEE 802.16, which supports different transport technologies including Internet Protocol version 4 (IPv4), IPv6, Ethernet and Asynchronous Transfer Mode (ATM). The WiMAX 802.16e supports power saving and sleep modes to extend the battery life of mobile devices and it also supports hard and soft handoffs to provide users with seamless connections.

Another study (Hoymann 2005) specifies four different Physical (PHY) layers but considers only the Orthogonal Frequency Division Multiplex (OFDM) layer. An overview of the OFDM based transmission mode of the WiMAX IEEE 802.16 standard is presented. The MAC and PHY layer are described in detail. The MAC layer configurations with different levels of robustness are also analyzed.

A study (Sayenko *et al.* 2006) presents the estimation for WiMAX MAC header overhead to reserve sufficient amount of slots for the constant-rate applications. This study presents several simulation scenarios to demonstrate how the scheduling solution allocates resources in various cases. The scheduling solution was based on the round-robin scheduling. The simulation scenarios run in ns-2. It could be more significant if the study could provide comparative results against other wireless technologies MAC header overhead to add value to the results.

Jain et al. (2001) propose a multi-channel Carrier Sense Multiple Excess (CSMA) with collision avoidance technique. This study evaluates the performance of receiver based channel selection, comparing with IEEE 802.11 Distributed Coordination Function (DCF) using ns-2 simulator. However, at a given time, one packet can be transmitted only on any channel, but multiple packets can be received at various channels at the same time.

A study (Chatterjee *et al.* 2007) uses Forward Error Correction (FEC) and Automatic Repeat Request (ARQ) to support streaming services and studied the problem of real time streaming media over WiMAX and exploited the flexibility features in MAC layer of 802.16a standard. The authors proposed the size of MAC packet data units to make adaptive to the instantaneous wireless channel condition.

Rong et al. (2007) propose an integrated adaptive power allocation (APA) - call admission control (CAC) downlink resource management framework for OFDM-TDD based multiservice network by taking into account the service provider and subscriber. This study developed algorithm of both.

Niyato and Hossain (2007) propose a pricing model for adaptive bandwidth sharing in an integrated Wi-Fi/WiMAX network. Game theory has been used to analyze and obtain pricing for bandwidth sharing between a WiMAX base station and Wi-Fi access point routers.

Hung-Yu et al. (2005) discussed the interference issues and proposed an efficient approach for utilization of WiMAX mesh through design of multi-hop routing and scheduling algorithm scheme. This scheme considered both traffic load demand and interference conditions. Simulation results show that the proposed schemes effectively improved the network throughput performance in IEEE 802.16 mesh networks and achieved high spectral utilization.

A scheduling mechanism and a routing algorithm is developed (Liqun *et al.* 2005) to maximize the spatial reuse in wireless mesh networks and to achieve better network throughput and spectral efficiency. The model requires the receiver to be free of interference and considers the interference range as equal to the communication range.

Huang et al. (2007) describe that a good scheduling control is critical to support mixed VOIP (Voice Over IP) and for non-real time services in mobile WiMAX as defined in the 802.16e standards which provides mobile support in cellular deployments.

Fourty et al. (2005) examined and classified wireless networks. This study suggests that WiMAX made the possibility to obtain a connectivity of the similar type as the rented lines used by network's operators for the Internet or telephony transport T1 (for American suppliers) or E1 (for European suppliers).

A study (Kejie *et al.* 2007) discusses common requirement of security in WiMAX and possible attack to WiMAX networks. This study suggests that secure network must satisfy and address the requirement for confidentiality, authentication, integrity, and availability.

Another study (Hoymann *et al.* 2006) is based on Wireless OFDM Networks which relates to the FIREWORKS project. This study analyses the characteristics to improve the IEEE 802.16 standards. This study also recommends designing of Radio Resource management (RRM) algorithms.

A study (Qiang *et al.* 2007) compares delay performance of two bandwidth request mechanisms which are detailed in the 802.16 standard, random access vs. polling. This study shows that the polling mode provides better QoS performance than random access mesh mode.

In above studies (Jing and Raychaudhuri 2006) investigate the feasibility of spectrum coexistence between the Wi-Fi IEEE 802.11b standard and the WiMAX IEEE 802.16a standard. Ghosh et al. (2005) mention the IEEE 802.16 standard and its advantages, Hoymann (2005) specifies four different Physical (PHY) layers, and another study (Sayenko et al. 2006) presents the estimation for WiMAX MAC header overhead.

Most of other studies provide generic information regarding WiMAX frequency and range. Some of the studies are narrative description of the WiMAX's standards and MAC layers. However, a study (Qiang et al. 2007) compares the interference issues and testing results on delay and data rate. Therefore, there is a requirement to study interoperability issues and Quality of Service requirements for WiMAX in Hybrid Wireless Networks.

2.3.3 3rd Generation (3G), Long Term Evolution (LTE) and LTE Advanced

The High Speed Packet Access family (HSPA) (Masmoudi and Tabbane 2009) and the Long Term Evolution advanced (Akyildiz *et al.* 2010) are new technologies that allow users with option of various access. The development of LTE Advanced can be seen from the evolution of 3G services that were developed using UMTS/W-CDMA technology. The maximum downlink speed is to be 1Gbps (Rinne and Tirkkonen 2010), the maximum uplink speed 500 Mbps, latency 5 ms and access methodology OFDMA/SC-FDMA.

Liangshan and Dongyan (2005) discusses and explained the main differences between Wi-Fi, WiMAX, 3G and their associated standards. This study presents SWOT (Strengths, weaknesses, opportunities and threats) analysis for these technologies and analyses future trends. The authors recommend that the three technologies can cooperate while competing at the same time. However, the consideration is not given to 4G technology. This study only discusses the WiMAX standard in detail but little information are provided for WLAN and the 3G standard.

Chen et al. (2005) propose an all IPv6 service architecture consisting of cellular network and wireless network. A General Packet Radio Service (GPRS) Wireless local Area Networks (WLAN) interworking gateway with an IPv6 facility has been designed. The performance examined queue length, system throughput, loss rate and delay. The study lacks finding and recommendations to address the issues.

Another study (Calvagna *et al.* 2003) develops a mobility framework that extends to seamlessly manage roaming into GPRS access network every time when the mobile host is not within the range from any Wi-Fi domain. This study presents a middleware which is designed to cope with the problem to provide uninterrupted wireless IP connectivity to users moving between remote Wi-Fi domains, seamlessly switching between Cellular IP (CIP) like Wi-Fi mobility and GPRS roaming. However, various advancements have been made like High Speed Packet Access (HSPA), Universal Mobile Telecommunications Services (UMTS), 3G LTE, and now 4G LTE Advanced (Rinne and Tirkkonen 2010).

A study (Iwamura *et al.* 2010) discusses carrier aggregation structure in 3rd Generation Partnership Project (3GPP) LTE advanced with reference to LTE release 10 to scale the system bandwidth beyond 20 MHz up to 100 MHz as compared to 3G PP LTE release 8/9 for 1.4, 3, 5, 10, 15 and 20 MHz bandwidth. Another study (Akyildiz *et al.* 2010) discusses the process for evolution from 3G, 3.5G to LTE advanced and highlights 3GPP network architecture. The challenges for design and management of LTE advanced are also highlighted; and performance for uplink and downlink speed and peak data rates for LTE advanced and IMT advanced are compared.

Nasri et al. (2010) presents the dynamic modulation and coding scheme and performed channel quality using LTE simulator by conducting Matlab based simulations.

Although literature review for 3G, 3GPP2 and 4G LTE advanced technologies are carried out, was not part of this research due to unavailability of these technologies in test scenarios.

2.3.4 SCPC (VSAT), DVB-S/S2 and DVB-RCS,

Very Small Aperture Terminals (VSAT) are fixed satellite antennas that provide reliable communication for data, voice, teleconference service and fax amongst geographically distributed (Al-Wakeel and Al-Wakeel 2000). A VSAT setup consists of three components namely Master Earth Station, Remote Earth station, and Geostationary Satellite. The Master Earth Station (HUB), is the network control centre for the VSAT network, which includes a large six-meter antenna, an independent backup power system, and a regulated air conditioning system. A VSAT station consists of two parts, the outdoor unit (ODU) and the indoor unit (IDU). The outdoor unit is the VSAT interface to the satellite, and the indoor unit is the interface to the user's terminals or LANs. An outdoor unit consist of the antenna, the transmitting amplifier, the low-noise receiver, the frequency synthesizer and the up-and-down converters.

The satellite communication industry is deploying low cost VSAT for data, voice and video communication (Stergioulas *et al.* 2008). VSAT technology is a proven solution for those who are interested in independent communications connecting a number of remote areas/sites. VSAT technologies offer satellite-based broadband internet, data, voice, video, and teleconferencing services (Stergioulas *et al.* 2008). VSAT provides private and public network communications. Generally VSAT operates in C and Ku band.

VSAT networks offer cost effective satellite-based network services which are capable of supporting the internet, Local Area Networking (LAN), audio, video, communications, and provide strongly, dependable communications classified or unrestricted.

Digital Video Broadcast for Satellite DVB-S and its successor DVB-S2 are both largely broadcast technologies for television and radio programmes, but they are also widely used in data communications.

DVB-S/S2 is a broadcast technology which was invented to deliver television signals. Therefore, it is always unidirectional and meant to serve several users from one central uplink. There is no specific return channel defined within DVB-S/S2, but there are several different possibilities for a bidirectional satellite link with a DVB-S/S2 forward link to the user's VSAT station. The DVB-S2 works with QPSK, 8PSK, 16APSK, and 32APSK modulations (Morello and Mignone 2006) in order to work properly on the nonlinear satellite channel.

The Table 2.3 shows a comparison of the two technologies (DVB-S verses DVB-S2).

Table 2.3: DVB-S & S2 standards

Standards	***DVB-S***	***DVB-S2***
Modulation	QPSK	SPSK,8PSK,16APSK,32APSK
Coding Scheme	Viterbi and Reed Solomon	LDPC(Low Density Parity Code) and BCH (Bose-Chaudhuri-Hocquenghem)
Coding Rates	1/2,2/3,3/4.5/6,7/8	1/4,1/3,2/5,1/2,3/5,8/9,9/10
Roll Off	0.35	0.2-4.5 bits Hz
Spectral Efficiency	1.0- 1.75 Bits/Hz	0.5-4.5 bits/Hz
Throughput	45 Mbps	65 Mbps
System Capacity	540 Mbps	780 Mbps

The primary use of DVB-S/S2 is the unidirectional delivery of signals from one uplink to many end users (Bennett *et al.* 2005). However, DVB-RCS is only an example for a return channel, others, like PSTN or satellite modems are available “as well”. DVB-S can transmit data extremely cost effectively, especially when a vast number of receivers are reached by the signal at the same time. Therefore, DVB-S has advantages whenever one content should reach many interested parties. This is absolutely true for radio and television broadcasts, but also for IP related teleteaching programmes which depend on basic internet services like chat as well as VoIP connects telephone calls serve as the return link. The DVB-S2 is designed for numerous satellite applications such as standard definition TV, HDTV, interactive services, internet, digital TV News gathering (DSNG), TV distribution to terrestrial VHF, UHF, and Data Contents distribution (Morello and Reimers 2004).

Digital Video Broadcasting–Return Channel through Satellite (DVB-RCS) is a satellite based communications compliant system defined in ETSI EN 301 790 (ETSI 2009).

The DVB-RCS is a geostationary earth orbit satellite interactive network which provides interactive broadband access via geostationary satellites. It offers the possibility to implicitly host the required return channel on the same medium. It is an open standard for providing two-way broadband access over satellite. The Forward Link carries communication from a gateway, via satellite, to satellite interactive terminals (SITs) and is based on the DVB/MPEG data format standard (Alagoz 2004) . The Return Link carries communication from SITs to the gateway and uses a Multi Frequency Time-Division Multiple Access (MF-TDMA) scheme, carrying ATM or MPEG cells allowing a two way exchange of data (Bennett *et al.* 2005). The forward link is in DVB-S format, whereas the return link is based on MF-TDMA.

DVB-RCS is a star topology network (Song *et al.* 2006) that utilizes broadband satellite access in order to achieve interactive communication (Chini *et al.* 2006) between Satellite Interactive Terminals (SITs) and the DVB-RCS Hub (gateway) of the network. This technology permits the provision of telecommunications services based on a wide frequency spectrum. The network was designed to meet a variety of broadband interactive services and multimedia applications.

There are three main core parts of a DVB-RCS network: the Satellite Interactive Terminals (SITs), the DVB-RCS Hub (gateway), and the satellite (Hassan *et al.* 2006). The Ground Segment of the DVB-RCS network is the central node of the network. The Ground Segment (hub station) comprises of the Forward link sub system, the Return Link subsystem and the IP router.

DVB-RCS capacity of the Forward Link (FL), namely the communication link from the DVB-RCS HUB station to the Satellite Terminals, is usually greater than the capacity in the Return Link (RL), namely the communication link from the Satellite Terminals to the DVB-RCS HUB station.

The applied services to the end-users are based on Internet Protocol (IP) communication and every Satellite Terminal can support a Local Area Network (LAN) comprising up to 254 devices with IP interface. The Return Link Sub-System (RLSS) is part of the gateway responsible for receiving return link communication from SITs or other destination. The RLSS also manages all aspects of return link communication, including SIT logon, SIT synchronization, and allocation of return link capacity. The RLSS uses the forward link to send control information to SITs. SITs are available for the C, Ku, Ka and X bands. DVB-RCS is a network that uses broadband satellite access in order to achieve interactive communication.

A study (Kim 2006) discusses three sections of wireless communication systems: key technologies, main standardization trends and implementation issues. The author describes the main difference between Wi-Fi, WiMAX technologies and describes that there are four core standards of Digital Video Broadcast (DVB) such as DVB-S (Digital Video Broadcast through Satellite television and Internet), DVB-H (Handheld), DVB-C (Cable), and DVB-T (Terrestrial). However, DVB-S has been superseded by DVB-S2 (Morello and Mignone 2006) and merged with DVB-RCS (de la Cuesta *et al.* 2009) for forward link. This study is limited to Korea.

Chini et al. (2006) discuss that the DVB-S was conceived for primary and secondary distribution Fixed Satellite Service (FSS) and Broadcast Satellite Service (BSS), mostly operated in the Ku band. This system is designed to provide a direct reception from the satellite (Direct-To-Home, DTH) for both a single user with an integrated receiver-decoder and a common access. Results show that variation on the intended clients has an impact on the protocol and signature levels as well as on the application and platform levels of service interoperability. However, this study does not elaborate nature of the impact.

Andersen et al. (2006) give a technical overview of the system architecture of an L-band and Ku-band hybrid solution for the provision of asymmetrical broadband services to mobile terminals, based on integrating DVB-S broadcast channels operating at Ku-band with the Inmarsat Mobile Packet Data Service (MPDS) system operating at L-band. This study concludes that short-term solution could utilize L-band and C-band or L-band and Ku-band hybrid solutions to bridge the gap between the existing L-band systems and future C-band, Ku-band or Ka-band systems for broadband mobile satellite services. Although they have discussed DVB-RCS but has not considered DVB-RCS Forward and Return Links. The

DVB-RCS Forward Link carries communication from a gateway, via satellite, to satellite interactive terminals (SITs) and is based on the DVB/MPEG standard. The Return Link carries communication from SITs to the gateway and uses a multi-frequency time-division multiple access (MF-TDMA) scheme carrying ATM or MPEG cells.

A study (comstream 2005) discusses DVB-S, DVB-S2 and DVB-RCS standards and their associated advantages. This study describes that DM240-S2 modulator can operate for DVB-S and DVB-S2 modes without adding additional software or hardware. Another study (Morello and Reimers 2004) discusses the DVB-S2 history, development phases, and modulation formats. It is observed that the competition might be re-opening with terrestrial infrastructures such as ADSL (Asynchronous Digital Subscriber Line) and cable modems in the rural areas in the near future.

Song et al. (2006) present the design of a mobile broadband interactive satellite access technology system (MoBISAT) that is based on DVB-S/DVB-RCS standard. MoBISAT system is described as a solution of mobile broadband interactive satellite multimedia service. The system is implemented for up to 80Mbps transmission based on DVB-S for forward link and up to 10Mbps with DVB-RCS for a return link for Ku/Ka band service. The key factors of hub and mobile terminal are addressed for the implementation of the MoBISAT. (Bennett *et al.* 2005) describes DVB-RCS implementations within Global Broadcast Service (GBS) IP architecture.

Hassan et al (2006) argues that DVB-RCS is given consideration over DVB-S/S2. Simulation results show that with “limited power” the higher rate FEC like 1/2 or 2/3 should be used to achieve high efficiency. If there is power to spare then the use of lower rate FEC like 3/4 or 5/6 is more efficient. For achieving higher data rates lower code rates 5/6 and 7/8 should be used and for robustness against noise higher code rates like 1/2 and 2/3 shall be used. For achieving higher data rates lower code rates 5/6 and 7/8 should be used. Single implementations will be reliable for less critical services but for achieving optimum performance concatenated approach shall be used. This also depends on the transponder capabilities.

Al-Wakeel and Al-Wakeel (2000) describe that VSATs (Very Small Aperture Terminals) has made feasibility to build low cost Multimedia on Demand (MMOD) services and these services can be installed in remote areas where there is no existing of any telephone infrastructures.

A study (Giambene and Kota 2006) argues satellite network architectures should be fully IP based, support DVB, and return channel protocols for DVB-S, DVB-S2 and DVB-RCS. As satellite services can never compete with the terrestrial on the basis of costs. However satellite technologies have the strength of providing services across large geographical dispersed areas where terrestrial networks cannot reach or are unavailable.

McSparron et al. (2006) discusses the specific implementation challenges faced when combining the use of the DVB-S and DVB-S2 standards, and the benefits that can be achieved in practice such as 30% transmission efficiency by DVB-S2 and advanced modulation schemes such as 16APSK and 32APSK. This study identifies that the major advantage of Ka band is low-service cost from the point of end user.

Lucke et al (2006) present and analyse cross-layer aspects for QoS scheduling on the forward link by means of simulation. A performance comparison between DVB-S and DVB-S2 has shown a significant gain achieved by the proposed scheme Smart Fade Mitigation Technique (FMT). The gain through FMT amounts to approximately 6–7% for the traffic scenario considered. For the considered fading scenario, the overall improvement in efficiency of DVB-S2 over DVB-S was calculated by a factor of 2 to almost 2.5. On the return link, several approaches for access burst structures have been presented and their impact on link efficiency through padding and encapsulation overhead was simulated. Efficiency and benefits of QoS for real-time, low-jitter applications were examined in detail.

A study (Vieira *et al.* 2006) proposes a novel cross-layer design that allows utilizing channel-related knowledge to the packet scheduling of the forward link to provide tuneable fairness. This study focused on the forward link of a GEO (Geostationary Earth Orbit) and Multibeam Broadband Satellite (MBS) system. A methodology to introduce a tuneable fairness parameter for bandwidth allocation in satellite systems has been presented. Another study (Costabile *et al.* 2004) proposes a modular system for QoS management over DVB-RCS satellite platforms. The main aim of this study was the standardization efforts towards better QoS management policies on DVB-RCS return channels. A dynamic traffic management strategy for the return channel of a DVB-RCS satellite system has been presented. A flexible traffic management strategy is designed by mapping applications onto differentiated service classes, on the basis of objective values of parameters, such as call delivery delay variation, packet loss and jitter. This study proposes priority based assignment policies depending on the status of the connections and verifies the throughput at a high level.

Pace et al. (2004) designs and test a DVB-RCS architecture using a multi-spot beam geostationary satellite. Simulation results shows that by using the Connection Admission Control (CAC) algorithm it is possible to obtain a given source peak data rate increase (up to the 50%) over the return link without the necessity of again performing the CAC procedure. This also guaranteed QoS for different types of supported services.

Alagoz (2004) investigates the relevance of integrating scene length characteristics of Moving Pictures Expert Group (MPEG) coded video bit streams into a Direct Broadcast Satellite (DBS) network with Return Channel System (DVB-RCS). The analysis relies on extensive set of simulations. The results show that the scene length characteristics can be incorporated into the network design while a few MPEG video sources are multiplexed on the same link.

A study (Skinnemoen *et al.* 2004) focuses on three main network management topics, namely: handling terminals user profiles; capacity requests; and Service Level Agreement (SLA). Fault events from network elements and applications in the gateway are reported from Simple Network Management Protocol (SNMP) agents within the network elements to a HP Open View Network Node Manager. Some of the main advantages of standardization, including cost-efficiency of a DVB-RCS system, are also highlighted.

Another study (Lee *et al.* 2005) discusses the performance enhancing mechanisms for supporting real time services on DVB-RCS environments. Results show that DVB-RCS system uses TCP over satellite (TCPSAT). The main disadvantages of using GEO satellite in network communication are that current interworking protocols do not quickly adapt to the available bandwidth when traversing a network with a long delay which is approximately 590 ms. The authors in this study have not considered UDP delay over satellite and comparison between TCP and UDP protocols over a satellite link.

Most of the previous studies regarding DVB-RCS, S/S2 and SCPC discuss the history of DVB-S, DVB-S2, DVB-RCS standardisation trends and their advantages (comstream 2005; Chini *et al.* 2006; Kim 2006; Morello and Mignone 2006). Andersen et al (2006) give a technical overview of the various satellite frequency bands. Song et al. (2006) present the design which is based on DVB-S/DVB-RCS standard. A study (Costabile *et al.* 2004) proposes a modular system for QoS management over the DVB-RCS return channel satellite platforms. However, the main aim of this study was to contribute to the standardization research efforts towards effective QoS management policies over DVB- RCS return channel. There is a study (Lee *et al.* 2005) which has carried out some testing for DVB-RCS on delay but this study is limited to test TCP only.

2.3.5 Comparison of Wireless Technologies

In this section, comparison of various wireless technologies is shown. Table 2.4 gives a general overview of the Hybrid wireless technologies, standards, modulation techniques, data rate, bandwidth and frequency.

Table 2.4: General overview of hybrid wireless technologies

Technology	***Standards***	***Multiple Access and Modulation coding***	***Data rate***	***Bandwidth***	***Frequency***
1G	AMPS	FDMA,TDMA	10 kbps maximum	2.4-3 KHz	400-800 MHz
2G	GSM	TDMA,CDMA	100kbps maximum	9.6-14.4 KHz	850/900/1800/1900 MHz
2.5 G	IS 95-B GPRS,IS 95C EDGE	8PSK(8 PHASE SHIFT KEYING)	2Mbps	384-1 MHz	850/950/1800/1900 MHz
3G PP2 3G	CDMA2000 IMT 2000 UMTS	OFDM, OFDMA, SDMA HSPDA, WCDMA	42 Mbps	1- 2 MHz	450/ 700 /800 /900 /700/1800/1900/ 2100 MHz. 850/1900/2100 MHz
3G LTE	LTE	OFDMA for Downlink SC-FDMA for uplink Support FDD and TDD duplexing, DL/UL Modulation QPSK,16QAM,64 QAM	100 Mbps down link 50 Mbps uplink	1.4-20 MHz (1.4,3,5,10,15, 20)MHz	3.5 GHz
Wi-Fi	802.11	CCK in 802.11b OFDM with data modulation BPSK,QPSK, 16QAM,64QAM, DSSS/FH, MiMo for 802.11 n	54 Mbps Up to 320 Mbps in 802.11n	20 MHz	2.412-2.484 5.15-5.25 GHz
WiMAX	802.16	OFDMA multiple access for uplink and down link , TDD duplexing, Data modulation QPSK,16QAM,64 QAM	100 Mbps	5-20 MHz (5,7,8.75, and 10 MHz for Mobile	2-66 GHz in 802.16 2-11 GHz in 802.16a Mobile 3.4 GHz
DVB-S		QPSK	45 Mbps	Upto 540 MHz Capacity	L band,C band
DVB-S2		QPSK,8PSK,16APSK,32APSK	65 Mbps	Upto 780 MHz Capacity	Ka, Ku band
DVB-RCS		MFTDMA	65 Mbps	Up to 500 MHz capacity	Ka band, Ku Band

Wi-Fi and WiMAX offer higher data rates as compared to other wireless technologies, including 3G. The Wi-Fi currently offers data rates up to 54 Mbps, which will increase, to 320 Mbps in the new IEEE802.11n standards.

WiMAX is a metropolitan area network with many technical features such as easy networking, flexibility, scalability, and cost effectiveness, which not only provides wireless access, but also serves as a wireless expanding for wired network access. The coverage area of WiMAX is around 30 to 50 kilometres for the Non line of sight (NLOS). It provides superior data rates up to 100 Mbps in 20 MHz bandwidth. The range of WiMAX can reduce 8-20 km when there are obstacles (lower than the limits as described in standard IEEE 802.16-2004 standard). WiMAX IEEE 802.16d technology is adapted for both core and access networks and provides QoS management for each flow at the MAC level, in the traditional terms of jitter, latency, and throughput.

The major difference between the Wi-Fi IEEE 802.11 standard and the WiMAX IEEE 802.16 standard is that WiMAX 802.16 uses both external Reed-Solomon block code concatenated and inner convolutional code. The MAC Layer of IEEE 802.16 was designed to meet the requirements of high data rate with a various level of QoS requirements. The signalling and bandwidth allocation of WiMAX was designed to accommodate several terminals per channel. The WiMAX standard allows each station to be shared by multiple users. The services required by users can vary in terms of bandwidth and latency requirements, which demand that the MAC layer is flexible and capable over a vast range of different data traffic. The WiMAX was designed time-division multiplex (TDM) voice and data techniques, Internet Protocol (IP) connectivity, and voice over IP (VoIP). The WiMAX MAC layer was designed to provide sparsely distributed stations with high data rates. WiMAX links are considered a natural backhaul medium for Wi-Fi hotspots.

The MAC of 802.16 also supports different transport technologies such as Internet Protocol version 4 (IPv4), IPv6, Ethernet, and Asynchronous Transfer Mode

(ATM).This allows service providers to use WiMAX independently of the transport technology they support.

Very Small Aperture Terminal (VSAT) networks offer satellite-based services which support broadband internet, data, LAN, voice and teleconference communications and can also provide powerful, value-added, dependable private and public network communications.

Other Satellite technologies can be used as distribution networks such as DVB-S which only allows a unidirectional links, and DBS-RCS supports bi-directional links. Most current satellite links used for IP data distribution are based on SCPC DVB-S carriers. During the last few years DVB-RCS was introduced to the market, therefore near future, DVB-RCS compliant systems are becoming popular, while other technologies will evolve towards broader applications, firstly DVB-S2, and secondly on-board processor (OBP)-based systems. The DVB-RCS operates mainly ku bands with transmit frequency band 14 to 14.5 GHz and receive frequency band 10.7 to 12.2 GHz.

A DVB-RCS compliant solution seems to be the most reasonable in terms of performance, capability, and flexibility. Since potentially many users have to be served, a fixed channel allocation is not feasible. The important advantage of DVB-RCS is that it allows on-demand assignment possibly for large terminal populations on a single shared resource.

2.3.6 Interoperability Issues and Quality of Service (QoS) Requirements

Satellite communications enforce particular constraints as compared to terrestrial systems in terms of bandwidth, delays, jitters, latency, and packet loss. For end-to-end Interoperability issues and Quality of Service (QoS) requirements, many technical challenges are to be addressed due to rapidly advancement in standards, protocols, and architectures of wireless communication technologies, with reference to both the ISO/OSI model and the Internet protocol suite.

Sabiguero et al. (2007) present a solution based on combined use of network virtualization and machine virtualization. This solution solves the problems that allow deploying several configuration scenarios with fixed hardware configuration. Interoperability has been defined as the ability to provide successful communications between end-users across a mixed environment of different networks, domains, facilities, and equipments, from different manufacturers and providers (Sugarbroad 1990).

Veer and Wiles (2008) describe all types of interoperability such as technical interoperability usually associated with hardware/software components and systems which allow machine-to-machine communication.

Syntactical interoperability (Jing *et al.* 2007) is usually associated with data formats. Semantic interoperability (Abdalla 2003) is associated with the ideas and concerns of human. Organizational interoperability (Rauffet *et al.* 2009) is the ability of organizations to communicate effectively and transfer data using a variety of different information systems over widely different instructions, possibly different geographic area and cultures.

Viswanth and Obraczka (2006) introduced two interoperability mechanisms: the flooding-based interoperability approach and the facilitator-based interoperability approach. The Qualnet as the simulation setup is used for simulation. It was observed that flooding-based interoperability exhibits higher reliability than the facilitator based approach. The simulation results show that flooding-based interoperability technique has the advantages of being simple in terms of implementation. However, it was observed that facilitator-based interoperability is more suitable for video conferencing.

Another study (Abuelma'atti *et al.* 2006) addresses the interoperability problem in wireless network appliance and describes the architecture of such network. It was envisaged that interoperability is a problem which exists in all OSI seven layers. For truly wireless network appliances systems all devices need to move and communicate seamless and should have plug and play environment. A test bed for home-based network gateway is also proposed with entire domain that is interoperable but this is not possible until there is any standardization of hardware and software.

Al-Gizawi et al. (2005) propose interoperability algorithms which consist of terminal type, traffic specification, speed, user preference, throughput, load balance, and number of handovers. An overall performance is proposed by means of software simulation platform and simulation results are presented in form of percentage of unsatisfied users, percentage of dropped handovers and user throughput. This study also discusses the cost function and suggests that inter-working platform should follow better cost, higher bandwidth, more capacity and enhanced QoS.

A study (Cullen *et al.* 1992) proposes a Global System for Mobile communication (GSM) space segment because of the propagation problems associated with low elevation angle mobile communications, the provision of high minimum elevation angles by the space segment is very desirable. The GEO orbit does not provide this high elevation angle coverage of Europe and so would require high system link margins. The use of Lower LEO/MEO (Low Earth Orbit/Medium Earth Orbit) orbits would overcome this problem due to their lower free space loss but because of the high numbers of the satellite required, constellation efficiency for European is low. Highly Elliptic Orbits (HE0) can be used to provide optimum coverage of mid-latitude regions. It is concluded that two compatible systems will compliments each other and allow for the quick, effective and complete coverage for ISDN mobile communication services.

Shave (2002) describes the advantages and challenges to achieve satellite system interoperability for 2.5 and 3 GHz. It is stated that a key feature of the Mobile Packet Data Services (MPDS) network is the satellite access protocols, which dynamically allocates bandwidth to a mobile only when data is sent. The interoperability within the signalling plane is important obstacle to gain full interworking between networks. This study also discusses that the interoperable system should support a wide range of subscribers, high-data rates, better call and data charges/cost. Therefore, there is a requirement of roaming services between different terrestrial and satellite networks.

Athanasopoulos et al. (2006) take into account the problem of interoperability among heterogeneous types of services on Web (Cheng *et al.* 2009), Grid (Kertesz and Kacsuk 2009) and P2P (Rezende *et al.* 2009) architectures. A generic service model has been developed that can leverage interoperability among Web, Grid and P2P services. An architecture that was selected for the development of the Generic Service Model (GeSMO) is a layered one. It is recommended that difference on the intended clients has an impact on the signature and protocol levels as well as on the platform and application dimensions of service interoperability. However, the nature of impact has not been elaborated in this study.

Jenson (2003) conducts a study for interoperability of wireless communication technologies, devices and tools for interoperable mobile wireless internet /web application. Data is collected by visiting the related websites and used email for follow up questions and responses. The cross-examination is conducted among the mobile devices with their respective GSM/GPRS, CMDA 2000, Wi-Fi and Bluetooth technologies. Six major programming tools are compared, namely: Nokia Mobile Internet Toolkit 3.1, Ericsson Mobile Internet Toolbox, Motorola Wireless Studio, Palm Wireless Tools, IBM Web Sphere Everyplace Suite, and Microsoft Mobile Internet Toolkit. It is recommended that web applications interoperable on various mobile wireless communication devices can be developed.

Dibuz and Kremer (2006) present a method that re-uses an existing Interoperability Test Suite (ITS) to produce a Conformance Test Suite (CTS). ITS has been used as a base to write from scratch as compared to a conventional CTS.

An analysis of the two test suites is conducted by comparing them after identifying the common parts.

A study (Evans *et al.* 2005) discusses on integration of satellite communication with terrestrial networks and proposes that there is need for integration of satellite systems with terrestrial systems as satellite systems cannot exist in isolation except in niche areas. This study establishes that Satellite has the edge in air and sea due to wide area coverage and speed to deliver new services but there is a need to integrate satellite-terrestrial technologies so that integrated services are viable, more efficient, and cost effective. This study concludes that satellite communication cannot exist as isolation except in niche area where terrestrial communication is impossible. However, it does not consider any interoperability problems.

Another study (Chitre and Henderson 1995) discusses the integration of satellite and terrestrial networks communication, the role of satellite in future communication, many of its advantages including rapid deployment, remote coverage, access to rural areas, bandwidth on demands and affordable rate etc. Some administrative and technical issues for seamless integration of satellite and terrestrial network such as terrestrial signalling and many national and international standards are also mentioned. This study discusses the low cost terminals, cost-effective broadcast and multipoint services, the role of satellite communication in e-commerce, healthcare, education, and multimedia. Recommendations are made that there is a requirement to develop new standards to overcome these communication barriers. However, the architecture of new solution is not described.

Skinnemoen and Tork (2002) discuss the key technologies that are being developed such as high-power generation, on-board processing, advanced antenna technologies which is space craft antenna specially developing phased array antenna technology and new broadband, multimedia services and applications. Some systems are piloting a combination of Ku and Ka-band technologies, with

Ka-band for the return link and Ku for the forward link. In addition, there are already filings for systems in the Q and V bands, which are in the 40/50 GHz range. The authors have discussed many advantages of satellite broadband including broad coverage, availability and economy. This study lacks the information on microwave frequency bands and channels size. The information on broadcasting and multicasting is very limited.

A study (Salhani *et al.* 2009) focuses on class of service mapping and call admission control procedures in WiMAX/DVB-RCS cooperative system by taking into account and share both satellite terrestrial network systems. This study discusses five categories of capacities request class-of-service based on operator requirement namely Continuous Rate Assignment; Rate Based Dynamic Capacity; Volume Based Dynamic Capacity; Absolute Volume Based Dynamic Capacity and Free Capacity Assignment; however, this study did not consider user's requirement.

Centonza and McCan (2006) propose a hybrid network solution for WiMAX and DVB- RCS. The proposed architecture employs DVB-RCS as a backhaul for content delivery to WiMAX domains. The proposed solution should provide connectivity and mobility, avoiding costly infrastructure and QoS management. However, it lacks discussion on end-to-end interoperability issues and QoS requirements.

Raina (2004) discusses the quality of service to subscribers and a concept of network quality manager tool to monitor quality of experience, quality of service and key quality indicator for users. This study ensures that QoS can be set for each subscriber user group. However, this study does not consider the cost implications for different level of Quality of Services.

Meawad and Stubbs (2006) present the design of a prototype for generic interoperability framework with Virtual Learning Environments (VLEs) for large scale deployment. Results and observations cover the system's usability and its importance for both tutors and students. The authors agree that the interoperability system should be cost effective. However, this study lacks in-depth analysis of interoperability among various learning platforms.

Loguinov and Radha (2002) conduct few experiments which shows end-to-end delays without any problem to real-time application. The end-to-end delays less than 100 ms are enough to support interactive streaming, which is possible with DSL and cable modems. This study suggests that with broadband access at home, the performance of real-time streaming largely depend on end-to-end congestion control in the streaming protocol, retransmission scheme, or delay jitter. However, it does not address the bandwidth requirements, throughput and interoperability issues on Satellite terrestrial hybrid wireless networks.

Yoshida et al. (1999) propose a hierarchical structure with three layers, namely: the network, system and application layers. The prominent feature of the three-layer structure is that in each layer, the introduction of new functionality, modification and expansion of scale are easily achieved. Each layer can share the network‘s components according to specific applications. The merit of this concept is that the user’s viewpoint is emphasized by including contents and their communications in the application layer. However, communications among different networks are not analysed in detail.

Zhu et al. (2006) propose a suitable reliable protocol which is used in the end-to-end (E2E) scheme at link layer. This study is validated by simulation with the ns-2 simulator. The main conclusion is that TCP splitting generally outperforms end-to-end (E2E) scheme but the end-to-end delay is dominated by terrestrial part where buffer size is limited at intermediate node, the E2E scheme is preferred. However, throughput and delay performance for satellite link is not considered in this study.

Most of the previous studies provide advantages, and generic information about integration of satellite and terrestrial networks, few studies have discussed bandwidth, delay, or throughput. A study (Abuelma'atti *et al.* 2006) addresses the interoperability problem in wireless network appliance and considered network appliance interoperability architecture, (Jenson 2003) conducts a study for interoperability of mobile wireless internet. Another study (Chitre and Henderson 1995) discusses the integration of satellite and terrestrial networks communication, remote coverage, access to rural areas, and bandwidth on demand. Loguinov and Radha (2002) conducts few experiments showing large end-to-end delays for terrestrial wireless networks. A study (Al-Gizawi *et al.* 2005) proposed new algorithm for interoperability, another study (Meawad and Stubbs 2006) presents the design of a prototype for generic interoperability framework.

2.4 Analysis and Discussion

Most of the previous studies discuss Wi-Fi, WiMAX, DVB-RCS, SCPC and 3G technologies as well as their standards, operating frequency, bandwidth, data rate, basic configuration and coverage. Some of the studies (Sun *et al.* 2005; Rajasekhar *et al.* 2006) conduct and implement an in-depth evaluation on the effectiveness of Adhoc Probe which is an end-to-end path capacity and address the concept of capacity to hop count ratio, which is used in computing QoS paths.

Jing and Raychaudhuri (2006) investigate the feasibility of spectrum coexistence between the Wi-Fi IEEE 802.11b standard and the WiMAX IEEE 802.16a standard. Ghosh et al. (2005) mention the IEEE 802.16 standard and its advantages, Hoymann (2005) specifies four different Physical (PHY) layers, (Sayenko et al. 2006) present the estimation for WiMAX MAC header overhead. However, a study (Qiang et al. 2007) compares the interference issues and testing results on delay and data rate for WiMAX.

Few other studies (comstream 2005; Chini *et al.* 2006; Kim 2006; Morello and Mignone 2006) discuss the history of DVB-S, DVB-S2, DVB-RCS standardisation trends and their advantages. Anderson et al. (2006) give a technical overview of the various satellite frequency bands. Another study (Song et al. 2006) presents the design which is based on DVB-S/DVB-RCS standard.

Costabile et al. (2004) propose a modular system for QoS management over the DVB-RCS return channel satellite platforms. Lee et al. (2005) have carried out some delay testing for DVB-RCS only.

Few of the studies (Henry and Hui 2002; Hung-Yu *et al.* 2005; Kejie *et al.* 2007; Suitor 2011) discuss power, interference, social issues, security problems, physical and MAC layers design and development issues. Some of the studies provide little more than a narrative description of standards and detail of MAC layers (Hoene *et al.* 2003). Zhu et al. (2006) conduct test for issues such as delay, bandwidth, capacity, data rate, and throughput at individual technology level.

Most of the previous studies regarding interoperability provide generic information about the integration of satellite and terrestrial networks; however some studies (Chitre and Henderson 1995; Jenson 2003; Abuelma'atti *et al.* 2006) have discussed the interoperability problem in wireless network appliance and considered network interoperability architecture, remote coverage, access to rural areas and bandwidth on demands.

Few of studies (Al-Gizawi *et al.* 2005; Meawad and Stubbs 2006) proposed new algorithm for interoperability, design of a prototype and algorithm for generic interoperability framework.

It is evident and clear that, none of the previous studies cover all aspects for end-to-end interoperability issues and QoS requirements such as bandwidth, jitter, latency, packet loss, link performance, TCP and UDP performance, unicast and multicast performance, for Wi-Fi, WiMAX, DVB-RCS, DVB-S/S2, SCPC on Hybrid wireless networks.

Therefore, it is concluded that there is a need for a study which should examine and address end-to-end interoperability issues and evaluate Quality of Service requirements on hybrid wireless networks both satellite broadband and terrestrial wireless technologies. The study should cover all interoperability issues such as bandwidth, delays, jitter, latency, packet loss, throughput, TCP performance, UDP performance, unicast and multicast services and availability.

2.5. Summary

This chapter reviews, discusses and compares the various wireless communication technologies and their associated standards. Wi-Fi, WiMAX, SCPC, DVB-S/S2 and DVB-RCS standards, and their characteristics such as associated modulation and coding techniques, bandwidth, data rate, speed, and frequency were discussed and compared in detail. Comparative results for all these technologies have been summarised.

This literature review describes and discusses the previous research studies for wireless technologies. All of these studies are regarding refer to individual technology information, standards, advantages and disadvantages. Some of the study discusses issues at Physical and Mac layers. Few of the studies raised concerned for bandwidth, delay, jitter, packet loss, latency, throughput, capacity, connectivity, interference, coverage, some aspects of security and other issues at individual level.

While comparing the satellite and terrestrial technologies, the satellite communication has advantages of wide-area coverage and easily deployment but the only satellite communication cannot compete and capture the entire communication market until it is integrated with terrestrial networks due to cost implications, although satellite can still retain exclusive status in the maritime and aeronautical markets due to its distinctive coverage feature. As the demand for broadband and multimedia services is currently growing rapidly, satellite technologies need to be integrated with terrestrial to become more efficient in their delivery and enable the exploitation of new services in densely populated big cities and rural areas with spread population. Satellite communication systems are quite costly compared to terrestrial wireless communication systems. Therefore, it is imperative to use terrestrial where it is possible, and provide interoperability with Satellite systems. To provide full coverage in all these areas, suitable hybrid network access architectures can be developed by integrating satellite and terrestrial wireless technologies.

In order to meet to the need for communication “anytime, anywhere with anyone”, the end-to-end interoperability in wireless communication technologies in hybrid networks is becoming a critical issue for new research in this area. Interoperability of wireless communication technologies in hybrid networks is likely to be an attractive scenario in the near future for many designers, users and operators.

Satellite and Terrestrial technologies inherently have different characteristics, such as delay, jitter, packet loss, latency, quality of service, throughput and availability. Therefore, the end-to-end interoperability issues and QoS requirements of wireless technologies in hybrid networks present a number of challenges. Many different studies agree that standardisation plays a key role in ensuring interoperability in a hybrid scenario where different technologies, networks and services coexist.

It is also acknowledged by many researchers that interoperable systems should result in broad coverage, higher volumes of traffic, better quality of service and lower cost. Different operators and users are experiencing an increasing demand of interoperable solutions in order to fulfil their requirements of flexibility in deployment of their infrastructures, possibly using equipment provided by different vendors and integrating different technologies. For wireless technologies in hybrid networks, all interoperability issues and QoS requirements as well regulatory regimes are required to be taken into account and to be addressed so that different wireless technologies should interoperable without any special efforts. To achieve these objectives an in-depth study, investigation, testing and evaluation have been carried out for bandwidth requirement, causes of delays and jitters, latency, packet loss, throughput, availability, TCP, UDP and link performance. Due to a lack of previous research on end-to-end Interoperability of emerging wireless technologies and QoS requirements, this research focuses on end-to-end interoperability issues and QoS requirements, proposes a testing methodology using common software test tools to assess bandwidth, delays, jitter, latency, packet loss, throughput, multicast and unicast services, TCP and UDP performance, link performance and availability testing. The testing was conducted using various live hybrid network scenarios and nodes. The test bed of this study was the Broadband Access Satellite Enabled Education (BASE²) project which was deployed to supports the different modes of learning such as live virtual classroom, video conference, offline asynchronous learning, collaborative learning, individual learning, educational content generation, management and delivery over different network technologies to a large number of sites for isolated agrarian communities in Greece and Cyprus as well as maritime communities on ships. The end-to-end broadband infrastructure was an integration of broadband terrestrial networks with satellite broadband technologies using DVB/DVB-RCS, SCPC (VSAT), Wi-Fi, and WiMAX.

The CLIX management and delivery frameworks and Isabel application were used in this study are discussed in Chapter 3. The test tools are discussed and compared in Chapter 4. The test plan and test bed network Architecture are discussed in Chapter 5. The testing was carried out and results are shown is Chapter 6. The results were analysed for Interoperability issues and QoS requirements such as bandwidth, delays, jitter, latency, packet loss, throughput, multicast and unicast services, TCP and UDP performance, link performance and availability testing. Recommendations are given in Chapter 7 on the design and implementation of interoperable hybrid wireless networks.

Chapter 3

Network Architectures, Quality of Service and Applications

3.1 Overview

Computer networking communication is mainly based upon a protocol suite and the OSI model. There are different protocols at each layer from physical to the application layer. The application, presentation and session layers are the upper layers of the OSI model. Software in these upper layers performs application specific functions such as data formatting and connection management. The transport, network, data link and physical layers are lower layers which provide more system specific functions, like routing, addressing and flow control (Howes and Weaver 1989). The functionality of each layer is shown in figure 3.1.

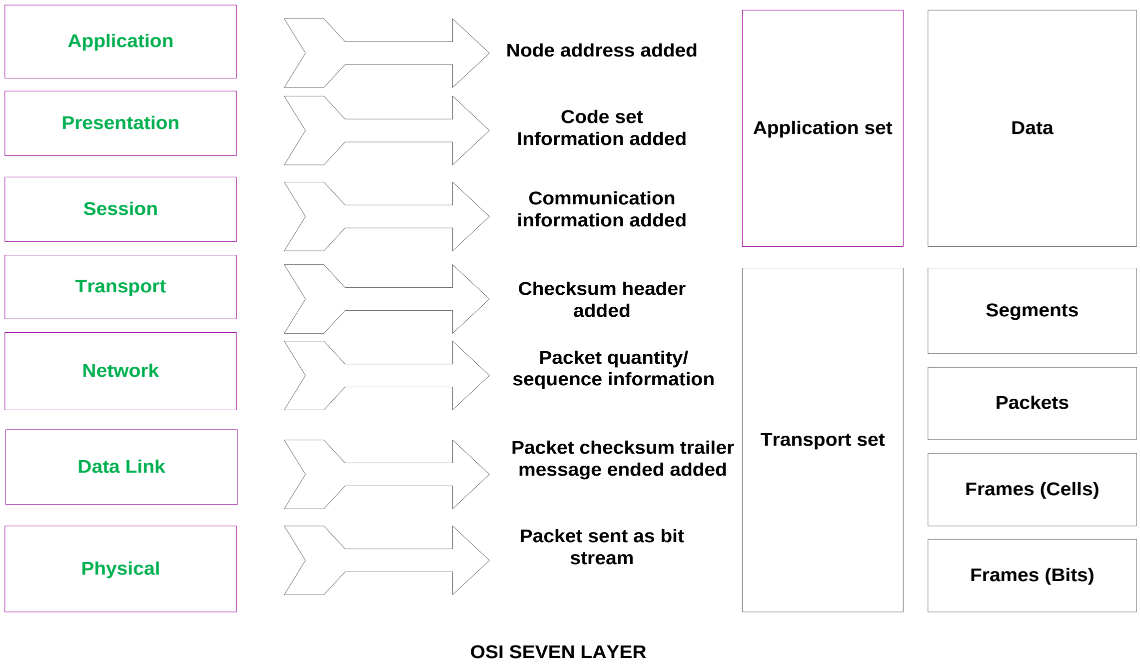


Figure 3.1: Overview of OSI Seven Layer Model

The aim of this research is to study interoperability issues and Quality of service requirements in Hybrid Wireless Networks. Interoperability issues and QoS requirements are the major issues and challenges among various technologies. Figure 3.2 shows the interoperability of OSI seven layers.

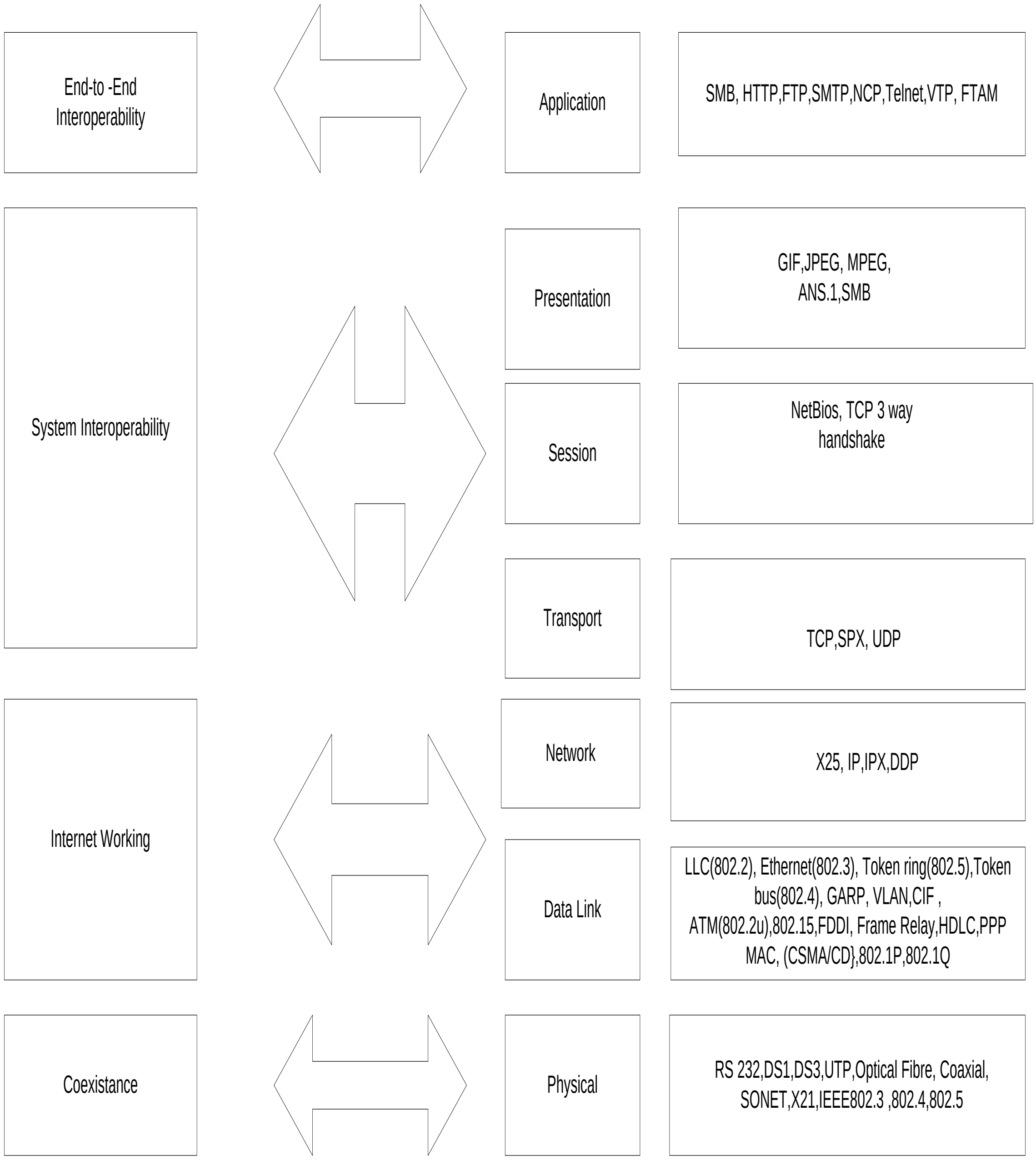


Figure 3.2: OSI Seven Layer Interoperability

Interoperability from ETSI's technical committee TISPAN (Veer and Wiles 2008) Page-5 is defined as " the ability of equipment from different manufacturers to communicate together on the same infrastructure (same system), or another while roaming". In other words, when two or more entities are engaged to perform a specific task effectively by behaving accurately as in our case communication protocol is called interoperability or Interoperability is the ability of two systems, to interoperate using the same communication protocols.

Al-Gizawi *et al.* (2005) defines “interoperability as the capability of a heterogeneous network to support seamless mobility (roaming) between different access radio technologies, while maintaining user’s minimum QoS requirements”.

Sugarbroad (1990) defines “Interoperability is the ability to provide successful communications between end-users (a service provider is considered as an end-user) across a mixed environment of different domains, equipments, facilities, and networks from different manufacturers and providers”.

Veer and Wiles (2008) defines all types of interoperability such as technical interoperability, which is usually, associated with hardware/software components, system and platforms interoperability which enables machine-to-machine communication.

Syntactical interoperability (Jing *et al.* 2007) is usually associated with data formats. Semantic interoperability (Abdalla 2003) is associated with the ideas and concerns of human. Organizational interoperability (Rauffet *et al.* 2009) is the ability of organizations to communicate effectively and transfer data using a variety of information systems over different instructions, possibly different geographic area and cultures.

The above studies define various types of interoperability therefore the different devices and systems while working on various standards and protocols may have end -to- end Interoperability issues and QoS requirements. Interoperability should be studied at different layers, including physical, data link, network, transport, session, presentation, application layer or any custom layer or sub layer or end-to-end between various technologies and systems.

The figure 3.3 shows the protocols Architecture and comparison of TCP/IP, 3GPP2 and WAP Suites and its interoperation within the OSI layers. TCP/IP Protocol is based on five layer model as compared to others model which are mostly based on seven layers. This means that there are variations of different protocols/layers.

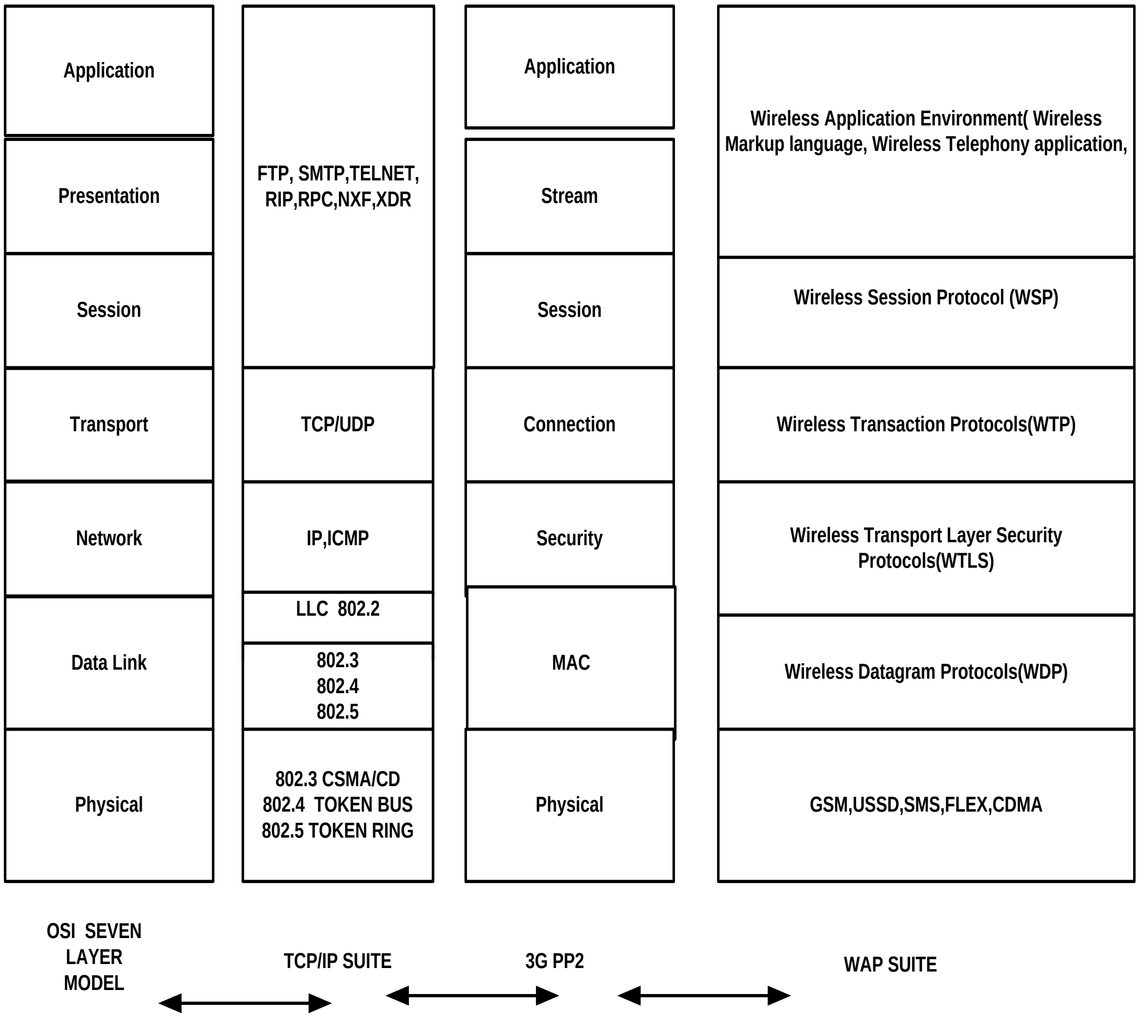


Figure 3.3: OSI, TCP/IP, 3GPP2 and WAP SUITE Model

This Chapter is organized as following;

Section 3.2 reviews the interoperability of wireless communication technologies (DVB-RCS, WiMAX and Wi-Fi). Section 3.3 describes QoS in Wi-Fi, WiMAX, DVB-RCS, DVB-S/S2, and SCPC. Section 3.4 discusses hybrid network communication services such as virtual classroom services for learning, Learning Content Management Services (LCMS), teleconference service and webinar/webcast services. Section 3.5 presents the methodology for consideration and describes various applications and platforms such as Skype, MSN Messenger, NetMeeting, Clix, Isabel and collaboration tools such as Remote Desktop Publishing and Virtual Network Connection. Section 3.6 gives an overview of multicast and unicast services and finally Section 3.7 gives summary of this chapter.

3.2 Interoperability Review

Hybrid wireless technological advances have rapidly growing in recent years. As a result of this, cost has decreased but the proliferation of networked devices has been increased, including its infrastructure particularly Wi-Fi and WiMAX in our daily lives. Some of research in this area is listed as follows: Personal Area Network Connectivity (Negus *et al.* 2000), IrDA Standards (Siep *et al.* 2000; Williams 2000), IEEE 802.11 specifications (IEEE 802.11J, 1999; IEEE 802.11 2010), IEEE 802.16 specifications (Dekleva *et al.* 2007; IEEE 802.11 2010), Bluetooth specifications (Bhagwat 2001; Bluetoothspefication 2001; Ferro and Potorti 2005), 3Gpp2 specifications (Veltri *et al.* 2006; ThirdGeneration(3GPP2)), 3G standard (Goodman and Myers 2005), and Home RF Communication Standards (Negus *et al.* 2000). These standards are used as guidelines. However, there is a limited research for the end-to-end interoperability issues and QoS requirements on hybrid wireless networks (satellite and terrestrial).

One of the earlier interoperability studies is conducted by (Cullen *et al.* 1992), who proposed a space segment which provides Global Standard for Mobile Communications (GSM)-compatible, mobile communications in the European region. This has been discussed in Chapter 2.

A study (Yue *et al.* 2008) investigates Cyclic Delay Diversity (CDD) transmit diversity scheme on Digital Video Broadcasting Handheld (DVB-H) networks and proposed improvements in terms of QoS. Alam and Wu (2007) provide end-to-end delay measurement analysis for instant messaging relay nodes. This study demonstrate SIP message problem for congestion control only and proposes a new model. Another study (Thompson 2011) discusses the next generation virtual worlds.

Serif and Ghinea (2005) verify that interoperability and coexistence of different communication technologies can still be a challenge for the research Community. The results of these studies emphasise the need for applications and infrastructures that are easy to set up and manage.

As (Abuelma'atti *et al.* 2006) Architecture confirms that the interoperability could be investigated at various levels and layers of the systems and devices. Therefore, our aim is to study end-to-end interoperability issues and QoS requirements such as delay, jitter, latency, packet loss, bandwidth, throughput and other issues including application communication protocol mismatch in hybrid wireless networks. The layered reference model for Wi-Fi, WiMAX, SCPC, and DVB-RCS protocols suite is presented in figure 3.4. Some of the layers of these models are similar to the layers of OSI reference model. The following figure shows the relationship to the OSI reference model.

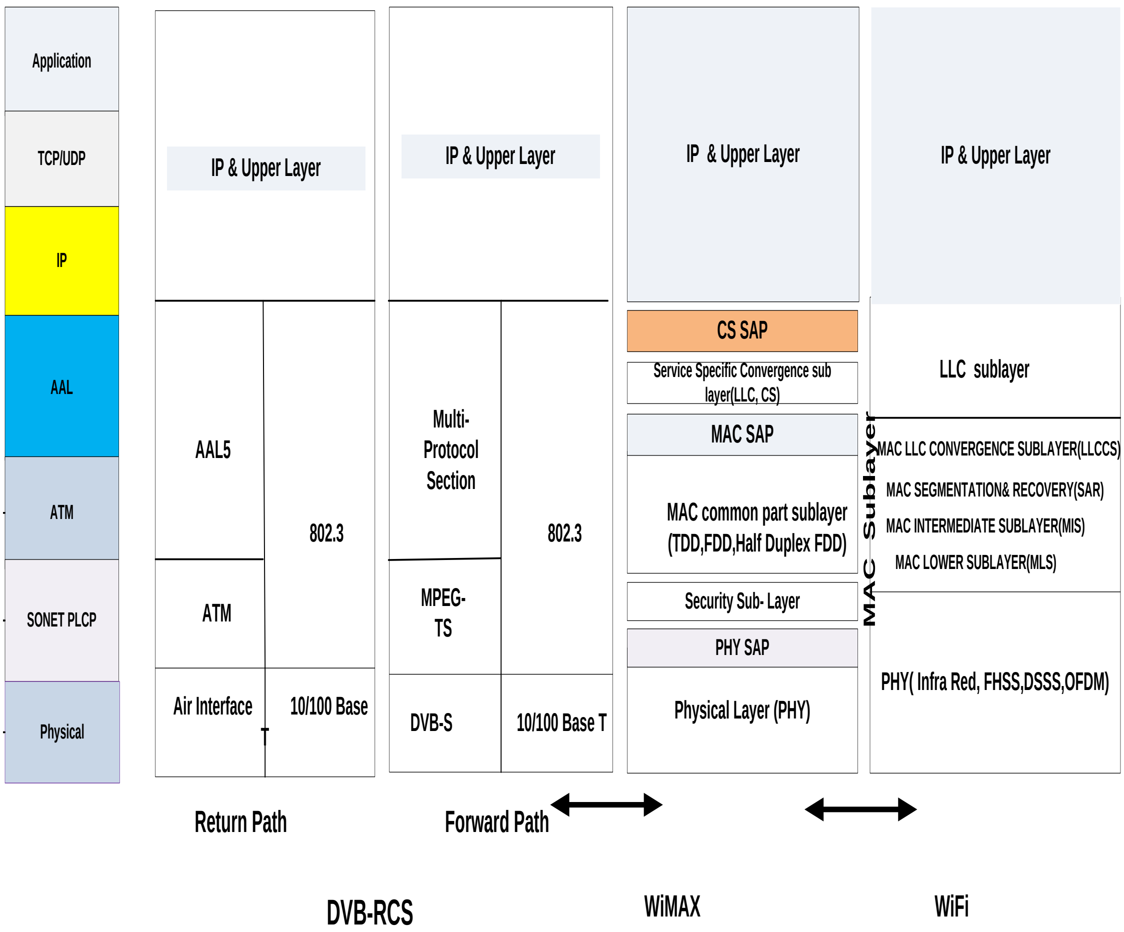


Figure 3.4: DVB-RCS, WIMAX and Wi-Fi Protocol Architectures

3.3 Quality of Service requirements

The integration of Satellite and Terrestrial wireless networks is normally evaluated on the basis of Interoperability, Quality of Service (QoS), availability and cost. QoS refers to the ability of networks to provide superior services to selected network traffic using various technologies and to prioritize different applications to a certain level of performance. The Real-time applications have inherently strict QoS requirements and real time operation is vital for multimedia applications such as audio and video conference which require specified bandwidth, delay and jitter guarantee (Romdhani *et al.* 2003).
QoS provision in hybrid Satellite-Terrestrial wireless network is a challenging issue. QoS in wireless networks is very complicated and difficult to predict (Papadimitriou *et al.* 2010).

The Satellite system has several inherent constraints such as bandwidth, long congestion and delay. The Satellite often suffers round trip time delay (RTT) and path losses.

The goal of QoS is to provide bandwidth to avoid congestion, control jitter and latency, manage queuing, and priority of traffic required by real time and interactive applications. Unfortunately adding bandwidth is not a lasting solution. There are other factors to manage, such as bandwidth, end-to-end delay, delay jitter, latency, packet loss and throughput to achieve required QoS. The bandwidth usage particularly depends on the type of the traffic/protocols used for data. Some protocols, which have bigger header size, use more bandwidth as compared to other types of protocols.

The future requirements from wireless networks is to provide pervasive ubiquitous coverage across diverse technologies offering a wide range of services with variable bandwidth and QoS, anytime, anywhere with anyone (Evans *et al.* 2005). However, the interoperability of hybrid wireless networks imposes a number of challenges which affects QoS. Differences in QoS properties between Satellite and Terrestrial wireless networks and applications have a considerable effect on the quality as well as network interoperability. In ideal conditions throughput should be guaranteed and there should be low delay, jitter, and packet loss.

There are two major categories of QoS requirements, user level QoS and network level QoS requirement. The user level QoS is subjective, such as speed, audio loudness level, interrupts, cross talk, echo, the responsiveness of voice or sound, quality of streaming audio, video, smoothness, cost, security, confidentiality and authentication etc. This is by and large known as Quality of Experience (QoE) and Quality of Perception (QoP). This is also called Total Customer Experience (TCE), which measures human user opinion of data, voice and video quality. The network level QoS requirement is objective and quantitative, comprising characteristics such as bandwidth, delay, jitter (delay variation), latency, throughput, reliability, priority, Bit Error Rate, response time, signal to noise ratio, packet losses and flow of data rate for Internet Protocol (IP), Session Initiation Protocol (SIP), File Transfer Protocol (FTP), Hypertext Transfer Protocol (HTTP), performance of Transmission Control Protocol (TCP), User Datagram Protocol (UDP) and network availability.

The QoS parameters can be measured at the packet level, session level, connection level and network level. It is possible to prioritise and provide different level of services for different class of applications in emerging 4G and WiMAX technologies and standards.

In wireless hybrid networks, end-to-end QoS requirements are very diverse, for promising applications, such as real time voice, video and teleconferencing. There are various factors which affect performance such as packet loss, out of order packet, duplicate packet, queuing, network congestion, fading, weather condition, noisy environment, throughput, latency, delay, jitter and error rate. The end-to-end packet loss can be caused by delay, congestion or the erroneous loss in the wireless part (Hao *et al.* 2005). In satellite communication, the packet loss is due to several reasons, including jitter and noise in the satellite channel itself (Viswanth and Obraczka 2006).

The capacity of the network also affects performance. The greater the capacity, the more likely greater performance, although overall performance, also depends on other factors, including latency (Jain and Dovrolis 2003). Latency describes the normal processing time from the time the packet is received from a source until the time the packet is forwarded to another destination (Sharma *et al.* 2006). Typical latency for GEO (Geostationary Earth Orbit) one way is 275 ms from earth station to earth station. Excessive latency makes it difficult for interactive or real-time applications such as video conferencing, and remote instrument control.

The end-to-end QoS functionalities are QoS control, resource management, and bandwidth on demand (Crescenzo *et al.* 2008) and end-to-end reliability which may not, always be required (Fabrice Arnal *et al.* 2008).

The end-to-end QoS is a challenge in hybrid network for unicast, multicast and broadcast services and the requirement is that the different networks have to be available and reachable.

QoS and availability are essential aspects to consider. Availability means that the system is ready for immediate use. There are several approaches to increase the availability in network design and redundancy plan for systems or components failure, such as Virtual Router Redundancy Protocol (VRRP) and IP Multipathing (IPMP) used for fault tolerance and local spreading for network interface cards (NICs). Miloucheva et al. (2009) studied various approaches on QoS advances for mobile applications and proposed advance resource allocation architecture to provide seamless handover for QoS aware applications. Sudhaakar et al. (2009) proposed a distributed MAC scheme to provide QoS by retransmitting each of its frames over a number of times.

Mean Time Between Failure (MTBF) is a unit to measure reliability (Cisco 2009). Reliability analysis is based on calculating the network availability, accuracy, and the causes of any network outages. The quantitative aspects commonly used are called Mean Time To Failure (MTTF), Mean Time to Repair/Recovery (MTTR) and Mean Time Between Failure (MTBF).

Network service availability is defined as any fraction of time the network is available from a specified group of measurement agents to a specified group of test points. In addition, a node is considered to be reachable from a measurement agent if that agent can send packets to the node and, within a short, predefined time interval, receive acknowledgment from the node that the packet was received. For example, if each measurement sample consists of multiple pings, the test point is considered reachable from the measurement agent if the latter receives at least one acknowledgment from the test point. In satellite service, availability is usually less as compared to the terrestrial network, particularly due to inherent delay, less capacity and weather conditions.

In general, IP connections are calculated towards an overall average-year availability of 99.9 percent or higher. This means, one should expect a maximum outage of 8.5 hours in an average year. Outages are in general affected by broken parts, software instabilities or weather influence.

The weather is the critical factor in some countries due to the heavy thunderstorms. Therefore, relation to rainfall, some countries need a link margin, which is at least 3 dB higher, i.e. if a margin of 5 dBs is required in Central Europe, then at least 8 dB margin is required in Greece to obtain the same availability figures.

The need to study and analyse QoS in various networks is derived by several factors. Most of these factors point to a gap in the design and un-expected growth of these networks and their utilization (Cisco 2008).

- *Growth of load and needs unexpectedly.*
- *Strict control is needed on the planned as well as newly arise requirements.*
- *To identify and quarantine un-wanted traffic.*
- *To set up a policy of priorities as not all current and future requirements are not of the same importance.*

Service prioritization for critical user, customer and application is essential as compared to non-critical users, customers and applications.

Many critical applications such as Enterprise resource planning (ERP), voice over Internet Protocol (VOIP) and remote application servers, require guaranteed bandwidth.

There are few algorithms, which have been proposed for congestion avoidance by providing buffer control, and allowing TCP to obstruct or throttle back before buffers are shattered. For example Cisco has proposed the Weighted Random Early Detection (WRED) algorithm (Ming-Jye *et al.* 2008) for congestion and buffering management.

A study (Xylomenos *et al.* 2001) discussed the TCP/IP protocol in wireless links and reviewed the wireless link characteristics (Libnik *et al.* 2010). Another study (Alam and Wu 2007) proposed the use of Session Initiation Protocol (SIP) for congestion aware handover in the heterogeneous network which is based on signal strength and network status which is not dependent on access technology but this has been used to maintain Quality of Experience (QoE) for VOIP. In all cases, most of the work was done on the physical and network layer, but still lot of research work is required to improve the end-to-end Quality of Service (QoS) requirements for hybrid wireless networks. The following section describes the QoS for Wi-Fi, WiMAX, DVB-RCS and SPCS.

3.3.1 QoS in Wi-Fi

Wi-Fi is designed for indoor short range communication for high data rate. The Wi-Fi and its associated standards have been discussed and compared in chapter 2. Wi-Fi is based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) technique (Peng and Cheng 2006) which manages the transmission of data over the WLAN to avoid multiple computers transmitting at the same time.

The important technique of Wi-Fi ’s access are Distributed Coordination Function (DCF) protocols (Laddomada *et al.* 2010) and Point Coordination Function (PCF) (Der-Jiunn and Hsu-Chun 2005). The DCF is based on the CSMA/CA which attempts to present collisions by using explicit acknowledge (ACK). The IEEE 802.11-based Wireless Local Area Network (WLAN) was not originally designed to support delay-sensitive traffic. DCF mode in the IEEE 802.11 standard delivers Best Effort (BE) service (Alturki *et al.* 2009) and there is no guarantee for service in terms of delay and bandwidth. This is suitable for data and other non-real time traffic. The access point grants access to the medium to an individual station by polling the station during the contention free period by using

PCF which is described as an optional in the IEEE802.11 standard. Initially the 802.11 network was not suitable for real time application.

The QoS in Wi-Fi relates to bandwidth (Sameh *et al.* 2010) with maximum rates specified in IEEE Wi-Fi 802.11 standards. The IEEE 802.11 MAC layer provides Cyclic Redundancy Code (CRC) and packet fragmentation which improves its performance. Several extensions to IEEE 802.11 standards have been added to the enhancement of QoS in WLAN but still improvement in QoS is required. The IEEE 802.11 a/b/g standards do not provide any specific control over resource allocation and default policy might not suitable for hybrid wireless networks. Both IEEE 802.11 b/g do not address the congestion problem, as there is no admission control and bandwidth control to maintain high quality video link. When there is congestion, jitter occurs and as a result packets are lost (Cunningham *et al.* 2009). Wi-Fi packets are variable in length with payload ranging from 0 to 2304 bytes (Ferro and Potorti 2005). Different headers and frame sizes are used in 802.11 standards.

The IEEE 802.11e standard uses three different frames and headers with 4 Byte Cyclic Redundancy Check (CRC) for error detection (Tzu-Chieh and Ming-Ju 2005). This standard introduces Enhanced Distributed Channel Access (EDCA) and Hybrid Coordination function controlled Access (HCCA) to provide differentiated level of QoS according to access categories.

The IEEE 802.11e standard implements QoS mechanism at some extent to allow real time audio and video traffic, instead of best effort. IEEE 802.11e provides an enhancement to the MAC layer's EDCF to prioritize traffic and addresses the QoS requirements (Yuxia and Wong 2006).

There is dual speed mode Wi-Fi available which is based on fair time, not fair access, but this doesn’t solve all QoS issues. The 802.11n is a new emerging standard based on Multiple input Multiple output (MIMO) (Yuxia and Wong 2009), which provides more than ten times throughput and higher data rate (maximum up to 600 Mbps), as compared to IEEE 802.11 a/b/g, and reduced fading (Paul and Ogunfunmi 2009).

Various studies have been conducted to address QoS of Wi-Fi and allocation of resources issues. Vittorio and Bello (2010) propose a solution called contention window adapter, a technique which reduces the number of collisions to use the channel more efficiently for real time traffic. Heusse et al (2003) mentioned that the heterogeneous data rate can affect the throughput, but authors don't provide any definite solution. Mishra et al. (2003) studied handoff process at link layer of 802.11 and observed that latency can affect the QoS for many applications.

A study (Tianji *et al.* 2006) pointed out that the overhead of MAC is the basis of inefficiency of MAC and addressed this issue by designing new MAC scheme called "Aggregation with Fragment Retransmission (AFR)" by aggregating packets from upper layer to frames. In all of these studies, there is no such study which addresses end-to-end QoS requirements.

The Wi-Fi used in our this study for testing of QoS requirement supports up to 108 MBit/s upload/download channels.

3.3.2 QoS in WiMAX

The WiMAX standards (IEEE802.16-2001 2004) defined well the QoS (Alavi *et al.* 2005) and proposed several QoS mechanisms for guaranteed services for data, voice and video to meet strict QoS criteria such as delay, jitter and throughput. The WiMAX IEEE 802.16a standard relies on a request protocol to access. The protocol employs Time Division multiplexed (TDM) data streams on the downlink (DL) and Time Division Multiple Access (TDMA) on the uplink (UL) which supports delay sensitive traffic.

The WiMAX IEEE 802.16-2004 standard provides mainly QoS functionality. In IEEE 802.16e, additional QoS has added to support real- time applications with variable bit rate. The WiMAX IEEE 802.16 standard provides broadband and real-time multimedia services with flexibility and QoS at a satisfactory level (Fen *et al.* 2009) for increased mobility and seamless roaming. WiMAX architecture supports point to point, point to multipoint and ubiquitous coverage which is defined in MAC protocol layer by 802.16 standards.

In WiMAX each user is given a provision for QoS allocated as a part of user profile. Once, the user is connected with WiMAX, Authentication, Authorization, and Accounting (AAA) server provides QoS as per parameters of user profile (Adibi *et al.* 2006). Each traffic can also be assigned as specific QoS characteristics which allow efficient traffic management specifically for video calling.

A study (Qiang *et al.* 2007) compares two bandwidth request mechanisms, namely random and polling, and shows that both mechanisms have merits and demerits, for example random access outperforms polling when polling request is low, but performance degrades when channel is congested. Another study (Vinel *et al.* 2006) investigated, the efficient bandwidth request mechanisms for IEEE 802.16 using modelling and compared the efficiency of the polling scheme to decide overall system performance.

In IEEE 802.16, 2009 multicast services at medium and physical layer provides a better framework for multicasting data to terminals using multicast connection and achieves higher data rates (IEEE802.11 2010). Multicast and Broadcast services are added to mobile WiMAX based on IEEE 802.16.2009 air interface standards (Zhang *et al.* 2010) which provide efficient mechanisms to send content to multiple users by using shared radio resources. The IEEE 802.16m standard was proposed to provide more capacity, reliability, support coverage, throughput, latency, handover, mobility up to 350 km/hour, bandwidth and efficient resource management with better QoS in mobile environment (Papapanagiotou *et al.* 2009; Ahmadi 2011).

The five QoS categories are included such as UGS (Unsolicited Grant Service), rtPS (real time Polling Service), nrtPS (Non real-time Polling Service, Extended rtPS and BE (Best Effort) (IEEE802.16 2006). The rtPS and BE are common service. The rtPS supports real time service flows while, Best Effort there is no priority for service and available on required basis.

The OFDMA multiple access technique used in WiMAX is based on time and frequency; therefore multiple users can use separate sub channels for multiple access for QoS support. WiMAX allows for an allocation of QoS-resources on a per flow basis. WiMAX competes and complements both Wi-Fi and 3G in range and data respectively. The WiMAX used in the test bed for our study during testing supports up to 70 MBit/s upload/download channels.

3.3.3 QoS in DVB-RCS and S/S2

Digital Video Broadcasting (DVB) is the major standard for audiovisual and data transmissions by satellite, cable and terrestrial means (Reljin and Sugaris 2009). It defines how MPEG-Packets, which contain payload, are transmitted and modulated. DVB is the backbone of digital television programme distribution, mainly due to the possibility to offer cost effective receivers to the mass market.

For the main three different media, three different modulation approaches have been available since 1995, which became necessary due to the variable noise and transmission environments. In DVB-S, the QPSK scheme (Cardarilli *et al.* 2006) is used for modulation, mainly due to two non-linear amplifiers (Uplink: Earth station, Downlink: Satellite) which make it rather difficult to use higher modulation techniques, i.e. amplitude dependent ones, with the same Error Area Margins for all transmitted symbols at the same time. In QPSK, two symbols are transmitted per time frame.

To produce a low BER with enough margins, to cope with e.g. varying weather conditions, Forward Error Correction is added. This is necessary because the signal is mostly below 10 dB Eb/No when using low gain antenna systems such as VSATs / DVB-RCS terminals. DVB's FEC consists of two code types, an outer code calculated with Reed Solomon (204,188) which is similar within each DVB transmission type and an inner code, which is a Viterbi stream with Rates 1/2 to 7/8, in general. While transponder units on satellites are often 27, 33 or 36 MHz wide, transmitted symbol rates of 22000 kSymbps or 27500 kSymbps for a DVB-S multiplex are very common. Generally, MPEG-2 compressed television channels produce between 2 and 6 Mbits/s throughputs, the range for radio channels is between 64 kbps and 384 kbps. Considering a net rate of 38.015 Mbps for a standard TV multiplex of 27500 kSymbps, it seems obvious that several digital TV, and Radio programs can be transmitted over such a multiplex channel. Therefore, the cost of the transmission for a digital TV station has dropped significantly since the switch from analogue to digital. Analogue modulated satellite transponders were able to host only one program, a digital multiplex carries up to ten channels.

In comparison with regular TV channels, one HDTV program requires a minimum of 15 Mbps per channel, i.e. more than two-times the bandwidth in a multiplex. Hence, only two HDTV programs could be put into a DVB-S multiplex, which increases the price per channel ratio on the transponder dramatically.

This has been a key driver for the development of DVB-S2 (Morello and Mignone 2006). While DVB-S is bound to QPSK, the maximum transmission rate on a 36 MHz transponder cannot exceed 60 Mbps. DVB-S2 offers, for the same MPEG-2 transport stream, higher modulation techniques that produce significant higher throughput rates. In comparison with the circumstances that led to QPSK in the mid-90s with fairly low expected Eb/No values, satellite peak EIRP was raised by nearly 10 dB for national beams or by approximately four to eight dB for pan-European or continental footprint areas, mainly due to the availability of heavy payload launchers and higher solar array efficiency for larger on-board-power availability. This factor leads directly to higher Eb/No reception values per transponder. An additional increase of the margin was introduced by the development and integration of Turbo coding.

While analyzing the disadvantages of the Viterbi algorithm with its random burst error behaviour due to its stream flow characteristics, it became necessary to frame its structure and fix initial conditions to achieve a significant gain in BER versus C/N or Eb/No values. In 2000, when Turbo codes were finally integrated into satellite modems, missing standardization and cross-company compatibility forced every turbo code integrator into its own niche market. Even more, different models of satellite modems of the same manufacturer did not work with each other while having turbo codes enabled.

DVB-S2 specifies, in parallel with QPSK as downward compatibility, 8PSK, 16APSK and 32APSK modulation techniques which equals three, four or even five Bits per Symbol, leading to a maximum data transmission rate of well above 100 Mbps per transponder. After a simulation of seven different options including several turbo code alternatives, the technical DVB committee decided to use Low Density Parity Check (LDPC) as inner FEC code. As outer code, Reed Solomon was replaced by Bose-Chaudhuri-Hocquenghem (BCH) coding, mostly because of the complexity of LDPC and the significant simplicity of BCH in comparison with Reed Solomon in keeping the processing requirements as low as possible. Both coding algorithms are concatenated in the same manner as it was with Viterbi / Reed Solomon in DVB-S before.

The LDPC code is defined for different code rates with a block size of 64800 that allows a transmission to the theoretical limits. There are around 300000 messages are processed for DVB-S2. These immense data processing and storage requirements are a challenge for the decoder hardware, which fulfills the specified throughput of 255 MBit/s for base station applications.

DVB-S2 offers additionally VCM (Variable Coding and Modulation) and ACM (Adaptive Coding and Modulation) as bidirectional communication methods to provide reception information from the receivers. Chipsets for these DVB-S2 add-ons are currently in production and testing. On one hand, broadcast services, which are covered today by DVB-S, have added flexibility when VCM enables different levels of protection for each service. There are backwards compatible broadcast services (BC-BS) for added interoperability with DVB-S decoders and more optimized NBC-BS, i.e. non-backwards compatible broadcast service.

The interactive services are being designed to operate with existing digital video broadcasting return channel standards (e.g.DVB RCS, SCPC). The DVB-S2 standard operates in constant coding and modulation (CCM) and Adaptive Coding and Modulation (ACM). ACM enables each receiving station in the system to control the traffic addressed to it on a frame-to-frame basis. This is an important development inside DVB-S2. Hence, due to DVB-S2 ACM; the efficiency of the DVB carrier is significantly raised.

Point-to-Multipoint star topology network data transmissions, like DVB-RCS operations, rely mostly on a large hub station antenna and several hundreds of small VSAT systems. The reason for such a topology structure is economy because of the price per remote terminal. In general, for KU-Band, hub station antennas are 3.7 meters or bigger while VSAT reflector sizes vary between 98 centimeters to 2.4 meters in diameter, depending on the required link availability numbers, weather influence and technical satellite data.The DVB-S system used during our study test and trial uses 8 shared MBit/s DVB-S forward channel and supports five Shiron VSAT systems with up to 384 kBit/s return link per station.

The DVB-RCS standard is a bidirectional satellite communication system. DVB-RCS is designed to support all IP applications effectively and also supports QoS at system as well as terminal levels. Each terminal can have several virtual channel assignments with different QoS parameters. QoS-provision on the shared RCS upstream link while keeping the latency low is an inflexible problem. The DVB-RCS specification defines a variety of bandwidth allocation mechanisms. Traffic prioritization is available with some DVB-RCS vendors (IEE802.11e 2005).

The DVB-RCS standards support different capacity requirement categories, which are, Rate Based Dynamic Capacity (RBDC); Continuous Rate Assignment (CRA); Free Capacity Assignment (FCA); Volume Based Dynamic Capacity (VBDC) and Absolute Volume Based Dynamic Capacity (AVBDC).

DVB provides overall bandwidth and data throughput in the Forward channel in the range of 1.5-45 Mbps. For each Return channel, this value is 0.016-2048 kbps, equalizing to 0.02-2.2 MHz over a 0.98-2.4 m antenna transmitting at 1-4 Watts. DVB-RCS has scalable bandwidth generally from 64 kbps to 8 Mbps on the return link. The Hub generally utilizes 3.7-9 m antenna. The DVB-RCS system used for testing in this study was configurable up to 45 Mbits/s downstream and up to 2 Mbps upstream.

3.3.4 QOS in SCPC

In SCPC connections with VSATs, a satellite modem is connected with another one in a point-to-point link with a service provider, comparable with a terrestrial telephone modem connection. Since data rate, frequencies and code rates are fixed and cannot be changed in the satellite system unless someone is told to do so, the transmission rate is rather inflexible and largely unused, but also at guaranteed rates for the paying customer when necessarily needed. It is the most expensive use of satellite communication since both carriers cannot be shared with other users, but for many companies, it is the only option for them to stay in contact with a very remote site that is not covered with a DVB-RCS provider.

VSAT satellite communication consists of TDMA-based hubless networking with relatively large antenna systems with average power transmitter ratings. Classical satellite Quality of Service features apply for bidirectional satellite communication; however, due to its TDMA structure, buffering prior to transmission might occur when transmission slots are not available at that particular point in time due to lower prioritization.

In general, system load and adverse weather conditions can lead to the same situation. Modulation schemes vary from QPSK to higher QAM or APSK, as well as coding schemes from Sequential, Viterbi to Turbo Codes. Hence, different availability numbers are achievable in comparison with modulation, coding, bandwidth, dish sizes in the network, satellite parameters and weather conditions. The VSAT system used in our study for testing supports 2 MBit/s upload/download channel.

3.4 Hybrid Network Services

There are various software applications with different protocols, available to use for audio, video, web conferencing, whiteboard, VOIP, application sharing, application remote control and instant messaging. In this sub section, different services such as virtual class room, learning content management service, teleconference services and webinar/webcast services are discussed.

3.4.1 Virtual Classroom Service

A virtual classroom provides a distributed learning service environment at any time, any place with any pace (Tsekeridou *et al.* 2008). The service may include the following applications:

- *Isabel*
- *NetMeeting*
- *Broadcast/multicast e-learning service*
- *Learning content management service*
- *Real-time Audio Video*
- *Chat and Conference*

- *Collaboration tools*

In the context of this service, a tutor based in a studio or in a lecture theatre provides the lesson to the learners. In case of tele-education room, each is equipped with TV, VCRs, video projectors and PCs (installed with the required receiver cards), receiving satellite antennas and hubs or wireless access points, microphones and speakers. There are few contemporary 3D virtual learning interactive boards available which provide multi-user platforms.

3.4.2. Learning Content Management System (LCMS) Service

The LCMS service facilitates the authoring, communication and learning content management, including multimedia content adaptation, from the use of system administration but at the same time it provides personalised access to combined learning objects in a learning path that suits best to learners of the participating communities (Stergioulas *et al.* 2008; Serif *et al.* 2009). The services include web-based LCMS Service.

The access to this service is provided using wireless-enabled PCs over a broadband wireless network interlinked with a core satellite-based communication system. Communication is accomplished using IP protocols (IPv4, IPv6).

3.4.3 Tele-conference Service

This service can be delivered using any type of satellite network infrastructure to tele-education halls/classrooms, which can be further transmitted to learner's PCs over a broadband wireless network interlinked with a core satellite network.

Each classroom can be equipped with presentation boards (flipcharts, white/black-boards, etc.), video cameras, microphones, speaker systems, DVD/VCR player, a TV set/projector, a satellite set-top box and one or two PCs (equipped with the required receiver cards, for remote video display function as well as for local presenting/sharing or running shared applications). This service can be used for unicast, multicast or broadcast distribution. For teleconferencing, the following tools and applications can be used.

- *Chat*

- *File Transfer*
- *Program/Application Sharing*
- *Shared Notepad*
- *Remote Desktop Sharing*

3.4.4 Webinar/Webcast Service

Many research companies are using these services to enable live lecture or pre-recorded transmission of material from one site to the rest of remote sites (Serif *et al.* 2009). The content could be sent unicast if it is a point-to-point communication, otherwise multicasting is used. Since this is an unidirectional transmission, the high propagation delays of satellite networks do not affect performance. However, a low packet inter-arrival delay (jitter) is desired. The service can be delivered using any type of satellite network that can transmit to learner's PCs over a broadband wireless network interlinked with a core satellite network. For Webinar /Webcast, the following applications are popular:

- *Real Player*
- *Windows Media Player*
- *Apple QuickTime*
- *VLC*

3.5 Applications and Platforms

The IEEE Learning Object Metadata (LOM) family of standards specifies a conceptual material which defines the structure of a metadata for a learning object. A learning object is defined as any object (digital or not) that can be used for learning and teaching. It is not necessary that each program is suitable for platform at hand (Kraan 2007).

The Shareable Content Object Reference Model (SCORM) (Services 2003) is a framework and defines a Web-based learning Content Aggregation Model, a Run-Time Environment, as well as Sequencing and Navigation for learning objects (Bohl *et al.* 2002; Chew 2008). SCORM is a collection of specifications selected from various sources to provide complete e- learning suite capabilities which enables interoperability, accessibility and readability of Web-based learning content (Shih *et al.* 2007).

A numbers of tools and applications are available with different protocols and standards. The characteristics of satellite-based communications offer many challenges in terms of the protocols and applications for real-time interactive communication and collaboration tools. The main challenge is excessive delay, inherent to geostationary satellite links. This delay prevents the use of a reliable transport protocol like TCP for the transport of real-time interactive media communication. In addition, some satellite networks can be behind the network address translators (NAT) or may limited IP (firewalled) connectivity to the public Internet, primarily to prevent denial of service (DoS) attacks. In the following section, key applications such as Skype, MSN messenger, Microsoft Netmeeting, Clix and Isabel are discussed and compared.

3.5.1 Skype

Skype is similar to other applications like Yahoo and MSN Messengers, but it has the edge in most cases which can handle the presence of NAT boxes successfully, achieving in this way a high call completion rate. The Skype protocol is proprietary (Baset and Schulzrinne 2006) unlike the Session Initiation Protocol (SIP) which is extensively used for communication over internet protocols for unicast and multicast services.

Skype nodes use a variant of the standard NAT traversal called, Simple Traversal utilities for NAT (STUN) and Traversal Using Relay (TURN) technology. At the transport level, Skype uses TCP for signalling and both UDP and TCP for media traffic. The major advantage of Skype is that it implements STUN and TURN servers in the node itself to handle NAT, unlike the specific server configuration in other architectures and specifications like Session Initiation Protocol (SIP) and H.323 (Basicevic *et al.* 2008; Voznak 2008).

In order to avoid firewalls, Skype randomly chooses the port numbers, although it also opens TCP ports 443 and 80, if possible to use. The lack of explicit service port number imposes a requirement of maintaining a list of super node IP addresses and port pairs in the host's cache, which the Skype client builds and refreshes frequently.

The call setup procedure depends if the clients include public unrestricted IP addresses or if one or both are behind a NAT box. If both users are on machines with public IP addresses whose traffic is not restricted by firewalls, the call setup signalling information is exchanged directly between them over TCP.

Considering the above limitations, Skype is not the best choice for the testing scenarios for hybrid wireless network. It is difficult to control the traffic pattern of the Skype clients, as most of the logic behind the Skype is handled by Skype-managed back end servers or by external super nodes. Skype application also depends upon internet to login on the public authentication services. The fact is that, in some scenarios, Skype itself decides to send media communication over TCP which makes it unsuited for satellite communications.

3.5.2 MSN Messenger

The protocol used by this network, called Mobile Status Notification Protocol (MSNP), is proprietary. There are two main types of servers in the MSN Messenger software network: Notification Servers (NS) and Switchboards (SB). NS handles existence information and performs other services such as notifying the user regarding new email in his in box allowing the user to create new switchboard sessions. The SB handles instant messaging between two or more users to a connection to a shared switchboard session. The Switchboard is also used to deliver other kind of messages such as invitations to services such as file transfers. The recently released version of MSN Messenger, rebranded as

Windows Live Messenger, claims to have better support for NAT-traversal (msn.org 2008).

3.5.3 Microsoft NetMeeting

Windows Netmeeting is a H.323-compliant voice and videoconferencing application which was incorporated in the Windows operating system (ITU 2007). H.323 protocol suite is a set of communication standards of the International Telecommunications Union for multimedia communication over packet-based networks, mostly IP based (ITU 2009). It is based on the H.320 technology for circuit-switched lines, such as ISDN but optimized for Internet.

H.323 specifies components that combine to provide a complete communication service (Microsoft 2009): H.225.0 defines a layer which formats the transmitted voice, video, data and control streams to be transmitted and received over the network. On IP, it uses the IETF's Real Time Protocol (RTP) and Real Time Control Protocol (RTCP) for logical framing, sequence numbering and error detection. It also includes registration and status control (RAS) and admission, to communicate with the gatekeeper.

Q.931 is a protocol that defines how H.323 layer interacts with its peer layers. It is a protocol that goes over H.225.0 and is a link layer protocol that is used to establish connections and framing data. Q.931 also provides a method for the definition of logical channels inside of a large channel.

H.245 provides the call control mechanism which allows H.323 terminal to connect to each other. That means it provides standard procedures for establishing audio and video connections, so this standard defines signalling, flow control, and channelling for messages, requests, and commands. It also includes codec selection and arrangement.

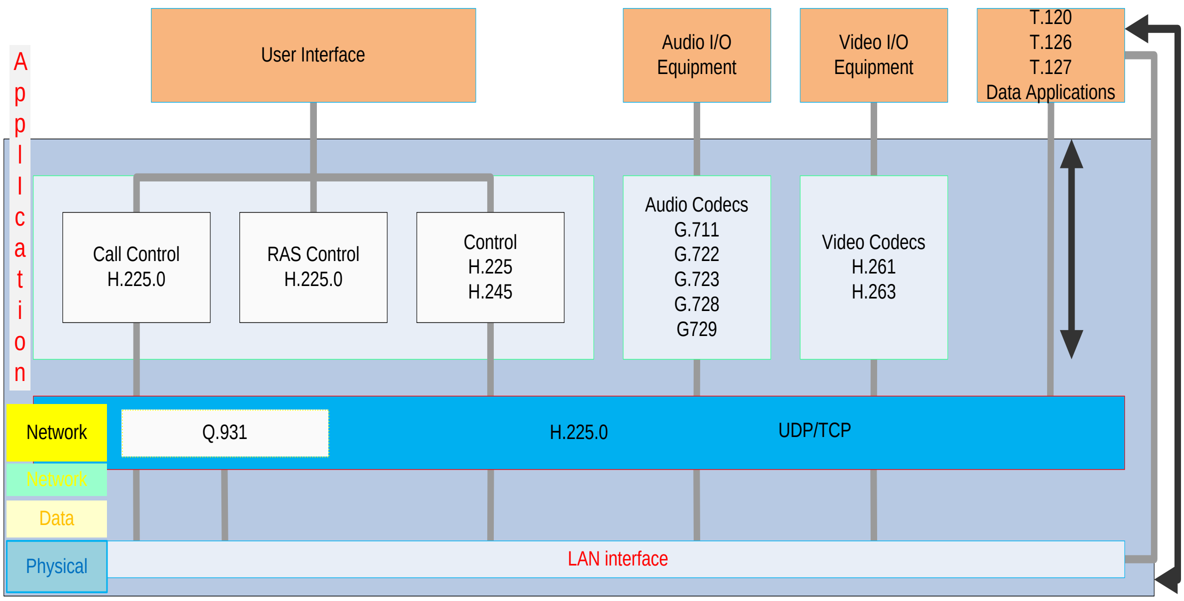


Figure 3.5: H.323 Architecture

The control plane of H.323 is composed by the H.225.0, Q.931 and H.245 recommendations. The H.323 specification requires codecs for audio and video: H.261, H263 for video and G.711, G.722, G.723, G.728, G.729 for audio.

T.120 provides support for real-time, multipoint data communications which can be used as the mechanism for packaging and sending data in H.323 (ITU 2007). It uses the H.225.0 layer to transmit and receive packets. The data conferencing products compliant with T.120 can interoperate with H.323 counterparts.

H.323 protocols are not NAT friendly. This means that they cannot function properly if a part of the nodes are behind a NAT box. In order to use H.323 under above circumstances, an application level Gateway is required to be deployed in the NAT box.

Based on the merits and protocol architecture, Netmeeting is found more suitable for the testing scenarios during our test/trail because it is a standards-based solution with well-known behaviour. In addition, it does not depend on external servers operated by third parties and accessible via the Internet. However, Netmeeting has limited support for some operating systems, such as Windows Vista and Window 7.

3.5.4 The Clix Platform

Corporate Learning and Information exchange (CLIX) is a commercial software application (developed by IMC AG) with which the user controls all information and learning knowledge processes via their browser in the intranet or internet, in real-time.

CLIX is based on a scalable, multi-layer client server architecture, which allows distributed data management and a distributed application operation. This also allows the graphical user interface (GUI) to integrate corporate design.

The CLIX platform is a web-based eLearning system that makes heavy use of the HTTP – protocol and, therefore, TCP – protocol for accessing static web content as well as for streaming learning video content. TCP is the underlying transport protocol used by CLIX. As long as there is two-way IP connectivity, TCP provides reliable communication.

3.5.5 Isabel Application

The Isabel computer support cooperative work (CSCW) application is a group of collaboration tool for the Internet, that uses TCP-UDP/IP protocols (Quemada *et al.* 2005). Isabel supports the attainment of distributed meetings, classrooms, by using a service concept which has an effective management of multipoint configurations (Quemada *et al.* 2005; Isabel 2007; Serif *et al.* 2009).

An Isabel terminal is a computer where the Isabel application is deployed with all the necessary hardware to run Isabel. The Isabel application provides enhanced support to distributed multimedia services (De Miguel *et al.* 1994). The Isabel session topology is tree-based- it consist a root, interconnecting nodes and final nodes.

Isabel runs on top of its own overlay network. This overlay is composed of a media delivery layer and an interaction mode control layer. The content delivery overlay network can be configured to make use of the variety of network protocols and services existing today, such as unicast, multicast, IPv4 and IPv6. Each Isabel module (master, interactive terminal, etc.) includes a flow server. The flow server is the core element used to construct the media transport layer. The flow server can be run as a remote platform module for performance reasons or can have several always-on entry points to the platform.

The Isabel function are (Agora Systems 2005; Isabel 2007) as follows:
The service proxy function allows the access of a participant to the session through another terminal or flow server in places where no direct IP connectivity exists with the master of the session.

The MCU function allows setting up of multipoint configurations over IP unicast.
The gateway functions to connect IP multicast, IP unicast, IPv4 and IPv6, etc.
Traffic shaping, limiting and merging functions. Each Isabel terminal includes a flow server inside that can be used for a service proxy, as an MCU or a gateway for other participants of the session. However, dedicated flow servers are recommended for performance reasons as it provides an always on entry point to sessions.

An Isabel Terminal can be connected as a normal PC, provided there is enough bandwidth to connect the session. The necessary bandwidth required to connect an Isabel Session is not a built-in parameter and can be decided by the session administrator. The bandwidths range from 128kbps to 10Mbps.

As Isabel uses TCP/IP-UDP/IP protocols, the terminal must have an IP address. It is important to keep the Bidirectional UDP from 53009 to 53032 ports open in the routers. Most of the configuration can be set up as required by the hybrid network testing and operating scenario such as audio and video codecs, error recovery features, and transfer topology. Isabel can handle the presence of NAT boxes with a flow server installed in the border of the network, and if the parent of a node is sited at an IP public address, a node behind a NAT box can join the session. In addition, it can handle heterogeneous network seamlessly for both multicast and unicast services simultaneously. In addition, most of the traffic (except signalling) goes over UDP, which is the most suitable protocol to transport media flows over high-delay satellite links. That is why Isabel was selected for our test bed during testing scenarios.

Isabel includes a whiteboard and desktop sharing mode. The whiteboard runs on the Isabel own overlay network and it is transported via TCP. The desktop sharing mode is based on Virtual Network Connection Remote Frame Buffer/ (VNC/RFB) to connect to any machine with a VNC server. The Isabel system can run its own VNC server, or an external server.

The distribution of the VNC desktop sharing mode in Isabel can be made using the NEREDA mode or the Shared Display mode. The NEREDA mode distributes VNC data persistently using an overlay Transmission Control Protocol (TCP) distribution tree. Shared Display encodes the desktop contents with the current video codec and distributes it through the User Datagram Protocol (UDP).

3.5.6 Remote Desktop Protocol (RDP)

RDP is a proprietary application and desktop sharing protocol developed by Microsoft. It is based on the ITU-T.120 protocol family of standards (ITU 2007). It implements separate virtual channels for carrying communication and data presentation.

RDP implements its own video driver on the server in order to encode display output in RDP messages that are transported over TCP. Client messages are sent to the server, which uses a virtual keyboard and mouse to process that messages coming from the client (RDP 2008).

RDP is a protocol which share application in Windows operating systems. However, the protocol data transported over TCP can be a concern over high-delay satellite links.

UDP and IP Multicast are the main transport protocols used by ISABEL. From the application's point of view, IP Multicast packets are treated similar to UDP packets. However, IP Multicast routing is different from IPv4 routing and often causes problems in real-world scenarios. Another issue is that WiMAX and Wi-Fi technologies treat multicast/broadcast packets, unlike in Ethernet, therefore the maximum bandwidth available to reach a node may vary significantly. In order to reach all nodes in a cell the more robust, but also less efficient modulation is used, which inherently limits the available bandwidth. For Wi-Fi, bandwidth limit can be as low as 1 Mbps for 802.11b and 6 Mbps for 802.11a/g. Since Wi-Fi 802.11g is used during test trial of this study and total multicast bandwidth requirements, are less than 2Mbps, therefore this should not be an issue.

The IEEE 802.16d standard/specifications, however, limit multicast traffic to 64kbps. It is essentially treated like management traffic. The simplest solution is to use ISABEL unicast<->multicast converters in front of the WiMAX sector controller and behind each subscriber station.

3.5.7 Virtual Network Connection/Remote Frame Buffer (VNC/RFB)

VNC is a desktop sharing software and protocol that allows a user to control remotely another computer over a TCP/IP network. HTTP (one port) and VNC allow VNC Server to provide VNC viewers for sessions through a TCP port (VNC 2008).

RFB is the open protocol that VNC uses. Its architecture distinguishes two kinds of endpoints: the remote endpoint is the RFB client or viewer, and the RFB server is the endpoint where the display originates. It is a point-to-point and point-to-multipoint system, as many clients can connect to the same server simultaneously (Ricardson 2005).

VNC/RFB is flexible and can be fine-tuned to various network scenarios. As in RDP, the protocol information is transported over TCP, and the screen updates are pooled by the client could be an issue over high-delay satellite links. VNC was used during test/trial of our research for SCPC Ship test scenario. The test plan is detailed in chapter 5 and testing is mentioned in chapter 6.

3.6 Multicast and Unicast Services

3.6.1 Multicast Services.

Multicast is a bandwidth conserving technology (Neto *et al.* 2007) that reduces network traffic by delivering a single stream of information or data to multiple destinations. This is an important application for DVB-RCS. Multicast applications can be sent from multipoint to multipoint, point to multipoint, and multipoint to point (Crescenzo *et al.* 2008). Since satellite system is designed for broadcast, therefore, it has the inherent ability to support multicast. IP multicasting over the satellite (Awal *et al.* 2005) provides large content of information from a single source to multiple receivers consuming minimum bandwidth. Multicasting is an important element (Guvenc *et al.* 2008) for emergency communication.

Interoperability of multicast routing protocols studies are carried out in wireless adhoc networks and investigated flooding-based interoperability and facilitator-assisted interoperability (Viswanth and Obraczka 2006). However, their study does not address any of the QoS issues for hybrid networks. The major advantage of the multicast service is that it provides vast data delivery from a single source to multiple receivers consuming minimum bandwidth. Therefore, our research in hybrid wireless network builds on previous studies and makes a system new contemporary to previous aspect.

3.6.2 Unicast Services.

Unicast is defined as the delivery of a single stream of information or data to a single recipient. For reliable unicast transfer applications, TCP protocol is used which is reliable and easily understood protocol. TCP can be modified to provide information to various users and use multicast when available to send data reliably to multiple recipient (Jeacole *et al.* 2005).

3.7 Summary

This chapter discusses and compares network architectures, with reference to OSI models and reviews the interoperability issues and QoS requirements in Wi-Fi, WiMAX, DVB-RCS, and SCPC. The various applications and platforms such as Skype, MSN Messenger, NetMeeting, Clix, and Isabel are described and compared. The Clix platform and Isabel application was used in our study during the testing of hybrid wireless network. The collaboration tools specifically Remote Desktop Publishing and Virtual Network Connection are also described. A brief description of unicast and multicast services is given at the end.

Chapter 4

Interoperability Issues and Testing Tools

4.1 Overview

In recent years, research on the design, development and deployment of wireless networks and applications has proliferated due to their flexibility and availability. Although there is a basic requirement for all communication technologies, systems, devices and applications should be interoperable with each other; however, there are many unresolved interoperability issues and QoS problems such as delay, jitter, packet loss, bandwidth, throughput, latency and interconnectivity.

Although various types of testing tools are available, but there is no single approach or methodology tool that deals with every type of testing and applications support. Therefore, it would be beneficial to propose a generic testing methodology that can meet all of the testing requirements in hybrid networks. Testing requirements include interoperability, performance, system availability and Quality of Service.

Based on review of various studies of hybrid wireless technologies interoperability issues. This chapter covers all interoperability issues and QoS testing requirements for hybrid wireless network's scenarios and issues with particular emphasis given to delay, jitter, packet loss, bandwidth, latency, connectivity, throughput and performance.

Jain and Dovrolis (2003) in their study discussed end-to-end available bandwidth methodology and test tools for measurement of bandwidth only such as path char, p char, p probe, netime and pathrate. A study (Shahbazian and Christensen 2004) describes a Time series Generator (TS Gen) tool to generate a time series to model video packet loss phenomena by capturing both the moment and autocorrelation signatures of frame loss in a video stream.

Claffy and Dovrolis (2008) in their project title "Bandwidth Estimation: measurement methodologies and applications" discussed capacity, bandwidth and throughput and used pathrate, path char, and P char test tools for bandwidth testing only. Sharma et al. (2006) presents Netvigator, a network proximity/latency analysis tool that use clustering method to locate the closest node to a given node.

A study (Bi. *et al.* 2002) in their measurement methodology selected open source trace route to measure and analyze the end-to-end path and developed Posip based on ping to measure delay. Some other researchers used ping and ethereal only.

Another study (Williamson 2001) recommended a specialized network card to use for each type of network on which traffic data is to be measured. However, in this research the author discussed all common tools, did a comparison of functionality so that the research community can select based on their measurement requirements.

This chapter is organized as follows: Section 4.2 discusses generic test consideration and interoperability requirement such as delay, jitter, latency, bandwidth, packet loss, and throughput. Section 4.3 describes interoperability and QoS failure analysis testing which includes failure analysis, availability testing, unicast and multicast testing. Section 4.4 describes and explains detail overview for testing tools including consideration of those test tools, which we have used in our study. Finally, section 4.5 gives conclusion.

4.2 Generic test considerations and Interoperability requirements

Interoperability testing measures to ensure correct communication of two or more devices/systems according to technical specifications which are necessary to ensure successful integration, supporting various communication protocols. It is important to consider which network protocol is to be used for measurement. The important network protocols are Transmission Control Protocol/Internet Protocol (TCP/IP) and User Datagram Protocol (UDP). The TCP is a common protocol used to transmit data between computers network when reliability matters. The UDP is a protocol used to transmit data between computers network when efficiency and timeliness are more important than reliability. The TCP and UDP protocols are layered on top of IP.

In the following subsection, interoperability issues and Quality of Service requirement are described and discussed.

4.2.1 Delay

Delay is the amount of time it takes a packet or frame of data to travel from the source to the destination. In other words, delay is the amount of time elapsed from point to point in a network. Delay can be measured for either one way or round trip. In most cases, delay can vary over the course of a voice stream.

Table 4.1 below (Kawalek 1995; Blakowski and Steinmetz 1996; Stergioulas *et al.* 2008; Serif *et al.* 2009) introduces the acceptable delay on audio/video content streamed via a teleconferencing application. This shows different types of media and acceptable delay for wireless communication in hybrid networks.

Table 4.1: An acceptable level of delay for random media

Media		***Mode/Application***	***Quality of Service***
Video	Animation	Correlated	+/- 120ms
	Audio	Lip synchronization	+/- 80ms
	Image	Overlay	+/- 240ms
		Non-overlay	+/- 500ms
	Text	Overlay	+/- 240ms
		Non-overlay	+/- 500ms
Audio	Animation	Event correlation [e.g. dancing]	+/- 80ms
	Audio	Tightly coupled [stereo]	+/- 11 µs
		Loosely coupled [dialogue mode with various participants]	+/- 120ms
		Loosely coupled [e.g. background music]	+/- 500ms
	Image	Tightly coupled [e.g. music with notes]	+/- 5ms
		Loosely coupled [e.g. slide show]	+/- 500ms
	Text	Text annotations	+/- 240ms
	Pointer	Audio related to the item to which the pointer shows	-500ms, +750ms

4.2.2 Round-trip delay

The Round-trip delay or Time (RTT) is defined as the interval between the time when a measurement agent application sends a signal pulse or packet to a node and the time it receives an acknowledgment that the packet or pulse was received by the particular node. The delay time depends on various factors such as a communication medium, data transfer rate, number of nodes, traffic on each node, packet loss, bandwidth, miss-configuration, redirection and interference. The

Roundtrip delay includes queuing delays but does not contain any system lookup times by the measurement application (Serif *et al.* 2009).

4.2.3 Jitter

Jitter is defined by ITU (International telecommunication union) as "short-term variations of digital signals from their ideal position in time" (Chin and Cantoni 1998). System stability can be determined by tracking errors over an extended period of time (Agilent 2007). "Jitter describes the inter packet gap for streaming applications"(Cisco 2005). If packets arrived at different times with different inter-packet interval timing, then jitter is high and audio and video quality are low (Cisco 2005).

Inter-arrival Jitter is defined as the deviation of packet trip time from subsequent datagram. Systems that rely on fixed, reliable packet arrivals can experience significant problems when jitter increases. Voice and streaming applications significantly degrade with increasing amounts of jitter.

4.2.4 Packet loss

Packet loss is defined as the fraction of packets sent from a source measurement agent to a sink measurement agent that does not arrive at their destination. This includes those packets which are not received as well as those packets which are without acknowledgments (TCP only) or not received within a predefined round-trip delay (Stergioulas *et al.* 2008). The effect of packet loss can create jitter, error, and gaps in speech, broken images or even complete absence of received signal. In a satellite network, packet loss is caused either by the sender or receiver equipment or intermediate network elements such as routers, delay, bandwidth allocation policies or satellite equipment (Awal *et al.* 2005).

While several enhancements in TCP/IP and UDP/IP protocols have been carried out to reduce the packet loss, nevertheless it is still a challenge to improve the current number of packet loss.

The table 4.2 below (Kawalek 1995; Blakowski and Steinmetz 1996; Stergioulas *et al.* 2008; Serif *et al.* 2009) highlights the acceptable and unacceptable loss for audio and video content.

Table 4.2: Probability of losing frames

	Probability of losing frames			
	Audio		Video	
	Task dependent ratings	Task independent ratings	Task dependent ratings	Task independent ratings
Good	0%	<%4	0%	19%
Still Acceptable	10%	6%	99%	51%
Poorest Quality [slight below acceptable]	22.5%	10%		64%

4.2.5 Bandwidth

Bandwidth is the data rate (in a given time period) transfer. In computer networking bandwidth represents to the data rate or the volume of the connection (Shahbazian and Christensen 2004). The greater the network capacity, it is more likely that the performance of networks will be greater, though overall performance depends on other factors including saving bandwidth, and avoiding end-to-end delay, delay jitter, latency, packet loss and throughput to obtain required QoS.

There are different definitions of bandwidth for different applications. In computer networks, the amount of information carried from one source to another in a time period is called bandwidth. A network with an available bandwidth is one that is able to provide enough information to justify the succession of images in a presentation. The bandwidth measurement tools are used to determine the amount of available resources along a path or available packets/bytes per second. The bandwidth is also calculated by downloading or uploading the amount of data in a specific period.

The bandwidth measurement tools are used to assess and calculate the amount of available resources along a path or available packets/bytes per second. There are various open source tools for bandwidth estimation available but most of these measure capacity rather than the available bandwidth. Useful tools for bandwidth measurement are path char, path load and delphi (Jain and Dovrolis 2003).

The path load test tool was mainly used for available bandwidth in multiple testing environments. Iperf is also an open source and reliable command line tool for measuring server's bandwidth. Spice work is another open source bandwidth-monitoring tool. The author of this study has used Iperf test tool for measuring bandwidth and PRTG test tool for monitoring bandwidth during test/trail of this study.

4.2.6 Latency

Latency describes the normal processing time of packet from the time it is received until the time packet is forwarded (Cisco 2009). Latency is measured by calculating roundtrip time. Data switches and routers have generally less than 1ms latency. Some voice packets converted from Analogue to Digital may takes up to 20ms. Audio latency is the time required for a computer or system to collect information from an audio input into a program and copy this to the audio output (Milijevic and Semicondutor 2008).

The network latency causes significant performance issue (Joung 2003), the slower the network information enters and leaves the system, the system will be over loaded.

The low latency is unnoticeable time between request and response. Low latency is useful for multimedia and real time applications. Latency less than 100 ms do not affect audio quality, but latency more than 200ms is very much noticeable for multimedia and real time audio and video conferencing. "When interactivity and timeliness are critical and vital, low latency communications are without reliance on excessive packet retransmission necessary" (Chiew *et al.* 2005).

4.3. Interoperability and QoS Failure Analysis Testing

System reliability and availability are very important aspects to consider. Reliability is the ability of the systems, devices and components to perform their function without failure. Availability means the system is available as well as ready for immediate use. MTBF is a common unit to measure reliability (Cisco 2009).

Reliability analysis is based on calculating the network availability and analysing the causes of any network outages. Some of the quantitative that are used for this purpose are known as Mean Time Between Failures (MTBF), Mean Time To Failure (MTTF), and Mean Time To Recover (MTTR).

It is determined by the formula:

MTBF = MTTF + MTTR (Abbasi *et al.* 2008).

4.3.1 Availability Testing

Network resource availability is defined as the fraction of time the network is available from a specified group of measurement agents to a specified group of test points. In addition, a node is considered to be reachable from a measurement agent if that agent can send packets to the node and, within a short, predefined time interval, receive acknowledgment from the node that the packet was received. For example, if each measurement sample consists of multiple pings, the test point is considered reachable from the measurement agent if the latter receives at least one acknowledgment from the test point. Network availability shows that the system is available and is operational for any immediate use, which can be calculated by the formula (Abbasi *et al.* 2008):

Availability (A) = MTTF / MTBF

High network availability is a critical requirement for some users and service providers.

4.3.2 Time of day Testing [Plot of delay (or loss) as a function of time]

The time intervals plotted dictate the nature of the plot. If the measurement interval was 30 minutes, this baseline would also be called a 30-minute baseline since the x-axis would be demarcated into 30-minute intervals (12:00 a.m., 12:30 a.m.). A Sample for the 24 hours would produce plots with useful, yet different information on performance. Such a plot may not add a valid baseline since performance at instants in time can vary widely due to different traffic loads condition such as light, medium and heavy traffic.

4.3.3 Daily Testing [Plot of delay (or loss) as a function of a day]

For each day of the week, all the sample sets are aggregated to calculate the statistics for that day. The number of days' or weeks worth of data to aggregate also depends on data availability and perceived relevance. A 24 hour daily testing for a period of a week produces a large amount of data.

4.3.4 Performance measurement

Performance measurement is used for capacity planning, bandwidth utilization, packet loss, and network circuit performance (Cisco 2005). Large-scale transfers of data requires high throughput and low loss, while voice and video applications demand low latency, delay jitter and packet loss (Abbasi *et al.* 2008). In communication networks, queuing, latency, and jitter affect performance.

For best performance the service should be available on a 24 hour basis and should be robust; however, this cannot be guaranteed for the tools provided by third-party providers. Since in satellite communication, there is an inherent delay which affects performance. However, a low packet inter-arrival delay (jitter) is desired. The satellite network should have enough bandwidth reserved during the transmission.

4.3.5 Throughput measurement

The throughput of any system is the maximum packet-forwarding rate or capacity so that the system will not fail to forward any received packets. Any packets that are not forwarded are considered lost packets. This can significantly affect the performance of the system.

The throughput of communication links is measured in bits, Kilobits, Megabits and Gigabits per second. There are various open source test tools available to test the throughput such as D-ITG, Wireshark, httperf and Iperf. These test tools were used during test/trial of this study.

4.3.6 IP fragmentation

Streaming media is one of the more established source of IP fragmentation. While not problematic in small amounts, IP fragmentation becomes a problem when facing large amounts of traffic/data and out-of-order arrivals. When it is combined with packet loss, both can increase network problems (Joung 2003), such as the loss of one fragment results in the discarding of all the other relative fragments.

4.3.7 Link layer characteristics

There is a wide range of technologies used for the link layer, like Ethernet, Token ring, GPRS, ISDN, PPP 802.11 for WLAN, and 802.16 WiMAX. The author of this study focuses to capture the network traffic and traces of wireless networks for Wi-Fi, WiMAX and DVB-RCS links to address end-to-end interoperability issues.

4.3.8 Packet Sniffer

Packet ‘‘sniffing’’ refers to capturing network traffic. This is performed by placing a network interface card, the interface card will ignore its assigned address and receive all frames and packets (Shorey *et al.* 2006). This functionality checks all data packets travelling through a particular network and analyzes the network packets to find out the IP addresses, protocols of the source and target machine. The TCPdump test tool was used to capture the packets during test/trial of our study.

4.3.9 End-to-end network measurement

Network traffic measurement including end-to-end network measurement provides a way to go “under the hood", to understand what is working properly on a network (Williamson 2001). A network researcher can obtain detailed information regarding the transmission of packets on the network and its content using network measurement software or hardware. In this way, end-to-end network interoperability issues and QoS requirements can be calculated.

4.3.10 Traffic generation testing

Traffic generation tools inject traffic into the system to determine bandwidth, delay, jitter, packet loss, jitter, and latency. Most of traffic generation tools are also used for load testing. The background traffic is generated in order to be used in tests under various load conditions.

For traffic generation test tool, sender and receiver devices are installed on both sides of the network nodes. The traffic is injected into the network topology, at speeds up to line rate, and traffic generation tool is able to receive traffic. On the receiving side, the test tool prepares statistics and captures the traffic for analysis. The author of this study used D-ITG, Mgen, and U2m for traffic generation during test/trial of this study.

4.3.11 Unicast and multicast testing

Multicast is a bandwidth-conserving and saving technology which reduces flow of network traffic by delivering a single stream of data or information to multiple destinations (Cisco Systems 2000). Multicast applications can be send from multipoint to point, point to multipoint, and multipoint to multipoint (Neto *et al.* 2007; Crescenzo *et al.* 2008).

IP multicasting over satellite provides extensive information content from a single source to multiple destinations consuming minimum bandwidth (Awal *et al.* 2005).

Unicast is delivering a single stream of data to a single recipient. The major advantage of the multicast services is that it provides vast information delivery from a single source to multiple receivers consuming minimum bandwidth. For reliable unicast transfer applications, TCP is used, which is a reliable and well-understood protocol. TCP can be modified to carry information to multiple receivers at a time and use multicast when available data to send reliably for multiple recipients (Jeacole et al. 2005). The author of this study has tested the performance of multicast and unicast traffic using mgen, httperf and u2m network analyser.

4.4 Network Test Tools

There are several hardware and software test tools for network testing. Some important hardware tools are Fluke networks Opti View Series II Integrated Network Analyzer, HP open view Network Node Manager, MS269XA Series Signal Analyzers, Air Magnet's Handheld Analyzer, Tektronic WCA11G Signal Analysis, The Agilent MX Series spectrum analyzer, ML2480A and ML2490A Series Power Meters, MG3700A Vector Signal Generator, MS2781B Signature Signal Analyzer, MT8222A BTS Master, Network Traffic Analyzer, NavTel IW95000 ATM, Anritsu various signal generators, Spectrum and Vector Network Analyzer.

The popular open source software test tools are Beacon/multibeacon, Ethereal/ Wireshark, httperf, Iperf, Kismet, Mgen, MRTG, Netmeter, OWAMP, PRTG, Pathchar, Pathload, Pathrate, Pchar, Ping, m ping, SNMP, TCP dump, TCP trace, Trace mate, Tracepath, and Trace route etc.

The above mentioned software tools are discussed and compared in the following subsections and the testing methodology is presented in next chapter.

4.4.1 Beacon /Multicast Beacon

Beacon is a software test tool to measure and monitor the performance of an H.323 video conference session. It helps end-users and conference-operators to provide information necessary to troubleshoot H.323 protocol performance in the network and at the host (ITU 2009; Multicastbeacon 2011). Beacon uses distributed client/server network architecture. The client refers to the end nodes, and the server refers as a core-node. Testing and monitoring between end nodes is achieved using a number of core-nodes in a test path (Beacon 2008).

The NLANR/DAST Multicast Beacon analysis tool is a multicast software analysis tool, which is written in Perl and uses the Real-time Transport (RTP) protocol to provide useful information regarding multicast group's connectivity characteristics (Multicastbeacon 2011).

The Multicast Beacon provides measurement for multicast traffic in a group. It relies on multicast for distributing voice, video and other data across the network.

4.4.2 Wireshark and Ethereal

The Wireshark/ Ethereal tool is a network testing software tool used for analysis, troubleshooting, protocol development, trace analysis, GUI network protocol analysis and generating several result statistics (Orebaugh *et al.* 2004; Orebaugh *et al.* 2006). Wireshark /Ethereal allows interactive analysis of packet from a live network or from an already saved capture data from tcpdump or other compatible source (Sharpe *et al.* 2008). IO graphs which are user configurable graphs are created from Wireshark.

Wireshark /Ethereal's original capture file format is the libpcap, which is the common format used by tcpdump and various other tools. It can also read captured files from numerous other format tools such as Sun snoop, Microsoft's Network monitor, and Novell's LANanalyser.

Wireshark /Ethereal has a rich display filter to view the reconstructed data or packet stream of a TCP session (Vinel *et al.* 2006). Ethereal/Wireshark Pcap utility is used to capture packets and allows the network tester to see all traffic being run over the network. Ethereal/Wireshark runs on UNIX, Linux, Solaris, NetBSD, Mac OS X, and Windows.

Live packets/information can be read from various protocol including the important Ethernet, Token-Ring, and IEEE 802.11, etc. Files can be exported to many other captured programs. Captured network packets may be browsed via a GUI, or via tethereal/T shark program. Capture files can be edited using command-line switches to the "editcap" program. Lippcap is packet-capturing software. The wireshark source code and binary kits for some programs are available at Http://www.wireshark.org. Tshark is a command line tool of wireshark. This researcher has used wireshark tool to capture of traces in addition to tcpdump during testing/trial. Wireshark was used for analyses of tcpdump and tshark captured files during analysis of our test results.

4.4.3 Httperf

Httperf is used to measure web server performance. It generates various HTTP workloads to measure server performance. The distinctive characteristics of httperf are its robustness, sustain server overload, extensibility to new workload generators and performance measurements (httperf 2008). The author of this study has used httperf during test/trial of this study.

4.4.4 Iperf

Iperf is a test tool that enables the user to measure TCP and UDP protocol characteristics in terms of bandwidth, delay, jitter and rate loss (Softpedia 2008).

This test tool during testing try to throttle network with UDP or TCP packets to realize the maximum transfer throughput between two nodes without monitoring in-between nodes. It is possible to determine the amount of data rate transferred or time in which the test is performed. The user can specify a TCP default window size for the client or server is running. Iperf requires the program to run at both the sending and receiving ends of a network.

Some of the associated attributes of this tool related to TCP are that it measures bandwidth and supports TCP window size via socket buffers. The Client and Server can have multiple simultaneous connections.

The UDP related are that the client can create UDP streams of certain bandwidth, measure packet loss, measure delay jitter, multicast capable, multi-threaded if threads are available. The client and server can have multiple simultaneous connections; can run for a specified time, and produce periodic delay jitter, bandwidth, and loss reports.

4.4.5 Kismet

Kismet is an IEEE 802.11 layer2 wireless network sniffer and intrusion detection software test tool. Kismet works with any wireless network card, which supports raw monitoring mode, and can sniff all the IEEE 802.11b, 802.11a, 802.11g and 802.11n standards and traffic devices. Kismet identifies the network by passively collecting packets (Kismet 2010).

4.4.6 Mgen

The Multi-Generator test tool set is known as Mgen. Mgen measures performance using UDP/IP unicast and multicast traffic (Mgen 2008).

Mgen test tool is a set of tools, which is used for generating traffic, receiving traffic, and analyzing results. It can run in command-line mode or in graphical mode. Mgen was used to measure unicast and multicast performance in our study.

4.4.7 Netmeter

Netmeter is a small network bandwidth monitoring application for UNIX /Linux operating systems. This test tool provides an integrated GUI for a set of tools that allow the measurement of Quality of Service parameters over IPv4 and IPv6 networks.

4.4.8 One-Way Active Measurement Protocol (OWAMP)

One Way Active Measurement Protocol (OWAMP) application is used to measure one-way latencies between hosts (OWAMP 2009). One-way measurement solves the problem faced by roundtrip measurement to isolate the specific path problems of direction and show the direction of network traffic congestion. OWAMP uses control protocol session by using traditional client and server (OWAMP 2009). OWAMP is designed to be deployed on many systems (OWAMP 2009).

4.4.9 Pathchar tool

The pathchar tool is used for estimating network link capacities and network traffic or protocol latencies along an Internet path (Williamson 2001).

4.4.10 Pathrate

Pathrate is a capacity estimation tool (Pathrate.org 2008). Pathrate sends packet pairs to uncover the bandwidth distribution, and selects the local mode which corresponds to the capacity. Path rate uses UDP for transferring probing packets. Additionally, path rate establishes a TCP connection (Dovrolis *et al.* 2004) between the sender and the receiver.

4.4.11 Pathload

Pathload is a tool used for measurement of end-to-end capacity and available bandwidth. This tool estimates the available bandwidth of network path.

4.4.12 Ping

Ping is an essential tool used to determine the network connections. Ping uses the Internet Control Message Protocol (ICMP). Ping packets are sent from source to a target destination at fixed intervals. The packets are considered lost by the sender if the response expected from the target are not received within a specified time (Sommers *et al.* 2005).

A small packet contains 64 bytes - 56 data bytes and 8 bytes is sent through the network to a specific IP address. The response-requesting computer sends the packet and waits for a return packet. Packet is received if the connections are accomplished. Ping also tells the roundtrip time of the packets. Ping tests the connectivity to a remote computer and reports the round-trip-time (RTT) to that computer by placing a time stamp in each packet and receiving echo back. Ping assign a sequence number on each packet and reports the sequence numbers of the packets it receives back. Thus, it can be established that if packets are duplicated or dropped. Ping reports other ICMP messages if a router is declaring the target host unreachable (Activeexpert 2010).

4.4.13 MRTG and SNMP

The Multi Router Traffic Grapher (MRTG) test tool is used to monitor the traffic load on different links. This tool graphically represents the data from SNMP agents to SNMP manager. It generates HTML pages with GIF graphics inbound and outbound traffic in network interfaces in real time. MRTG is based on pearl and works with Linux, Unix and Windows (Oetiker 2001).

Simple Network Management Protocol (SNMP) test tool is used for managing network devices and interpret traffic counters such as routers, switches, and servers (Breitgand *et al.* 2002). This consists of network management standards, including an application level protocol and data objects. A system administrator can run a manager tool, which communicates with SNMP agents. Network service providers have the ability to monitor passively network nodes within their network for packet loss on routers (Sommers *et al.* 2005) using SNMP.

4.4.14 TCPdump

TCPdump is a command line debugging tool used in computer networks (TCPdump 2007). It allows the network tester to capture the packets on a computer over a network (Hong *et al.* 2005) to debug applications one is developing which utilize the network for communications. TCPdump is used to perform the following tests (uCertify 2005):

- *To debug the system by determining whether all necessary routing is performing properly and allowing the tester to pinpoint the source of an issue.*
- *To intercept and display communication of another user or computer.*

Many analysis tools can read tcpdump files for the analysis of captured packets (Williamson 2001), along with filtering of captured traffic streams based on a specific host.

The captured packets/data is analyzed using various software including wireshark. The author of this study has used tcpdump for capturing the packets during test/trial of this study.

4.4.15 TCP Trace

The TCP trace software tool is used for analysis of TCPdump files. Several popular packet captured programs such as tcpdump, HP Net Metrix, and WinDump etc. can be used by TCP trace files. The Tcptrace can provide various details such as the number of segments, elapsed time, bytes sent and received, round trip times, throughput and latency. It can also provide a set of graphs for further analysis.

4.4.16 Trace route

The traceroute command is an easy trace software tool used to discover network connectivity (Tateishi *et al.* 2009). Traceroute uses the ICMP protocol implemented at the routers to process an ICMP response. High delays at a hop usually signal that the router is very busy. Not all applications experience these delays because routers may treat the application's IP packets differently. In addition, some routers simply block the ICMP packets if the router is down or that the remote host is unreachable. The traceroute utility (Williamson 2001) is used for determining Internet routing paths. The Tracepath can also be used in Linux based systems. The author of this study has used traceroute for connectivity and round trip time (RTT) during test/trial of this study.

4.4.17 Traffic Monitoring Tool: Paessler Router Traffic Grapher

Paessler Router Traffic Grapher (PRTG) is an easy-to-use software test tool used for monitoring and classifying network bandwidth usage. The users are provided usage trends for network devices. It is used for bandwidth usage monitoring and for monitoring of many other aspects of a network such as CPU usages, disk usages, and temperatures.

PRTG provides extensive bandwidth and network usage data. This data helps to improve the network efficiency. This information helps in avoiding bandwidth and server performance bottlenecks and also helps in finding out which applications or servers used the allocated bandwidth. PRTG software is designed to run on a system to records the network usage.

PRTG (Stergioulas *et al.* 2008) uses the three most common bandwidth data acquisition methods, which are SNMP, Packet sniffer and Netflow. The author of this study has used PRTG for bandwidth monitoring during test/trial of this study.

4.4.18 u2m – UDP/RTP packet analyzer

'u2m' is a tool developed by FOKUS Germany that can be used to evaluate RTP streams. This tool has been modified to understand ISABEL's multicast streams, which contain multiple audio/video streams from different sources within one IP Multicast group as a multiplex. 'u2m' provides a dump of the RTP header of packets received and performs per-stream analysis in order to calculate individual bandwidth and loss statistics.

4.4.19 D-ITG

'D-ITG' is a traffic measurement tool which consists of three parts namely a traffic generator/sender, receiver, and traffic analyzer. A traffic generator (sender) creates one or more data streams simultaneously. Data streams consist of UDP, TCP, DCCP or SCTP sessions.

A traffic receiver on the other side just collects the received data and packets in a log file. The log file can be analyzed with the decoding component, giving bandwidth, delay, jitter, and loss statistics accumulated for a time interval or raw jitter and delay times for each packet. Since D-ITG did not measure the round trip time, the synchronization between the stations/nodes is required. For this purpose, the clocks of the stations/nodes are required to be synchronized via NTP to a local time server before and after the measurement. A linear drift of the clocks was assumed during the measurements. The author of this study has used D-ITG during test/trial of this study using synchronized clocks via NTP to a local time server.

4.4.20 Traffic generation Tool

Traffic generation tools inject traffic into the system to determine network characteristics such as delay, jitter, latency and loss. Traffic generation tools are used for load testing. The realistic background traffic is generated which is normally based on a real network scenario.

The traffic is generally injected into network sender side and it is received on the receiver side for analysis and statistics.

There are different test tools available for traffic generation. Some useful tools for traffic generation are Mgen (Richling *et al.* 2002), Network traffic generator and monitor etc. Both Mgen and PRTG have been used during testing in our test bed in the present study.

Table 4.3: Comparison of Test Tools

Software Tools	Delay /RTT	Jitter	Packet Loss	Bandwidth	Throughput	Latency	Performance of Multicast and Unicast services	Traffic generation	Traffic capture	Traffic Analysis
Beacon/ multibeacon							✔			
D-ITG	✔	✔	✔	✔	✔	✔		✔		✔
Ethereal/ Wiresharh			✔	✔	✔				✔	✔
httperf		✔	✔		✔		✔			
Iperf	✔	✔	✔	✔	✔	✔				
Kismet									✔	
Mgen traffic generator				✔			✔	✔	✔	
Netmeter				✔						
OWAMP	✔ (0ne way)									
PRTG				✔				✔		
Pathchar				✔		✔				
Pathload				✔						
Pathrate				✔						
Ping, m ping	✔									
SNMP										✔
TCPdump	✔	✔	✔		✔	✔				
TCPtrace	✔				✔					
Traceroute	✔	✔								

4.5 Summary

This chapter described various interoperability issues and Quality of Service requirements in term of bandwidth, delay, jitter, throughput, network performance, packet loss, latency, availability and interconnectivity. This chapter also describes common test tools and compared them in terms of their functionality. All possible interoperability issues and QoS requirements, such as bandwidth, delay, jitter, throughput, network performance, packet loss, latency, availability and interconnectivity for wireless communication technologies in hybrid networks have been explained in detail. Some of the important software testing tools have been described, which include:multibeacon, Ethereal/Wireshark, httperf, Iperf, kismet, Mgen, MRTG, PRTG, Netmeter, OWMP, Pathchar, Pathload, Pathrate, Ping, SNMP, TCPdump, TCP trace, and Trace route.

A functionality comparison table was produced for these test tools, presented at the end of this chapter. Most of these test tools have been used in our study during test. The advantage of using these tools is that these software tools are available from open source, flexible, easy-to-use and economical.

As expected it is concluded that there is not a single software testing suite available which is sufficient to measure all interoperability issues and Quality of Service requirements, such as bandwidth, delay, jitter, latency, packet loss, throughput, networks performance, and availability for wireless communication technologies in hybrid network. However, in practice, interoperability testing requires a considered combination of these test tools.

Chapter 5

Research and Testing Methodology

5.1 Overview

This chapter discusses the research methodology adopted, the research process and the testing methodology with its various phases and its suitability for the study. The test setup and architecture are also described. The test setup described was used in our study during testing.

5.2 Research Methodology

This research was initiated by researching and scoping of the problem area; and then by conducting literature reviews in the area of interoperability of wireless communication technologies in hybrid networks. The research methodology process is shown in Figure 5.1. Several search engines, relevant databases and journals have been searched as detailed in Annex A.

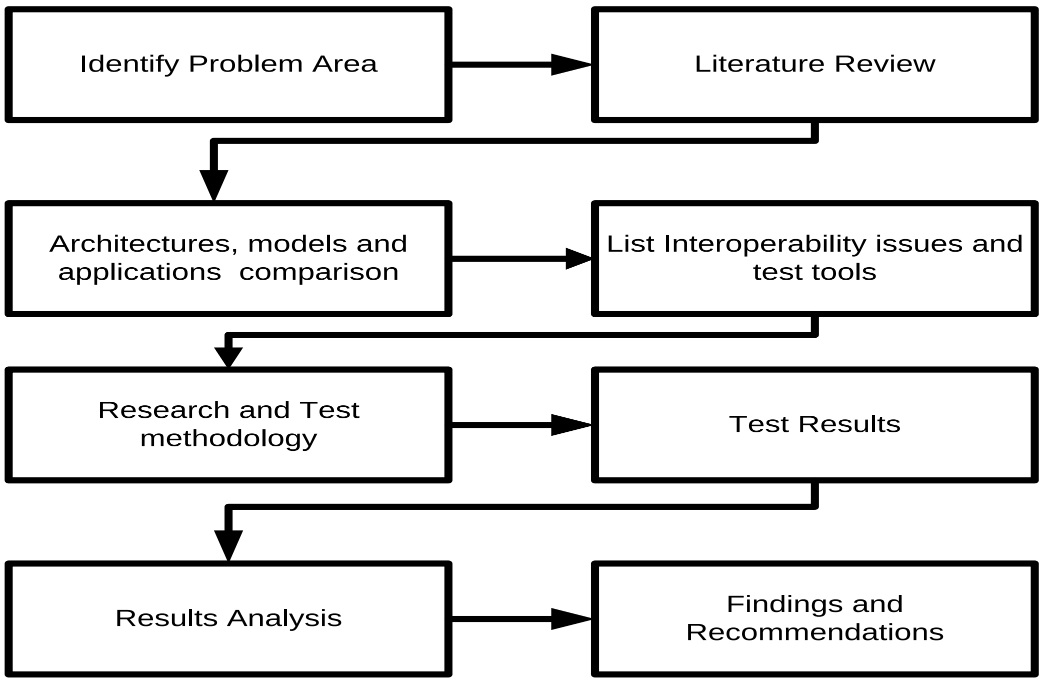


Figure 5.1: Research Methodology Process

Several research studies were reviewed and analysed for Wi-Fi, WiMAX, DVB-RCS, S/S2, SCPC (VSAT), and interoperability issues and QoS requirements on hybrid wireless networks in Chapter 2. Based on the descriptive and critical reviews of previous studies and analytical comparisons, it was established that there is limited research in the area of interoperability of wireless technologies in hybrid network. This lead to a systematic investigation of problem, and to clear objectives of research, which are stated in Chapter 1.

In the next step, various wireless technologies, standards, protocols, platforms and architectures were compared. Interoperability issues and Quality of Service requirements such as bandwidth, delay, jitter, latency, packet loss, throughput and availability are described and discussed. The important test tools such as Ethereal/Wireshark, httperf, Iperf, Kismet, Mgen, Netmeter, OWAMP, PRTG, Pathchar, Pathrate, Ping, SNMP, TCPdump, TCPtrace, and Trace route are described and compared. Most of these software based testing tools are open source, flexible, easy-to-use and economical.
The selection of the test tools can be made from a comparison list, detailed in chapter 4 to test the interoperability issues and QoS requirements for a hybrid network. It is more appropraite to use atleast two different test tools for the testing of same issue to prove the accuaracy for outcome of the result. A test process flow diagram (as shown in this Chapter figure 5.1) below was created and a detailed plan to test wireless technologies in hybrid networks was also developed which was implemented during test/trail of BASE² Project and is shown at the end of this chapter.

The motivation of the research is the rapid evolution of wireless technologies characterized by heterogeneous satellite terrestrial wireless network which fulfills one of our objective to prepare a comprehensive test plan and methodology which allows testing of a wide range of hybrid wireless networks in various realistic application scenarios. One of the main contribution is to address the interoperability issues and QoS requirements.

The significance of this research is that it is based on a clear theoretical investigation of the issues and facts, instead of any hypothetical scenarios or simulations which enabled us to devise a test plan and methodology for empirical data gathering by testing and considering the interoperability issues for measurement using proven test tools as mentioned above on various live test scenarios of a test bed. The test bed was a project supported by EU in e-learning for remote areas using the Clix platform and the Isabel application.

This chapter outlines the test tools selected and all possible test scenarios that are significant for this study. The merit of the selected test tools is that they are open source, and there is no requirement for any special type of test equipment or any type of network card.

Detailed testing is carried out using various important software test tools, and results are obtained, which are then analysed for interoperability issues and QoS requirements in the form of charts, graphs and tables (as shown in Chapter 6). The merit of this research is systematic investigation of issues and carring out test accordingly. The recommendations are given at the end of this research.

5.3 Test Plan and Process

This section outlines the test plan and test process flow diagram of the testing methodology.

5.3.1 Test plan outline

This section discusses various processes and steps to consider for the testing of wireless communication technologies in hybrid networks; these includes, testing schedule, time estimate, test scenario and configuration, type of test, objective of test, selection of test tools, test procedure, exclusion of testing, test results, analysis of results and rerun of the results.

Testing Schedule [When]: Testing schedule is an important factor which determines the dates of testing. A test priority list is prepared and an individual test plan is required to set the time of the day or day of week. This should depend on the individual test requirement, priorities and specifications.

Time estimate for test (How long/duration): Each type of test it is required to plan the duration required to carry out the test. The test estimate covers load condition and traffic pattern. It depends upon the traffic flow, performance requirements for the duration and load level at different times. The time estimate can vary significantly from a few minutes, to hours, days or weeks.

Test scenario, configuration and setup: The test scenario specifies what the network architecture is and what the traffic flow or scenario will be, such as Hub to DVB-RCS , Hub to DVB-RCS-WiMAX/Wi-Fi, DVB-RCS to Hub, DVB-RCS to DVB-RCS SIT, and SCPC (VSAT) to SIT and /Wi-Fi including full measurement of node-to-node or site-to-site. Hybrid network supporting subsets of servers need full combinations of Star, mesh or tree topology.

Type of test: The type of test is required to specify the objectives of test.

Objective/purpose of the test (Why): The objective should be clear for what will be tested and what to look for, e.g., bandwidth, delay, jitter, packet loss, latency, throughput, application level error, network level error, system error, end-to-end performance, availability and QoS.

Test Tools: Selection of test tools is very important. Deciding which test tool to use is very important in order to get the desired result.

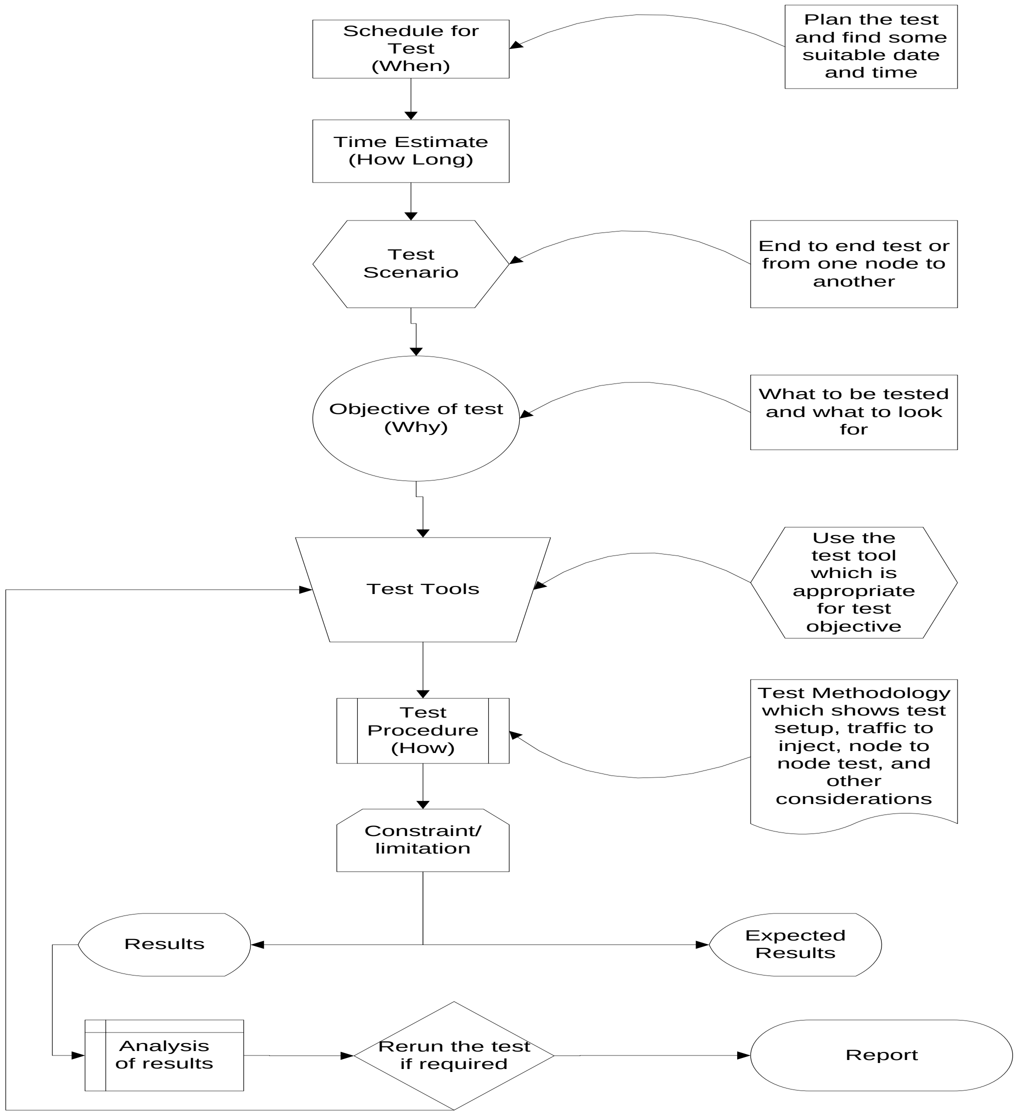


Figure 5.2 Flow diagram of the test process

There may be several tools that test same property. In this case it is good practise to use two or more different test tools and compare the results.

Test Procedure/Methodology (How): Testing methodology covers step-by-step instructions, including detail of traffic to inject/generation/IP Packet generation for active test, node to test, type of platform (e.g.Clix and Isabel), unicast/multicast based scenario, TCP, UDP test, uplink/down link considerations and command line SSH session, etc. The test methodology should include all test parameter/ configuration and setup for each test, which is required to be carried out as per required parameter and specification. This may also include some baseline or pass/fail criteria.

Constraint/limitation-if-any during testing: This stage exercises the limitations of each test. The limitations show what is covered or excluded from the test.

Expected results: The acceptable parameters are determined and whether these parameters are within the acceptable range and satisfy the specifications or requirement.

Actual Results: The actual results are the outcomes from the testing.

Analysis of results: The analysis of results is very important, in order to see whether objectives of testing are met or not, and what the deviations or variations between expected results and actual results are. This lists the reconfigurable requirements and their variation and demonstrates whether or not the variations are within an acceptable level.

Rerun the tests: There might be some results which do not meet the baseline or acceptable level or are not within the specifications, and so there might be a big gap between expected results and actual results. In this case, a rerun of test will be required to resolve any errors or issues and to find the difference between expected and actual results.

5.4 Research Methodology considerations and Test plan

The objective of this test methodology is to plan a test which allows to test end-to-end interoperability issues and Quality of Service requirements such as bandwidth, delays, jitter, latency, packet loss, throughput, TCP performance, UDP performance, unicast and multicast services and availability on Hybrid (both satellite broadband and terrestrial wireless) Networks.

This methodology recommends proven test tools on live test scenarios with various nodes. Various suitable test tools required for interoperability testing of Hybrid Networks were discussed and used during testing such as D-ITG, Ethereal/Wireshark, httperf, Iperf, Mgen, Ping, PRTG, Tcpdump, Trace route, and U2m network analyser etc. The test tools were selected from the comparison list as detailed in Chapter 4.

The different internet-based applications such as Skype, MSN Messenger, NetMeeting, Clix, Isabel and collaboration tools (such as Remote Desktop Publishing and Virtual Network Connection testing) were compared and analysed, keeping in view the requirement for hybrid network test scenarios and test setup configurations. The applications were selected based on review and consideration as detailed in Chapter 3.

Considering the specific characteristics of satellite networks and acknowledging Skype's non-transparent and non-adjustable network traffic management system, which is handled either by Skype-managed backend servers or by external super nodes, it can be clearly said that Skype was not the most suitable application to be utilized on the hybrid wireless network infrastructures.

The MSN Messenger application is a proprietary service dependent which requires Internet access. As a result, being proprietary protocol makes it impossible to have any control over the traffic patterns of MSN Messenger clients to suit to the satellite network requirements, such as delay. Netmeeting was one of the most suitable proprietary applications due to its well-known communication standards and third party operated servers. However, this application has limitation in many ways since it was not possible to make changes to its network handling procedures to suit testing requirements. On the other hand, Isabel was very flexible application that can be fine-tuned and its behaviour was very deterministic. Due to its easily configurable network settings, audio and video codecs, transfer topology, error recovery features and different interaction modes, Isabel was the most suitable teleconferencing application that could be used to realize the scenarios on satellite-based hybrid wireless networks.

The test bed of this study was a network developed from the Broadband Access Satellite Enabled Education (BASE² EU) project, which was designed and deployed a hybrid satellite and terrestrial-based network infrastructure to support distance learning for geographically isolated communities. In particular, BASE² focused on the empowerment (enabling learning) of the agrarian and maritime geographically isolated communities. The BASE² project aims was connecting remote communities to knowledge by identifying, designing and deploying e-Learning services over an integrating broadband terrestrial networks with satellite broadband technologies for isolated agrarian and maritime communities in Greece and Cyprus. The BASE² project involved twelve agrarian community sites (10 in Greece and 2 in Cyprus) were active with full network and service deployment. The test bed scenarios include DVB-RCS Satellite only, SCPC Satellite and WiMAX and finally Satellite, WiMAX and Wi-Fi. The BASE² network architecture was supporting educational content generation, management and delivery systems for different modes of learning (live virtual classroom, video conference, offline asynchronous learning, collaborative learning, individual learning, educational content generation, management and delivery) over different network technologies to a large number of sites. The overall objective of this project was the implementation of an end-to-end system for tele-education applications.

The end-to-end broadband infrastructure was integrated both terrestrial and satellite technologies using DVB/DVB-RCS, SCPC(VSAT), Wi-Fi, and WiMAX. The CLIX management and delivery frameworks and Isabel application were used as discussed in Chapter 3.

A test plan was developed as shown below, and implemented. As the test plan was generated after finding the gaps in literature review and considering various end-to-end interoperability issues such as bandwidth, delay, jitter, latency, packet loss, throughput, TCP, UDP Performance, link characteristics, connectivity and availability issues, therefore, this test plan can be used as a generic plan and testing methodology for hybrid wireless networks to test DVB-RCS, SCPC, WiMAX, and Wi-Fi scenarios and nodes.

This testing methodology covers all possible realistic test scenarios on real hybrid networks comprising variable numbers of nodes such as Hub to SIT, SIT to Hub,

Hub to WiMAX/Wi-Fi, WiMAX /Wi-Fi to Hub, and Wi-Fi/WiMAX to WiMAX, which is a robust approach included in our research methodology.
The test time is considered for minutes, hours, days, or weeks and month. The expected results criteria were selected on literature review and product specifications. The following are some of the baseline for acceptable level criteria.

The DVB-RCS and SCPC round trip time (RTT) is expected (an ideal condition) around 500 ms due to speed of light, as geostationary satellite positioned at a constant height of about 36000 Km above the earth. The Wi-Fi used in our study supports 6-54 Mbits/s. The WiMAX used in the test bed for our study during testing supports up to 70 MBit/s upload/download channels. The minimum value is 6 Mbits/s. The DVB-RCS system used for this study during testing was configurable up to 45 Mbits/s downstream, and up to 2 Mbps upstream. The VSAT system used in our study for testing was supporting 2 MBit/s upload/download channel.

Table 5.1 shows the test plan for testing interoperability requirements for wireless communication technologies in hybrid network. This test plan covers all possible test scenarios in hybrid networks. The testing is suggested for delay, jitter, packet loss, bandwidth, throughput, availability, performance of multicast, unicast, and stress test. The suggestions for the test plan, mentioned in Table 5.1, are based on the evidence collected from the literature. The performance of Wi-Fi is expected at least 6 Mbps and WiMAX 6 Mbps as per their specifications. The test plan has been implemented in BASE² project during test/trial. The results are shown in Chapter 6 which proves the validity of this plan.

Table 5.1: Test plan

Test time estimate	*Test scenarios including list of sites and node*	*Test type*	*Objective /purpose (what and why)*	*Test Tool*
20 minutes	DVB-RCS(Hub) to DVB-RCS(SIT), DVB-RCS(Hub) to WiMAX/Wi-Fi, WiMAX to WiMAX/ Wi-Fi, DVB-RCS(SIT) to DVB-RCS (Hub), Wi-Fi/WiMAX to DVB-RCS (Hub), SCPC (Hub) to SCPC (SIT /Wi-Fi)	Delay	Connectivity, Audio and video quality	Ping, Traceroute
20 minutes	DVB-RCS(Hub) to DVB-RCS(SIT), DVB-RCS(Hub)to WiMAX/Wi-Fi, WiMAX to WiMAX/ Wi-Fi, DVB-RCS(SIT) to DVB-RCS (Hub), Wi-Fi/WiMAX to DVB-RCS (Hub), SCPC (Hub) to SCPC (SIT /Wi-Fi)	Jitter	To measure overall jitter in wireless communication on hybrid network	Iperf, Tcpdump, Mgen, D-ITG
20 minutes	DVB-RCS(Hub) to DVB-RCS(SIT), DVB-RCS(Hub)to WiMAX/Wi-Fi, WiMAX to WiMAX/ Wi-Fi, DVB-RCS(SIT) to DVB-RCS (Hub), Wi-Fi/WiMAX to DVB-RCS (Hub), SCPC (Hub) to SCPC (SIT /Wi-Fi)	Packet Loss	To measure TCP, UDP packet loss	Tcpdump, Iperf, Ethereal/Wireshark for packet analysis
20 minutes	DVB-RCS(Hub) to DVB-RCS(SIT), DVB-RCS(Hub)to WiMAX/Wi-Fi, WiMAX to WiMAX/ Wi-Fi, DVB-RCS(SIT) to DVB-RCS (Hub), Wi-Fi/WiMAX to DVB-RCS (Hub),	Bandwidth	Look for bandwidth utilization, The system should guaranty the minimum bandwidth requirement.	Iperf, PRTG, Wireshark, Mgen

	SCPC (Hub) to SCPC (SIT /Wi-Fi)			
20 minutes	DVB-RCS(Hub) to DVB-RCS(SIT), DVB-RCS(Hub)to WiMAX/Wi-Fi, WiMAX to WiMAX/ Wi-Fi, DVB-RCS(SIT) to DVB-RCS (Hub), Wi-Fi/WiMAX to DVB-RCS (Hub), SCPC (Hub) to SCPC (SIT /Wi-Fi)	Throughput (delay ,Jitter, latency, data rate)	Throughput for video conference and live lectures, TCP and UDP performance	Iperf, httperf, TCP dump, Wireshark, Mgen
1 day, 3 days, 30 days	SIT to DVB-RCS (Hub)	Availability testing	To test the system is available and robust	ICMP ping, u2m
60 minutes	SIT to DVB-RCS (Hub) DVB-RCS (Hub) to SITs	Performance of multicast and unicast sessions	Measure capacity required for unicast and multicast session	Ethereal/tethereal, mgen, httperf u2m network Analyser
20 minutes	DVB-RCS(Hub) to DVB-RCS(SIT), DVB-RCS(Hub)to WiMAX/Wi-Fi, WiMAX to WiMAX/ Wi-Fi, DVB-RCS(SIT) to DVB-RCS (Hub), Wi-Fi/WiMAX to DVB-RCS (Hub), SCPC (Hub) to SCPC (SIT /Wi-Fi)	Latency	To measure for both satellite and terrestrial	Mgen, DITG
4 hours and 24 hours	DVB-RCS, WiMAX, Wi-Fi	To measure the link performance for jitter packet loss, latency, load and unload condition	To measure different applications such as text, voice, video and application sharing	D-ITG Httperf

5.5 Test setup and Scheme for DVB-RCS and SCPC (VSAT)

The testing setup layout with all possible scenarios for testing the interoperability of wireless communication in DVB-RCS hybrid networks is shown in Figure 5.3 and the SCPC (VSAT) network is shown in Figure 5.4. This test setup has been implemented for the BASE² project for both pre-trial and final interoperability testing. In this first test setup, Satellite technology DVB-RCS and Terrestrial wireless technologies, such as WiMAX and Wi-Fi were integrated, tested and analysed. In second test setup Satellite SCPC (VSAT) and Wi-Fi were integrated, tested and analysed. The testing results are presented, analysed and are shown in Chapter 6.

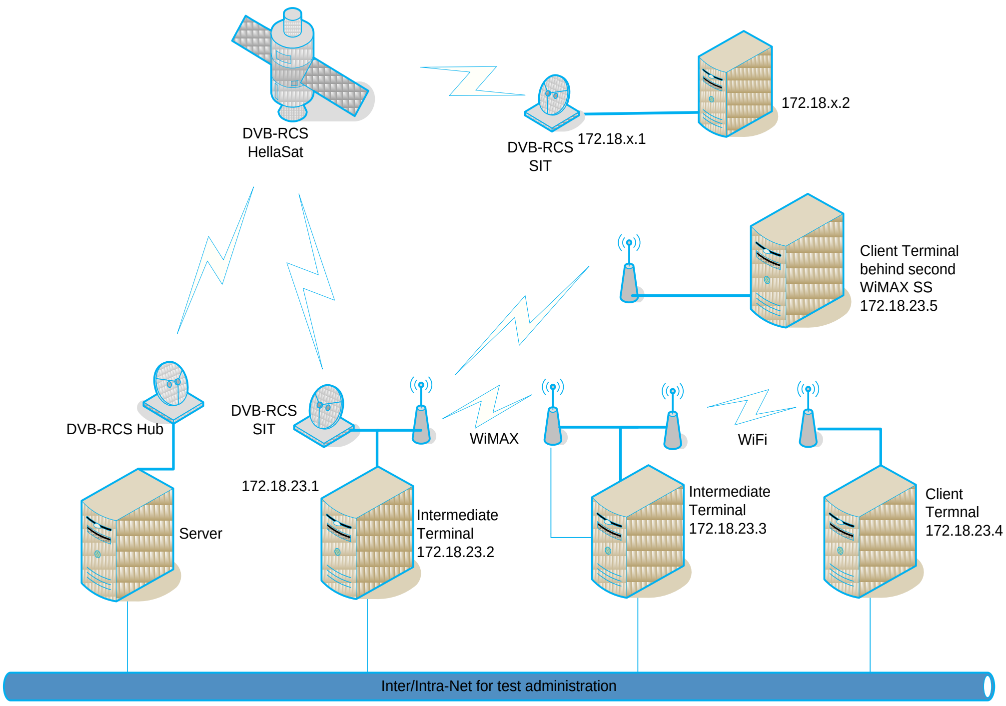


Figure 5.3. Hybrid Network DVB-RCS Test Scenarios

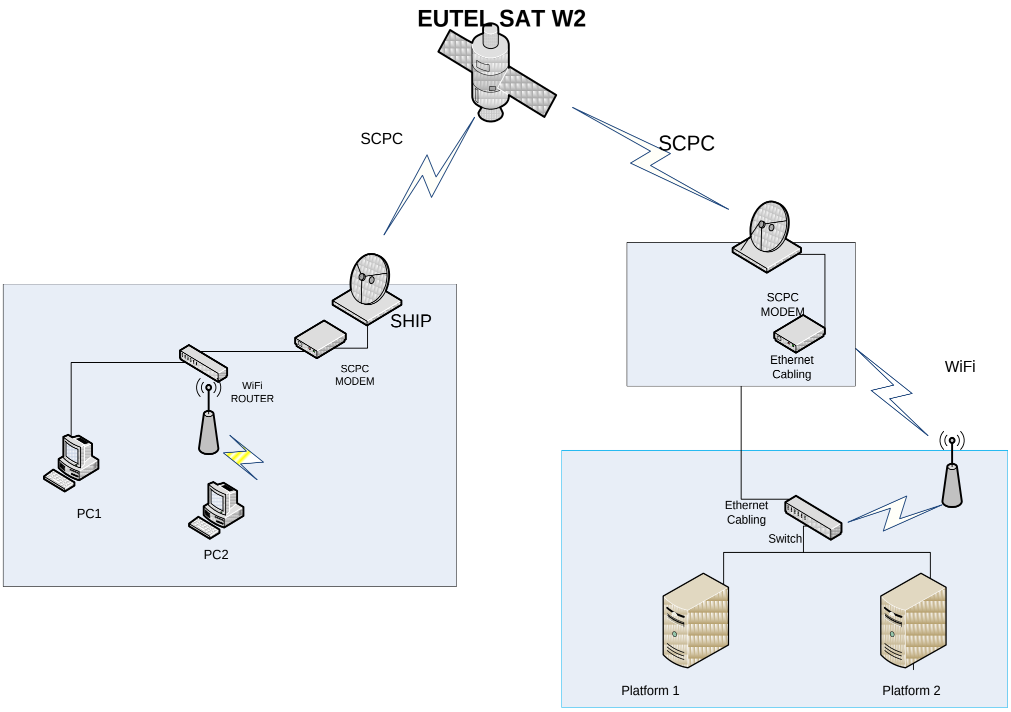


Figure 5.4. SCPC Test Scenarios

5.6 Summary

This chapter describes the research methodology, test process and testing methodology, along with the test setup and architecture. The testing methodology is based on finding of literature review, applications consideration and QoS requirements and product specifications for Wi-Fi, WiMAX, DVB-RCS and SCPC. The methodology considers all possible test scenarios for Hybrid Networks and allows testing for bandwidth, delay, jitter, latency packet loss, throughput, availability and performance of multicast and unicast services.

A test flow process diagram and testing methodology was produced using most of the common test tools. This methodology addresses effectively the testing of interoperability issues and can facilitate the analysis of QoS requirements in Hybrid Wireless Networks.

Chapter 6

Interoperability and Quality of Service Testing and Evaluation

6.1 Overview

Hybrid wireless communication technologies are becoming more common and complex. Satellite communication services are available in the form of different platforms, as, when and where required, with inherent properties of broadcasting, multicasting and capability of high speed internet connectivity. However, satellite communications alone cannot compete commercially in the communications market, unless it is integrated with terrestrial networks. Due to the advantage of availability, satellite communication networks are more and more getting integrated into the infrastructure of modern terrestrial communication networks and are becoming popular for the delivery of education contents, for teleconferencing, as well as for data, information and entertainment services.

Existing research on the interoperability of hybrid wireless networks and its testing has been limited. Previous studies have been discussed in the literature review (in Chapter 2). The interoperability and Quality of Service requirements in Wi-Fi, WiMAX, DVB-RCS, and SCPC (VSAT) are described and discussed in Chapter 3. The CLIX management and delivery frameworks and Isabel application were used in this study are discussed also in Chapter 3. The test tools were discussed and compared in Chapter 4. The test plan and testing methodology were produced in Chapter 5, covering comprehensively the interoperability issues and QoS requirements.

With the ever increasing demand for new services and applications, it is becoming essential that the wireless network architecture should seamlessly interoperate with new and existing technologies, protocols and standards particularly at end-to-end level. There is still a significant gap in end-to-end interoperability, and

Quality of Service (QoS) requirements such as delay, jitter, packet loss, latency, bandwidth, throughput measurement, availability testing, Transmission Control Protocol (TCP), User Datagram Protocol (UDP) performance and performance of multicast and unicast services. This thesis, attempts to cover all these issues and develops a methodology to address the interoperability and QoS issues. The testing of physical characteristics of hybrid networks, such as signal strength, attenuation, distortion level and signal to noise etc., were not part of this study.

A series of tests were carried out on a typical hybrid network test bed in order to establish the end-to-end interoperability requirements for the heterogeneous Satellite Terrestrial wireless communication technologies on various scenarios by using measurement and analysis tools as detailed in chapter 4, following the testing methodology as explained in chapter 5. This test plan and testing methodology was developed specifically in order to consider end-to-end interoperability issues and QoS requirements.

The objective of the testing was to investigate and identify any end-to-end interoperability issues and Quality of Service requirements in hybrid wireless networks.

The test bed of this study was the BASE² project aims at identifying, designing and deploying e-Learning services over an integrating broadband terrestrial networks with satellite broadband technologies for isolated agrarian communities in Greece and Cyprus as well as maritime communities on ships. The BASE² network architecture supports objectives of the project by deploying and operating educational content generation, management and delivery systems for different modes of learning such as live virtual classroom, video conference, offline asynchronous learning, over different network technologies to a large number of sites. The end-to-end broadband infrastructure was an integration of broadband terrestrial networks with satellite broadband technologies using DVB/DVB-RCS, Wi-Fi, WiMAX, and SCPC (VSAT). The main objective of this project was the implementation of an end-to-end network for tele-education applications.

The motivation was to identify the needed capabilities and main challenges for the seamless operation of the emerging heterogeneous satellite-terrestrial wireless networks.

The main contribution of this research was comparison and selection of test tools, a test plan and testing methodology of how to obtain and analyse testing results in hybrid networks, through real implementation, testing and demonstration of findings using available/existing live hybrid networks which employ DVB-RCS, WiMAX, Wi-Fi and, SCPC (VSAT) and were designed to deliver e-learning services to remote areas. The obtained results are not limited to tele-education services and can be generalised for all hybrid satellite-terrestrial networks (having similar characteristics). In these tests, various traffic data, packets and traces were captured. The results - in terms of end-to-end interoperability issues and QoS requirements such as delay, jitter, latency, bandwidth, throughput, TCP, UDP performance, link characteristics, availability testing, unicast and multicast testing are presented, discussed and analysed in this chapter.

This chapter is organised as follows: Section 6.2 shows preliminary interoperability testing locations, scenarios, setup and evaluation procedures. Section 6.3 shows analysis of the results for interoperability issues for Wi-Fi, WiMAX, DVB-RCS and SCPC. Section 6.4 presents the final interoperability test setup, test machines, link configurations and scenarios for Wi-Fi, WiMAX, DVB-RCS and SCPC. Section 6.5 shows the test measurements, as well as evaluates and analyses the test results for long and short term measurements, delay, jitter, latency, bandwidth, RTP analysis, unicast, multicast, TCP, UDP, HTTP web site access, throughput measurements and availability testing. Finally, Section 6.6 summarises and concludes this chapter. Table 6.0 shows the breakdown detail of various sections, test scenarios, test types, objective to test and test tools.

Table 6.0: Detail Breakdown of Test Results

Sections	*Test Scenarios*	*Test type*	*Objective*	*Test Tools*
6.3.1 to 6.3.4	DVB-RCS (Hub) to Wi-Fi and WiMAX) (Figures 6.1 to 6.4)	Delay(RTT), Jitter, Bandwidth, Packet Loss, TCP, UDP, throughput	Connectivity, Audio, Video quality, TCP traffic, UDP traffic	Iperf, Ping, PRTG
6.3.5	SCPC Hub to SIT (Figure 6.10)	Delay	Connectivity	Ping, Traceroute
6.5	DVB-RCS (Hub) to DVB-RCS (SIT) DVB-RCS (Hub) to Wi-Fi and WiMAX, WiMAX to WiMAX/Wi-Fi (Figure 6.8)	Delay, Jitter	Connectivity	Ping
6.5.1 to 6.5.2	DVB-RCS, WiMAX, Wi-Fi	Delay, Jitter, Packet Loss, Latency	Long term and short term measurements (Link characteristics with or without load	D-ITG
6.5.3	DVB-RCS(Hub) to DVB-RCS (SIT), DVB-RCS(Hub)to WiMAX/Wi-Fi, WiMAX to WiMAX/ Wi-Fi, DVB-RCS (SIT) to DVB-RCS (Hub), Wi-Fi/WiMAX to DVB-RCS (Hub)	Delay, Jitter, Packet Loss, Bandwidth, Latency, Throughput	Various traffic analysis	TCPdump, T shark, Wireshark, Ping
6.5.4	DVB-RCS (Hub) to DVB-RCS (SITs)	Bandwidth, Jitter, Loss	RTP analysis for real life session	U2m Network analyser

6.5.5	DVB-RCS (Hub) to DVB-RCS (SITs) DVB-RCS (SITs) to DVB-RCS (Figure 6.9)	Bandwidth, Packet loss	Unicast and Multicast measurements analysis	Httperf, U2m network analyser, Mgen, Wireshark
6.5.6	DVB-RCS	Throughput, RTT	TCP measurement	Httperf, Gnuplot
6.5.7	DVB-RCS	Throughput	Performance of Sustained HTTP/TCP	Ping, Gnuplot
6.5.8	DVB-RCS	RTT	HTTP web access/ Load times	Httperf, Firefox
6.5.9	DVB-RCS(Hub) to DVB-RCS (SITs) + WiMAX/ Wi-Fi	Latency, Rate , Loss, jitter	UDP measurements	U2m analyzer, Mgen
6.5.10	DVB-RCS (Hub) to SITs	Packet Loss, RTT	Availability measurements	Ping
6.5.11	DVB-RCS (Hub) to SITs	Jitter, Latency, Rate	Satellite Throughput measurement	U2m, Gnuplot
6.5.12	SCPC (Hub) to SIT	Delay, Packet loss, Bandwidth, Jitter, Data Rate	SCPC testing results for Ship	TCPdump, Ping, Wireshark

6.2 Preliminary Interoperability Testing Locations, Scenarios, Setup and Evaluation

6.2.1 Testing Locations and Scenarios

The overall networking infrastructure comprises two parts: an access part and a core part. The core network consists of a satellite-based wide-area infrastructure, and a possible terrestrial wireless extension via WiMAX 802.16. The access networks use Wi-Fi 802.11 technology and connect to the core network.

The network platform is deployed at various sites in Greece. The preliminary tests took place at OTE (Greek Telecommunication Organization) and HAI (Hellenic Aerospace Industry) Greece premises. In this pre-test phase, only a Digital Video Broadcasting Return Channel via Satellite (DVB-RCS) connection was available. For the preliminary test of the DVB-RCS platform at OTE premises at Psalidi Attikis, Greece, scenarios were as following:

1st scenario: DVB-RCS – WiMAX – Internet interconnection with satellite hub, server and 1 site (Fig 6.1)
2nd scenario: DVB-RCS – Internet interconnection with satellite hub, server and 2 sites (Fig 6.2)

For the preliminary test of the DVB-RCS platform at HAI premises (at Oinofyta Attikis), the following scenarios were involved:
3rd scenario: DVB-RCS with satellite hub, server and 3 sites (Fig 6.3)
4th scenario: DVB-RCS – Wi-Fi- interconnection with satellite hub, server and 1 site (Fig 6.4)

6.2.2 Physical Setup

The Network points were setup at OTE and HAI facilities.
The Network Points for OTE Facilities were:
Satellite Interactive Terminal (SIT1)/Personal Computer (PC1): NCSR facilities,

- *SIT2/PC2: OTE facilities, and*
- *HUB/PC-HUB: OTE facilities.*

Three PCs were set up, and the assigned bandwidth capacity for this satellite network was 512Kbps Forward link / 512Kbps Return link
The Network points at HAI Facilities, four PCs were set up:

- *SIT1/PC1: HAI facilities,*
- *SIT2/PC2: HAI facilities,*
- *SIT3/PC3: HAI facilities, and*
- *HUB/PC-HUB: HAI facilities*

The assigned bandwidth capacity for this satellite network was 1024Kbps Forward link / 1024Kbps Return link.

6.2.3 Network Evaluation Procedure

A number of tests were performed at OTE premises in order to verify the full IP unicast connectivity, which included sending unicast IP packets from each to the rest of the PCs, measuring the delay and jitter (just ICMP echo request/reply packets) and verifying the maximum transmission unit (MTU) of the link between each PC.

Once the connectivity tests were successfully completed, Isabel platform evaluation tests were conducted. In this phase of evaluations, an unicast session with the PC-HUB acting as flow server and a multicast session between four PCs were established. Both of the scenarios, the traffic characteristics and bandwidth were measured to identify the characteristics of the application traffic, and to evaluate the amount of traffic flow in a session (audio/video/signalling). The sent and received traffic were compared to determine the packet loss figures.

Additionally, the response statistics of the Ping tool were used to find out the minimum and maximum round-trip delay (min/max), the average round-trip delay on the link (avg), and the mean deviation of the round-trip delay (jitter). These measurements were used to characterize the delay and jitter properties of a specific path.

The Iperf test tool was used to test the TCP throughput of the link under default settings and to verify the link's bandwidth suitability for UDP traffic under congested conditions, by sending 1470 byte packets.

Iperf servers and clients were setup to evaluate the multicast scenarios, so that the application can be used as a virtual "multicast ping" and also to confirm the basic IP multicast connectivity.

6.3 Preliminary Interoperability Test Results

6.3.1 Scenario 1: DVB-RCS – WIMAX

This scenario used two PC terminals: one at NCSR Greece connected via Internet to DVB-RCS HUB station at OTE facilities at Maroussi Greece and the other one located at OTE premises at Psalidi Greece, attached to remote DVB-RCS Satellite Interactive Terminal (SIT) via WiMAX. Both the SIT at Psalidi and the HUB station at Maroussi used a 512/512 Kbps Forward/Return link to connect to the Hellas Sat satellite as shown (Figure 6.1). The WiMAX connection was non Line of Sight (NLOS).

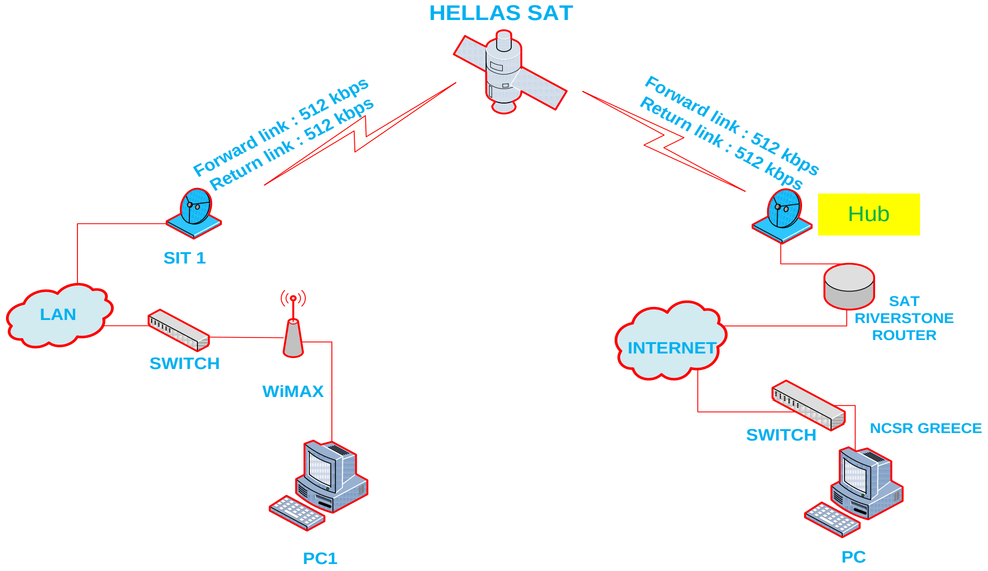


Figure 6.1: DVB-RCS WiMAX Network topology of the 1st Scenario

Running the Iperf tests with bandwidth as the test parameter, it was observed for the UDP sessions (although 512 Kbps were allocated in both the forward and return channel) that datagrams could only be successfully sent to their destination when the bandwidth parameter was 480 Kbps maximum. For bandwidth-parameter values more than 480 Kbps, datagrams failed to reach the destination. The detail of measurements are shown in Appendix B.

This means that 480 Kbps is the limit to operate effectively out of 512 Kbps allocated bandwidth. In the case of the first scenario with the given bandwidth allocated, no problem was encountered. The Isabel application worked unobstructed, with all the default settings for audio and video, providing a very satisfying media quality.

On the other hand, the unavoidable time delay of 600 ms due to satellite caused no significant disturbance of communication that could degrade the level of multimedia quality. The presence of WiMAX did not impose any significant time delay in the system other than the default satellite delay.

6.3.2 Scenario 2: DVB-RCS

This scenario consists of three PC terminals: one at NCSR connected via Internet to DVB-RCS HUB station (SAT Riverstone Router) at OTE facilities at Maroussi and the other two located at OTE premises at Psalidi Greece, attached directly to two remote DVB-RCS Satellite Interactive Terminals (SIT), i.e. Psalidi 1 and Psalidi 2. Both SITs at Psalidi and the HUB station at Maroussi used a 512/512 Kbps Forward/Return link to connect to Hellas Sat as shown (Figure 6.2).

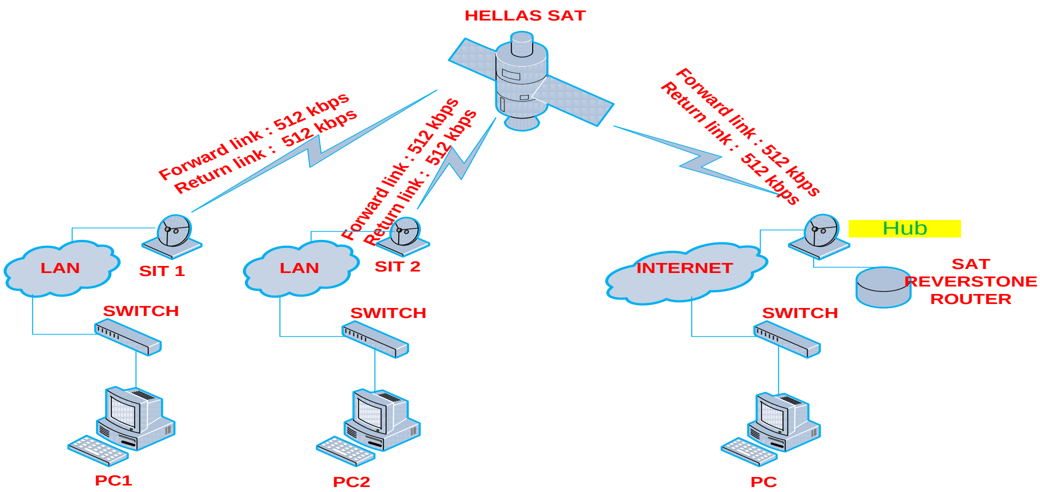


Figure 6.2: DVB-RCS Network Topology of the 2nd Scenario

In this scenario, the Ping tool was used to check the connectivity between PSALIDI 1 and PSALIDI 2 terminals. The communication between the two terminals was successful with 1838 ms delay. An Iperf test was run for TCP analysis using 16.0 Kbytes window size between server PSALIDI 2 and client PSALIDI 1, server NCSR and client PSALIDI 1, server PSALIDI 2 and client

PSALIDI 1. All of the combinations reported having successful connectivity between each other.

This scenario produced significantly different results from the first scenario. This was due changes made to the network parameters on the Isabel default settings to enhance the transferred media quality. However, the results observed were worse, instead of being better, with satellite time delays resulting as high as 5 seconds, which was much higher than the results obtained using the default Isabel setup.

High quality media streams also have a tendency to increase time delays. This is mainly caused when the used bandwidth for the session media stream reaches the limit of allocated values, i.e. 512 kbps, it surpasses the channel bandwidth limit, as further delay is added to the standard satellite delay of approximately 600 ms (300ms Earth-Satellite plus 300ms Satellite-Earth). In this case, with only 512 Kbps allocated bandwidth for both forward and return link and each site producing approximately 300 kbps streams, maximum capacity was exceeded and substantial delays made communication not possible. As a result, the audio and video had a lot of interruptions, in most cases there was no video image at all. The system itself became unstable and frequently collapsed. This was due to an overload situation and packets were delayed due to excessively large packets queues. The detail results are shown in Appendix B.

Overall, this scenario was complicated due to the specific restriction of the allocated bandwidth of 512Kbps. This problem was overcome by allocating more bandwidth, as proved in the following tests that took place at the HAI premises, that used an uplink/downlink capacity of 1024/1024Kbps. Video streams of high quality were sent unobstructed and with acceptable time delays by allocating what seems to be the proper bandwidth of 1Mbps.

6.3.3 Scenario 3: DVB-RCS

This scenario consists of four PC-terminals. Three of them were attached to remote DVB-RCS Satellite Interactive Terminals, and the fourth was attached to the DVB-RCS HUB station, all located at HAI facilities at Oinofyta Attikis Greece (Figure 6.3).

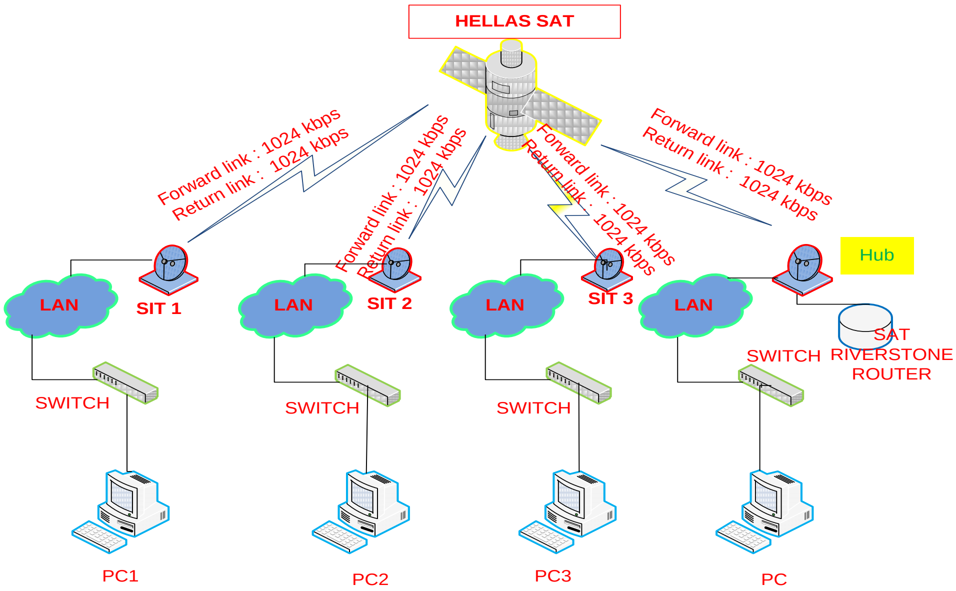


Figure 6.3: DVB-RCS Network topology of the 3rd Scenario

In this scenario, the three-site connectivity was examined by looking at packet loss ratios for all links. 50 packets were exchanged among various sites and delivered successfully with 0% packet loss except for PC2 to PC1, PC2 to PC3, and PC2 to Hub packets where packet loss was 5 to 10 %. These losses may be due to physical error. From Hub to PCs, from PC1 to PC2, and from PC1 to PC3, there were no losses. This shows that the network connectivity for this scenario was maintained efficiently. The detail results are shown in Appendix B.

Tests for delay and jitter were performed. The results for the delay and jitter measurements show that the total round trip time (RTT) was within the range of 3000 ms. The minimum, maximum and average RTT for all paths in the scenario was calculated. The RTT 3000 ms was considered to be a significant amount of delay. This was a foreseeable result, due to the fact that satellite links added a significant time delay. With regards to RTT average, the delay variation among the paths was ranging from as low as 670ms up to 2200ms.

During both unicast and multicast sessions, packet capturing applications were running on both ends of the network and packet transmission was recorded for 20 minutes.

Even though the default multimedia quality setting was changed on the Isabel platform, having allocated twice the bandwidth of what was used in the OTE interoperability tests, 1024kbps for both forward and return link, no particular problems emerged, although various sites were communicating simultaneously. The Isabel application worked efficiently by providing a satisfying media quality. On the other hand, the time delay of 600ms-1000ms was observed due to satellite delay.

The captured packets were divided into four distinct groups, namely Audio, Video, Other UDP traffic and TCP signalling for analysis. The analysis of the results show that the highest packet loss occurred in UDP traffic between the HUB station and the PC terminals (1 and 3).

However, the packet losses in audio video traffic between the HUB station and the PC terminals were in the range of 0.02% and 0.04%.

Additionally, during the conference sessions, the network bandwidth was monitored using the Paessler Router Traffic Grapher (PRTG) application.

6.3.4 Scenario 4: DVB-RCS – Wi-Fi

This final scenario consists of two PC- terminals. One was attached to a remote DVB-RCS Satellite Interactive Terminal via Wi-Fi, and the other was attached to the SAT Riverstone DVB-RCS HUB station (Figure 6.4).

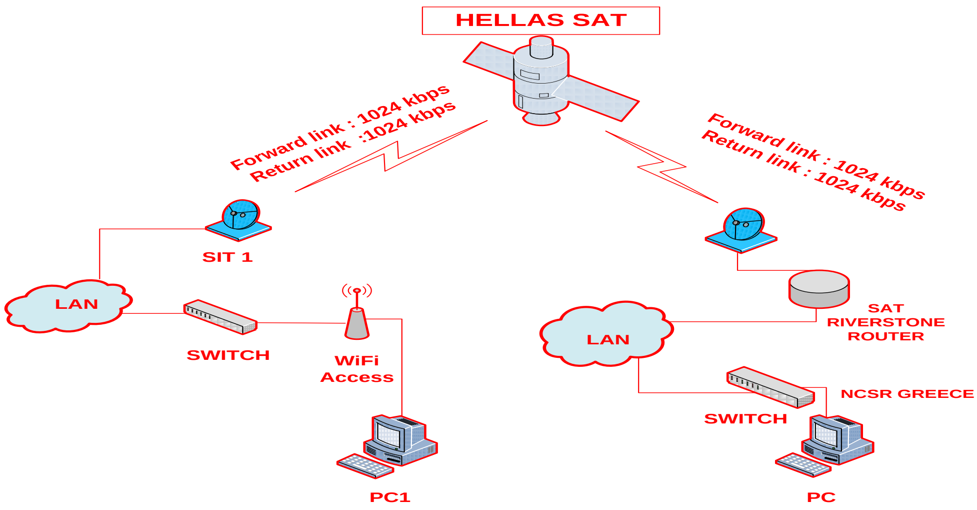


Figure 6.4: DVB-RCS Wi-Fi Network topology of the 4th Scenario

This scenario's aim was to investigate connectivity and delay issues, in hybrid networks where satellite networks are linked to Wi-Fi local area networks. In terms of connectivity, 50 packets were exchanged among the various sites and all were delivered successfully, with 0% packet loss.

However, with Wi-Fi, mobility is considered to be a crucial issue for measuring the time delay. To assess the mobility effect on delay measurements, a portable laptop (a client) connected via Wi-Fi to PC1, located out of the operating range (100 metres as per Wi-Fi specification) coverage and time delays increased up to 1 sec. The mobile user then returned within the Wi-Fi range and time delays returned to normal, i.e. satellite average delays of approximately 600ms. The detail results are shown in Appendix B.

Twice the bandwidth used at OTE interoperability tests was allocated, and there was only one site connected to the Hub, therefore, no problem was encountered during the session with all the default settings for audio and video providing a very satisfying media quality. It is vital to reveal that the presence of Wi-Fi equipment did not impose any significant time delay in the system other than the default satellite delay of 600ms. It is therefore concluded that the main bottleneck was only the satellite link in DVB-RCS interconnection terrestrial wireless link

(Wi-Fi in this scenario, and WiMAX in the 1st scenario). The bandwidth problem was also noted in the 2nd scenario.

The client with a mobile laptop connected via Wi-Fi to SIT1, was moving around within the range of the Wi-Fi antenna during the Ping measurements. During tests, the Wi-Fi connection was established from a non Line of Sight (NLOS) location. At distances outside the range of Wi-Fi coverage, time delays increased up to 1 second. The maximum connectivity range for Wi-Fi was approximately 100 meters.

6.3.5 SCPC for Maritime Scenario

Single Channel per Carrier (SCPC) provides a simple means to connect VSAT stations. Two dedicated satellite channels (forward and return) are provided between each VSAT station. A mesh topology allows direct (single hop) communication between each station.

The maritime scenario was based on SCPC, which is the simplest satellite modem technology behaving like a leased line connection. IP unicast and multicast traffic can easily be encapsulated. Some basic tests using ping and traceroute were carried out from the NCSR premises in Athens to the vessels 'uplink station to verify the connectivity, modem and link operation (see figure 6.10).The test configuration was set up as follows:

Subnet 172.16.22.0/24 Network behind SCPC modem at NCSR

Subnet 10.254.9.0/25 Network behind SCPC modem at provider

The PC at NCSR was at IP address 172.16.22.2.

The 'traceroute' output shows the IP traffic being routed across the satellite modem, which can be inferred from the Round Trip Time (RTT) of about 570 ms (Figure 6.5):

Traceroute to 10.254.9.35 (10.254.9.35), 30 hops max, 40 byte packets

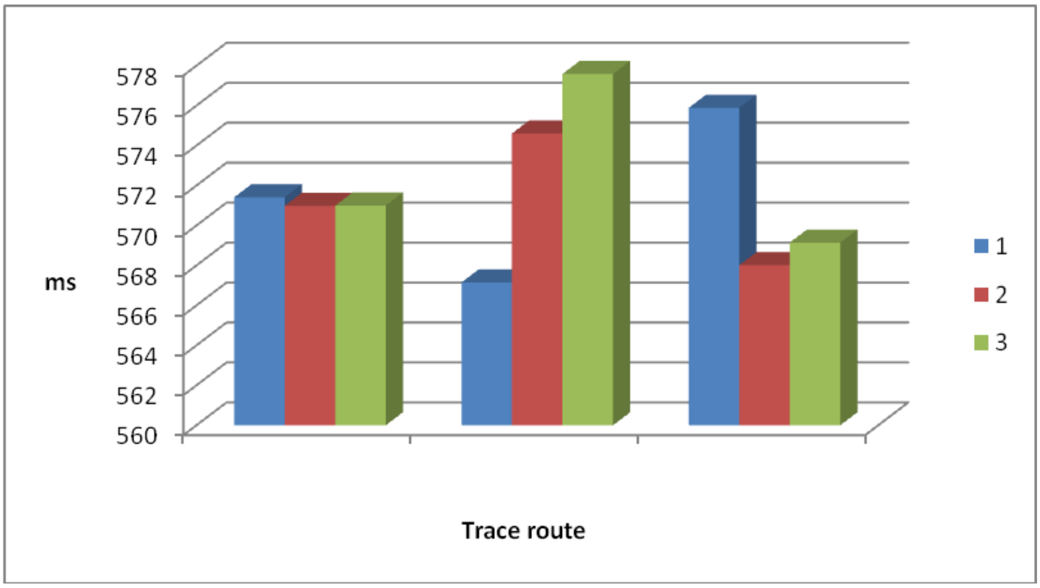


Figure 6.5: Trace route   SCPC RTT

The ‘ping’ output shows a relatively small variance in the RTT which is typical for a non-shared SCPC link (Figure 6.6):

PING 10.254.9.35 (10.254.9.35) 56(84) bytes of data.

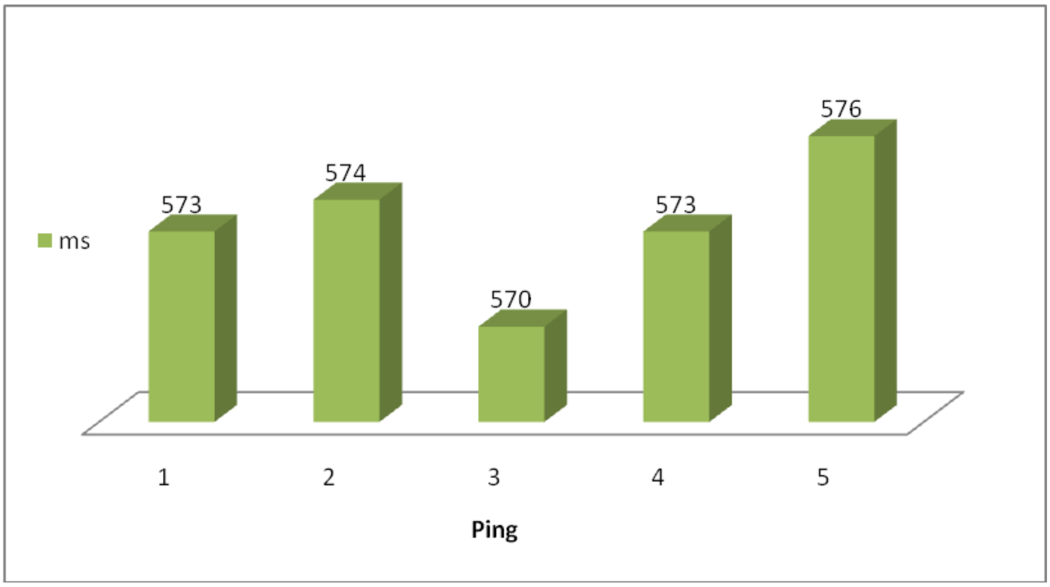


Figure 6.6: Ping Statistics SCPC RTT

5 packets transmitted, 5 received, 0% packet loss, time 4024ms

rtt min/avg/max/mdev = 570.044/573.447/576.170/1.973 ms

Further tests were conducted at NCSR, Greece and Fokus, Germany in April-May 2008. The results, including raw data, IP header, and TCP or UDP header, are discussed and analysed in the following sections.

6.4 Final Interoperability testing

Further interoperability testing was carried out to determine the end-to-end interoperability issues and QoS requirements of heterogeneous satellite-terrestrial networks for seamless operation. The testing examined the interoperability requirements in heterogeneous satellite, terrestrial wireless network technology.

All testing and measurements were carried out at different sites and nodes as detailed in the test methodology (chapter 5). Testing parameters were set and verified by using different test tools, such as ping, tcpdump, Distributed Internet Traffic Generator (D-ITG), PRTG, mgen, Iperf, Httperf, wireshark/tashark, and focus u2m NET analyzer. These tests assess the end-to-end interoperability issues and QoS requirements such as bandwidth, delay, jitter, packet loss (loss data rate), latency, throughput, unicast and multicast, TCP, UDP, HTTP measurements, connectivity and network availability. Most of the tests were conducted at Fokus Germany, HAI Greece and NCSR Greece.

A number of tests have also been run for multiple remote sites. These sites were tested for both audio and video streams for both with-load and without-load conditions. The packets were captured on flow server boxes at various sites, and results were analysed with the wide range measurements of statistics which is shown below. The results include delay time, detail of conversation, details of source and destination, protocol hierarchy, error warning, and checksum details.

The network architecture depicted below supports hybrid wireless network technologies. This model provides end-to-end integrated services and applications that can be delivered to a large number of sites. Figure 6.7 presents the DVB-RCS network architecture (Abbasi *et al.* 2008; Abbasi *et al.* 2008; Stergioulas *et al.* 2008)

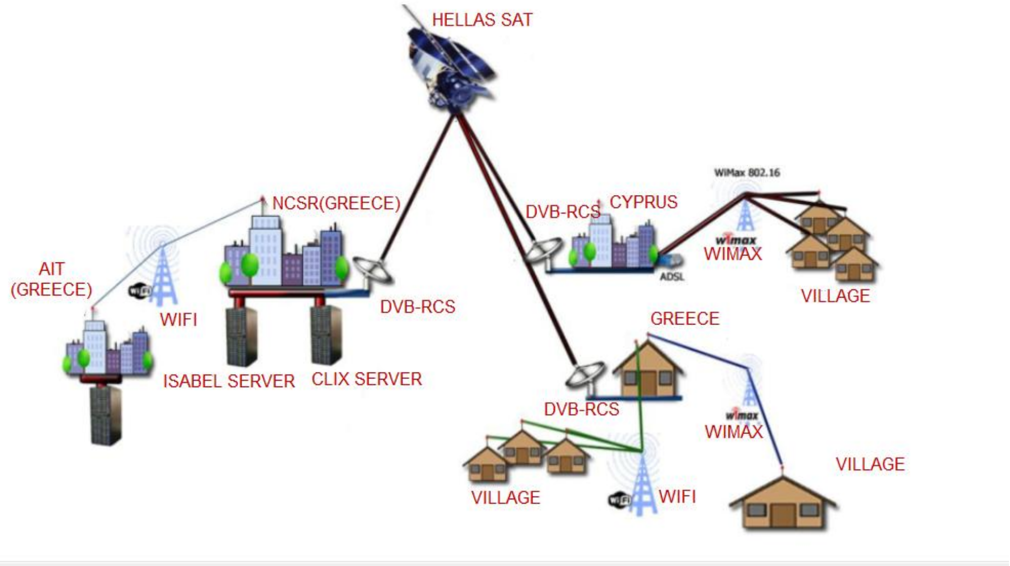


Figure 6.7: Hybrid Wireless Network - Satellite (DVB-RCS) and Terrestrial (Wi-Fi, WiMAX) - Architecture.

The following sections describe the configuration of the test computers, the configuration of the different network topologies and the test setup.

6.4.1 Configuration of the Test Computers

All the computers used in the trial were Linux-based Pentium4/Core2 (or faster) PCs running recent versions of their respective distribution (Debian-derived Ubuntu in most cases). The test platform relied on either DVB-RCS or SCPC as the backhaul satellite technology. Terrestrial connectivity was established via WiMAX and Wi-Fi.

6.4.2 DVB-RCS-based link configuration

The DVB-RCS-based scenario was implemented to connect the twelve sites. The ten sites were located in Greece and two in Cyrus. At each site, a DVB-RCS remote terminal was installed to create the backhaul network. An additional remote terminal was set up at the FOKUS site in Germany to connect the local WiMAX/Wi-Fi test bed with the DVB-RCS backhaul network. User PCs were connected to the remote terminal via WiMAX and/or Wi-Fi.

The DVB-RCS system used for testing was configurable up to 45 Mbps downstream, and up to 2 Mbps upstream as per own (proprietary) specifications. However, the DVB-RCS system was configured to 1Mbps per downstream flow. The upstream bandwidth was set to 128 kbps to support audio streams and to 320 kbps to support both audio and video streams. The return channel access strategy was set to Continuous Rate Assignment (CRA), which ensures low latency and jitter.

The WiMAX link was configured for up to 6 Mbps real-time polling traffic in both directions, which yields a RTT of about 30ms and a jitter of about 10 ms. The 6 Mbps bandwidth allocation, allows enough space for even multiple streams to be requested in parallel.

Wi-Fi was configured to automatically select the optimal link speed. The minimum data rate according to the IEEE 802.11a/g standards is 6 Mbps, which was also the default rate for multicast transmissions. Latency and jitter for Wi-Fi were just a few milliseconds.

The bandwidth requirements were chosen after some initial testing, as a trade-off between satellite bandwidth costs and the audio-visual quality of the content:

- *Downstream: 1Mbps per audio/video stream*
- *Upstream: 128 kbps per audio stream*

The minimum bandwidth required for the unobstructed transmission of video streams was found from experiments to be 1 Mbps downlink/ 1 Mbps uplink, when running the ISABEL application over the satellite (DVB-RCS) network. When running the ISABEL application over an end-to-end terrestrial network, the bandwidth could be reduced to 384 Kbps / 384 Kbps without any distinguishable reduction of quality. However, capacities less than 384 Kbps symmetrical over a terrestrial network, or 1 Mbps symmetrical uplink and downlink over the satellite network, presented some problems while the ISABEL connection was sometimes forced to terminate. The bandwidths below the minimum capacities (e.g. 384 kbps), the audio and video streams significantly reduce and result in very poor Quality of Perception (QoP).

The one-way latency has a physical minimum of around 300 ms for geostationary satellites. The technology-introduced latency should be kept to a minimum in order to maintain a sensitivity of interactivity (Ghosh *et al.* 2005). Jitter should be kept as low as possible since the size/duration to the de-jitter buffers adds to the latency of the signal. Maximum and sustained TCP throughput was measured using httperf.

Each test was run five times and the average and deviation were calculated and reported. The results were visualized with gnuplot. The FireFox extension FasterFox was used to measure the access/download times of a static website hosted on a machine at HAI's DVB-RCS hub station. This test was repeated five times and the average time was calculated. This test indicates how smooth the web surfing experience is behind the examined link.

Command line tools like wget or puf did not provide satisfactory results since they either support HTTP pipelining or parallel fetching, but not the combination of both. Hence, the FasterFox plug-in was chosen.

6.4.3 DVB-RCS, WiMAX and Wi-Fi

A number of tests were carried out using various test scenarios using different nodes. The testing aim was to evaluate end-to-end interoperability issues and QoS related requirements for the various architecture scenarios. The testing scenario was Satellite only; Satellite and WiMAX; and Satellite, WiMAX and Wi-Fi. Figure 6.8 below depicts the main mechanism of the DVB-RCS-based network topology. The sites in Greece and Cyprus were configured similarly to the test bed at FOKUS at Germany. The FOKUS test bed machines, however, have additional wire-line Internet connectivity to keep test administration and management traffic separate from the actual test traffic flows and allow remote access. Each remote terminal (SIT) provides an IPv4 class-C subnet in the range 172.17.x.0/24. A Linux-based PC was used on SIT 172.18.x.2. Those PCs were typically used for network performance measurements, as well.

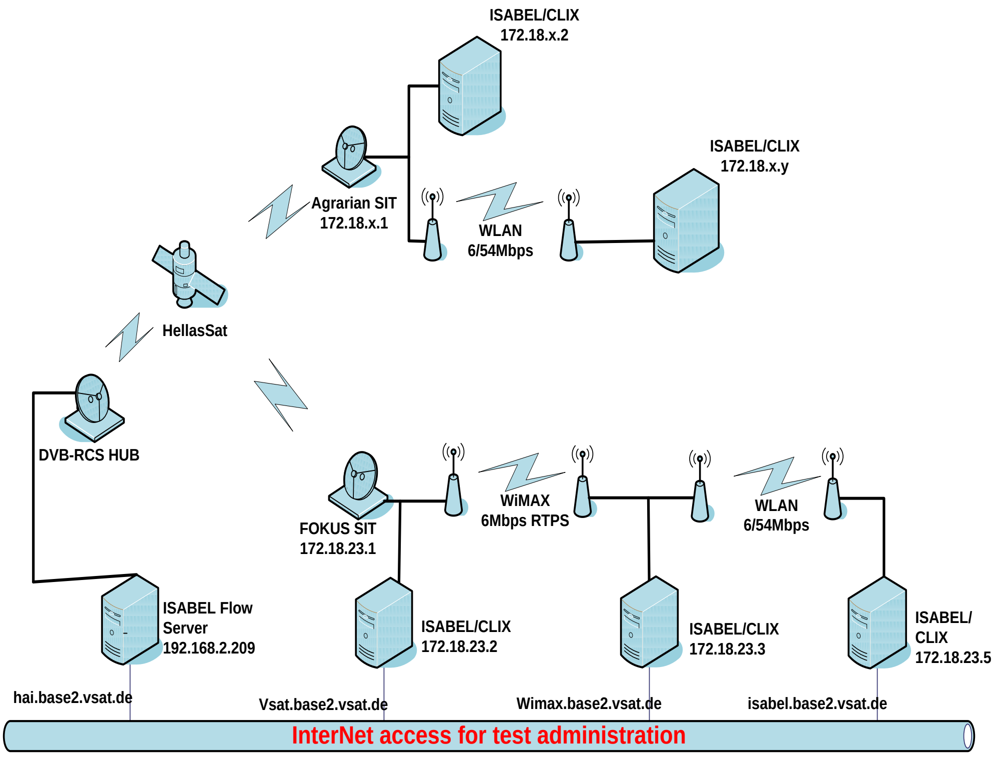


Figure 6.8: Actual DVB-RCS Test setup

6.4.4 DVB-RCS Scenario (Multicast)

In this scenario a mix of unicast and multicast traffic was sent. The network was divided in two remote legs. One composed of the satellite remote terminals, the satellite itself, and the DBV-RCS hub. The other leg was on the public internet to where both NCSR (Greece) and HAI (Greece) were connected (Figure 6.9). An Isabel flow server as joining these two legs, which acted as a gateway.

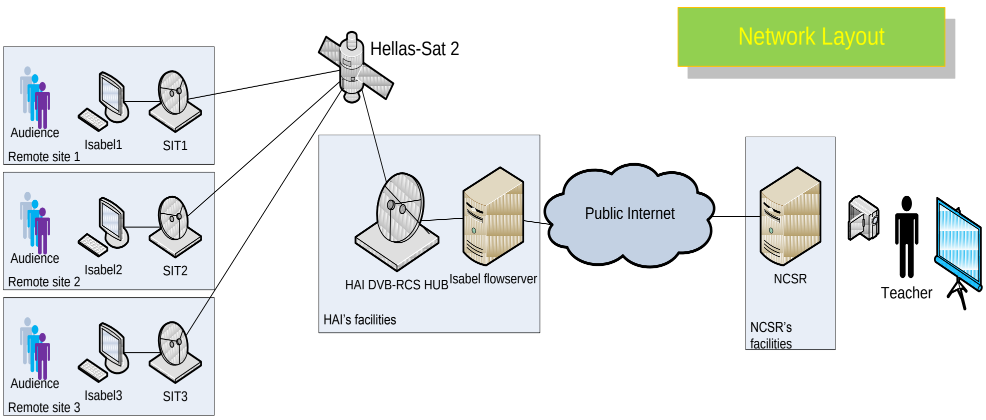


Figure 6.9: Network layout of multicast DVB-RCS scenario

In this scenario the nodes were: NCSR, Flow Server (FS), SIT1, SIT2, and SIT3. The total 5 tests with different test configurations were carried out and are referred test 1 to 5 in the test results (see section 6.5).

The Isabel overlay network connects each remote station (Isabel1, Isabel2 and Isabel3 in figure 6.9) to the Isabel flow server using multicast traffic that traverses the satellite IP network (low volume Isabel signalling traffic uses unicast).These three Isabel remote stations were located at HAI's facilities in Greece. The traffic was carried by the public internet from the flow server at HAI to NCSR Greece, (Isabel terminal). As there was only unicast connectivity in this leg, unicast was the only choice to connect NCSR to the flow server at HAI.

Flows generated by satellite terminals Isabel 1, 2, 3 travel to the Isabel flow server and the rest of the satellite terminals using IP multicast. Physically the flows travelled through the terminal's uplink and arrived to the DVB-RCS hub. The DVB-RCS hub replicate each multicast packet: one copy sent back to the satellite via the multicast uplink and received by the rest of the satellite terminals, the second copy sent through the local LAN to the Isabel flow server. This second copy was received by the flow server and relayed over the public Internet to NCSR Greece using IP unicast. Flows generated at NCSR were sent through the public Internet to the flow server, then sent through the satellite multicast uplink once, and received by all the satellite remote terminals.

This scenario saves much bandwidth in comparison with the unicast service. Over the satellite leg, those packets that went from the hub to the receiving stations were sent once and received by all the satellite terminals, instead of being replicated at the hub and uploaded many times. In this way, the bandwidth used for each flow does not rise by adding new terminals to the scenario.

6.4.5 DVB-RCS + WiMAX nodes and DVB-RCS + Wi-Fi nodes testing scenarios

In this testing scenario, a remote Isabel terminal was attached to WiMAX/ Wi-Fi links. The WiMAX/Wi-Fi links bridged to the Ethernet network of the DVB-RCS SIT. All the traffic between the SIT to the station was relayed through the WiMAX/Wi-Fi link. In this way, the WiMAX link was a seamless extension to the SIT, making the properties of these scenarios similar to the ones without WiMAX/Wi-Fi.

6.4.6 SCPC link configuration and testing setup

The Single Channel Per Carrier (SCPC) connections with Very Small Aperture Terminals (VSATs) are used to provide the satellite links. A satellite modem is connected with a point to point link with a service provider, comparable with a terrestrial telephone modem connection. The remote/maritime community members can access this service from public tele-education halls/classrooms. The access to the service was facilitated using wireless-enabled PCs over a broadband wireless network interlinked with a core satellite-based communication system.

A VSAT network enables multiple remote sites to communicate with a centralized location or network operations centre. VSAT terminals were installed at the Ship and are connected to the hub or to another VSAT terminal via a satellite link. A satellite modem was connected with another one in a point-to-point link with a service provider, comparable with a terrestrial telephone modem connection. The SCPC based scenario is used to connect the ship during the testing, as shown in figure 6.10. The VSAT-SCPC was configured with 512K bandwidth divided in 256kbps upstream and 256kbps downstream.

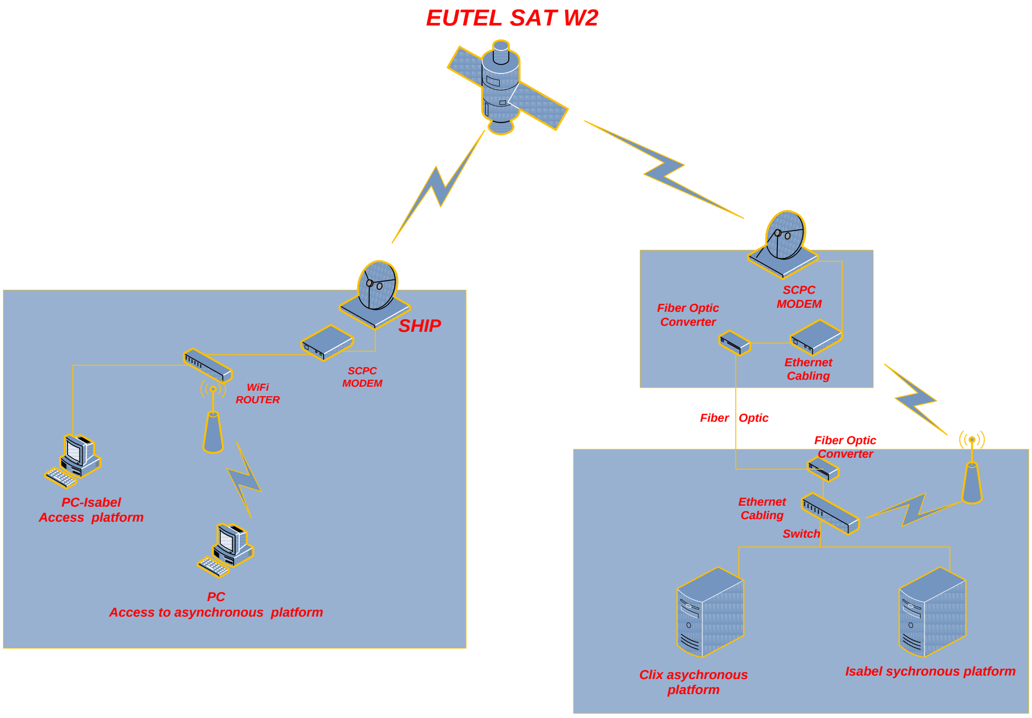


Figure 6.10: SCPC network architecture.

For SCPC, testing was carried out between HAI server and Bluestar 2 ferry. The SCPC network was tested to examine the interoperability issues in VSAT networks. Testing parameters were set up and verified by using different test tools, such as tcpdump, Distributed Internet Traffic Generator (D-ITG), ethereal/ethereal, wireshark/tashark for interoperability and performance measurements such as bandwidth, delay, jitter, packet loss, latency, throughput, and connectivity.

In these scenarios, Isabel PC at NCSR was connected through SCPC modem on NCSR side to Isabel PC on ship through SCPC modem.

The testing was carried out using different bandwidth settings such as 128k, 256k, 512k and 1MB for stationary or moving ships. The test configurations were set as follows;

- *128k/128k stationary ship*
- *256k/256k stationary ship*
- *512k/512k stationary ship*

- *1MB/1MB stationary ship*
- *512k/512 k small window stationary ship*
- *512k /512k large window stationary ship*
- *256k/256k moving ship*
- *512k/512k moving ship*

Tcpdump and tethereal command test tools were used to capture the data, files, packets and traces during. The Wireshark test tool was used for the analysis of the data.

IO graphs which are user configurable graphs were created from Wireshark. The results show that there were few packets which were lost; however, minor amounts of jitter were found in all these captured files. The jitter was around 60ms for traffic up to 512k in ship measurements. However, from 512k window captured packet analysis, it was found that there was high jitter and several packets were lost. The reason was that the system bandwidth was not supporting for this test the 512k (large) window scenario. From the analysis of the IO graphs, no considerable delay is found.

6.5 DVB-RCS and SCPC Result Analysis

The following section shows measurement and result analysis for both DVB-RCS scenarios as shown in the above figure 6.8, figure 6.9 and SCPC maritime scenarios as shown in figure 6.10. The traffic was generated to check the network and application performance. The first scenario examined where DVB-RCS satellite technology was used in the backbone of the network. In this scenario, WiMAX and Wi-Fi links were also available. The results of this DVB-RCS (DVB-RCS to Wi-Fi/WiMAX) scenario are shown in sub sections 6.5.1 to 6.5.11.

The sub section 6.5.12 shows measurement taken during testing sessions on ship, where SCPC–VSAT satellite technology used are presented. In DVB-RCS and SCPC scenarios (with various nodes), packet loss, delay, jitter, link characteristics, TCP, HTTP, UDP, availability and throughput results were examined for end-to-end interoperability.

To check the connectivity and delay initially a ping test was carried out between Wi-Fi-WiMAX, WiMAX-WiMAX and DVB-RCS Hub and various sites. The following figures (Figures 6.11-6.13) show the RTT (Round-Trip Time) results. The average RTT between Wi-Fi and WiMAX is 0.025 ms, and between DVB-RCS to WiMAX is 27 ms. The overall RTT was around 30ms.

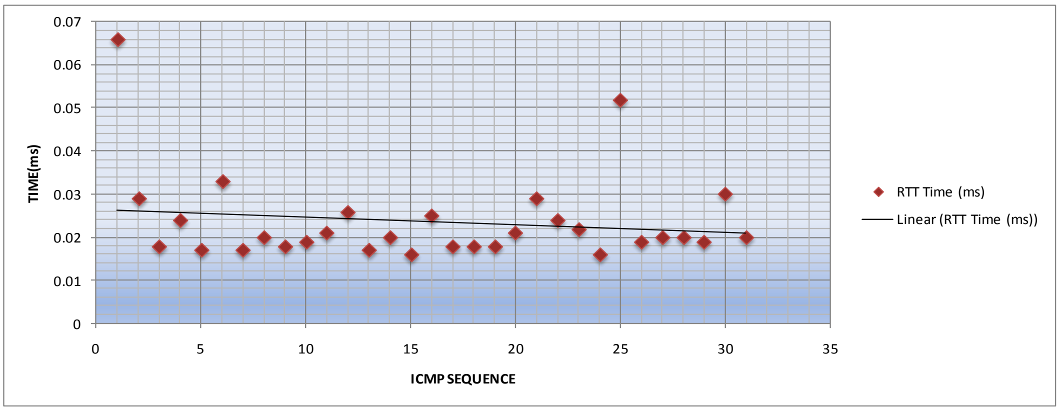


Figure 6.11: RTT WiMAX-Wi-Fi Ping statistics

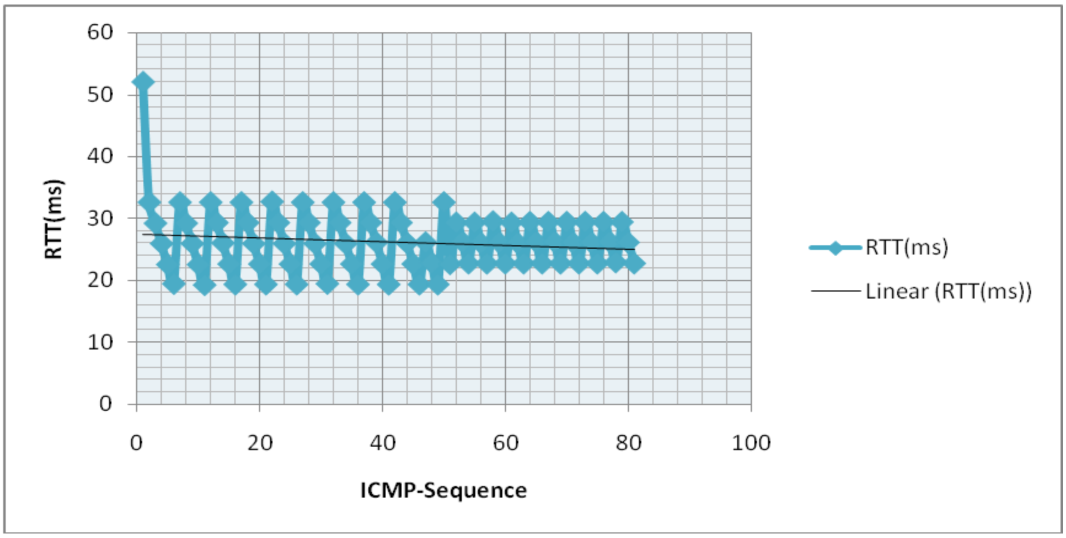


Figure 6.12: RTT DVB-RCS SIT to WiMAX Ping statistics

The RTT from DVB-RCS to SITs located at villages was around 600ms as shown in figure 6.13. Again in this scenario, there were a few peaks at the start of the ping response. Figure 6.13 shows that the jitter was minimal.

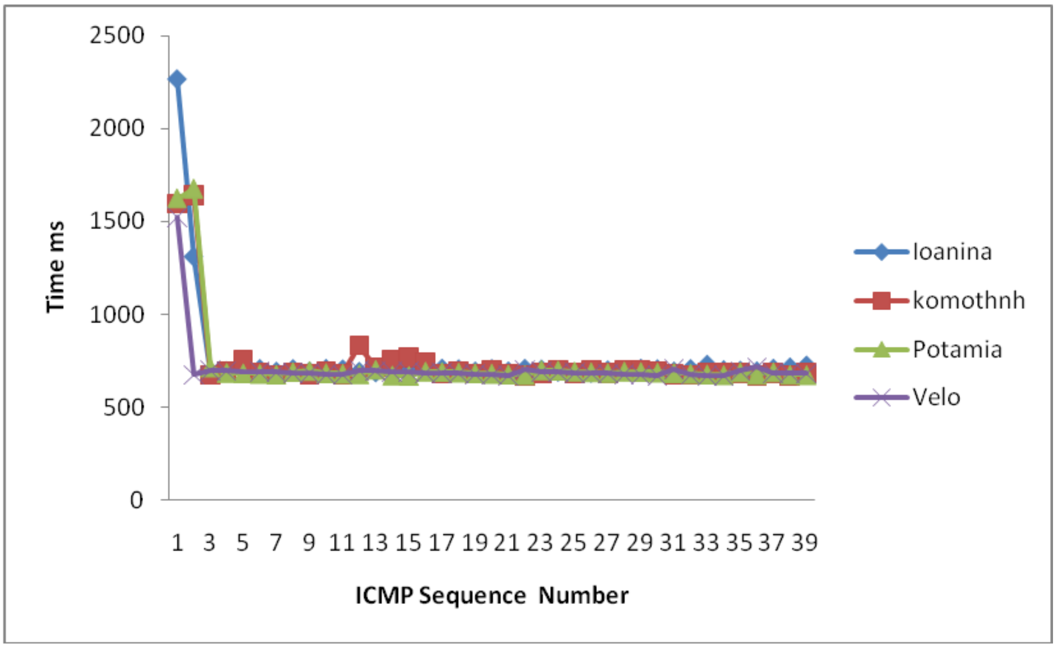


Figure 6.13: DVB-RCS-SITS Ping Statistics(Villages, Velo, Potamia, Komothnh, Ioanina)

6.5.1 Long Term Measurements

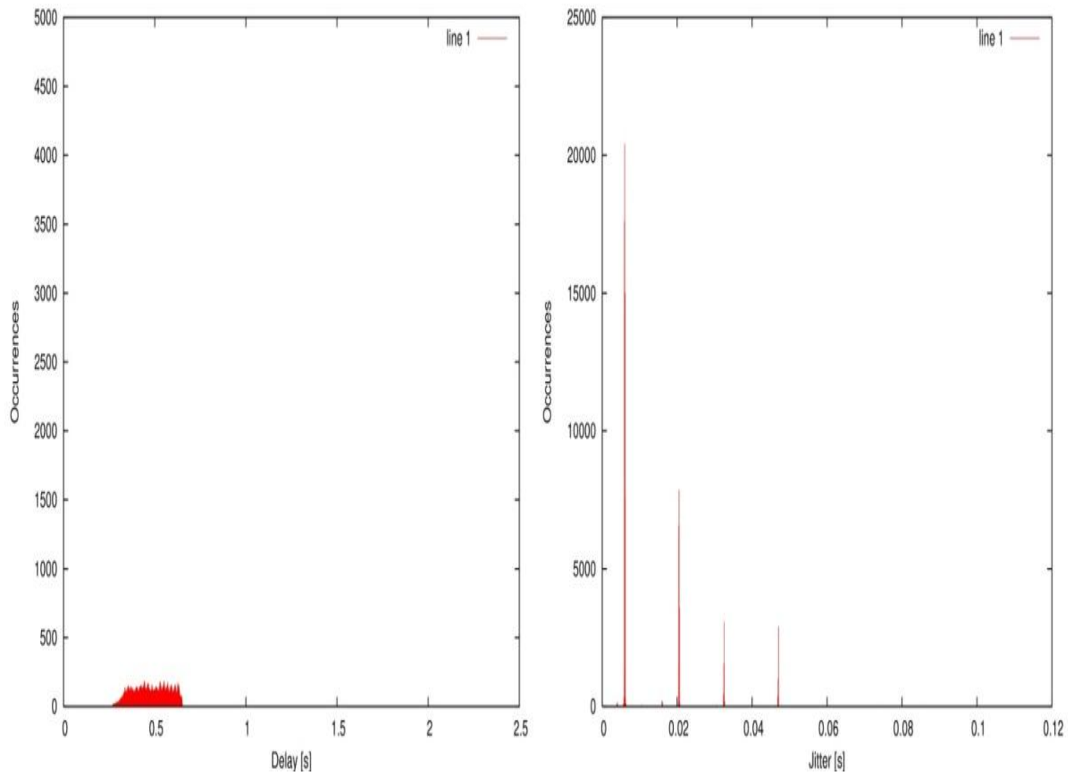
To determine the long-term stability of each link, a data stream with a small bandwidth of 100 bytes per packet, 10 packets per second was sent with D-ITG over two periods of 5 hours and 24 hours respectively. The packets were sent with the UDP Protocol to avoid retransmissions and congestion on the transport protocol layer.

In ideal links, there should not be any packet loss, delay, latency and jitter. However, real links suffer some of these issues, due to variation caused by time slots, and hardware retransmissions etc.

The following sub sections show long term and short- term measurements for Satellite DVB-RCS, WiMAX and Wi-Fi links which were visualized with dot-box and histogram plots. The dot plots give a complete overview of regularities and therefore are also valued for showing outliers. Histogram and box plots visualise statistical parameters like distribution of samples, first, second and third quartile. The whiskers of the box plot specify the minimum and maximum values, as long as they are less than 1.5 IQR (Interquartile range). Otherwise they are ignored as outliers.

6.5.1.1 DVB-RCS Satellites

The following figure 6.14 shows an example of delay and jitter plots of measurements for the duration of five hours. In the dot plots, there were only few outliers within the range of 0.3s to 0.6s delay. The jitter values group in small areas at approximately 0.005, 0.02, 0.032 and 0.048s.



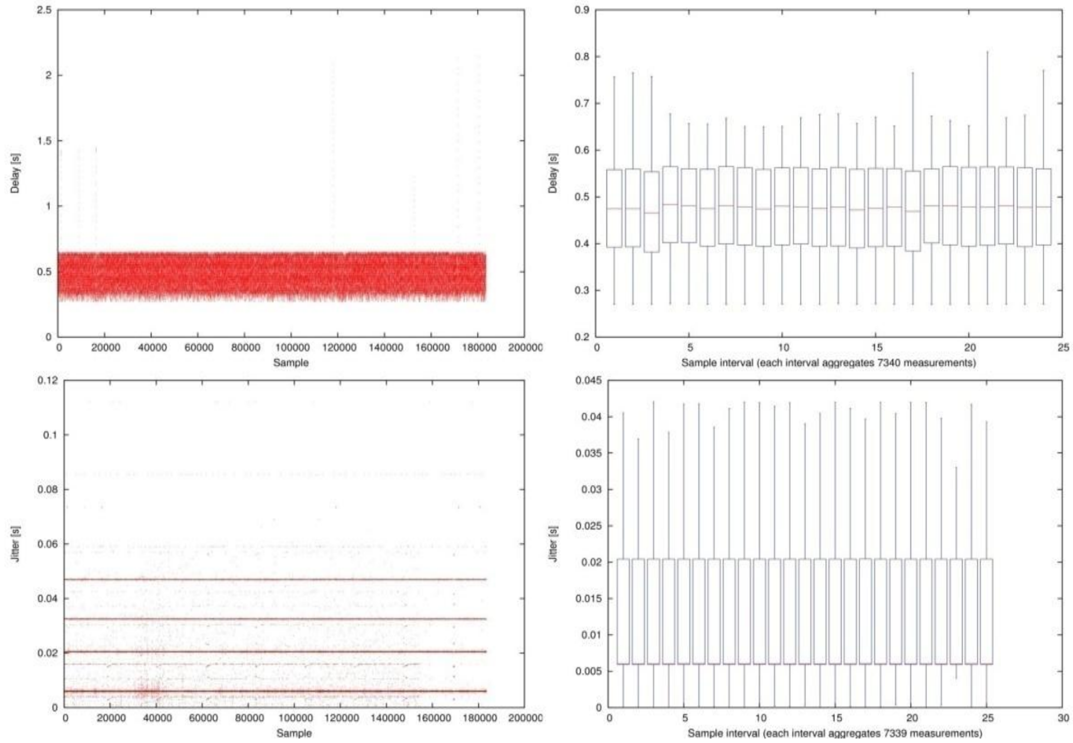


Figure 6.14: Delay and Jitter plots and Histogram of a 5 hours measurement via Satellite

These values show minimum jitter. This can be seen in the histogram and in the dot plot. The cause for this is unknown. Since the measurement with 1 Mbit/s load below does not show this effect, it was considered as irrelevant.

6.5.1.2 WiMAX Best Effort (BE)

In Best Effort (BE) services, network traffic is processed as quickly as possible but there is no guarantee for delivery and its timing. Graphs in the following figure 6.15 indicate the delay and jitter of a WiMAX link set to Best Effort (BE) characteristics. The duration of the test was 24 hours. The delay values are from 0.03s to 0.045s was recognised (figure.6.15).

There are several regions with increased outliers delimited. It is understood that this was due to hardware retransmissions caused by temporary disturbances in the transmission medium. The jitter was from 0.02 s to 0.027 s. Since the amount of these outliers was small, these cannot be seen in the box plots. Therefore, the jitter in this scenario was minimal.

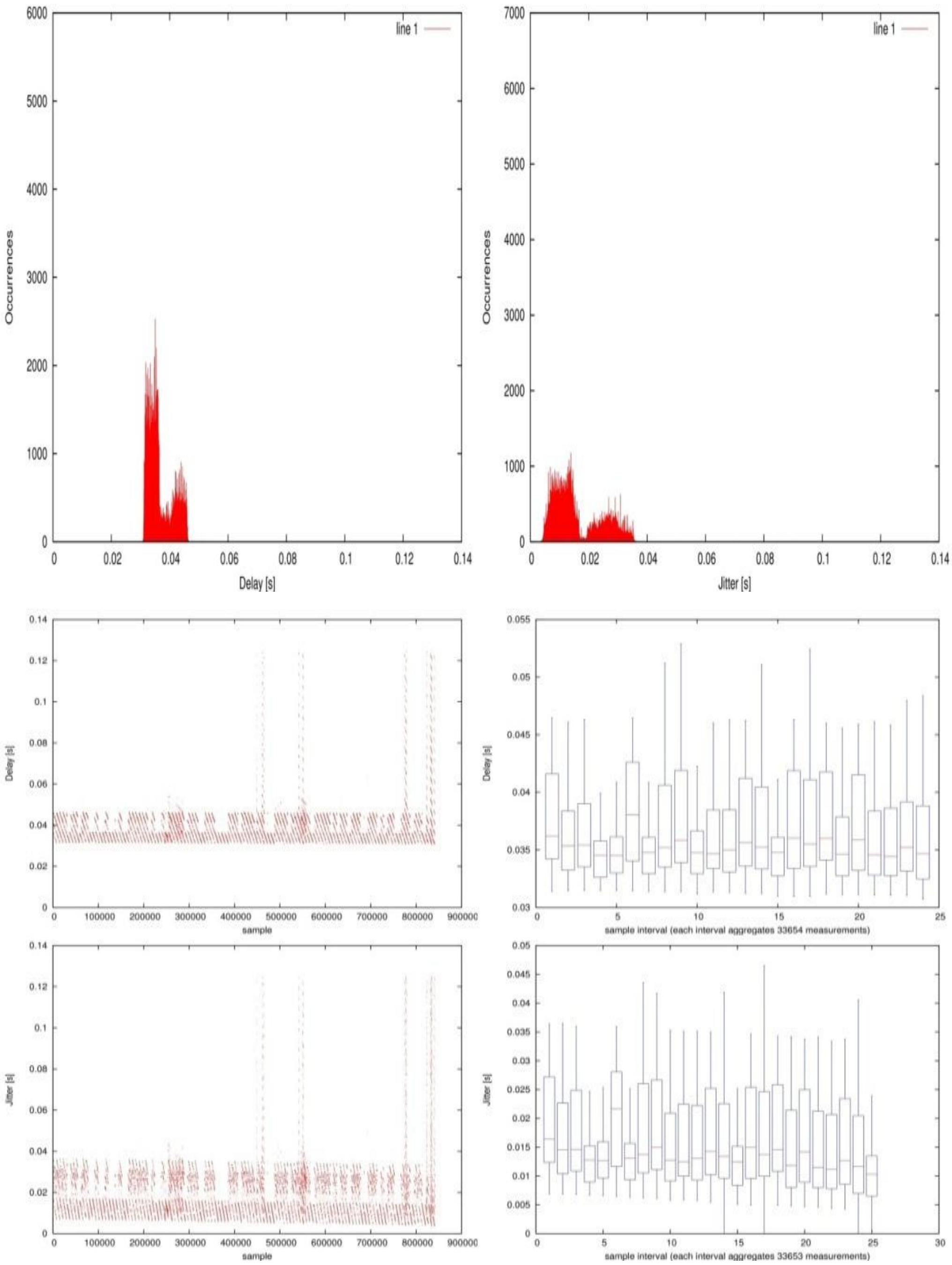


Figure 6.15: Delay and Jitter plots and histogram of a 24h measurement via WiMAX BE

6.5.1.3 WiMAX Real Time   Pooling Service (rtPS)

The following figure 6.16 shows delay and jitter plots of a WiMAX link with Real Time Pooling Service (rtPS) class. Since the variance of the delay was similar to the BE characteristic, jitter values were significantly better.

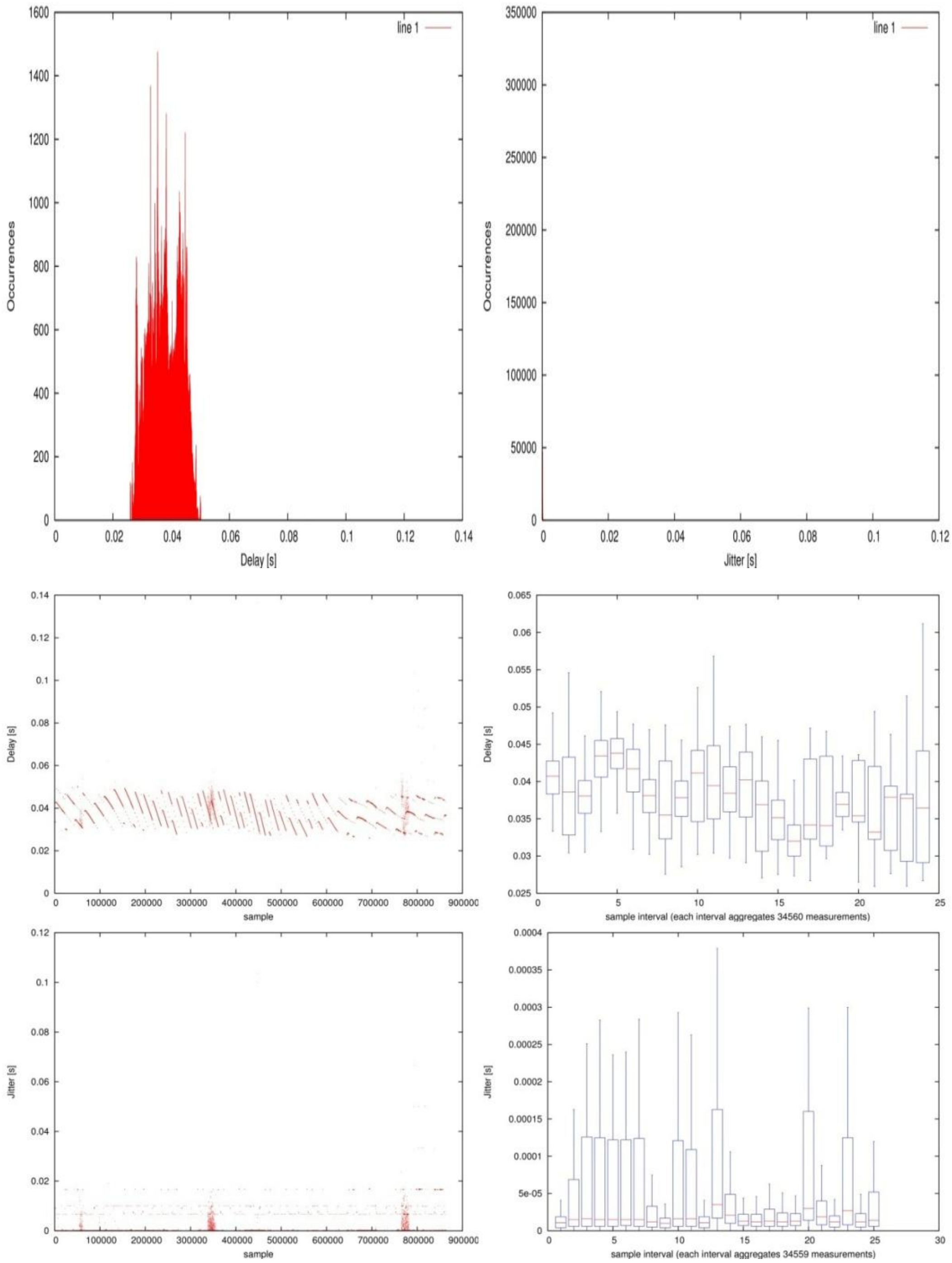


Figure 6.16: Delay and Jitter plots and Histograms of 24-hour measurements via WiMAX rtPS

6.5.1.4 Wireless LAN (Wi-Fi)

The duration of WLAN link measurement test was 24 hours. Figure 6.17 shows delay and jitter plots and histogram for this test. In the dot plots depicting the delay, most of the values range between 0.002s and 0.003s. The magnitude of the delays (ranging between 0.003s and 0.04s) are large relative to the delays between 0.002 and 0.003s. Therefore they are counted as outliers and are not shown in the box plots.

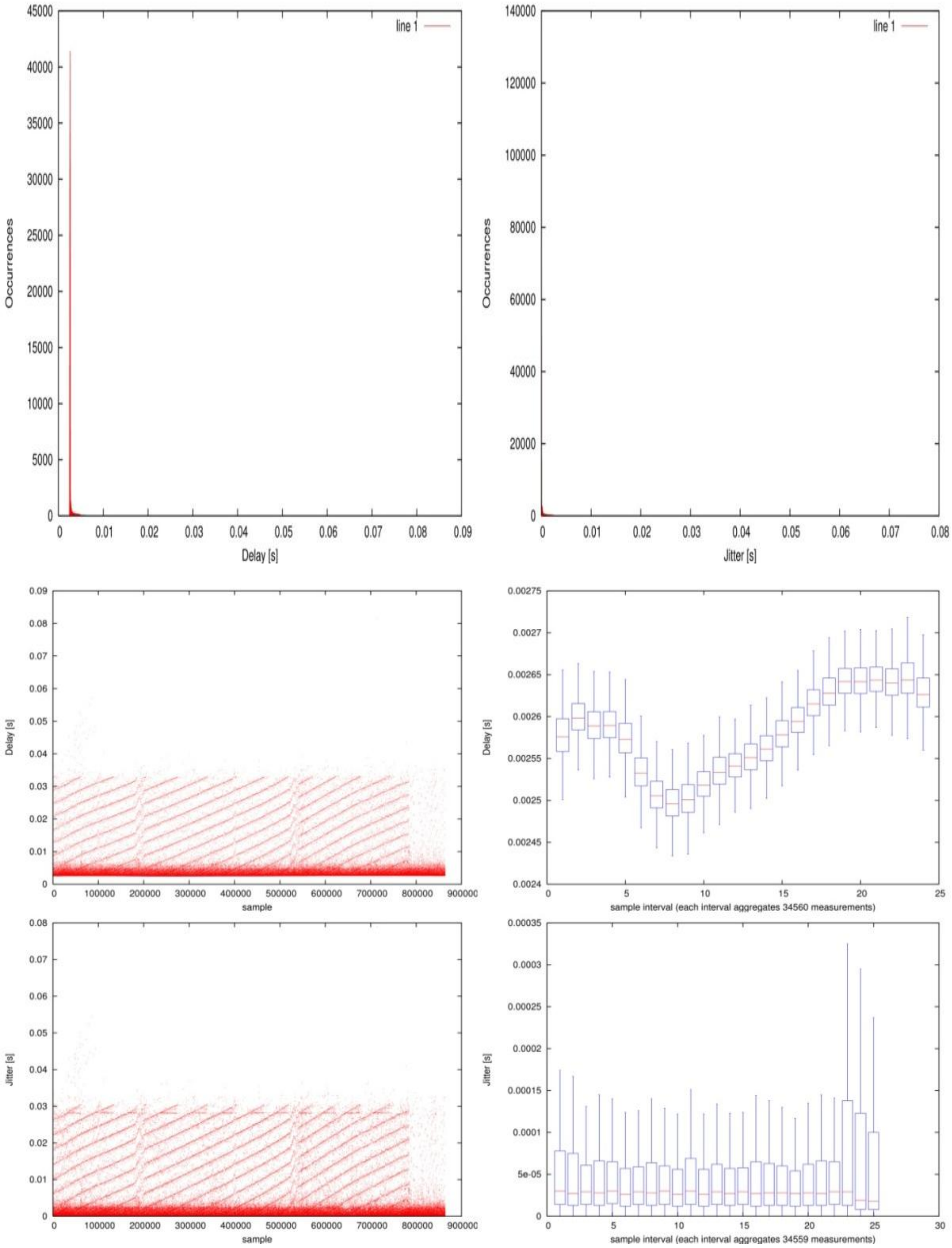


Figure 6.17: Delay and jitter plots and Histogram of a 24h measurement of Wireless LAN

These outliers are most likely caused by the retransmissions of the Wireless LAN hardware. The box plot of the delay samples show unevenness. This is caused by non-linear drift of the station clocks. Since this drift amounts to only 0.00015s, it can be considered as immaterial.

6.5.1.5 WiMAX rtPS vs. BE

The following figures 6.18 and 6.19 compare WiMAX BE and rtPS class of service. In this test, a WiMAX sector controller with an available Bandwidth of 10 Mbit/s serves two clients, one with rtPS class and one with BE class. Each client tries to download 6 Mbit/s TCP from the Sector controller.

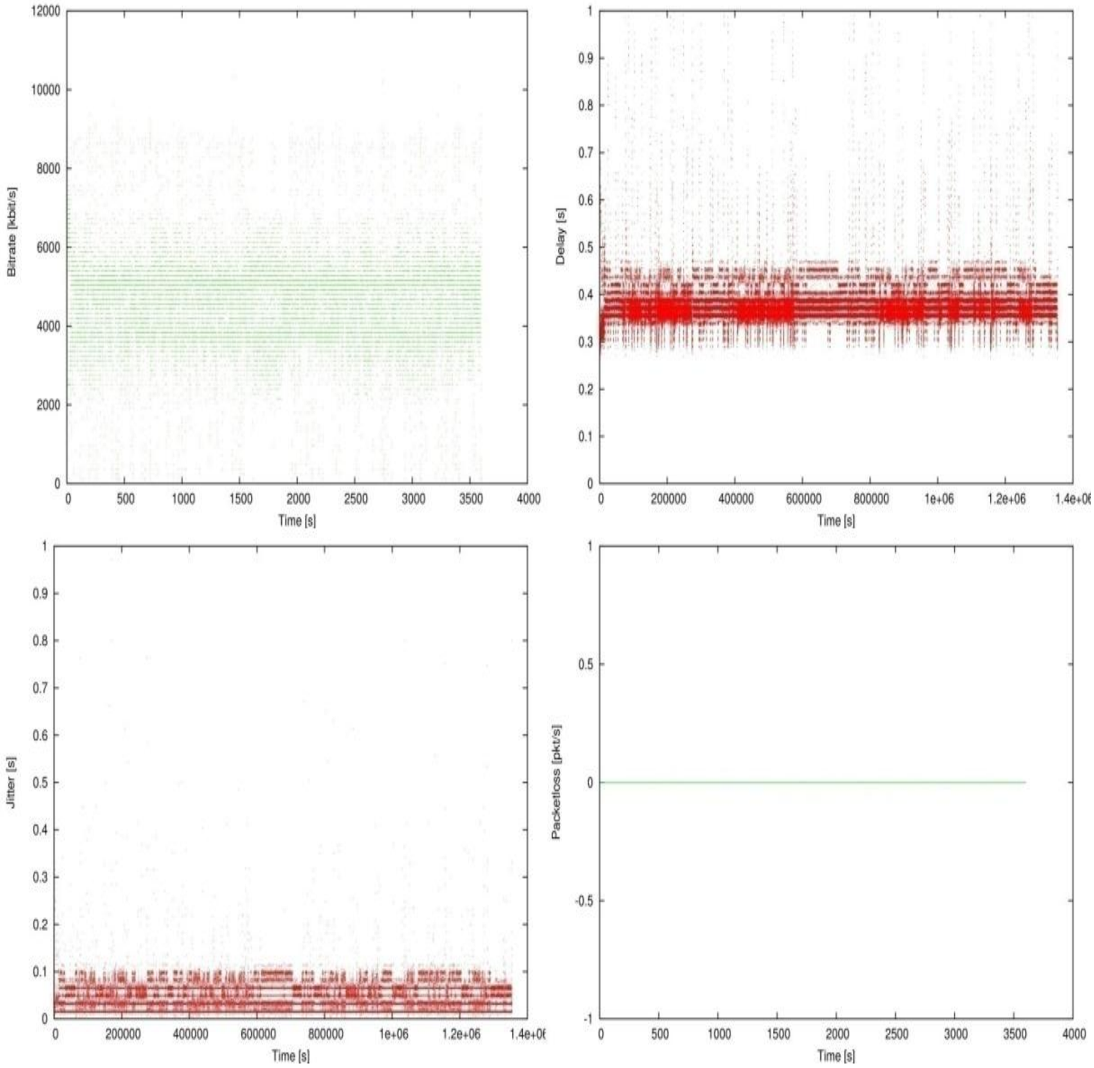


Figure 6.18: Jitter and Delay plots of an oversaturated WiMAX BE link

The delay is 0.3-0.4 s for BE and the maximum jitter is 0.1 s as shown in figure 6.19. For the rtPS, the delay is 0.2-0.3 s. Therefore, the rtPS has a significant better performance than the BE. As the packet loss plots indicate, both links did not experience any packet loss. This is due to the TCP that retransmits lost packets and rescales the TCP Window in order to avoid congestion. The bit rate plots show both rtPS and BE services reach more or less the same bit rate. It is observed that the rtPS link has a significant lower jitter and delay values as compared to BE.

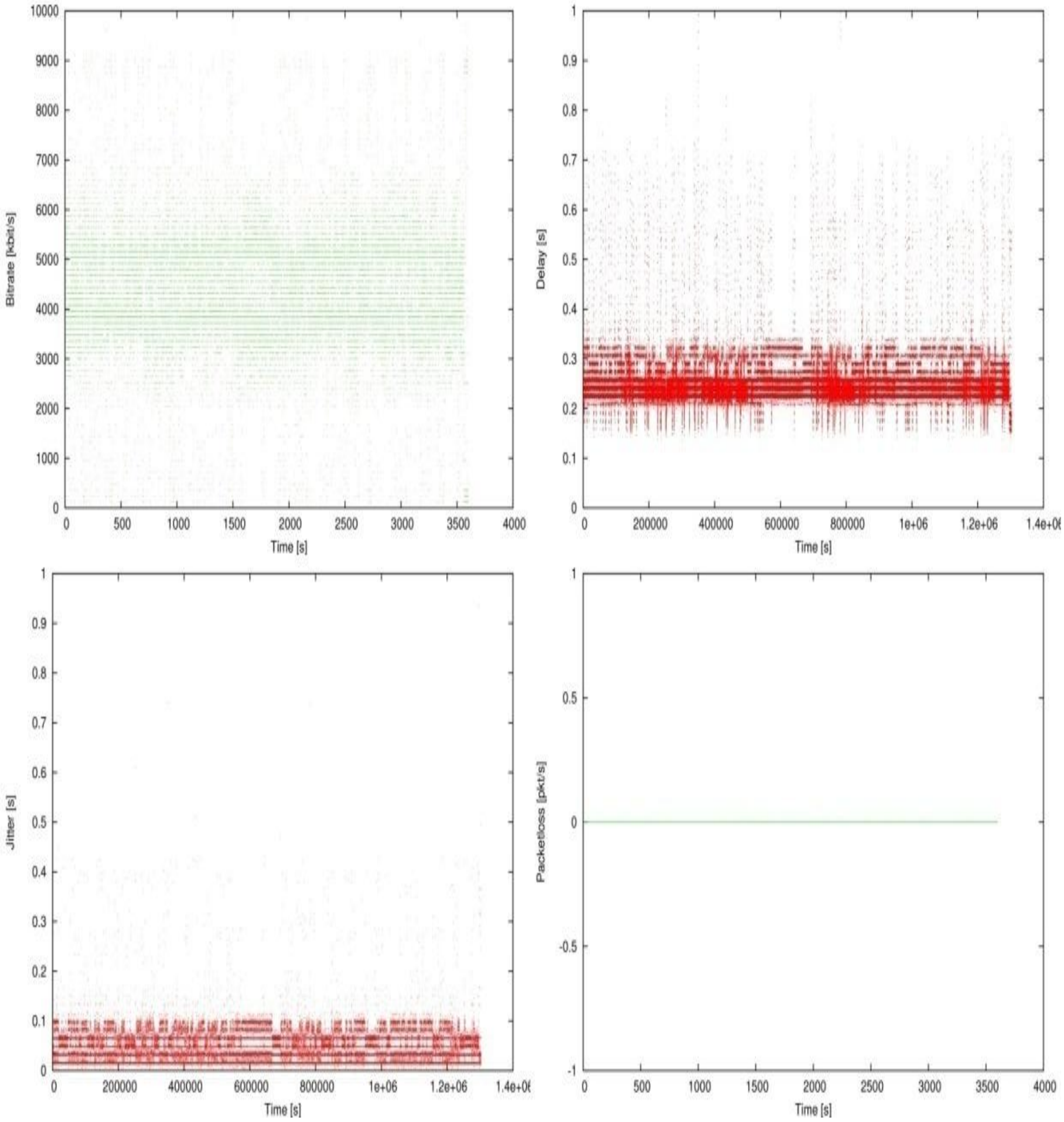


Figure 6.19: Jitter and Delay plots of an oversaturated WiMAX rtPS link

6.5.2 Short-term Measurements with load

The short term load test for DVB-RCS, WiIMAX, Wi-Fi links were performed on the same links as the long term series. These tests were limited to one and four hours respectively due to large amount of data. The following sub sections show the measurements of the short term series of tests for DVB-RCS, WiMAX and Wi-Fi Links. For these testing, a UDP data stream of 88 packets per second with 1500 byte per second was generated. This data rate was underlink saturation conditions. The Packet loss was very minor. Delay and jitter were expected to be similar to those of the long-term test series, however the higher packet rate level effects may be due to the power saving and timing algorithms.

6.5.2.1 Satellite (DVB-RCS)

The satellite link with load as shown in figures 6.20-6.21 behaves interestingly much the same as without load.

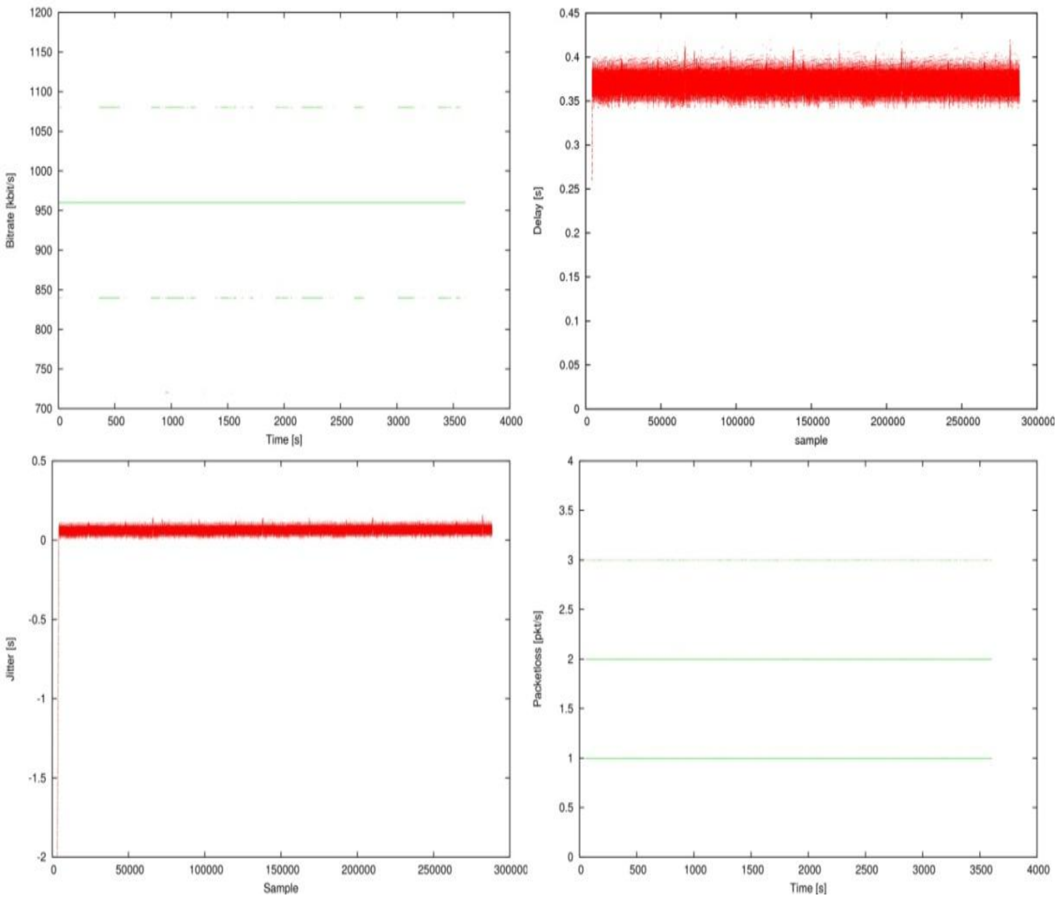


Figure 6.20: Delay and jitter plots of a satellite link with 1 Mbps load, 1hour duration

It shows that the variance of delay is smaller; however the jitter variance is higher. By comparing the histograms of the satellite measurements, it is noticed that spikes only appear in the unloaded scenario. There is also a drastic increase in packet loss due to 1Mbp traffic load.

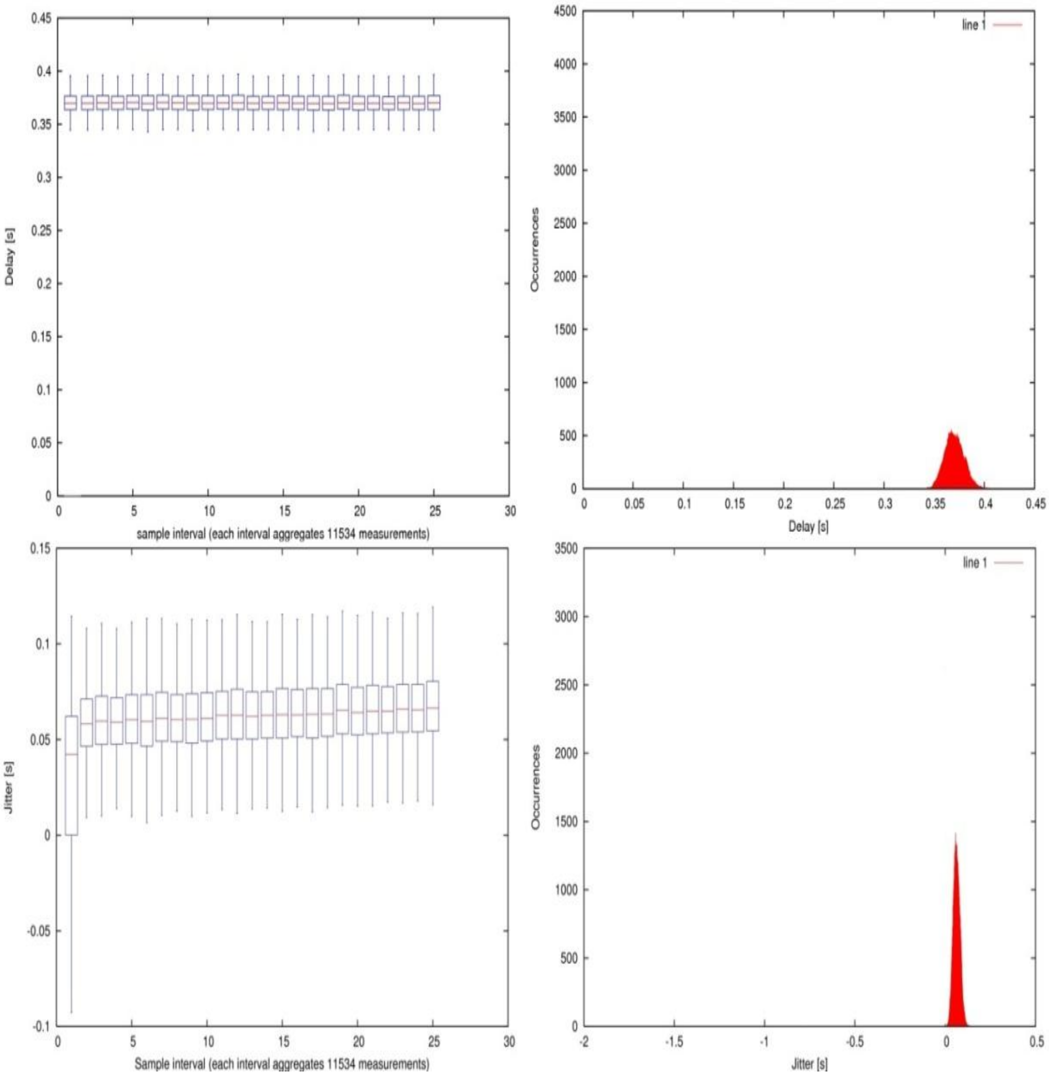


Figure 6.21: Histogram and Box plot of satellite link with 1 Mbps load, 1hour

6.5.2.2 WiMAX

The following figure 6.22 and figure 6.23 show the delay, jitter and packet loss for the cases of WiMAX BE and rtPS. The results show that both delay and jitter behave similarly. The delay and jitter values are comparable to the unloaded link.

Both links did not show any packet loss. Under these circumstances, it is irrelevant which of the two - BE or rtPS–characteristic/class was chosen.

6.5.2.2.1 WiMAX BE

The figure 6.22 below shows delay and jitter for the WiMAX BE characteristic with 1MB load. The delay is between 0.025 and 0.045 s and the maximum jitter is 0.015 s.

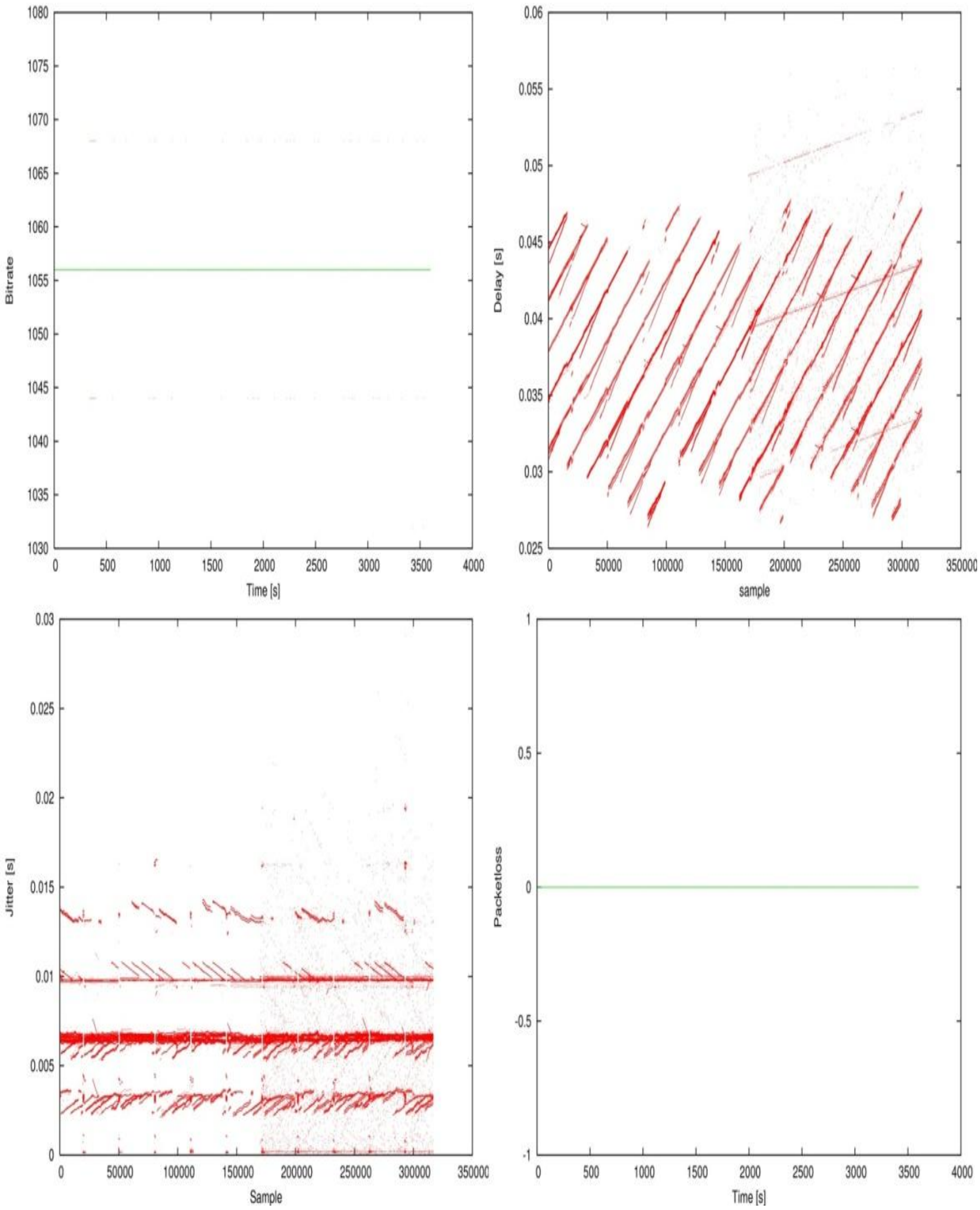


Figure 6.22: Delay and jitter plots of a WiMAX BE link with 1 Mbps load, 1hour

6.5.2.2.2 WiMAX Real Time Pooling Service (rtPS)

Figure 6.23 shows the delay and jitter plots of a WiMAX rtPS link with 1 Mbps load, 1hour. The delay is between 0.035 to 0.045s.The jitter is 0.015 s maximum.

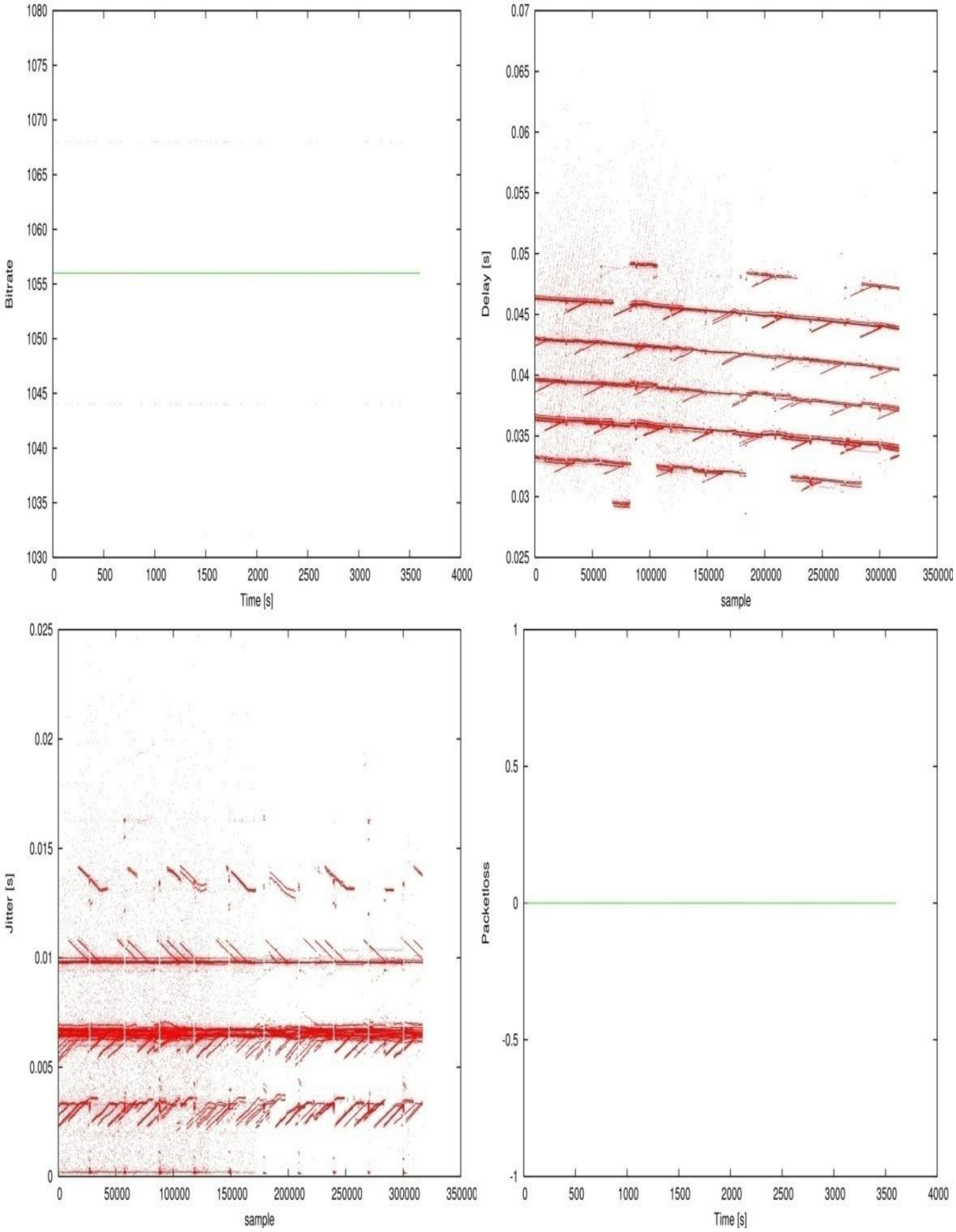


Figure 6.23: Delay and jitter plots of a WiMAX rtPS link with 1 Mbps load, 1hour

6.5.2.3 Wi-Fi (Wireless LAN) Link

The Wi-Fi (wireless LAN) link shows, as it has in the long-term measurement, the best delay and jitter values. The delay dot plot shows the outlier percentage is higher than in the long-term measurement (Figure 6.24). Again, this is due to the retransmissions done by the hardware. The packet loss is noticeable but the occurrence is for very short period and almost irrelevant and inconsequential.

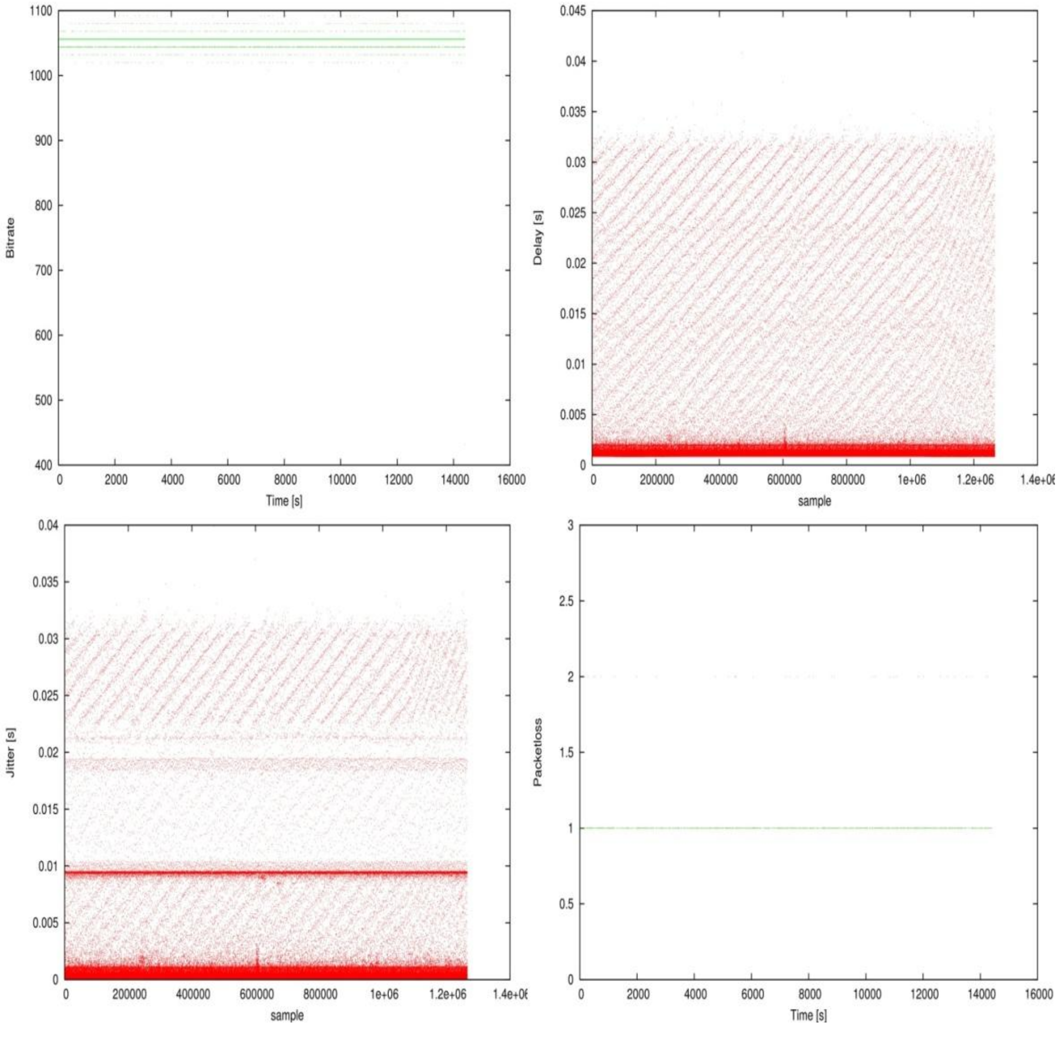


Figure 6.24: Delay and jitter plots of a wireless LAN link with 1 Mbit/s load, 1hour

6.5.3 DVB-RCS, WiMAX, Wi-Fi Various Traffic Analysis

The traffic was also captured using tcpdump tool and analysed by using the Wirshark protocol analysis tool. The following figure 6.25 shows an example of screen shorts from Wireshark tool which shows packets /tick. The traffic was filtered for TCP and UDP. While analysing the traffic there were few messages for “UDP checksum incorrect”. This was due to wireshark analysis which examines the outgoing packets before they exit, therefore these packets were considered as normal. Some TCP protocol shows the message “Packet size limited during capture message”. This message appears when TCP size is truncated to 96 Bytes. The flow of traffic was smooth and there was no packet loss which can be seen.

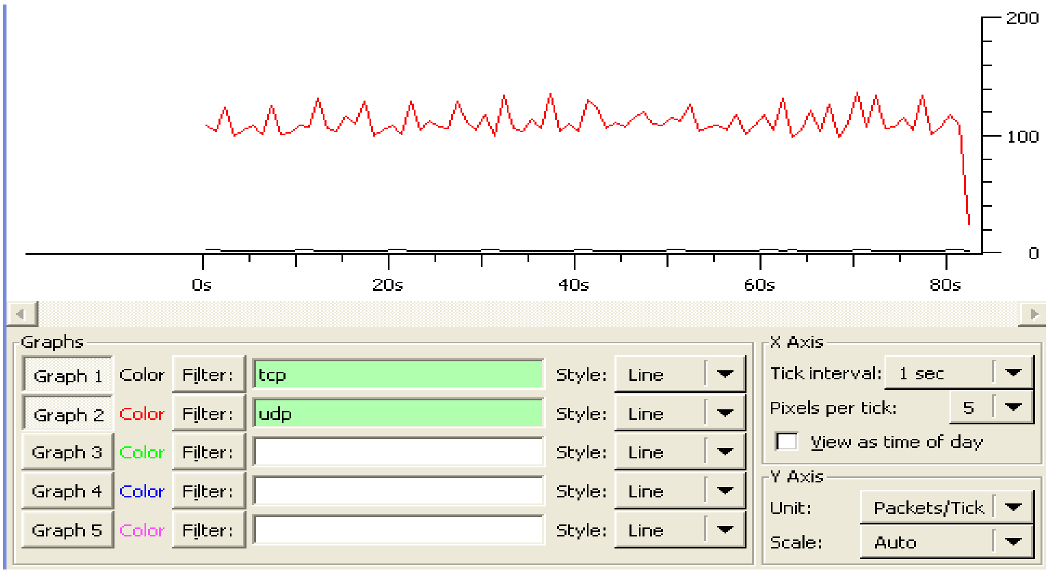
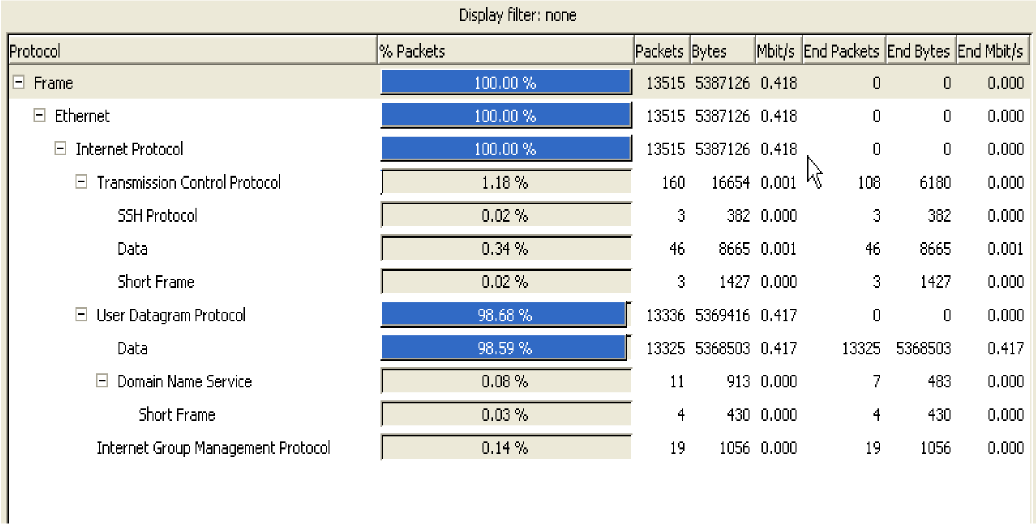


Figure 6.25: I/O graph of HAI Site

Figure 6.26 is an example of protocol hierarchy which shows the protocol hierarchy of packet captured during live DVB-RCS scenarios. This gives an overview of types of traffic i.e. TCP, UDP, HTTP, percentage of packets, number of packets for each protocol to see the type of traffic, bandwidth Mbits/s, end packets , end bytes and end Mbits/s. In all cases around 98 % are UDP protocols and the remaining 2% are TCP and Address Resolution protocol.



Display filter: none

Protocol	% Packets	Packets	Bytes	Mbit/s	End Packets	End Bytes	End Mbit/s
Frame	100.00 %	13515	5387126	0.418	0	0	0.000
Ethernet	100.00 %	13515	5387126	0.418	0	0	0.000
Internet Protocol	100.00 %	13515	5387126	0.418	0	0	0.000
Transmission Control Protocol	1.18 %	160	16654	0.001	108	6180	0.000
SSH Protocol	0.02 %	3	382	0.000	3	382	0.000
Data	0.34 %	46	8665	0.001	46	8665	0.001
Short Frame	0.02 %	3	1427	0.000	3	1427	0.000
User Datagram Protocol	98.68 %	13336	5369416	0.417	0	0	0.000
Data	98.59 %	13325	5368503	0.417	13325	5368503	0.417
Domain Name Service	0.08 %	11	913	0.000	7	483	0.000
Short Frame	0.03 %	4	430	0.000	4	430	0.000
Internet Group Management Protocol	0.14 %	19	1056	0.000	19	1056	0.000

Figure 6.26: Traffic flow Protocol percentage Chania Site

Table 6.1 is created from Wireshark traffic analysis. This shows a comparison among average packets/sec, average packet size in Bytes, Average Bytes/sec and average Mbits/sec for various sites. The average packets/sec range from 76.34 to 131.44. The average packet sizes range from 145.055 to 871.763 bytes. The Average data rate is 0.316-0.457 Mbits/s.

Table 6.1: Network Traffic size comparison

SIT	***Average Packets/sec***	***Average Packet Size Bytes***	***Average Bytes/sec***	***Average Mbits/sec***
HAI	111.45	366.851	40887.399	0.327
AIT	112.217	465.153	52198.069	0.418
Potamia	76.347	517.313	39495.202	0.316
Velo	122.276	871.763	56681.866	0.453
Chania	131.07	398.603	52245.008	0.418

The following figures 6.27 and 6.28 were calculated from packet lengths screen shot taken via the Wireshark tool, which were captured during analysis of final interoperability test between various sites. The majority of the traffic packet length is between 160-319 for DVB-RCS scenarios. There was no data size which was less than 39 packets and more than 1280 packets.

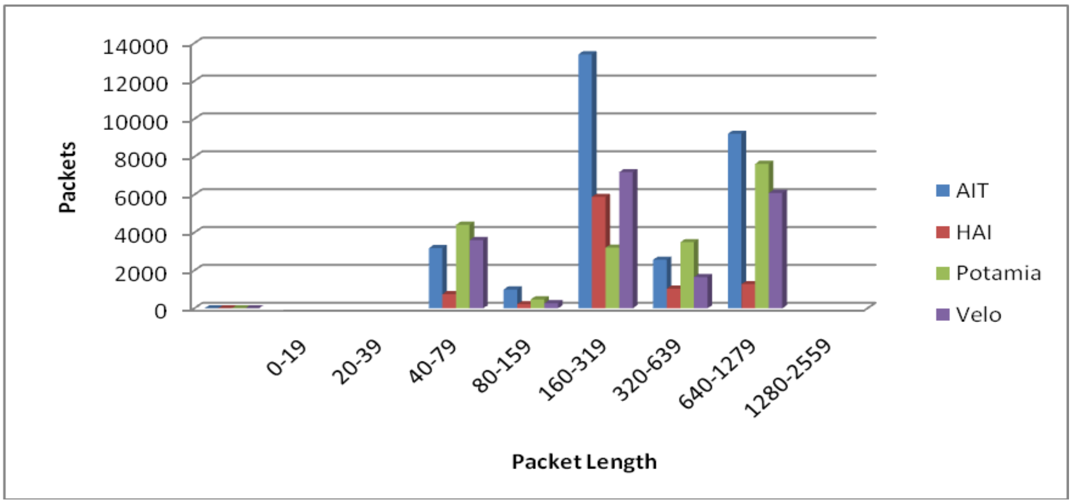


Figure 6.27: Packet Length for DVB-RCS

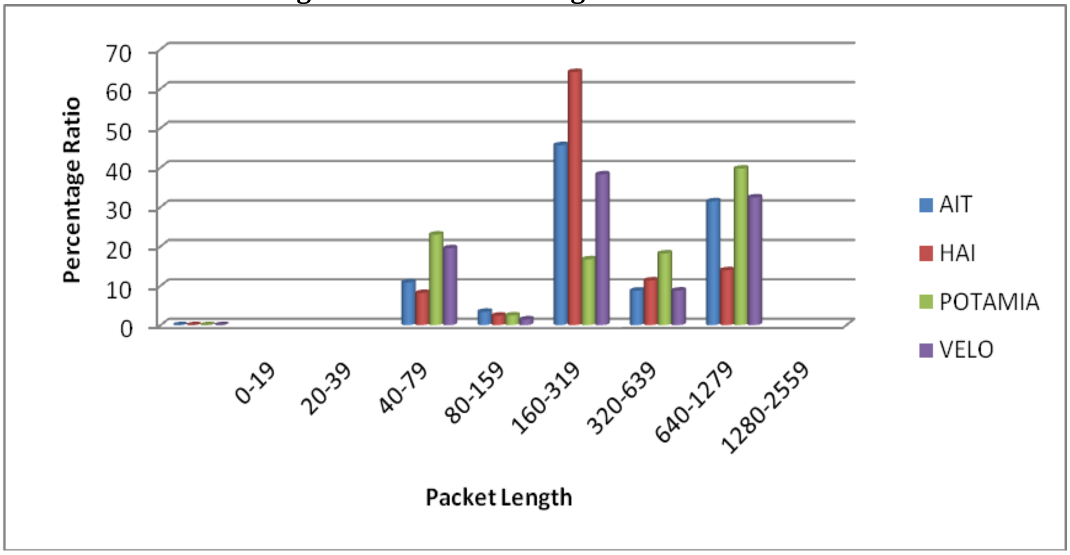


Figure 6.28: DVB-RCS Packet Length Percentage Ratio

6.5.4 RTP Analysis

The purpose of this RTP traffic analysis is to verify that the bandwidth, jitter and loss figures of a real-life session between multiple participating sites are within acceptable limits. The DVB-RCS return channel is most critical link in the system where all participating sites send their traffic back to the ISABEL flow server which is located at HAI's hub station. The 'u2m' analyzer tool has been setup to run on the flow server to capture the characteristics of the individual incoming streams.

The following figure 6.29 visualizes the RTP stream statistics of a teleconference between three sites as per DVB-RCS scenario shown in figure 6.8. Further passively listening participating sites do not add to the traffic load, due to the multicast distribution of the content. The top row represents the bandwidth, jitter and loss figures of the audio streams, while the bottom row represents the video stream figures. The jitter statistics of the video streams have a limited significance since the video frames are not sent at fixed intervals. The inter-arrival jitter is smaller than 100 ms assuming a frame rate of 10 frames per second and at least one RTP packet per frame. Higher numbers indicate that jitter is present during transmission. Participants SITE1 and SITE2 were permanently active while participant SITE3 had its audio channel muted for most of the time and only communicated occasionally.

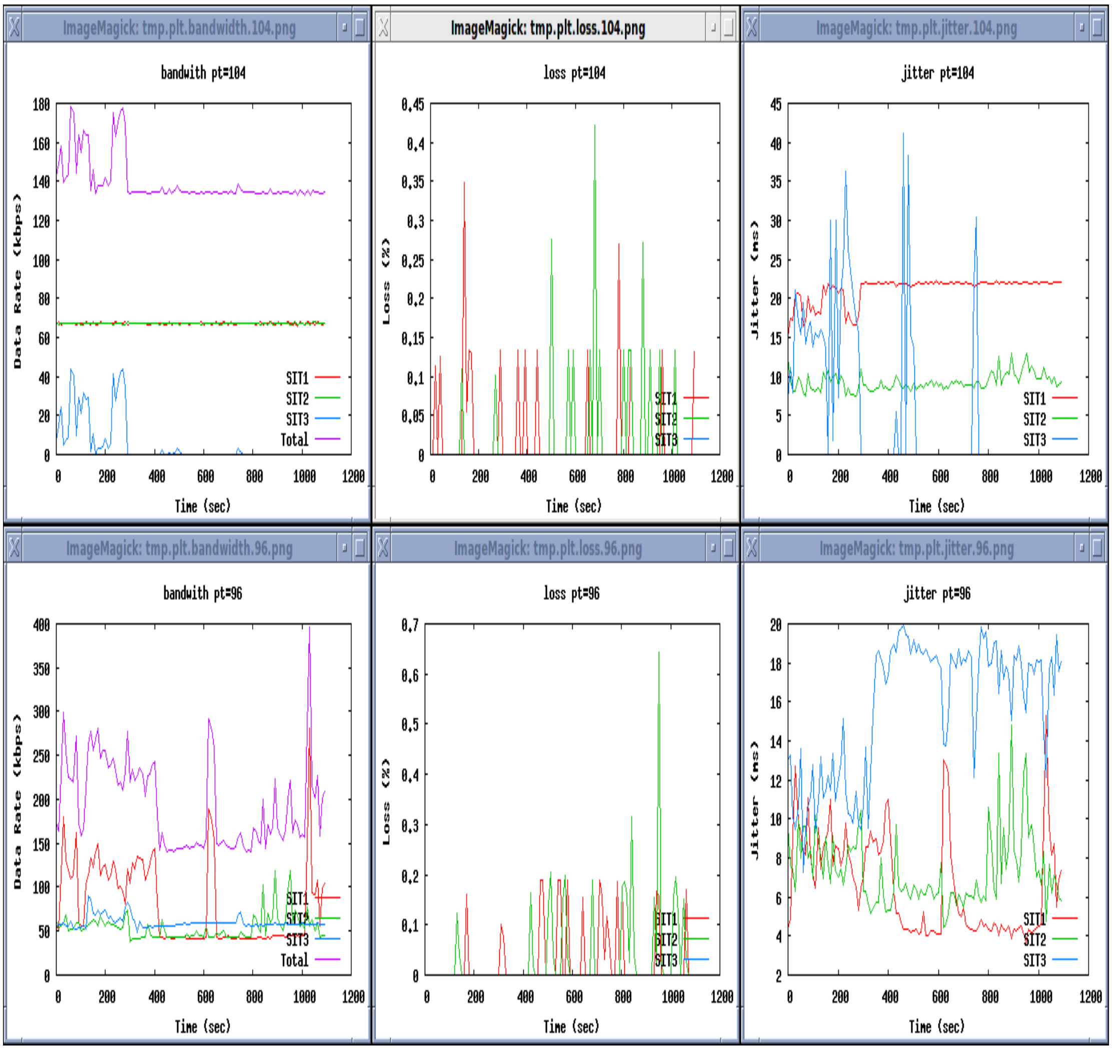


Figure 6.29: DVB-RCS Scenario RTP stream statistics

Adding up the bandwidth of the audio and video streams, it is shown that the total amount does not exceed the 1Mbps limit. The Packet loss rate (averaged) is less than 0.2%, with a few occasional spikes. Those spikes could be caused by transmission errors in any of the involved devices. The jitter of the audio signal is below 20ms, while the jitter of the video signal is below 50ms, which shows that there is no considerable jitter on the transmission links. Hence, the satellite link under real- load performs well.

6.5.5 Unicast and Multicast Measurements Analysis

The distribution of IP Multicast traffic across the DVB-RCS Scenario is thoroughly tested with the DVB-RCS test scenario shown above in figure 6.9. The five nodes – namely NCSR, Flow Server, SIT1, SIT2 and SIT3 are tested for unicast and multicast testing. The traffic was sent from NCSR Greece, to SIT1, and flow server. The traffic was also sent from Flow server to NCSR, SIT1, and SIT2 and SIT3 and from SIT1 to Flow server, NCSR, SIT2 and SIT3. Overall five different types of tests were performed (from Test 1 to Test 5) using various scenarios. The traffic was sent from all terminals one by one (NCSR, SIT1, SIT2, SIT3, from flow server to DVB-RCS and from flow server to NCSR).

The size of audio and video for each test is given in table below for Test 1 to Test 5.

Table 6.2: Audio and Video data rate

Test Number	Teacher Video kbps	Teacher Audio kbps	SIT1 Video kbps	SIT1 Audio kbps	SIT2 Video kbps	SIT2 Audio kbps	SIT3 Video kbps	SIT3 Audio kbps
test 1	582	78	16	40	20	0	12	0
test 2	337	78	165	78	23	0	8	0
test 3	183	78	123	75	121	0	120	0
test 4	364	78	144	76	0	0	0	0
test 5	223	78	22	78	23	0	8	0

Where the value is 0 for audio and video, the audio channel is muted and video are not participating. The following figure 6.30 gives examples showing the bandwidth of traffic over time. The traffic from NCSR and Flow server is utilizing the same bandwidth (between 500 to 1500 kbps). However, traffic sent from SIT1 is utilising up to 130kbps. The results of IP Unicast and Multicast traffic are shown below in Table 6.3. The result shows that the traffic can be sent and received in both directions i.e. from the HUB to the SIT and return. No packet failure was encountered during those measurements. This confirms that the results of UDP measurements with regards to datagram traffic forwarding across the DVB-RCS platform. During testing, four videos were displayed on screen.

The following figure 6.30 shows the bandwidth utilised during test 1 DVB-RCS Scenario. The teacher at NCSR was using bandwidth of 582 kbps for video and 78 kbps for audio.

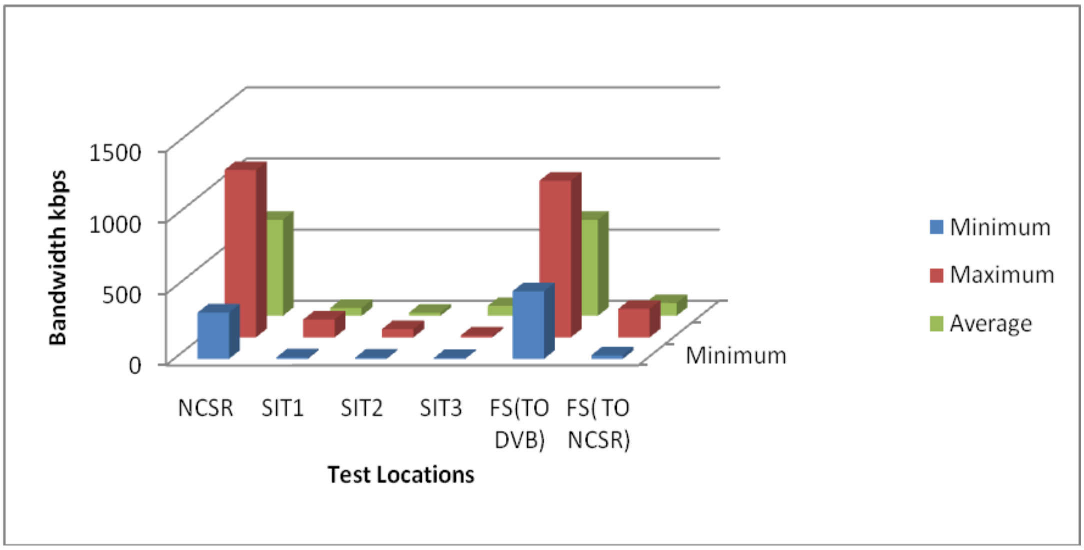


Figure 6.30: Bandwidth for Test 1

The following readings in Table 6.3 is an example of traffic from NCSR, SIT1, SIT2, SIT3 to DVB-RCS flow server and from DVB-RCS flow server to NCSR, SIT1, SIT2, SIT3 as per DVB-RCS scenario figure 6.9. The unicast and multicast traffic are shown in Table 6.3 with total no of packets, data size in bytes, average packet size (APS), Packets per second (PPS) and bandwidth in kilobits per second (kbps). This shows that there were no losses or duplicate packets that could have been seen. The node MCNAT does not exist, but it represents the multicast packets received from the SITs with a translated source address.

Table 6.3: Traffic Matrix from NCSR, SIT1, SIT2, and SIT3 to the flow server.

Source	***Destination***	***Packets***	***Bytes***	***Aps***	***PPS***	***Kbps***
Received (Unicast)						
NCSR	Flow Server	41928	27599489	658.26	126.29	665.05
SIT1	Flow Server	66	6501	98.5	0.2	0.16
SIT2	Flow Server	66	6534	99	0.2	0.16
SIT3	Flow Server	66	6534	99	0.2	0.16
		42126	27619058	655.63	126.89	665.52
Received (Multicast)						
MCNAT	MC2	5316	1989384	374.23	16.01	47.94
MCNAT	MC5	496	30482	61.46	1.49	0.73
MCNAT	MC1	5755	1660614	288.55	17.33	40.01
MCNAT	MC4	201	12060	60	0.61	0.29
MCNAT	MC6	201	12060	60	0.61	0.29
MCNAT	MC3	201	12060	60	0.61	0.29
		12170	3716660	305.4	36.66	89.56
Total received:		54296	31335718	577.13	163.54	755.08
Sent(Unicast)						
Flow Server	NCSR	11730	3788276	322.96	35.33	91.28
Flow Server	SIT2	66	5742	87	0.2	0.14
Flow Server	SIT1	66	5709	86.5	0.2	0.14
Flow Server	SIT3	66	5742	87	0.2	0.14
		11928	3805469	319.04	35.93	91.7
Sent (Multicast)						
Flow Server	MC2	29536	24168127	818.26	88.96	582.36
Flow Server	MC1	11071	3254874	294	33.35	78.43
Flow Server	MC5	692	42904	62	2.08	1.03
		41299	27465905	665.05	124.39	661.83
Total sent:		53227	31271374	587.51	160.32	753.53
Aggregated traffic (received + sent)		107523	62607092	582.27	323.86	1508.6

Packet length histograms were produced to analyse the traffic from NCSR Greece to SIT1, and from the flow server to the DVB-RCS Hub. These histograms show that about 75 % of the total traffic (24000 packets) is in the range of 641-960 bytes and 50 % traffic (12500 packets)) is in range of 161-320 bytes. There is no traffic which is less than 40 bytes. For SIT1, about 75 % of the traffic (6000 packets) is in the range of 161-320 bytes, while 10 % traffic (about 800 packets) are in the range of 641-960 bytes. Furthermore, there are no packets that have less than 40 packets in these scenarios.

The following figure 6.31 shows an example of the packet length histograms for the test 1. Most packets have packet length in the ranges 161-320 and 641-960.

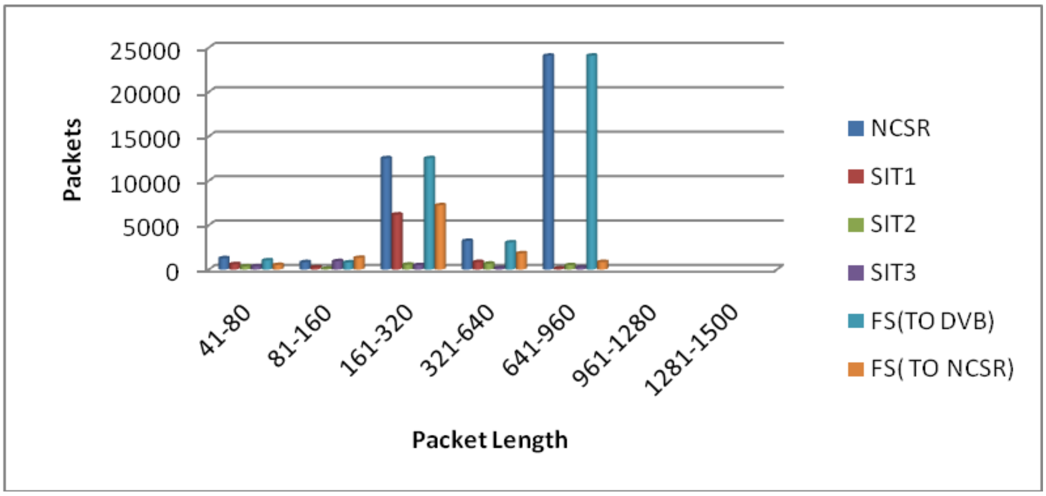


Figure 6.31: Histogram of Packet length (in bytes) for test 1 scenario

The following figure 6.32 is an example of minimum, 25%, 50%, 75% and maximum packet length quartiles.

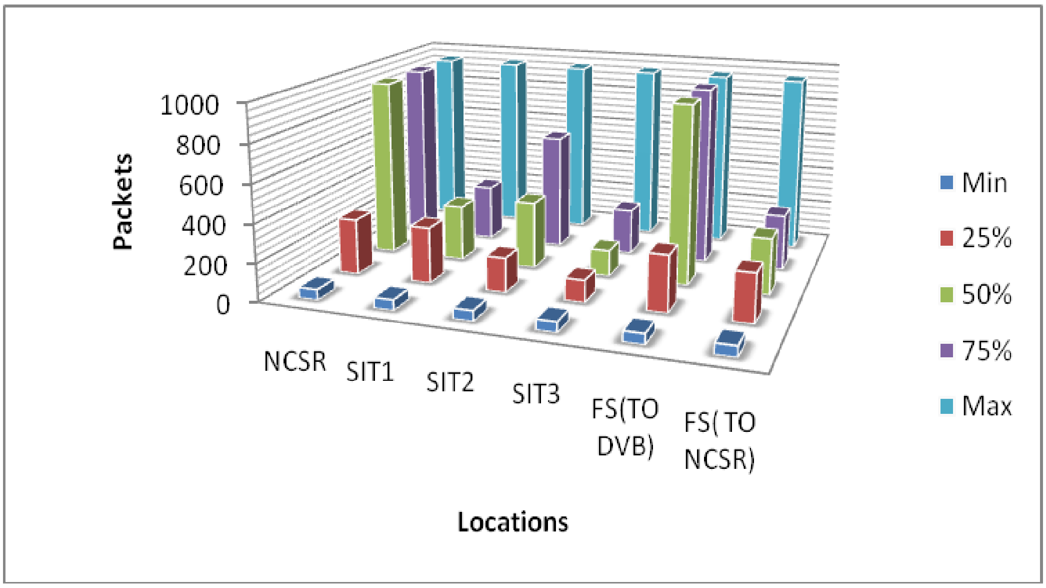


Figure 6.32: Packet length quartiles for test 1

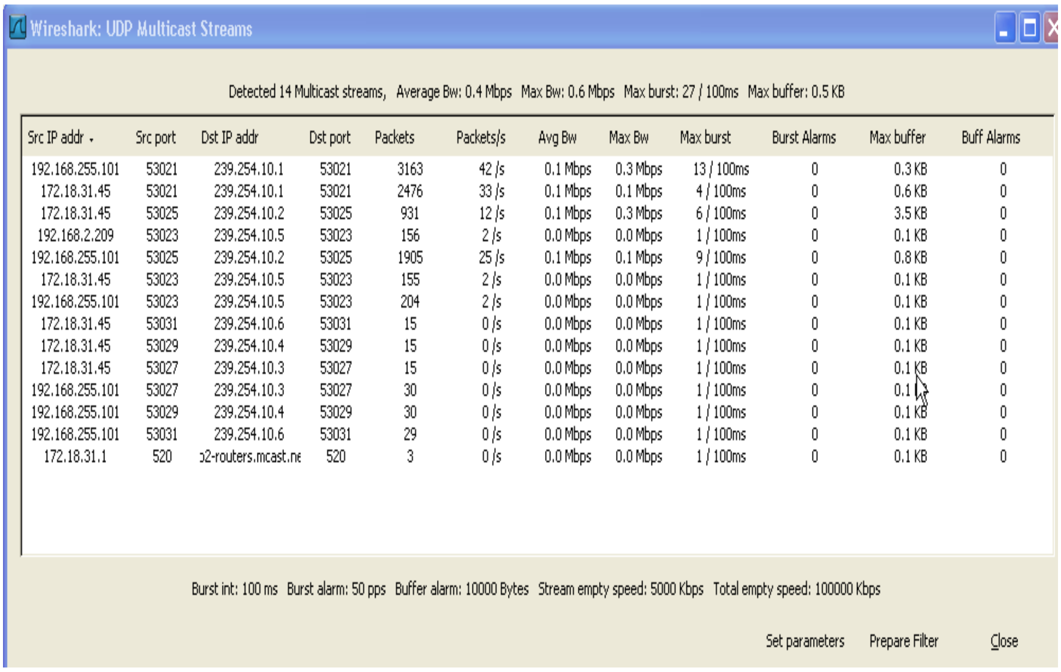
Examples of multicast audio and video traffic are shown in Table 6.4 below, including total numbers of packets, packet/sec, average size of data, bandwidth utilised and data rate. The difference in Table 6.4 for bytes of Flow Server is related to Ethernet padding on short frames.

Table 6.4: Results for IP Multicast traffic sent from SIT1 to Flow Server (FS), SIT2, and SIT3

Video	***Start sequence :45377***	***End Sequence: 47103***			***SSRC: 8043***
	Packets	Packets/sec	Average Size	Bytes	Mbps
Sent from SIT1	1727	5.23	372.15	642702	0.016
Received by FS	1727	5.23	372.15	642707	0.016
Received by SIT2	1727	5.23	372.15	642707	0.016
Received by SIT3	1727	5.23	372.15	642707	0.016
Audio	Start sequence:	End sequence:33674			SSRC: 8043
	Packets	Packets/sec	Average Size	Bytes	Mbps
Sent from SIT1	5587	16.94	294	1642578	0.040
Received by FS	5587	16.91	294	1642578	0.040
Received by SIT2	5587	16.91	294	1642578	0.040
Received by SIT3	5587	16.91	294	1642578	0.040

It is found that packet loss was the same for all scenarios. However, DVB-RCS does have a considerable round trip time delay. This characteristic is unavoidable on a satellite link. This is particularly a problem for services such as multimedia and video conferencing. DVB-RCS is a highly asymmetric service, since large Mbps-sized bandwidth is required on downlink and more throughput on the individual uplink. However, on the uplink sector, only narrowband communications can be established. The bandwidth can’t be shared between users and due to this; the cost also cannot be divided. A small jitter was recorded, but that can be reduced by introducing clever traffic prioritization and carrier management on the return channel, and throughput can be optimized at the same time.

The following figure 6.33 shows multicast stream analysis for an example of traffic from the DVB-RCS Flow Server to various sites and from the Hub to the ship SCPC. The multicast stream was detected with Average Bandwidth of 0.1 Mbps to Maximum Bandwidth of 0.6 Mbps. The maximum buffer was 0.5 kB. The ship SCPC multicast streaming results were not encouraging due to the limitation on available bandwidth.



Wireshark: UDP Multicast Streams

Detected 14 Multicast streams, Average Bw: 0.4 Mbps Max Bw: 0.6 Mbps Max burst: 27 / 100ms Max buffer: 0.5 KB

Src IP addr	Src port	Dst IP addr	Dst port	Packets	Packets/s	Avg Bw	Max Bw	Max burst	Burst Alarms	Max buffer	Buff Alarms
192.168.255.101	53021	239.254.10.1	53021	3163	42 /s	0.1 Mbps	0.3 Mbps	13 / 100ms	0	0.3 KB	0
172.18.31.45	53021	239.254.10.1	53021	2476	33 /s	0.1 Mbps	0.1 Mbps	4 / 100ms	0	0.6 KB	0
172.18.31.45	53025	239.254.10.2	53025	931	12 /s	0.1 Mbps	0.3 Mbps	6 / 100ms	0	3.5 KB	0
192.168.2.209	53023	239.254.10.5	53023	156	2 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0
192.168.255.101	53025	239.254.10.2	53025	1905	25 /s	0.1 Mbps	0.1 Mbps	9 / 100ms	0	0.8 KB	0
172.18.31.45	53023	239.254.10.5	53023	155	2 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0
192.168.255.101	53023	239.254.10.5	53023	204	2 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0
172.18.31.45	53031	239.254.10.6	53031	15	0 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0
172.18.31.45	53029	239.254.10.4	53029	15	0 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0
172.18.31.45	53027	239.254.10.3	53027	15	0 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0
192.168.255.101	53027	239.254.10.3	53027	30	0 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1	0
192.168.255.101	53029	239.254.10.4	53029	30	0 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0
192.168.255.101	53031	239.254.10.6	53031	29	0 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0
172.18.31.1	520	ɔ2-routers.mcast.ne	520	3	0 /s	0.0 Mbps	0.0 Mbps	1 / 100ms	0	0.1 KB	0

Burst int: 100 ms Burst alarm: 50 pps Buffer alarm: 10000 Bytes Stream empty speed: 5000 Kbps Total empty speed: 100000 Kbps

Set parameters Prepare Filter Close

Figure 6.33: Multicast stream Analysis from HAI Sites

6.5.6 TCP measurements

TCP tests were performed for the DVB-RCS scenario. TCP performs poorly over links with long RTTs (long fat pipe problem). The performance limitations are caused by the TCP flow control mechanisms, which are slow start, window size limitations and congestion control. The main limiting factor was the standard TCP window size limit of 64kBytes. Assuming an RTT of about 600ms, the maximum sustained throughput was capped at about 110Kbyte/s. The current maximum window size to be utilized was controlled by the TCP congestion avoidance algorithm. They, too, are often RTT driven. The classic ‘Reno’ is a good example (Abbasi *et al.* 2008).

Modern TCP stacks usually support Window Scaling, which allows the TCP window to grow exponentially to a maximum of 1 GByte. A number of less RTT-dependent (i.e.Vegas) or even satellite-optimized (i.e.Hybla) congestion avoidance algorithms have been proposed, but are not yet widely used.

Also since slow start is dependent on the RTT it takes about 2-3 seconds for a TCP connection to reach its maximum throughput. Therefore, short transfers, for example HTTP transactions, suffer from a significant drop in throughput. Similar limitations apply to recovery after packet loss or congestion.

Figure 6.34 (Abbasi *et al.* 2008) illustrates the effect of 'slow start' at the beginning of the transfer and after a packet loss or congestion over a plain satellite link. The throughput limit of about 200 kByte/s is determined by the bandwidth of DVB-S channel.

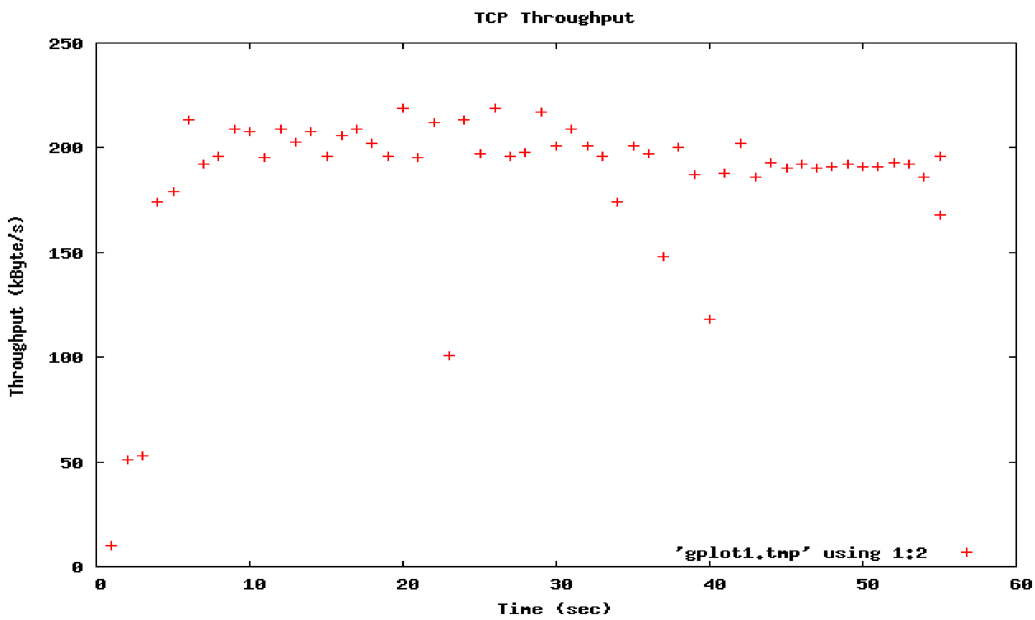


Figure 6.34: DVB-S (Shiron InterSky): Plain TCP (BIC – Binary Increase)

To overcome those TCP limitations, most satellite hub stations feature so-called PEPs (Performance Enhancing Proxy), which moderate the TCP traffic across the satellite link in order to minimize the performance degradation.

The DVB-RCS hub station at HAI has such a PEP built in, while the DVB-S Shiron InterSky hub station at FOKUS operates without PEP.

Figures 6.35 (Abbasi *et al.* 2008) shows the typical TCP throughput for HAI DVB-RCS hub with the PEP enabled. The bandwidth limit of about 120 kByte/s was configured at the hub station. The slow start effect was almost eliminated and the throughput was fairly constant.

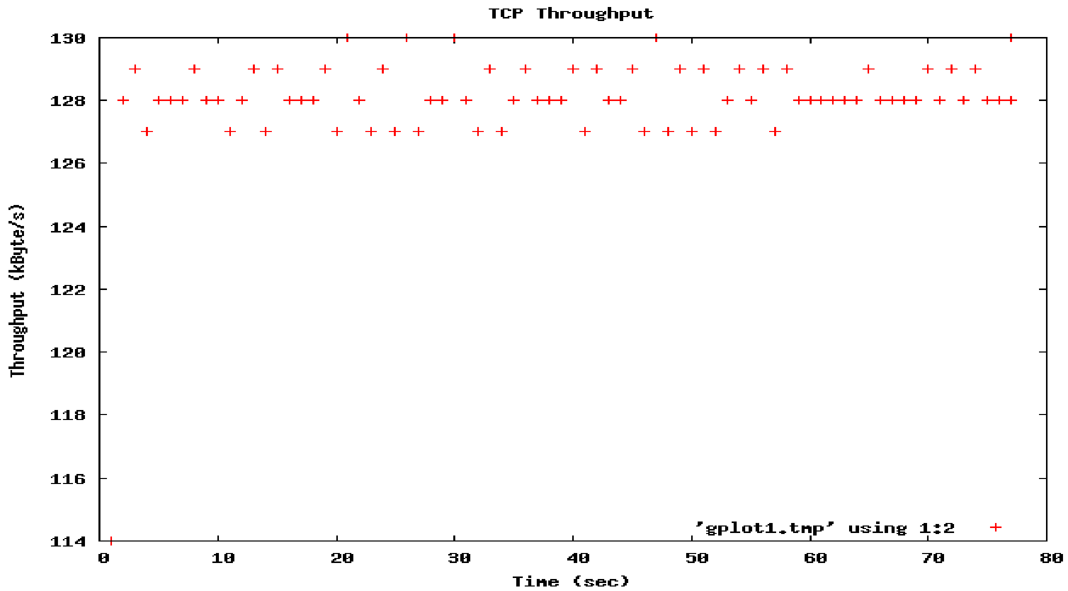


Figure 6.35: TCP Throughput HAI hub station (PEP enabled)

6.5.7 Sustained HTTP/TCP throughput

Table 6.5 below shows the httperf results of the measurements across various link types. Each test was repeated five times and the average throughput was calculated. RTT values, as reported by 'ping', are also shown (Abbasi *et al.* 2008).

Table 6.5: Sustained HTTP/TCP Throughput

Link Type	***RTT/ms***	***M1/ kbps***	***M2/ kbps***	***M3/ kbps***	***M4/ kbps***	***M5/ kbps***	***Avg./ kbps***
Shiron 2.5Mbps DVB-S	645	1363	1520	1485	1354	1342	1412
WiMAX	43	5649	5805	5654	5617	5537	5652
Wi-Fi	1	23051	23256	22969	23994	23790	23412
SIT	647	1060	1060	1060	1060	1059	1060
SIT + WiMAX	688	1060	1060	1060	1060	1060	1060
SIT + WiMAX + Wi-Fi	691	1054	1060	1060	1060	1060	1058

The above results show that the TCP throughput over the satellite links was significantly lower compared to wireless technologies. Hence, as expected, the satellite link was the bottleneck delivering a sustained throughput of more than 1Mbps.The results show that WiMAX and/or Wi-Fi links behind the SIT have no measurable impact on the throughput.

6.5.8 HTTP website access/load times

For this test, a directory containing 48 JPEG images with an average size of 6500 Bytes was downloaded and displayed with FireFox's FasterFox plugin. This test simulates the access of the static part of the CLIX web interface and aims to provide a feeling for the 'experienced smoothness'.

Table 6.6 below (Abbasi *et al.* 2008) shows the results of the measurements across various link types. Each test was repeated five times. The time elapsed as reported by FasterFox was noted and the average time was calculated. RTT values, as reported by 'ping', are also shown.

Table 6.6: HTTP website access/load times

Link Type	*RTT/ms*	*M1/s*	*M2/s*	*M3/s*	*M4/s*	*M5/s*	*Avg./s*
SIT (default)	647	21.5	21.5	21.4	21.4	21.5	21.5
SIT (optimized)	647	15.2	18.5	17.4	16.4	17.7	16.9
Shiron (optimized)	645	23.4	24.5	22.8	23.5	20.8	23.0
SIT + WiMAX (optimized)	688	17.8	16.6	18.5	17.4	14.9	17.0
SIT + WiMAX + Wi-Fi (optimized)	691	18.6	14.3	17.7	15.4	17.3	16.7
SIT (optimized, cached)	647	3.5	3.5	3.6	3.6	3.5	3.54

The results show that website access/load times increase with higher RTTs. This is expected since the time to complete a transaction is directly depending on the RTT. Parallel fetching and HTTP pipelining, where possible, can soften the impact a bit (default vs. optimized). Still, web browsing behind a satellite link feels rather sluggish. Compared to the un-accelerated Shiron DVB Link, the PEP module used at the HAI hub station does seems to improve the situation quite a bit. Turning on local caching on the FireFox browser reduces the access times for static pages significantly. It is therefore recommended to use local caches/proxies at the remote sites in order to speed up the CLIX website access.

6.5.9 UDP Measurements

To analyze the datagram forwarding behaviour, 'mgen' was used to generate UDP traffic across the various links. On the forward channel, 950 kbps were used for the video stream while 96 kbps were used for the audio stream. On the return channel, only the 96 kbps audio stream was sent (Abbasi *et al.* 2008). In order to calculate accurate latency times, the clocks on the two test machines were synchronized using the Network Time Protocol (NTP) protocol before the test. After the test, the clocks were synchronized again and a possible (linear) deviation was corrected for. This functionality is part of the Net Analyzer package as developed by FOKUS.

The measurements show that the main impact in terms of packet trip time and jitter was caused by the satellite links and the WiMAX connection. The impact of Wi-Fi is within the acceptance band. The graphical summary of the measurements using the HAI DVB-RCS platform with Continuous Rate Assignment (CRA) turned on in connection with WiMAX and Wi-Fi is presented below. For comparison, the results taken using the Shiron DVB-S InterSky platform with no traffic moderation are also shown.

The moderated (guaranteed) nature of the link provided by the HAI platform produces much cleaner results. There was no packet loss, and jitter was around 25ms, which was relatively low for a shared satellite channel.

The Shiron InterSky system did not provide a clean traffic pattern on the forward (DVB-S) channel. The return channel was fairly clean, except for a few latency spikes, which were related to an issue with the current firmware configuration. The Shiron InterSky DVB-S system uses FDMA on the return channel and therefore, it essentially provides a dynamically established, but guaranteed, non-shared link. It is therefore, worth noting that the TDMA DVB-RCS system with CRA enabled provides a return channel with almost identical characteristics as an FDMA system.

It should also be noted that other traffic was present on the Shiron InterSky DVB-S when the above tests were run. Therefore, the results of the Shiron InterSky DVB-S system could have been improved by using a traffic shaper.

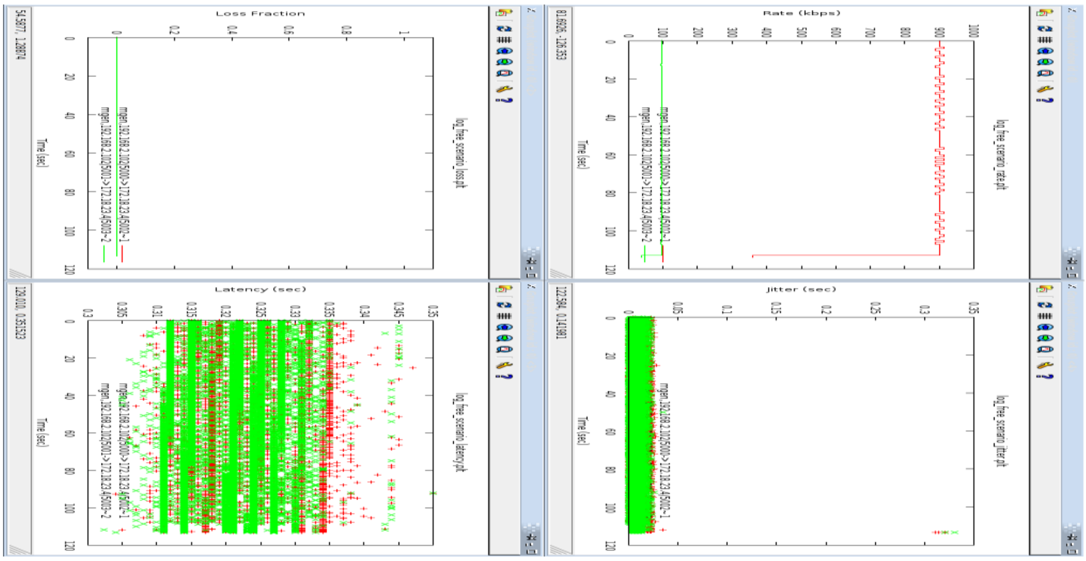


Figure 6.36: DVB-RCS scenario: HUB-to-SIT UDP Results

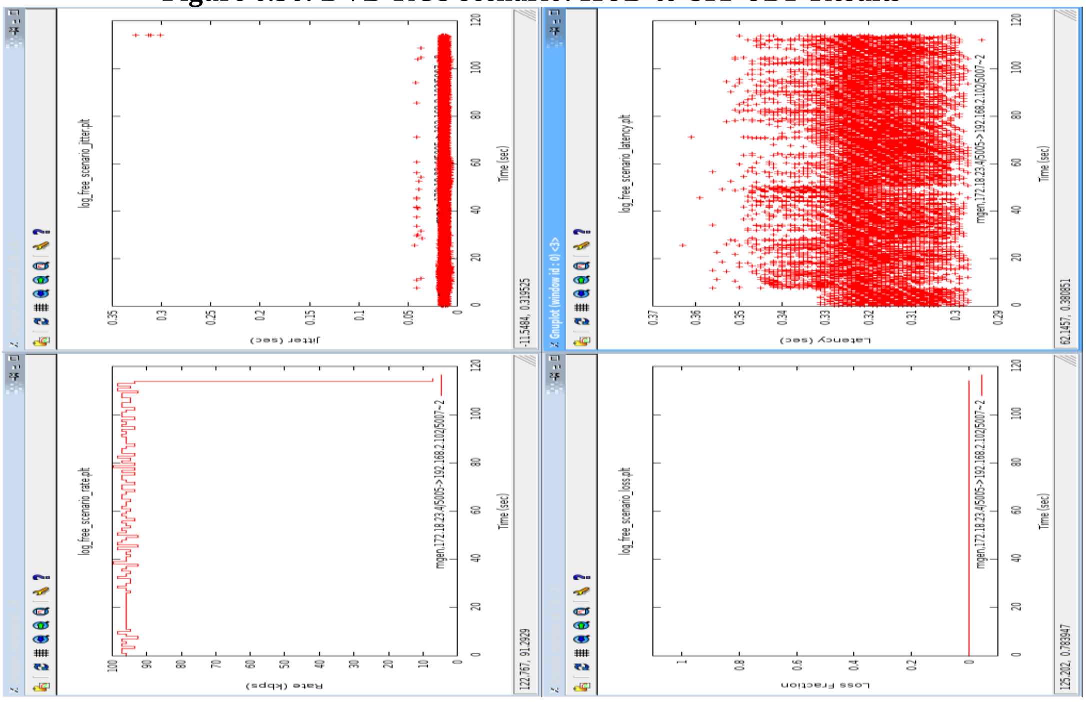


Figure 6.37: DVB-RCS scenario: SIT-to-HUB UDP Results

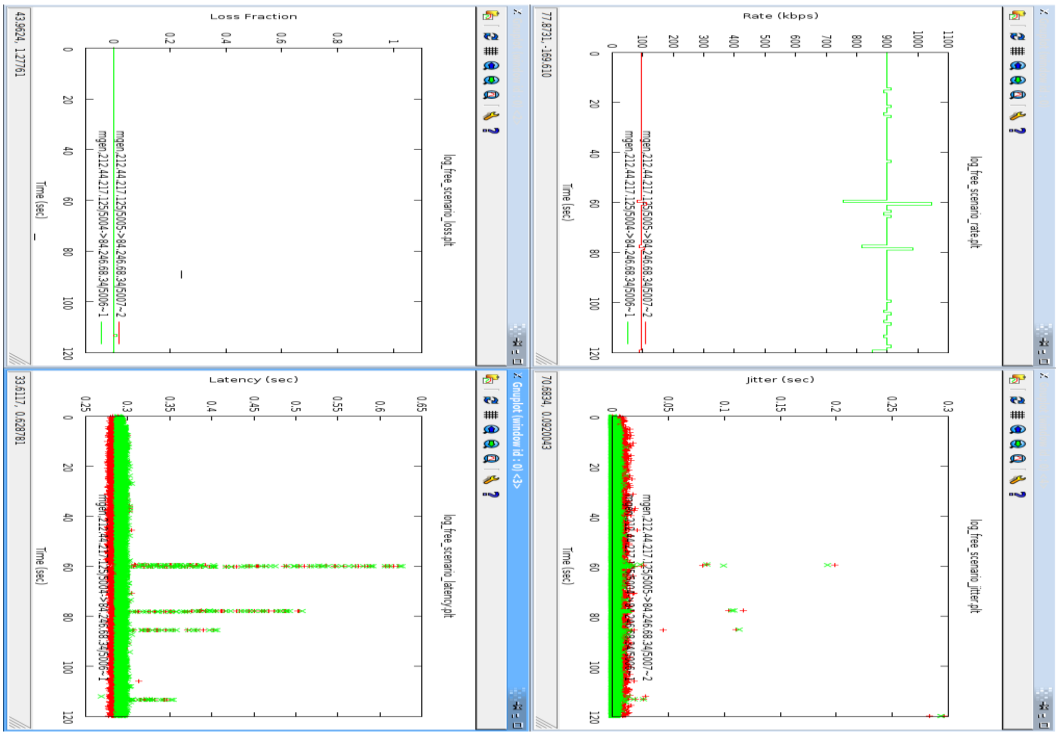


Figure 6.38: DVB-S scenario (using Shiron InterSky): HUB-to-SIT UDP results

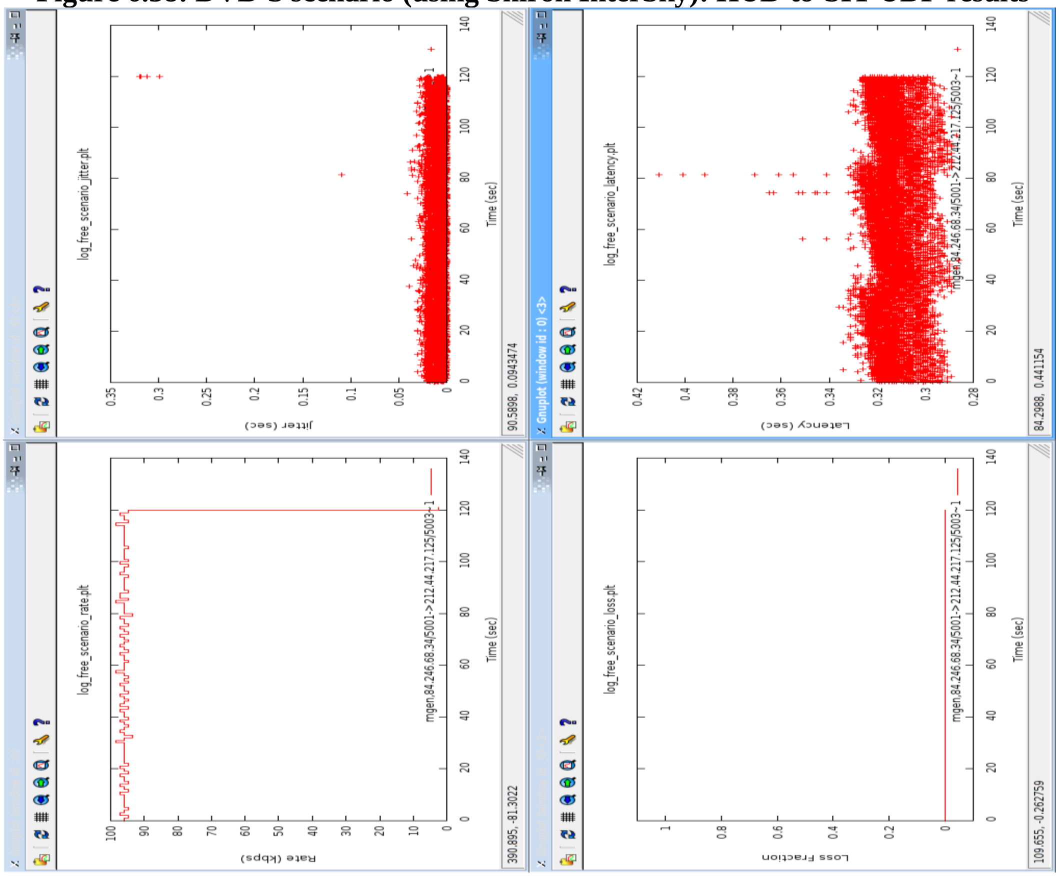


Figure 6.39: DVB-S (using Shiron InterSky): SIT-to-HUB UDP results

The above figures 6.36 to 6.39 (Abbasi *et al.* 2008) show that on an idle link loss was minimal, the RTT was sustained and therefore, the inter-arrival jitter was very little. The forward and return channels of a DVB-RCS system have very different characteristics, especially concerning jitter. Proper hub station configuration can hide those quite well so that the system performs nicely. If the link was to be shared with other traffic (web access), QoS enforcement (i.e. DiffServ) should be configured to prioritize the conferencing traffic over other traffic in order to guarantee an uninterrupted session.

6.5.10 Availability Measurements

In this subsection, availability figures for a DVB-RCS IDU test scenario are presented. The availability was monitored using standard ping software, with the central node being located at the DVB-RCS HUB. The software was based on the analysis of ICMP ECHO/ECHO_REPLY packets. It sends an ICMP ECHO packet to a SIT and waits for a corresponding reply. The ping application was set up with time-out value of a few seconds, after which the software stops waiting for the reply packet to arrive and issues a failure notification (Abbasi *et al.* 2008).

For this test, at a given interval, an ICMP_ECHO packet was sent from the HUB to the SIT and the percentage of not received ICMP_ECHO_REPLY packets was calculated.

Figure 6.40 (Abbasi *et al.* 2008) is a 30-day graph of a SIT. The ping response times were fairly constant and the ping failures were rather low. The short 100% outage results from the unit being powered-down for a couple of hours.

The result shows that out of total 3179 requests, 3094 were good (97.3%) and 85 were failed (2.7%), therefore the availability was 98.4899%, and downtime was 1.5101% in terms of time. The request time was 612 ms average in 5 minute interval.

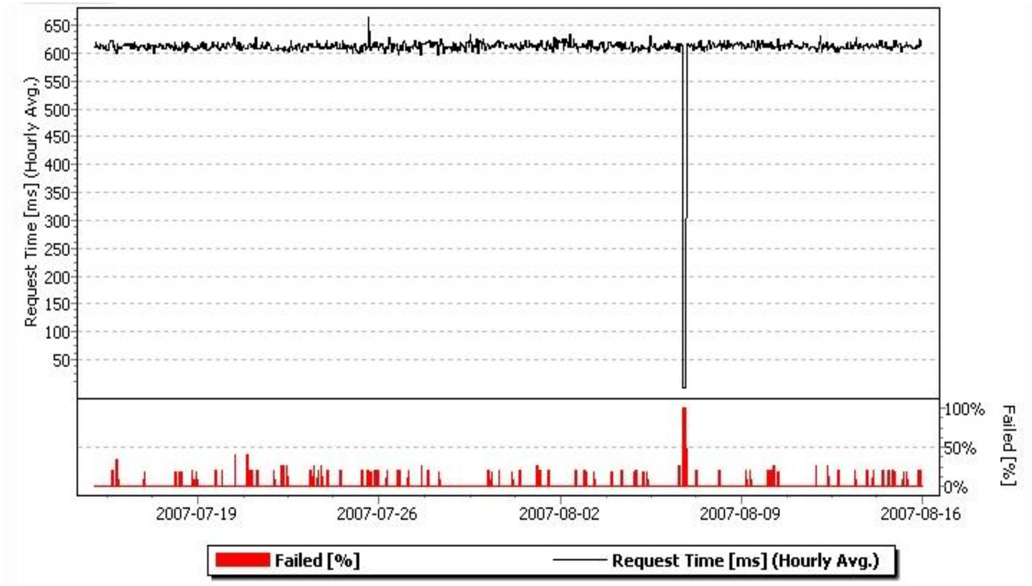


Figure 6.40: 30-day graph of SIT

Figure 6.41 below shows the ping response time and availability of a SIT. In that time, ISABEL-related measurements were taken. During the period, the response times increased because there was no special QoS traffic Class set up for ICMP traffic at the DVB-RCS hub. Therefore, the ping packets were queued as normal (Best Effort) traffic while the ISABEL traffic was prioritized.

From Figure 6.41 below, it can be seen that from the total 808 requests, 794 were good (98.2%), and 14 were failed (1.8%). The availability was 99.4167%, and the downtime was 0.5833%. This is based on ping reply packets average request time of 738 ms that do not return on time, in the form of single echo.

This can be explained considering a number of reasons, which include responsiveness of a SIT's traffic queue to incoming traffic, or the lack of incoming traffic (according to different capacity request mechanisms CRA, RBDC, VDBC, ADVBC assigned to the SITs).

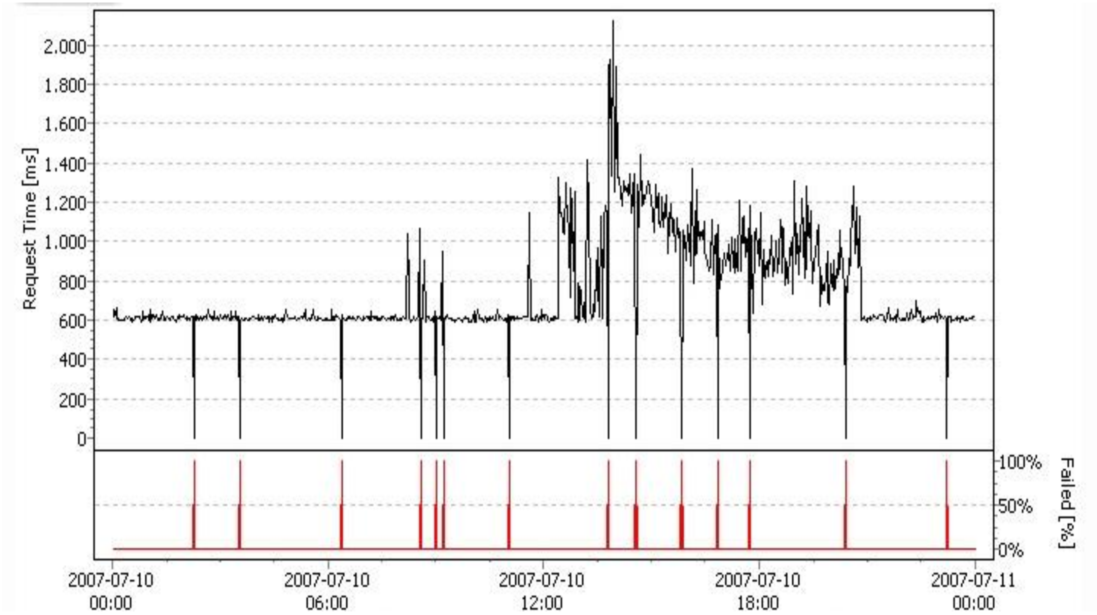


Figure 6.41: 1-day graph of Ping for DVB-RCS SIT

Figure 6.42 (Abbasi *et al.* 2008) shows a 3-day graph of the ping response times of a SIT. The results show that from a total of 2365 requests, 2319 were good (98.1%) and 46 were failed (1.9%), with one minute interval. The availability was 99. 5961% and downtime was 0.4039% in terms of time. The average request time was 819 ms.

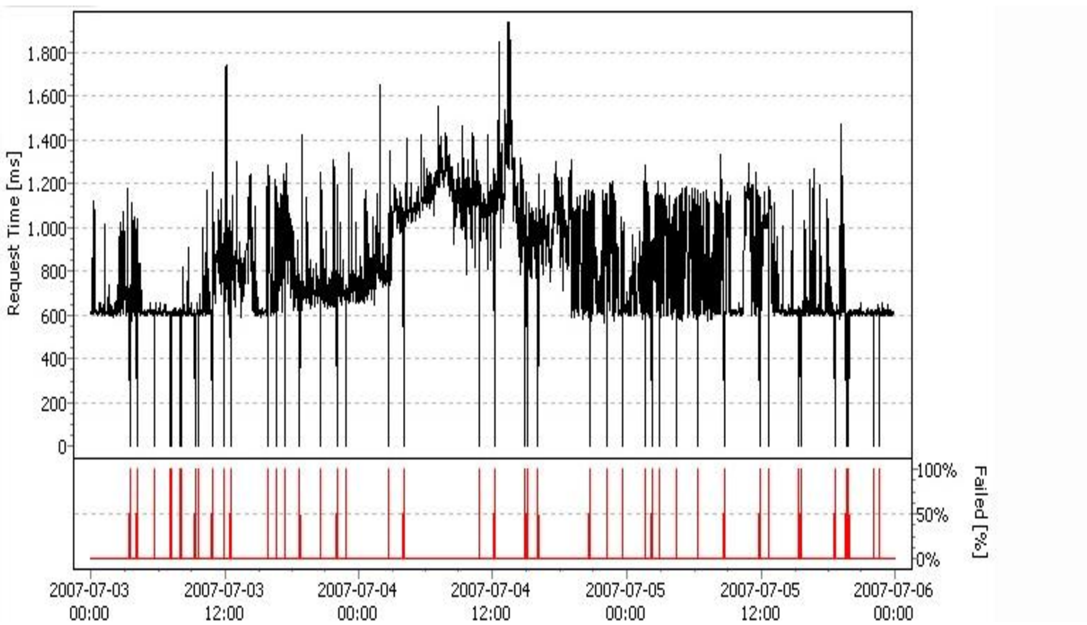


Figure 6.42: 3-day graph of a DVB-RCS SIT

6.5.11 Satellite Throughput Measurement

To determine the QoS requirement in a network, throughput analysis was performed. Throughput and latency are two important factors for network performance measurement and Quality of Service requirements. A number of screen shots were taken during testing. The following figures 6.43 and 6.44 show two examples of the screen shots taken during testing.

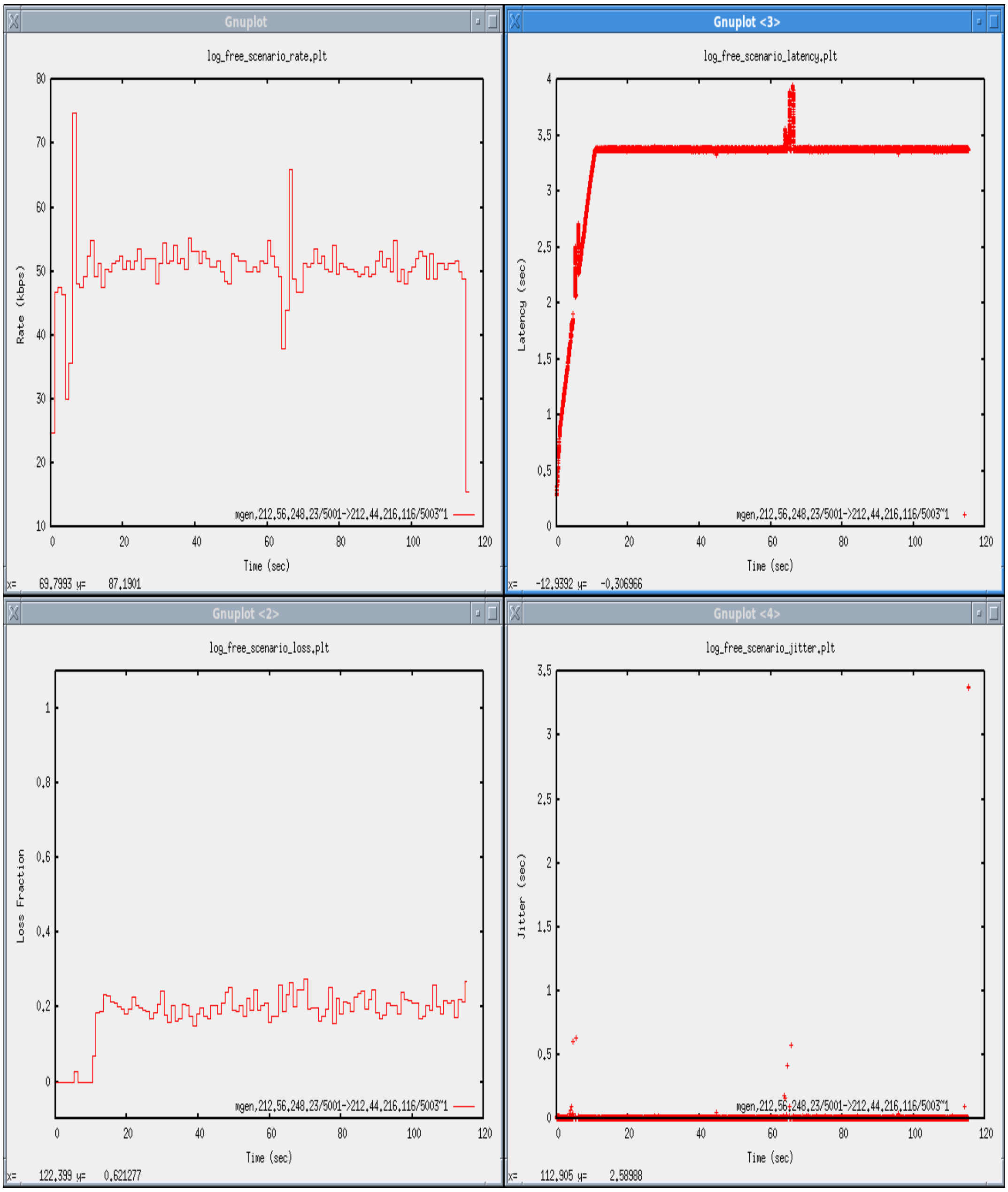


Figure 6.43: Satellite throughput from DVB-RCS (Hub) to Satellite Interactive Terminal

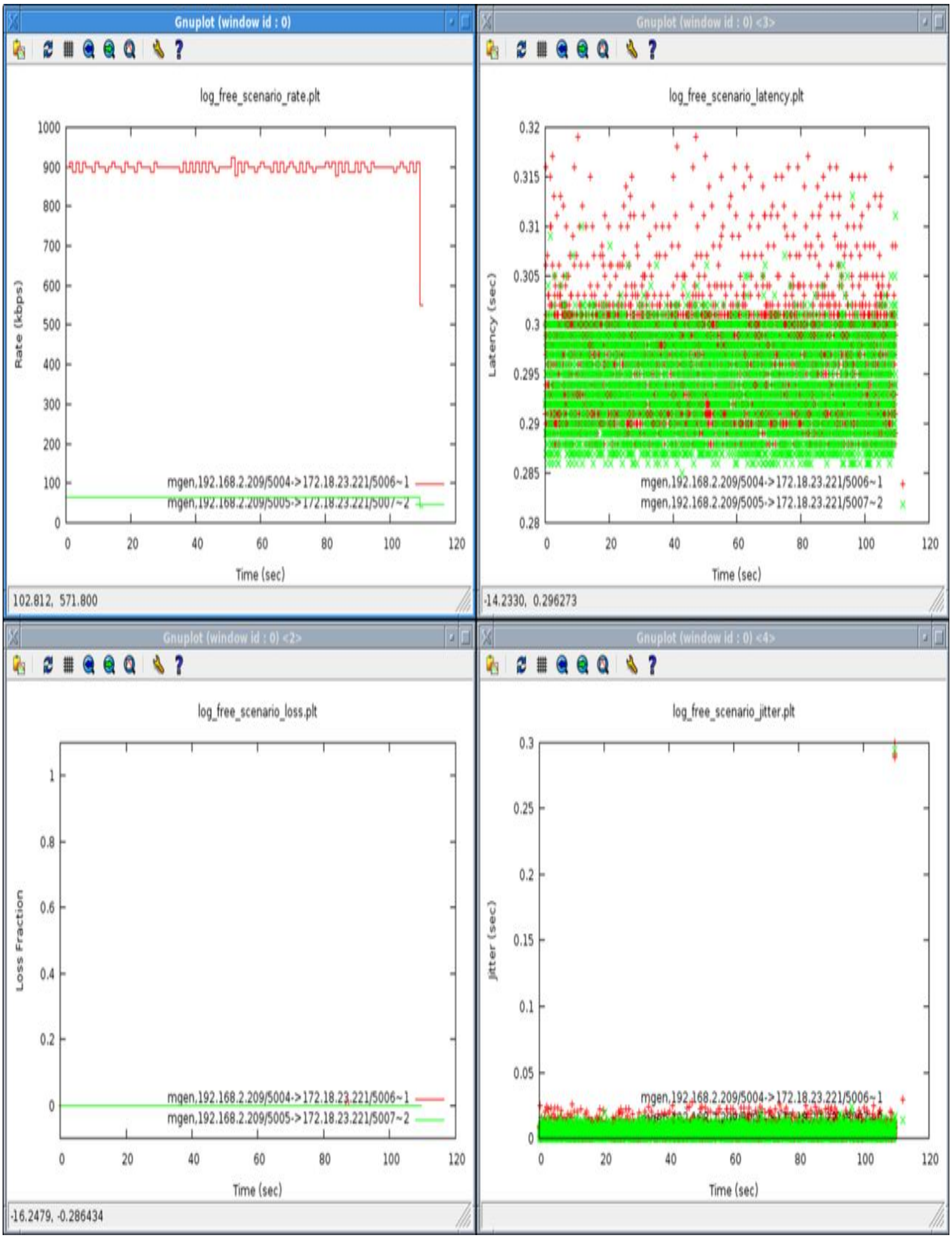


Figure 6.44: Satellite Interactive Terminal (SIT) to DVB-RCS (Hub)

These results indicate different values of throughput for various DVB-RCS scenarios and setup. The packet loss was low in UDP. The throughput data was taken from various site, recorded and compared in Table 6.7 below.

Table 6.7: Throughput measurement comparison for DVB-RCS various sites and nodes

Scenario	***Rate (Kbps)***	***Latency (sec)***	***Jitter(Sec)***	***Loss friction***
Hub to SIT	900-925	0.3-0.35	0.025	0
Hub to SIT	890-910	0.28-0.31	0.025	0
Hub to SIT- CRA	875-925	0.285-0.305	0.015	0
Hub to SIT- Shiron	890-910	0.28-0.3	0.02	0
Hub To SIT WiMAX-Wi-Fi - CRA	900-910	0.3-0.335	0.025	0
Hub to SIT WiMAX-Wi-Fi - Shiron	900-910	0.25-0.3	0.015	0
HAI to NCSR	900-925	0.28-0.31	0.025	0
HAI to Focus	890-910	0.285-0.3025	0.025	0
Focus to Hai	55-65	0.28-0.31	0.025	0-0.1
NCSR to HAI	62-66	0.28-0.31	0.025	0-0.02
SIT to Hub	75-90	0.70-0.78	0.05	0.1-0.2
SIT to Hub –CRA	95-98	0.29-0.315	0.012	0
SIT to Hub- Shiron	95-98	0.3-0.35	0.025	0
SIT to Hub - WiMAX-Wi-Fi - CRA	95-98	0.28-0.35	0.015	0
SIT to Hub WiMAX-Wi-Fi - Shiron	95-98	0.28-0.33	0.025	0

Table 6.7 shows data throughput, latency, jitter and loss between various DVB-RCS sites and nodes. The data rate was in the ranges of 55-100 kbps and 890-910 kbps. The latency was between 0.28 and 0.35 sec except for two scenarios where latency was within 0.3-0.5 sec and 1-3.5 sec, with few spikes which indicate that latency was extremely high for short durations, and packets were lost. The jitter was between 0 and 0.02 sec, except for a few spikes (Figure 6.45). The loss was minimum (Figure 6.46). This shows that there were no significant variations in packets, and that reduced jitter contributes significantly to better quality audio and video.

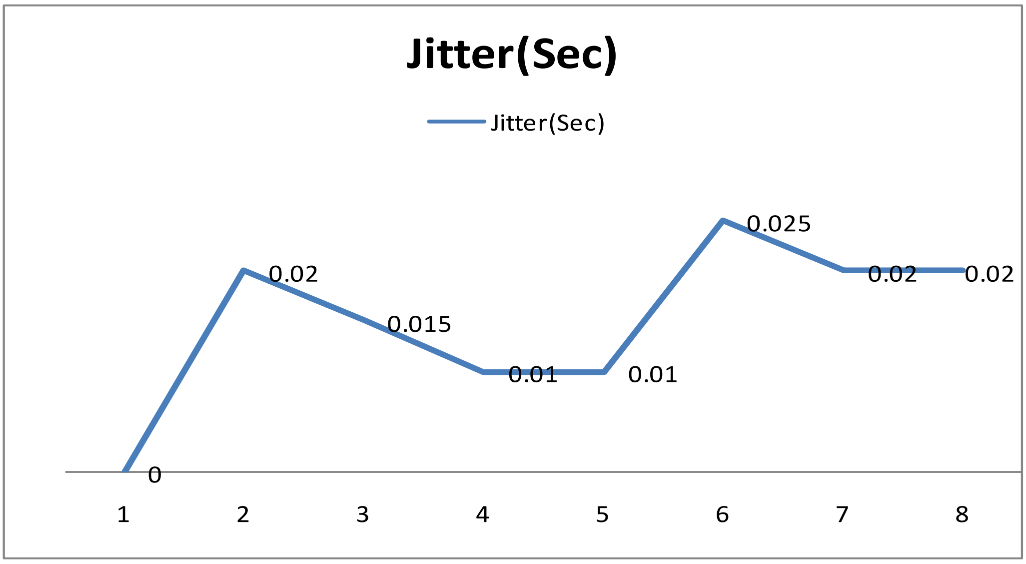


Figure 6.45: DVB-RCS Jitter

The loss fraction was within the range 0-0.02. This shows maximum losses are less than 98 %.

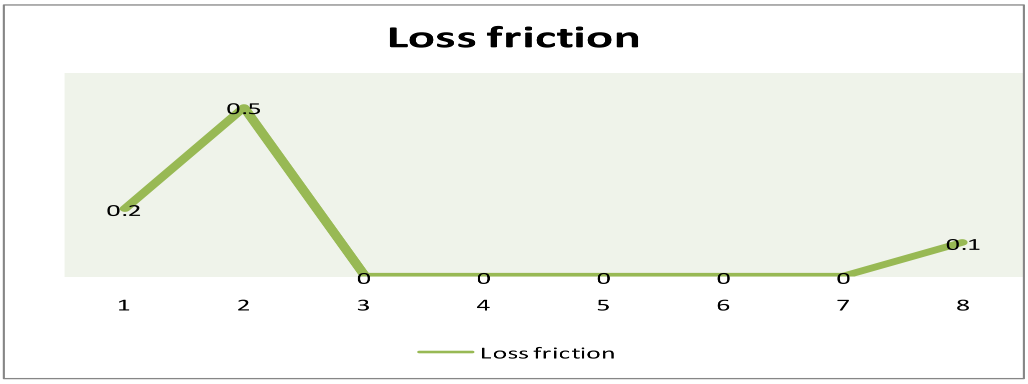


Figure 6.46: DVB-RCS Loss Friction

Figures 6.47, 6.48, and 6.49 below are the screen shots of DVB-RCS scenarios from Gnuplot, taken during testing at different timings between NCSR, HAI Greece and Fokus Germany, for measurements of jitter, data rate, and percentage loss. These figures show that the jitter was around 15ms, except for a few peaks at the Fokus site. The total data rate was around 250 kbps, except for some peak cases which were close to 450 kbps. The loss was less than 1% at NCSR and HAI Greece. However, higher losses can be seen at Fokus, but these losses did not relate to Bandwidth or Jitter. This might be due to the physical properties of the network.

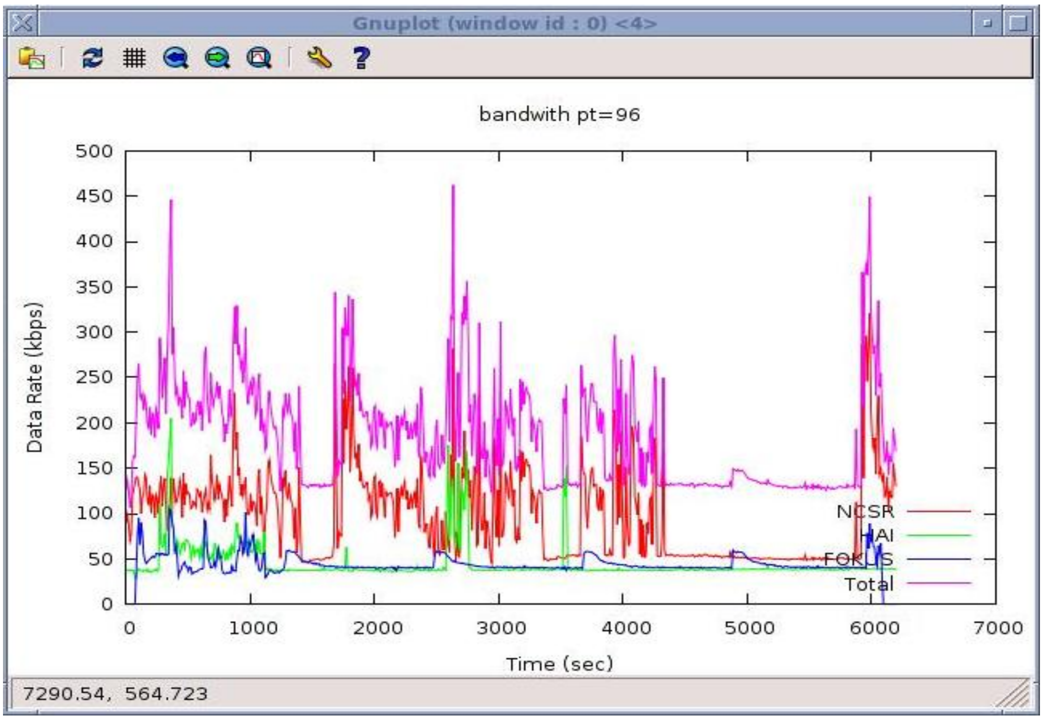


Figure 6.47: Jitter comparison through Gnuplot

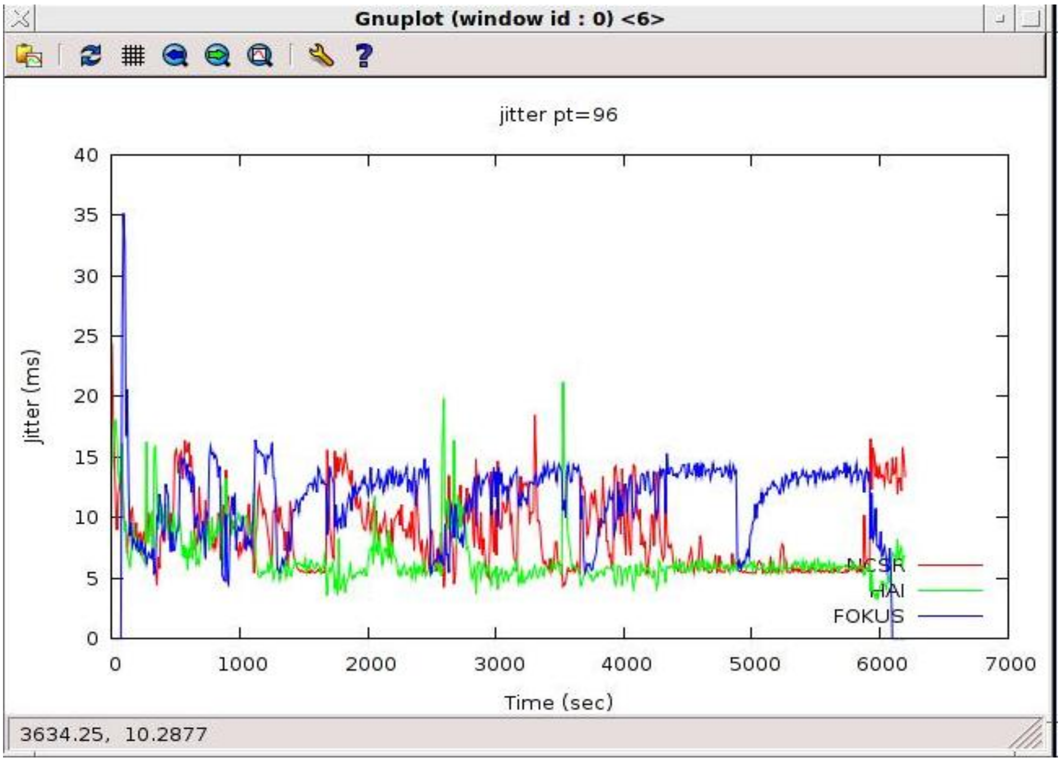


Figure 6.48: Data rate comparison through Gnuplot

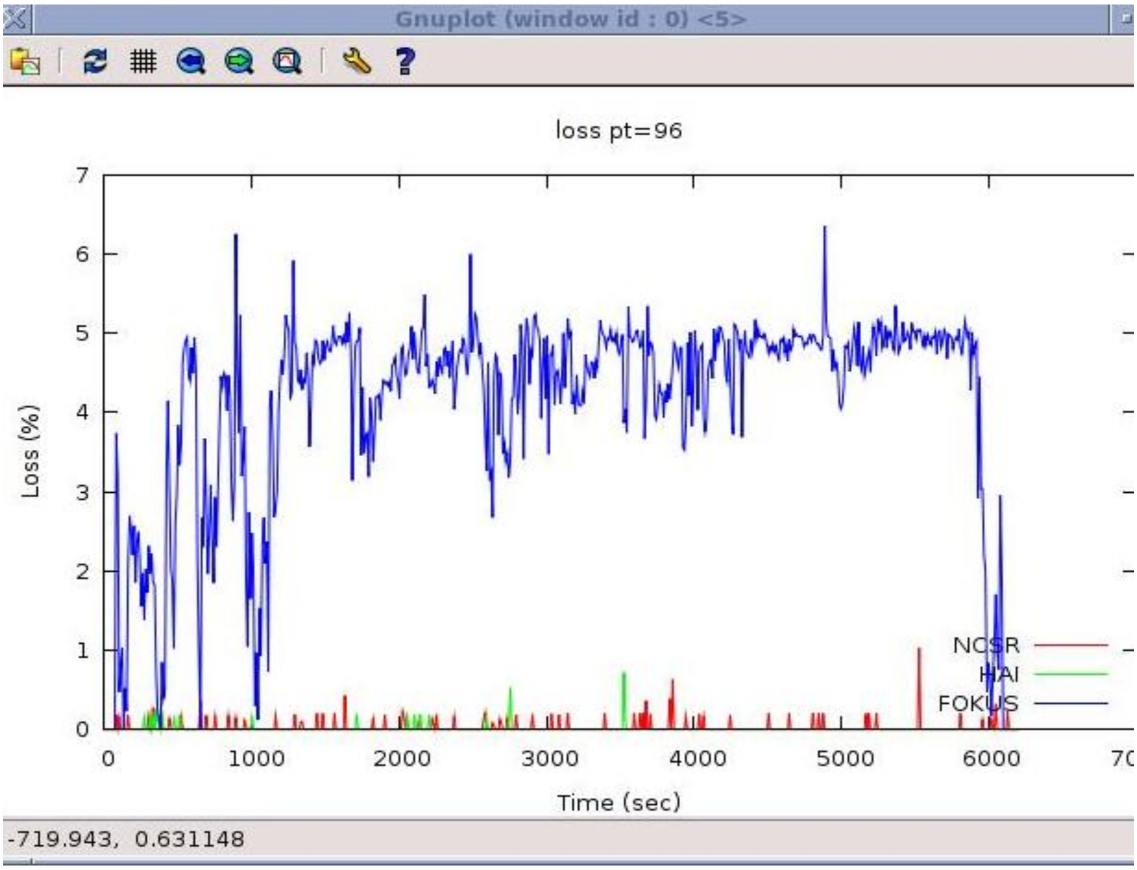


Figure 6.49: Loss (%) comparison through Gnuplot

A comparison shows that jitter, data rate and losses shown in above figures are almost the same with the results in Table 6.7. Therefore, it is concluded that these values can be used for Hybrid wireless network with confidence.

6.5.12 SCPC testing results for the Maritime scenario

Network performance tests were carried out to evaluate the performance of Hybrid Networks in the Maritime Scenario (SCPC shown in figure 6.10). Tcpdump and tethereal test tools were used to capture the traffic. The Wireshark test tool was used for the analysis of the tcpdump and tethereal captured files and data. All raw data with HTTP, IP, UDP and TCP headers were analysed and Charts, Conversation Tables, IO graphs and RTP were prepared. No packet loss or significant jitter was noticed in all these captured traffic/ files/data. Jitter is around 60ms for traffic up to 512k in ship measurements as shown below. However, it has been observed that there were several packets which were lost on many occasions and the jitter was long for the 512k (big) window and 1MB windows for RTP analysis. The reason for this is that the system bandwidth supported windows only up to 512k for this test scenario. In the analysis of the IO graphs, there were no noticeable gaps, nor could any extra delay and drop down problem be noticed.

The results also show the average number of bytes per flow, number of packets per flow, duration of flow, average time of flow, histogram of packet size, bandwidth usage per protocol and application, packet loss, maximum and mean jitter. From the analysis of the packet expert information from Wireshark, some “UDP bad checksum\incorrect” messages were found. Analysis of the data shows that the most of the packets are in the range between 39 and 1280.

The following figure 6.50 shows an example of IO graphs taken through Wireshark. While comparing the 256k and the 1MB cases in the IO graphs, differences in jitter and throughput can easily be observed.

These results show that few packets are lost. Minimum jitter is found in all these captured files. Jitter is around 60ms for traffic up to 512k in ship measurements. However, in 512k big window traces analysis, it is observed that there is a long jitter and some packets are also lost. The reason is that the system bandwidth was not sufficient to support this test scenario. IO graphs show no significant gap in time. Furthermore, there is no considerable delay also or drop down problem which can be observed in SCPC ship testing scenario.

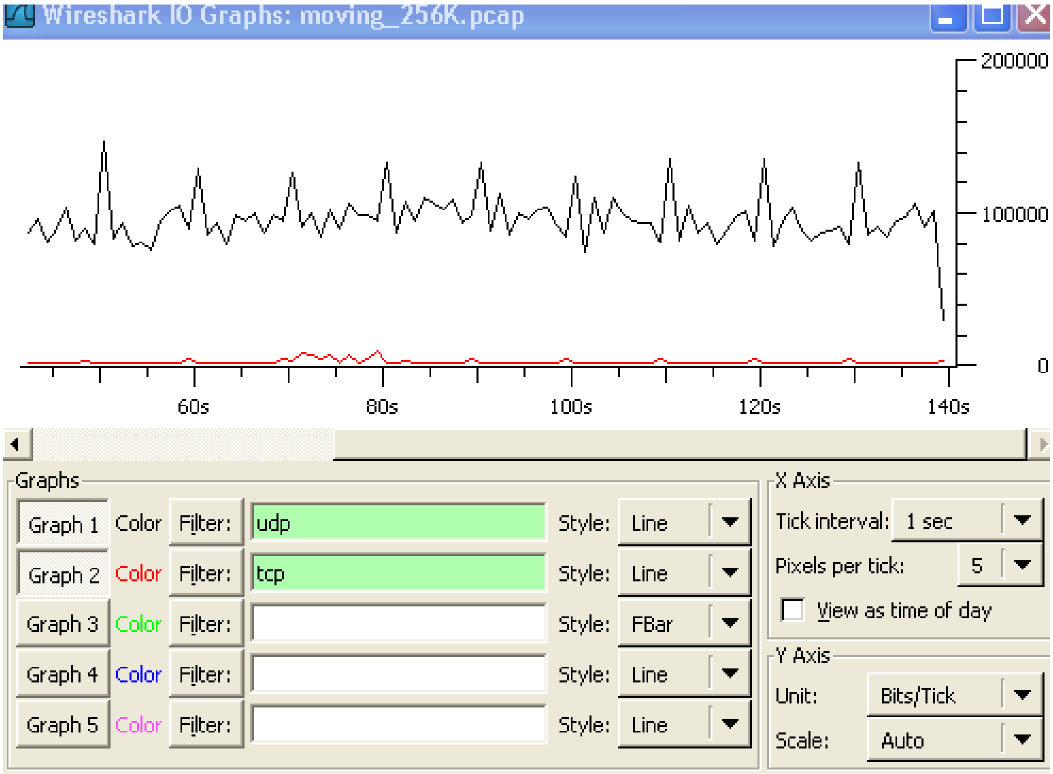


Figure 6.50: 256k window moving ship Wireshark IO Statistics

A comparison of the SCPC results showed that the traffic was smooth for stationary small windows up to 512k. But significant losses were recorded for the cases of 512k moving ship and 1M window stationary ship.

Figure 6.51 below shows an example of round trip time (RTT) for both stationary ship and moving ship cases, as analysed by wireshark. The RTT is around 575 ms for all types of traffic. These results are almost the same with the results of the initial ping and trace route tests done during the preliminary interoperability testing phase. Although the delay affects the quality of video signal – an effect which cannot be avoided due to satellite distance - but this delay is acceptable for VSAT communication.

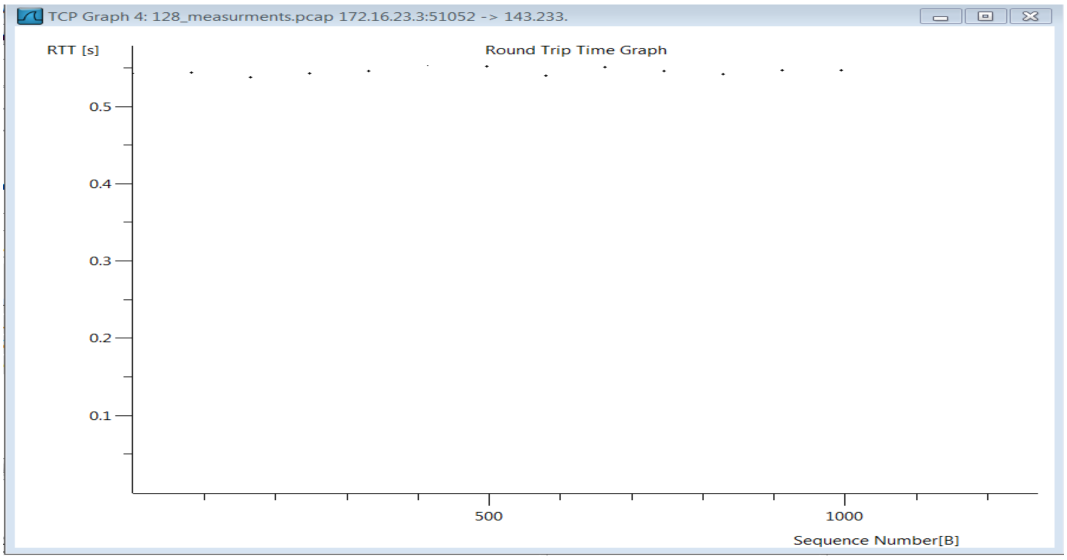


Figure 6.51: 128k stationary ship roundtrip time

Figure 6.52 below shows a comparison for percentage of type of packets protocol taken from various wireshark screen shots. The UDP protocols are from 98.43 to 99.12 % and TCP protocols are 0.70 percent. The remaining protocols are spanning tree protocol, link layer discovery protocol, and address resolution protocol. There are no losses in UDP and other protocols except in TCP. The loss can only be seen in TCP protocol, which is due to congestion and bandwidth restrictions. This loss is only 0.32% which is within the acceptable range in this scenario.

Protocol	% Packets	Packets	Bytes	Mbit/s	End Packets	End Bytes	End Mbit/s
Frame	100.00 %	9336	1354231	0.071	0	0	0.000
Ethernet	100.00 %	9336	1354231	0.071	0	0	0.000
Internet Protocol	99.12 %	9254	1348686	0.070	0	0	0.000
Transmission Control Protocol	0.70 %	65	6670	0.000	33	2178	0.000
SSH Protocol	0.02 %	2	292	0.000	2	292	0.000
Short Frame	0.32 %	30	4200	0.000	30	4200	0.000
User Datagram Protocol	98.43 %	9189	1342016	0.070	0	0	0.000
Data	98.43 %	9189	1342016	0.070	9189	1342016	0.070
Logical-Link Control	0.81 %	76	4560	0.000	0	0	0.000
Spanning Tree Protocol	0.81 %	76	4560	0.000	76	4560	0.000
Link Layer Discovery Protocol	0.05 %	5	925	0.000	0	0	0.000
Short Frame	0.05 %	5	925	0.000	5	925	0.000
Address Resolution Protocol	0.01 %	1	60	0.000	1	60	0.000

Figure 6.52: 128k SCPC Protocols ratio

Table 6.8 shows results from Wireshark traffic analysis for SCPC. This shows a comparison in terms of average packets/sec, average packet size in Bytes, Average Bytes/sec and average M bits/sec for various sites. The average packets/sec ranges from 60.882 to 113.44. The lowest value is for 128k ship SCPC and the highest is for 512k big window SCPC. The average sizes range from 145.055 to 871.763 bytes. Again the smallest value is for the 128k window SCPC. The average bytes/sec range from 8831.282 to 56681.886. In all cases, these values are low for the 128k small window case.

Table 6.8: SCPC Network Traffic size comparison

SIT	Avg. Packets/sec	Average Packet Size Bytes	Average Bytes/sec	Average M bits/sec
128K SHIP	60.882	145.055	8831.282	0.071
256K STATIONARY	70.618	164.465	11614.235	0.093
256 K MOVING	66.69	184.946	12334.143	0.099
512K MOVING	110.946	216.831	24056.528	0.192
512K STATIONARY	109.583	172.307	18881.898	0.151
512K BIG WINDOW	113.44	301.941	34252.232	0.274

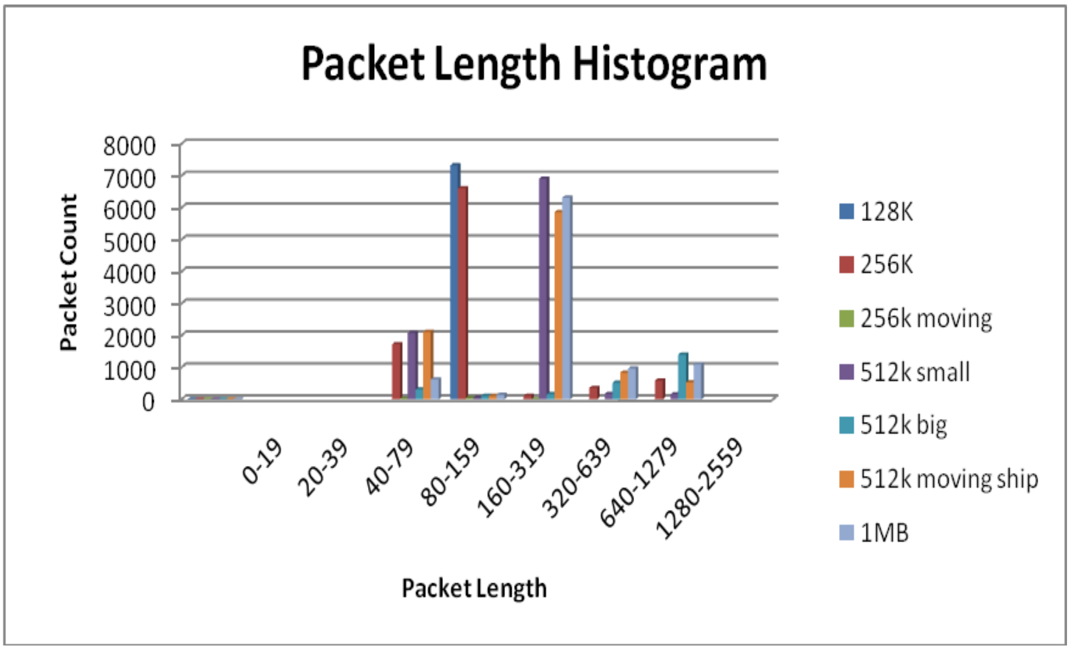


Figure 6.53: Packet Length histogram for SCPC

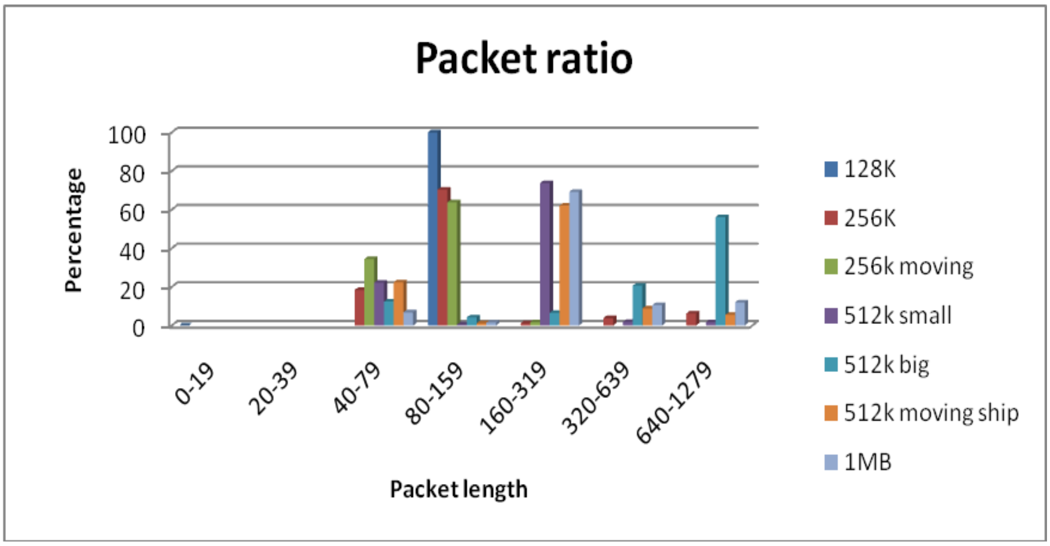


Figure 6.54: Packet Percentage ratio for SCPC

Figure 6.54 shows the packet ratio for SCPC. It can be seen that the majority of packets (60% to 100%) were between 80 and 319 bytes in size. The RTT was also tested in this SCPC scenario and all values were around 575 ms, which were similar, to the results shown in section 6.3.5 during the initial test for this scenario.

6.6 SUMMARY

A series of testing was carried out in order to establish the end-to-end interoperability issues and Quality of Service requirements for the heterogeneous Satellite Terrestrial wireless communication technologies. The measurements were taken from various scenarios and nodes. The important test tools such as ping, Iperf, httperf, traceroute, U2m, DITG, TCPdump, wireshark, and Tshark, were used for measurement and result analysis. The testing fulfils the objectives to conduct the test and to identify any end-to-end interoperability issues and to ensure that QoS requirements are assessed.

The result and its analysis presented are divided into two parts; preliminary test and final interoperability test, the summary is as following.

The evaluation of multicast connections indicates that the distinction is to be taken into account while Isabel application supports multicast bi-directional sessions

(ASM model), i.e. any host can send and receive multicast packets of data. From the particular satellite communication point of view, a multicast session was implemented basically as uni-directional (SSM model). This is because in specific DVB-RCS implementation, a multicast distribution tree was generated, with its “root” from the sending station behind SIT to hub and the “branches” from the hub to the receiving hosts behind the rest of SITs. The flow of data in this scheme is from root to branches. In case of another host behind a SIT is willing to transmit its own flow of data; another “tree” must be generated with new host to its root and other hosts to its branches. In this scenario many multicast distribution trees can be supported in network SSM mode, but ASM mode will have problems.

In the IP-level, Isabel multicast sessions were supported when the session server was appointed to a standard multicast group and all clients willing to participate in this session must join the same multicast group – thus all the stations (server and clients) were configured with the same multicast IP group address. On the other hand, with the particular DVB-RCS network multicast communication implemented here a multicast distribution tree was created with a specific source S sending to a group G, and hosts-receivers joining channel (S,G) will receive packets from source S and not any other source.

Keeping in view the above, it is analysed that the support of Isabel and specific DVB-RCS network was difficult in the multicast bi-directional scenario; however, multicast can easily support uni-directionally flow.

Another constraint observed from results was that when using Isabel over HAI’s Greece DVB-RCS system, Isabel was configured to use different multicast groups for video, voice, blackboard, file transfer etc. but the remote SITs used by HAI can receive any multicast group traffic from the Forward link (From Central-Hub to remote Sit), while allowing traffic from their LAN towards only one multicast group, back up to their Return link (remote SIT to central-hub), thus permitting only one kind of traffic flow (video traffic that was only originated from an Isabel client and to be streamed towards other clients).

All links (DVB-RCS, WiMAX, Wi-Fi) have proven that they were stable over a long period, i.e. compared to the duration of a typical session. All links indicate stability of network as values for delay and jitter were constant over time. Since all links show a similar behaviour whether they experience load or not, therefore all links with and without load were stable.

Since an analysis of a measurement with D-ITG consumes a massive amount of memory proportional to the duration of the measurement, the duration of each measurement was limited.

To summarise, it has been analysed that in the case of one-site connecting to the satellite hub, the use of DVB-RCS alone or interconnected with Wi-Fi/WiMAX equipment makes no significant difference, regardless of the bandwidth allocated as in testing scenarios as it was 512 and 1024Kbps. In such a scenario, there was no need to make any changes on the default audio and video settings of platform as it produces very high multimedia quality for the user even with the standard satellite time delay of 600ms.
However, in case of two or more sites communicating simultaneously with application server, the time delays increase. The audio and video quality also decreases due to limitations imposed by the allocated bandwidth. It is analysed that in the presence of two sites, a Forward/Return link of 512Kbps proved insufficient, where each site produced a media stream of 300 Kbps approximately. For this very same reason with three sites was a success, as each site required 300kbps stream for a relatively high quality communication which resulted in overall bandwidth requirement well within the 1Mbps bandwidth limit assigned. The overall conclusion is that in order to send high quality and cost effective video streams with acceptable time delays, the proper bandwidth is required to be allocated according to the number of sites that can be served in the specific network infrastructure.

Extensive tests have been performed in order to assess the end-to-end interoperability issues and Quality of service requirements, and performance of the hybrid wireless network infrastructure during final interoperability testing.
To prove the performance of each link long term stability for the duration of 5 and 24 hours and short term stability with 1Mbits/s load for 1 hour were tested. First the Satellite link was tested for delay and jitter measurement with duration of five hours. In the dot plots, there are only few outliers within the range of 0.3s to 0.6s delays. The jitter values group in small areas at approximately 0.005, 0.02, 0.032 and 0.048s. The cause for this was unclear but the effect is unique for the satellite link. Since the measurement with 1 Mbit/s load does not show this effect, it can be considered as irrelevant.

The WiMAX link was tested for both BE and rtPS services/characteristics.
The WiMAX link was set to both BE and rtPS characteristics and carried out test for 24 hour for each setting simultaneously. In WiMAX BE the delay values was from 0.03s up to 0.045s. There are several regions with increased outliers delimited. It can be assumed, that this is due to temporary disturbances in the transmission medium. Since the amount of these outliers was small, they did not show up in the box plots.

For WiMAX rtPS characteristic the variance of the delay was similar to the BE characteristic, However, jitter values are significantly better.
The Wi-Fi was also tested for 24hours. Most of the values range between 0.002s and 0.003s in dot plots of the delay. The quantity of the delays between 0.003s and 0.04s is small relative to the delays between 0.002 and 0.003s. These values are counted as outliers and did not show up in the box plots. These outliers are most likes caused by the retransmits of the Wireless LAN hardware. The box plot of the delay samples shows waviness. This was caused by non-linear drift of the stations clocks. Since this drift ranges only 0.00015s it was considered as irrelevant.

The WiMAX RTPS versus BE characteristics were compared. In this test a WiMAX sector controller with an available Bandwidth of 10 Mbits/s serves two Clients, one with rtPS characteristic and one with BE characteristic. As the packet loss plots indicate both links didn't experience any packet loss. This was due to the TCP Transport protocol that retransmits lost packets and rescales the TCP Window in order to avoid congestion. As the bit rate plots show, both characteristics reach more or less the same bit rate. By comparing the histogram and box plots of both characteristics, it is found that the rtPS link has a significant lower jitter and delay distribution.

The short term measurement with load was done for one and four hours due to the huge amount of data. The UDP data stream with 88 packets per second with 1500 byte per second was generated from D-ITG. This data rate was under the saturation of the links. However the packet loss was very little. Delay and jitter was expected to be similar to the long-term series, however the higher packet rate may level effects caused by power saving or timing algorithms.

The Satellite was tested with 1 Mb/s load. The satellite link with load behaves pretty much the same as without load. It shows that the variance of delay was smaller; however the jitter variance was higher. It can be noticed that the spikes were only shown in the unloaded scenario. There was also noticeable packet loss.
The WiMAX characteristics both delay and jitter behaves similar to without loaded link. Both BE and rtPS links did not show any packet loss. Under this circumstances it is insubstantial if the BE or the rtPS service is chosen.

The wireless LAN Wi-Fi link the delay dot plot the outlier percentage is higher than in the long-term measurement. Again, this was due to retransmits done by the hardware. There was little amount of packet loss but it was almost insignificant.
The wireless LAN Wi-Fi link the delay dot plot the outlier percentage is higher than in the long-term measurement. Again, this was due to retransmits done by the hardware. There was little amount of packet loss but it was almost negligible.

It is concluded that the WiMAX IEEE802.16 can send IP data symmetrically in both directions. For WiMAX, average Round trip time was calculated between 20-40 ms between sector controller and subscriber station. The maximum throughput was around 23 Mbps in both directions.

Satellite services face significant round trip time delay of approximately 640 to 680 ms when communicating over the geostationary satellite. In some cases it was noted even worse, more than 1200 ms from Server to SIT and back. Since RCS is a highly asymmetric service, broadband requirements can be met when delivering through downlink towards the target area. In some of the initial packets it was even worst. Although due to clever communication prioritization on the return channel, the jitter of the round trip time was minimized and throughput numbers were optimized at the same time but this could not provide better to QoS requirements.

The latency problems were also noted in few results but were at acceptable level of around 100 ms for TCP and UDP traffic. It is analysed that the UDP traffic was found more suitable due to connectionless properties and small header bytes as compared to TCP. As TCP protocol requires handshaking for end- to- end communications, TCP packet losses were far less but can be improved by forward correction error. For link layer interoperability DVB-RCS return link is required to be available.

The biggest issues for a satellite links are either weather conditions or transmission delay. Weather conditions are usually accounted for by the link budget calculations. i.e. how much signal attenuation is expected due to heavy rain, which affects network availability? One of the major issues found was the time delay which was in addition to normal RTT time. This can be a particularly a problem for services such as video conferencing, online gaming services, or simply some forms of tele-commuting or multimedia services.

QoS parameters were measured such as data rate, bandwidth, delay, jitter, latency, packet loss and throughput. Testing was carried out for unicast and multicast services requirement. A combination of measurements were taken for Hybrid wireless networks mainly Wi-Fi/WiMAX with Satellite DVB-RCS and SCPC VSAT technologies. The results presented and analysed for Hybrid wireless network infrastructure in terms of low jitter, low packet loss and sufficient sustained bandwidth.

The result shows that the performance of unicast and multicast services over DVB-RCS was fantastic. There were no packet drops, duplicates or unordered packets have been found in the captured traces. However, in some scenarios few QoS issues such as bandwidth, throughput and delay was observed during result analysis. These issues were more in TCP traffic as compared to UDP.

The network availability test was also carried out for DVB-RCS Scenarios. It was found that in worst case the availability was 98.5%.

Chapter 7

Conclusions and Recommendations

7.1 Overview

Heterogeneous satellite terrestrial wireless technologies are becoming more common and popular due to the advantages of satellite technology inherent properties of broadcasting, multicasting and capability of high speed internet connection. However, satellite communication alone cannot compete and capture the whole communication market, unless it is integrated with terrestrial networks. Due to the advantage of availability, satellite communication networks are being integrated into the infrastructure of the modern terrestrial communication networks and are becoming popular for the delivery of education contents, tele conferences, data, information, entertainment and emergency services.

This chapter provides a summary of the thesis and the conclusions from the research carried out. Moreover, research findings, limitations and avenues for future research are discussed, along with the extent to which the technologies would be applicable and adaptable.

7.2 Research Summary

The aim of this research is to examine end-to-end interoperability issues and QoS requirements by using a wide range of real network scenarios of Satellite and Terrestrial Wireless Networks, as opposed to using simulation tools and models. The results are compiled, compared, and analysed in detail using various test tools to make a significant contribution in terms of study, test plan, testing methodology and implementation.

A comprehensive overview of wireless communication technologies has been presented in Chapter 1, which details the aim, objectives, research questions, and research rationale.

The literature review, supported by critical analysis, is presented in Chapter 2. It identifies gaps in previous studies and explores new avenues, so that the research aim, objectives and questions can be addressed effectively. Several studies in the literature discuss Wi-Fi, WiMAX, DVB-RCS, SCPC and 3G standards, and their characteristics such as operating frequency, bandwidth, data rate, basic configuration, coverage, power, interference, social issues, security problems, physical and MAC layers design and development. Some of the previous studies concern research on an individual technology or layer as an independent entity, while little if any cross-layer research has been done. Some of the studies were limited to link layer characteristics and TCP performance only. However, none of the previous studies include systematically all issues of end-to-end interoperability and QoS requirements, such as bandwidth, jitter, latency, packet loss, link performance, TCP, and UDP performance for end-to-end level on Hybrid wireless networks.

It was established that there is a scarcity of research in the area of end-to-end interoperability issues and Quality of Service requirements for Hybrid wireless network.

This provided the main motivation behind this thesis - namely, to address an area of research which covers end-to-end interoperability issues and Quality of Service requirements on hybrid wireless networks, including both satellite broadband and terrestrial wireless technologies, in terms of bandwidth, delays, jitter, latency, packet loss, throughput, TCP performance, UDP performance, link performance, unicast and multicast services, and availability. The wireless communication technologies and their associated standards have been described and discussed. A comparison of the various wireless technologies, standards, protocols, and models was also carried out in this chapter.

Various applications and platforms such as Skype, msn, net meeting, clix and the Isabel platform were studied and compared in Chapter 3. The chapters 2 and 3 meet the first objective of this research: "***to study and compare, technologies, standards, and architectures of hybrid wireless networks***".

The interoperability issues, such as bandwidth, delays, jitters, latency, packet loss, throughput, availability, and link layer performance, were described and discussed in detail in chapter 4. This chapter also gives a comprehensive description and comparison of network testing tools, such as Ethereal/Wireshark, httperf, I Perf, Kismet, Mgen, Netmeter, OWAMP, PRTG, Pathchar, Pathrate, Pchar, Ping, SNMP, TCPdump, TCPtrace, and Trace route etc. All these software-based testing tools are open-source, flexible, easy to use and economical/affordable. By using a combination of these tools, the interoperability issues and QoS requirements for hybrid networks can be tested properly. A comparison of these test tools in terms of functionality is provided at the end of Chapter 4. This achieves the second research objective: "***to investigate the interoperability issues and compare test tools for Hybrid Wireless Networks***".

A comprehensive and generic test plan and testing methodology were devised to address the end-to-end interoperability issues and QoS requirements. These were described in chapter 5. This testing methodology is suitable for testing all the common interoperability issues and QoS requirements in Hybrid networks. Chapter 5 also outlines test procedures and lists of the test processes in order to use common test tools available to test end-to-end interoperability issues and QoS requirements. A detailed test plan and testing methodology was devised for delay, jitter, packet loss, bandwidth, throughput, availability, performance of audio and video session, multicast and unicast performance, and stress test.

This testing methodology is based on all test tools functionality and test scenarios with expected results. The proposed tests address the important issues, such as delay, jitter, packet loss, bandwidth, throughput, availability, performance of audio and video session, multicast and unicast performance, and stress test. Possible combination of scenarios and nodes were proposed, such as Hub to SIT, SIT to Hub, Hub to WiMAX/Wi-Fi, WiMAX /Wi-Fi to Hub, and Wi-Fi/WiMAX to WiMAX. It was established that the roundtrip delay for DVB-RCS should be 500-650 ms. For WiMAX, it should be <60ms and Wi-Fi <20ms. The overall jitter should be +/- 50 ms. The packet loss should be less than 1 % and availability should be 99%. The UDP should reach 99% of the available capacity. The performance of Wi-Fi should be in the range of 4-24 Mbps and WiMAX at least 6 Mbps. The acceptable audio loss should be 22.5 % and video 10%. The latency for audio and video should be less than 20ms. This achieves the third research objective: "***to prepare a comprehensive test plan and methodology which allows testing of all possible Hybrid wireless network scenarios***".

One of the key features of this research was to capture packets and traces from the test bed of the network developed for the BASE² EU project (supported by the Aeronautics and Space programme of European Commission), which has designed and deployed a hybrid, satellite and wireless-based broadband network infrastructure on land (agrarian scenario – remote locations) and at sea (maritime scenario - ships). The second feature was using real time live scenarios of various hybrid wireless network (heterogeneous satellite terrestrial) nodes, instead of using simulation and models. Most of the previous research has relied on simulation, using software simulation tools, such as the ns-2 Simulator.

The testing was carried out in two phases; namely, the preliminary and the final interoperability test phases. The files, data, and packets were captured. Most of the tests were conducted on site, and only a few tests were conducted remotely. Test tools such as Ping, Iperf, httperf, Mgen, MRTG, U2m, TCPdump, Trace route and wireshark, were used to test bandwidth, packet loss, latency, throughput, link performance, TCP and UDP performance, load and availability testing.

The results were analysed and discussed in Chapter 6, covering end-to-end interoperability issues and QoS requirements respectively. It was found that results taken via using various types of testing tools did not significantly influence the measurement of the result. Analysis of results highlighted a few issues with bandwidth, delay, jitter, packet loss, throughput and link performance, but overall hybrid wireless networks were interoperating smoothly. There are slightly more interoperability issues to do with the satellite link. This achieves the fourth research objective "***Analyse results and investigate interoperability issues and QoS requirements***".

It is concluded that the Hybrid wireless technologies are flexible and scalable, and provide better performance in terms of coverage. Hence, combining Satellite and terrestrial wireless network fulfils the requirement in terms of end-to-end interoperability and QoS subject to some conditions. Finally, based on the findings of the investigation, a set of recommendations were derived and proposed for hybrid wireless networks. This fulfils the fifth and final objective of this research.

7.3 Main Findings

This research investigated the end-to-end interoperability issues, and QoS requirements of wireless communication technologies in hybrid networks, including Satellite (DVB-RCS and SCPC) and Terrestrial (Wi-Fi and WiMAX) Wireless Networks. A wide range of tests was performed to investigate the interoperability issues and QoS requirements for Hybrid wireless networks. The following are the main findings;

- *The RTT for DVB-RCS was calculated between 640-690 ms (mili second), and RTT for SCPC was around 575 ms. The latency problems were also noted in few results but were acceptable level of around 100 ms for TCP and UDP traffic. The latency of around 250-300 ms for TCP traffic across the satellite channel was found in some scenario which caused further delay.*
- *TCP perceives this RTT (delay), as an awful link performance and automatically set the window size for data packets small and move up the number of requests for acknowledgement of successful transmission. The performance result was degraded in two ways. First, throughput was reduced, which slows the communication. Second, communication session establishment using TCP was lethargic. In a worst-case scenario, with multiple users active simultaneously, the performance of a typical Internet speed same as that of a 56 Kbit/s modem.*

- *The TCP traffic was found more delay constrained as compared to UDP. The UDP has minimal transmission delay due to less connection setup process, flow control method and retransmission, but it is unreliable. TCP was identified reliable, but responsible for queuing behaviour. Especially this was a problem for multimedia and live traffic which software functions well in terrestrial networks, including WiMAX and WLANs, but its performance over the satellite link is limited. This is because the Transmission Control Protocol (TCP), primarily was designed to work with the delays on terrestrial networks. This Protocol was not designed for wireless particularly satellite networks. Therefore, its performance is poor with satellite communication, due to this reason delays on satellite system are greater.*
- *UDP traffic was found more appropriate due to connectionless properties and small header bytes as compared to TCP. As TCP protocol requires handshaking for end-to-end communications.*
- *It was found during testing that, in the case of one site connecting to the satellite hub, the use of DVB-RCS alone or interconnected with Wi-Fi/WiMAX equipment makes no significant difference, regardless of the bandwidth allocated 512 kbps and 1024 Kbps. The audio and video multimedia quality for the user was exceptionally good even with typical satellite relay time delay of 600 ms. However, in a case of two or more sites communicating simultaneously by using 512 kbps, there was a bandwidth issue on the satellite link side and time delay was also increased. The 512 kbps for forward and return link proved inadequate for two sites. While allocating 1024 kbps for three sites, each having 300 kbps of stream received a good quality video stream although the overall bandwidth was within 1 Mbps. Therefore, proper bandwidth is required to be allocated according to the number of sites that can be served in the specific network infrastructure. However, there was no link layer issue found in DVB-RCS, WiMAX and Wi-Fi.*

- *Netmeeting was found a bit better, but this application was also not considered to be a long-term solution due to insufficient support by Microsoft on the Vista and Window 7 operating systems.*
- *The Isabel software platform was found more suitable for hybrid wireless networks, due to the use of standard TCP-UDP/IP protocols. This was found to be particularly suitable, due to the fact that the link bandwidth can be adjusted (by the session organizer) within the range from 128 kbps to 10 Mbps for superior audio and video quality.*
- *The WiMAX RTT was found 20-40 ms, and on average it was 30 ms. For Wi-Fi, RTT was found 1-2 ms. There was a small proportion of packet in Wi-Fi which delay was between 3ms and 4 ms. This was most likely caused by the retransmissions or hardware problem of the Wireless LAN. The short term measurement with load was completed and found that the packet loss was very little, which may be caused by power saving or timing algorithms. The wireless LAN Wi-Fi link the delay was higher than in the long-term measurement. Again, this was due to retransmits done by the hardware. There was little amount of packet loss but it was almost negligible.*
- *During interoperability test analysis it was found that there was a relationship between end-to-end delay and packet loss. The end-to-end delay was depending on transmission delay and link propagation delay. The processing and queuing delay was also affecting end-to-end delay. The greater end-to-end delay, the more packets were lost; as a result it consumes more bandwidth. In case bandwidth was less, the delay, jitter and packet loss were more due to less capacity. These values were taken for mean, minimum and maximum which are more beneficial for design and operation.*
- *The WiMAX used was designed from 70 Mbps upload channel. The WiMAX falls back to 23 Mbps. The WiMAX characteristics both delay and jitter behaves similar to unloaded link. Both BE and rtPS classes did not show any packet loss. Under this circumstances it was insubstantial if the BE or the rtPS characteristic was to be chosen. For WiMAX rtPS*

characteristic the variance of the delay was similar to the BE characteristic. By comparing the WiMAX BE and rtPS both characteristics, it was found that the rtPS class has a significant lower jitter and delay distribution.

- *When transmitting multicast data, a Wi-Fi system falls back to its lowest basic data rate, which yields for Wi-Fi standard 802.11a/g, 6 Mbps theoretical and about 3 Mbps actual throughput, as specified in IEEE 802.11a/b/g.*
- *It was found that there was no special QoS setting required to be configured on WiMAX or WLAN. The Integrated System (DVB-RCS and WiMAX/Wi-Fi) was available 98.5%, and the QoS are satisfied.*
- *The DVB-RCS used during the testing supports 45 Mbps downstream, and up to 2 Mbps upstream for 4 users and up to 1 Mbps upstream and 8 users simultaneously. However, the maximum bandwidth available during testing was 2 Mbps including overheads. The DVB-RCS system supports SITs at different carrier user bit rates and code types: 144 kbps, 384 kbps, 1.024 Mbps and 2.048 Mbps for Concatenated Coding and 256 kbps, 512 kbps, 1.024 Mbps and 2.048 Mbps for Turbo coding. The total throughput capacity assigned to a SIT is the sum of CRA, RBDC, VBDC and FCA and must be in multiples of 16 Kbps. This total capacity should not be larger than the carrier user bit rate of the SIT. The maximum bit rate at which a SIT is capable of transmitting depends on physical characteristics of the SIT such as the size of the satellite dish and power of the SIT transmitter. For the DVB-RCS, each 128 Kbps was allocated to the return link and 1 Mbps to the forward link for each SIT. The return links have a contention ratio of 1:1, i.e. the capacity allocation scheme was 128 Kbps CRA, which ensures low latency and jitter.*

7.4 Recommendations

A signal over a satellite travels up and down a path of some 70,000 to 90,000 km, which depends upon the geographical locations of Ground segments (Satellite Gateway (Hub) and Satellite Interactive Terminals (SITs)). The distance is around the same as twice perimeter of the Earth at Equator. This is the main reason of the difficulty. The communication signal travelled at the speed of light, takes 240 to 300 mili seconds (ms) from the SIT via the satellite. An acknowledgement from the (Hub) to the SIT must retrace the path. So the total delay, a packet is sent until the Satellite Interactive Terminal (SIT)/ Hub gets acknowledgement of its successful reception, is 500 to 600 ms. The latency of around 250-300 ms across the satellite channel was found in some scenario which caused additional delay as discussed in the above section.

To overcome the delay issue, the following recommendations are proposed:

- *It is recommended to add small buffer buffering delay at the receiving side of the satellite link, which should enough to absorb delay by employing some form of buffering approach, similar to which were used in (Cottle and Ehrenberg 2010; Jain and Raina 2011).*
- *It is suggested to configure all end user entities (hosts, servers, terminals, PCs) for the longer latency of the satellite path. This recommendation requires modification of all equipments including router that may never communicate via satellite which looks difficult.*
- *Another suggestion is to manage the satellite gateway and the Terminals it serves to overcome the longer latency (PEP – Performance Enhancing Proxy). As satellite has inherent delay issues, this can also control by looking changes in TCP and UDP protocols for long delays.*
- *There are some built in delay compensator has been designed which can be useful to avoid satellite delay (Telesat 2011).*
- *The design of protocol accelerator is recommended which can arrange handshake for TCP locally at terrestrial leg. The Automatic Retransmission Request (ARQ) technique is also in developing stage which can reduce the authentication time by sending data in block of*

windows with one acknowledgement request for authentication instead of separate for each packet.

- *Satellite delay can also be reduced by adjusting the clock at both transmitter and receiver end and recalculating the whole packets instead of forwarding the TCP and IP header. In short the delay can be reduced by reducing queuing delay, increasing bandwidth, and forwarding the traffic as fast as possible.*
- *Bandwidth guaranteed mechanism is to be devised for acknowledgements (ACKs) particularly for real-time video transmission over hybrid wireless communication network. These types of protocol are required to be designed which can help for TCP Traffic to reduce the packet loss and wireless link error.*
- *QoS packets can be prioritized by a number of criteria generated by applications themselves or system by making provisioning of more capacity that may be sufficient for the expected peak traffic load. Streaming video and voice communication uses the RTP. The IP, UDP, and RTP packet headers are to be compressed. This saves a significant amount of bandwidth in the case of slow links and when a large number of multimedia streams are sent.*
- *Flow control is another option which becomes essential in case the application which is receiving the data is reading it more slowly than the sending application.*
- *QoS routing is a special type of routing in which different connections are assigned different priorities. For video conferencing fast delivery of packets are required to ensure images and voice appears continuous and smooth as delay can seriously affect QoS. The resource reservation protocol (RTRP) and the real time streaming protocol (RTSP) are used to create a connection and request certain guaranty minimum data rate. The QoS in the alternate routing paths if available can reduce the system blocking and make better use of network resources.*
- *The latency problem can be improved by fast handover, authentication process and transmission/congestion losses.*

- *Forward Error Correction (FEC) methods can be used to reduce the latency. Packet loss and improve reliability.*
- *For Satellite traffic another option for delay control is to design and add delay Stamp on each packet at both sending and receiving ends.*
- *The delay and packet loss can also be controlled using various methods such as traffic classification, traffic prioritization, congestion control and bandwidth allocation or similar approach proposed in studies (Chan et al. 2010; Guerrero and Labrador 2010; Libnik et al. 2010).*
- *As there can be variance in bandwidth requirement such as peak, average and low, therefore real time traffic can be given guaranteed over normal traffic.*
- *High quality communication can be provided over a Best Effort services as tested for WiMAX in our test bed. In this way complex QoS mechanism can be avoided. For link layer interoperability DVB-RCS return link is required to be available.*

7.5 Research Contributions

Most of the previous research studies in hybrid wireless technologies were focused on physical layer characteristics and its implication on individual technology interoperability issues and QoS requirement. This thesis has investigated end-to-end interoperability issues and QoS requirements in Hybrid Wireless Networks. The novelty of this research is the use of several live scenarios instead of using simulation tools. The contributions of this thesis are theory, methodology and implementation.

7.5.1 Theory

This thesis discusses and compares various interoperability issues such as bandwidth, delay, jitter, latency, packet loss, throughput and availability testing. The standards, protocol suite/ models and architectures for Wi-Fi, WiMAX, DVB-RCS, SCPC, platforms and applications are discussed and compared. The comprehensive description and comparison of various common test tools were carried out. All these software based test tools are available as open source, flexible, easy to use and are economical. This comparison list can be used as a guideline for selection and comparison. These test tools can be used to test all possible interoperability issues and assess QoS requirements in any hybrid network.

7.5.2 Methodology

A detailed testing methodology was devised for delay, jitter, packet loss, bandwidth, throughput, availability, performance of audio and video session, multicast and unicast performance, TCP and UDP performance and stress test.
The work on this methodology is more practical than theoretical nature and could be directly transferred to practice. This testing methodology is suitable for testing all the common interoperability issues and QoS requirements in Hybrid networks using common test tools available to test end-to-end interoperability issues and QoS requirements.

This test methodology covers test process, test plan outlines and generic test plans for all possible test scenarios in hybrid networks such as DVB-RCS (Hub) to Satellite Interactive Terminals (SIT), SIT to DVB-RCS (Hub), DVB-RCS (Hub) to WiMAX/Wi-Fi, WiMAX /Wi-Fi to DVB-RCS (Hub), Wi-Fi/WiMAX to WiMAX, SCPC to SIT.

7.5.3 Implementation

The results are compiled, compared, and analysed with detail using various test tools to make significant contribution in terms of study, test plan, testing methodology and implementation. This methodology can be used for design aid. The findings and recommendations can be used to improve network design and its implementation.

These research contributions will also have an impact on the deployment of end-to-end Heterogeneous Satellite Terrestrial Broadband Wireless Communication technologies efficiently to remote area providing better QoS.

7.6 Research Limitations

This research covers all end-to-end interoperability issues and QoS requirements for Hybrid Wireless Networks i.e. satellite (DVB-RCS, SCPC) and terrestrial (WiMAX, Wi-Fi). Numerous tests with various scenarios were conducted, and results were analysed for end-to-end interoperability issues, and QoS requirements. However, this research has the following limitations.

The testing of mobile wireless technology, such as 3G and beyond could not done due to unavailability of these technologies during test/trail.

The Satellite DVB-S2 technology which is a successor of DVB-S was not tested also due to unavailability of this technology during test/trial.

The performance and failure of network element of each technology such as CPU, RAM, modulator, demodulator, encoder, decoder, and router failure were also not carried out.

The testing and assessment of other external factors on interoperability issues and Quality of Service requirements such as weather condition, signal to noise ratio, interference, attenuation, and Quality of Perception were also not covered in this research, since they were outside its scope and its particular aim.

7.7 Future Research

In this thesis end-to-end Interoperability issues and QoS requirements for Hybrid Wireless Networks i.e. Satellite (DVB-RCS, SCPC) and Terrestrial (Wi-Fi, WiMAX) are studied, discussed, tested and analysed. Numerous tests with diverse scenarios were conducted and results were analysed for end-to-end interoperability issues and QoS requirements. 3G technology was beyond the scope of this research and hence it could not be tested.

Research on current issues between Satellite and Terrestrial wired networks which may also include 3G and Optical Fibre is another topic for research.

Further interoperability issues and QoS requirements testing can be done at the Physical and MAC Layers for each technology for evaluation and comparison to distinguish issues at physical, MAC and link layers.

The performance of CPU, RAM, modulator, demodulator, encoder, decoder, switches and routers can also be considered for future research which can cover issues in wireless links and congestion in order to discriminate the causes. A strategy for such testing will need to be devised.

There is an imperative relation between network performance, availability and network security. Wireless security is an enormous, vital and challenging topic for research in modern wireless communication technologies. Moreover, wireless networks have substantial security issues which can be covered for future research. There is some progress on security in homogeneous networks, but hybrid-heterogeneous, wireless networks require different security architectures due to different security architectures used within each network, so this interoperability study could lay a sound basis for future research in this area of research.

Appendix A: Inclusion and exclusion

Inclusion	***Exclusion***	***Keywords list, and data selected***
IEEE/IET Electronic Library	3G and LTE	Interoperability
ACM Digital Library	4G	Wireless communications
Compendex/INSPEC	weather satellite	Hybrid Networks
Web of Science	Before year 2000 except few	Heterogeneous Networks
Web of Knowledge	GPS	Satellite Broadband
Science Direct	Blue tooth	Satellite Networks
Computer Abstract International		VSAT
Scopus		DVB-RCS
SPIE Digital Library		DVB-S/S2
SpringerLink		WiMAX
Wiley Interscience		Wi-Fi
WiMAX Forum		SCPC
Wi-Fi Forum		QoS
IPV6 forum		Bandwidth
Wireless world research forum		Delay
Black well Synergy		Jitter
International Journal of wireless information networks (Springer)		Latency
Wireless personal communications (Springer)		Broadband Internet
Wireless Communications and Mobile computing (Wiley)		TDMA
International Journal of Satellite Communications & networking (Wiley)		OFDMA
International Journal of Satellite Communications		Packet Loss
The international journal of computer & telecommunication networking		Wireless Standards
International Journal of wireless & mobile communications		Throughput
International Journal of Communication Networks and Distributed Systems (Inderscience)		End-to-end delay
IEEE Wireless Communication		SPSC(Single Channel Per Carrier)
IEEE Communications Magazine		MMDS
IEEE Personal Communications		LMDS
IEEE Journal on selected areas in		MPEG

communications		
IEEE Transactions on Wireless Communications		QPSK modulation
IEEE/ACM Transaction on Mobile computing		Cross layer protocols
IEEE Transactions on computing		Adaptive modulation & coding
IEEE Internet computing		MIMO
ACM Transaction on computer system		SCOFDMA
ACM Computer communication		Isabel Platform
Wireless communication & Mobile Computing		Clix Platform
Computer networks		Reed Solomon
Computer communication		Viterbi
Cambridge journals online		Convolutional Coding
Oxford Journal		

Appendix B: Preliminary Test Results

Appendix C: DVB-RCS Unicast and Multicast Testing

Appendix D: Satellite Throughput Measurements

Appendix E: DVB-RCS Wireshark Conversation

Appendix F: DVB-RCS Traffic Analysis

Appendix G: SCPC Ship Traffic Analysis

Appendix H: SCPC Ship Wireshark Conversations

References

Abbasi, M., Stergioulas, L. K., El-Haddadeh, R., Kretschmer, M., Pitsilis, V., Tsiolis, I., Zagkos, D. and Hatziefremidis, A. (2008)."Interoperability and testing in broadband satellite networks." Signal Processing for Space Communications, 2008. SPSC 2008. 10th International Workshop on.

Abbasi, M., Stergioulas, L. K., Kretschmer, M., Pitsilis, V., Khalid, Z. and Khan, N. (2008)."Heterogeneous satellite-terrestrial technologies: Quality of service and availability testing." Emerging Technologies, 2008. ICET 2008. 4th International Conference on.

Abdalla, K. F. (2003)."A model for semantic interoperability using XML." Systems and Information Engineering Design Symposium, 2003 IEEE.

Abuelma'atti, O., Merabti, M. and Askwith, B. (2006)."A wireless networked appliances interoperability architecture." Wireless Pervasive Computing, 2006 1st International Symposium on.

Activeexpert. (2010). "ActiveXperts PING backgrounds (PING is part of the ActiveSocket Toolkit)." Retrieved July 2010, from http://www.activexperts.com/activsocket/toolkits/ping/.

Adibi, A., Bin, L., Pin-Han, H., Agnew, G. B. and Erfani, S. (2006)."Authentication Authorization and Accounting (AAA) Schemes in WiMAX." Electro/information Technology, 2006 IEEE International Conference on.

Agilent. (2007). "Jitter measurement." white paper. Retrieved July, 2007, from http://www.agilent.com.

Agora Systems , S. A. (2005, November 2005). " Introduction to Isabel 4.9 documentation overview." Accessed on http://www.agora-2000.com/ Retrieved Septemeber, 2008, from http://www.agora-2000.com/.

Ahmadi, S. (2011). "The IEEE 802.16m Physical Layer (Part I)." Mobile WiMAX. Oxford, Academic Press**:** 335-487.

Akyildiz, I. F., Gutierrez-Estevez, D. M. and Reyes, E. C. (2010). "The evolution to 4G cellular systems: LTE-Advanced." Physical Communication **In Press, Corrected Proof**.

Akyildiz, I. F., Gutierrez-Estevez, D. M. and Reyes, E. C. (2010). "The evolution to 4G cellular systems: LTE-Advanced." Physical Communication **3**(4): 217-244.

Al-Gizawi, T., Peppas, K., Axiotis, D. I., Protonotarios, E. N. and Lazarakis, F. (2005). "Interoperability criteria, mechanisms, and evaluation of system performance for transparently interoperating WLAN and UMTS-HSDPA networks." Network, IEEE **19**(4): 66-72.

Al-Khusaibi, H., Al-Wardi, F., Firas, S. and Weidong, X. (2006)."Experiment and Analysis on the Comparison of the IEEE 802.11a and 802.11g Wireless Local Area Networks." Electro/information Technology, 2006 IEEE International Conference on.

Al-Wakeel, S. S. and Al-Wakeel, M. M. (2000)."An architecture design of a VSAT satellite network for multimedia on demand services." Wireless Communications and Networking Conference, 2000. WCNC. 2000 IEEE.

Alagoz, F. (2004). "Integrating the scene length characteristics of MPEG video bitstreams into a direct broadcast satellite network with return channel

system." International Journal of Satellite Communications and Networking **22**(2): 217-229.
Alam, M. T. and Wu, Z. d. (2007). "End-to-end delay measurement for instant messaging relay nodes." Ubiquitous Computing and Communication Journal **2**(2).
Alavi, H. S., Mojdeh, M. and Yazdani, N. (2005)."A Quality of Service Architecture for IEEE 802.16 Standards." Communications, 2005 Asia-Pacific Conference on.
Almuhtadi, W. (2005). Rural/Remote WiFi Wireless Broadband System." Proceedings of the Second Annual Conference on Wireless On-demand Network Systems and Services, IEEE Computer Society**:** 180-188.
Alturki, R., Nwizege, K., Mehmood, R. and Faisal, M. (2009)."End to End Wireless Multimedia Service Modelling over a Metropolitan Area Network." Computer Modelling and Simulation, 2009. UKSIM '09. 11th International Conference on.
Andersen, B. R., Gangaas, O. and Andenæs, J. (2006). "A DVB/Inmarsat hybrid architecture for asymmetrical broadband mobile satellite services." International Journal of Satellite Communications and Networking **24**(2): 119-136.
Ashtaiwi, A. and Hassanein, H. (2010). "MIMO-Based Collision Avoidance in IEEE 802.11e Networks." Vehicular Technology, IEEE Transactions on **59**(3): 1076-1086.
Athanasopoulos, G., Tsalgatidou, A. and Pantazoglou, M. (2006)."Interoperability among Heterogeneous Services." Services Computing, 2006. SCC '06. IEEE International Conference on.
Awal, M. A., Kanchanasut, K. and Tsuchimoto, Y. (2005). "Multicast Packet Loss Measurement and analysis over Unidirectional satellite Network." k.Cho and P Jacquet(Eds.) AINTEC 2005 **LNCS 3837**(Journal Article): 254-268.
Bahr, M. (2006). Proposed routing for IEEE 802.11s WLAN mesh networks." Proceedings of the 2nd annual international workshop on Wireless internet. Boston, Massachusetts, ACM**:** 5.
Baset, S. A. and Schulzrinne, H. G. (2006)."An Analysis of the Skype Peer-to-Peer Internet Telephony Protocol." INFOCOM 2006. 25th IEEE International Conference on Computer Communications. Proceedings.
Basicevic, I., Popovic, M. and Kukolj, D. (2008)."Comparison of SIP and H.323 Protocols." The Third international conference on Digital Telecommunications, 2008. ICDT '08

Beacon. (2008). "Multicast beacon." Retrieved May, 2008, from http://www.beacon.ncsa.uiuc.edu.
Bekkers, R., Verspagen, B. and Smits, J. (2002). "Intellectual property rights and standardization: the case of GSM." Telecommunications Policy **26**(3-4): 171-188.
Bennett, B., Holt, C., Skowrunski, M., Summers, E. and Hamilton, B. A. (2005)."DVB-S2 benefits for military broadcast systems." Military Communications Conference, 2005. MILCOM 2005. IEEE.
Bennett, B., Quock, K., Skowrunski, M., Difrancisco, M. and Hamilton, B. A. (2005). Digital video broadcast return channel satellite (DVB-RCS)

architectures and applications for the department of defence (DoD) " Military Communications Conference , MILCOM 2005 IEEE. **1:** 113-119.

Bhagwat, P. (2001). "Bluetooth: technology for short-range wireless apps." Internet Computing, IEEE **5**(3): 96-103.

Bi., J., Wu., Q. and Li., Z. (2002). Packet Delay and Packet Loss in the Internet." Proceedings of the Seventh International Symposium on Computers and Communications (ISCC'02), IEEE Computer Society**:** 3.

Blakowski, G. and Steinmetz, R. (1996). "A media synchronization survey: reference model, specification, and case studies." Selected Areas in Communications, IEEE Journal on **14**(1): 5-35.

Bluetoothspefication. (2001). "Bluetooth Specification Version 1.1, The Bluetooth Special Interest Group (SIG)." Retrieved May, 2008, from http://www.bluetooth.org.

Bohl, O., Scheuhase, J., Sengler, R. and Winand, U. (2002)."The sharable content object reference model (SCORM) - a critical review." Computers in Education, 2002. Proceedings. International Conference on.

Breitgand, D., Raz, D. and Shavitt, Y. (2002). "SNMP GetPrev: an efficient way to browse large MIB tables." Selected Areas in Communications, IEEE Journal on **20**(4): 656-667.

Calvagna, A., Morabito, G. and La Corte, A. (2003)."WiFi bridge: wireless mobility framework supporting session continuity." Pervasive Computing and Communications, 2003. (PerCom 2003). Proceedings of the First IEEE International Conference on.

Calvagna, A., Morabito, G. and Pappalardo, A. (2003)."WiFi mobility framework supporting GPRS roaming: design and implementation." Communications, 2003. ICC '03. IEEE International Conference on.

Cardarilli, G. C., Del Re, A., Re, M. and Simone, L. (2006)."Optimized QPSK modulator for DVB-S applications." Circuits and Systems, 2006. ISCAS 2006. Proceedings. 2006 IEEE International Symposium on.

Celebi, B., Dericiogullari, B. and Bitirim, Y. (2007)."Performance Evaluation of IEEE 802.11b, IEEE 802.11g and GPRS/EDGE Based on Query Retrieval Time." Wireless and Mobile Communications, 2007. ICWMC '07. Third International Conference on.

Centonza, A. and McCann, S. (2006)."Architectural and Protocol Structure for Composite DVB- RCS/IEEE 802.16 Platforms." Digital Video Broadcasting over Satellite: Present and Future, 2006. The Institution of Engineering and Technology Seminar on.

Chan, Y.-C., Lin, C.-L., Chan, C.-T. and Ho, C.-Y. (2010). "CODE TCP: A competitive delay-based TCP." Computer Communications **33**(9): 1013-1029.

Chatterjee, M., Sengupta, S. and Ganguly, S. (2007). "Feedback-based real-time streaming over WiMax " Wireless Communications, IEEE **14**(1): 64-71.

Chen, J.-L., Chen, W.-H. and Kuo, S.-Y. (2005). "All-IPv6 service interworking gateway." Int. J. Netw. Manag. **15**(2): 135-147.

Cheng, Z., Keqing, H. and Bing, L. (2009)."A Framework to Support Interoperability among Web Service Registries." Computer and Information Science, 2009. ICIS 2009. Eighth IEEE/ACIS International Conference on.

Chew, L. K. (2008). "Scorn 3rd edition specification and intelligent tutoring system – The difference and similarities." Syenthesis Journal 2008: 79-86.
Chiew, T. K., Ferre, P., Agrafiotis, D., Molina, A., Nix, A. R. and Bull, D. R. (2005)."Cross-layer WLAN measurement and link analysis for low latency error resilient wireless video transmission." Consumer Electronics, 2005. ICCE. 2005 Digest of Technical Papers. International Conference on.
Chin, J. and Cantoni, A. (1998). "Phase jitter≡timing jitter?" IEEE Communication letters **2**(2): 54-56.
Chini, P., Giambene, G., Bartolini, D., Luglio, M. and Roseti, C. (2006). "Dynamic resource allocation based on a TCP-MAC cross-layer approach for DVB-RCS satellite networks." International Journal of Satellite Communications and Networking **24**(5): 367-385.
Chitre, D. M. and Henderson, T. (1995)."Seamless integration of satellite and terrestrial networks." Digital Satellite Communications, 1995., Tenth International Conference on.
Cisco. (2005). "Capacity and performance management:Best practices." Cisco documents ID 20769, White paper Retrieved May 2008, ID 20769.
Cisco. (2008). "Network QoS using Cisco HOW TO." Retrieved June, 2008, from http://opalsoft.net/qos.
Cisco. (2009). "Network availability: How much do you need ?how do you get it?" Retrieved May 2009, from http://www.cisco.com.
Cisco Systems, I. (2000). "Internet Protocol Multicast." Retrieved July 2008, from http://www.pluscom.ru/cisco_product/cc/td/doc/cisintwk/ito_doc/ipmulti.htm.
Claffy, K. C. and Dovrolis, C. (2008, May 2008). "Bandwidth Estimation:measurement methodologies and applications." Retrieved June, 2008, from http://www.caida.org/projects/bwest/.
comstream, R. (2005). "DVB-S2 radyne comstream DM240." web accessed WP017, Rev 1.3, January 2005. Retrieved 17 April 2008.
Corvaja, R. (2006). "QoS Analysis in Overlay Bluetooth-WiFi Networks with Profile-Based Vertical Handover." Mobile Computing, IEEE Transactions on **5**(12): 1679-1690.
Costabile, M., Follino, C., lera, A. and Molinaro, A. (2004)."QoS differentiation in DVB-RCS multimedia platforms." Personal, Indoor and Mobile Radio Communications, 2004. PIMRC 2004. 15th IEEE International Symposium on.
Cottle, W. B. and Ehrenberg, R. G. (2010)."Strategies for integrating very high latency modems into a deployed MILSATCOM network while assuring high levels of reliability." MILITARY COMMUNICATIONS CONFERENCE, 2010 - MILCOM 2010.
Crescenzo, M. D., Guainella, E. and Sansone, C. (2008). "Multicast aware QoS in Next generation networks." Retrieved May 2008.
Cullen, C. J., Tafazolli, R. and Evans, B. G. (1992)."Satellite/cellular interoperability for Europe." Land Mobile Satellite Systems, IEE Colloquium on.

Cunningham, G., Perry, P., Murphy, J. and Murphy, L. (2009). "Seamless Handover of IPTV Streams in a Wireless LAN Network." Broadcasting, IEEE Transactions on **55**(4): 796-801.

Dao, N. T. and Malaney, R. A. (2007)."Throughput Performance of Saturated 802.11g Networks." Wireless Broadband and Ultra Wideband Communications, 2007. AusWireless 2007. The 2nd International Conference on.

de la Cuesta, B., Yun, A. and Solano, A. (2009)."DVB-RCS systems in the NGN convergence framework." Satellite and Space Communications, 2009. IWSSC 2009. International Workshop on.

De Miguel, T., Pavon, S., Salvachua, J., Vives, J., Alonso, P., Fernandez-Amigo, J., Acuna, C., Yamamoto, L., Lagarto, V. and Vastos, J. (1994). "ISABEL Experimental distributed cooperative work application over broadband networks." Multimedia: Advanced Teleservices and High-Speed Communication Architectures. Ralf Steinmetz, Springer Berlin / Heidelberg. **868:** 353-362.

Dekleva, S., Shim, J., Varshney, U. and and Knoerzer, G. (2007). "Evolution and emerging issues in mobile wireless networks." Communication of the ACM **50**(6): 38-43.

Der-Jiunn, D. and Hsu-Chun, Y. (2005). "Quality-of-service provisioning system for multimedia transmission in IEEE 802.11 wireless LANs." Selected Areas in Communications, IEEE Journal on **23**(6): 1240-1252.

Dibuz, S. and Kremer, P. (2006)."An easy way to test interoperability and conformance." Testbeds and Research Infrastructures for the Development of Networks and Communities, 2006. TRIDENTCOM 2006. 2nd International Conference on.

Dovrolis, C., Ramanathan, P. and Moore, D. (2004). "Packet dispersion techniques and a capacity estimation methodology." IEEE/ACM Transactions on Networking **12**(6): 963-977.

Eklund, C., Marks, R. B., Stanwood, K. L. and Wang, S. (2002). "IEEE standard 802.16: a technical overview of the WirelessMAN air interface for broadband wireless access." Communications Magazine, IEEE **40**(6): 98-107.

ETSI. (2009). "Digital Video Broadcasting (DVB), “Interaction channel for satellite distribution systems”, ETSI EN 301 790 V1.5.1 (2005-09)." DVB BLUE BOOK a130. Retrieved May, 2010.

Evans, B., Werner, M., Lutz, E., Bousquet, M., Corazza, G. E., Maral, G. and Rumeau, R. (2005). "Integration of satellite and terrestrial systems in future multimedia communications." Wireless Communications, IEEE **12**(5): 72-80.

Fabrice Arnal, Ana Bolea-Alamanac, Michel Bousquet, Laurent Castanet, Laurent Claverotte, Laurent Dairaine, Ricardo Gutierrez-Galvan and Gerard Maral. (2008). "Reliable multicast transport protocols performances in emulated satellite environnement taking into account an adaptive physical layer for the geocast system."

Fen, H., She, J., Pin-Han, H. and Xuemin, S. (2009). "A flexible resource allocation and scheduling framework for non-real-time polling service in

IEEE 802.16 networks." Wireless Communications, IEEE Transactions on **8**(2): 766-775.

Ferro, E. and Potorti, F. (2005). "Bluetooth and Wi-Fi wireless protocols: a survey and a comparison." Wireless Communications, IEEE **12**(1): 12-26.

Fong, B., Ansari, N., Fong, A. C. M. and Hong, G. Y. (2004). "On the scalability of fixed broadband wireless access network deployment." Communications Magazine, IEEE **42**(9): S12-S18.

Fourty, N., Val, T., Fraisse, P. and Mercier, J. J. (2005)."Comparative analysis of new high data rate wireless communication technologies "From Wi-Fi to WiMAX"." Autonomic and Autonomous Systems and International Conference on Networking and Services, 2005. ICAS-ICNS 2005. Joint International Conference on.

Ghosh, A., Wolter, D. R., Andrews, J. G. and Chen, R. (2005). "Broadband wireless access with WiMax/802.16: current performance benchmarks and future potential." Communications Magazine, IEEE **43**(2): 129-136.

Ghribi, B. and Logrippo, L. (2000). "Understanding GPRS: the GSM packet radio service." Computer Networks **34**(5): 763-779.

Giambene, G. and Kota, S. (2006). "Cross-layer protocol optimization for satellite communications networks: a survey " International Journal of Satellite Communications and Networking **24**(5): 323-341.

Goodman, D. J. and Myers, R. A. (2005)."3G cellular standards and patents." Wireless Networks, Communications and Mobile Computing, 2005 International Conference on.

Gross, J., Emmelmann, M., Puñal, O. and Wolisz, A. (2009). "Enhancing IEEE 802.11a/n with dynamic single-user OFDM adaptation." Performance Evaluation **66**(3-5): 240-257.

Guerrero, C. D. and Labrador, M. A. (2010). "On the applicability of available bandwidth estimation techniques and tools." Computer Communications **33**(1): 11-22.

Guvenc, I., Kozat, U. C., Moo-Ryong, J., Watanabe, F. and Chia-Chin, C. (2008). "Reliable multicast and broadcast services in relay-based emergency communications." Wireless Communications, IEEE **15**(3): 40-47.

Hao, Y., Chuang, L., Jin-Jun, Z., Bo, L. and Qiang, N. (2005). "Effective Video Multicast over Wireless Internet: Rate Allocation and End-System Based Adaptation." IEICE Trans Commun (Inst Electron Inf Commun Eng) **E88-B**(NO.4): 1395-1402.

Hassan, S., Linfoot, S. L. and Al-Akaidi, M. M. (2006). "An analysis on the performance of the forward and return interaction channels in DVB-RCS." Consumer Electronics, IEEE Transactions on **52**(2): 371-376.

Haug, T. (2002). "A commentary on standardization practices: lessons from the NMT and GSM mobile telephone standards histories." Telecommunications Policy **26**(3-4): 101-107.

Henry, P. S. and Hui, L. (2002). "WiFi: what's next?" Communications Magazine, IEEE **40**(12): 66-72.

Heusse, M., Rousseau, F., Berger-Sabbatel, G. and Duda, A. (2003)."Performance anomaly of 802.11b." INFOCOM 2003. Twenty-Second Annual Joint Conference of the IEEE Computer and Communications. IEEE Societies.

Hoene, C., Gunther, A. and Wolisz, A. (2003)."Measuring the impact of slow user motion on packet loss and delay over IEEE 802.11b wireless links." Local Computer Networks, 2003. LCN '03. Proceedings. 28th Annual IEEE International Conference on.

Hong, S.-S., Wong, F., Wu, S. F., Lilja, B., Yohansson, T. Y., Johnson, H. and Nelsson, A. (2005). "TCPtransform: Property-Oriented TCP Traffic Transformation." Intrusion and Malware Detection and Vulnerability Assessment. Klaus Julisch and Christopher Kruegel, Springer Berlin / Heidelberg. **3548:** 222-240.

Howes, N. R. and Weaver, A. C. (1989). "Measurements of Ada overhead in OSI-style communications systems." Software Engineering, IEEE Transactions on **15**(12): 1507-1517.

Hoymann, C. (2005). "Analysis and performance evaluation of the OFDM-based metropolitan area network IEEE 802.16." Comput. Netw. **49**(3): 341-363.

Hoymann, C., Dallas, P., Valkanas, A., Gosteau, J., Noguet, D. and Hoshyar, R. (2006). Flexible Relay Wireless OFDM-based Networks**:** 5.

httperf. (2008). "Httperf test tool " Retrieved 20 Septemeber, 2008, from http://www.hpl.hp.com/research/linux/httperf/.

Huang, C., Juan, H.-H., Lin, M.-S. and Chang, C.-J. (2007). "Radio resource management of heterogeneous services in mobile WiMAX systems [Radio Resource Management and Protocol Engineering for IEEE 802.16]." Wireless Communications, IEEE **14**(1): 20-26.

Hung-Yu, W., Ganguly, S., Izmailov, R. and Haas, Z. J. (2005)."Interference-aware IEEE 802.16 WiMax mesh networks." Vehicular Technology Conference, 2005. VTC 2005-Spring. 2005 IEEE 61st.

IEE802.11e. (2005). "IEEE Std 802.11eT, Part 11: Wireless LAN Medium Access Control(MAC) and Physical Layer (PHY) specifications; Amendment 8: Medium Access Control(MAC) Quality of Service Enhancements." January 2007. Retrieved 17 April 2008.

IEEE802.11 (2009). "IEEE Draft STANDARD for Information Technology--Telecommunications and information exchange between systems--Local and metropolitan area networks--Specific requirements Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications Amendment 7: Interworking with External Networks." IEEE Unapproved Draft Std P802.11u/D5.0, Feb 2009.

IEEE802.11 (2010). "IEEE Draft Amendment Standard for Local and Metropolitan Area Networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems - Advanced Air Interface." IEEE P802.16m/D7 July 2010: 1-932.

IEEE802.11 (2010). "IEEE Draft Standard for Information Technology-Telecommunications and information exchange between systems-Local and metropolitan area networks-Specific requirements-Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications-Amendment 10: Mesh Networking." IEEE Std P802.11s/D5.0 April 2010: 1-305.

IEEE802.11j (1999). "Unapproved Draft Standard for Information Technology-Telecommunications and information exchange between systems- Local and metropolitan area network- Specific requirements Part 11: Wireless

LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications. (This document reflects the combining of the 2003 Edition of 802.11 plus the 802.11g, 802.11h, 802.11i and 802.11j Amendments) (Revision of IEEE Std 802.11-1999) (Superseded by P802.11-REVma_D5.0)." IEEE Std P802.11REV-ma/D4.0.

IEEE802.11n (2010). "IEEE Draft Standard for Information Technology - Telecommunications and Information Exchange Between Systems - Local and Metropolitan Area Networks - Specific Requirements - Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications." IEEE Draft P802.11-REVmb/D3.0, March 2010 (Revision of IEEE Std 802.11-2007, as amended by IEEE Std 802.11k-2008, IEEE Std 802.11r-2008, IEEE Std 802.11y-2008, IEEE Std 802.11w-2009 and IEEE Std 802.11n-2009): 1-2228.

IEEE802.16-2001 (2004). "Unapproved Draft IEEE Standard for Local and metropolitan area networks Corrigendum to IEEE Standard for Local and Metropolitan Area Networks-Part 16: Air Interface for Fixed Broadband Wireless Access Systems (Revision of IEEE Std 802.16-2001; IEEE Std 802.16c-2002, and IEEE std 802.16a-2003)." IEEE Std P802.16/REVd/D5.

IEEE802.16 (2003). "IEEE Standard for Local and Metropolitan Area Networks --- Part 16: Air Interface for Fixed Broadband Wireless Access Systems--- Amendment 2: Medium Access Control Modifications and Additional Physical Layer Specifications for 2-11 GHz." IEEE Std 802.16a-2003 (Amendment to IEEE Std 802.16-2001): 0_1-292.

IEEE802.16 (2006). "IEEE Standard for Local and Metropolitan Area Networks Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems Amendment 2: Physical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed Bands and Corrigendum 1." IEEE Std 802.16e-2005 and IEEE Std 802.16-2004/Cor 1-2005 (Amendment and Corrigendum to IEEE Std 802.16-2004): 0_1-822.

ieee.org. (2009). "IEEE Standard for Local and metropolitan area networks—Part 16: Air Interface for Fixed Broadband Wireless Access Systems." Retrieved May, 2009, from http://standards.ieee.org/getieee802/802.16.html.

Isabel. (2007). "ISABEL Technical Documentation." Retrieved May, 2007, from http://www.agora-2000.com/products/ISABEL/documentation.html.

ITU. (2007). "T.120:Data protocol for multimedia conferencing " Retrieved April, 2007.

ITU. (2009). "H.323 System implementation guide" Implementors' Guide for Recommendations of the H.323 System (Packet-based multimedia communications systems): H.323, H.225.0, H.245, H.246, H.283, H.341, H.450 Series, H.460 Series, and H.500 Series " Retrieved August, 2009.

Iwamura, M., Etemad, K., Mo-Han, F., Nory, R. and Love, R. (2010). "Carrier aggregation framework in 3GPP LTE-advanced [WiMAX/LTE Update]." Communications Magazine, IEEE **48**(8): 60-67.

Jain, M. and Dovrolis, C. (2003). "End-to-end available bandwidth: measurement methodology, dynamics, and relation with TCP throughput." IEEE/ACM Trans. Netw. **11**(4): 537-549.

Jain, N., Das, S. R. and Nasipuri, A. (2001)."A multichannel CSMA MAC protocol with receiver-based channel selection for multihop wireless networks." Computer Communications and Networks, 2001. Proceedings. Tenth International Conference on.

Jain, S. and Raina, G. (2011)."An experimental evaluation of CUBIC TCP in a small buffer regime." Communications (NCC), 2011 National Conference on.

Jeacole, K., Crowcroft, J., Barcellos, M. P. and Pettini, S. (2005). Hybrid Reliable Multicast with TCP-XM " Proceedings of the 2005 ACM conference on Emerging network experiment and technology Toulouse, France**:** 177-187.

Jenson, D. Z. J. (2003)."Interoperability of wireless communication technologies and mobile internet application development tools." OSRA 2003 conference Advancing Technologies, Las Vegas, USA.

Jing, N., Xinli, Z. and Lijun, Z. (2007)."A Semantic Web Service-Oriented Model for E-Commerce." Service Systems and Service Management, 2007 International Conference on.

Jing, X. and Raychaudhuri, D. (2006). "Spectrum co-existence of IEEE 802.11b and 802.16a networks using reactive and proactive etiquette policies." Mob. Netw. Appl. **11**(4): 539-554.

Joung, P. (2003). "General Network Performance testing methodology." SPIRENT Communications Retrieved July 2008, from http://www.spirentcom/enterprise.

Kalavakunta, R. and Kripalani, A. (2005)."Evolution of mobile broadband access technologies and services -considerations and solutions for smooth migration from 2G to 3G networks." Personal Wireless Communications, 2005. ICPWC 2005. 2005 IEEE International Conference on.

Kawalek, J. (1995)."A user perspective for QoS management." Proceedings of 3rd International Conference on Intelligence in Broadband Services and Network (IS & N '95, Crete, Greece).

Kejie, L., Yi, Q. and Hsiao-Hwa, C. (2007). "WIRELESS BROADBAND ACCESS: WIMAX AND BEYOND - A Secure and Service-Oriented Network Control Framework for WiMAX Networks." Communications Magazine, IEEE **45**(5): 124-130.

Kertesz, A. and Kacsuk, P. (2009). "Grid Interoperability Solutions in Grid Resource Management." Systems Journal, IEEE **3**(1): 131-141.

Kim, K.-H. (2006). Key technologies for the next generation wireless communications." Proceedings of the 4th international conference on Hardware/software codesign and system synthesis. Seoul, Korea, ACM**:** 266-269.

Kismet. (2010). "Kismet test tool " Retrieved January 2010, from http://www.kismetwireless.net.

Kraan, W. (2007). "No one standard will suit all, The Centre for Educational Technology Interoperability Standards, Presentation available online." Retrieved July, 2007, from metadata.cetis.ac.uk/content/20030513175232

Kyasanur, P. and Vaidya, N. H. (2006). "Routing and link-layer protocols for multi-channel multi-interface ad hoc wireless networks." SIGMOBILE Mob. Comput. Commun. Rev. **10**(1): 31-43.

Laddomada, M., Mesiti, F., Mondin, M. and Daneshgaran, F. (2010). "On the throughput performance of multirate IEEE 802.11 networks with variable-loaded stations: analysis, modeling, and a novel proportional fairness criterion." Wireless Communications, IEEE Transactions on **9**(5): 1594-1607.

LaRoche, P. and Zincir-Heywood, A. N. (2006)."Genetic programming based WiFi data link layer attack detection." Communication Networks and Services Research Conference, 2006. CNSR 2006. Proceedings of the 4th Annual.

Lee, N.-K., Chae, S.-H., Oh, D.-G. and Lee, H.-J. (2005). "Advanced Performance Enhancing Mechanisms for supporting Real_time Services on DVB-RCS System Enviroments." IEEE TRANSACTIONS **E88-B**(Journal Article): 2777-2783.

Li, Y., Long, K.-P. and Zhao, W.-L. (2006). "Defining and maximizing -a novel performance parameter for IEEE 802.11 DCF: Research Articles." Int. J. Commun. Syst. **19**(9): 977-992.

Liangshan, M. and Dongyan, J. (2005)."The Competition and Cooperation of WiMAX, WLAN and 3G." Mobile Technology, Applications and Systems, 2005 2nd International Conference on.

Libnik, R., Svigelj, A. and Kandus, G. (2010). "A novel SIP based procedure for congestion aware handover in heterogeneous networks." Computer Communications **33**(18): 2176-2184.

Lindemann, C., Lohmann, M. and Thümmler, A. (2002). "Adaptive performance management for universal mobile telecommunications system networks." Computer Networks **38**(4): 477-496.

Liqun, F., Zhigang, C. and Pingyi, F. (2005)."Spatial reuse in IEEE 802.16 based wireless mesh networks." Communications and Information Technology, 2005. ISCIT 2005. IEEE International Symposium on.

Liu, H. and Zarki, M. E. (2006). "An adaptive delay and synchronization control scheme for Wi-Fi based audio/video conferencing." Wirel. Netw. **12**(4): 511-522.

Liu, W., Zehua, G., Feng, G., Di, T. and Ronghua, Z. (2009)."Performance analysis of IEEE 802.11a in non-saturation conditions." Network Infrastructure and Digital Content, 2009. IC-NIDC 2009. IEEE International Conference on.

Loguinov, D. and Radha, H. (2002). "Large-scale experimental study of Internet performance using video traffic." SIGCOMM Comput. Commun. Rev. **32**(1): 7-19.

Lorchat, J. and Noel, T. (2006). "Overcoming the IEEE 802.11 paradox for realtime multimedia traffic." Comput. Commun. **29**(17): 3507-3515.

Lucke, O., Jahn, A. and Werner, M. (2006). "MAC and encapsulation efficiency of satellite DVB using fade mitigation techniques." International Journal of Satellite Communications and Networking **24**(6): 521-559.

Lyytinen, K. and Fomin, V. V. (2002). "Achieving high momentum in the evolution of wireless infrastructures: the battle over the 1G solutions." Telecommunications Policy **26**(3-4): 149-170.

Mahasukhon, P., Sharif, H., Hempel, M., Zhou, T., Wang, W. and Chen, H. H. (2009). "IEEE 802.11b based ad hoc networking and its performance in mobile channels." Communications, IET **3**(5): 689-699.

Masadeh, S. R., Aljawarneh, S., Turab, N. and Abuerrub, A. M. (2010)."A comparison of data encryption algorithms with the proposed algorithm: Wireless security." Networked Computing and Advanced Information Management (NCM), 2010 Sixth International Conference on.

Masmoudi, A. and Tabbane, S. (2009)."Optimized dimensioning methods for HSPA based Beyond 3G mobile networks." Information Infrastructure Symposium, 2009. GIIS '09. Global.

McSparron, N., Cote, M., Lambert, M. and Erup, L. (2006)."Implementation Challenges and Synergistic Benefits of DVB-S2 & DVB-RCS." Digital Video Broadcasting over Satellite: Present and Future, 2006. The Institution of Engineering and Technology Seminar on.

Meawad, F. E. and Stubbs, G. (2006). A Framework for Interoperability with VLEs for Large Scale Deployment of Mobile Learning." Proceedings of the Fourth IEEE International Workshop on Wireless, Mobile and Ubiquitous Technology in Education, IEEE Computer Society**:** 13-17.

Mgen. (2008). "mgen users guide." Version 4.2. Retrieved April, 2008, from http://pf.itd.nrl.navy.mil/mgen/mgen.html.

Microsoft. (2009). "Understanding the H.323 protocol." Retrieved July, 2009, from http://www.microsoft.com/windows/NetMeeting/Corp/reskit/Chapter11/default.asp.

Milijevic, S. and Semicondutor, Z. (2008). "Jitter generation and measurement with off-the-self test equipment." Retrieved May 2008.

Miloucheva, I., D. Hetzer, R. Pascotto and Jonas, K. (2009). "Resource reservation in advance for QoS based mobile applications." International review on computer and software papers. Retrieved Web Page, 2009, from http://www.praiseworthyprize.com/IRECOS.

Ming-Jye, S., Park, K. I. and Mak, T. (2008)."Analysis of adaptive WRED and CBWFQ algorithms on tactical edge." Military Communications Conference, 2008. MILCOM 2008. IEEE.

Mishra, A., Shin, M. and Arbaugh, W. (2003). "An empirical analysis of the IEEE 802.11 MAC layer hand off process." ACM SIGCOMM Computer Communication Review, April **33** (2): 93-102.

Morello, A. and Mignone, V. (2006). "DVB-S2: The Second Generation Standard for Satellite Broad-Band Services." Proceedings of the IEEE **94**(1): 210-227.

Morello, A. and Reimers, U. (2004). "DVB-S2, the second generation standard for satellite broadcasting and unicasting." International Journal of Satellite Communications and Networking(22): 249-268.

msn.org. (2008). "Detailed analysis of the MSN Messenger protocol " March 2003 Version 2.1. Retrieved January, 2008, from http://www.hypothetic.org/docs/msn/general/overview.php.

Multicastbeacon. (2011). Retrieved May, 2011, from http://hpc.isti.cnr.it/accessgrid/beacon/.

Nasri, R., Rakotomanana, E., Affes, S., Ste, x and phenne, A. (2010)."On the evaluation of the LTE-advanced proposal within the Canadian evaluation group (CEG) initiative: Preliminary work results." Communications (QBSC), 2010 25th Biennial Symposium on.

Negus, K. J., Stephens, A. P. and Lansford, J. (2000). "HomeRF: wireless networking for the connected home." Personal Communications, IEEE **7**(1): 20-27.

Neto, A., Cerqueira, E., Rissato, A., Monteiro, E. and Mendes, P. (2007)."A Resource Reservation Protocol Supporting QoS-aware Multicast Trees for Next Generation Networks." Computers and Communications, 2007. ISCC 2007. 12th IEEE Symposium on.

Niyato, D. and Hossain, E. (2007). "Integration of WiMAX and WiFi: optimal pricing for bandwidth sharing." IEEE Communications Magazine **45**(5): 140-146.

Oetiker, T. (2001). "Monitoring your IT gear: the MRTG story." IT Professional **3**(6): 44-48.

Olteanu, A. and Yang, X. (2010). "Security overhead and performance for aggregation with fragment retransmission (AFR) in very high-speed wireless 802.11 LANs." Wireless Communications, IEEE Transactions on **9**(1): 218-226.

Orebaugh, A., Morris, G., Warnicke, E. and Ramirez, G. (2004). "Introducing Network Analysis." Ethereal Packet Sniffing. Rockland, Syngress**:** 1-38.

Orebaugh, A., Ramirez, G., Burke, J., Pesce, L., Wright, J. and Morris, G. (2006). "Wireshark and Ethereal." Wireshark & Ethereal Network Protocol Analyzer Toolkit. Rockland, Syngress**:** 523-540.

OWAMP. (2009, 23 January). "One Way Active Measurement Protocols (OWAMP) test tool " Version 1.3. Retrieved Web Page, 2009, from http://e2epi.internet2.edu/owamp/.

OWAMP. (2009). "One Way Ping." Retrieved 15 May, 2009.

Pace, P., Aloi, G. and Marano, S. (2004)."Performance analysis of connection admission control scheme in a DVB-RCS satellite system." Personal, Indoor and Mobile Radio Communications, 2004. PIMRC 2004. 15th IEEE International Symposium on.

Papadimitriou, P., Tsaoussidis, V. and Zhang, C. (2010). "End-to-end loss differentiation for video streaming with wireless link errors." Telecommunication Systems **43**(3): 295-312.

Papapanagiotou, I., Toumpakaris, D., Jungwon, L. and Devetsikiotis, M. (2009). "A survey on next generation mobile WiMAX networks: objectives, features and technical challenges." Communications Surveys & Tutorials, IEEE **11**(4): 3-18.

Pathrate.org. (2008). "Path rate and path load test tool." Retrieved March, 2008, from http:// www.Pathrate.org.

Paul, T. K. and Ogunfunmi, T. (2009). "Evolution, insights and challenges of the PHY layer for the emerging ieee 802.11n amendment." Communications Surveys & Tutorials, IEEE **11**(4): 131-150.

Peng, J. and Cheng, L. (2006)."Revisiting Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA)." Information Sciences and Systems, 2006 40th Annual Conference on.

Potter, B. (2003). "Wireless security's future." Security & Privacy, IEEE **1**(4): 68-72.

Qiang, N., Vinel, A., Yang, X., Turlikov, A. and Tao, J. (2007). "WIRELESS BROADBAND ACCESS: WIMAX AND BEYOND - Investigation of Bandwidth Request Mechanisms under Point-to-Multipoint Mode of WiMAX Networks." Communications Magazine, IEEE **45**(5): 132-138.

Quemada, J., de Miguel, T., Pavon, S., Huecas, G., Robles, T., Salvachua, J., Ortiz, D. A. A., Sirvent, V., Escribano, F. and Sedano, J. (2005)."Isabel: an application for real time collaboration with a flexible floor control." Collaborative Computing: Networking, Applications and Worksharing, 2005 International Conference on.

Raina, S. (2004)."Quality of service to subscriber." Telecommunications Quality of Services: The Business of Success, 2004. QoS 2004. IEE.

Rajasekhar, S., Khalil, I. and Tari, Z. (2006)."Probabilistic QoS Routing in WiFi P2P Networks." Advanced Information Networking and Applications, 2006. AINA 2006. 20th International Conference on.

Rao, Y. S., Wing-Cheong, Y. and Kripalani, A. (2000)."Third-generation (3G) radio access standards." Communication Technology Proceedings, 2000. WCC - ICCT 2000. International Conference on.

Rauffet, P., Da Cunha, C. and Bernard, A. (2009)."Designing and Managing Organizational Interoperability with Organizational Capabilities and Roadmaps." Interoperability for Enterprise Software and Applications China, 2009. IESA '09. International Conference on.

RDP. (2008). "RDP: Remote Desktop Protocol Features and Performance, Microsoft " Retrieved 2008, 2008, from http://www.microsoft.com/technet/prodtechnol/Win2KTS/evaluate/featfunc/rdpfperf.mspx.

Reljin, I. S. and Sugaris, A. N. (2009)."DVB standards development." Telecommunication in Modern Satellite, Cable, and Broadcasting Services, 2009. TELSIKS '09. 9th International Conference on.

Rezende, E. S., Dodonov, E., Ulson, R. S., Cavenaghi, M. A. and Lobato, R. S. (2009)."Towards Interoperability in P2P World: An Indexing Middleware for Multi-protocol Peer-to-Peer Data Sharing." Internet and Web Applications and Services, 2009. ICIW '09. Fourth International Conference on.

Ricardson, T. (2005). "The RFB Protocol." Retrieved June, 2008, from http://www.realvnc.com/docs/rfbproto.pdf.

Richling, J., Werner, M. and Popova-Zeugmann, L. (2002)."Automatic composition of timed Petri net specifications for a real-time architecture." Robotics and Automation, 2002. Proceedings. ICRA '02. IEEE International Conference on.

Rinne, M. and Tirkkonen, O. (2010). "LTE, the radio technology path towards 4G." Computer Communications **33**(16): 1894-1906.

Romdhani, L., Qiang, N. and Turletti, T. (2003)."Adaptive EDCF: enhanced service differentiation for IEEE 802.11 wireless ad-hoc networks."

Wireless Communications and Networking, 2003. WCNC 2003. 2003 IEEE.

Rong, B., Qian, Y. and Lu, K. (2007). "Integrated Downlink Resource Management for Multiservice WiMAX Networks." Mobile Computing, IEEE Transactions on **6**(6): 621-632.

Sabiguero, A., Baire, A., Boutet, A. and and, C. V. (2007). "Virtualized Interoperability Testing : Application toIPv6 Network Mobility." International Federation or Information Processing **lncs 4785**(Journal Article): 187-190.

Salhani, M., Dhaou, R. and Beylot, A. L. (2009). QoS mapping and connection admission control in the WiMAX - DVB-RCS access network." Proceedings of the 4th ACM workshop on Performance monitoring and measurement of heterogeneous wireless and wired networks. Tenerife, Canary Islands, Spain, ACM**:** 94-98.

Sameh, A., Wagh, S. and Salama, Q. (2010)."Dealing with Quality of Service in Hybrid Wired-Wireless Networks." Network Applications Protocols and Services (NETAPPS), 2010 Second International Conference on.

Sayenko, A., Alanen, O., Karhula, J. and Hamalainen, T. (2006). Ensuring the QoS requirements in 802.16 scheduling." Proceedings of the 9th ACM international symposium on Modeling analysis and simulation of wireless and mobile systems. Terromolinos, Spain, ACM**:** 108-117.

Serif, T. and Ghinea, G. (2005). "HMD versus PDA: a comparative study of the user out-of-box experience." Personal Ubiquitous Comput. **9**(4): 238-249.

Serif, T., Ghinea, G., Stergioulas, L., Chen, S., Tiropanis, T. and Tsekeridou, S. (2009). "Satellite-based delivery of educational content to geographically isolated communities: a service based approach." Personal and Ubiquitous Computing **13**(3): 229-241.

Services, F. I. (March 3, 2003). " Shareable Content Object Reference Model (SCORM)
"Retrieved May 2010, from http://www.marketresearch.com/product/display.asp?productid=887982.

Shah, S. I. (2008)."UMTS: High Speed Packet Access (HSPA) Technology." Networking and Communications Conference, 2008. INCC 2008. IEEE International.

Shahbazian, J. and Christensen, K. J. (2004). "TSGen: a tool for modeling of frame loss in streaming video." Int. J. Netw. Manag. **14**(5): 315-327.

Sharma, P., Xu, Z., Banerjee, S. and Lee, S.-J. (2006). "Estimating network proximity and latency." SIGCOMM Comput. Commun. Rev. **36**(3): 39-50.

Sharpe, R., Warnicke, E. and Lamping, U. (2008). "Ethereal users guide 18189 for Ethereal 0.10.14." Retrieved June, 2008, from http://www.ethereal.com.

Shave, N. P. (2002)."Roaming between satellite and terrestrial networks." 3G Mobile Communication Technologies, 2002. Third International Conference on (Conf. Publ. No. 489).

Shih, T. K., Te-Hua, W., Chih-Yung, C., Tai-Chien, K. and Hamilton, D. (2007). "Ubiquitous e-Learning With Multimodal Multimedia Devices." Multimedia, IEEE Transactions on **9**(3): 487-499.

Shorey, R., Ananda, A., Chan, M. C. and Ooi, W. T. (2006). Mobile, Wireless, and Sensor Networks: Technology, Applications, and Future Directions. Chapter 1 measuring wireless LANs. Wiley on line.

Siep, T. M., Gifford, I. C., Braley, R. C. and Heile, R. F. (2000). "Paving the way for personal area network standards: an overview of the IEEE P802.15 Working Group for Wireless Personal Area Networks." Personal Communications, IEEE **7**(1): 37-43.

Skinnemoen, H., Leirvik, R., Hetland, J., Fanebust, H. and Paxal, V. (2004). "Interactive IP-network via satellite DVB-RCS." Selected Areas in Communications, IEEE Journal on **22**(3): 508-517.

Skinnemoen, H. and Tork, H. (2002)."Standardization activities within broadband satellite multimedia." Communications, 2002. ICC 2002. IEEE International Conference on.

Skordoulis, D., Qiang, N., Hsiao-Hwa, C., Stephens, A. P., Changwen, L. and Jamalipour, A. (2008). "IEEE 802.11n MAC frame aggregation mechanisms for next-generation high-throughput WLANs." Wireless Communications, IEEE **15**(1): 40-47.

Sommers, J., Barford, P., Duffield, N. and Ron, A. (2005)."Improving Accuracy in end-to-end packet loss measurement." SIGCOMM'05,Proceedings of the 2005 conference on Applications, technologies, architectures, and protocols for computer communications ACM.

Song, Y.-J., Kim, P.-S., Oh, D.-G., Jeon, S.-I. and Lee, H.-J. (2006). "Development of mobile broadband interactive satellite access system for Ku/Ka band." International Journal of Satellite Communications and Networking **24**(2): 101-117.

Stergioulas, L. K., Abbasi, M., Pitsilis, V., Constantin, M. and Kretschmer, M. (2008)."Satellite-enabled education for geographically isolated communities of farmers and maritime workers, The BASE² Project." International Conference on Bridging the Digital Divide in Rural Communities: Practical Solutions and Policies, Athen, Greece.

Sudhaakar, R. S., Seokhoon, Y., Jia, Z. and Chunming, Q. (2009). "A novel Qos-aware MAC scheme using optimal retransmission for wireless networks." Wireless Communications, IEEE Transactions on **8**(5): 2230-2235.

Sugarbroad, I. (1990). "An OSI-based interoperability architecture for managing hybrid networks." Communications Magazine, IEEE **28**(3): 61-69.

Suitor, K. (2011). "What WiMax forum certified products will bring to Wi-Fi." Retrieved January.

Sun, T., Yang, G., Chen, L.-J., Sanadidi, M. Y. and Gerla, M. (2005). A measurement study of path capacity in 802.11b based wireless networks." Papers presented at the 2005 workshop on Wireless traffic measurements and modeling. Seattle, Washington, USENIX Association**:** 31-37.

Tateishi, N., Seto, S. and Seshake, H. (2009). "High-Speed Traceroute Method for Large Scale Network." Management Enabling the Future Internet for Changing Business and New Computing Services. Choong Hong, Toshio Tonouchi, Yan Ma and Chi-Shih Chao, Springer Berlin / Heidelberg. **5787:** 51-60.

TCPdump. (15 May 2007). "tcpdump test tool." Retrieved May, 2007, from http//:www.tcpdump.org.

Telesat. (2011). "Delay and response times." Retrieved May, 2011, from http://old.telesat.com/satellites/delay-response-times-e.asp.

Theodoros, T. and Kostantinos, V. (2007)."WiMAX network planning and system's performance evaluation." 8th IEEE Wireless Communications and Networking Conference, Kowloon, China, IEEE.

ThirdGeneration(3GPP2). (2009). "3G partnership project p2." Retrieved September 2010, from http://www.3gpp2.org/.

Thompson, C. W. (2011). "Next-Generation Virtual Worlds: Architecture, Status, and Directions." Internet Computing, IEEE **15**(1): 60-65.

Tianji, L., Qiang, N., Malone, D., Leith, D., Yang, X. and Turletti, T. (2006)."A new MAC scheme for very high-speed WLANs." World of Wireless, Mobile and Multimedia Networks, 2006. WoWMoM 2006. International Symposium on a.

Tsekeridou, S., Rorris, T., Rorris, D., Makropoulos, C., Serif, T. and Stergioulas, L. (2008). "Satellite-enabled educational services specification and requirements analysis based on user feedback." Int. J. Knowledge and learning **4**(2/3,2008): 272-284.

Tzu-Chieh, T. and Ming-Ju, W. (2005)."An analytical model for IEEE 802.11e EDCA." Communications, 2005. ICC 2005. 2005 IEEE International Conference on.

uCertify. (2005). "What is tcpdump? | uCertify Articles Retrieved from http://www.ucertify.com/article/what-is-tcpdump.html
" Retrieved July 2010, from http://www.ucertify.com/article/what-is-tcpdump.html

Veer, H. v. d. and Wiles, A. (2008) "Achieving Technical Interoperability -the ETSI Approach." Standard for Business ETSI **white paper no 3**.

Veltri, L., Salsano, S. and Martiniello, G. (2006)."Wireless LAN-3G Integration: Unified Mechanisms for Secure Authentication based on SIP." Communications, 2006. ICC '06. IEEE International Conference on.

Vieira, F., Castro, M. A. V. and Granandos, G. S. (2006). "A tunable-fairness Cross-layer scheduler for DVB-S2." Int. J. Satell. Commun. Network. Retrieved May, 2008.

Vinel, A., Ying, Z., Qiang, N. and Lyakhov, A. (2006)."WLC22-4: Efficient Request Mechanism Usage in IEEE 802.16." Global Telecommunications Conference, 2006. GLOBECOM '06. IEEE.

Viswanth, K. and Obraczka, K. (2006). " Interoperability of multicast routing protocols in wireless adhoc networks." wireless communications and mobile computing-Special issue on adhoc wireless network **6**(2): 225-234.

Vittorio, S. and Bello, L. L. (2010) "An approach to enhance the QoS support to real time traffic on IEEE 802.11e networks " White paper.

VNC. (2008). "Real VNC." Retrieved Web Page, 2008, from http://www.realvnc.com/support/documentation.html.

Voznak, M. (2008). "Comparison of H323 and SIP Protocol specification." Retrieved May 2008.

Wei, R. (2006). "Wi-Fi powered WLAN: when built, who will use It? Exploring predictors of wireless internet adoption in the workplace." Journal computer-mediated communication **12**(1): 155-175.

WiFi.org. (2011). "WiFi." Retrieved May, 2011, from “http://www.wi-fi.org”.

Williams, S. (2000). "IrDA: past, present and future." Personal Communications, IEEE **7**(1): 11-19.

Williamson, C. (2001). "Internet traffic measurement." Internet Computing, IEEE **5**(6): 70-74.

WiMAXforum. (2008). "WiMAX." Retrieved May, 2008, from http://www.wimaxforum.org.

Xiangpeng, J. and Raychaudhuri, D. (2006). "Spectrum co-existence of IEEE 802.11b and 802.16a networks using reactive and proactive etiquette policies." (Journal Article): 539-554.

Xylomenos, G., Polyzos, G. C., Mahonen, P. and Saaranen, M. (2001). "TCP performance issues over wireless links." Communications Magazine, IEEE **39**(4): 52-58.

Yoshida, S., Kimura, H., Inoue, Y., Masamura, T. and Yamauchi, N. (1999). "Interactive multimedia communication systems for next-generation education using asymmetrical satellite and terrestrial networks." Communications Magazine, IEEE **37**(3): 102-106.

Yue, Z., Chunhui, Z., Cosmas, J., Kok-Keong, L., Owens, T., Di Bari, R., Lostanlen, Y. and Bard, M. (2008). "Analysis of DVB-H Network Coverage With the Application of Transmit Diversity." Broadcasting, IEEE Transactions on **54**(3): 568-577.

Yuxia, L. and Wong, V. W. S. (2006)."Saturation throughput of IEEE 802.11e EDCA based on mean value analysis." Wireless Communications and Networking Conference, 2006. WCNC 2006. IEEE.

Yuxia, L. and Wong, V. W. S. (2009). "Cross-Layer Design of MIMO-Enabled WLANs With Network Utility Maximization." Vehicular Technology, IEEE Transactions on **58**(5): 2443-2456.

Zhang, Y., Ansari, N. and sunoda, H. (2010). "Wireless telemedicine services over integrated IEEE 802.11/WLAN and IEEE 802.16/WiMAX networks." Wireless Communications, IEEE **17**(1): pp. 30-36.

Zhu, J., Roy, S. and Kim, J. H. (2006). "Performance modelling of TCP enhancements in terrestrial-satellite hybrid networks." IEEE/ACM Trans. Netw. **14**(4): 753-766.